Active Noise Control In Truck Cabin

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Abstract

This Master of Science Thesis is a collaboration between ITS (Institution of Telecommunication and Signal processing) at Blekinge Institute of Technology (Blekinge Tekniska Högskola) and KALMAR Industries AB. The thesis is based on examinations of sound fields in a truck cabin. The main technical issue was to investigate how an active noise control system could work in a truck cabin, and also how effective it would be. Through recordings of noise in the cabin 16 microphones were used in seven different layers. The analysis of the noise and sound field visualization were performed in MATLAB. After analysis of the total RMS value of each microphone before and after the approximated narrow/broad-band filters were applied, a attenuation of 3dBA was possible. The implementation of adaptive algorithm showed that, the result did not quite compare to the approximations, due to the effect of the badly estimated control paths. Better control paths would probably result in a likely decrease as for the approximation.
Acknowledgement

We would like to start by thanking our friends and families for all the support. Also a big thanks to our examiners Sven Johansson and Mathias Winberg, both from ITS, for all their help and guidance through this entire master thesis, and Anders Brandt, Saven Edutech AB, for his answers about problems with the measurement system I-DEAS. Last but not least, we would like to thank our supervisor from Kalmar Industries AB, Börje Svensson for all his help and support during the measurement in Ljungby. Also a big thank to the staff at Kalmar Industries who was involved with this master thesis. We could not have done this thesis without you.

Thank You All!!!!
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Chapter 1

Introduction

1.1 Background of the thesis

During recent years a much more consideration have been put to the comfort in truck cabins. This have grown to be an important part in producing better trucks. That is why Kalmar Industries are interested to investigate if active noise control can be effective in a truck cabin. At the moment, the most ordinary solution is to isolate outside noise in the cabin by passive noise reduction techniques. One negative side of passive noise reduction is that it is most effective at higher frequencies. It can also be very costly if you have to redesign an already existing passive system.

Active attenuation on the other hand is based on generating an anti-noise of equal amplitude and opposite phase. Combined with the primary noise, thus resulting in the cancellation of both noises. It has been tested in aeroplanes such as SAAB 2000, helicopters and submarines with very good results.

The assignment of this master thesis consists of two primary parts. First there is the sound field measurements and analysis, then the active noise control measurements and analysis. Both those two parts are measured at three different revolutions: 1500RPM, 1900RPM and maximal RPM. The data collection is then used for sound field analysis and active noise control calculations. This report consists of every step in the analysis, from the recordings to the results. Chapter one treats briefly the background of the thesis along with information about Kalmar Industries AB and the examined truck. The second chapter describes the sound field measurements and analysis, while the third chapter handles approximations of active narrow- and broad band systems. Chapter four consists of sound field measurement before/after extra passive reduction materials have been applied in the standard truck cabin. The fifth chapter deals with the active noise control measurements, theories, analysis and simulations. The summary and conclusions are gathered in chapter six and last of the report consists of bibliography and appendix.
1.2 Kalmar Industries AB

Kalmar is a global provider of heavy duty materials handling equipment and services to ports, terminals and demanding industrial customers. Kalmar focuses on supplying handling solutions that enable customers to operate with a high level of efficiency and reliability.

Kalmar leads development in heavy industrial, container and trailer handling with many well known trademarks in their brief-case, as Kalmar, Sisu, Ottawa and Magnum. They offer a unique breadth of products, especially when it comes to diesel, LPG and electric fork lift trucks.

There are over 65,000 Kalmar machines in service in more than 140 countries around the world. You can find them in every industry and transport system, and in all imaginable climate and environment. Every fourth container or trailer transfer at terminals around the world is handled by a Kalmar machine.

Manufacturing plants are located in Sweden (as well as the head office), in Finland, in the USA, in the Netherlands and in Estonia.

Figure 1.1: Some of Kalmar industries products.

1.3 Specification of the examined truck

<table>
<thead>
<tr>
<th>Specification</th>
<th>Details</th>
</tr>
</thead>
<tbody>
<tr>
<td>Truck model:</td>
<td>DCE90-6</td>
</tr>
<tr>
<td>Engine:</td>
<td>Volvo TAD620VE</td>
</tr>
<tr>
<td>Gearbox:</td>
<td>Clark 1207FT20315</td>
</tr>
<tr>
<td>Maximum revs (RPM):</td>
<td>2430 RPM</td>
</tr>
<tr>
<td>Idling revs:</td>
<td>760 RPM</td>
</tr>
<tr>
<td>Radiator:</td>
<td>Nissen</td>
</tr>
<tr>
<td>Fan:</td>
<td>Volvo standard fan / 9 blades, shift up 1.12 times</td>
</tr>
<tr>
<td>Hydraulic pump:</td>
<td>Parker triple pump / 10 cogs, shift up 1.10 times</td>
</tr>
<tr>
<td>Cabin:</td>
<td>Completely standard</td>
</tr>
<tr>
<td>Running time:</td>
<td>75 hours</td>
</tr>
<tr>
<td>Weight truck:</td>
<td>15360 kilogram</td>
</tr>
<tr>
<td>Lift capacity:</td>
<td>9000 kg on 600mm from the center of gravity</td>
</tr>
</tbody>
</table>

Table 1.1: Specification of the examined truck used.
Chapter 2

Sound Field In The Truck Cabin

2.1 Sound Field Measurements

One of the most important parts of this thesis is the sound field measurements in the truck cabin. The primary purpose with sound field measurements is to determine the dominating frequency components, and also find out what is generating the acoustics field and which level the acoustic field has. The intention of finding the sound fields dominating frequencies is to be able to identify the noise sources, that are dependent on the revolutions. Due to lack of time identification were only done for the three most common sources, which is the engine, the fan and the hydraulic pump. These sources could later on be used to obtain reference signals to the active system. Last in this chapter, Section 2.3, shows the sound maps which gives an overview on how the sound is distributed in the truck cabin. These analysis are later on very important for the ANC-system.

The truck that was examined, was located at Ljungby, where all the recordings took place. First a simple solution of attaching the microphones had to be solved. A sort of suspension device had to be placed in the cabin for the microphones. The device became a grid, and was provided by KALMAR, which was attached to the ceiling in the cabin.

Figure 2.1: A grid for the microphones, which was attached to the ceiling in the cabin.
The software that was used for measurements in this thesis was SDRC I-DEAS installed on an ordinary PC. Attached to that PC was a VXI HP front end, to which 16 channels could be connected, shown in figure 2.2. During the sound field measurements, microphones were attached to all these channels. The microphones that were used were PCB, model 130a10, which are ICP-transducers. After testing so that the whole system worked, the microphones were then calibrated. The calibrator used, were a Larson-Davis model CA250, which calibrates on a 114dB high peak at 250Hz.

The first thing that had to be considered before attaching the microphones in the cabin, was the wavelength. Because if two microphones were placed at a wavelength distance between each other, they will give the same result. Say for example that they are both placed in a node, the sound pressure level would be very low. By counting the wavelength it would be easy to know what distance it should be between two microphones.

\[ \lambda = \frac{c}{f} \]  

Where \( f \) is approximately 600Hz (the essential bandwidth) and \( c \) is the acoustic velocity in air (340m/s), which gave a wavelength of approximately 57cm. By knowing the wavelength a rectangular matrix of microphones could be attached to the grid.
Chapter 2. Sound Field In The Truck Cabin

Figure 2.3: Microphone array inside the cabin.

The microphones were placed 10cm above the cabin floor, were the first layer was decided to be. With a distance of 20cm between each layer, recordings were able for seven layers in the cabin. Layer six, which was 110cm above the floor, represented the height, where a truck drivers head is (see figure 2.4). Measurements were then done for each of the seven layers, at both closed and opened cabin windows and also for three different number of revolutions.

Figure 2.4: Microphone layers inside the cabin.

Those were as following: 1500RPM\(^1\), 1900RPM and maximal (2340)RPM. In I-DEAS the data was saved as time data, because then once recordings are done, it is possible to extract data both into the time- and frequency domain.

\(^1\)Rotation Per Minute
The settings that were used in I-DEAS during the recordings were:

<table>
<thead>
<tr>
<th>Setting</th>
<th>Value</th>
</tr>
</thead>
<tbody>
<tr>
<td>Fs</td>
<td>25600 Hz</td>
</tr>
<tr>
<td>Spectral lines</td>
<td>3201</td>
</tr>
<tr>
<td>Frame size</td>
<td>8192 sample</td>
</tr>
<tr>
<td>Number of mean values</td>
<td>200</td>
</tr>
<tr>
<td>Window</td>
<td>Hanning broad</td>
</tr>
<tr>
<td>Trigger</td>
<td>Free Run</td>
</tr>
</tbody>
</table>

Table 2.1: Measurement settings in I-DEAS.

The interesting bandwidth for an active noise control is only up to about 600Hz, but to see how it affects the total sound pressure level, it was necessary to increase the bandwidth during recordings. The 600Hz boundary is due to the wavelengths. At low frequencies the wavelengths are larger, than for higher frequencies. The active noise control interfere the disturbing sound field, with a “mirror” of it. To gain the best interference, the two sound fields have to be aligned over time and space[5]. When the wavelengths are larger, this alignment will be easier to accomplish. A good check is done by controlling that no higher frequencies has a larger sound pressure level than the ones in 0-600Hz. If there is a frequency above 600Hz that has a much higher sound level, active noise control is often useless, for those frequencies. Because even though the lower frequencies have been decreased, it have not effected the total sound pressure level, which the ear catches.

The window Hanning broad was used, because this window is most suitable for noise. There was a thing to remember using Hanning broad, which is that Hanning broad has a wide main lobe. Normally for ordinary Hanning, there is a amplitude compensation factor[2] of \(A_w = 0.5\) (see eq. 2.3), but for Hanning broad \(A_w\) had to be counted. This was done later on in the thesis.

The number of mean values was set to 200 because that resulted in time data of about one minute, which was more than necessary for analysis.

When all the data had been recorded for the seven layers, they had to be exported first into unv-files from I-DEAS. Exporting was done both into time data and frequency data. Only the exported frequency data was used for analysis, while the time data was used in backup purpose.

Before the saved time- and frequency data could be imported in Matlab and analyzed, a transformation of the universal-file (which is the exported file from I-DEAS) into a math-file had to be done. The transformation was made with a Matlab toolbox called unv. This toolbox has been developed by Anders Brandt, Saven EduTech AB. When processed the unv-file, a math-file with the same filename was created. If loading the math-file in Matlab, a matrix with all the essential data forms was created. The number of columns in the matrix depends on how many microphones that have been used. For example in this thesis, a number of 16 microphones was used and the microphone that was connected to channel one, ends up in column one in the matrix.

### 2.2 Analysis and results of Sound field Measurements

Due to limited time in the thesis, analysis were only done for the case with the cabin windows closed. By studying the auto (power) spectrum \(G_{xx}(f)\) for several microphones, will give a conclusion on how the dominating frequencies are distributed over the bandwidth. When examining the auto spectrum, it was obvious that the amplitude of the sound level had decreased considerably. The reason for this is when importing a file in Matlab, there had to be a consideration of the SPL\(^2\)[3]. The exported files in the frequency domain from I-DEAS are the squared power and in linear scale. This have to be transform into logarithmic scale(dB)

\[
L_p = 10 \log_{10} \frac{p^2}{p_{ref}^2} \tag{2.2}
\]

were \(p_{ref} = 20\mu Pa\) is the reference level for sound pressure. By knowing that Hanning broad had been used, the decrease factor had to be counted.

\(^2\)Sound Pressure Level
Calculating the amplitude compensation factor $A_w$ with

$$A_w = \frac{N}{\sum_{n=0}^{N-1} w(n)}$$

(2.3)

where $w(n)$ is the time-window function and $N$ is the frame size. Another way to “find” $A_w$ is that in I-DEAS, there is a function where one can display the auto spectrum along with the total RMS $^3$, which is the square root of the integral of the auto spectrum, $G_{xx}(f)$, over the total bandwidth.

$$\text{RMS}_x = \sqrt{\int G_{xx}(f) df}$$

(2.4)

The RMS-value is a power-value and measures how “large” a dynamic signal is. Comparing that value with the total RMS value for the Matlab vector, a decrease factor of 3.126 was given. That factor was applied for several different microphones at different layers. Comparing every value with the one’s from I-DEAS resulted in that they all matched.

Figure 2.5: Auto spectrum of microphone 13 at 10cm above the floor, for the three different number of revolutions.

The figure 2.5 displays the auto spectrum over the entire measurement bandwidth. As seen in the figure, no dominating frequencies were found above 600Hz, therefore a closer look at the essential bandwidth could take place. Because the number of spectral lines used were 3201 and a bandwidth of 10000Hz, it is 3,124 Hz between each frequency sample.

$^3$Root Mean Square
Chapter 2. Sound Field In The Truck Cabin

Figure 2.6: Auto spectrum of microphone 13 up to 600Hz, 10cm above the floor, at 1500RPM, 1900RPM and Maximal RPM.

Fig. 2.6 illustrates the auto spectrum for each of the three different RPMs. It will be easy to examine every dominating frequency and determine if it is depended of the revolutions from the engine, or not. For example, in the figure 2.6 a tonal component is found at approximately 180Hz for each of the different RPMs. That makes it independent on the number of revolutions, but for some other frequencies, there are a dependency and will be illustrated further on in the thesis. By knowing that a frequency depends on the engines revolutions in some way, a comparison between the frequencies and the different orders from the engine, the fan and the hydraulic pump can be done. This would specify which frequencies originates from those devices.
2.2.1 A-Weighted Auto spectrum

Examining an auto spectrum of a signal gives a very reliable answer on where the dominating frequencies exists, because it displays the squared effect in log scale. But it will not show how the human ear experience the sound. By applying a weighting to the signal, one can make a resemblance on how the ear apprehend the sound. The weighting in this thesis has been done in I-DEAS, else weighting can be done by interpolating values from a table with the ones in the spectra. There are three different weights, A-weighting, B-weighting and C-weighting. This thesis has used A-weighting, which has been developed from the 40-phon curves and is thought to be used for low SPL.

![A-weighted spectrum for mic 13, 10cm above floor at maximal RPM](image)

Figure 2.7: A-weighted spectrum (dBA) for mic 13, 10cm above the floor at maximal RPM.

A-weighted SPL is a weighted SPL that is adjusted for the apprehension of the human ear, and has non-linear behaviour in amplitude. It attenuates frequencies below 1000Hz[3], where it reduces frequencies to compensate for the human ears lower sensitivity at low frequencies.
2.2.2 Dominating frequencies at 1500RPM

Looking at the figure 2.6, it displays several tonal components, and therefore by writing a Matlab script that shows the most dominating frequencies, histogram can be depicted. The boundary for a dominating frequency was set to: peak of 4dB. This script was only done for the layer 110cm above the floor, which represented the ear, because the active system should be configured to work best in this height.

![Figure 2.8: Histogram of dominating frequencies 4dB or higher at 1500RPM.](image)

The number of microphones in figure 2.8 illustrates how many microphones in the layer one special frequency has occurred. At 1500RPM there are 4 tonal components that almost occur for every microphone. If these would be compared to some of the orders from the engine, and also be depended on the number of revolutions, an active narrowband system could be applied on them. By comparing these frequencies with a table of the frequencies from the engine at different orders, would give a conclusion what kind of system should be most suitable.

<table>
<thead>
<tr>
<th>Frequency (Hz)</th>
<th>25</th>
<th>50</th>
<th>75</th>
<th>100</th>
<th>125</th>
<th>150</th>
</tr>
</thead>
<tbody>
<tr>
<td>175</td>
<td>200</td>
<td>225</td>
<td>250</td>
<td>275</td>
<td>300</td>
<td></td>
</tr>
<tr>
<td>325</td>
<td>350</td>
<td>375</td>
<td>400</td>
<td>425</td>
<td>450</td>
<td></td>
</tr>
<tr>
<td>475</td>
<td>500</td>
<td>525</td>
<td>550</td>
<td>575</td>
<td>600</td>
<td></td>
</tr>
</tbody>
</table>

Table 2.2: Frequencies (Hz) of the 24th first orders from the engine at 1500RPM.

Comparing the dominating frequencies at 1500RPM to the frequencies of the engine orders, it is obvious that order 1, 3, 6, 9, 12, 13, 17, 18, 21 from the engine exists among the frequencies. Before making the same analysis at the other revolutions, an equal table must be done for both the fan and the hydraulic pump. The fan has nine blades and rotates 1.12 times faster than the engine, while the hydraulic pump rotates 1.10 times faster and has 10 teeth, resulting in following frequencies for the different orders.

<table>
<thead>
<tr>
<th>Frequency (Hz)</th>
<th>28</th>
<th>56</th>
<th>84</th>
<th>112</th>
<th>140</th>
<th>168</th>
</tr>
</thead>
<tbody>
<tr>
<td>196</td>
<td>224</td>
<td>252</td>
<td>280</td>
<td>308</td>
<td>336</td>
<td></td>
</tr>
<tr>
<td>364</td>
<td>392</td>
<td>420</td>
<td>448</td>
<td>476</td>
<td>504</td>
<td></td>
</tr>
<tr>
<td>532</td>
<td>560</td>
<td>588</td>
<td>616</td>
<td>644</td>
<td>672</td>
<td></td>
</tr>
</tbody>
</table>

Table 2.3: Frequencies (Hz) of the 24th first orders from the fan at 1500RPM.

For the fan there were only two orders, that matched the tonal components. Those were 12 and 13. One
aspect of that the fan did not contribute to so many frequencies could be that for 1500RPM, by counting the numbers of agreeing orders from the engine, makes the engine the dominant source. This was quite obvious when listening to the sound during the recordings.

<table>
<thead>
<tr>
<th></th>
<th>27.5</th>
<th>55</th>
<th>82.5</th>
<th>110</th>
<th>137.5</th>
<th>165</th>
</tr>
</thead>
<tbody>
<tr>
<td>192.5</td>
<td>220</td>
<td>247.5</td>
<td>275</td>
<td>302.5</td>
<td>330</td>
<td></td>
</tr>
<tr>
<td>357.5</td>
<td>385</td>
<td>412.5</td>
<td>440</td>
<td>467.5</td>
<td>495</td>
<td></td>
</tr>
<tr>
<td>522.5</td>
<td>550</td>
<td>577.5</td>
<td>605</td>
<td>632.5</td>
<td>660</td>
<td></td>
</tr>
</tbody>
</table>

Table 2.4: Frequencies (Hz) of the 24th first orders from the hydraulic pump at 1500RPM.

Result from the table 2.4 with the orders from the hydraulic pump led to only one agreement and that were order 13th. By putting all the agreeing orders from the different sources together, it would be easy to see if all the frequencies have arisen from these three sources.

To sum up the identification of the dominating frequencies at 1500RPM, figure 2.8 illustrates the tonal components, marked with the sources from which they have arisen. E is the engine, F the fan and H the hydraulic pump. As seen there are two frequencies that occurs for many microphones, which are not identified. These frequencies are quite hard to tell where they arise from, it could be anything that vibrates in the cabin for example a window or something else rotating. More clarity about these tonal components could be drawn after the active measurements, where accelerometers are placed on different vibration sources.
2.2.3 Dominating frequencies at 1900RPM

Using the script for processing the most dominant frequencies at 1900RPM, resulted in figure 2.9.

![Histogram of dominating frequencies 4dB or higher at 1900RPM.](image)

Figure 2.9: Histogram of dominating frequencies 4dB or higher at 1900RPM.

It is obvious that there are a couple of more tonal components who occurs at 1900RPM, compared to 1500RPM. The reason for this could be several different things, but one likely aspect, would be that for higher rotations more vibrations should occur. A closer look at the most dominant sources was done in order to make more detailed conclusion. During the sound recordings for 1900RPM, one could clearly hear that the fan became more dominant than at 1500RPM. By looking at the frequencies for the different orders from the fan, may result in successfully comparing with the dominating frequencies.

<table>
<thead>
<tr>
<th>Frequency (Hz)</th>
</tr>
</thead>
<tbody>
<tr>
<td>31.7</td>
</tr>
<tr>
<td>63.3</td>
</tr>
<tr>
<td>95</td>
</tr>
<tr>
<td>126.7</td>
</tr>
<tr>
<td>158.3</td>
</tr>
<tr>
<td>190</td>
</tr>
<tr>
<td>221.7</td>
</tr>
<tr>
<td>253.3</td>
</tr>
<tr>
<td>285</td>
</tr>
<tr>
<td>316.7</td>
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<td>348.3</td>
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<td>380</td>
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<td>411.7</td>
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<td>443.3</td>
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<tr>
<td>475</td>
</tr>
<tr>
<td>506.7</td>
</tr>
<tr>
<td>538.3</td>
</tr>
<tr>
<td>570</td>
</tr>
</tbody>
</table>

Table 2.5: Frequencies (Hz) of the 18th first orders of the engine at 1900RPM.

Examining the figure 2.9, results in that order: 3, 6, 12 from the engine, are sources to 3 of the frequencies. Comparing with the previous RPM, all the orders existed there to. If that would be the results for maximal RPM to, these orders could be used as reference signals for an active system, valid for all RPMs.

<table>
<thead>
<tr>
<th>Frequency (Hz)</th>
</tr>
</thead>
<tbody>
<tr>
<td>35.5</td>
</tr>
<tr>
<td>70.9</td>
</tr>
<tr>
<td>106.4</td>
</tr>
<tr>
<td>141.9</td>
</tr>
<tr>
<td>177.3</td>
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<td>212.8</td>
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<td>248.3</td>
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<td>283.7</td>
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<td>319.2</td>
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<td>390.1</td>
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<td>425.6</td>
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<td>496.5</td>
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<tr>
<td>532</td>
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<tr>
<td>567.5</td>
</tr>
<tr>
<td>602.9</td>
</tr>
<tr>
<td>638.4</td>
</tr>
</tbody>
</table>

Table 2.6: Frequencies (Hz) of the 18th first orders from the fan at 1900RPM.

For the fan orders: 4,16 compares to the frequencies in figure 2.9, while at 1500RPM there were two different orders. That was a negative result. Doing this comparison between the different revolutions, should hopefully lead to an agreement of the orders, but at different frequencies. It can be quite hard to explain this problem, but reality can sometimes be that hard, when you analyse complex systems. One issue could be that when studying the auto spectrum of an microphone, it contains mostly broadband noise. A lot of these tonal components in the broadband noise lies near the boundary for being included in the MATLAB-script that calculates the dominating frequencies.
Running the script for different rotations, they maybe become included for some cases.

<table>
<thead>
<tr>
<th>34.8</th>
<th>69.7</th>
<th>104.5</th>
<th>139.3</th>
<th>174.2</th>
<th>209</th>
</tr>
</thead>
<tbody>
<tr>
<td>243.8</td>
<td>278.7</td>
<td>313.5</td>
<td>348.3</td>
<td>383.2</td>
<td>418</td>
</tr>
<tr>
<td>452.8</td>
<td>487.7</td>
<td>522.5</td>
<td>557.3</td>
<td>592.2</td>
<td>627</td>
</tr>
</tbody>
</table>

Table 2.7: Frequencies (Hz) of the 18th first orders from the hydraulic pump at 1900RPM.

At 1900RPM, the frequencies that arrised from the hydraulic pump, were the ones at the orders 5, 13 for the pump. Order 13 also existed at 1500RPM.

To summarize the identification of the dominating frequencies at 1900RPM, led to that orders 3, 6, 12 from the engine, order 4, 16 from the fan and finally order 5 and 13 from the hydraulic pump were found.
2.2.4 Dominating frequencies at Maximal RPM

Finally, the examination of the dominating frequencies were done for maximal RPM and the result is illustrated in figure 2.10.

![Figure 2.10: Histogram of dominating frequencies 4dB or higher at maximal RPM.](image)

For the maximal revolutions, most of the frequencies that have been included in the boundary 4dB or higher, exists for nearly every microphone. If there can be an identification of these tonal components, it will lead to a better effect of the active system.

<table>
<thead>
<tr>
<th>Frequency (Hz)</th>
<th>39</th>
<th>78</th>
<th>117</th>
<th>156</th>
<th>195</th>
<th>234</th>
</tr>
</thead>
<tbody>
<tr>
<td>Frequencies</td>
<td>273</td>
<td>312</td>
<td>351</td>
<td>390</td>
<td>429</td>
<td>468</td>
</tr>
<tr>
<td></td>
<td>507</td>
<td>546</td>
<td>585</td>
<td>624</td>
<td>663</td>
<td>702</td>
</tr>
</tbody>
</table>

Table 2.8: Frequencies (Hz) of the 18th first orders of the engine at maximal RPM.

From table 2.8 there can be seen that order: 3, 6, 12 from the engine contributes to three of the frequencies at maximal number of revolutions. Finally, comparing the orders from the engine that contributes to the frequencies, results in that order: 3, 6, 12 exists for every different RPM.

<table>
<thead>
<tr>
<th>Frequency (Hz)</th>
<th>43.7</th>
<th>87.4</th>
<th>131</th>
<th>174.7</th>
<th>218.4</th>
<th>262.1</th>
</tr>
</thead>
<tbody>
<tr>
<td>Frequencies</td>
<td>305.8</td>
<td>349.4</td>
<td>393.1</td>
<td>436.8</td>
<td>480.5</td>
<td>524.2</td>
</tr>
<tr>
<td></td>
<td>567.8</td>
<td>611.5</td>
<td>655.2</td>
<td>698.9</td>
<td>742.6</td>
<td>786.2</td>
</tr>
</tbody>
</table>

Table 2.9: Frequencies (Hz) of the 18th first orders from the fan at maximal RPM.

Examining the orders from the fan, led to that none of the orders compared to the frequencies. By looking at the orders from the fan at all the different revolutions, one can draw a conclusion that the fan is not a practical source for an active narrowband system.
<table>
<thead>
<tr>
<th>42.9</th>
<th>85.8</th>
<th>128.7</th>
<th>171.6</th>
<th>214.5</th>
<th>257.4</th>
</tr>
</thead>
<tbody>
<tr>
<td>300.3</td>
<td>343.2</td>
<td>386.1</td>
<td>429</td>
<td>471.9</td>
<td>514.8</td>
</tr>
<tr>
<td>557.7</td>
<td>600.6</td>
<td>643.5</td>
<td>686.4</td>
<td>729.3</td>
<td>772.2</td>
</tr>
</tbody>
</table>

Table 2.10: Frequencies (Hz) of the 18th first orders from the hydraulic pump at maximal RPM.

From the hydraulic pump at maximal RPM, there are order 6, 13 that compare to the dominating frequencies. At the cases with the lower revolutions, order 13 also contributed to a tonal component. Therefore, this order from the hydraulic pump could be useful as a reference source to an active system.

### 2.2.5 Summary of the dominating frequencies

To make a summary of the identification for the frequencies at the three different revolutions, order 3, 6, 12 from the engine and order 13 from the hydraulic pump would be useful for an active narrowband system. Using these four orders as reference signals to an active system, would probably not lead to a significant total decrease. This is the conclusion of looking at the earlier auto spectrum, where there are a lot more than four high tonal components.
2.3 Calculation of Sound maps and A-weighted Sound maps

To get an overview of the sound field in the truck cabin, sound maps could be depicted. When making a sound map, first there has to be done a calculation of the total SPL-value for each one of the microphones in the grid. Having calculated the auto spectrum, the total SPL-value can be formulated as a sum of all the samples in the vector,

\[
L_{\text{Tot}} = 10 \log \left( \sum_{i=1}^{l} 10^{L_{pi}/10} \right)
\]

(2.5)

where \( i \) is each of the samples in the vector. Formula 2.5 is applied on both the ordinary auto spectrum and the A-weighted auto spectrum. Once the total SPL-value has been calculated for every microphone at a particular layer, interpolation in 2-dimension took place. When using the predefined function for interpolation in MATLAB, it will interpolate over the range between microphones in X-direction and Y-direction along with the total SPL-value for the microphones.

![Soundmap of Autospectrum, 10cm above the cabin floor at maximal RPM](image)

Figure 2.11: Sound map in dB of layer 10cm above cabin floor at maximal RPM.

The figure 2.11 shows the matrix of all 16 microphones, which are displayed like circles. This particular sound map displays the sound field at the layer 10cm above the cabin floor for maximal RPM. The axis in the figure symbolizes the walls of the cabin. The range of the axis are according to the physical scale. Studying the sound map above, make it obvious that the highest SPL are located at the rear corners of the cabin floor. Therefore, beside applying an active noise control system in the truck, passive reduction can be applied in these corners to regain lower SPL.
By making a similar sound map as fig. 2.11, but for A-weighted spectrum instead, gives a good answer on how the human ear experience that particular layer in the cabin.
Chapter 3

Active Narrow- and Broadband approximation

In this chapter a rough approximation on an active narrow- and broadband system will be done. Doing an approximation of an active noise control system, will only give a hint on how an actual active noise control system would effect the sound field or more directly the microphones in the grid. There are two quite different approaches to an active system: a narrowband system or a broadband system. Approximations of these two systems are not solved the same way as for an actual system, where a LMS-algorithm along with an adaptive FIR-filter are used. But this will be explained further on in the thesis.

3.1 Active Narrowband approximation

When implementing an active narrowband solution it will attenuate, specific frequencies, particular reference tonal components. If an active narrowband system would be used for the examined truck in this thesis, there has to be frequencies that are dependent on the RPM. By having the conclusion in mind from the identification of the dominating frequencies in chapter 2.2.5, orders 3, 6, 12 from the engine and order 13 from the hydraulic pump will be used as reference signals. Normally a tachometer signal is recorded from the object and then those reference frequencies will be generated from that signal. For an actual active system, these 4 tonal components will be decreased at about 15-20dB by the LMS algorithm. A calculation of this decrease was done in MATLAB, by the use of Notch filters.

![Active Narrowband Approximation](image.png)

Figure 3.1: Approximation of an active narrowband system for an A-weighted spectrum (dBA) of microphone 6.

As seen in figure 3.1, the frequencies for the different orders have been decreased with about 15dB.
This particular A-weighted spectrum represent microphone 6 in the sound field grid. Even though not all of the orders exists for every microphone, these orders will still be decreased, for simplicity, that is. This is illustrated for the second order of the engine (234Hz) in the figure. By calculation of the total RMS-value before and after the approximation of the active narrowband system, gave the following result:

<p>| | |</p>
<table>
<thead>
<tr>
<th></th>
<th></th>
</tr>
</thead>
<tbody>
<tr>
<td><strong>Before</strong></td>
<td><strong>78.7dBA</strong></td>
</tr>
<tr>
<td><strong>After</strong></td>
<td><strong>77.6dBA</strong></td>
</tr>
</tbody>
</table>

Table 3.1: RMS-value, before and after active narrowband approximation of bandwidth 0-10kHz, for microphone nr. 6.

The values in table 3.1 are for the A-weighted spectrum from microphone 6 in the grid, which represent the right ear of a truck driver. By studying the A-weighted spectrums in figure 3.1 they mostly consist of broadband noise, and a few high frequencies. But when decreasing these frequencies, it will not have a major effect on the total RMS-value, as seen in the table. Therefor an examination of an active broadband solution should be done, which may give better result of decreasing the total RMS-value.

### 3.2 Active Broadband approximation

Estimation of an active broadband system can be done by decreasing the lowest frequency with 20dB and then at 600Hz and up, 0dB decrease. Between those two boundaries there should be a linear reduction of the amplitude decrease.

![Active broadband filter](image)

Figure 3.2: Active broadband filter.

Figure 3.2 illustrates how an active broadband system would effect the spectrum. It should be reminded that this is only a hint on how it could effect the spectra, it is not reality.
Applying the broadband-filter on the A-weighted microphone signal is illustrated in the figure below.

![Active Broadband Approximation](image)

Figure 3.3: Approximation of an active broadband system for an A-weighted spectrum (dBA) of microphone 6.

When making an actual broadband system, accelerometers or microphones are normally being used as reference signals, instead of a tachometer signal as in the narrowband system. To gain the effect of the broadband filter displayed in the figure 3.3, it is necessary to have decent loudspeakers, that can work for very low frequencies in a real system. Because of all the broadband noise that the spectrum contains, it will probably be best to use an active system with broadband solution and have an accelerometer as reference signal. For a narrowband system, there has to be a lot of frequencies generated to decrease all the tonal components and this will end up in a very complex system. But before drawing any conclusions, the RMS-value for the broadband system has to be considered.

<table>
<thead>
<tr>
<th>Before</th>
<th>78.7dBA</th>
</tr>
</thead>
<tbody>
<tr>
<td>After</td>
<td>77.6dBA</td>
</tr>
</tbody>
</table>

Table 3.2: RMS-value, before and after active broadband approximation of bandwidth 0-10kHz, for microphone nr. 6.

The difference between the two RMS-values are practical the same as for the narrowband approximation. It can be noticed that the RMS-values differentiates for each of the microphones. There can be a difference up to 3dBA between the RMS-value before and after the decrease.

To sum up the approximated systems, the mean value of all the microphones RMS-values have been calculated to see which system gives the best “total” effect.

<table>
<thead>
<tr>
<th>Before</th>
<th>80.1dBA</th>
</tr>
</thead>
<tbody>
<tr>
<td>Narrowband</td>
<td>77.7dBA</td>
</tr>
<tr>
<td>Broadband</td>
<td>77.5dBA</td>
</tr>
</tbody>
</table>

Table 3.3: Mean RMS-value of all microphones in head-layer, before the approximated systems and after the two systems, for bandwidth 0-10kHz.

In table 3.3 there is a total decrease of 3dBA for both an active narrowband system and an active broadband system. The question is, when comparing the figures 3.1 and 3.3 with the tables 3.1 and 3.2, why the differences of the RMS-value are minimal from the tables between the two filtering systems, comparing with the big difference one can see in the figures. The answer is that the figures only shows a bandwidth between 0-600 Hz, when from the tables the RMS-value is a result of bandwidth 0-10kHz, which was the total measurement bandwidth.
Chapter 4

Passive Noise Reduction

As the sound maps in Chapter 2.3 show, the highest SPL are located in the rear corners at the floor. Taking a closer look at these corners there are for instance gaps between the cabin floor and the doors, vibrating metal between the plastic panel located in the back of the cabin and a hole in the floor where the steering cables came through. Before implementing an active noise control system, it would be interesting to investigate how a decrease of the SPL can be obtained with passive techniques by using the sound maps as design tools. By using ordinary passive reduction material in all the places mentioned above, would hopefully lead to a positive result. To best view the effect of the passive reduction, a comparison between the A-weighted sound map before and after applying the extra passive materials was done.

Figure 4.1: Sound map of A-weighted noise spectrum (dBA) before extra passive reduction material in the rear corners of the cabin floor. The sound field is measured 10cm above the cabin floor.
As seen in the figures 4.1 and 4.2, there are a big difference between them. But having investigated the effect of the passive noise reduction for the layer were the head of the driver would be, led to no change in the SPL. It should be noticed that the recordings and construction of the passive reduction was done under great time pressure, so by doing a more advanced passive reduction system would probably lead to better results. The passive reduction were also applied during the measurements for the active noise control system further on in the thesis.
Chapter 5  

Active Noise Control  

5.1 Introduction  

The basic idea of active noise control is that there are two sound fields, which interfere with each other, thus result in a cancellation of them both. The primary sound field is the normal sound in the truck cabin, and the secondary field is the cancellation sound generated by the active noise control system and transmitted from the loudspeakers in the cabin. When implementing a feedforward ANC-system into the truck cabin, it is essential that there have been an identification of noise sources. These sources are then used to obtain reference signals to the active system. The reference signals contains the frequency components we want to reduce in the noise. The reference signals are processed in the ANC-system and are used to generate the secondary sound field or the “anti-noise”. This secondary sound field has the same amplitude but opposite phase to the primary sound field. This will result in a cancellation of both the two sound fields.

![ANC-system schematic](image)

Figure 5.1: A schematic figure of the feedforward ANC-system in the cabin.

ANC-systems are most effective at low frequencies (below 500/600Hz), while passive systems are effective at high frequencies. This is linked to the wavelength of the sound. To achieve the most effective ANC-system the primary and secondary sound fields have to be exactly aligned over an acoustic wavelength. Due to the alignment, active systems works best at low frequencies, where the wavelengths of the acoustic are large compared to the area where the noise shall be cancelled. One can also describe active noise control as that the system creates an inverse or mirror of the disturbing sound field. This mirror will cancel the primary sound field by the principle destructive interference. The way that this works is that a sound wave is a series of high-pressure and low-pressure. If the high-pressure part of one wave, lines up with the low-pressure of another wave, the two waves interfere destructively and there is no more pressure (no more sound). This matching

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\(^1\text{Active Noise Control}\)
must occur both in time and in space, which is a tricky problem. Using passive reduction on the other hand gives the most efficient result at high frequencies. For low frequencies, passive materials have to be thicker and massive, because they have to reduce larger wavelengths. This results in bulky and heavy passive absorbers. Therefore to decrease the overall sound, a combination of passive and active absorbers are probably the best to achieve total cancellation of the sound.

In an ANC-system there is an adaptive control filter and an adaptive algorithm. The control filter is for example an adaptive FIR\(^2\)-filter. An adaptive FIR-filter has adjustable weights, which means that it takes consideration to changes of the noise and by the adaptive algorithm update the weights after these changes.

5.2 Sound Field Measurements for the ANC-system Evaluation

Before simulation of a real ANC-system can take place, active measurements must be done. The most important things to reconsider when making these new measurements are that one have to record the sound around the truck drivers ear, and reference signals from the noise sources. The reference signals can be for example a tachometer signal or vibration signal from the engine. There also have to be control microphones or error microphones, placed on suitable positions in the truck cabin.

![Figure 5.2: The positions for the control microphones used in the evaluation of the ANC-system.](image)

The ANC-system generates a zone of reduced noise around these control microphones. The control microphones are also used when implementing an actual active system in the truck cabin, so they have to be placed on a non-disturbing area for the driver. During these measurements they were placed behind the drivers head and at the ceiling of the cabin, see figure 5.2. They record the residual noise in the cabin to the ANC-system. This residual noise is the result of the primary and secondary sound field combined together. They are used to adjust the control filter in the ANC-system to gain high reduction. To control changes in the noise for example different revolutions of the engine, the system has to be adaptive. An adaptive system has adjustable weights, which means that it updates the weights all the time, with the goal to reduce the noise from the control microphones. The control microphones are primary used to record changes in the characteristic of the noise in the cabin. For example if the revolutions have changed, the ANC-system will no longer decrease the dominating frequencies. By using the control microphones and measure the “new” noise, results in cancellation ones again.

\(^2\)Finite Impulse Response
Figure 5.3: Microphone grid of the sound field measurements for the ANC-system evaluation.

As seen in figure 5.3 the microphone layer is located at the drivers head and microphone 12 and 8, symbolizes the drivers ears. The ANC-system was implemented for 4 control microphones and those were as following: 7, 10, 11, 14, where microphones 7 and 11 are located behind the drivers head and the other two control microphones are placed at the ceiling of the cabin, above the drivers head.

Figure 5.4: Microphones in the cabin during the active measurements.

The number on the microphones in figure 5.3 symbolizes the channels on which they were connected to in I-DEAS. The first four channels are not included in the figure.

<table>
<thead>
<tr>
<th>Channel</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>1</td>
<td>Tachometer signal from the engine</td>
</tr>
<tr>
<td>2</td>
<td>Accelerometer attached to the engine</td>
</tr>
<tr>
<td>3</td>
<td>Accelerometer attached to the rear window of the cabin</td>
</tr>
<tr>
<td>4</td>
<td>Microphone placed outside the back of the cabin</td>
</tr>
</tbody>
</table>

Table 5.1: Possible reference signals for the ANC-system.

Table 5.1 displays the recordings of the four first channels in I-DEAS. These four channels in table 5.1 are the different recorded sources that can be used as a reference signal for an active system. From the tachometer signal, different sinusoidal reference signals related to the engine rpm can be generated and used by the ANC controller. The accelerometer on the engine contains the frequencies of the engines vibrations and the accelerometer on the window contains vibrations near the drivers head. Finally the microphone outside the cabin will cover the noise outside the back of the cabin. Channel 3 and 4 are placed at these particular locations because as a result of the sound maps earlier in the thesis (see Chapter 2.3 and Appendix A), where the
results were that the highest SPL was located at the rear of the cabin. Measurements were done for 1500RPM, 1900RPM and maximal RPM with a bandwidth of 1000Hz.

Beside making measurements of the noise in the cabin and recording reference signals, the control paths had to be recorded. The control path includes the characteristics of the acoustic path between loudspeaker $l$ and microphone $m$ as well as such electronic equipment as amplifiers, D/A- and A/D-converters, anti-aliasing and reconstruction filters and the loudspeaker and control microphone. The ANC-controller used in this work is based on off-line measured control paths, i.e. before the ANC-system is switched on. For an ANC-system as the one in this thesis with 4 loudspeakers and 12 microphones in the cabin, a total of 48 frequency response functions have to be measured.

![Loudspeaker positions in the cabin for the ANC-system. Loudspeaker 1 and 2 are located at the ceiling and loudspeaker 3 and 4 are located on top of the fascia panel.](image)

The control paths are recorded by generating noise through the loudspeakers and measure that noise in the microphones. To gain best estimation of the control paths, a chirp is more suitable to generate and record. Because of the deterministic characteristic, only one mean value can be enough to gain estimation. Compared to the noise where a lot of mean values have to be calculated. Unfortunately the chirp is not included as a source signal in I-DEAS and therefore could not be generated. These control paths will be applied further on in the adaptive algorithm used in the ANC-system.
5.3 Active Noise Control System

Having made all the measurements for an active system, an implementation is possible. The thesis has focused primary on the case with 1500RPM which is the revolution of the engine under normally conditions. Figure 5.6 displays a simplified figure of a feedforward active noise control system. \(x(n)\) is the reference signal, which in this thesis can be 4 different signals; tachometer signal, vibration signal from the engine, the rear window, or the exterior reference microphone signal. These references will further on be tested, to see which one is the most suitable to use. \(H(z)\) in this overview is the physical path from the reference signal or source signal into the cabin. After passing through \(H(z)\), the noise in the cabin is recorded, this is referred to \(d(n)\). By having \(x(n)\) as reference to an adaptive FIR-filter \(W(z)\), it will by adjusting the weights, subtract the output of the filter from \(d(n)\). One can also say that the output of the adaptive filter is a mirror of the primary sound field \(d(n)\) in the cabin. When these two sound fields lines up in time and space, a cancellation of them both takes place. After the substraction an error signal \(e(n) = d(n) - y(n)\) remains, which goes back into the algorithm. When the optimal filter coefficients of the adaptive FIR-filter are reached, the error signal has become minimal and smaller values on the error is not achievable.

![Figure 5.6: A simplified figure of a feedforward active noise control system.](image)

5.4 Adaptive Algorithm

Studying a truck in action, it is obvious that the noise will not have the same constant character. By increasing the RPM, immediately the character has changed. Because of this there is no possibility to use ordinary fixed digital FIR and IIR filters. Instead one can use an adaptive filter, since the adaptive filter has adjustable weights and take consideration to these changes. There are two different types of active reduction, feedforward and feedback control[9]. The one used in this thesis is the ANC feedforward control. It uses a reference signal to the feedforward controller. This signal can be a tachometer signal from the engine of the truck. The output signal from the feedforward controller is then fed to the loudspeaker with the same amplitude, but opposite phase, resulting in cancellation of the disturbing noise in the cabin. The noise that still remains is called the error signal \(e(n)\) and will be recorded in a control microphone and fed back to the active noise controller. The controller uses an adaptive algorithm to adjust the filterweights in the controller so that minimum error signal is obtained. Most of the ANC-systems minimizes the mean square value of the error signal, because this is what the human ear responds to. It has been a great development of both adaptive FIR- and IIR-filter for adaptive filtering, but the FIR-filter is the most practical and most common used. The FIR-filter has high stability, but that also depends on which algorithm that is being used. In reality adaptive feedforward control has been implemented in a SAAB 2000 aircraft to control four harmonics of the blade-passing frequency. There has also been experiment to reduce low-frequency engine noise inside cars. These adaptive algorithms depends on that an estimate of the gradient is used to receive optimal parameters. The algorithm that has been used in this thesis, is the least mean square (LMS)[9].
5.4.1 Least Mean Square (LMS)

The LMS algorithm is the far most common algorithm\[9\]. Since the algorithm is easy to implement and very robust. The LMS algorithm is a gradient based algorithm and uses a direct estimate of a cost function to be minimized. In this thesis the LMS algorithm is presented for a single loudspeaker and control microphone, then expanded to a multi-channel system consisting of several loudspeakers and control microphones.

Assume that the cost function to be minimized is given by the squared error signal $e^2(n)$,

$$\xi(n) = e^2(n).$$

The error signal $e(n) = d(n) - x^T(n)w(n)$ where the primary noise is given by $d(n)$ and the reference signal vector $x(n)$ and weight vector $w(n)$ are given by:

$$x(n) = [x(n), x(n-1), ..., x(n-L+1)]^T$$
$$w(n) = [w_0, w_1, ..., w_{L-1}]^T.$$

Here $L$ is the length of the FIR filter $w(n)$.

This results in adjustments of the weights for the adaptive filter by:

$$w(n + 1) = w(n) - \mu \widehat{\nabla} \xi(n) \quad (5.1)$$

where $\widehat{\nabla} \xi(n)$ is the gradient estimate to the cost function $\xi(n)$, $(\widehat{\nabla} \xi(n) = \frac{\partial \xi(n)}{\partial w(n)})$. The gradient “points” in the direction in which the error increases the most, and when minimizing this error, one has to go the opposite direction $(-\widehat{\nabla} \xi(n))$, which is the steepest descent. The steepest descent is the basic idea of some adaptive algorithms, which is to minimize the error. The LMS algorithm is one of the adaptive algorithms that is based on the steepest descent.

![Figure 5.7: The cost function $\xi(n) = e^2(n)$.](image)

One thing to consider when minimizing the error, is the step size $\mu$. The step size $\mu$ will affect the rate at which the weight vector moves down the bowl. It must have a positive number, else the weight will move up and increase the error. For small values on the step size the movement down the bowl will be slow, but the filter weights are updated more accurately. By increasing $\mu$ will result in faster convergence, but the negative side is that the algorithm will never reach minimum (circle around the minimum point) and the remaining error will be larger than for smaller values on $\mu$.

There is an upper limit on how large $\mu$ can be and if exceeding that limit will lead to divergence of the algorithm.

The steepest descent algorithm can be summarized as follows\[4\]:

1. Initialize the steepest descent algorithm. The weights are often set to zero.
2. Evaluate the gradient of $\xi(n)$ at the current estimate $\widehat{\nabla} \xi(n)$
3. Update the estimate at time $n$ by adding a correction that is formed by taking a step of size $\mu$ in the negative gradient direction
\[ w(n + 1) = w(n) - \mu \hat{\nabla} \xi(n) \]

4. Go back to (2) and repeat the progress.

An explicit expression of the gradient estimate is given by
\[ \hat{\nabla} \xi(n) = \frac{\partial \xi(n)}{\partial w(n)} = e(n) \frac{\partial e(n)}{\partial w(n)} = -e(n)x(n) \]  

(5.2)

where
\[ e(n) = d(n) - y(n) = d(n) - w^T(n)x(n) \]

Because of that the gradient estimate is an “unbiased”, the mean value of the gradient estimate results in the true gradient \( E[\hat{\nabla} \xi(n)] = \nabla \xi(n) \).

The general formula for the LMS-algorithm is according to the equation:
\[ w(n + 1) = w(n) + \mu e(n)x(n) \]  

(5.3)

The adaptive filter converges in the mean to the Wiener-Hopf equation
\[ w_{opt} = R^{-1} x \]

(5.4)

where the optimal solution for the weights \( w_{opt} \). Here \( R \) is a \( L \times L \) Hermitian Toeplitz matrix of autocorrelations of the reference signal \( x(n) \), \( R = E[x(n)x^T(n)] \), and \( r_{dx} \), the vector of cross-correlations between the noise signal \( d(n) \) and the reference signal \( x(n) \), \( r_{dx} = E[d(n)x(n)] \). The convergence of the adaptive filter takes place when the step size \( \mu \) is in the range:
\[ 0 < \mu < \frac{2}{\lambda_{max}} \]  

(5.5)

where \( \lambda_{max} \) is the maximum eigenvalue of the autocorrelation matrix \( R \). When the autocorrelation matrix increases, the value on \( \lambda_{max} \) also increases. This results in that with a longer adaptive controller filter, the allowed value for \( \mu \) will become smaller. A more profound way to calculate the boundaries of \( \mu \) is:
\[ 0 < \mu < \frac{2}{(L + 1)E[x^2(n)]} = \mu_{max} \]  

(5.6)

where \( E[x^2(n)] \) is the power of the reference signal \( x(n) \).

**The Filtered – x LMS Algorithm**:

As told earlier, this thesis uses an adaptive digital FIR-filter as a controller. From a filter like that the output is given by:
\[ y(n) = w^T(n)x(n) \]  

(5.7)

where
\[ x(n) = [x_n, x_{n-1}, \ldots, x_{n-L+1}]^T \]  

(5.8)

\[ w(n) = [w_0, w_1, \ldots, w_{L-1}]^T \]  

(5.9)

\( x \) is the reference signal vector and \( w \) is the filter weight vector, which has a length of \( L \). Taking consideration to the control paths, which was the transfer functions from the controller output (loudspeakers), to the error input (control microphones), the output from the filter will be affected of this with:
\[ y_{new}(n) = y^T(n)H_{cp} \]  

(5.10)

where \( y(n) \) is
\[ y(n) = [y_n, y_{n-1}, \ldots, y_{n-J+1}]^T \]  

(5.11)

which can be described as the controller output signal, \( J \) samples back. The control path vector containing \( J \) coefficients is:
\[ H_{cp} = [h_0, h_1, \ldots, h_{J-1}]^T \]  

(5.12)
The control path vector has been chosen with a length of 300 and this will be shown further on in the thesis. After the control paths have been applied, the error signals can be calculated. This is done by:

\[ e = d_{psf}(n) - y_{new}(n) \]  \hspace{1cm} (5.13)

where \( d_{psf}(n) \) = the primary sound field. The equation 5.13 can also be formulated as

\[ e(n) = d_{psf}(n) - w^T(n)x_{filtered-x}(n) \]  \hspace{1cm} (5.14)

where \( x_{filtered-x}(n) = [x_{filtered-x}(n), x_{filtered-x}(n-1), \ldots, x_{filtered-x}(n-J+1)]^T \) and each element is given by

\[ x_{filtered-x}(n) = x(n)H_{cp} \]

For an ordinary LMS algorithm the error is calculated by:

\[ e(n) = d(n) - w^T(n)x(n) \]  \hspace{1cm} (5.15)

Comparing the two different ways of the error calculation, leads to that for the filtered-x the error is calculated from the filtered reference signal, instead of the originally reference signal as is done in an ordinary LMS-algorithm. Otherwise there are no difference between the two algorithm, that is why the filtered algorithm is called the filtered-x LMS[9]. It uses the same gradient estimate and the formula for updating the filter coefficients are as following:

\[ w(n+1) = w(n) + \mu e_k x_{filtered-x}(n) \]  \hspace{1cm} (5.16)

where the algorithm is stable for a step size in the range

\[ 0 < \mu < \frac{2}{(L+1)E[x^2_{filtered-x}(n)]]} \]  \hspace{1cm} (5.17)

The equation 5.16, correspond an ANC-system consisting of a single loudspeaker and microphone. This equation can be generalized to a multi-channel ANC-system[9] consisting of \( L \) loudspeakers and \( M \) microphones, resulting in following cost function

\[ \xi(n) = \sum_{m=0}^{M} e_m^2(n) \]  \hspace{1cm} (5.18)

and the new equation for updating the filter coefficients

\[ w_l(n+1) = w_l(n) + \mu \sum_{m=0}^{M} x_{filtered-x}(n)e_m(n) \]  \hspace{1cm} (5.19)

where \( x_{filtered-x}(n) \) is the filtered reference vector, i.e the reference signal \( x(n) \) filtered with the control path between loudspeaker \( l \) and microphone \( m, H_{cp}^{lm}: \)

\[ x_{filtered-x}(n) = [x_{filtered-x}(n), x_{filtered-x}(n-1), \ldots, x_{filtered-x}(n-J+1)] \]  \hspace{1cm} (5.20)

\[ x_{filtered-x}(n) = x^T(n)H_{cp}^{lm} \]  \hspace{1cm} (5.21)
5.5 Implementation of ANC-system

5.5.1 Filter length and Step size

When using the LMS-algorithm, it is necessary to decide the filter length $L$ and the step size $\mu$. These two variables can either be experimentally found or calculated by formulas.

As shown in figure 5.8, there is a way to decide the filter length by making a figure with different filter lengths and for every filter length calculate the mean square error $E[e^2(n)]$. The figure will then display how the LMS algorithm converge for different filter lengths. The algorithm has converged when the mean square error has reached the minimum value and stabilized at that value.

The filter length that was chosen in this thesis were 200. As seen in the figure at about filter length 200, the MSE\(^3\) has stabilized.

The step size on the other hand were tested, and best results were achieved at the step size $0.5 \times \mu_{\text{max}}$, where $\mu_{\text{max}}$ is the maximal step size.

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\(^3\)Mean Square Error
5.5.2 Coherence

Having the step size and filter length, the LMS-algorithm can be tested. The most easiest way to test the LMS, is where there is no control path. If the algorithm has converged, it will decrease the frequencies where the coherence between $x(n)$ and $d(n)$ was high[9]. The coherence $\gamma_{dx}^2(f)$ is a measure of the linear relation between the recorded noise in the cabin $d(n)$ and the reference signal $x(n)$.

The coherence can be estimated by:

$$\gamma_{dx}^2(f) = \frac{|G_{dx}(f)|^2}{G_{xx}(f)G_{dd}(f)} , 0 \leq \gamma^2(f) \leq 1$$  \hspace{1cm} (5.22)

where $G_{xx}(f)$ is the auto power spectrum of the reference signal $x(n)$ and $G_{dd}(f)$ the auto power spectrum of the recorded noise in the cabin $d(n)$. Finally $G_{dx}(f)$ is the cross spectrum between the recorded noise and the reference signal[9].

![Coherence between engine acc. and mic7](image)

Figure 5.9: Coherence between engine accelerometer and the control microphone 7. In the figure coherence corresponding to the engine orders 1, 3, 12 and 24 are marked.

The figure 5.9 displays the coherence between the accelerometer attached near the engine and microphone 7 in the cabin, which is one of the two control microphones behind the drivers head. The coherence is calculated for 1500RPM. As shown in the figure, at the orders: 1, 3, 12 and 24 from the engine results in high coherence, which means that there is almost a total linear relation. A perfect linear relation has coherence equal to 1. This compares quite well to the earlier identification of the dominating frequencies. Those were order 3, 6 and 12 from the engine. If having the engine accelerometer as the reference signal into the active system, will result in that those frequencies that had high coherence in the figure above will decrease the most, if the control path would be equal to one.
During the earlier examinations of the noise in the truck cabin showed that the sound in the cabin consisted of mostly broadband noise. Because the engine accelerometer only records the distinct vibrations from the engine, another accelerometer was placed on the window behind the drivers head. Using this accelerometer as a reference, it may result in a satisfying decrease of the broadband noise. To view how much of the vibrations in the window exists in the cabin noise, the coherence between the accelerometer and the control microphone 7 were calculated.

The result of the coherence between the window accelerometer and the same control microphone as in figure 5.9 is displayed in figure 5.10. When comparing the two coherence figures, the difference between them is that the accelerometer attached to the window gives high coherence with more frequencies than the engine accelerometer. In figure 5.10 there is a lot of the broadband noise that has quite high coherence with the window accelerometer, compared to figure 5.9 where there are a few distinct frequencies that has high coherence. For a realization of an ANC-system, the accelerometer attached to the window would probably be most suitable. Because if there can be a decrease of the broadband noise, the total effect would likely be more obvious to the human ear, than decreasing the frequencies of the engine.
5.5.3 ANC without control path

When the ANC-system was implemented, it was examined in different aspects, so that the results given from the system were credible. This has to be controlled before applying the control path to the ANC-system. Due to lack of time during the thesis, only realization of the MIMO\textsuperscript{4} systems with 2 speakers and 2 control microphones, 2 speakers and 4 control microphones were done. If there had been more time, then realization of ANC-system with 4 speakers and 4 control microphones could have been done. The larger ANC system that is being used, the larger zone of sound field can be covered and decreased, larger a resulting in that zone of reduced noise is obtained.

Figure 5.11: ANC-system consisting of 2 loudspeakers and 2 control microphones (2 × 2 system).

<table>
<thead>
<tr>
<th>d1</th>
<th>Control microphone 7</th>
</tr>
</thead>
<tbody>
<tr>
<td>d2</td>
<td>Control microphone 11</td>
</tr>
<tr>
<td>e1</td>
<td>Error signal at microphone 7 from ANC</td>
</tr>
<tr>
<td>e2</td>
<td>Error signal at microphone 11 from ANC</td>
</tr>
</tbody>
</table>

Table 5.2: Signal description for the 2 × 2 system.

In figure 5.11 the signals y11, y12, y21 and y22 symbolizes the control paths between each loudspeaker and each microphone. During this section of the thesis, these signals are equal to one, but further on they will be applied with the measured control paths.

\textsuperscript{4}Multiple Input Multiple Output\cite{9}
The basic thing to examine, if the implemented ANC-system worked, was to study the recorded noise \( d(n) \) (uncontrolled noise) and the error \( e(n) \) (which is the noise after the active reduction) in the time-domain. All the examinations during the implementation was done for both with the window accelerometer as a reference signal and the engine accelerometer as reference signal.

Figure 5.12: Time-domain of \( d_1 \) (black, ANC off) and \( e_1 \) (red, ANC on) with engine accelerometer as reference signal.

Figure 5.13: Time-domain of \( d_1 \) (black, ANC off) and \( e_1 \) (red, ANC on) with window accelerometer as reference signal.

As can be seen in both figures above there are a big difference between them. The clearly best result is given when having the window accelerometer as reference signal. Only figures with \( d_1 \) and \( e_1 \) are shown and examined in the text, all other figures are attached in the appendix.
A more profound way to examine the ANC-system is to study the auto spectrum of $d_1$ and $e_1$, for the two different references. The result of the auto spectrum, at the frequencies where the coherence is high (see figures 5.9 and 5.10), would lead to that $e_1$ should be reduced with some decibels from the originally noise signal $d_1$, as shown in figures 5.14 and 5.15 below.

![Figure 5.14: Auto spectrum of $d_1$ (black, ANC off) and $e_1$ (red, ANC on) with engine accelerometer as reference signal.](image1)

![Figure 5.15: Auto spectrum of $d_1$ (black, ANC off) and $e_1$ (red, ANC on) with window accelerometer as reference signal.](image2)

As can be seen when comparing the figures above, with the coherence figures, the frequencies with high coherence, are also the ones which decreases the most. If the estimate of the control path would be good, which means that it will have high linear transference between the loudspeakers and control microphones, then the auto spectrum above would look almost the same for applied control path. This will however not be the case in reality and further on in the thesis also shown.
5.5.4 ANC with control path

Figure 5.16 displays the scheme of the ANC-system, when the control path has been implemented.

![Figure 5.16: Scheme over the ANC-system with control path.](image)

The control path is represented as $c$ in the figure. When applying the path to the system, the reference signal into the algorithm also has to be filtered with the path. This algorithm is generally recalled as filtered-x LMS.

In the computer evaluation of the ANC-system $c = \hat{c}$.

When applying the control path to the implemented and fully functional ANC-system, there are a few things to consider. The length of the control path has to be decided. This is done by studying the impulse response and examining when it has stabilized at zero. In figure 5.17 the control path between speaker 1 in the cabin and control microphone 7 is shown.

![Figure 5.17: Impulse response of the control path between control microphone 7 and loudspeaker 1.](image)

The length of the control path has been selected to 300. This value can be reduced, but at 300 the impulse response has certainly stabilized.
Besides the impulse response, the coherence of the control path can be displayed. The coherence of the control paths shows the coherence between the loudspeakers and the microphones. Figure 5.18 displays the coherence of the control path (which also can be described as the linear relation) between control microphone 7 and loudspeaker 1.

![Coherence between speak1/mic7](image)

Figure 5.18: Coherence of the control path between control microphone 7 and loudspeaker 1.

A perfect relation shall give a coherence around one for every frequency, but this is not the case in the figure. There are many explanations of the not so high coherence. For example the recordings of the control path, have been effected with noise from the surroundings in the building where the truck was located. Another aspect of the coherence could be that the loudspeakers used in the cabin, only covered frequencies above around 60Hz. Frequencies below this area will give very bad coherence, which can be seen in figure 5.14. Maybe if the estimations of the control paths were done with a generated chirp, results would be better. But due to the broadband noise, the chirp would probably not result in a much bigger decrease than the one in this thesis with generated noise.
Further on the same examinations as in section 5.5.3 can be done, which is to display figures of the signal before and after the ANC-system in the time- and frequency-domain. For a very good estimation of the control paths, result in section 5.5.3 can be reached. But after studying the control paths it was obvious that this will not be the case.

Figure 5.19: Time-domain of $d_1$ (black, ANC off) and $e_1$ (red, ANC on) with engine accelerometer as reference signal.

Figure 5.20: Time-domain of $d_1$ (black, ANC off) and $e_1$ (red, ANC on) with window accelerometer as reference signal.

When comparing the figures above to the time-domain figure in section 5.5.3, there are a big difference. The result with the applied control path is considerable worse than without the path.
Also figures of the auto spectrum has to be displayed, to see the effect from the control path.

![Auto spectrum of $d_1$ (black, ANC off) and $e_1$ (red, ANC on) with engine accelerometer as reference signal.](image)

**Figure 5.21:** Auto spectrum of $d_1$ (black, ANC off) and $e_1$ (red, ANC on) with engine accelerometer as reference signal.

![Auto spectrum of $d_1$ (black, ANC off) and $e_1$ (red, ANC on) with window accelerometer as reference signal.](image)

**Figure 5.22:** Auto spectrum of $d_1$ (black, ANC off) and $e_1$ (red, ANC on) with window accelerometer as reference signal.

Studying the auto spectrum and comparing them to the ones in section 5.5.3, there can be seen that the lower frequencies have not been decreased at all. This comes from the effect of the control path and the bad coherence at these frequencies. To reduce these frequencies, the loudspeakers have to be exchanged with ones that covers frequencies of about 30Hz and higher.
Chapter 6

Result and Summary

A summary of every part of this report will be that:

In the first part of the thesis, where the sound field measurements and analysis of those measurements were done, led to that the recorded noise contained mostly of broadband noise. The measurements were made for 1500RPM, 1900RPM and maximal RPM. From the sound maps, there could be seen that the highest SPL of the noise were located in the rear of the cabin.

Then an identification of the most dominating frequencies was performed, and gave the result that orders 3, 6, and 12 from the engine and order 13 from the hydraulic pump were the sources that contributed to those frequencies and existed for every different number of revolutions.

The next part in the report were the active narrow- and broadband approximations. Those approximations gave a hint on how results of an actual ANC-system would be. The results were that for both active narrow- and broadband approximation, a decrease of 3dBA was possible.

Then due to the earlier sound maps, the highest SPL were at the rear corners in the cabin, passive noise reduction was applied at those corners. The passive reduction gave a big decrease of the SPL in the sound field close to the floor of the cabin. But the passive materials that were implemented in the cabin, did not affect the sound field at the level where a truck drivers head would be.

Finally the ANC-system was simulated in MATLAB. Results of the simulation were quite satisfying before the control paths were applied to the system. The best reference signal of the ones measured in the thesis, were the accelerometer attached to the rear window.

To sum up this master thesis one can say that active noise control does not have any greater effect of the disturbing sound in the truck cabin. But maybe it can be used as a higher quality cabin than a normal one. If Kalmar Industries AB, decides to aim for an implementation of an ANC-system, then there are a few things that have to be improved.

For once, and probably the thing that will contribute to the greatest improvement are the control paths. To estimate better control paths, a chirp is probably necessary to use as a signal during the measurements of the paths. The other thing to gain better control paths, is the loudspeakers. These speakers that was used in this thesis and those that Kalmar industries AB have implemented in their cabins are far to poor for use in an ANC-system. There probably has to be an investment in really special loudspeakers that covers very low frequencies and can generate noise of high SPL for those frequencies.

Another thing to improve is the source signal, or the reference signal to the ANC-system. The result in this thesis is clearly, that the best reference signal, was the accelerometer attached to the window. But to gain better results, there has to be found a reference signal, that has a higher correlation with the disturbing sound field in the cabin, than the ones recorded in this thesis. Due to the time limitation for the thesis, this was not possible to do. Another aspect of gaining better results, is to use multiple reference signals, but this will lead to a more complex ANC-system and control algorithms.
Bibliography


Appendix A

A-weighted sound maps

A.1 A-weighted sound maps for 1500RPM
A.1.1 Layer 1

Figure A.1: Sound map in dBA of layer 10cm above cabin floor.
A.1.2 Layer 2

Figure A.2: Sound map in dBA of layer 30cm above cabin floor.

A.1.3 Layer 3

Figure A.3: Sound map in dBA of layer 50cm above cabin floor.
A.1.4 Layer 4

Figure A.4: Sound map in dBA of layer 70cm above cabin floor.

A.1.5 Layer 5

Figure A.5: Sound map in dBA of layer 90cm above cabin floor.
A.1.6 Layer 6

Figure A.6: Sound map in dBA of layer 110cm above cabin floor.

A.1.7 Layer 7

Figure A.7: Sound map in dBA of layer 130cm above cabin floor.
A.2 A-weighted sound maps for 1900RPM

A.2.1 Layer 1

Figure A.8: Sound map in dBA of layer 10cm above cabin floor.

A.2.2 Layer 2

Figure A.9: Sound map in dBA of layer 30cm above cabin floor.
A.2.3 Layer 3

Figure A.10: Sound map in dBA of layer 50cm above cabin floor.

A.2.4 Layer 4

Figure A.11: Sound map in dBA of layer 70cm above cabin floor.
A.2.5 Layer 5

Figure A.12: Sound map in dBA of layer 90cm above cabin floor.

A.2.6 Layer 6

Figure A.13: Sound map in dBA of layer 110cm above cabin floor.
Appendix A. A-weighted sound maps

A.2.7 Layer 7

Figure A.14: Sound map in dBA of layer 130cm above cabin floor.

A.3 A-weighted sound maps for maximal RPM

A.3.1 Layer 1

Figure A.15: Sound map in dBA of layer 10cm above cabin floor.
A.3.2 Layer 2

![Sound map in dBA of layer 30cm above cabin floor.](image)

Figure A.16: Sound map in dBA of layer 30cm above cabin floor.

A.3.3 Layer 3

![Sound map in dBA of layer 50cm above cabin floor.](image)

Figure A.17: Sound map in dBA of layer 50cm above cabin floor.
A.3.4 Layer 4

Figure A.18: Sound map in dBA of layer 70cm above cabin floor.

A.3.5 Layer 5

Figure A.19: Sound map in dBA of layer 90cm above cabin floor.
A.3.6 Layer 6

Figure A.20: Sound map in dBA of layer 110cm above cabin floor.

A.3.7 Layer 7

Figure A.21: Sound map in dBA of layer 130cm above cabin floor.
Appendix B

A-weighted Sound maps after Passive Noise Reduction

B.1 A-weighted sound maps for 1500RPM after passive noise reduction

B.1.1 Layer 1

Figure B.1: Sound map in dBA of layer 10cm above cabin floor after passive noise reduction.
Appendix B. A-weighted Sound maps after Passive Noise Reduction

B.1.2 Layer 2

Figure B.2: Sound map in dBA of layer 110cm above cabin floor after passive noise reduction.

B.2 A-weighted sound maps for 1900RPM after passive noise reduction

B.2.1 Layer 1

Figure B.3: Sound map in dBA of layer 10cm above cabin floor after passive noise reduction.
B.2.2 Layer 2

![Sound map in dBA of layer 110cm above cabin floor after passive noise reduction.](image)

Figure B.4: Sound map in dBA of layer 110cm above cabin floor after passive noise reduction.

B.3 A-weighted sound maps for maximal RPM after passive noise reduction

B.3.1 Layer 1

![Sound map in dBA of layer 10cm above cabin floor after passive noise reduction.](image)

Figure B.5: Sound map in dBA of layer 10cm above cabin floor after passive noise reduction.
Figure B.6: Sound map in dBA of layer 110cm above cabin floor after passive noise reduction.
Appendix C

Active Noise Control with 2 loudspeakers and 4 control microphones for 1500RPM

C.1 Engine accelerometer as reference signal

![Graph](attachment:image.png)

Figure C.1: Auto spectrum of $d_1$ (black, ANC off) and $e_1$ (red, ANC on) which is the control microphone 7.
Figure C.2: Auto spectrum of $d_2$ (black, ANC off) and $e_2$ (red, ANC on) which is the control microphone 11.

Figure C.3: Auto spectrum of $d_3$ (black, ANC off) and $e_3$ (red, ANC on) which is the control microphone 10.
Appendix C. Active Noise Control with 2 loudspeakers and 4 control microphones for 1500RPM

C.2 Window accelerometer as reference signal

Figure C.4: Auto spectrum of $d_4$ (black, ANC off) and $e_4$ (red, ANC on) which is the control microphone 14.

Figure C.5: Auto spectrum of $d_1$ (black, ANC off) and $e_1$ (red, ANC on) which is the control microphone 7.
Figure C.6: Auto spectrum of $d_2$ (black, ANC off) and $e_2$ (red, ANC on) which is the control microphone 11.

Figure C.7: Auto spectrum of $d_3$ (black, ANC off) and $e_3$ (red, ANC on) which is the control microphone 10.
Figure C.8: Auto spectrum of $d_4$ (black, ANC off) and $e_4$ (red, ANC on) which is the control microphone 14.