Acoustic Echo Cancelling with Microphone Arrays

by

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Abstract

This report describes a novel method to perform acoustic echo cancelling with microphone arrays. The method employs a digital self-calibrating microphone system. The on-site calibration process is a simple indirect calibration which adapts in each specific case to the environment and electronic equipment. The method also continuously takes into account environmental disturbances such as car engine noise and fan noise. The method is primarily aimed at handsfree mobile telephones, by suppressing the handsfree loudspeaker and car noise simultaneously. The report also contains an extensive evaluation in a car.
Chapter 1

Introduction

Echo related problems are very common in telephone systems. Speech originates from the far-end speaker and echoes back, see figure 1.1, with a time delay causing perception problems. Perception will be further impaired when the speaker is situated in a noisy car in handsfree mode.

A method to decrease the echo from the far-end speaker during handsfree communication is echo cancellation (EC) [1], and a considerable amount of effort has been spent in this field. Most work has been devoted to single microphone solutions [2], although some papers propose systems with two microphones [3, 4]. The acoustic echo canceller which is introduced in this paper is not based on conventional array theory [5, 6], but relies on an indirect calibration [7]. The system is aimed at mobile hands-free communication systems, and it therefore also takes into

![Diagram of two-way, handsfree communication, conversation between the far- and near-end speaker.](image)
account the near field in a small enclosure. The near field is difficult to describe in an a priori model; and this is the reason for employing gathered information from the real target and jammer positions. Obviously these gathered signals contain useful information about the acoustic environment as well as electronic equipment, such as microphones, amplifiers, A/D-converters and anti-aliasing filters. Other important information is microphone element geometry and other spatial and spectral information inherent in the gathered target and jammer signals. Instead of calculating a large number of statistical a priori information from the gathered data, we will merely use the signals as they appear during the operating phase.

**Advantages of the new microphone array EC-method:**

- Allows spatial selection/suppression of targets/jammers.
- Allows simple on-site calibration.
- Allows standard quality microphones.
- Allows arbitrarily choice of placement and number of microphones.
- Well suited to handsfree mobile communication systems.
- Adapts to the actual acoustic environment.
- Controls speech distortion.
- Robust.
Chapter 2

Working Scheme for the Echo Cancelling Adaptive Beamformer

The working scheme for the beamformer can be divided into two phases: phase 1, which is the gathering phase; and phase 2, which is the continuous filtering and adaptation phase. During phase 2 the beamformer utilizes the calibration signals gathered in phase 1.

2.1 Phase 1 - The Gathering Process

The gathering phase takes place on-site in the actual environment by emitting representative sequences from each jammer and target position. Array signals for each sequence and channel are memorized in a digital memory for subsequent use.

For the target signal, this can be performed by letting a loudspeaker emit coloured noise, or by letting the near-end speaker read a representative sequence from a desired position in the car (see figure 2.1). The procedure is repeated for the handsfree loudspeaker (see figure 2.2). All signals are also stored in the memory. The gathered, multi-channel, memorized signals will later be used to form the input and reference for the echo canceller in the operating phase. Since a-priori modelling almost always leads to the loss of some information, the main idea in this straight-forward solution is that the calibration signals themselves “is the best tutor.” The calibration signals contain information on the acoustical environment, variations in the electrical equipment and spatial and frequency responses.

During and after the calibration phase some restrictions are, however, imposed:

- The microphone elements and placement can be chosen arbitrarily, but must not be altered or moved unless a new calibration is made.
Figure 2.1: Target calibration signals recording.
Acoustic Echo Cancelling with Microphone Arrays.

Figure 2.2: Jammer calibration signals recording.
• The equipment must be time invariant.
• A fair signal to noise ratio during gathering is necessary.

2.2 Calibration signals

The gathered calibration signals must arrive from the desired/unwanted (target or jammer) positions, and the signals must contain approximately the same spectral content as a real target/jammer signal. There are different methods to facilitate this. A very simple approach is to collect human utterances from the target position. A different kind of calibration procedure is to place loudspeakers in the desired/unwanted positions and let the loudspeakers emit, one at a time, flat or coloured noise, or gathered speech.

2.3 Phase 2 - The Operating Mode

A telephone conversation can be divided into four modes: Idle, Receive, Transmit and Double Talk mode.

2.3.1 Idle/I-mode

During the I-mode only car noise exists, i.e. there is no speech from the near or far-end speaker.

<table>
<thead>
<tr>
<th>Near-end speaker:</th>
<th>quiet</th>
</tr>
</thead>
<tbody>
<tr>
<td>Far-end speaker:</td>
<td>quiet</td>
</tr>
<tr>
<td>Upper beamformer:</td>
<td>Copy of LB</td>
</tr>
<tr>
<td>Lower beamformer:</td>
<td>Adaptation on</td>
</tr>
</tbody>
</table>

Environmental noise impinges on each of the microphone elements in the microphone array, and is added with virtual near- and far-end speech signals, i.e. the memorized signals. Note that the noise signals have passed through the same electronic equipment as the memorized signals; and the virtual gathered speech signals represent a speaker talking from the correct environment/position and a handsfree loudspeaker, gathered with high signal- to noise-ratios. These three sets of signals are mixed and comprise the inputs to the adaptive filters. The desired signal for the adaptive filters is formed by a suitable combination of the memorized target signals only. The adaptive algorithm, in this case a Least Mean Square (LMS) algorithm, now has access to all information needed to adapt the filters. It is capable of enhancing the near-end speaker, and suppressing the far-end acoustic echo and car noise both in the spatial- and frequency-domain.
The coefficients are continuously copied into a fixed beamformer/filter which produces the output. Note that the inputs to this beamformer contain only signals coming from the microphone elements.

### 2.3.2 Transmit/TX-mode

During the TX-mode only near-end speech and car noise are present.

- Near-end speaker: **talking**
- Far-end speaker: **quiet**
- Upper beamformer: Fixed
- Lower beamformer: Adaptation off

When near-end speech is detected, the adaptation is turned off. Incoming microphone signals from the microphone array are now processed by the fixed upper beamformer, using the latest filter coefficients adapted to the latest actual situation. The filter coefficients also suit in all probability the actual disturbance situation.

### 2.3.3 Receive/RX-mode

During the RX-mode, only far-end loudspeaker signals and car noise are present.

- Near-end speaker: **quiet**
- Far-end speaker: **talking**
- Upper beamformer: Copy of LB
- Lower beamformer (LB): Adaptation on

The algorithm behaviour is similar to the I-mode described above. The main difference is that the microphone signals in this mode consist of real noise and speech from the far-end speaker.

### 2.3.4 Double Talk/DT-mode

During the DT-mode, near- and far-end speech as well as real noise are present.

- Near-end speaker: **talking**
- Far-end speaker: **talking**
- Upper beamformer: Fixed
- Lower beamformer: Adaptation off

In double talk mode, no adaptation is made; the beamformer coefficients are fixed, and the output from the upper beamformer is transmitted.
2.4 Calibration Signal Levels

During phase 2 the balance (see figure 2.3) between the memorized inputs and microphone input to the adapting beamformer, can be controlled by:

1. Factor $\alpha \Rightarrow$ Near-end memorized speech signal amplification/attenuation.
2. Factor $\beta \Rightarrow$ Far-end memorized speech signal amplification/attenuation.
3. Factor $\gamma \Rightarrow$ Incoming environment noise amplification/attenuation for signals from the microphone array.

The mix between the components will control the adaptive filter suppression or amplification of sources both in frequency and spatial domain. A large value of $\gamma$ and $\beta$ versus $\alpha$ will cause the adaptive filter to emphasize cancellation of both the far-end speech and/or car noise. However, a too large suppression will yield degradation of the near-end speech signal. A suitable choice of $\alpha$, $\beta$ and $\gamma$ are values which vary around the true signal levels with a maximum of $\pm 10dB$. Extreme choices of $\alpha$, $\beta$ and $\gamma$ can possibly have advantages in extreme situations.
Figure 2.3: Balance between inputs during phase 2.
Chapter 3

Evaluation Conditions

3.1 Car Environment

The performance evaluation of the acoustic echo cancelling beamformer was made in a Volvo 940 GL station wagon. Data have been gathered on a multichannel DAT-recorder with a sample rate at 12 kHz, and with 5 kHz bandwidth. In order to facilitate simultaneous driving and recording, a loudspeaker mounted in the passenger seat to simulate a normal speaker.

3.2 Microphone configurations

Two microphone brands were evaluated: a standard hands-free mobile telephone microphone Ericsson RLC 509 11/03 RA, and a high quality microphone from Sennheiser. The Ericsson microphones were mounted flat on the visor, while the Sennheiser microphones were mounted 20 mm below the visor with a fixture.

The distance between the speaker position and microphones was 330 mm (Ericsson) and 350 mm (Sennheiser) respectively. Two different microphone geometries were evaluated.

- Linear array, see figures 3.1, 3.2, 3.3
- Non-linear array, see figures 3.4, 3.5, 3.6
Figure 3.1: Linear microphone geometry with six microphones. The distance between elements is 50 mm.

Figure 3.2: Placement of the Ericsson microphones, linear mount.

Figure 3.3: Placement of the Sennheiser microphones, linear mount.
Figure 3.4: Non-Linear microphone geometry, with six microphone. The distance between the element is 50 mm.

Figure 3.5: Placement of the Ericsson microphones, non-linear mount.

Figure 3.6: Placement of the Sennheiser microphones, non-linear mount.
Chapter 4

Implementation

The data gathered from the multichannel DAT recorder was converted into Matlab format. The evaluation part of the report is based on this data, and processed by Matlab. The results will be compared with an exhaustive real-time DSP evaluation in a later report. The main difference is the fix-point implementation on the real-time DSP-system, the DSP 900.

4.1 Algorithm Implementation

The main part of the algorithm is written in Matlab code and has been divided into two phases: the adapting phase and filtering phase. The adapting phase is the most time-consuming part, and is thus written in C and compiled into C-Mex\(^1\) file format. Calculations has been carried out on Sun Sparc Station 20. During the calculation process most of the time is taken up by overhead for the workstation handling large amounts of data. The adaptation phase requires for four seconds of data, with six parallel microphone channels and concurrent LMS algorithms updating 256 taps each, approximately 90 sec. The corresponding FIR-filtering phase, however, needs only about 4 seconds.

4.2 Real Time Implementation

The adaptive Acoustic Echo Canceller is also implemented on a multi-processor TMS 320C25 DSP system, the DSP 900 [8]. The DSP 900 system is equipped with eight TMS 320C25 DSPs which in this implementation the system is configured to serve six A/D converters and four D/A converters simultaneously. The system is controlled from an ordinary Lap Top type PC, and acts as a host computer and

\(^1\)MEX Stands for Matlab Executable and are a dynamically linked subroutine which can be automatically loaded and executed from the Matlab interpreter.
operator interface. Real-time evaluation from this system will be presented in a later report.
Chapter 5

Evaluation

Performance evaluation from the Volvo is illustrated in Appendix A figs. A.1-A.55

Common information for all plots is:

- Near-end speech, coming from the target position is denoted “Speaker”.
- Far-end speech echo, i.e. handsfree loudspeaker or jammer speech, is denoted “Echo”.
- The calibration signals are either Flat Noise, Coloured noise or Human Speech.
- All sequences are 7 seconds, and optionally merged together, i.e. a new sequence starts at 0,7,14... seconds
  - The near-end speaker is active from 1.5 to 3.5 seconds
  - The far-end speaker is active from 4.5 to 6.5 seconds
  - For the remaining time, only background noise is present unless otherwise declared.
- All figures begin with a 7 second sequence with an unadapted single microphone signal, i.e. a plain unfiltered single channel microphone signal.
- The results are presented as short time (20 ms) power estimates in $dB$.
- All signals are within telephone bandwidth (300-3400 Hz).

Before each evaluation, there is a table containing information on the test and data, such as number of microphones, microphone configurations, environmental conditions, as well as calibration and evaluation conditions.

During phase 2, the Memory-Signal-to-Interference Ratio $MSIR$ is defined by:
\[ MSIR = 10 \log \left( \frac{\sum_{m=0}^{M} \sum_{n=1}^{N} (\alpha \cdot s_{T_m}[n])^2}{\sum_{m=0}^{M} \sum_{n=1}^{N} (\beta \cdot s_{J_m}[n])^2} \right), \]

where \( s_T \) and \( s_J \) denote the memorized calibration signals for target and jammer respectively. In the handsfree mode this corresponds to the speaker, usually placed in the front seat; and the handsfree loudspeaker, directed towards the driver. The number of microphones in the array \( M \) is in the interval \([1 \ldots 6]\), while the number of samples, \( N \), is \([1 \ldots 48000]\) in the memorized calibration sequence.

In addition, the \( MSNR \) for the calibration phase is defined as:

\[ MSNR = 10 \log \left( \frac{\sum_{m=0}^{M} \sum_{n=1}^{N} (\alpha \cdot s_{T}[n])^2}{\sum_{m=0}^{M} \sum_{n=1}^{N} (\gamma \cdot s_{D}[n])^2} \right), \]

where \( s_T \) and \( s_D \) denote the memorized calibration signals for the target and the actual ambient disturbances. In a car environment \( s_D \) corresponds to environmental car noise which in this evaluation includes radio music, fan noise or noise from a turned down side window.

Observe that during the filtering phase, in the evaluation, the true signal levels in the car were used, with approximately \( SNR = 0 dB \) and \( SIR = 0 dB \) for flat noise. These are always the nominal values in this evaluation.

The sound files corresponding to all figures in the following chapters are available in .mat- and .wav- format from http://www.hk-r.se/ish5/

### 5.1 Calibration Process - Amount of Filter Taps

In this section, the number of filter taps in the filters relative to degree of suppression is presented (see figs. A.1-A.8). The analysis is performed using Ericsson and Sennheiser microphones using two and six microphones. The evaluation is performed with the sample rate 12000 Hz. Note that with a 8000 Hz implementation, the number of taps could be reduced by 30%.

<table>
<thead>
<tr>
<th>No. of microphones:</th>
<th>2, 6</th>
</tr>
</thead>
<tbody>
<tr>
<td>Geometry:</td>
<td>Linear</td>
</tr>
<tr>
<td><strong>Amount of taps:</strong></td>
<td>16, 32, 64, 128, 256, 512, 1024, 2048</td>
</tr>
<tr>
<td>Calibration Signals:</td>
<td>flat noise</td>
</tr>
<tr>
<td>Calibration signal level ( MSIR ):</td>
<td>0 dB</td>
</tr>
<tr>
<td>Adaptation coefficient ( \mu ):</td>
<td>0.01</td>
</tr>
</tbody>
</table>
In this evaluation, the acoustic echo suppression is shown for 2, and 6 microphones. Both the Ericsson and Sennheiser implementations attain their “optimum” with 256 filter taps. The suppression of the handsfree loudspeaker using 2 microphones is 18 dB (Ericsson) and 17 dB (Sennheiser). When using 6 microphones, the improvement is 24 dB (Ericsson), and 19 dB (Sennheiser).

5.2 Different Calibration Process Signals

The ability to reduce the influence of far-end speech echo for different choices of calibration signals has also been evaluated (see figs. A.9-A.10).

- White Noise: Band limited white noise. Band limit 5000Hz.
- Coloured Noise: Band limited speech coloured white noise. Band limit 5000Hz.
- Mixed human speech: Sentences added from a human speaker.

<table>
<thead>
<tr>
<th>No. of microphones:</th>
<th>6</th>
</tr>
</thead>
<tbody>
<tr>
<td>Geometry</td>
<td>Linear</td>
</tr>
<tr>
<td>Amount of taps:</td>
<td>256</td>
</tr>
<tr>
<td><strong>Calibration Signals:</strong></td>
<td><strong>flat noise</strong></td>
</tr>
<tr>
<td></td>
<td><strong>coloured noise</strong></td>
</tr>
<tr>
<td></td>
<td><strong>mixed human speech</strong></td>
</tr>
<tr>
<td>Calibration signal level $MSIR$:</td>
<td>0 dB</td>
</tr>
<tr>
<td>Adaptation coefficient $\mu$:</td>
<td>0.01</td>
</tr>
</tbody>
</table>

All calibration signals yield good suppression of the handsfree loudspeaker. For the Ericsson microphone, the best choice of calibration signal appears to be flat noise; while the Sennheiser microphones perform best when using speech coloured noise.

5.3 Calibration Process - Calibration Signal Level

The mix between the calibration components will cause the adaptive filter to amplify or suppress sources differently both in frequency and spatial domain. In this section the $MSIR$ value is evaluated (see figs. A.11-A.14).
<table>
<thead>
<tr>
<th>Part B</th>
</tr>
</thead>
</table>

### Table 1: Microphone Setup Parameters

<table>
<thead>
<tr>
<th>Parameter</th>
<th>Value</th>
</tr>
</thead>
<tbody>
<tr>
<td>No. of microphones:</td>
<td>2 and 6</td>
</tr>
<tr>
<td>Geometry:</td>
<td>Linear</td>
</tr>
<tr>
<td>Amount of taps:</td>
<td>256</td>
</tr>
<tr>
<td>Calibration Signals:</td>
<td>flat noise</td>
</tr>
<tr>
<td>Calibration signal level $MSIR$:</td>
<td>-15 dB, -5 dB, 0 dB, +5 dB, +15 dB</td>
</tr>
<tr>
<td>Adaptation coefficient $\mu$</td>
<td>0.01</td>
</tr>
</tbody>
</table>

In this evaluation it is necessary to examine subjectively the results by listening. A $MSIR < 0$ dB will give a notable degradation of the near-end speech. When $MSIR = 0$, the degradation of the near-end speech is low, and the suppression of the handsfree loudspeaker is considerable. We have chosen this as a subjective optimum.

### 5.4 Convergence of the Adaptive Filter vs. Adaptation Coefficient

In this section, the influence of the adaptation coefficient $\mu$ of a Normalized-LMS algorithm used during phase 1, is evaluated (see figs. A.15-A.16).

<table>
<thead>
<tr>
<th>Parameter</th>
<th>Value</th>
</tr>
</thead>
<tbody>
<tr>
<td>No. of microphones:</td>
<td>6</td>
</tr>
<tr>
<td>Geometry:</td>
<td>Linear</td>
</tr>
<tr>
<td>Amount of taps:</td>
<td>256</td>
</tr>
<tr>
<td>Calibration Signals:</td>
<td>flat noise</td>
</tr>
<tr>
<td>Calibration signal level $MSIR$:</td>
<td>0 dB</td>
</tr>
<tr>
<td>Adaptation coefficient $\mu$</td>
<td>0.001-0.08</td>
</tr>
</tbody>
</table>

The results for Ericsson and Sennheiser are similar. With a $\mu < 0.01$ the adaptation is too slow and for $\mu > 0.01$ stability problems can occur. The default value $\mu = 0.01$ is a good choice in this evaluation to ensure robust convergence.

### 5.5 Microphone Array Configuration

In this section, different subsets of the two array geometries, linear and non-linear, are evaluated (see figs. A.17-A.44).
Acoustic Echo Cancelling with Microphone Arrays.

**Number of Microphones and Microphone Geometry**

<table>
<thead>
<tr>
<th>No. of microphones</th>
<th>1, 2, 3, 4, 5, 6 (subsets of Geometry)</th>
</tr>
</thead>
<tbody>
<tr>
<td>Geometry</td>
<td>Linear and Non-Linear</td>
</tr>
<tr>
<td>Amount of taps</td>
<td>256</td>
</tr>
<tr>
<td>Calibration Signal</td>
<td>flat noise</td>
</tr>
<tr>
<td>Calibration signal level $MSIR$</td>
<td>0 dB</td>
</tr>
<tr>
<td>Adaptation coefficient $\mu$</td>
<td>0.01</td>
</tr>
</tbody>
</table>

The evaluation indicates that the important factors are:

1. The array aperture.
2. Number of microphones.

We observe that 3 microphones seems to be sufficient. This is, however, only true when judging from short time power estimates. The speech quality and distortion are subjectively further improved when the number of microphones is increased.

### 5.6 Disturbance Situations

Four different disturbance situations have been evaluated (see figs. A.45-A.55):

1. Car noise. Car engine noise, tire noise and wind noise transmitted into the coupé (90km/h).
2. Car noise and fan noise.
3. Car noise and music.
4. Car noise and side window down

<table>
<thead>
<tr>
<th>No. of microphones</th>
<th>2 and 6</th>
</tr>
</thead>
<tbody>
<tr>
<td>Geometry</td>
<td>Linear and Non-Linear</td>
</tr>
<tr>
<td>Amount of taps</td>
<td>256</td>
</tr>
<tr>
<td>Calibration Signal</td>
<td>flat_noise</td>
</tr>
<tr>
<td>Calibration signal level $MSIR$</td>
<td>0 dB</td>
</tr>
<tr>
<td>Calibration signal level $MSNR$</td>
<td>0 dB</td>
</tr>
<tr>
<td>Disturbance Situation</td>
<td>Car noise (90km/h)</td>
</tr>
<tr>
<td></td>
<td>Car noise (90km/h) and fan noise</td>
</tr>
<tr>
<td></td>
<td>Car noise (90km/h) and music</td>
</tr>
<tr>
<td></td>
<td>Car noise (90km/h) and side window down</td>
</tr>
<tr>
<td>Adaptation coefficient $\mu$</td>
<td>0.01</td>
</tr>
</tbody>
</table>
The array is surprisingly good at maintaining the acoustic echo cancelling ability even in the presence of environmental disturbance. In addition, the array suppresses the environmental disturbances by $7 - 10 \, \text{dB}$. 
Chapter 6

Summary and Conclusions

A troublesome part of microphone array implementations is the calibration phase. The on-site calibrated acoustic echo canceller in this report employs a self-calibrating process, which does not rely on any a priori model. The acoustic echo canceller gives excellent suppression of the handsfree loudspeaker in a car compartment, as well as at the same time substantial suppression of environmental disturbance. The placement of the microphones seems to be an important factor, while the microphone quality is less important.

The acoustic echo canceller yields good suppression (24\(dB\) with 6 microphones and 256 filter taps) of the handsfree loudspeaker in a car environment. During situations when environmental noise, such as car noise, also exists, the handsfree suppression is reduced only slightly. At the same time, the suppression of the background noise is significant (7 \(- 10dB\)). Important considerations include:

- Good suppression of the handsfree loudspeaker is reached with only two microphones (19\(dB\) with 2 microphones and 256 filter taps).
- Target distortion decreases when increasing the number of microphones is increased.
- The acoustic echo cancelling method also gives good suppression of the ambient noise in the car.
- The calibration signals can be either coloured noise, flat noise or human speech. None of these variations had much influence on results.
- The placement of the microphones is important.
- The method is well suited for low-cost microphones.
Bibliography


Appendix A

Figures-Evaluation

Figure A.1: Number of filter taps - Ericsson with 2 microphones
Figure A.2: Number of filter taps - Ericsson with 2 microphones

Figure A.3: Number of filter taps - Ericsson with 6 microphones
Figure A.4: Number of filter taps - Ericsson with 6 microphones

Figure A.5: Number of filter taps - Sennheiser with 2 microphones
Figure A.6: Number of filter taps - Sennheiser with 2 microphones

Figure A.7: Number of filter taps - Sennheiser with 6 microphones
Figure A.8: Number of filter taps - Sennheiser with 6 microphones

Figure A.9: Different Calibration Signals - Ericsson with 6 microphones
Figure A.10: Different Calibration Signals - Sennheiser with 6 microphones

Figure A.11: Calibration Signal Level - Ericsson with 2 microphones
Figure A.12: Calibration Signal Level - Ericsson with 6 microphones

Figure A.13: Calibration Signal Level - Sennheiser with 2 microphones
Figure A.14: Calibration Signal Level - Sennheiser with 6 microphones

Figure A.15: Convergence of the Adaptive Filter - Ericsson with 6 microphones
Acoustic Echo Cancelling with Microphone Arrays.

Figure A.16: Convergence of the Adaptive Filter - Sennheiser with 6 microphones

Figure A.17: Array Configuration Geometry 1 - Ericsson with 1 microphone
Figure A.18: Array Configuration Geometry 1 - Ericsson with 2 microphones

Figure A.19: Array Configuration Geometry 1 - Ericsson with 3 microphones
Figure A.20: Array Configuration Geometry 1 - Ericsson with 4 microphones

Figure A.21: Array Configuration Geometry 1 - Ericsson with 5 microphones
Figure A.22: Array Configuration Geometry 1 - Ericsson with 6 microphones

Figure A.23: Array Configuration Geometry 1 - Ericsson summary
Figure A.24: Array Configuration Geometry 2 - Ericsson with 1 microphone

Figure A.25: Array Configuration Geometry 2 - Ericsson with 2 microphones
Figure A.26: Array Configuration Geometry 2 - Ericsson with 3 microphones

Figure A.27: Array Configuration Geometry 2 - Ericsson with 4 microphones
Figure A.28: Array Configuration Geometry 2 - Ericsson with 5 microphones

Figure A.29: Array Configuration Geometry 2 - Ericsson with 6 microphones
Figure A.30: Array Configuration Geometry 2 - Ericsson summary

Figure A.31: Array Configuration Geometry 1 - Sennheiser with 1 microphone
Figure A.32: Array Configuration Geometry 1 - Sennheiser with 2 microphones

Figure A.33: Array Configuration Geometry 1 - Sennheiser with 3 microphones
Figure A.34: Array Configuration Geometry 1 - Sennheiser with 4 microphones

Figure A.35: Array Configuration Geometry 1 - Sennheiser with 5 microphones
Figure A.36: Array Configuration Geometry 1 - Sennheiser with 6 microphones

Figure A.37: Array Configuration Geometry 1 - Sennheiser summary
Figure A.38: Array Configuration Geometry 2 - Sennheiser with 1 microphone

Figure A.39: Array Configuration Geometry 2 - Sennheiser with 2 microphones
Figure A.40: Array Configuration Geometry 2 - Sennheiser with 3 microphones

Figure A.41: Array Configuration Geometry 2 - Sennheiser with 4 microphones
Figure A.42: Array Configuration Geometry 2 - Sennheiser with 5 microphones

Figure A.43: Array Configuration Geometry 2 - Sennheiser with 6 microphones
Acoustic Echo Cancelling with Microphone Arrays.

Figure A.44: Array Configuration Geometry 2 - Sennheiser summary

Figure A.45: Car Cabin Noise (90km/h) - Ericsson Geometry 1
Figure A.46: Car Cabin Noise (90km/h) - Ericsson Geometry 2

Figure A.47: Car (90km/h) and Fan Noise - Ericsson Geometry 1
Acoustic Echo Cancelling with Microphone Arrays.

Figure A.48: Car (90km/h) and Fan Noise - Ericsson Geometry 2

Figure A.49: Car (90km/h) and Music - Ericsson Geometry 1
Figure A.50: Car (90km/h) and Music - Ericsson Geometry 2

Figure A.51: Car (90km/h) and Side Window Noise - Ericsson Geometry 1
Figure A.52: Car Cabin Noise (90km/h) - Sennheiser Geometry 1

Figure A.53: Car (90km/h) and Fan Noise - Sennheiser Geometry 1
Figure A.54: Car (90km/h) and Music - Sennheiser Geometry 1

Figure A.55: Car (90km/h) and Side Window Noise - Sennheiser Geometry 1