Silence is golden – Implementation of a noise cancelling office chair

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ABSTRACT
Elevated noise levels are commonplace in today’s office environment and may compromise attention and work performance or even affect health. This work aims at reducing acoustic immissions by integrating an active noise control feature into an office chair. To obtain a reasonable noise cancellation, several challenges have to be considered. First, noise cancellation has to be achieved at the user's ears (virtual microphone), rather than at the noise measurement location (physical microphone). This requires estimating the noise at the virtual location, which is generally referred to as virtual microphone algorithm. Second, head movements of the user as well as the position of the noise source have to be detected. Third, all mechatronic parts should be integrated into the chair, ruling out the use of a reference microphone. This paper describes the necessary measurement techniques and algorithms to implement a noise-cancelling office chair. To achieve adequate noise cancellation, stereo operation with two actuators is advisable. Different stereo algorithms are examined. A measurement principle based on ultrasonic sensors as well as the inherent algorithms to detect the user's head position are described. Finally, this paper describes the implementation of the concepts in experiment and simulation and provides results describing the achieved performance.

Keywords: Active Noise Control, Remote Microphone Technique, Stereo Operation
I-INCE Classification of Subjects Number(s): 38.2

1. INTRODUCTION
In today's office environment, acoustical immissions may affect attention, productivity or even health of office workers. This work aims at improving acoustical working conditions by integrating an active noise control (ANC) system into an office chair. Overviews on general ANC techniques are given in (1) and in (2). This paper presents some of the specialized techniques needed to implement a noise cancelling office chair. This particular application poses several challenges, which are described in the following.

For the ease of use, all mechatronic parts including loudspeaker, microphones, controller and power supply should be integrated into the office chair, see Figure 1. This implies that there is no reference microphone, which may lead to a forecast lead time for the ANC controller. In contrast, common ANC applications such as in acoustic ducts (3) or in cars (4) rely on a reference microphone.

Figure 1 – Integration of signal processing, sensors and actuators in the head rest

Acoustic noise emerging from various sources and arriving from all directions should be cancelled, which calls for an adaptive ANC system. Noise cancellation is required around the ears of the user (virtual microphones) rather than at the error (physical) microphones, which demands for an

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estimation of the disturbance at the virtual microphone location. These estimation techniques are generally called virtual microphone algorithms. In reference (5), an overview of different virtual microphone algorithms is presented. The problem of a moving virtual microphone in particular is addressed in reference (6). In reference (7), a description of the Virtual Microphone Arrangement (VMA) and the Remote Microphone Technique (RMT) is given and the performance of these algorithms is compared to conventional ANC.

When applying the RMT algorithm, a model of the transfer path from the physical to the virtual microphone $\hat{H}(z)$ is employed, see Figure 2. Clearly, this model depends on the user's head position and on the direction of arrival (DOA) of the acoustic noise. Therefore, head position and direction of arrival have to be estimated during operation of the noise cancelling chair. This paper presents a methodology for estimating the head position based on ultrasonic sensors.

In the application at hand, situations may arise where the transfer function $\hat{H}(z)$ becomes non-causal. This is the case if a noise source is located in front of the user, and the noise is measured at the physical microphones only after it should have been cancelled at the virtual microphones. In references (8) and (9), the Delayed Remote Microphone Technique (DRMT) is proposed, which yields satisfying noise attenuation even in non-causal situations. In Section 2.4, the DRMT algorithm is summarized.

To obtain acceptable noise attenuation, stereo operation with two actuators is preferred. The mutual interference between the two actuators has to be considered in the ANC algorithm. In reference (10), the virtual microphone control (VMC) is described and different variants of stereo ANC are compared. This paper evaluates several variants of how to expand the DRMT algorithm to the stereo application scenario and gives results concerning the achievable performance.

The paper is organized as follows: Section 2 outlines the general aspects of this work, including model identification and hardware setup. Section 3 proposes a measurement principle and an algorithm to estimate the user's head position. In Section 4, several variants of the DRMT stereo algorithm are presented and their performance is evaluated.

## 2. GENERAL ASPECTS

The following section describes the experimental and simulation setup and reviews the DRMT algorithm.

### 2.1 Methodology

In this work, a combined approach based on both simulations and experiments was chosen in order to benefit as much as possible from the respective strengths, which is a common approach in the area of ANC. The advantages of simulations are among others the possibility to perform rapid controller prototyping, investigation of various setups, allowing for a thorough comparison of different situations at a low cost. On the other hand, the advantages of experiments are more reliable results, ongoing validation during the development process and automatic and inevitable consideration of imperfections (e.g. nonlinearities of loudspeakers or echoes).

### 2.2 Experimental Setup

The experiments were performed in a semi-anechoic chamber (dimensions $3.5 \times 3.5 \times 2.2$ m). Multiple noise sources were positioned throughout the chamber. An in-ear microphone was placed in a dummy (i.e. at the virtual microphone location), for performance measurements and for identification of transfer paths. Rugged and versatile hardware was employed for performance measurements (Bruel & Kjær microphones and data acquisition unit PULSE 3560-C, sampling rate 51.2 kHz) and controller implementation (National Instruments cRIO-9076 with integrated real-time controller and Field Programmable Gate Array (FPGA) and analog input and output modules NI 9234 and 9263, respectively). The chosen sampling rate on the FPGA is 20 kHz.

The most important experimental results used in the simulations are the dynamical models of the transfer paths between different components. Using a finite impulse response (FIR) filter is particularly advantageous because it is simple to implement and there exist powerful and proven algorithms in order to determine an FIR during operation. The least-mean-squares (LMS) algorithm as described in reference (1) is used in the controller, but also for initially identifying FIRs. To this end, white noise is generated and output via the loudspeakers, the microphone signal is recorded, and finally the FIR filter is identified using the LMS algorithm.
2.3 Simulation Setup

Figure 2 gives an overview of the identified transfer functions in the experimental setup. The secondary paths and the transfer functions \( \hat{H}(z) \) are used in simulation and in the control algorithm, the primary paths in simulation only. As previously mentioned, the transfer functions were identified using FIR filters, and \( S(z) \) denotes estimated transfer paths. Thus, no first-principle models of the components (e.g. loudspeakers and microphones) were required.

![Figure 2 – Overview of Transfer Functions.](image)

MATLAB/Simulink without any specific toolboxes was used in order to simulate both the physical system and the ANC controller. The sampling rate used for the discrete-time simulation was identical to the one used in the FPGA, i.e. 20 kHz. Emphasis was put on the match between signals obtained in experiments and simulations. Continuous cross-comparison allows for fast validation and debugging.

2.4 The DRMT Mono Algorithm

To accomplish noise attenuation even in non-causal situations, the DRMT algorithm has been proposed, see references (8) and (9). This algorithm is able to achieve noise attenuation even if the disturbance signal \( d(n) \) is measured at physical error microphones \( e_p(n) \) after it should have been cancelled at the virtual microphones \( e_v(n) \). In the following subsection, the equations for the mono use case with one physical microphone and one loudspeaker are given.

The filter coefficients \( W(z) \) of the ANC controller in the presence of a secondary path are identified using the normalized leaky Filtered-x-LMS (FxLMS) algorithm, see reference (1). The filter coefficients \( W(z) \) of the ANC controller are given in Eq. (1), where \( e(n) \) is the error signal, \( \nu \) is the leakage factor, \( \mu(n) = \alpha / (L \cdot P_e(n)) \) is the variable step size, \( L \) is the filter length and \( P_e(n) \) an estimate of the power of \( x(n) \).

\[
w(n+1) = \nu \cdot w(n) + \mu(n) \cdot x'(n) \cdot e(n)
\] \hspace{1cm} (1)

The filtered reference signal \( x'(n) \) is calculated by filtering the reference signal \( x(n) = \hat{d}_p(n) \) with the estimate of the physical secondary path \( \hat{S}_p(z) \), see Eq. (2):

\[
x'(n) = z^{-N} \left( \hat{S}_p(z) * x(n) \right)
\] \hspace{1cm} (2)

where * refers to the convolution operation. The disturbance signal at the physical microphone location \( \hat{d}_p(n) \) is estimated as in Eq. (3):

\[
\hat{d}_p(n) = e_p(n) - \hat{S}_p(z) * y(n)
\] \hspace{1cm} (3)

In a non-causal situation, the transfer function \( \hat{H}(z) \) is separated into a causal part \( \hat{H}_0(z) \) and a negative dead time \( z^{-N} \), with \( N \in \mathbb{N}_0 \):

\[
\hat{H}(z) = \hat{H}_0(z) \cdot z^{-N}
\] \hspace{1cm} (4)
In the DRMT ANC algorithm, the non-causal part of \( \hat{H}(z) \) is omitted. This results in a delayed estimation of the disturbance at the virtual microphone:

\[
\hat{d}_v(n - N) = H_0(n) * \hat{d}_p(n)
\]  
(5)

The estimation of the error at the virtual microphone \( \hat{e}_v(n) \) is given in Eq. (6). To account for the delayed disturbance estimation \( \hat{d}_v(n - N) \), the influence of the controller output and the reference signal \( x'(n) \) are also delayed by \( z^{-N} \), see Eqs. (2) and (6).

\[
\hat{e}_v(n - N) = \hat{d}_v(n - N) + z^{-N} \cdot (\hat{S}_v(n) * y(n))
\]  
(6)

As previously mentioned, the transfer function \( \hat{H}(z) \) depends on the DOA of the acoustic noise. Assuming that there is one dominant acoustic noise source in a semi-circle, the DOA may be estimated using the existing instrumentation, namely the left and right physical error microphones. The delay between these two signals is calculated by means of cross correlation. For estimating the DOA in all directions, further instrumentation and algorithms are required.

### 3. ESTIMATION OF THE HEAD POSITION

As mentioned previously, the transfer function \( \hat{H}(z) \), which describes the transfer between the physical microphone location and the virtual microphone location (i.e. the user's ear), varies in dependence of the user's head position and rotation. Furthermore, noise cancellation is highly local, i.e. effective noise cancellation only occurs in a narrow region around the target point. Further away from the target point, the resulting noise level may even be higher compared to the situation without active noise control. For these reasons, it is highly important to possess precise estimates of the positions of the user's ears.

#### 3.1 Approaches for head position estimation

Several approaches based on different sensor technologies that may yield such estimates have been developed and investigated, see reference (12). The two most promising paths are the following ones:

- image recognition based on cameras installed on either side or on top of the office chair
- ultrasonic ranging based on several ultrasonic sensors installed in the headrest

Although the image recognition approach appears more suitable in terms of detection of the ear positions than the ultrasonic ranging approach, it raises doubts in terms of user acceptance. Therefore, the latter approach was pursued. Also, knowledge of the user's head position (as opposed to the positions of the ears) was considered sufficient for the time being.

#### 3.2 Ultrasonic ranging

Easy-to-use low-cost off-the-shelf ultrasonic sensors (HC-SR 04 by Cytron Technologies) that both send and receive ultrasonic signals have been selected. The minimum ranging distance is of particular importance and is 2 cm in the current case, which is low enough for the typical use case. Figure 1 gives an impression of the low-key integration of the sensors in the headrest.

![Figure 3 – Illustration of the basic idea of ultrasonic ranging](image)
Although the different ultrasonic sensors could be used individually in order to measure the closest distance of the user's head to a particular sensor, the basic idea used for ultrasonic ranging is more elaborate. It consists of making use of several ultrasonic sensors in a combined way which could be referred to as sensor fusion.

Figure 3, which only considers the two-dimensional situation in the x-y plane for the sake of simplicity, illustrates this idea. The three ultrasonic sensors which are positioned on the same vertical (z-) position are indicated with squares on the lower limit. The user's head is modelled as a circle of known diameter whose center is the point M. Let us now suppose that the sensor emits an ultrasonic ranging signal which is reflected by the head and received by the middle sensor. The distance travelled by the signal (which is the duration between emission and reception multiplied by the speed of sound), along with the positions of the two mentioned ultrasonic sensors, defines an ellipse which is tangent to the circle defining the user's head. Such an ellipse exists for all possible combinations of ultrasonic sensors. Figure 3 shows the resulting three ellipses for the three ultrasonic sensors used.

Since all ultrasonic sensors make use of the same frequency, it is advisable to follow a multiplexing approach, i.e. to ensure that all sensors emit at distinct points in time. The sensors already contain functions such as the generation of a pulse train and the measurement of the time elapsed between sending and receiving of such a pulse train. Since multiple sensors have to operate in a synchronized manner in order to put the sensor fusion idea into practice, these functions were implemented once again on FPGA.

Reference (12) contains a simple, yet effective iterative algorithm that yields the position of the center of the head in a few iterations starting from a wide range of initial head position estimates. This algorithm was implemented on the same hardware as the ANC algorithm. Experiments carried out with a dummy in realistic circumstances suggest that the estimate of the head position is precise to approximately 2-3 cm.

4. STEREO OPERATION

To achieve an optimal sound attenuation at the ears of the user in this application, stereo operation of the ANC system is preferred. Stereo operation implies the use of two loudspeakers and two physical error microphones integrated in the headrest.

A general problem of stereo ANC is the mutual interference of the controllers over loudspeakers and error microphones. This interference strongly depends on the head position, which is noticeable in the amplitude of the cross-path transfer functions, and may lead to instability of the ANC system.

This section evaluates several variants of how to expand the DRMT algorithm to the stereo application scenario, see Section 2.1. Each of these algorithms varies in implementation complexity and required hardware resources, and they are therefore investigated in a simulation setup. The examined algorithms are given in Section 4.2 and the results are given in Section 4.3.

4.1 Simulation Scenarios

Figure 4 shows the simulation setup of the acoustical domain, which is the same for all algorithms. The interface between the acoustical and the signal processing domain is defined by the control output of the left and right loudspeaker \( y_L(n) \) and \( y_R(n) \) and the signals of the left and right physical error microphone \( e_{pL}(n) \) and \( e_{pR}(n) \). The sound level at the users ears is given by the signals of the left and right virtual error microphone \( e_{vL}(n) \) and \( e_{vR}(n) \). These signals are only used for performance comparison and not in the control algorithm. The acoustical disturbance signal is given by \( d(n) \). The disturbance \( d(n) \) comprises 20 linearly spaced harmonics, ranging from 100 to 1000 Hz.
To evaluate the stereo algorithms under different conditions, two parameters in the ANC scenario are modified. First, the head position of the user is altered, moving from a standard position ($hd0$) to a position 10 cm further away from the headrest ($hd10$). This increases especially the amplitude of the cross-over virtual secondary paths $S_{v,rl}(z)$ and $S_{v,lr}(z)$. Second, the direction of the noise source relative to the users sitting position is altered from the 12 o’clock position directly in front of the user ($agl0$) to a position under an angle of 50° clockwise relative to the 12 o’clock position ($agl50$). All transfer functions for these different scenarios have been identified in the laboratory setup. Figure 5 shows a sketch of the different scenarios.

Figure 5 – Overview of ANC scenarios

The evaluation of the different stereo ANC algorithms should be carried out in a realistic scenario which delivers reproducible results. Therefore, all transfer functions in Figure 2 are identified in the laboratory setup and are then used in the simulation software. This leads to results which are highly comparable. Figure 6 shows examples of identified transfer functions. On the left-hand side, the virtual secondary path $\hat{S}_{v,rr}(z)$ from right source to right virtual microphone is plotted for standard and further forward head position. The dead time at the beginning of the impulse response increases with increasing head distance. On the right-hand side, the cross-over virtual secondary path $\hat{S}_{v,lp}(z)$ from right source to left virtual microphone is plotted. As the head distance increases, there is a stronger coupling between the two sides because the head acts as an acoustic shield. The amplitude of the impulse response is significantly smaller for cross-over transfer paths.
4.2 Stereo ANC Algorithms

In the following subsections, different stereo ANC algorithms using the Delayed Remote Microphone Technique are presented and their performance is analyzed. Table 1 gives an overview of the evaluated algorithms.

<table>
<thead>
<tr>
<th>Algorithm name</th>
<th>No. of adaptive filters</th>
<th>Signal estimation</th>
<th>FxLMS type</th>
</tr>
</thead>
<tbody>
<tr>
<td>2W2</td>
<td>2</td>
<td>Direct path signal estimator</td>
<td>SISO</td>
</tr>
<tr>
<td>2W4</td>
<td>2</td>
<td>Direct and cross path signal estimator</td>
<td>SISO</td>
</tr>
<tr>
<td>4WSISO</td>
<td>4</td>
<td>Direct and cross path signal estimator</td>
<td>SISO</td>
</tr>
<tr>
<td>4WMIMO</td>
<td>4</td>
<td>Direct and cross path signal estimator</td>
<td>MIMO</td>
</tr>
</tbody>
</table>

4.2.1 2W2

The most straightforward method to obtain a stereo ANC algorithm is to duplicate the structure of the mono algorithm. This results in two SISO (single-input, single-output) systems with the physical error microphone signal $e_p(n)$ as input and the loudspeaker signal $y(n)$ as control output for the left and right side, respectively. These two systems are independent of each other, virtual and physical cross-over transfer paths are not considered.

Figure 7 – DRMT Stereo 2W2
Figure 8 shows the general structure of the 2W2 stereo algorithm with the calculation of the adaptive filter coefficients $W_{rr}(z)$ and $W_{ll}(z)$ and the controller outputs $y_r(n)$ and $y_l(n)$ by means of the FxLMS algorithm. Figure 8 shows the direct path signal estimator, which estimates the disturbance signals $\hat{d}_{p,r}(n)$ at the physical microphone locations and the error signals $\hat{e}_r(n-N)$ at the virtual microphone location. In this case, the cross-over physical and virtual secondary paths are not considered.

Since there is a transfer function $\hat{H}_d(z)$ for the right and the left side, there are also two different negative dead times $z^{-N}$. For example, the transfer function $H(z)$ may be causal for one side and non-causal for the other side. Therefore, the parameter $N$ is determined for each side, resulting in $N_r$ and $N_l$. This applies to all of the following algorithms and is, for the reason of simplicity, not marked in the algorithm diagrams. The controller outputs are given in Eqs. (7) and (8):

$$y_r(n) = W_{rr}(z) \star \hat{d}_{p,r}(n)$$

(7)

$$y_l(n) = W_{ll}(z) \star \hat{d}_{p,l}(n)$$

(8)

### 4.2.2 2W4

The DRMT stereo 2W4 algorithm comprises two adaptive filters as well. In contrast to the 2W2 algorithm, the virtual and physical cross-over transfer paths are now considered in the estimation of the disturbance and error signals, see Figure 10. The disturbance at the physical microphone location is given in Eq. (9):

$$\hat{d}_{p,r}(n) = e_{p,r}(n) - \hat{S}_{p,r}(z) \star y_r(n) - \hat{S}_{p,rr}(z) \star y_r(n)$$

(9)

The estimation of the error at the virtual microphone location is given in Eq. (10):

$$\hat{e}_{v,r}(n-N) = \hat{d}_{v,r}(n-N) + z^{-N} \left( \hat{S}_{v,r}(z) \star y_r(n) + \hat{S}_{v,vr}(z) \star y_r(n) \right)$$

(10)

The calculation of the controller outputs is equivalent to the 2W2 algorithm, see Eqs. (7) and (8), as well as the general structure, see Figure 9.
4.2.3 4WSISO

The DRMT stereo 4WSISO algorithm employs four adaptive filters, one each for the two direct transfer paths $W_{rl}(z)$ and $W_{lr}(z)$ and one each for the two cross-over transfer paths $W_{rr}(z)$ and $W_{ll}(z)$. To identify the coefficients of the filters, the SISO FxLMS algorithm given in Eq. (1) is applied. Each of the filters is adapted to minimize one error signal only, either at the left or the right virtual microphone location.

The estimation of the disturbance and error signals is equivalent as in the $2W4$ structure, see direct and cross path signal estimator in Figure 10, but the controller outputs are now given by Eqs. (11) and (12). Figure 11 shows the general structure of the 4WSISO algorithm.

\[
y_j(n) = W_{jj}(z) \ast \hat{d}_{p,j}(n) + W_{jr}(z) \ast \hat{d}_{p,r}(n)
\]

\[
y_j(n) = W_{rl}(z) \ast \hat{d}_{p,r}(n) + W_{rr}(z) \ast \hat{d}_{p,r}(n)
\]

4.2.4 4WMIMO

The DRMT stereo 4WMIMO architecture involves four separate MIMO (multiple-input, multiple-output) FxLMS filters, see reference (11). The stereo RMT algorithm from (7) has been
adapted to the DRMT algorithm. Using the MIMO FxLMS algorithm, the coefficients of the adaptive filters are updated as follows:

$$w_{p}(n+1) = v \cdot w_{p}(n) - \mu \cdot z^{-N} \left( \hat{S}_{v,p}(z) \ast \hat{d}_{p,r}(n) \right) \hat{e}_{r,l}(n-N) - \mu \cdot z^{-N} \left( \hat{S}_{v,r,r}(z) \ast \hat{d}_{p,l}(n) \right) \hat{e}_{r,v}(n-N)$$  \hspace{1cm} (13)

$$w_{b}(n+1) = v \cdot w_{b}(n) - \mu \cdot z^{-N} \left( \hat{S}_{v,b}(z) \ast \hat{d}_{p,r}(n) \right) \hat{e}_{r,l}(n-N) - \mu \cdot z^{-N} \left( \hat{S}_{v,r,r}(z) \ast \hat{d}_{p,l}(n) \right) \hat{e}_{r,v}(n-N)$$  \hspace{1cm} (14)

$$w_{j}(n+1) = v \cdot w_{j}(n) - \mu \cdot z^{-N} \left( \hat{S}_{v,j}(z) \ast \hat{d}_{p,r}(n) \right) \hat{e}_{r,l}(n-N) - \mu \cdot z^{-N} \left( \hat{S}_{v,r,r}(z) \ast \hat{d}_{p,l}(n) \right) \hat{e}_{r,v}(n-N)$$  \hspace{1cm} (15)

$$w_{g}(n+1) = v \cdot w_{g}(n) - \mu \cdot z^{-N} \left( \hat{S}_{v,g}(z) \ast \hat{d}_{p,r}(n) \right) \hat{e}_{r,l}(n-N) - \mu \cdot z^{-N} \left( \hat{S}_{v,r,r}(z) \ast \hat{d}_{p,l}(n) \right) \hat{e}_{r,v}(n-N)$$  \hspace{1cm} (16)

The controller outputs are calculated according to Eqs. (11) and (12). The estimation of the disturbance and error signals is equivalent as in the 2W4 and 4WSISO structure, see direct and cross path signal estimator in Figure 10. Figure 12 shows the general structure of the 4WMIMO algorithm.

In contrast to the 4WSISO algorithm, each of the filters is adapted to minimize both of the error signals at the virtual locations \( \hat{e}_{r,l}(n-N) \) and \( \hat{e}_{r,v}(n-N) \).

![Figure 12 – DRMT Stereo W4MIMO](image)

**4.3 Results**

To evaluate the performance of the above algorithms, simulations have been performed under the scenarios described in Section 4.1. The following parameters have been chosen:

- Filter length \( L = 350 \)
- Leakage factor \( \nu = 1 - 1 \times 10^{-5} \)
- Normalized step size \( \alpha = 0.004 \)

The performance criterion in Eq. (17) evaluates the attenuation between the root mean square (RMS) of the disturbance at the virtual location \( d_{r}(n) \), which corresponds to switching the ANC controller off, and the RMS of the remaining error at the virtual location \( e_{r}(n) \) when the controller is switched on.

$$\text{atten} = 20 \cdot \log_{10} \left( \frac{1}{N} \sum_{n=1}^{N} d_{r}(n)^2 \right) \frac{\sqrt{1/N \sum_{n=1}^{N} e_{r}(n)^2}}{}$$  \hspace{1cm} (17)
Figure 13 and Figure 14 show the noise attenuation at the virtual microphone location for the left and right side for two different scenarios. For the scenario with noise from the side and a standard head position (hd0, agl50), the attenuation at the right ear is generally good, except for very low frequencies, where the frequency response of the loudspeaker does not allow for a better attenuation. At the left ear, the attenuation is similar for the 2W4, 4WSISO and 4WMIMO algorithms. For the 2W2 algorithm, the attenuation is worse and there is even an amplification of the noise level for certain frequencies.

For the scenario with noise from the front and a head position further ahead (hd10, agl0), the noise attenuation is generally acceptable for the 2W4, 4WSISO and 4WMIMO algorithms, in particular because this is a non-causal situation for the ANC controller. For the 2W2 algorithm, noise attenuation is generally poor, because with this head position, there is more coupling between the two sides, and the cross-path transfer functions are not considered in the 2W2 algorithm.

These simulations have shown that 2W4, 4WSISO and 4WMIMO algorithms yield similar noise attenuation performance for the scenarios considered. The 2W2 algorithm however performs substantially worse, resulting even in noise amplification for certain cases. Taking into consideration the implementation complexity and the required hardware resources, the algorithm 2W4 is best suited for the implementation.

The 2W4 algorithm has been implemented on FPGA and tested in the laboratory setup. The experiments have shown satisfactory correlation between simulation and measurements. However, a detailed description of the experiments is beyond the scope of this paper.

5. CONCLUSIONS

If some kind of remote microphone technique is applied to achieve noise attenuation at a location apart from the physical error microphone, the knowledge of the user's head position is crucial. This paper delineates a measurement principle and an algorithm to estimate the user's head position. Further, different variants of the DRMT stereo algorithm are evaluated. It is shown that taking into account the cross-path transfer functions in the signal estimator is reasonable, but the use of four instead of two adaptive filters or even the application of the MIMO FxLMS algorithm does not result in better noise.
attenuation in the scenarios considered.

Ongoing work is conducted to optimize the interaction of the different subsystems, namely the control algorithm, the estimation of the user's head position and the estimation of the DOA. Another promising way forward is the use of wireless reference microphones placed near known noise sources to further improve the noise attenuation in non-causal situations.

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