ON THE PERFORMANCE OF A LOCAL ACTIVE NOISE CONTROL SYSTEM

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ABSTRACT

This paper presents a multichannel active system for the local control of sound around the headrest on the back of a seat placed inside an enclosure. The size of the zones of quiet produced makes the system practical only at relatively low frequencies [1]. Finally, some results of cancellation for narrowband and broadband noise are presented. Two different system configurations algorithms have been tested on the adaptive controller. Both of them show similar results, but the new algorithm based on the minimization of the maximum error signal power, has shown computational saving and higher speed convergence than the multiple channel least squares algorithm.

1. INTRODUCTION

Noise reduction in a local system represents a successful application of active noise cancellation in enclosures [2]. The acoustic field is controlled in a limited area, dependent on error microphones-secondary sources separation (a sphere of $\lambda/10$ radious around each sensor if the error microphone is further than about $\lambda/2$ from the secondary source). If the microphones are placed closer, a greater zone of quiet can be obtained although the acoustic presure can increase in other points of the room [3][4].

This paper is organised as follows. We first comment different multichannel algorithms for active noise control [5], the multiple error LMS algorithm (multichannel version of the filtered-X LMS algorithm)[1] and the Least Maximum Mean Squares[2] algorithm. Then, we describe the local noise control prototype. Finally we show the experimental results obtained in the performance of the local ANC system using different algorithms and different acoustic environments. We will focus our attention on the study of the zones of quiet obtained.

2. ADAPTIVE CONTROL ALGORITHMS

The multichannel active noise control system model used is described in [3]. Figure 1 shows the block diagram of

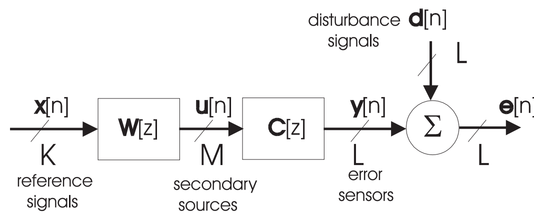


Figure 1: Multichannel pure feedforward ANC system. The system has M secondary sources, K reference signals and L error sensors. Block C represents a matrix of LxM error paths and block $\mathbf{W}$ is a matrix of KxM control filters.

the model. As a first approach, the multichannel version of the filtered-X LMS algorithm was used for the controller (MELMS algorithm) [6]. The output of the *l-th* error sensor can be written as,

$$e_l[n] = d_l[n] + \sum_{k=1}^{K}\sum_{m=1}^{M}\sum_{j=0}^{J-1} c_{lmj} \sum_{i=0}^{I-1} w_{mki} x_k[n-i-j] \tag{1}$$

Equation 1 illustrates the linear relationship between the error signal and the controller coefficients and can be expressed in matrix form as,

$$\mathbf{e}[n] = \mathbf{d}[n] + \mathbf{R}[n]\mathbf{w} \tag{2}$$

where $\mathbf{R}[n]$ is the filtered reference signals matrix. The iterative expression to update the coefficients vector is given by

$$\mathbf{w}[n+1] = \mathbf{w}[n] - \alpha \sum_{l=1}^{L} \mathbf{R}_l^T[n] e_l[n] \tag{3}$$

where α is the convergence parameter and $\mathbf{R}_l[n]$ corresponds to the *l-th* row of the filtered reference signals matrix $\mathbf{R}[n]$.

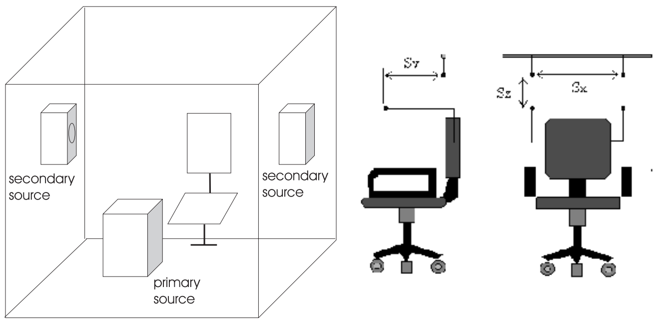
There exists a kind of algorithms called minimax type which were studied in [7][8]. The idea behind a minimax type algorithm in active control is to balance the acoustic field after control, to achieve this desire it is needed to define first which measure of the acoustic field is wanted to balance and then apply a minimax strategy of minimisation using this measure. If it is minimised the maximum of the mean

[1] MELMS
[2] LMMS

squared values of the error signals the algorithm is called LMMS [7][8][9][10]. The iterative algorithm can be found as

$$\mathbf{w}[n+1] = \mathbf{w}[n] - \alpha \mathbf{R}_b^T[n] e_b[n] \tag{4}$$

subscript b denotes the error signal with maximum mean squared value for a given value of the control vector. The performance of the LMMS algorithm on the controller has been also tested.



a) b)

Figure 2: a) setting position of sources in the enclosure; b) seat and microphones layout.

3. PROTOTYPE DESCRIPTION

The local ANC system was tested in two different acoustic environments: a rectangular enclosure with internal dimensions 4.35mx3.28mx2.96m and a semi-anechoic chamber. For the results presented here, a single 210-mm-diam loudspeaker was used as the primary source and two loudspeakers more were used as the control sources. In the reverberant room the primary source was located at the position (0.85,1.5,1)m and the secondary sources were located at the positions (2.4,0.75,1.3)m and (2.4,2.3,1.3)m. A seat was placed at the position (3.25,1.6,1.2)m, just in front of the primary source and an array of up to four microphones was mounted on the headrest of a seat, as shown on figure 2.

The frequency range of interest was 40-150Hz in the reverberant enclosure and 40-240Hz in the semi-anechoic chamber. The signals to be cancelled were car engine noise and random noise. The sampling rate used was 500Hz.

4. EXPERIMENTS

Experiments were carried out in the two environments described before. Different configurations have been chosen to demonstrate the ability in creating quiet zones of the ANC system. As the quiet zones are three-dimensional areas in space, the results are shown on different planes (different z coordinates). The zones of quiet are obtained sampling an aproximated area of 700x500 mm^2 around the headrest of the seat with a monitor microphone. The objective is to obtain a zone of quiet centered in the listener´s head position.

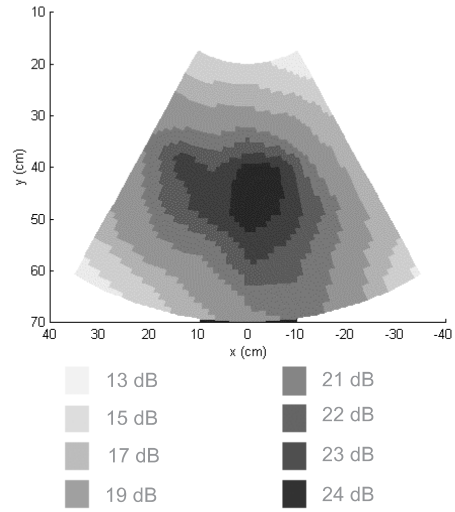


Figure 3: Attenuations achieved in a 1:2:4 system in the $z = 20cm$ plane after the ANC system operation using the MELMS algorithm in order to cancel car engine noise.

4.1. Reverberant Room

Consider a control system with 1 primary source, 2 secondary sources and 4 error sensors (1:2:4), and the MELMS algorithm on the controller. The noise to be cancelled is engine noise (obtained at 1200rpm: harmonics of 20Hz). The microphones are situated in $S_x = 30cm$, $S_y = 40cm$ and $S_z = 40cm$. The whole sources are set in a plane of about 1.30cm in height ($z = 20cm$). Figure 3 shows the attenuations measured in that plane which correponds to the listener's head position. Different planes of measure got worse noise level reduction. If the listener moved his head away from the area centered around the sources plane it could be appreciated that the noise level increases.

Using engine noise signals, tests were performed using the MELMS and LMMS algorithms. Microphones were situated on the loudspeakers diaphragm ($1.20cm$ in height) in order to achieve optimal results, but not all of them received the same acoustic pressure. Noise reduction levels achieved using both algorithms are shown in figure 4. Circles show relative error microphones position. The zones of quiet obtained after ANC system operation using both algorithms are represented on figure 5. No remarkable differences were perceived on the quiet noise areas obtained. On table 1 it can be seen that as the MELMS algorithm strives on reducing acoustic pressure at microphones with higher levels (the right ones), the LMMS algorithm searchs for achieving more equal distribution. In a random noise sound field con-

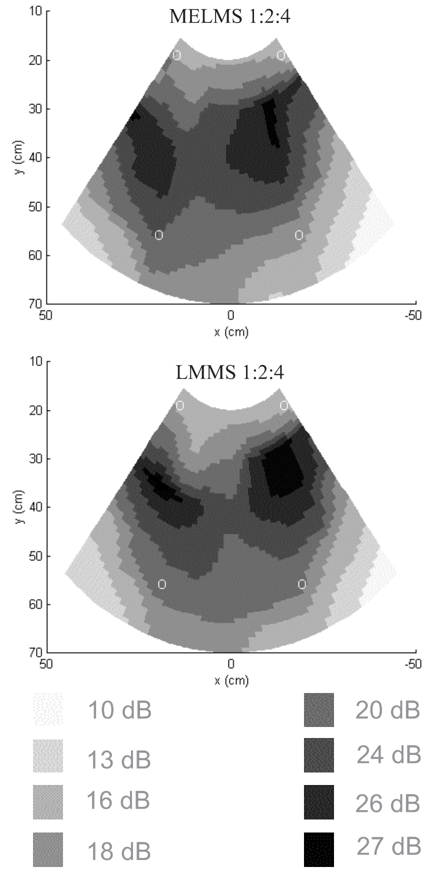


Figure 4: Attenuation levels obtained in a 1:2:4 system in the sensors plane using the MELMS and LMMS algorithms. Car engine noise.

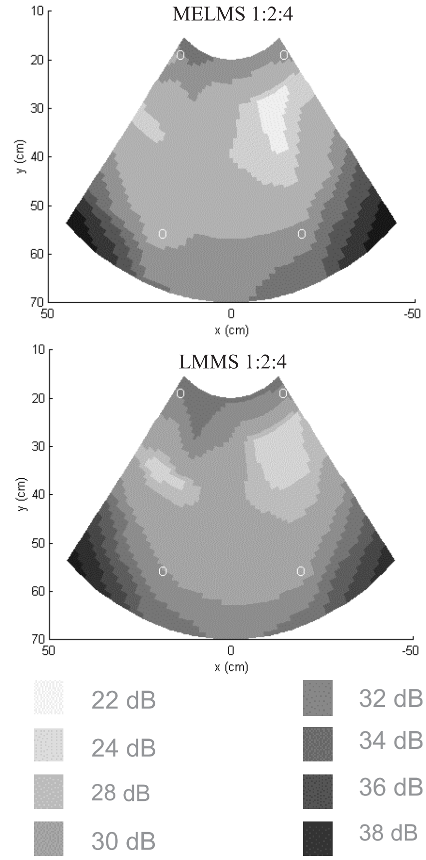


Figure 5: Zones of quiet obtained in a 1:2:4 system in the sensors plane using the MELMS and LMMS algorithms. Car engine noise. (Arbitrary units).

clusions are not so easy to be obtained because algorithm convergence is more difficult. Note the stability condition of both algorithms is ensured with people moving inside the room.

4.2. Semi-anechoic Chamber

A complete set of measurements was also carried out in the semi-anechoic chamber. The noise reduction on this room improves compared to the reverberant room due to the acoustic field simplicity of the semi-anechoic chamber. Figure 6 shows the cancellation of random noise in a 1:2:2 system in both environments obtained on a sensor position. The use of more error microphones gives the greatest zone of quiet. Despite increasing the frequency range of work it has been achieved attenuations higher than 30dB at some frequencies. Figure 7 shows the zones of quiet cancelling engine noise with a 1:2:4 system using filters increasing the cutt-off frequency (240Hz). Despite increasing the range of work it has achieved zones of quiet higher than 20dB.

5. CONCLUSIONS

A local ANC system has been developed, and the performance measured with a microphones array using car engine noise and random noise reproduced in two different environments (a reverberation room and a semi-anechoic chamber). We have introduced a minimax type algorithm (the LMMS one) which despite its simplicity has shown a good convergence and stability behaviour compared with the MELMS algorithm. Very similar areas of quiet were observed using both algorithms and they appear fairly robust

Microphone	MELMS	LMMS
front right	550	720
front left	850	600
rear right	550	800
rear left	850	800

Table 1: Acoustic pressure levels measured in microphones after cancellation (arbitrary units).

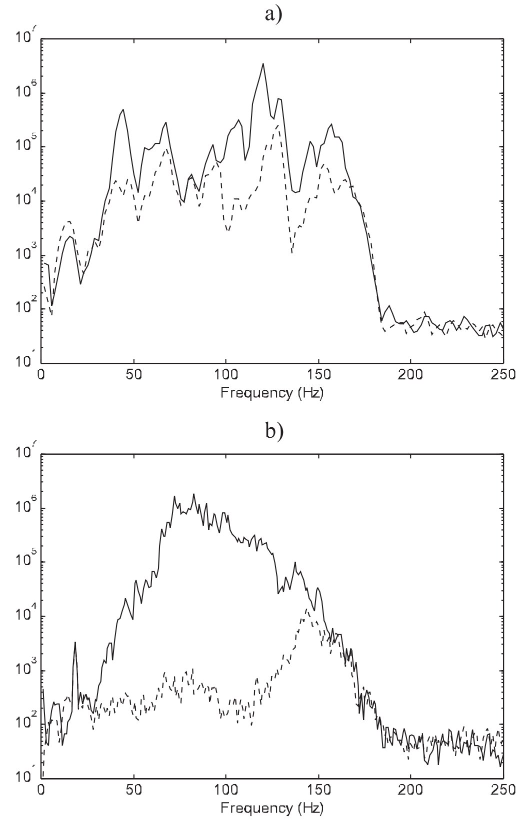


Figure 6: Power spectral density of the signal measured at one error sensor in a 1:2:2 system before the ANC system operation (solid line), after the ANC system operation (dashed line) using the MELMS; a) on the reverberant room, b) on semi-anechoic chamber. Cutt-off frequency of the filters:150 Hz. Random noise. (Arbitrary Units).

against changes in zeros of the enclosures transfer functions (i.e. people moving inside the enclosure).

One of the significant features of the results is that the shape of the zone of quiet is quite dependent on sources interaction (source placement). The current focus of research lies in performing a practical comparison of the LMMS algorithm and fast algorithms for ANC.

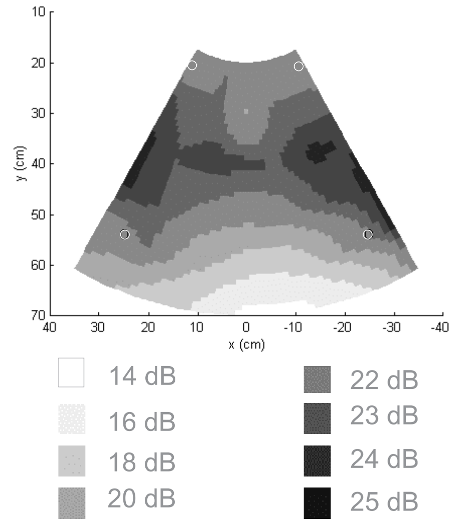


Figure 7: Attenuation levels measured in a 1:2:4 ANC system on semi-anechoic chamber using the MELMS algorithm. Engine noise.

6. REFERENCES

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