MULTIPLE CHANNEL ACTIVE NOISE CONTROL

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Abstract. *High level of acoustic noise can be the cause of human ear injuries. Active noise control has been used as optimum solution where ear defender cannot provide sufficient noise attenuation. Three multiple-channel active noise control techniques are studied in this work: feedforward, feedback and hybrid control systems. An acoustic duct is used in order to validate the control proposed methods. The algorithms are implemented in a dSPACE data acquisition system. The results are presented and discussed to multi-ton and white noise excitation.*

Keywords*: Active Control, Noise, Multiple channel.*

1. Introduction

There are a lot of researches that intent to increase the quality of life of the people. The researches are focused in questions that affect health and comfort. In the case of engineering, acoustical control has been mobilized a great number of researches. There are two large fields in this area: Passive Noise Control and Active Noise Control (ANC).

Active Noise Control uses electro acoustic or electromechanical transducers in order to cancel the noise based on superposition wave principle. The fundamental idea is to apply an anti-noise with the equal amplitude and opposite phase from the primary noise. The cancellation is achieved when both noise are combined (Nelson *et al.*, 1992; Hansen, 1997).

The idea is simple, but there are some obstacles to apply active noise control. Acoustical systems have characteristics of non-linearity and non-stationary that disturbs the performance of controller (Kuo *et al.*, 1999). To compensate these problems, adaptive controls were proposed. These controllers are adaptive filters that adjust their coefficients to minimize the noise of system (Goodwin *et al.*, 1984; Clarkson, 1993).

A common mechanism used to adjust the coefficients is the least-mean-square (LMS) algorithm (Widrow,1985), where finite impulse response (FIR) filters and infinite impulse response (IIR) filters are used.

There are two principal methodologies in active noise control: the first uses an input perturbation signal measured by a reference sensor (feedforward), the other one estimates this perturbation though the output system (feedback), dispensing the input sensor.

However it has situations where the physical systems to be controlled are more complex or large dimension, needing complex and sophisticated systems of control. In this context that the study of multiple channel active noise control is inserted.

In this paper, are considered the implementation of a multiple channel active noise control feedforward, a controller multiple channel feedback and finally a hybrid configuration that uses both concepts simultaneously.

To evaluate this proposal, an experimental duct workbench was built and the algorithms were implemented in a software Matlab-Simulink® using a system of acquisition of data dSPACE®. The experimental results are presented and discussed.

2. Multiple Channel Active Noise Control System

To get good noise attenuation in a large enclosure or free space via active noise control, a multiple-channel active noise controls must be used. The next sections present the different configuration of multiple-channel ANC system.

2.1. Multiple Channel ANC Feedforward System

The structure and block diagram of a multiple channel acoustic ANC system is illustrated in Fig. 1:

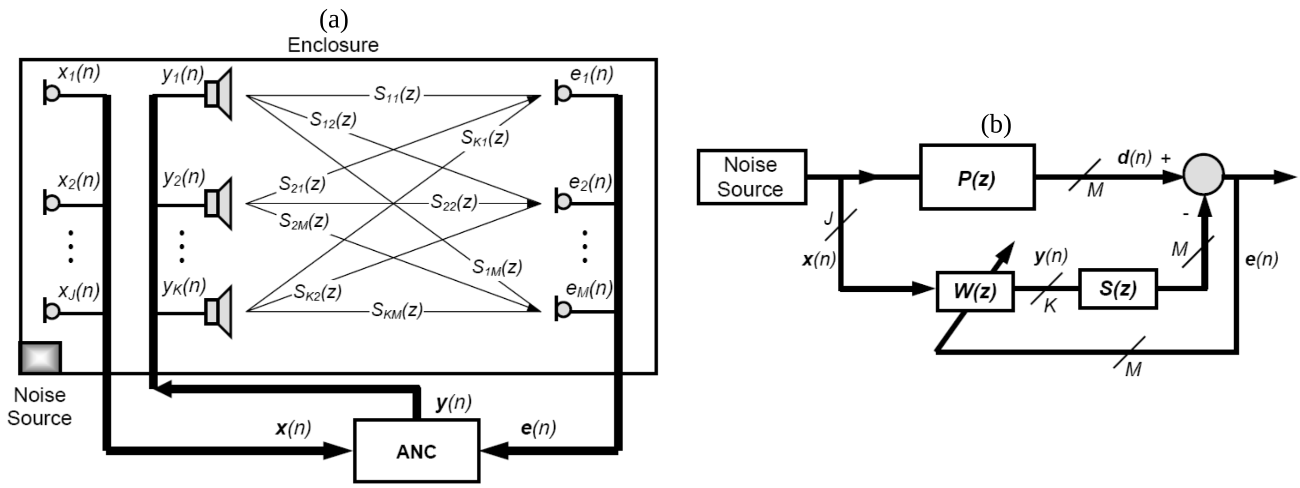

Figure 1. Structure and block diagram of a multiple channel acoustic of ANC

The multiple channel system employs (J) reference sensors to form the reference signal vector (x_J). The multiple channels system generates (K) signals of canceling (y_K) for the corresponding secondary sources that are distributed in diverse points of the environment to be controlled. Too, (M) error sensors (e_M) are distributed over desired locations to measures the residual noise components (Kuo *et al.*, 1999).

From Figure 1(b), $P(z)$ represents the transfer function between reference sensors and the error sensors, called, primary path. $S(z)$ represents the transfer function between canceling loudspeakers and the error sensors, called secondary path. $W(z)$ are the filters used to estimate an unknown plant $P(z)$, the primary path.

The elements of matrix W, that are the filter coefficients of estimated plant, are represented by $w(n) \equiv [w_1^T(n)\; w_2^T(n) \cdots w_K^T(n)]^T$ where $w_k(n) \equiv [w_{k,1}^T(n)\; w_{k,2}^T(n) \cdots w_{k,J}^T(n)]^T$.

Each controller $w_{k,j}(n)$ represents a set of coefficients of the FIR filter modeling the primary path from the reference sensors to the secondary sources: $w_{k,j}(n) \equiv [w_{k,j,1}(n)\; w_{k,j,2}(n) \cdots w_{k,j,L}(n)]^T$ and L is the filter order. The output signal of the FIR filters $y_k(n)$ are computed as:

$$y_k(n) = \sum_{j=1}^{J} w_{k,j}^T(n) x_j(n), \qquad k = 1, 2, \ldots, K \tag{1}$$

where the J channel reference signals can be expressed in a vector: $x(n) \equiv [x_1^T(n)\; x_2^T(n) \cdots x_J^T(n)]^T$. Each $x_j(n)$ is the references signals vector of length L: $x_j(n) \equiv [x_j(n)\; x_j(n-1) \cdots x_j(n-L+1)]^T$.

From Fig. 1 (b), the residual error is expressed as:

$$e(n) = d(n) + s(n) * [w^T(n) x(n)] \tag{2}$$

Assuming a cost function $\xi = \sum_{m=1}^{M} E\{e_m^2(n)\} = E\{e^T(n) e(n)\}$ the filter $W(z)$ is evaluated according with a least mean square (LMS) algorithm to minimize the sum of the instantaneous squared errors $\xi(n) \approx e^T(n) e(n)$.

So, the LMS adaptive uses the steepest descent algorithm (Kuo *et al.*, 1999) to adjust the coefficients of the adaptive FIR filters in order to minimize the cost function $\xi(n)$:

$$w(n+1) = w(n) - \frac{\mu}{2} \frac{\partial \xi(n)}{\partial w(n)} \tag{3}$$

where μ is the adaptive step size (Minguez, 1998). The gradient of the Eq. (3) is approximated by:

$$\frac{\partial \xi(n)}{\partial w(n)} \approx 2[s^T(n) * x(n)]^T e(n) = 2x'(n)e(n) \tag{4}$$

Substituting (4) into (3) we have the final expression of the filtered-X least mean square (FXLMS) algorithm:

$$w(n+1) = w(n) - \mu x'(n)e(n) \tag{5}$$

The Eq. (5) can be expanded to $k = 1, 2, \cdots, K$ and to $j = 1, 2, \cdots, J$. Getting the equation for a multiple-reference/multiple-output system FXLMS:

$$w_{kj}(n+1) = w_{kj}(n) - \mu \sum_{m=1}^{M} x'_{jkm}(n) e_m(n) \tag{6}$$

where $x'_{jkm}(n) \equiv \hat{s}_{mk}(n) * x_j(n)$ are the reference signal vectors filtered by the secondary path estimate $\hat{S}_{mk}(n)$.

In this work will be used a control algorithm with a single-reference/multiple-output. The FXLMS algorithm for this configuration is expressed as (Minguez, 1998):

$$w_k(n+1) = w_k(n) - \mu \sum_{m=1}^{M} x'_{km}(n) e_m(n), \qquad k = 1, 2, \ldots, K \tag{7}$$

where $x'_{km}(n) \equiv \hat{s}_{mk}(n) * x(n)$ to $k = 1, 2, \cdots, K$ and $m = 1, 2, \cdots, M$. The Figure (2) shows the block diagram for a system with single-reference, $K = 2$ signals output and $M = 2$ signals errors.

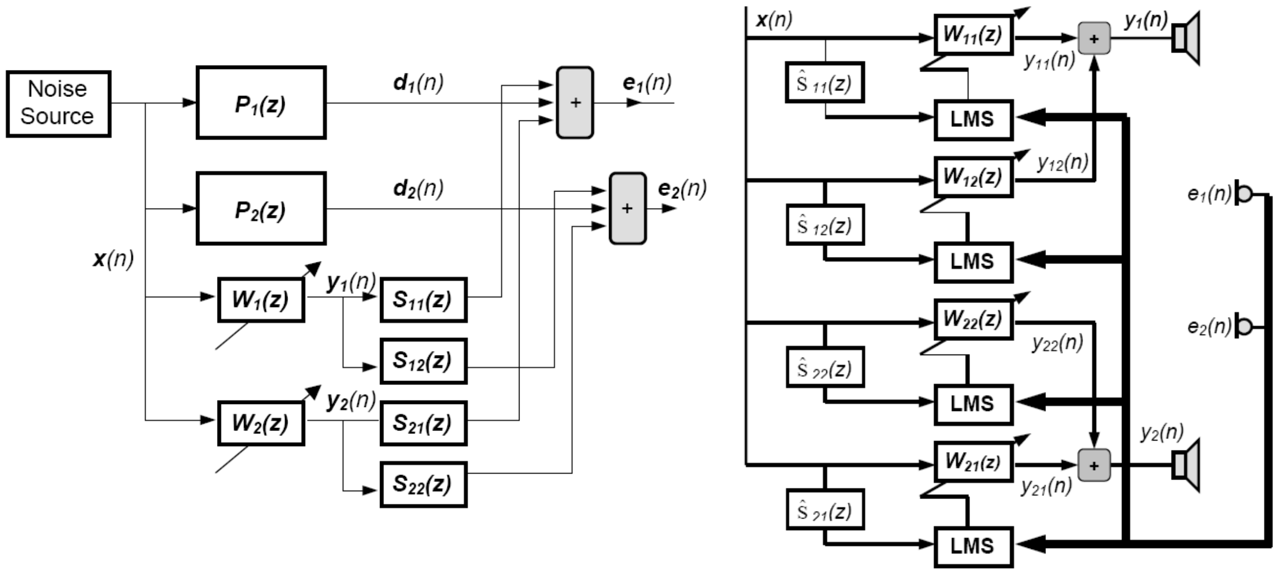

Figure 2. Multiple-Channel Control System: single-reference/two-output

In practical active noise control applications, $S(z)$ is unknown and must be estimated by an additional filter $\hat{S}(z)$. Widrow, 1985; Ericksson, 1989; fan 1990 and Bao 1993, developed some online identification techniques.

However, offline identification techniques, using a training process, could be used. In Nuñez *et al.*, (2004) is presented the application of this technique for active vibration control.

2.2. Multiple-Channel ANC Feedback System

The basic idea of a feedback ANC is to estimate the primary noise that is used as a reference signal $x(n)$ for the ANC FIR filter (Kuo, 1999; Nuñez *et al.*, 2004).

The diagram block of multiple-channel feedback ANC system is illustrated in Fig. (3).

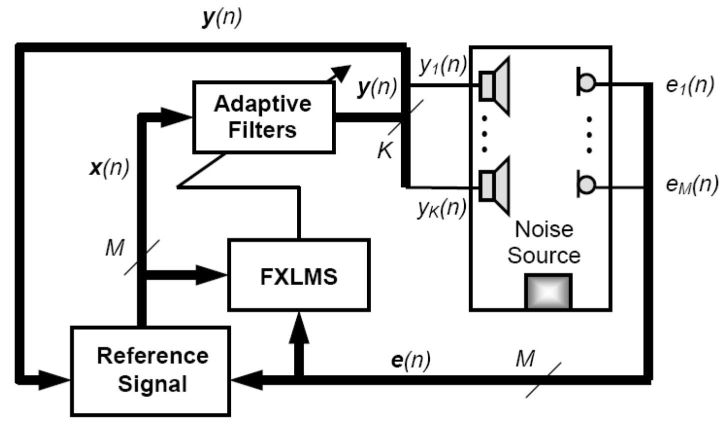

Figure 3. Block diagram of feedback ANC system

The reference signal uses K secondary signals $y_k(n)$, M error signals $e_m(n)$, and $M \times K$ secondary path estimates $\hat{S}_{mk}(z)$ to generate M reference signals $x_m(n)$ for the corresponding $K \times M$ adaptive filters, $W_{km}(z)$. The reference signals estimates are expressed as:

$$x_m(n) = e_m(n) - \sum_{k=1}^{K} \hat{s}_{mk}(n) * y_k(n), \qquad m = 1, 2, \ldots, M \tag{8}$$

where $\hat{s}_{mk}(n)$ represents the impulse response from secondary path and * denotes linear convolution. The coefficients are updates by the multiple channel FXLMS algorithm discussed in section 2.1.

2.3. Multiple-channel Hybrid Active Noise Control System

The feedforward control discussed before uses two sensors: the reference sensor and the error sensor. The input sensor measures the primary noise that will be cancelled while the error sensor monitors the performance of the active control.

The feedback control uses only an error sensor in order to generate the control signal. The combination of two methodologies is denominated hybrid active noise control (Kuo, 1999). Figure (4) illustrates this kind of system.

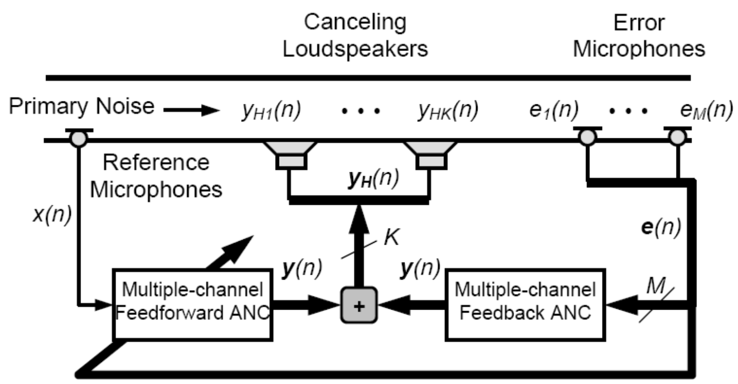

Figure 4. Multiple Channel Hybrid Active Noise Control System

In theory, the feedforward structure attenuates primary noise that is correlated with reference signal, while the feedback cancels the components of he primary noise that are not observed by the reference sensor.

3. Experimental Results

To experimentally evaluate these distinct control systems showed, an experimental PVC duct workbench was built and instrumented with loudspeakers (actuators) and microphones (sensors).

The loudspeakers used on this paper was a Bravoxx model BA6SS that have 60 Watts RMS of maximum power. Microphone electrets were used as sensors. Figure (5) shows a picture of the experimental workbench.

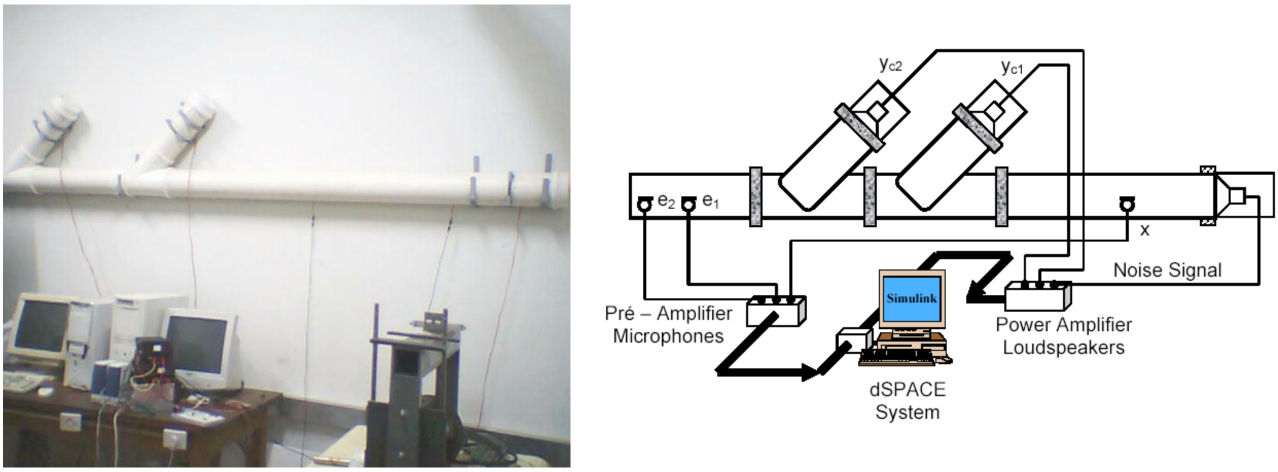

Figure 5. View of experimental setup

The PVC duct showed on picture has 4.0m of length and diameter of 0.15m. The primary noise loudspeaker was placed on edged and the canceling loudspeakers were placed 2.12m and 3.12m respectively.

The reference microphones were placed 0.25m and finally the error microphones were placed 3.5m and 3.86m, both away from noise loudspeaker.

Control algorithms were implemented using Matlab-Simulink® and tested in real time using dSPACE® platform. The sampling frequency adopted was 2.0 KHz in according with Nyquist Criterion.

To evaluate the performance of the control algorithms, the physical system was excited by a disturbance composed for four sine functions with frequencies of 150, 250, 350 e 450 Hz. The disturbance signal was generated for Simulink®.

The adaptive step (μ) based on input power was used and limited in 10% of the maximum (Nuñez *at al.*, 2004). In the next section, the experimental results obtained with the setup described before are presented.

Controle Multicanal Feedforward:

Figure (6) shows the spectrum of the error signal at the error microphones (e_1 , e_2) with the multiple-channel ANC system turned on (dashed line) and turned off (solid line). FIR filters of order 50 were used in this experiment.

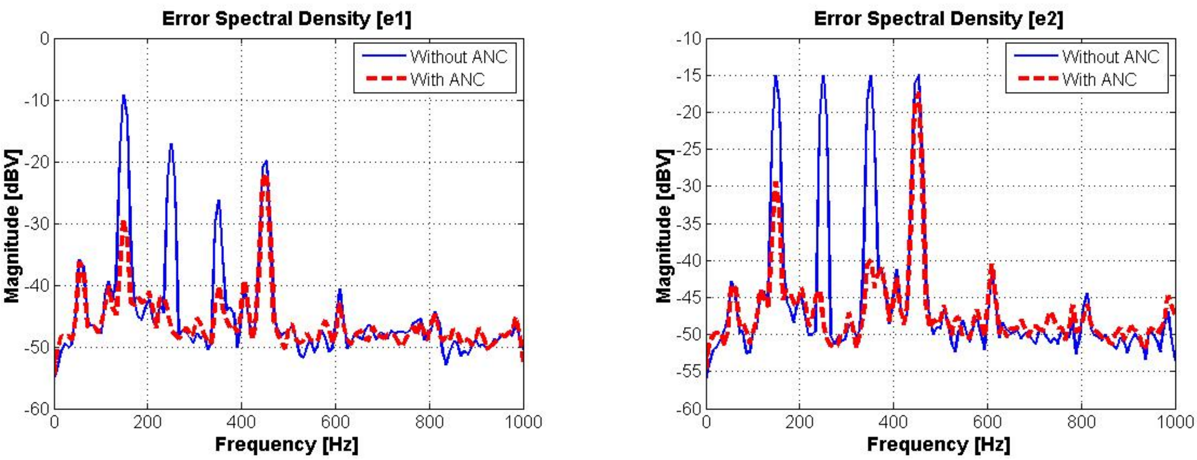

Figure 6. PSD – Multiple-Channel Feedforward ANC

The proposal multiple-channel feedforward showed efficient for the frequencies 150-350 Hz. In this experimental test, the amplitude in 450 Hz is reduced with a long time, but without the same efficiency and velocity of the others frequencies. This behavior can be improved with the increase of the step size (μ), but the algorithm can be instable.

In Fig. 7 are shown the error spectral densities for a white noise excitation. This test was accomplished for to evaluate the robustness and performance of the multiple channel system. In the analyzed frequency band the system shown a good performance.

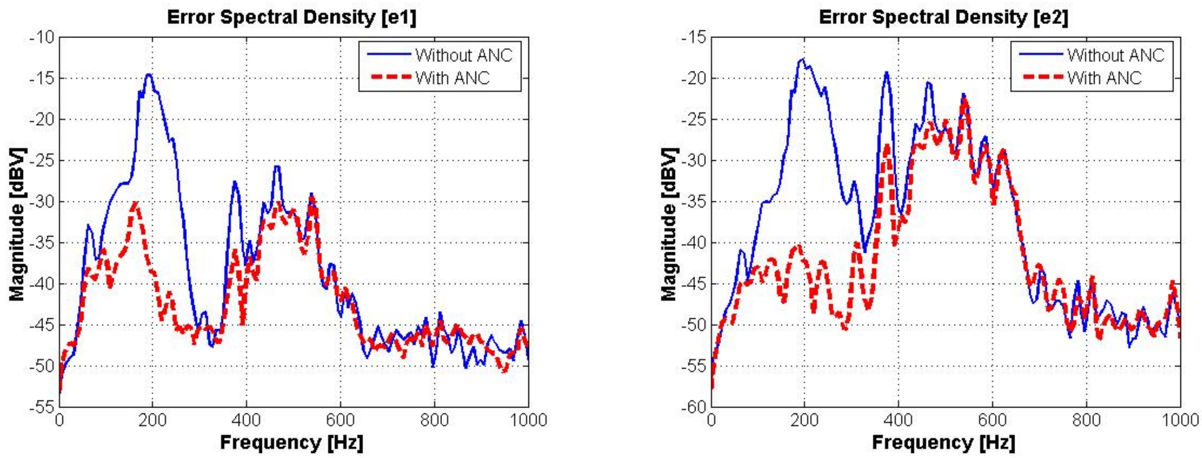

Figure 7. PSD – Multiple-Channel Feedforward ANC – White Noise

Multiple-channel Feedback Control:

The MFXLMS feedback results are shown in Fig. (8). FIR filters of order 25 were used in this test due to the dSPACE software limitation. The controller efficiency is reduced with little coefficient numbers.

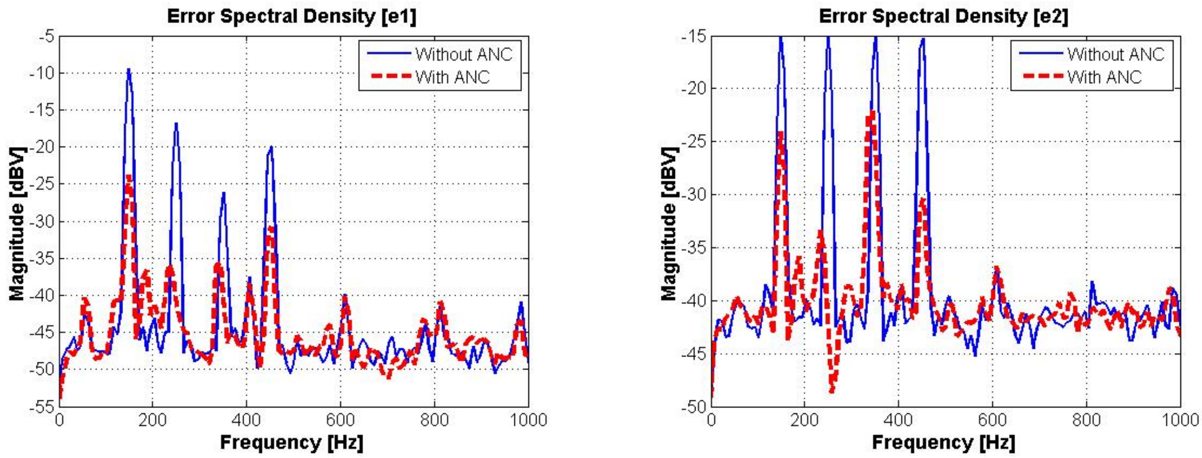

Figure 8. PSD – Multiple-Channel Feedback ANC

Multiple-Channel Hybrid Control:

The error spectral densities and the error microphone signal for multiple-channel hybrid control are shown in Fig. (9) and (10).

Only 15 coefficients for each FIR filter were used in this test, because the dSPACE memory is limited. Although, with reduced coefficient numbers, the hybrid control had a good performance.

Comparing the results in Fig. (9) and (10) with the results in Fig. (7) and (8), can be observed that the hybrid control had an optimal performance.

The system response frequencies for a white noise excitation are shown in Fig. (11). The lowed control efficiency is due to the reduced coefficient numbers (only 15 coefficients).

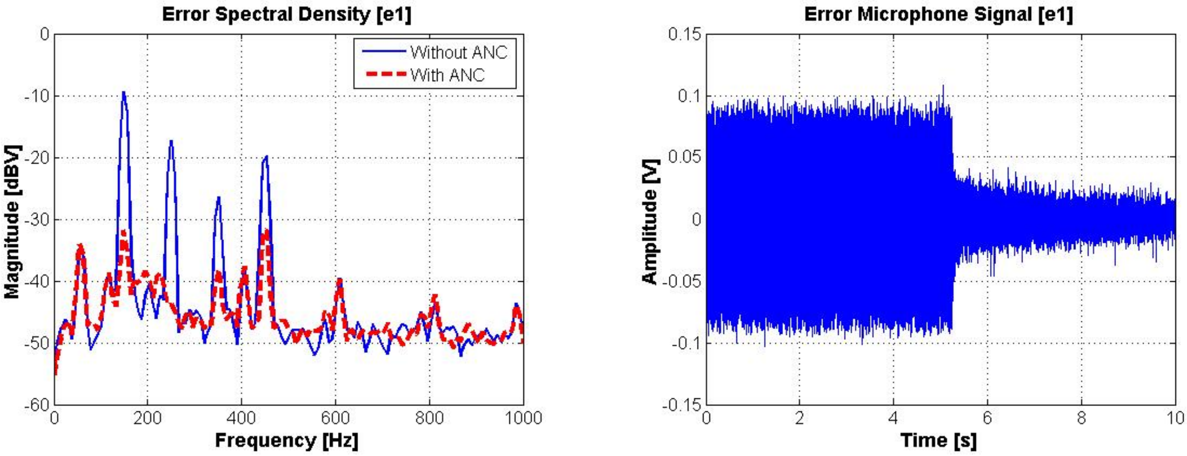

Figure 9. PSD – Multiple-Channel Hybrid ANC – Sensor [1]

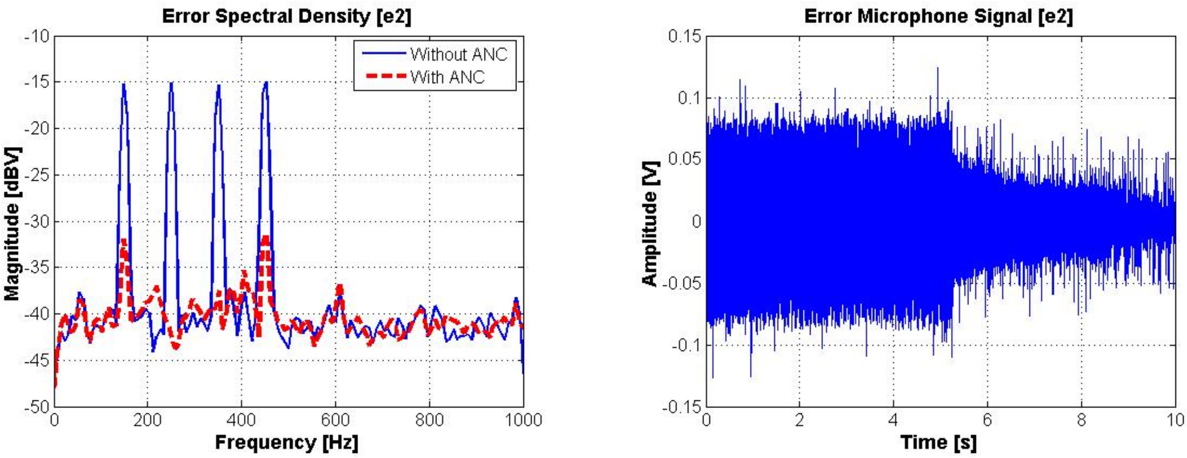

Figure 10. PSD –Multiple-Channel Hybrid ANC – Sensor [2]

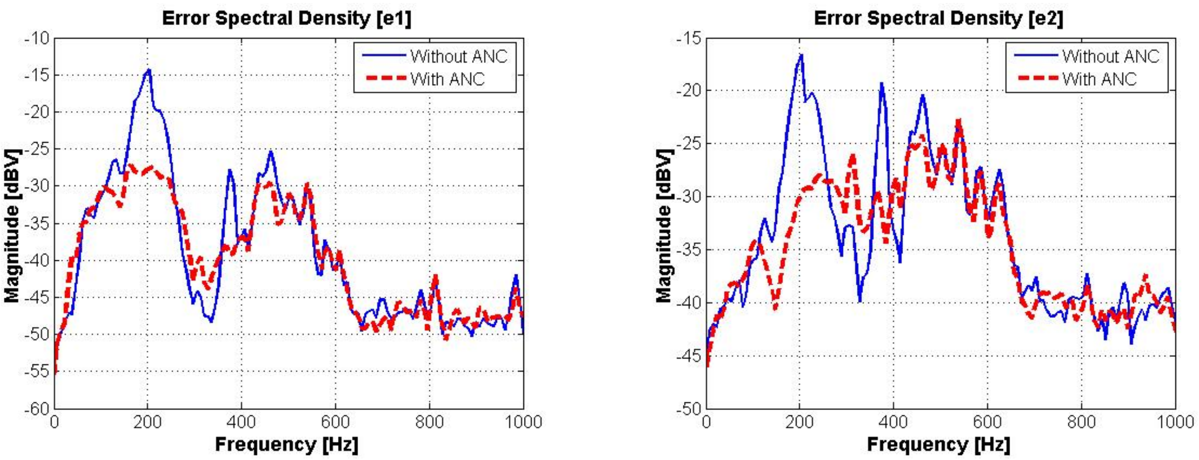

Figure 11. PSD –Multiple-Channel Hybrid ANC – White Noise

4. Conclusions

This work points to several important conclusions about different techniques of multiple-channel ANC systems applied to acoustic ducts.

In these projects are necessary to estimate a lot of secondary paths due to the several sensors and actuators used in the control process. In this case, the tuning of the controllers is more complex and vulnerable to errors, because the performance of these controllers depends highly of the estimated secondary paths.

The step size is very important for the filters convergence. The application of an adaptive step size based on the power of the filtered reference signal is a good choice for the stability of the algorithms.

The multiple-channel control algorithms are a good alternative to attenuate multiple tons and white noise. The size of the algorithms is a big problem in these control systems. The increase of the variable number's in the software causes this. In future works the online modeling of the secondary paths should be coupled to the control systems studied in this work.

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