ACOUSTIC ACTIVE NOISE CONTROL SYSTEM

Paulo Lopes, Bruno Santos, Miguel Bento and Moisés Piedade

INESC – Sistemas de Processamento de Sinal
Rua Alves Redol, nº 9, 1000 Lisboa, PORTUGAL
Tel: +351-1-3100210, Fax: +351-1- 3145843
email: paclopes@eniac.inesc.pt

SUMMARY

Acoustic active noise control [1] is a technique in which ambient noise is acquired by a microphone, processed, and emitted by a loudspeaker with opposite phase from the original. When the two signals are summed the noise level will be reduced. This paper describes the design, realization and experimental results of an acoustic broadband feedforward active noise control system.

INTRODUCTION

Acoustic noise reduction using passive techniques requires the use of devices relatively larger than the acoustic wavelength [2]. This constrain implies that, for these systems to have a reasonable response at low frequencies, they must be quite large, heavy, and expensive. Active noise control (ANC), where loudspeakers are used to reduce environment noise, has shown significant results in overcoming these problems. In ANC, an anti-noise of opposite phase is combined with the primary noise, resulting in cancellation of both noises. These systems are effective at low frequencies (where the passive techniques are inefficient) thus serving as a supplement to conventional passive techniques.

THE ACOUSTIC ACTIVE NOISE CONTROL SYSTEM

Figure 1 illustrates a single channel active noise control system. It is composed by a signal processor, two microphones (reference and error) and a loudspeaker. It works by modeling the *primary path*, $P(z)$, from the reference microphone to the error microphone and the *secondary path*, $S(z)$, from loudspeaker to the error microphone, with adaptive filters.

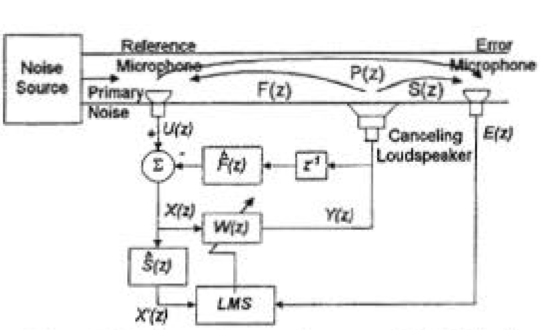


Figure 1. Active noise control system with feedback neutralization using FXLMS algorithm.

Classical noise reduction methods are usually simpler than the referred ones because there is no secondary path; the noise is subtracted directly from the original signal. In these systems, the basic approach is to use an adaptive digital FIR (finite impulse response) filter, and the LMS algorithm. In ANC, the classical LMS algorithm must be adapted to cope with the secondary path. One of the most common algorithms that accomplish this, is the FXLMS algorithm [3, 4, 5]. When an acoustic sensor is used for acquiring the reference signal, this signal is corrupted by the anti-noise generated by the loudspeaker, producing acoustic feedback. A feedback neutralization filter, $\hat{F}(z)$, as is shown in Figure 1, can solve this problem [6, 7].

In Figure 1, *P(z)*, *S(z)* and *F(z)* represent the transfer functions of the primary, secondary and feedback paths, respectively. $\hat{S}(z)$ and $\hat{F}(z)$ represent FIR models for *S(z)* and *F(z)*. These models may be previously determined (off-line) by system identification using the LMS algorithm and a white noise generator, with appropriate DSP algorithms.

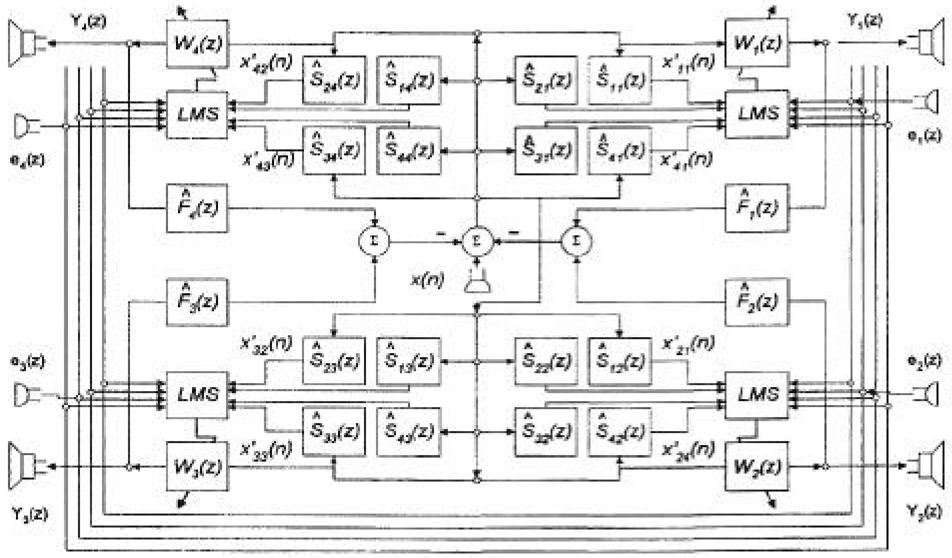


Figure 2. Four-Channel FXLMS algorithm with feedback neutralization.

In practice, in a single channel system with a single error microphone and a single secondary loudspeaker, noise reduction can only be achieved in a small region (about ¼ of the wavelength) surrounding the error microphone. To obtain a larger cancellation region it is required to use a multi-channel ANC [8]. We have decided to build a system based on four noise canceling loudspeakers, excited by the signals produced by the algorithm represented in Figure 2. This system uses the muti-channel FXLMS algorithm represented in Eq. 1.

$$w_k(n+1) = w_k(n) + \mu \cdot \sum_{m=1}^{M} x'_{km}(n) \cdot e_m(n), \quad x'_{km}(n) = \hat{s}_{mk}(n) * x(n) = \sum_{i=0}^{I-1} \hat{s}_{mk,i}(n) \cdot x(n-i),$$
$$k = 1, 2, \ldots, 4. \qquad\qquad m = 1, 2, \ldots, 4 \text{ and } k = 1, 2, \ldots, 4$$

(Eq. 1)

The system was implemented as is shown in Figure 3 [9, 10], with a TMS320C50 16 bits fixed-point DSP. It is possible to identify the four error microphones and canceling loudspeakers. The noise to eliminate is created by a loudspeaker and acquired by the reference microphones. Every acquired signal is filtered and sampled by the A/D converters; then, using the sampled signals, the DSP calculates the anti-noise signals and sends them to the D/A converters. The D/A output signals are then processed by the reconstruction filters and fed to the loudspeakers by the power amplifiers. The sampling rate was 4 kHz, and the FIR filters had 32 coefficients.

EXPERIMENTAL RESULTS

In order to obtain some experimental results we arranged the acoustics systems represented in Figure 4. In the experiments a pre-recorded noise from the interior of a bus was fed to the noise source loudspeaker and the Active Noise Control would try to eliminate it using the secondary loudspeakers, error microphones and the reference microphone.

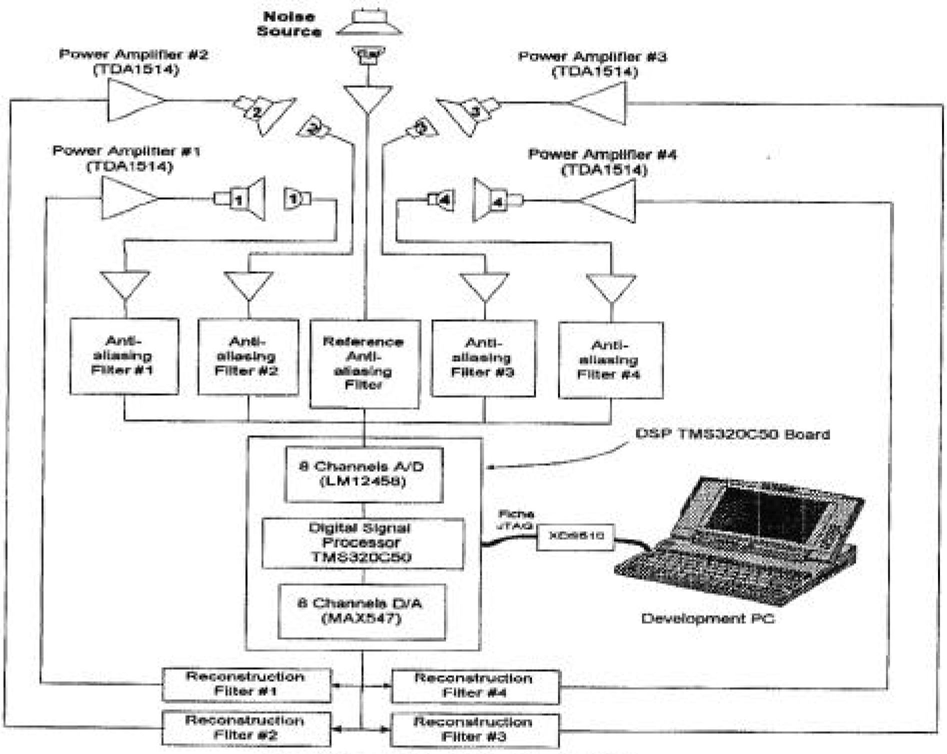


Figure 3. The Active Noise Control System.

The arrangements try to be realistic. We tried to make them possible to implemented inside of an automobile. The configurations presented are the result of several experiments in order to obtain a good configuration. Nevertheless, they follow the basic principles referred in the following lines.

To ensure causality, the distance from the reference microphone to the error microphone should be greater than the distance from the loudspeakers to the error microphones. To get a good reference signal, the reference microphone should be close to the noise source. Finally, the reference microphone should be as close as possible to the error microphones to increase the correlation between the two signals.

In Table 1, the values of total noise reduction measured at the several error microphones, in two similar experiments, are represented. We achieve a mean 8 dB of noise cancellation at the error microphones. In Figure 5, we show the performance of the ANC system in time and frequency domains.

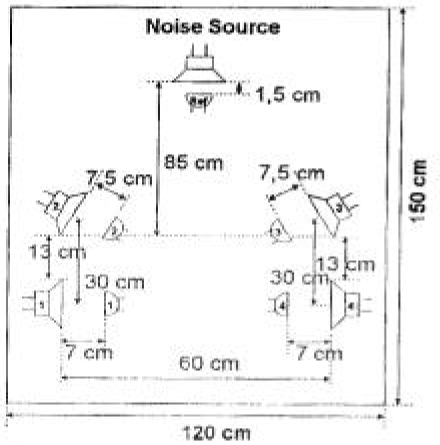
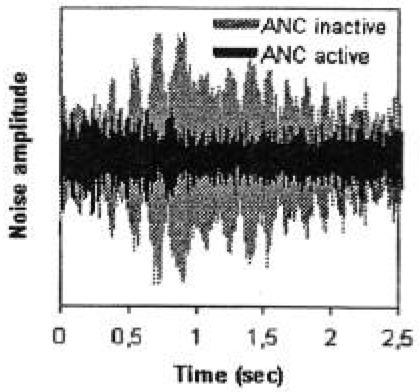


Figure 4. Spatial configuration of system.

	Measure 1	Measure 2
Mic. 1	7,13 dB	8,61 dB
Mic. 2	6,41 dB	6,94 dB
Mic. 3	8,82 dB	9,48 dB
Mic. 4	8,50 dB	9,30 dB

Table 1. Total noise reduction in the error microphones



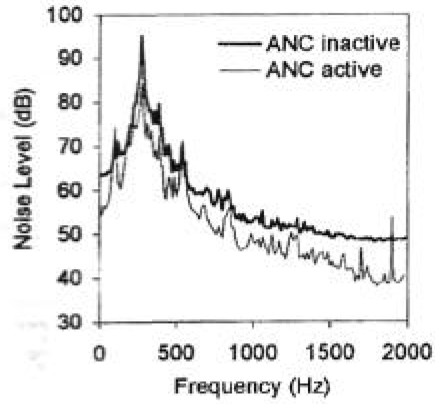


Figure 5. Noise signal in time domain and noise power spectrum at an error microphone. A 9-dB total noise reduction was obtained.

CONCLUSION

The work presented in this paper consists in the development of a four-channel acoustic feedforward active noise control system. We designed and assembled the system. The performance of the developed software was tested against simulation results and, finally, experimental results were obtained, in a real environment.

We have achieved a mean 8-dB total noise reduction at the error microphones in the 100 Hz to 2 kHz range. The major issues holding down noise reduction are, in order of relevance: causality problems, limited FIR filter size, poor FXLMS algorithm convergence, acoustic field dispersion, non-linear effects and system dynamic range.

In order to this algorithms to be applicable in real world they must be highly stable; they have to sustain strong changes in the environment like people getting in and out of the quit zone or the sudden opening of a window. This didn't represent a serious problem. We met this specifications throw carefully selection of the adaptive step size.

With the four-channel system, we have achieved a 50-cm wide quiet zone (for low frequency noises produced by a running car). Due to causality problems the reference microphone has to stay considerably far from the quiet zone, this can cause correlation problems. In practice, we have used 80-cm distance.

REFERENCES

[1] S. M. Kuo, D. R. Morgan, "Active noise control systems: algorithms and DSP implementations," Wiley, 1996.

[2] L. J. Eriksson, M. C. Allie, and C. D. Bremigan, "Active noise control using adaptive digital signal processing," in Proc. ICASSP, 1988, pp. 1594-2597.

[3] D. R. Morgan, "An analysis of multiple correlation cancellation loops with a filter in the auxiliary path," IEEE Trans. Acoustic Speech, Signal Processing, ASSP-28, 456-467, Aug. 1980.

[4] B. Widrow, D. Shur, and S. Shaffer, "On adaptive inverse control," in Proc. 15th Asilomar Conf., 1981, pp. 185-189.

[5] J. C. Burgess, "Active adaptive sound control in a duct: A computer simulation," J. Acoustic Soc. Am., 70, 715-726, Sept. 1981.

[6] M. M. Sondhi and W, Kellermann, "Adaptive echo cancellation for speech signals," in Advances in Speech Signal Processing, Chap. 11, S. Furui and M. Sondhi, eds., New York: Marcel Dekker, 1992.

[7] S. M. Kuo and J. Chen, "Multiple-microphone acoustic echo cancellation system with the partial adaptive process," Digit. Signal Process., 3, 54-63, Jan. 1993.

[8] S. J. Elliott, I. M. Stothers, and P. A. Nelson, "A multiple error LMS algorithm and its application to the active control of sound and vibration," IEEE Trans. Acoustical, Speech, Signal Processing, ASSP-35, 1423-1434, Oct. 1987.

[9] P. Lopes, B. Santos, M. Bento, "Cancelador de Ruído Acústico", graduation project, Instituto Superior Técnico, Portugal, 1997.

[10] P. Lopes, B. Santos, M. Bento, M. Piedade, "Active Noise Control System", Recpad98 Proceedings, IST, Lisboa, Portugal, March 1988.