Acoustic noise cancellation in closed environments- A design proposal

E.A. Lara-Alemán and F. Sandoval-Ibarra

Abstract—**Noise is a determining factor in stress-related ailments, and progressively affects the population's sense of hearing. In practice, because noise is typical of power-operated systems, its effect can be minimized if the power of the wave front is below the hearing threshold. That is a proposal commonly reported in the literature and with various variants is focused on designing the controlled alteration of the communication channel. The system thus developed is an active noise controller. In this contribution, experimental results show the feasibility of developing a channel controller using the TAdsp28335 card. With this proposal, two purposes were achieved: to develop a low-cost system, and to determine the requirements of the Active Noise Controller to develop optimal processing algorithms for real time operation. Experimental results show that the best solution is the FxLMS algorithm, with online estimation. It was observed that if the ANC system is excited with an electrical signal of $1.0V_{pp}$, at a frequency of 100 Hz, the power of that signal is reduced by constructing an anti-signal, achieving a value of $220mV_{pp}$ (which is equal to the intrinsic noise of the FxLMS algorithm), in 9s.**

Keywords—**Active noise, digital filters, digital signal processing, hearing loss**

I. Introduction

Exposure to noise causes hearing damage, and around the world, modern society suffer from acoustic trauma. Therefore, because hearing disorder is a global health problem, in literature we found several strategies to minimize the effect of noise on the population. In practice, a source of noise is a determining factor in health problems when noise travels to the individual, and this interaction becomes recurrent. Thus, who favors the transit of noise is the communication channel. Therefore, a strategy oriented to the care of hearing is to alter the traffic of noise while it transit through to the Communication Channel (CC). The controlled alteration of the CC possible by building an anti-noise signal, so that the interaction of both signals gives rise to a low amplitude signal.

The World Health Organization (WHO) recommends 65 dB as the upper limit range that acoustic noise must reach in order to avoid hearing loss. In this contribution, it is of interest not only to satisfy the recommendation of the WHO, but also

E. O. Lara-Alemán

Universidad Autónoma de Guadalajara /Cinvestav-Unidad Guadalajara
Mexico

F. Sandoval-Ibarra

Cinvestav-Unidad Guadalajara
Mexico

to minimize the noise power to develop a free noise zone. To do this, we first limit the problem to the human audible range, and that on average is from 20- to 20-kHz. In practice, there are several solutions to minimize the noise power, and their effectiveness is in well-defined frequency ranges. For example, passive solutions are useful at frequencies greater than 500 Hz, but al low frequencies the solution is an active network [1]-[4], that is, integrating software and hardware in an Active Noise Control system (ANC). In this contribution, the solution we proposed is an ANC based on a Digital Signal Processor (DSP). The organization of this contribution is as follows. The problem definition is described in Section II, where basic concepts and common architectures for reducing noise are widely analyzed. The Section III presents experimental results of our proposal based on the card Tadsp28335, where the advantages of several adaptive algorithms are analyzed. The characteristics of the selected electroacoustic elements are also presented in the same section. The conclusions of this work are given in Section IV.

II. Definition of the problem

There are two types of acoustic noise in the environment, broadband and narrowband noise. Broadband noise is mainly by turbulence. It is random noise and distributes its energy evenly across the frequency spectrum. Narrowband noise, on the other hand, is quasi-periodic noise and concentrates most of its energy at certain frequencies. There are two approaches to controlling acoustic noise: passive and active. The passive approach uses barriers, cages, and silencers to attenuate the noise power (passive silencers use the *Impedance Change* caused by a combination of tubes and baffles), as well as the *Energy Loss* caused by the noise propagation in channels bounded by sound absorbing materials. This passive solution is useful in a wide frequency range, however it is expensive for the materials it uses, it is large an ineffective as mentioned at low frequencies. Active solutions, by contrast, include an electroacoustic device that cancels noise by generating an anti-noise signal, of equal magnitude but opposite phase. That is, the combination of both the noise and anti-noise signal cancel each other out. It is clear that the effectiveness of noise cancellation depends on the magnitude and phase accuracy of the anti-noise signal. That noise cancellation approach is the so-called Primary Noise Cancellation. Active noise attenuation is a very attractive approach to achieve high percentages of noise reduction using a small dimensions proposal, and particularly at frequencies below 500 Hz. Therefore, to make noise cancellation effective, it is necessary to generate the anti-noise signal and for this, a DSP is used. In practice, because the acoustic characteristics of the noise source and the communication channel are not constant, both the noise source and the anti-noise signal are non-stationary signals. Thus, an active noise control system is adaptive in order to copy the changing characteristics of the acoustic environment, i.e. the communication channel. Remember

that in digital signal processing, there is a kind of adaptive system where the coefficients of a digital filter are adjusted in order to minimize the error between the desired signal and the actual one. The idea here is to build the anti-noise signal at time t_0, so that it not only corresponds to the noise signal with phase change, but when both interact at time t_1 ($>t_0$) they canceled each other out. In 1981, Burgess proposed the one-channel noise cancellation approach, based on adaptive filters [4]. Then, it was in the 1980s that the design of the active noise control was favored by the development of DSPs, and by the development of adaptive signal processing algorithms as well. The basic blocks of an ANC system are shown in Fig. 1, where $x(t)$, $y(t)$ and $e(t)$ are functions representing the reference, anti-noise and error, respectively. Therefore, if we have a DSP, and adaptive algorithms can be proposed, the problem is to define the architecture to develop the ANC system. Now to cancel broadband noise it is required to know the noise source in order to generate the anti-noise signal. Here the primary noise measurement will be used as a reference to the attenuator so that the primary noise is cancelled by the noise emitted by a horn. The best result will occur when the digital controller correctly models magnitude and phase. Noise cancellation systems, on the other hand, are based on two approaches: those using a feedforward control or using a feedback control. The operation is as follows. With feedforward control (see Fig. 2), the noise signal is sensed, the anti-noise signal $y(n)$ is generated, and while performing these tasks the noise $d(n)$ travels a distance D_1 where signals cancel each other. In practice, a sensor microphone will be at $D=D_0$ and the speaker at $D=D_1$. In this proposal, the error signal is required by the controller to modify the adaptive filter coefficients, and run the task until the error magnitude is minimized. In feedback control, on the other hand, the active controller cancels the noise signal without using a reference one (see Fig. 3). Here, the prediction of the noise signal is used, so it is only possible to attenuate periodic noise or narrow band noise. For all practical purposes, regardless of the control selected, any implementation must consider the nature of the communication channel. In an ANC system there are two channels (see Fig. 4), where $P(z)$ is the physical medium through which the noise $d(t)$ travels to the so-called free noise zone. The other one, $S(z)$, is the physical medium through which the anti-noise $y'(t)$ travels from the speaker to the microphone that senses the error $e(t)$. It should not be forgotten that environmental changes (mainly humidity and temperature) modify the performance of both communication channels. These variations are correctly compensated with adaptive filters, and with the appropriate algorithms, their coefficients are adapted to minimize the error. Once the system error decreases, it is expected that the coefficients converge to the optimal values, and with them represent the additive inverse of both the primary CC and the adequate estimate of the secondary CC. In practice, the adaptive filter that calculates the additive inverse of the primary CC is the Controller, and uses adaptive algorithms for its correct operation. Several of these algorithms for estimating the additive inverse of the primary CC can be found in the literature. The most used, due its simplicity, is LMS [5], and its variants such as the LMS family [6] and RLS algorithms [7]-[10]. However, when the ANC system includes the estimation of the secondary CC, algorithms for adaptive X filters are required [11]-[13]. By evaluating the performance of the reported algorithms, we decided to synthesize the FxLMS algorithm [14] because the most

important thing, in addition to minimizing the error, is to attenuate the total noise of the system under development.
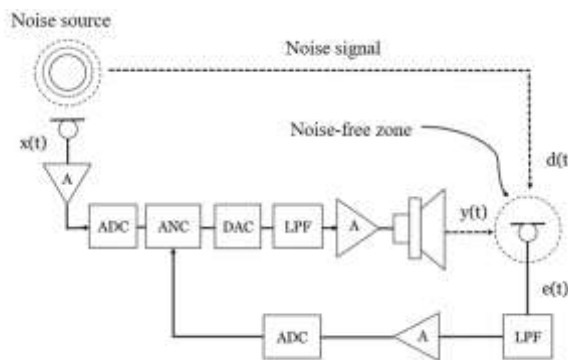

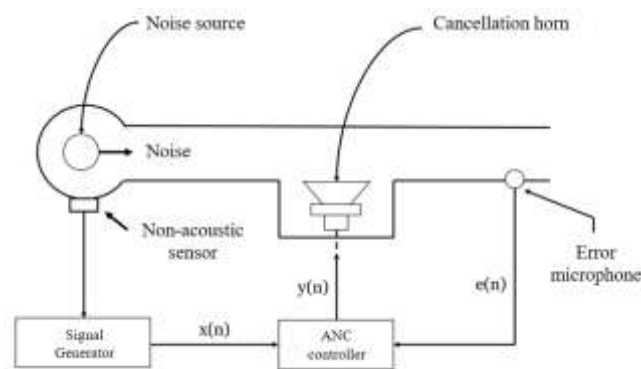Figure 1. Basic blocks of an active noise cancellation system.

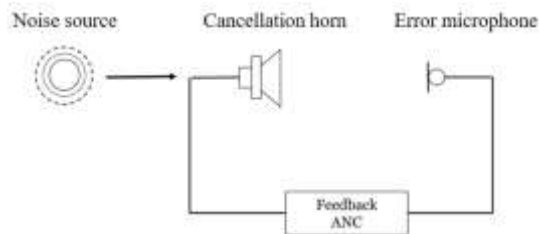Figure 2. Narrowband ANC with pre-feed.

Figure 3. Feedback control for cancelling periodic noise.

III. Performance testing

The ANC design includes the TAdsp28335 card to emulate acoustic channels with 16-coefficient digital filters. The parts of the system under design are shown in Fig.1, where noise sources of frequencies less than 500 Hz are used. The electrical validation of the controller was performed with the adaptive FxLMS algorithm (with on-line estimation) added to the block diagram shown in Fig. 5, where $P(z)$ and $S(z)$ are digital low-pass filters with cutoff frequencies at 300 Hz and 200 Hz, respectively, and $Y(z)$ is an all-pass filter that adds a delay equivalent to 16 samples. The input $x(n)$ is a $1.0V_{pp}$ sine signal with a frequency of 100 Hz and an offset of 1.0V, while the values of the convergence factors, μ_1 and μ_2, are 1.0×10^{-9} and 1.0×10^{-11}, respectively. Here, it is important to remember that the LMS algorithm model is $w_k(n+1)=w_k(n)+\mu e(n)x_k(n)$, where $w_k(n+1)$ is the next coefficient of the filter $w_k(n)$ is the actual coefficient, μ is the convergence factor, $x_k(n)$ is the input

signal and $e(n)$ is the error. However, when the ANC system includes the secondary CC, the estimation is on-line to detect its changes. This fact implies using another adaptive filter, and obtaining its optimal coefficients depends on the correct selection of the convergence factor. That explains why, in the ANC system design, two convergence factors are used. Then, for the electrical test, the noise generator is implemented by using random numbers, with values ranging from -180 to 180. Experimental measurements were captured with the Agilent DSO-X 3034A oscilloscope. It is observed that the system is capable of cancelling the input signal $x(n)$ by applying the anti-noise signal $y'(n)$, in approximately 80s. The error $e(n)$ is the curve with smallest amplitude in Fig. 6, where $0.1256V_{pp}$ corresponds to an attenuation of -18dB. In practice, the error $e(n)$ had already reached its minimum value e_{min} in 2s. However, because the highest values of μ_1 and μ_2 the value of e_{min} is reached in less time, a card-based design of experiments was done in order to determine the best values of μ_1 and μ_2. Table I shows the convergence time to reach the e_{min} value for the proposed variations, where ⊠ represents the combination of values that do not provide a solution. With these performance values, the designer can choose the most suitable ones according to the application under development.

In this work, the NPA415-O (PULSE®) omnidirectional microphone was chosen, and the SC 5.9 (Visaton®) speaker was our choice because it generates an average power of 10W, and its sensitivity is 84 dB (at 1.0 W) for a 1.0m distance. Additionally, a pair of preamplifiers were required to adapt the error signal and the reference one; microphones sense both signals. The preamplifiers are based on operational amplifiers, one to amplify the signal up to 33.4 dB and the other to adapt the signal to a level of 1.6 V. In addition, the TPA3125D2N power amplifier was selected to link the cancellation speaker with the controller DAC, and to amplify the power of the controller output signal up to 10 W. The TPA3125D2N is a 2-channel D-type audio amplifier that is configured to provide 20 dB gain. It is also of interest to know how the filter order affects the processing time. Therefore, an adaptive filter was synthetized in the same card, and the order was varied from 16 to 48 coefficients. It was observed that there is a linear relationship between the delay generated by the filter, δC, and its order, N. The results of the experiment shows that the total delay ($\delta E = \delta t + \delta C$) must be less than or equal to the noise transit time t_1, where the filter delay is approximately modeled by $\delta C \approx 0.625N + 25$.

IV. Conclusions

In this work, the correct functioning of the FxLMS algorithm with online estimation was verified. It was shown that the TPA3125D2N card has the frequency and resource requirements to satisfy the described application [15]. If the ANC system is stimulated with an electrical signal of $1.0V_{pp}$, at a frequency of 100 Hz, the system is capable of reducing this signal by means of an anti-noise signal, obtaining and error of 220 mV_{pp} (-13.15dB), in a time of 9s. This error includes the card floor noise, which is mainly thermal noise. For Comparison Purposes the authors are considering the Nucleo F446RE. Both cards are capable of processing digital signals at speeds in the order of hundreds of MHz. Both cards are optimal for implementing the driver.

However, while the TAdsp28335 card has advantages in memory capacity, the Nucleo card has a processing speed larger and includes more bits for D/A conversion; its processing speed is better to cancel acoustic noise [16]. Therefore, future work includes evaluating the various design considerations imposed by the ANC, without forgetting the relevance of low cost.
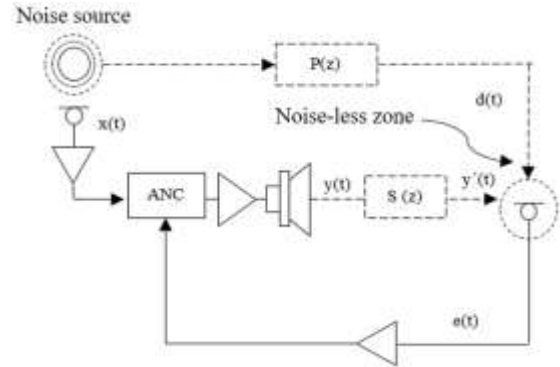

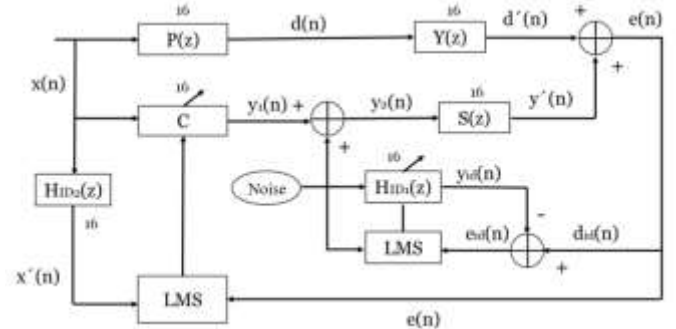Figure 4. Communication channels in an ANC system.

Figure 5. Block diagram of the ANC system.

Figure 6. Measurements of $e(n)$ and $x(n)$ in the ANC system.

TABLE I. FxLMS ALGORITHM CONVERGENCE TIME

μ_1	μ_2						
	10^{-14}	10^{-13}	10^{-12}	10^{-11}	10^{-10}	10^{-9}	10^{-8}
10^{-13}	>120s	>120s	>120s	>120s	>120s	>120s	⊠
10^{-12}							
10^{-11}						⊠	
10^{-10}					70s	27s	
10^{-9}				80s	24s		
10^{-8}			90s	30s	9s	⊠	
10^{-7}		85s	30s	13s	⊠		

Acknowledgment

References

[1] P. de Aquino Morais Junior *et al*, "Characterization of Automobile Acoustic Insulation Performance based on Artificial Neural Networks via Sensitivity Analysis," IEEE Latin America Transactions, vol. 16, no. 1, pp. 222-231, January 2018.

[2] H. Vu and K. Chen, "A 1.4 mW low-power feed-back FxLMS ANC VLSI design for in-ear headphones," International Symposium on VLSI Design, Automation and Test, Hsinchu, pp. 1-4, 2016

[3] F. A. Gonzalez *et al*, "Adaptive Narrowband ANC, Design and Implementation Issues," IEEE Latin America Transactions, vol. 9, no. 4, pp. 438-444, July 2011.

[4] S. Hirose and Y. Kajikawa, "Effectiveness of headrest ANC system with virtual sensing technique for factory noise," Asia-Pacific Signal and Information Processing Association Annual Summit and Conference, Kuala Lumpur, pp. 464-468, 2017.

[5] B. Widrow, "Adaptive filters I: Fundamentals," Stanford Electronics Labs., Stanford, Calif., Rept. SEL-66-126 ('Tech. Rept. 6764-6), December 1966.

[6] A. G. Troshin, S. Cha and D. Kim, "Evaluation and parametric study of LMS algorithms family for active noise control barriers international conference on control, automation and systems," 11th International Conference on Control, Automation and Systems, Gyeonggi-do, pp. 1656-1659, 2011.

[7] E. Eweda, "Comparison of RLS, LMS, and sign algorithms for tracking randomly time-varying channels," IEEE Transactions on Signal Processing, vol. 42, no. 11, pp. 2937-2944, November 1994.

[8] E. Eweda, "Transient performance degradation of the LMS, RLS, sign, signed regressor, and sign-sign algorithms with data correlation," IEEE Transactions on Circuits and Systems II: Analog and Digital Signal Processing, vol. 46, no. 8, pp. 1055-1062, August 1999.

[9] R. Merched, "Extended RLS lattice adaptive filters," IEEE Transactions on Signal Processing, vol. 51, no. 9, pp. 2294-2309, September 2003.

[10] Yegui Xiao, Liying Ma, K. Khorasani, A. Ikuta and Li Xu, "A filtered-X RLS based narrowband active noise control system in the presence of frequency mismatch," IEEE International Symposium on Circuits and Systems, Kobe, pp. 260-263 (Vol. 1), 2005.

[11] Iman Tabatabaei Ardekani, Waleed H. Abdulla, "FxLMS-based Active Noise Control: A Quick Review", APSIPA ASC 2011 Xi'an, China.

[12] M. Rupp and A. H. Sayed, "Two variants of the FxLMS algorithm," Proc. of Workshop on Applications of Signal Processing to Audio and Accoustics, New Paltz, pp. 123-126, 1995.

[13] Edgar Lopez-Gaudana et all, "A Hybrid Active Noise Cancelling with Secondary Path Modeling", IEEE, 978-1-4244-2167-1/08-2008.

[14] C. Burgess, Active adaptive sound control in a duct: A computer simulation, J. Acoustical Soc. Amer., vol. 70,pp. 715726, September 1981.

[15] E.O. Lara Alemán, F. Sandoval-Ibarra, S. Ortega-Cisneros, Jorge Rivera y J. R. Loo-Yau, "Factibilidad de la tarjeta TAdsp28335 como controlador en un sistema orientado a la cancelación de ruido acústico", Proc. of the 3er. Coloquio en Electrónica Analógica y Digital, Zapopan, pp. 1-4, November 2019 (In Spanish).

[16] E. O. Lara-Alemán, "Diseño e implementación de un sistema ANC para cancelar ruido de banda angosta en un rango de frecuencia de 100 Hz a 1.5 kHz," D. Sc. Thesis, Cinvestav-Guadalajara Unit, January 2020 (In Spanish)

E. O. Lara-Alemán received the BS degree in Mecatronics from Instituto Tecnológico de Celaya (ITC), and MSc and DSc degrees in Electronics from Centro de Investigación y de Estudios Avanzados (CINVESTAV), Guadalajara Unit, Jalisco, Mexico, in 2015 and 2020, respectively.

His research interests are on the efficient design and implementation of digital signal processing algorithms, digital systems design and analysis of algorithms for canceling acoustic noise.

F. Sandoval-Ibarra received the BS degree in Physics Electronics from Facultad de Ciencias, Universidad Autónoma de San Luis Potosí (UASLP), San Luis Potosí, Mexico, in 1988, and MSc and DSc degrees in Electronics from Instituto Nacional de Astrofísica, Óptica y Electrónica (INAOE), Puebla, Mexico, in 1991 and 1998, respectively.

He was a researcher at Laboratorio de Microelectrónica, INAOE, from 1991 to 1997, where he developed wet-etching techniques for designing silicon-based microsensors. In 1997, he was with CNM-Bellaterra (Spain), as a visiting researcher being involved in the development of surface micromachining techniques to design a fully-integrated microphone. He was a full-time Professor at Instituto Tecnológico de Puebla (ITP), in 1998. Since 1999, he has been a Professor and Researcher at Cinvestav, Guadalajara Unit (GU). From 2015 to 2020, he was the Coordinator of the Electronic Design Group at GU. His research areas include the design of low-noise analog integrated CMOS circuits, mixed-mode circuits design for RFID applications and the physical modelling of electronic noise using digital signal processing algorithms. He is currently working on the design of A/D converters based on Hybrid ΣΔ Low-Pass Modulators for multi-standard applications.

Dr. Sandoval-Ibarra is a member of Sistema Nacional de Investigadores (SNI), Mexico.