Design and analysis of a digital active noise control system for headphones implemented in an Arduino compatible microcontroller

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Abstract - *Active noise control can be a powerful tool when dealing with problematic low frequencies tones or general broadband noise. The primary goal of this paper is to expose the design methods and analyze the performance measurements of an Arduino compatible active noise control system for headphones. The measured performance was impaired by the poor coupling between the headphone and the pinna of the head and torso simulator used in the measurements. For this reason, the complete measurement set was carried out with and without the artificial pinna (in this first step only the results without the pinna are discussed). The algorithm used for this research was the FXLMS (Filtered Least Mean Squared), which is a variation of the Least Mean Square algorithm (LMS). Although the system was designed to attenuate broadband noise, great results were obtained for both broadband and tonal noise, showing that is possible to adapt a headphone to substantially attenuate noise with easy to use microcontrollers.*

Keywords: *headphones, Arduino, acoustics, FXLMS, active noise control, Teensy 3.6.*

Introduction

Active Noise Control (ANC) can be a useful tool when reducing excessive and/or undesired Sound Pressure Levels (SPL). This technique is based on the wave superposition principle and it works by generating an anti-noise that cancels the primary noise when they are added (in the same sound field) [1].

Passive noise control methods, like porous materials, barriers, and enclosures, usually aren't able to reduce significantly SPL at lower frequencies, unless they are very large and bulky, which may be impossible to implement due to space limitations. On the other hand, according to Kuo e Morgan [2], ANC works mainly in the low-frequency range. This fact arises from the difficulty of adjusting the anti-noise's phase for high frequencies.

The application of this method in headphones has been used in the field of aviation, specifically in communication devices used by pilots, to ensure good communication even with loud wind and engine noise. Furthermore, in the past few years, several headphones designed for audio and music have been developed with this ANC system, the goal in this situation is to make possible listening to songs in good quality even in loud environments such as train stations and airports.

The main purpose of this article is to evaluate the performance of a digital active noise control system that was implemented in a Teensy 3.6 board. This board has a ARM cortex M4 processor and can be programmed with the same programming language and Integrated Development Environment (IDE) utilized for Arduino boards, which can be useful to people who are used to work with them. Moreover, this is also interesting to new programmers, since there are several easy-to-use libraries and the company's forum provides good tutorials and code examples.

The additional microphone preamplifier and headphone driver circuits that were necessary to the development of this project are going to be published soon in a more detailed paper (discussing circuit diagrams and codes).

Materials and methods

In this section, the necessary theories regarding the digital active noise control technique used are briefly explained. The specification of the measurement systems, as well as the hardware used, is given.

1 Feedback active noise control

According to Elliot e Nelson [1], one of the most successfully use of feedback ANC control has been applied to the design of wideband noise control systems for headphones. A feedback control system, firstly described by Olson e May [3], is based upon an error microphone in the position where the reduction is desired and a loudspeaker close to the microphone. A diagram of such configuration can be observed in Figure 1. The controller purpose is to generate a signal to the loudspeaker that minimizes the error captured by the microphone.

The major problem of this configuration is that the phase distortions due to the circuitry, acoustic path between microphone and loudspeaker and the loudspeaker itself can impair the performance of the controller and even generate positive feedback at higher frequencies (if the phase delays reach more than 180 degrees). Thus, these secondary systems have to be accounted for and the distance between the microphone and the loudspeaker (source-receiver) should be small as possible.

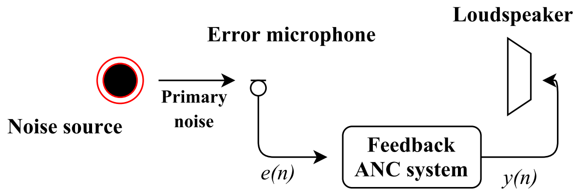

Figure 1 – Basic model of a feedback active noise control system (adapted from [2]).

2 FXLMS algorithm

The algorithm used for this research was the FXLMS (Filtered Least Mean Squared), which is a variation of the Least Mean Square algorithm (LMS). The method is based on an adaptive finite impulse filter that varies its coefficients in order to minimize the square of the error measurement [2] (or to minimize the variance of error signal). A complete diagram o the algorithm can be found in Figure 2.

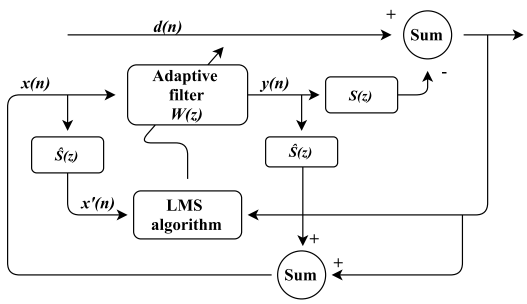

Figure 2 – Diagram of the FXLMS algorithm (adapted from [2]).

The equations related to the filter and the adaptive algorithm will be shown throughout this section (further reading about LMS may be found in [4]). Herein, vectors and matrices will be denoted by uppercase letters and scalar quantities denoted by lowercase letters. The FIR filter is defined as a vector $W(n)$ with L coefficients and the input vector $X(n)$ is defined as a vector of the same size, where $x(n)$ represent the current input value, $x(n-1)$ the immediately past input value and so on and so forth. The cited vectors can be denoted by

$$W(n) = [w_0(n)\ w_1(n)\ ...\ w_{L-1}(n)]\ , \qquad (1)$$

and

$$X(n) = [x(n)\ x(n-1)\ ...\ x(n-L+1)]\ . \qquad (2)$$

For each discrete value of time n, the error is given by the microphone's measurement.

The output of the filter can be computed as a real time convolution between the filter's impulse response and the input vector (Equation 3). This convolution can also be expressed as the vector product of the transposed version of vector $W(n)$ and the input vector. Accordingly,

$$y(n) = \sum_{i=0}^{L-1} w_i(n)\, x(n-i) = W^T(n)\, X(n)\ . \qquad (3)$$

There is a significant amount of phase and amplitude distortion between the exit and the input of the controller, called secondary path. These distortions are given by the loudspeaker, preamplifier, A/D and D/A converters, microphone and acoustic path between the microphone and the loudspeaker. The sum of these systems' influence is denoted by $S(z)$ in the diagram (the z is the derived by the use of Z-transform). If these distortions are not taken into account, the algorithm might become unstable. Therefore, an estimate of the secondary path ($\hat{S}(z)$) is performed to adjust the LMS[1] algorithm, hence, becoming FXLMS.

The Equation 4 is responsible for updating the adaptive filter's coefficients in order to minimize the instantaneous squared error. The step size, represented by μ, coordinates the rate in which the algorithm converges [5], therefore,

$$w(n+1) = w(n) + \mu\, x'(n)\, e(n)\ , \qquad (4)$$

where $e(n)$ is the error (considering a threshold) and the $\{\cdot\}'$ indicates that a sample has passed through the $\hat{S}(z)$.

Since there is only one microphone that measures the error, it is necessary to estimate the primary noise. Thus, the estimate of a given $\hat{S}(z)$ of size M yields

$$x(n) \equiv \hat{d}(n) = e(n) + \sum_{m=0}^{M-1} \hat{s}_m\, y(n-m)\ . \qquad (5)$$

Finally, the primary noise and the secondary path estimates are convoluted to generate the signal that is used to update the filter's coefficients. The convolution can be expressed by

$$x'(n) = \sum_{m=0}^{M-1} \hat{s}_m\, x(n-m)\ . \qquad (6)$$

In practice, the algorithm is not able to reach the exactly the optimal solution. However, it achieves a fairly close point. The measure of how close the solution reaches the optimum is called misadjustment. If the step size is small, the algorithm will

[1] The use of FXLMS was needed to overcome instabilities in the LMS approach.

take longer to converge but the solution will get closer to the optimum, thus, the misadjustment value will be smaller. If μ is greater, the opposite event occurs, a fast convergence shall be expected, but after convergence, the solution will be far (or less close) to the desired value than with a small step size [6].

The shown algorithm is implemented in a streaming format. At each input sample converted by the controller, an output sample must be computed and emitted before the next input sample is gathered. Therefore, the sampling period must be higher than the time it takes the processor to compute the equations above (thereby decreasing the sampling frequency).

In this study, the sampling frequency was not set as constant. At the codes loaded in the boards, each input sample is obtained as soon as the output sample is converted into analog value. Consequently, the sampling rate varies with the size of the filter. For this reason, there is a trade-off between filter size and sampling rate. If the filter size is too small, it might not be possible to achieve the necessary impulse response needed for a good performance. However, if the size of the filter is too wide, the sampling frequency, as well as the maximum frequency of analysis, is accordingly reduced. A continued study will consider fixed sampling rate.

3 Secondary path estimation

The secondary path estimation is obtained based on the system identification technique. According to Morgan and Kuo [5], the basic idea behind the system identification procedure is to construct a model based upon a measurement of the signal produced by the system. The diagram of the secondary path estimation can be consulted in Figure 3.

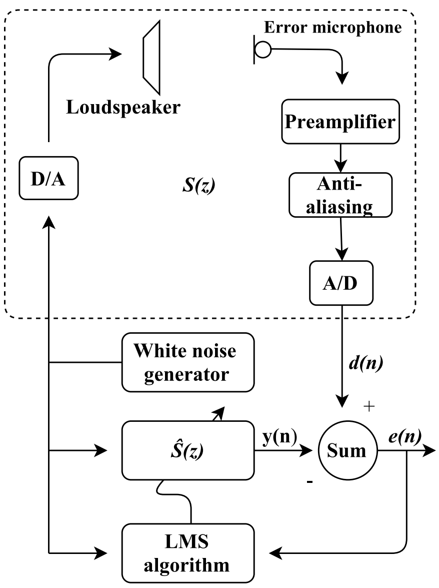

Figure 3 – Secondary path estimation diagram (adapted from [2]).

An input signal $x(n)$, usually a wideband signal (such as a white noise), is generated by the processor and it serves as inputs to both adaptive filter and the secondary path.

The output of $S(z)$, expressed in the diagram by $d(n)$, and the output of the adaptive filter $\hat{S}(z)$, expressed by $y(n)$, are subtracted to generate an error signal $e(n)$. The error, as well as the input signal, is used by the minimization algorithm to adjust the filter to minimize the difference of outputs.

When the error reaches its minimum (or threshold), the IR of the adaptive filter is (in an optimum way) emulating the impulse response of $S(z)$.

4 Hardware and measurement

The performance measurements were made with a head and torso simulator (HATS) Type 4128C from B&K. The experiment was carried out inside a reverberation chamber with an omnidirectional sound source (see Figure 4). The signals tested were white noise and a 500 Hz tonal noise. The data was collected with and without sound reproduction from the dodecahedron.

Figure 4 – Instrumentation used to obtain the performance measurements.

During the measurements, it was noticed that the HATS's ear was too big for the chosen headphone, and, subsequently, the coupling was unsatisfactory. The measurements were then carried out with and without the ears, for comparison purposes. Later, it was possible to realize that the bad coupling has impaired the performance of the control system. Therefore, only the results measured without the ear will be shown herein.

The headphone used throughout this experiment was the AKG K44 and the error microphone was the JLI-61A. The microphone was placed next to the HATS's microphone and connected to the preamplifier circuit that was designed for the controller. The headphone was set to the simulator and connected to the controller's output.

A variety of measurements were collected for both, white noise and the 500 Hz tone, varying the filter length for a given (fast converging) value of μ. The best results are shown in the next section.

Results

Considering the narrowband test, the 500 Hz tone has achieved the best result out of a filter length of 50. The SPL reduction obtained was quite large, almost reaching a 40 dB reduction, as can be seen in Figure 5. Nonetheless, more measurements and analysis must be carried out to determine whether the gain noticed in the frequency of 1000 Hz is due to harmonic distortions (THD) of the preamplifier or to the system itself.

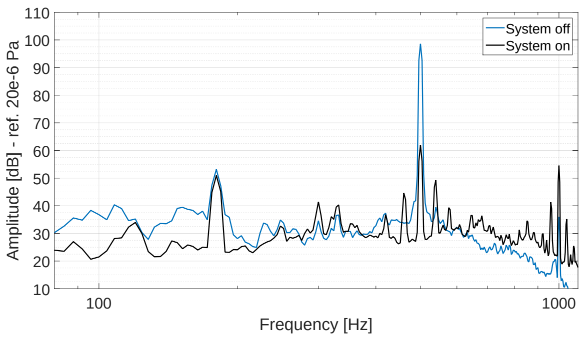

Figure 5 – Performance of the ANC system with a 50 taps filter for a 500Hz tone.

The results obtained for the wideband noise were not as expressive as for the tonal noise, as per Figure 6 (note that the amplitude scale is different from Figure 5 to better compare on/off situations). Nevertheless, some reduction was obtained for a frequency range from 50 Hz up to 700 Hz. This result was obtained with a 15 taps filter. Although the small filter length might be limiting, a smaller length means a larger sampling rate. For the filters used to the tone and wideband noise measurements, the sampling rates obtained were 25 kHz and 45 kHz, respectively.

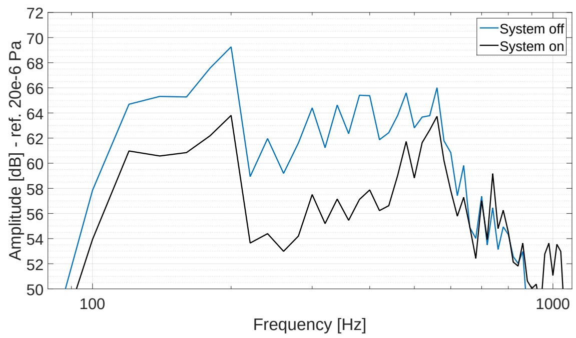

Figure 6 – Performance of the ANC system with a 15 taps filter for white noise.

It is important to notice that these results were achieved for a large value of μ (convergence time below 1 s). Therefore, better results could be obtained with lower values of this parameter at the expense of a slower convergence time.

According to some measurements performed by Brent Butterworth [7], commercial headphones with active noise control can reach 20 to 25 dB of reduction in a frequency range from 20 Hz to about 800 Hz (for wideband noise). Therefore, several improvements must be made to the system presented in this paper in order to accomplish similar results as commercial noise-canceling headphones.

Discussions and conclusions

Although this system is still far from achieving the same reduction as the commercial solutions, satisfactory results were obtained for both noise tests: tonal and wideband. Even better results may be reached with further developments of the preamplifier circuit and by the correct adjustments of the filter size and the value of the step size. In addition, the problem regarding the bad coupling between the headphone and the measuring instrumentation may be solved by changing the headphone model or obtaining a smaller ear for the head and torso simulator. Further studies are still in development.

References

[1] S. ELLIOTT and P. NELSON, "Active noise control," *IEEE SIGNAL PROCESSING MAGAZINE*, 1993.

[2] S. M. KUO and D. R. MORGAN, "Active noise control: A tutorial review," *PROCEEDINGS OF THE IEEE*, 1999.

[3] H. F. OLSON and E. G. MAY, "Electronic sound absorber," *THE JOURNAL OF THE ACOUSTICAL SOCIETY OF AMERICA*, 1953.

[4] P. M. Clarkson, *Optimal and Adaptive Signal Processing: A Volume in the Electronic Engineering Systems Series: 3*. CRC-Press, 1993.

[5] S. M. Kuo and D. R. Morgan, "Active noise control systems – algorithms and dsp implementation," *Wiley Series in Telecommunications and Signal Processing*, 2009.

[6] J. A. A. Jr and S. L. Netto, *QRD-RLS Adaptive Filtering*. Springer, 2009.

[7] B. Butterworth, "Review and measurements: Bose QC25 noise-cancelling headphone," 2017. Available in `https://www.lifewire.com/bose-qc25-review-specs-3134560`. [Accessed Nov. 2017].