External Noise Cancelling: Design of a noise canceling prototype for an urban environment

Francescolli Anticona
Universidad de Ingeniería y Tecnología - UTEC
Lima, Perú
gustavo.anticona@utec.edu.pe

José Del Maestro
Universidad de Ingeniería y Tecnología - UTEC
Lima, Perú
jose.delmaestro@utec.edu.pe

Moises Nuñez Ochoa
Universidad de Ingeniería y Tecnología - UTEC
Lima, Perú
mnunez@utec.edu.pe

***Abstract*—Noise pollution comes mainly from three main sources of noise such as traffic, industry and commerce, and high-decibel events. In the future, people who are exposed to this noise will suffer severe hearing damage. Today there are many solutions to this problem, as technological advances are constantly changing and are fundamental to solve the problems that disturb the calmness of humankind. There are several ways to deal with this, the main attraction being noise reduction. Noise reduction success depends on the needs of the end-user and the applications that can be given to noise-canceling devices. The purpose of this project is to analyze the three most used algorithms such as the Least Mean Square (LMS), Normalized Least Mean Square (NLMS), and Recursive Least Square (RLS) for noise removal in an urban environment. It is possible to reduce the noise in different scenarios, being the Normalized Least Mean Square (NLMS) the algorithm is the most optimal in all the results. This noise reduction within the analysis makes possible its future application in physical devices that have the characteristic of reducing noise and achieving the well-being of users.**

***Index Terms*—Noise pollution, decibel, active noise control, IoT, HRTF, LMS, NLMS, RLS**

I. Introduction

The principle of noise pollution refers to the level of noise in the environment that is intended to disturb, damage, or affect human health and well-being. Noise alters the stability of people and, over time, can have a variety of physical consequences. Among the major health consequences caused by human exposure to high noise levels are stress, high blood pressure, dizziness, insomnia, speech difficulties, and hearing loss. Certain groups of people, such as the chronically ill and the elderly, tend to rest more than others and are more sensitive to noise than others. In addition, this phenomenon can particularly affect children and their ability to learn, such as hearing loss, interrupted sleep, stress-related disorders, and psychological consequences, such as memory loss, severe depression, and panic attacks. [4]

To understand how noise cancellation occurs, you first need to understand how sound travels through our ear canal to reach our brain, and how we interpret all the sounds around us. The source of sound emission travels through waves and they pass through a conduit called the narrow ear canal. These waves reach the eardrum and cause the eardrum to vibrate, which in turn makes three small bones called the Hammer, Anvil, and Stirrup vibrate respectively. These little bones amplify sound and send these waves to the Cochlear. The Cochlea is full of fluid, which is why it generates vibrations causing this fluid to ripple. Inside the Cochlea are impregnated at the base Stereocilia similar to hair strands, the set of Stereocilia oscillate according to the movement of the waves and are known as hair cells. The hair cells are responsible for carrying the information obtained to the auditory nerve and generating the vibrations received in electrical impulses to be processed by the brain. Hair cells located in the cochlea respond to different sound frequencies. The hair cells located at the base or widest sector of the cochlea manage to detect the highest-pitched sounds such as a flute or a bagpipe. The hair cells located in the middle sector of the spiral manage to detect progressively less acute sounds such as a saxophone or a cello. Finally, in the upper part or dome of the spiral, it is possible to detect much lower sounds such as a bass or an oboe. [3]

The intensity of different sounds is measured in decibels (dB). The hearing threshold, measured in dB, starts at zero (0) dB and reaches a maximum level of 120 dB (this is the stimulus level at which people begin to experience pain and/or discomfort). The WHO (World Health Organization) recommends not exceeding 55 decibels (dB) during the day. Noise sources can be mobile (e.g., airplanes, ships, trains, vehicles) or stationary (e.g., industrial sites, construction sites, commercial areas). According to a noise level study conducted by the Universidad Federico Villarreal de Miraflores, Figure 1 shows that vehicle horns and sirens are the main sources of noise emissions. [1]

As we can see in the following graph, the main sources of noise in residential areas are mobile sources, which are closely related to traffic schedules in the main points of the city. Some technologies passively cancel noise, such as sound-absorbing materials (e.g., anti-noise walls, baffles, weather-stripping), earplugs, hearing protectors, etc. This technology is effective in reducing noise; however, these materials are

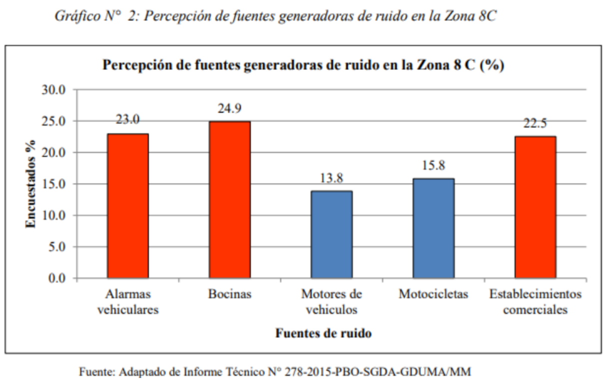

Fig. 1. Main sources of noise in Miraflores. [1]

large and expensive, and it is necessary to know how to install in the field. The interaction between the head, ears, and shoulders is the result of a transformation between the eardrum and the sound source to perceive the sound around us. Active Noise Control (ANC) systems are a more precise and effective solution to eliminate noise that is harmful to people's health. An algorithm is still being developed to reduce noise as much as possible. For this reason, ANC headphones are marketed with active noise control, which improves the well-being of listeners by allowing them to better concentrate on their activities.

The ANC system is a method based on the principle of superposition, where an additional source produces an opposite phase of the same magnitude as the main source to eliminate unwanted noise. Most of these ANC systems use adaptive filters, one of the most commonly used is the finite impulse response (FIR) filter. According to Floyd Freeston, the ANC (Active Noise Control) principle treats the head as a rigid sphere and analyzes it by placing two sound pressure bars at the ears 180 degrees apart. When the sound source is in one ear, the interaural level difference (ILD) ranges from 500 Hz to 4 kHz. The interaural time difference (ITD) is the difference between the left and right ear, and it is important to know the location of these sound sources to indicate their direction in front of the head. ITD, ILD, and transforms are basic human functions for localizing surrounding sound sources. [2]

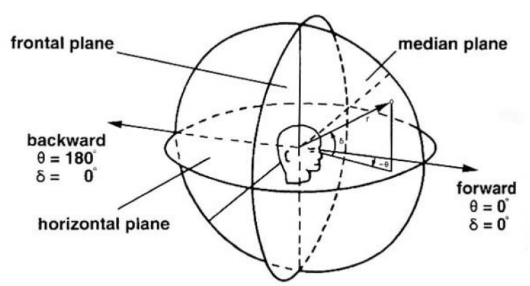

Fig. 2. ANC-Head reference in space. [2]

After understanding how ANC works and how it affects us as end users. We can see different uses of this concept in different projects that have been done in different specific fields. For example where FIR filters were tested for their performance to cancel snoring. According to Kajikawa(2012) states that "Snoring is an acoustic phenomenon generated by the vibration of tissue structures due to the obstruction of the upper respiratory tract during sleep, and is a prominent problem in modern society" (p.11) The application of the ANC system to reduce snoring noise can be seen in the following image.

Fig. 3. Experimental setup for the snore ANC inside an acoustic chamber. [10]

Another case is an incubator using a hybrid ANC system. In addition, intrauterine sounds can reduce stress, improve infant neurological development, and reduce infant stress. For this reason, an ANC system with a sound that can reproduce intrauterine sounds for infants is being developed.

Fig. 4. Hatchery using a hybrid ANC system. [10]

Some projects are much more competitive in terms of the applications they have, such as implementing IoT in their features. MUTE is a project from the University of Illinois that uses everything we have seen so far. LANC (Lookahead Noise Cancellation Algorithm) is used to transmit the wireless signal faster than the sound to eliminate noise. The basic algorithm is used by the receiver receiving the audio signal to calculate the anti-noise signal and finally play it back to the speaker. The appeal of this project is that they offer an IoT theme.

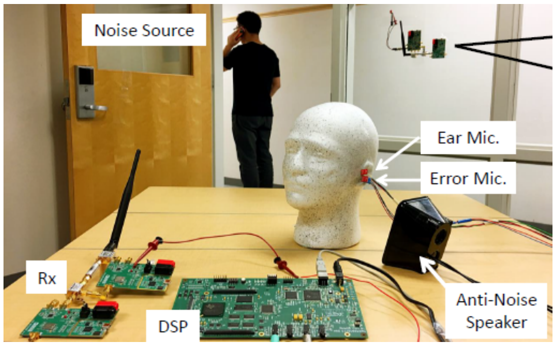

Fig. 5. MUTE's experimental platform. [8]

This study compared MUTE with ANC headphones and the results favored MUTE. The advantage of using a wireless signal is that it is completely beneficial for sound detection. IoT devices can listen to surrounding noises and transmit signals that are sent via radio. With this technology in mind, it has been shown to outperform ANC headsets and the new technology is well on its way to improving this noise problem in our environment. [8]

II. Technology Overview

Active noise cancellation (ANC) systems have at least two microphones and a loudspeaker (Figure 6). The microphone closest to the eardrum is called the error microphone Me (Microphone Error), and the one farther away from the ear is called the reference microphone Mr (Microphone Reference). The speakers are close to me and are called noise cancellers. The ambient noise reaches the master first, then me, and finally the eardrum. The purpose of the DSP is to get the sound from Mr, calculate the attenuator, and reproduce it through the speakers so that the attenuator eliminates the ambient noise at Me.

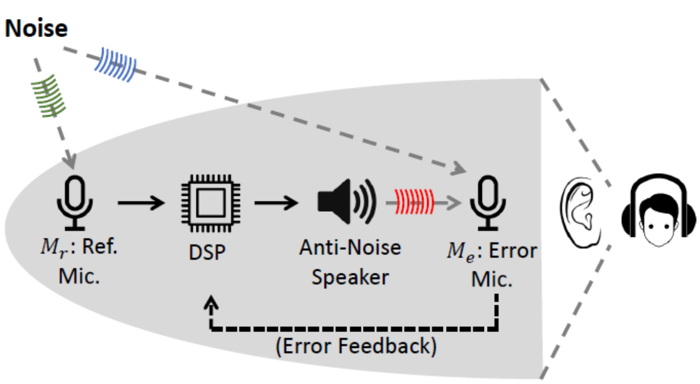

Fig. 6. ANC operating architecture. [8]

In practical applications, the characteristics of the noise source, and the acoustic environment are changing, and therefore the frequency content, amplitude, and phase of the primary noise are also changing. are also changing. Noise reduction performance depends mainly on the accuracy of the amplitude and phase of the anti-noise generated by a signal processing algorithm. signal processing algorithm. To deal with these time-varying problems, most ANC systems use adaptive filters to track these variations and unknown plants. The most commonly used adaptive filters are realized using a finite impulse response (FIR) filter with the least mean square (LMS) algorithm.

The ANC control structure is generally classified into two classes: feedforward control and feedback control. In the case of feedforward control, it is assumed that a reference noise is available for the adaptive filter. Feedforward ANC systems can be classified as broadband or narrowband depending on the type of primary noise that can be reduced. In the case of broadband feedforward control, a reference sensor (e.g. a microphone) detects a reference noise and therefore the noise that correlates with the reference noise can be reduced.

To better understand how the feedforward ANC model works, extracted from the Matlab repository: "The following figure illustrates a classic example of feedforward ANC. A noise source at the entrance of a duct, such as a fan, is "canceled" by a loudspeaker. The noise source b(n) is measured with a reference microphone, and the signal present at the output of the system is monitored with an error microphone, e(n). Note that the smaller the distance between the reference microphone and the loudspeaker, the faster the ANC must be able to compute and playback the "anti-noise"."

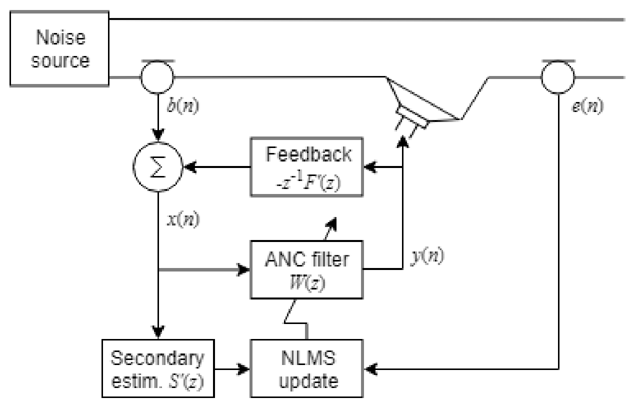

Fig. 7. Initial parameters for audios.

III. Methodology

In the following section, we will show the methodology we will use. The purpose of this paper is to show a design in Matlab that simulates the functions of the physical DSP, which must acoustically isolate the noise perceived in different environments once determined the algorithm for the DSP to work, we will determine the manufacturing cost of this equipment. To perform these two activities it is necessary to investigate the state of the art of current solutions for noise cancellation, mainly of the LMS, and NLMS algorithms, once this research is done, we will perform the corresponding tests by collecting samples of acoustic noise at different points of Lima, which are: Miraflores, Via Expresa, Javier Prado, Barranco, and Acho. Once this data collection is finished, we will make a comparative table explaining which is the most appropriate algorithm for each specific area.

A. LMS Algorithm

The Least Mean Square algorithm or its acronym known as LMS (Least Mean Square) was used for the first time by Widrow and Hoff in 1960 and is currently the most widely used adaptive filtering algorithm. This algorithm uses a stochastic approximation to calculate the gradient of the least square function. This algorithm requires a large number of iterations to achieve convergence, it even requires a larger number of coefficients to work. The error computation is inferior to other adaptive algorithms, but it is much easier to understand mathematically than other algorithms. It presents linear characteristics so that the system does not crash, which is why it is widely used as an adaptive filter algorithm. In general, the LMS algorithm converges better than other algorithms such as RLS, CMA, and DMI.

```
function [e, y, w] = alg_LMS(d, x, mu, M)

Ns = length(d);
if (Ns <= M)
    print('error: La longitud de la Señal
    es menor que el orden del filtro');
    return;
end
if (Ns ~= length(x))
    print('error: La Señal de entrada y
    la Señal de referencia tienen
    diferentes longitudes');
    return;
end

x = x;
xx = zeros(M,1);
w1 = zeros(M,1);
y = zeros(Ns,1);
e = zeros(Ns,1);

for n = 1:Ns
    xx = [xx(2:M);x(n)];
    y(n) = w1' * xx;
    e(n) = d(n) - y(n);
    w1 = w1 + mu * e(n) * xx;
    w(:,n) = w1;
end
end
```

B. NLMS Algorithm

The Normalized LMS algorithm (Zhao, et al, 2013; Jamel, 2013) aims to make the convergence independent of the power of the input signal, therefore, it is more robust than the LMS algorithm. This algorithm uses the maximum slope method, where the estimation of the gradient is very noisy, and the convergence factor presents a compromise between convergence and precision, that is, it varies over time. The NLMS is the most convenient algorithm when the appropriate convergence factor is unknown to achieve optimal filter performance since it takes into account the power of the reference signal, ensuring the convergence of the algorithm. The NLMS the algorithm generally converges much faster than the LMS with a very small number of extra operations; however, it has been shown to be unstable in its regular form when the convergence factor tends to approach the acceptable upper limit, especially in the initial stages.

The equations that specify the NLMS algorithm are summarized in equation 17, which, when substituted into equation 11, define the expression of equation 18: Where and are constant parameters. In general, for both methods, a smooth estimate of power occurs when the (effective) window length is larger, but then the filter will not be able to respond to sudden power changes (Zhao, Et. Al, 2013). Intuitively, in the NLMS algorithm, the number of operations is increased with respect to the LMS, when calculating the power of the input signal. However, the complexity of the NLMS algorithm is relatively low of the order O(3N), which implies that in each iteration, updating a coefficient requires 3N +1 multiplications and 3N additions.

```
function [e,y, w] = alg_NLMS(d, x, mu, M, a)
Ns = length(d);
if (Ns <= M)
    print('error: La longitud de la Señal
    es menor que el orden del filtro');
    return;
end
if (Ns ~= length(x))
    print('error: La Señal de entrada y
    la Señal de referencia tienen
    diferentes longitudes');
    return;
end

x = x;
xx = zeros( M,1);
w1 = zeros( M,1);
y = zeros(Ns,1);
e = zeros(Ns,1);

for n = 1:Ns
    xx = [xx(2:M);x(n)];
    y(n) = w1' * xx;
    k = mu/(a + xx'*xx);
    e(n) = d(n) - y(n);
    w1 = w1 + k * e(n) * xx;
    w(:,n) = w1;
end
```

```
end
```

C. RLS Algorithm

The RLS or recursive least square algorithm is a derivative of the least square algorithm. One of its advantages is that it does not need to perform many iterations to converge, even performing more complex mathematical operations than the main least-squares algorithm. The Kalman filter is essential for this algorithm to reach the solution. This filter is based on identifying the hidden state of a linear dynamic system. Because it is a recursive algorithm, it works in real-time taking into account the current input measurements.

```
function [e, y, w] = alg_RLS(d, x,lamda,M)

Ns = length(d);
if (Ns <= M)
    print('error: La longitud de la señal
    es menor que el orden del filtro');
    return;
end
if (Ns ~= length(x))
    print('error: La señal de entrada
    y la señal de referencia tienen
    diferentes longitudes');
    return;
end


I = eye(M);
a = 0.01;
p = a * I;

x = x;
w1 = zeros(M,1);
y = zeros(Ns, 1);
e = zeros(Ns, 1);
xx = zeros(M,1);

for n = 1:Ns
    xx = [x(n); xx(1:M-1)];
    k = (p * xx) ./ (lamda + xx' * p * xx);
    y(n) = xx'*w1;
    e(n) = d(n) - y(n);
    w1 = w1 + k * e(n);
    p = (p - k * xx' * p) ./ lamda;
    w(:,n) = w1;
end
end
```

IV. Results

To test the NLMS, LMS, and RLS algorithms, we will work with 5 different noises collected at different points of the capital to see which is the most appropriate algorithm for these points. As a first step, we must obtain the signal of the collected audio, and then pass them through the DSP.FILTER which gives us a signal that we will call fsignal, and we complete the following parameters.

Once we have finished placing the initial parameters for each algorithm, we wait for each algorithm to perform the corresponding function with the values we have entered. The results obtained will be shown below.

A. ANC with Matlab functions

1) Noise from AV. Javier Prado in front of La Cultura Station:
Date: Friday, November 11
Time: 6 pm
Best filtering: NLMS

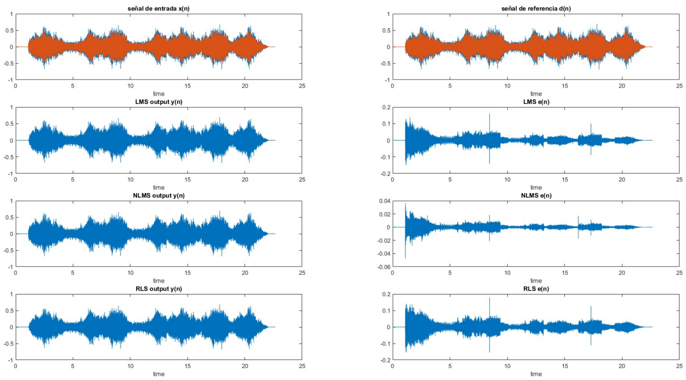

Fig. 8. LMS, NLMS, RLS graphs for noise 1.

Algorithm	Operating time
LMS	1.741042 seconds.
NLMS	1.699327 seconds.
RLS	3.490445 seconds.

2) Noise from Puente Santa Anita - Panamericana Norte:
Date: Thursday, November 10
Time: 3 pm
Best filtering: NLMS

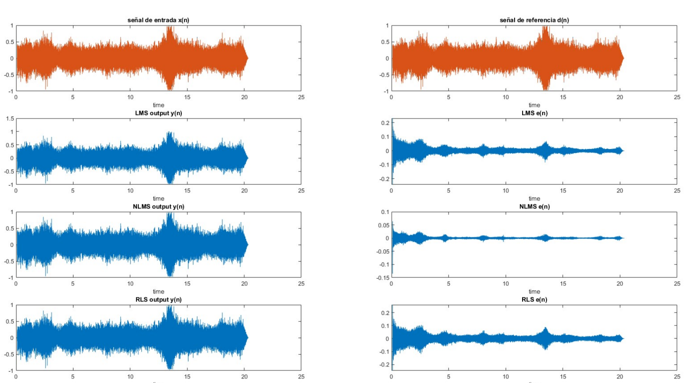

Fig. 9. LMS,NLMS,RLS graphs for noise 2.

Algorithm	Operating time
LMS	1.728273 seconds.
NLMS	1.826116 seconds.
RLS	3.625372 seconds.

3) Noise from Puente Atocongo - Panamericana Sur:
Date: Thursday, November 17
Time: 9pm
Best filtering: NLMS

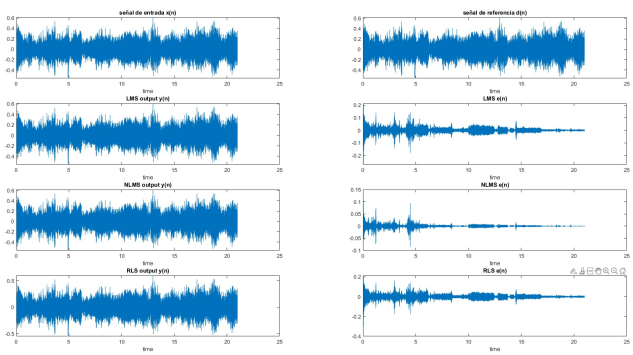

Fig. 10. LMS,NLMS,RLS graphs for noise 3.

Algorithm	Operating time
LMS	1.709559 seconds.
NLMS	1.546547 seconds.
RLS	3.405095 seconds.

4) Noise from Barranco - Chipoco Stadium:
Date: Saturday, November 19
Time: 9pm
Best filtering: NLMS

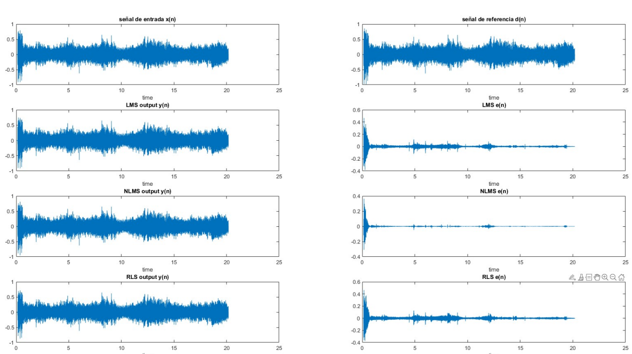

Fig. 11. LMS,NLMS,RLS graphs for noise 4.

Algorithm	Operating time
LMS	1.469076 seconds.
NLMS	1.821409 seconds.
RLS	3.420428 seconds.

5) Noise from Av. Los Heroes - Ciudad de Dios:
Date: Friday, November 18
Time: 8pm
Best filtering: NLMS

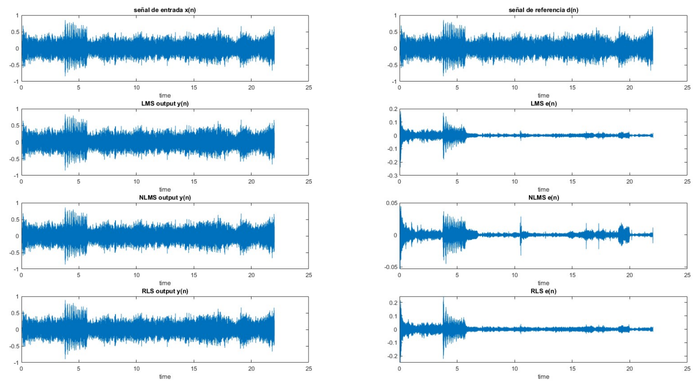

Fig. 12. LMS,NLMS,RLS graphs for noise 5.

Algorithm	Operating time
LMS	2.094554 seconds.
NLMS	1.882329 seconds.
RLS	3.668571 seconds.

B. ANC with simulink

As we can see in these 5 tests the algorithm that has the best performance in terms of sound filtering is the normalized least mean square. To corroborate this we will compare NLMS with LMS in Simulink which follows the same principle as the code presented in figure 8.

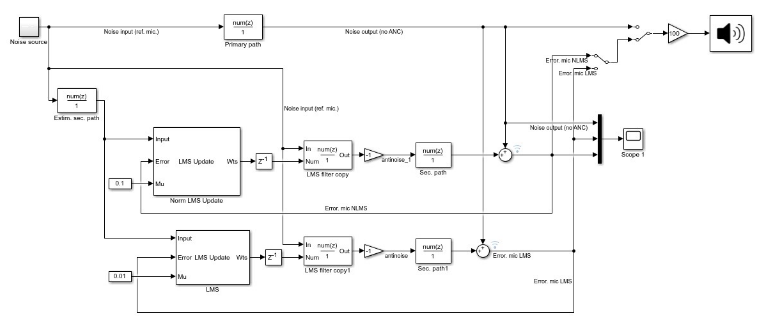

Fig. 13. ANC model in Simulink.

Note: To interpret the following graph it must be taken into account that NLMS is the red signal, LMS is the blue signal and the input signal is yellow.

V. Discussions

It was decided to work on a Matlab .m file to have the code ready to be converted to C++ and uploaded to the DSP that is planned to be purchased in the future for future field tests. For this next phase of implementation, we plan to purchase the AS3410-EQFP-500 equipment, which has a low investment cost of 0.82 dollars and we can give a similar added value to the MUTE project by adding IoT technology to our DSP. By obtaining more noise data from various strategic points of the capital we can generate a new sound map that will give

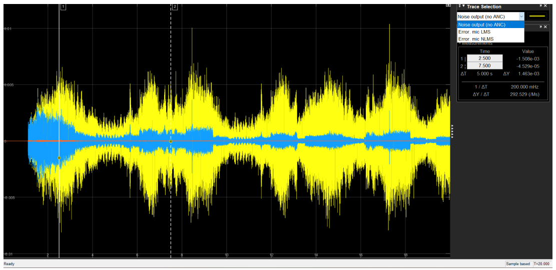

Fig. 14. LMS and NLMS in Simulink for the noise 1.

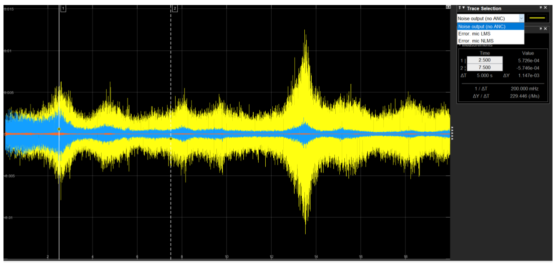

Fig. 15. LMS and NLMS in Simulink from Puente Santa Anita - Panamericana Norte.

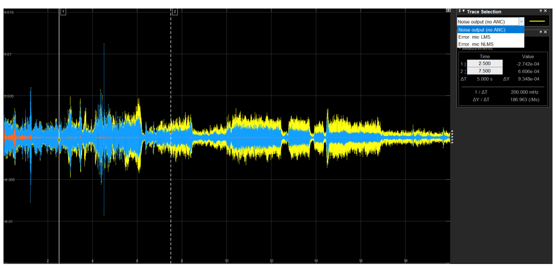

Fig. 16. LMS and NLMS in Simulink from Puente Atocongo - Panamericana Sur.

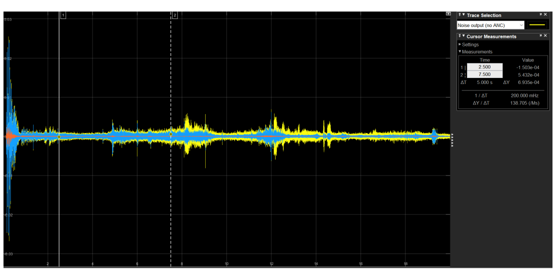

Fig. 17. LMS and NLMS in Simulink from Barranco - Chipoco Stadium.

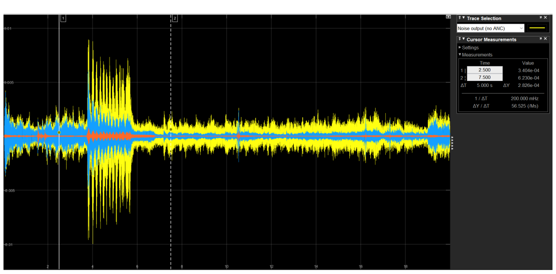

Fig. 18. LMS and NLMS in Simulink from Av. Los Heroes - Ciudad de Dios.

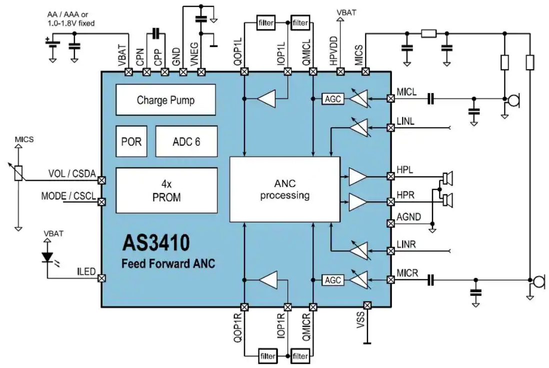

Fig. 19. AS3410-EQFP-500. [13]

us a comparison of how the perception of mobile and static noise has changed in various parts of Lima. If you want to be more accurate in terms of the data we are going to record, you should consider acquiring a sound level meter which has among its various functions to record the noise and store it in a memory or file with which you can work later. This noise cancellation device can be applicable for both mobile sources such as traffic noise and fixed sources, for example, a pump in an industrial plant. We tested the noise signal in a pump of a water recirculation plant.

A. *Pump noise - Lab 415*

Best filtering: NLMS

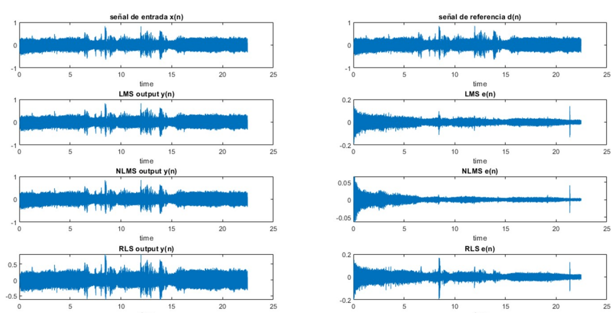

Fig. 20. LMS,NLMS,RLS graphs for noise pump.

Algorithm	Operating time
LMS	1.678208 seconds.
NLMS	1.567725 seconds.
RLS	3.640288 seconds.

VI. Conclusions

It was possible to prove with different samples that the LMS, NLMS, and RLMS algorithms can attenuate the noise of an audio signal, besides making the error decrease better and converge more efficiently and quickly, each one using its different operating variables. It is worth mentioning that the

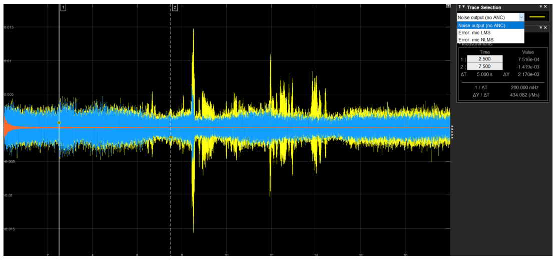

Fig. 21. LMS and NLMS in Simulink from a Pump.

disadvantage of the use of the LMS algorithm is its convergence speed, as well as its sensitivity to the ill-conditioning of the matrix input. On the other hand, the fast RLS algorithms are fast in converging but present instability problems and their implementation in C++ needs much more time investment if we want to use this filter for our physical prototype. Therefore, the NLMS algorithm is the most suitable to implement in the DSP we plan to acquire for future tests. The algorithms presented in this paper are the most used and the ones that best filter noise compared to other algorithms such as CMA, DMI, etc. By understanding how we can reduce noise, it is possible to continue the research and add IoT to anticipate the noise and make the functionality of our project more efficient. Our work is available for all to review at the link below: https://github.com/FrancesAn/ExternalNoiseCancelling

REFERENCES

[1] Bach. Tito Moya, Ewonny, ”Estimation of Noise Pollution due to environmental noise in zone 8 C of the district of Miraflores - Lima”, Thesis, Federico Villarreal National University, Lima, Peru, 2017.

[2] F. Firestone. The phase difference and amplitude ratio at the ears due to a source of pure tone. J. Acoust. Soc. Am., 2, 260-268, 1930.

[3] National Institute on Deafness and Other Communication Disorders (2022, March 16). How Do We Hear? [online]. Available: https://www.nidcd.nih.gov/health/how-do-we-hear

[4] Zufesha, A. Nusrat, A. (2019, marzo). Assessment of noise pollution in commercial and residential areas and its impact on the health/surrounding Environment: A Review. International Research Journal of Engineering and Technology (IRJET), 6(3), 2395-0072. https://www.irjet.net/archives/V6/i3/IRJET-V6I31306.pdf

[5] Gomez, P., Rivera, M., Galeano, B. Escobar, N. (2019). Technological innovation in the design process of an acoustic panel to the hospital environment. IOP Publishing, 1257(012012). https://doi.org/10.1088/1742-6596/1257/1/012012

[6] Ferraz, F. (2016). Assessment of Indoor and Outdoor Noise Pollution at a University Hospital Based on Acoustic Measurements and Noise Mapping. Open Journal of Acoustics, 06(04), 71-85. https://doi.org/10.4236/oja.2016.64006

[7] Studying of noise pollution for selected classrooms in Tikrit university-Iraq. (2012, 1 junio). Journal of Environmental Studies, 8(1), 1-7. https://doi.org/10.21608/jesj.2012.189126

[8] Sheng Shen, Nirupam Roy, Junfeng Guan, Haitham Hassanieh, and Romit Roy Choudhury. 2018. MUTE: Bringing IoT to Noise Cancellation. In ACM SIGCOMM. 282–296.

[9] KINSLER, L.E., FREY, A.R., COPPENS, A.B. Y SANDERS, J.V., (1992) “Fundamentos de Acústica” Editorial Limusa, S.A., Noriega Eds., Mexico.

[10] Kajikawa, Y. and Woon-Seng, G. and Kuo, S. M. ”Recent advances on active noise control: open issues and innovative applications”, Thesis, Cambridge University Press, Cambridgeshire, England, 2012.

[11] Yang, Z. and Roy Choudhury, R. Personalizing head related transfer functions for earables. pp. 137–150, 08 2021.

[12] 2018. Wireless noise cancelling headphones SHB8850NC/27 — Philips. Retrieved January 29, 2018

[13] AMS-OSRAM. (2013, Oct 03). AS3410/30 Activate Noise Cancellation ICs [Online]. Available: https://www.digikey.fr/en/product-highlight/a/ams/as341030-active-noise-cancellation-ics

[14] AMS-OSRAM. (2013, Oct 03). AS3410/30 Activate Noise Cancellation ICs [Online]. Available: http