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SINGLE-CHANNEL ACTIVE NOISE CANCELLER USING THE FIXED POINT DSP APLLIED TO THE CONTROL OF NOISE COMPRESSOR

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Abstract-: This paper reports the performance of a single-channel active noise compressor cancellation system. The active noise canceller (ANC) was developed based on a PC-hosted Analog Devices Inc., a low-cost 16-bit fixed-point Digital Signal Processor (DSP), mounted in a two-channel analog I/O board connected to the real world via standard consumer audio components. The algorithm implemented here was the single-channel filtered-X LMS. In the first part of this paper, the noise generated by the compressor will be characterized, followed by the selection of a coherent reference sensor for the ANC implementation. Finally, the noise attenuation achieved with the application of the ANC is presented.

Keywords--cancellation system, Digital Signal Processor (DSP), compressor

I. INTRODUCTION

Factory acoustic noise problems become more and more evident as increased numbers of industrial equipment such as engines, blowers, fans, transformers, and compressors are in use. These machines operate continuously in industrial settings, generating high levels of acoustic noise that can significantly impact workers' health and overall productivity. The exposure of workers to excessive noise can lead to various physical, psychological, and emotional problems. Physically, prolonged exposure to high noise levels has been associated with hearing loss, tinnitus, elevated blood pressure, and cardiovascular complications. Psychologically, workers in noisy environments may experience increased stress, anxiety, and fatigue, which can, in turn, lead to reduced cognitive function and concentration [1] e [2]. Emotionally, excessive noise exposure can contribute to irritability, decreased job satisfaction, and overall lower quality of life. These factors highlight the urgent need for effective noise control measures in industrial workplaces.

In the most part of industrialized countries, regulation and standard associations have selected permissible values of maximum noise that can exist in industrial environments, and there is a general tendency to decrease this level [3]. Regulatory agencies such as the Occupational Safety and Health Administration (OSHA) in the United States and the European Agency for Safety and Health at Work (EU-OSHA) have implemented strict guidelines to limit workers' exposure to high noise levels. These regulations often specify maximum allowable noise levels over an eight-hour work shift and recommend engineering controls or personal protective equipment (PPE) if noise levels exceed safe limits. Companies that fail to comply with these regulations may face legal consequences, including fines and operational restrictions. Despite these measures, achieving effective noise control remains a complex challenge, especially in environments where high-powered industrial machinery operates continuously.

Traditional methods of suppressing acoustic noise using passive sound absorbers, enclosures, barriers, and silencers generally do not work well at low frequencies. This is due to the fact that the acoustic wavelengths of low frequencies become larger than the thickness of a typical acoustic absorber. In other words, low-frequency noise has long wavelengths, which makes it difficult for conventional sound-absorbing materials to effectively attenuate the noise. Thereby, in order to have an effective passive method in the attenuation of these low frequencies, the absorber must be large, bulky, and heavy, becoming therefore costly and ineffective. Additionally, the implementation of passive noise control solutions in industrial environments can be impractical due to space constraints, high installation costs, and the need for frequent maintenance. As a result, industries often seek alternative methods that offer more efficient and adaptable noise reduction capabilities.

One of the most promising solutions for addressing industrial noise challenges is active noise cancellation (ANC). The active noise canceller (ANC) is generally formed by an electroacoustic or electromechanical system that cancels the primary (unwanted) noise based on destructive interference between the primary sound field and a secondary one controlled [4]. This secondary source is responsible for generating an anti-noise signal of equal amplitude and opposite phase to the primary noise. When these two signals interact, they effectively cancel each other out, leading to significant noise reduction in the target environment.

Typically, an ANC system is composed of two main parts: a physical system and an electronic controller. The physical system generally consists of microphones, which capture the incoming noise, and speakers, which generate the anti-noise signal. The electronic controller processes the noise signal and determines the appropriate anti-noise output. Since many characteristics of acoustic noise and industrial environments are time-varying, the control system must, therefore, be adaptive to continuously maximize noise attenuation [3]. This adaptability is crucial in industrial applications, where noise levels fluctuate due to variations in machine operation, changes in equipment positioning, and other environmental factors.

The development of digital signal processors (DSPs) in the 1980s enabled the low-cost implementation of powerful adaptive algorithms and their applications, including ANCs [5]. DSP technology revolutionized ANC systems by allowing for real-time signal processing, rapid filter coefficient adjustments, and improved noise cancellation performance. Modern ANC systems leverage high-performance DSPs to analyze noise patterns, optimize anti-noise generation, and minimize computational latency. The availability of affordable and efficient DSP technology has facilitated the widespread adoption of ANC systems in industrial applications, consumer electronics, and automotive noise control.

The ANCs are generally based on adaptive filters that adjust their coefficients to minimize an error signal and can be realized as a transversal filter, finite impulse response (FIR), recursive filter, infinite impulse response (IIR), lattice filter, and transform domain filter [5]. These adaptive filters are essential for dynamically adjusting the noise cancellation response in real-time, ensuring that the ANC system remains effective under changing noise conditions. The most common structure, and the one utilized in this work, is the transversal filter using the least mean square (LMS) algorithm [7]. The LMS algorithm is widely used due to its simplicity, computational efficiency, and effectiveness in continuously updating filter coefficients based on incoming noise signals. By iteratively minimizing the mean squared error between the desired and actual output signals, the LMS algorithm ensures that the ANC system maintains optimal performance in varying acoustic environments.

Given the increasing need for efficient noise control solutions in industrial settings, ongoing research continues to explore new advancements in ANC technology. Future developments may include the integration of machine learning techniques to enhance real-time noise prediction and adaptation, as well as the use of advanced sensors and actuators to improve ANC system performance. As industries strive to create safer and more comfortable working conditions, ANC technology is expected to play a vital role in reducing occupational noise exposure and mitigating its negative effects on workers' health and well-being.

II. NOISE CHARACTERIZATION

With the purpose of selecting the best ANC technique to be implemented, it is crucial to perform a detailed characterization of the noise we aim to cancel. This process involves analyzing the spectral properties of the noise source and identifying an appropriate reference sensor that can capture a highly correlated signal without interference from other environmental sounds. The selection of a good noise reference is fundamental to ensuring the effectiveness of the ANC system, as it directly influences the adaptive filter’s ability to generate an accurate anti-noise signal. A good noise reference sensor is defined as one positioned strategically near the noise source—in this case, a compressor—that can sense a signal strongly correlated with the environmental noise and remains unaffected by extraneous sound sources [8] [9].

To characterize the noise, a piezoelectric microphone was positioned two meters from the compressor to capture the primary noise signal. The signal detected by the microphone was amplified to enhance its strength, filtered through an anti-aliasing filter to remove unwanted high-frequency components, and then digitized for further analysis. The digitized signal was recorded in a computer file and subjected to in-depth signal processing to extract relevant noise characteristics.

The noise analysis involved the estimation of two primary functions: the power spectrum function (using the Welch method) and the correlation function [10]. The power spectrum estimation was conducted to determine whether the noise was predominantly narrowband or broadband. If the noise exhibited a narrowband profile, further analysis was performed to identify the specific frequency components that contained the highest energy levels. The correlation function, on the other hand, was used to evaluate how quickly the correlation between successive samples decreased with increasing lag, providing insights into the predictability of the noise signal [11].

The results obtained from the power spectrum estimation revealed that the majority of the energy was concentrated in four distinct frequencies: 60 Hz, 120 Hz, 180 Hz, and 590 Hz. Among these, the dominant frequency was 60 Hz, which registered a signal level of 38 dB, while the remaining tones exhibited lower intensities: 11 dB at 120 Hz, 10 dB at 180 Hz, and 3 dB at 590 Hz. This indicates that the noise generated by the compressor was primarily tonal, with energy concentrated at discrete frequencies rather than being distributed over a broad range.

The correlation function estimation further confirmed the presence of strong tonal components and a gradual decline in correlation as lag increased. At lag = 0 samples, the correlation coefficient was 1, indicating perfect self-correlation. As lag increased to 1000 samples, the correlation coefficient dropped to 0.7, demonstrating a slow decay and reinforcing the observation that the noise signal exhibited a highly structured periodic pattern. Given that the signal was sampled at 2000 Hz, this slow decay in correlation suggests that the noise retains significant predictability over time, which is advantageous for ANC applications.

A. Selection of the Reference Noise Sensor

To determine the most suitable reference sensor for ANC implementation, two different sensors were evaluated:

1. A piezoelectric microphone, positioned close to the compressor.

2. An accelerometer, physically attached to the compressor housing.

The performance of these two sensors was compared using four key analytical methods:

- Power spectrum estimation of each sensor's signal.

- Correlation function estimation to measure the predictability of each signal.

- Cross-correlation function estimation between each sensor's signal and the environmental noise signal captured by the microphone positioned in the industrial space.

- Coherence function estimation, which assesses the linear dependence between each sensor's signal and the environmental noise [12].

B. Analysis of Sensor Performance

The accelerometer, which was fixed directly onto the compressor, demonstrated superior performance as a reference sensor. Its correlation function closely matched the environmental noise signal, indicating that it captured the core characteristics of the compressor noise with minimal distortion. Furthermore, its power spectrum estimation revealed frequency components that were highly consistent with those found in the environmental noise, spanning the 0–600 Hz range.

The cross-correlation function between the accelerometer's signal and the environmental noise exhibited a high correlation level of 0.7 at a lag of 1000 samples, demonstrating a strong relationship between the two signals. Additionally, the coherence function estimation showed a high degree of linear dependence between the accelerometer's signal and the environmental noise in the key tonal frequencies. For example, at 60 Hz, the coherence value was 1, indicating near-perfect correlation. This result confirms that the accelerometer effectively captures a clean and accurate reference noise signal, making it an ideal choice for ANC implementation.

On the other hand, the microphone positioned near the compressor exhibited several limitations. The power spectrum density analysis revealed the presence of additional tones that did not appear in the environmental noise spectrum. This suggests that the microphone signal contained extraneous frequency components, likely introduced by sensor saturation or unwanted interference from nearby sources.

The correlation function and cross-correlation function analysis of the microphone signal showed a steeper decline in correlation levels as lag increased. This indicates that the microphone's signal was less stable and more affected by external disturbances compared to the accelerometer. Furthermore, the coherence function between the microphone's signal and the environmental noise was significantly lower than that of the accelerometer, particularly in the critical tonal frequencies. This suggests that the microphone's signal had a weaker linear dependence with the environmental noise, making it a less reliable reference for ANC purposes.

III. SYSTEM DEVELOPMENT

Active noise cancellation (ANC) systems are designed to mitigate unwanted noise by generating an anti-noise signal that destructively interferes with the original sound. These systems can be classified into two primary categories:

1. Feed-forward control, where a coherent reference noise input is available and is detected before it propagates toward the secondary source. This approach allows the ANC system to anticipate and counteract noise before it reaches the target environment.

2. Feedback control, where the active noise controller attempts to cancel the noise without the benefit of an upstream reference input. In this method, the system relies on an error microphone to continuously monitor residual noise and adjust the anti-noise signal accordingly [12].

Feed-forward ANC systems are often preferred in industrial applications, particularly when a stable and reliable reference noise signal can be obtained. These systems typically provide faster response times and higher noise reduction efficiency in environments with predictable noise patterns. In contrast, feedback ANC systems are more suitable for applications where the noise characteristics vary unpredictably, such as in enclosed spaces where reverberation effects are significant.

In this work, a coherent noise reference was obtained from an accelerometer fixed on the compressor. This accelerometer captured vibrations and noise emissions directly from the source, providing a highly correlated reference signal for the ANC system. Based on this feature, an ANC feed-forward structure was chosen as the most effective approach for noise cancellation in this application [13].

A. Feed-Forward ANC System Design

The Fig. 1, illustrates a single-channel feed-forward ANC system applied to active noise control in a duct. In this system, a microphone positioned near the noise source detects the reference input noise signal. This signal is then processed by the ANC system, which generates a control signal aimed at canceling the acoustic noise downstream in the duct.

The system consists of several key components:

- Reference microphone: Captures the noise signal before it propagates further.
- ANC processor: Utilizes adaptive filtering algorithms to compute the appropriate anti-noise signal.
- Loudspeaker: Converts the electric control signal into an acoustic control signal, generating the anti-noise wave.
- Error microphone: Monitors the residual noise level and provides feedback for performance evaluation.

By using this configuration, the ANC system can effectively attenuate low-frequency noise, which is traditionally difficult to suppress with passive noise control methods. The feed-forward structure ensures that the system responds proactively, reducing the overall noise level before it reaches the listener or workspace [13] e [15].

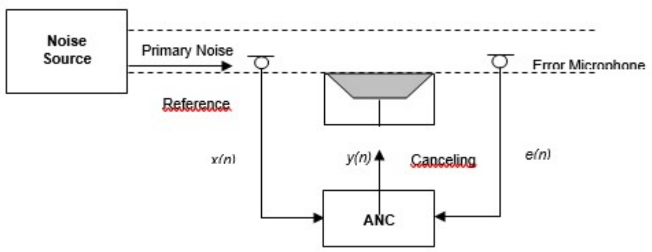

Fig. 1. Simgle-clannel feed-forwand ANS Sistem

B. Implementation of the Adaptive Algorithm

The Filtered-X LMS (FXLMS) algorithm was implemented in the ANC controller to optimize the performance of the noise cancellation process. This algorithm is an extension of the well-known Least Mean Square (LMS) adaptive algorithm, which is widely used in ANC applications due to its simplicity and efficiency. However, modifications were necessary to accommodate the complex transfer functions involved in an active noise control system [17].

The FXLMS algorithm with online secondary-path modeling was implemented using the additive random noise technique [15]. This enhancement was necessary due to the influence of various transfer functions associated with the adaptive filter, including:

1. Digital-to-analog (D/A) converter
2. Reconstruction filter
3. Power amplifier
4. Loudspeaker
5. Error microphone
6. Preamplifier
7. Anti-aliasing filter
8. Analog-to-digital (A/D) converter
9. Acoustic path between the loudspeaker and the error sensor (secondary path transfer function)

Each of these components introduces delays and distortions that must be accounted for in the adaptive filtering process. The FXLMS algorithm addresses this by incorporating an estimated model of the secondary path transfer function into its computations. This ensures that the input signal is properly filtered before updating the LMS weights, leading to more accurate noise cancellation [16].

The update equation for the FXLMS algorithm is represented as follows, equation (1).

$$\mathbf{w}(n+1) = \mathbf{w}(n) + \mu \mathbf{x}'(n) e(n) \qquad (1)$$

Where: $\mathbf{x}'(n) = [\hat{\mathbf{s}}(n)^{\mathbf{T}} \mathbf{x}(n) \;\; \hat{\mathbf{s}}^{\mathbf{T}}(n-1)\mathbf{x}(n-1)$

$\hat{\mathbf{s}}(n-L+1)^{\mathbf{T}} \mathbf{x}(n-L+1)]^{T}$;

$\hat{\mathbf{s}}(n) = [\hat{s}_0(n) \; \hat{s}_1(n) \ldots \; \hat{s}_{K-1}(n)]^{T}$;

$\mathbf{x}(n-i) = [x(n-i) \; x(n-i+1) \; \ldots \; x(n-i-K+1)]^{T}$.

and:
x'(n) is the response of the convolution between the input x(n) and the estimated impulse response of the secondary path $\hat{s}(n)$;

K is the number of weights utilized in this secondary path transfer function estimation.

The algorithm implemented in the low cost 16 bits fixed point DSP, of the Analog Devices Inc., was a modification of this one known as leaky FXLMS, and that is capable to deal with finite precision problems of digital implementation [18] [19].

IV. RESULTS

A.. Hardware Implementation

The ANC system was implemented using a low-cost 16-bit fixed-point DSP from Analog Devices Inc.. Fixed-point DSPs are often preferred over floating-point alternatives in cost-sensitive industrial applications due to their lower power consumption and affordability. However, one of the challenges of using fixed-point DSPs is the finite precision of numerical computations, which can introduce quantization errors and limit the accuracy of the adaptive filter, fig 2.

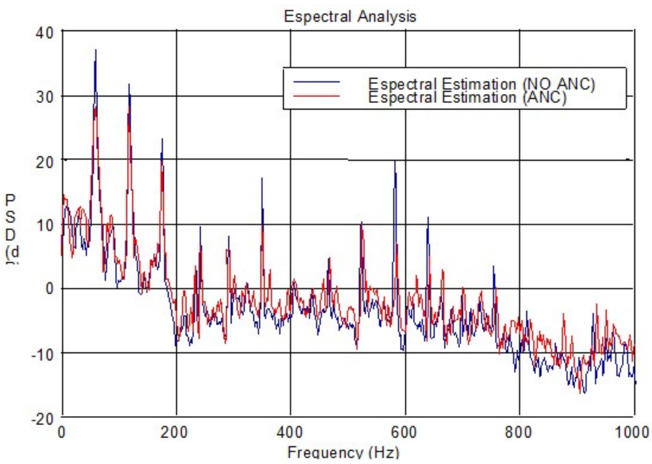

Fig. 2 - ANC performance

This research used solid state accelerometer, capacitance and signal conditioning, the ADLX05, Analog Devices Inc., with a high reliability, and low cost [11].

V.CONCLUSIONS

In this study, a real-time Active Noise Cancellation (ANC) system was successfully developed using a low-cost 16-bit fixed-point DSP and a capacitive accelerometer with integrated signal conditioning. The system utilized the Filtered-X LMS (FXLMS) algorithm, which was tailored to handle the complexities of the acoustic path, including the various components that introduce delays and distortions. This adaptation, along with the implementation of a leaky FXLMS modification, ensured that the fixed-point DSP could manage finite precision issues, such as quantization errors, without compromising the performance of the adaptive filter.

The experimental results validated the effectiveness of the system in reducing low-frequency acoustic noise, particularly from industrial piston compressors. The ANC system demonstrated substantial noise attenuation, achieving performance levels that are crucial for environments where noise exceeds 85 dB(A)—the upper limit for occupational exposure. The results highlight the potential of using fixed-point DSP-based ANC solutions in industrial settings, offering both a cost-effective and reliable approach to mitigating noise pollution.

This research reinforces the viability of implementing fixed-point DSPs in real-time ANC systems for industrial applications. Moving forward, improvements could include optimizing multi-channel ANC systems, refining the adaptive filtering algorithms, and integrating machine learning techniques to enhance noise cancellation in more dynamic acoustic environments. These developments would further bolster the performance and scalability of ANC solutions in diverse industrial noise control applications.

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