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SPEECH ENHANCEMENT USING ADAPTIVE ALGORITHMS FOR NOISY SIGNALS

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Abstract: speech enhancement is one of the techniques used to improve the quality of a speech signal. Noise is an important factor that effect any kind of speech signal. To remove the noise present in the speech signal different adaptive algorithms are used. In this paper, the performance is estimated by using adaptive algorithms like LMS, NLMS and AP algorithms respectively. The adaptive filtering is observed for noises at different frequencies such as 0db, 5db, 10d and 15db. These algorithms are used to analyze the speech signal quality. The effectiveness of these algorithms is simulated by using MATLAB 2020b software tool.

Keywords: Speech Enhancement, Noise, Adaptive Algorithm, LMS, NLMS, APA.

1.INTRODUCTION

Speech, as a primary mode of human communication, is susceptible to degradation when confronted with environmental noise, interference, or suboptimal recording conditions. In scenarios ranging from telecommunications and audio processing to voice recognition systems, the presence of noise can impede the intelligibility and quality of speech signals [1-3]. Speech enhancement emerges as a pivotal signal processing technique designed to alleviate this challenge, striving to extract and restore the clarity of speech while mitigating the impact of unwanted noise. This introductory discourse unveils the essence of speech enhancement, its significance, methodologies, and the broader implications it holds in diverse domains.

In real-world settings, speech signals often coexist with various forms of noise—ambient sounds, electronic interference, background chatter—disrupting the seamless exchange of information. Such noise not only degrades the intelligibility of speech but also hampers the accurate interpretation of spoken content by both humans and machines. As a result, the need for techniques capable of enhancing speech in the presence of noise becomes paramount.

At its core, speech enhancement aims to restore the clarity and fidelity of speech signals that have been compromised by noise. Unlike noise cancellation, which seeks to remove noise entirely, speech enhancement focuses on preserving the essential features of the speech signal while minimizing the perceptual impact of noise. The goal is to optimize the signal-to-noise ratio, allowing the intended speech content to emerge prominently.

2.ADAPTIVE FILTERING

Adaptive filtering is a powerful signal processing technique used to enhance the quality of signals by reducing unwanted noise, interference, or distortions [4-5].

Unlike traditional fixed filters, adaptive filters have the unique ability to self-adjust their filter coefficients based on the input signal and a desired reference signal [6]. This dynamic adjustment allows them to continually optimize their performance, making them well-suited for applications in which signal characteristics change over time. Adaptive filtering is widely employed in various domains, including telecommunications, audio processing, image processing, control systems, and biomedical signal analysis [7-9].

At the core of adaptive filtering is the concept of an error signal, which quantifies the difference between the desired (reference) signal and the actual filter output. Adaptive algorithms [10-11] use this error signal to iteratively update the filter's coefficients, aiming to minimize the error and bring the filter output as close as possible to the desired signal. The choice of the adaptive algorithm, step size (adaptation rate), and filter order are key factors that influence the filter's performance in terms of convergence speed and overall effectiveness.

One prominent example of adaptive filtering is noise cancellation in audio applications. In such cases, an adaptive filter analyzes an input signal containing noise and the desired audio signal. By continuously adjusting its coefficients, the filter can effectively attenuate or remove the noise component, resulting in a cleaner audio signal. This technology is commonly used in headphones, hearing aids, and telecommunications to improve audio quality and intelligibility, particularly in noisy environments. Adaptive filtering's ability to adapt to real-time changes in signal conditions makes it an invaluable tool for enhancing the performance of various systems and devices.

2.1 Adaptive Noise Canceller

An adaptive noise canceller (ANC) [12] is a sophisticated signal processing system designed to mitigate unwanted noise from an audio signal in real-time. It operates on the principle of adaptive filtering, where it continuously analyzes the noisy input signal and generates an anti-noise signal that has the opposite phase of the unwanted noise. When the anti-noise signal is combined with the original input, it cancels out the noise, resulting in a cleaner and more intelligible audio signal. ANC technology has gained significant popularity in various applications where noise reduction is critical, such as headphones, hearing aids, telecommunications, and automotive audio systems.

The core component of an ANC system is the adaptive filter, which continuously adjusts its coefficients based on the error signal—the difference between the desired noise-free signal and the current filter output. This adaptation is performed iteratively, allowing the filter to track changes in the noise environment and improve noise cancellation performance over time. ANC systems often employ one or more microphones to capture both the noisy input signal and a reference signal that primarily contains the noise. The reference signal is used as a template for the anti-noise generation process.

One notable application of ANC technology is in noise-cancelling headphones as shown in fig and earphones. These devices use built-in microphones to capture ambient sounds and then generate

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anti-noise signals to counteract the noise, creating a quiet listening environment. ANC is particularly valuable in scenarios where users want to enjoy audio content without being disturbed by external noise, such as during travel or in noisy public spaces. As technology continues to advance, ANC systems are becoming increasingly sophisticated, delivering remarkable noise reduction capabilities and enhancing overall audio experiences across a wide range of consumer and industrial applications.

Fig 1: Adaptive Noise Cancelling Headphones

3. ADAPTIVE ALGORITHM

Adaptive algorithms [13] are a class of computational techniques employed in various fields, including signal processing, machine learning, and control systems. What sets them apart is their ability to self-adjust or adapt their parameters based on changing input data or environmental conditions. These algorithms use feedback mechanisms to continuously update their internal parameters, striving to optimize their performance and minimize errors.

3.1 LEAST MEAN SQUARE ALGORITHM

The Least Mean Squares (LMS) algorithm [14] is a widely used adaptive filtering technique employed to iteratively adjust filter coefficients to minimize the mean squared error between the filter's output and a desired signal. It operates on the principle of gradient descent, where it calculates the gradient of the error with respect to the filter coefficients and updates these coefficients in small steps to reduce the error. LMS is particularly popular in applications like noise cancellation and equalization, where it efficiently adapts to changing input conditions and provides real-time improvements in signal quality.

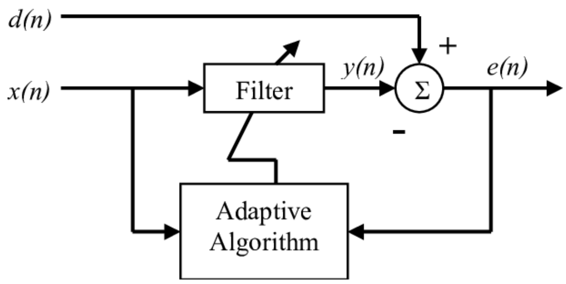

Fig 2: Block Diagram of ANC

The Least Mean Squares (LMS) algorithm is a widely used adaptive filtering algorithm used to minimize the mean squared error between the desired signal and the filter output by iteratively updating the filter coefficients. The core equation for the LMS algorithm can be expressed as follows:

1. Weight Update Equation:

The weight update equation for the LMS algorithm is given as:

$$w(n+1) = w(n) + \mu\, e(n)\, x(n) \quad (1)$$

- w(n+1) is the updated filter coefficient vector at iteration (n+1).

- $w(n)$ is the filter coefficient vector at iteration n.

- μ is the step size or adaptation rate, a small positive constant.

- $e(n)$ is the error signal at iteration n, which is the difference between the desired signal and the filter output at that iteration.

- $x(n)$ is the input signal vector at iteration n.

2. Filter Output:

The filter output at each iteration n is calculated as:

$$y(n) = w^T(n)\, x(n) \qquad (2)$$

- $y(n)$ is the filter output at iteration n.

- $w^T(n)$ is the transpose of the filter coefficient vector at iteration n.

- $x(n)$ is the input signal vector at iteration n.

These equations represent the fundamental operations of the LMS algorithm, where the filter coefficients are iteratively updated based on the error signal and input signal to minimize the mean squared error, thereby adapting to changing input conditions and optimizing the filter's performance.

3.2 NORMALIZED LEAST MEAN SQUARE ALGORITHM

The Normalized Least Mean Squares (NLMS) [15-16] algorithm is an adaptive filtering technique that builds upon the basic LMS algorithm by normalizing the step size during coefficient updates. NLMS is particularly useful in scenarios where the input signal's power level varies significantly. Instead of using a fixed step size, NLMS adapts the step size based on the current input signal, making it a more robust and versatile algorithm.

In NLMS, the step size μ is divided by a normalization factor that is proportional to the power of the input signal. This normalization process ensures that larger input signals receive smaller step sizes, preventing excessive updates and instability. As a result, NLMS converges efficiently in varying signal conditions and is less sensitive to the choice of the step size. This makes it a popular choice in applications such as adaptive noise cancellation, channel equalization, and adaptive beamforming, where adaptive filtering is employed to enhance signal quality while adapting to changing environments.

Weight Update Equation:

$$w(n+1) = w(n)+\mu\, e(n)x(n) \qquad (3)$$

where μ is calculated as

$$\mu = \frac{\alpha}{c+x^T(n)x(n)} \qquad (4)$$

3.3 AFFFINE PROJECTION ALGORITHM

The Affine Projection (AP) [17] algorithm is an advanced adaptive filtering technique used to minimize the mean squared error between the desired signal and the filter output. AP builds upon the principles of the LMS and Normalized LMS algorithms but introduces a more extensive filter memory and additional parameters, making it suitable for applications with challenging adaptive filtering requirements.

What sets the AP algorithm apart is its ability to adapt to both linear and nonlinear signal characteristics. It employs a tapped

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delay line to capture delayed versions of the input signal, providing a more comprehensive view of the signal's history. By using a combination of past input samples and their corresponding filter coefficients, the AP algorithm can handle applications with complex signal dynamics and non-stationary environments, making it valuable in scenarios like channel equalization, echo cancellation, and acoustic modeling. AP's adaptability to a wide range of signal conditions makes it a versatile tool in various domains, including telecommunications, audio processing, and control systems.

The step size value for AP algorithm is given by

$$A(n) = [x(n)]^T \qquad (5)$$

$$\mu(n) = [A(n)[A(n)]^T]^{-1} \qquad (6)$$

by substituting in Equation 1 the weight update equation is obtained

$$w(n+1) = w(n) + [A(n)[A(n)]^T]^{-1} \; [e(n)x(n)] \qquad (7)$$

4 SIMULATION RESULTS

The simulation results of ANC system for different adaptive filter algorithms at babble, airport, street and car noise are added to the original clean sp02 signal. The experiments conducted aimed to assess the performance of a speech-denoising algorithm when applied to noisy data. To create a comprehensive evaluation, a range of Signal-to-Noise Ratio (SNR) scenarios, including 0dB, 5dB, 10dB, and 15dB, was considered [18]. These SNR levels were achieved by combining clean speech samples with noise signals sourced from well-known databases like NOIZEUS and TIMIT, encompassing diverse real-world noise environments such as car noise, airport noise, street noise, and babble noise.

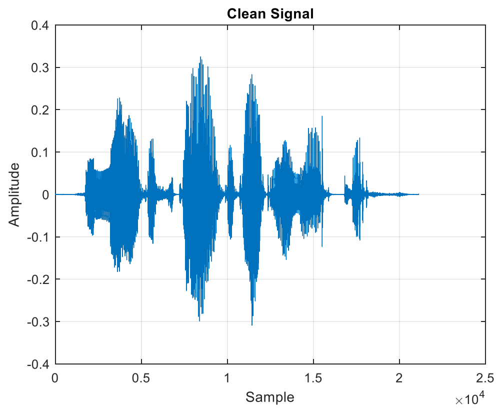

Fig 3: Clean signal

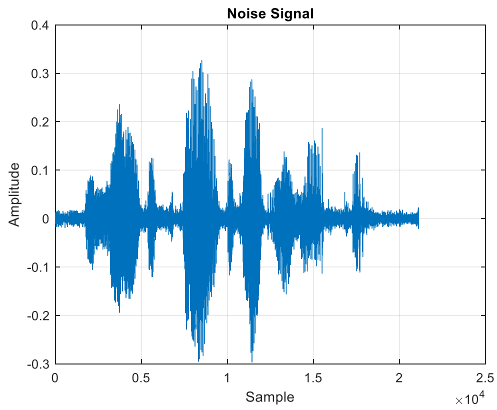

Fig 4: Noise signal

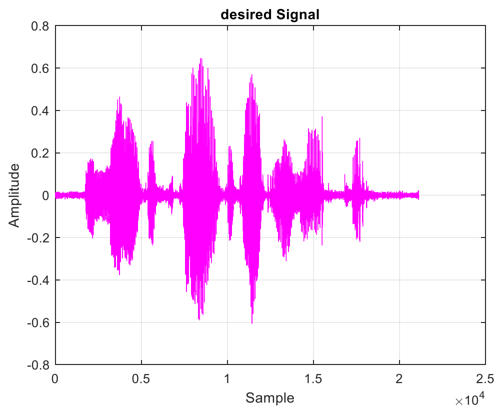

Fig 5: Desired Signal

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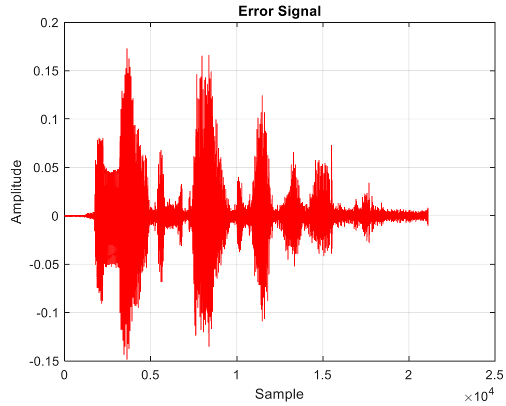

Fig 6: Error Signal of LMS Algorithm

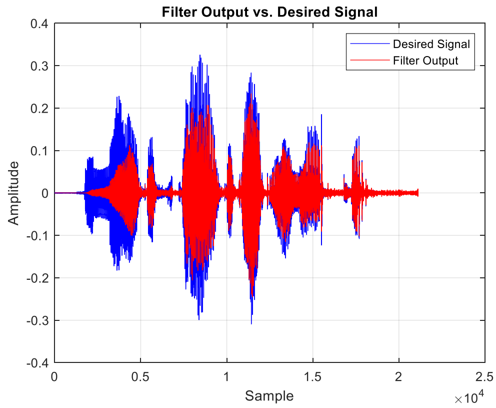

Fig 7: Filter Output

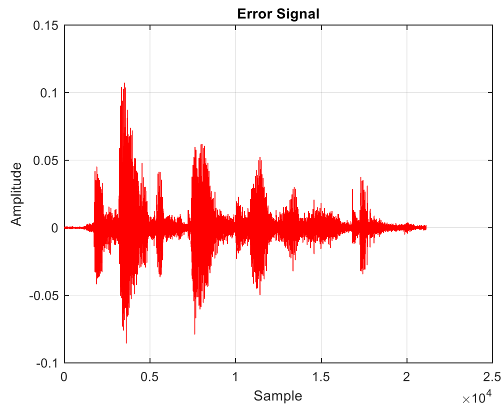

Fig 8: Error Signal of NLMS Algorithm

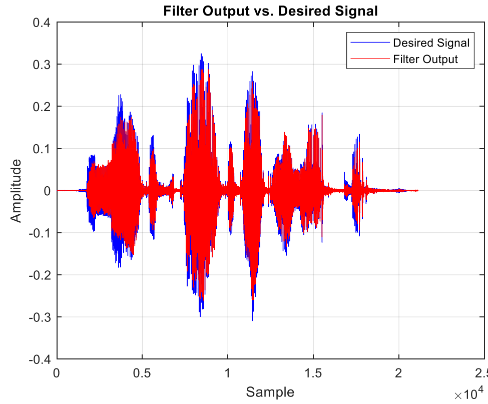

Fig 9: Filter Output

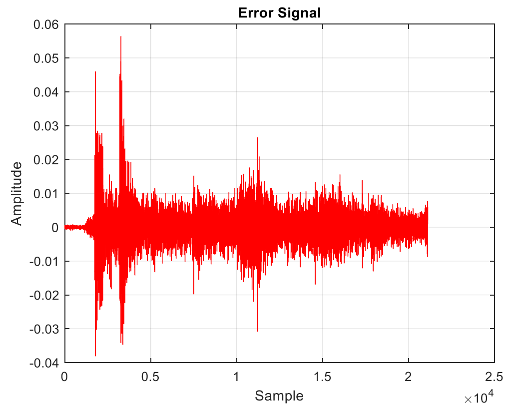

Fig 10: Error Signal of AP Algorithm

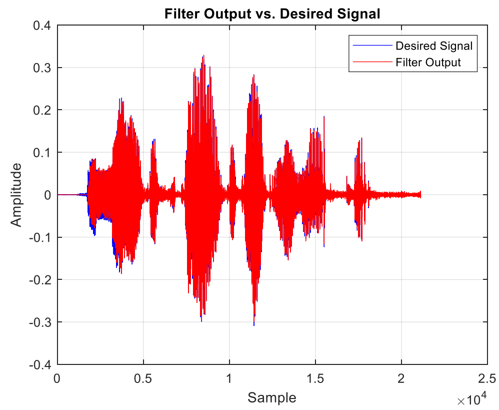

Fig 11: Filter Output

Then the signals performance parameters [19] are measured such as SNR, MSE and RMSE values as shown in the below tables

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Noise signals		Car Noise (In dB)	Airport Noise (In dB)	Street Noise (In dB)	Babble Noise (In dB)
LMS	0dB	0.0047721	0.14635	-0.69469	0.29976
	5dB	3.3247	3.2617	3.3726	3.5935
	10dB	5.1369	5.1447	5.2013	5.1261
	15dB	5.8417	5.8481	5.8049	5.8953
NLMS	0dB	0.11478	0.15577	-0.28876	0.32287
	5dB	4.704	4.7493	4.8521	4.9267
	10dB	8.5956	8.6585	8.8111	8.8067
	15dB	11.843	12.0354	11.612	12.0825
APA	0dB	0.5581	0.57389	0.34798	0.74057
	5dB	6.2307	6.3784	6.1711	6.3767
	10dB	12.1396	11.9796	11.9318	12.1107
	15dB	17.2658	17.0913	17.1999	17.2842

Table 1: SNR values

Noise signals		Car Noise	Airport Noise	Street Noise	Babble Noise
LMS	0dB	0.00063491	0.00065029	0.00055963	0.00067293
	5dB	0.00051568	0.00051559	0.00051284	0.00053413
	10dB	0.00046917	0.00047106	0.00046961	0.0004708
	15dB	0.00045203	0.00045243	0.00045015	0.00045429
NLMS	0dB	0.00034544	0.00035906	0.00031557	0.00038856
	5dB	0.00021116	0.00021397	0.00022082	0.00022389
	10dB	0.00014358	0.00014033	0.00012457	0.00014159
	15dB	9.2995e-05	8.8279e-05	9.4735e-05	9.4231e-05
APA	0dB	0.00019306	0.00019504	0.00016696	0.00022089
	5dB	8.9167e-05	9.5772e-05	8.5026e-05	9.5673e-05
	10dB	4.2359e-05	4.0891e-05	3.8501e-05	4.2647e-05
	15dB	1.9666e-05	1.9331e-05	1.9375e-05	2.0004e-05

Table 2: MSE values

Noise signals		Car Noise	Airport Noise	Street Noise	Babble Noise
LMS	0dB	0.025197	0.025501	0.023656	0.025941
	5dB	0.022709	0.022707	0.022646	0.023111
	10dB	0.02166	0.021704	0.021671	0.021698
	15dB	0.021261	0.02127	0.021217	0.021314
NLMS	0dB	0.018586	0.018949	0.017764	0.019712
	5dB	0.014531	0.014628	0.01486	0.014963
	10dB	0.011983	0.011846	0.011161	0.011899
	15dB	0.0096434	0.0093957	0.0097332	0.0097073
APA	0dB	0.013895	0.013966	0.012921	0.014863
	5dB	0.0094428	0.0097863	0.009221	0.0097812
	10dB	0.0065084	0.0063946	0.0062049	0.0065304
	15dB	0.0044346	0.0043968	0.0044017	0.0044726

Table 3: RMSE values

From the tables we can conclude that the Highest SNR can be seen at babble noise for Affine projection algorithm. As we know that the we can get the original with lowest noise at babble noise.

CONCLUSION

In summary, the analysis of the data presented in the table strongly supports the conclusion that the APA outperforms both the LMS and NLMS algorithms across key performance metrics, including Signal-to-Noise Ratio (SNR), Mean Squared Error (MSE), and Root Mean Squared Error (RMSE). The consistently superior values obtained by APA in these metrics demonstrate its remarkable proficiency in signal estimation and noise reduction.

APA's notably higher SNR values indicate a more substantial Signal-to-Noise Ratio, implying its capability to provide clearer and more discernible signals. This quality is particularly valuable in applications where signal fidelity is paramount. Furthermore, the lower MSE and RMSE values achieved by APA signify its heightened accuracy in predicting and minimizing the errors between estimated and actual values, underscoring its reliability in signal processing tasks.

In conclusion, based on the available data and the comprehensive evaluation of performance metrics, it can be confidently asserted that the Affine Projection Algorithm stands as a superior choice when prioritizing signal quality, estimation precision, and noise mitigation in adaptive filtering applications, surpassing the LMS and NLMS Algorithms in these aspects.

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