Analysis of Audio Filtering Algorithms for Noise Cancellation

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Abstract

Real-time signal noise reduction poses a significant challenge in communication systems. To combat intrusive background noise during portable device conversations, an active noise cancellation (ANC) is employed. ANC generates anti-noise signals using techniques such as Least Mean Square (LMS) and Normalized Least Mean Square (NLMS) algorithms, Recursive Least Squares (RLS) and modified Recursive Least Squares (M-RLS) for effective reduction of unwanted sound and enhancing the output signal quality. The study includes various real-time noise signals to create anti-noise responses, successfully negating corresponding noise in input signals, which include sine waves, live voices, and music. Through a dedicated micro-controller device and electronic network, an "Anti-noise" wave is introduced to achieve active noise control (ANC). This research identifies ANC module specifications and effectively transmits them through the ANC library, demonstrating the efficiency of the proposed noise reduction technique.

Keywords: ANC, LMS, NLMS, RLS, M-RLS

1. Introduction

The use of active noise cancellation (ANC) [1-6] technologies has completely changed how humans interact with sound. ANC aggressively cancels out outside noise by utilizing speakers and microphones, resulting in a much more enjoyable listening experience. Even

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though ANC has a wide range of uses, it has proven to be very useful in the automotive industry. The use of ANC in automobiles has increased recently as automakers explore methods to enhance the driving experience for their customers. Using ANC in cars has several advantages, including reduced driver fatigue, better audio quality, and a smooth ride. One of ANC's key benefits for automobiles is that it helps in reducing driver fatigue. Driving long distances can be tiresome, and the noise from the outside world can make it even more so. . ANC can aid in reducing this noise, making it simpler for drivers to focus on the road in front of them. For lengthy trips, where driver weariness may pose a safety risk, this can be especially crucial. The ability to enhance audio quality is another advantage of ANC in automobiles. Listening to music or phone calls can frequently be challenging, especially in noisy environments. A better, more pleasurable listening experience can be achieved by ANC's assistance in reducing this outside noise. This is important for vehicles because external noise might originate from the engine, tyres, wind, and various other sources . Last but not least, ANC also helps to have a smooth journey. The comfort of the ride can be greatly hindered by noise from roads, which can be a significant source of outside noise in vehicles. Using ANC to help block out this noise will result in a much peaceful ride. This can significantly improve the enjoyment of driving, especially for individuals who spend a lot of time in their cars. While utilizing ANC in cars has many advantages, there are some obstacles that must be solved as well. The fact that ANC uses a lot of computing power is one of the greatest problems. This might be challenging in an automobile, where space is limited and power consumption needs to be carefully managed. An additional problem is that ANC may interfere with other car sensors and systems, like hands-free call microphones. Despite these challenges, ANC's importance in the automotive industry is growing. ANC is probably going to appear in more and more new automobiles as automakers try to enhance the driving experience for their customers. In the realm of automobiles, active noise cancellation (ANC) has a lot to offer, whether it's lowering driver fatigue, enhancing audio quality, or offering a comfortable and smooth ride.

The problem lies in real-time signal noise reduction, a pervasive issue in communication systems. Unwanted background noise disrupts conversations on portable devices, negatively affecting voice quality and user experience. The need for effective noise reduction techniques, such as ANC, is evident. As external noise can exacerbate driver fatigue in the automotive sector, ANC is critical to ensuring safer and more pleasant driving experiences.

The primary objective of this research is to develop and evaluate an active noise cancellation system, utilizing algorithms such as the Least Mean Square (LMS) and Normalized Least Mean Square (NLMS) for real-time signal noise reduction. By doing so, the research aims to provide a noise-free signal output during conversations on portable devices, significantly enhancing the listening experience.

Furthermore, this study aims to address the specific challenges associated with implementing ANC in the automotive sector. The research aims is to create an efficient ANC system that reduces driver fatigue, enhances audio quality, and ensures a peaceful ride, thereby improving the overall driving experience. By achieving these objectives, the research seeks to contribute to the broader adoption of ANC in various applications, offering a solution to the pervasive issue of real-time signal noise reduction.

2. ANC Algorithm

A noise cancelling device is used by Active Noise Cancellation (ANC) [9] to lessen undesirable background noise. A speaker inside the earphone cancels out loud noises from the outside by neutralizing soundwaves, an ANC chipset inverts soundwaves, and microphones both inside and outside the earphone and listens to sounds. Similar to adding -2 inside and subtracting +2 outside to get zero. Active noise cancellation can be processed through the ANC chipset using either:

- Feed Forward ANC: The microphone is placed outside the earphone in the feed forward ANC.
- Feedback ANC: The microphone is placed inside the earphone in the feedback ANC.
- Hybrid ANC: The feedback and the feed forward are combined to create a hybrid ANC.

In proposed study has used LMS, NLMS, RLS [11,12] and M-RLS type of \ Active Noise Cancellation methods which are discussed below.

A. LMS Algorithm [7,8]

In machine learning and signal processing, the LMS (Least Mean Squares) technique is a popular adaptive filter [10, 13-16] algorithm. By applying an adaptive filter to decrease the discrepancy across the expected output and the system's actual output, it is possible to estimate unknown systems or filters. based on the error signal, the LMS algorithm iteratively changes the filter coefficients until the error is minimized. It is a stochastic gradient descent approach.

The formulated representation of the LMS algorithm is given by

$$w(n+1) = w(n) + \mu\, e(n)\, x(n) \qquad \textbf{(1)}$$

Where e(n) is the error signal, which is the difference between the planned output d(n) and the actual output y(n), and x(n) is the input vector at iteration n. Where w(n) is the filter coefficient vector at iteration n.

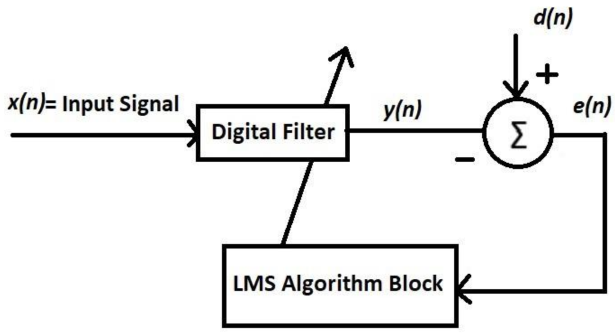

Figure 1. LMS Algorithm Architecture [7]

The LMS technique is simply implementable in both hardware and software and is computationally efficient. The selection of the starting filter coefficients and the step size, however, might have an impact on the performance.

The LMS (Least Mean Squares) method is an adaptive filter that has a wide variety of uses in machine learning and signal processing.

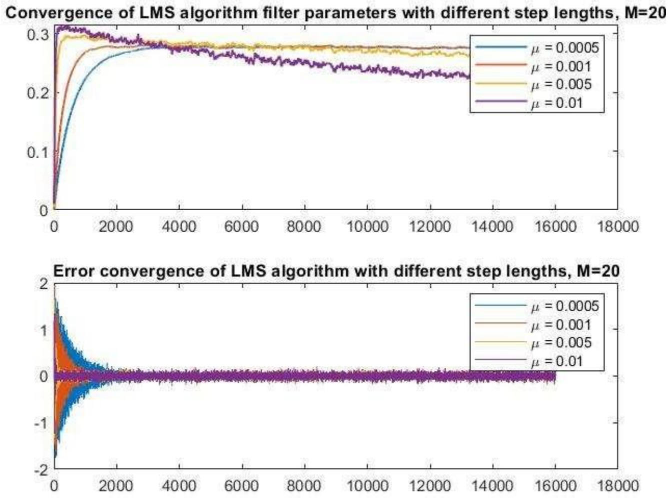

Figure 2. Convergence of LMS Algorithm [7]

B. NLMS Algorithm

In signal processing, the Normalised Least Mean Squares (NLMS) technique is a popular adaptive filter algorithm. It is a variation of the LMS (Least Mean Squares) method that boosts performance by using a normalised version of the filter coefficients. The mean squared error (MSE) between a desired signal and a linear filter's output is reduced by using the NLMS algorithm to modify the coefficients of the filter. It is often employed in tasks like channel estimation, equalization, and noise cancellation.

A popular adaptive filter technique for noise reduction and signal amplification applications is the Normalised Least Mean Squares (NLMS) algorithm. It is a particular kind of gradient descent method that iteratively changes the filter coefficients.

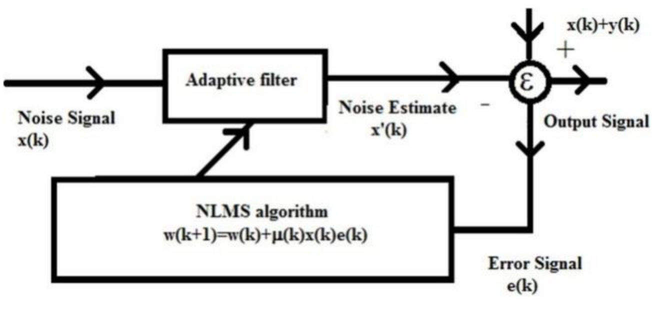

Figure 3. NLMS Architecture [8]

The working of NLMS algorithm is as follows:

1. Set the filter coefficients to a starting point, usually zero.
2. To create the output signal, take a sample of the input signal and run it through the filter.
3. Determine the difference in error between the intended signal and the output signal.
4. Use the following formula to update the filter coefficients:

$$w(n+1) = w(n) + \mu * e(n) * x(n) / (||x(n)||^2 + \delta) \quad \textbf{(2)}$$

Where $||x(n)||^2$ is the energy of the input signal at time n, w(n) is the vector of filter coefficients at time n, is the step size, e(n) is the error at time n, x(n) is the input signal at time n, and is a tiny constant to prevent division by zero.

- For every sample of the input signal, repeat steps 2-4.

In order to preserve stability as the filter converges to the ideal coefficients, the NLMS algorithm modifies the step size. Although convergence occurs more slowly with smaller step sizes, the filter is more stable. The NLMS method is a computationally effective and comparatively simple algorithm. However, it could occasionally experience a sluggish convergence and perform poorly. An adaptive filter technique with several uses in machine learning and signal processing is the NLMS (Normalised Least Mean Squares) algorithm. To guarantee the correct operation of the NLMS algorithm, however, a few requirements must be met:

- Stability of the Input Signal: The NLMS method assumes that the input signal is steady over time. In other words, the input signal's statistical characteristics need to hold true across time. The NLMS method may not converge or may converge slowly in non-stationary input signals.
- Small Step Size: The NLMS algorithm's step size parameter controls how frequently the filter coefficients are changed. The method may become unstable and the filter coefficients may fluctuate or diverge if the step size is set to a value that is too big. Therefore, it is better to have a short step size to provide consistent convergence.

- Normalization of the Input Signal: Prior to applying the update rule, the NLMS algorithm normalizes the input signal. By doing this, the algorithm is made to be resilient to changes in the input signal's scale. Inaccurate estimations might be generated by the algorithm if the input signal is not normalized.

- Gaussian Distribution of the Noise: The NLMS method takes the white, Gaussian-distributed noise in the system as a given. The algorithm may not converge or may converge to poor solutions if the noise is non-Gaussian.

- Linear System: The NLMS algorithm is made to calculate a linear system's coefficients. Poor performance might result from the algorithm's inability to effectively simulate a nonlinear system. The NLMS algorithm is a well-liked option in practical applications since it is less susceptible to the selection of step size than the LMS algorithm

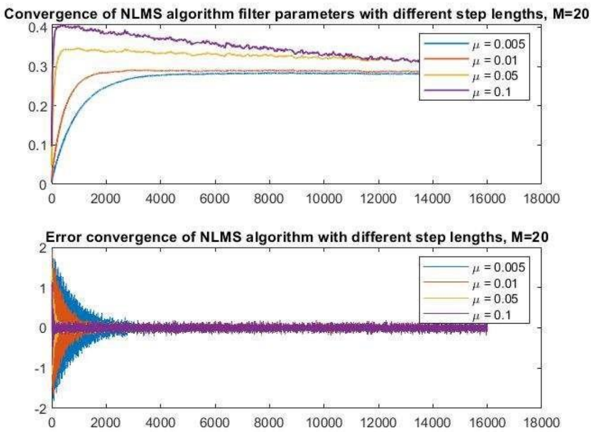

Figure 4. Convergence of NLMS Algorithm [17]

C. RLS Algorithm

Recursive Least Squares, or RLS, is an adaptive filter technique used in machine learning and signal processing. By recursively updating the estimates in light of fresh information, it is used to estimate the parameters of a linear system. In more precise words, the RLS method modifies the parameter estimates by, subject to a set of restrictions, minimizing a weighted sum of the squared errors between the expected and actual outputs. The algorithm updates the estimates and the related weight matrix, which symbolizes the uncertainty in the estimations, using a recursive formula.

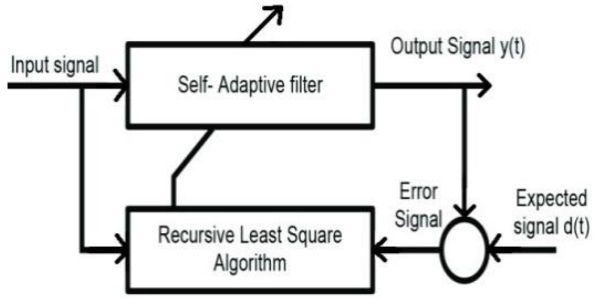

Figure 5. RLS Architecture [8]

The RLS method offers a better trade-off between convergence speed and steady-state performance than the LMS algorithm, although being computationally more costly. It is frequently employed in applications like adaptive equalization, channel estimation, and noise cancellation where quick convergence and strong tracking performance are needed. The RLS algorithm is ideal for online applications because it can adapt to changing system parameters over time and is available in a streaming format. The regularization value that is used as well as the numerical stability of the recursive update equations, nevertheless, can have an impact on the performance of the system.

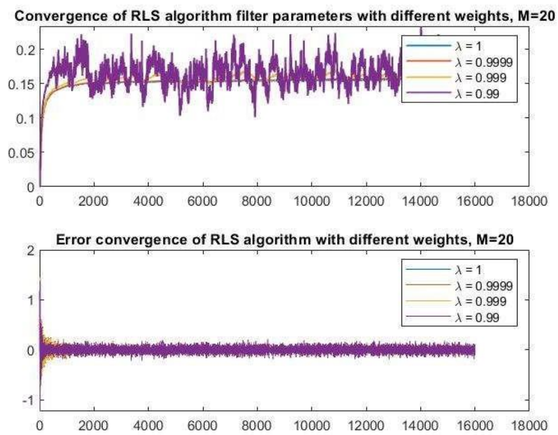

Figure 6. Convergence of RLS Algorithm

The Recursive Least Squares (RLS) algorithm for audio noise filtering is an adaptive filtering technique that updates filter coefficients and the inverse of the covariance matrix in real-time. For each incoming audio sample x(k) and the corresponding desired signal d(k), it computes the filter output y(k) and calculates the error e(k). The filter coefficients W(k) and the inverse covariance matrix P(k) are updated as follows:

$$y(k) = W(k)^T * x(k) \quad \textbf{(3)}$$

$$e(k) = d(k) - y(k) \quad \textbf{(4)}$$

$$K(k) = P(k) * x(k) / (1 + x(k)^T * P(k) * x(k)) \quad \textbf{(5)}$$

$$W(k+1) = W(k) + K(k) * e(k) \quad \textbf{(6)}$$

$$P(k+1) = P(k) - (P(k) * x(k) * x(k)^T * P(k)) / (1 + x(k)^T * P(k) * x(k)) \quad \textbf{(7)}$$

In the above equation (5), K(k) is the Kalman gain, P(k) is the inverse correlation matrix, x(k) is the input vector at iteration k, w(k) is the filter coefficients at iteration k, and e(k) is the error.

D. Modified RLS(M-RLS) Algorithm

In a variety of applications, signal processing techniques are essential for lowering noise and enhancing the quality of real-time signals. Adaptive filtering tasks, such as active noise cancellation, have been successfully completed using the Recursive Least Mean Square (RLMS) algorithm. This study will examine the Modified Recursive Least Mean Square algorithm, its advantages, and its use in communication systems to achieve greater noise reduction.

The improved and modified RLMS algorithm(M-RLS) continually calculates the properties of the background noise in active noise cancellation based on the input signals. The noise components are then eliminated by combining the original signal with an anti-noise signal that is produced next. By ensuring that the anti-noise signal appropriately reflects the noise characteristics, this adaptive filtering technique effectively reduces noise and improves signal-to-noise ratio.

In terms of convergence speed, stability, and adaptability, the modified Recursive Least Mean Square algorithm significantly outperforms the traditional RLMS algorithm. Its use in active noise cancellation systems, in particular, allows for the creation of anti-noise signals that successfully block background noise. As a result, communication systems, particularly mobile ones, may transmit signals that are crystal clear and noise-free, improving user experience overall. The performance of the modified RLMS algorithm is still being improved, and it is being investigated for use in a variety of signal processing applications.

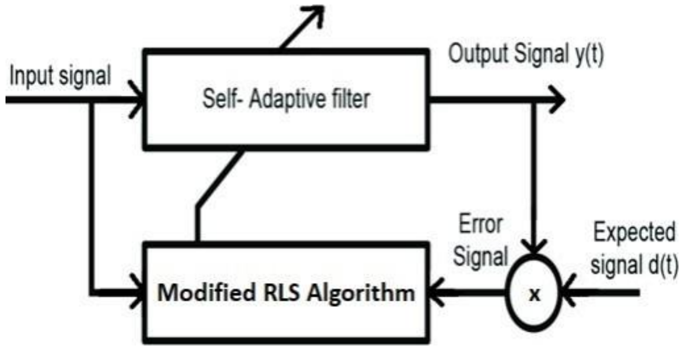

Figure 7. M-RLS Algorithm [8]

In the proposed M-RLS algorithm, the equations (3) and (4) hold well for filter output y(k) and error e(k) but the filter coefficients W(k) and the inverse covariance matrix P(k) changes as per the equations given below.

$$W(k+1) = W(k) + P(k) * x(k) * e(k) / (1 + x(k)^T * P(k) * x(k)) \quad \textbf{(8)}$$

$$P(k+1) = P(k) - (P(k) * x(k) * x(k)^T * P(k)) / (1 + x(k)^T * P(k) * x(k)) \quad \textbf{(9)}$$

M-RLS effectively reduces noise in audio signals and is particularly suitable for applications where noise profiles vary over time, ensuring that the output signal is a cleaner representation of the desired audio.

3. Hardware for M-RLS

This section provides the hardware descriptions for implementation of M-RLS for real time applications

A. OEM –TRAVEO II Microcontroller

In the proposed research, a microcontroller plays a pivotal role in integrating with the Active Noise Cancellation (ANC) system. Specifically, the microcontroller used is the OEM –

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TRAVEO II Microcontroller, which is a 32-bit TRAVEO™ T2G Arm® Cortex® Microcontroller. Here's how the microcontroller is utilized and integrated into the ANC system:

1. **Signal Processing**: The microcontroller is responsible for processing and controlling the ANC system. It serves as the central processing unit for executing the ANC algorithms and managing the adaptive filtering required for noise cancellation. The microcontroller executes the core ANC algorithms such as the Least Mean Square (LMS), Normalized Least Mean Square (NLMS), and other related algorithms to generate the anti-noise signal.

2. **Data Acquisition**: The microcontroller interfaces with the input components of the ANC system, such as the microphones that detect the input audio signals. It collects data from these input sensors, which include the sounds from the external environment and within the earphones.

3. **Filter Coefficients Control**: The microcontroller manages and updates the filter coefficients of the ANC system in real-time. These filter coefficients are modified to create the anti-noise signal, which is phase-inverted to cancel out the unwanted external sounds.

4. **Adaptive Control**: The microcontroller implements adaptive control strategies, such as adjusting the step size of the adaptive algorithms, which ensures the efficient and stable operation of the ANC system. These controls are essential to optimize the noise cancellation process.

5. **ANC Chipset Integration**: The microcontroller cooperates with the ANC chipset, which inverts the sound waves based on the data and control commands sent by the microcontroller. The chipset is an integral part of the ANC system and works in tandem with the microcontroller to achieve noise cancellation.

6. **Speaker Management**: The microcontroller also controls the speakers within the earphones. These speakers generate the anti-phase sound waves that cancel out the external noise. The microcontroller synchronizes the speaker's output with the generated anti-noise signal to achieve effective noise cancellation.

7. **Power and Resource Management**: It's noteworthy that the microcontroller is chosen based on its capabilities to handle the ANC processing tasks with efficient power usage. Given the constraints in the automotive environment, where space and power resources are limited, the microcontroller must be carefully selected for optimal performance.

By integrating the microcontroller with the ANC system, the proposed idea aims to develop an efficient and practical solution for real-time noise cancellation, particularly in the context of the automotive sector. The microcontroller ensures that the ANC system operates effectively to reduce driver fatigue, enhance audio quality, and provide a comfortable and more enjoyable driving experience. This integration of the microcontroller is a critical component in achieving the stated objectives of the research.

B. Main CPU

– Execution of application software
– Up to 160-MHz operation
• MICROPHONE – to detect the input audio signal
• SPEAKERS – for output

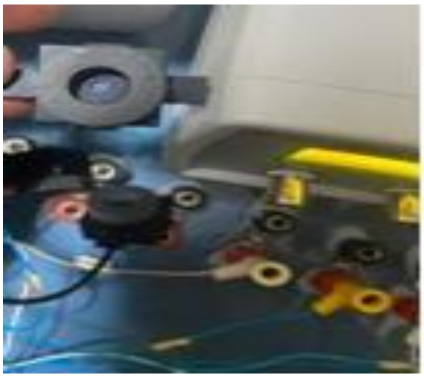

Figure 8. MIC

Figure 9. Speakers

4. Results and Discussion

Fig(10).Shows the sample sinusoidal input signal from signal generator. In (a) shows normal sinusoidal signal and (b) shows the anti phased signal (180^0 out of phase sinusoidal signal). If dealing with a noisy audio signal, a comparable technique involves creating an anti-phased signal to eliminate the noise.

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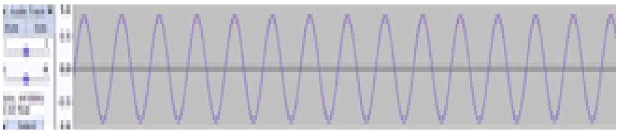
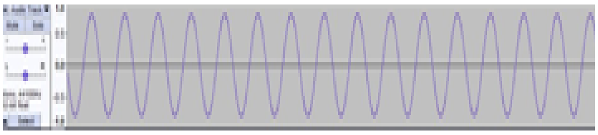
The simulation tools used is MATLAB software



(a) Input Signal



(b) Output Signal (Anti Phase)

Figure 10. Input and Output Signal

Table1.Shows the performance comparison of various ANC methods discussed in this study using simulation tool. The input signal used to obtain the parameter listed in Table1. is "audio.wav" file which is recorded audio signal. The noise signal considered for analysis is additive white Gaussian noise. The noisy audio input signal and the convergence of M-RLS algorithm have been provided in Fig (11). From the Fig (11), it is evident that the convergence rate is very faster for the proposed M-RLS algorithm compared to other ANC algorithms mentioned in this work.

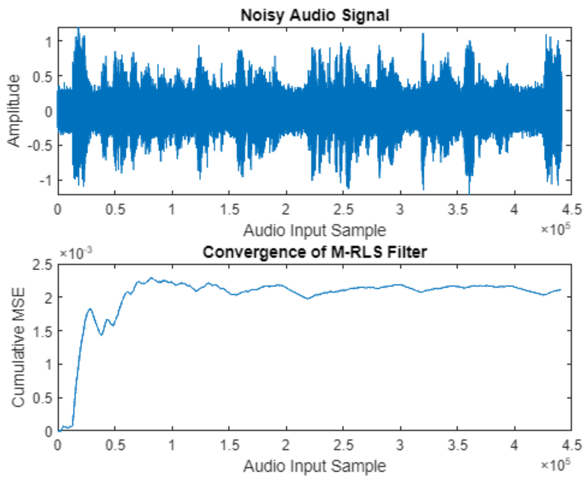


Figure 11. Noisy Audio Input Signal and the Convergence of Proposed M-RLS Algorithm

Table1. MATLAB Simulation Results of M-RLS Algorithm vs other Noise Cancellation Algorithm

Parameters	**Name of the ANC algorithms (LMS,NLMS,RLS)**	**M-RLS algorithm**
Efficiency	Simple and less efficient	Complexity higher but

		more efficient
Convergence rate	Slow convergence	Fast convergence
Real time audio filtering	Not suitable	Works on real time audio
Processing time	LMS- 0.126 seconds NLMS- 0.178 seconds RLS- 0.184 seconds	0.0227 seconds
Bit error rate	Higher	Lesser
SNR(dB)	LMS- 18.8033 NLMS- 18.6332 RLS- 18.9688	19.0676

The Table.1 above illustrates that the M-RLS algorithm outperforms all other algorithms utilized in this study, offering a remarkable 19.0676 dB signal-to-noise ratio (SNR) with an exceptionally short execution time of 0.0227 seconds. Fig 12 visually represents the SNR in decibels and execution time in seconds obtained through MATLAB simulation.

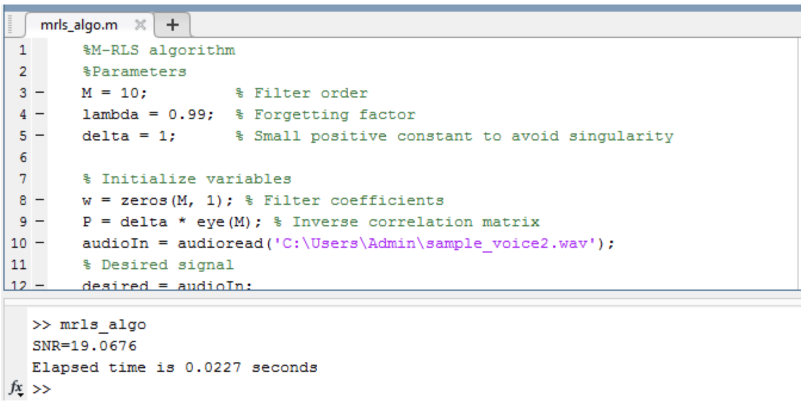


Figure 12. MATLAB simulation Output of M-RLS Algorithm

The hardware results acquired with microphones and speakers are not displayed in a tabular format due to the necessity for multiple trials to attain a noise-free audio signal. Therefore, taking the algorithm to the hardware implementation stage can be a pivotal progression, bridging the gap between a theoretical concept and a practical application.

5. Conclusion

In conclusion, this study delved into the comprehensive assessment of several audio filtering algorithms designed for noise cancellation, with a primary objective of tackling the persistent issue of real-time signal noise mitigation within communication systems. The central thrust of this research revolved around enhancing voice quality and optimizing the auditory experience in portable devices and automobiles, where the presence of disruptive background noise is a notable challenge.

The outcomes, as depicted in Table 1, unequivocally indicate that the proposed M-RLS algorithm outperforms its counterparts in the realm of active noise cancellation (ANC). It excels in parameters crucial to achieving superior performance, including signal-to-noise ratio, overall processing time, and the intricacy of its implementation. These findings underscore the efficacy and promise of the M-RLS algorithm as a key advancement in the pursuit of improved noise reduction technologies, with substantial implications for enhancing user experiences across various audio communication platforms.

6. Future Work

Future work for this study involves validating the proposed M-RLS algorithm through hardware implementation, optimizing its performance for resource-constrained environments, exploring adaptability in dynamic noise profiles, integrating it into diverse applications, and conducting user experience studies and industry collaborations. These collective efforts form a holistic roadmap for advancing the field of active noise cancellation, enhancing its real-world applicability and impact.

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