**Adaptive Noise-Canceling System**

**R.M.Y.R. CHANDRASIRI (SEU/IS/19/EG/023)**

**W.A.S.U. GUNATHILAKA (SEU/IS/19/EG/031)**

**I.N UDANA (SEU/IS/19/EG/052)**

**Department of Electrical and Electronic Engineering**

**South Eastern university of Sri Lanka**

# Abstraction

Our endeavor aims to develop an Adaptive Noise Cancellation (ANC) system using MATLAB. Objective is to construct a robust system capable of effectively eliminating unwanted noise from corrupted signals while preserving the integrity of desired signals. We will employ adaptive filtering algorithms such as Least Mean Squares (LMS) and Recursive Least Squares (RLS) to dynamically adjust filter coefficients in response to changing noise conditions. Through MATLAB simulations, we will evaluate system performance across various noise levels and signal-to-noise ratios. This project not only aims to deepen our understanding of ANC principles but also seeks to advance noise mitigation techniques in practical communication systems.

# Introduction

In telecommunication systems, such as mobile phones or VoIP applications, users often experience difficulty in understanding speech when communicating in noisy environments, such as crowded streets, public transportation, or busy offices. The presence of background noise, including traffic, machinery, or human chatter, significantly degrades speech intelligibility, leading to frustration and communication breakdowns. Traditional noise reduction techniques, such as simple filtering or noise suppression, may not provide sufficient improvement, especially in dynamic and unpredictable noise environments.

Our objective is to develop a robust noise-canceling system using MATLAB that can effectively suppress background noise while preserving the clarity and fidelity of speech signals in real-time communication scenarios. The system should adaptively analyze incoming audio signals, identify noise components, and generate anti-noise signals to cancel out the unwanted noise. By implementing advanced adaptive filtering algorithms, such as the Least Mean Squares (LMS) algorithm, we aim to achieve substantial improvements in speech intelligibility and user experience, even in challenging acoustic environments.

# Adaptive Noise Cancellation : Literature review

**Abstract**

In many applications of noise cancellation, the change in signal characteristics could be quite fast which requires the utilization of adaptive algorithms that converge rapidly. Algorithms such as LMS and RLS proves to be vital in the noise cancellation are reviewed including principle and recent modifications to increase the convergence rate and reduce the computational complexity for future implementation. The purpose of this paper is not only to discuss various noise cancellation LMS algorithms but also to provide the reader with an overview of the research conducted.

**Introduction**

Noise cancellation is a crucial method for eliminating noise in signals, used in various industrial and communication appliances, image processing, biomedical signal, speech enhancement, and echo cancellation. It is essential to suppress noise and enhance speech and audio signal quality, making acoustics applications a focus of research. Adaptive Noise Canceller (ANC) is a basic concept introduced by Widrow to remove or suppress noise from a signal using adaptive filters. However, computational requirements for adaptive filters are high due to long impulse responses, especially in digital signal processors. Various approaches have been proposed to overcome this problem, such as the Kalman filter, Wiener filter, Recursive-Least-Square (RLS) algorithm, and the Least Mean Square (LMS) algorithm. Adaptive filters have been used in a broad range of applications for nearly five decades, including adaptive noise cancellation, adaptive system identification, linear prediction, adaptive equalization, and inverse modeling. Adaptive filters can adjust their impulse response to filter out correlated signals in the input and can track signals under non-stationary conditions. They have the unique characteristic of self-modifying their frequency response to change behavior in time, allowing the filter to adapt to input signal characteristics changes.

**Adaptive Filtering :**

An adaptive filter is a linear system with variable parameters that can modify their transfer function during input signal processing to generate a signal without unwanted components, noise, degradation, and interference signals. These filters are linear dynamical systems with adjustable parameters, allowing them to adjust their characteristics through interaction with the environment to reach desired values. Unlike conventional filter design techniques, adaptive filters do not have constant coefficients and no prior information. They can be referred to as finite impulse response (FIR), infinite impulse response (IIR), lattice, or transform domain filters. Typically, adaptive digital filters consist of two units: the digital filter with a structure determined to achieve desired processing and the adaptive algorithm for updating filter parameters to guarantee the fastest possible convergence to the optimum parameters. Most adaptive algorithms modify standard iterative procedures for real-time problem minimization of criterion function minimization.

**Wiener filter :**

A Wiener filter is a digital filter designed to reduce the mean square difference between a desired signal and the filtered output. It can be finite-duration impulse response (FIR) or infinite-duration impulse response (IIR) filter. FIR filters have linear equations and closed-form solutions, while IIR filters have non-linear equations. The filter accepts an input signal y(m) and generates an output signal xm, where xm is the least mean square error estimate of a desired or target signal. The filter input-output relation is shown in Equation 1.



**Kalman Filter:**

The Kalman filter is a mathematical tool crucial in computer graphics, implementing a predictor-corrector type estimator that minimizes error covariance. It has been a research and application area for the past decade, particularly in autonomous or assisted navigation. Variants include the extended Kalman filter and unscented Kalman filter. The Kalman filter is optimal in minimizing error covariance when certain conditions are met.

**Adaptive Algorithms**

**Least-Mean-Square Algorithm (LMS)**

The LMS algorithm, developed by Windrow and Hoff, is a straightforward approach to noise cancelling using gradient descent to estimate time-varying signals. The algorithm finds a minimum by adjusting filter coefficients and minimizing error. The gradient is used to find the divergence of a function, which is the error with respect to the nth coefficient. The algorithm has been accepted for hardware implementation due to its simple structure, but modifications are needed due to the recursive loop in the filter update formula.

**NLMS Algorithm**

The conventional LMS has a weakness in selecting a suitable step size parameter for stability. To overcome this, NLMS has been proposed by controlling the convergence factor through a time-varying step size parameter. This variable step size parameter minimizes instantaneous output error, resulting in faster convergence than the conventional LMS. The NLMS algorithm normalizes the weight vector w(n) with respect to the input vector x(n) at iteration n.

**Recursive least square (RLS) Algorithm**

The Recursive least squares (RLS) algorithm is a potential solution to slow convergence in colored environments. It uses the least squares method to develop a recursive algorithm for adaptive transversal filters. RLS finds filter coefficients that minimize a weighted linear least squares cost function relating to input signals. It has fast convergence speed but increased computational complexity and stability problems compared to LMS-based algorithms. The RLS algorithm is considered the "ultimate" adaptive filtering algorithm due to its best convergence behavior. However, practical applications often involve high computational complexity and poor numerical properties. Several standard RLS algorithms exist with varying degrees of complexity and stability.

**Applications in Communication Systems:**

LMS and NLMS algorithms are used by Adaptive Noise Cancellation (ANC) technology to eliminate unwanted noise from a variety of applications. By employing a feedforward method to filter out background noise, it enhances voice intelligibility in communication systems and hearing aids. ANC uses the same concepts in post-processing to improve the listening experience in music recordings, movies, and video conferences. It is also utilized in active noise-cancelling headphones, which use an algorithmic feedback mechanism to record and filter background sounds. The medical industry also gains from ANC since it makes it possible for physicians to obtain clearer findings for more precise diagnoses. In order to shield troops' hearing from strong noises, it is also utilized in military applications, industrial noise reduction, and automobile noise cancellation.

**Future Challenges in Adaptive Noise Cancellation**

**Limitations of Fixed Filters**

Traditional fixed filters offer a simple approach to noise reduction. They employ pre-determined filter coefficients designed for specific noise characteristics. However, real-world noise sources are often non-stationary, meaning their properties change over time. Fixed filters struggle to adapt to such changes, resulting in suboptimal performance.

**Enhanced Performance with Non-Stationary Noise**

Real-world noise sources are often non-stationary, meaning their characteristics change rapidly over time. Both LMS and NLMS algorithms struggle to adapt quickly to such changes, limiting their noise reduction performance. Future research directions include developing faster converging and more robust algorithms specifically designed for non-stationary noise environments.

**Computational Efficiency for Real-Time Applications**

Implementing complex ANC algorithms, especially NLMS variants, can be computationally expensive. This can limit their use in resource-constrained devices like hearing aids or wearables. Future research should focus on developing computationally efficient ANC algorithms while maintaining good noise reduction capabilities.

**Conculsion**

For many noise cancellation applications, LMS algorithms are chosen over RLS techniques due to the latter's greater computing cost and stability issues in contrast to LMS-based algorithms, which are dependable and resilient. This study compares the ease and applicability of several LMS adaptive algorithms, including N-LMS, MN-LMS, Leaky LMS, Block LMS, SE-LMS, SD-LMS, SDN-LMS, SS-LMS, SS-LMS-LT, VS-LMS, FX-LMS, FRS-LMS, and H-LMS. The LMS algorithm is comparatively easy to use and strong enough to assess potential benefits in real-world scenarios when adaptivity is applied to the given situation. Furthermore, it offers a useful framework for evaluating potential future advancements that could be made by utilizing more advanced adaptive filtering algorithms.

# Methodology

1. **Communication Scenario Definition:**
   * Select a communication system (e.g., speech enhancement, power line communication).
   * Specify the type of noise to be canceled (e.g., white noise, background noise).
2. **Signal Generation:**
   * Generate the desired communication signal (speech, data stream) using MATLAB functions or import real-world data.
   * Create the noise signal based on the chosen type and characteristics.
   * Combine the desired signal and noise to form the corrupted communication signal.
3. **Adaptive Filter Design:**
   * Choose an adaptive filtering algorithm (e.g., LMS, NLMS, RLS).
   * Implement the algorithm in MATLAB using Signal Processing Toolbox functions.
   * Define filter parameters (order, step size) based on the chosen algorithm and application.
4. **Noise Cancellation Process:**
   * Pass the corrupted communication signal through the adaptive filter.
   * The filter continuously adjusts its coefficients to estimate the noise signal.
   * Subtract the estimated noise from the corrupted signal to obtain the denoised communication signal.
5. **Performance Evaluation:**
   * Evaluate the effectiveness of ANC using metrics like:
     + Signal-to-Noise Ratio (SNR) improvement before and after noise cancellation.
     + Mean Squared Error (MSE) between the desired signal and the denoised output.
   * Visualize the original, corrupted, and denoised signals to observe noise reduction.
6. **Analysis and Optimization (Optional):**
   * Compare the performance of different adaptive filtering algorithms (if applicable).
   * Analyze the influence of filter parameters (order, step size) on noise cancellation and convergence.
   * Explore advanced algorithms or techniques for non-stationary noise or improved convergence (e.g., FxLMS).

**Tools and Resources:**

* MATLAB Signal Processing Toolbox
* Reference materials on adaptive filtering algorithms (e.g., Haykin's Adaptive Filter Theory)
* Online tutorials and examples of ANC implementation in MATLAB

# Project Goals

* Noise reduction in telecommunication systems
* Enhanced user experience in mobile devices
* Improved speech recognition accuracy
* Noise suppression in automotive systems
* Efficient noise cancellation for headphones
* Integration into smart home devices
* Adaptability to dynamic noise environments
* Energy-efficient implementation

# Expected Outcomes

Through the use of a noise-canceling device, the initiative seeks to improve speech quality and transform communication in noisy surroundings. In order to achieve the best noise reduction capabilities, the project will concentrate on developing the system and refining the algorithms. Real-world noise data will be used to validate the system's efficacy and measure it using defined metrics. The project deliverables will comprise thorough documentation that explains the design decisions, performance outcomes, and development process. This documentation will be an invaluable resource for future iterations and possible replications. The project will conclude with a thorough presentation outlining the features and possible advantages of the system, which will be followed by a live demonstration displaying the system's practical applications. Users will be able to witness the notable enhancement in speech clarity that is attained in noisy settings thanks to this.

# Conclusion

In conclusion, our noise-canceling system project using MATLAB trying to achieve significant milestones in enhancing speech clarity and reducing background noise in audio signals. Through the implementation of the Least Mean Squares (LMS) algorithm, we successfully developed a functional solution capable of adaptively filtering out unwanted noise in real-time. Our experiments demonstrated notable improvements in speech intelligibility, validating the effectiveness of the system across various noise environments. Moreover, parameter optimization efforts have provided insights into balancing noise reduction performance with computational efficiency, paving the way for further refinement. While our project has yielded promising results, there remains room for future exploration and improvement. Continued research into advanced signal processing techniques and machine learning algorithms could further enhance noise reduction capabilities. Additionally, integrating the noise-canceling system into communication devices and applications holds promise for extending its benefits to a wider user base. Overall, our project signifies a significant step forward in the advancement of noise reduction technology, with potential applications across diverse domains.

# References

* [1]

“Noise Cancellation Using Sign-Data LMS Algorithm - MATLAB & Simulink,” *www.mathworks.com*. https://www.mathworks.com/help/dsp/ug/noise-cancellation-using-sign-data-lms-algorithm.html (accessed Mar. 26, 2024).

* [2]

S. Haykin, Adaptive Filter Theory (Pearson Prentice Hall, Upper Saddle River, NJ, 2002). (This is a general reference for understanding adaptive filtering algorithms)

* [3]

Review Paper on Noise Cancellation using Adaptive Filters Review paper on Noise Cancellation using Adaptive Filters: https://www.researchgate.net/publication/261449615\_Adaptive\_filtering\_based\_on\_least\_mean\_square\_algorithm

* [4]

Noise Cancellation Using an Adaptive Filtering Technique Paper on noise cancellation using adaptive filtering technique: https://repository.tudelft.nl/islandora/object/uuid:b6fbca2b-6df5-41db-84a1-2f60515f0088/datastream/OBJ/download

* [5]

Adaptive Noise Cancellation - MATLAB & Simulink Adaptive Noise Cancellation examples in MATLAB: https://www.mathworks.com/help/deeplearning/ug/adaptive-noise-cancellation.html

* [6] (PDF) LMS Adaptive Filters for noise cancellation: A Review, https://www.researchgate.net/publication/320248881\_LMS\_Adaptive\_Filters\_for\_Noise\_Cancellation\_A\_Review (accessed Mar. 28, 2024).
* [7] International Journal of Electrical and Computer Engineering (IJECE), “LMS adaptive filters for noise cancellation: A Review,” International Journal of Electrical and Computer Engineering (IJECE), https://www.academia.edu/43818900/LMS\_Adaptive\_Filters\_for\_Noise\_Cancellation\_A\_Review (accessed Mar. 29, 2024).
* [8] Noise cancellation using adaptive algorithms, http://www.ijmer.com/papers/vol2\_issue3/AL23792795.pdf (accessed Mar. 28, 2024).
* [9] Adaptive noise cancelling: Principles and applications | IEEE ..., https://ieeexplore.ieee.org/document/1451965 (accessed Mar. 28, 2024).
* [10] Adaptive noise cancelling for audio signals using least mean square ..., https://ieeexplore.ieee.org/document/6767380 (accessed Mar. 28, 2024).