Project Title : **Leaky LMS algorithm based low complexity Adaptive Noise Cancellation**

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ABSTRACT:

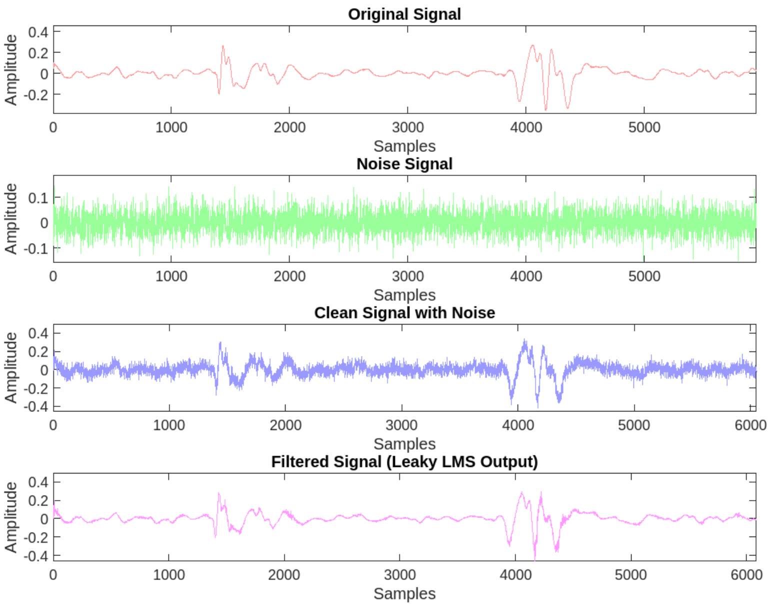
This project aims to focus on the denoising and analysis of acoustic signals, specifically speech signals, by implementing the Leaky Least Mean Squares (LMS) Adaptive Filtering Algorithm. Speech signals play a critical role in diagnosing speech and neurological disorders, as they often reflect subtle changes in an individual's voice that indicate potential health issues. However, these signals are highly vulnerable to noise, which significantly degrades their quality and reduces the effectiveness of speech-processing systems. While existing techniques like wavelet transforms, deep learning, and Empirical Mode Decomposition (EMD) have been utilized for noise reduction, they tend to be computationally expensive and unsuitable for real-time applications. The core objective of this project is to develop a computationally efficient Leaky LMS adaptive filtering algorithm that effectively mitigates noise while preserving the integrity of the original speech signal. This approach seeks to improve the clarity of speech signals, ensuring that critical diagnostic information is maintained. By offering a more efficient and real-time solution, this project addresses the limitations of current noise-reduction methods and contributes to more effective speech-based diagnostics in medical applications.

Objectives	To design and implement a computationally efficient Leaky LMS adaptive filtering algorithm for denoising.
Knowledge acquired from the courses studied	LMS algorithm and its variations such as Leaky LMS algorithm. Implementation of LMS algorithm and LEAKY LMS algorithm using MATLAB
Realistic Constraints	Isolating a clean reference noise signal is challenging. Unpredictable noise can degrade signal processing Slow convergence in noisy environments delays results.
Standards to be referred/followed	IEEE 802.11 (Wi-Fi Standards): Defines methods for signal processing in wireless communication, ensuring effective filtering and noise management in networks. IEC 61672 (Electroacoustic Measurement Standards): Provides guidelines for measuring the performance of audio and acoustic signal processing equipment, which is essential for filter accuracy. ISO 9001 (Quality Management): Ensures that signal processing systems, including adaptive filters, meet consistent quality standards for reliability and performance.

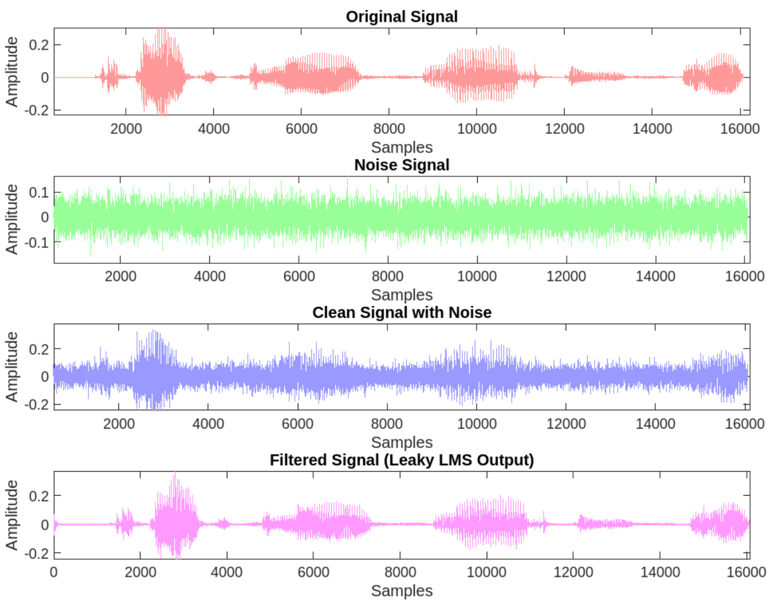
Multidisciplinary tasks involved	Basic Sciences for Mathematics. Computational and IT field for MATLAB. Biomedical Engineering for the input dataset of PCG signals.
Deliverables/ Outcomes	Software interface with MATLAB.

RESULTS:

DENOISING OF PCG SIGNAL WITH GAUSSIAN NOISE USING MULTISTAGE LEAKY LMS :

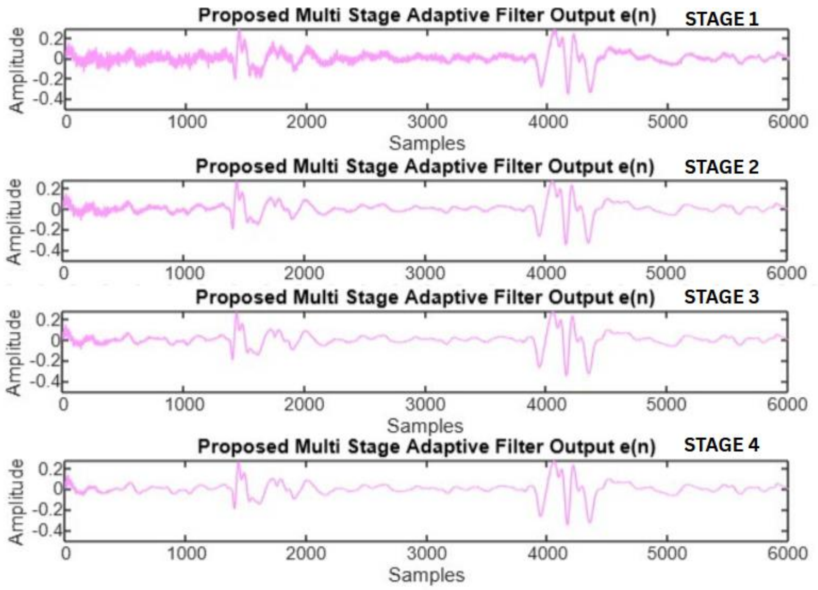


DENOISING OF SPEECH SIGNAL WITH GAUSSIAN NOISE USING MULTISTAGE LEAKY LMS :

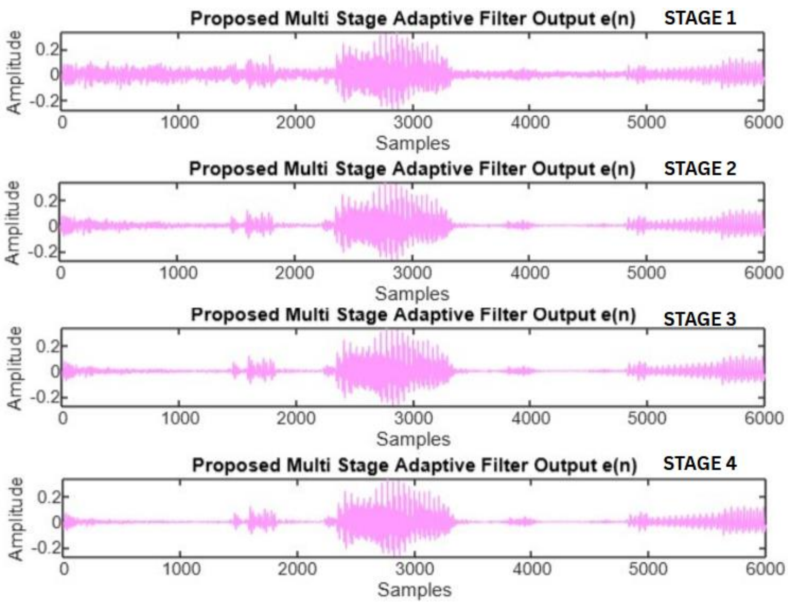


COMPARISION OF STAGES:

PCG SIGNAL:



SPEECH :



References

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