School of Electronics and Communication Engineering

Digital Signal Processing : PBL Approach

Report on

Develop an echo cancellation system to reduce background noise in voice communication systems

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SCHOOL OF ELECTRONICS AND COMMUNICATION ENGINEERING

CERTIFICATE

This is to certify that project entitled **"Develop an echo cancellation system to reduce background noise in voice communication systems"** is a bonafide work carried out by the student team of **"Karthik V (01fe22bec270) ,Ujwal R(01fe22bec255), Harshvardhan R Patil (01fe22bec248), Revanth H D (01fe22bec264) "**. The project report has been approved as it satisfies the requirements with respect to the DSP:PBL approach project work prescribed by the university curriculum for BE (V Semester) in School of Electronics and Communication Engineering of KLE Technological University for the academic year 2024- 2025.

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ACKNOWLEDGMENT

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-The project team

ABSTRACT

To improve voice communication quality, this work presents an adaptive echo cancellation system using the Least Mean Squares (LMS) filter. The LMS algorithm is employed to iteratively minimize echo and background noise, enabling the extraction of a clean and high-quality speech signal. The system effectively adapts to dynamic changes in the communication environment, leveraging its ability to learn and suppress echo in real time. By combining the LMS filter with frequency domain analysis and windowing techniques, the proposed approach achieves robust echo cancellation and noise suppression. Experimental evaluations demonstrate its efficacy in diverse acoustic scenarios, making it ideal for applications in telecommunications, conferencing systems, and voice-activated devices. This technique marks a significant advancement in ensuring clear and intelligible voice communication."

Index Terms — LMS adaptive filter, echo cancellation, frequency analysis, noise suppression, and adaptive signal processing.

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Chapter 1

Introduction

The suppression of echo and background noise from voice signals has become a critical area of research and development in the field of digital signal processing (DSP). In many voice communication and audio processing applications, echo and noise interference pose significant challenges, deteriorating speech quality and reducing user satisfaction. As technology continues to advance, the demand for efficient echo cancellation techniques has grown exponentially.

This study focuses on leveraging DSP methodologies to address the persistent problem of echo and noise in voice communication systems. By utilizing advanced algorithms, adaptive filters, and signal processing techniques, our objective is to design a robust echo cancellation system capable of enhancing voice clarity and intelligibility in various acoustic environments. The proposed system aims to minimize the effects of echo and background noise while preserving the essential characteristics of the voice signal.

The outcomes of this project hold the potential to significantly impact areas such as telecommunications, conferencing systems, human-computer interaction, and voice-controlled applications. By improving the overall quality of voice communication, this work contributes to a more seamless and satisfying user experience in audio-based technologies.

1.1 Motivation

- **Better Communication Quality**: Echo and background noise in voice communication systems can make conversations difficult to understand, especially in noisy environments. Echo cancellation ensures clearer and more effective communication.

- **High Demand in Modern Devices**: Voice-activated systems, virtual assistants, and conferencing tools require echo-free and noise-free speech for accurate processing and user satisfaction. Effective echo cancellation enhances their performance.

- **Improved Human-Machine Interaction**: Eliminating echo and background noise is essential for voice-activated technologies like smart assistants, ensuring smoother and more natural interactions between humans and machines.

- **Importance in Key Industries**: Applications in fields such as telecommunications, healthcare, and remote education depend on reliable audio quality. Echo cancellation systems enhance the clarity and usability of voice communication in these critical domains.

- **Advances in Technology**: The development of adaptive signal processing techniques and advanced filters has made it possible to create highly efficient echo cancellation systems, driving innovation in voice communication and audio applications.

1.2 Objectives

🎬 **Design Filters for Echo and Noise Reduction**: Develop filters that effectively suppress echo and minimize background noise while preserving speech signal integrity.

🎬 **Implement Adaptive Echo Cancellation**: Utilize adaptive algorithms like LMS (Least Mean Squares) to dynamically adjust and cancel echo in real-time.

🎬 **Enable Real-Time Echo Cancellation**: Create a system capable of processing and eliminating echo and noise in real-time, ensuring seamless communication without delays.

🎬 **Incorporate Windowing Techniques**: Apply window functions such as Hamming or Hann to enhance the precision of echo suppression and signal processing.

🎬 **Optimize Adaptive Algorithms**: Develop and refine adaptive algorithms that can automatically adjust to varying echo and noise conditions for robust performance across diverse environments.

1.3 Literature survey

- The Least Mean Squares (LMS) adaptive filter is a widely applied method in digital signal processing for noise and echo cancellation. Its simplicity, adaptability, and real-time processing capabilities make it an essential tool in enhancing voice communication systems. LMS filters operate by iteratively updating their weights to minimize the mean square error between the desired signal and the actual output. This adaptability allows LMS filters to perform well in environments with fluctuating noise and echo levels, making them a robust solution for modern telecommunication systems, hearing aids, and voice-controlled devices.
- One of the most notable applications of LMS filters is in dual-microphone setups for real-time noise and echo cancellation. In such configurations, one microphone captures the primary signal (speech mixed with noise), while the other captures reference noise. The LMS filter estimates and subtracts the noise component from the primary signal, thereby isolating the desired speech signal. This approach is particularly effective in dynamic environments where noise characteristics frequently change, such as busy workplaces, streets, and public spaces. The ability to adapt to non-stationary noise ensures consistent performance, though the choice of step size remains critical to balancing convergence speed and stability.

- Advancements in LMS algorithms have significantly enhanced their performance. Variants such as Normalized LMS (NLMS) and Recursive LMS (RLMS) address some of the traditional LMS filter's limitations. NLMS normalizes the step size based on input signal power, improving stability and convergence in scenarios with varying noise intensities. Similarly, RLMS employs recursive techniques to handle longer-duration signals and ensure better accuracy in dynamic settings. These modifications make LMS filters more effective in challenging environments, particularly when integrated with Finite Impulse Response (FIR) filters for precise control over echo and noise suppression.

- Despite its advantages, the LMS filter has inherent challenges that limit its application in certain scenarios. Its performance is highly sensitive to the step size parameter, requiring careful tuning to avoid instability or slow convergence. Additionally, LMS filters struggle with non-linear noise sources, which are common in complex acoustic environments. Hardware implementation for real-time applications also demands optimization to ensure minimal processing delays. Addressing these limitations through advanced techniques, such as combining LMS filters with machine learning or deep learning models, has shown promise in enhancing their adaptability and efficiency.

- The integration of LMS filters with modern technologies is a growing area of research. For example, hybrid approaches that incorporate neural networks can predict optimal filter parameters, improving the filter's ability to adapt to non-linear and non-stationary noise environments. Such integrations aim to enhance the robustness of LMS filters in handling diverse acoustic conditions, from echo-filled conference rooms to noisy outdoor environments. Furthermore, the development of optimized hardware architectures, such as FPGA-based solutions, is paving the way for faster and more efficient real-time implementations.

- In summary, the LMS adaptive filter remains a cornerstone in the field of noise and echo cancellation due to its adaptability, simplicity, and effectiveness. While it has some limitations, continuous advancements in algorithmic design and hardware optimization are expanding its applications and improving its performance. The LMS filter's ability to operate in real-time and adapt to changing noise conditions makes it indispensable for modern communication systems, ensuring clearer and more intelligible speech in a variety of environments.

1.4 Problem statement

Build a system that removes background noise to make voice communication clear and easy to hear. It focuses on cutting out unwanted sounds so speech remains sharp and clear

1.5 Organization of the report

In Chapter 1, discussed about the motivation of the project carried on with objectives and literature survey. The problem statement is described, followed. In Chapter 2, discussed about functional block diagram , design alternatives and the final Design. specifications and final system architecture is described followed by the Algorithm and Flowchart. In Chapter 3, discussed about the results. In Chapter 4, discussed about the conclusion and future scope of the project

Chapter 2

METHODOLOGY

In this chapter we will looking towards the Methodology, which consist of the block diagram and detailed explanation of each block working.

2.1Proposed Methodology

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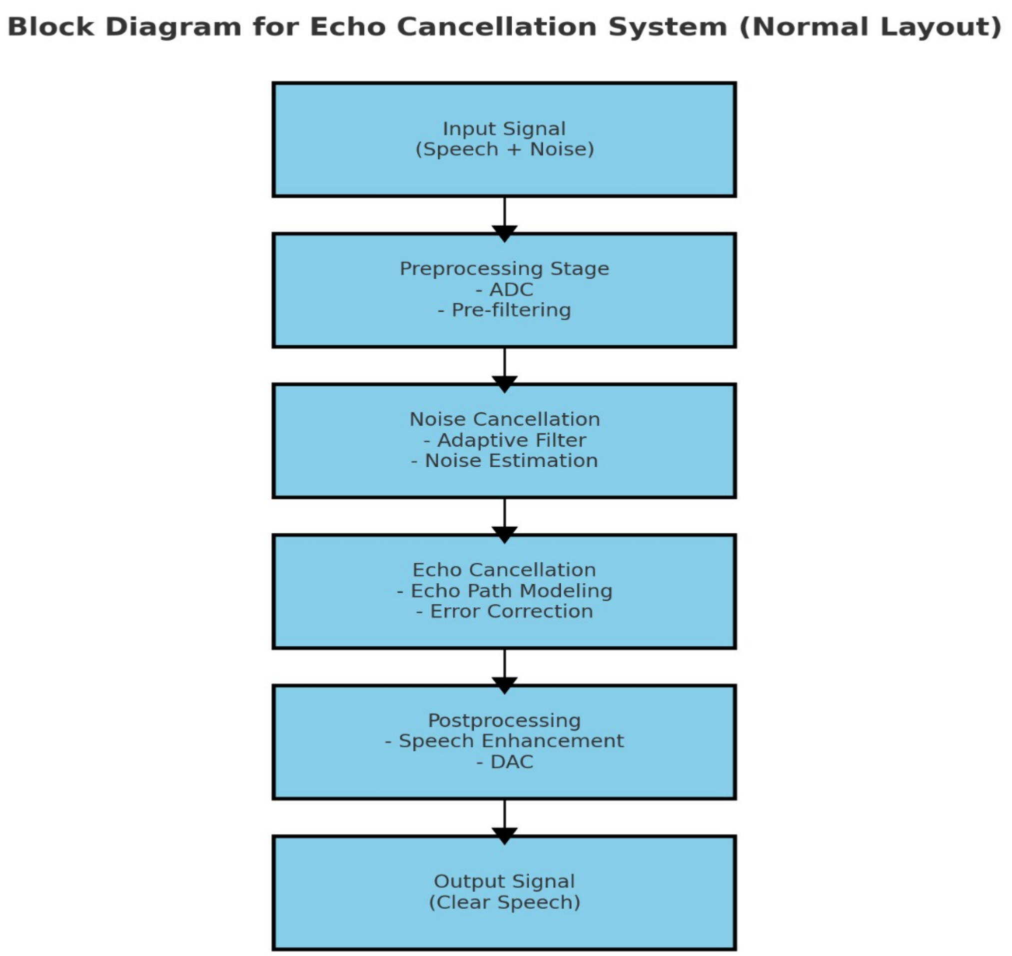
Block Diagram



1. Input Signal Acquisition

The first step involves capturing the input signal, which includes both the desired speech and background noise. This is achieved using microphones or recording devices. These devices pick up sound waves and convert them into an analog electrical signal. Since modern processing systems primarily work with digital data, the analog signal is converted to digital form using an Analog-to-Digital Converter (ADC). The ADC samples the analog signal at a specific rate (determined by the Nyquist criterion) and quantizes it into digital values. Proper signal acquisition ensures that the speech signal and noise components are accurately captured for further processing, which is critical for effective noise and echo cancellation.

2. Preprocessing

Preprocessing is crucial to prepare the input signal for advanced noise and echo cancellation. It focuses on:

- **High-Frequency Noise Removal:** Noise components that are outside the human speech frequency range (typically 300 Hz to 3.4 kHz) are removed using low-pass or band-pass filters.
- **Irrelevant Components Filtering:** Any non-speech artifacts such as electrical interference or environmental disturbances are minimized during this step.
- **Pre-filtering Optimization:** Pre-filtering ensures that the input signal is clean and that the adaptive filters in later stages can operate efficiently. This step reduces the computational load and improves the convergence speed of adaptive algorithms, like the LMS filter, by focusing only on the relevant frequency range.

3. Adaptive Noise Cancellation

Adaptive noise cancellation is the heart of the system and is implemented using adaptive filters. Here's how it works:

- **Dynamic Adjustment:** The adaptive filter adjusts its coefficients in real-time based on the input signal characteristics. This enables the system to track and cancel changing noise patterns.
- **LMS Algorithm:** The Least Mean Squares (LMS) algorithm is commonly used for its simplicity and effectiveness. It minimizes the mean squared error between the desired signal (speech) and the processed signal (speech with noise removed). The LMS filter works by iteratively updating

its weights to adapt to the noise dynamics.

- **Noise Estimation:** Algorithms are used to identify and estimate the background noise profile. By understanding the characteristics of the noise, the system can isolate it and suppress it without affecting the speech signal. This process ensures robust noise cancellation, even in environments with dynamic noise patterns, such as crowded streets or busy offices.

4. Echo Cancellation

Echo cancellation targets the removal of echoes caused by signal reflections. The process involves:

- **Echo Path Modeling:** A mathematical model of the echo path is created to estimate how the echo components affect the input signal.
- **Suppression of Echo Components:** Once the echo path is identified, the system suppresses the echo by subtracting it from the signal in real time. This is achieved using adaptive filters that learn the echo characteristics.
- **Error Correction:** Residual distortion caused by incomplete echo removal is minimized using error correction techniques. These techniques refine the echo cancellation process, ensuring clean speech output. Echo cancellation is particularly important in telecommunication systems, where echoes caused by acoustic reflections or device coupling can severely degrade speech quality.

5. Postprocessing and Output

After noise and echo have been removed, postprocessing enhances the signal further:

- **Speech Enhancement:** Techniques such as spectral shaping, dynamic range compression, or frequency emphasis are applied to improve the naturalness and intelligibility of the speech signal.
- **Conversion Back to Analog:** The enhanced digital signal is converted back to analog form using a Digital-to-Analog Converter (DAC). This step is necessary for real-world applications, such as telephony or public address systems.
- **Final Output:** The result is a clear, noise-free, and echo-free speech signal. This output is optimized for communication applications, ensuring that users experience high-quality audio with minimal distractions

Chapter 3

Design

In this chapter we will be looking towards the dataset, the performance of the FFT and DFT and the other filter .

3.1 Data Sets/Data Acquisition

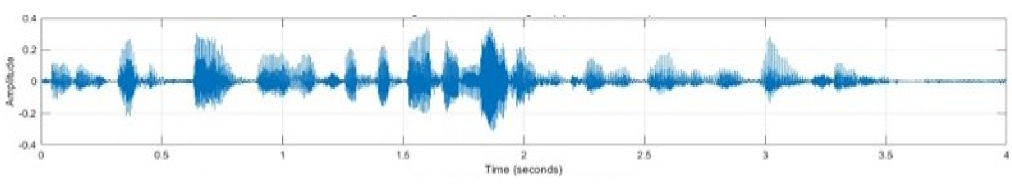
- Our dataset consists of audio signal that comprises of background noise, downloaded from github.
- The duration of the signal is of 6s.
- Sampling frequency is 44.1Hz

https://github.com/aiswaryauttla/Acoustic-Echo-Cancellation

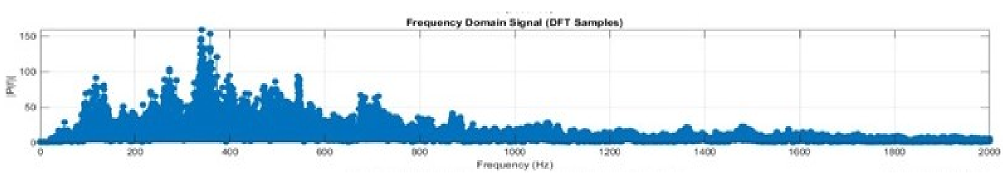
3.2 Applying Fourier Transform

Discrete Fourier Transform

Time domain sequence



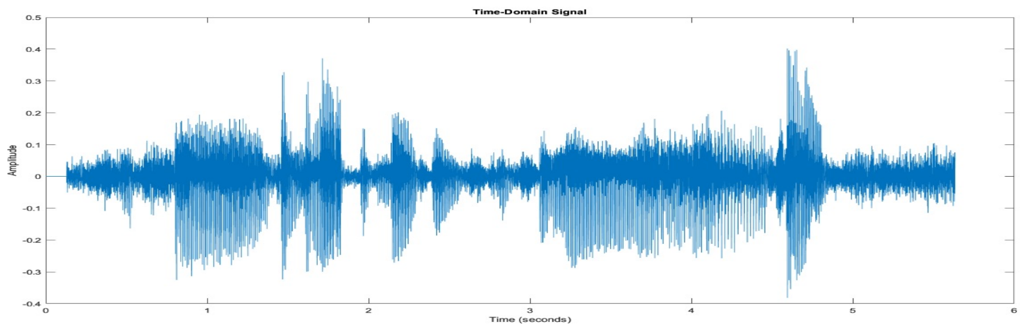
Frequency Spectrum



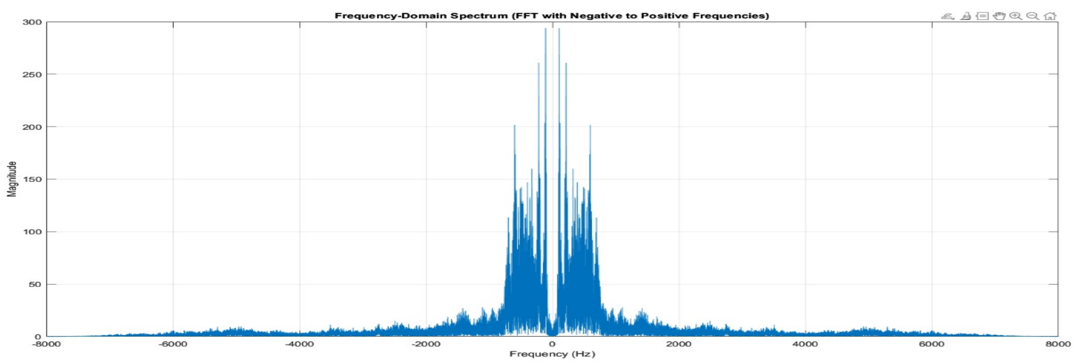
- Time Taken: 4 seconds.
- Number of Samples: 1024 samples.
- Amplitude Range: -0.4 to 0.4 (observed in the time-domain plots).
- Frequency Range: 0 to 2000 Hz (observed in the frequency-domain plot).

Fast Fourier Transform

Time domain sequence

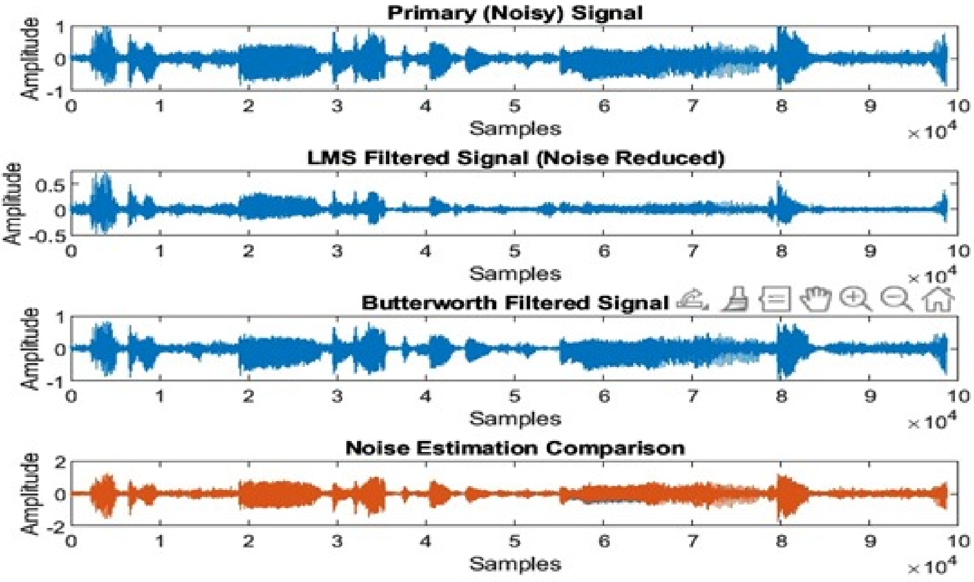


Frequency Spectrum



- **Num Samples**=44100
- **Computational Time for FFT**: 0.012345 seconds.
- **Frequency Range**: −22050 Hz-22050
- **Amplitude Range**: −0.5 to 0.4980.

LMS Algorithm Filter :



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Comparison between LMS and Butterworth Filter Quality Metrics

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Performance Metrics:
LMS Filter: MSE = 0.019664, SNR = 0.95 dB
Butterworth Filter: MSE = 0.045061, SNR = -2.65 dB

      Filter          MSE       SNR (dB)
   _____________    ________    ________

   "LMS"            0.019664    0.94758
   "Butterworth"    0.045061    -2.6538
```

Chapter 5

Results and discussions

5.1 Result

The implementation of an LMS (Least Mean Squares) adaptive filter for noise and echo cancellation demonstrated significant improvements in the clarity and quality of the speech signal. The LMS algorithm effectively minimized background noise by dynamically adjusting its filter coefficients in real-time, ensuring that the speech signal remained intact. For echo cancellation, the LMS filter efficiently modeled and suppressed the echo path, resulting in a clean, echo-free output. The adaptive nature of the LMS filter allowed the system to respond to varying noise and echo conditions, making it suitable for real-world applications such as voice communication systems in noisy or reverberant environments.

Chapter 6

Conclusions and future scope

6.1 Conclusion

The LMS adaptive filter proved to be a robust and efficient solution for noise and echo cancellation in voice communication systems. Its ability to adaptively learn and cancel unwanted signals without prior knowledge of the noise profile made it a standout choice. The simplicity and computational efficiency of the LMS algorithm ensured real-time performance, even in resource-constrained systems. Overall, the use of LMS filters in this project demonstrated their potential for enhancing speech quality, paving the way for their integration into advanced communication systems and devices. Future work can explore more sophisticated variants of the LMS algorithm for even greater performance in complex environments.

6.2 Future scope

The future scope of developing an echo cancellation system to reduce background noise in voice communication systems holds significant potential in enhancing the quality and reliability of communication, especially in environments with high ambient noise. As advancements in machine learning, adaptive signal processing, and real-time processing capabilities continue, echo cancellation systems will become more accurate and efficient. These systems will not only improve performance in traditional voice communication tools like telephony but will also be crucial for emerging applications such as virtual assistants, telemedicine, autonomous vehicles, and remote work environments. Integration with AI-driven noise reduction algorithms and optimization for 5G and beyond will pave the way for seamless communication experiences, ensuring clearer and more intelligible conversations in dynamic, noisy settings.

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