Digital Signal Processing: PBL approach

Theme: Speech Processing

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

Course: Digital Signal Processing

Course Code: 23EECC303

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Credits: 4 [2-0-2]

Hours/Week: 6

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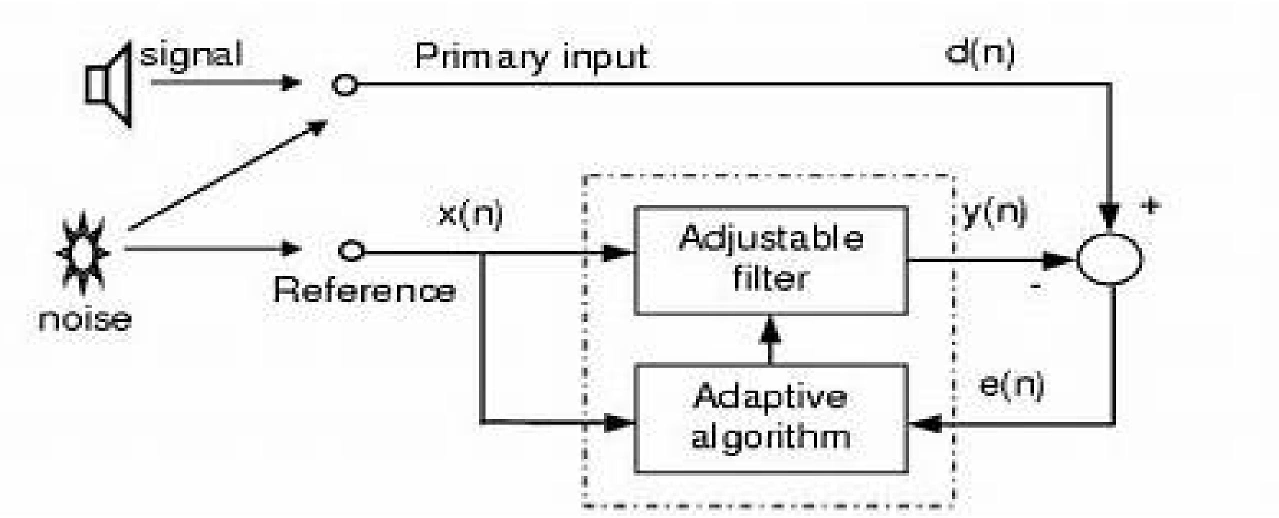
Content

- Problem Statement
- Introduction
- Literature Survey
- Functional Block Diagram
- Data Sets/Data Acquisition
- Methodology
- Result of DFT
- Result of FFT
- LMS Algorithm Filter
- Results and Observations
- References

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



Introduction

- Background noise cancellation is a technique in audio processing that reduces unwanted ambient sounds, making the main audio signal clearer.

- By isolating the primary speech, this technology improves communication quality, especially in noisy environments.

- This approach enhances clarity, making conversations sound more natural and easing listener fatigue in both live and recorded audio settings

Literature Survey

Sl.No	Title	Methodology/ Algorithm	Merits	Demerits	Gaps
01	Adaptive Noise Cancellation for Voice Communication Systems Using Reconfigurable Filters	Butterworth and RLS/NLMS Filters Combination: Noise filtering is performed using Butterworth filters, followed by refinement using Recursive Least Squares (RLS) or Normalized Least Mean Squares (NLMS) algorithms to handle buzzing, static, and crackling noises	Improved Signal-to-Noise Ratio (SNR) and Mean Square Error (MSE) Reduction: Demonstrated through simulation, achieving better audio quality by efficiently suppressing various noise types Dynamic Adaptability: RLS and NLMS filters provide effective handling of dynamic and changing noise conditions	Higher Complexity and Computational Cost: Combining Butterworth and adaptive filters like RLS or NLMS can lead to increased complexity and computational requirements. Limited Tracking in Highly Variable Conditions: Some algorithms, such as RLS, may struggle to adapt quickly to rapid system changes	Scalability for Real-Time Applications: The methodology's performance under real-time processing conditions and high-speed applications may be limited. Lack of Comprehensive Testing for Diverse Environments: The system primarily focuses on buzzing, static, and crackling noise; further testing and optimization are needed for other noise scenarios

Literature Survey

Sl.No	Title	Methodology/ Algorithm	Merits	Demerits	Gaps
02	Real time background noise cancellation in end user device	adaptive Least Mean Squares (LMS) algorithm :The LMS algorithm adaptively filters out the noise by estimating it and subtracting it from the primary microphone's signal A modified form of LMS updates weight vectors using a correlation-based approach, allowing for real-time noise estimation and subtraction with minimized signal delay	Effective real-time background noise cancellation, providing clear communication even in noisy environments daptable to dynamic noise environments due to the adaptive nature of the algorithm, offering broad application in mobile devices and public communication systems	Performance highly dependent on hardware processing capabilities; slower hardware can introduce unacceptable delays Non-ideal matching between the two microphones can lead to imperfect noise cancellation, causing residual noise in the output signal	The need for more robust methods to handle varying environmental conditions and types of noise (e.g., colored noise vs. white noise) for greater accuracy Limited scalability to higher-order filters due to computational demands, potentially limiting effectiveness in complex environments

Literature Survey

Sl. No	Title	Methodology/ Algorithm	Merits	Demerits	Gaps
03.	A New Simple Adaptive Noise Cancellation Scheme Based On ALE and NLMS Filter	Adaptive Filtering: This technique uses adaptive filters, such as LMS (Least Mean Squares) Spectral Subtraction: Spectral subtraction removes noise by estimating the noise spectrum and subtracting it from the signal, helping isolate the primary voice. This method often leverages the frequency domain, which is effective in separating voice from noise and reverberation.	By canceling out echoes, these systems make the primary voice clearer, allowing users to communicate without distractions from background noise Adaptive filtering adjusts in real time, making it suitable for dynamic environments where noise and echo patterns constantly change, like busy offices or outdoor settings.	Spectral subtraction can sometimes lead to "musical noise" or residual artifacts, which degrade the quality of the output and may make the voice sound unnatural.	Limited Robustness in Highly Dynamic Environments Inadequate Performance in Low Signal-to-Noise Ratios (SNR)

Literature Survey

Sl. No	Title	Methodology/ Algorithm	Merits	Demerits	Gaps
04.	A Novel Background Noise Estimation in Adverse Environments	Adaptive Filtering (e.g., LMS Algorithm) Uses real-time adjustments to minimize echo by learning from incoming signals. Deep Learning-Based Noise Suppression Leverages neural networks to distinguish between voice signals and background noise, offering high precision.	Quick convergence in moderate noise environments. Suitable for real-time applications with lower computational demands. High accuracy in differentiating speech from complex background noises. Improves overall communication clarity significantly in challenging scenarios.	Struggles with non-linear echoes or sudden changes in background noise. Performance degrades in highly dynamic acoustic environments. Requires significant computational resources and hardware capabilities.	Limited effectiveness in handling multiple noise sources or overlapping conversations. Generalization challenges in unseen environments or with diverse accents.

Literature Survey

Sl.No	Title	Methodology/ Algorithm	Merits	Demerits	Gaps
05.	Reconfigurable Filter Design for Multiband Noise Cancellation.	Butterworth filters, Recursive Least Squares (RLS) algorithm, and Normalized Least Mean Squares (NLMS) algorithm to address buzzing, static, and crackling noises in audio signals.	• . Improved Performance Parameters. • Adaptive Noise Handling. • Efficient Noise Cancellation. • Versatility.	• Computational Complexity. • Dependency on Filter Tuning • Limited Noise Types. • Thresholding Limitations.	• Lack of Real-Time Validation. • Hardware Implementation Missing. • Broader Noise Environments

Literature Survey

Sl.No	Title	Methodology/ Algorithm	Merits	Demerits	Gaps
06.	Enhancing the Quality of Voice Communications by Acoustic Noise Cancellation (ANC) using a Low cost Adaptive Algorithm based Fast Fourier Transform (FFT) and circular convolution.	FFT and circular convolution.	Low Computation-al Complexity, Improved Signal-to Noise Ratio (SNR).	Limited Testing Scenarios,SNR Variability.	Limited Algorith-m Comparison,Ap plication to Non-Stationary Noise.

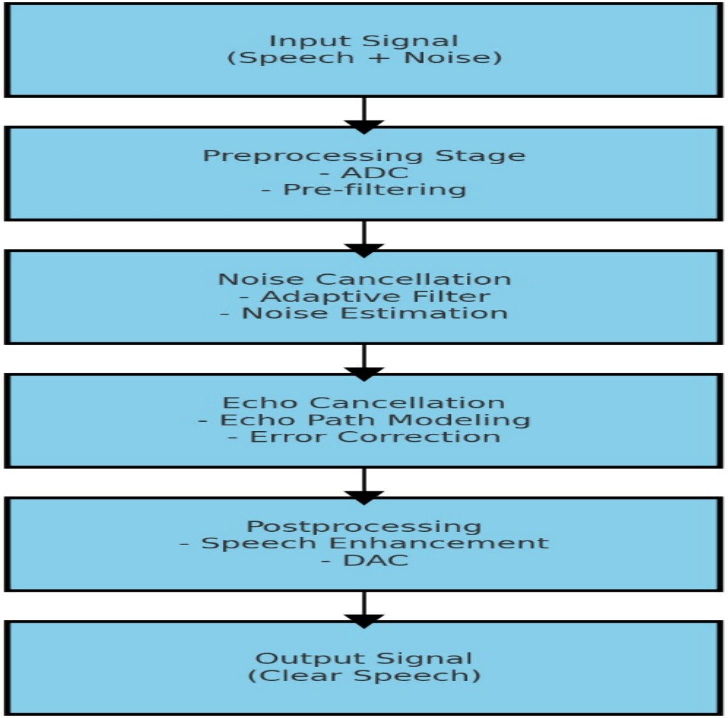
Adaptive Filtering Techniques:-

Least Mean Square (LMS) Algorithm: This algorithm is one of the simplest and most commonly used adaptive filtering techniques for AEC.

Normalized LMS (NLMS) Algorithm: A variation of LMS, which dynamically adjusts the step size, providing better stability and performance.

Functional Block Diagram

Block Diagram for Echo Cancellation System (Normal Layout)



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.

Methodology

•Reading Input Signal:
The input signal, containing both speech and noise, is captured using a microphone or recording device. It is then digitized via an Analog-to-Digital Converter (ADC) for further processing.

•Preprocessing:
High-frequency noise and irrelevant components are removed through pre-filtering. The resulting signal is optimized for adaptive noise cancellation.

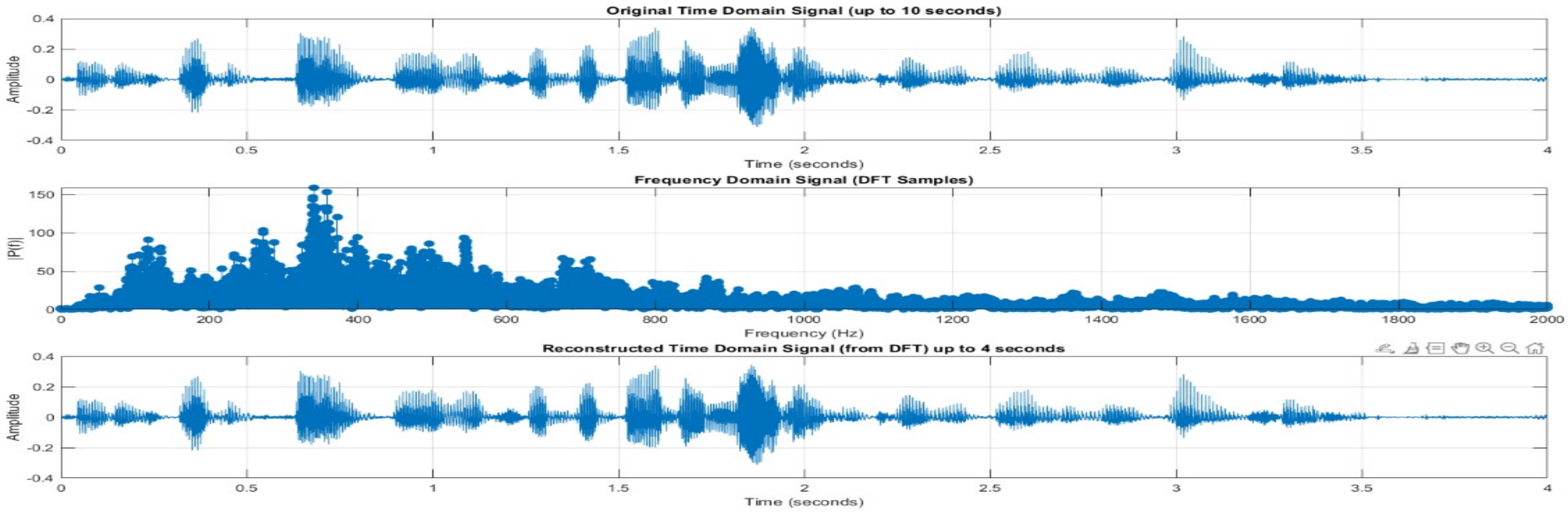
•Noise Cancellation:
An adaptive filter isolates noise by dynamically adjusting its coefficients in real-time. Noise estimation algorithms enhance this process by identifying and eliminating background noise effectively.

•**Echo Cancellation:**
Echo path modeling estimates and suppresses echo components within the signal. Error correction methods minimize residual distortion, ensuring clean echo removal.

•**Postprocessing:**
Speech enhancement techniques improve signal clarity and intelligibility. The enhanced signal is then converted back to analog using a Digital-to-Analog Converter (DAC).

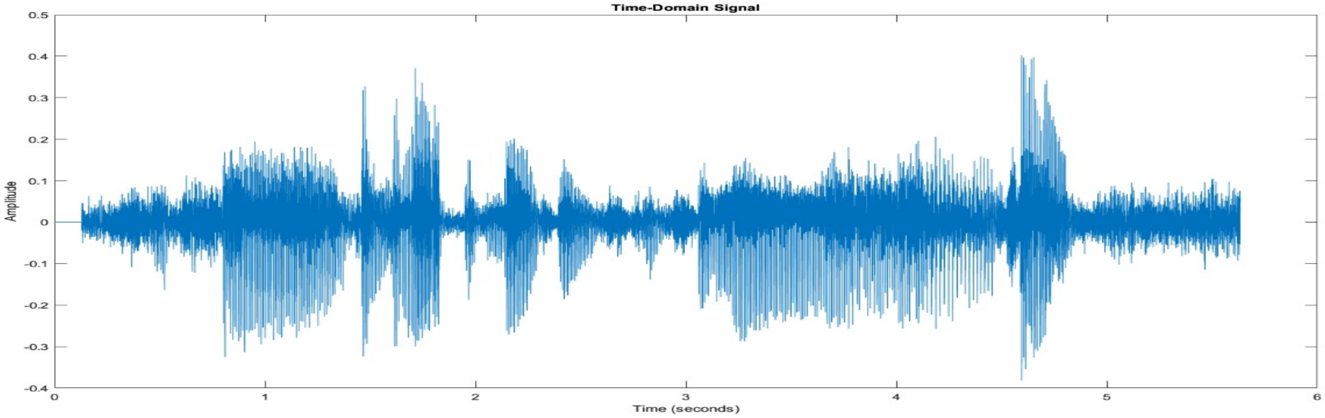
•**Output Signal:**
The final output is a clear, noise- and echo-free speech signal, ready for communication or other applications.

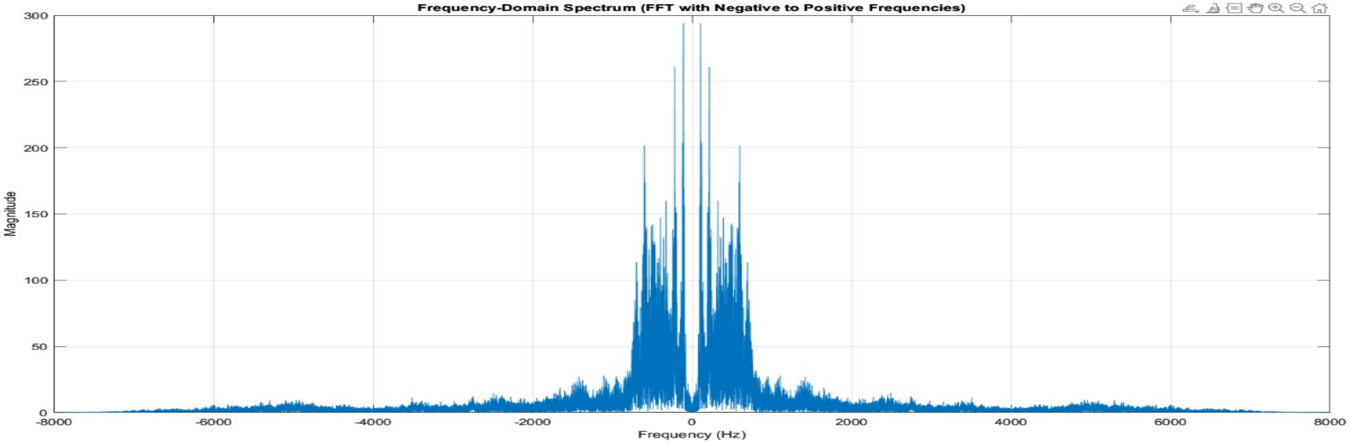
Result of DFT



- **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).

Result of FFT



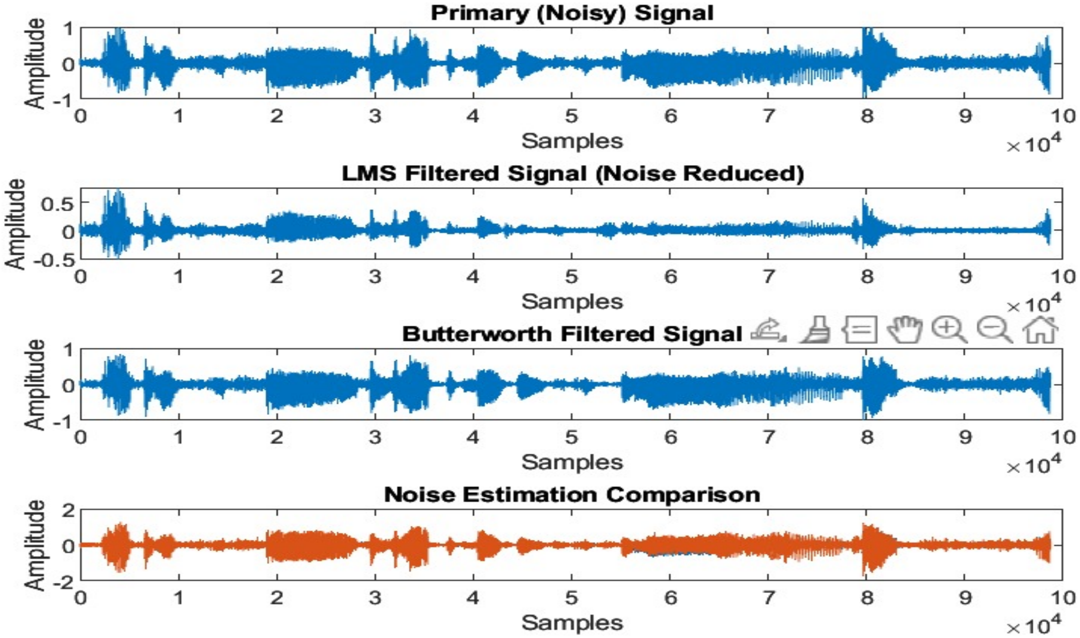


Num Samples=44100

Computational Time for FFT: 0.012345 seconds.

- **Frequency Range**: −22050 Hz-22050
- **Amplitude Range**: −0.5 to 0.4980.

LMS Algorithm Filter :



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
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Filter	MSE	SNR (dB)
"LMS"	0.019664	0.94758
"Butterworth"	0.045061	-2.6538

Additional Work :

We analyzed a signal containing both voice and background instrumental music, applying an LMS filter with consistent filter parameters to evaluate its performance and draw inferences on its effectiveness across different types of noisy signals.

Primary Noisy Signal :

LMS filter signal

Thank you