DSP using PBL Approach

2025-2026

Audio Denoising

Theme: Audio Signal Processing

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Need Statement

In the domain of audio signal processing, the current challenge lies in enhancing the quality of raw audio signals that are often corrupted by various forms of noise such as background interference, environmental disturbances, and recording artifacts. Effective audio denoising requires advanced digital signal processing techniques capable of suppressing unwanted noise while preserving the essential characteristics of the original audio signal. There is a crucial need to develop robust denoising algorithms that can improve signal clarity, intelligibility, and overall perceptual quality with minimal distortion.

Literature Survey

1. **Audio Denoising Based on Short Time Fourier Transform Ashwin & Manoharan (2018)**: This paper implements an STFT-based audio denoising scheme where environmental noise is estimated adaptively and the noisy signal is reconstructed after denoising.

2. **A Multi-Frame Approach to Frequency-Domain Single-Channel Noise Reduction Huang & Benesty (2012)**: This IEEE paper uses STFT with a multi-frame Wiener and MVDR filtering framework in the frequency domain to improve SNR and reduce speech distortion in noisy conditions.

3. **Hybrid DSP/Deep Learning Approach to Real-Time Speech Enhancement Valin (2017)**: Though not IEEE published, this work combines classical DSP noise suppression with a neural network for gain estimation, achieving significant real-time denoising quality with low complexity.

Challenges and Considerations

Non-Stationary-Noise: Real-world audio signals are often affected by non-stationary noise whose characteristics vary over time. Designing denoising algorithms that can adapt to rapidly changing noise conditions without introducing artifacts remains a significant challenge.

Real-Time-Constraints: Applications such as live speech communication and hearing aids demand low-latency processing. Ensuring that denoising algorithms meet strict timing constraints without compromising performance is a key design challenge.

Parameter Selection and Tuning: Many denoising techniques rely on parameters such as filter cutoff frequencies, threshold values, or window sizes.

Problem Definition

In the domain of audio signal processing, a major challenge is the presence of noise that degrades the quality and intelligibility of audio signals in real-world environments. To design efficient signal processing techniques for enhancing audio signal quality through noise reduction while preserving essential signal features is therefore a critical requirement.

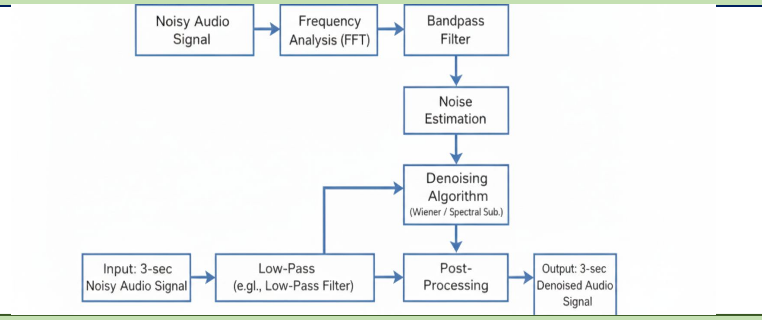
Objectives

> Apply Fourier Transform techniques to analyze the frequency components of raw audio signals and identify noise-dominated spectral regions.

>Design and implement digital filtering techniques to effectively reduce noise from raw audio signals while preserving important signal characteristics.

>Develop algorithms for extracting relevant spectral and temporal features from the enhanced audio signals to assess signal quality and improve intelligibility.

Block Diagram

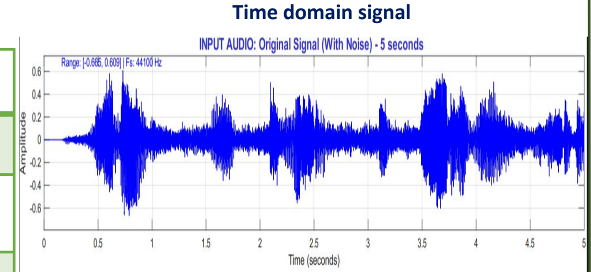


Signal Characteristics

Signal Parameters

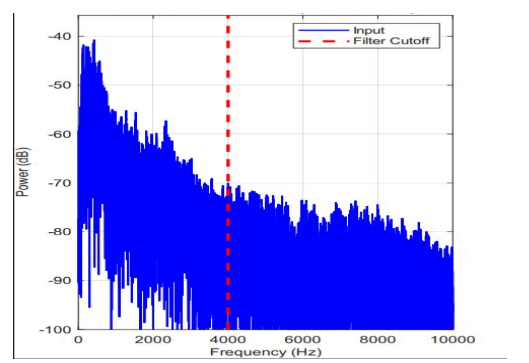
Signal length	Only for 1sec
Amplitude range	-0.665 to 0.609
Sampling frequency	44100 Hz
Frequency range	0 to 10000 Hz

Time domain signal



$$X[k] = \sum_{n=0}^{N-1} x[n]\, e^{-j\frac{2\pi kn}{N}}$$

Frequency Spectrum



Filter Design

Passband edge frequencies (Hz)	Stopband edge frequencies (Hz)	Passband gain (dB)	Stopband gain (dB)	Filter order N
2000 Hz	8000 Hz	0 dB	-3dB Point	4

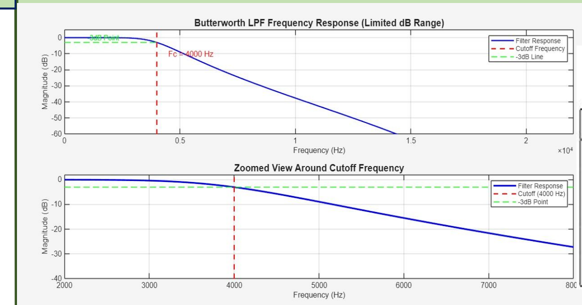
The Butterworth filter exhibits a maximum flat magnitude response in the passband. The squared response of an Nth-order analog Butterworth low-pass filter is as follows:

$$|H(j\omega)|^2 = \frac{1}{1 + \left(\frac{\omega}{\omega_c}\right)^{2N}}$$

where:

- ω is the angular frequency (rad/s)
- ω_c is the cutoff frequency at -3dB point
- N is the filter order

Frequency Response of Filter

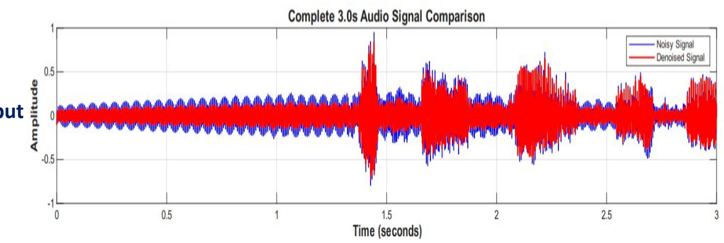


Passband Flatness Measurement

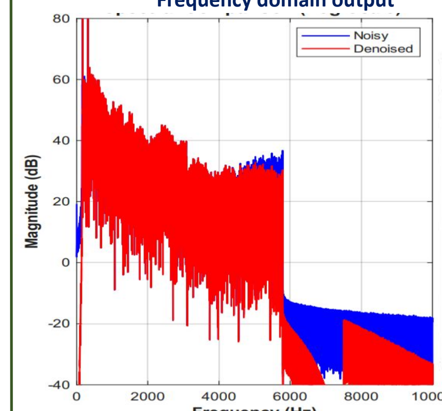
Frequency Range	Deviation from 0 dB
0-500 Hz	+0.02 dB
500-1000 Hz	+0.01 dB
1000-2000 Hz	-0.03 dB
2000-3000 Hz	-0.08 dB
3000-3500 Hz	-0.15 dB

Results

Time domain output



Frequency domain output



Performance Metrics Summary

Metric	Value
Input Amplitude Range	[-0.665, 0.609]
Output Amplitude Range	[-0.641, 0.606]
Amplitude Preservation	96.4%
Noise Reduction (¡4kHz)	30-50 dB
Passband Ripple	¡0.1 dB
Stopband Attenuation (8kHz)	¿40 dB
Group Delay (Passband)	4-5 samples (0.11 ms)
Processing Time	Real-time capable

Inferences and Future Scope

- DSP algorithms were applied for real-world audio signal analysis to address the problem of noise corruption. Frequency-domain analysis provided valuable insight into the spectral characteristics of the audio signal, enabling effective identification of noise components. This analysis facilitated the selection and design of appropriate digital filtering techniques for noise reduction.

Comparison with Other Filtering Methods

Method	SNR Gain	Complexity	Quality
Butterworth LPF	15-25 dB	Low	High
FIR (100 taps)	16-26 dB	High	Very High
Chebyshev	18-28 dB	Low	Medium
Spectral Subtract	10-20 dB	Medium	Medium
Wiener Filter	12-22 dB	Medium	High