Audio Denoising using Spectral Gating

Cluster Innovation Centre
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Semester Long Project submitted for the paper of "**Semester Long Innovation Project**"

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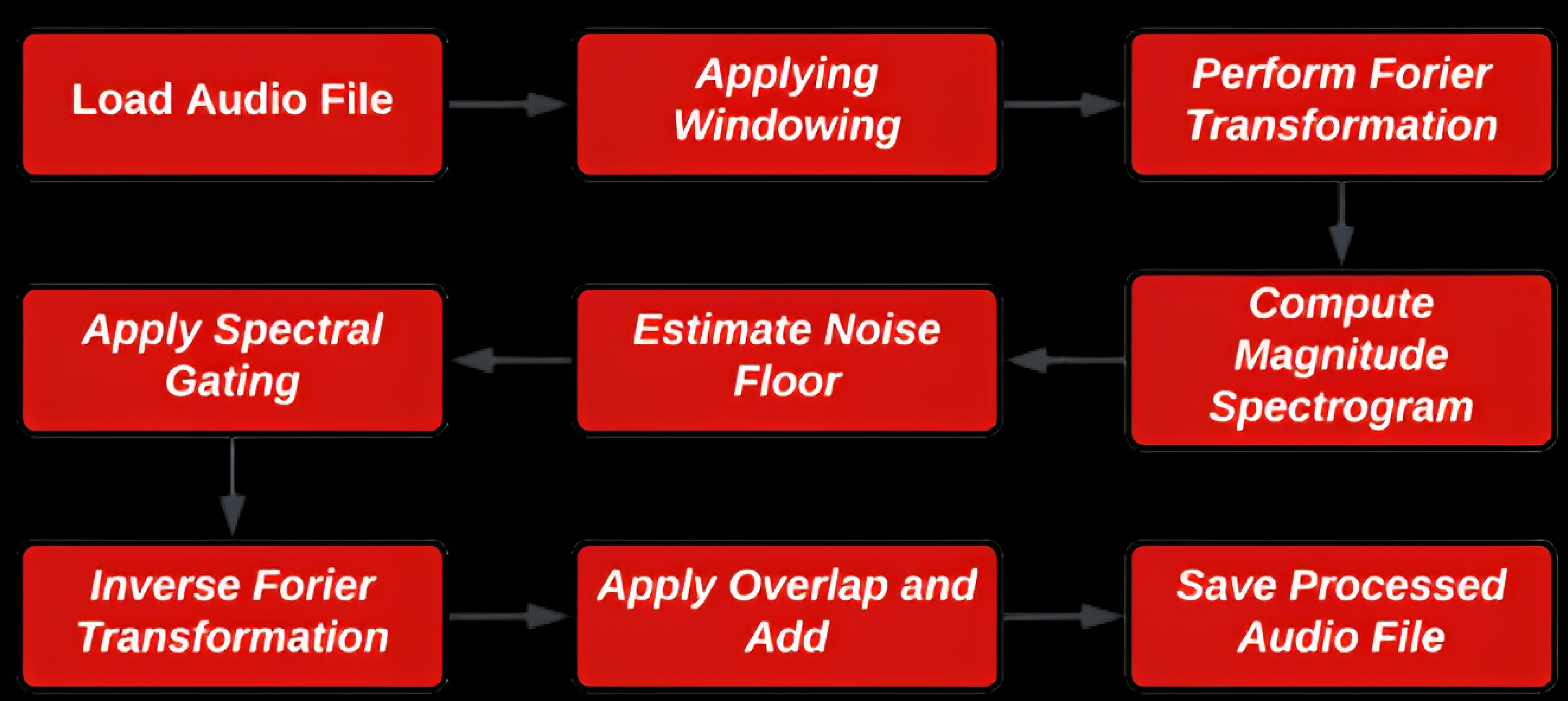
Abstract

In this project, we developed an algorithm for noise reduction in audio files using Fourier Transform techniques. The algorithm involves a series of pre-processing steps including framing, windowing, and Fast Fourier Transform (FFT) to convert the audio signal into the frequency domain. Then, a threshold-based approach is applied to suppress the noise components, followed by inverse FFT to obtain the denoised audio signal. The performance of the algorithm is evaluated using Signal-to-Noise Ratio (SNR) as the quantitative metric.

Methodology

The Spectral Gating and FFT algorithm are implemented in Python using several functions and classes. The main goal of the algorithm is to reduce noise from an input signal by applying spectral gating techniques. The key steps and processes involved in the algorithm are as follows:

• Smoothing Filter Generation
• Reading and Padding Input Signal
• Fast Fourier Transform (FFT)
• Filtering the Input Signal
• Iterating over Chunks• Noise Reduction
• Non-Stationary and Stationary Noise Reduction



Dataset

- Our Own Generated Audio Files
  We created our own set of audio files for testing the algorithm. These files were generated using a custom script that simulated various environmental conditions, such as different levels of background noise, reverberation, and interference.
- Clean and Noisy Parallel Speech dataset
  We also used an online published dataset for evaluating the performance of our algorithm. This dataset, provided by the University of Edinburgh, consists of pairs of clean speech signals and their corresponding noisy versions.

References

1. Kumar, E. S., Surya, K. J., Varma, K. Y., Akash, A., & Reddy, K. N. (2023). Noise Reduction in Audio File Using Spectral Gatting and FFT by Python Modules.
2. Valentini-Botinhao, Cassia. (2016). Noisy speech database for training speech enhancement algorithms and TTS models, [dataset]. University of Edinburgh. School of Informatics. Centre for Speech Technology Research (CSTR).
3. Digital processing of speech signals. Prentice-Hall, Inc. Ephraim, Y., & Malah, D. (1984).

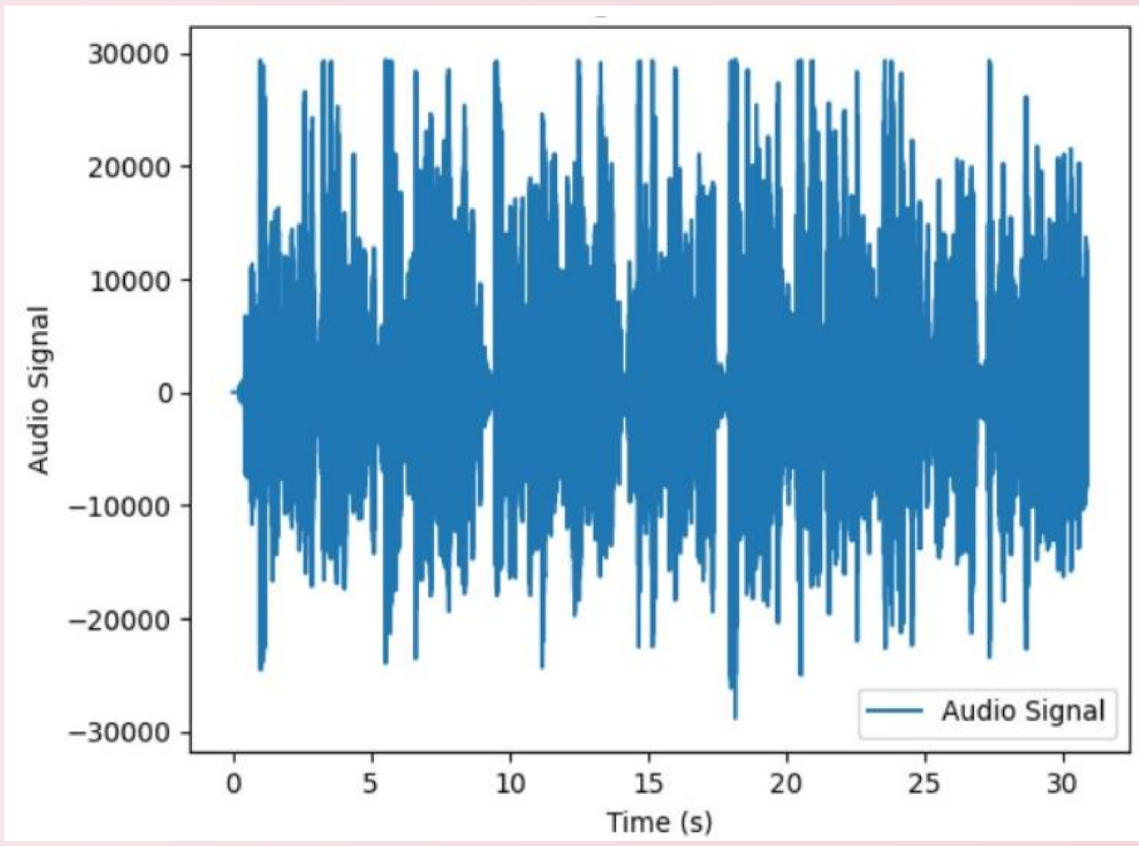
Previous Algorithms

- The **LMS filter** operates in the time domain and adjusts its filter coefficients iteratively to minimize the error between the filtered signal and the desired signal.
- The **Kalman filter** estimates a system's state using a model of its dynamics and noisy observations. It works in the time domain, making it useful for audio denoising, where signal and noise characteristics can vary over time. The filter can handle nonlinear and time-varying systems

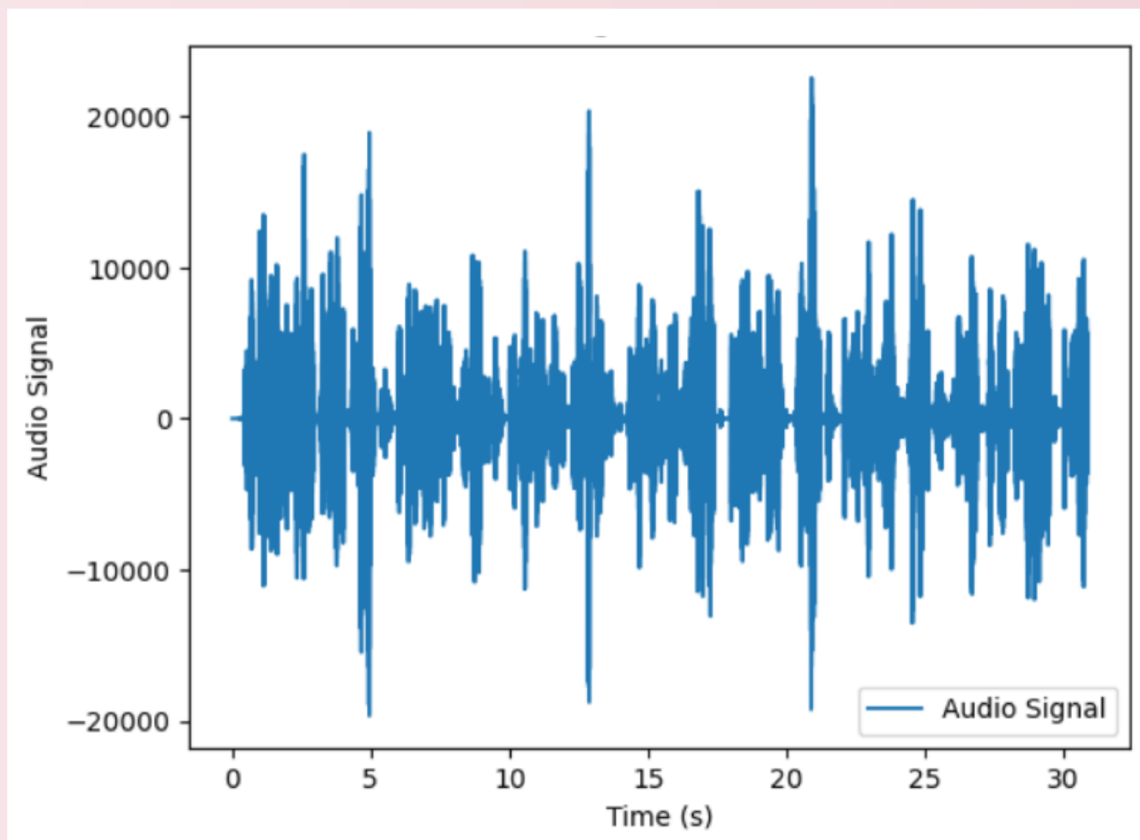
Results

The performance of our algorithm was evaluated through experiments on our own generated audio files and a publicly available dataset from the University of Edinburgh. Quantitative evaluation was conducted using Signal-to-Noise Ratio (SNR) as the chosen metric to assess the effectiveness of our algorithm in reducing noise in audio files.

NOISE REDUCTION ALGOTIHM	SNR VALUES
LMS Filter	2
Kalman Filter	6
Spectral Gatting	14
Improved Spectral Gatting	15.48



Noisy Audio



Reduced Audio

Conclusion

Our project aimed at developing an algorithm for noise reduction in audio files Our algorithm showed significant improvements in reducing noise and enhancing the quality of audio signals, as evidenced by the quantitative evaluation using SNR values and visual comparison of altitude vs. time graphs.