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Noise Reduction in Audio File Using Spectral Gatting and FFT by Python Modules

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Abstract. Future telecommunication will more and more rely on proper noise reduction techniques with less convolution. Speech communication, speech augmentation, and speech transmission have quite a great challenge regarding noise in audio transmissions. Therefore, the most effective noise cancellation technique should be developed such that it is coherent and active noise cancellation and rapid transmission rate, and a significant proportion of the noise is suppressed. Fast Fourier Transmission (FTT) and Spectral gating techniques are a handful of algorithms that have been often used during the reduction of noise. These algorithms have indeed been put into effect to suppress any noise, especially noise from fans in the room, horn sounds and dog barks on the streets. The intended audio is generated to use the recorded audio signal, which will then be handled using the aforementioned techniques to provide the clearest, least-noisy audio possible. To determine whether the system the method is and how well the noise in the audio signal is being reduced, compare the SNR levels of the raw audio file and the noise-reduced audio signal. For variable SNR in a very noisy environment, noise cancellation and noise reduction techniques of an audio signal are essential. When compared to the many models and research articles currently in existence, this one has been concluded that this methodology will be much more productive.

Keywords. Telecommunication, Spectral Gating, Fast Fourier Transformation, Non-stationary noise, Amplitude, Threshold value.

1. Introduction

In contrast to the traditional methods to get rid of the unwanted/unnecessary sound in audio which meant to decrease a large range of noise origins like huge background noise, glitch, echo, and breeze. In process of degrading quality and intelligibility of speech in audio background noise plays a key denominator role. We proposed various ML models like spectral gating for enhancing creative sound design and Fast Fourier Transformation (FFT) to seek out the set of cycle speeds, amplitudes, and phases to match signals. To beat this traditional method, we proposed a mathematical approach to decreasing background noise[1-2]. Experimental results demonstrate that the proposed framework can do significant improvements in the mathematical approach[3-5]. It's also interesting to see that the proposed approach reduces non-stationary noise (continuously updates the estimated noise threshold over time). Also, we identify the only efficient method to scale back noise which supplies better results with efficient accuracy than existing methods[6-7].

In this modern era, communication plays a vital role. For better communication, we need both ends of cooperation for a lethal conversation. Nowadays, we are much accustomed to podcasts on platforms like Spotify, YouTube, etc for everybody's relaxation. For these kinds, we want an honest and prime quality of audio with no background noises to enjoy it. This paper presents a spectral gating and FFT algorithm based on noise reduction. Which changes according to time.To reduce the noise in the audio to understand the context clearly with any disturbances from external sources. Communication is important to exchange any information, and the information need to be exact to minimize the errors[8-11], so it is in our best interest to reduce these kind of situations and improve the commutation standards thus reducing the errors automatically.

As per our research and knowledge, Neural network approaches for audio signals were in the initial stage, and it does require a couple of years and high-end computational speed to reinforce it. Hence, it was in the initial stage we are enthusiastic to develop a user-friendly mathematical approach that is cost-friendly and numerable with better results[8]. Here we processed some implementation parts by creating a website to convert noisy audio files to a great extent noise cancellation files. We will be working with spectral gating and Fast Fourier transformation methods using noise reduce library in python. Justification to the proposed method is the SNR ratio is higher, the better the signal quality. Spectral Gatting gives the highest signal-to-noise ratio[Table1][2]. compared to among existence techniques for our experiment. Therefore, it is an effective method for noise reduction in audio.

The paper is organized as follows: Section II depicts the Literature Survey made for understanding various factors of the noise cancellation techniques. Section III defines the proposed methodology system. Section IV explains the implementation and results. Section V describes the conclusions and future work that can be done.

2. Sections in Paper

2.1. Proposed Methodology

- Spectral Gatting
- Fast Fourier Transformation
- Spectral gating is an uncommon filter effect that is used as a tool for creating sound design. It divides the incoming signal into two frequency ranges one above and another below a center frequency and bandwidth parameters. The spectral gating algorithm uses a noise gate to set the above-mentioned threshold gates. The signal which crosses above the lower gate threshold will be recorded and so the lower gate is also referred to as the open gate. If the signal crosses the specified upper threshold the gate closes and the signal stops recording, so the upper gate is referred to as a closed gate[5]. These gates are helpful to take the signal between the two gates and the signal below or above is taken as noise. So only the required or noise-reduced audio will be produced as output.
- There are two types of signals: stationary and non-stationary. Stationary signals have the time period as constant, that is the signal doesn't change concerning time[9]. Non-stationary signals change concerning time, human voices are a form of non-stationary signals[10]. So the open and close gate threshold needs to be

changed over time. This threshold is set by calculating the frequency of the signal over time[11].

- Noise reduction algorithm in python that diminishes the Speech, bioacoustic, and physiological signals are all examples of time-domain signals that comprise noise. It is centered on a technique known as "spectral gating" that takes the shape of a Noise Gate[8]. It operates by monitoring the frequency bands of a signal's spectrogram and imposing a noise threshold value (or gate) to each band. The frequency-varying threshold is used to process a mask that gates noise below it[7].
- Two inputs are required by the algorithm:
- The stereotypical noise of an audio clip makes up a noise audio clip.
- It is necessary to remove the noise from a signal audio clip in addition to the signal.
- The fast Fourier transform (FFT) is a computationally faster way to calculate the DFT and has good time complexity. Fast Fourier transform (FFT) is the foremost tool and is rapidly used in signal processing. FFT results of each frame data. Zoom FFT analysis can bring forth high res in the frequency domain and inspect vibration signals of a certain frequency band explicitly.
- FFT-based feature generation hinges on the decomposition of the composite signals to smaller transforms. The decomposed signals are consolidated to find the resulting transform signal. Low frequencies are eliminated from the procured signal using FFT and noises are unfastened after the application of the inverse FFT. As preceded before, FFTs are used in many DSP applications. It is therefore interesting to erupt an FFT processor as an extensively usable VLSI building block. As intended to be flexible so that the processor can be used in a variety of applications without considerable revamp, the performance in terms of computational throughput, word length, and transform length should be easily modifiable.
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- Sequence steps of the algorithm:
- An FFT is built and applied to the disorderly audio clip.
- Statistics are calculated using the noise's FFT (in terms of frequency)
- Relying on the noise statistics, a threshold is set.
- Over the signal, an FFT is calculated.
- A mask is generated by contrasting the signal's FFT to the threshold.
- The mask is homogenized over intensity and frequency domain with the aid of a filter.
- The mask is transposed and fed to the signal's FFT.
- DISADVANTAGES OF EXISTING SYSTEM :

- Can only reduce a specific single type of noise.
- Need for different algorithms to deal with different types of noises[8].
- Cannot reduce background noise to a high extent.
- The model may not reduce the clear audio of the second person in case both are near the device.

- The model may not work in case the audio signal frequency is too high or too low.

- ADVANTAGES OF OUR PROPOSED SYSTEM :
- It deals with any type of noise, so it is easy to reduce any type of noise with a single algorithm.
- This algorithm is used to deal with human speech as it can deal with Non-Stationary as well as Stationary signals.
- Reduce the noise in the algorithm to almost zero extents.
- The model effectively reduces the background noise caused by many factors to hear the context of the audio more clearly.
- The model helps the user to exchange information without any disturbance even in the crowded places like subways, by reducing the noise.

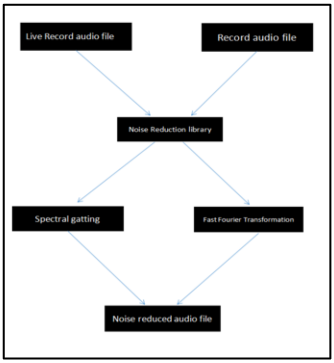

Figure 1. Block diagram

2.2. Implementation and Results

In every filter, the first plot shows the original audio, which contains the noise, the second plot shows the removal of stationary noise, and the third plot shows the removal of non-stationary noise.
The Noise Reduction System is designed to clean the audio file and deliver the de-noised audio file to the user. This reduction system provides an intuitive interface for enhancing the audio files with noise inclusion. This system contains a block to upload the audio file into the interface shown in Figure 3.
The foremost thing that is done after the upload of the audio file, is our system will analyze the audio file based on the proposed algorithms i.e, Spectral gating and FFT where the de-Noising takes place as shown in Figure.3, Figure.4, and Figure.5 respectively. The following Figureures represent the original file noise and stationary noise removal as well as the non-stationary noise removal.

Figure 2. The user interface to upload the audio file

2.3. Collation

Noise Reduction Algorithms were implemented and it was seen that the best SNR ratio was obtained for spectral gating and the Fast Fourier Transformation Method. SNR ratio is given as follows in Table1:
SNR = Power of the Signal/power of the noise

Noise Reeduction Algorithms	SNR Values
LMS filter	2
Kalman filter	6
Spectral Gatting	14

Table 1. SNR values

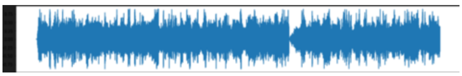

Figure 3. Original noise

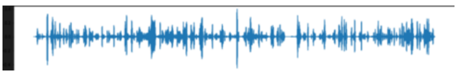

Figure 4. Stationary noise removal

Figure 5. Non-stationary noise removal

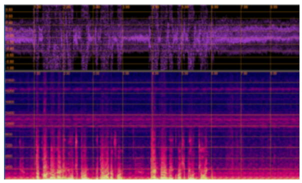

Figure 6. Strength of the audio file

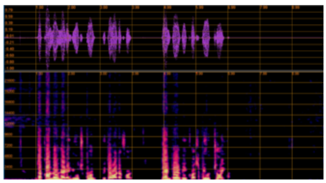

Figure 7. Noise removal strength

2.4. Conclusion and future work

In this study, we presented a method for reducing noise signals of different frequencies in audio recordings utilizing Spectral Gating and the Fast Fourier Transform (FFT). A clear audio clip that is presumably enjoyable to listen to is produced by the spectral gating technique. Just too much noise obscures the data due to the high signal-to-noise ratio. The proposed technique keeps the rest of the audio sample intact while suppressing the loud area. When compared to other conventional approaches, it also

outperforms them in terms of quality. As a result, the algorithm's effectiveness must be evaluated before removing noise from any audio stream. It is therefore a successful method of noise reduction. In the future, this methodology can be coupled with others to enhance productivity.
We can clearly hear the audio utilizing our proposed technique, which is more efficient in this research than that of other conventional methods. Listening without any unwanted noise in an audio file has a substantial impact on the audio community. We kept rising the SNR ratio in order to enhance the audio quality in comparison to the other main traditional protocols.

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