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Exploring Sound Generation and Processing in Mechatronic Systems through Digital Signal Processing: A MATLAB-Based Investigation

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ABSTRACT: Digital signal processing (DSP) encompasses various applications such as convolution, censoring, and filtering. These techniques find relevance in diverse fields, including computer games consoles, tone generation for mobile cellular devices, and theatrical performances. For computer engineers facing limited memory budgets in their devices, generating new sounds by convolving with existing sounds or noises becomes a significant challenge. The objective of this paper is to elucidate and demonstrate the generation of sounds through sound and recorded sound processing. These operations form the foundation of DSP, particularly digital filtering. Sound processing involves modifying prerecorded sound clips using different types of digital filters and processors. The entire process, including sound recording and processing, will be implemented and tested using MATLAB, the widely adopted technical computing software. This project aims to provide valuable archival material that complements the learning and expertise of computer engineering students and specialists.

KEYWORDS: Digital signal processing, Sound generation, Audio classification, Auditory scene analysis

INTRODUCTION

The human sense of hearing provides us rich information about our environment with respect to the locations and characteristics of sound-producing objects [1]. For example, we can assimilate the sounds of birds twittering outside the window and trace moving objects in the distance while following the lyrics of a song over the radio accompanied by multiple instruments. The human auditory system is capable of processing the complex sound mixtures reaching our ears and forming high-level abstractions of the environment through the analysis and grouping of measured sensory inputs [2]. This process, known as auditory scene analysis, has significant implications in applications such as speech recognition in noise, automatic music transcription, and multimedia data search and retrieval [3]. Machine realization of this functionality, involving sound source separation and classification, has become increasingly valuable in various domains [4].

Audio classification is an essential component of audiovisual data handling with applications in digital libraries, professional media production, education, entertainment, and surveillance [5]. Speech and speaker recognition have received substantial research attention in the audio retrieval field [6]. Moreover, the exponential growth of digital music archives on the internet has highlighted the need for advanced methods of nonlinear browsing and retrieval, particularly in the domain of music [7]. Effective audio indexing enhances the retrieval of audiovisual material, as listening to audio clips can facilitate navigation more efficiently than video scene analysis alone. Audio classification also plays a crucial role in audio compression systems by matching the compression method to the audio type, such as speech or music, thereby optimizing coding and transmission [8].

In this project, we aim to explore fundamental methods for audio signal processing, primarily focusing on the analysis of audio and recorded sounds using MATLAB [9]. We will discuss the general properties of audio signals and investigate techniques such as convolving noise signals with original recorded sounds [10].

Characteristics of Audio Signals

The range of human hearing spans from 20 Hz to 20 kHz, enabling us to perceive a broad spectrum of frequencies. Intensity, measured in decibels (dB), is another critical aspect of sound. The threshold of hearing is typically around 0 dB, while sound levels exceeding 120 dB can cause discomfort or even pain [11].

Audio signals can be classified into various categories based on their characteristics. Broadly, sounds can be classified as pleasant or unpleasant. However, these categories alone fail to capture the diverse nature of sound. Hence, we adopt natural English nouns, such as speech, music, noise, natural sound, and artificial sound, to create a multidimensional hierarchy of audio classification
(Figure 1) [12]. The characteristics of audio signals vary over time, and they can be represented in either the time domain or the frequency domain. Our perception of sound is associated with specific spectral or temporal properties of the sound. As a result, a comprehensive representation of sound

requires the incorporation of features from both the time domain and the frequency domain

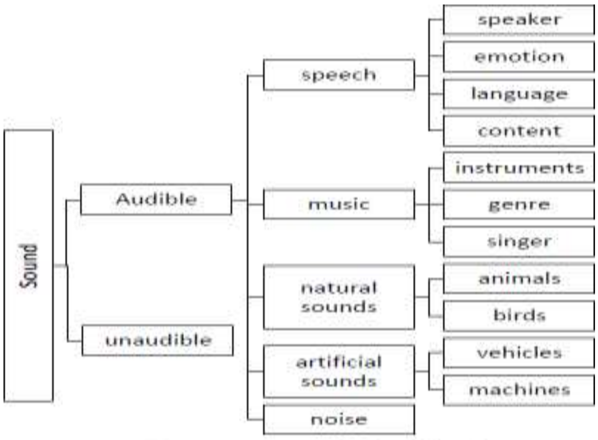
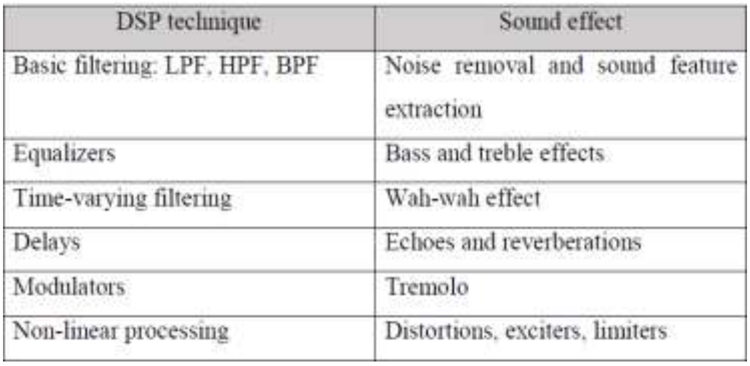


Figure 1. Sound Classification

Sound Processing

Sound manipulation involves altering and enhancing sounds to create more engaging effects for a variety of applications. Different digital signal processing (DSP) techniques can be categorized based on how they process sound. Table 1 provides an overview of various DSP techniques along with their corresponding sound effects.

Table 1: DSP techniques and the possible resulting sound effects



DSP technique	Sound effect
Basic filtering: LPF, HPF, BPF	Noise removal and sound feature extraction
Equalizers	Bass and treble effects
Time-varying filtering	Wah-wah effect
Delays	Echoes and reverberations
Modulators	Tremolo
Non-linear processing	Distortions, exciters, limiters

Basic filtering focuses on specific frequency bands by either accentuating them or attenuating others. This type of frequency-selective filtering is commonly used for noise removal or extracting specific features like pitch. However, compared to other DSP techniques, basic filtering has limitations in generating more captivating sound effects.

Equalizers, on the other hand, amplify or reduce specific frequency bands while keeping others unchanged. This allows for desired sound effects such as emphasizing bass or treble. Time-varying filters, such as variable-center-frequency band-pass filters, produce the popular "wah-wah" effect, often used in electric guitars to mimic a human-like vocal sound.

Delays introduce echoes and reverberations, which are commonly desired in concert halls and theaters. These effects are typically created using both IIR and FIR digital filters.

Modulators generate the tremolo effect by modulating the amplitude of the sound signal with a low-frequency sinusoidal signal. The carrier signal is the sound itself, while the baseband signal is the low-frequency modulation signal.

Non-linear processing techniques, such as limiters, compressors, and expanders, are employed to control high sound peaks or enhance low signal levels, resulting in more dynamic and vibrant sound characteristics.

The objectives of this paper include:

1. Recording a voice signal.
2. Applying the convolution technique.
3. Implementing signal censoring.
4. Filtering the voice signal to remove noise.

Convolution technique in mathematics

The convolution theorem states that, under suitable conditions, the Fourier transform of the convolution of two signals is equivalent to the pointwise multiplication of their Fourier transforms. This means that convolution in one domain (e.g., time domain) corresponds to pointwise multiplication in the other domain (e.g., frequency domain). When working with dynamic measurements and digital signals, the convolution integral plays a crucial role and is closely related to the Fourier transform. It exploits a mathematical property of the Fourier transform, making it a widely used and convenient tool in various fields and applications.

Understanding the convolution integral is essential before applying it. To facilitate comprehension, this document aims to provide a detailed explanation of the theorem along with relevant examples.

Basic definition Mathematically

convolution is defined as the integral of one function multiplied by another function, with respect to a specific variable (e.g., time, frequency, or space). It is important to note that convolution yields a new function with the same variable domain as the original functions. Convolution is typically represented by a circle with a cross in the center, as depicted in Figure 2.

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It is worth mentioning that convolution is commutative, meaning that the order of functions being convolved does not affect the result

$$f(x) \otimes g(x) = \int_{-\infty}^{\infty} f(x) \cdot g(u-x) = \int_{-\infty}^{\infty} g(x) \cdot f(u-x)$$

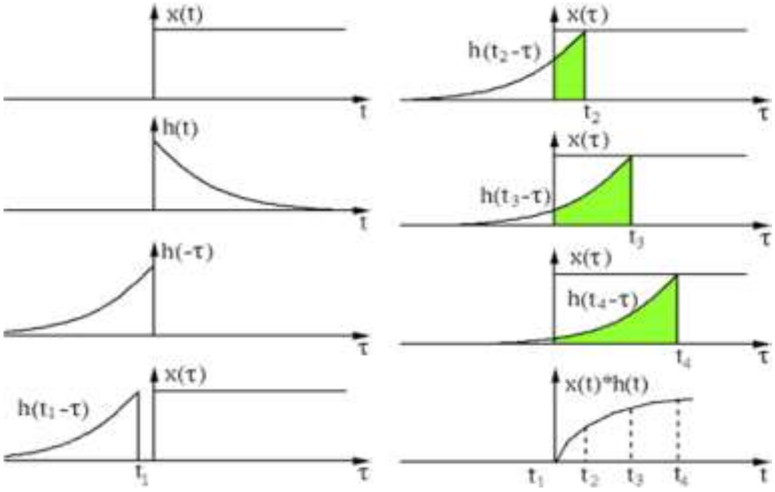

Figure 2. Illustration of the convolution process

RESULTS AND DISCUSSION

In this paper we relied on MATLAB to do the analysis and the processes as well as wedid not use any of the toolboxes embedded in MATLAB. We coded the whole project with the well-known and common codes and scripts**.** In addition to that, we used to different codes to process the signal we need.

Explanation of the code

First part we started with a simple code by recording one of the group member voice to say our basic information and recorder by the computer then convert it to wav form and let the computer to read it through Matlab. Then we plotted it in the form of one-dimensional form, After we plotted it, we convolve it with a function made by ourselves through Matlab to see the effects of the second function to the original function which is ouroriginal function and plotted it in the same form of the previous one. Then we did the censoring part which is to hide part of the signal like the name orany other part we can choose. In this part, we opt to choose to hide the name part and we plotted it.

Second part We created another Matlab code to record any of the project group voice to record the following statement “Do you want to record your voice?Yes or NO” This code will record that statement and listen (sound) to it then save it in a file , in case we will need after and we can plot the signal as well.

1. After we record that statement, we will have an if condition to check whether we want to record another statement or not. In this step a prompt message will appear toask the user “Please enter Yes or No” for the recording. IF the user answers YES
then another voice message will appear “**PLEASE RECORD YOUR VOICE FOR FIVE SECONDS AFTER THE TONE**”
2. After the voice message finished, atone sound will appear
3. After that, we will record a message from the user which is “**DO NOTLET THE STRESS GET THE BEST OF YOU**”
After we record this message (sound), a text message appears saying “**ENDOF THE RECORDING**”
4. The recorded file will be saved in a file in a form of *double-pression Array.*
5. IF the condition is NO, then we will upload a music from the computer and play.

THE RESULTS

The result of the first Matlab code that we recorder one of our groupmember's voice and personal information.

Figure 3 represents the original signal without any effects. We plotted the figure inone dimensional form

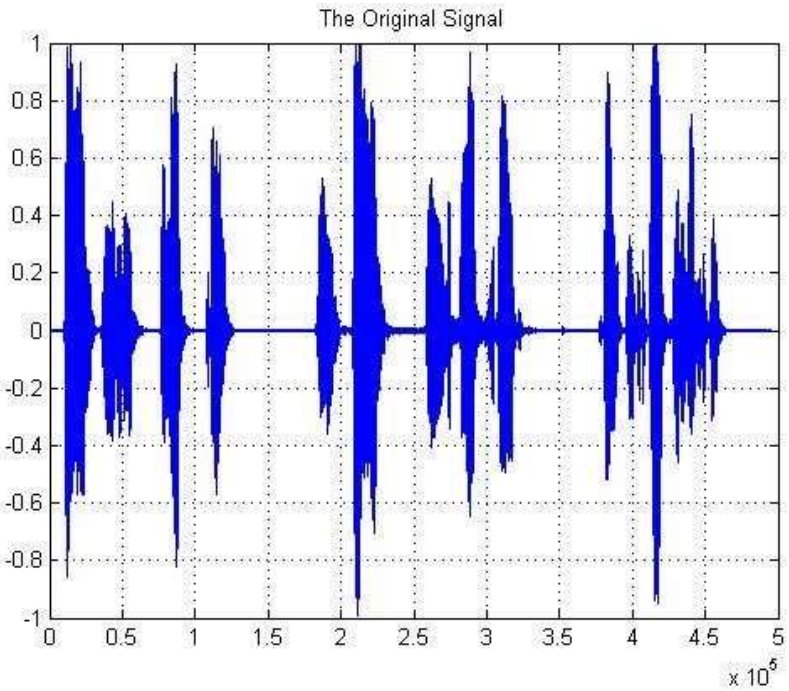

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Figure 3. the original voice signal

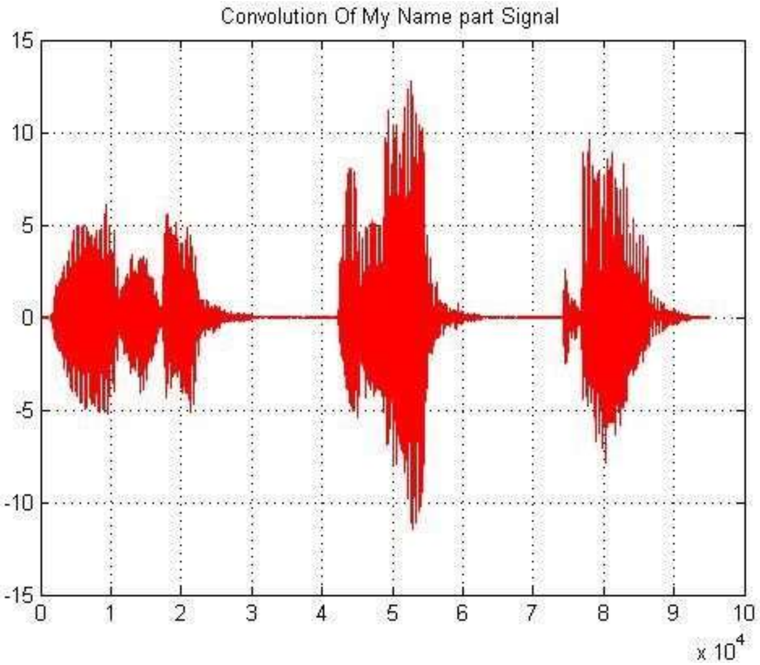

Figure 4. the convolution with the original signal

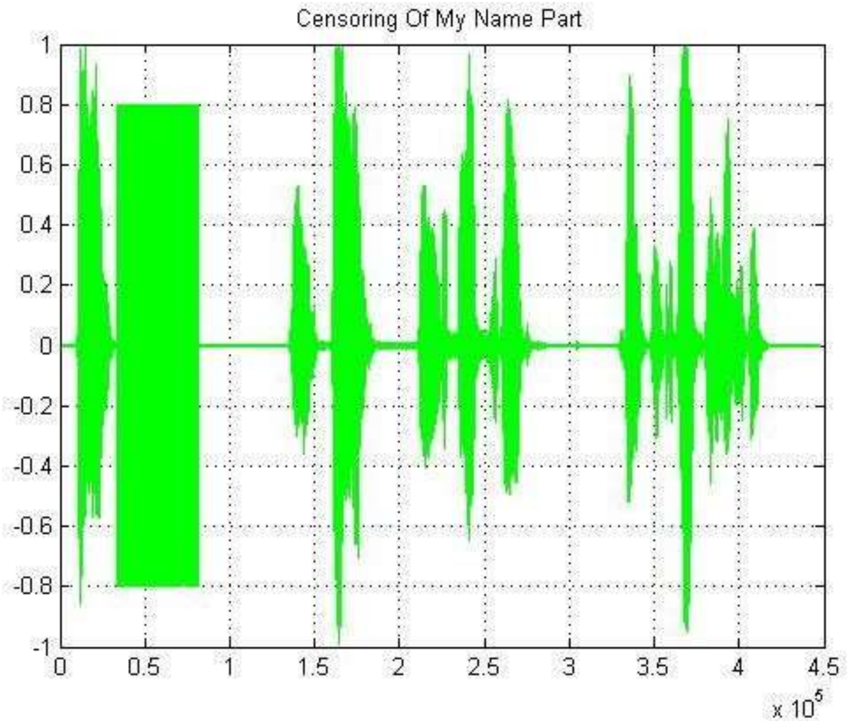

Figure 4. Censoring part of the signal

In this part we are going to display the results from step 2 in 4.2 section of thischapter as follow:
If We pressed **YES ,** In this step, we first converted the voice signal to a double-precession array asfollow:

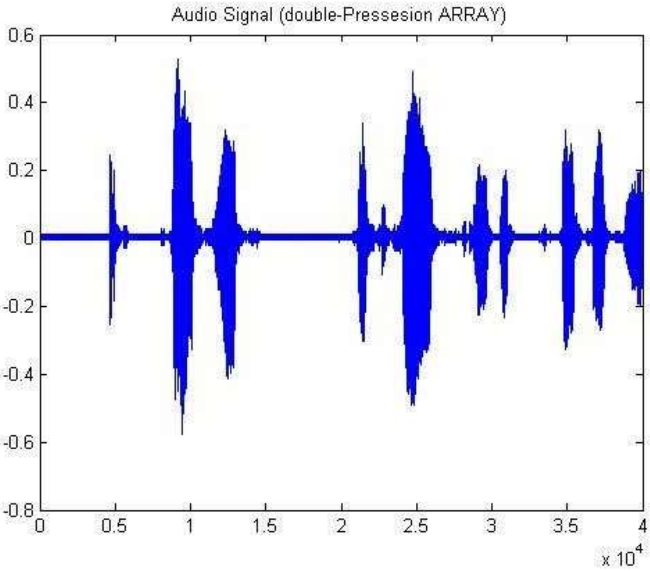

Figure 5. the original voice signal

After that, we added a white gaussian noise with two magnitudes 5Db and 10dB tothe pure music signal and can be seen the figure below:

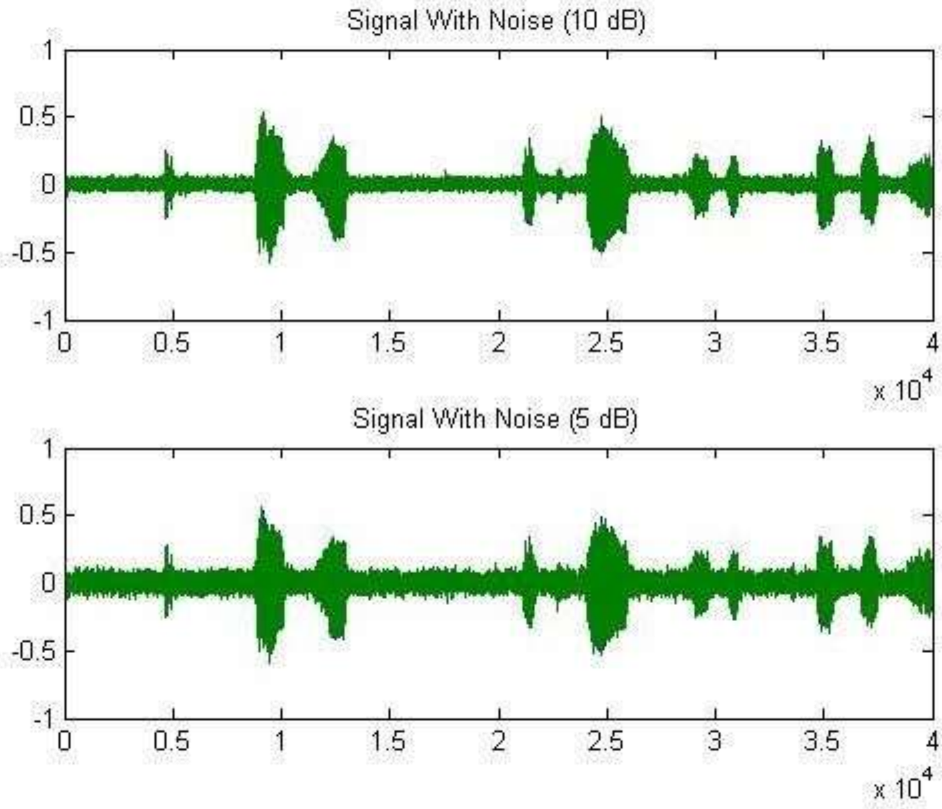

Figure 6. Signal with noise

We wanted to remove the added noise from the signal. To do so, we used IIRfilter to remove the noise from the original signal and we calculated the filtered noise and magnitude and the phase of the signal as shown in the figures below:

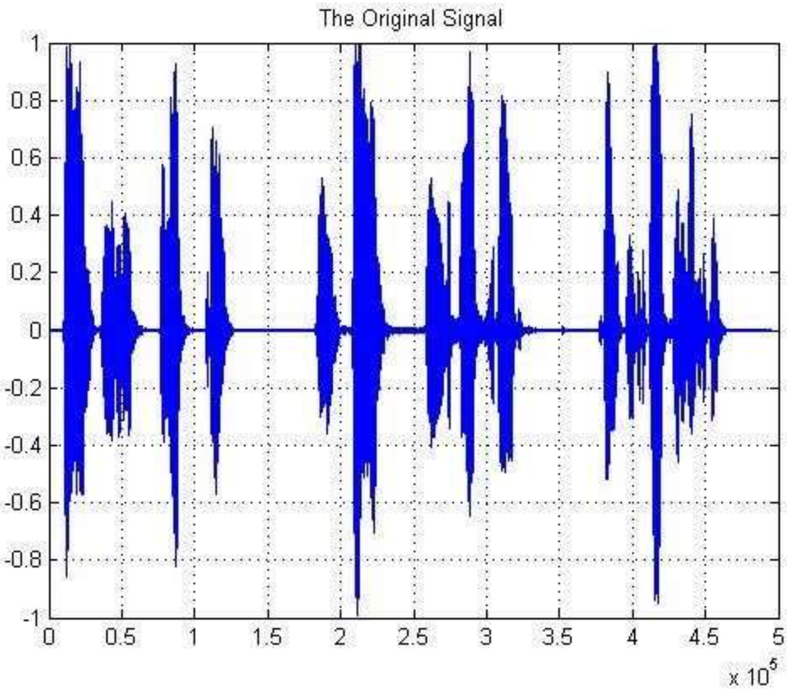

Figure 7. Filtered noise signal

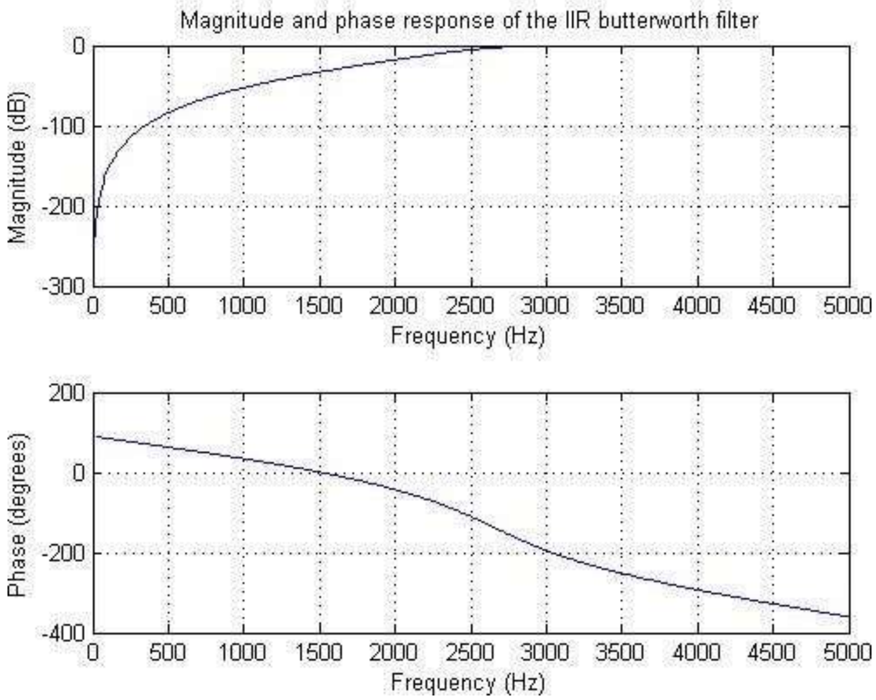

Figure 8. The magnitude and the phase of the voice signal

If We pressed **NO,** In this step, we first converted the music signal to one dimensional signal as follow:

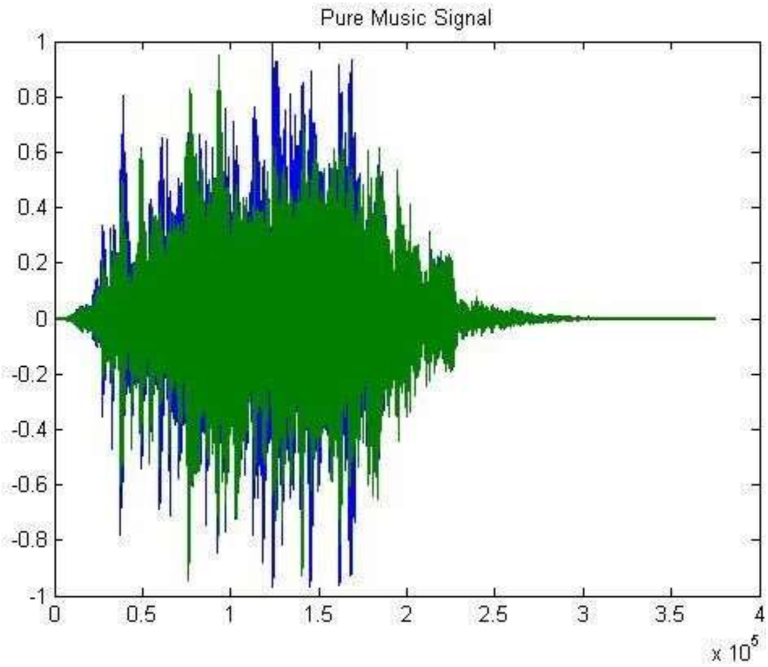

Figure 9. the music signal

After that, we added a white gaussian noise with two magnitudes 5Db and 10dB tothe pure music signal and can be seen the figure below:

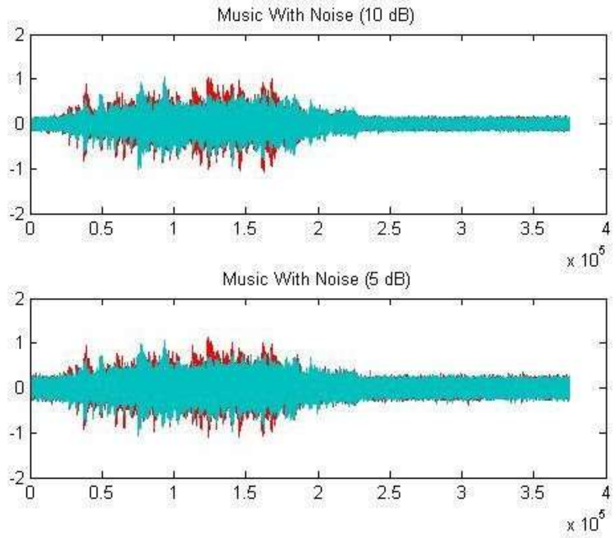

Figure 10. the music with the noise

We wanted to remove the added noise from the signal. To do so, we used IIRfilter to remove the noise from the original signal and we calculated the filtered noise and magnitude and the phase of the signal as shown in the figures below:

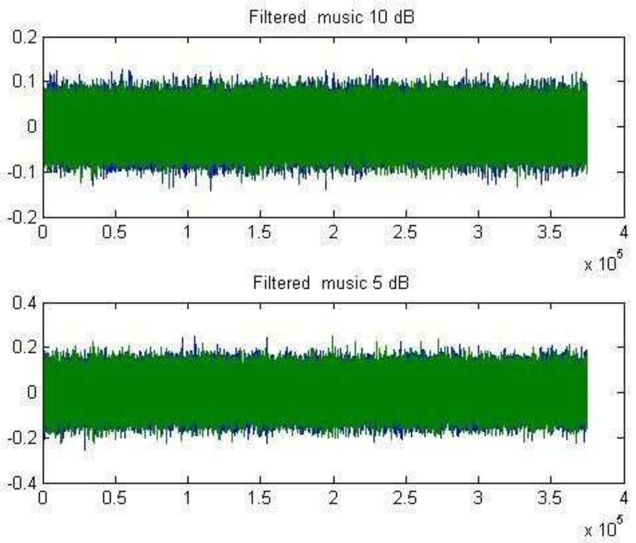

Figure 11. Filtered noise from the signals

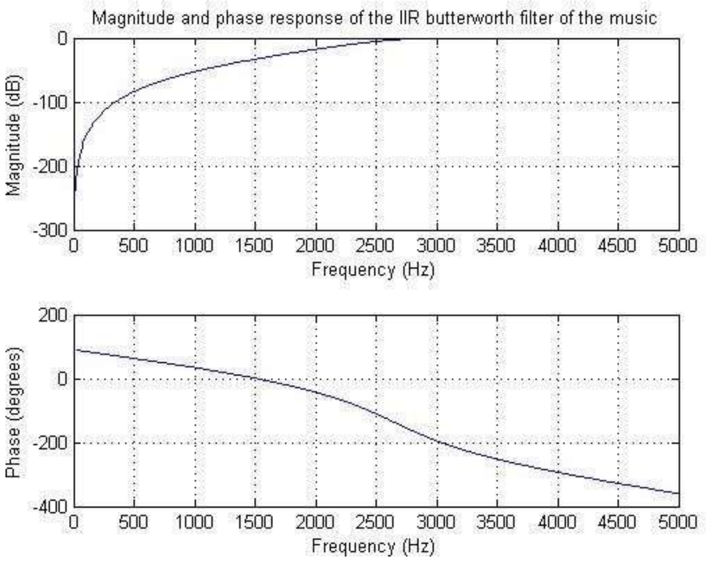

Figure 12. The magnitude and the phase of the signal

CONCLUSION

In our study, we have explored and simulated various sound processing techniques with the aim of generating sound suitable for diverse applications. Specifically, we have examined the impact of altering the dB magnitude of noise added to the signal and experimented with inputting pre-existing music from the computer. Furthermore, we successfully eliminated the noise component from both the signal and the music. Throughout these processes, digital filtering emerged as a pivotal element in sound processing. The findings of this research present exciting possibilities for future investigations focused on generating impact sounds using digital filters. For instance, by developing simple filters that selectively attenuate specific portions of the frequency spectrum, we can introduce variations in impact sounds that go beyond the capabilities of modal synthesis alone. It is important to note that digital filtering should be regarded as a complement to modal synthesis rather than a replacement for it.

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