# Removing Silence and Noise using Audio Framing

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**Abstract -** *The development in electronic music has upheld the advancement of sound testing frameworks. Along these lines, for any sound examining framework the way toward making various examples utilizing diverse time spaces is the most significant advance. In this paper, we have built up a straightforward calculation for making tests of the first sound record. This calculation is actualized just by utilizing the recipe for changing over sign into a persistent sign expelling all the clear edges and clamor. All the reproduction tests were done utilizing MATLAB where the technique delivered moderately great outcomes. This paper gives a point by point presentation of recorded sound examining, balance and impact age.*

***Key Words***: **Audio, MATLAB, Noise Reduction, Digital Signal, Audio Files.**

## 1. INTRODUCTION

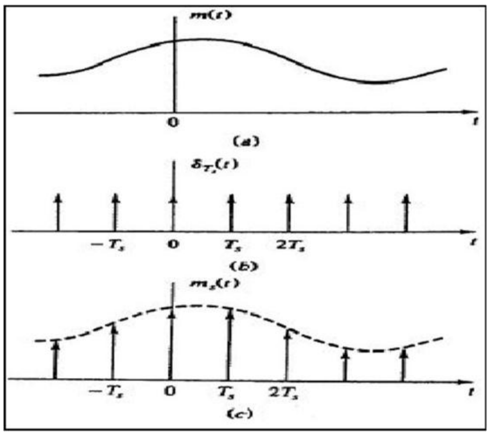
Sound inspecting has turned out to be one of the most significant thing in the present music industry. A similar sound record can be utilized to make various examples of sign which are discrete in time. This encourages numerous music makers to utilize other individuals' work and change it with the assistance of inspecting method and produce music which sounds totally unique however yet is the equivalent without being gotten for copyright infringement. Impact age is the following enormous thing which help in adding cool sound impacts to music and produce various varieties of a similar sound.

### 1.1 PROJECT OVERVIEW

The starting point for audio sampling consists of analog audio data arising from input wave file which contains music audio in raw form. This audio data, which are supposed to be available as standard audio file, contains the continuous audio signal which is used to create samples that are of discrete in time form. Pre-Processing of Speech signal is significant in the applications where quietness or foundation commotion is totally bothersome. Applications like Speech and Speaker Recognition needs proficient component extraction methods from discourse signal where the vast majority of the voiced part contains Speech or Speaker explicit attributes.

### 1.2 LITERATURE SURVEY

A sample rate conversion system and its method uses a digital signal processor (DSP) and a separate sample rate conversion circuit (SRC) to perform many stream conversion and put together of different rate input audio streams. The sample rate conversion system converts data, such as multiple streams of digital audio data sampled at different rates, and performs interpolation, decimation, FIR filtering, and mixing of multiple streams of data using the separate SRC. The SRC uses two bifacial I/O reminiscences for alternately storing input and output information as part of a sample rate converter. When the sample rate converter writes output to one of the bidirectional memories, it has the option of summing the data with the data already stored in the same I/O memory. Therefore, a separate digital signal processor can use the sample rate converter circuit to perform a number of blending for the multiple streams.




In [1] author used Short Time Energy and Statistical method for composite silence removal technique. The performance of the projected formula is compared with the Short Time Energy (STE) formula and therefore statistical method with given Signal to Noise Ratio (SNR). A comparison between the speaker identification rate including and excluding the silence removal technique shows around 20% increase in identification rate.

In [2] author used Time Frequency Transforms for signal decomposition in setting frame theory and provides new insights in application to data analysis. He investigates some interaction of these tools, both theoretically and numerical experiments in order to characterize the signal and their frame transforms. He also used a concepts of persistent homology as an important new subfield for computational topology also formulations of time frequency analysis in frame theory.

In [3] author used automatically segment the speech signal into silence, voiced and unvoiced regions which are very beneficial in increasing the accuracy and performance of recognition systems. Proposed system is based on three necessary characteristics of speech signal namely Zero Crossing Rate, Short Time Energy and Fundamental Frequency.

## 2. PROPOSED SYSTEM

Speech is a non-stationary signal, meaning that its statistical properties are not constant across time. Instead, we want to extract spectral features from a small window of speech that characterizes a particular sub phone and for which we can make the (rough) assumption that the signal is stationary.

When harmonic components of a signal are known, the signal can be presented in a different way that highlights its frequency content rather than its time domain content.

Introducing the third axis of frequency perpendicular to the amplitude-time plane the harmonic components can be plotted in the plane that corresponds to their frequencies.

Firstly, an audio with a gap of silence is given as input. This gap of silence needs to be eliminated. In order to eliminate this frame by frame analysis is carried out. Due to this, changes are observed in the amplitude of the signal. The final output we get is with changes in amplitude. This output does not have the gaps of silence that are present in the original audio.

The final output is a continuous audio with no silence gaps. It is a clear and continuous output audio. This is the output that is desired. Thus, the silence gap in an audio is successfully removed giving a continuous audio clip without any pauses or gaps of silence as output.

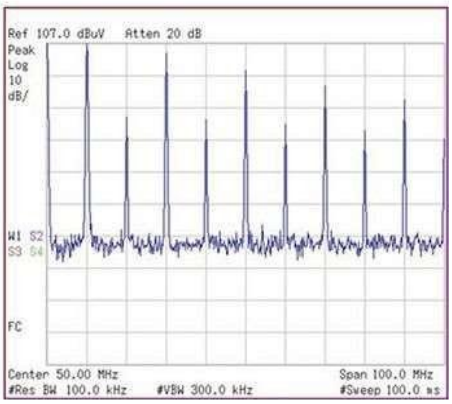



Fig 2.Signal with Noise

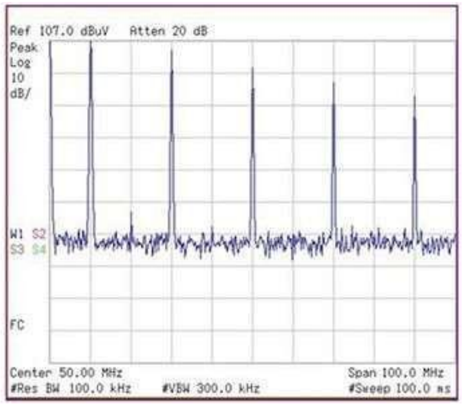
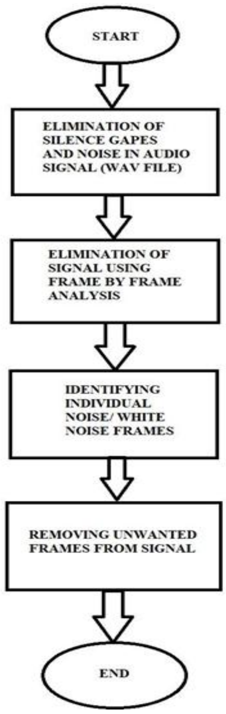



Fig 3. Signal without Noise

## 3. SYSTEM DESIGN

Firstly, an audio with a gap of silence is given as input. This gap of silence needs to be eliminated. In order to eliminate this frame by frame analysis is carried out. Due to this, changes are observed in the amplitude of the signal. The final output we get is with changes in amplitude. This output does not have the gaps of silence that are present in the original audio. The final output is a continuous audio with no silence gaps. It is a clear and continuous output audio. This is the output that is desired. Thus, the silence gap in an audio is successfully removed giving a continuous audio clip without any pauses or gaps of silence as output. The flowchart for the proposed system has been shown below where in the audio signal is analyzed, divided into frames and then how the silence or the noise from the audio signal is removed.




**Diag -1**: Proposed System

There has not been made use of any such kind of special algorithms in our project instead we have just used the inbuilt matlab functions to modify the audio analog signal which is in wave format and create samples which are discrete in time. the different functions and libraries that we have used from the MATLAB are listed below.

(i) audioread()- audio read(filename) reads data from the file named filename, and returns sampled data, y, and a sample rate for that data, Fs.

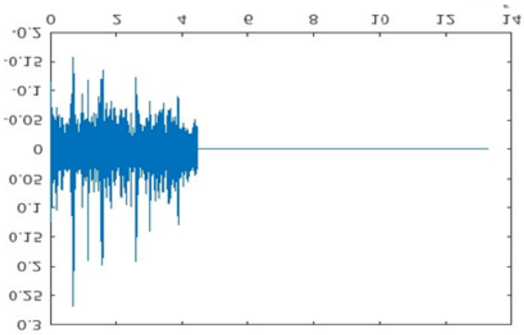
(ii) plot(signal)- plot() returns a column vector of chart line objects. Use h to modify the properties of a specific chart line after it is created.

(iii) figure- When you specify this argument, MATLAB searches for an existing figure in which the Number property is equal to n If no figure exists therewith property price, MATLAB creates a replacement figure and sets its variety property to n . By default, the amount property price is displayed within the title of the figure.

(iv) sound(signal)- sound (MATLAB Functions) sound(y,Fs) sends the signal in vector y (with sample frequency Fs ) to the speaker on PC and most UNIX platforms. Values in y are assumed to be in the range.

## 4. EXPERIMENTATION & RESULT ANALYSIS

The simulation studies involves the analysis of various audio signals and analyze various parameters using MATLAB. In this project, Silence and Noise of an input signal is removed by using MATLAB software. And it gives the resultant output without any noise and silence. This process can also be implemented by using LMS loop adaptive filter algorithm for getting better accuracy.



**Diag -1**: Proposed System

## 5. CONCLUSIONS

Using MATLAB software, we are able to get a clear audio output without any noise or silence gaps. The given input audio has gaps of silence or noise. By using Matlab, the code that is written using it analysis the audio clip using the concept of frame by frame analysis. Due to this, the silence gap or noise in the signal is searched and when found it is eliminated. Once the silence gap or noise has been successfully removed, we get a continuous audio clip. This is the desired output and is successfully achieved.

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