Supervised Learning for Predictive Audio Noise Filter using Look-Back Method

Sachin M. Narangale, G. N. Shinde

***Abstract*- This paper proposes a prototype algorithm for noise removal in the audio files. The predictive noise filter identification technique used here is a novel technique of noise removal. The objective is to understand the effect of supervised noise removal function. The predictive noise filter execution on audio files is reducing the noise by understanding the noise of previous instance or bit of the audio file. The noise filter removes noise of the audio signal. Experiments were conducted by designing a prototype with modifying the Noise removal function of the software which uses the look-back method. In the look-back method, the current pointer refers the previous location's noise. The novel code reduces the noise larger than already existing function. The quantitative analysis of experimented prototype reduces the abrupt noise up to the average of 3.2463% than the original noise removal module.**

***Index Terms*- Abrupt Noise, Noise, Noise Removal, Predictive noise, Supervised Noise Removal**

I. Introduction

NOISE if processed, and utilized for audio processing, can yield optimized outputs in the field of audio programming. Audio programming has become a vast and important sector of multimedia communication. Audio retrieval or searching algorithms can be refined with the effective implementation of noise removal algorithms.

In audio noise removal, some filters are effective for low-light noise removal with edge preservation and comparably low computational load [1]. In audio programming, noise removal is a key factor. As noise is an unwanted audio signal, there are various sources and types of noise. Electronic noise is a thermal noise. The noise generated from revolving machines, vehicles, rain, etc. is an acoustic noise. Due to some electronic equipment electromagnetic noise gets generated. Electrostatic noise is generated by presence of voltage. These and more types of noise are removed using noise removal functions. Some distortion or fading can occur due to communication distortion. Due to low capability of network technology and bandwidth quantization noise and loosing data packets can occur. This may result in signal loss. Signal distortion means unwanted change in a signal which can be a proper audio signal [2]. The novel technology proposed in this paper can be implemented for linear and non-linear approach for noise removal of images [3]. To record the audio, audio recording devices and the software are required. All recording devices, either analogue or digital, have traits which make them susceptible to noise [4]. Professional and technically perfect audio recording devices can avoid entry of noise in the audio signal. It can be observed that, numerous filter designs have been implemented in communication systems to reduce and eventually eliminate the effects of incoming background noise, as well as to enhance speech intelligibility [5,9].

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II. Prototype Algorithm

Following is the stepwise algorithm for the noise removal function. This algorithm demonstrates the supervised learning for predictive audio noise removal. The filter uses look-back method.

ALGORITHM: Supervised Noise removal function using predictive audio noise filter:

START
STEP 1. Get the audio track to be processed
STEP 2. Get the noise profile
STEP 3. Train the Noise removal function
a. Select the noise of previous instance
b. Learn from the previous noise
c. Define the noise removal to be applied
STEP 4. Apply the prototype (Apply noise removal function)
STEP 5. Analyze normal, noise removed and experimented files
STEP 6. Implement output to the frequency-amplitude spectrum
STOP.

III. Plot Spectrum Analysis

The plot spectrum analysis of frequency (Hz) and the amplitude dB values (sample values of a song "constant_ringing_sleigh_bells.mp3") are presented in Table 1: Plot Spectrum Analysis of the audio file. The first column of the table contains the frequency values in Hz of the audio tracks. The second column contains the amplitude values of normal audio track in dB. The third column contains the amplitude values of the noise removed audio track. This noise removal function is the existing noise removal function. The fourth and last column is the representation of the amplitude values of the experimented audio track.

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TABLE 1 PLOT SPECTRUM ANALYSIS

Frequency (Hz)	NORMAL (dB)	NOISE REMOVED (dB)	EXP (dB)
86.1328	4893.9275	8587.8430	7682.3449
172.2656	4991.7472	8768.9481	7941.0367
258.3984	5599.9082	9569.03901	8913.6939
344.5312	5735.6251	9747.28669	9588.2468
430.6640	5821.6590	9860.25883	9777.9720
516.7968	5930.9246	10001.7350	9883.5669
602.9296	5798.1953	9829.73796	9716.7457
689.0625	5686.5305	9684.36808	9586.5870
775.1953	5613.0392	9588.27078	9482.0049
861.3281	5549.8578	9505.64746	9403.0583
947.4609	5549.6134	9505.38266	9405.4051
1033.593	5588.1612	9555.82770	9414.0611
1119.726	5523.9839	9471.84029	9226.5939
1205.859	5199.5066	9045.41297	8696.8587
1291.992	3926.5604	7338.02570	6994.1359
1378.125	3376.2052	6578.04130	6315.1743

IV. SCREENSHOTS

The screenshots captured in the experiment are shown below.

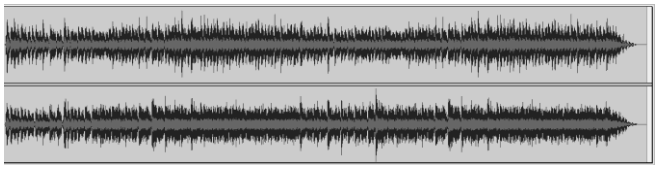

Figure 1: NORMAL track panel of the audio file

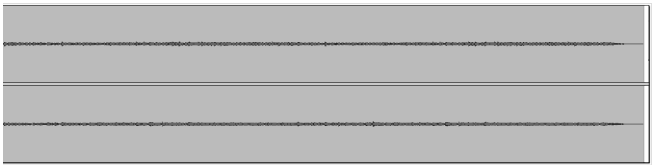

Figure 2: NOISE REMOVED track panel of the audio file

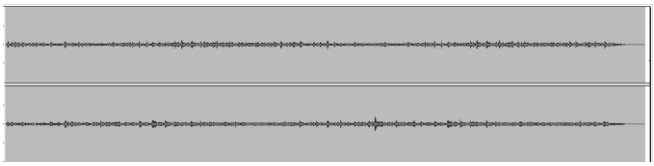

Figure 3: EXPERIMENTED track panel of the audio file

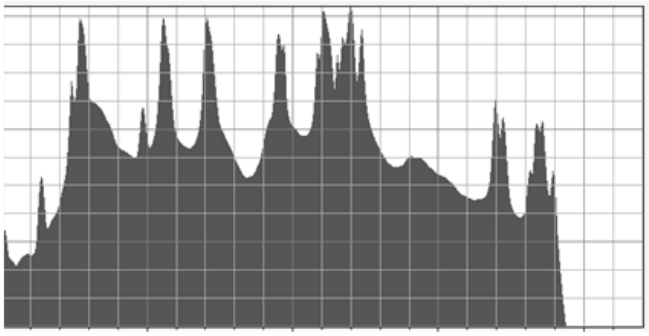

Figure 4: NORMAL window of the Plot spectrum of the audio file

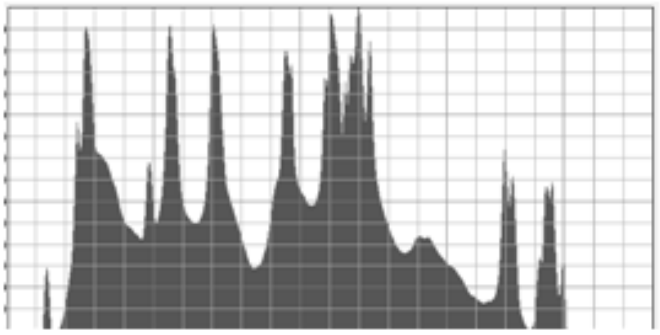

Figure 5: NOISE REMOVED window of the Plot spectrum of the audio file

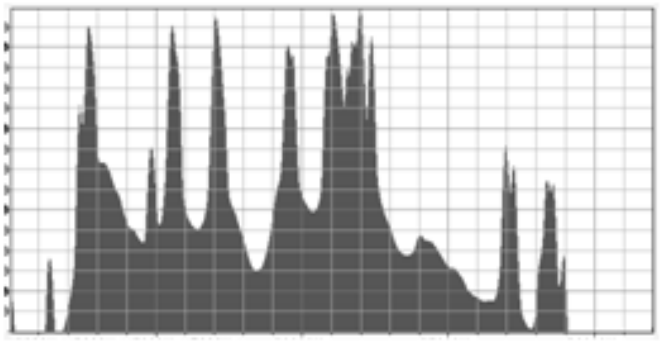

Figure 6: EXPERIMENTED window of the Plot spectrum of the audio file

Figure 1 is the normal track panel of audio file. The audio file may contain the noise. The audio programming software has to be used for the removal of noise. The noise removal module of the audio editing software seeks the noise profile to be selected from the audio file. Then, it sets the amplitude range for the noise removal. Once the noise is removed, the track panel looks like figure 2. This noise removal module removes the noise. The proposed prototype, when experimented, for the audio file generates the track panel shown in figure 3. Figure 4, 5 and 6 are the screenshots of window of the plot spectrum in normal, noise removed and experimented audio tracks respectively.

The spectrum analysis figures of the different audio files in normal, noise removed and experimented files are shown below.

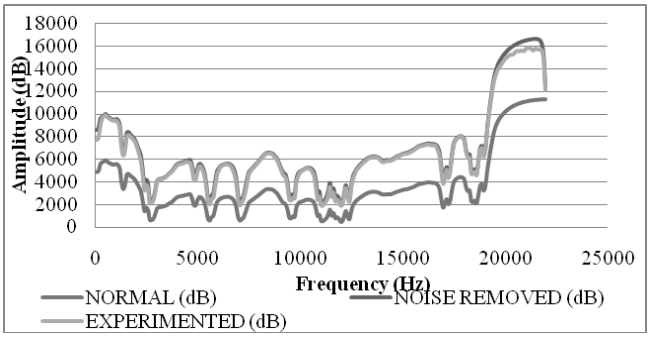

Figure 7: Spectrum analysis of the audio file "Constant_ringing_sleigh_bells.mp3"

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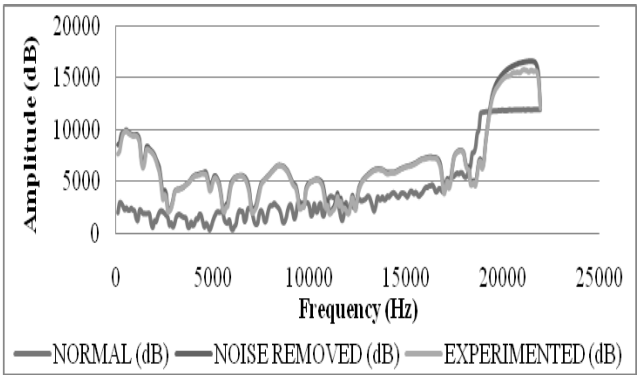

Figure 8: Spectrum analysis of the audio file "Fast_electronic_beeper.mp3"

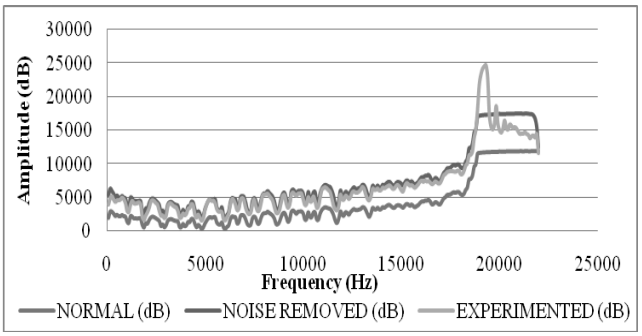

Figure 9: Spectrum analysis of the audio file "Fire_bell.mp3"

Figure 7 represents the frequency vs amplitude Spectrum analysis of Constant_ringing_sleigh_bells.mp3 audio file. Frequency is shown on the x axis and amplitude is shown on y axis. A small difference can be observed in the noise removed and proposed prototype's amplitude values.

V. Experimental setup

This research experiments were executed on open source freeware Ubuntu, a Linux desktop operating system. For audio recording and editing an open source, free and cross-platform software Audacity was used. The features of Audacity in this research are used for prototype designing related to noise removal. For GUI between Audacity and Ubuntu a C++ library wxWidgets was used. wxWidgets facilitates developers with a single code base for creating cross-platform applications for Windows, Mac OS X, Linux and other platforms. A desktop system with hardware configuration 2GB RAM, i3 processor, 3.08GHz and 350GB HDD was used for experiments.

VI. Results and Discussions

Data recording gets corrupted due to the excessive noise. Medical instruments which record the audio signals are largely affected due to noise. The ECG (electrocardiogram) is the electrical signal which is important in identifying the state of cardiac patients. As the mobile phone users are becoming more and more daily, there is a need of proper communication. At the listeners' side, speech enhancement programs or noise removal programs are must. Due to the equipments and environment lot of noise gets generated. Sometimes the sound level can go below threshold audio level. The noise removal modules remove the noise in abrupt way.

From the experiments and the analysis, it can be clearly seen in above figures, the audio files were processed for the noise removal function in Audacity. After understanding the noise instance in the track, the experimented program with proposed prototype has been executed on the audio file. The results of the experimented prototype show that, the noise is removed from the files and at the same time, there is a smoothening effect on the audio files. There is a minute distance between the plot spectrum of noise removal and experimented prototype's plot spectrum. As shown in the figure 7, 8 and 9 there is a small difference in the noise removed and experimented files.

VII. Conclusion

In this paper, a prototype algorithm for Smoothening the noise in the audio files has been proposed. The predictive noise filter identification technique used here is a novel technique of noise removal. A novel method of supervised noise removal function is demonstrated. The predictive noise filter execution on audio files reduces the noise of the audio files. The experimented prototype reduces the abrupt noise up to the average of 3.2463% than the original noise removal module. The prototype is doing this by understanding the noise of previous instance or bit of the audio file which is look-back method. The prototype can be implemented for noise removal. It can be effectively implemented in audio retrieval programs, smoothening of the audio files, audio programming.

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