A Systematic Algorithm for Denoising Audio Signal Using Savitzky - Golay Method

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Abstract— Audio signal noise reduction is a great task to acquiring noiseless sounds in the noisy environment. But, naturally the audio signals are naturally very noise, which is produced from analog or digital recorders. In this research paper, an efficient noiseless audio reduction system using Savitzky-Golay method is proposed for eliminating unwanted echoes, ripples, etc. Three different audio signals are taken for this work. First one is a human voice in the mobile conversation, which is used for telecommunication purpose. Second is one vehicle sound for usages in security applications. The last one is a music instrumental sound, which used in recording lab. Finally, the proposed system will be compared with the existing noise filters and proved their efficiency with high accuracy, high signal to noise ratio and low elapsed time.

Keywords*- Audio Signal, Digital Filters, Digtal Audio Processing, MATLAB, Savitzky-Golay Algorithm,*

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I. INTRODUCTION

Audio is one of the most important data of multimedia, which has been intensively used in the everyday application, in the internet, telephone communication and teleconferencing, etc. Because of human error or machine error, the audio signals are naturally noisy (such as echo, reverberation, white noise, narrow band noise, etc.). But, we need a noiseless environment for the communication. So, the audio noise reduction system is an important one in the studies of Audio Signal Processing. Many filters and algorithms are proposed by the researchers. In this research paper, an efficient audio signal noise reduction system is proposed by using Savitzky-Golay algorithm. First, the existing methods will be reviewed and its performance will be analyzed by calculating Accuracy, Signal to Noise Ratio and its existing time. Then finally our proposed method will be explained and its performance will be analyzed for the various types of audio signals. The properties of the proposed method will be compared with existing algorithms which should be more efficient than them.

II. REVIEW AND PERFORMANCE ANALYSIS

Echo Cancellation is an important step in the audio signal noise reduction problems. A design of accurate echo cancellation [2] has been proposed for the telecommunication applications. That technique is easily executed and gives the perfect results for canceling unwanted peaks of signals. Adaptive filtering algorithms were proposed for Echo cancellation [3, 4]. These results proved that LMS algorithm is the best for channel equalization respectively. By using Wavelet transform [5], the noise had been reduced from the free speech signal. In that work, the wavelet is based on thresholding and denoising functions. There were hard thresholding is more comfortable for the unwanted ripples. The spectral subtraction [6] has been used to remove the noises in speech signals.

The ripples are removed in frequency domain only, not in the time domain. The SNR value is calculated as 0.3371dB, which proves the lesser amount noise ratio. The wavelet transform is also combined with the Kalman Filter [7] for reducing the spectral distortion which is measured by Itakura-Saito distance. From the readings, Kalman filtering with wavelets provides the denoised system. The noise signals are transformed by using Discrete Wavelet Transform [9], which is also in wavelet domain with hard and soft thresholding.

Apart from these methods, some filters are commonly used for the noise reduction process. They are the Gaussian filter, Butterworth filter, Elliptical filter, Chebyshev I and II filters and Comb filter, etc. For eliminating the noises in ECG signals [10] using wavelets, the performance of Butterworth and Chebyshev filters had been analyzed. In that work also proposes the both filters have the small variation in their performance. In another one paper [11], the Chebyshev II filter is implemented using XSG for reducing the ECG signals, which filter showed the good performance when its area and power will be considered. From these reviews, the Butterworth filter is the more efficient one. But we should analyze their performance for reducing their drawbacks in our proposed method. The speech signal is taken as our input signal for this analysis and the outputs are as follows:

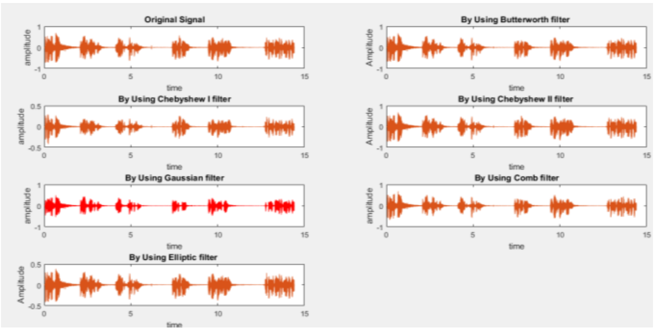

Figure 1. Results for the various filters

Table 1. Performance analysis of Filters from the Results

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S. No	Name of the Filter	Accuracy	SNR (dB)	Execution Time (Sec)
1.	Butterworth	0.93	32.15	1.39
2.	Gaussian	0.96	36.13	5.72
3.	Chebyshev I	0.62	17.19	2.12
4.	Chebyshev II	0.81	26.58	1.46
5.	Comb	0.73	13.91	4.87
6.	Elliptical	0.91	17.44	3.07

Fig.1 gives the results from the various type of filters. The accuracy, SNR (dB) and Execution time are tabulated in Table 1. From the literature review and performance analysis, the Gaussian filter has the lowest number of ripples when compare to the other filters but it takes more time for its execution process (5.72 Sec). Butterworth filter has the maximum flat response, phase distortion is moderated and highly accurate filter. An elliptical filter has a sharp cutoff slope when compared to the others. But, it has many ripples in its output signal. Also, it has a non-linear phase response. Chebyshev I filter has a cut off frequency nearer to Butterworth filter but very lower accurate filter. Chebyshev II filter has the better rate of attenuation than others. But, Comb filter has many ripples and executed very slow. So, the proposed system should be reducing the drawbacks of existing noise reduction methods.

III. METHODOLOGY

3.1. Block Diagram

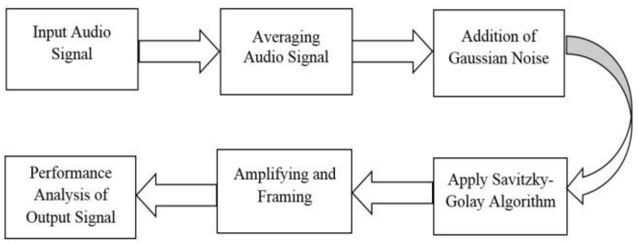

Figure 2. Block Diagram of Audio Noise Reduction System

The working process of audio signal noise reduction system is explained in Fig.2. First, the input signal is taken from our database which are a human voice, vehicle sound and music instrument sound. Then the signal will be averaged to equalize every samples. Then, the Gaussian noise will be added to the signal for more accuracy. After that, the Savitzky – Golay algorithm will be used for smoothing the audio signal to reducing the noise elements. The filtered signal should be amplified for converting the lower signals into higher signals. More numbers of sample will be presented, so output signal must be framed for an accurate survey. Finally, the output noiseless signals of proposed method will be analyzed by using mathematical descriptions.

3.2. Proposed Method

Step 1: The input audio signal *I (t, a)* is derived in the time domain as follows,

$$I(t_n, a_n) = \sum_{n=1}^{N} \frac{1}{f_s} . n \quad (1)$$

In Equation (1), t_n and a_n are the time and amplitude values respectively in the n^{th} sample. N is the length of the signal and f_s is sampling frequency.

Step 2: The Input signal should be averaged for hiding or avoiding the intensity of high-frequency noise signal and increase the strength of the original audio signal. The input signal $I (t_n, a_n)$ is averaged as follows,

$$I_{AVG}(t_n, a_n) = \frac{I(t_n + 1, a_n) + I(t_n, a_n) + I(t_n - 1, a_n)}{3} \quad (2)$$

In Equation (2), $I_{AVG} (t_n, a_n)$ is averaged audio signal. This step will increase the PSNR and accuracy of the output signal.

Step 3: For accurate values of the output signal, we can add the Gaussian Noise with the audio signal. First, we could design the Gaussian window of size L, which is described as

$$w(n) = e^{\frac{-1}{2}} \left(\frac{I_A(t_n, a_n) - (L - 1/2)}{\sigma \, (L - 1)/2} \right)^2 \quad (3)$$

In Equation (3), σ is the standard deviation. The windowing function will be convoluted with Gaussian Filter.

$$I_N = \frac{1}{\sigma \sqrt{2\pi}} e^{\frac{-I_A(t_n, a_n)^2}{2\sigma^2}} . w(n) \quad (4)$$

In Equation (4), I_N is the audio signal with Gaussian noise. It gives more precious results and easily executed using mathematical descriptions.

Step 4: Now the noised signal will be filtered by using Savitzky-Golay algorithm. The preservation of time series (Local Maxima and Minima) is a great advancement by using SG Algorithm. For SG algorithm, the input audio signal is taken as the polynomials I_{SG} and expressed as follows in Equation (5).

$$I_{SG} = \sum_{n=1}^{N} C_n \, (a_n t_n)^k \quad (5)$$

The filtered signal I_F is the convolution of noisy signal and SG algorithm polynomial as follows in Equation (6).

$$I_F = I_N \otimes I_{SG} \quad (6)$$

Step 5: Amplification is the process of conversion from lower signals into the higher signal, which factors are usually expressed in decibel (dB). The amplification of the audio signal is derived as follows,

$$I_A(t_n, a_n) = I_F . \frac{std(a_n)}{std(a_n) * \sqrt{10^{dB\,Value/10}}} \quad (7)$$

In Equation (7), $I_A (t_n, a_n)$ is the amplified signal, *std* refers standard deviation value and dB value will be given as our need.

Step 6: After the amplification, the audio signal should be analyzed. But, many numbers of sample are presented in the audio signal. So, the signal will be divided into several frames for accurate analysis. Therefore the signal is divided into 50 frames, which corresponds to *x* samples. The framing window length is taken as $f_s/50$. Then every framed windows are denoted as $W_1, W_2 \ldots W_{50}$ and described as follows,

$$W_n = I_A(t_1, a_1) + I_A(t_2, a_2) \ldots I_z(t_z, a_z) \tag{8}$$

In Equation (8), I_z (t_z, a_z) is the final edge signal of the selected frame. $I_A(t_n, a_n)$ is the final output signal and it will be analyzed by using frame by frame using (8).

IV. RESULTS AND DISCUSSIONS

In this research paper, three types of sounds are taken. They are the human voice, truck sound and Piano sounds, which are taken as the input audio signals. MATLAB R2017b is used as the software platform. The original noisy audio signals and resultant noiseless audio signals are plotted as following figures.

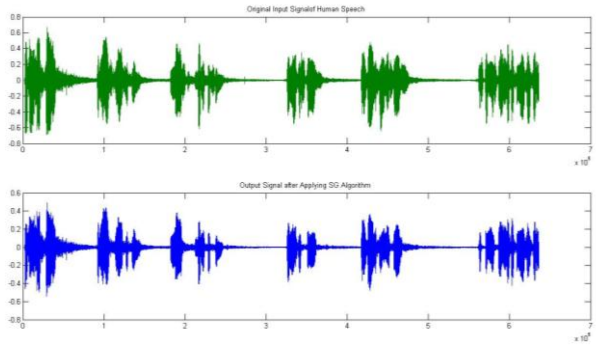

Figure 3. Input and output Audio signals for Human Voice

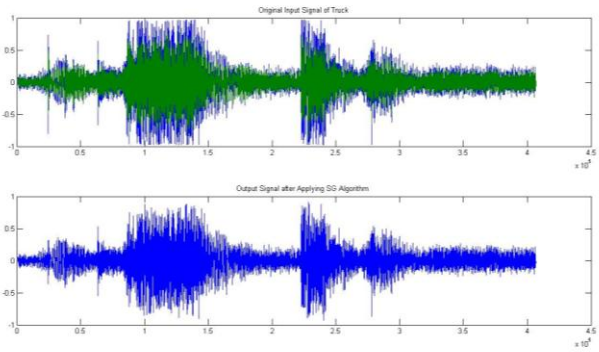

Figure 4. Input and output Audio signals for Truck

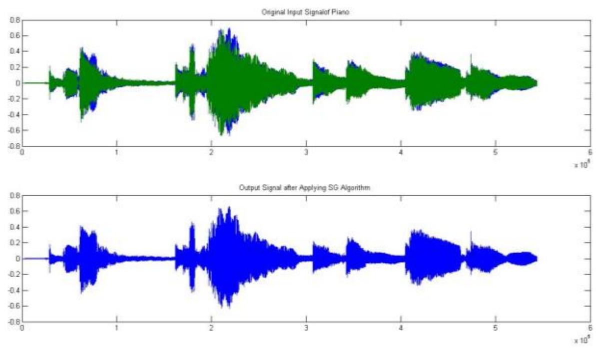

Figure 5. Input and output Audio signals for Piano

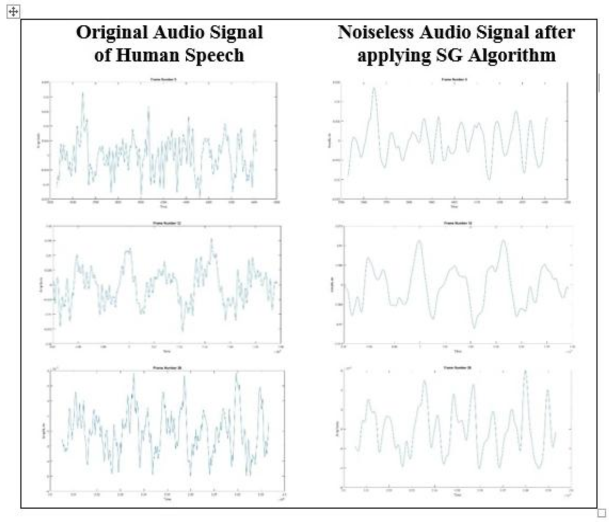

Figure 6. Input and Output Audio signals for Human Speech Frame Numbers 5, 12 and 26

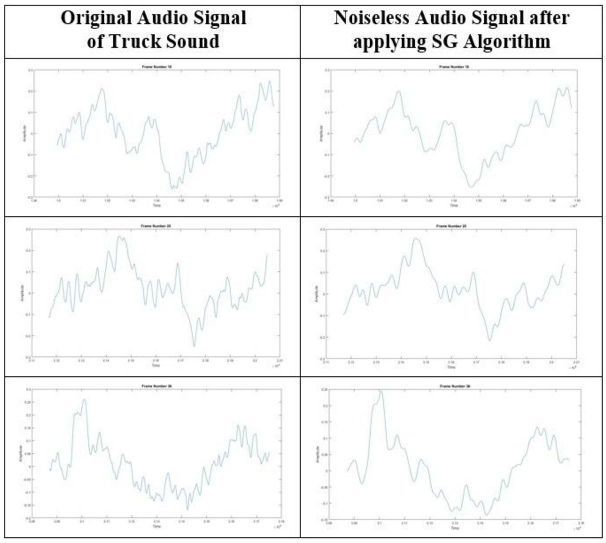

Figure 7. Input and Output Audio signals for Human Speech Frame Numbers 18, 25 and 36

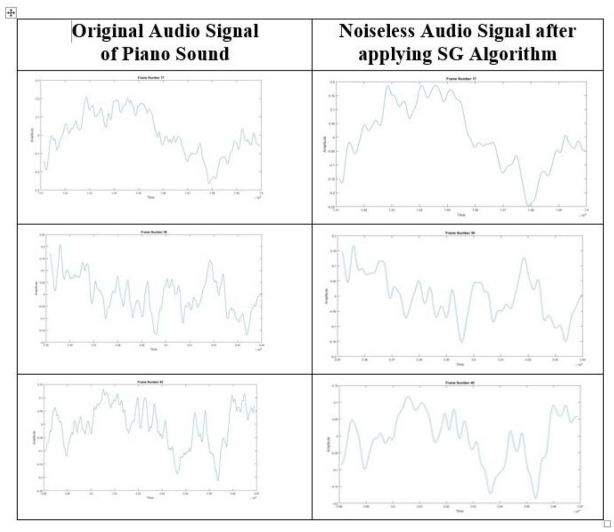

Figure 8. Input and Output Audio signals for Human Speech Frame Numbers 17, 39 and 45

Table 2. Performance analysis of the Proposed Method

Type of Audio Signal	Frame Number n=1,2…50	Accuracy	SNR (dB)	Execution Time (Sec)
Human Voice	Frame No. 05	0.92	30.11	0.65
	Frame No. 12	0.97	36.83	0.48
	Frame No. 26	0.89	20.14	1.63
Truck Sound	Frame No. 18	0.88	28.32	1.47
	Frame No. 25	0.95	34.41	0.56
	Frame No. 36	0.91	32.11	0.41
Piano Sound	Frame No. 17	0.82	29.47	0.78
	Frame No. 39	0.87	31.34	0.63
	Frame No. 45	0.96	37.23	1.58

In Figures 3, 4, and 5, the input and output audio signals are plotted for a human voice, truck sound and piano sound respectively. But there many numbers of samples which are divided into 50 frames. For performance analysis, three frames are selected randomly for every audio signal. The framed signals of inputs and outputs are plotted in Figures 6, 7 and 8. Comparatively, the accuracy is greater than 0.82 and higher value of accuracy is achieved in Frame number 12 (0.97) in a human voice. For the readings of Signal to Noise ratio values, the maximum reached value is 36.83 dB and all SNR values are the higher than existing noise reduction methods. The proposed method is rapidly executed within minimum 0.41 seconds and maximum 1.63 seconds. Hence, the proposed method has the low elapsed time which makes the quick response from possible outcomes.

V. CONCLUSION

The audio signal noise reduction method using Savitzky-Golay algorithm is more efficient than exiting methods. From the results, the proposed system is highly accurate one, noiseless background and rapidly executed. This system will be used to enhancing the strength of audio signals and more gain for telecommunication, internet and security applications. In our future work, this method will be demonstrated in the hardware environment and MATLAB Simulink.

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