A Review for Reduction of Noise by Wavelet Transform in Audio Signals

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Abstract - *Noise reduction performing on the acoustic signals presents a kind of techniques that is useful to reduce the noise or unnecessary signal (also called unwanted signal) from the original signals. There are many noise reduction techniques which have been proposed for the removal of noises from the digital audio signals. However, the effectiveness of those techniques is fewer. In this paper, the review on the noise reduction techniques for the acoustic signals is discussed regarding on the use of wavelet transform. It provides an overview to reduce the unwanted noise and gives much better outcomes for the future research works. In addition, the implementation of the experimental results is also presented with the wavelet-based denoising technique.*

Key Words: Acoustic signals, Denoising, Wavelet Transform, Filters, Thresholding, Additive Gaussian White noise.

1. INTRODUCTION

Most of the images and audio signals are mostly affected by noise during transmission, capturing, and storage. The denoising had already been a traditional difficulty within the processing of images and signals. The term "denoising" is the extraction of a signal from a mixture of signal and noise to enhance the audio signals. A variety of denoising techniques have been proposed so far, among this, wavelet-based denoising provides a superior performance, due to its properties such as scarcity, energy compaction, and multi-resolution structure. In this paper, we discuss the literature reviews on the reduction of noise using by some Wavelet Transform techniques in audio signal processing and investigate the usage of double density discrete wavelet transform (DD-DWT), which based on one scaling function and two wavelet functions for signal denoising, that is exposed by the illustration of the conducted experiments comparing with the differences between the noisy and denoised of a sample signal.

1.1 History of Wavelet

Wavelet transform has been studied extensively in recent years as a promising tool for noise reduction. It has attractive properties so it is used for signal processing. It consists of a set of basis function that can be used to analyze signals in both time and frequency domains simultaneously. Since presence of noise in signals and images restricts one's ability to obtain information it has to be removed. Discrete Wavelet Transform is a type of wavelet transform which has one scaling function and one wavelet function [1].

There are a lot of wavelet families. The literatures that relate to the wavelet transform are being summarized according to the histories of the types of wavelet families [2] as shown in Table 1. In these wavelet families, the "Daubechies" are the most widely used in acoustic based recognition problems. Haar wavelet (similar to Daubechies db1) is one of the oldest and simplest wavelet. The daubechies names are written as dbN where N is the order of the family [3].

Wavelet transform divides the input signal into two components: one is the approximation of the signal "low-frequency subband" and the other defining the details "high-frequency subband". In general, most of noisy components concentrate in the high frequency subband; while the main information components concentrate in the low frequency subband [4].

1.2 Type and Categories of Noise

The digital form of audio, images, and video has become the commercial standard in the past decade. During digitization noises may be present in the system. These noises may affect original signal. There are many types and sources of noise or distortions. They are included in [5]:

- Electronic noise such as thermal noise and shot noise.
- Acoustic noise emanating from moving, vibrating or colliding sources such as revolving machines, moving vehicles, keyboard clicks, wind and rain.
- Electromagnetic noise that can interfere with the transmission and reception of voice, image and data over the radio-frequency spectrum.
- Electrostatic noise generated by the presence of a voltage.
- Communication channel distortion and fading and
- Quantization noise and lost data packets due to network congestion.
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Signal distortion is the term often used to describe a systematic undesirable change in a signal and refers to changes in a signal from the non-ideal characteristics of the communication channel, signal fading reverberations, echo, and multipath reflections and missing samples [6]. Depending on its frequency, spectrum or time characteristics, a noise process is further classified into several categories [7] illustrated in Figure 1:

1) White noise: Purely random noise has an impulse autocorrelation function and a flat power spectrum. White noise theoretically contains all frequencies in equal power.
2) Narrow band noise: It is a noise process with a narrow bandwidth such as 50/60 Hz from the electricity supply.
3) Colored noise: It is non-white noise or any wideband noise whose spectrum has a non flat shape. Examples are Pink noise, Brownian noise and autoregressive noise.
4) Impulsive noise: It consists of short-duration pulses of random amplitude, time of occurrence and duration.
5) Transient noise pulses: It consists of relatively long duration noise pulses such as clicks, burst noise etc.

Table -1: History of Wavelet Families

Year	Name of Mathematician	Types of Wavelet
1910	Alfrd Haar	Haar families
1981	Jean Morlet	concept of Wavelet
1984	Jean Morlet and Alex Grossman	term of "Wavelet"
1985	Yves Meyer	Orthogonal Wavelet
1988	Stephane Mallat and Meyer	concept of Multi-resolution
1988	Ingrid Daubechies	Compact Support Orthogonal Wavelet
1989	Mallat	Fast Wavelet Transform

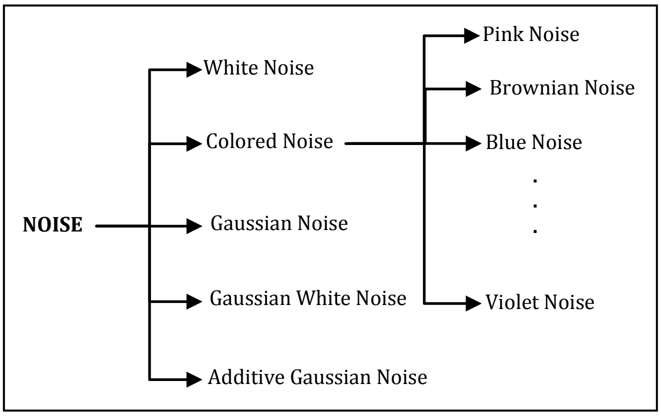

Fig -1: Some Categories of Noise

2. SOME STUDIES ON LITERATURE SURVEY

The objective of audio denoising is attenuating the noise, while recovering the underlying signals. It is presented in many applications such as music and speech restoration etc. There has been a fair amount of research on wavelet-based signal denoising. From the beginning of wavelet transforms in signal processing, it is noticed that the wavelet thresholding focuses an attention in removing noise from signals and images. To remove the wavelet coefficients smaller than given amplitude and to transform the data back into the original domain, the method has to decompose the noisy data into an orthogonal wavelet basis. A nonlinear thresholding estimator can compute in an orthogonal basis such as Fourier or cosine.

In denoising of the audio signals, the denoised signal obtained after performing wavelet transformation is not totally free from noise, some remains of noise left or some other kinds of noise gets introduced by the transformation that is present in the output signal. Several techniques have been proposed so far for the removal of noise from an audio signal, yet, the efficiency remains an issue as well as they have some drawbacks in general [8].

B. JaiShankar and K. Duraiswamy [8] have attempted an audio signal denoising technique that based on an improved block matching technique in transformation domain. This is achieved by the transformation with a clear representation of the input signal so that the noise can be removed well by reconstruction of the signal. A biorthogonal 1.5 wavelet transform is used for the transformation process. A multi dimensional signal vector is generated from the transformed signal vector and the original vector signal is reconstructed by applying inverse transform. The signal to noise ratio (SNR) is comparatively higher than SNR level of the noisy input signal thus increasing the quality of the signal. Michael T. Johnson et al. [9] have demonstrated the application of the Bionic Wavelet Transform (BWT), an adaptive wavelet transform derived from a non-linear auditory model of the cochlea, to the task of speech signal enhancement. Results measured objectively by Signal-to-Noise ratio and Segmental SNR and subjectively by Mean Opinion Score were given for additive white Gaussian noise as well as four different types of realistic noise environments. C. Mohan Rao et al. [10] have presented a new adaptive filter whose coefficients are dynamically changing with an evolutionary computation algorithm and hence reducing the noise.

Priyanka Khattar, Dr. Amrita Rai and Mr. Subodh Tripathi [5] have developed the audio quality of de-noised signals that is determined based on mean square error, peak signal to noise ratio and cross correlation. In that paper, the comparison among Daubechies and Haar wavelet is performed and Daubechies 10 performs the best result because they have a peak signal to noise ratio. Nishan Singh and Dr. Vijay Laxmi [11] have focused on audio signals corrupted with white Gaussian noise which is especially hard to remove because it is located in all frequencies. In that paper, Discrete Wavelet Transform (DWT) is used to transform noisy audio signal in wavelet domain. And, their work has been modified by changing universal thresholding of coefficients which results with better audio signal. In these various parameters such as SNR, Elapsed Time, and Threshold value is analyzed on various types of wavelet techniques similar to Coiflet, Daubechies, Symlet etc.

Nanshan Li et al. [12] have proposed an audio denoising algorithm on the basis of adaptive wavelet soft-threshold which is based on the gain factor of linear filter system in the wavelet domain and the wavelet coefficients teager energy operator in order to progress the effect of the content-based songs retrieval system. Their algorithm integrated the gain factor of linear filter system and nonlinear energy operator with conventional wavelet soft-threshold function. Experiments demonstrated that their algorithm had important outcome in inhibiting Gaussian white noise and pink noise in audio samples and enhancing the accuracy of the songs retrieval system.

In the research work [13], the authors have proposed the acoustics-based vehicle type classification system with a new wavelet-based denoising technique called "Denoised MFCC" that has been organized into the addition of Double-Density Dual-Tree Complex Discrete Wavelet Transform (DDC-DWT) to the Mel-Frequency Cepstral Coefficient (MFCC). This proposed denoising method is based solely on excluding the noisy parameters transmitted by the original signal within the process of the conventional MFCC. Even the similarities between features of vehicle dataset are high; the proposed system can be able to adequately detect the type of vehicles. According to the experimental result of that system, it can be observed that this proposed Denoised MFCC method is improved the classification accuracy and the enhanced features of the vehicle type classification system.

Based on this literature survey, it is clear that many approaches exist for audio denoising using wavelet transform but still there is required the purpose of other audio denoising techniques such as adaptive filtering, type and value of thresholding, two-dimensional and three-dimensional fast discrete wavelet transform, etc to improve the performance of the signal features and noise reduction.

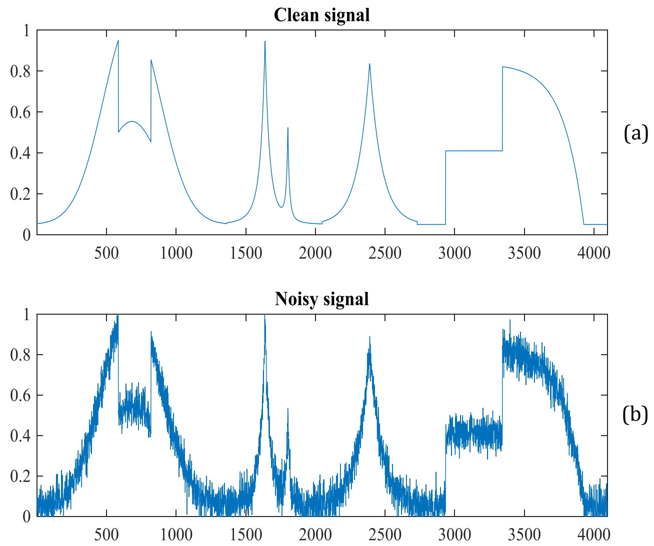

Fig -2: (a) The Primary Input Signal; (b) The Input Noisy Signal corrupted by AGW Noise

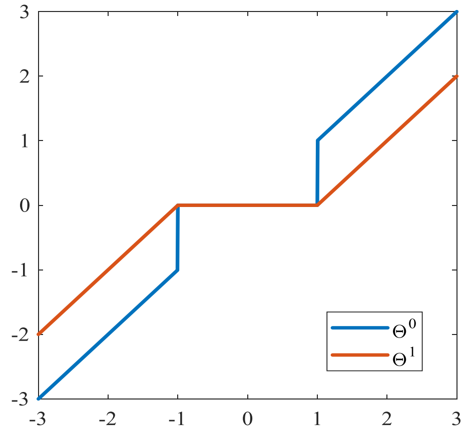

Fig -3: The Linearly Space Points between Theta_0 and Theta_1

3. IMPLEMENTATION ON WAVELET-BASED DENOISING TECHNIQUE

In this section, the implementation of the wavelet-based denoising technique is presented with the differences between the noisy and denoised of a sample signal. The experimental results on the signal denoising are performed in MATLAB application. For testing the performance of the wavelet-based denoising, a sample monophonic wave file called "piece-regular" is taken as the input. The first step is loading the input signal and making a noise. Here, a simple Additive Gaussian White (AGW) noise is considered for the testing purpose. This AGW noise was generated and added to the input signal. The signal-to-noise ratio (SNR) of the noisy audio signal was generated on the calculation of normally distributed random numbers on the size of input signal between 0.05 and 0.95 plus the input signal with the product at a noise level σ of 0.0500. The input clean and noisy signal with respect to its length N=4000 is represented in Figure 2 (a) and (b).

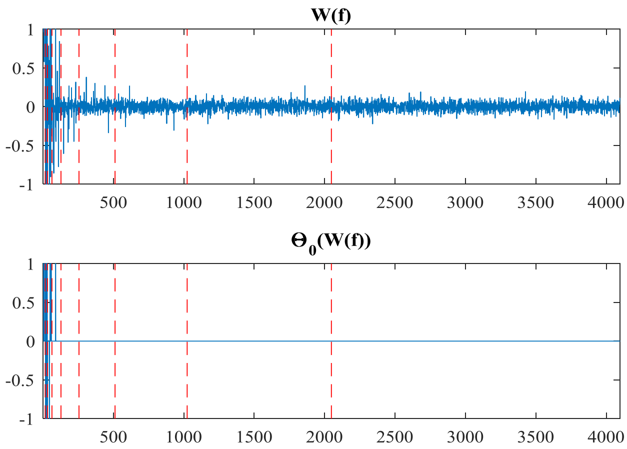

Fig -4: The Wavelet Coefficients W(f) and the Hard Thresholded Coefficients $\Theta_0(W(f))$

To compute the wavelet coefficients W(f), the hard and soft thresholding values of Theta_0 (θ_0) and Theta_1 (θ_1) are set up to produce the linearly space points between -3 and 3 and the demonstration of these values are exposed in Figure 3. The wavelet coefficients W(f) and the hard thresholded coefficients θ_0(W(f)) are also illustrated in Figure 4. Note that how the wavelet coefficients are contaminated by a small amplitude of Gaussian white noise, most of which are remove W by thresholding. In this computation, the forward orthogonal wavelet transform is used to compute these coefficients and the inverse wavelet transform are applied to reconstruct the denoised signal from the thresholded coefficients with the final estimator. As per this conducted result, the final experiments of the difference appearances between the noisy signal and the denoised signal are obtained as shown in Figure 5.

4. CONCLUSION

Most of audio signals can be corrupted by the different types of noise during the audio data acquisition process when attempting to get these input signals. The development of reducing such noise from the audio signals is audio denoising. In this paper, the audio denoising techniques based on wavelet transform are conferred among the other noise reduction techniques for the audio and image signals. According to the survey, the paper can be concluded that the signal denoising techniques must be necessitated to obtain the efficient and improved features of signal and to avoid the producing route of distortion by noise. Additionally, in this paper, the Wavelet Transform for denoising audio signal corrupted with AGW noise have been implemented by the conducted results of experimentation. Audio denoising is performed in wavelet domain by thresholding wavelet coefficients. It is shown that soft or hard thresholded coefficients can be applied in order to get the greatest performance especially for an existing level of noise.

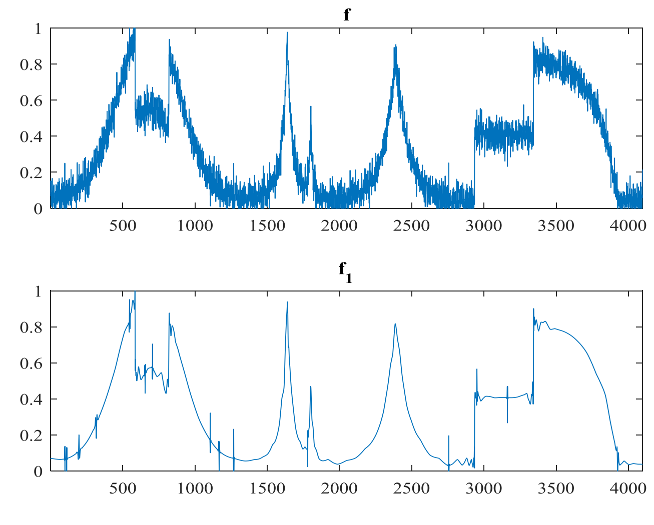

Fig -5: The Differences between the Noisy Signal *f* and the Denoised Signal f_1

REFERENCES

[1] F. Luisier, T. Blu, and M. Unser, "A New SURE Approach to Image Denoising: Inter-scale Orthonormal Wavelet Thresholding," IEEE Trans. Image Process, Vol.16, No. 3, Mar. 2007, pp. 593-606.

[2] C. L. Liu, "A Tutorial of the Wavelet Transform," Feb. 2010.

[3] A. M. Amruta, and S. Shashikant, "Advanced Speaker Recognition," International Journal of Advances in Engineering & Technology, Jul. 2012.

[4] Olkkonen, and Hannu, "Discrete Wavelet Transforms: Algorithms and Applications," In Tech, Aug. 2011.

[5] K. Priyanka, Dr. R. Amrita and Mr. T. Subodh, "Audio De-noising using Wavelet Transform," International Research Journal of Engineering and Technology (IRJET), Vol. 3, Issue 7, Jul. 2016, pp. 489-493.

[6] K. P. Obulesu and P. U. Kumar, "Implementation of Time Frequency Block Thresholding Algorithm in Audio Noise Reduction," International Journal of Science, Engineering and Technology Research, Jul. 2013, pp. 1513-1520.

[7] K. Jashanpreet, B. Seema, and K. Sunil, "A Review: Audio Noise Reduction and Various Techniques," International Journal of Advances in Science Engineering and Technology, Vol. 3, Issue 3, Jul. 2015, pp. 132-135.

[8] B. JaiShankar and K. Duraiswamy, "Audio Denoising using Wavelet Transform," International Journal of Advances in Engineering & Technology (IJAET), Vol. 2, Issue 1, Jan. 2012, pp. 419-425.

[9] T. J. Michael, Y. Xiaolong, and R. Yao, "Speech Signal Enhancement through Adaptive Wavelet Thresholding," Speech Communications, Vol. 49, No. 2, 2007, pp. 123-133.

[10] C. R. Mohan, Dr. B. C. Stephen, and Dr. M. N. Giri, "A Variation of LMS Algorithm for Noise Cancellation," International Journal of Advanced Research in Computer and Communication Engineering, Vol. 2, Issue 7, ISSN (Print): 2319-5940, Jul. 2013.

[11] S. Nishan and Dr. L. Vijay, "Audio Noise Reduction from Audio Signals and Speech Signals," International Journal of Computer Science Trends and Technology (IJCST), Vol. 2, Issue 5, Sep-Oct. 2014.

[12] L. Nanshan and Z. Mingquan, "Audio Denoising Algorithm Based on Adaptive Wavelet Soft-Threshold of Gain Factor and Teager Energy Operator," Proceedings of IEEE International Conference on Computer Science and Software Engineering, Vol. 1, 2008, pp. 787-790.

[13] N. T. Lwin, W. Aung and N. O. Htet, "Vehicle Type Classification Based On Acoustic Signals Using Denoised MFCC," Proceedings of 2018 IEEE International Conference on Information Communication and Signal Processing (ICSP), Sep. 2018, pp. 113-117.