International Research Journal of Engineering and Technology (IRJET) e-ISSN: 2395 -0056
Volume: 03 Issue: 07 | July-2016 www.irjet.net p-ISSN: 2395-0072

Audio De-noising using Wavelet Transform

Priyanka Khattar[1], Dr. Amrita Rai[2],Mr. Subodh Tripathi[3]

[1]M.Tech Scholar, Department of Electronics and Communication Engineering(Specialization in Signal Processing),Rawal Institute of Engineering and Technology, Faridabad, India

[2] Professor, Department of Electronics and Communication Engineering Rawal Institute of Engineering and Technology, Faridabad, India

[3] Professor, Department of Electronics and Communication Engineering Rawal Institute of Engineering and Technology, Faridabad, India

---***---

Abstract - *Removing noise from audio signal is a challenging task. Although different techniques exist but since sources of noise is also unlimited so effectiveness of these algorithms is not perfect. In this paper, an audio de-noising technique based on wavelet transformation is proposed. De-noising will be performed in the transformation domain and the improvement in de-noising is achieved by using various types of wavelet transform. The comparison among 2 types of wavelets families: Daubechies and Haar is performed. The audio quality of de-noised signals is determined based on mean square error, peak signal to noise ratio and cross correlation. Based on this research it is observed that Daubechies 10 provides best results compared to others.*

Key Words: Audio denoising; Discrete wavelet Transform; Peak Signal to Noise Ratio; Mean Square Error.

1.INTRODUCTION

Noise reduction in speech signals is a field of study to recover an original signal from its noise corrupted signal. The noise can be of various types like white, impulsive or even other types of noise usually found in speech signals. Over the past decades, the removal of this noise from audio signals has become an area of interest of several researchers around the world, since the presence of noise can significantly degrade the quality and intelligibility of these signals. Many studies have been conducted since the sixties, with the goal of developing algorithms for improving the quality of audio and speech. Michael T. Johnson et al. [1] 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. Mohammed Bahoura and Jean Rouat [2] have proposed a new speech enhancement method based on time and scale adaptation of wavelet thresholds. Ching-Ta and Hsiao-Chuan Wang [3] have proposed a method based on critical-band decomposition which converts a noisy signal into wavelet coefficients (WCs), and enhanced the WCs by subtracting a threshold from noisy WCs in each subband. Time frequency based analysis of speech signal has been introduced by Marián Képesia and Luis Weruaga [4]. In this Short-Time Fan-Chirp Transform (FChT) was defined univocally. Nanshan Li et al. [5] 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 trigger energy operator in order to progress the effect of the content-based songs retrieval system. Eric Martin [6] have introduces an adaptive audio block thresholding algorithm. The denoising parameters are computed according to the time-frequency regularity of the audio signal using the SURE (Stein Unbiased Risk Estimate) theorem. B. JaiShankar and K. Duraiswamy [7] have introduced the noises present in communication channels are disturbing and the recovery of the original signals from the path without any noise is very difficult task. This is achieved by denoising techniques that remove noises from a digital signal. C Mohan Rao, Dr. B Stephen Charles, Dr. M N Giri Prasad [8] has presented a new adaptive filter whose coefficients are dynamically changing with an evolutionary computation algorithm and hence reducing the noise.

Based on this literature survey it is clear that many approaches exist for audio denoising but still there is lack of general purpose audio denoising techniques. In order to improve this wavelet based approached is used.

Rest of paper organized as section II represents detailed methodology and section III represents experimental results and finally section IV concludes the paper.

2. METHODOLOGY USED

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

1. **Electronic noise** such as thermal noise and shot noise.
2. **Acoustic noise** emanating from moving, vibrating or colliding sources such as revolving Machines, moving vehicles, keyboard clicks, wind and rain.
3. **Electromagnetic noise** that can interfere with the transmission and reception of voice, image and data over the radio-frequency spectrum.
4. **Electrostatic noise** generated by the presence of a voltage.
5. Communication channel distortion and fading and
6. Quantization noise and lost data packets due to network congestion

Depending on its frequency, spectrum or time characteristics, a noise process [11-13] is further classified into several categories:

i.**White noise**: Purely random noise has an impulse autocorrelation function and a flat power spectrum. White noise theoretically contains all frequencies in equal power.

ii**. Band-limited white noise**: Similar to white noise, this is a noise with a flat power spectrum and a limited bandwidth that usually covers the limited spectrum of the device or the signal of interest. The autocorrelation of this noise is sinkshaped.

iii. **Narrowband noise**: It is a noise process with a narrow bandwidth such as 50/60 Hz from the electricity supply.

iv. C**olored noise**: It is non-white noise or any wideband noise whose spectrum has a non-flat shape. Examples are pink noise, brown noise and autoregressive noise.

v.**Impulsive noise**: Consists of short-duration pulses of random amplitude, time of occurrence and duration.

vi. **Transient noise pulses:** Consist of relatively long duration noise pulses such as clicks, burst noise etc. The detailed model used in this thesis is described in this section. For simulating this MATLAB 2015 [13] is used. The steps are as shown in the Figure 1.

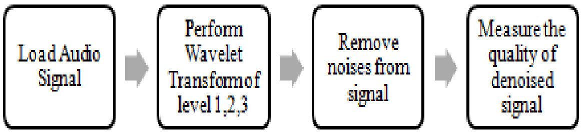

Figure 1 : Steps used for audio denoising.

The audio signal is subjected to many kinds of distortions as it passes through stages such as processing, compressing and transmitting. Measuring the quality of the image is a complicated task as human opinion is affected by physical and psychological parameters. There are many techniques for measuring the quality of the signal. MSE, PSNR, SNR [14] are the most commonly used audio quality measures.

PSNR is an abbreviation for peak signal-to-noise ratio (PSNR). It is defined as the ratio between maximum possible power of a signal and the power of distorting noise that affects the quality of the signal. The PSNR is usually expressed in terms of logarithmic decibel scale using eq. 1.

The mathematical representation of the PSNRis as follows:

$$PSNR : \text{......................... Eq. 1}$$

where the MSE (Mean Squared Error) is:

$$MSE = (\frac{1}{(m*n)} * sum(sum)\text{.............Eq. 2}$$

f represents the matrix of original signal
g represents the matrix of degraded signal
m represents the number of rows of signal
n represents the number of columns of signal
MAXf is the maximum signal value that exists in original signal which is known to be good signal.

3. Experimental Results

The experiments are performed in MATLAB using wave menu simulator available in MATLAB. The detailed method is described in chapter three. The sample processing is shown in Figure 2.

.

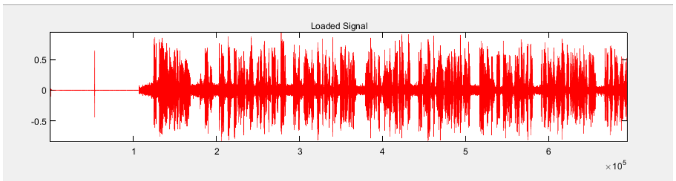

Figure 2(A) : Input Signal

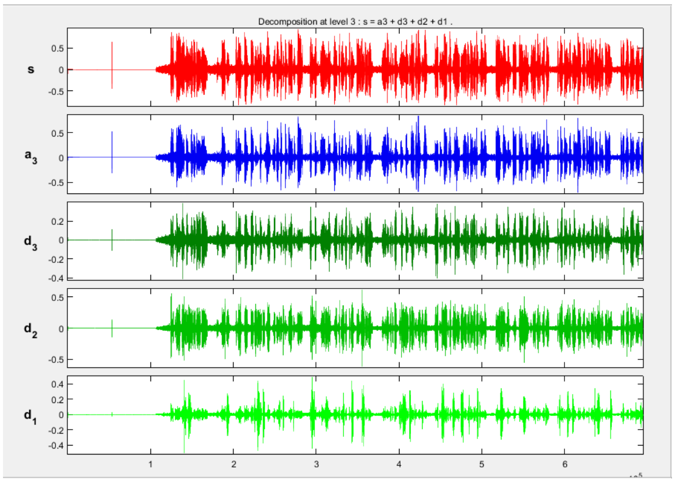

Figure 2(B) : Analysis of input signal (Haar Wavelet)

The Haar wavelet decomposition is shown in Figure 3.

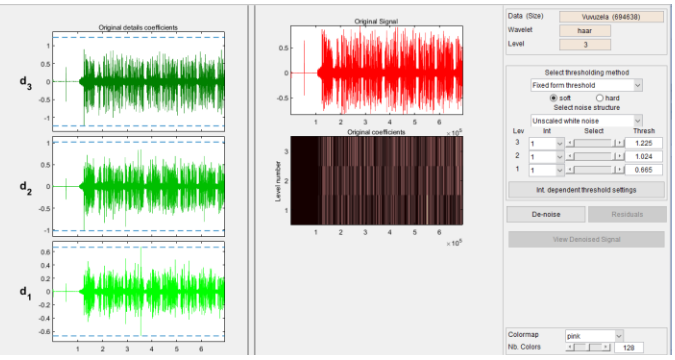

Figure 3 : Haar wavelet decomposition of signal.

Detailed coefficient is shown in Figure 4.

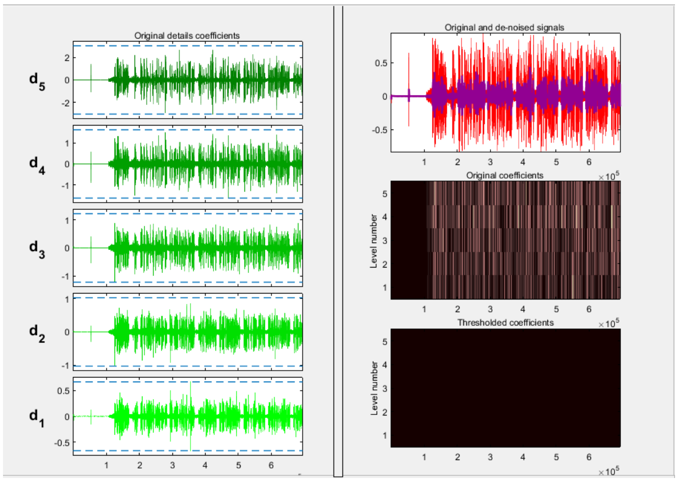

Figure 4 : Detailed Coefficients of De-noised signal (Haar Wavelet)

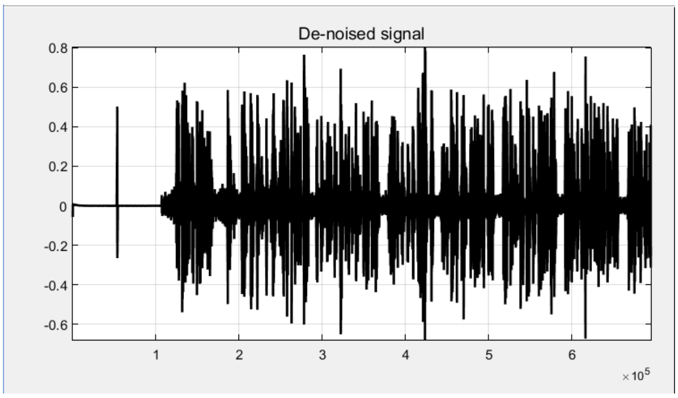

Figure 5 : De-noised signal (Haar Wavelet)

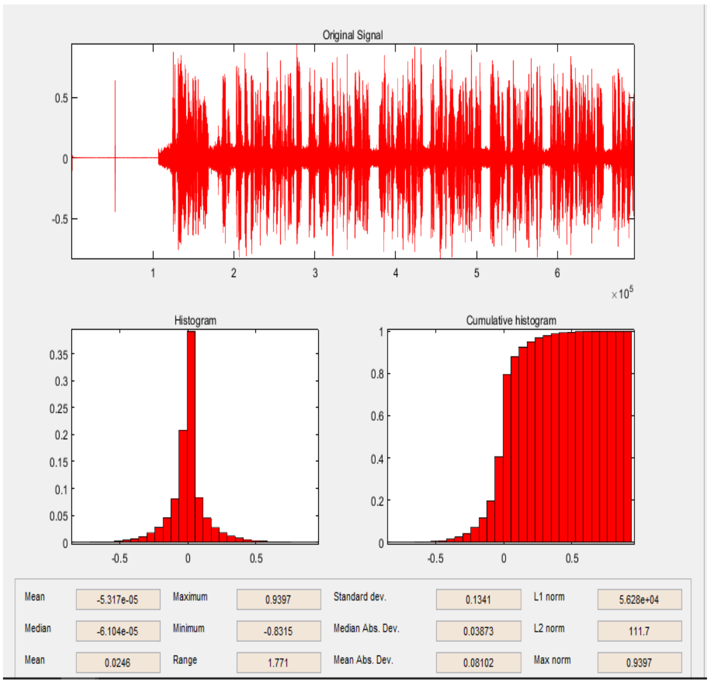

Figure 6 : Statistical analysis of signals (Haar Wavelet)

Figure 5 shows the de-noised signal. The Figure 6 shows the statistical analysis of input signal.

Similarly we perform this for Daubechies Wavelet. Further the input signal being the same we perform analysis of the input signal is shown in Figure 7 by using Daubechies (db) Wavelet.

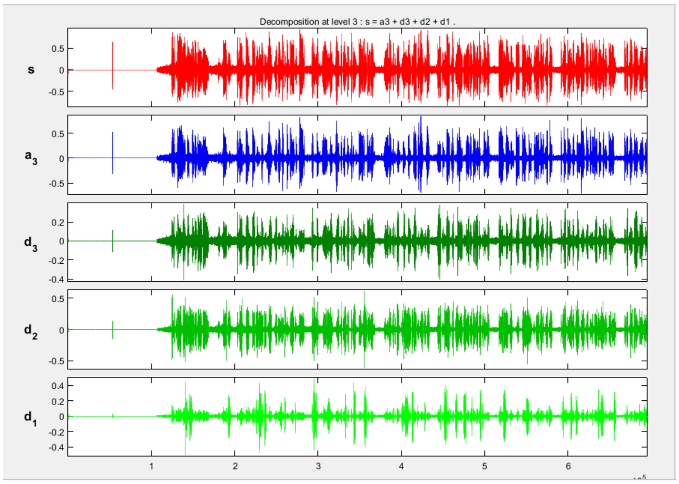

Figure 7 : Analysis of input signal (Daubechies Wavelet)

The db wavelet decomposition is shown in Figure 8.

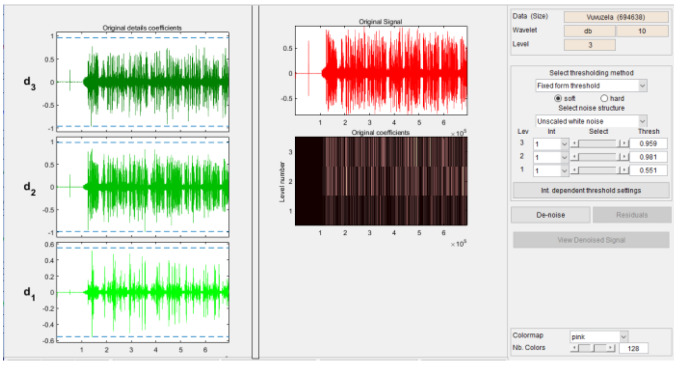

Figure 8 : Daubechies Wavelet decomposition of the signal

Detailed coefficients are shown in Figure 9.

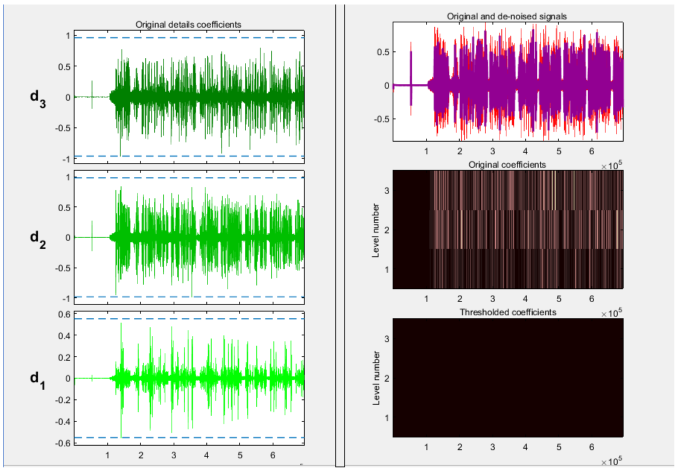

Figure 9 : Detailed Coefficients of De-noised Signal (dbWavelet)

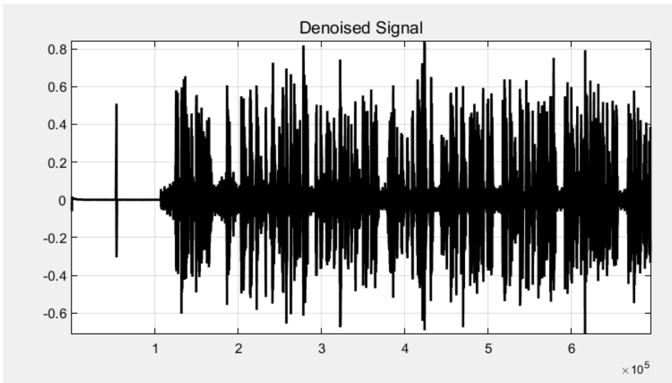

Figure 10 : De-noised Signal (db Wavelet)

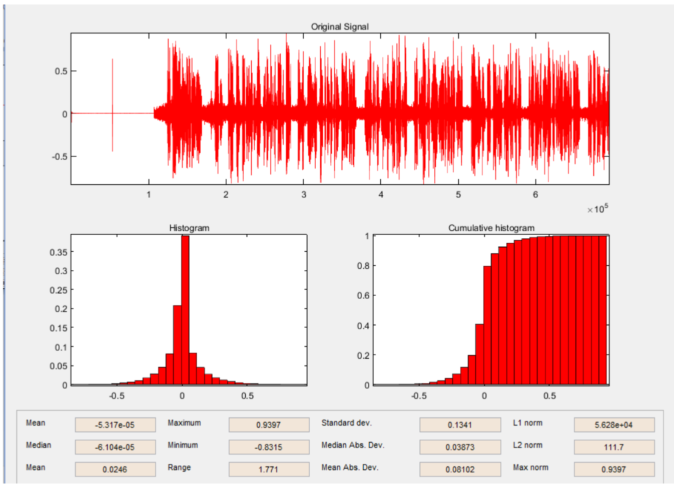

Figure 11 : Statistical Analysis of Signal (db Wavelet)

Figure 10 depicts the de-noised signal. The Figure 11marks the statistical analysis of input signals.

3. CONCLUSIONS

In this research the practical approach of how to put wavelets into practice for a noisy audio data to improve clarity and signal retrieval has been discussed. Fourier transform assumes the signal is stationary, but audio signal is always non-stationary. To overcome this wavelet is used. The comparison among Daubechies and Haar wavelet is performed. The audio quality of de-noised signals is determined based on mean square error, peak signal to noise ratio and cross correlation. Daubechies 10 performs the best result because they have a peak signal to noise ratio.

Further to the scope of this research we can analyse the signal and de-noise it by using various other wavelets available for example: Coiflets , Biorthogonal , Symlets, BiorSplines etc.

REFERENCES

[1] Michael T. Johnson, Xiaolong Yuan and Yao Ren, 2007. "Speech Signal Enhancement through Adaptive Wavelet Thresholding", Speech Communications, Vol. 49, No. 2, pp: 123-133.

[2] Mohammed Bahoura and Jean Rouat, 2006. "Wavelet speech enhancement based on time–scale adaptation", Speech Communication, Vol. 48, No. 12, pp: 1620-1637.

[3] Ching-Ta and Hsiao-Chuan Wang, 2003. "Enhancement of single channel speech based on masking property and wavelet transform", Speech Communication, Vol. 41, No 2-3, pp: 409- 427.

[4] Marián Képesia and Luis Weruaga, 2006. "Adaptive chirp-based time–frequency analysis of speech signals", Speech Communication, Vol. 48, No. 5, pp: 474-492.

[5] Nanshan Li and Mingquan Zhou, 2008. "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, pp: 787-790.

[6] Eric Martin," Audio denoising algorithm with block thresholding Published in Image Processing On Line on ISSN 2105-1232.

[7] B. JaiShankar1 and K. Duraiswamy —audio denoising using wavelet transform International Journal of Advances in Engineering & Technology, Jan 2012. ISSN: 2231-1963

[8] C Mohan Rao1, Dr. B Stephen Charles —A Variation of LMS Algorithm for Noise Cancellation" International Journal of Advanced Research in Computer and Communication Engineering Vol. 2, Issue 7, July 2013 ISSN (Print) : 2319-5940 .

[9] Sharma, Kavita, and Prateek Haksar. "Speech Denoising Using Different Types of Filters." *International Journal of Engineering Research and Applications Vol* 2 (2012): 809-811.

[10] Pižurica, Aleksandra, et al. "A versatile wavelet domain noise filtration technique for medical imaging." *Medical Imaging, IEEE Transactions on* 22.3 (2003): 323-331.

[11] Cano, Pedro, et al. "A review of algorithms for audio fingerprinting." *Multimedia Signal Processing, 2002 IEEE Workshop on*. IEEE, 2002.

[12] JaiShankar, B., and K. Duraiswamy. "AUDIO DENOISING USING WAVELET TRANSFORM." *International Journal of Advances in Engineering & Technology* 2.1 (2012): 419.

[13] http://in.mathworks.com/ [MATLAB 2015]

[14] Bilal, Ifra, and Rajiv Kumar. "Audio Steganography using QR Decomposition and Fast Fourier Transform." *Indian Journal of Science and Technology* 8.1 (2015).