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Audio Noise Reduction using Discrete Wavelet Transformation

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***Abstract* - Discourse signal investigation is one of the critical ranges of exploration in mixed media applications. Computerized channels successfully decrease the undesirable higher or lower request recurrence segments in a discourse signal. In this paper the discourse upgrade is performed utilizing diverse advanced channels. Essential straight channels and DWT with thresholding and sorts of wavelet are utilized to denoised the sound flags and improve discourse and sound sign quality. Our fundamental target is to decrease clamor from framework which is intensely reliant on the particular connection and application. As, we need to expand the understandability or enhance the general discourse discernment quality. Subsequent to considering and examining, we have reasoned that Noise lessening innovation is gone for decreasing undesirable encompassing sound, and is executed through two distinct systems with parameters, for example, Noise SNR, Denoise SNR and the Time to diminish the clamor for loud flags for uprooting commotion. We can dissect the denoised flag by sign to clamor proportion (SNR), Threshold values and slipped by time investigation. In the DWT db10 wavelet with delicate limit is best right now db10 soft edge, in DWT delicate edge results are has been best when contrasted with hard limit.**

Keywords: Noise SNR, De-noise SNR, DWT Denoising, db10, Thresholding

I. INTRODUCTION

Sound commotion lessening framework is the framework that is utilized to expel the clamor from the sound signs. Sound commotion lessening frameworks can be partitioned into two essential methodologies. The principal methodology is the corresponding sort which includes compacting the sound flag in some very much characterized way before it is recorded (essentially on tape). On playback, the resulting adjusting augmentation of the sound sign which restores the first element range, in the meantime has the impact of drawing nearer the repeat tape commotion (included amid recording) more remote beneath the top sign level—and ideally underneath the limit of hearing. The second approach is the single-finished or non-integral sort which uses systems to lessen the clamor level officially display in the source material—basically a playback just commotion lessening framework [4]. This methodology is utilized by the LM1894 incorporated circuit, composed extraordinarily for the decrease of perceptible clamor in practically any sound source. Clamor lessening is the procedure of expelling commotion from a sign. All soundtrack gadgets, both simple or advanced, have characteristics which make them helpless against commotion. Commotion can be irregular or background noise no dependability, or steady clamor presented by the gadget's component or handling calculations. There is an Active commotion control (ANC), otherwise called clamor scratch-off, or dynamic commotion decrease (ANR), is a technique for lessening undesirable and natural sound by the expansion of a second solid particularly intended to wipe out the first [7]. Sound is a weight wave or we can say sound is the simple flags that are prepared by recurrence, which comprises of a pressure stage and a rarefaction stage.

A. TYPES OF NOİSES

There are many types and sources of noise or distortions and they include:

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 [10]. Depending on its frequency, spectrum or time characteristics, a noise process is classified into several categories:

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) 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 sinc-shaped.
3)Narrowband noise: It is a noise process with a narrow bandwidth such as 50/60 Hz from the electricity supply.
4)Colored 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.
5)Impulsive noise: Consists of short-duration pulses of random amplitude, time of occurrence and duration.
6)Transient noise pulses: Consist of relatively long duration noise pulses such as clicks, burst noise etc.

B. *INTRODUCTION TO WAVELET TRANSFORM*

Wavelet transform consists of a set of basic functions that can be used to analyze signals in both time and frequency domains simultaneously. This analysis is accomplished by the use of a scalable window to cover the time-frequency plane, providing a convenient means for the analyzing of non-stationary signal that is often found in most application [8].

Wavelet analysis adopts a wavelet prototype function known as the mother wavelet given as:

$$\psi(\tau, s) = \frac{1}{\sqrt{s}}\psi\left(\frac{t-\tau}{s}\right) \tag{1.4}$$

This mother wavelet in turns generates a set of basic functions known as child wavelets through recursive scaling and translation.

Where, s reflects the scale or width of a basis function,

τ is the translation that specifies its translated position on the time axis,

$\psi\left(\frac{t-\tau}{s}\right)$ is the mother wavelet,

$\frac{1}{\sqrt{s}}$ is the normalized factor used to ensure energy across different scale remains the same[10].

C. *DWT ALGORİTHM FOR SİGNAL ENHANCEMENT*

Step 1: Load an original wave signal.

Step 2: Noise is added to the original wave signal read in above step using the Gaussian noise and produces the noisy wave signal.

Step 3: The Gaussian original wave signal on which logarithmic transform is performed firstly.

$$\text{Log } J(x, y) = \log I(x, y) + \log \eta(x, y)$$

Step 4: A multilevel decomposition is performed on the log transformed signal using wavelet transform.

Step 5: Apply the wavelet types.

Step 6: Apply thresholding to the noisy coefficients using bayes shrinkage method.

Step 7: After the decomposed signal coefficients are thresholded using the thresholding technique, denoised image is reconstructed as IR(x,y) using inverse wavelet transforms- IDWT.

Now apply the filter based on statistics estimated from a local neighborhood around each pixel. Filter reconstructed image IR(x,y) according to following formula:

$$I(x,y) = \mu + \frac{(\sigma^2 - v^2)(I_s(x,y) - \mu)}{\sigma^2}$$

Where, μ is the local mean, σ2 the variance in 3x3 neighborhoods around each pixel and v2 is the average of all estimated variances of each pixel in the neighborhood.

Step 8: Take exponent of the signal obtained in above step and obtained the denoised signal.

Step 9: Now we get the denoised signal and different parameters.

II. RESULTS

The implementation and simulation of these filter and wavelet using different algorithm have been done using MATLAB environment and their responses have been studied. The different filters values are given below:

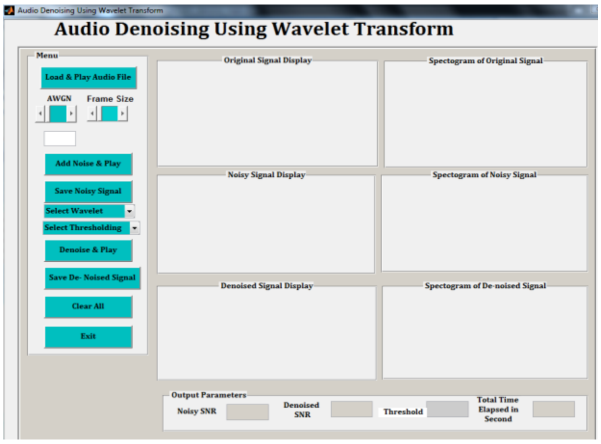

Figure 1: DWT GUI starting window

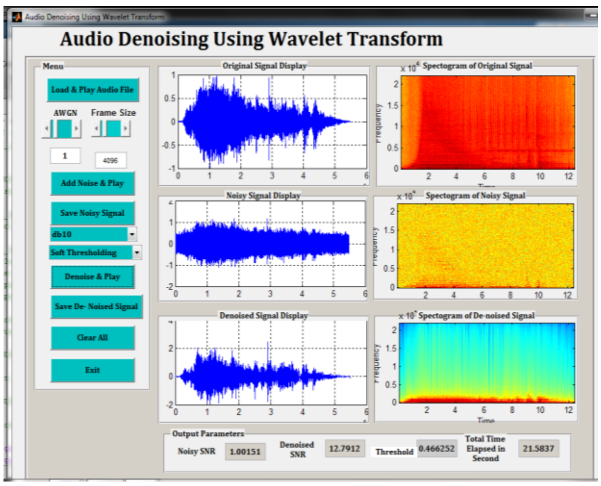

Figure 2: Denoised Signal with db10 wavelet

Table 1: Using Coif5 wavelet type with Soft Thresholding

Name of Signal	Noise SNR value	Denoisd SNR value	Threshold	Total Time Elapsed value(in sec.)
1N.wav	1.00741	10.1623	0.31724	9.5625
2N.wav	1.00339	10.5671	0.19695	6.0578
3N.wav	1.00151	12.6169	0.42760	17.995

Table 2 : Using Coif5 wavelet type with Hard Thresholding

Name of Signal	Noise SNR value	Denoised SNR value	Threshold	Total Time Elapsed value(in sec.)
1N.wav	0.998506	9.40588	0.18301	6.7981
2N.wav	0.999931	8.83714	0.17812	5.5622
3N.wav	0.994037	10.5652	0.41943	18.410

Table 3 : Using sym4 wavelet type with soft Thresholding

Name of Signal	Noise SNR value	Denoised SNR value	Threshold	Total Time Elapsed value(in sec.)
1N.wav	1.00741	10.3243	0.32104	7.15381
2N.wav	1.00339	10.9	0.19967	5.82927
3N.wav	0.994037	10.5805	0.41939	20.5543

Table 4 : Using sym4 wavelet type with Hard Thresholding

Name of Signal	Noise SNR value	Denoised SNR value	Threshold	Total Time Elapsed value(in sec.)
1N.wav	0.998506	9.50313	0.28225	6.7188
2N.wav	0.999931	9.04367	0.24028	5.1908
3N.wav	1.00151	10.6222	0.41898	19.319

Table 4 : Using sym8 wavelet type with soft Thresholding

Name of Signal	Noise SNR value	Denoised SNR value	Threshold	Total Time Elapsed value(in sec.)
1N.wav	1.00741	10.3255	0.39351	7.1942
2N.wav	1.00339	11.2604	0.27785	5.8679
3N.wav	1.00151	12.7862	0.45529	20.144

Table 4 : Using sym8 wavelet type with Hard Thresholding

Name of Signal	Noise SNR value	Denoised SNR value	Threshold	Total Time Elapsed value(in sec.)
1N.wav	0.99850	9.48901	0.27542	6.771
2N.wav	0.99993	8.8662	0.19024	5.289
3N.wav	0.99403	10.5758	0.43073	22.24

Table 4 : Using db9 wavelet type with soft Thresholding

Name of Signal	Noise SNR value	Denoised SNR value	Threshold	Total Time Elapsed value(in sec.)
1N.wav	1.00741	10.3475	0.39005	7.26304
2N.wav	1.00339	11.1282	0.26310	5.871
3N.wav	1.00151	12.7492	0.45050	18.9165

Table 4 : Using db9 wavelet type with Hard Thresholding

Name of Signal	Noise SNR value	Denoised SNR value	Threshold	Total Time Elapsed value(in sec)
1N.wav	0.998506	9.53768	0.33744	7.6531
2N.wav	0.999931	9.02351	0.25700	5.4105
3N.wav	0.994037	10.5459	0.385582	17.011

Table 4 : Using db10 wavelet type with soft Thresholding

Name of Signal	Noise SNR value	Denoised SNR value	Threshold	Total Time Elapsed value(in sec.)
1N.wav	1.00741	10.4215	0.43597	7.24155
2N.wav	1.00339	11.3326	0.30263	5.93694
3N.wav	1.00151	12.7912	0.46625	21.5837

Table 4 : Using db10 wavelet type with Hard Thresholding

Name of Signal	Noise SNR value	Denoised SNR value	Threshold	Total Time Elapsed value(in sec.)
1N.wav	0.998506	9.54482	0.35058	6.6764
2N.wav	0.999931	8.815	0.16662	5.5559
3N.wav	0.994037	10.5432	0.39357	16.898

III. CONCLUSION AND FUTURE WORK

We used wavelet transform for denoising speech signal corrupted with Gaussian noise. Speech denoising is performed in wavelet domain by different types of wavelet with different thresholding. By using this we can get the better results of de-noising, especially for low level noise. During different analysis we found that soft thresholding is better than hard thresholding because soft thresholding gives better results than hard thresholding. Higher threshold removes noise well, but the part of original signal is also removed with the noise. It is generally not possible to filter out all the noise without affecting the original signal. We can analyze the denoised signal by signal to noise ratio (SNR), Threshold values and elapsed time analysis. In the DWT Coif5, db9, db10, sym4 and sym8 wavlet with hard threshold and soft threshold is implemented and compared with each others. In this db10 wavelet with soft threshold is best as compared to other wavelet. In DWT soft threshold results are has been best as compared to hard threshold.

Future work might involve a real time implementation of the system so that the maximum noise is reduced form the audio signals and videos. In the future anybody can extent the order of the different filters and works on higher amplitude signals. They can calculate the efficiency of the filters that they have to implement. In the DWT we are using coif5,db9,db10, sym4 and sym8 with hard and soft threshold but in the future different types of wavelet is implemented with different types of thresholding techniques or hybrid techniques is designed with the help of filters and wavelets and thresholding techniques. Other things in future the results may be improved in the filters and DWT techniques.

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