RESEARCH ARTICLE OPEN ACCESS

Audio Noise Reduction from Audio Signals and Speech Signals

Nishan Singh[1], Dr. Vijay Laxmi[2]
Research Scholar[1], Associate Professor[2]
GKU, Talwandi Sabo, Bathinda
Punjab - India

ABSTRACT

Speech signal analysis is one of the important areas of research in multimedia applications. Discrete Wavelet technique is effectively reduces the unwanted higher or lower order frequency components in a speech signal. Wavelet-based algorithm for audio de-noising is worked out. We focused on audio signals corrupted with white Gaussian noise which is especially hard to remove because it is located in all frequencies. We use Discrete Wavelet transform (DWT) to transform noisy audio signal in wavelet domain. It is assumed that high amplitude DWT coefficients represent signal, and low amplitude coefficients represent noise. Using thresholding of coefficients and transforming them back to time domain it is possible to get audio signal with less noise. Our work has been modified by changing universal thresholding of coefficients which results with better audio signal. In this various parameters such as SNR, Elapsed Time, and Threshold value is analyzed on various types of wavelet techniques alike Coiflet, Daubechies, Symlet etc. In all these, best Daubechies as compared to SNR is more for Denoising and Elapsed Time is less than others for Soft thresholding. In using hard thresholding Symlet wavelet also works better than coiflet and Daubechies is best for all. Efficiency is 98.3 for de-noising audio signals which also gives us better results than various filters.

Keywords:- white Gaussian noise; Thresholding, Speech enhancement, DWT, coefficients, Daubechies

I. INTRODUCTION

Wavelet Filters are the manipulation of the amplitude and/or phase response of a signal according to their frequency. These are the basic components of all signal processing and -telecommunication systems. There are two kinds of wavelet filters- fixed and tunable. Fixed filters are those in which passband frequencies and stopband frequencies are fixed whereas in case of tunable filters, passband and stopband frequencies are variable. These frequencies can be changed according to the requirement of the applications. Tunable digital wavelet filters are widely employed in telecommunications, medical electronic, digital audio equipment and control systems. These wavelet filters are also known as variable digital filters [17]. Tunable digital filters are used in telecommunication system in the front end of a receiver to select a particular band of frequencies. In medical electronics, tunable notch filters are used to suppress the power line interference [11].

The bases for the design of the tunable digital filters are the spectral transformation [8] [18]. It is basically used to modify the characteristics of a filter to meet new specifications without repeating the filter design procedure. This modification is done by changing a Low pass(LP) digital filters to Low pass(LP) filters with different cutoff frequencies or to a High pass(HP), Band pass(BP) or Band stop(BS) filters.

The variable Band pass (BP) and Band stop (BS) filters are used to eliminate and retrieve some narrow band signals.

AUDİO NOİSE REDUCTİON

Audio noise reduction system is the system that is used to remove the noise from the audio signals. Audio noise reduction systems can be divided into two basic approaches. The first approach is the complementary type which involves compressing the audio signal in some well-defined manner before it is recorded (primarily on tape). The second approach is the single-ended or non-complementary type which utilizes techniques to reduce the noise level already present in the source material—in essence a playback only noise reduction system [18]. This approach is used by the LM1894 integrated circuit, designed specifically for the reduction of audible noise in virtually any audio source. Noise reduction is the process of removing noise from a signal.

All recording devices, both analogue or digital, have traits which make them susceptible to noise. Noise can be random or white noise with no coherence, or coherent noise introduced by the device's mechanism or processing algorithms. Their is a Active noise control (ANC), also known as noise cancellation, or active noise reduction (ANR), is a method for reducing unwanted and unprocessed sound by the addition of a second sound specifically designed to cancel the first[26]. Sound is a pressure wave or we can say sound is the analog signals that are processed according to their frequency, which consists of a compression phase and a rarefaction phase. A noise-cancellation speaker emits a sound wave with the same amplitude but with inverted phase (also known as anti phase) to the original sound. The waves combine to form a new wave, in a process called interference, and effectively cancel each other out - an effect which is called phase cancellation.

Modern active noise control is generally achieved through the use of analog circuits or digital signal processing. An Adaptive algorithms are designed to analyze the waveform of the background no neural noise, then based on the specific algorithm generate a signal that will either phase shift or invert the polarity of the original signal. This anti phase is then amplified and a transducer creates a sound wave directly proportional to the amplitude of the original waveform, creating destructive interference [8]. This effectively reduces the volume of the perceivable noise. The transducer emitting the noise cancellation signal may be located at the location where sound attenuation is wanted (e.g. the user's ear/any music/headphone sound). This requires a much lower power level for cancellation but is effective only for a single user.

II. FLOW OF WORK CARRIED OUT

Algorithm:
Discrete Wavelet transforms types filter Algorithm

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.
Log J(x, y) = log I(x, y) + log ɳ(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 Soft or Hard 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 $I_R(x,y)$ using inverse wavelet transforms- IDWT.

Now apply the filter based on statistics estimated from a local neighborhood around each pixel. Filter reconstructed image $I_R(x,y)$ 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, σ^2 the variance in 3x3 neighborhoods around each pixel and v^2 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.
Step10: Stop.

Results:
Results with GUI images is shown below and analyzed in which various parameters noisy SNR, Denoised SNR, Elapsed time, Efficiency, Threshold values are calculated.

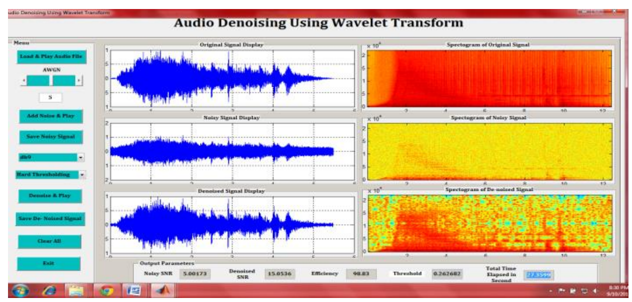

Figure 1: window with Denoised signal using db9 technique at Hard threshold Technique

Db9 type wavelet for de-noising is shown in above figure with Hard threshold for selecting and the denoised spectrogram signal and sound signal is displayed with parameters values.
meters values.

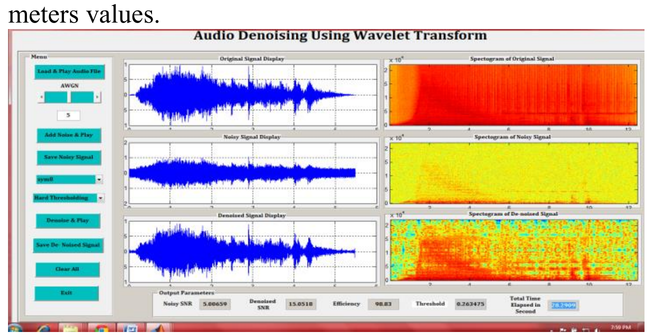

Figure 2: window with Denoised signal using sym8 technique at Hard threshold Technique

Sym4 type wavelet for de-noising is shown in above figure with Hard threshold for selecting and the denoised spectrogram signal and sound signal is displayed with parameters values.

Table 1: Comparison of wavelet transforms techniques with AWGN as 5 at Soft threshold Technique

Sr. No.	Types of Wavelet	Noisy SNR	Denoised SNR	Total Elapsed time in seconds	Threshold
1	Coif5	4.00151	12.8456	32.8652	0.294682
2	Db9	5.00151	13.1161	33.7916	0.270804
3	Db10	4.99404	13.5224	28.9395	0.247432
4	Sym4	5.00659	13.2289	28.3885	0.243985
5	Sym8	5.00659	13.2202	27.7899	0.263475

Above table 1 shows the Noisy SNR value, Denoised SNR, Total Elapsed Time ad Threshold Value. In above, comparison is made between various wavelet types filter in which Coif5 has less Denoised SNR value. Db9 & Db10 is best compared to all other wavelet filters shown as in various parameter values such as Denoised SNR and Total Elapsed time i.e reduced time for Soft thresholding type.

Table 2: Comparison of wavelet transforms techniques with AWGN as 5 at hard threshold technique

Sr. No.	Types of Wavelet	Noisy SNR	Denoised SNR	Total Elapsed time in seconds	Threshold
1	Coif5	4.99911	14.9132	42.2529	0.224576
2	Db9	5.00173	15.0536	27.3599	0.262682
3	Db10	4.99452	15.0841	28.1884	0.251217
4	Sym4	5.00548	14.7002	28.3055	0.231492
5	Sym8	5.00368	15.0206	29.1295	0.268752

Above table 2 shows the Noisy SNR value, Denoised SNR, Total Elapsed Time ad Threshold Value. In above, comparison is made between various wavelet types filter in which Coif5 and Sym4 has less Denoised SNR value. Db9, Db10 & Sym8 is best compared to all other wavelet filters shown as in various parameter values such as Denoised SNR and Total Elapsed time i.e reduced time for Hard thresholding type.

III. CONCLUSION & FUTURE WORK

From the above results the DWT Coif wavlet with hard threshold and soft threshold and Sym4 hard and soft threshold is implemented and compared with each others. In this Coif wavelet with soft threshold is best as compared to coif hard threshold and Sym4 wavelet with hard and soft threshold.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 coif and sym4 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.

REFERENCES

[1] Abdulmunim Matheel E., Abass Rabab F. "Novel Video Denoising Using 3-D Transformation Techniques "International Journal of Engineering and Advanced Technology (IJEAT) ISSN: 2249 – 8958, Volume-2, Issue-5, June 2013 .

[2] Aggarwal Rajeev" Noise Reduction of Speech Signal using Wavelet Transform with Modified Universal Threshold "International Journal of Computer Applications (0975 – 8887) Volume 20– No.5, April 2011.

[3] Anju , Mamta Katiyar " Design of Butterworth and Chebyshev1 Lowpass Filter for Equalized Group Delay" International Journal of Advanced Research in Computer Science and Software Engineering, Volume 2, Issue 5, ISSN: 2277 128X, May 2012.

[4] Chakraborty Subhadeep,etl." Design of IIR Digital Highpass Butterworth Filter using Analog to Digital Mapping Technique" International Journal of Computer Applications (0975 – 8887) Volume 52 – No. 7, August 2012.

[5] CHANG S, KWON Y,YANG S,et al. "Speech enhancement fornon-stationary noise environment by adaptive wavelet packet".Proceedings of IEEE International Conference on Acoustics Speech and Signal Processing,2002:61-564.

[6] Direkoglu C., Nixon M. S., "Image based multiscale shape description using Gaussian filter", IEEE Indian Conference on Computer Vision, Graphics and Image Processing 673–678,May 2008.

[7] Direkoglu C., Nixon M. S., "Shape classification using multiscale Fourier-based description in 2-D space", IEEE International Conference on Signal Processing 820–823,June 2008.

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

[9] Jebastine J. , Rani Dr. B. Sheela "design and implementation of noise free Audio speech signal using fast block least Mean square algorithm" Signal & Image Processing : An International Journal (SIPIJ) Vol.3, No.3, June 2012.

[10] Kumar R.senthill , 'Performance Improvement in the bivariate models by using Modified Marginal Varince of noisy observation for image-denoising applications',World Academy of Science,Engineering and Technology, ISSN 1307-6884 Volume 5, April 2005.

[11] Martin Eric,*" Audio denoising algorithm with block thresholding*" Published in Image Processing On Line on YYYY {MM {DD.ISSN 2105-1232.

[12] Moon. Sang-Hyun "Importance of Phase Information in Speech Enhancement". 2010 International conference on complex, intelligentand software intensive systems, page :770-773.

[13] Murphy J. and Godsill S., "Joint bayesian removal of impulse and background noise," in Acoustics, Speech and Signal Processing

(ICASSP),2011 IEEE International Conference on. IEEE, 2011, pp. 261–264.

[14] Niedzwiecki M. and Ciolek M., "Elimination of impulsive disturbances from archive audio signals using bidirectional processing," Audio, Speech, and Language Processing, IEEE Transactions on, vol. 21, no. 5,pp. 1046–1059, 2013.

[15] OBULESU K.P., KUMAR P. UDAY "implementation of time frequency block thresholding algorithm in audio noise reduction "ISSN: 2278 – 7798 International Journal of Science, Engineering and Technology Research (IJSETR) Volume 2, Issue 7, July 2013 .

[16] Prabhakar Sujata, etl. "Characteristics of Tunable Digital Filters" International Journal of Advanced Research in Computer Science and Software Engineering, Volume 3, ISSN: 2277 128X, Issue 8, August 2013.

[17] Ramli Roshahliza M., Noor Ali O. Abid, and Samad Salina Abdul *"A Review of Adaptive Line Enhancers for Noise Cancellation"* Australian Journal of Basic and Applied Sciences, 6(6): 337-352, ISSN 1991-8178, 2012.