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Audio Noise Reduction Using Discrete Wavelet Transform and Filters

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Abstract
In present paper, audio noise reduction in '.wav' format audio using discrete wavelet transform and various filters was attempted. The sound in the format '.wav' in the given window is firstly uploaded. The sound is listened which appear to be noisy due to Gaussian Noise. In the GUI, the desired wavelet type filters & Non-Linear filters are chosen for de-noising. After selecting wavelet & non-linear filters, different types of threshold rules are also selected. The soft or hard threshold is selected. Finally, parameters values with all sound playing such as original, Noisy & De-noised sound are obtained. This window shows filtered sound along with the graphs and various details about the parameters such as Noisy SNR, De-noised SNR, Elapsed Time, Efficiency, and Threshold values.

Keywords
Gaussian Noise; SNR; Threshold; GUI; Filters; De-noising.

I. Introduction
Noise control is an active or passive means of reducing sound emissions for personal comfort, environmental considerations or legal compliance. Active noise control is sound reduction using a power source. Passive noise control is sound reduction by noise-isolating materials such as insulation, sound-absorbing tiles, or a muffler rather than a power source.

Filters

Filters are networks that process signals in a frequency-dependent manner. The basic concept of a filter can be explained by examining the frequency dependent nature of the impedance of capacitors and inductors [1]. Filters can be classified as Median filters and FIR filters. **Median filtering** is a common step in image processing and audio de-noising. Median filter is a well-used nonlinear filter. The median filter is also called the order specific filter because it is based on statistics derived from ordering the elements of a set rather than taking the means. This filter is popular for reducing noise. **FIR filters** are digital filters with finite impulse response. They are also known as non-recursive digital filters as they do not have the feedback (a recursive part of a filter), even though recursive algorithms can be used for FIR filter realization.

Types of noise

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 machines, moving vehicles, keyboard clicks, wind and rain.
3. Electromagnetic noise that can interfere with the transmission and reception of voice.
4. 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. Literature Review
Rao and Charles (2013) designed a new adaptive filter whose coefficients are dynamically changing with an evolutionary computation algorithm and hence reducing the noise.

Obulesu and Kumar (2013) developed a non-diagonal method is used in which Block parameters are automatically adjusted to the nature of the audio signal by minimizing a Stein estimator, calculated analytically from noisy signal values.

Sudha et al. (2009) proposed translation invariant with Wiener filter to solve the shift variance problem and use this method to de-noise a noisy video with Gaussian white noise.

Sharma and Pyara (2013) proposed a robust DWPT based adaptive bock algorithm with modified threshold for de-noising the sounds of musical instruments shehnai, dafli and flute.

Sampat and Vithalani (2013) developed a noisy speech wav file having additive white Gaussian noise, used to de-noised the features of two stage hard threshold, soft threshold.

Shankar and Duraiswamy (2012) proposed an audio de-noising technique based on wavelet transformation is proposed.

III. Proposed Methodology
Wav file format

Waveform Audio File Format (WAVE, or more commonly known as WAV due to its filename extension),(also, but rarely, named, Audio for Windows) is a Microsoft and IBM audio file format standard for storing an audio bit stream on PCs. It is an application of the Resource Interchange File Format (RIFF) bit stream format method for storing data in "chunks", and thus is also close to the 8SVX and the AIFF format used on Amiga and Macintosh computers, respectively[7]. It is the main format used on Windows systems for raw and typically uncompressed audio. The usual bit stream encoding is the linear pulse-code modulation (LPCM) format [2].

Wavelet Based De-noising System

There has been a fair amount of research on wavelet-based image denoising. The paper published by Donoho and Johnstone (1994), developed a theoretical framework for denoising signals using Discrete Wavelet Transform (DWT).The method consists of applying the DWT to the original data, thresholding the detailed wavelet coefficients and inverse transforming the set of threshold coefficients to obtain the de-noised signal.

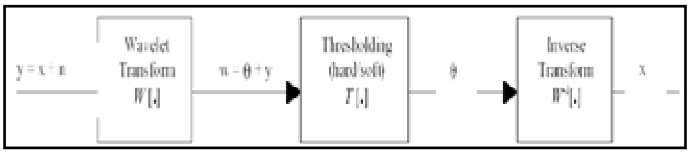

Fig 1: Block diagram of DWT based denoising framework

Given a noisy signal $y = x + n$ where x is the desired signal and n is independent and identically distributed (*i.i.d*) Gaussian noise $N(0,$

σ^2), y is first decomposed into a set of wavelet coefficients w = W[y] consisting of the desired coefficient q and noise coefficient n**.** By applying a suitable threshold value T to the wavelet coefficients, the desired coefficient q = T[w] can be obtained; lastly an inverse transform on the desired coefficient q will generate the denoised signal x=w^{-1}[ɵ] [10].

The steps of proposed methodology for the current work are detailed below.

1. Load an original wave signal.
2. Noise is added to the original wave signal read in above step using the Gaussian noise and produces the noisy wave signal.
3. The Gaussian original wave signal on which logarithmic transform is performed firstly. Log J(x, y) = log I(x, y) + log □(x, y)
4. Apply the median Filter or Adaptive FIR Filter as a wavelet technique.
5. Select the Threshold Rule form four types 'rigrsure', 'heursure', 'sqtwolog', 'minimaxi' to be applied on above step.
6. Apply Soft or Hard thresholding to the noisy coefficients using Bayes shrinkage method.
7. After the decomposed signal coefficients are threshold using the thresholding technique, de-noised image is reconstructed as $I_R(x, y)$ using inverse wavelet transforms-IDWT.
8. Take exponent of the signal obtained in above step and obtained the de-noised signal.
9. Now we get the de-noised signal and different parameters.
10. Stop.

Threshold Rules

Various thrushold rules to be followed are discussed below.

1. '**rigrsure**', adaptive threshold selection using principle of Stein's Unbiased Risk Estimate.
2. '**heursure**', heuristic variant of the first option.
3. '**sqtwolog**', threshold is sqrt (2*log(length(X))).
4. '**minimaxi**', minimax thresholding.

IV. Results And Discussions

The GUI in Fig 1 shows the audio de-noising screen with four buttons. First button for selecting to run the wavelet transform types filter as for loading sound. Same way second and third button is used for Selecting Median filter and FIR filter to run and load sound respectively. Fourth Button is used for Exiting from main window.

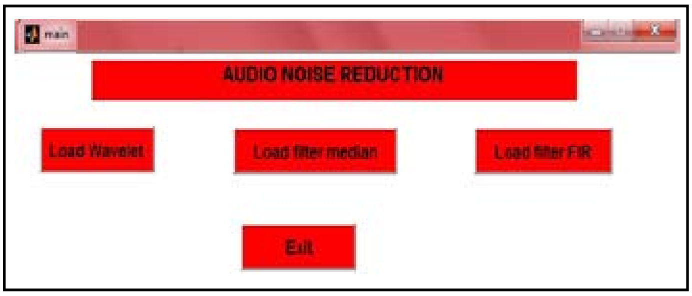

Fig 2: GUI window for audio denoising

The running programming during which threshold rule is selected from four types such as: **heursure, rigrsure, minimaxi, sqtwolog** for all wavelet type filter as well as this selection rule is applied on FIR and median filter. Table 1-4 shows the soft thresholding for heursure, rigrsure, minimaxi and sqtwolog respectively.

Table 1: Wavelet transforms techniques filters Soft threshold (Heursure)

Sr	Techniques Names		Noisy SNR	Denoised SNR	Total Elapsed time in seconds	Efficiency	**Error**
1		**Coif5**	4.9989	7.2745	1.025	9.883	7.9678
2		**Db13**	5.0056	7.2822	1.072	9.883	7.9583
3	**Wavelet types filters**	**Db40**	4.9815	7.3480	1.090	9.883	8.0815
4		**Sym13**	5.0063	7.2487	1.046	9.883	7.9963
5		**Sym21**	5.0011	7.3102	1.063	9.883	8.0631
6		**Haar**	4.9957	6.9848	1.020	9.883	3.8645
7	**Median Filter**		5.0113	-8.9215	1.031	2.436	56.65
8	**Adaptive Filter**		4.9959	-12.4716	9.913	2.424	92.9557

Table 2: Wavelet transforms techniques filters Soft threshold (rigrsure)

Sr.	**Techniques Names**		**Noisy SNR**	**Denoised SNR**	**Total Elapsed time in seconds**	**Efficiency**	**Error**
1		**Coif5**	5.0000	7.9505	1.0768	9.883	4.3448
2		**Db13**	4.3654	7.9763	1.0508	9.883	4.3654
3	**Wavelet types filters**	**Db40**	4.4580	7.9026	1.0475	9.883	4.4580
4		**Sym13**	5.0013	7.8660	1.2370	9.883	4.4683
5		**Sym21**	5.0097	7.9052	1.0598	9.883	4.3854
6		**Haar**	5.0162	7.0367	1.1773	9.883	3.8753
7	**Median Filter**		4.9948	-8.9012	1.0671	2.4396	56.4114
8	**Adaptive Filter**		5.0066	-12.4124	1.0068	2.4355	92.3099

Table 3: Wavelet transforms techniques filters Soft threshold (minimaxi)

Sr.	**Techniques Names**		**Noisy SNR**	**Denoised SNR**	**Total Elapsed time in seconds**	**Efficiency**	**Error**
1		**Coif5**	5.0004	5.7566	1.0404	9.883	11.8288
2		**Db13**	4.9961	5.7421	1.0234	9.883	11.2550
3	**Wavelet types filters**	**Db40**	4.9941	5.6982	1.0273	9.883	10.8930
4		**Sym13**	5.0053	5.6940	1.0404	9.883	11.4900
5		**Sym21**	5.0017	5.6985	1.0628	9.883	10.9019
6		**Haar**	4.9948	4.6878	1.0089	9.883	11.8180
7	**Median Filter**		5.0002	-2.4274	9.9603	3.9262	44.7721
8	**Adaptive Filter**		5.0044	-8.7479	1.0060	3.7747	79.8343

Table 4: Wavelet transforms techniques filters Soft threshold (sqtwolog)

Sr	**Techniques Names**		**Noisy SNR**	**Denoised SNR**	**Total Elapsed time in seconds**	**Efficiency**	**Error**
1		**Coif5**	5.0038	5.5238	1.0499	9.883	13.2221
2		**Db13**	5.0138	5.5213	1.0244	9.883	12.6952
3	**Wavelet types filters**	**Db40**	5.0007	5.4968	1.0401	9.883	12.1008
4		**Sym13**	5.0030	5.4977	1.0273	9.883	13.2518
5		**Sym21**	5.0002	5.5043	1.0409	9.883	12.3099
6		**Haar**	4.9992	4.5176	1.0217	9.883	13.3807
7	**Median Filter**		5.0001	-0.2024	1.0112	4.884	46.7132
8	**Adaptive Filter**		5.0074	-7.5908	9.9891	4.670	74.2744

Table 1 – Table 4 shows the noisy signal, de-noised signal, total elapsed time, efficiency & error. Results in the tables show the comparison between various wavelet filters. The efficiency of the nonlinear filters such as Median and adaptive filter is better. FIR filter is much better than median filter using all rules of threshold with soft thresholding technique as for as de-noised SNR is concerned .Median filter has best elapsed time.

Similarly table 5- table 8 shows the hard thresholding for heursure, rigrsure, minimaxi and sqtwolog respectively.

Table 5: Wavelet transforms techniques filters at hard threshold (Heursure)

Sr.	Techniques Names		Noisy SNR	Denoised SNR	Total Elapsed time in seconds	Efficiency	Error
1	Wavelet types filters	Coif5	5.0036	5.6984	1.0209	9.883	7.7119
2		Db13	4.9889	5.6558	1.0687	9.883	7.3189
3		Db40	5.0041	5.8447	1.0817	9.883	7.1366
4		Sym13	4.9953	5.7956	1.0691	9.883	7.2063
5		Sym21	4.9929	5.7819	1.0433	9.883	7.6671
6		Haar	4.9909	5.6148	1.0267	9.883	3.7119
7	Median Filter		5.0019	-10.1219	9.9459	2.2977	60.5725
8	Adaptive Filter		5.0075	-13.2547	1.0050	2.2864	85.7571

Table 6: Wavelet transforms techniques filters at Hard threshold (rigrsure)

Sr.	Techniques Names		Noisy SNR	Denoised SNR	Total Elapsed time in seconds	Efficiency	Error
1	Wavelet types filters	Coif5	4.9932	6.2422	1.0126	9.883	4.1840
2		Db13	4.9815	6.2506	1.0189	9.883	4.2276
3		Db40	4.9969	6.3403	1.0431	9.883	4.3007
4		Sym13	4.9992	6.3556	1.0232	9.883	4.3757
5		Sym21	4.9820	6.2223	1.0783	9.883	4.2796
6		Haar	5.0019	5.6129	1.0062	9.883	3.6953
7	Median Filter		4.9950	-10.1079	1.0081	2.299	55.1438
8	Adaptive Filter		5.0023	-13.2477	9.9748	2.287	86.8983

Table 7: Wavelet transforms techniques filters at Hard threshold (minimaxi)

Sr.	Techniques Names		Noisy SNR	Denoised SNR	Total Elapsed time in seconds	Efficiency	Error
1	Wavelet types filters	Coif5	4.9931	6.2370	9.9860	9.883	10.0516
2		Db13	4.9971	6.2162	1.0069	9.883	10.1564
3		Db40	4.9878	6.1427	1.0247	9.883	10.4425
4		Sym13	5.0057	6.1756	1.0099	9.883	9.9839
5		Sym21	5.0046	6.1563	1.0534	9.883	10.0051
6		Haar	5.0098	5.0658	1.0229	9.883	10.5136
7	Median Filter		4.9933	-8.5241	1.0233	2.5128	56.8349
8	Adaptive Filter		4.9997	-12.1923	9.8894	2.4794	87.4360

Table 8: Wavelet transforms techniques filters at Hard threshold (sqtwolog)

Sr.	Techniques Names		Noisy SNR	Denoised SNR	Total Elapsed time in seconds	Efficiency	Error
1	Wavelet types filters	Coif5	5.0037	5.7397	1.0549	9.883	13.6504
2		Db13	5.0009	5.7277	1.0166	9.883	11.7803
3		Db40	5.0090	5.6764	1.0715	9.883	12.0671
4		Sym13	4.9969	5.6925	1.0282	9.883	11.4247
5		Sym21	5.0101	5.6898	1.0489	9.883	11.3478
6		Haar	5.0034	4.6541	1.0067	9.883	12.7270
7	Median Filter		4.9958	-7.6250	1.0014	2.681	59.0814
8	Adaptive Filter		4.9955	-11.5455	1.0875	2.624	90.2294

The table 5- table 8 shows Noisy SNR value, De-noised SNR, Total Elapsed Time, Efficiency and Error. The results in the tables show the comparison between various wavelet types filter, the efficiency of the Non-Linear filters such as Median & Adaptive FIR is better. Adaptive FIR Filter is much better than Median Filter using all rules of threshold with hard thresholding technique as far as denoised SNR value is concerned. Finally it can be demonstrated that median filter has best total elapsed time i.e. reduced time for hard thresholding type. According to the rule selection, adaptive FIR filter takes more time to denoised signals in Heursure, Sqtwolog but rigrsure and minimaxi has less time elapsed than Median filter.

V. Conclusion and Future Work

Speech de-noising is performed in wavelet domain by different types of wavelet & Filters with different thresholding. During different analysis, 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. We can analyze the de-noised signal by signal to noise ratio (SNR), Efficiency and elapsed time analysis.

From the above results, the Adaptive FIR filter is the best as compared to the other filters because Denoised SNR is Low and ellapsed time is also less as compared to other filters but the Median Filter on other hand also results next to it. Wavelet Types filter results in efficency only as best but not through Denoised and Elapsed time both in soft and Hard thresholding technique used.

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 future, anybody can extent the order of the different filters and works on higher amplitude signals. In DWT we are using coif, Haar, Db and sym4 with hard and soft threshold but in future different types of wavelet & filters can be implemented with hybrid techniques of threshold rule. Other things in future the results may be improved in the filters and DWT techniques.

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