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Efficient Speech De-noising Algorithm using Multi-level Discrete Wavelet Transform and Thresholding

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ABSTRACT

Noise is introduced in speech signal due to various unavoidable reasons which makes the speech less intelligible. De-noising is an ongoing research area from past many decades. De-noising is achieved by different type of filtering techniques. Performance of the particular de-noising algorithm depends on the characteristics of the filter used. In this paper, an efficient speech denoising technique using Discrete Wavelet Transform (DWT) and thresholding is proposed. To get optimal de-noising, multilevel DWT is used. This method separates the noise components present in the noisy speech and further noise components are suppressed by thresholding of DWT coefficients. Clean speech signal is reconstructed by performing Inverse Discrete Wavelet Transform (IDWT). The results show that good quality de-noised signal is obtained using Haar wavelet with Mini-Maxi thresholding technique. On the other hand, low error rate is obtained using dB13 wavelet with Rigsure thresholding technique. DB13 and Sym13 wavelets with Rigsure thresholding technique provide good tradeoff between error rate and the quality of the de-noised signal.

Key words: Speech Processing, Discrete Wavelet Transform, Thresholding, Signal to Noise Ratio etc.

1. INTRODUCTION.

The processing of speech signal is a part of Digital Signal Processing where digital algorithms are used to process the speech. But any speech signal is analogous in nature which is digitized by suitable ADC architecture [1] for before processing. Due to the various design issues, noises are introduced in the digitized speech signal. Normally the noises can be added in the signal due to some external noise sources and sometime due to the effect of communication channel. All those above mentioned noise degrades the speech quality depends upon the intensity of noise. As a result it is essential to develop algorithmic model to minimize the noises from noisy speech signal. For this denoising normally the use of filter banks are popular among most of the existing filter bank, the filter band of Discrete Wavelet Transform (DWT) [2] has more capabilities to denoise any signal. In this paper, an efficient speech denoising algorithm is prepared which denoise any speech signal using multilevel DWT and soft thresholding operation.

2. LITERATURE SURVEYS

The existing techniques used to denoise speech signal are explained in this section briefly.

In this paper [3], a new method has been proposed that deals with dual channel speech enhancement methods utilize the coherence function determine from the input signals without prior noise statistics. Discrete Wavelet Transform (DWT) is make use of a gain function. This method gave the good result when speech is perverted by various noise types when applied in an domain where interfering speakers are present.

In this paper [4], they presented the usage of denoising wavelet on speech input of MFCC (Mel Frequency Cepstral Coefficient) feature extraction method. The denoising process using wavelet transformation is used to enhance the MFCC shows on noisy signals. They used 120 speech data, with 30 data were used as the reference, and the other 90 were used as the testing data. These methods using wavelet transformation are able to boost the accuracy of the speech recognition system on input signals with SNR of 0-10 dB.

In this paper [5], to remove the noise on the audio signal Discrete wavelet transform ¸ based algorithm has been used. Both hard and soft thresholding are used for denoising. This method gave the good and efficient results and can be used real-time processing.

In this paper [6], to estimate the nature of noise power spectrum of the E-DATE algorithm is subsequent by using DWT instead of STFT. The novel method recovered STFT of the input speech signal by wavelet decomposition of the framed signal pursed by thresholding of the accurate coefficients. This method gave best progress in the objective evaluation measures such as SNR and PESQ-MOS scores.

In this paper [7], they proposed speech compression using Discrete Cosine Transformation (DCT) and the Discrete Wavelet Transform (DWT), then speech denoising using packet wavelet. To calculate the quality of the acoustic compression by using Peak Signal to Noise Ratio (PSNR) and mean square error. Marvelous results are obtained with Wavelet transform corelated to DCT. DWT has given good results estimate to DCT technique where about 50 dB is obtained for PSNR.

In this paper [8], they proposed speech enhancement appraise utilizing a sequence of wavelet thresholding and notch filter. This proposed method was successful in solving signal distortion issues caused by fixed frequency and white Gaussian noise, and also increase the speech quality for real time.

In this paper [9], a novel two-stage processing proposal for single-channel speech dereverberation and denoising to enhance the spectrum of the noisy reverberant signal is proposed. The novelty of proposed algorithm is decomposing the RIRs into two parts to build a two-stage processing scheme for enhancing speech from the noisy situations. The advantage of our proposed algorithm is that it is more efficient to enhance the speech and more time-saving by dividing the long RIRs into two parts. It improved the enhanced speech clarity and good quality.

In this paper [10], they proposed enhanced speech denoising method for low SNR speech signal, which is integrated wavelet threshold and MMSE-LSA (minimum mean square error short time log spectral amplitude estimation). Compared to the wavelet threshold denoising and EMD based wavelet threshold denoising methods, the suggested algorithm gave the better performances on speech intelligibility in simulations.

In this paper [11], a discrete wavelet packet transform algorithm is used for speech signal denoising. Both hard and soft thresholding are applied and noisy speech signal samples corrupted by white Gaussian noise from 0dB to +15dB are denoised. They concluded that soft thresholding method is most efficient rather than hard thresholding for output SNR value.

In this paper [12], a speech denoising method based on the principal component review is proposed. This De-noising algorithm build on principal component analysis achieved a greater effect. And the signal waveform of the principal component denoising technique was more full and more close to the original speech signal.

3. PROPOSED ALGORITHM

The block diagram of the proposed algorithm is shown in Figure 1. The Pre-processing unit is used to perform framing of the total audio signal and to increase the amplitude; this audio signal is multiplied by a constant factor. These enhanced signals are fed to the wavelet decomposition unit which generates 3-level user defined wavelet coefficients namely approximation and detailed coefficients respectively. Since the detailed coefficients are containing most of the noisy components of the input signal, user defined thresholding is applied to those coefficients without approximation coefficients. Now 3-level inverse DWT is applied to the approximation coefficients and modified detailed coefficients to generate de-noised signal.

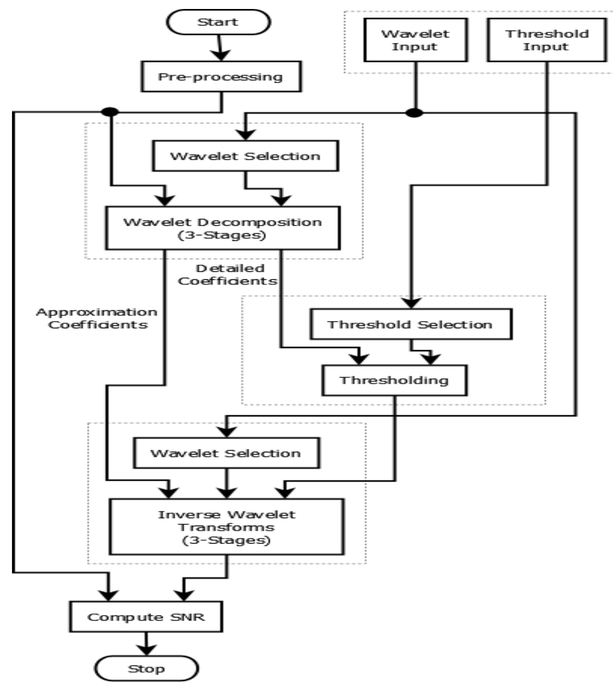

Figure. 1: Block Diagram of Proposed Speech Denoising Algorithm

3.1. Pre-Processing

To make the signal more suitable for processing the input speech is pre processed by framing and increment of amplitude.

3.1.1. Framing

The speech signal changes with time slowly is being its property gives variety processing methods for short-time in which speech signals short segment are isolated and handled as if they were sustained sound's short segments with fixed properties. Speech signal is changeable with time and not steady, but this change is slow. Thus it can be into the speech signal one by one in short section, this named frame processed [13]. The common speech's frame lasts 10-30 ms [1].

3.1.2. Increment of Amplitude

Normally, for denoising the amplitude of the input speech is not sufficiently high. So to increase the amplitude we multiply by some constant factors [14].

3.2. Discrete Wavelet Transform (DWT)

The research on wavelet de-noising has been done extensively. This approach is disseminated on the transform domain. In this method, the DWT of a signal is determined and then the calculated wavelets are passed through a threshold testing. In this case, the coefficients that are lesser than a certain value are eliminated. To reconstruct the signal these calculated coefficients are used. By using this method it is possible to remove noise with less loss of details. The signals energy co-efficient values will be comparatively large if a signal has its energy concentrated in a small number of wavelet co-efficient related to the noise that has its energy spread over a large number of co-efficient [15]. In conventional Fourier based signal processing, the spectrum of the signal is simulated to have little overlap with spectrum of the noise and hence a linear time-invariant filtering is employed. The signal with Fourier spectra overlap the linear filtering approach cannot independent noise from signal. In DWT analysis, is totally different. The perception in this case is based on the inferences of the amplitude, instead of location, of the spectra of the signal to be as other as possible for that of the noise. This allows separating signals or eliminates noise by clipping, thresholding and shrinking of the amplitude of the co-efficient. It is the localizing or concentrating properties of the wavelet transform that makes it particularly reasonable when used with this non linear filtering technique [16]. Thresholding offers a low pass and smoother version of the original noisy signal. The objective is to defeat the additive noise W(k) from the signal X(k).

3.3. Thresholding

The noise reduction can be done by soft and hard thresholding methods. These are efficient for noise reduction. The performance of denoising depends on the thresholding estimation used for the required application for speech enhancement. Hard thresholding incorporate the establishing to zero the coefficients whose absolute values are less than the threshold, otherwise, the coefficients value is not modified [16]. The equation is used to implement this block is shown in eq (1).

$$\mathbf{f_S} = \begin{cases} \mathbf{sgn(C(n))(C(n) - T)}; \mathbf{for|C(n)| \geq T} \\ \mathbf{0} \qquad ; \mathbf{Otherwise} \end{cases} \quad (1)$$

Where, C(n) represents the coefficients and T thresholding value.

Soft thresholding is a technique, based on establishing to zero coefficients whose absolute values are less than a threshold, otherwise, the coefficients value is modified, as shown in eq. (2), [16]. The thresholding result gives a equal to the value of a sign function which multiplies the subtraction value between a coefficient and threshold T. If the co-efficient is greater than zero, equal zero or less than zero the sign function returns 1, 0, -1 respectively.

$$f_S = \begin{cases} \mathbf{sgn(C(n))(C(n) - T)}; \mathbf{for|C(n)| \geq T} \\ \mathbf{0} \qquad ; \boldsymbol{Otherwise} \end{cases} \quad (2)$$

4. PERFORMANCE PARAMETERS

The parameters are used to calculate the performance [17] of the proposed algorithm is discussed in this section.

4.1. Increment in Segmental SNR

As the speech signal is non- stationary, there is a random fluctuation in the energy of the speech signal. Hence evaluation of speech quality performed considering the whole signal as the one may not be accurate. Hence SNR of the each segment of frame is computed separately and combined to form segmental SNR. This test can be conducted either in time-domain or in transform domain. Segmental SNR can be calculated by the relation as,

$$\mathbf{SNR_{seg}} = \left(\frac{10}{l}\right) \sum_{i=0}^{l-1} \mathbf{log_{10}} \left(\frac{\sum_{n=N_i}^{N_i+N-1} x^2(n)}{\sum_{n=N_i}^{N_i+N-1} (x(n) - \hat{x}(n))^2} \right) \quad (3)$$

Where, N is the frame length.

l is the number of frames.

x(n) is the original noisy speech.

x^(n) is the processed speech signal.

After getting the segmental SNR value we need to calculate the increment in SNR of the processed speech signal from noisy speech signal which is termed as the increment in the segmental SNR. During the silent intervals there may be the possibility of getting negative values; this is the major limitations of this evaluation method.

4.2. Log Likelihood Ratio (LLR)

This test is linear predictive coding based objective measures test. In LLR test tilt in phase between the spectrum of clean speech and processed speech is computed. This value gives the distortion introduced during processing method. Usually the value of LLR is less than 2. Equation for the calculation of LLR is given by

$$\mathbf{D_{LLR}(a_e, a_c)} = \mathbf{log_{10}} \left(\frac{a_e R_c a_e^T}{a_c R_c a_c^T} \right) \quad (4)$$

Where, a_c is the LPC vector of clean speech signal.

a_e is the LPC vector of enhanced or processed speech signal .

R_c is the autocorrelation matrix of clean speech signal.

4.3. Itakura–Saito Spectral Distance (ISD)

Itakura–Saito spectral distance is also a PLC based Objectives measures test of evaluation. It gives difference in the spectral envelope of enhanced speech signal and clean speech signal. Usually it's value is less than 100. Equation to calculated ISD is given as,

$$\mathbf{D_{ISD}(a_e, a_c)} = \frac{G_c}{G_e}\left(\frac{a_e R_c a_e^T}{a_c R_c a_c^T}\right) + \log_{10}\left(\frac{G_c}{G_e}\right) - 1 \quad (5)$$

Where, G_e and G_c are the Linear predictive coding gains of processed speech and clean speech respectively.

5. SOFTWARE SIMULATIONS

The wavelet decomposition of the speech signal with 3-level discrete wavelet transform is shown in Figure 2. From the figure, it can be seen that most of the noises are present in the detailed coefficient bands only.

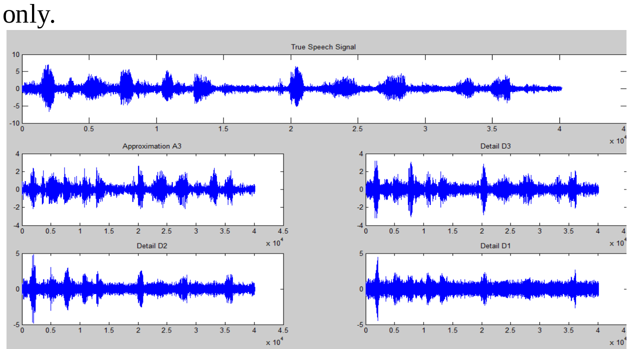

Figure 2: Wavelet Decomposition of Audio Signal.

The detailed coefficients are denoised by thresholding techniques. The pictorial representation of those denoised coefficients is shown in Figure 3 which shows most of the noises present in those bands are eliminated.

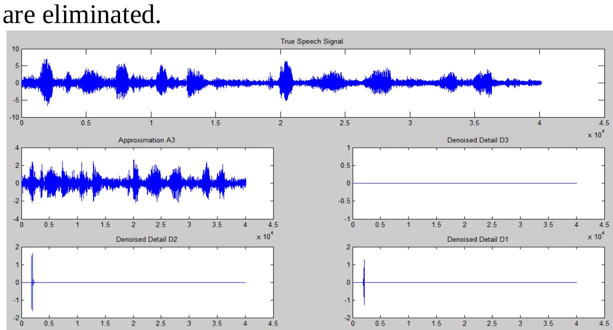

Figure 3: Thresholding Operation on each Sub-band coefficients.

The time domain graph of input speech signal, noisy speech signal and denoised version of the corresponding version of that signal is shown in Figure 5 from which it can be seen that the denoised signal is almost nearer to the original signal.

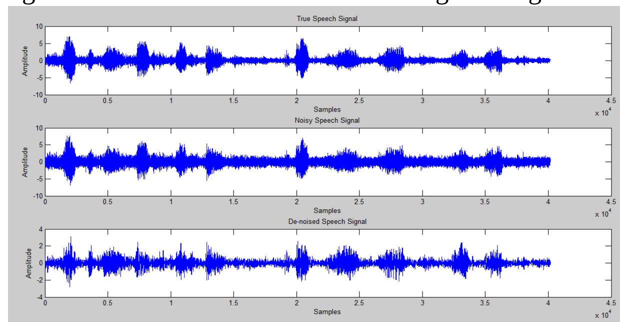

Figure 4: Signal comparisons in time domain

The frequency domain graph of input speech, noisy speech and denoised speech signal is given in Fig.6 which is known as Spectogram [1]. Spectrogram is the graphical representation of the relative energy concentration of the signal at each frequency as the function of time. In spectrogram time is represented by x-axis and frequency is represented as y-axis. Color in spectrogram represents the energy or magnitude of the signal. If darker the color higher will be the energy or magnitude. Silent periods or regions with low energy are represented by white color.

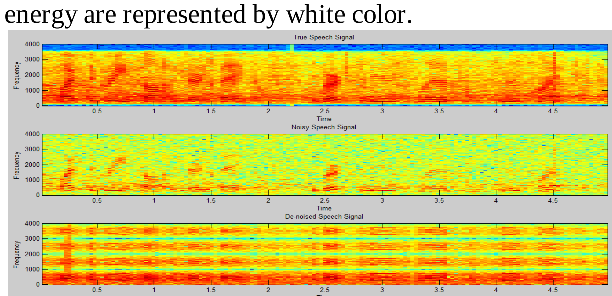

Figure 5: Signal Comparisons in frequency domain.

6. PERFORMANCE ANALYSIS.

The proposed algorithm is to analyze the performance of standard voice signal with different types of noises (i.e., airport, Babble, Restaurant and AWGN noises) at 5 dB level are considered using which various performance parameters are calculated for different wavelets and thresholding techniques. The calculated values are tabulated in Table 1 to Table 4 for various different noises respectively. By comparing those values, it can be concluded that the good quality denoised signal is generated by Haar DWT with MiniMaxi thresholding technique whereas the low error rate is generated by dB13 with Rigsure threshold. Similarly, both dB13 and Sym13 DWT with Rigsure threshold provide good tradeoff between error rate and the quality of the denoised signal.

Table 1: Performance comparison of proposed method with different DWT and Thresholding techniques for Airport noise at 5 dB level.

Sl. No.	Wavelets		SNR Values (dB)			
			Heursure Threshold	**Rigsure Threshold**	**Minimaxi Threshold**	**Sqtwolog Threshold**
1	**dB13**	**Noisy**	4.3331	4.3331	4.3331	4.3331
		Denoised	2.5329	3.2489	2.5996	2.5329
		Error	5.6756	4.3676	5.2668	5.6756
		dB SNR	3.6209	3.7927	3.6332	3.6209
		LLR	1.8299	1.7498	1.8347	1.8299
		ISD	29.0506	29.0506	29.0506	29.0506
2	**dB40**	**Noisy**	4.3331	4.3331	4.3331	4.3331
		Denoised	2.7916	3.2934	2.8173	2.7916
		Error	6.0518	4.8168	5.8866	6.0518
		dB SNR	3.8839	3.9746	3.8845	3.8839
		LLR	1.8474	1.7789	1.8556	1.8474
		ISD	37.8936	37.8936	37.8936	37.8936
3	**Sym13**	**Noisy**	4.3331	4.3331	4.3331	4.3331
		Denoised	2.4982	3.0991	2.5543	2.4982
		Error	5.5703	4.9698	5.5703	5.5703
		dB SNR	3.5819	3.7191	3.5867	3.5819
		LLR	1.8390	1.7807	1.8394	1.8390
		ISD	55.4097	55.4097	55.4097	55.4097
4	**Sym21**	**Noisy**	4.3331	4.3331	4.3331	4.3331
		Denoised	2.5444	3.0745	2.5770	2.5444
		Error	5.6200	4.7806	5.6200	5.6200
		dB SNR	3.6036	3.7102	3.6055	3.6036
		LLR	1.8216	1.7622	1.8290	1.8216
		ISD	47.4445	47.4445	47.4445	47.4445
5	**Haar**	**Noisy**	4.3331	4.3331	4.3331	4.3331
		Denoised	1.6676	2.1084	1.6837	1.6676
		Error	7.3564	5.7574	7.2740	7.3564
		dB SNR	2.5674	2.6861	2.5682	2.5674
		LLR	5.1381	4.7092	5.0909	5.1381
		ISD	1.0307e+03	1.0307e+03	1.0307e+03	1.0307e+03
6	**CDF-5/3**	**Noisy**	4.3331	4.3331	4.3331	4.3331
		Denoised	2.3417	3.0523	2.4418	2.3417
		Error	5.7735	4.8217	5.5749	5.7735
		dB SNR	3.4482	3.6450	3.4679	3.4482
		LLR	1.9330	1.8868	1.9273	1.9330
		ISD	61.4680	61.4680	61.4680	61.4680

Table 2: Performance comparison of proposed method with different DWT and Thresholding techniques for Babble noise at 5 dB level

Sl. No.	Wavelets		SNR Values (dB)			
			Heursure Threshold	**Rigsure Threshold**	**Minimaxi Threshold**	**Sqtwolog Threshold**
1	**dB13**	**Noisy**	4.3331	4.3331	4.3331	4.3331
		Denoised	2.3033	2.9233	2.3881	2.3033
		Error	5.8646	4.6860	5.1354	5.8646
		dB SNR	4.7662	4.9209	4.7863	4.7662
		LLR	1.8749	1.7911	1.8755	1.8749
		ISD	18.5315	18.5315	18.5315	18.5315
2	**dB40**	**Noisy**	4.3331	4.3331	4.3331	4.3331
		Denoised	2.5262	3.1839	2.5647	2.5262
		Error	6.1712	4.2038	6.0113	6.1712
		dB SNR	5.0124	5.1307	5.0119	5.0124
		LLR	1.8980	1.8358	1.88975	1.8980
		ISD	26.4986	26.4986	26.4986	26.4986
3	**Sym13**	**Noisy**	4.3331	4.3331	4.3331	4.3331
		Denoised	2.2739	2.8147	2.3297	2.2739
		Error	6.7365	4.8233	5.9855	6.7365
		dB SNR	4.7346	4.8881	4.7433	4.7346
		LLR	1.8613	1.8195	1.8489	1.8613
		ISD	19.0066	19.0066	19.0066	19.0066
4	**Sym21**	**Noisy**	4.3331	4.3331	4.3331	4.3331
		Denoised	2.3097	2.7800	2.4310	2.3097
		Error	6.4586	4.7059	6.4586	6.4586
		dB SNR	4.7734	4.8885	4.7772	4.7734
		LLR	1.8366	1.7797	1.8316	1.8366
		ISD	19.0671	19.0671	19.0671	19.0671
5	**Haar**	**Noisy**	4.3331	4.3331	4.3331	4.3331
		Denoised	1.5070	1.8788	1.5193	1.5070
		Error	6.7880	5.2374	6.6036	6.7880
		dB SNR	3.7438	3.8393	3.7445	3.7438
		LLR	5.2466	4.6983	5.1933	5.22486
		ISD	1.3761e+03	1.3761e+03	1.3761e+03	1.3761e+03
6	**CDF-5/3**	**Noisy**	4.3331	4.3331	4.3331	4.3331
		Denoised	2.1252	2.6178	2.2423	2.1252
		Error	5,7416	5.1658	5.3782	5.7416
		dB SNR	4.5519	4.7036	4.5756	4.5519
		LLR	1.9488	1.9542	1.9500	1.9486
		ISD	18.4565	18.4565	18.4565	18.4565

Table 3: Performance comparison of proposed method with different DWT and Thresholding techniques for Restaurant noise at 5 dB level

Sl. No.	Wavelets		SNR Values (dB)			
			Heursure Threshold	**Rigsure Threshold**	**Minimaxi Threshold**	**Sqtwolog Threshold**
1	**dB13**	**Noisy**	4.3332	4.3332	4.3332	4.3332
		Denoised	2.2277	2.6200	2.2779	2.2277
		Error	5.9572	4.8869	5.1175	5.5972
		dB SNR	3.0550	3.1538	3.0623	3.0550
		LLR	1.8061	1.7409	1.8159	1.8061
		ISD	18.2559	18.2559	18.2559	18.2559
2	**dB40**	**Noisy**	4.3332	4.3332	4.3332	4.3332
		Denoised	2.5021	2.9978	2.5184	2.5021
		Error	4.9483	4.7046	4.8829	4.9483
		dB SNR	3.2239	3.3023	3.2222	3.2239
		LLR	1.8508	1.7379	1.8571	1.8508
		ISD	23.7675	23.7675	23.7675	23.7675
3	**Sym13**	**Noisy**	4.3332	4.3332	4.3332	4.3332
		Denoised	2.2104	2.5577	2.2490	2.2104
		Error	5.8442	5.1877	5.8442	5.8442
		dB SNR	3.0286	3.1033	3.0296	3.0286
		LLR	1.7956	1.7319	1.7997	1.7956
		ISD	18.5354	18.5354	18.5354	18.5354
4	**Sym21**	**Noisy**	4.3332	4.3332	4.3332	4.3332
		Denoised	2.2530	2.5473	2.2738	2.2530
		Error	5.6572	5.1874	5.6572	5.6572
		dB SNR	3.0543	3.1145	3.0550	3.0543
		LLR	1.7844	1.7375	1.7900	1.7844
		ISD	17.8116	17.8116	17.8116	17.8116
5	**Haar**	**Noisy**	4.3332	4.3332	4.3332	4.3332
		Denoised	1.5475	1.9685	1.5605	1.5475
		Error	6.9865	5.2592	6.9865	6.9865
		dB SNR	1.9039	2.0080	1.9050	1.9039
		LLR	5.1566	4.5718	5.1137	5.1566
		ISD	496.9675	496.9675	496.9675	496.9675
6	**CDF-5/3**	**Noisy**	4.3332	4.3332	4.3332	4.3332
		Denoised	2.0265	2.5509	2.1113	2.0265
		Error	6.5067	4.9841	5.7010	6.5067
		dB SNR	2.8927	3.0224	2.9046	2.8927
		LLR	1.9143	1.8380	1.9223	1.9143
		ISD	19.4655	19.4655	19.4655	19.4655

Table 4: Performance comparison of proposed method with different DWT and Thresholding techniques for AWGN noise at 5 dB level.

Sl. No.	Wavelets		SNR Values (dB)			
			Heursure Threshold	Rigsure Threshold	Minimaxi Threshold	Sqtwolog Threshold
1	dB13	Noisy	5.0528	5.0771	4.9444	5.0415
		Denoised	2.9928	3.6459	3.0671	3.0219
		Error	6.1239	4.7587	5.4657	6.2382
		dB SNR	2.6963	2.8519	2.5319	2.1830
		LLR	1.8841	1.8389	1.8880	1.8985
		ISD	23.0547	33.9508	16.1831	16.8005
2	dB40	Noisy	4.9853	4.9574	4.9552	5.0575
		Denoised	3.2619	3.8526	3.3088	3.2823
		Error	4.8973	4.5959	5.0539	5.4761
		dB SNR	2.8192	3.1513	2.3918	3.1741
		LLR	1.8940	1.8130	1.8985	1.8866
		ISD	18.2230	16.8270	21.9806	17.0408
3	Sym13	Noisy	4.9830	5.0142	4.9709	4.9447
		Denoised	2.9663	3.6715	3.0502	2.9654
		Error	5.7618	4.8609	6.0317	5.6131
		dB SNR	2.6255	3.0103	2.1327	2.8281
		LLR	1.8979	1.8317	1.9094	1.9146
		ISD	15.5540	23.6341	19.6048	46.8193
4	Sym21	Noisy	5.0528	5.0771	4.9444	5.0415
		Denoised	3.0035	3.5197	3.0307	3.0326
		Error	5.8141	5.0927	5.5581	5.9266
		dB SNR	2.6957	2.8003	2.5296	2.1905
		LLR	1.8657	1.7980	1.8733	1.8790
		ISD	23.9142	28.7090	16.3954	16.8440
5	Haar	Noisy	4.9853	4.9574	4.9552	5.0875
		Denoised	1.9800	2.3033	2.0084	1.9865
		Error	6.3999	5.6192	6.3466	6.5461
		dB SNR	1.4725	1.7734	1.0811	1.8013
		LLR	3.7762	3.4089	3.8202	3.8309
		ISD	44.7705	177.4307	65.2203	25.3849
6	CDF-5/3	Noisy	4.9830	4.9709	5.0142	4.9447
		Denoised	2.8744	3.7832	3.0099	2.8795
		Error	6.0499	4.7732	5.0142	4.9447
		dB SNR	2.3890	2.1269	2.5648	2.5573
		LLR	1.9704	1.9340	1.9873	2.0176
		ISD	16.4213	28.1999	47.5399	31.5203

7. CONCLUSION

In this research article, an efficient speech denosing algorithm based on Discrete Wavelet Transform and Thresholding is proposed. For proper processing, the whole speech signal is divided into smaller frames which are then processed by different DWT-IDWT technique to separate noise components. The thresholding block minimizes those noise components by which the denoised signal is generated through inverse DWT. Due to the uses of multilevel DWT-IDWT and thresholding the proposed algorithm can be able to denoise speech better than existing which is proved in the comparison table.

8. FUTURE SCOPE

In future, higher order filter banks with dynamic characteristics of taping factors will be considered for more accurate denoising.

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CONFLICT OF INTEREST

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