A New Speech Enhancment algorithm in Hearing Aid based on Wavelet Transform

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Abstract:- Voice Enhancement systems are used to remove background inference in a speech signal and are become an main component of modern hearing aid. In everyday life the speech communication is vivid uses for the hearing impaired and the numerous other applications. Speech is the fundamental means of human communication. After over thirty years of research enhancement algorithm. Which offer superior noise reduction over current methods? All speech enhancements suffer from distortion for the residual noise due to imperfect noise removal. It is always require to perform denoising in voice processing system operating in highly noise because of wavelet transform is one of the popular techniques used in signal enhancement, In the present paper wavelet thresholding and wavelet packet thresholding method have been used to decrease the noise from the voice signal. A simple threshold method is presented to compute the optimum threshold value. Mean square error(MSE) at different values of SNR is computed to method like traditional speech subtraction, wiener filtering method, spectral subtraction with MMSE etc.The result obtained is compared with the other voice enhancement algorithm given in various reference papers. In comparison to other traditional methods we get improved result in terms of SNR and MSE.Simulation done in MATLAB platform.

Keywords:- *Discrete wavelet Transform, Minimum mean-square error, Signal to Noise Ratio, Wavelet packet transforms*

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I. INTRODUCTION

Improvement of voice signal is obliged in voice recogionization and voice enhancement system. Speech enhancement is required when the quality of signal is poor at receiver side or speech releted application. For example, hearing impaired person require enhancement of completly normal speech with their hearing capabilities. voice signals produced in a room generate reverberations, which may be quite obvious when a single channel telephone system is used.

Whenever voice is recorded by a single channel microphone,then background unwanted noise is also carried by it. So This noise depends on the environment such as computer fan noise, car engine noise to factory noise. The aim of basic voice enhancement is to suppress or mainly remove the background noise and improving the quality and/or intelligibility of the speech.

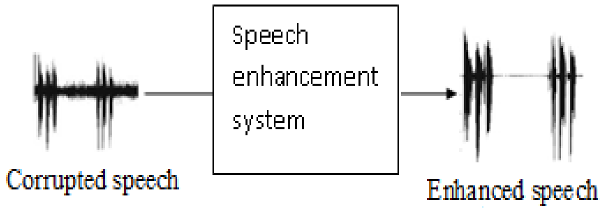

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Fig 1.Basic overview of a speech enhancement system

II. WAVELET DECOMPOSITION

The new signal, S, passes through two balancing filters and come out as two signals. If we in fact perform this process on a real signal, we end with twice as much data as we started with. Suppose that the new signal S consists of 1000 samples of data. Then the estimate and the detail will each have 1000 samples, for a total of 2000. To right this problem, we present the impulse of down sampling. This easily means throwing away every second data point.

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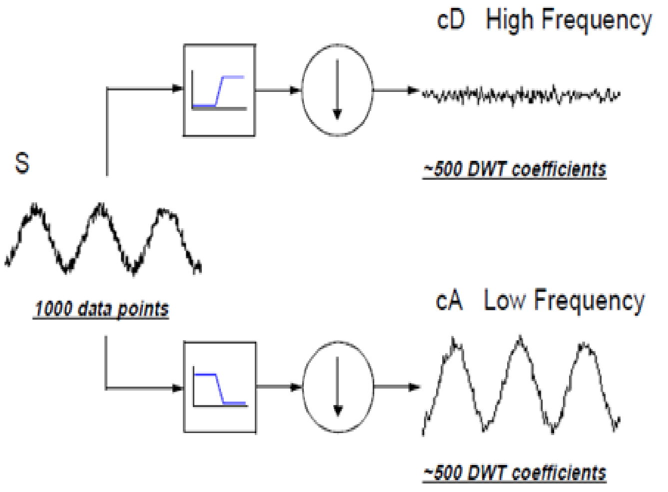


Fig 2. Wavelet Decomposition with filters and Down sampling

III. RECONSTRUCTION STEP

Before reconstruction apply any threshold technique the detail coefficients to remove the noise from it.

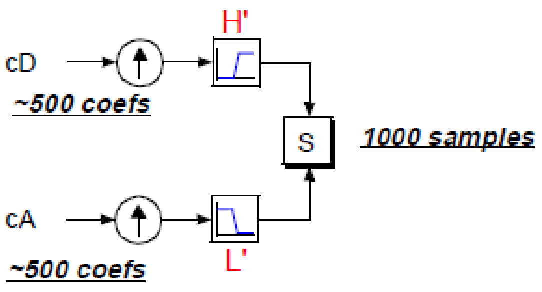


Figure 3 Wavelet Reconstruction: Approximation and Detail coefficient.

It is too likely to reconstruct the approximations and details themselves from their coefficient vectors. let's think how we would reconstruct the first-horizontal approximation A1 from the coefficient vector cA1. We pass the coefficient vector cA1 through the same process we used to reconstruct the original signal. However, instead of combining it with the horizontal-one detail cD1, we feed in a vector of zeros in place of the details. The process yields a reconstructed approximation A1, which has the same length as the new signal S and which is a real approximation of it. Similarly, we can reconstruct the first-level detail D1, using the similar process

The reconstructed details and approximations are true basic of the new signal. We find signal by combining the approximation A_1 and detail D_1.
Coefficient vectors cA1 and cD1 were created by down sampling, control aliasing distortion, and are simply half the length of the new signal and hence they cannot exactly be combined to reproduce the signal. It is requirement to reconstruct the approximations and details before combining them. Continue this method to the part of a multi-level examination; we locate that similar relationships hold for all the reconstructed signal basic. There are some ways to reassemble the new signal.

IV. WAVELET PACKET TRANSFORM

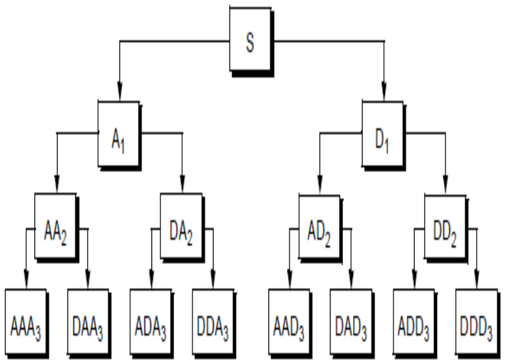


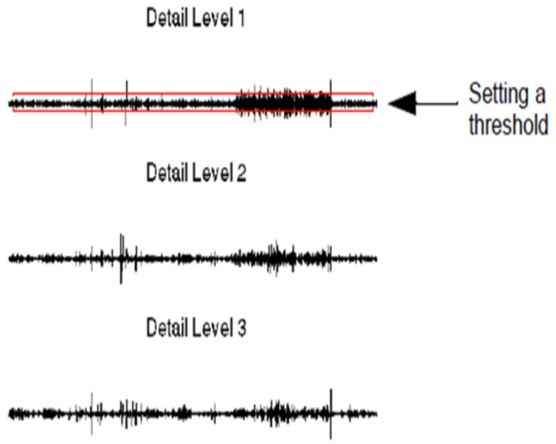
Fig 5. wavelet packet

In wavelet packet analysis, the details clearly the approximations can be split. This yields 2^n another ways to set the signal. We use an entropy-based criterion to select the most proper decomposition of a given signal. This means we seem at each join of the decomposition tree and measure the information to be gained by performing each split. Particular wavelet packet decomposition gives a provider of bases from which you can appear for the best statement with respect to a design object. This can be accomplishing by finding the "best tree" based on an entropy criterion. Simple and efficient method exists for both wavelet packet decomposition and best possible decomposition selection.

V. THRESHOLDING

Best possible de-noising need a more proper method called thresholding. Previous to the reconstruction or production process we use one of the thresholding methods. This involves discarding only the portion of the details that exceeds a sure limit. This is the portion where noise is removed from coefficients. Wavelet Packet analysis exemplify the application of Risk (SURE) as a principle.For choosing a threshold to be used for de-noising. This technique calls for setting the threshold T to

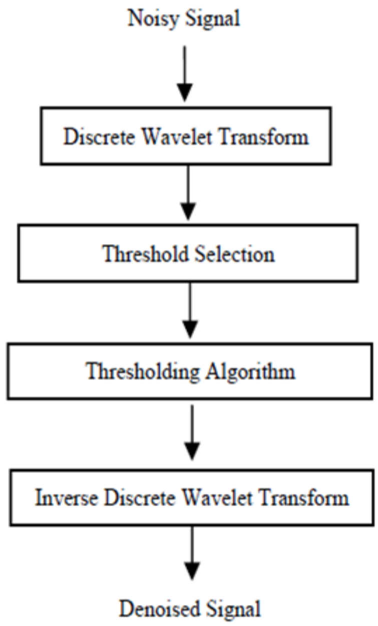
$$T = \sqrt{2\log_e(n\log_2(n))}$$



VI. INVERSE WAVELET TRANSFORMS

we perform the synthesis method for denoising the signal, which provides two main functions, Filtering and Up sampling. Up sampling is the process of lengthening a signal component by inserting zeros between samples.

VII. IMPLEMENTATION METHODOLOGY



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A. Performance Evaluation:

Evaluation measurement can be done by subjective quality tests and objective quality tests. Objective measures are based on mathematical comparison of the original and enhanced time domain signals. The majority of objective quality testes calculate time domain quality of the signal in word of a numerical distance measure. The signal to noise ratio (SNR) is the mainly widely used method to calculated time domain

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signal excellence. It is calculated as the ratio of the signal to noise power in db. We apply our method to enhance the noisy signal. And the test of our algorithm is measured and compared with [1]. We present our result in terms of SNR and MSE of the Denoised Signal which is implemented in MATLAB.

$$\mathbf{SNR} = \mathbf{10log_{10}}\left\{\frac{\sum(\mathbf{x})^2}{\sum(\mathbf{x}-\mathbf{x_d})^2}\right\}$$

$$\mathbf{MSE} = \frac{\mathbf{1}}{\mathbf{N}}\left\{\sum(\mathbf{x}-\mathbf{x_d})^2\right\}$$

Where x is speech signal, is denoised signal (Enhanced signal) and N is the no. of samples of speech files.

B. SIMULATION AND RESULTS:

The results have been resolute in the form of comparison table. After the comparison tables, a graphical statement has also been done for a quick examination of results. All the method have been tested for all the understood standard voice signals collected from internet open source database. Here we test our result in terms of SNR and MSE of denoise signl.

C. Comparison of Babble Noise at various dB level:

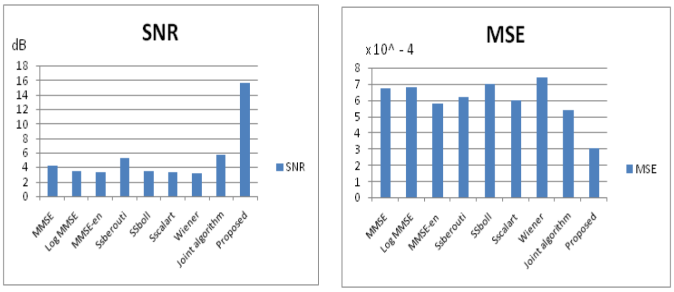
Table 1 Babble Noise at various dB levels

Sr. No.	Babble Noise (dB)	SNR (dB)	MSE
1	0	10.51	0.0008
2	5	15.51	0.00025
3	10	20.57	0.00008
4	15	25.39	0.00003

Table2 Comparison with Reference Paper [1]

Comparison	Algorithm	SNR (dB) (Babble Noise 5 dB)	MSE
Reference Paper [1]	Wavelet Based Algorithm	5.8	0.0005
Proposed Work	Wavelet Pocket Based Algorithm	15.53	0.00025

SNR and MSE comparison with [1]



VIII. RESULTS AND DISCUSSION

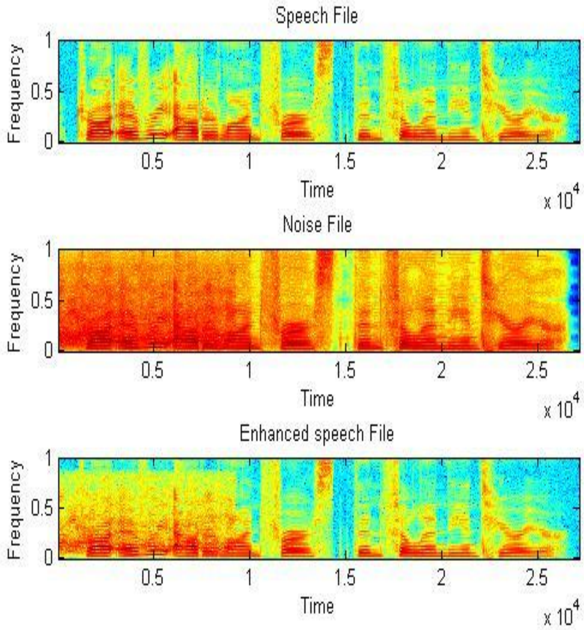


Fig 6. Signal spectrogram with 0db babble noise

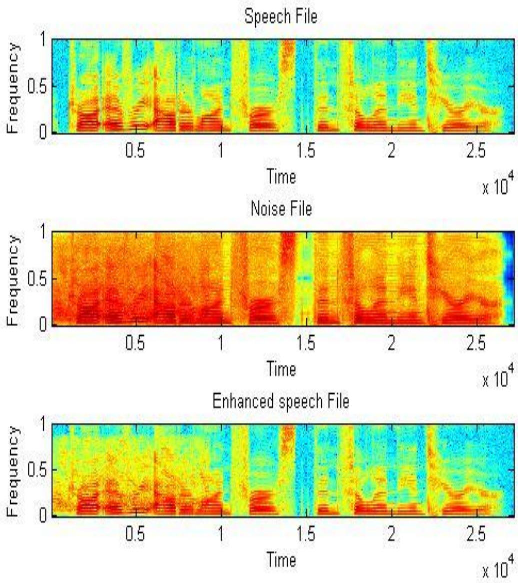


Fig 7. Signal spectrogram at 5db babble noise

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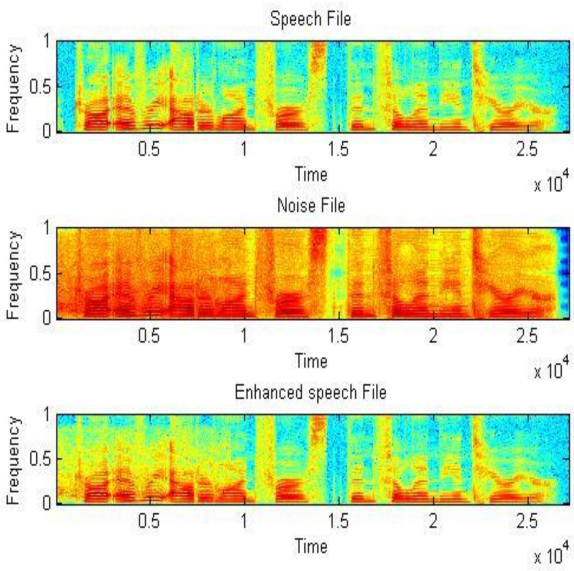

Fig 8. Signal spectrogram at 10 db babble noise

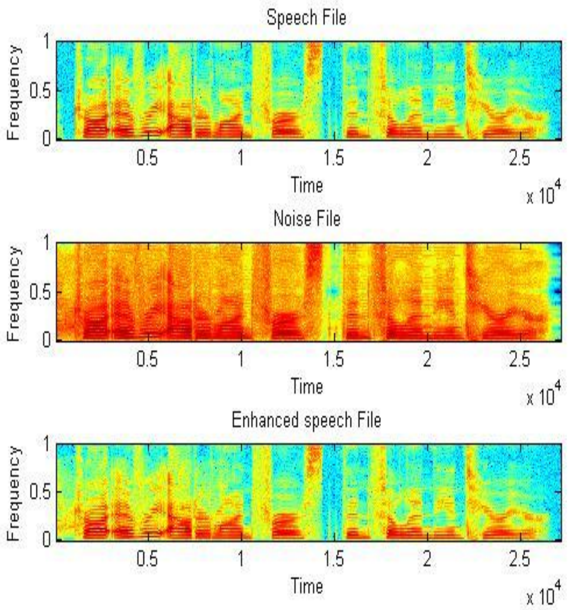

Fig 9 Signal Spectrogramplot at 15 dB babble Noise

CONCLUSION

In this paper, with wavelet packets we have a greater variety of options for decomposing the signal. we have presented various approaches based on a single channel for voice processing application. Here we use Coiflet and Daubechies wavelet that improves the SNR of denoise signal as compared to other. From the results we conclude that Discrete Wavelet Decomposition technique and wavelet packet transform with the use of different thresholding techniques improves the SNR and hence reduces the MSE as compared to other speech enhancement techniques. so, these methods are to improve quality and intelligibility and increasing performance of human perception and speech production system.

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