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Speech Denoising using Wavelet Transform

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Abstract: *This paper presents about Wavelet Transform used in Embedded Signal processing for speech signal noise cancellation. Filters are used to filter out particular fixed frequency, but in Wavelet transform is taking the overlapped windowed frames of input signal transforming it from time domain to frequency domain. Wavelet Transform we can decompose the signal into two values approximate and detail components. Approximate component is having high scale and low frequency while detail component is having low scale and high frequency. Databases of clean speech and Noise speech can be downloaded freely from NOIZEUS, and SpEAR database. Results are obtained using clean speech from NOIZEUS database, and different noises such as Airport Noise, Babble Noise, Restaurant Noise, Exhibition Noise, Station Noise, Street Noise, Car Noise and Train Noise at different dB levels such as 0dB, 5dB, 10dB, 15dB and Computation Time, Signal to Noise Ratio and Mean square error is obtained.*
Keyword: *Wavelet Transform family such as Haar, Daubechies, Coiflet, Symlet.*

I. Introduction

Signals used for information transfer from one place to another place. Signals may be in the form of light, sound, image, etc. Signals used as a carrier in the form of wireless communication, sound signal as the human being used to speak to another human being, image used to observe and understand. If the signals contributions are important in our life then their processing is also important. The signal gets corrupted by any type of unwanted signal called as interfering signal causing reduction in the quality of information. In order to preserve signal for further processing in non-causal operation such as storing, comparison, identifying, verification, research purpose.

Technically the signal is categorized in continuous, discrete, static, dynamic, deterministic, non-deterministic, causal, non-causal types. They are distinguishing the signal for research purpose. As we know the Meteorological system basically senses the parameters such as the temperature, humidity, UV radiation, for forecasting purpose. A seismic wave from shaking of the plates of the earth denotes the intensity of earthquake occurrence. Music that we listen are combinations of vowels and consonants that we used to mix voice of the singer with effects of the musical instruments.

Non deterministic signals are generally interfering the signal properties such as the Signal power, Signal MSE, Signal SNR that we able to predict for the need for restoration and regeneration of signals. For research purpose we undergo these consideration for getting the quality of speech and consideration of approach towards accuracy. As the goal of signal restoration from noisy environment is gained by Shannon-Fano Algorithm, Signals in the presence of Noise.

Signal need medium for travel towards the destination. Medium is full of contamination from unwanted sources of noises. That’s why the signal processing is required at the transmitting side and the receiving side. Most of the communication devices are embedded with this processing system. Such as the smartphones are embedded with more than one microphone in order to cancel the similar occurring data.

Dynamic system using memory used to store the previous values and updation on the previous values by finding and identifying old values are also a concept of signal identification and updation used in the software updation. So where is the area that you can say that signal is not used here. High and low levels of voltage indicates the On and Off of the system or in binary 1‘s or 0’s in the electronic circuits for communication.

Challenge of Signal Processing is to made signal easily understood to the listener or receiver. If the generated signal in form of light then receiver will be able to decode or read that sent signal is the challenge. Features present in the signal are properly extracted by Feature Extraction Methods so that signals in the form of Feature or template are properly identified or stored for further identification.

In Signal Processing, Filter is a process or electronic device used to remove unwanted features or components from signal. Filtering is a part of Signal Processing where it is used to remove or suppress unwanted parameters partially or completely. Means removing some frequency components such as background Noise and suppress interfering signals in order to filter out information signal. Filters not necessarily work in

Frequency domain but in Image Processing there are other filtering parameter too. Correlation removes some of the components without going in the Frequency domain by adding some redundancy and substraction.[1]

Filters are classified as: Linear or Non-Linear , Time Invariant or Time Variant, Analog or Digital, Discrete Time (Sampled) or Continuous Time, Passive(Using R, L,C) or Active(Using Op-Amp) Type of Continuous Time Filter, Infinite Impulse Response or Finite Impulse Response, Causal or Not Causal. But there are two types of digital Filter on the basis of Impulse Response of the Filter: Infinite Impulse Response , Finite impulse Response .[1]

FIR Filter Design carried out in three methods: Window Method, Frequency Sampling Method, Optimal Filter Design Method. The Window method basically begins with a desired unit sample response which is then truncated by a Finite Duration Window. In the Frequency Sampling Method the frequency response of the FIR filter is specified in terms of the samples of desired Frequency Response. Optimal Filter Design Method is available with an Algorithmic Design Procedure which generates Optimum Equiripple FIR Filter design. IIR filter Design methods classified as: The Impulse Invariant Method, The Bilinear Transformation Method. [1]

In case of Analog Circuits the Filter consists of Resistors, Capacitors, and Inductors, which are bulky in comparison to the Integrated Circuits that are why Analog is transformed to Digital. Analog work as it is transformed to Digital work is a kind of Challenge. Analog Radio is transformed to Software Defined Radio. Different signals have different parameters to suppress interfering signals or Noise so the challenge is removing this unwanted parameters make the Filter more Precious in Filtering. [1]

II. Literature Review

Zhanfeng Wang[01] in his paper shows comparison between Fourier transform and Discrete Wavelet packet transform in speech denoise process. Wavelet Packet coefficients collects parameters such as time and frequency threshold. Noise cancellation can be done by zeroing certain wavelet packet coefficients and thresholding.

The vibration of mechanical system in running is collected in database and is used to detect early fault. To detect early fault from vibration signal in presence of Noise is very important. Small wavelet correlation filtering denoise is proposed. Signal is threshold to remove vibration signal and noise is removed to improve SNR of signal. Compared to tradition method his method is more effective according to author Lixia et al.[02]

Non-stationary sensor signals undergo denoising process within Mechatronics system to detect signal for analysis. As the sensors actually transducer generates its energy depending on the environment condition. Saber et al., proposed in their paper that non-stationary signal is processed by Wavelet Transform. Different efficient architectures for the implementation of Wavelet Transform are explained. They notice that the better result of denoising is obtained with Lifting architecture. The mother wavelet having close attribute to the original signal is selected for denoising along with number of levels defines decisive factor. After that signal is thresholded to remove unwanted noise and filtering signal decomposition and reconstruction is done. Wavelet shrinkage method is applied to reduce noise from sensor signals in order to understand signal for specific operation.[3]

III. Proposed Work And Problem Definition

Study of the Wavelet Transform family and its application for noise cancellation in speech signal. As the Algorithms are developing day by day and their parameters such as the Computation Time Signal to noise Ratio, Mean Square Error is certainly being improving at each Algorithm for achieving Accuracy and Precision. In Wavelet Transform here is to perform Speech enhancement that is to remove Train Noise, Car Noise, different types of Noise from the Input Signal.

IV. Block Diagram Of Noise Cancellation Using Wavelet Transform:

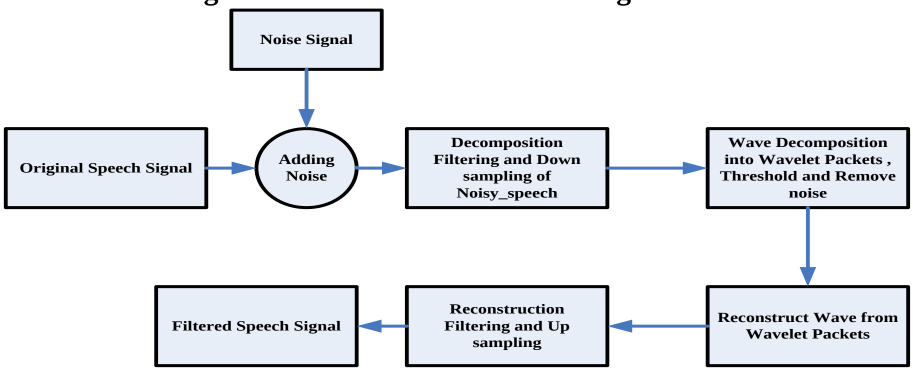

Fig 1: Block diagram of Wavelet Transform

Wavelet Transform:

The integral wavelet transform is the integral transform defined as:

$$[W_\psi f](a,b) = \frac{1}{\sqrt{|a|}} \int_{-\infty}^{\infty} \psi\left(\frac{x-b}{a}\right) f(x)dx \quad \text{---- (1)}$$

The wavelet coefficients C_{jk} are then given by

$$C_{jk} = [W_\psi f](2^{-j}, k2^{-j}) \quad \text{------- (2)}$$

Here, $a = 2^{-j}$ is called the binary dilation or dyadic dilation, and $b = k2^{-j}$ is the binary or dyadic position.

Basic idea:

The idea of wavelet transforms is that the shifting should allow only changes in time extension, but not shape. Changes in time function leads to corresponding changes in frequency function. Based on the Heisenberg's uncertainty principle of signal processing,

$$\Delta t \Delta \omega \geq \frac{1}{2} \quad \text{-------(3)}$$

Where t represents time and ω angular frequency (ω = 2πf, where f is temporal frequency).

Higher the resolution in time, lower the resolution in frequency. The larger the extension of the analysis windows is chosen, the larger is the value of Δt.

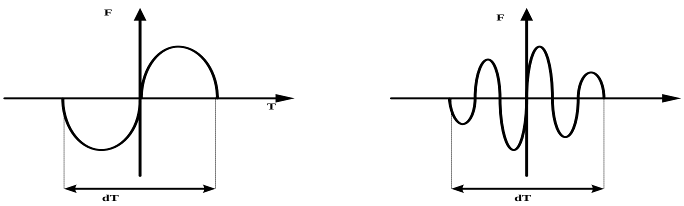

Fig 2: Shows greater variation in time leads to smaller variation in frequency

When Δt is large,
1. Bad time resolution
2. Good frequency resolution
3. Low frequency, large scaling factor

When Δt is small
1. Good time resolution
2. Bad frequency resolution
3. High frequency, small scaling factor

The basis function Ψ can be regarded as an impulse response of a system with which the function x (t) has been filtered. The transformed signal provides information about the time and the frequency. Wavelet transform contains information of both time and frequency as that of STFT but provides higher frequency analysis. The limit to decompose wavelet packet is greater than or equal to 0.25 of the product of Time and Frequency parameters in wavelet domain. The Time Domain, Frequency Domain, Short-Time Fourier Transform, and Wavelet Transform as shown below:

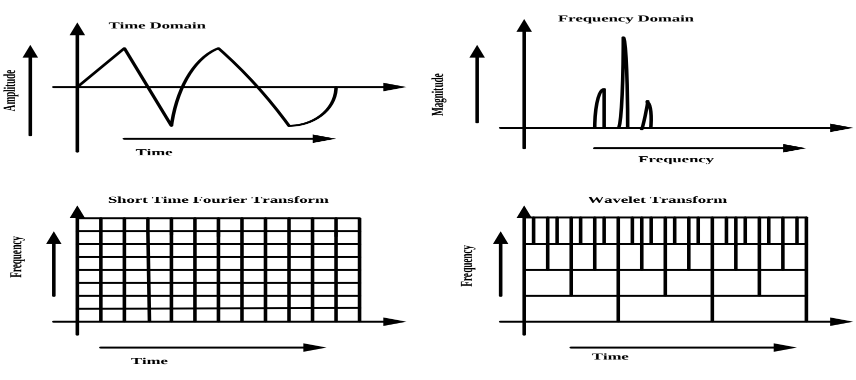

Fig 3: Different domain of signal representation

Family of Wavelets:
- Haar
- Daubechies
- Coeflit
- Symlet

Working of Wavelet:
Approximations: High-scale, low-frequency components of the signal.
Details: low-scale, high-frequency components.

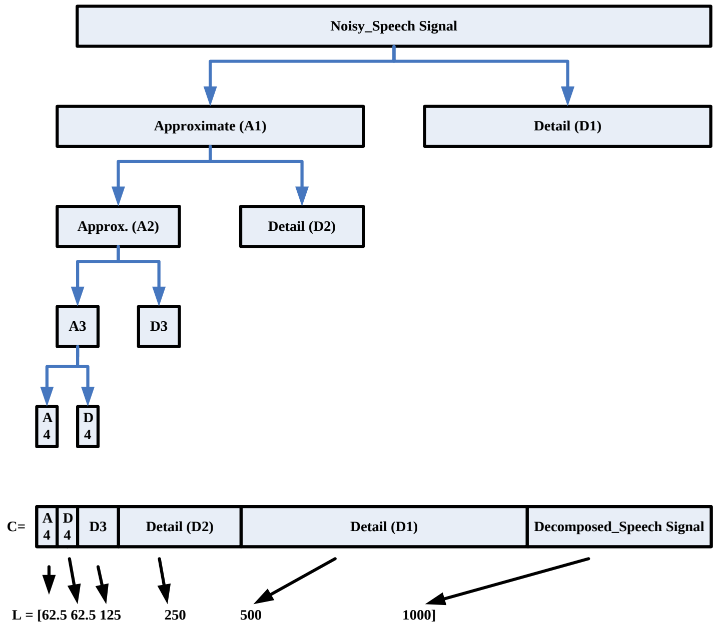

Fig 4: Wavelet Decomposition at each level and the values stored inside matrix c and l used to decompose signal

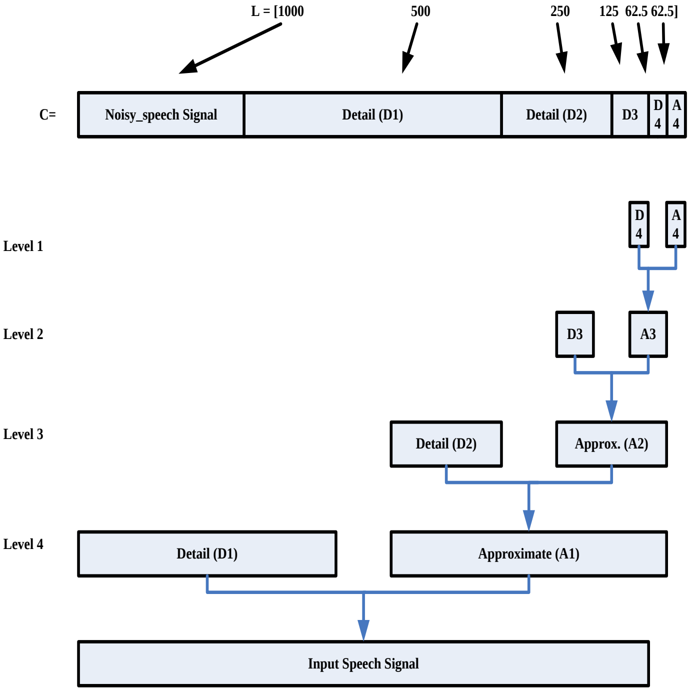

Fig 5: Wavelet Reconstruction at each level and the values stored inside matrix c and l used to reconstruct signal

V. Results Obtained

Wavelet	Airport Noise at 0_dB			Airport Noise at 5_dB			Airport Noise at 10_dB		
	Time	SNR	MSE	Time	SNR	MSE	Time	SNR	MSE
haar	0.013488	40.93719	0.005235	0.01074	40.78622	0.005157	0.010223	40.39437	0.005251
dB4	0.011593	49.022	0.004296	0.012418	49.34385	0.004198	0.012404	49.34202	0.004267
coif4	0.012381	46.95297	0.004196	0.012339	47.23515	0.004092	0.029835	47.175	0.004168
sym8	0.012252	45.80252	0.004201	0.013413	45.96552	0.004099	0.012164	46.01083	0.004173

Table 1: Results of different Airport noises at different dB levels noise from NOIZEUS database.

Wavelet	Station Noise at 0_dB			Station Noise at 5_dB			Station Noise at 10_dB		
	Time	SNR	MSE	Time	SNR	MSE	Time	SNR	MSE
haar	0.010415	40.7844	0.005344	0.011564	40.52455	0.005293	0.010478	40.57647	0.005274
dB4	0.012586	48.9662	0.004366	0.012085	49.48061	0.004314	0.012325	49.44188	0.004284
coif4	0.012758	46.76361	0.004271	0.012744	47.3016	0.004214	0.012357	47.34634	0.004185
sym8	0.012346	46.17209	0.004277	0.012419	46.07436	0.004221	0.013303	46.10705	0.004194

Table 2: Results of different Station noises at different dB levels noise from NOIZEUS database.

Wavelet	Restaurant Noise at 0_dB			Restaurant Noise at 5_dB			Restaurant Noise at 10_dB		
	Time	SNR	MSE	Time	SNR	MSE	Time	SNR	MSE
haar	0.011683	40.07699	0.00536	0.010521	40.64618	0.005296	0.010803	40.59932	0.00527
dB4	0.012849	49.13997	0.004397	0.012483	49.4618	0.004319	0.011977	49.36709	0.004284
coif4	0.012632	46.78156	0.004304	0.012197	47.30007	0.004218	0.012405	47.18433	0.004183
sym8	0.012489	45.95856	0.004312	0.012933	46.06491	0.004222	0.01249	45.94093	0.004189

Table 3: Results of different Restaurant noises at different dB levels noise from NOIZEUS database.

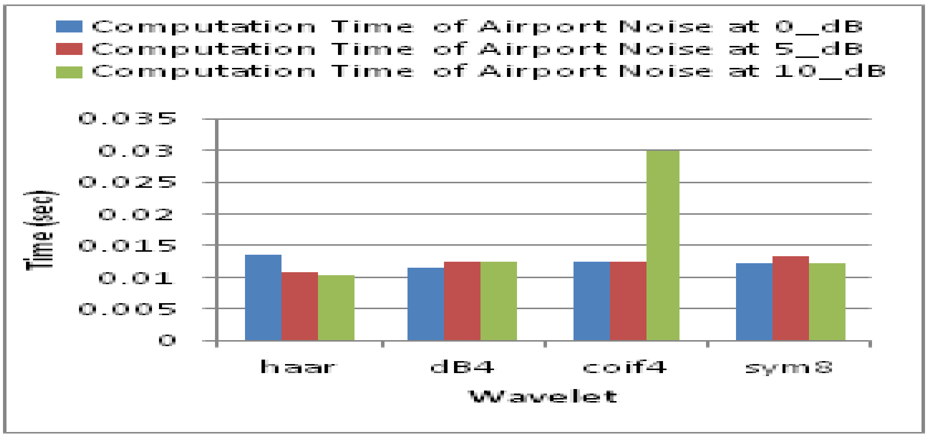

Fig 6: Comparison of various noise levels vs Computation time of different Wavelet families.

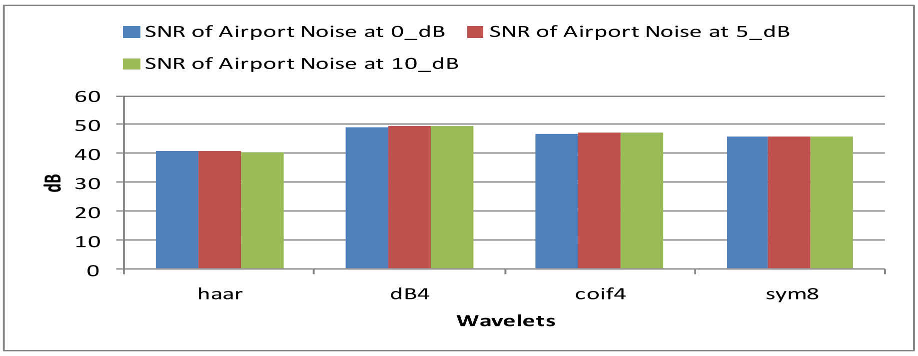

Fig 7: Comparison of various Noise levels vs Computation Time of different Wavelet Families

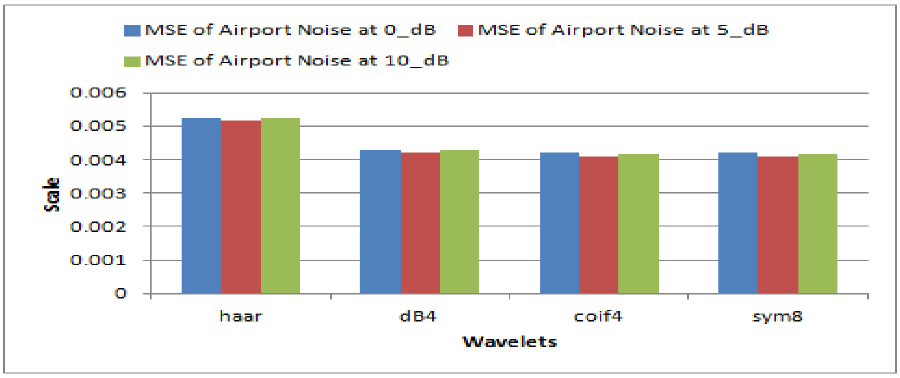

Fig 8: Comparison of various noise levels vs Computation Time of different Wavelet Families.

VI. Conclusion

Thus we can conclude that the Wavelet Transform Family can be used for fast processing such as compression and denoising by thresholding speech signal at each level of iteration.

The results obtained in graph (1) shows that the haar Wavelet is taking less computation time than sym8, dB4 and coif4 for Airport Noise at 0, 5 and 10 dB. Graph (2) dB4 is having high SNR value as compared to coif4, sym8 and haar. From graph (3) we conclude that coif4 is having less mean square error, as compared to sym8, dB4 and haar.

From Table (1), (2) and (3) by removing different noises(Airport noise, Station noise, Restaurant noise) at various dB(0dB, 5dB and 10dB) using wavelet transform family(haar, dB4, coif4, sym8) we obtain minimum better result in removing Station noise as compared Airport noise and Restaurant noise.

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