Journal of **N**onlinear **A**nalysis and **O**ptimization
Vol. 15, Issue. 1 : 2024
ISSN : **1906-9685**

VERBAL COMMUNICATION: NOISE REDUCTION AND CHARACTERISTICS ANALYSIS USING WAVELET AND THRESHOLD DERIVATIVE METHOD

R Vijaya, Saichand Proddutiri, Akutota Vinaykumar, Assistant Professor, Dept. of Electronics and Communication Engineering , Brilliant Institute of Engineering and Technology, Hyderabad, Telangana, India
Jammudi Venkatesh, Student, Dept. of Electronics and Communication Engineering , Brilliant Institute of Engineering and Technology, Hyderabad, Telangana, India

ABSTRACT
Noise reduction is a very difficult task but it has been a subject of intense research papers in recent years. Audio signal processing system is often suffering from noise troubles. Noise will seriously affect the quality of audio and even lead to audio processing failure. Simulation and results are performed in MATLAB R2012. In order to extract the desired speech signal from its corrupted signal, speech processing and communication systems are applied for effective noise reduction techniques. To avoid this noise troubles and quality degrades, noise reduction and characteristic analysis using discrete wavelet and threshold derivative method are used which improves the signal quality and to achieve high signal to noise ratio (SNR).
Keywords: Audio signal processing; Discrete wavelet; Threshold derivative; SNR.

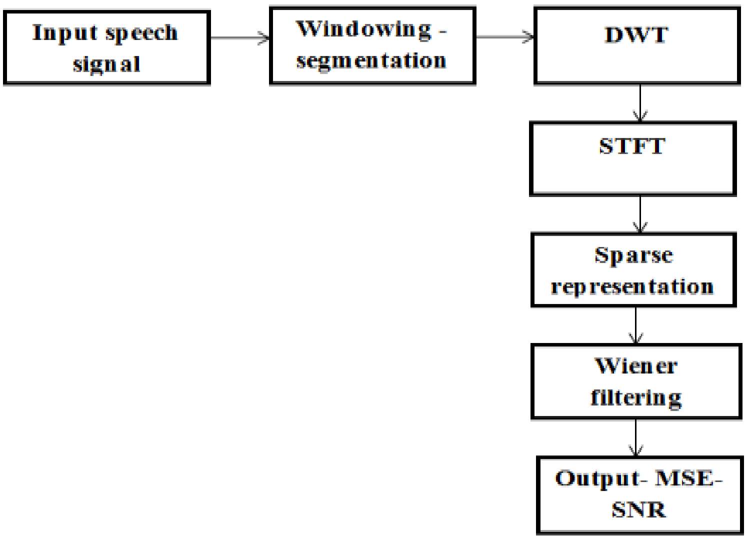
1. INTRODUCTION
Speech communications are used in our day to day life. A speaker, a listener and various communication devices are included in every case of speech communication. Historically the sounds of spoken language have been studied at two different levels: (1) *phonetic* components of spoken words, e.g., vowel and consonant sounds, and (2) *acoustic* wave patterns. Everywhere speech communication takes place such as hospitals, theatres, on roads, classrooms, at meetings etc. [3] The field of engineering consist of speech de-noising that studies the method used to recover the original signal from the corrupted signal by different types of noises. Noises may be classified into many forms such as white noise, pink noise, babble noise and many other types of environmental noises. For speech de-noising wavelet methods are mostly used.
The passive acoustic approach is the classical approach towards speech signal. The wiener filter generates a minimum mean square estimate of speech communication and also estimates speech and noise spectra. To adjust the deduction and removal methods, short time fourier transform is used to compensate the excepted time varying behaviour of the signal and noise. [2] Finally, after removing the noise from the noisy speech, the noise-reduced speech back to time domain using the inverse fast fourier transform (IFFT) is reconstructed. Noise removal can be successfully implemented in frequency domain rather than time domain. Thus the speech signal is reconstructed and noise had been effectively reduced. By calculating the speech to noise ratio, the results are accomplished by the statistical evaluation.
Techniques of frames averaging are applied to overlapping lengths of the data buffers and hamming windows SNR is improved.
For speech de-noising, various wavelet based methods have been proposed. One of the speech de-noising procedure is to remove noise in the wavelet split coefficient method by shrinking the wavelet coefficient in the domain of wavelet. Each wavelet coefficient of the signal is compared with the average threshold value in thresholding method. It is set to zero if the coefficients is smaller than the threshold value. Else the amplitude is reduced slightly. For de-noising a signal, both soft and hard thresholding are used. [1] Using wavelets the components containing the noise is determined and

those elements are removed to acquire a noise free signal.

2. Wavelet Analysis

The dependent variable is the amplitude and the independent variable is time in the time domain signal. Frequency content contains most of the hidden information. The frequency information can be obtained very easily using wavelet transform which is not possible by working in a time domain. Using fourier transform and short time fourier transform analysis of a continuous signal is done, but satisfactory results are not gained. To get better results wavelet transform analysis is done. For affixed time frequency resolution short time fourier transform is used and for multi resolution technique, wavelet transform is used. [4] Wavelet transform analysis can perform local analysis which serves as a great advantage. Wavelet analysis can express the signal appearance such as breakdown points, discontinuities etc., that other analysis techniques cannot express. On using multi resolution analysis, at High frequencies the signal has good time resolutions and poor frequency resolution. At low frequencies the signal has poor time resolution and good frequency resolution. In practical applications, it is assumed that low frequency occurs for entire duration and high frequency in short interval. When frequency and coordinates are independent parameters, two dimensional sweep of a one dimensional signal is kept under consideration.

3. Block Diagram



Windowing

When frequency content of a signal is computed, errors can and do arise when we take a limited-duration snapshot of a signal that actually lasts for a longer time. [2] Windowing is a way to reduce these errors, though it cannot eliminate them completely.

Hamming window

This modified cosine taper starts at 0.08, rises to 1 in the middle of the period, and then goes smoothly back down to 0.08 at the end.

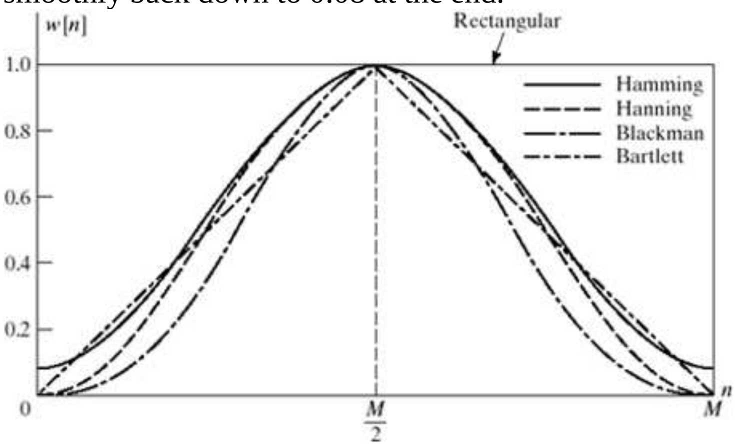


Figure .1 windowing comparison graph

http://doi.org/10.36893/JNAO.2024.V15I1.024-027

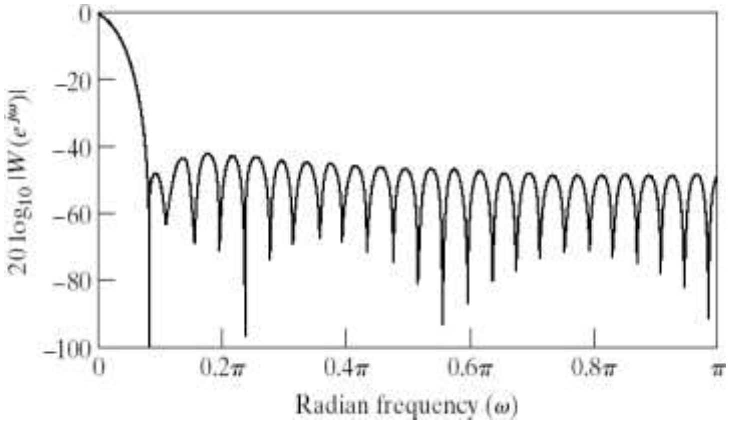


Figure no. 2 characteristics of hamming window

Segmentation

The concept of the voice sequence in order to classify the speech into voiced and unvoiced(noise) frames. This is accomplished by dividing a speech signal of your choice into short frames and by computing the average power of each frame. The speech in a particular frame is then declared to be voiced if its average power exceeds a threshold level that is chosen by the user. Otherwise it is declared unvoiced.

4. *Discrete Wavelet Transform (DWT)*

A multi resolution analysis is performed by contraction and dilation of wavelet functions and discrete wavelet transform using continuous wavelet transform. Multi resolution filter banks and special wavelet filters are used in DWT for the analysis and reconstruction of signal. [5] Fourier transform contains the frequency content of the signal, but the time at which the frequency component will occur is unknown. To avoid this problem STFT and wavelet transform are performed as it analyzes the signal at different frequency and different resolution. DWT sufficiently reduces the computation time and provides sufficient information for both analysis and synthesis. [1] It details the information with coarse approximation on decomposing the signal.

$$\phi(x) = \sum_{k=-\infty}^{\infty} a_k \phi(Sx - k).$$

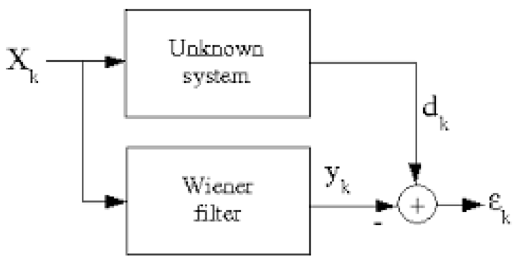
Short Time Fourier Transform (STFT)

Using STFT, the sinusoidal frequency and phase content is determined. There are two types of STFT (1) continuous time STFT (2) discrete time STFT. Here, continuous STFT is used for better analysis of the signal. Both time and frequency is the function of STFT.

$$\mathbf{STFT}\{x[n]\}(m,\omega) \equiv X(m,\omega) = \sum_{n=-\infty}^{\infty} x[n]w[n-m]e^{-j\omega n}$$

Weiner Filter

The main aim of weiner filter is to compute statistical estimate. Winer filter are usually applied in the frequency domain.



The Wiener filter is:

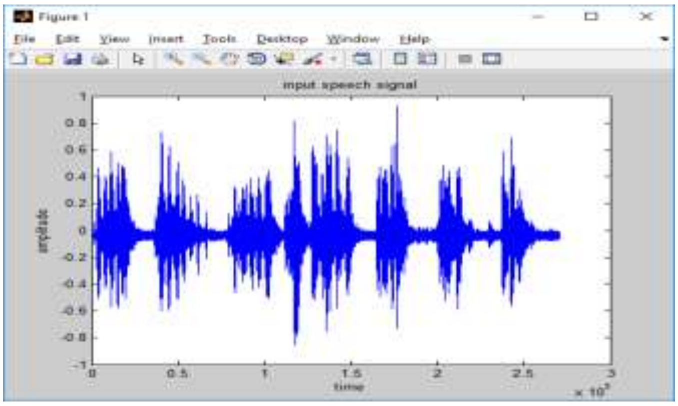
$$G(u,v) = \frac{H^*(u,v)P_s(u,v)}{|H(u,v)|^2 P_s(u,v) + P_n(u,v)}$$

http://doi.org/10.36893/JNAO.2024.V15I1.024-027

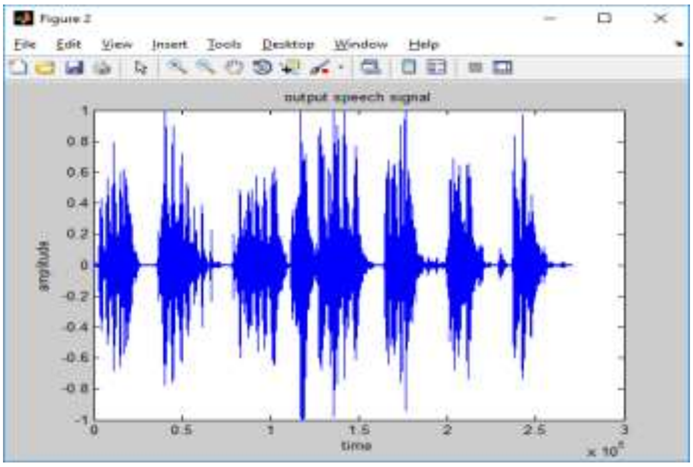
H(u, v)- Fourier transform of point spread function (PSF)
$P_S(u, v)$- power spectrum of signal process, obtained by taking the fourier transform of the signal autocorrelation. $P_n(u, v)$- power spectrum of noise spectrum, obtained by taking the fourier transform of the noise autocorrelation.

5. Simulation Results

Input signal



Output Signal



CONCLUSION

In this paper, wavelet transform and threshold derivative method are used for noise reduction and characteristics analysis. Noise is reduced in wavelet coefficients. On doing different analysis it is found that soft thresholding gives better results than hard thresholding. Thus we can obtain higher SNR ratio of about 7dB. Error rate is low and low complexity with signal estimation.

REFERENCES

1. Marvin R. Sambur, "Adaptive Noise Cancelling for Speech Signals", *IEEE Transactions on Acoustics, Speech andSignal Processing*, Vol. ASSSP-26, NO.5, October,1978.
2. Robert C. Maher, "Audio Enhancement using Nonlinear Time- Frequency Filtering", *AES 26th International Conference*, Denver, Colorado, USA, 2005 July-7-9.
3. Rajeev Aggarwal, Jai Karan Singh, Vijay Kumar Gupta, "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.
4. Lakshmikanth. S, Natraj. K. R, Rekha .K.R, "Noise Cancellation in Speech Signal Processing", *International Journal of Advanced Research in Computer and Communication Engineering*, vol.3, issue 1, January 2014.
5. Marc Karam, Hasan F.Khazaal, Heshmat Aglan, Cliston Cole, "Noise Removal in Speech Processing using Spectral Subtraction", *Journal of Signal and Information Processing*, 2014, 5, 32-41.
http://doi.org/10.36893/JNAO.2024.V15I1.024-027