

Speech Signal Noise Reduction by Wavelets

Roopali Goel, Ritesh Jain

Abstract— *Speech plays an important role in multimedia system. Speech enhancement is to remove noise from speech for multimedia systems. Noise act as a disturbance in any form of communication which degrades the quality of the information signal. Generally transmission and receiving signals are often corrupted by noise which can cause severe problems for downstream processing and user perception. Therefore an automated removal of noise would be an invaluable first stage for many signal processing tasks. Denoising has long been a focus of research and yet there always remains room for improvement. There are so many ways to improve the signal quality or to regenerate the signal. In this paper we have present a method for speech signal denoising using different wavelets. In this we will demonstrate the usefulness of wavelets to reduce noise in a model system where Gaussian noise is inserted into an audio signal.*

Index Terms—*About four key words or phrases in alphabetical order, separated by commas.*

I. INTRODUCTION

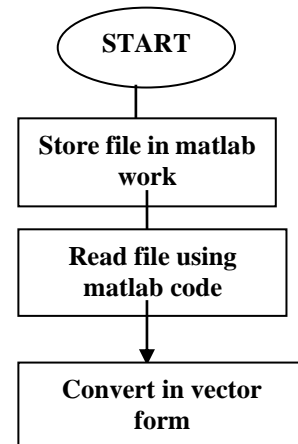
After automated system wavelets come into an existence as a popular tool for speech processing, such as speech analysis, pitch detection and speech recognition. Wavelets proven to be successful front end processors for speech recognition, by using the time resolution of the wavelet transform. For the speech recognition, the mother wavelet is based on the Hanning window. The recognition performance depends on the coverage of the frequency domain. The goal for good speech recognition is to increase the bandwidth of a wavelet without significantly affecting the time resolution. This can be done by compounding wavelets white noise is the most difficult to detect and to remove. The correlated(harmonic) signals results in larger coefficients than the uncorrelated noise. The noise can easily be removed by discarding small coefficients. White noise can be handled either by hard and soft thresholding. Hard thresholding smoothes the signal by reducing the wavelet coefficients by a quantity equal to the threshold value and modifies the signal energy . For the denoising of the signal it is assumed that the noise can be approximated by a Gaussian distribution. The speech components will have large values compared to the noise. The computation of the coefficients is done using a multi-resolution wavelet filter bank. The filter choice depends on the noise level and other parameters. For a good denoising result, a good threshold level has to be estimated. The wavelet function and the decomposition level also play an important role in the quality of the denoise signal .Recently, various wavelet based methods have been proposed for the purpose of speech denoising. The wavelet split coefficient method is a speech denoising procedure to

remove noise by shrinking the wavelet coefficients in the wavelet domain. The method is based on thresholding in the signal that each wavelet coefficient of the signal is compared to a given threshold. Using wavelets to remove noise from a signal requires identifying which components contain the noise, and then reconstructing the signal without those components.

II. PROBLEM STATEMENT

Review The basic idea behind the project is to estimate the uncorrupted speech from the distorted or noisy speech signal and sine signal, and is also referred to as speech “denoising”. There are various methods to help restore speech from noisy distortions. Selecting the appropriate method plays a very important role in getting the desired speech. The denoising methods tend to be problem specific. For example, a method that is used to denoise esophageal speech may not be suitable for denoising Emd. A sine signal and audio signal is taken and white Gaussian noise is added to it. This would be given as an input to the denoising algorithm, which produces an speech signal close to the original high quality speech signal. Selecting a wavelet that has compact support in both time and frequency in addition to significant number of vanishing moments is essential for an algorithm. Several criteria can be used in selecting an optimal wavelet function. The objective is to minimize reconstructed error variance and maximize signal to noise ratio (SNR). Optimum wavelets can be selected based on the energy conservation properties in the approximation part of the coefficients. Wavelets with more vanishing moments should be selected as it provides better reconstruction quality and introduce less distortion into processed speech and concentrate more signal energy in few coefficients. Computational complexity of DWT increases with the number of vanishing moments and hence for real time applications it cannot be suggested with high number of vanishing moments

ADDING an Audio File in MATLAB



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*Correspondence Author(s)

Roopali Goel, Computer Science, CET-IILM-AHL, Greater Noida, India.
Ritesh Jain, Computer Science, Guru Nanak Dev Polytechnic, Delhi, India.

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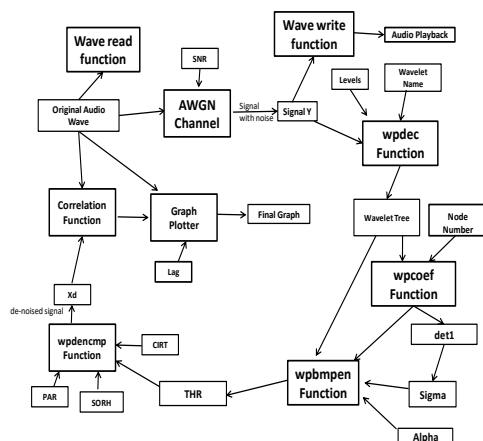
III. IMPLEMENTATION WITH MATLAB

In wavelets, there are two types of frequency components that is low frequency components and high frequency components. Low frequency components responsible for most of the information, while high frequency component give knowledge of few features. The signal is decomposed to form a tree and a function is applied to calculate the coefficients of a tree. Also, the number of tree nodes that gives higher performance is generated. A threshold value is obtained which is used to perform denoising. Then the denoised signal is obtained and compared with the original signal. A correlation function is applied between the original and denoised signal for the purpose.

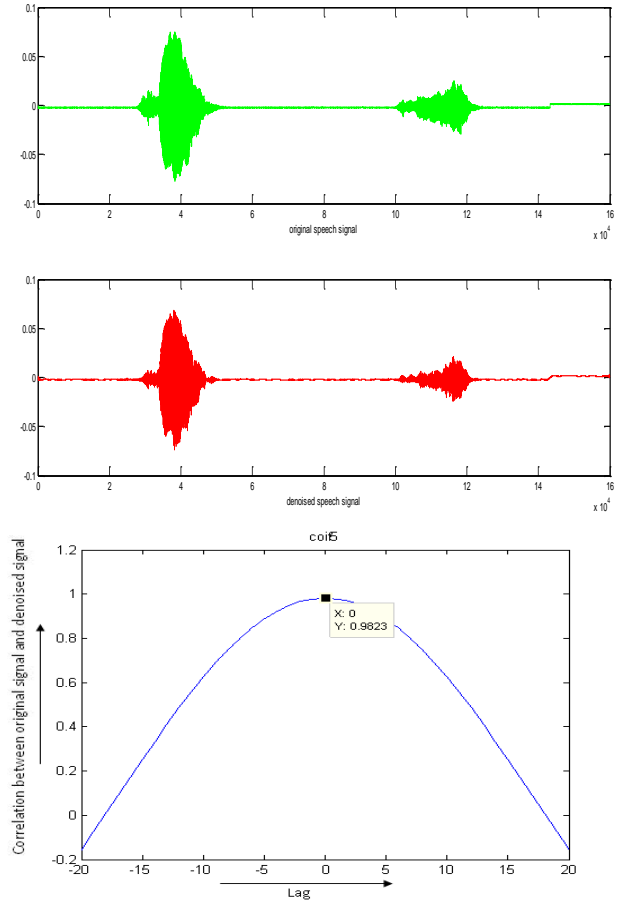
A. Algorithm for denoising of an audio signal and comparing the correlation of original signal and de-noised signal

- 1) First define the time interval K and angular frequency W.
- 2) Define a variable H which represents the argument of sine function ,
- 3) Read audio signal using WAVREAD command and define it x. obtain x1 in terms of vector form of x .sample x1 to get x2, Where x2 will be the original signal.
- 4) Add white Gaussian noise to the original signal to generate Y defining SNR ratio.
- 5) Use WAVWRITE command to listen to audio signal y
- 6) Define the wavelet name and level which you intend to use.
- 7) Use the wpdec command which is used for creating a tree.
- 8) Generate the coefficients of the decomposition of a signal using wpccoef command.
- 9) wpbmpen command generates a global threshold.
- 10) Sigma is the standard deviation of white Gaussian noise used in calculating threshold.
- 11) Alpha is a tuning parameter for the penalty term used in threshold.
- 12) Wpdencmp provides decompression or denoising of a signal.
- 13) SORH stands for soft and hard thresholding.
- 14) Decomposition performed using entropy criterion is performed by string CRIT and parameter PAR.
- 15) Xd is the denoised signal.
- 16) Calculate the correlation D Between the original signal and denoised signal.
- 17) Plot the original signal, denoised signal and signal with noise with time interval k.
- 18) Calculate the correlation between the original and denoised signal.
- 19) Code can be performed with different levels

BLOCK DIAGRAM



IV. EXPERIMENTAL RESULTS



V. CONCLUSION

Denoising of speech signals has been achieved successfully using wavelets. This paper provides a practical approach on how noisy audio (in wavelet form) incorporated with white Gaussian noise can be denoised. By using the coiflet wavelet, the author has presented a novel denoising method. A high correlation value above 98% at lag value of zero has been achieved.

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AUTHOR PROFILE



Ms. Roopali Goel is working as a Asst. prof in the department of CSE in CET-IILM-AHL Greater Noida, UP, India. She is also pursuing her M.Tech in CS (final year) from Jamia Hamdard University Delhi. Her area of interest is Cloud Computing and Wireless Networks.



Mr. Ritesh Jain is working as a Lecturer in the department of CSE in GURU NANAK DEV POLYTECHNIC, ROHINI, Delhi, India. He is also pursuing M.Tech in CS (final year) from Jamia Hamdard University Delhi