

ANALYSIS OF NOISE SUPPRESSION BY>NNLMS AND ADAPTIVE FILTERS

Sujeet Dubey¹, J P Upadhyay², Sachin Singh³

¹M.Tech Student, EC Deptt, SRIST JBP

²Guide, Asstt Prof., EC Deptt, SRIST JBP

³Co-Guide, Asstt.Prof.,EC Deptt,SRIST JBP

Abstract - This paper reviews the past and the recent research on Adaptive Filter algorithms based on adaptive noise cancellation systems. In many applications of noise cancellation, the change in signal characteristics could be quite fast which requires the utilization of adaptive algorithms that converge rapidly. Algorithms such as LMS and RLS proves to be vital in the noise cancellation are reviewed including principle and recent modifications to increase the convergence rate and reduce the computational complexity for future implementation. The purpose of this paper is not only to discuss various noise cancellation LMS algorithms but also to provide the reader with an overview of the research conducted. The main idea of de noising algorithm based on wavelet adaptive threshold is that speech signals should be packet transformed to get the wavelet coefficients used in optimal wavelet. Since the signal and noise have different relevance, there will be different attenuations in wavelet decomposition process. based on above characteristics, the appropriate threshold can be calculated by a new threshold function and the minimum mean square algorithm, even if the noise coefficients can be removed and the signal coefficients can be saved. here both hard and soft thresholding are used for denoising. analysis is done on noisy speech signals corrupted by babble noise at 0 db, 5db, 10db and 15 db SNR levels.

Key Words: Noise, denoising, thresholding, LMS,>NNLMS.

1.INTRODUCTION

The problem of controlling the noise level has become the focus of a vast amount of research over the years. If accurate information of the signals to be processed is available, the designer can easily choose the most appropriate algorithm to process the signal. When dealing with signals whose statistical properties are unknown, fixed algorithms do not process these signals efficiently. The solution is to use an adaptive filter that automatically changes its characteristics by optimizing the internal parameters. The adaptive filter has the property that its frequency response is adjustable or modifiable automatically to improve its performance in accordance with some criterion, allowing the filter to adapt to changes in the input signal characteristics. Because of their

self adjusting performance and in- built flexibility, adaptive filters are used in many diverse applications such as echo cancellation, radar signal processing, navigation systems, and equalization of communication channels.

It is well known to any scientist and engineer who work with a real world data that signals do not exist without noise, which may be negligible (i.e. high SNR) under certain conditions. However, there are many cases, in which the noise corrupts the signals in a significant manner, and it must be removed from the data in order to proceed with further data analysis. The process of noise removal is generally referred to as signal denoising or simply denoising. Example of a noisy signal and its denoised version can be seen in Figure 1. It can be seen that the noise adds high-frequency components to the original signal which is smooth. This is a characteristic effect of noise.

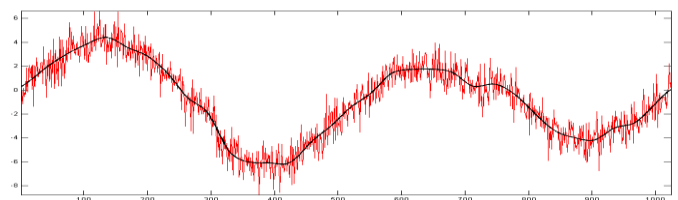


Fig 1: Noisy sine and its denoised version (solid line)

2. WAVELETS

A wavelet is a waveform of effectively limited duration that has an average value of zero. In Comparing with sine waves, wavelets are the basis of Fourier analysis. Sinusoids do not have limited duration, they extend from minus to plus infinity. And where sinusoids are smooth and predictable, wavelets tend to be irregular and asymmetric. Fourier analysis consists of breaking up a signal into sine waves of various frequencies. Similarly, wavelet analysis is the breaking up of a signal into shifted and scaled versions of the original (or mother) wavelet.

The term *wavelet* was first introduced by Jean Morlet while working on the analysis of signals for seismic analysis on oil-related projects. Before Morlet's work remarkable contributions were developed by Haarcitepand Zweig in 1975. After the work of Morlet and Grossmann on the definition of the continous wavelet transforms (CWT), several developing

have followed. The work of researchers as Stromberg, Duabechies, Mallat and Newland, among others, has pushed forward the theoretical frontiers of wavelets-based orthogonal decomposition and also augmented the scope of possible application fields.



Fig 2: Daubechies wavelet

Based on a particular wavelet, it is possible to define a *wavelet expansion*. A wavelet expansion is the representation of a signal in terms of an orthogonal collection of real-valued functions generated by applying suitable transformations to the original given wavelet. These functions are called “daughter” wavelets while the original wavelet is dubbed “mother” wavelet, acknowledging its function as source of the orthogonal collection.

2.1 SOFT AND HARD THRESHOLDING

The soft and hard thresholding methods are used to estimate wavelet coefficients in wavelet threshold denoising. [1] Hard thresholding zeros out small coefficients, resulting in an efficient representation. Soft thresholding softens the coefficients exceeding the threshold by lowering them by the threshold value. When thresholding is applied, no perfect reconstruction of the original signal is possible. Hard thresholding can be described as the usual process of setting to zero the elements whose absolute values are lower than the threshold. The hard threshold signal is x if $|x| > \text{thr}$ and is 0 if $|x| \leq \text{thr}$, where „thr“ is a threshold value. Soft thresholding is an extension of hard thresholding, first setting to zero the elements whose absolute values are lower than the threshold, and then shrinking the nonzero coefficients towards 0. If $|x| > \text{thr}$, soft threshold signal is $(\text{sign}(x) \cdot (|x| - \text{thr}))$. And if $|x| \leq \text{thr}$, soft threshold signal is 0. Hard thresholding is the simplest method but soft thresholding has nice mathematical properties and gives better denoising performance.

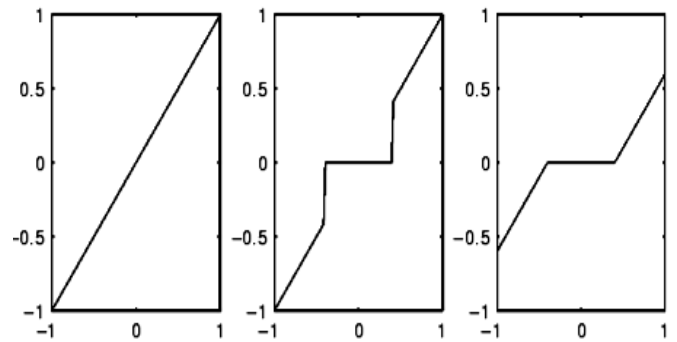


Fig 2.1: Hard and Soft thresholding

2.2 ADAPTIVE FILTER

Adaptive filter is widely used as noise canceller. In an adaptive noise canceller two input signals, $d(k)$ and $x(k)$, are applied simultaneously to the adaptive filter. The signal $d(k)$ is the contaminated signal containing both the desired signal, $s(k)$ and the noise $n(k)$, assumed uncorrelated with each other. The signal, $x(k)$ is a measure of the contaminating signal which is correlated in sole way with $n(k)$, $x(k)$ is processed by the digital filter to produce an estimate $y(k)$, of $n(k)$. An estimate of the desired signal, $e(k)$ is then obtained by subtracting the digital filter output $y(k)$, from the contaminated signal [5].

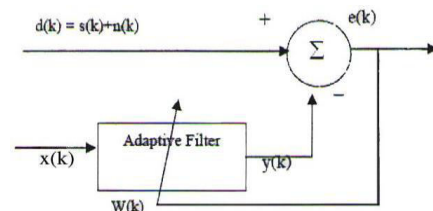


Fig2.2: Adaptive filter as noise canceller

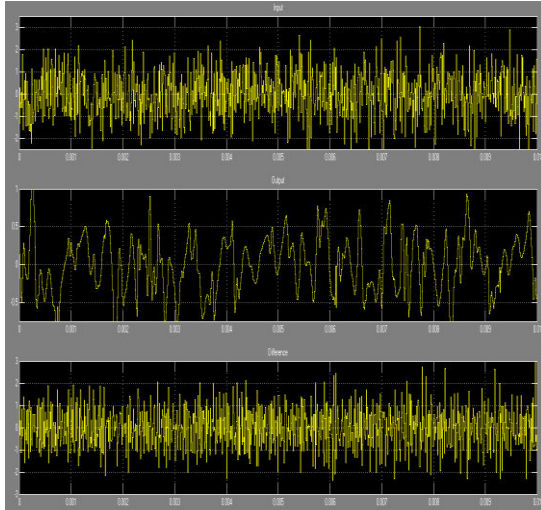
3. PROPOSED METHOD

Normalized LMS (NLMS) algorithm is another class of adaptive algorithm used to train the coefficients of the adaptive filter. This algorithm takes into account variation in the signal level at the filter output and selecting the normalized step size parameter that results in a stable as well as fast converging algorithm. The variable step can be written as-

$$\mu e(k) = \frac{\mu}{p + x^t(k) \times (k)}$$

Here μ is fixed convergence factor to control small adjustment. The parameter p is set to avoid denominator being too small and step size parameter.

4. CONCLUSION



In this noise suppression using wavelet and new adaptive NNLMs filter is performed. In the first simulation the RMS value is 0.8530. In the second simulation the RMS value is 0.0030. In both the simulations we may visualize the signal noise suppression. Smaller the value of RMS, better the noise reduction. Hence we may conclude that we are approaching towards the original signal with help of new adaptive NNLMs filter.

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