

Noise Cancellation using Adaptive Filter in the signal of Saxophone Musical Instrument

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ABSTRACT

In the paper attempt has been made to reduce the noise present in the signal of the musical instrument called saxophone. For active noise present in the speech signals, many methods are present and got success in reducing the noise. As noise which is introduced from any source will always create trouble and reducing of noise is of most importance and difficult. Simple filters are not able to deal with fast varying noise signals so we have to adopt the other method. In the present work the attempt has been made to cancel the gaussian noise present in the signals of the musical instrument saxophone. Here first we show that adaptive filters may be used for noise cancellation in musical instrument and using Least mean square (LMS) and Normalized Least mean square (NLMS) adaptive algorithm gaussian noise are reduced. We get the optimistic result. The work is done with the help of MATLAB software.

1. Introduction

The introduction of digital computers, microprocessor, microcontroller, and specific digital signal processor, the digital filters are often used for the filtering of the signals [3]. The infinite impulse response (IIR) filters are not practically used because of infinite memory size and introduce instability in the filter response[1]. Noise cancellation can be done with the finite impulse response (FIR) filters. But due to rapid variation in the characteristics of the noise simple FIR filters are not able to give the desired output. So we have to shift to adaptive filters. Adaptive filters have the capability to update their parameter in accordance to the varying environment. A speech signal has a bandwidth of 4KHz [2]. It can be defined as the waves of air pressed out of the lungs and coming from the mouth and nasals. Speech is single dimensional signal, time is independent variable and it is random in nature. But the musical signals are multidimensional. Music elements include Pitch and melody, Harmony and chords, Rhythm, Texture, Timbre or "tone color", Expression, Form like Strophic form, Binary form, Ternary form, Rondo form, etc.

The theoretical analysis of noise reduction, speech distortion, and SNR improvement of optimal filtering noise-reduction techniques including the time-domain causal Wiener filter, the subspace method, and the frequency-domain subband Wiener filter are done by J. Chen, J. Benesty, and Y. (Arden) Huang[7]. The different types of noise signals require different types of noise removing methods. K. Kirankumar, M. Suneel introduce a hybrid technique for reducing the noise from the speech signals which are corrupted by noise in multi environments like street, airport, car and train[8]. Least Mean

Squares (LMS), Normalized Least Mean Squares (NLMS) and Recursive Least Squares (RLS) algorithm are widely used for adaptive noise cancellation. Sayed, A. Hadei, and M. Iotfizad propose use of the FAP and FEDS algorithms in noise cancellation for speech enhancement[9].

Signals are varying with time, for fast varying signals fixed filters are not suitable for noise cancellation for that signal adaptive filtering technique is used. Niranjana D and Ashwini B have put on the efforts in design and implementation of LMS and Averaging algorithm[10]. Mamba'us Sa'adah, Diah Puspito Wulandari and Yoyon Kusnendar Suprpto conducted the study for noise removal from gamelan instruments by least-mean-square (LMS). In proposed work the desired signal which is the signal from saxophone instrument is interfered with the gaussian noise and using LMS and NLMS adaptive filter algorithm noise is reduced. The result obtained through matlab software[11].

2. Fundamental Theory:

Saxophone : Saxophone was invented in 1840 by Adolphe Sax. It lies in the family of woodwind instrument which is mostly made up of brass. It is played by a single reed mouthpiece. An oscillating reed is used to produce sound and note pitch is controlled by covering holes in the body tube. By changing the effective length of tube the resonant frequency of the air column changes which are present in the body. By pressing the keys holes are covered or uncovered by the player. The Saxophones are used in classical music, orchestras, military bands, marching bands, jazz and contemporary music. The frequency range of trumpet is 125 Hz to 1 KHz[4][5],[6].

Noise:

Noise is defined as unwanted signals which is present in the information signals. Noise are get in the signals mostly in the medium of propagation of signals. The noise in digital communication cause the binary bits inversion due to which error get generated in the received signals and caused the false information to be decoded. In analog communication due to increase in noise power caused decreased in signal to noise ratio hence system suffers and quality of the system get degraded. So either digital communication or analog communication, noise affect the signal and overall system performance falls down. So it is atmost important to cancel or decrease the noise if present in the received signals. Noise is classify depending on its occurrence. There are many types of noise like white noise, pink noise, impulsive noise etc. These noise have to considered in dealing with noise in communication system.

Gaussian noise is noise which exists in every frequency spectrum. It has probability density function (PDF) is like a Gaussian distribution hence name Gaussian Noise. As it has continuous distribution so called white noise.

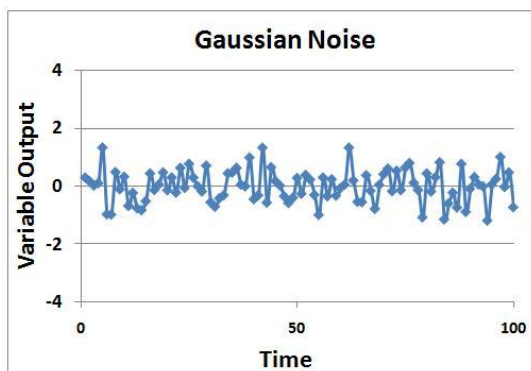


Fig 1. Gaussian noise

Pink Noise is having power spectral density inversely proportional to the frequency. In it power in each octave is constant. The name is pink noise because of the appearance of pink visible colour.

Impulse noise is a instantaneous noise like a spike having sharp sound like clicks and pop

As all the processing of the signals are done using computer or DSP processor thus we need to convert the analog signals to digital using analog to digital converter or any other method but have to follow the naquist criteria in that sampling frequency should be twice the maximum frequency of the input signals[2]. Due to conversion process and requirement of memory the digital filtering process is expensive than the analog filtering process [12]. But as far as performance level is concerned digital filter performance is superior to other methods [3].

3. Previous Methods:

The Ray Dolby has introduced the speech noise reduction technique. In this the signals amplitudes of are increase by

four bands at the time of encoding and then decreases at the time of decoding the signals. Spectral subtraction systems are also employed [3].

A. Wiener Filter

In this analysis is done in discrete time domain form. Noise auto correlation and shape of signal are present then wiener filter will reduced the noise and improve the signal. In this filter mean square error of estimated random process and desired random process are minimized.

B. Gaussian Window

Gaussian window is used for the non-stationary signal like speech whose properties changing from time to time [13]. In these signals is estimate of future samples are obtained by dividing a window of size 20m. Then convolution is perform using the Gaussian filter, then smoothing is done for achieving clean and sharpen speech signal.

C. Kalman Filter

In Kalman filter estimate the future samples from the first one in an iterative way [14]. For reliable communication it is important that signals received should be free from noise which is not possible. Noise is always present in the signals received to minimized noise from corrupted signal filtering process is done to improve the accuracy and performance of the communication system. But it is very difficult to filter out the noise from the desired signal. Many algorithms have been developed for noise cancellation. Most of the algorithms are based on kalman filtering. This filter uses mean square error to compare between estimated signal and desired signal.

4. Adaptive Filtering:

The concept of adaptive filtering was first introduced by Widrow [15], the Adaptive Noise Canceller (ANC) reduced the noise present in the desired signal with the help of adaptive filters. To generate the error help of reference signal is taken. The generated error signal is used to update the filter weight to get the required signal.

The simple FIR filter will not able to response the quick changes in the characteristics of the noise signals in the noisy environment thus we have to opt for the other options to deal with the rapid changes of the noise signals. Due to adapting capability of the adaptive filters according to the environment, we find suitable, use of adaptive filters for handling the fast changing characteristics of noise. For the optimal filtering it is required by the filter to adopt according to the environment and then vary the filters weight to give desired signal at the output. For adopting the environment feedback system is used by the adaptive filters.

The reference signal is used to learn about the environment which acts as a feedback. This reference signal will help to generate the error signal. The error signal is responsible to make changes in the weight of the filters parameter. This error signal is used to change the transfer

function of the filter and because of the change in the transfer function the impulse response of the filter change which cause change in the desired signal at the output of the filter. Actually our aim is to cancel out the effect of the noise which is present in the desired signal at output of the filter, this noise is called the background noise. To remove or to reduce this background noise from the desired signal we make use of reference signal. The reference signal is the noise signal which is correlated with the noise signal that is added with the desired signal at the input of the filter. As in many cases the noise that interfered with the desired signal is the gaussian noise since gaussian noise is present in all the frequency spectrum. In our case the desired signal is the signal from the saxophone musical instrument and with it the noise get added, so we need to reduce or cancel the effect of noise with the help of the adaptive filter. The adaptive filter make uses of adaptive algorithms for getting the desired results.

In adaptive filtering system continuous track of the environment is kept so that different parameters of the filter get change to maintain the quality and performance of the desired signals. It is time-varying process.

5. Adaptive Algorithm:

To obtained the desired signal from the noise interfered signal for the speech many algorithms were implemented. Here in our work we are implementing LMS and NLMS adaptive algorithm for reducing and cancelling the noise effect of noise signal from the signal obtained from the saxophone musical instrument .Widrow and Hoff in 1959 introduce an adaptive algorithm. It consists of an iterative process to minimize mean square error. In this method there is no need of calculating correlation function and there are no matrix inversions. This algorithm is a linear adaptive filtering algorithm having two basic process:

a) Filtering process:

Consist of (i) computing the output of a transversal filter produced by a set of tap inputs, and (ii) generating an estimation error by comparing this output to a desired response.

b) Adaptive process:

consists of automatic varying of the tap weights of the filter in accordance with the estimation error.

Implementation of the LMS Algorithm :

Each iteration of the LMS algorithm requires distinct steps in this order:

- 1.Filter output : $y[n]$
- 2.Estimation error : $e[n] = d[n] - y[n]$
- 3.Tap-weight adaptation :

LMS Algorithm:

By assuming autocorrelation matrix Γ_M and cross correlation vector Y_d are known, then coefficients can be computed iteratively as,

$$h_M(n+1) = h_M(n) + 1/2\Delta(n)s(n), n=0,1,\dots$$

to obtain the minimum of $J(h)$. Where,

$h_M(n)$ – vector of coefficient at the nth iteration.

$\Delta(n)$ – step size at the nth iteration.

$s(n)$ – direction for the nth iteration.

$h_M(0)$ is chosen arbitrarily.

For minimization of $J(h_M)$ with Γ_M and Y_d are known uses gradient vectors. Steepest Descent search method are used. In that the direction vector is $S(n) = -g(n)$ where $g(n)$ is the gradient vector at the nth iteration. The recursive algorithm is

$$h_M(n+1) = h_M(n) - 1/2\Delta(n)g(n)$$

and by substituting for $g(n)$ we get,

$$h_M(n+1) = (I - \Delta(n)\Gamma_M)h_M(n) + \Delta(n)Y_d$$

$h_M(n)$ converges to h_{opt} as $n \rightarrow \infty$, the sequences of step size $\Delta(n)$ should be absolutely summable with $\Delta n \rightarrow 0$ as $n \rightarrow \infty$, also as $n \rightarrow \infty$, $g(n) \rightarrow 0$.

NLMS Algorithm:

Parameters:

p = filter order

μ = step size

Initialization: $\hat{h}(0) = \text{zeros}(p)$

Computation : $n = 0, 1, 2, \dots$

$X(n) = [x(n), x(n-1), \dots, x(n-p+1)]^T$

$e(n) = d(n) - \hat{h}^T(n)X(n)$

$\hat{h}(n+1) = \hat{h}(n) + \mu e(n)X(n)/X^T(n)X(n)$

6. Experiment Set Up:

For the result and waveforms, LMS and NLMS filter are implemented on the MATLAB having Filter order = 8 and 32. For LMS algorithm step size taken is 0.008. The programs for the filters used is in off-line mode. The instrument used is saxophone. In first step the signal from the instrument is taken then the signal is mixed with the Gaussian noise and afterwards noise is minimized from the mixed signals. The waveforms of instrument signal, noise signal, mixed signals and recovered signal is shown in the result. Also PSNR is measured for both the filters.

7. Result:

For LMS Filter with Gaussian Noise :

The instrument sound signal is the input to the filter which is taken from the musical instrument saxophone , the input signal is added with Gaussian noise, mixed signal consists of instrument signal and added noise signal and the recovered signal is the desired output signal .

1] Tab size = 8 , mu = 0.008 Out_psnr = 17.32

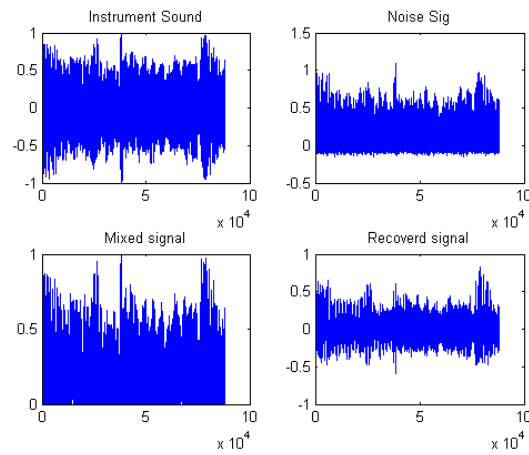


Fig.2 Signals for LMS filter with Tab size 8.

2] Tab size = 32 , mu = 0.008 Out_psnr = 18.14

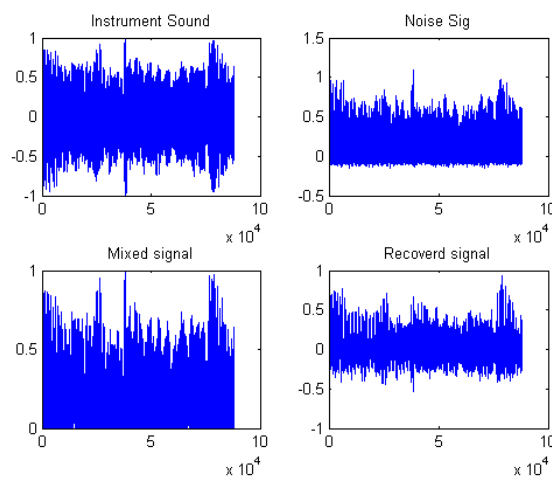


Fig.3 Signals for LMS filter with Tab size 32.

For NLMS Filter with Gaussian Noise :

The instrument sound signal is the input to the filter which is taken from the musical instrument saxophone , the input

signal is added with Gaussian noise, mixed signal consists of instrument signal and added noise signal and the recovered signal is the desired output signal .

1] Tab size = 8 , mu = 0.008 Out_psnr = 18.34

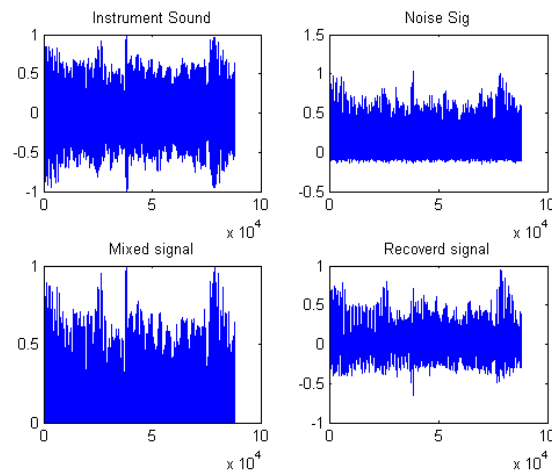


Fig 4 Signals for N LMS filter with Tab size 8.

2] Tab size = 32 , mu = 0.008 Out_psnr = 19.19

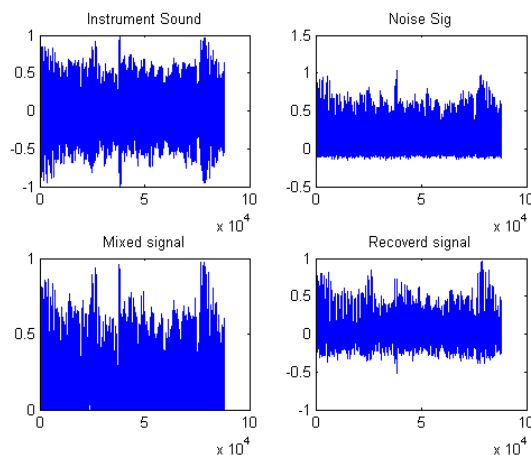


Fig 5 Signals for N LMS filter with Tab size 32.

Table:

Filter Name	Noise	Out-PSNR Tab Size =8	Out-PSNR Tab Size = 32
LMS	Gaussian	17.32	18.14
NLMS	Gaussian	18.34	19.19

Table 1 Results

Table 1 shows the filter used, tab size and the value of PSNR.

8. Conclusion:

From the result obtained we can say that the adaptive filters can be used for noise cancelling in musical instrument signal. The tab size affects the output of adaptive filter. And as in result table shows that as compare to LMS, NLMS is having performance better.

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