Noise Cancellation from Signal of Tabla Musical Instrument with the help of Adaptive Filters.

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ABSTRACT:
In the present work noise cancellation is done from the signals obtained from the musical instrument through the adaptive filter. The musical instrument used is tabla. The noise cancellation using adaptive filter is an alternative method for estimating signals which are get corrupted be additive noise or any interference. By applying adaptive noise cancellation, we have advantage that we do not required any prior estimate of signal or noise and we are able to minimize the required noise level which will be very difficult to obtain by any other signal processing methods. The additive noise which are corrupting our signals are white Gaussian noise and pink noise. The signals of musical instrument is mixed with the noise signals and the results are investigated for different adaptive filter algorithms which include Least Mean Square (LMS) and Normalized Least Mean Square (NLMS). While investigating the output from the adaptive filters, different parameter like output MSE, MAE, PSNR, SNR and filter order are recorded Input and output waveform of musical instruments signal, noise signal, mixed signal and filtered signal are observed. The computer simulations is carried out using MATLAB software and results shows the noise get reduced using adaptive filters from musical instruments signal.

Keywords: Adaptive filter, Noise cancellation, Gaussian noise, Pink noise, Adaptive algorithm, Least mean square (LMS), Normalized Least Mean Square (NLMS) algorithm.

I. INTRODUCTION:

Filter is the system which is designed to obtain the required signal of our interest from the noisy environment. Basically filters allow to pass the required signal of our interest and providing the attenuation to the unwanted signals. The unwanted signals or noise may present in different components of communication system which may be analog or digital systems. 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) Huangin[1]. The different types of noise signals require different types 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 [2]. For speech signal. Least Mean Square (LMS) and Normalized-Least Mean Square (NLMS) algorithms are used frequently but step size selection is difficult and length of filter too play important role for these features Mugdha Dewasthale and R.D. Kharadkar proposed High Performance Self Tuning (HPST) adaptive filter algorithm [3] They have suggested the selection of length of adaptive filter is based on the distance between two microphones in the ANC system and using HPST adaptive filter algorithm is used to adaptively determine the step size. 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. Niranjan D and Ashwini Bhave put on the efforts in design and implementation of LMS and Averaging algorithm. They use adaptive filter method because it requires less memory as compared to non- adaptive filter. They tested the system for different frequencies of signals. The result shows that adaptive filter adopts the changes in the input frequencies. The result also shows the role of step size for maintaining system stability [4]. In case of identical instrument, Noise reduction is very key challenge for instrument recognition, music processing and instrument analysis. Optimal filtering technique can reduce the level of noise that is present in the instrument signal. Mamba'us Sa'adah, Diah Puspito Wulandari and Yoyon Kusnendar Suprapto conducted the study for noise removal from gamelan instruments by least-mean-square (LMS). In doing study they mixed additive white Gaussian noise with the original signal and performance of the designed adaptive filter is evaluated based on the mean square error. They found that performance of Least Mean Square is satisfactory and is viable to be applied in gamelan signal. [5]

II. FUNDAMENTAL THEORY:

The transmission of the signals between transmitter and receiver is possible through the communication channel or also called medium. The transmission of the signal can be possible through wire lines, wireless or fiber optic channels. The optical disk, magnetic tapes and disk etc. are also used for transmitting the information as they can also carry data. But each and every communication channel consists of some inherent problems. They include,

- Signal attenuation: Due to internal resistance of the channel and fading of the signal, the signal attenuation in the channel occurs.
- Amplitude and phase distortion: Due to the nonlinear characteristics of the communication channel the transmitted signal is distorted in amplitude and phase.
- Additive noise interference: Due to internal solid state devices and resistors etc, used to implement communication system additive noise interference is produced.
- Multipath distortion: In wireless communication channels the multipath distortion occurs mostly due the signals coming from different paths and which tend to interfere with each other.

In designing a communication system may be analog or digital, engineer usually faces several restrictions or limitations. These are noise limitation, band with limitations and equipment limitation.
Here we focus on noise limitation. As we know that noise is defined as an unwanted form of energy which tends to interfere with the transmission and reception of the desired signals in communication systems. The noise cannot be eliminated completely. However, the effect of noise on desired signal can be minimized with the help of several techniques.

Noise is the major problem in any communication that may regress the speech or music signal during the transmission. Generally noise is classified into many ways according to source and relation with the receiver. According to source there are two types of noises External noise and Internal noise. In case of external noise, the noise sources are external to the receiver, where as in case of internal noise, the noise generated within the receiver. They may be taken as subjectively and can be suppressed or minimized by designing a proper noise cancellation system. [6][7]

The random signal may be considered as white Noise if it has been observed that the signal is having a flat spectrum over the range of frequencies that has been under consideration. As a human ear can hear the frequencies between 20Hz to 20 kHz, so in this case the frequencies under consideration is audio signal range i.e. 20Hz to 20 kHz.

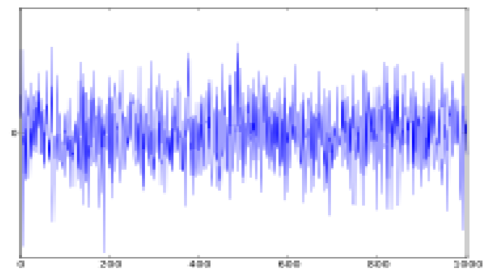

Fig.1Guassian Noise

Pink noise is a signal or a process which is having power spectral density (power per frequency interval) inversely proportional to the frequency of the signal. In pink noise in every Octave the noise energy is having equal amounts. Octave means it if we have a double the frequency. The name pink comes from the appearance of visible light with this power spectrum.

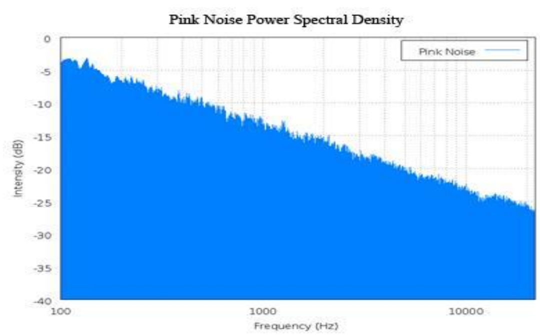

Fig. 2 Pink Noise

Musical instrument Tabla:

In the Indian subcontinent the table which is a pair of twin hand drums is most commonly used musical instrument. From 18th century, table is the principal percussion instrument. It is played by solo player but accompanied with other instrument and vocals will become larger ensembles. It is played in popular and folk music performances in Hindustani classical music India, Bangladesh, Pakistan, Nepal, Afghanista n and Sri Lanka. It is important instrument in bhakti devotional tradition of Hinduism and Sikhism while singing bhajan and kirtan. Also main qawali instrument by Sufi musicians. It used in dance performance like Kathak. Tabla instrument have two drums with different sizes and shape. The drums are made uo of hollowed out wood, clay or metal. The drum called daya is smaller drum which is used to create treble and tonal sounds. The drum bayan is larger drum which create bass sound. Both drums are laced with hoops, thongs and wooden dowels on its sides. For tighten the tension of the membranes for tuning the drums dowels and hoops are used. the fingers and palms are used extensively for playing the table. The playing technique is complex in nature. The wide variety of different sounds and rhythms are being created with the table. The frequency range of trumpet is 100 Hz to 3 KHz. [8][9][10]

III. ADAPTIVE FILTERING:

The filters are frequency selective devices. The two major types of digital filters are FIR (Finite Impulse Response) and IIR (Infinite Impulse Response) filters. Generally, the filter specification will be a desired frequency response. The inverse fourier transform of the frequency response will be the impulse response of the filter and it will be an finite duration signal. The digital filters designed by choosing finite samples of impulse response are called FIR filter and the filters designed by considering all the infinite samples are called IIR filters. Since an FIR filter is designed from the finite samples of impulse response, the direct design of FIR filter is possible in which the transfer function of the filter is obtained by taking Z-transform of impulse response. Mathematically, the filter design is design of transfer function of the filter. As an IIR filter is designed by considering/preserving the finite samples of impulse response, the direct design of IIR filter is not possible. Therefore, the IIR filter is designed via analog filter. As in the fast moving signals in case of the noise signal it is not possible with the FIR or IIR filter to update the filter coefficient so fast so we required the adaptive filters. Adaptive filter is one that is self-designing and relies on recursive algorithm for its operation. It can perform satisfactorily in an environment where complete knowledge of relevant signal is not known.

The algorithm starts with some predetermined sets of initial condition about the environment and if environment is stationary the algorithm convert to optimum wiener solution in some statistical sense after successive adoption cycle of the algorithm. And if environment is non- stationary then the algorithm offers a tracking capability.

The direct consequence of recursive algorithm is that the parameters of an adaptive filter become data dependent. Hence we can say that the adaptive filter is in reality a non-linear system as it does not obey the principle of superposition. The adaptive filters are classified as linear or non-linear. It is linear when input output obeys the principle of superposition otherwise nonlinear.

IV. ADAPTIVE NOISE CANCELLATION:

Adaptive noise cancellation is widely used to improve the Signal to Noise Ratio (SNR) of a signal by removing noise from the received signal. In this configuration the input x(n), a noise source $N_1(n)$, is compared with a desired signal d(n), which consists of a signal s(n) corrupted by another noise $N_0(n)$. The adaptive filter coefficients adapt to cause the error signal to be a noiseless version of the signal s(n) as shown in Fig.3.

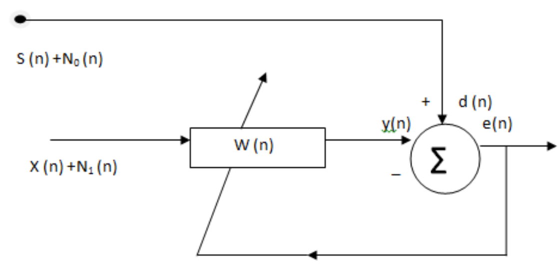

Fig.3 Adaptive noise cancellation configuration

The noise signals for this configuration need to be uncorrelated to the signal s(n). In addition, the noise sources must be correlated to each other in some way, preferably equal, to get the best results. The error signal should converge to the signal s(n), but it will not converge to the exact signal. In other words, the difference between the signal s(n) and the error signal e(n) will always be greater than zero. The only option is to minimize the difference between those two signals using certain error minimization techniques [11].

V. ADAPTIVE ALGORITHM:

LMS:
There are many algorithms used to adjust the coefficients of the digital filter in order to match the desired response as well as possible. But mostly used method is Least mean square (LMS) algorithm. This is introduced by Widrow and Hoff in 1959 is an

adaptive algorithm. It incorporates an iterative procedure leads to the minimum mean square error. The least mean square algorithm is a linear adaptive filtering algorithm that consists of two basic process :

a] Filtering process:
This involves (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:
This involves the automatic adjustment of the tap weights of the filter in accordance with the estimation error. [12]
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 :
As it is based on the instantaneous values of cost function which is given by
$E (w) =1/2\ e^2(n)$

The error signal e(n) is expressed as
$e (n) =d (n)-x^T(n) w(n)$
Where X_i – is input vector $\overline{X_i}$= ($x_1, x_2 \ldots\ldots x_n$)
and w_i = weight rector $\overline{w_i}$= ($w_1, w_2 \ldots\ldots w_n$)
Weight estimating for next iteration is given by
$\hat{w}_i(n+1) = \hat{w}_i(n) + \mu x(n)\ e(n)$, i = 0, 1, ….. m-1
Where, m is length of the FIR filter.
μ- the constant step size parameter.
Under these assumption LMS convergence in the mean square provided that the value of μ lies in;
$0 < \mu < \frac{2}{\lambda_{max}}$

NLMS:
Normalized Least Mean Square Algorithm The main drawback of the LMS is that it is sensitive to the scaling of its input x(n) . This makes it very difficult to choose a learning rate μ that guarantees stability of the algorithm. NLMS solves problems by normalizing with the power of the input. The weight estimation in LMS algorithm for the next iteration of the filter is determined by equation;
$\hat{W}(n+1) = \hat{w}(n) + \mu\ x(n)\ e(n)$.
Here μ is the step size parameter governed by designer.
The source or input vector $\vec{x}(n)$ and estimation of error e(n) is done with help of iterative cycle n.
As input vector x(n) is responsible for variation in the filter directly and if large x(n) at the input then LMS has to suffer from gradient noise amplification. Hence a normalized LMS algorithm is used. In the NLMS for the n+1 iteration variation in weight of the filter is normalized with respect to the square Euclidean norm of the input vector x(n) at the iteration n-cycle. Hence named as normalized.
NLMS has similar structure but difference in weight adjustment variation.

Weight estimating for next iteration is given by

$$\hat{w}(n+1) = \hat{w}(n) + \frac{\tilde{\mu}}{||X(n)||^2}\ x(n)\ e^*(n)$$

For computing weight of the filter in NLMS the desired equation is equation no (10). In that the product x(n)e*(n) is normalized w.r.t. Squared Euclidean of input x(n). [12]

VI. EXPERIMENT SET UP:

For the result and waveforms, LMS and NLMS filter are implemented on the MATLAB having Filter order = 8. For LMS algorithm step size taken is 0.008. The instrument used is tabla. In first step the signal from the instrument is taken then the signal is mixed with the Guassian and pink 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 output MSE, MAE, PSNR and SNR is measured for both the filters.

VII. RESULT:

Input and Output waveform:

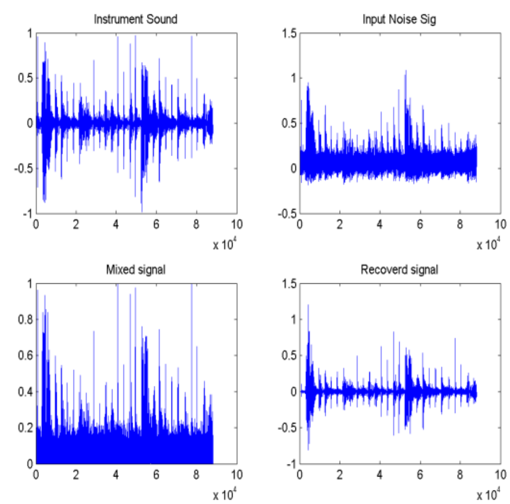

Figure No.4 Instrument Tabla, Tap 8, Noise Gaussian, Filter LMS

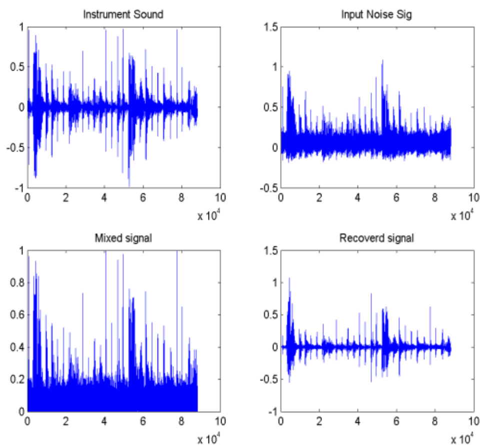

Figure No.5 Instrument Tabla, Tap 8, Noise Gaussian, Filter NLMS

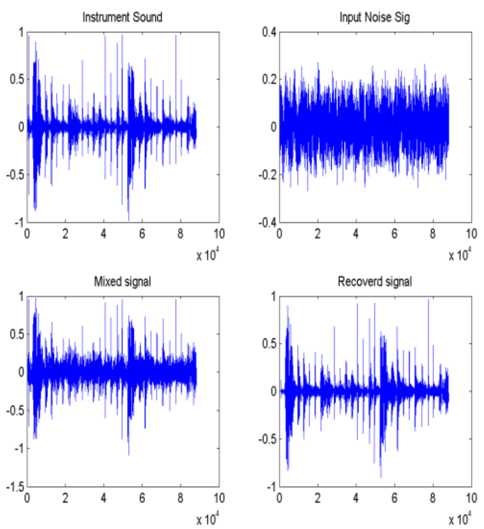

Figure No.6 Instrument Tabla, Tap 8, Noise Pink, Filter LMS

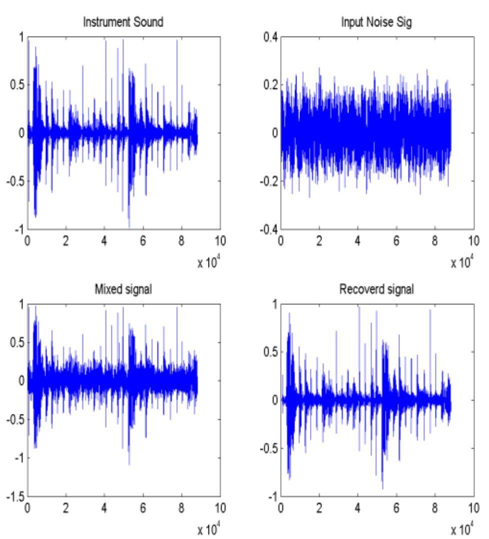

Figure No.7 Instrument Tabla, Tap 8, Noise Pink, Filter NLMS

Table :

Filter Tab Size =8

S N	Noise Type	Filter Type	MSE_ OUT	MAE_ OUT	PSNR OUT	SNR_ OUT
1	Gaussian	LMS	0.0078	0.05967	21.06	0.49
2	Gaussian	NLMS	0.0069	0.05632	21.56	-0.30
3	Pink	LMS	0.0046	0.00346	23.32	4.90
4	Pink	NLMS	0.0043	0.00329	23.57	5.40

VIII. CONCLUSION:

From the result obtained from both the adaptive filter LMS and NLMS algorithms, it can be concluded that the noise cancellation is possible using adaptive filters for the musical instrument tabla. From the result we can also say that output PSNR of NLMS is than LMS algorithm but it take more time to give output.

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