Cancellation of Noise using Adaptive Filter in the Signal of Trumpet musical Instrument

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ABSTRACT

As far as receiver end of the communication system is concerned all the received signals are unpredictable and random in nature. The received signals may be of any type speech , computer data, video and music signals all carry useful information. The noise which is unwanted signal interfered with the useful signal in the medium most of the time and will very difficult to decode at the receiver end. The noise guassian noise which is present in all the frequency range is random in nature. So it has to deal properly otherwise false information is convey. Many methods are implemented earlier. In the present work we have implemented adaptive filter using adaptive algorithm Least mean square (LMS) and Normalized Least mean square (NLMS) to cancel or minimize the effect of the guassian noise present in the signals of Trumpet musical instrument. The work is done using matlab software. Wave form and result which include output PSNR is measured.

1. Introduction

Speech signals are having alternative(repetition) random sequences in nature. Speech signals have bandwidth of 4 khz[2]. In speech signal power distribution i.e. PSD is at low frequency end. The musical signal is considered as the signals obtained from the musical instruments or from the singer. In musical signal the basic component is tone. All tones come from a resonance frequency. The highest frequency a human's sound system limits to achieve is 4168 Hz. Musical instruments cover more frequencies than audible bandwidth which is limited to 20 kHz. In the processing of the speech or musical signals use of digital computers, microprocessor, microcontroller, and specific digital signal processor are in practice. For filtering of the signals, the digital filters are often used[15].The infinite impulse response (IIR) filters are not practically [1] so for noise cancellation can be done with the finite impulse response (FIR) filters. But rapid variation in the characteristics of the noise, simple FIR filters are not suitable. Hence we have to use adaptive filters.

S Dixit, D Nagaria had investigated to provide solution for improving the performance of noise canceller with filter of parameters. They obtained the result by adaptive algorithm having variable step size and different initial weight of the filters. By doing this they got high convergence speed. The innovative concept for adaptive noise cancellation (ANC) used is cascaded form of least-mean-square (LMS) adaptive filters[3].

Elimination of background noise from the main signal of speech can be done with the help of adaptive filter is the proposed by Meera V. Boramanikar1 & Anjali C. Pise. In their work they used adaptive filtering algorithms LMS and NLMS for controlling the adaptive noise[4].

The statistical characteristics of signal and noise are usually unknown and very hard to fix the digital filter coefficient. For dealing with this problem Ying He, HongHe, LiLi, Yi Wu have design and simulate LMS and the RLS algorithms. They compare and analyze the result and give advantage and disadvantage of both algorithms. They simulates the adaptive filter with MATLAB [5].

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 B have put on the efforts in design and implementation of LMS and Averaging algorithm[6]. Mamba'usSa'adah, DiahPuspitoWulandari and YoyonKusnendarSuprapto 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[7].

2. Fundamental Theory:

Trumpet :

For classical and Jazz ensembles the most often used brass instrument is trumpet. It ranges from piccolo trumpet with highest to bass trumpet one octave below. In battle or hunting trumpet have been used as a signaling instrument in late 14th or early 15th century. Now Trumpets found use in art music styles, orchestras, concert bands , jazz and in popular music. Trumpet is played when we blow air via small opening of mouth just like closed lips , produces sound like buzzing which starts standing wave vibration in the air column which is present in the instrument. It is mostly constructed of brass tubing with twice bent in rounded rectangular shape. In the modern trumpet pitch changing is possible while changing the length of tubing. The trumpet consists of eight combinations of three valves of piston type or rotary type .For lowering the pitch of the trumpet, the valve has to be engaged which in turn increases the tubing and we obtained the desired pitch. The frequency range of trumpet is 170 Hz to 1 KHz [8][9][10].

Noise:

The noise which is having the properties of the random vaiable. A random variable may be discrete or continuous. Cumulative distribution function enables us to have the probabilistic description of a random variable which include both discrete or continuous. Probability density function is an alternative description of the probabilistic description of a random variable. The concept of a random or stochastic process is the logical extension of the random variables. The stochastic process has been necessary for statistical analysis of random signals encountered in communication systems such as speech signal, video signal, digital data end electrical noise. These signals are function of time and random in nature. The random or stochastic process whose statistical characteristics do not change with the time is called stationary stochastic process. The noise process representing the noise in a medium is a stationary process because its statistics like mean value, mean square value do not change with time. Mean, autocorrelation, covariance of a random or stochastic process are important statistical parameters that provide the partial description of the distribution of the process. These parameters plays key roles for characterizing the stochastic process whose probability density function cannot be determined directly. The process may not be stationary in the strict sense, yet it may have a constant mean value , a constant autocorrelation function that is independent of the time shift and finite autocorrelation function at zero time lag. Such process is called wide sense stationary process. The properties of power spectral density of wide sense stationary process are similar to those of the periodic signals.

The communication channel or medium has some inherent problems like signal attenuation, amplitude and phase distortion, multipath distortion and additive noise interference. While designing a communication system an engineer has to face several limitation. These includes noise limitation, bandwidth limitation and equipment limitation. We focus on the noise limitation.

The noise is defined as an unwanted form of energy which tend to interfere with the desired signals in the communication system. It cannot be nullify completely , but we can minimized the effect of noise in the desired signal with the help of several techniques. Noise can be classify as a external and internal noise. Example of external noise are atmospheric noise, industrial noise etc. and that of internal noise is thermal noise, shot noise etc.

The effect of noise at the receiver will affect the ability to identify the intended or desired signals correctly and thus limits information transmission. If noise power is small then it is unnoticeable and is the case in the most communication. But if noise power is large then noise severely limits the capabilities of communication system.

Gaussian noise is to be consider as a random signal who is having equal intensity at all frequencies which gives constant power spectral density. It has normal distribution having zero mean.

3. Adaptive Filter:

To extract information of prescribed quantity of our interest from the noisy data we used filter or estimator. The filtering and estimation theory have application in communication , radar, biomedical engineering and financial engineering etc. In any application noise is the unwanted signal and it impaired the desired signal. There is three kinds of estimation - filtering, smoothing and prediction are exists. Linear optimum filters are of two types linear and nonlinear. The filter is linear if filtering, smoothing and prediction quantity is a linear function of input of the filter otherwise not linear filter.

In the linear filtering for the statistical approach mean and correlation functions of information signal and additive noise must be present for minimizing the effect of noise from the noisy data. The difference between desired response and actual output of the filter is called error signal. To minimized the mean square value of the error signal is useful approach for filter optimization problem. For stationary inputs the resulting solution is known as wiener filter which is optimum in the mean square error sense But non stationary signal wiener filter is not applicable for such cases Kalman filter found solution. Both Wiener and Kalman filter is well developed for continuous and discrete time signals.

For wiener filter priori information of statics of data which is to processed is required and should match with the characteristics of input data otherwise the design of the filter is not optimum. In non optimum case filter design used is estimate and plug approach. Were first estimate the statistical parameters of the signal and then result is plug into nonrecursive formula for computing the filters parameters. When real time operation is involved this procedure required excessive elaborated and costly hardware. Thus to overcome the limitation adaptive filters are use.

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 set of the initial conditions about the environment and if environment is stationary the algorithm converges to optimum wiener solution in some statistical sense after successive adaption cycles of the algorithm. And if the environment is non stationary then the algorithm offers a trackling capability.

The direct consequence of the recursive algorithm is that the parameters of an adaptive filter become data dependent. Hence we can say that the adaptive filter is a non linear system as it does not obey the principle of superposition. The choice of algorithm is done on the basis of rate of convergence, misadjustment, tracking, robustness, computational requirement, structure and numerical properties.

4. Adaptive Algorithm:

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 trumpet musical instrument.. They consists of an iterative process to minimize mean square error. The algorithms 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 ΓM and cross correlation vector Yd are known, then coefficients can be computed iteratively as,

hM(n+1) = hM(n)+1/2Δ(n)s(n), n=0,1,.......

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

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

Δ(n) – step size at the nth iteration.

s(n) – direction for the nth iteration.

hM(0) is chosen arbitrarily.

For minimization of J(hM) with ΓM and Yd 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

hM(n+1) = hM(n)-1/2Δ(n)g(n)

and by substituting for g(n) we get,

hM(n+1) = (I- Δ(n) ΓM)hM(n)+Δ(n)Yd

hM(n) converges to hopt as n→ ∞, the sequences of step size Δ(n) should be absolutely summable with Δ n → 0 as n→ ∞, also as n→ ∞, g(n) → 0.

NLMS Algorithm:

Parameters:

p= filter order

μ= step size

Initialization: ĥ(0) = zeros (p)

Computation : n= 0,1,2,….

X(n)= [x(n), x(n-1), … x(n-p-1)]T

e(n)= d(n) - ĥ H (n)X(n)

ĥ(n+1) = ĥ(n) + μe*(n)X(n)/X H (n)X(n)

5. 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 trumpet. In first step the signal from the instrument is taken then the signal is mixed with the Guassian noise and afterwards noise is minimized from the mixed signals. The waveforms of instrument signal, noise signal, mixed signals and recovered signalis shown in the result. Also PSNR is measured for both the filters.

6. Result:

For LMS Filter with Gaussian Noise :

The instrument sound signal is the input to the filter which is taken from the musical instrument trumpet, 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 = 20.0225

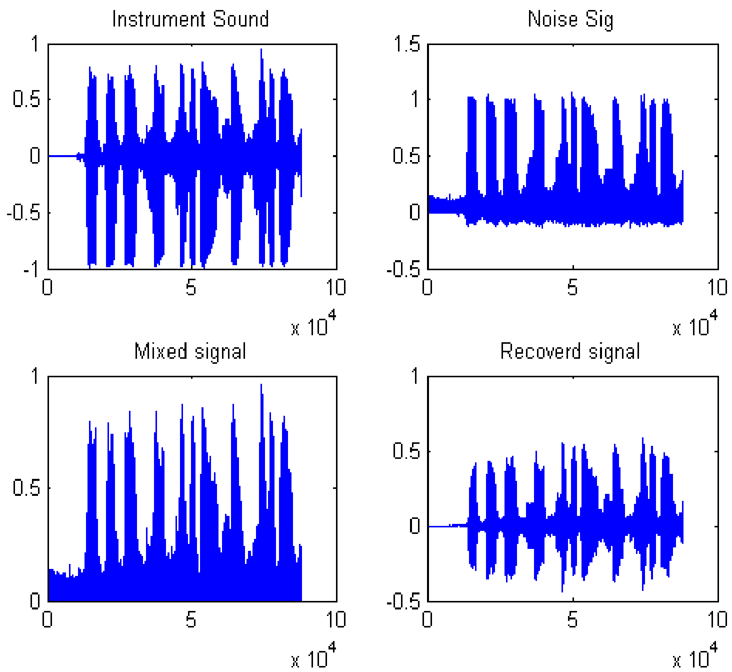

Fig.1Signals for LMS filter with Tab size 8.

2] Tab size = 32 , mu = 0.008 Out_psnr = 20.8226

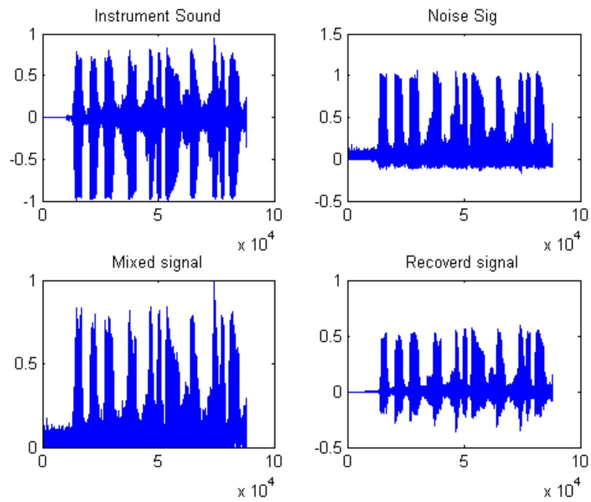

Fig.2Signals for LMS filter with Tab size 32.

For NLMS Filter with Gaussian Noise :

The instrument sound signal is the input to the filter which is taken from the musical instrument trumpet , 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 = 20.08

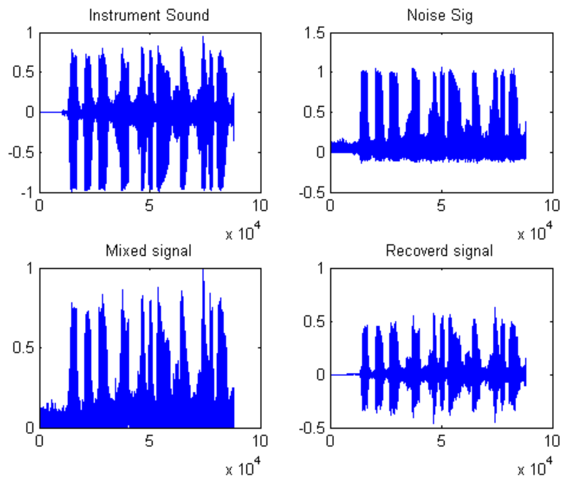

Fig3Signals for N LMS filter with Tab size 8.

2] Tab size = 32 , mu = 0.008 Out_psnr = 20.823

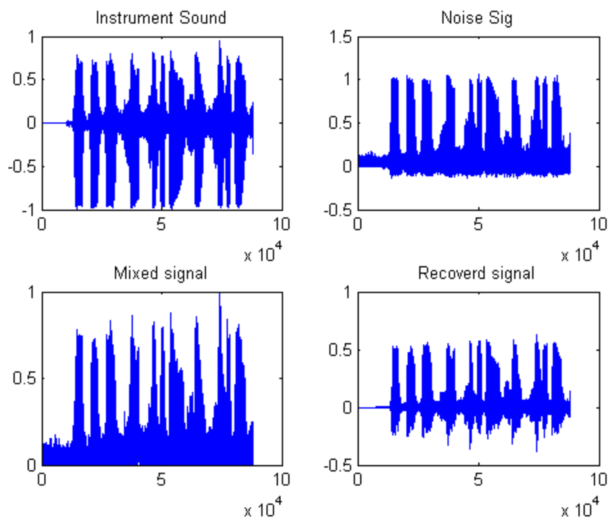

Fig4 Signals for N LMS filter with Tab size 32.

Table:1

Filter Name	Noise	Out-PSNR Tab Size =8	Out-PSNR Tab Size = 32
LMS	Gaussian	20.0225	20.8226
NLMS	Gaussian	20.08	20.823

Table 1 Results

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

7. Conclusion:

From the result it is concluded that NLMS filter has high output PSNR than LMS filter. The Tab size for the filters is very important parameter for considering in design according to the application with stability constraint. Also we can say that adaptive filter can be used for noise cancellation from the signals of musical instruments.

References:

[1] S.Haykin, “Adaptive filter theory”, 5th Edition, Prentice Hall.

[2] Cox, Richard V. "Current methods of speech coding." International journal of high speed electronics and systems8.01 (1997): 13-68.

[3] Design and Analysis of Cascaded LMS Adaptive Filters for Noise Cancellation.S Dixit, D Nagaria Springer - February 2017, Volume 36, Issue 2, pp 742–766

[4] Adaptive Noise Cancellation in Speech Signal Meera V. Boramanikar& Anjali C. Pise Electronics & Telecommunication Department SKN Sinhgad College of Engineering,Korti, Pandharpur- 413304,India.

[5] YingHe, HongHe, LiLi, Yi Wu: The Applications and Simulation of Adaptive Filter in Noise Cancelling. International Conference on Computer Science and Software Engineering, © 2008 IEEE, Tianjin University of Technology, China.

[6] Niranjan, D., and B. Ashwini. "Noise cancellation in musical signals using adaptive filtering algorithms." 2017 International Conference on Innovative Mechanisms for Industry Applications (ICIMIA). IEEE, 2017.

[7] Sa'adah, Mamba'us, DiahPuspitoWulandari, and YoyonKusnendarSuprapto. "Noise Cancellation in Gamelan Signal by Using Least Mean Square Based Adaptive Filter." International Journal of Simulation--Systems, Science & Technology 19.3 (2018).

[8] www.orsymphony.org.

[9] "Trumpet, Brass Instrument". dsokids.com. Retrieved 3 May 2008.

[10] Trumpet". www.etymonline.com. Online Etymology Dictionary. Retrieved 20 May 2017.

[11] Principles of Communication Systems by Herbert Taub and Donald Schilling, 3rd Edition, McGraw Hill.

.[12] Manohar, Kotta, and Preeti Rao. "Speech enhancement in nonstationary noise

[13] Er. Gaurav Badhwar, “Noise Reduction using Kalman Filter”, International Journal of LatestTrends in Engineering and Technology, vol 7, issue 1, may 2016.

[14] Hoff, M. E., and B. Widrow. "Adaptive switching circuits." 1960 IRE WESCON Convention Recor. 1960.

[15]Kirankumar, K., M. Suneel, and P. Surendrakumar. "Hybrid Technique for Denoising Multi Environment Noise in Speech Processing." 2018 9th International Conference on Computing, Communication and Networking Technologies (ICCCNT). IEEE, 2018.