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An Iterative Pruning Approach of Neural Network for Proficient Noise Cancellation

Shashi Kant Dargar, Himanshu Purohit, S C Mahajan

Abstract*— For Active Noise Cancellations various algorithms run and result in standard output and better performance. Technique to pruning is related to prune nonfunctional neuron's (algorithm) from ANC network, which makes only best neurons responsible for noise cancellation. Neurons are classified as different algorithms. Performance of neurons depends upon instantaneous surrounding conditions. So a proficient novel approach for ANC has been proposed. The wiener filter based on least means squared (LMS) algorithm family is most sought after solution of ANC. This family includes LMS, NLMS, VSLMS, VSNLMS, VFXLMS, FX- sLMS and many more. Some of these are nonlinear algorithm, which provides better solution for non linear noisy environment. The components of the ANC systems like microphones and loudspeaker exhibit nonlinearities themselves. The non linear transfer function create worse situation. For example, FX-sLMS algorithm behaves well than the second order VFXLMS algorithm in conditions of non-minimum phase and most important is the mean square error. The classical approach to RBF implementation is to fix the number of hidden neurons based on some property of the input data, and estimate the weights connecting the hidden and output neurons using linear least square method.

***Index Terms*— Least Mean Square; RBF neural network; Artificial neural Network, Filter bank design, ANC**

I. INTRODUCTION

Echo is delayed and degraded version of original signal which travels back to its source after several reflections or because of some other reason. Nature of echo signal can be either acoustic or electrical, and in order to reduce its undesired effect we employ echo cancellers. Design of echo cancellers requires an application of adaptive filter theory. Echo cancellers must work within specific time limits so adaptive algorithms must provide fast convergence of filter parameters. We've been applying echo cancellers successfully for many years, but we always tend to improve them and increase their efficiency. The wiener filter based Least means squared (LMS) algorithm family is most sought after solution of ANC. This family includes LMS, Fx-LMS, VFx-LMS, FsLMS and many more . Then there are Kalman filter algorithms which are basically based on recursive least square algorithm. Some of these are nonlinear algorithm, which provides better solution for non linear noisy environment. The components of the ANC systems like microphones and loudspeaker exhibit nonlinearities themselves. The non linear transfer function of primary and secondary path may itself create worse situation. If there is only one reference microphone, one loudspeaker and one error microphone then the situation is termed as Single channel case in ANC.

Single channel is not sufficient for complex real time application, where noise is normally of multi order nonlinear characteristics. Recently, done work suggests several methods for the development of active control of noise process for a single channel case. Multichannel approach model is close to actual conditions of field or area where noise removal is required. Nonlinear Multichannel model with trigonometric expansion give better results than volterra series method [4]. The FsLMS algorithm behave good than second order VFXLMS algorithm in conditions.

II. BACKGROUND AND LITERATURE REVIEW

The traditional approach to acoustic noise control uses passive techniques such as enclosures, barriers, and silencers to attenuate the undesired noise. These passive silencers are valued for their high attenuation over a broad frequency range. However, they are relatively large, costly, and ineffective at low frequencies. Mechanical vibration is another related type of noise that commonly creates problems in all areas of transportation and manufacturing, as well as with many household appliances. Active Noise Control (ANC) involves an electro acoustic or electromechanical system that cancels the primary (unwanted) noise based on the principle of superposition of an antinoise wave of equal amplitude. S.M. Kuo and D. R. Morgan (1996) in their paper have emphasized the practical aspects of ANC systems in terms of adaptive algorithms and DSP implementations for real-world applications [4]. Widrow and Hoff (1960) developed the least mean square algorithm (LMS). This algorithms is a member of stochastic gradient algorithm algorithms, and because of robustness and low computational complexity, it has been used in wide spectrum of applications [6]. Least square solution is not very practical in the actual implementation of adaptive filters, this is because we know all the pas samples of the input signal, and as well the desired output signal must be available at every iteration. Hence RLS (Recursive least square) algorithm is based on the least square estimate of the filter coefficient and iterations. Performance with less computational complexity compared to the Second-order VFXLMS algorithm [9], [11]. Debi Prasad das and G. panda in 2004 has shown a new approach for nonlinear processes using Filtered-s LMS (FsLMS). They proved that using a nonlinear controller for a linear device can achieve superior performance when the secondary path is liner non minimum phase and the reference noise is non Gaussian and predictable.

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FSLMS offers a computational advantage [10]. Efficient adaptive Volterra filters for ANC with linear secondary path entitled paper 2004 developed an algorithm called Volterra filtered-x LMS (VFXLMS) algorithm Lu Yingwei, Narashiman Sundararajan, Fellow, IEEE, and P. Saratchandran, Senior Member, IEEE in 1998 presented paper on Performance Evaluation of a Sequential Minimal Radial Basis Function (RBF) Neural Network Learning Algorithm [1].

In this researchers have begun to examine the use of radial basis functions RBF's for solving function approximation and pattern classification problems. In the classical approach to RBF network implementation, the basic functions are usually chosen as Gaussian and the number of hidden units is fixed a priori based on some properties of the input data. The algorithm is suitable for sequential learning and is based on the idea that the number of hidden units should correspond to the complexity of the underlying function as reflected in the observed data [12]. The outcomes of the literature survey can be summarized as follows: The exponential increase of noise pollution and ineffectiveness of passive techniques for noise mitigation have led to the development of the ANC Systems. Active noise cancellation methods are the current topics for research. ANC cancels the unwanted noise by generating antinoise of equal amplitude and opposite phase through the secondary sources.

III. IMPLEMENTATION

The experimental work has been done on MATLAB and Texas's dsk kit TMS3206713 series. DSK 6713 is audio video application processor, capable to perform real time processing on signals. Various program based on individual algorithm were written first e.g. LMS, NLMS, FSLMS, VFXLMS, RLS. Each of them is considered as a separate hidden neuron of network.

In case two primary noise signal is chosen to be a logistic chaotic type $x(n+1)=\lambda x(n)[1-x(n)]$ $\lambda=4$, $x(0)=0,.9$[2]

All the neurons were allowed to run for 500 iterations, and respective obtained results were compared with standard threshold value.

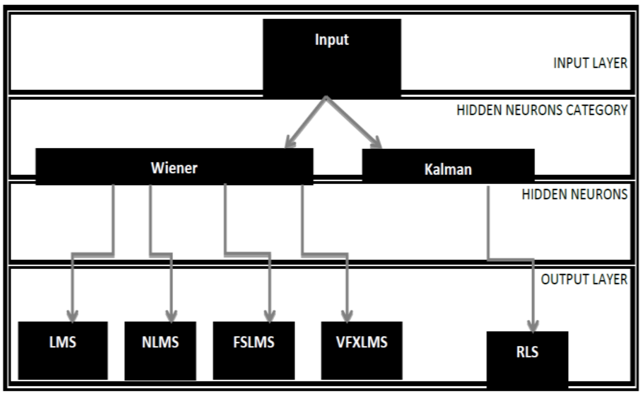

Figure 1. ANN based ANC model

LMS algorithm weight update equation is as

$$w(n+1)=w(n)-\mu e(n)v(n) \qquad (1)$$

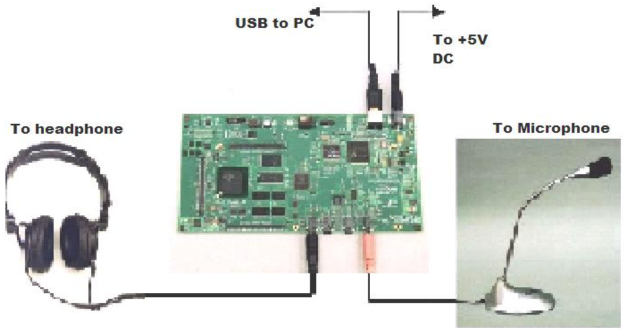

Figure 2. Noise Cancellation TMS320DSK6713

$v(n)=x(n)*a(n)$ where $a(n)$ is approximation of secondary path transfer function . $w(n)$ if filter weight , μ is step size factor[2]. The calculation of threshold depends upon number of parameter and it may vary according to surrounding situation and instantaneous noise characteristics. We had adopted threshold parameter $Ø_k$ Which was based on following equations.

For each input X_n

$$Ø_k(X_n) = \exp(-1/\sigma^2(\|X_N - \mu_K\|^2)) \qquad (2)$$

Where σ is variance, μ is is weight factor of particular neuron [5].

This approach is based on pruning technique of ANN. We are using this technique to prune less effective neurons from network and finally running systems with successful neuron only. In different conditions different neuron may give better results. The survival chances of neurons depend upon its algorithm efficiency in given circumstances. The step $\mu$was size varied factor between .005 to .1, depending upon the optimization required between complexity and amount of noise cancellation.

IV. RESULT AND ANALYSIS

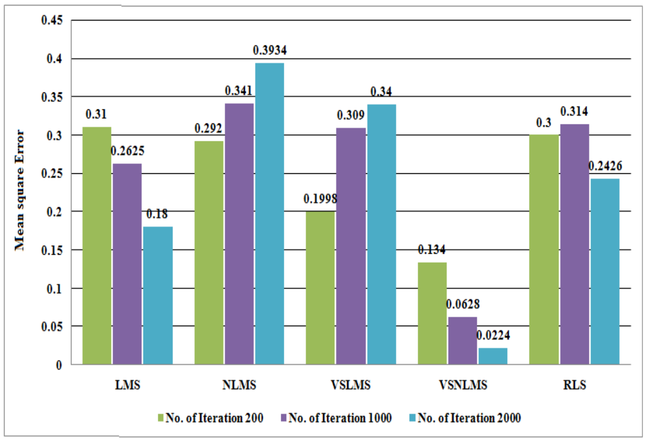

Figure 3. Graphical analysis of MSE for algorithms for number of iterations=200, 1000 and 2000.

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Applied Algorithm	No. of Iteration 200	No. of Iteration 1000	No. of Iteration 2000
LMS	0.31	0.2625	0.18
NLMS	0.292	0.341	0.3934
VSLMS	0.1998	0.309	0.34
VSNLMS	0.134	0.0628	0.0224
RLS	0.3	0.314	0.2426

Table1. MSE of Various algorithms for number of iterations=200, 1000 and 2000.

V. COMPUTATIONAL COMPLEXITY

Application of pruning criteria to eliminate non efficient neuron where neuron is the different algorithm, for the selection of proficient algorithm can be selected for ANC. The performance of the algorithm depends upon the type of input. We found that (refer table 5.1) VSNLMS is doing better corresponding to the input 'son'. Here we used the pruning criteria for deciding threshold value [1]. And the simplified formula is given as

$$\varphi_k(x) = \exp\left(-\left(\frac{1}{\sigma^2}\right) \parallel x - \mu_k \parallel^2\right) \qquad (3)$$

where $\mu_k(x)$ is response of k_{th} hidden unit.
In our case, implementation of the concept of RBF neural network approach can be used after simplification $\sigma = 1$, and mean square error μ variance is normalized by factor 0.05 Threshold criteria results in selection of the efficiently working neuron as efficient algorithm can be given in simplified manner as :

$$Th = MSE - \left(\frac{1}{20}\right) average\ mean \qquad (4)$$

Where *Th* is the threshold value and *MSE* is Mean square error *1/20* is the normalizing factor.

Algorithm	Iterations	MSE	Threshold	Deviation	Survived
LMS	200	0.31	Th=MSE-0.05[mean]	0.297655	0.010055
	1000	0.2625		0.250155	
	2000	0.18		0.167655	
NLMS	200	0.292		0.279655	
	1000	0.341		0.328655	
	2000	0.3934		0.381055	
VSLMS	200	0.1998		0.187455	
	1000	0.309		0.296655	
	2000	0.34	Mean of the MSE obtained 0.2469	0.327655	
VSNLMS	200	0.134		0.121655	
	1000	0.0628		0.050455	
	2000	0.0224		**0.010055**	
RLS	200	0.3		0.287655	
	1000	0.314		0.301655	
	2000	0.2426		0.230255	

Table2. Application of threshold criteria for MSE obtained by various algorithms.

VI. CONCLUSION AND FUTURE WORK

It was found out those in particular noisy conditions most suitable algorithm can be sort out using our pruning method. This method can also be applied to noise cancellation, vibration cancellation and signal estimation problems. Mean of the MSE obtained 0.2469 deviations from mean when calculated; it is observed that VSNLMS is performing better than the other utilized application for ANC of "Song" input for no. of iterations 2000 as the threshold criteria results in (0.0224-0.2469= 0.010055). 0.010055 is the least value amongst the all applied algorithm, hence pruning of all other algorithm inculcate the selection of that algorithm as survival neuron. Next good performing algorithm is LMS itself for 2000 iterations. The surprisingly good performance of the LMS is due to linear nature of the 'song' input noise. Here in this project have tried to reduce such weed signals by using basic adaptive approach. But in real time applications which involve thousands of such weed signals and knowing the fact that noise, at any cost, cannot be cancelled to cent percent, this basic approach cannot do us a good favour and keeps this research topic active in upcoming years. The future task is to apply this novel concept for Kalman and Wiener filter's family algorithm simultaneously. It is also being tried to implement it for outer noise removal in automobiles and noise cancellation in head phones. Noise performance can be upgraded by finding proper step factor and threshold value. The algorithm level up gradation of threshold and pruning criteria may improve performance.

VII. ACKNOWLEDGMENT

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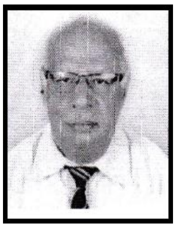

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