IOSR Journal of Computer Engineering (IOSR-JCE)
e-ISSN: 2278-0661,p-ISSN: 2278-8727, Volume 17, Issue 6, Ver. III (Nov – Dec. 2015), PP 22-28
www.iosrjournals.org

Adaptive Digital Filter Design for Linear Noise Cancellation Using Neural Networks

Nikhil Verma[1], Rajesh Mehra[2]
[1](*Department of ECE, National Institute of Technical Teachers Training & Research, Chandigarh, India)*
[2](*Department of ECE, National Institute of Technical Teachers Training & Research, Chandigarh, India)*

Abstract: *Noise is the most serious issue in the filters and adaptive filters are subjected to this unwanted component. This paper deals with the problem of the adaptive noise and various adaptive algorithms functions which when implemented practically shows that the noise is cancelled or removed by the neural network approach using the exact random basis function. The adaptive filters are used to control the noise and it has a linear input and output characteristics. This approach is done so as to get the minimum possible error so that to obtain the error free desired signal. The designed filter will reduce this noise from measured signal by a reference signal which is highly correlated with the noise signal. This approach gives excellent result for this signal processing technique that removes or eliminates the linear noise from the different functions. The simulation results are also mentioned so as to gives a vivid idea of reduced noise using neural networks algorithm.*

Keywords: *Adaptive filter, Exact Random Basis Function (RBF), Linear Noise Cancellation, Neural Networks.*

I. Introduction

Filters are the devices that remove some unwanted components from the unwanted signal to obtain the desired signal. Adaptive digital filters are self learning filters, whereby an FIR (or IIR) is designed based on the characteristics of input signals. No other frequency response information or specification information is available. There are a large number of applications suitable for the implementation of adaptive digital filters. The noise cancellation is the most common practical application of adaptive filters [1]. An adaptive filter is a linear filter which has a transfer function and that transfer function is controlled by the changeable parameters and that parameters are optimized by the use of different algorithms. In this paper, the algorithm used is the neural networks algorithm. The adaptive filter emergences since the 1960s and its theory is continuously developing and improving [1]. Adaptive filters have the capability to manage the impulse response to remove out the correlated signal in the input. They require little or no a prior knowledge of the signal and noise characteristics [2].

The block diagram of the adaptive digital filter is shown:

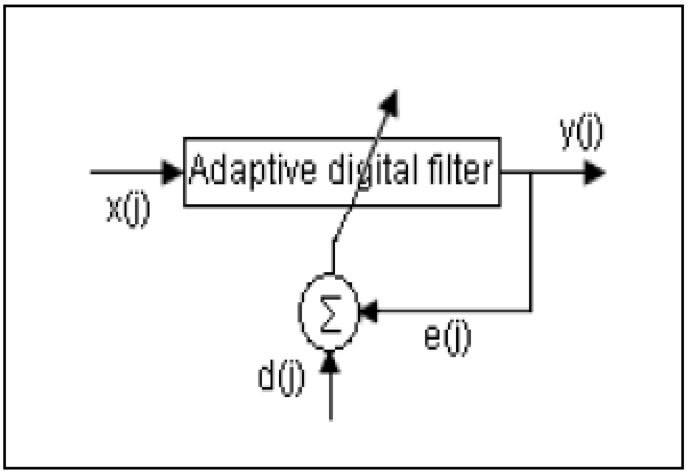

Fig. No.1 Adaptive digital filter [1]

Adaptive filters are widely used in telecommunications, control systems, radar systems, and in other systems where minimal information is available about the incoming signal. Due to the variety of implementation options for adaptive filters, many aspects of adaptive filter design, as well as the development of some of the adaptive algorithms, are governed by the applications themselves.

II. Adaptive Noise Cancellation

An adaptive filter is a filter that self handles the transfer function according to an optimized algorithm. Because of the nexus of optimizing algorithm, most adaptive filters are digital filters that perform digital signal processing and show their performance based on the input signal. Some of the applications require adaptive coefficients since some parameters of desired processing operation are not known in advance [3]. In such cases, this filters uses feedback to refine the values of filter coefficients and somewhat its frequency response.

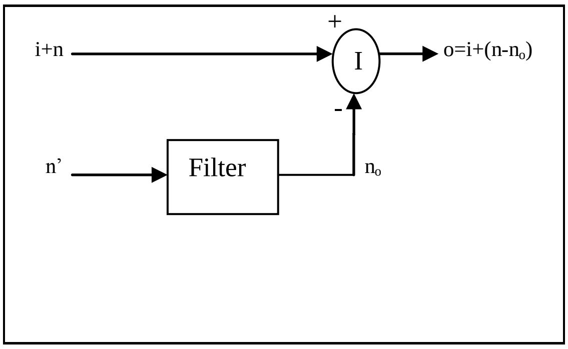

Fig. No. 2 Adaptive noise concept

It makes use of an auxiliary or reference input which contains a correlated estimate of the noise to be cancelled. The reference can be obtained by placing one or more sensors in the noise field where the signal is absent or its strength is weak enough. Subtracting noise from a received signal involves the risk of distorting the signal and if done improperly, it may lead to an increase in the noise level. This requires that the noise estimate n_o should be an exact replica of n. If it were possible to know the relationship between n and n_o, or the characteristics of the channels transmitting noise from the noise source to the primary and reference inputs are known, it would be possible to make n_o a close estimate of n by designing a fixed filter. However, since the characteristics of the transmission paths are not known and are unpredictable, filtering and subtraction are controlled by an adaptive process. Hence an adaptive filter is used that is capable of adjusting its impulse response to minimize an error signal, which is dependent on the filter output.

Due to repeatedly changing of noise over a time period and overlapping of frequencies of noise and signal, adaptive filtering is a necessity nowadays. They have proved to be a useful resource for real time applications when there is less time for statistical estimation. The ability of adaptive filters to operate satisfactorily is unknown and possibly time-varying environments without user intervention and improving their performance during operation by learning statistical characteristics from current signal observations has made them more efficient [4]. A typical structure of a noise cancelation system is shown in figure (3) where additive noise, n (k), corrupts the information signal, s (k), resulting in the noise signal, d (k). The noise and information signals are assumed to be uncorrelated. The principle of noise cancelation is based on the assumption that both the noisy signal d (k), and a filtered or distorted measurement of the noise, named reference noise x (k), are available. Noise x (k) is considered to pass through a channel with a transfer function T(.) [2]. Noise Cancellation is a variation of optimal filtering that involves producing an estimate of the noise by filtering the reference input and then subtracting this noise estimate from the primary input containing both signal and noise. An adaptive filter is a filter that self adjusts its transfer function according to an optimizing algorithm. Because of the complexity of optimizing algorithm most adaptive filters are digital filters that perform digital signal processing and adapt their performance based on the input signal. For some applications adaptive coefficients are required since some parameters of desired processing operation are not known in advance.

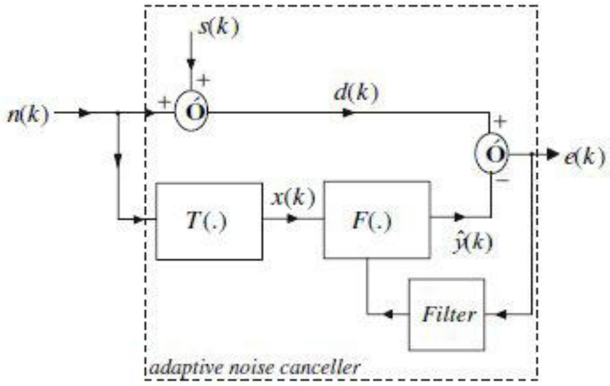

Fig. No.3 Adaptive Noise Canceller [2]

In such situations adaptive filters which uses feedback are used to refine the values of filter coefficients and hence its frequency response. Generally speaking, the adapting process involves the use of a cost function, which is a criterion for optimum performance of the filter (for example, minimizing the noise component of the input), to feed an algorithm, which determines how to modify the filter coefficients to minimize the cost on the next iteration. The error signal or cost function is the difference between the desired and the estimated signal.

III. Nn Based Adaptive Filters

Neural Networks (NN), in general, is a highly interconnected network of a large number of processing elements called a neuron in an architecture inspired by the brain. An NN can be massively parallel and therefore is said to exhibit parallel distributed processing. The architecture here we are using is RBF (Radial basis function). This type of network usually has only one hidden layer with special "neurons". Each of these "neurons" responds only to the inputs signals close to the stored pattern. The behavior of this "neuron" significantly differs from the biological neuron. In this "neuron", excitation is not a function of the weighted sum of the input signals. Instead, the distance between the input and stored pattern is computed. If this distance is zero, then the "neuron" responds with a maximum output magnitude equal to one. This "neuron" is capable of recognizing certain patterns and generating output signals being functions of a similarity.

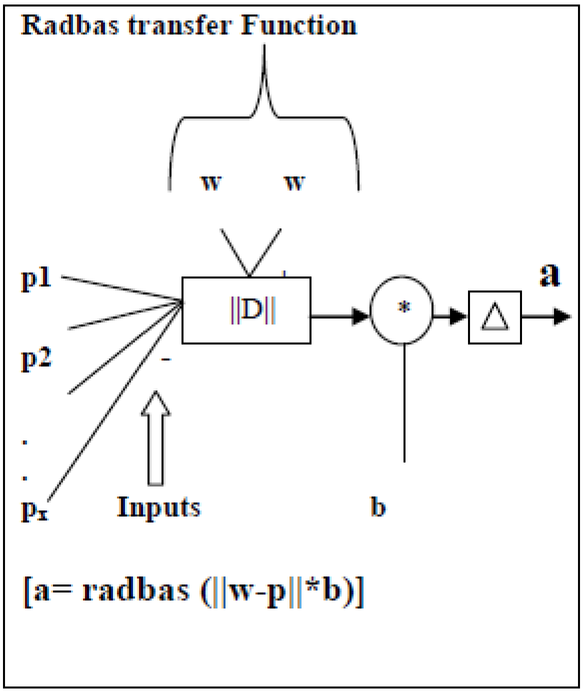

Fig.No.4 The random function block diagram

Neural network types vary from those with only one or two layers of single direction logic, to complicated multi–input many directional feedback loops and layers. On the whole, these systems use algorithms in their programming to determine control and organization of their functions. Most systems use "weights" to change the parameters of the throughput and the varying connections to the neurons. The neural networks can be autonomous and learn by input from outside or even self-teaching from written-in rules. The net input to the radbas transfer function is the vector distance between its weight vector w and the input vector p, multiplied by the bias *b*. The (d) box in this figure accepts the input vector p and the single row input weight matrix, and produces the dot product of the two. The expression for the net input of a radbas neuron is different from that of other neurons. The random basis function is used to minimize the error rate so as to obtain signal which has both the input and output closer to each other.

The transfer function for a radial basis neuron is: $\text{Radbas}(n)= e^{-n^2}$

The radial basis function architecture uses the both the radial basis layer and the linear layer. The (newrbe) function layer diagram is shown:

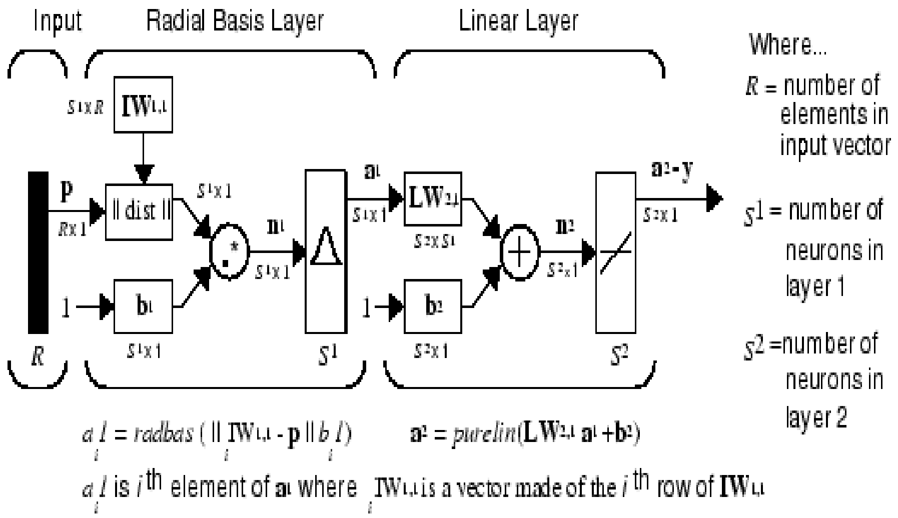

Fig. No.5 Exact Radial basis function algorithm[7]

The newrbe function overshadows the newlin function as it gets the more error free signal that is close to the desired signal. The newrbe function creates a two-layer network. The first layer has radbas neurons, and calculates its weighted inputs with dist and its net input with net product. The second layer has purelin neurons, and calculates its weighted input with dot product and its net inputs with net sum. Both layers have biases. The architecture of the MLP with radial basis function (RBF) units, also called an RBF network, follows the definition of the feed-forward, but it is restricted to only containing one hidden layer of units whose output depends on a distance between the centre of an arbitrary transfer function and a given pattern. The multi-layer perceptron (MLP) is a universal function approximator, as proven by the universal approximation theorem. However, the proof is not constructive regarding the number of neurons required or the settings of the weights. The parameters of an adaptive filter are updated in each iterative step and hence it becomes data dependent.

The adaptive filter is used when the parameters are not fixed or the specifications are unknown. Therefore, this implies the nonlinearity feature of the filter, as it fails to follow the principle of superposition and homogeneity. An adaptive filter is linear if the input output relation obeys the above principles and the filter parameters are fixed. As the parameters change continuously in order to meet a performance requirement, the adaptive filters are time varying in nature. In this sense, we can interpret an adaptive filter as a filter that performs the approximation step on-line. The performance criterion requires the existence of a reference signal that is usually hidden in the approximation step of fixed-filter design. These filters are used because of ease of stability and simplicity in implementation without any adjustment. Adaptive filtering, which concerns the choice of structures and algorithms for a filter that has its parameters (or coefficients) adapted, in order to improve a prescribed performance. [8]

IV. Simulation Results

The simulated result shows the difference between the usages of the different functions to attain the main purpose of the minimum error concept.

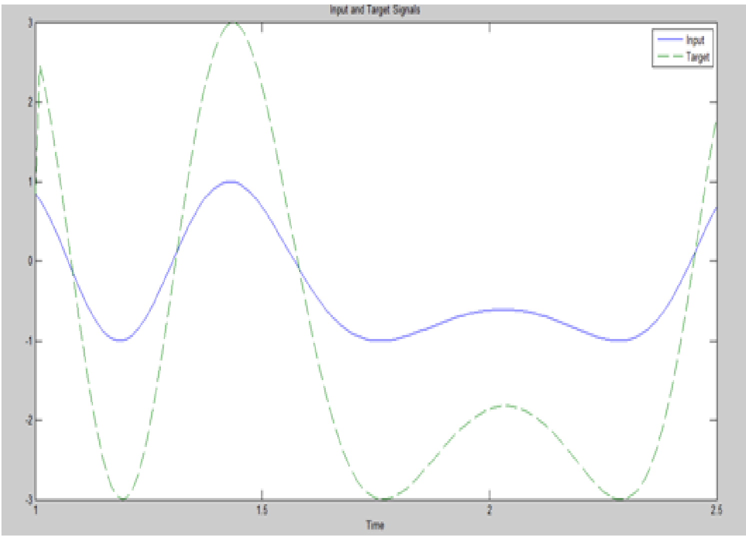

Fig. No.6 The input vs target waveforms.

The above graph shows the input given to the adaptive filter and the desired signal or the target that should be achieved. This graph depicts the newlin function (the linear layer function). The newlin function used here now shows the error estimation in the below graph.

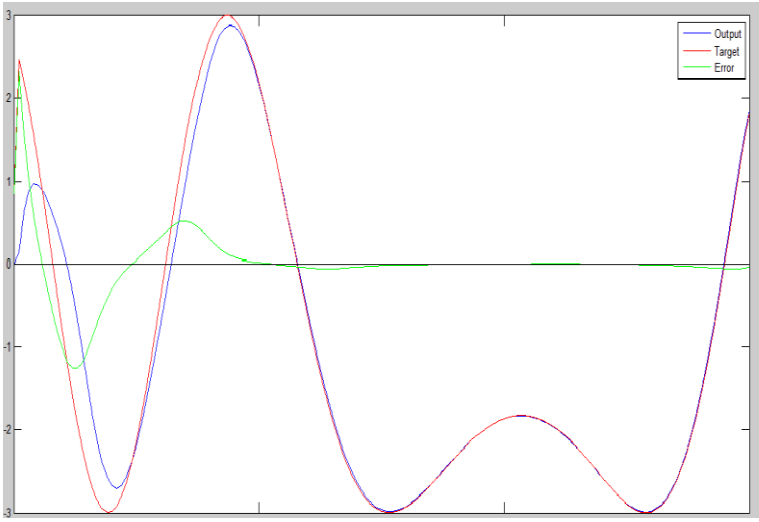

Fig. No.7 The error waveform in linear function.

The graph here shows that when the linear layer function is used to remove the linear noise then the error is more than the desired value. Here the output is shown and in the previous diagram the input is also depicted. The ouput is close to the desired signal or target but the error rate is high in this case. As the motive to reduce the error or get a minimum possible error. This function is reducing the linear noise and also bridging down the gap between the signals but if error has to be reduced by more margin then some other function is required. The function that has have been discussing in the whole paper i.e. the exact random basis function. The newrbe function reduces the error to least possible value as shown in the graph below:

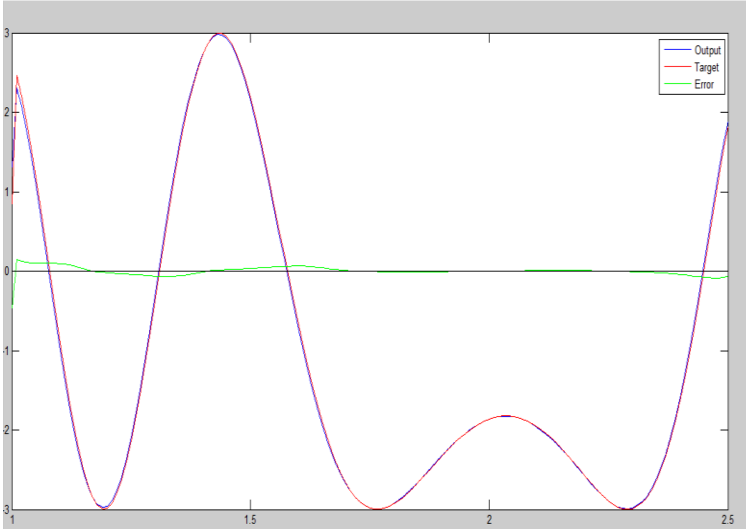

Fig. No. 8 The error waveform in randon basis function.

The result shows that the Random basis function in which the output is nearly equal to the target making the signal error approximately close to zero. The input is same as in case of the linear function but the ouput and the target signal are close to each other making the error value to be least possible. The function here defines that the error is minimized by the huge level. Now, the noise level is reduced considerably well in this case.

V. Conclusion

The adaptive noise cancellation is done by the neural networks algorithm that is defined by many functions but here is a usage of two of the functions i.e. the linear layer function and the random basis approximation. The above results have shown that the random basis function is more preferred due to its lower value of error. The random basis function shows that the output signal much closer to the target or desired signal. The major issue i.e. to get the minimum possible error or to remove the noise level to the considerable low level has been achieved. So, that the signal obtained should resembles the near or same level as the desired one. As depicted the error or unwanted signal is removed by a great factor so as to obtain the signal that is noise free or error free and that obtained siganl can be used for other applications.

Acknowledgements

The author would like to thank Director and Head of Electronics and Communication Engineering Department, National Institute of Technical Teachers' Training & Research, Chandigarh, India and for their constant inspirations, support and helpful suggestions throughout this research work.

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