AFNCS: An Intelligent Adaptive Filter Based Noise Cancellation System for Mobile Communications

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Abstract: ***Noise in an audio signal has become major problem and hence mobile communication systems are demanding noise-free signal. In order to achieve noise-free signal various research communities have provided significant techniques. Adaptive noise cancellation (ANC) is a kind of technique which helps in estimation of un-wanted signal and remove them from corrupted signal. This paper introduces an Intelligent Adaptive Filter Based Noise Cancellation System (AFNCS) that incorporates a hybrid back propagation learning for the adaptive noise cancellation in mobile applications. The proposed hybrid algorithm consists all the significant features of Least Mean Square (LMS) and Gradient Adaptive Lattice (GAL) algorithms. The performance analysis of the method is performed by considering convergence complexity and Bit Error Rate (BER) parameters. The outcomes suggests the errors are reduced significantly when the number of epochs are increased. Also, incorporation of less hidden layers resulted in negligible computational delay along with effective utilization of memory.***

Index Terms - Adaptive Filter, Noise Cancellation, Convergence Complexity, Bit Error Rate (BER), Adaptive noise cancellation, Mobile Communication.

I. INTRODUCTION

In wired or wireless communication systems, noise cancellation has become a prime concern and is considered an open research problem in current era of mass communication of data all over the world. The mechanism and technologies are invented in recent past for noise cancellation from the noise-containing desired signal. The existence of noise in the desired signal may distorts the received signal in random manner which may affect several sources. From the researches of Riahi Manesh et al. [1] and Quadri et al. [2] found that some of the sources like a) non-linearity exist in RF frontend, b) existence of time-varying thermal noise at receiver end and c) noise interference from adjacent environment. In addition, several other factors are affecting the received signal such as crosstalk and electromagnetic interference. Over the past few decades, several denoising techniques are addressed in Quadri et al. [3], Tandra and Sahai [4] and Zeng et al. [5] are divided as gradient-descent Adaptive Filter Algorithm (AFA) and non-gradient AFA. The Gradient descent is a steepest descent which is a multivariate optimization approach and is initialized by assigning initial value and negative gradient to achieve desired local minimum. Earlier various algorithms were provided to identify this desired signal. The Least Mean Square (LMS) algorithm was considered as significant with respect to need of computational efficiency and storage ability at low convergence speed. Also, a normalized LMS algorithm was considered at moderate sped of convergence but it was very slow for colored input signals.

Further, a Recursive least squares (RLS) algorithm was considered an effect with respect to high speed of convergence and tracking ability. However, RLS yields high computational cost. This paper aims to design an Artificial Neural Network based hybrid backpropagation algorithm to achieve better noise cancellation through adaptive control. This paper is categorized as: Section 2- Related works, Section 3 gives the Proposed Solution, Section 4 relates the algorithm implementation, and Section 5 presents the discussion of obtained results while Section 6 conveys the conclusion of the paper.

II. RELATED WORKS

The prior section involves with the discussion of the most recent researches subjected with the mobile applications with different techniques adopted to offer better performance by considering BER, SNR, noise cancellation, interference cancellation, correlation, computational complexity as performance parameters etc. Today, the exponential growth of the mobile applications usage is creating huge traffic in wireless broadband technologies. In Alom and Lee [2010] [6], comparative analysis of beamforming algorithm is performed with respect to correlation values at different angle of signal arrival by using backpropagation algorithm. The work towards achieving accurate results for noise cancellation, Echo cancellation and equalization is found in Kayode et al. (2014) [7] that implements the LMS based adaptive filtering technique for digital audio signals. The [7] simulation result come up with accurate and desired input/output signal by noise signal removal.

Similarly, an experimentation work is observed in Mohammed and Jafar (2014) [8] that adopts the adaptive noise cancellation for health monitoring applications. Here, adaptive filters and modified LMS algorithms were used to eliminate the interference and noise from the mobile phones respectively. A work considering the mobile traffic issue is found in Yoon et al. (2014) [9] and have offered a multicast resource allocation scheme for 4G networks. The experimental analysis of [9] is incorporated with the modulation and coding scheme by considering average PSNR as performance parameter with significant improvement than existing schemes. The implementation of Error Back Propagation (EBP) is found in Roy and Rodrigues (2014) [10] for noise cancellation in the echo corrupted signal by building correlation with pure signal. Here, the voice/speech data are trained using ANN and analyzed [10] performance with respect to SNR, Echo tracking, Error tracking, Echo variance from the recovered pure signal. Most of the researches were incorporated with an aim of interference cancellation schemes for the mobile communication and are perform well under synchronous environment which are not been handled with GMSK modulation schemes. The solution to this scenario is presented in Ruder et al. (2015) [11] with proper modifications in the conventional GSM systems towards the effects of asynchronous co-channel interferences and complexity. The outcomes of the modified GSM system [11] gives the robust results against such asynchronous interferences than unmodified GSM systems.

A significant research towards the phase noise cancellation in shortest range of communications is observed in Zhang et al. (2016) [12] that correlate the quantization of noise cancellation with Least Mean Square (LMS) algorithm yielding high immunity and linearity. This [12] approach has achieved area optimization as well as power optimization with high data rates. Towards the noise removal for real-time sinusoidal signals in healthcare application Kelly et al. (2016) [13] have offered an adaptive filter approach which tracks the sinusoidal frequency and achieved narrow bandwidth results. The experimentation was done with electro-Cortico-Graphic (ECoG) neural data consisting of power line noise and outcomes with enhanced SNR. Similarly, Chilipi et al. (2016) [14] worked for distributed power generation system through adaptive filters for noise/harmonics cancellations to achieve power quality enhancement.

Further, Garcia et al. (2016) [15] used adaptive filter to have noise suppression for echo cancellation, channel equalization, array beamforming in surveillance, tracking and target localization applications. A review work on noise cancellation using adaptive filter based LMS algorithm is presented in Dixit and Nagaria (2017) [16] by considering computational complexity and convergence rate. The similar direction of research is performed by Zheng et al. (2017) [17] and introduced the robust approach by utilizing the adaptive filtering algorithms for acoustic echo cancellations. The [17] approach outcomes with performance improvement with respect to sparsity of impulse response as well as differentiable cost function.

Similarly, Zhang et al. (2017) [18] have come up with a robust solution for echo cancellation application by using LMS algorithm. The outcomes of the simulation using Monte Carlo suggests that the proposed [18] gives robustness in different environments. With an intention to suppression of residual self-interference, Ahmed and Tsimenidis (2018) [19] have used adaptive mean squared error filter for iterative decoding/detective. Here, both the Rayleigh fading and AWGN channels were considered for modulation and its simulation outcomes highlights the system performance with respect to BER, SNR as performance parameters. A novel implementation of adaptive filtering algorithm is found in Menguc (2018) [20] for quaternion-valued least-mean kurtosis (QLMK) method. This [20] method is significant with applicability for wide range of noise signals cancellation and improvement in convergence, steady-state error. On analyzing the existing researches it is observed that very rare works are incorporated with adaptive approach for noise cancellation. Among these, most of the works adaptive approaches are implemented only the LMS algorithms. Most of the researches were used backpropagation or feedforward propagation approach for noise cancellation and minimize noise error rate. This paper introduces an artificial neural network based adaptive approach which incorporates hybrid backpropagation for noise cancellation by using different algorithms like LMS, Gradient adaptive algorithm and hybrid adaptive algorithm. The proposed Adaptive Filter Based Noise Cancellation System (AFNCS) considers the convergence error and computational complexity for performance analysis.

III. Proposed Solution

For noise cancellation various Gradient Adaptive Lattice (GAL) and LMS algorithms are used. Recently, the hybrid adaptive algorithms with neural networks have gained popularity in cancelling the noise available in communication system. The working principle of the proposed intelligent adaptive filter based noise cancellation system (AFNCS) is the continuation of prior work (Kumar et al. [21]) which is further empirically designed and simulated to enhance the performance of the input synthetic signal with respect to noise cancellation. In this, a hybrid backpropagation algorithm is introduced by which learning of multi-layer network is achieved. The noise analysis of the system is performed by using Artificial Neural Network (ANN). This intelligent hybrid backpropagation algorithm involves both GAL and LMS algorithms. The prime aim of the proposed intelligent AFNCS is to acquire a signal from reference signal and output noisy signal. The signal noise is eliminated by subtracting the reference signal and noisy signal with original signal. The use of AFNCS can significantly restore the original signal by eliminating the noise by using adaptive control and adjustment of weights through ANN. The Figure 1, indicates the block representation of the AFNCS which intakes the input signal i(t) and generates output signal O(t) by using adaptive system and reference signal R(t). Finally, the error signal e(t) is computed by finding the difference among reference signal and output signal as given in Eq. (1).

$$e(t) = R(t) - O(t) \quad \text{....Eq. (1)}$$

Where t represents number of iterations

The adaption of hybrid algorithm considers this error signal e(t) to generate a function for execution. This function performs the computation of desired filter coefficients. The minimized error rate indicates that output signal is similar as that of original signal. Here backpropagation algorithms are used to evaluate the error rate of each neurons. The Figure 2, highlights the structural model of backpropagation network. The network is composed of three layers like input, output and hidden layer. The hidden layer is exist in between input and output layer which incorporates both the layers. The overall backpropagation network is affected by one neuron error. The network allows audio or speech signal to propagate via ANN and provides output signal. As given in Eq. (1) the error results of the output layer is computed and this error is forwarded back to input layer through hidden layer until the desired output is obtained.

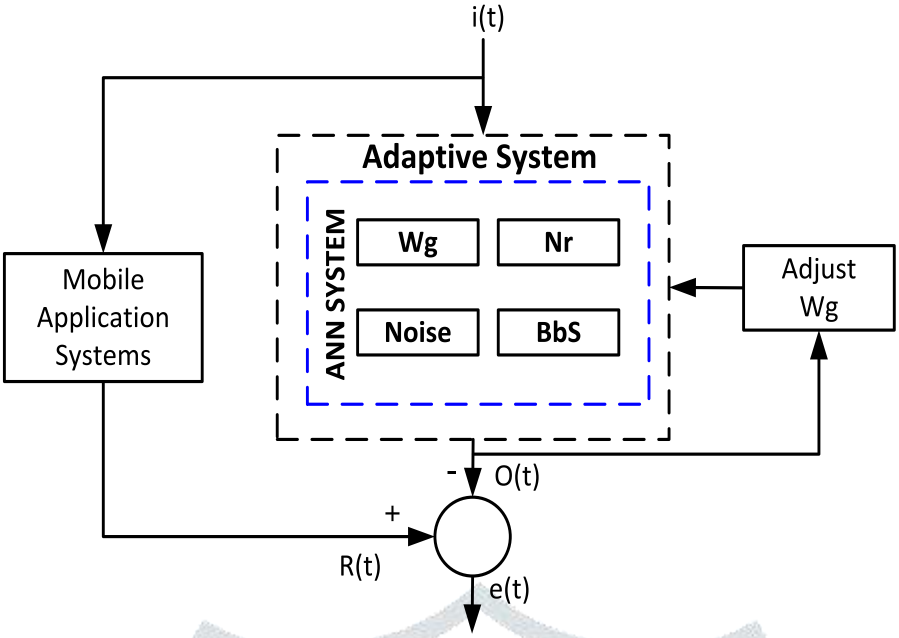

Figure 1. Proposed Adaptive Filter Based Noise Cancellation System (AFNCS)

Further, to minimize its error signal, adjustment of weight is performed for each neurons. The proposed hybrid algorithm combines both the backpropagation algorithms of LMS and GAL which helps to tackle slow convergence. The proposed AFNCS adopts adaptive filtering for implementation of ANN and also adopts a control system for adjustment of adaptive filtering parameters. The elements connection is trained with ANN by weight adjustment. The output of ANN is obtained by using Eq. (2). The table 1, indicates the parameters used in design.

$$ANN_{out} = \sum i(t) \times W_g \quad \text{....Eq. (2)}$$

Table 1. Parameters used in intelligent AFNCS system

Parameter	Description
i(t)	*Input Signal*
R(t)	*Reference Signal*
O(t)	*Output Signal*
e(t)	*Error Signal*
Wg	*Weight*
Nr	*Neurons*
BbS	*Baseband Signals*
Th	*Threshold (0 to 1)*
$ANN_{\_out}$	*Output of ANN*

All inputs are accompanied by a weights.If, $\sum W_g \geq Th$ then the output of ANN is 1 as given in Eq. (3)

i.e., $ANN_{out} = 1$ ...Eq. (3)

However, if $\sum W_g < Th$, then the output of ANN is 0 as given in Eq. (4)

i.e., $ANN_{out} = 0$ ...Eq. (4)

To get the desired output signal ANN to adjust the weights with respect to input samples. The formation of the ANN system is done by considering three layers such as the output layer, hidden layer, and the input layer. The audio signal is fed to input layer having neurons. The hidden layers minimization reduces the error rate to achieve the desired output. The output layer does the competition of neuron nodes based on the requirement of output.

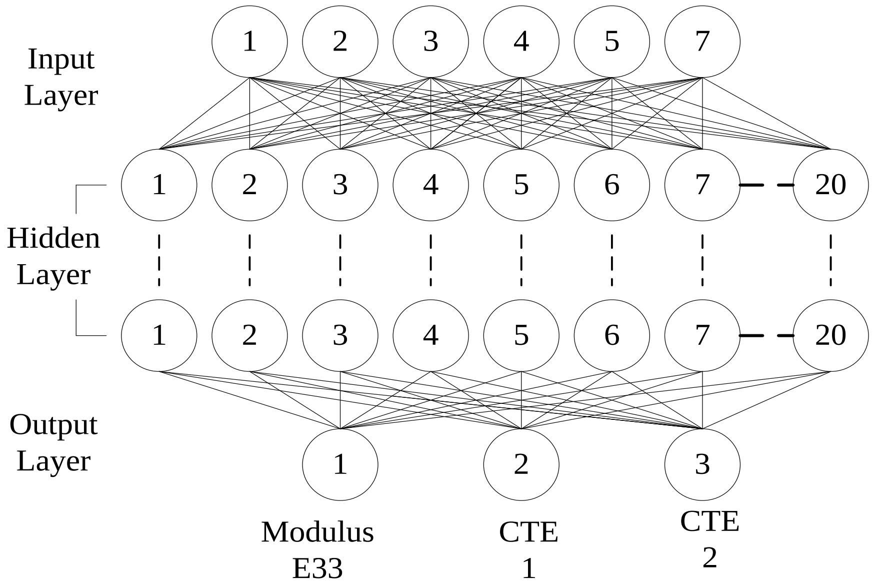

Figure 2. Layer diagram of ANN

In the proposed ANFCS implements the LMS mechanism to compute the instantaneous value of the gradient vector. Then, the minimization of the MSE is performed by varying the filter weights. For each iteration of the adaptive filter, the optimized Weiner solution is obtained which is referenced with Eq. (5) where n, St, Ws(n), I(n) represents the time, step size, adaptive filter coefficient and input vector respectively.

$$Ws(n+1) = Ws(n) + (S_t \times e(n) \times I(n) \quad \text{...Eq. (5)}$$

In order to avoid the instability, output divergence and convergence time, the optimal value of St is selected. Further, to reduce the error, negative gradient functions are considered.

The Eq. (6) and (7) gives the computation of e (n) and I (n) respectively.

$$e(n) = R(n) - (W_g(n) \times I(n) \quad \text{...Eq. (6)}$$

$$I(n) = [i(n) \times i(n-1)....x(n-) - (W_g(n) \times i(n+1)^T \quad \text{...Eq.(7)}$$

Then the LMS Algorithm is described as,

$$O(n) = \sum_{i=0}^{M-1} Wi(n) \times i(n-i) \quad \text{...Eq. (8)}$$

$$e(n) = R(n) - O(n) \quad \text{...Eq. (9)}$$

$$Wi(n+1) = w_i(n) + (St(n)^* \times i(n-i) \quad \text{...Eq. (10)}$$

$$O(n) = w_0(n)^* \times i(n) \quad \text{...Eq. (11)}$$

Where, $W(n) = [w(n) \times w_1(n)....(W_{IL-1}(n)^T$ is a coefficient vector.

In the proposed method, the convergence rate of error signal increases with the value of St. LMS mechanism is adopted in the proposed method because of its easier implementation, simple computational, dynamic usage of memory ability and is performed by adjusting filter coefficients for error minimization.

IV. ALGORITHM

In a numerical computing environment proposed AFNCS is modelled by means of soft computation based algorithm design and implementation. The system specifications which are required to know the AFNCS algorithm performance includes a 64-bit operating system, an x64-based processor supported with 4.00 GB installed memory (RAM), where the processor type is Intel® Core™ i-8250U CPU@1.60GHz 1.80GHz. The following algorithm exhibits the steps associated with AFNCS design goals to achieve cost-effective adaptive noise cancellation from a sinusoidal signal.

Proposed AFNCS Algorithm

Input: i(t), R(t)
Output: e (t), O (t)
1. **START**
2. ***Initialize*** W_g,N_r, B_bS,i (t), R (t) where $t \in Z^+$ which is iteration number
3. Initialize hybrid adaptive Filter Configuration with LMS and GAL
4*.* *for (i = 1:t)*
5. ***Enable*** ANN
6. ***Compute*** O (t) = f_{AFNCS} (i(t))
7. *Compare* O(t) with R(t)
8. e(t) = R(t)- O(t)
9. $\text{Training}_{\text{Input}}[a_{ij}]_{12x4,}\text{Training}_{\text{Output}}\ [b_{ij}]_{12x\ 1}$
10. Initialize hidden neurons, epochs
11. Validate the number of rows equality among input data and output data vector, if mismatch error.
12. Compute $\mu_{mean} = \frac{1}{n}\sum_{i=1,j=1}^{n}[aij]12x\ 4$ where training vector is made up of n scalar observations
13. Compute σ_{ip} = standard deviation of SNR from AFNCS at different dB's.
14. start Learning
15 for (iteration =1: epochs)
16. Compute learning rate
17. Adjust weight *hidden* →*Output*
18. Adjust weight *input* →*hidden*
19. *Normalize inputs and output vectors from (0 to 1) using ANN*
20. Compute error e(t) = R(t)- O(t)
21. end
22. end
23. **END**

The above algorithm represents the computational steps associated with the proposed AFNCS which combines the strength and significant features of adaptive algorithms such GAL and LMS for the purpose of accomplishing multi-layer perceptron network. The proposed system incorporates a hybrid back propagation learning for the adaptive noise cancellation in mobile applications. The proposed hybrid algorithm consists of all the significant features of LMS and GAL algorithms.

The above algorithm clearly shows that how incorporating ANN based adaptive learning AFNCS significantly reduces the error rate of an input speech signal from a soft-computing viewpoint. It also applies weight-adjustment in ANN which influences lower convergence performance with less iterative steps driven by non-recursive functions. The algorithm is designed simulated in a numeral computing platform, and the performance of the proposed AFNCS has been justified with respect to two different parameters such as processing time(s) and Bit Error Rate (BER) from both complexity and signal quality viewpoint.

V. Results Analysis

This section talks about the experimental outcome obtained after simulating the proposed AFNCS in a numerical computing environment. The proposed AFNCS considers training input of a vector [aij]12x 3 with SNR values from different adaptive filters and further it assessed the mentioned AFNCS algorithm which further results in Training output = [aij]12x 1, which is a predicted/estimated SNR values for different signals of the proposed method at different dBs. The figure 3 exhibits the percentage of total error obtained from the output error signal, where the observation carried out shows that error has been reduced to a significant extent while the number of epochs increased to 5000 iterations.

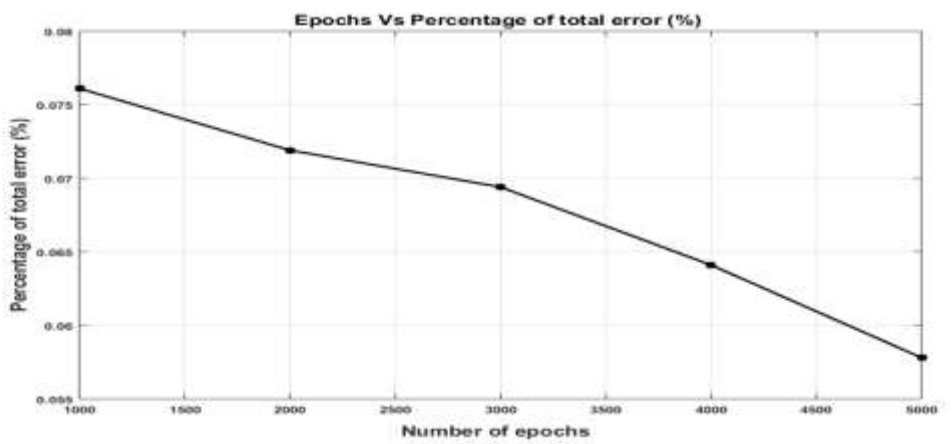

Figure 3. Evaluation of BER computation after assessing AFNCS

It is also closely observed during the numerical computation that, number of hidden layers which having a relationship with computational delay. As the number of hidden layers increases delay will increase. i.e., less the number of hidden layers less will be computational overhead, which results in faster response. It also lead to the effective utilization of system memory. Error decreases as the number of iteration increases up to 5000.Afterthen error percentage increases as we keep increasing iterations. The Comparison Table-2 shows a comparative performance analysis where the outcome of SNR from simulating the proposed AFNCS has been compared with RLS, FTF and GAL algorithms with respect to error [22]. The comparative performance analysis considered three different types of the signals viz., Chirp, Sinusoidal, Saw tooth and Audio with different SNR values of 30dB, 10dB.

Table 2. Comparative Performance Analysis

At 30dB	**RLS[22]**	**FTF[22]**	**GAL[22]**	**AFNCS**
Chirp	13.9296	24.7287	68.7400	9.7591
Sinusoidal	14.5297	14.5130	72.7454	7.7829
Sawtooth	12.7979	12.7880	71.4740	7.5157
Audio	13.0794	25.5130	46.6549	8.2704
At 10dB	**RLS**	**FTF**	**GAL**	**AFNCS**
Chirp	9.7591	8.3550	19.3497	8.3550
Sinusoidal	7.7829	7.7703	18.1702	7.7703
Sawtooth	7.5157	7.5087	20.4481	7.5087
Audio	8.2704	9.3365	10.0652	9.3365

A closer look into the table 2 depicts that the proposed AFNCS accomplishes better outcome as it yields very less output signal error as compared to the conventional baselines. It is also observed that incorporation of less hidden layers resulted in negligible computational delay along with effective utilization of memory.

VI. Conclusion

This paper introduces an intelligent adaptive filter based noise cancellation system (AFNCS) by using LMS and GAL algorithms. The AFNCS is modelled in the numerical computing environment by means of soft computation based algorithm design and implementation. The performance of the proposed AFNCS is performed by considering convergence complexity and BER as a performance parameter. From the analysis of the outcomes, it is found that the errors are reduced significantly when the number of epochs are increased. From computational numerical data, it is found that as the number of the hidden layers increases the delay will also increase and have a faster response the number of hidden layers should be less. The performance analysis of AFNCS considered for signals such as Chirp, Sinusoidal, Saw tooth, and Audio at an SNR values of 30dB and 10 dB are compared with Ferdouse et al., [22]. From, the comparative analysis it is found that the AFNCS has achieved better results than existing method accomplishes very less output signal error as compared to the conventional baselines. Also, incorporation of less hidden layers resulted in negligible computational delay along with effective utilization of memory.

VII. Acknowledgment

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