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Designing of an Adaptive Filter in Digital Hearing-Aids for Noise Cancellation

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Abstract: ***Noise is very much influencing the intelligibility of the speech with deafness. Because from countless years the limitation has been tended to from a couple of facets of view. The problem has been analyzed through filtering to advanced adaptive filtering methods. To plan a versatile channel for Hearing-Aid application by proposing any modernization of the conventional algorithms or the hybrid of two or more algorithms. This will have to be feasible by analyzing various adaptive filtering algorithms by simulation and converging of the advantages of any of the two adaptive filtering algorithms and in this way to cancel the internal noise and all acoustic echo issues and accomplishing Minimum Mean Square Error (MMSE) in an audio signal.***

Keywords: Adaptive Filter, LMS, NLMS, RLS, MSE

1. INTRODUCTION

Hearing-Aids make sounds louder and clearer. The problem actually in the individual with deafness is not with the audibility but with the noise. Digital Hearing-Aids are available in the marked ranges from 10,000 to lakhs. Even though the optimal solution has not been found out. The research has been started since 1970s and still the researches are going on. The major issue with the Hearing-Aid users is the amplification of noise along with the original and that makes them discomfort and it causes Headache too. And they not likely to use their hearing-aid in their daily life because of this issue. That motivates to work with adaptive noise cancellation method and is employed to achieve environment noise free.

2. ADAPTIVE NOISE CANCELLATION (ANC)

In the adaptive noise cancellation process, the ultimate goal is to reduce the noise in the input signal. In the figure desired signal (D) is generated from the signal source (S) which is added with the additional noise (N1). And the adaptive filter is placed that the only input signal to the filter is noise (N2).The signals (N1) and (N2) are correlated. The error signal (E) is generated which is the difference of the desired signal (D) and the filter output (Y). The error signal (E) forms the negative feedback loop and sent into the filter for automatic adjustment of variable weights to cancel the interference of noise. The optimal solution is system output (E) is composed by the signal (S) alone leaving the noise signal (N1) [1]-[8].

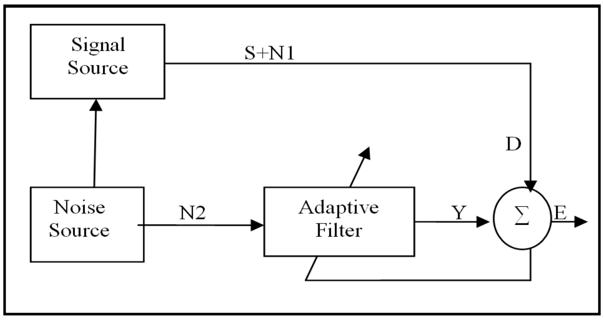

Figure 1 Block Diagram of Adaptive Noise Cancellation

3. ADAPTIVE FILTERING ALGORITHMS

Adaptive Filter is that it utilizes the channel parameters of a minute back to naturally alter the channel parameters of the present minute, to adjust to the factual properties that flag and clamor obscure or irregular change keeping in mind the end goal to accomplish ideal channel. In view of inside and out investigation of versatile channel, in light of the slightest mean squares calculation and recursive minimum squares are connected to the versatile channel innovation to the commotion, and through the reproduction comes about demonstrate that its execution is typically much superior to anything utilizing ordinary strategies intended to channel settled [2]

The purported versatile channel, is the utilization of the aftereffect of the channel parameters a minute back, naturally alter the channel parameters of the present minute, to adjust to the obscure flag and clamor so as to accomplish ideal sifting.

2.1 Least Mean Square Algorithm (LMS)

In light of the technique for steepest plunge, move towards the base on the mistake surface to get to least and it requires the inclination of the blunder surface to be known. Most prevalent adjustment calculation is LMS which is gotten from steepest plunge and it doesn't oblige angle to be known, it is assessed at each emphasis [3]-[7]. The LMS Algorithm comprises of two essential procedures [2],

- Filtering process- calculate the yield of FIR channel by convolving information and taps, Calculate estimation mistake by contrasting the yield with sought flag.
- Adaptation Process- prepare Adjust tap weights in light of the estimation mistake.

Filter Output,

$$y[n] = \sum_{k=0}^{M-1} u[n-k]\hat{w}_k[n] \quad (1)$$

Error Estimation,

$$e[n] = d[n] - y[n] \quad (2)$$

Weight-Updation,

$$\hat{W}(n+1) = \hat{W}(n) + 2\mu e*(n)u(n) \quad (3)$$

3.1 Normalized Least Mean Square Algorithm (NLMS)

The normalized LMS (NLMS) algorithm is a modified form of the standard LMS algorithm. The difference is the adaptive step-size. The adaptive weight control mechanism is adaption of step-size in which it depends on Squared Euclidean normalization of input signal and due to this the product of error signal and input signal get normalized [4]-[9].

Filter Output,

$$y[n] = \sum_{k=0}^{M-1} u[n-k]\hat{w}_k[n] \quad (4)$$

Error Estimation,

$$e[n] = d[n] - y[n] \quad (5)$$

Weight-Updation,

$$\hat{W}(n+1) = \hat{W}(n) + 2\mu(n)e*(n)u(n) \quad (6)$$

Time-Varying Step-Size Parameter,

$$\mu(n) = \frac{\tilde{\mu}}{\|u(n)\|^2} \quad (7)$$

4.1 Recursive Least Square Algorithm (RLS)

The Recursive Least squares (RLS) versatile channel is a calculation which recursively finds the channel coefficients that limit a weighted straight minimum squares cost work identifying with the info signals. The RLS calculations are known for their magnificent execution when working in time changing conditions yet at the cost of an expanded computational multifaceted nature and some strength issues [5]

It is differed with the LMS algorithm from the weight equation, it recursively least square the error by the weight update equation considering intermediate gain factor which includes forgetting factor (lambda). It considers the error variations from the distant past and it recursively reduce the least square error [6].

Filter Output,

$$\overline{y}_{n-1}(n) = \overline{w}^{T}(n-1)x(n) \quad (8)$$

Error Estimation,

$$\overline{e}_{n-1}(n) = d(n) - \overline{y}_{n-1}(n) \quad (9)$$

Weight-Updation,

$$w(n) = \overline{w}^{T}(n-1) + k(n)\overline{e}_{n-1}(n) \quad (10)$$

Intermediate-Gain Factor,

$$k(n) = \frac{u(n)}{(\lambda + x^{T}(n)u(n))} \quad (11)$$

$$u(n) = \overline{w}_{\lambda}^{-1}(n-1)x(n) \quad (12)$$

4. SIMULATION RESULTS

The first part of simulation shows the input signal, addition of input signal with the noise, error estimate and finally filter output. The related input and output are shown in figures from Figure.2 through Figure.7. Figure.2 shows the input and output for LMS algorithm, Figure.3 for NLMS algorithm, Figure.4 for RLS [3]-[9].

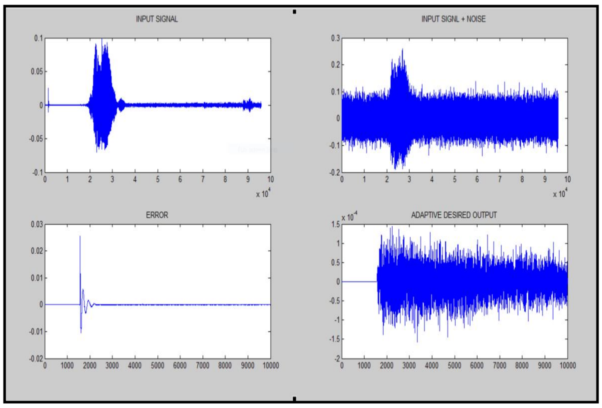

Figure 2 Noise cancellation using LMS

subplot(1) shows the original audio signal which is given as an input signal and sampled at a frequency of Fs=44100 Hz. Subplot(2) is adding of noise with the input signal.

Here the noise is Additive White Gaussian Noise(AWGN) to cancel out the ambient or background noise. Subplot(3) shows the Error Estimate, which is the difference of the original and desired signal. And Subplot(4) shows the Adaptive Filter Output in response to an input signal.

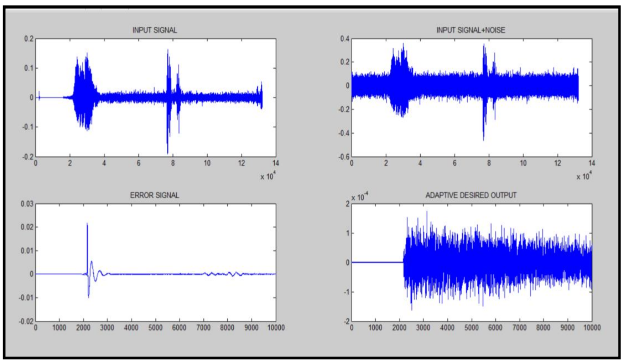

Figure 3 Noise cancellation using NLMS

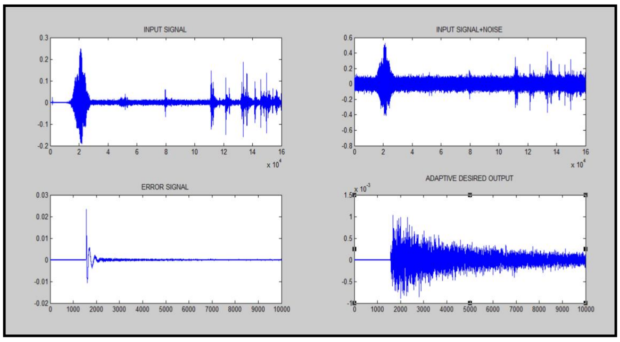

Figure 4 Noise cancellation using RLS

Second some portion of reproduction demonstrates the execution examination of traditional calculations lms, nlms and rls. In this work, step-sizes are considered for both LMS and NLMS and an forgetting factor for RLS. For these conditions, the mean square errors are compared for different algorithms as shown in figure.5,6,7.

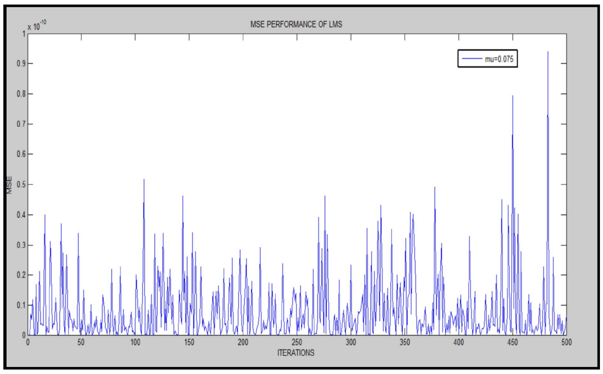

Figure 5 MSE of LMS Algorithm

MSE of LMS algorithm is depicted in Fig.5. The result shows that the MSE performance of LMS Algorithm between Mean Square Error (MSE) and Number of Iterations for the various step-size mu=0.07 is considered as an experimental setup. Considering 500 iterations the LMS shows slow convergence and it may take further more iterations to converge and it is less stable.

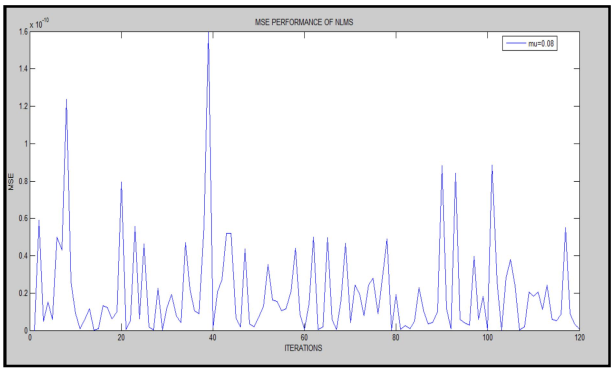

Figure 6 MSE of NLMS Algorithm

Fig.6 depicts MSE of NLMS. Considering 120 iterations it starts to converge as for (mu=0.08) it doesn't show any high peak after 40^{th} iteration and NLMS is stable when compared with LMS.

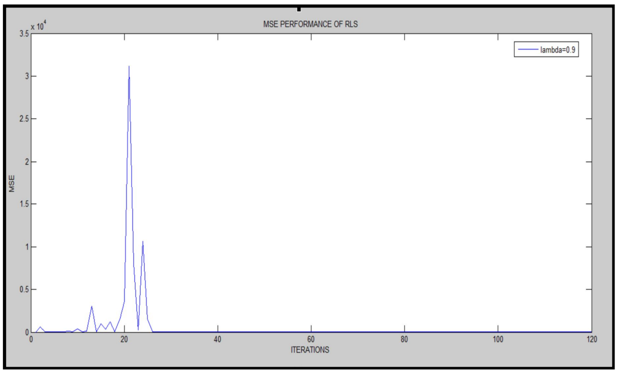

Figure 7 MSE of RLS Algorithm

Fig.7 shows the MSE performance of RLS Algorithm between Mean Square Error (MSE) and Number of Iterations for the various forgetting factor (lambda), {lambda=0.9} is considered as an experimental setup.

The experiment is repeated for different iterations starting from 10,50,100,etc., Considering 120 iterations it starts to converge faster. As for all forgetting factors it gets stable around 30^{th} iteration. And RLS is highly stable when compared with LMS and NLMS

The Mean Square Error in terms of simulation results also confirms that the RLS gives lowest MSE and also fast convergence while cancelling noise.The conclusion drawn is RLS outperforms both LMS and NLMS in noise cancellation of an audio signal [10].

Table 1: MSE Comparison of LMS, NLMS and RLS

S.no	Algorithms	Min (MSE)	Max (MSE)	Complexity
1	LMS	0	5.7448e-04	LESS
2	NLMS	0	5.3581e-04	MODERATE
3	RLS	0	3.3528e-08	HIGH

It is inferred from the Table 1, the performance of RLS adaptive algorithm is high as compared to other algorithms due to the less Mean-Square Error (MSE). And LMS shows less complexity compared to the other two algorithms. The Mean Square Error value of NLMS shows slight variations when compared to the LMS.The values which are tabulated are got from the simulation results of algorithm in MATLAB. Stability is high with the Minimum Mean Square Error (MMSE). It is also clear that the RLS has the Minimum Mean Square Error compared to the other two algorithms

5. CONCLUSION

To approve the proposed display broad recreations were done for routine calculations in Matlab. Error signal(e) is to be calculated which is the difference of the desired signal and the filter output. Thus the performance analysis is done for all conventional algorithms considering (MSE) Versus Number of Iterations. LMS shows slow convergence taking more than 500 iterations to become stable. NLMS takes around 100 iterations which is faster than NLMS. And RLS is pretty faster than the other two takes 30 iterations. From the simulations it is inferred that LMS is simple but with slow convergence, RLS shows faster convergence, higher stability and Minimum MSE but with the higher computational complexity. Thus the hybrid model is needed to overcome the limitations of conventional algorithms. In the similar way analysis should be done for proposed hybrid model and the results shows that the proposed algorithm may be best suited in terms of MMSE, number of iterations over boisterous environment.

6. FUTURE WORK

Moreover adaptive sifting calculations are dissected and to check the execution examination of Mean Square Error. Along these lines to propose versatile channel for Hearing-Aid applications by modernization in the customary calculations or mixture of at least two algorithms. It is relied upon to accomplish Minimum Mean Square Error (MMSE) in the proposed calculation by cancellation of noise in an audio signal.

References

[1] Sarthak Panda, Mihir Narayan Mohanty, “Performance Analysis of LMS based Algorithms used for Impulsive Noise Cancellation,” In International Conference on Circuit, Power and Computing Technologies [ICCPCT], 2016.

[2] Simon Haykin, “Adaptive Filter Theory,” Prentice Hall, ISBN 81-317-0869-1, 2002.

[3] Yin Wu, Wei Zhao, Qin Zhang, “The Improvement of Audio Noise Reduction System Based on LMS Algorithm,” In International Conference on Computer Sciences and Applications, 2013.

[4] Jafar Ramadhan Mohammed, “A New Simple Adaptive Noise Cancellation Scheme Based On ALE and NLMS Filter,” In Fifth Annual Conference on Communication Networks and Services Research (CNSR'07).

[5] Alina Mirza, S.Mehak Kabir, Muhamad Arslan, Shahzad Sheihk, “Enhanced Impulsive Noise Cancellation based on SSRLS,” In IEEE International Conference on Computer, Communication and Control Technology (14CT 2015) April 21-23 in Imperial Kuching Hotel, Kuching, Sarawak, Malaysia, 2015.

[6] Md. Zulfiquar , Ali Bhotto, Andreas Antoniou, “New Improved Recursive Least-Squares Adaptive-Filtering Algorithms,” In IEEE Transactions On Circuit and systems—SI: Regular Papers ,Vol.60, No.6, June2013.

[7] Xu Yanhong, Zhang Ze, “Design the adaptive noise canceller based on an improved LMS algorithm and realize it by DSP ,” In Fifth International Conference on Intelligent Computation Technology and Automation, 2012.

[8] Sheng Zhang, Jiashu Zhang, and Hongyu Han, “Robust Shrinkage Normalized Sign Algorithm in Impulsive Noise Environment,” DOI 10.1109/TCSII.2016.2546905, In IEEE Transactions on Circuits and Systems II: Express Briefs. (unpublished)

[9] Ms. Mugdha. M. Dewasthale, “Improved NLMS Algorithm with Fixed Step Size and Filter Length using Adaptive Weight Updation for Acoustic Noise Cancellation,” In Annual IEEE India Conference (INDICON), 2014.

[10] Jyothi Dhimani, Shadab Ahmad, Kuldeep Gulia, “Comparison between Adaptive filter Algorithms (LMS, NLMS and RLS),” In International Journal of Science, Engineering and Technology Research (IJSETR) Volume 2, Issue 5, May 2013.

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