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Adaptive Speech Noise Cancellation Using Least Mean Square Algorithm

Dr.Y. Amar Babu[1,] M. Chandu[2,] M. Raja[3,] P. Sowjanya[4,] S. Bhargav[5]

[1]*Professor, Department of Electronics and Communication Engineering, Lakireddy Balireddy College of Engineering, L.B.Reddy Nagar, Mylavaram-521230, Andhra Pradesh, India.*
[2-5]*UG Scholar, Department of Electronics and Communication Engineering, Lakireddy Balireddy College of Engineering, L.B.Reddy Nagar, Mylavaram-521230, Andhra Pradesh, India.*

Abstract
In any communication noise is always has been playing an undesirable performance. Irrespective of the type of the signals (i.e., Speech, Audio/Video, Image) that we acquire at the receiver side is affected by noise. Noise is an unwanted portion of the signals. During transmission signals gets effected by noise. So, we have to cancel out the noise content which modifies the transmitted version during processing, storage and conversions. In voice communication recorded/live speech data signal transmitted will be very useless, unless a perfect noise cancellation system is adopted. There are many kinds of such mechanisms. In this paper we approached and adopted an adaptive noise cancellation, and by implementing the three algorithms (LMS, JAYA, PSO) noise eliminated to some extent and the gain percentage from the base paper is 35%.ss Adaptive filtering applications serves their best in Interference and noise cancellations.

Keywords: Speech Noise cancellation, LMS, JAYA, PSO, Adaptive Filter Application, Noise cancellation Mechanism.

1. INTRODUCTION

In this world content is the king and information is the wealth. Whatever the kind of information is supposed to communicate well between the source point to destination point. In the midway the content suffers due to many reasons. In Signal processing we knew the betterment of digital signal processing over analog signal processing. Because of the convenient in storage, transmission and conversions of the digital signals, instead of analog one digital is preferred and processed. But it is also suffering during these conversions in source encoding and channel encoding. Signal gets effected because of bit errors. And also, in processing of the transmitted signal it is gets corrupted because of the added noise. In Signal Processing field the most challenging question is how to cancel out the noise which corrupts the speech from surroundings.
In this paper, section-2 illustrate the related work done and base of the paper. And section-3 will give the architecture of the proposed work. Further section-4 shows the implementations of the algorithms in the working model and section-5 will give the conclusion and possible future scope of the current working model.

2. RELATED WORK

In signal processing, noise is the common term for undesirable (and, obscure) alterations that a signal may suffers during capturing, storage, transmission, preparing, or conversion. In some cases, the word is additionally used to mean signals that are irregular and carrying no helpful data; regardless of whether they are not meddling with different signals or may have been presented deliberately, as in comfort noise. Noise cancellation, the recuperation of the first signal from the noise ruined one, is an extremely shared objective in the plan of signal processing frameworks, particularly in filters. Noise may emerge in signs important to different scientific and specialized fields, regularly with explicit highlights: Noise (sound), for example, "hiss's" or "humm's", in sound signs. Background Noise, because of false sounds during the capturing of the signal. Comfort noise, added to voice correspondences to fill quiet gaps in communication. Electromagnetically-excited noise, audible noise because of electromagnetic vibrations in frameworks including electromagnetic fields. Noise (radio), for example, "static", in radio transmissions. Noise (gadgets) in electrical signs. Ground noise, showing up at the ground terminal of sound equipment.
For example, let us assume that two people are communicating by means of a mobile. because of the air flow and the sounds which are produced by other people to do their own work like opening or knocking a door, vehicle horns and engine sounds. These all sounds also captured by the micro phone of the person who is talking from source side. These are not necessary to the other person who is listening from the other end. Due to the intensive noise power speech signal is gets effected. If the SNR is unity or less than that of unity it is not at all possible to hear the speech. If it is the greater than unity we can easily reconstruct the speech signal. In general, human conversation is in the order of 40-60 db. If the surrounding noises levels are more than or equal to 82.5 dB the speech signal suffers severe noise problem. Studies in Germany [7] and other industrialized countries made surveys on noise just because of the levels of noise is far greater than 85 dB in those areas. There by the workers who ever working in that intensive noise conditions suffering from the hearing problems. Some of the workers who

are working for hours and years getting permanent deafness. It is becoming a major problem in workers lives. So, it makes the engineers to work on noise cancellation [6] devices and mechanisms. There are many kinds of mechanisms and devices too. People made ear muffles and noise less machines or by putting the machines in a sound blocking chambers. but every kind of invention worked well for only live noise cancellations. If the situation comes for signal processing microphones are capturing every kind of disturbances in the surroundings. It's been a problem in signal processing and the research is ongoing rigorously. Even a processed signal also gets corrupted by noise [8]. So, the noise content is more in the desired signal rather than the original signal. For that it is needed to cancel out the noise by using some mechanisms. There are many kinds of mechanisms like Such as Passive noise cancellation, Active noise cancellation, [2] Digital noise cancellation, Audio signal noise cancellation, Additive noise cancellation and adaptive noise cancellation mechanisms. For every mechanism they have their own way of cancelling the noise. We adopted the adaptive noise cancellation mechanism and by using three algorithms (LMS, JAYA & PSO). The very first one very well-known algorithm but the thing is implementation of the algorithm in a better way to cancel out the noise. And the remaining two are of bio-inspired algorithms. Already traditional and conventional optimization techniques such as Newton Raphson Method, Simpson and Trapezoidal methods are in existence. But the creatures that are living on this earth have their own and efficient intelligence to do the things on their own in an efficient way. So, people by observing them and yielding the intelligence of creatures to algorithms and serving the best in optimization of any real time functions. So, in this paper JAYA and Particle Swarm Optimization techniques are used to optimize the noise content at the receiver side.

3. ARCHITECTURE OF THE WORK

Figure 1 shows the sketch of proposed architecture and the input parameters to the Noise cancellation system are the live speech, noise which is generated randomly and the reference noise which is to be generated as per the length of noise signal which will be forwarded to Adaptive filter. Speech [9] signal will be corrupted by adding the noise as of the length the of I/P speech signal. Now we can create the noise corrupted speech signal. Initializing the adaptive filter weights and row vector based on the order of the filter. We can get the adaptive filter [1] O/P as the multiplication of row vector and the inverse of the weighted vector. Initially the row vector will be loaded with reference noise samples. Referring the noise corrupted signal as the primary I/P signal and compute the difference of primary and O/P of adaptive filter and notate this O/P as Difference O/P. It will be used to get the new weights. Three algorithms having their own way of approaches and weight updating mechanisms. Difference O/P will be placed to get the new weights in updating model equations. And by calculating the Mean Square Error (MSE) for every algorithm as the computation takes place on speech signal S(i) and the desired signal or Difference signal D(i). i.e., *Error = S(i)-D(i)*. By squaring of the error, we can get square error and to compute the MSE the following equation will be used and are placed in MATLAB.
MSE = Sum (square_error)/ {No. Of samples of the i/p – order of the filter – 1}
From this we can easily calculate the MSE in dB *MSE_dB = 20*(Log10(MSE)).*

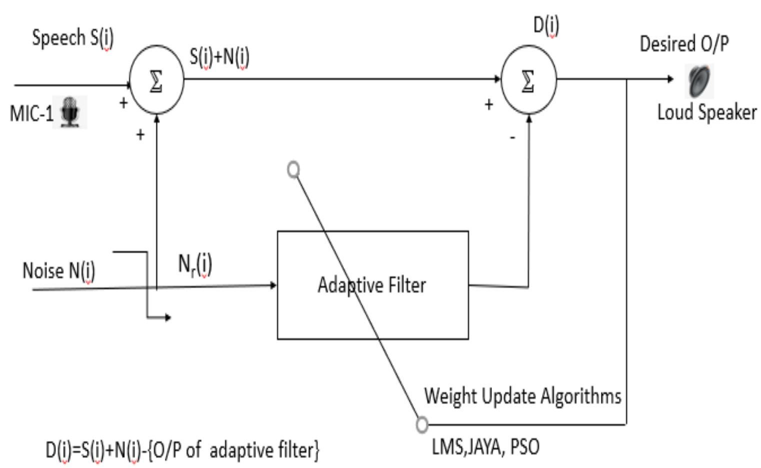

Figure 1: Architecture of Adaptive Noise Cancellation System

4. IMPLEMENTATION OF ALGORITHMS

ASNC using LMS Algorithm:

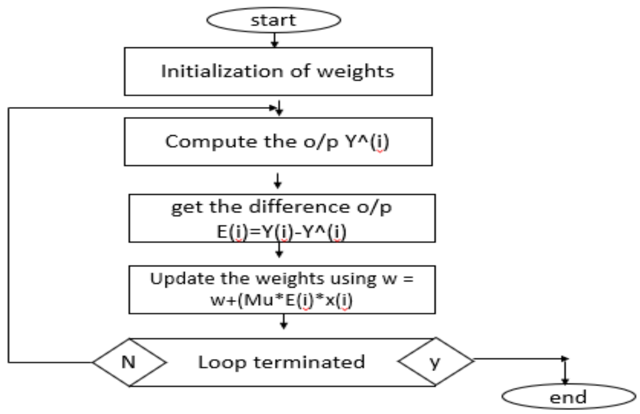

Figure 2: Flow chart of LMS Algorithm

A best method for noise cancellation which was developed by widrow and Hoff. This algorithm makes use of gradient steepest descent method which is used to find a minimum. By taking proper steps in the direction of negative of gradient and it can perform so by adjusting the filter weights in order to minimize the error. Here it will converge the filter weights with the physical system. And it has some correlation with system identification which needs to converge with the original weights. But in case of noise cancellation or interference cancellation no need of physical system. Weights of the adaptive filter [3,10] are adjusted such that the o/p of the filter to cancel the noise in the primary speech signal
$S(i)$ = Speech signal,$N(i)$ = Noise signal and *primary i/p* = $S(i)+N(i)$. Generate a reference noise based on the length of the noise signal $N_r(i)=rand*N(i)$. Now the difference signal will be D(i) = Primary – {O/p of adaptive filter}. This D(i) will be used to find out the new weights so that at the end of computation adjusted weights will cancels out the noise from primary speech so that we can here original or captured signal. The following Equations will be used in LMS algorithm,
New weights $w_{new} = w_{old}+(\mu*D(i)*U(i))$
Where U(i) is the row vector used to load the reference noise samples.

ASNC JAYA Algorithm:

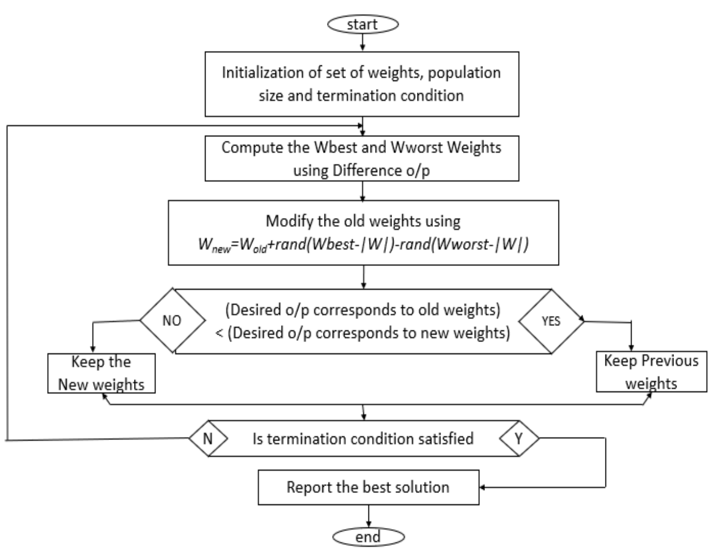

Figure 3: Flow chart of JAYA Algorithm.

In JAYA algorithm initialization of set of weights will be generated randomly. Whatever the process followed for LMS will be same for the JAYA[4,10] but computation will be done for all the weights, there by a chance of comparison among the final solutions which will serves the best. Based on that indicating the best and worst weights in accordance with the desired solutions. So, finally the best weights will serve and placed to cancel out the noise. And the weight update equation for JAYA is asfollows$W_{new}=W_{old}+r1(Wbest-|W|)-r2(Wworst-|W|)$ Where r1, r2 are the random values.

ASNC PSO Algorithm:

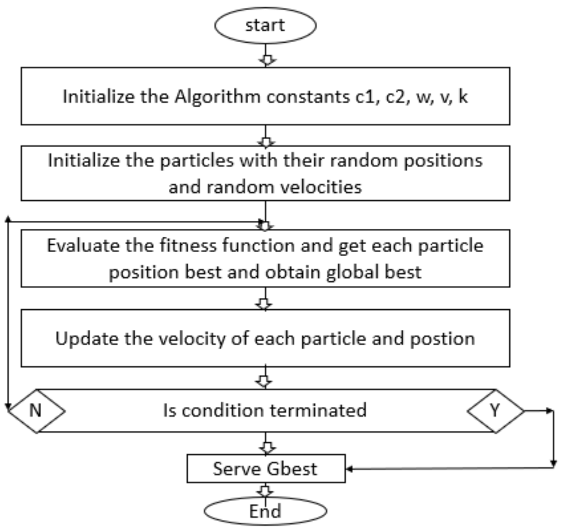

Figure 4: Flow chart of PSO Algorithm.

PSO algorithm is derived from the swarms like birds, ants, fishes. To search out their food they will communicate among themselves. In the same way the communication provided among the weights such that best weight modifies the old weights so in less time the whole process will be done. And in very less time PSO [5] serves the best weights and the same process will be done here as the process followed in the earlier two algorithms. Here the new weights will be developed by using the following equation i.e.,

$W_{new}=w*k+(c1*r1)*(Wbest-|W|)-(c2*r2)*(GWbest-|W|).$

RESULTS

In LMS Algorithm it is clear that gradually the noise content is removing from sample to sample. In the fig.5 two lines drawn on primary and desired showing the removal of noise content from primary speech signal and original signal strength is enhanced.

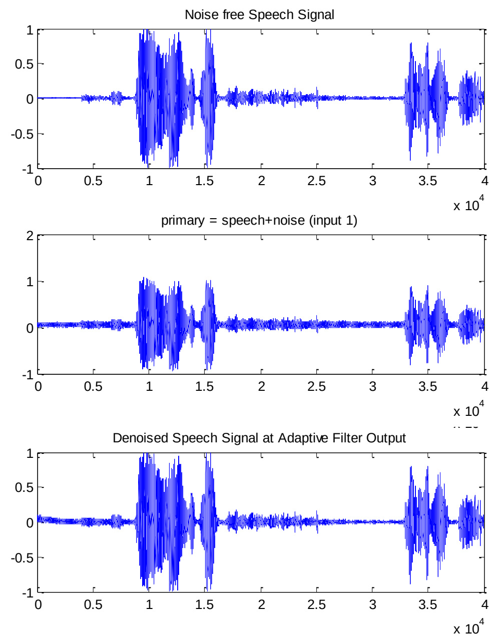

Figure 5: [LMS] Signal graphs for 3.5seconds live speech.

By observing the JAYA based noise cancellation from the fig.6 primary is full of noise and the noise cancelled in the desired signal and also the original signal is amplified in desired one.

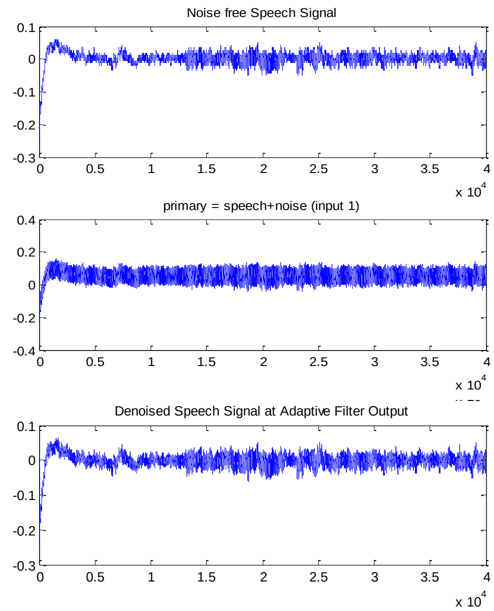

Figure 6: [JAYA] Signal graphs for 3.5seconds live speech.

From the fig.7 it is clear that by using PSO we can cancel the noise to greater extent. And the signal strength is amplified in the reconstructed signal.

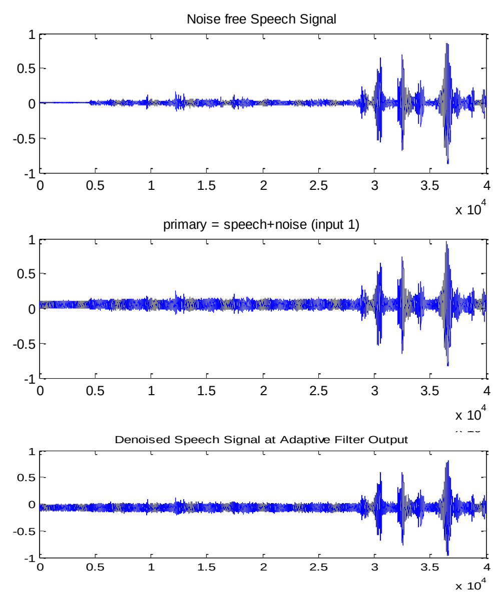

Figure 7: [PSO] Signal graphs for 3.5seconds live speech.

Algorithm	Samples	MSE	Execution time in secs
LMS	24000	0.0002	6.632187
JAYA	24000	0.0001	23.34288
PSO	24000	0.0057	7.837653

Table1: parameter based algorithmic results with loop size 500.

S.NO	Step Size	Loop Size	MSE	Ref[3]
1	0.003	622	0.0016	0.056
2	0.004	537	0.0014	0.045
3	0.005	489	0.0012	0.037
4	0.006	468	0.0010	0.029

Table2: parameter based LMS algorithm MSE variations.

On the whole it is clear that JAYA is time taking process to cancel out the noise because of searching out the best weights to forward for the cancellation process. And even though the PSO serving best than the JAYA and computing set of weights the MSE is less than that of LMS. So, as a result of comparison table-1, LMS gave the best MSE, that's why it is tested with different step and loop sizes. But the limitation is on LMS is all about the row vector size. If it is increases then the convergence also increases, but computational complexity increases. Even though it is more advantageous as it is simple to perform.

5. CONCLUSION

This paper presented an adaptive noise cancellation system (ASNC)using LMS, JAYA, PSO algorithms. By implementing these algorithms in digital signal processing mainly in noise cancellation and in interference cancellation a proper communication will be achieved. Proper mechanism is needed and that prompt the perfect noise cancellation. Already the cancellation of noise that is added in capturing challenging the signal processing. In that kind of scenario if the processed signal needed to be perfect as captured one. Otherwise it will have severe noise content at the receiver side and the intensity is more than that of the captured one. Hence by using these algorithms the noise that is adding to the captured signal due to transmission, processing, storage and conversions in digital domain eliminated by implementing this adaptive filter mechanism. Moreover, rather than the conventional LMS and several kinds of LMS, Bio-Inspired algorithms like JAYA, PSO and many more gives their best in adaptive speech noise cancellation mechanism. Not only for speech noise cancellation it can be useful in military and medical field applications. The whole concept can be applied on DSP processors like TMS320C6748 to investigate by means of hardware.

REFERENCES:

[1]D.Nicole, R.romulus, "Noise Canceling in audio signal with Adaptive Filter"University of Oradea,Vol.45,Number 6,2004,pp 599-602.

[2] S.J.Elliott & P.A.Nelson,"Active noise control," IEEE Signal Processing Mag., vol. 10, no. 4, pp. 12-35, 1993.

[3] Patel Shobhit,Pandya Killol, Bhalani Jaymin, Kosta Yogesh "Adaptive Noise Cancellation Using Least Mean Square algorithm" IJSAT Volume I, Issue I, (Oct-Nov.) 2010 pp 019-028.

[4] Rao, R.V., JAYA: A simple and new optimization algorithm for solving constrained and unconstrained optimization problems. International Journal ofIndustrial Engineering Computations, 7(2016), 19-34.

[5] Shi Y, Eberhart RC (1998) Parameter selection in particle swarm optimization. In: The proceedings of evolutionary programming VII (EP98), pp 591-600.

[6] S.A.Hadei," A family of Adaptive Filter Algorithms in noise cancelation for speech Enhancement", International Journal of Computer and Electrical and Engineering, VOL.2, No.2, April 2010, pp 307-315.

[7] Damberg W., Foss T. (1982). Statistics of frequent noise sources (in German: Statistik haufiger Larmquellen). Zeitschrift fiir Larmbekampfung 29, 1982, p. 55-59.

[8] John Wiley & Sons Ltd "Noise Interference with Oral Communications" © 1986 SCOPE.

[9] "Adaptive Signal Processing: next generation solutions" by Simon Haykin © 2010.

[10] "Computational Vision and Bio Inspired Computing" by D.Jude Hemanth and S.Smys Editors © Springer International Publishing 2018.