International Journal of Engineering Innovation & Research
Volume 3, Issue 2, ISSN: 2277 – 5668

Comparitive Study of Different Active Noise Cancellation Methods & Transfer Function Method

Prof. Ritu Chauhan
Department of Electronics & Communication
Technocrats Institute of Technology
Email: ritu_chauhan82@rediffmail.com

Kumkum Dubey
Department of Electronics & Communication
Technocrats Institute of Technology
Email: dubey.kumkum@gmail.com

***Abstract* – Today there is a need for high speed and efficient data transmission over the communication channels. It is a challenging task for the engineers and scientist to provide a reliable communication service by utilizing the available resources effectively in-spite many factors that distort the signal. The main objective of the digital communication system is to transmit symbols with minimum errors. During this process the transmitted data is distorted, due to the effects of linear and nonlinear distortions. Linear distortion includes inter-symbol interference (ISI), co-channel interference (CCI) in the presence of additive noise. Nonlinear distortions are caused due to the subsystem like amplifiers, modulator and demodulator along with nature of the medium. For removing these distortions different types of techniques are developed. In this paper we have to discuss some nonlinear techniques like genetic algorithm and Particle swarm optimization (PSO) algorithm & compared it. Here we use transfer function method for making these algorithm become more effective.**

***Keywords* – Inter-Symbol Interference, Distortion, Non-Linear.**

I. Introduction

Present days high speed & quality is a major issue in communication. Lots of techniques are developed for improving the quality of signal in communication field. For proper communication, it is necessary that information is received at the receiver without any distortion. But due to the presence of noise distortion take place. For transmission of acoustic signal this distortion is not tolerable. For removing distortion it is necessary that noise will be cancelled out. Different methods are used for cancelling the noise one of the famous method is active noise control (ANC) method [5]. In ANC method an antinoise signal is generated, whose magnitude is similar to the noise signal but its phase is opposite to the noise signal. When antinoise and noise signal are combined then destructive interference take place and noise is cancelled out. This scheme contains a reference microphone which sampled noise to be cancelled, an electronic control unit to process the input signal and generate control signal. This control signal is given to the loudspeaker and finally loudspeaker generate antinoise signal and antinoise signal get mixes with noise signal and cancelled it. If some noise is remaining out then it is treated as an error signal and it is absorbed by the microphone it act as feedback signal to the controller. The controller can adjust itself so that generating such type of antinoise signal so that it can cancelled out noise completely and error become zero. Controller contain digital filter which synthesis the antinoise signal. The performance of digital filter is affected by the type of filter, filter weight values are adjusted by using different algorithm. Large numbers of algorithms are developed for cancellation of noise. Like LMS, Filtered-X LMS algorithm, FLANN etc. LMS is one of the simplest algorithms but its convergence speed is low. So that it is updated and filtered-X LMS algorithm is used for removing noise of linear environment. But these methods are suitable for non-linear environment, hence other methods were developed for non-linear environment these include volterra series, memory polynomial filters, FLANN filter etc. FLANN is one of the successful methods for non-linear environment. But this method increases computation complexity of the system .For removing the limitation of FLANN genetic algorithm is used with it. Genetic algorithm and PSO algorithm are the best algorithm for removing noise. Since their convergence speed is high in comparison to other algorithm and they remove the requirement of secondary path thus complexity in computation reduces.

In this we don't use neural network. Here noise is calculated from transfer function of channel. This is a simplest method for calculating noise. It can measure accurately and fastly. Since neural network is a complex method, so make it simple transfer function method is used.

II. Adaptive Algorithm

Adaptive filters are used for removing noise from the original signal, these filters vary their coefficient according to the variation in the communication properties of the communication channel. These filters automatically adapt time –varying properties of the communication channel.

A. Principal of Adaptive Algorithm

When the noise is mixes with the original then adaptive filter remove this noise by subtracting the same amount of noise from the distorted signal. But it is difficult to generate the same amount of noise from any noise source .Hence estimated noise signal is used. This estimated noise signal is generated by a noise source and filter which is linearly related with the noise of distorted signal. By subtracting this estimated noise signal which is generated by filter from distorted signal, desired signal is obtained. If estimated noise signal is more close to real noise of distorted signal then more desired signal is obtained. This process of adaptive filter is known as active noise cancellation. Advantage of adaptive filter over any digital filter is that adaptive filter change filter coefficients according to the variation in the input signal, environment and output signal and channel characteristic. Different types of adaptive algorithms are used for varying the filter coefficients. Some of them are described in following section.

B. LMS Algorithm

Least mean square algorithm was introduced by Widrow and Hoff in 1959. It uses a gradient-based method of steepest decent. The steepest decent relies on the slope at any point on the surface to provide the best direction to reach the lower point on the surface and find the bottom of the surface. The steepest decent direction gives the greatest change in the elevation of the surface of the cost function for a given step. In LMS algorithm cost function is the mean square error. The location of the lowest point of the surface define the optimum values of the filter coefficients. The magnitude of the cost functions parameter is the proportional to the magnitude of the slope of the cost function. In LMS algorithm at each step the coefficients are updated by finding the gradient of the mean square error. If MSE-gradient is positive , then it implies that error would keep increasing positively. If the same coefficients is used for further method, hence value of the coefficients are reduces for further steps and vice versa. This algorithm perform following operation for updating the coefficients:-

(1) Calculate the output signal y(n) from the FIR filter.

$$y(n)= u^T(n).\ w(n) \qquad (1)$$

where u(n) the filter input vector, and $u(n)=[x(n)\ x(n-1) \ldots\ldots x(n-N+1)]^T$, w(n) filter coefficient vector $w(n)= [w_0(n)\ w_1(n)\ldots\ldots w_{N-1}(n)]^T$

(2) Calculate the error signal e(n).

$$e(n)=d(n) - y(n) \qquad (2)$$

(3) Updates the filter coefficient by using following equation

$$W(n)=w(n)+u*u(n)*e(n) \qquad (3)$$

Where u is the step size of the adaptive filter, w (n) is the filter coefficient, u(n) is the filter input vector. It is necessary that ‘u’ is chosen properly because convergence rate depend on ‘u’. Convergence speed of this algorithm is slow.

C. Filtered -X-LMS algorithm

This algorithm is used in order to improve the stability & performance of system. It is called filtered-X-LMS algorithm [4] because it require a filtered version of the reference signal as input. This filtering action is taken place through the secondary path transfer function. Secondary path is the path taken by anti noise from output loudspeaker to microphone. The characteristics of path is estimated through its transfer function. So that it help in calculating an output signal which destructively interfere with noise signal. The secondary path include delay in the signal transmission which is reduced by algorithm. Due to delay phase shift is introduced in the secondary path. This phase shift is reduces by including the transfer function of secondary in the output equation. This is involved in filtered –X-LMS algorithm.

The output of the adaptive filter is

$$y(n)= w^t(n)\ x(n)$$

Where w (n) = [w0(n) w1(n) · · · wL−1(n)]T, is the coefficient vector of the adaptive filter W(z), x(n) =[x(n)x(n-1)_ _ _ _ _ _ _ _ _ _x(n-L+1)] T is the reference signal. The output of the adaptive filter y(n) is filtered through the secondary path S(z) and is subtracted from the primary noise d(n) to generate error.

$$d(n) =P(n)*x(n)$$

$$y'(n)=s(n)*y(n)$$

P(n) is the primary path response.
S(n) is the secondary path response.
e(n) is the error.

The updated filter coefficient is find out from the following equation:-

$$W(n+1) =w(n) +\mu e(n)x'(n)$$

where μ is the step size.

This algorithm calculate the “slope” of the error surface and hence calculate weight that will cause the error to move down the slope to a smaller value when the slope is reduced to zero, the algorithm was to investigate the limits of the convergence & the influence on the obtainable noise reduction. The disadvantage of this algorithm is that here estimation of secondary path take place which is complicated & selection of proper step size is required.

IV. Evolutionary Algorithm

These algorithms are stochastic search methods that mimic the metaphor of natural biological evolution. Evolutionary algorithm based method are used in communication system for channel equalization. This algorithm based on the principal of survival of the fittest to produce better and better approximations to a solution. At each generation a new set of approximations is created by the process of selecting individual according to their level of fitness in the problem domain and combine them together for generating new individuals. This process leads to the evolution of population of individual that are better suited to their environment. Evolutionary algorithms involve selection, recombination, migration, locality selection and neighborhood.

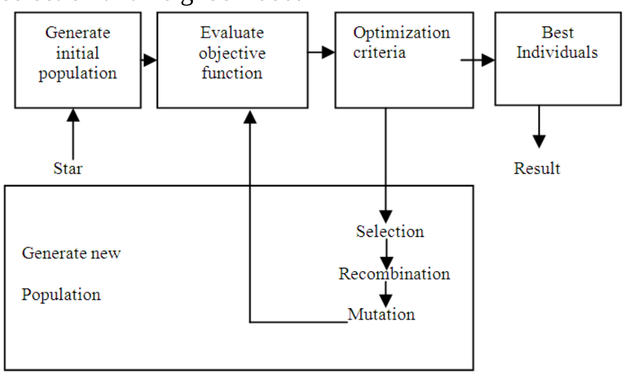

Fig.1. Structure of a single population evolutionary algorithm

At the beginning a number of individual are randomly initialized. The objective function is then evaluated for these individuals and the initial generation is produced. If the optimization criteria are not met the creation of new generation starts. Individuals are selected according to their fitness for the production of off-spring . Parents is recombined to production off-spring. All offspring will be mutated with a certain probability. The fitness of the offspring is then computed .The offspring are inserted into the population replacing the parents, producing a new generation. This cycle is performed until the optimization criteria is reach. It contain following algorithms:-

A. Genetic Algorithm

This is the most popular type of evolutionary algorithm. Genetic algorithm is based upon the process of natural selection and doesn't require gradient statistics. This algorithm is able to find a global error minimum. The genetic algorithm [11] with small population size and high mutation rates can find a good solution fastly. This algorithm is started with a set of solution called population. Solution from one population are taken and used to form a new population. The new population is better than old population. Solutions which are selected to form new solutions are selected according to their fitness. This process is repeated until some condition is satisfied. In genetic algorithm, a population is evolved toward better solution. Each individual solution has a set of properties which can be mutated and altered, solution are represented in binary as string of 0s and 1s. The process of randomly generated population at the starting of algorithm is known as iteration is called a generation. Finally it involved all the steps of evolutionary algorithm that means selection of individual on the basis of fitness value and then replacement of old individual until the optimum solution is achieved.

Limitation:

1. Evaluation of fitness function for complex problem is complicated. Finding the optimal solution to complex high dimensional, multimodal problems require very expensive fitness function evaluation.
2. Genetic algorithm is not suitable for complex function. Especially when the number of elements which are exposed to mutation is large.
3. Genetic algorithm may have a tendency to converge towards local optima or rather than the global optimum of the problem.

B. Particle swarm optimization

It was first developed in 1995 by Eberhart and Kennedy rooted on the notion of swarm intelligence of insects, birds etc. This algorithm attempts to mimic the natural process of group communication of individual knowledge that occurs when such swarms, flock, migrate, forage etc. in order to achieve such optimum property such as configuration or location. Similar to evolution algorithm , conventional PSO begins with a random population of individuals; here termed a swarm of particles. The conventional PSO algorithm begins by initializing a random swarm of M particles each having unknown parameters to be optimized. At each epoch, the fitness of each particle is evaluated according to the selected fitness function. The algorithm stores and progressively replaces the most fit parameter of each particle(pbest, i=1,2,3-----M) as well as a single most fit particle (gbest) as better fit parameters are encountered. The trajectory of each particle is influenced in a direction determined by the previous velocity and the location of gbest and pbesti. Each particles previous position(pbesti) and the swarms overall best position (gbest) are meant to represent the notion of individual experience memory and group knowledge of a " leader or queen", respectively that emerges during the natural swarming process. The acceleration constants are typically chosen in the range ε(0,2) and serve dual purposes in the algorithm. For one they control the relative influence toward gbest and pbesti respectively by scaling each resulting distance vector. Secondly, the two acceleration coefficients combined form what is analogous to the step size of an algorithm. The new particle coordinates can lie anywhere within the bounded region, depending upon the weights and random components associated with each vector. The particle update bounds are basically composed of all of the bounded regions for each vector with the addition of previous velocity component. When a new gbest is encountered during the update process all other particles begin to swarm toward the new gbest, continuing the directed global search along the way. When all of the particles in the swarm have converged to gbest, the gbest parameters characterize the minimum error solution obtained by the algorithm.

C. Application PSO method in ANC system

In the ANC system the electrical control signal is generated in a sample by sample basis and is converted to acoustic signal using a loudspeaker system. The control signal is generated by passing the reference microphone signal through ANC system. A linear ANC system consists of a finite impulse response filter. The objective of the tuning algorithm in an ANC system is to optimize the FIR filter coefficients of the ANC system such that received signal in the error microphone is minimized. PSO algorithm cannot be directly used in the ANC system as the performance of the algorithm depends on a set of error samples and not on the instantaneous value of the error microphone. In addition the conventional PSO based optimization, the same data set is used to the particles respectively in every generation. The objective of the PSO based ANC algorithm is to minimize the mean square error that is sensed by the error microphone. Let us consider a coefficient vector of P adaptive filters as population which is represented as a set of particles in the PSO terminology. The PSO algorithm is initialized with a population of random solutions. In this case the coefficient vectors of the P adaptive filters are used as the initial random solutions. Let this set be represented as

$$W = \begin{pmatrix} w_1^1 & w_1^2 \ldots\ldots\ldots\ldots\ldots\ldots w_1^P \\ w_2^1 & w_2^2 \ldots\ldots\ldots\ldots\ldots\ldots w_2^P \\ w_N^1 & w_N^2 \ldots\ldots\ldots\ldots\ldots\ldots w_N^P \end{pmatrix}$$

Each column of W in 1 eq. is a potential initial solution and is called a particle, this represents the coefficient vector of one of the P adaptive filters. The noise signal received by the reference microphone is fed to each of these filters which produce $y_1(n), y_2(n) \ldots\ldots\ldots\ldots\ldots y_p(n)$ as output at different time frames. The P outputs act as the control signal in the ANC which pass through the loudspeaker system to generate the antinoise signal set represented as $d_1^{\wedge}(n), d_2^{\wedge}(n) \ldots\ldots\ldots d_p^{\wedge}(n)$. It is clarified here that the set $\{y_1(n), y_2(n) \ldots\ldots\ldots y_p(n)\}$ consists of electrical signals where as noise signals are represented by $\{d_1(n), d_2(n) \ldots\ldots d_P(n)\}$. A set of residual error signals $\{e_1(n), e_2(n) \ldots\ldots\ldots e_p(n)\}$ is received by the error microphone. These error signal set is a result of the acoustical superposition of noise and antinoise signal. Thus error signals are given by $e_P(n)= d(n) + d_P^{\wedge}(n)$

where p= 1,2,3----------P, n denotes the sample index of the input signal which ranges from 1 to M and M is the total number of samples used in each generation. The mean square error of each of these P errors, represents the fitness of each particle stored in the PSO processor. In the PSO terminologies, the particle is flown through the problem space with a velocity in the Kth iteration are denoted by $w_i(k)$ and $v_i(k)$ respectively. The position of the ith particle with the smallest mean square error in the previous position is recorded and represented by the symbol pbest i and its position is represented as $w_{pbest\ i}$. The best among all the particle is represented by the symbol gbest and its position is represented as w_{gbest}. The PSO algorithm updates the velocity and position of each particle toward its w_{pbest} and w_{gbest} positions at each step according to the following update equations:-

$v_i(k)= v_i(k-1) + r_1 [w_{pbesti} – w_i(k)]+ r2 [w_{gbest} – w_i(k)]$.

$W_i(k) = w_i(k-1) + v_i(k)$

Where r1 and r2 are two random numbers that are generated independently in the range [0,1]. The basic PSO algorithm uses two random vectors for r1 and r2; however we use r1 and r2 as random numbers instead , due to reduced complexity. $V_i(k)$ is a new velocity for each particle based on its previous velocity $v_i(k-1)$, the particle's position at which the best fitness has been achieved w_{pbest} and the best positions of each particle achieved so far among the neighbors w_{gbest}.

Advantages of PSO:

1) PSO has memory i.e. every particle remembers its best solution as well as the group's best solution.

2) Its initial population is maintained fixed throughout the execution of the algorithm and so there is no need for applying operators to the population.

3) This process is both time & memory storage consuming.

4) Another key advantage of PSO is the ease of implementation in both the contest of coding and parameter selection.

Limitations:-

1) If the new gbest particle is an outlying particle with respect to the swarm, the rest of the swarm can tend to move toward the new gbest from the same direction.

2) Particles closer to gbest will tend to quickly converge on it and become stagnant while the other more distant particles continue to search. A stagnant particle is essentially useless because its fitness continues to be evaluated but it no longer contributes to the search.

Further researches will take place for removing these limitations of PSO.

V. Proposed Methodology

Transfer function is measured here for calculating noise. For measuring transfer function a pilot signal is sent through the channel and then ratio of fourier transform of input signal and output signal is calculated that gives us transfer function of the channel. Thus by calculating the transfer function we calculate the behavior of channel. We follow the change occur in the behavior of channel through its transfer function. This change is take place due to noise effect.

Noise at observed microphone = Transfer function noise to observed microphone*noise signal.

Noise at error microphone = Transfer function noise to error Microphone* noise signal.

After calculating noise from transfer function we find out the difference between the noise and antinoise signal. If noise is not fully cancelled by antinoise signal then coefficients of PSO algorithm are updated so that it generate such type of antinoise signal which cancel the noise completely.

VI. Conclusion

Finally we concluded that cancellation of noise is very simple in industries if we calculated its transfer function without using any neural network method we easily find out the approximate value of noise and then generate its antinoise signal which cancelled it. This method is very suitable for removing non-linear distortions. For making PSO algorithm more effective we calculate the transfer function at microphone and then update the coefficients of PSO. Since this method converge the PSO algorithm fastly and help it in generating an perfectly antinoise signal.

References

[1] S. U. H. Qureshi, "Adaptive Equalization," IEEE, vol. 73, pp. 1349–1387, September 1985

[2] J.Kennedy and R.C.Eberhat," Partical Swarm Optimizer ", IEEE Int. Conf. On Neural Networks,IV, 1942-1948. Piscataway, NJ:IEEE Service Center, 1995.

[3] C. Y. Chang and D. R. Chen, "Active noise cancellation without secondary path identification by using an adaptive genetic algorithm IEEE Trans. Instrum. Meas., vol. 59, no. 9, pp. 2315–2327, Sep. 2010.

[4] S. C. Douglas, "Fast implementations of the filtered-X LMS and LMS algorithms for multichannel active noise control," *IEEE Trans. Speech Audio Process.*, vol. 7, no. 4, pp. 454–465, Jul. 1999.

[5] S. M. Kuo and D. R. Morgan, *Active Noise Control Systems—Algorithms and DSP Implementations*. New York: Wiley, 1996.

[6] C. A. Jacobson, C. R. Johnson, D. C. McCormick, and W. A. Sethares, "Stability of active noise control algorithms," *IEEE Signal Process. Lett.*, vol. 8, no. 3, pp. 74–76, Mar. 2001.

[7] M. Diego, A. Gonzalez, M. Ferrer, and G. Pinero, "An adaptive algorithms comparison for real multichannel active noise control," in Proc. 12th Proc. Eur. Signal Process. Conf., Sep. 2004, pp. 925–928.

[8] M. Young, The Technical Writer's Handbook. Mill Valley, CA: University Science, 1989.

[9] John G. Proakis, Dimitris G. Manolakis, "Digital Signal Preocessing", Pearson Publication, 2007.

[10] S. M. Kuo and D. R. Morgan, Active noise control systems: Algorithms and DSP implementations. New York: Wiley, 1996.

[11] L. Yao and W. A. Sethares, "Nonlinear parameter estimation via the genetic algorithm," IEEE Transactions on Signal Processing, vol. 42, April 1994.

[12] S.J. Elliott and P.A. Nelson. Active noise control. IEEE signal processing magazine, pages 12–35, October 1993.

[13] P. L. Feintuch, N. J. Bershad and A. K. Lo, "A frequency-domain model for filtered LMS algorithms - Stability analysis, design, and elimination of the training mode," IEEE Transactions on Signal Processing, vol. 41, pp. 1518-1531, Apr. 1993.

[14] D.R Morgan. An analysis of multiple correlation cancellation loops with a filter in the auxiliary path. IEEE Transactions on Acoustics, Speech and Signal Processing, ASSP-28(4):454–467, August 1980.

[15] C.C. Boucher, S.J. Elliott, and P.A. Nelson. Effect of errors in the plant model on the performance of algorithms for adaptive feedforward control. In Proceedings of Institution of Electrical Engineers, volume 138,pages313–319,1991.

Author's Profile

Ritu Chauhan

is currently working as a Professor in Electronics and Communication department, Technocrats Institute of Science & Technology college, Bhopal. She had completed her B.E. in Electronics & Communication branch from MPCT College Gwalior and M.Tech. in Communication Control & Networking from MITS college Gwalior.

Kumkum Dubey

received her B.E. degree in Electronics and Communication from R.G.P.V. University, Bhopal in 2011. She is currently pursuing M.Tech. in Digital Communication from Technocrats Institute of Science and Technology college, Bhopal. She qualified gate in 2011 with 93.4 percentile.