Pseudo Affine Projection Algorithm Based Noise Minimization from Speech Signals

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ABSTRACT
This paper presents the noise minimization from speech signal and how speech enhancement is done by using pseudo affine projection algorithm. Noise minimization is one of the major applications of the adaptive filters used in recent research areas. The Affine Projection algorithm and its variants are popular choice for noise minimization because of its fast convergence like recursive least square (RLS) and low complexity like least mean square (LMS) algorithm. The pseudo affine projection is a gradient type variant of affine projection algorithm with relaxed step-size conditions and less complexity which offers improved performance. The pseudo affine projection algorithm works successfully for local robustness properties of algorithms, as well as steady-state values of moderate to high accuracy specially when applied to long filter order. The maximum signal to noise ratio improvement (SNRI) achieved is 40.22dB and the minimum mean square error (MSE) achieved is 0.0031 at filter order 400 for input SNR of -20dB. The robustness of this algorithm is verified by evaluating it for various noises.

Keywords
Adaptive filter, Affine Projection (AP), SNRI, MSE, and Computational Complexity.

1. INTRODUCTION
Noise is an undesired signal that is available in the environment or coming from various sources that interferes with the desired signal. This results in critical issues in speech operated devices as it corrupts the desired signal. In speech operated devices, noise cancellation system is implemented to reduce active noise based on the optimization algorithms [1-3]. The noise can be of different types such as tape hiss, microphone hum, fighter plane noise (F16), Babble16 noise, car noise, pink noise, factory noise etc. When microphone picks a sound, noise in the sense of sound is also picked up other than the user's interest which results in degradation of speech quality. Due to this, serious problems are countered during the analysis of a speech signal.

The adaptive algorithms used for noise cancellation system is mainly of two types- LMS and RLS [4]. All other algorithms are derived from these two algorithms. The affine family of algorithms is derived by the LMS types of algorithms, the only difference is that in affine projection, multiple past input vector are used to update the filter coefficient, while in conventional LMS only the current input vector is used as an excitation for updating the filter coefficients [5-8]. The affine projection algorithm shows the improved performance in terms of SNR improvement, minimization in MSE and convergence rate. The Pseudo Affine Projection (PAP) Algorithm is a member of APA family algorithms which works on the basis of utilizing reasonable approximations of the original sample by sample operations of affine projection algorithm calculations. AP algorithms are more computationally efficient and more versatile in comparison to LMS family and RLS family algorithms.

The noise cancellation setup is shown in fig. 1 [9]. Here the desired signal d(n) is composed of the speech signal s(n) corrupted by adaptive noise n(n). The input to the adaptive filter is the reference noise n'(n). The noises n(n) and n'(n) taken are correlated. The filter output y(n) is subtracted by the desired signal d(n), generating an error signal e(n). This error signal e(n) is used to adjust the filter coefficients based on the adaptive algorithm used to minimize the noise added. The separation of noise signal from speech signal is required in order to improve the performance of communication system.

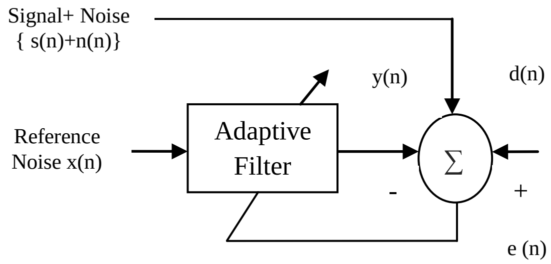

Fig 1: Adaptive Noise Canceller

2. PSEUDO AFFINE PROJECTION ALGORITHM (PAP)
An affine projection algorithm (APA) and its variants have been proposed in recent years [10-14]. The weight updation of APA algorithm depends on multiple recent past input vectors. where as in LMS and NLMS algorithms the past input vectors are not considered. The APA variables used in adaptive noise cancellation system are the excitation noise signal matrix for adaptive filter [*X(k)]*, which is L by M matrix, the desired signal is the corrupted speech signal *d(k).* Here, projection order of APA is *M* and the length of filter is *L*. The weight update equation for classical regularized APA is stated as

$$w(k) = w(k-1) + \mu X(k)[X^T(k)X(k) + \delta I]^{-1} e(k) \tag{1}$$

In equation (1), the step size parameter μ is having in the range of $0 \le \mu \le 1$, the error vector is calculated as $e(k) = d(k) - y(k)$ and the filter output is calculated as $y(k) = X^T(k)w(k-1)$, δ is called as the regularization parameter. While speeding up the convergence, the tracking capability of the algorithm gets restricted as the prediction order *M* increases for the large filter order *L*.

The pseudo affine projection (PAP) calculation is determined by utilizing the reasonable approximations of original Affine Projection algorithm. Considering a Gram-Schmidt procedure that changes the input vector $x(n-i)$ into its orthogonal form $x_i(n)$, $0<i<M-1$, where '*M*' denotes the order of projection [15-16]. By solving '*M*' linear prediction of orders of 0 through $M-1$, the transformation matrix *L* can be obtained. The pseudo affine projection algorithm using preprocessor can be expressed as:

$$e(n) = y(n) - X^T(n)w(n) \tag{2}$$

$$w(n+1) = w(n) + \mu \sum_{i=0}^{M-1} \frac{\varepsilon_i(n)}{\|x_i(n)\|_2 + \delta} x_i(n) \tag{3}$$

Where '$w(n)$' is $L\times1$ tap weight vector, $x(n)$ is input noise matrix of size $L\times M$, $y(n)$ is adaptive filter output vector of size $M\times1$, 'μ' is the step-size parameter and 'δ' is the regularization parameter. The term '$\varepsilon_i(n)$', denotes elements of the transformed error vector '$\varepsilon(n) = L^T(e(n))$.'

3. SIMULATION RESULTS

This section presents the simulation results of PAP algorithm for the ANC setup. Various performance parameters are evaluated which are needed for analyzing the ANC system. This application is basically a noise cancellation application of adaptive filter, where the noise introduced in the original clean signal is being cancelled using adaptive filter. For the simulation setup one clean speech sentence "YAHA SAI LAGHBAG PANCH MEAL DAKSHIN PASCHIM MAI KATGHAR GAON HAI" and a word "SHOONYA", is taken from Hindi Speech Database [17] and the various types of noises like "F16", "FACTORY", "CAR-NOISE" and "BABBLE16" which are taken from NOISEX-92 database [18] are used to corrupt above mentioned clean signals. In the experimental setup, the filter orders taken are 400, 410, 420 and 430. The corrupted version of speech signal is prepared at different input SNR levels of -5dB, -10dB and -20dB. The performance of the said algorithm is measured on the basis of signal to noise ratio (SNR), mean square error value (MSE), and robustness of the system. The SNR improvement parameter is used as a measure to compare the level of a desired signal to the level of the noise added and is expressed in decibel as-

$$SNR(dB) = 10\log_{10}\left(\frac{(SP-NP)}{NP}\right) \tag{4}$$

Where '*SP*' is the signal power calculated using output error signal and '*NP*' is the noise power calculated using estimated noise signal. The next performance parameter measured is the mean square error (MSE). For MSE calculation, first, the error signal is calculated for each iteration by subtracting the desired signal from filter output signal. Then the mean square value of error gives the MSE value. The third parameter measured is the robustness of the system which is verified by observing the output for different types of noises added with the same standard clean speech signal at various input SNR levels.

Table 1 Shows the analysis of SNR improvement when the speech word "SHOONYA" is corrupted with fighter plane (F16) noise. The signal is corrupted at input SNR levels of -5dB, -10dB and -20dB. It is observed that the highest SNR improvement of 40.2174dB is achieved for filter order 420 and input SNR -20dB. To check the robustness behavior of the system the same analysis is performed for the different speech signals corrupted with different types of noises. Table 2 to table 4 depicts the similar analysis done for speech word "SHOONY" and Speech sentence "YAHA SAI LAGHBAG PANCH MEAL DAKSHIN PASCHIM MAI KATGHAR GAON HAI" which are corrupted by babble16 and F16 noises for same input SNR levels and filter orders and it is observed that approximately same results is found in all cases. It is also clear that the algorithm performs well specially for higher noise conditions.

Fig. 2 to Fig. 5 Shows the comparison of output SNR achieved with different input SNR levels for all said speech and noise samples. Here, the plots are the results of ensemble averaging over 20 independent trials.

Table 1: SNR Improvement for speech signal "SHOONYA" corrupted with F16 noise.

Filter Order	Input SNR (-5dB)	Input SNR (-10dB)	Input SNR (-20dB)
400	9.3692	19.1713	38.9986
410	11.1618	20.6829	40.2189
420	11.1529	20.678	40.2174
430	11.1387	20.6702	40.2149

Table 2: SNR Improvement for speech signal "SHOONYA" corrupted with "Babble16" noise.

Filter Order	Input SNR (-5dB)	Input SNR (-10dB)	Input SNR (-20dB)
400	9.6502	19.3266	39.0296
410	10.689	20.3872	40.1266
420	10.6856	20.3853	40.126
430	10.6824	20.3835	40.1254

Table 3: SNR Improvement for speech signal "YAHA SAI LAGHBAG PANCH MEAL DAKSHIN PASCHIM MAI KATGHAR GAON HAI" corrupted with "Babble16" noise.

Filter Order	Input SNR (-5dB)	Input SNR (-10dB)	Input SNR (-20dB)
400	11.1735	20.6894	40.2209
410	9.3653	19.1693	38.9982
420	9.3631	19.1681	38.998
430	9.361	19.1668	38.9973

Table 4: SNR Improvement for speech signal "YAHA SAI LAGHBAG PANCH MEAL DAKSHIN PASCHIM MAI KATGHAR GAON HAI" corrupted with "F16" noise.

Filter Order	Input SNR (-5dB)	Input SNR (-10dB)	Input SNR (-20dB)
400	10.6923	20.389	40.1272
410	9.6465	19.3244	39.0287
420	9.6413	19.3214	39.0277
430	9.6379	19.3192	39.0265

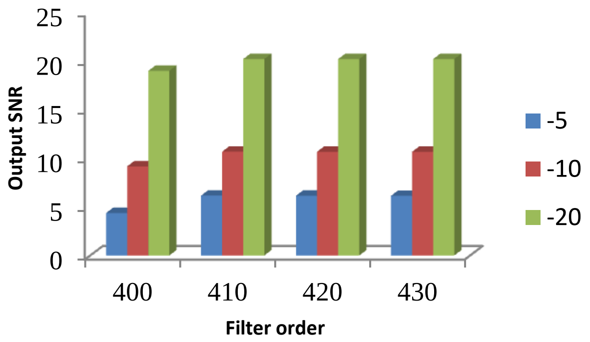

Fig 2: SNR Comparison for speech word signal "SHOONYA" corrupted with F16 noise for input SNR -5dB,-10dB and -20dB.

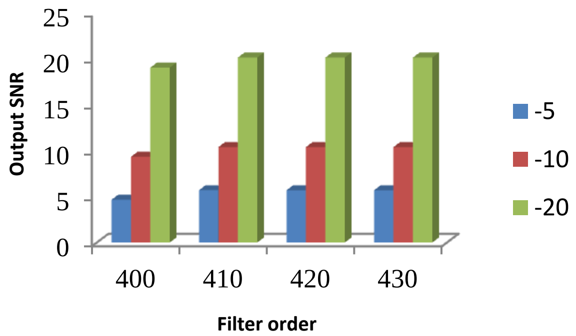

Fig 3: SNR Comparison for speech word signal "SHOONYA" corrupted with Babble16 noise for input SNR -5dB,-10dB and -20dB

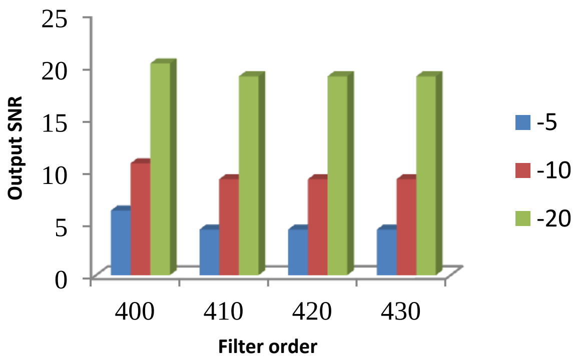

Fig 4: SNR Comparison for speech sentence "YAHA SAI LAGHBAG PANCH MEAL DAKSHIN PASCHIM MAI KATGHAR GAON HAI" corrupted with Babble16 noise for input SNR -5dB,-10dB and -20dB.

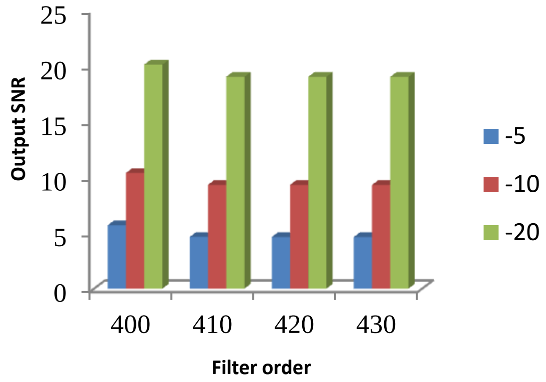

Fig 5: SNR Comparison for speech sentence "YAHA SAI LAGHBAG PANCH MEAL DAKSHIN PASCHIM MAI KATGHAR GAON HAI" corrupted with F16 noise for input SNR -5dB,-10dB and -20dB.

The analysis of mean square error (MSE) value for all said speech signals which are corrupted with different noises are done in table 5 to table 8. These analysis are also done for filter length 400, 410, 420 and 430 and for input SNRs of -5dB,-10dB and -20dB respectively. Since minimum mean square error value is majorly required for good performance, so from analysis result it is clear that minimum MSE is achieved in all cases. The minimum mean square error (MSE) achieved is 0.0031 at filter order 400 for input SNR of -5dB. Again, the robustness of this algorithm is verified by evaluating it for various noises. The results of MSE values obtained in all said cases are shown more clearly in fig. 6 to fig. 9.

Table 5: Mean square error (MSE) value for speech signal "SHOONYA" corrupted with F16 noise.

Filter Order	Input SNR (-5dB)	Input SNR (-10dB)	Input SNR (-20dB)
400	0.0103	0.0145	0.0242
410	0.0105	0.0146	0.0243
420	0.0105	0.0147	0.0243
430	0.0106	0.0148	0.0244

Table 6: Mean square error (MSE) value for speech signal "SHOONYA" corrupted with Babble16 noise.

Filter Order	Input SNR (-5dB)	Input SNR (-10dB)	Input SNR (-20dB)
400	0.0137	0.0157	0.0234
410	0.0137	0.0158	0.0234
420	0.0138	0.0158	0.0235
430	0.0138	0.0159	0.0236

Table 7: Mean square error (MSE) value for signal "YAHA SAI LAGHBAG PANCH MEAL DAKSHIN PASCHIM MAI KATGHAR GAON HAI" corrupted with "Babble16" noise.

Filter Order	Input SNR (-5dB)	Input SNR (-10dB)	Input SNR (-20dB)
400	0.0065	0.0068	0.0092
410	0.0065	0.0068	0.0093
420	0.0065	0.0069	0.0094
430	0.0066	0.0069	0.0094

Table 8: Mean square error (MSE) value for signal "YAHA SAI LAGHBAG PANCH MEAL DAKSHIN PASCHIM MAI KATGHAR GAON HAI" corrupted with "F16" noise

Filter Order	Input SNR (-5dB)	Input SNR (-10dB)	Input SNR (-20dB)
400	0.0031	0.0042	0.0094
410	0.0033	0.0042	0.0094
420	0.0032	0.0043	0.0095
430	0.0032	0.0043	0.0095

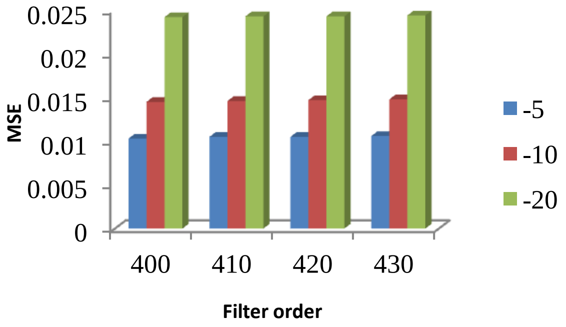

Fig 6: Mean Square Error (MSE) for speech word signal "SHOONYA" corrupted with F16 noise for input SNR -5dB,-10dB and -20dB

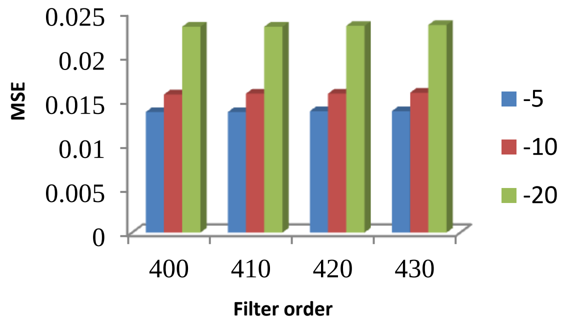

Fig. 7: Mean Square Error (MSE) speech word signal "SHOONYA" corrupted with Babble16 noise for input SNR -5dB,-10dB and -20dB

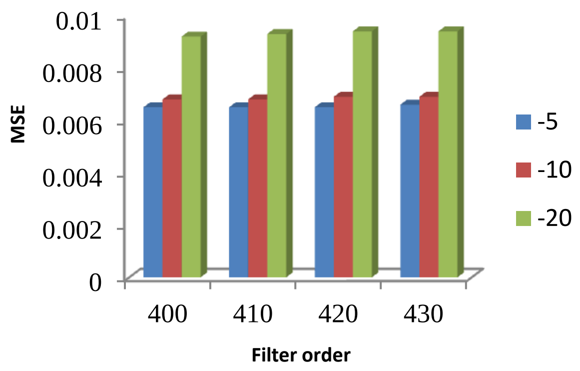

Fig 8: Mean Square Error (MSE) for speech sentence "YAHA SAI LAGHBAG PANCH MEAL DAKSHIN PASCHIM MAI KATGHAR GAON HAI" corrupted with Babble16 noise for input SNR -5dB,-10dB and -20dB

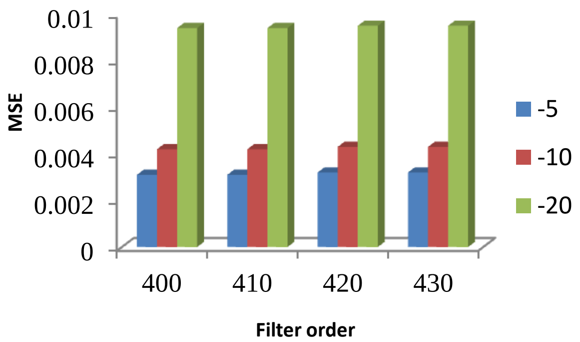

Fig 9: Mean Square Error (MSE) for speech sentence "YAHA SAI LAGHBAG PANCH MEAL DAKSHIN PASCHIM MAI KATGHAR GAON HAI" corrupted with F16 noise for input SNR -5dB,-10dB and -20dB

4. CONCLUSION

In this paper, the Pseudo Affine Projection Algorithm is implemented for adaptive noise cancellation system. The performance of this algorithm is measured in terms of signal to noise ratio improvement, the mean square error value and robustness. The result shows that the significant improvements are achieved in output SNR and mean square error. The PAP works well for variety of signals corrupted with variety of noises. The result also shows that the PAP outperforms especially for high noise conditions and for high filter orders.

5. REFERENCES

[1] Smith S. Haykin: Adaptive Filter Theory, 3rd ed. Upper Saddle River, NJ: Prentice Hall, (2002).

[2] Sayed, A.H.: Adaptive Filters. Wiley, Hoboken (2008).

[3] Davis, G, M, "Noise Reduction in Speech Applications", CRC Press LLC, ISBN 0-8493-0949-2, USA, 2002.

[4] D. K., Gupta V. K., and Chandra Mahesh, "Review Paper on Linear and Nonlinear Acoustic Echo Cancellation", Proceedings of the 3rd International Conference on Frontiers of Intelligent Computing: Theory and Applications (FICTA) 2015, Vol.2, pp. 465-473.

[5] K. Ozeki and T. Umeda, "An adaptive filtering algorithm using an orthogonal projection to an affine subspace and

its properties," *Electron. Commun. Jpn.*, vol. 67-A, no. 5, pp. 19–27, May 1984.

[6] H.-C. Shin and A. H. Sayed, "Mean-square performance of a family of affine projection algorithms," *IEEE Trans. Signal Process.*, vol. 52, no.1, pp. 90–102, Jan. 2004.

[7] S. L. Gay and J. Benesty, *Acoustic signal processing for telecommunication*. Second edition, Kubler Academic Publishers, 2001.

[8] Alberto Gonzalez , Miguel Ferrer , Felix Albu , and Maria de Diego, "Affine Projection Algorithm: Evolution to smart and fast algorithms and application" 20th European Signal Processing Conference, EURASIP, 2012 - ISSN 2076-1465.

[9] V. K. Gupta, D. K. Gupta, Mahesh Chandra, "Real time noise canceller using modified sigmoid function RLS algorithm", *proceedings of International Conference on Computational Vision and Robotic(ICCVR-2014)* 9th-10th, August, 2014 Bhubaneswar, INDIA, 2014.

[10] C. Paleologu, J. Benesty, "A Variable Step-Size Affine Projection Algorithm Designed for Acoustic Echo Cancellation*," IEEE Transactions on Audio, Speech, and Language Processing*, vol. 16, no. 8, pp. 1446-1478 2008.

[11] F. Bouteille, P. Scalart and M. Corazza, "Pseudo affine projection algorithm: new solution for adaptive identification," Eurospeech, 1999, vol. 1, pp. 427–430.

[12] Markus Rupp*, Senior Member, IEEE "*Pseudo Affine Projection Algorithms Robustness and Stability Analysis", 2011.

[13] F. Bouteille, P. Scalart and M. Corazza, "Pseudo affine projection algorithm: new solution for adaptive identification," Eurospeech, 1999, vol 1, pp. 427–430.

[14] Felix Albu, Martin Bouchard, and Yuriy Zakharov, "Pseudo Affine Projection Algorithms for Multichannel Active Noise Control" *IEEE.*

[15] J. Lee, Y.-C. Park and D. H. Youn, "Robust pseudo affine projection algorithm with variable step-size", ELECTRONICS LETTERS, vol. 44 no. 3, 31st January 2008.

[16] V. Myllylä and G. Schmidt, "Pseudo-optimal regularization for affine projection algorithms," in *Proc. IEEE Int. Conf. on Accoust., Speech, Signal Process. (ICASSP'02)*, Orlando, FL, May 2002, pp. 1917–1920.

[17] Samudravijaya, K et al. :Hindi Speech Database, Proc. ICSLP00, Beijing, China, CDROM 00192.pdf.

[18] Varga,A.; Steeneken, H. J. M.; Jones, D. :The noisex-92 study on the effect of additive noise on automatic speech recognition system, Reports of NATO Research Study Group (RSG.10) (1992).