International Journal of Informatics and Communication Technology (IJ-ICT)
Vol. 12, No. 1, April 2023, pp. 38~45
ISSN: 2252-8776, DOI: 10.11591/ijict.v12i1.pp38-45

Single line noise cancellation using derivative of normalized least mean square algorithm

Rathnakara Srinivasa Pandit, Udayashankara Veerappa
Department of Electronics and Instrumentation, S. J. College of Engineering, JSS Science and Technology University, Mysore, India

Article Info

Article history:

Received Jul 26, 2022
Revised Jul 31, 2022
Accepted Aug 9, 2022

Keywords:

EDNSS
EMSE
Misadjustment
NLMS
SNR
Speech enhancement

ABSTRACT

Suppression of noise in noisy speech signal is required in many speech enhancement applications like signal recording and transmission from one place to other. In this paper a novel single line noise cancellation system is proposed using derivative of normalized least mean spare algorithm. The proposed system has two phases. The first phase is generation of secondary reference signal from incoming primary signal itself at initial silence period and pause between two words, which is essential while adaptive filter using as noise canceller. Second phase is noise cancellation using proposed modified error data normalized step size (EDNSS) algorithm. The performance of the proposed algorithm is compared with normalized least mean square (NLMS) algorithm and original EDNSS algorithm using standard IEEE sentence (SP23) of Noizeus data base with different types of real-world noise at different level of signal to noise ratio (SNR). The output of proposed, NLMS and EDNSS algorithm are measured with output SNR, excessive mean square error (EMSE) and misadjustment (M). The results clearly illustrates that the proposed algorithm gives improved result over conventional NLMS and EDNSS algorithm. The speed of convergence is also maintained as same conventional NLMS algorithm.

Corresponding Author:

Rathnakara Srinivasa Pandit
Department of Electronics and Instrumentation, S. J. College of Engineering
JSS Science and Technology University
Campus Rd, University of Mysore Campus, Mysuru, Mysuru, Karnataka 570006, India
Email: rathnakara_s@sjce.ac.in

1. INTRODUCTION

The speech is very important way of communication in human being. But during recording and transmission of the speech many times corrupted by background noises, which reduces the quality and intelligibility of speech signal. When this corrupted signal is transmitted it becomes unpleasant to listen at receiving end. The background noise may be either stationary or non-stationary in nature. The suppression of stationary noise is not much complicated due to their statistical parameters are same with time for entire signal. But in case of non-stationary noise the statistical parameters like standard deviation, mean and amplitude distribution are changes with time [1].

From more than four decades researchers are focused in the area of noise estimation and developed the different speech enhancement algorithms as well as try to improve the intelligibility of the speech signal. Researchers are especially given more emphasis on different forms of adaptive filter like least mean square (LMS), leaky LMS, normalized (NLMS), and recursive (RLS) algorithm. Out of these algorithms NLMS and its derivative alogirthms are widely used for speech enhancement [2]. To enhance the speech signal in stationary noise environment either spectral subtraction or wiener filter is [3]–[5] suffieint. But enhancement of non-stationary signal like speech in non-stationary noise is very difficult and still active field of research.

When adaptive filter [6], [7] used as noise canceller it required primary signal along with secondary reference signal. The characteristics of secondary reference signal used in adaptive noise canceller must be correlated with noise that is present along with the primary signal. Researchers are worked in this field and developed various algorithms for speech/music discrimination, speech/noise discrimination, speech/song discrimination, and echo cancellation [8]–[14]. In single line noise canceller, the information about secondary reference is unknown and difficult to generate using mathematical procedure [12]. So, in this proposed method secondary reference signal is generated by primary signal itself by taking the information at initial silence period and pauses between the two words. The primary signal along with generated secondary signal is applied to an adaptive filter as primary and reference inputs. So, the proposed method has two stages: i) generation of secondary reference signal and ii) noise cancellation using an adaptive filter.

2. PROPOSED METHOD

2.1. Generation of secondary reference signal

The first step in the proposed algorithm is generation of reference signal. The steps followed to generate the secondary reference is signal is as follows:

Step 1: break the signals into frames of 0.01 seconds

The input signal can break into frames of 0.01 seconds using following procedure. Initially set the frame duration as 0.01 s and sampling frequency as 16,000 Hz. Then take the product of sampling frequency and frame duration. Finally calculate the number of frames by dividing total number of samples in the incoming signal by frame lengh and round off to nearest integer value.

Step 2: identifying signal frame and noise frame by thresholding method

Set the amplitude threshold depends on input noise level. In this work amplitude threshold fixed as 0.164 for 0 dB, 0.10 for 5 dB, 0.054 for 10 dB, and 0.043 for 15 dB. Then start with first frame and compare with threshold value, if the frame is less than threshold value segregates it as noise frame otherwise segregate as signal frame and repeat the same for all the frames.

Step 3: generation of secondary reference using pauses and initial silence period

Select the noise frame which gives information about initial silence period and pauses (approximately) in the speech signal. Start with noise present in the initial silence period and extend the same information until first pause then select noise frame extracted from first pause frame extend the same until next pause period and repeat the same for entire signal. This will be act as reference signal for adaptive filter.

The outcome of the procedure is illustrating in Figure 1. From top to bottom which consists of standard IEEE sentence (SP23) noisy speech signal corrupted with zero decibels babble noise as primary input, separated signal frame, separated noise frame, and completed reference signal. Then generated reference signal will be applied to adaptive noise canceller as secondary reference signal.

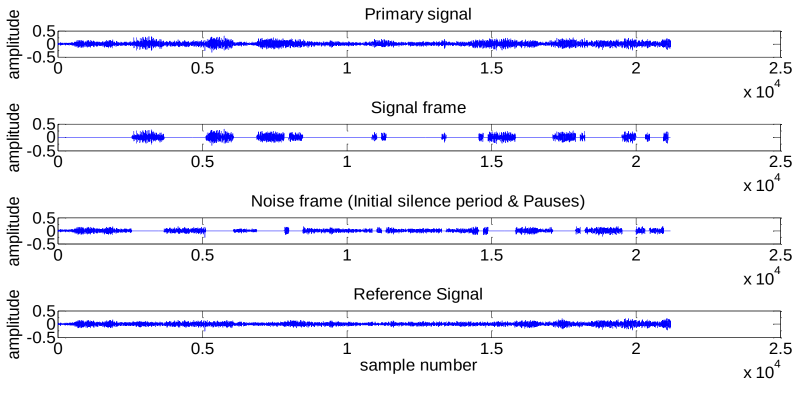

Figure 1. The primary signal, signal frame, noise frame, and the reference signal from ’0’ dB babble noise

2.2. Noise cancellation using adaptive filter

Next phase is noise cancellation using adaptive filter. The basic block diagram of adaptive noise canceller is shown in Figure 2. Where d(n) is primary input which consists of desired signal s(n) along with uncorrelated noise $n_0(n)$. The reference input x(n) is another noise $n_1(n)$ which is correlated with $n_0(n)$ and uncorrelated with desired signal s(n).

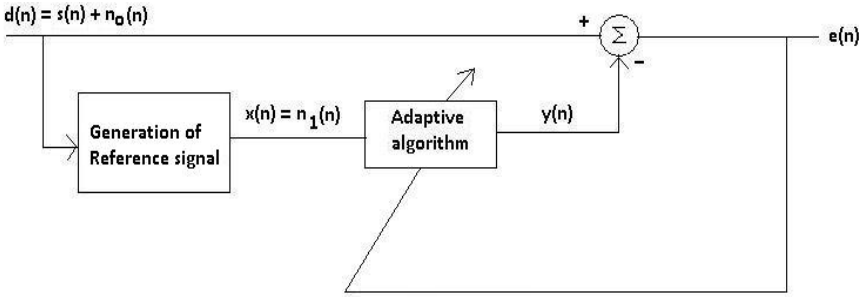

Figure 2. Single line adaptive noise canceller

The step size remains constant in the basic LMS algorithm for entire adaptive process but in NLMS algorithm the step size changes from one iteration to next iteration. In which it reduces slope overload problem and also rate of convergence of NLMS is faster than that of LMS filter [2], [15]. So, the method of varying the step size is focused by many researchers and developed different forms of NLMS algorithms [16]–[22]. In which error data normalized step size (EDNSS) [23] algorithm is also one type of NLMS derived algorithm. In this paper an algorithm is proposed which is modification of EDNSS algorithm. The output, error, and weight equation of conventional NLMS algorithms are shown in:

The output; $y(n) = w(n).xT(n)$ (1)

The error; $e(n) = d(n) - y(n)$ (2)

The weight (3):

$$w\ (n+1)\ = w\ (n)\ +\ \frac{\mu}{\acute{\epsilon}+||x(n)||^2}\ x(n)\ e(n) \tag{3}$$

where ‘w’ is the adaptive filter vector weight, ‘μ’ is step size and $||x(n)||^2$ is equal to $x(n) * xT(n)$, const ($\acute{\epsilon}$) is used in the denominator to prevent the division by a very small number. A modified form on NLMS algorithm introduced in called as EDNSS algorithm [19], [21], [22] and the weight update (4) and (5) is:

$$w(n+1) = w(n) + \frac{\mu}{\alpha\left||eL(n)|\right|^2+(1-\alpha)||x(n)||^2}x(n)e(n) \tag{4}$$

where $||eL(n)||2 = \sum_{i=o}^{L-1}|e(n-i)|^2$ (5)

The parameter ‘L’ represents fixed number of samples selected on the basis to get optimum response. The proposed algorithm may be considered as modified EDNSS algorithm in which adding error vector for ‘*n’* samples along with fixed samples ‘*L’* in denominator then (4) becomes:

$$w(n+1) = w(n) + \frac{\mu}{\alpha\left||eL(n)|\right|^2+(1+\alpha)\left||e(n)|\right|^2+1-\alpha)||x(n)||^2}x(n)e(n) \tag{6}$$

Where $||e\ (n)||2 = \sum_{i=o}^{n-1}|e(n-i)|^2$ (7)

The parameter ‘*n* ‘represents the total number of speech samples. The performance of the adaptive noise canceller may be described in terms of output signal to noise ratio (SNR), the excessive mean square error (EMSE) and misadjustment ‘M’ [22], [23]. The EMSE at the n^{th} iteration is defined by:

$$EMSE\ (n)\ = \frac{1}{L}\sum_{j=0}^{L-1}|e1(n-J)|^2 \tag{8}$$

where $e1(n) = e(n) - s(n)$ is the excess residual error. The steady state EMSEss [19] is defined by:

$$EMSE\ ss = \frac{1}{K-P}\sum_{n=P}^{K-1}EMSE(n) \tag{9}$$

where 'K' is the total number of samples of the speech signal in SP23 and 'P' is the number of samples after which the algorithm reaches steady state. The misadjustment as shown in:

$$M = \frac{EMSEss}{MSEmin} \tag{10}$$

where $MSE\ min = \frac{1}{K-P}\sum_{n=P}^{K-1}|s(n)|^2$ (11)

$$SNR\ (0), = \frac{Mean(Signal)^2}{Mean(Signal-Noise\ Reduced\ Signal)^2} \tag{12}$$

Interms of decibels

$$SNR\ (0), dB\ =\ 10log10\frac{Mean(Signal)^2}{Mean(Signal-Noise\ Reduced\ Signal)^2} \tag{13}$$

3. RESULTS

The experimentation is carried for speech signal "SP23" from Noizeus data base of male voice saying "stop whistling and watch the boys march". The original signal has 21,209 samples. The simulations are carried with K=21,209 for speech signal, N=10 (length of the filter), L=200 P=1, α=0.7, and μ=0.1.

The performance of the NLMS, EDNSS, and proposed method are tested with babble noise, exhibition noise, restaurant noise and airport noise at different level of SNR. Figure 3 illustrates the effect of the different noise level added to original speech signal. Figure 4 shows the primary signal, reference signal, output and the EMSE for NLMS with original speech corrupted with babble noise of '0' dB. Figure 5 shows the same for EDNSS algorithm and Figure 6 shows for proposed method. The proposed method gives better output SNR for all types of nonstationary noises at different level of input SNR compare to NLMS and EDNSS algorithms. This is clearly shown in Table 1. Figures 7–9 shows plot of EMSE in decibels for NLMS, EDNSS, and proposed algorithm (modified EDNSS algorithm). Compare to NLMS and EDNSS algorithms steady state EMSE and misadjustement decreases in proposed (modified EDNSS) algorithm. Table 2 demonstrates the steady state EMSEss in dB and misadjustement (M) for different algorithms.

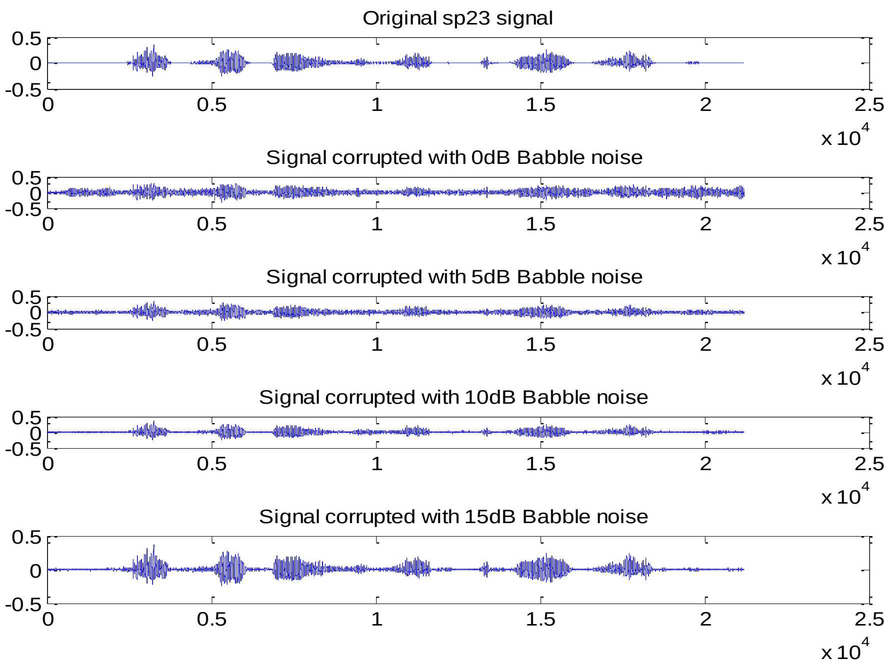

Figure 3. From top to bottom: original speech signal, babble noise corrupted speech at '0' dB, '5' dB, '10' dB, and '15' dB

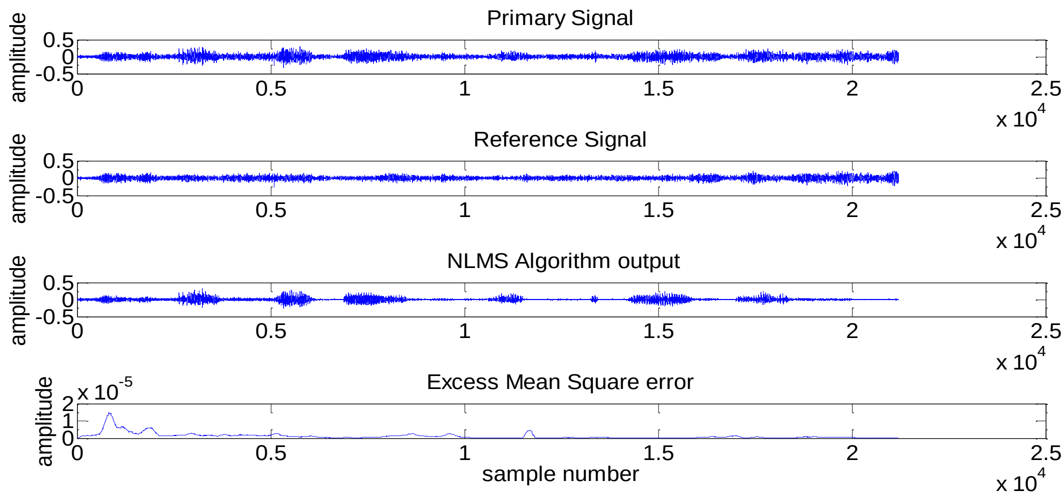

Figure 4. From top to bottom, primary signal, reference signal, output, and EMSE of NLMS algorithm

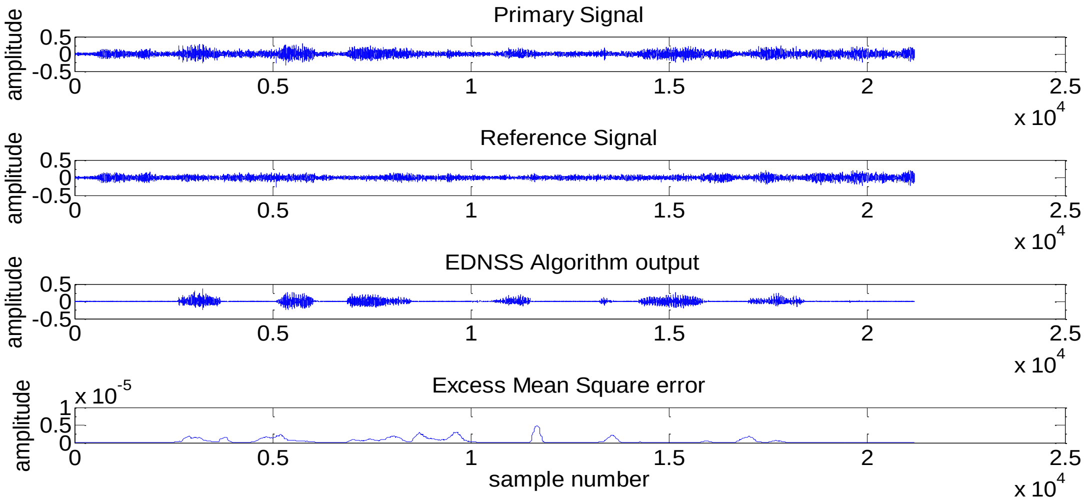

Figure 5. From top to bottom, primary signal, reference signal, output, and EMSE of EDNSS algorithm

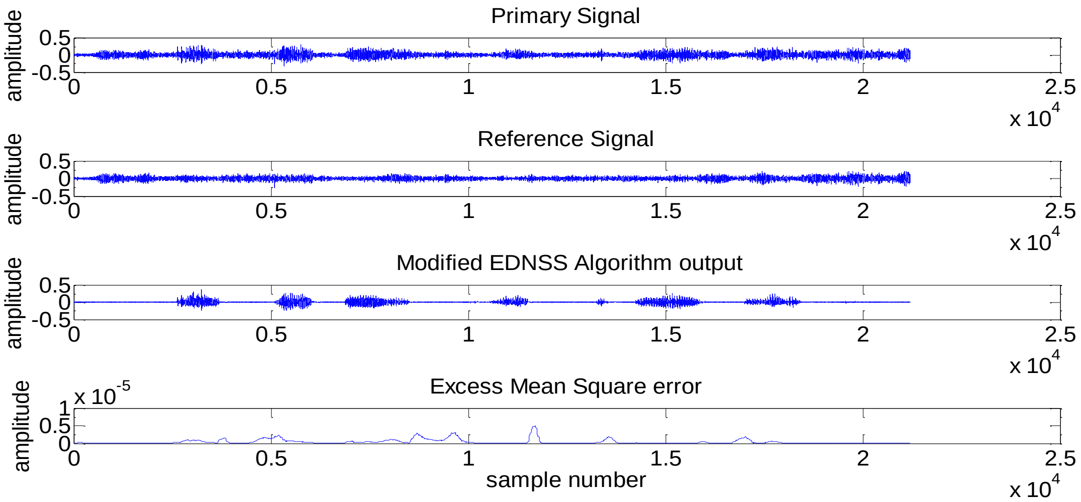

Figure 6. From top to bottom, primary signal, reference signal, output, and EMSE of modified EDNSS algorithm

Table 1. Performance of various algorithms for different types of noises at different input SNR

Type of noise	Input file name	Input SNR in dB	Output SNR in dB		
			NLMS	EDNSS	Proposed algorithm
Babble noise	sp23_babble_sn0	0	8.16	9.7	11.93
	sp23_babble_sn5	5	9.47	11.67	13.86
	sp23_babble_sn10	10	11.66	14.31	15.1
	sp23_babble_sn15	15	15.62	17.83	19.15
Restaurant noise	sp23_resturant_sn0	0	6.44	7.72	8.32
	sp23_resturant_sn5	5	9.0	11.75	12.53
	sp23_resturant_sn10	10	10.57	12.31	12.77
	sp23_resturant_sn15	15	14.98	17.41	18.7
Airport noise	sp23_airport_sn0	0	5.69	7.27	7.89
	sp23_airport_sn5	5	9.10	11.55	13.57
	sp23_airport_sn10	10	11.25	13.37	15.07
	sp23_airport_sn15	15	14.99	18.00	18.59
Exhibition noise	sp23_exhbition_sn0	0	4.07	4.84	5.72
	sp23_exhbition_sn5	5	8.64	12.28	14.4
	sp23_exhbition_sn10	10	11.65	16.41	17.52
	sp23_exhbition_sn15	15	14.84	18.36	19.21

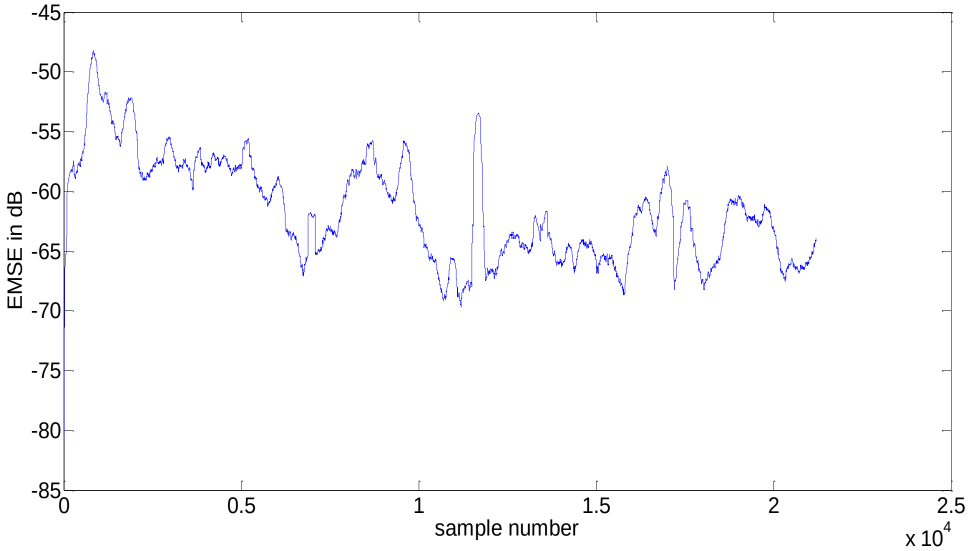

Figure 7. Excess mean square in decibels plot for NLMS algorithm

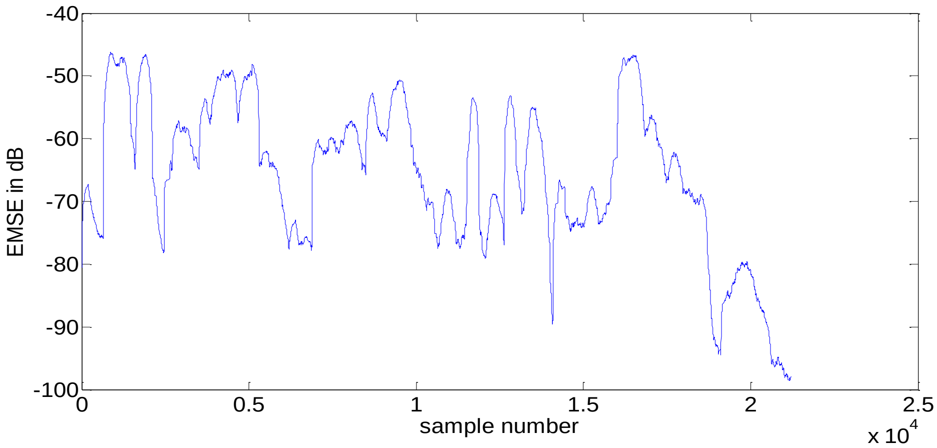

Figure 8. Excess mean square in decibels plot for EDNSS algorithm

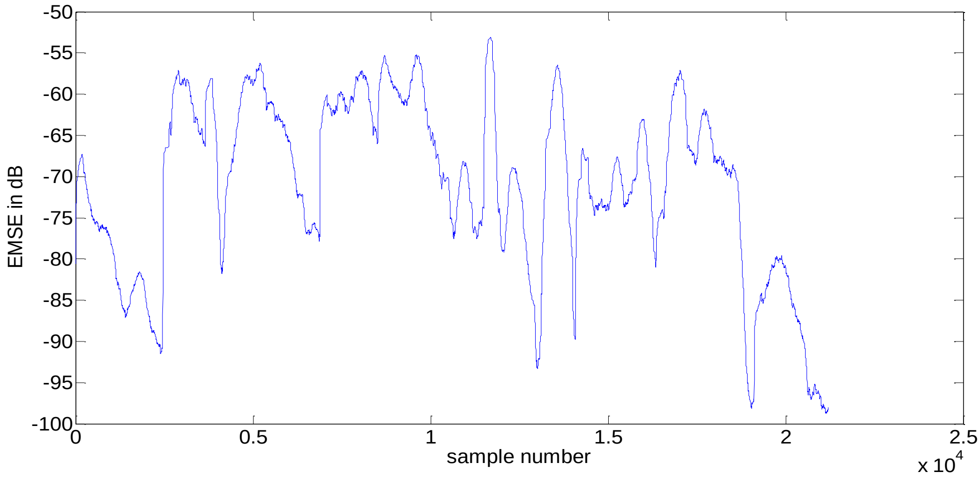

Figure 9. Excess mean square in decibels plot for modified EDNSS algorithm

Table 2. Comparison of EMSEss and misadjustment for different algorithms

Type of noise	Input file name	Input SNR in dB	NLMS M	NLMS EMSEss (dB)	EDNSS M	EDNSS EMSEss (dB)	Proposed algorithm M	Proposed algorithm EMSEss (dB)
Babble noise	sp23_babble_sn0	0	0.0761	-59.15	0.0536	-60.67	0.0254	-63.92
	sp23_babble_sn5	5	0.0564	-60.45	0.0340	-62.65	0.0205	-64.84
	sp23_babble_sn10	10	0.0341	-62.64	0.0185	-65.29	0.0154	-66.07
	sp23_babble_sn15	15	0.0137	-66.61	0.0082	-68.80	0.0011	-70.52
Restaurant noise	sp23_resturant_sn0	0	0.1133	-57.42	0.0845	-58.69	0.0736	-59.29
	sp23_resturant_sn5	5	0.0628	-59.98	0.0334	-62.72	0.0279	-63.50
	sp23_resturant_sn10	10	0.0436	-61.57	0.0283	-63.29	0.0284	-63.74
	sp23_resturant_sn15	15	0.0158	-65.94	0.0091	-68.38	0.0067	-69.67
Airport noise	sp23_airport_sn0	0	0.1348	-56.67	0.0937	-58.24	0.0813	-58.87
	sp23_airport_sn5	5	0.0614	-60.08	0.0350	-62.52	0.0221	-64.52
	sp23_airport_sn10	10	0.0371	-62.23	0.0300	-64.34	0.0155	-66.04
	sp23_airport_sn15	15	0.0158	-65.97	0.0079	-68.98	0.0069	-69.56
Exhibition noise	sp23_exhbition_sn0	0	0.0196	-55.04	0.1638	-55.81	0.1337	-56.70
	sp23_exhbition_sn5	5	0.0681	-59.62	0.0295	-63.26	0.0181	-65.38
	sp23_exhbition_sn10	10	0.0365	-62.34	0.0114	-67.38	0.0088	-68.49
	sp23_exhbition_sn15	15	0.0163	-65.83	0.0073	-69.34	0.0060	-70.18

4. CONCLUSION

In this paper single line noise cancellation system is proposed by generation of secondary reference signal and modified EDNSS algorithm. The results are demonstrated for different types of noises at different level of input SNR. The proposed algorithm improves output SNR, reduces misadjustment and steady state minimum mean square error especially in low SNR environment. The algorithm also increases speed of convergence. So, the proposed methodology can be used in speech enhancement applications like automatic noise canceller, echo removal, and noise cancellation in digital hearing aids when the information of secondary reference signal is unavailable.

REFERENCES

[1] B. Widrow, J. M. McCool, M. G. Larimore, and C. R. Johnson, "Stationary and nonstationary learning characteristics of the LMS adaptive filter," *Proceedings of the IEEE*, vol. 64, no. 8, pp. 1151–1162, 1976, doi: 10.1109/PROC.1976.10286.

[2] S. Li, S. Wu, Y. Wang, W. Guo, and Y. Zhou, "An improved NLMS algorithm based on speech enhancement," in *2015 IEEE Advanced Information Technology, Electronic and Automation Control Conference (IAEAC)*, Dec. 2015, pp. 896–899, doi: 10.1109/IAEAC.2015.7428686.

[3] S. Boll, "Suppression of acoustic noise in speech using spectral subtraction," *IEEE Transactions on Acoustics, Speech, and Signal Processing*, vol. 27, no. 2, pp. 113–120, Apr. 1979, doi: 10.1109/TASSP.1979.1163209.

[4] J. Kim and A. D. Poularikas, "Performance analysis of the adjusted step size nlms algorithm," in *Thirty-Sixth Southeastern Symposium on System Theory, 2004. Proceedings of the*, 2004, pp. 467–471, doi: 10.1109/SSST.2004.1295701.

[5] M. Shengqian, X. Guowei, M. Zhifeng, W. Shuping, and F. Manhong, "Research on adaptive noise canceller of an improvement LMS algorithm," in *2011 International Conference on Electronics, Communications and Control (ICECC)*, Sep. 2011, pp. 1611–1614, doi: 10.1109/ICECC.2011.6067870.

[6] S. S. Haykin, “Adaptive filter theory,” *Pearson Education India*, 2008.
[7] A. H. Sayed, *Fundamentals of adaptive filtering*. John Wiley & Sons, 2003.
[8] C. Paleologu, J. Benesty, S. L. Grant, and C. Osterwise, “Variable step-size NLMS algorithms designed for echo cancellation,” in *2009 Conference Record of the Forty-Third Asilomar Conference on Signals, Systems and Computers*, 2009, pp. 633–637, doi: 10.1109/ACSSC.2009.5469916.
[9] C. Panagiotakis and G. Tziritas, “A speech/music discriminator based on RMS and zero-crossings,” *IEEE Transactions on Multimedia*, vol. 7, no. 1, pp. 155–166, Feb. 2005, doi: 10.1109/TMM.2004.840604.
[10] A. R. Abu-el-quran, R. A. Goubran, and A. D. C. Chan, “Adaptive feature selection for speech/music classification,” in *2006 IEEE Workshop on Multimedia Signal Processing*, Oct. 2006, pp. 212–216, doi: 10.1109/MMSP.2006.285299.
[11] W.-H. Tsai and C.-H. Ma, “Speech and singing discrimination for audio data indexing,” in *2014 IEEE International Congress on Big Data*, Jun. 2014, pp. 276–280, doi: 10.1109/BigData.Congress.2014.138.
[12] C.-T. Lin, “Single-channel speech enhancement in variable noise-level environment,” *IEEE Transactions on Systems, Man, and Cybernetics-Part A: Systems and Humans*, vol. 33, no. 1, pp. 137–144, Jan. 2003, doi: 10.1109/TSMCA.2003.811115.
[13] P. C. Loizou and G. Kim, “Reasons why current speech-enhancement algorithms do not improve speech intelligibility and suggested solutions,” *IEEE Transactions on Audio, Speech, and Language Processing*, vol. 19, no. 1, pp. 47–56, Jan. 2011, doi: 10.1109/TASL.2010.2045180.
[14] S. A. Hadei and M. Lotfizad, “A family of adaptive filter algorithms in noise cancellation for speech enhancement,” *International Journal of Computer and Electrical Engineering*, pp. 307–315, 2010, doi: 10.7763/IJCEE.2010.V2.153.
[15] C. H. You, S. N. Koh, and S. Rahardja, “/spl beta/-order MMSE spectral amplitude estimation for speech enhancement,” *IEEE Transactions on Speech and Audio Processing*, vol. 13, no. 4, pp. 475–486, Jul. 2005, doi: 10.1109/TSA.2005.848883.
[16] R. .S, “System identification in speech signal using modified Kalman based normalized least mean square (NLMS) algorithm,” *International Journal of Advanced Research in Computer Science and Software Engineering*, vol. 7, no. 3, pp. 180–184, Mar. 2017, doi: 10.23956/ijarcsse/V7I3/0104.
[17] R. S and V. Udayashanakara, “Enhancement of speech signal using improved NLMS algorithm,” *Communication on Applied Electronics*, vol. 6, no. 9, pp. 34–37, 2017, doi: 10.5120/cae201765257.
[18] R. S and V. Udayashankara, “Estimation of noise in nonstationary signals using derivative of NLMS algorithm,” *International Journal of Image, Graphics and Signal Processing*, vol. 9, no. 8, pp. 9–16, Aug. 2017, doi: 10.5815/ijigsp.2017.08.02.
[19] Z. Ramadan and A. Poularikas, “Performance analysis of a new variable step-size lms algorithm with error nonlinearities,” in *Thirty-Sixth Southeastern Symposium on System Theory, 2004. Proceedings of the*, pp. 384–388, doi: 10.1109/SSST.2004.1295684.
[20] Z. Ramadan and A. Poularikas, “A robust variable step-size LMS algorithm using error-data normalization,” in *Proceedings. IEEE SoutheastCon, 2005.*, pp. 219–224, doi: 10.1109/SECON.2005.1423249.
[21] Z. Ramadan and A. Poularikas, “A variable step-size adaptive noise canceller using signal to noise ratio as the controlling factor,” in *Thirty-Sixth Southeastern Symposium on System Theory, 2004. Proceedings of the*, pp. 456–461, doi: 10.1109/SSST.2004.1295699.
[22] Z. Ramadan and A. Poularikas, “An adaptive noise canceller using error nonlinearities in the LMS adaptation,” in *IEEE SoutheastCon, 2004. Proceedings.*, pp. 359–364, doi: 10.1109/SECON.2004.1287943.
[23] A. D. Poularikas and Z. M. Ramadan, *Adaptive filtering primer with MATLAB®*. CRC Press, 2017.

BIOGRAPHIES OF AUTHORS

Rathnakara Srinivasa Pandit completed his Bachelor of Engineering in Instrumentation Technology and Master of Technology in Biomedical instrumentation from Sri Jayachamarajendra College of Engineering affiliated to University of Mysore, Karnataka, India. He obtained his Ph.D. in the area of speech progressing from University of Mysore. His area of interest includes speech and image processing. He has to his credit 30 conference and journal papers both at national and international level and guided more than 25 M.Tech projects. Currently he is working as Assistant professor in Department of Electronics and Instrumentation. Sri Jayachamarajendra College of Engineering, JSS Science, and Technology University, Mysore, Karnataka, India. He can be contacted at email: rathnakara_s@sjce.ac.in.

Udayashankara Veerappa completed his Bachelor of Engineering in Electronics and Communication from Sri Jayachamarajendra College of Engineering, Mysore Karnataka and obtained his M.E & Ph.D. degree from Indian Institute of Science (IISc) Bangalore. Currently he is working as professor in Department of Electronics and Instrumentation, at Sri Jayachamarajendra College of Engineering, Mysore, India. His research interests include rehabilitation engineering, digital signal processing, speech recognition, speech enhancement, and EEG analysis. He has authored more than 100 publications in National and International Journals and Conferences in these areas. He has authored three books, 8051 microcontrollers: hardware, software and applications, McGraw Hill-2009, real time digital signal processing, PHI-2010, modern digital signal processing, PHI-2012. He can be contacted at email: v_udayashanakara@sjce.ac.in.