Adaptive IIR Notch Filter with Modified Algorithm for Echo Cancellation

S. Sukhumalchayaphong and C. Benjangkaprasert
Faculty of Engineering, King Mongkut's Institute of Technology Ladkrabang, Bangkok, Thailand 10520
Email: sethawuit@yahoo.com, kbchawal@kmitl.ac.th

***Abstract*—In this paper an adaptive echo canceller is presented, based on adaptive IIR notch filter with a new algorithm, for the echo cancellation in telephone network. Where the new algorithm is employs the modified sign-regressor algorithm and works with the autocorrelation of the error signal to estimate the parameter, resulting in obtaining good estimate of parameter for the echo canceller. The simulation results show that the performance of the proposed canceller outperforms the canceller with the classical LMS, SR and RLS algorithms in terms of Echo Return Loss Enhancement (ERLE).**

***Index Terms*—echo canceller, adaptive algorithm, IIR notch filter**

I. INTRODUCTION

The applications of adaptive IIR notch filters are widely used for communication systems, radar, sonar, biomedical engineering and so on [1]-[7], because of its simplicity. The one challenging application is an echo canceller [4], [5], [8]-[17] in the long distance telephone network. Echo return loss and impulse response are important characteristics of an echo path for the design of echo cancellers. The performance of echo cancellers depends on the choice of the adaptive filtering algorithm. From the literature survey, it is found that the Least Mean Square (LMS) adaptive algorithm and the Sign-Regressor (SR) adaptive algorithm [1], [2] are the most commonly used technique for real time applications, due to its simplicity and numerical robustness. However, it suffers from the slow convergence since the echo path is usually very long and the speech signals are non-stationary and highly correlated and a large number of filter coefficients is needed. In order to improve the convergence rate and the performance of the system, the Recursive Least Square (RLS) algorithm is the best choice. However, the computational complexity of this algorithm also increases. Therefore, in this paper presented a new algorithm based on adaptive lattice form IIR notch filter that gives less computational complexity than the RLS algorithm for the echo canceller.

The paper is organized as follows. Section II, the structure of the echo canceller in the telephone network is presented. In Section III the adaptive lattice form structure IIR notch filter for adaptive echo canceller is presented. Section IV presents the attractive algorithms and the new algorithm. Section V describes the performance evaluate in the form of Echo Return Loss Enhancement (ERLE). Computer simulations for several situations to demonstrate the performance of the system and conclusions are given in Sections VI and VII, respectively.

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II. ECHO CANCELLER SCHEME

In the telecommunication system, a major source of impairment to speech quality is network echoes. The echo signal in the telephone network can be depicted in Fig. 1.

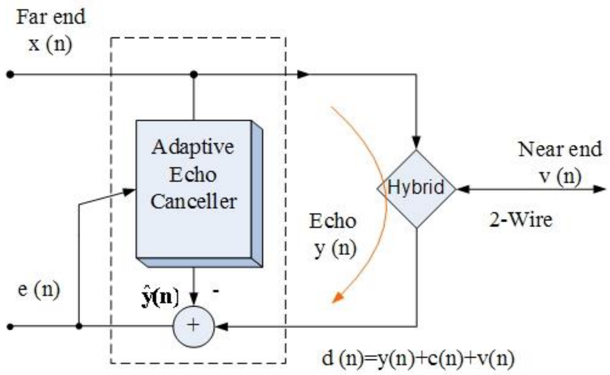

Figure 1. Adaptive echo cancellation in telephone network.

Consider the adaptive echo canceller described in Fig. 1, where $x(n)$ and $v(n)$ represent the far-end signal and near-end speech signal, respectively. Also $d(n)$, $c(n)$ and $\hat{y}(n)$ denote the received input signal to the echo canceller, background noise, and the estimated echo signal, respectively. That can be described as follows [4]:

$$\hat{y}(n) = W^T(n)X(n) \tag{1}$$

$$e(n) = d(n) - \hat{y}(n) \tag{2}$$

$$d(n) = v(n) + y(n) + c(n) \tag{3}$$

where $W^T(n)$ is the filter coefficient vector of the echo canceller, $X(n)$ is the input vector and $e(n)$ is the error signal.

We assume the linearity of a finite duration *(n)* of the echo impulse response path with length of *n* samples containing the noise $c(n)$ are input $x(n)$ and the

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estimate echo are $\hat{y}(n)$ all can be expressed as follows [4]:

$$Echo \quad path = \sum_{i=0}^{n} R_i \exp[-\frac{(i+1)}{880}]\delta(n-i) \qquad (4)$$

where n is the numbers of samples and R_i is a random number within –2, +2 and $\delta(n)$ is the Dirac function, respectively.

III. Adaptive IIR Notch Filter

The transfer function of adaptive echo canceller in the second-order adaptive IIR notch filter lattice form structure is given as follows [3]:

$$H(z) = \frac{1 + 2k_0(n)z^{-1} + z^{-2}}{1 + k_0(n)(1+\rho)z^{-1} + \rho z^{-2}} \qquad (5)$$

The output signal of Eq. (5) can be expressed in time domain as follows:

$$\hat{y}(n) = u(n) + 2\hat{k}_0(n)u(n-1) + u(n-2) \qquad (6)$$

$$u(n) = x(n) - \hat{k}_0(n)(1+\rho)u(n-1) - \rho u(n-2) \qquad (7)$$

where $\hat{k}_0(n)$ is filter coefficient vector of adaptive lattice form structure filter at time n, and ρ is the pole constrain factor close to but less than unity to ensure the stability of the filter, respectively. The larger parameter ρ , the narrower notch bandwidth is obtained.

IV. Adaptive Algorithms

In this section we will describe a family of the attractive algorithms and the new one.

A. *Sign-Regressor Algorithm*

The Sign-Regressor (SR) algorithm [1] is obtained from the conventional LMS algorithm by replacing the tap–input $x(n)$ with the signum function of $x(n)$ or sign [$x(n)$], where the sign function is applied to the input $x(n)$ on an element-by-element basis. The SR algorithm for updating $k_0(n)$ is summarized as follows:

$$k_0(n+1) = k_0(n) + 2\mu sign(x(n))e(n) \qquad (8)$$

where $x(n)$ is the input signal, The step size parameter μ is a small positive constant, sign is the Signum function and $e(n)$ is the estimation error signal, respectively.

B. *Recursive Least Square Algorithm*

The Recursive Least Square (RLS) algorithm [1], [2] solves the slow convergence speed of the SR algorithm by using the time average instead of the statistical average. The summary of RLS algorithm is described as follows:

$$\hat{d}(n) = w^T(n-1)x(n) \qquad (9)$$

$$k(n) = \frac{R^{-1}(n-1)x(n)}{\lambda + r^T(n)R^{-1}(n-1)x(n)} \qquad (10)$$

$$R^{-1}(n) = \frac{1}{\lambda}[R^{-1}(n-1) - k(n)x^T(n)R^{-1}(n-1)] \qquad (11)$$

$$k_0(n) = k_0(n-1) + k(n)e^*(n) \qquad (12)$$

where $e^*(n)$ is the complex conjugate of $e(n)$. An initialize as

$$k_0(0) = k(0) = \hat{d}(0) = 0 \qquad (13)$$

$$R^{-1}(0) = \delta I \qquad (14)$$

where I is identity matrix and δ is a real scalar less than but close to unity.

C. *A New Algorithm*

In this subsection, we will describes the Modified Sign-regressor Algorithm (MSA). The proposed technique is using the combination of the Signum function to the input of the filter $x(n)$ and the autocorrelation function of the estimation of the error signal to control the update function of the algorithm. Therefore, the weight update equation can be expressed as follows:

$$k_0(n) = k_0(n-1) + \mu sign\big(x(n)\big)e(n)(\frac{1-A(n)}{1+A(n)}) \qquad (15)$$

where $A(n)$ is given by

$$A(n) = 1 - \mu e(n-1) \qquad (16)$$

From (15) and (16), finally the proposed algorithm for the new echo canceller may be expressed as

$$k_0(n) = k_0(n-1) + \frac{\mu^2 sign\big(x(n)\big)e(n)e(n-1)}{2 - \mu e(n-1)} \qquad (17)$$

The step size parameter (μ) is a small positive constant, sign is Signum function, $x(n)$ is the input signal and $e(n)$ is the estimation of the error signal, respectively.

Table I shows the computational complexity per iteration of the proposed algorithm and the conventional SR, RLS algorithms.

TABLE I. The Complexity of the Algorithms

Algorithms	Multiplications	Additions	Sign
SR	2	2	1
RLS	7	4	-
New	6	3	1

V. Performance Evaluation

The most common measurement of adaptive echo canceller performance is echo return loss enhancement (ERLE) [1], which can be expressed as

$$ERLE(dB) = 10\log_{10}\frac{E[\hat{y}^2(n)]}{E[\hat{e}^2(n)]} \qquad (18)$$

The ERLE is simply the ratio of signal to the 'noise' that could not be cancelled from the signal. On the other hand, ERLE is the logarithm of the power ratio of original echo to echo residue. In this paper we evaluate the performance of echo canceller by ERLE simulation.

VI. SIMULATION RESULTS

In this section, the performance of the proposed adaptive echo canceller is compared with the LMS, SR and RLS adaptive algorithms applied to an echo cancellation in the telephone network. The main parameters for the cancellers are given as follows: $\hat{k}_0(0) = 0$, $\mu = 0.0001$, $\rho = 0.99$, data range N = 1,000 and with 100 independent runs.

Fig. 2 shows the echo impulse response that generated by (11), the signal was repeated at time (n) equal to 502.

Fig. 3 shows an example of background noise for the echo canceller with SNR = 30 dB and then changed to 20 dB at time (n) = 502. From Fig. 3, it is demonstrated that the background noise is inversely proportional to the SNR.

Fig. 4 shows the mean Echo Return Loss Enhancement (ERLE) comparison with the new algorithm and the conventional algorithms such as LMS, SR, and RLS algorithms where SNR ranging between 0 dB to 70 dB. Form the result in Fig. 4, it is evident that the new echo canceller gives the best mean ERLE.

Fig. 5 shows the average echo residue power comparison with the new algorithm and the conventional algorithms such as LMS, SR, and RLS algorithms in various SNR ranging between 0 dB to 70 dB. From Fig. 5, it is seen that the new algorithm for the echo canceller provides the lower error than the previous one.

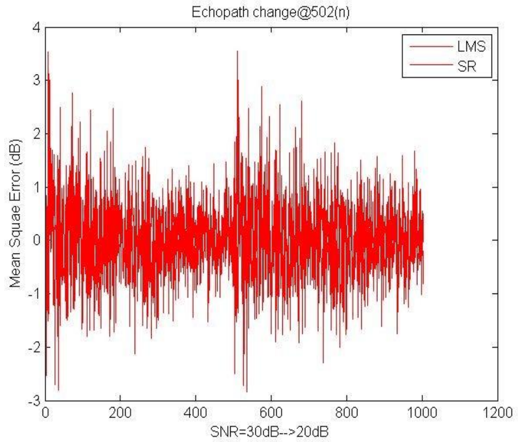

Figure 2. Input signal.

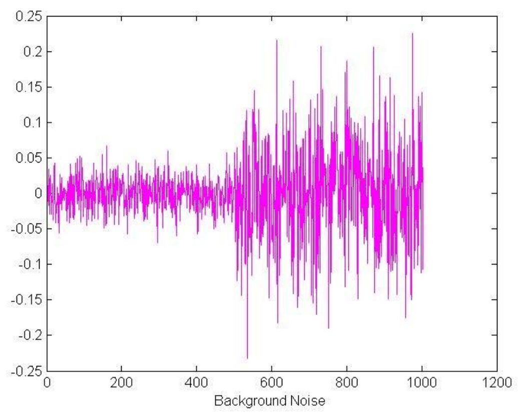

Figure 3. Background noise.

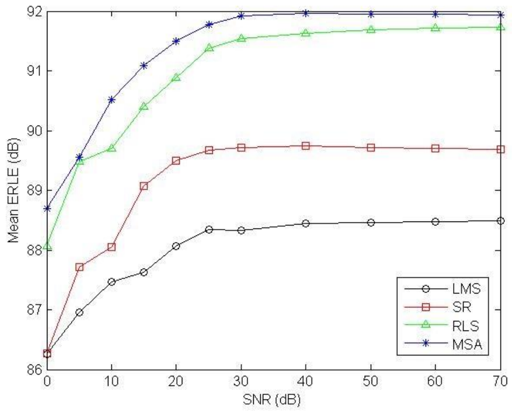

Figure 4. Mean ERLE.

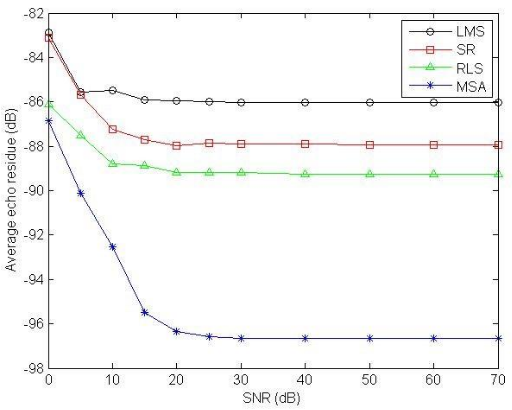

Figure 5. Average echo residue.

VII. CONCLUSIONS

In this paper, an adaptive echo canceller based on IIR notch filter with a new algorithm for echo cancellation in telephone network is proposed. The new algorithm employs the signum function to the conventional sign-regressor algorithm and autocorrelation function of the estimation of the error signal. As a result, it can be reduced the computational complexity of the adaptive algorithm when compared to the conventional RLS algorithm and the new echo canceller provides a good stability.

REFERENCES

[1] B. Farhang-Boroujeny, *Adaptive Filter: Theory and Applications*, Chichester: John Wiley, 1998.

[2] S. Haykin, *Adaptive Filter Theory*, 4th edition, Pearson, 2002.

[3] N. I. Cho, “On the adaptive lattice notch filter for the detection of sinusoidal,” *IEEE Trans. Circuits System*, vol. 40, no. 7, pp. 405-416, July 1993.

[4] C. Benjangkaprasert, S. Sukhumalwong, S. Teerasakworakun, and K. Janchitrapongvej, “The new variable step-size algorithm adaptive lattice structure for echo cancellation,” in *Proc. ICCAS*, Oct. 2003, pp. 2090-2092.

[5] J. H. Yoo, S. H. Cho, and D. H. Youn, “A Lattice/Transver-sal Joint (LTJ) structure for an acoustic echo canceller,” in *Proc. IEEE International Symposium on Circuits and Systems*, Apr.-May 1995, pp. 1090–1093.

[6] J. F. Chicharo and T. S. Ng, “Gradient-based adaptive IIR notch filtering for frequency estimation,” *IEEE Trans. Acoustic, Speech, Signal Processing*, vol. ASSP-38, pp. 769-777, May 1990.

[7] S. C. Pei and C. C. Tseng, "Adaptive IIR notch filter based on least mean *p*-power error criterion," *IEEE Trans. Circuit System-II*, vol. 40, pp. 525-529, Aug. 1993.
[8] T. Aboulnasr and K. Mayyas, "A robust variable step-size LMS-type algorithm analysis and simulation," *IEEE Trans. Signal Processing*, vol. 45, no. 3, pp. 631-639, March 1997.
[9] M. M. Sondhi, "The history of echo cancellation," *IEEE Signal Processing Magazine*, vol. 23, no. 5, pp. 95-98, Sep. 2006.
[10] C. Paleologu, "An efficient proportionate affine projection algorithm for echo cancellation," *IEEE Trans. Processing Letters*, vol. 17, no. 2, pp. 165-168, Feb. 2010.
[11] S. K. Roy and C. F. Rodriques, "Echo canceller using error back propagation algorithm," in *Proc. International Conference on Soft Computing and Machine Intelligence*, Sep. 2014, pp. 98-101.
[12] A. Kar and M. Chandra, "Dynamic tap-length estimation based low complexity acoustic echo canceller," in *Proc. International Conference on Emerging Trends in Science, Engineering and Technology*, Dec. 2012, pp. 339-343.
[13] Z. Yang, Y. R. Zheng, and S. L. Grant, "Proportionate affine projection sign algorithms for network echo cancellation," *IEEE Trans. on Audio, Speech, and Language Processing*, vol. 19, no. 8, pp. 2273-2284, Nov. 2011.
[14] P. P. Kadam, Z. Saquid, and A. Lahane, "Adaptive echo cancellation in VoIP network," in *Proc. IEEE International Conference on Engineering and Technology*, Mar. 2016, pp. 295-299.
[15] M. Z. Ikram, "Double-talk detection in acoustic echo cancellers using zero-crossings rate," in *Proc. IEEE International Conference on Acoustics, Speech and Signal Processing*, Apr. 2015, pp. 1121-1125.
[16] F. Zabini, M. Mazzotti, D. Dardari, G. Chiurco, and O. Andrisano, "Performance and stability analysis of echo cancellers based on training sequences," *IEEE Trans. on Broadcasting*, vol. 60, no. 3, pp. 437-451, Sept. 2014.
[17] ITU-T Recommendation G.168, *Digital Network Echo Canceller*, 2012.

Sethaphak Sukhumalchayaphong received the B.Ind.Tech. and M.Eng. degrees from King Mongkut's Institute of Technology Ladkrabang, Bangkok, Thailand in 1995 and 2007, respectively. He is currently pursuing the Ph.D. degree with the Department of Computer Engineering, Faculty Engineering, King Mongkut's Institute of Technology Ladkrabang. His research interests include wireless communication and signal processing.

Chawalit Benjangkaprasert received his B. Ind. Tech. and M.Eng. degrees from King Mongkut's Institute of Technology Ladkrabang, Bangkok, Thailand in 1987 and 1990, respectively. In 2006, he received Doctor degree in engineering from Tokai University, Japan. Since 1987 he has been with King Mongkut's Institute of Technology Ladkrabang, where he is now an associate professor in Faculty of Engineering. His current research interests are in filter and antenna designs, wireless communications and adaptive signal processing.