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ADAPTIVE ECHO CANCELLER USING LMS ALGORITHM

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Abstract-*An echo canceller is presented, using an adaptive filter with a modified LMS (Least Mean Square) algorithm, where this modification is achieved coding error on conventional LMS algorithm. Simulation results, show a better convergence speed than conventional LMS algorithm, furthermore, Coded Error algorithm presents less complexity for digital adaptive filters design, due to reduction of floating point operations.*

Keywords: *LMS, Echo Canceller, Adaptive Filter.*

I. INTRODUCTION

Communications systems development increases considerably obtaining more troubles as additive noise, signal interference and echo, therefore errors in data transmission are generated, but adaptive filter is an option to reduce these channel effects. Adaptive filters are systems with four terminals as showed in Fig 1, where *x* is the input signal, *d* is the wished signal, *y* is output signal filter and *e* is the output filter error.

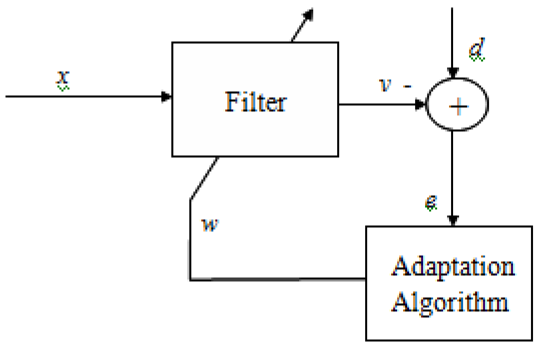

Fig. 1. General scheme for an adaptive filter.

Adaptive filters design technique may be digital, analog or mixed. Every technique presents advantages and disadvantages, for example, analog adaptive filters are very fast, but offset avoids to get the least error [1, 2]. Digital filters are slow but precise, because is necessary the use of a lot of components, due to floating point operations [3, 4]. Mixed design (analog and digital), offers a good compromise between precision and speed, but VLSI (Very Large Scale Integration) design is more complicated [5-7], because is necessary to separate analog and digital components inside the chip.

In this paper we present an echo canceller employing a modified LMS (Least Mean square) adaptive algorithm. The objectives are digital design reduction of an adaptive filter, making use of a low complexity algorithm and to achieve improvement in convergence speed. The proposed algorithm was probed using an echo canceller comparing results with LMS algorithm.

II. ADAPTIVE LMS ALGORITHM

Actually there are different adaptive algorithms like RLS (Recursive Least Square) or LMS .These algorithms works on time domain [3] and also exist frequency domain algorithms [9, 10]. The time domain algorithm often used is LMS, because its computational complexity lets an easy implementation on a chip.

The LMS algorithm is based on gradient search error, the mathematical expression is in,

$$w(T+1) = w(T)\ x(T)\ e(T) \qquad (1)$$

where *w(T+1)* are the next filter weights, *w(T)* are the current weights, *x(T)* is the sampled input signal, *e(T)* is the filter error and the convergence factor, which must satisfy :

$$1/N\, x^2(T) \qquad (2)$$

Where *N* is the filter taps number and *x2(T)* is the mean power estimated using some input samples. If convergence factor is high, thus speed convergence is fast but least error can't be gotten. Making use of a low convergence factor, the algorithm reaches to least error but on the other hand speed convergence is slow.

Some modifications has been done over LMS algorithm [11, 12], where sign input signal or sign error or both are considered, resulting an algorithm with less computational complexity. The main disadvantage on these modifications, is convergence speed reduction, because algorithm only employs signals polarity, and magnitude is completely unknown for algorithm. For this reason, we propose a modification on the LMS coding the error, thus the CE-LMS (Coded Error - LMS) is in [13, 14].

$$w(T+1) = w(T)x(T)\ C\,[e(T)] \qquad (3)$$

Where the coded error $C[e(T)]$ is obtained using :

$$C[e(T)] = round\{\underline{e(T)/\mathrm{Res}}\} \qquad (4)$$

where *round* is the result rounding and *Res* is coder resolution obtained with

$$\mathrm{Res} = Error_{max}/(2^n-1) \qquad (5)$$

where n is the coding bit number and *Errormax* is the probably maximum error, which can be obtained from peak of input signal.The coded error resulting is an integer number, thus floating point operations is lower and therefore digital design is less complicated. Furthermore, speed convergence is greater in CE-LMS compared with LMS, because error used in algorithm is greater than real, increasing adaptation steps.

III. SIMULATION RESULTS OF CE-LMS APPLIED ON A ECHO CANCELLER.

Echo is a feedback of delayed and attenuated data transmitted, which causes interference between transmitted data and the received data. There are two class of echoes:

1) *The acoustic echo:* produced by sound rebound, as produced on free-hands cell phones.

2) *Electrical echo:* Caused by impedance mismatch in hybrid and transmission channel. This echo is depicted in Fig. 2.

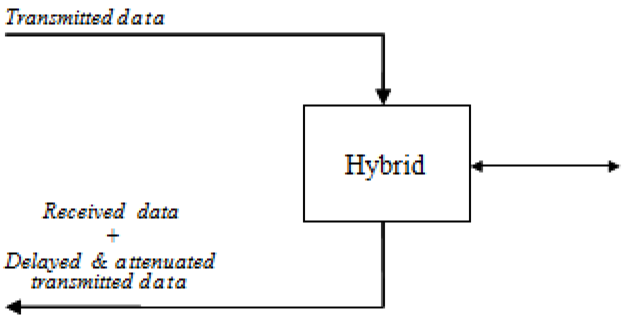

Fig. 2. Electrical echo generated in full duplex communications.

Effects caused by echo, can be reduced by an echo canceller (EC). An echo canceller (EC) generates a delayed and attenuated input signal for subtracting from received data as showed in Fig. 3

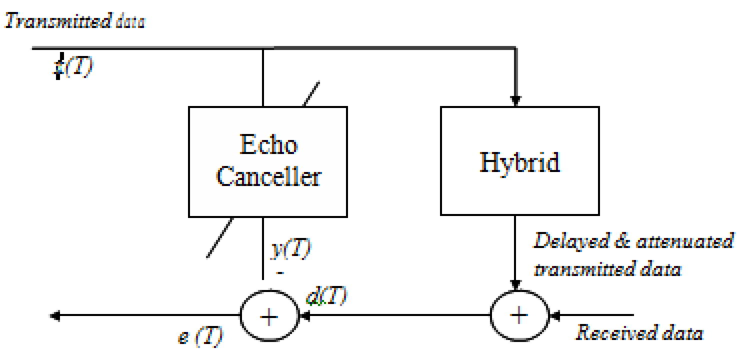

Fig. 3. Echo cancelling using adaptive filters.

Fig. 3, depicts an adaptive filter used on echo cancelling, where input signal is transmitted data, the sum of received and transmitted data is the desired signal, and the error is received data without echo. The results showed

above, were obtained using Matlab TM. These results shows speed convergence, output spectrum of EC-LMS and LMS algorithms, for a response comparison. Filter structure employed on EC, is a transversal filter also known as FIR filter (Finite Impulse Response). Furthermore convergence factor used in both algorithms is the same, for CE-LMS *Errormax* is considered equal to ninety percent of maximum input signal and an error coded with eight bits. An important notice is that transmitted data used is a sum of different tones and received data is a voice signal.
Mean Square Error (MSE) is used to observe convergence speed of adaptive filters. In Fig. 4, the MSE obtained for CE-LMS is faster than LMS, because EC-LMS needs less than one hundred iterations to get minimum error, while LMS requires a lot of iterations to find its minimum error. Speed convergence is not sufficient to measure adaptive filter quality is necessary to make a comparison between output and input signal power spectrums. With these parameters system effectiveness can be observed. In Fig 5, shows power spectrum for transmitted data, received data and error signal for LMS algorithm. Fig 5.shows that error spectrum contains transmitted data with power over zero decibels.

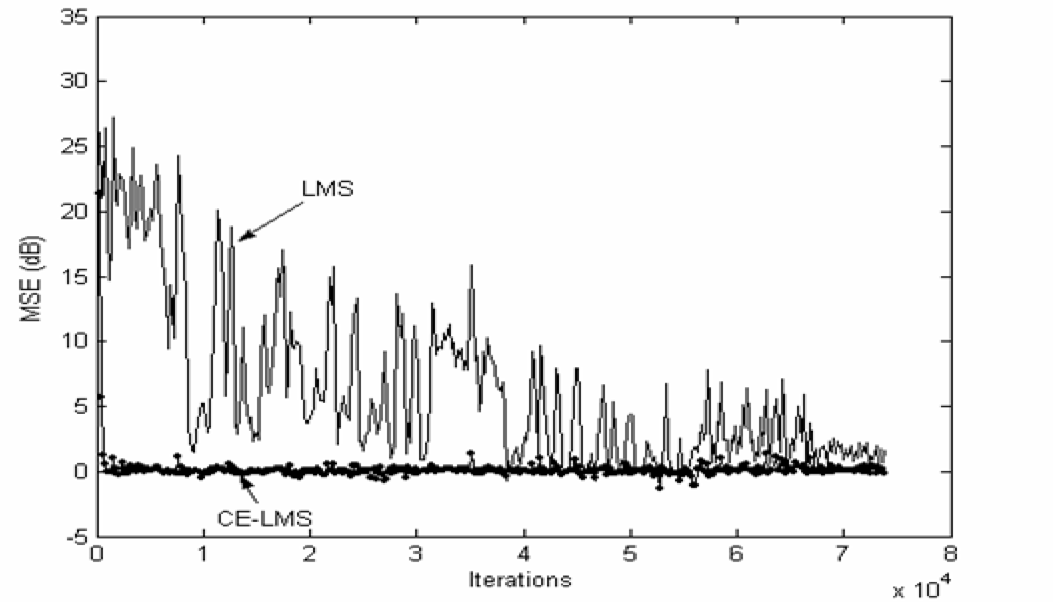

Fig. 4. Speed convergence for CE-LMS and LMS on a Echo Canceller.

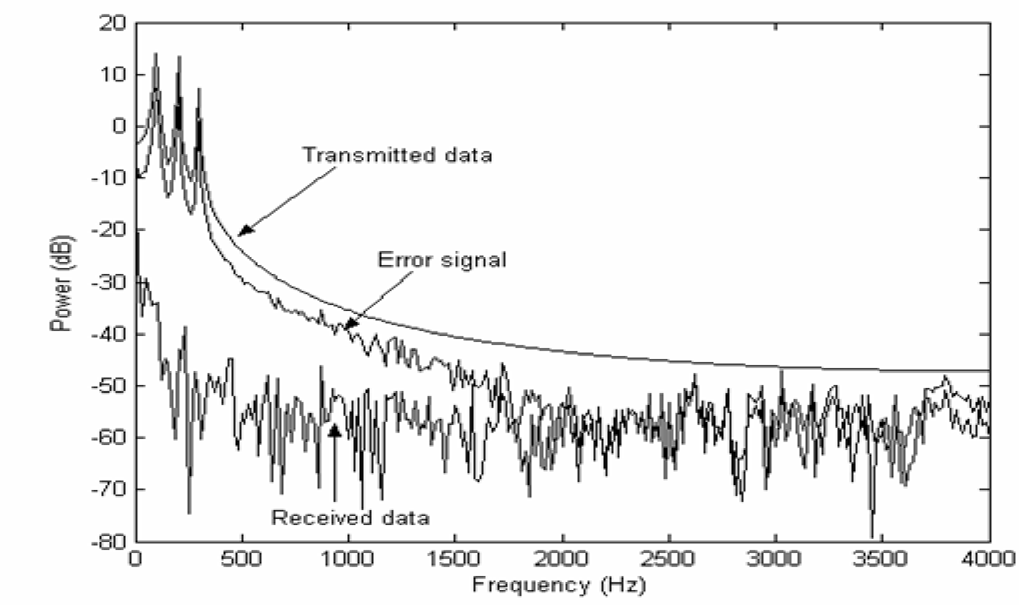

Fig. 5. Power spectrum for LMS algorithm.

In Fig. 6 the spectrum for CE-LMS algorithm is showed, where error signal spectrum is almost equal to received data. Thus, CE-LMS offers high speed convergence and a least error.

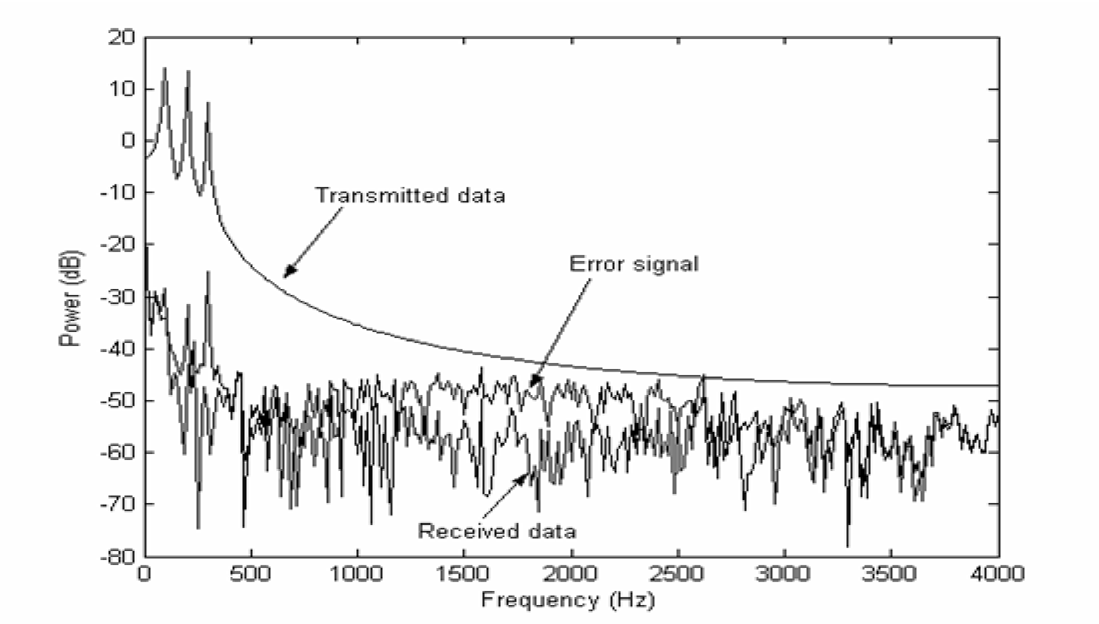

Fig. 6. Power spectrum for CE-LMS algorithm.

Using weights filter after training period, and eliminating adaptation, is another way to compare effectiveness algorithms. Fig. 7 depicts MSE for LMS and CE-LMS observing that CE-LMS gets least error.

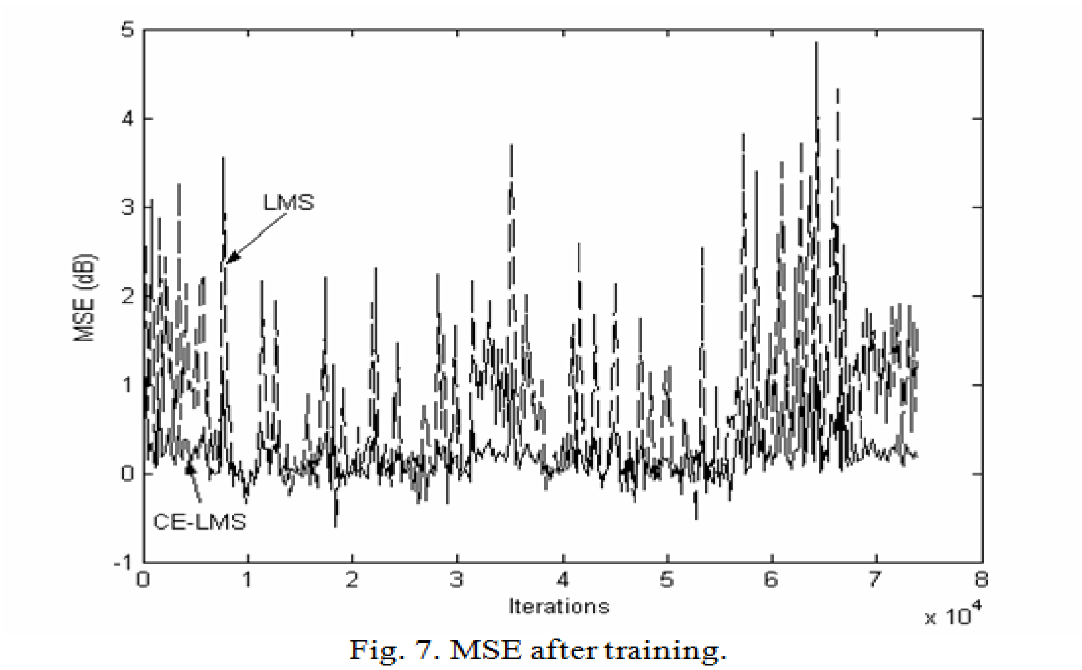

Fig. 7. MSE after training.

The ERLE (Echo Return Loss Enhacement), measures CE effectiveness. The ERLE is echo before and after cancellation.

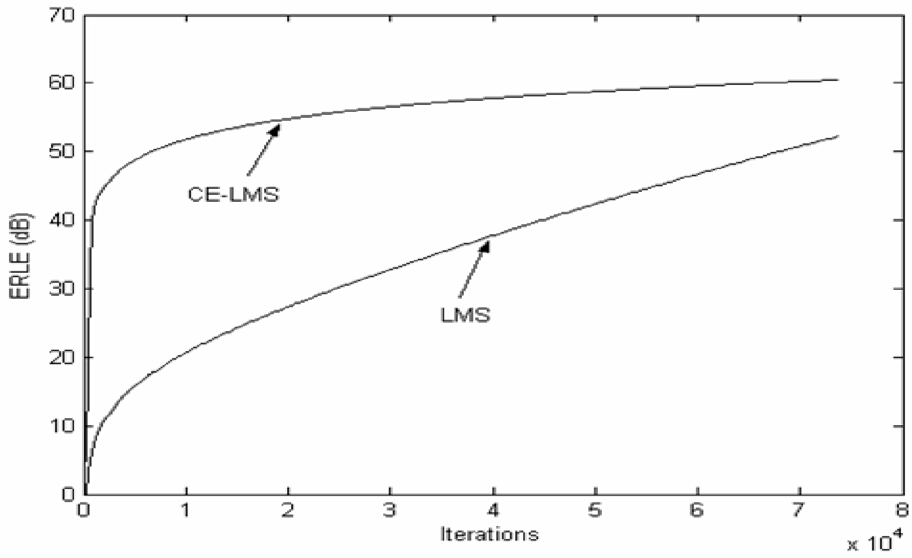

Fig. 8. ERLE for LMS and CE-LMS algorithm.

Fig. 8 depicts ERLE for LMS and the LMS proposed, showing that CE-LMS presents a better echo loss than LMS.

IV. DISCUSSION

The EC-LMS algorithm lets an easy digital design due to reduction of floating point operations, because input and error signals are integer numbers. An entirely implementation on fixed point is too complicated, due to convergence factor must satisfy condition presented in (2), where there is an inversely ratio, thus an integer convergence factor implies to make use of input and error with floating point performance.

V. CONCLUSION

The results obtained with Coded Error LMS algorithm show that it is an alternative for communications systems with high convergence speed.

REFERENCES

[1] Hector Pérez Meana, Mariko Nakano Miyatake, Luis Niño de Rivera, Juan Sánchez García; "Desarrollos recientes en sistemas adaptables analógicos"; Revista Científica; Vol. 5 No. 3; July- Sept 2001; pag. 113-129.

[2] Ayal Shoval, David A. Johns and W. Martin Snelgrove; "Comparison of DC Offset Effects in Four LMS Adaptive Algorithms"; IEEE Transactions on Circuits and Systems-II: Analog and Digital Signal Processing; Vol. 42, No. 3, March1995; pag. 176-185.

[3] Al Jackson; "ADSL For High-Speed Broadband Data Service"; Aerospace Conference, 1998. Proceedings., IEEE, Vol. 4, 21-28 March 1998; pag. 451–465.

[4] O'Neill; "A Low-Power 128-Tap Digital Adaptive Equalizer For Broadband Modems"; IEEE Journal of Solid State Circuits, Vol. 32, No. 11, November 1997; pag.: 1777-1789.

[5] Tai-Cheng Lee and Bezhad Razavi; "A 125 Mixed-Signal Echo Canceller for Gigabit Ethernet on Copper Wire"; IEEE Journal of Solid-State circuits, Vol. 36, No. 3, March 2001; pag. 366-373.

[6] Bhupendra K. Ahuja, Miles A. Copeland and Chong Hon Chan, "A Sampled Analog MOS LSI Adaptive Filter", IEEE Journal Of Solid-State Circuits, Vol. SC-14, No.1, February 1979; pag.148-154.

[7] Miguel Figueroa, Seth Bridges, David Hsu, and Chris Diorio, "A 19.2 GOPS Mixed-Signal Filter With Floating-Gate Adaptation", IEEE Journal of Solid State Circuits, Vol. 39, No.7, July 2004; pag. 1196-1201.

[8] Carusone, A., Johns, D.A.; "Analogue adaptive filters: past and present"; Circuits, Devices and Systems, IEEE Proceedings; Vol: 147, February 2000; pag. 82 – 90.

[9] Hanna Bogucka and Krzysztof Wesolowski; "Frequency- Domain Echo Cancellation in Digital Multicarrier Modulation Systems"; IEEE Transactions on Communications; Vol. 48, No.2, February 2000; pag. 333-342.

[10] Katleen Van Acker, Geert Leus, Marc Moneen; "Per Tone Equalization for DMT-Based Systems"; IEEE Transactions on Communications, Vol. 49, No. 1, January 2001, pag: 109-119.

[11] Shivaling S. Mahant-Shetti, Srinath Hosur and Alan Gatherer; "The Log-Log LMS Algorithm" Acoustics, Speech, and Signal Processing, 1997. ICASSP-97., 1997 IEEE International Conference on, 21-24 April 1997; pag. :2357-2360 vol.3.

[12] Bhupendra K. Ahuja, Miles A. Copeland and Chong H. Chan; "A new adaptive algorithm and its implementation in MOS LSI"; IEEE Journal of Solid State Circuits, Vol. SC-14, No. 4, August 1979; pag. 747-753.

[13] J. Velázquez López, Juan Carlos Sánchez García y Héctor Pérez Meana; "Algoritmo LMS con error codificado"; Proceedings CAS-Tour/ICED 2004; Veracruz México.

[14] J. Velázquez López, Juan Carlos Sánchez García y Héctor Pérez Meana; "Algoritmo LMS modificado para aplicaciones de filtrado adaptivo de rápida velocidad de convergencia", Proceedings CIECE 2005; Puebla México.

