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An improved LMS algorithm for single and doubletalk echo canceller implemented on Motorola DSP SC140

Daniel Silion, Dorin Panaitopol, Mircea Sorin Rusu[1]

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Keywords: echo canceller, LMS, GNGD.

I. INTRODUCTION

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In telecommunication networks, the combination of reflections from network components such as 2- to 4-wire converters, together with the signal processing and the transmission delay produce echo. Echo has a major effect on voice quality in telecommunication networks.

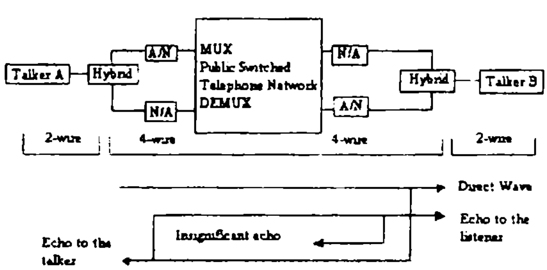

Fig.1. Echo apparition

B. *Problems involved*

Users may encounter difficulty in talking or listening over a telephone connection caused by the echo. It may also affect the transmission of voiceband data, fax and text over phone lines.

C. *Echo cancelling method*

Digital network echo cancellers are designed to eliminate echo reflected from the user and to allow successful transmission of voiceband data and fax.
Echo cancellers are devices placed in the 4-wire portion of a circuit (which may be an individual circuit path or a path carrying a multiplexed signal) and are used for reducing the echo by subtracting an estimated echo from the signal produced by hybrid circuit.

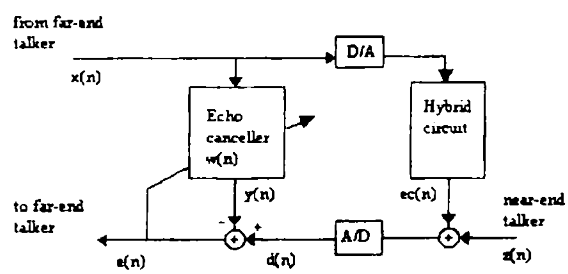

Fig.2. A simplified model of an Echo Canceller

In Fig. 2 it is represented a simplified model of an echo canceller, where $x(n)$ is the input signal vector (far-end signal) for the time instant n, $ec(n)$ is the echo, $z(n)$ is the near-end sequence ($z(n)=0$ in case of single-talk), $d(n)$ is a combination of the echo and the speech of the near-end talker, $y(n)$ is the adaptive filter's output, $w(n)$ is the filter coefficient (weight) vector and $e(n)$ is a combination of the reduced echo and $z(n)$.

II. ADAPTIVE ALGORITHMS FOR ECHO CANCELLATION

The adaptive algorithm is used to adjust filter's coefficients in order to minimize the echo, so in case

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of singletalk is desirable that $d(n)-y(n)=0$ and in case of doubletalk $d(n)-y(n)=z(n)\neq 0$.

A. *LMS (Least mean square algorithm)*

The least mean square algorithm is a simple, yet most frequently used, algorithm for adaptive finite-impulse response (FIR) filters. It is described by the following:

$$y(n)=x^T(n)\cdot w(n) \tag{1}$$

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

$$w(n+1)=w(n)+\mu\cdot e(n)\cdot x(n) \tag{3}$$

The parameter μ is the step size (learning rate) that defines how fast the algorithm is converging.
Ideally, we want an algorithm for which the speed of convergence is fast and the steady-state misadjustment is small when operating in a stationary environment, whereas in a nonstationary environment the algorithm should change the learning rate according to the dynamics of the input signal, so as to achieve as good a performance as possible.
Because of this, the normalized LMS (NLMS) algorithm has been introduced.

B. *NLMS (normalized LMS)*

The step size of NLMS was found to be $\eta(n)=\mu/\|x(n)\|_2^2$, $0<\mu<2$, where $\|.\|_2$ denotes the Euclidean norm. Equation used for the actualization of the filter's coefficients is described below:

$$w(n+1)=w(n)+\frac{\mu}{\varepsilon+\|x(n)\|^2}e(n)x(n) \tag{4}$$

In theory, value $\mu=1$ provides the fastest convergence, whereas in practice, the step size of the NLMS algorithm needs to be considerably smaller.
To preserve stability for close-to-zero input vectors, the optimal NLMS learning rate is usually modified as $\mu/\|x(n)\|_2^2\rightarrow\mu/(\|x(n)\|_2^2+\varepsilon)$, where ε is a small positive constant.

C. *GNGD (generalized normalized gradient descent algorithm)*

The GNGD represents an extension of the normalized least mean square (NLMS) algorithm by means of an additional gradient adaptive term in the denominator of the learning rate of NLMS. This way, GNGD adapts its learning rate according to the dynamics of the input signal, with the additional adaptive term compensating for the simplifications in the derivation of NLMS. The performance of GNGD is bounded from below by the performance of the NLMS, whereas it converges in environments where NLMS diverges. The GNGD is shown to be robust to significant variations of initial values of its parameters. Simulations in the prediction setting support the analysis.
The proposed GNGD algorithm is described by:

$$y(n)=x^T(n)\cdot w(n) \tag{5}$$

$$e(n)=d(n)-y(n) \tag{6}$$

$$w(n+1)=w(n)+\eta(n)e(n)x(n) \tag{7}$$

$$\eta(n)=\frac{\mu}{\|x(n)\|_2^2+\varepsilon(n)} \tag{8}$$

$$\varepsilon(n)=\varepsilon(n-1)-\rho\mu\frac{e(n)e(n-1)x^T(n)x(n-1)}{(\|x(n-1)\|_2^2+\varepsilon(n-1))^2} \tag{9}$$

where the $(.)^T$ is the vector transpose operator and $\|.\|_2$ denotes the Euclidean norm.

D. *Variable Step-Size NLMS*

The step-size μ governs the rate of convergence and the steadystate excess mean-square error. To meet the conflicting requirements of fast convergence and low misadjusment, the step-size needs to be controlled. In standard LMS, various variants for controlling the step-size have been proposed. The performance of these schemes is determined by how accurately they can estimate how far the filter is from optimal performance. In this paper, we have discussed three kinds of VSS-NLMS:

VSS-LMS:

$$\mu(n)=\alpha\mu(n-1)+\gamma e^2(n) \tag{10}$$

RVS-LMS:

$$\mu(n)=\alpha\mu(n-1)+\gamma p^2(n) \tag{11}$$

$$p(n)=\beta p(n-1)+(1-\beta)e(n)e(n-1) \tag{12}$$

VS-NLMS:

$$\mu(n)=\frac{\sigma_x^2(n)}{\sigma_e^2(n)+\zeta}\beta(n) \tag{13}$$

$$\sigma_x^2(n)=\alpha_1\sigma_x^2(n-1)+(1-\alpha_1)x^2(n) \tag{14}$$

$$\sigma_e^2(n)=\alpha_2\sigma_e^2(n-1)+(1-\alpha_2)e^2(n) \tag{15}$$

$$\beta(n)=\gamma\beta(n-1)+(1-\gamma)\frac{e^2(n)}{x^2(n)+\tau} \tag{16}$$

To preserve stability for close-to-zero input vectors, two very small positive constants ζ and τ were introduced as a necessity in the experiments.

III. REQUIREMENTS OF G168 RECOMMENDATION

Experiments were performed with ITU-G168 echo paths. The tests use special signals such as noise, tones, group 3 facsimile signals, and a subset of the composite source signals (CSS) consisting of the band-limited CSS with speech like power density spectrum and the band-limited CSS for double talk. The CSS emulates the characteristics of the speech, and its use as a test signal improves the ability to measure echo canceller performance for speech signals. In experiments were used the signals css_st, for far-end talker and css_dt, for near-end talker.

IV. SIMULATION RESULTS

Using MatLab simulations, different algorithms were compared by the point of view of convergence speed and echo attenuation. The echo signal was obtained using ITU-G168 echo paths, especially B1 path with N=64 elements.

The algorithms were studied in optimum conditions, the parameters for each algorithm were chosen by means of faster convergence and better attenuation.

In experiments, for NLMS and GNGD, value $\mu = 1$ provides the fastest convergence, whereas in practice, the step size of the NLMS algorithm needs to be considerably smaller. Whatever, the interest is to find a optimum algorithm by comparisons in the same conditions and environment.

The tests for GNGD were made using $\mu = 1$ and $\rho = 0.9$.

Parameters used for Variable Step-Size NLMS algorithms are shown in Table 1.

Table 1

VSS-LMS	RVS-LMS	VS-NLMS
$\mu_{max} = 1$	$\mu = 0.02 \ldots 1$	$\alpha_1 = 0.03$
$\mu_{min} = 0.02$	$\alpha = 0.995$	$\alpha_2 = 0.93$
$\alpha = 0.995$	$\beta = 0.99$	$\gamma = 0.90$
$\gamma = 0.015$	$\gamma = 0.013$	

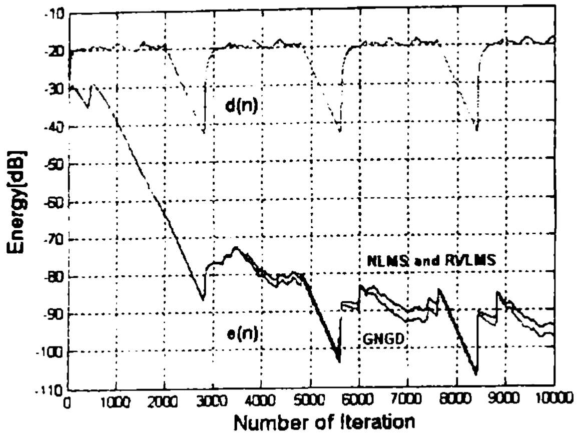

Fig.3. Comparison between GNGD, NLMS and RVLMS algorithms

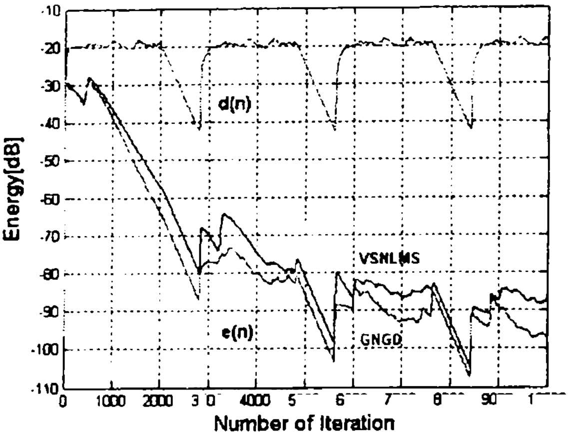

Fig.4. Comparison between GNGD and VSNLMS algorithm

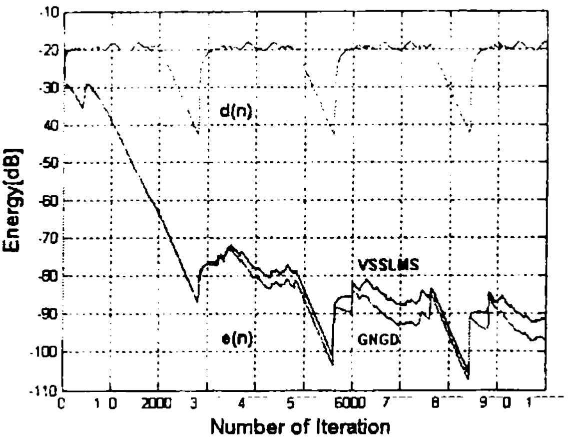

Fig.5. Comparison between GNGD and VSSLMS algorithm

As it is shown in figures presented below, some algorithms are proved to have a lower speed of convergence and others a lower attenuation of the echo, than GNGD.

In Fig. 6 are presented the test signal (CSS_ST), the echo signal and the error signal (the minimized echo) for GNGD algorithm. For single-talk case, $z(n) = 0$, so in this simulation, CSS_DT is not used. In this case, $d(n) = ec(n)$.

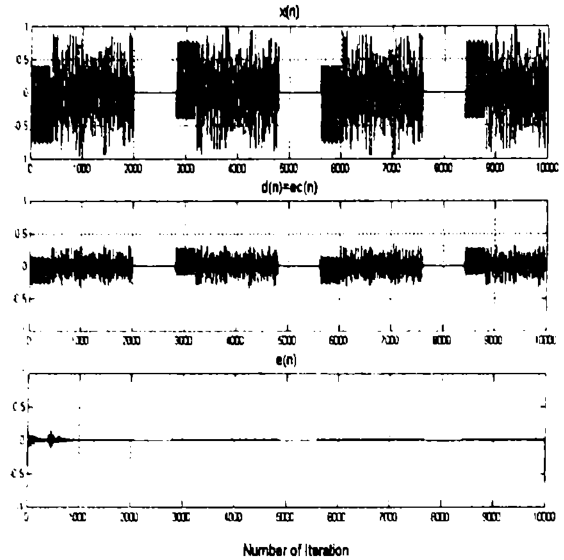

Fig.6. Signals for GNGD algorithm - single-talk

The analyse was also made in the case of doubletalk, were $z(n) \neq 0$. In this case, $d(n) = z(n) + ec(n)$.

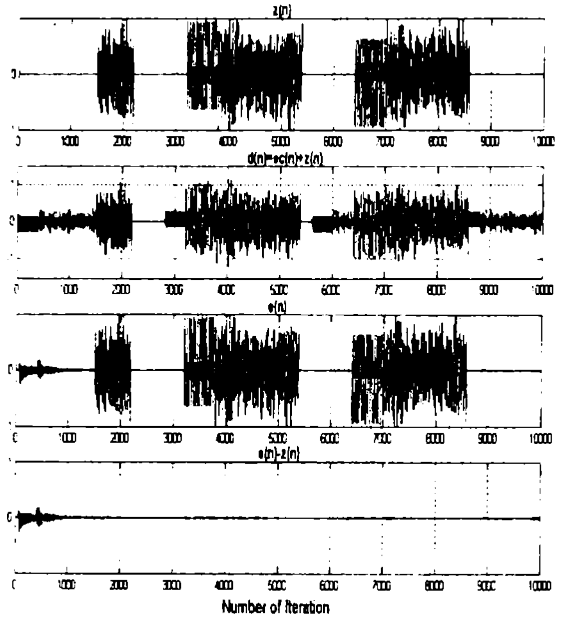

Fig 7 Signals for GNGD algorithm - double-talk

In case of double-talk, the equations used for the actualization of the filter's coefficients for NLMS (4), and for GNGD (7), (8), (9) were modified into:

NLMS:

$$w(n+1) = w(n) + \frac{\mu}{\varepsilon + N \cdot \sigma_x^2(n)} e(n)x(n) \qquad (4')$$

GNGD:

$$\eta(n) = \frac{\mu}{N \cdot \sigma_x^2(n) + \varepsilon(n)} \qquad (8')$$

$$\varepsilon(n) = \varepsilon(n-1) - \rho\mu \frac{e(n)e(n-1)x^T(n)x(n-1)}{(N \cdot \sigma_x^2(n-1) + \varepsilon(n-1))^2} \qquad (9')$$

where $\sigma_x^2(n) = (1-\alpha) \cdot \sigma_x^2(n-1) + \alpha \cdot x^2(n)$ (17)

As it can be seen below, GNGD behaves better than NLMS or the other studied algorithms even in the case of double-talk. In Fig. 8 were compared the echo attenuations in both cases.

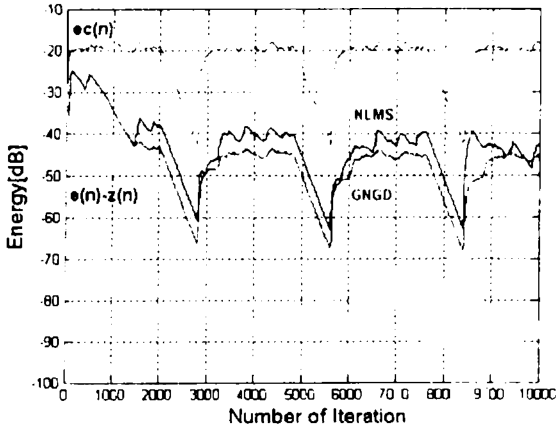

Fig.8 Comparison between GNGD and NLMS algorithm - double-talk

V. IMPLEMENTATION ON DSP

The real time algorithm implementation was tested on an Application Development System equipped with SC140 DSP and A/D, D/A codec at 8kHz sampling frequency. The program was optimized for the improved SC140's architecture (4 ALUs, dual data memory transfer) to reduce time execution cycles. The code implementation was performed using MatLab reference code. The results using fixed-point C specific instructions were similar to MatLab simulations.

VI. CONCLUSION

The error vector is used as a criterion to determine how close the adaptive filter is to optimum performance. The algorithms show improved system performance.

A generalized normalized gradient descent algorithm for

linear adaptive filters has been proposed for an echo cancellation system. It has been derived as an extension of the normalized least square algorithm where the learning rate comprises an additional adaptive factor, which stabilizes NLMS and makes GNGD suitable for filtering of nonlinear and nonstationary signals. Unlike the previously proposed gradient adaptive step size algorithms, GNGD has

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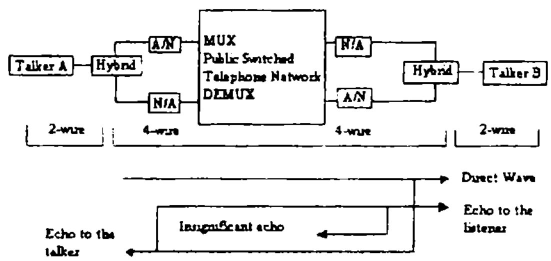

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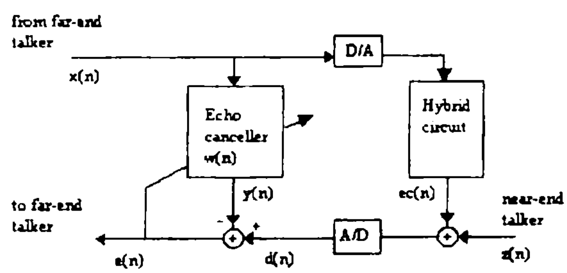

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been shown to be robust to the initialization of its parameters.

VII. ACKNOWLEDGMENT

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REFERENCES

[1] *ITU-T Recommendation G.168, Digital Network Echo Cancellers*, 2000.

[2] *ITU-T Recommendation P.56, Objective measurements of active speech level.*

[3] Danilo P. Mandic, *"A Generalized Normalized Gradient Descent Algorithm"*, IEEE Signal Processing Letters, Vol. 11, No. 2, February 2004.

[4] Hyun-Chool Shin, Ali H. Sayed, Woo-Jin Song, *"Variable Step-Size NLMS and Affine Projection Algorithms"*, IEEE Signal Processing Letters, Vol. 11, No. 2, February 2004.

[5] Silviu Ciochina, Cristian Negrescu, *"Adaptive systems"*, Editura Tehnica, Bucharest, 1999.

[6] Jianfeng Liu, *"A Novel Adaptation Scheme in the NLMS Algorithm for Echo Cancellation"*, IEEE Signal Processing Letters, Vol. 8, No. 1, Jan 2001.

[7] Tomas Gänsler, Steven L. Gay, M. Mohan Sondhi, Jacob Benesty, *"Double-Talk Robust Fast Converging Algorithms for Network Echo Cancellation"*, IEEE Transaction on Speech and Audio Processing, Vol. 8, No. 6, November 2000.

[8] *SC140 DSP Core Reference Manual*, Revision 3, November 2001.

[9] *SC100C Compiler User's Manual*, Revision 2, November 2001.

[10] *StarCore® SC140 Application Development Tutorial*, Revision 0, March 2003.