AN EFFICIENT MULTICHANNEL LINE ECHO CANCELER ALGORITHM FOR PSTN AND VoIP/VoDSL APPLICATIONS[§]

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ABSTRACT

This paper proposes an efficient multichannel line echo canceler algorithm with interchannel distribution of computations. A limited number of coefficient adaptations are distributed among channels depending on the degree of convergence and the input signal power to achieve up to 50% reduction of total computations. Coefficient adaptations are more frequently performed in the channels where convergence is behind others and there is a sufficient input power. As the index to the degree of convergence, an averaged sum of squared coefficients is used. Its gradient weighted with the input signal power is evaluated for interchannel distribution of coefficient adaptations. Simulation results with white Gaussian signals and speech signals demonstrate good convergence. The efficiency of coefficient adaptations is improved by 30% over the conventional algorithm. The computational savings can be used to accommodate more channels on the same chip, or to cover a longer echo-path with additional delays by codecs and/or ATM cell assembly/disassembly. It is also promising to analog interface in IADs and SOHO routers for VoIP/VoDSL applications.

1. INTRODUCTION

Line echo cancelers are widely employed for cancelling echoes generated at two-to-four-wire conversions (hybrid transformers) in central switching offices (CSOs) of the PSTN (plain switched telephone network). Each CSO accommodates multiple channels, thereby multiple echo cancelers are needed. Figure 1 shows a typical PSTN with five CSOs (A through E) as an example. CSO D has four independent lines (channels) with dedicated echo cancelers. Echo cancelers in the same CSO may be combined as a multichannel echo canceler [1]. Similar configuration can be found in IAD/SOHO[1] routers in VoIP/VoDSL[2] applications where 4 to 24 analog terminals are multiplexed [2].

Such a multichannel echo canceler can be implemented efficiently by sharing computations for coefficient adaptation among channels [3, 4] based on the random nature of call set-ups. The total number of coefficient adaptations is limited and distributed among channels depending on the degree of convergence. Coefficient adaptations are more frequently performed in the channels where convergence is behind others. It is reported [3] that the computational saving reaches 50%.

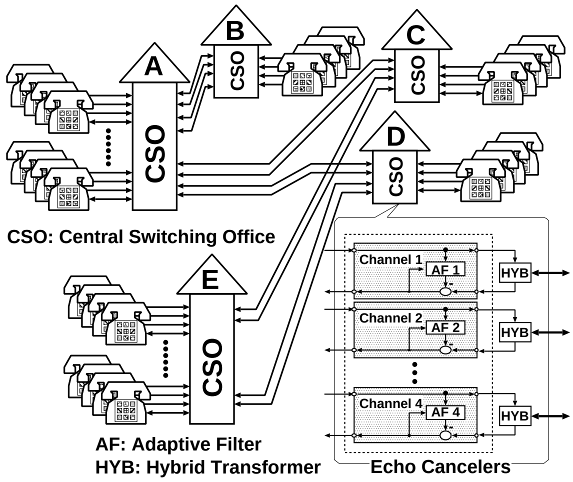

Figure 1: Typical PSTN with Echo Cancelers.

This algorithm assumes the normalized LMS (NLMS) algorithm [5] for coefficient adaptation because of its popularity. The step size for coefficient adaptation is normalized by the input signal power. To avoid divergence caused by a small input power, coefficient adaptation is generally skipped when the input signal power is smaller than a predetermined level. Another remedy for divergence is to add a small constant to the normalization factor [6]. In either case, a part of the assigned coefficient adaptations may not contribute to evolution of coefficients. A later section will demonstrate that such an inefficiency reaches 30%. Computational efficiency could be improved if these wasted adaptations were assigned to channels where coefficients are actually adapted.

This paper proposes an efficient multichannel line echo canceler algorithm with interchannel distribution of computations. The new algorithm evaluates the input signal upon distribution of coefficient adaptations so that no adaptations are assigned to silent channels. In the following section, the principle of the proposed algorithms is explained followed by its application to the terrestrial PSTN echo cancelers. The final section demonstrates improvement over wasted adaptations and the convergence characteristics of the proposed algorithms.

2. PROPOSED ALGORITHM

2.1. Interchannel Distribution of Adaptations

The proposed algorithm takes the input signal power into account in addition to the degree of convergence upon distribution of a

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[§] This research was conducted under a student training program arranged by NEC Corporation and Nagaoka University of Technology.

[1] IAD: Integrated Access Device, SOHO: Small Office, Home Office

[2] VoIP: Voice over Internet Protocol, VoDSL: Voice over Digital Subscriber Line

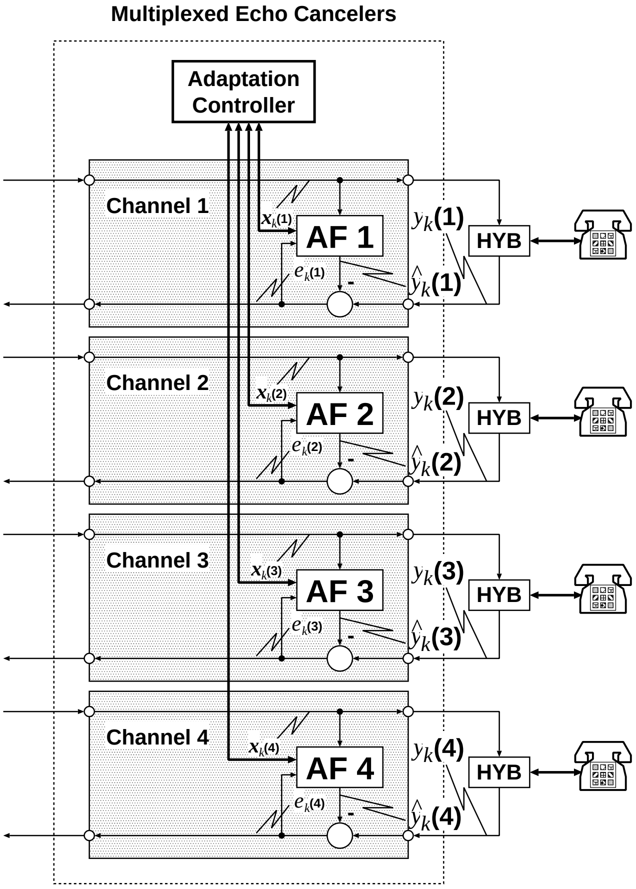

Figure 2: Blockdiagram of Multiplexed Echo Cancelers Equipped with the New Algorithm (M=4).

limited number of coefficient adaptations. The input signal power is first quantized with two levels, 0 and 1. The convergence index is multiplied by this quantized value to obtain a weighted convergence index. The number of coefficient adaptations is distributed based on the weighted convergence index. More coefficient adaptations are performed where they are highly demanded unless the input-signal power is small. To enable tracking of an echo-path drift, a minimum number of iterations are first assigned to M channels, where M is the number of channels. Then, the remainder will be distributed based on the weighted convergence index.

Figure 2 illustrates a blockdiagram of a multichannel echo canceler which implements this principle for a four-channel case ($M = 4$). There is an adaptation controller to collect information on the degree of convergence and the input signal power from the adaptive filter in each channel. The convergence index is multiplied by the quantized input-signal power to obtain the weighted convergence index. This principle leads to the following steps:

(I) A minimum number of adaptations, a_{min}, are distributed to each channel.

(II) The number of coefficient adaptations, $\bar{a}_{TTL}$, to be distributed based on the weighted convergence index, is calculated by

$$\bar{a}_{TTL} = a_{TTL} - a_{min} \cdot M, \tag{1}$$

where a_{TTL} is the original total available number of adaptations.

(III) The quantized input-signal power $\hat{P}_k(m)$, at the k-th iteration in the m-th channel, is calculated from the input signal power, $P_k(m)$, by

$$\hat{P}_k(m) = \begin{cases} 1 & for \quad P_k(m) \geq p_{th} \\ 0 & for \quad P_k(m) < p_{th} \end{cases}, \tag{2}$$

where p_{th} is a quantization level.

(IV) The convergence index, $\Delta_k(m)$, at the k-th iteration in channel m, is calculated by the following equation based on the convergence measure $d_k(m)$.

$$\Delta_k(m) = \left| \frac{d_k(m) - d_{k-1}(m)}{d_k(m)} \right| \tag{3}$$

$|\cdot|$ is an absolute-value operator. $\Delta_k(m)$ represents a gradient of $d_k(m)$.

(V) The weighted convergence index, $\bar{\Delta}_k(m)$, is calculated by

$$\bar{\Delta}_k(m) = \hat{P}_k(m) \cdot \Delta_k(m). \tag{4}$$

(VI) The number of coefficient adaptations, $a_k(m)$, at the k-th iteration for channel m, is calculated by the following equation.

$$a_k(m) = INT \left[\frac{\bar{a}_{TTL} \cdot \bar{\Delta}_k(m)}{\sum_{j=1}^{M} \bar{\Delta}_k(j)} \right] + a_{min} \tag{5}$$

$a_k(m)$ is given to channel m every D iterations of coefficient adaptation.

$INT[\cdot]$ is an operator to take the integer part of the argument. Because of this operator, the sum of $a_k(m)$ across the channels may be smaller than a_{TTL}. In such a case, the remainder will be redistributed to channels in proportion to the weighted convergence index. This redistribution will be repeated until the sum of $a_k(m)$ becomes equal to a_{TTL}. For $P_k(m) < p_{th}$, $a_k(m)$ becomes zero whatever the degree of convergence is. Therefore, wasted coefficient adaptations in channels with a small input power are fully utilized in other channels.

2.2. Convergence Measure

The proposed algorithm employs an averaged sum of squared coefficients [7] as the convergence measure instead of a time-varying step-size based on the autocorrelation of the error[3]. The time-varying step size is under influence by the echo autocorrelation when the coefficients are immature and thus, does not correctly reflect the convergence degree. The new measure grows as adaptation goes on and is finally saturated, faithfully reflecting the convergence. The convergence measure $d_k(m)$ is defined by

$$d_k(m) = \delta \cdot d_{k-1}(m) + (1 - \delta) \sum_{j=0}^{N-1} c_{j,k}^2(m), \tag{6}$$

where $c_{j,k}(m)$, δ, and N are the j-th coefficient at the k-th iteration in channel m, a positive constant satisfying $0 < \delta < 1$, and the number of taps, respectively. As $d_k(m)$ is saturated with convergence, $\Delta_k(m)$ in (3) approaches zero. Therefore, $\Delta_k(m)$ of the echo canceler, which is most advanced in terms of convergence, is the smallest and can be used as the convergence index in (5).

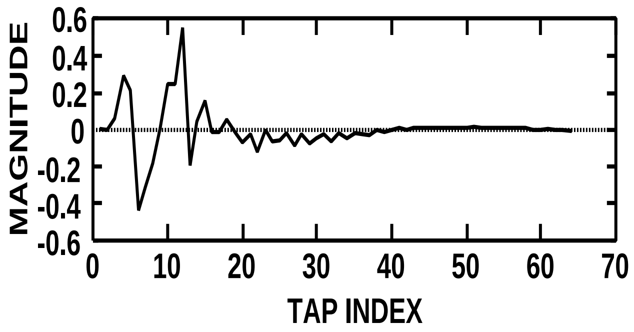

Figure 3: Impulse Response of the Echo Path.

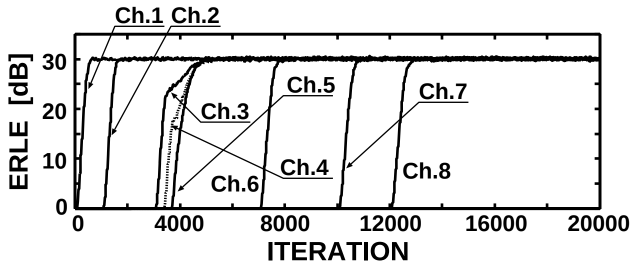

Figure 4: Convergence Characteristics (White Signals).

Table 1: Value of the Parameters.

a_{TTL}	2^5	D	2^5	μ	2^{-1}
N	64	δ	$1-2^{-7}$	p_{th}	2^{-5}

3. SIMULATION RESULTS

Performance in the single-talk was evaluated with white *Gaussian* signals and speech signals. Another white *Gaussian* signal with an echo-to-noise ratio of $-30dB$ was added as an additive noise. The impulse response depicted in Fig. 3 [8] was used for the echo path. The values of basic parameters are given in Tab. 1 where μ is the step size for coefficient adaptation. a_{TTL} was set equal to D for the severest case where the total available number of adaptations is equal to that for a single channel case. Selection of a_{TTL} is a trade-off between computations and convergence. Convergence was evaluated by $ERLE$ (echo return loss enhancement) which, at the k-th iteration in channel m, was calculated by

$$ERLE_k(m) = \frac{\sum_{i=k-N+1}^{k} y_i(m)^2}{\sum_{i=k-N+1}^{k} \{y_i(m) - \hat{y}_i(m)\}^2}. \qquad (7)$$

$y_k(m)$ and $\hat{y}_k(m)$ are the echo and the echo replica at the k-th iteration in channel m as defined in Fig. 2.

Figure 4 shows the convergence characteristics of the multi-channel echo canceler for white input signals. Channels 3 through 5 exhibit slower convergence than other channels. This is because they are competing in the convergence process and have to share a limited number of coefficient adaptations. All other channels can solely enjoy the whole available coefficient adaptations when they are in the convergence process. This is also understood from Fig. 5 where $10 \cdot \log_{10}$ of the weighted convergence index is shown

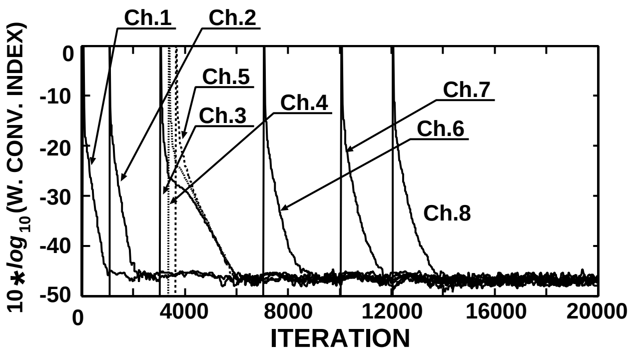

Figure 5: Weighted Convergence Index (White Signals).

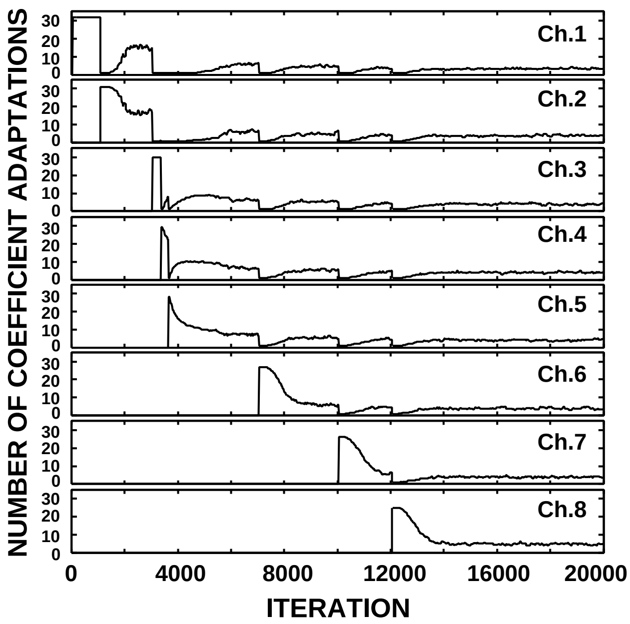

Figure 6: Distributed Coefficient Adaptations (White Signals).

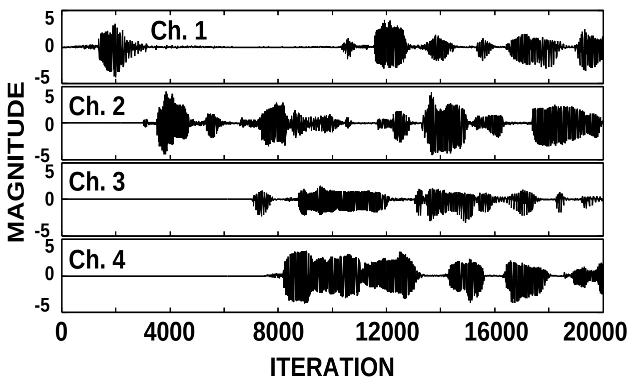

Figure 7: Speech Signals Used for Simulations.

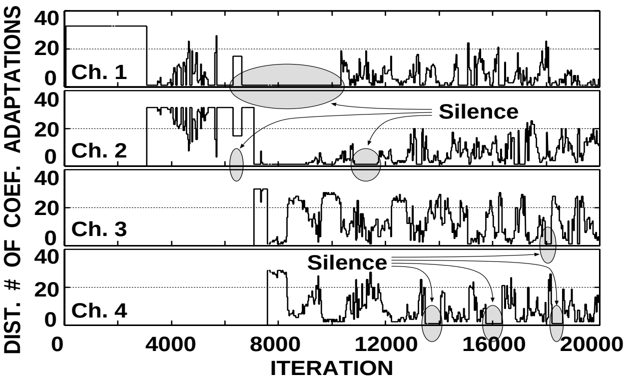

Figure 8: Distributed Coefficient Adaptations (Speech Signals).

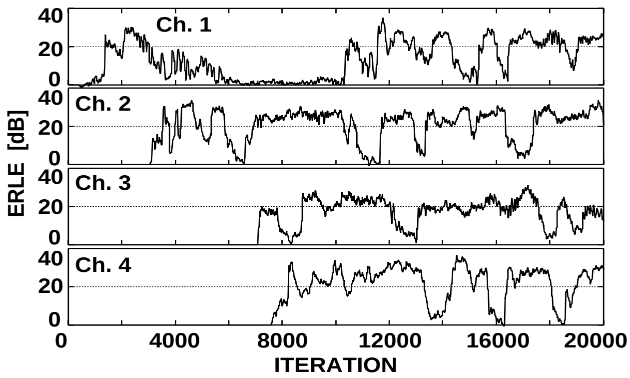

Figure 9: Convergence Characteristics (Speech Signals).

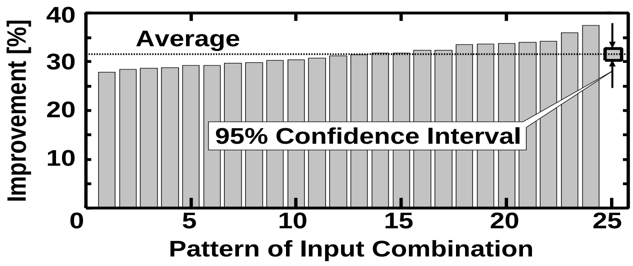

Figure 10: Improvement on Actual Coefficient Adaptations.

instead of the index itself for readability. The convergence is slow in the competing channels compared with other channels, thereby slow decrease of the weighted convergence index.

This result is reflected in Fig. 6 where the actual number of coefficient adaptations assigned to each channel is depicted. The maximum value is approximately equal to 32 in channel 1 whereas it is smaller in all other channels. This is because more than one channel were active simultaneously and the available coefficient adaptations, $D = 32$, had to be shared.

For speech signals depicted in Fig. 7, the distributed number of coefficient adaptations is shown in Fig. 8. Only four channels were used for readability of the graphs. No adaptations were assigned to the channel in a silent section as are marked by shaded areas. This is achieved by weighting of the convergence index by the input signal power. When all channels have a small power, the available coefficient adaptations are evenly distributed over active channels. This happens at around 6500 iterations as in Fig. 8. The corresponding convergence characteristics are shown in Fig. 9.

Shown in Fig. 10 is improvement of actual coefficient adaptation by the proposed algorithm and that by [3] for $M = 4$. 24 different combinations of speech input signals have been evaluated. The average improvement was 31.5% and its 95% confidence interval is depicted in the right most column of the figure. 31.5% of the coefficient adaptations by the proposed algorithm, on average, would be wasted if the convergence index had not been weighted by the input signal power. This fact clearly shows superiority of the proposed algorithm over the conventional algorithm [3].

4. CONCLUSION

An efficient multichannel line echo canceler algorithm with inter-channel distribution of computations has been proposed. A limited number of coefficient adaptations are distributed over channels based on the degree of convergence weighted by the input signal power for as much as 50% computational saving. As the convergence index, the gradient of an averaged sum of squared coefficients is used. Good convergence characteristics with white Gaussian signals and speech signals have been demonstrated by simulations. It has been shown that the efficiency of coefficient adaptations is improved by 30%. The proposed algorithm contributes to further cost reduction by accommodating more channels on the chip and/or by longer echo-path coverage in the PSTN and VoIP/VoDSL applications such as IADs and SOHO routers.

ACKNOWLEDGMENTS

The authors would like to thank Miho Hoshino of Chuo University for her contribution to the data in Fig. 10. They are also indebted to Dr. Kazunori Ozawa, formerly a Senior Manager of C&C Media Research Laboratories, NEC Corporation, and Prof. Shin-Ichi Hanaki of Nagaoka University of Technology for their efforts to coordinate the student training program.

5. REFERENCES

[1] D. L. Duttweiler, “A Twelve Channel Digital Echo Canceler,” IEEE Trans. on Com., vol.26, No.5, pp.647-653, May. 1978.

[2] D. Minoli and E. Minoli, “Delivering Voice over IP Networks,” John-Wiley and Sons, Inc., 1988.

[3] A. Sugiyama and G. Homma, “Computationally Efficient Echo Canceler Algorithms for Multiple Telephone Lines,” Proc. 14th DSP Symp., A6-5, pp.671-676, Nov. 1999.

[4] A. Sugiyama and G. Homma, “A Computationally Efficient Echo Canceler Algorithm for Multiple Telephone Lines,” Proc. IEEE DSP Workshop, Hunt, TX, Oct.15-18, 2000.

[5] G. C. Goodwin and K. S. Sin, “Adaptive Filtering, Prediction and Control,” Prentice-Hall, Info.and Sys. Sci. Ser., 1985.

[6] R. R. Bidmead and B. D. O. Anderson, “Performance of Adaptive Estimation Algorithms in Dependent Random Environments,” IEEE Trans. Automat. Contr., vol. AC-25, pp.788-794, Aug. 1980.

[7] A. Sugiyama, “An Adaptive-Step LMS Algorithm with Coefficient Value Evaluation,” Proc. 1991 Fall Conference of IEICE, A75, pp.1–75, Sep. 1991.

[8] T. Aboulnasr and K. Mayyas, “A Robust Variable Step-Size LMS-Type Algorithm,” IEEE Trans. on SP, vol.45, No.3, pp.631-639, Mar. 1997.