

A COMPUTATIONALLY EFFICIENT ECHO CANCELER ALGORITHM FOR MULTIPLE TELEPHONE LINES[§]

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ABSTRACT

—This paper proposes a computationally efficient echo canceler algorithm for multiple telephone lines. 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. Each adaptive filter is implemented as a sparse-tap adaptive filter, whose tap positions are adaptively controlled. As the index to the degree of convergence, tap concentration to dispersive regions is used. Assuming a floating-point signal processor, computational saving approaches 40% which is close to the original coefficient adaptation share, *ie.* 50%. This indicates 40% larger number of channels on the same chip, or 40% longer echo-path coverage with longer delays caused by codecs and/or ATM cell assembly/disassembly. Simulation results with white Gaussian signals and speech signals demonstrate good convergence.

1. INTRODUCTION

Line echo cancelers have been used to cancel echoes generated at two-to-four-wire conversions (hybrid transformers) in central switching offices (CSOs). Figure 1 shows a typical telephone network with five CSOs (A through E) as an example. At each CSO, multiple echo cancelers are needed. In case of CSO D, there are four independent lines (channels) with dedicated echo cancelers. Echo cancelers in the same CSO may be combined as a multichannel echo canceler which simultaneously cancels echoes in multichannels.

A twelve channel echo canceler, developed by Duttweiler[1], time-shared the multipliers for convolution in FIR (Finite Impulse Response) filtering and correlation in coefficient adaptation among twelve channels. Therefore, for a case of N -tap FIR filters in M channels, the necessary number of multipliers is reduced to $1/NM$ of the original. In those days, such a time-sharing manner was practical and effective because a single echo canceler was implemented by primitive logic chips. However, in today's DSP (digital signal processor) implementation, such time-sharing is automatically achieved in a single chip. Therefore, the total computations for multiplexed echo cancelers are simply proportional to the number of channels to be multiplexed. For further cost reduction, new techniques suitable for today's LSI technology are needed.

This paper proposes a computationally efficient echo canceler algorithm for multiple telephone lines. In the following section, the principle of the proposed algorithms is explained. The new algorithm is introduced and explained in details in Section 3. Section 4

[§] This research was conducted under student training programs arranged by NEC Corporation and Chuo University.

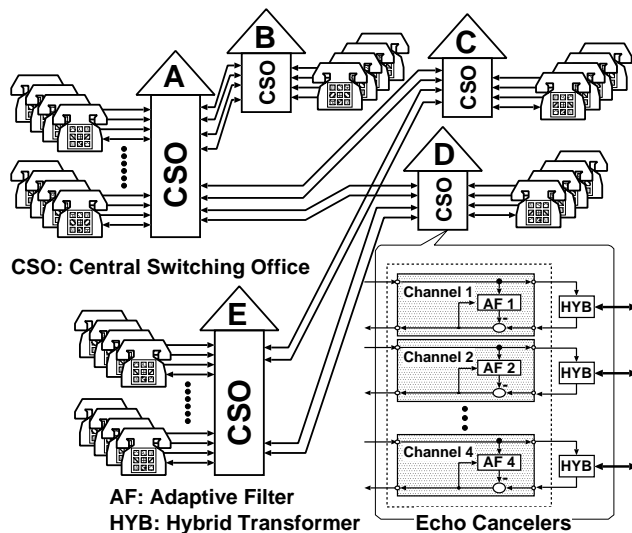


Figure 1: Typical Telephone Network with Echo Cancelers.

evaluates the computational saving in comparison with the straightforward implementation. Finally, in Section 5, the convergence characteristics of the proposed algorithm are demonstrated.

2. PRINCIPLE OF THE PROPOSED ALGORITHM

It is widely known that telephone calls are set-up following the *Poisson* distribution, or in other words, randomly [2]. This implies that coefficient adaptation in the multiplexed echo cancelers do not start operation at the same time and the degree of convergence is different each other. When an echo canceler is still in the convergence process, some others may have converged. Coefficient adaptation in those converged channels is not as important as in the channel where convergence is still in progress. This is because coefficients may not be adapted unless there is a change in the echo path. Therefore, it is possible to skip coefficient adaptation in the converged channels.

The principle of the proposed algorithms is to distribute a small number of coefficient adaptations among channels based on the degree of convergence. Figure 2 illustrates multiplexed echo cancelers which implement this novel principle for a four-channel case. There is an adaptation controller which collects information on the degree of convergence from the adaptive filter in each channel. The degrees of convergence are compared each other and more coefficient adaptations are performed where they are more needed. Though the principle is general, this paper focuses on satellite echo cancelers for a detailed algorithm. Satellite

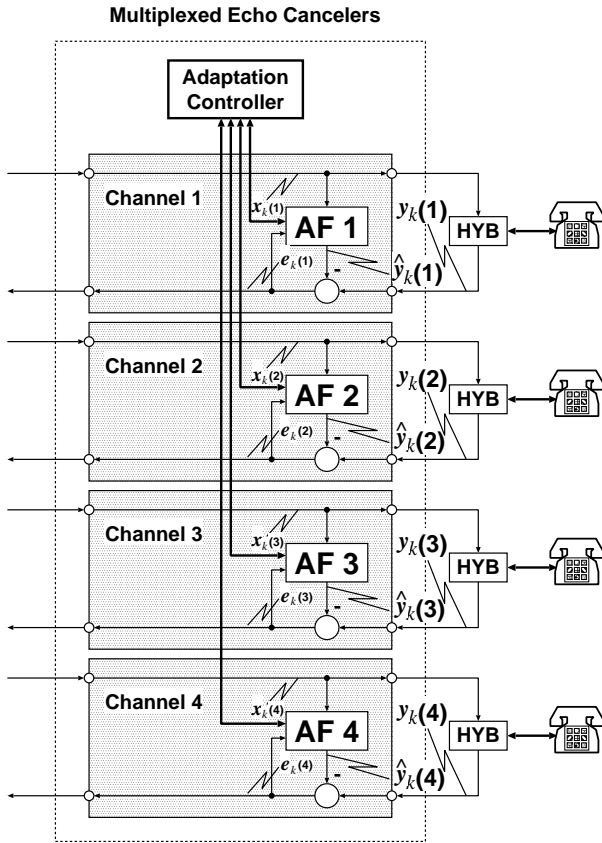


Figure 2: Blockdiagram of Multiplexed Echo Cancelers Equipped with the New Algorithm ($M=4$).

echo cancelers, which are commonly encountered in satellite links, model a long and sparse impulse response. Their round-trip delays can be as long as 600 *m.s.*

3. PROPOSED ALGORITHM

3.1. Algorithm in Each Channel[3]

For satellite echo cancellers, a sparse-tap adaptive FIR filter algorithm provides fast convergence and economical implementation [3, 4]. This algorithm controls the tap positions and coefficient values simultaneously so that a smaller number L of coefficients than the number N of delay elements can be placed where the impulse response of the echo path is significant.

Initially, coefficients are allocated to active taps based on the given initial condition and the indexes to inactive taps are kept in a queue. After Q coefficient adaptations, the index to the tap with the smallest absolute value of the coefficient is searched for and taken to be appended to the tail of the queue. The top index of the queue is taken out and used for the index to a new active tap position. This is how the tap-positions are exchanged and coefficients are located within dispersive regions.

A single constrained region is selected at a time from equisize subgroups of all possible tap positions. The selected constrained region hops from one subgroup to another along the tapped delay

line in order of its significance so that it covers multiple dispersive regions. The hopping order and the staying time for the subgroup are determined based on the sum of absolute coefficient values in each subgroup. They are redetermined every time all the subgroups have been selected once as the constrained region in the current hopping order.

Coefficients are updated by the Normalized LMS (NLMS) algorithm [5] as in

$$c_{k+1}(m) = c_k(m) + \mu \cdot \frac{e_k(m) \tilde{x}_k(m)}{|\tilde{x}_k(m)|^2}. \quad (1)$$

m represents the channel index and k is the number of iterations. μ is a fixed step size and $|\cdot|$ is an absolute-value operator. $\tilde{x}_k(m)$ is defined as a partial input signal vector whose elements are the input signal samples at the active taps.

3.2. Distribution of Coefficient Adaptations

The index to convergence has a significant impact on good distribution of coefficient adaptations. The proposed algorithm employs the largest value of the number of active taps, $q_k(m)$, among subgroups, as the convergence index $d_k(m)$. As adaptation goes on, it is increased and finally is almost saturated. Therefore, the larger is $q_k(m)$, the more advanced is the convergence. An average $\bar{q}_k(m)$ of $q_k(m)$ given by $\bar{q}_k(m) = \delta \cdot \bar{q}_{k-1}(m) + (1-\delta) \cdot q_k(m)$ is more desirable as the convergence index with smaller fluctuations. δ is a positive constant satisfying $0 < \delta < 1$. The number of coefficient adaptations $a_k(m)$ at the k -th iteration in the m -th channel is given by the following equation,

$$a_k(m) = \text{INT} \left[\frac{1/\bar{q}_k(m)}{\sum_{j=1}^M 1/\bar{q}_k(j)} \cdot T \right]. \quad (2)$$

In the designated coefficient adaptation period of D iterations, $a_k(m)$ adaptations are assigned to the m -th channel. $\text{INT}[\cdot]$ 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 D . In such a case, the remainder may be given to the channel whose convergence is most behind of others. Convergence of channel m is declared when a normalized change $\Delta_k(m)$ of $\bar{q}_k(m)$ given by

$$\Delta_k(m) = \frac{\bar{q}_k(m) - \bar{q}_{k-1}(m)}{\bar{q}_k(m)} \quad (3)$$

becomes smaller than a threshold Δ_{th} . When an echo canceler has converged, A minimum number of coefficient adaptations are assigned to converged channels in order to keep track with echo-path changes.

4. COMPUTATIONAL SAVING

Assuming ADSP-21020 (Analog Devices), FIR filtering with L sparse taps requires $4L + 10$ multiply-and-add (MPA) operations [5]. When coefficient adaptation and tap-position control are needed, the necessary MPA operations become $6.625L + 16$. This is because a combination of FIR filtering, power calculation (POW) for the NLMS algorithm, and 75% of the absolute-coefficient sum (SUM) is the heaviest load in a single instruction cycle [5]. In this case, FIR/POW takes $4L + 12$ MPAs and 75% of the SUM requires $2.625L + 4$ MPAs. When M_T channels are allowed to adapt coefficients, $(2.625L + 6)M_T + (4L + 10)M$

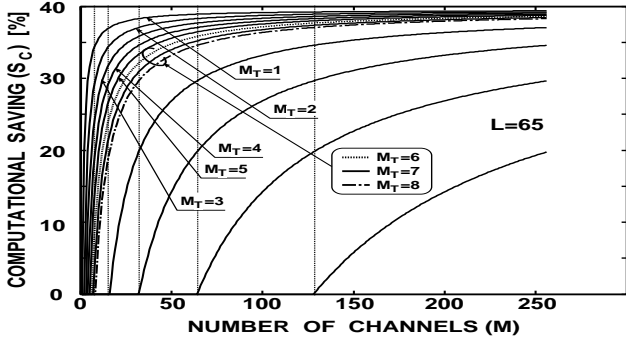


Figure 3: Computational Saving.

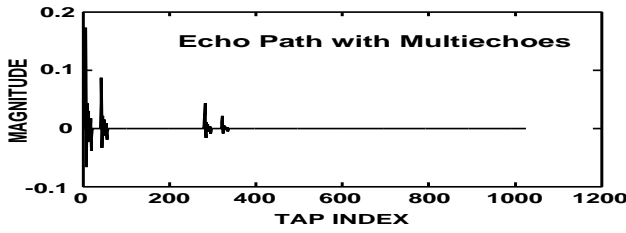


Figure 4: Impulse Response of the Echo Path.

MPA operations are consumed for the proposed multiplexed echo cancelers. Consequently, the computational saving S_C becomes

$$\begin{aligned}
 S_C &= 1 - \frac{(2.625L + 6)M_T + (4L + 10)M}{(6.625L + 16)M} \\
 &= \frac{(M - M_T) \cdot (2.625L + 6)}{(6.625L + 16)M}. \quad (4)
 \end{aligned}$$

Figure 3 illustrates the theoretical computational savings for various values of M_T and M , and $L=65$. For a parameter combination with $M=8$, $M_T=1$, whose convergence is evaluated in the following section, $S_C = 35\%$ is obtained from (4). For a larger M and L with $M_T=1$, S_C approaches 39.6%.

5. 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 -40 dB was added as an additive noise. *Linear-to- μ -law* and *μ -law-to-linear* conversions were incorporated before and after the echo path to model PCM (pulse code modulation) codecs.

The impulse response depicted in Fig. 4 [3] was used for the echo path. The values of basic parameters are given in Tab. 1. Other parameters were set as in [3].

Figure 5 depicts the *ERLE* (echo return loss enhancement) convergence of the multiplexed echo cancelers for white input signals. The echo canceler in channel 1 converged fastest among the others. Convergence in all other channels are slowed down because the echo cancelers in these channels had to share the available coefficient adaptations with others. The initial convergence in channel 2 is as quick as in channel 1. When channels 3 through 8 were activated while channel 2 had not converged, the convergence speed in channel 2 was significantly degraded.

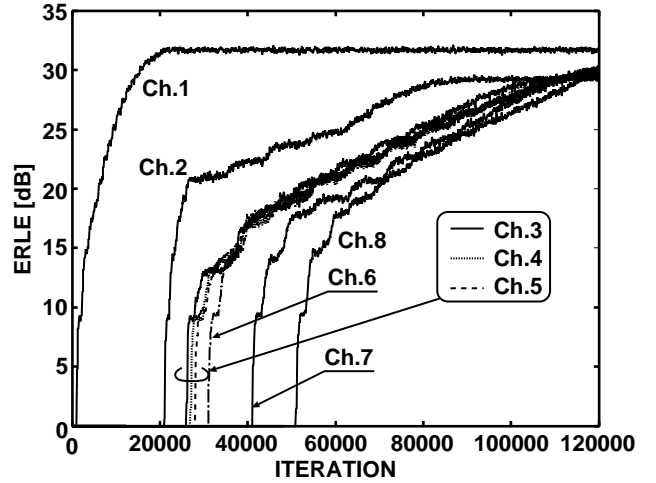


Figure 5: Convergence Characteristics (White Signals).

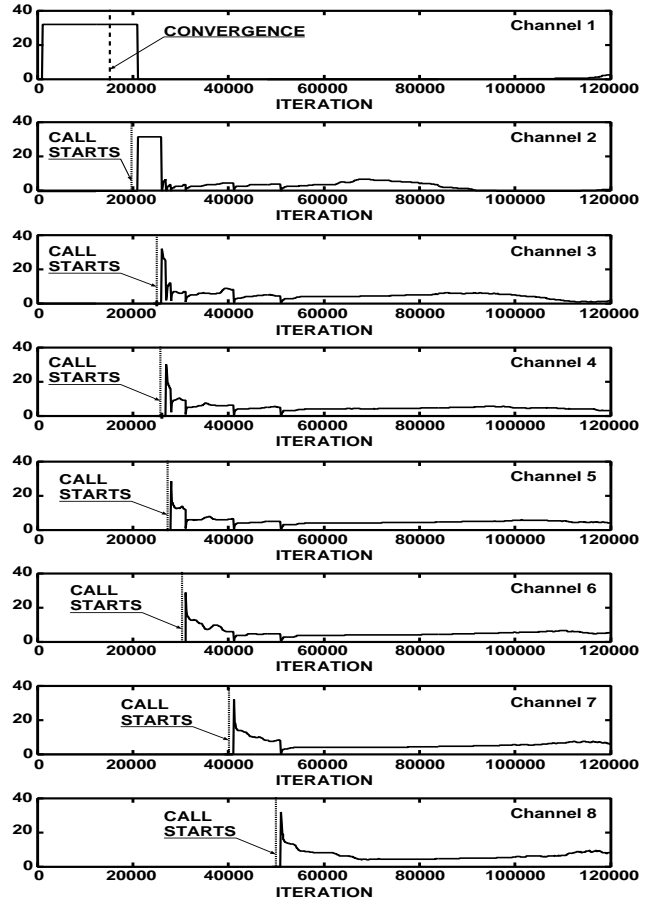


Figure 6: Distribution of Coefficient Adaptation (White Signals).

Tab. 1. Value of the Parameters.

N	1024	L	65	Q	2
μ	2^{-1}	δ	0.99	D	2^5

Shown in Fig. 6 are the actual number of coefficient adaptations distributed to each channel. It is again clear that the echo canceler in channel 1 monopolized coefficient adaptations for the first 20,000 iterations. It converged within 20,000 iterations, however, it kept using all the adaptations until there was another active channel.

With the speech signals shown in Fig. 7 as the input signal, the convergence for speech input was also evaluated. Only four channels were used in the simulations for readability of the graphs. Convergence characteristics for these speech signals are exhibited in Fig. 8. Figure 9 is the corresponding distribution of coefficient adaptations. Careful investigation of Figs. 8 and 9 reveals that faster convergence is achieved when there is only one active channel. Speed of convergence well agrees with the distribution of adaptations. *ERLEs* in channels 2 and 3 are slower in convergence compared with those in channels 1 and 4. This can be explained by the input signals in Fig. 7. The speech signals in channels 2 and 3 have more frequent silent sections, where coefficient adaptation is skipped and convergence is slowed down.

6. CONCLUSION

A computationally efficient echo canceler algorithm for multiple telephone lines has been proposed. Convergence in each channel is monitored and coefficient adaptation is controlled such that adaptive filters in less convergent channels are more frequently adapted with the total adaptations maintained constant. As the index to the degree of convergence, the largest value of the number of active taps are used. A detailed calculation has shown that the computational saving for a floating-point signal processor approaches 40% with a larger number of channels. Convergence characteristics with white Gaussian signals and speech signals have been demonstrated by simulation results. The proposed algorithm contributes to further cost reduction by accommodating more channels on the chip and/or by longer echo-path coverage.

ACKNOWLEDGMENTS

The authors would like to thank Shinpei Uno of Fujitsu Ltd and Masaki Suzuki of Chuo University for their contributions to simulation programs. They are indebted to Dr. Kazunori Ozawa and Dr. Takao Nishitani of NEC Corporation, for their support, continuous encouragement, and valuable comments. Efforts by Prof. Jinhui Chao of Chuo University in coordinating the student training program are also greatly appreciated.

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] R. B. Cooper, "Introduction to Queueing Theory," Second Ed., North Holland, 1981.
- [3] S. Ikeda and A. Sugiyama, "A Fast Convergence Algorithm for Sparse-Tap Adaptive FIR Filters for an Unknown Number of Multiple Echoes," in Proc. ICASSP'94, Vol.III, pp.41-44, Apr 1994.
- [4] A. Hirano, A. Sugiyama, and S. Ikeda, "DSP Implementation and Performance Evaluation of Sparse-Tap Adaptive FIR Filters with Tap-Position Control," in Proc. ICASSP'96, Vol.3, pp.1295- 1298, May 1996.
- [5] G. C. Goodwin and K. S. Sin, "Adaptive Filtering, Prediction and Control," Prentice-Hall, Info.and Sys. Sci. Ser., 1985.

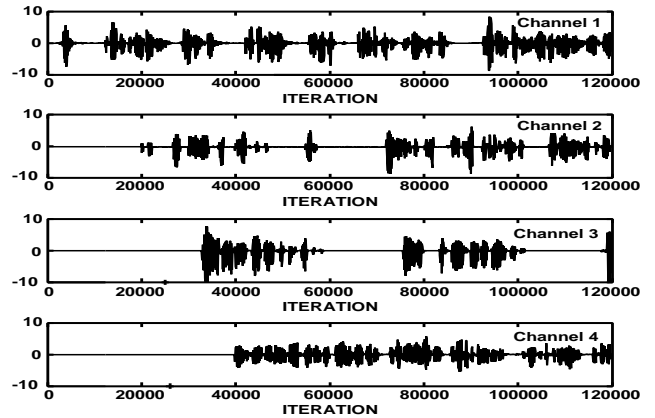


Figure 7: Speech Signals Used for Simulations.

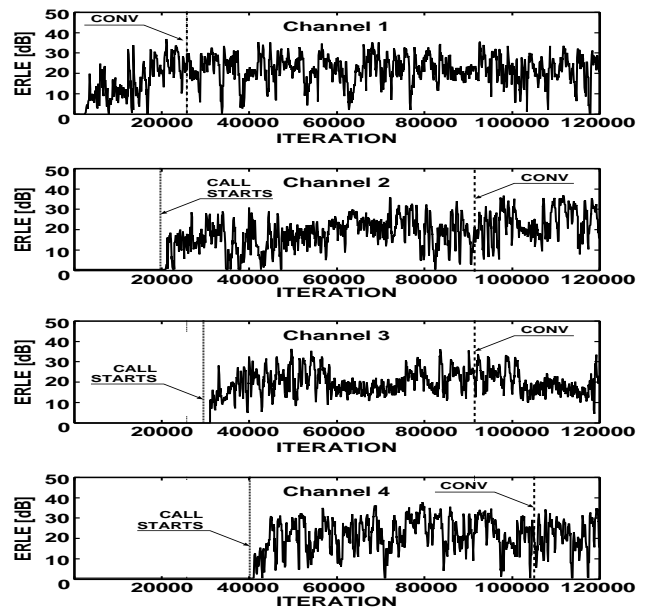


Figure 8: Convergence Characteristics (Speech Signals).

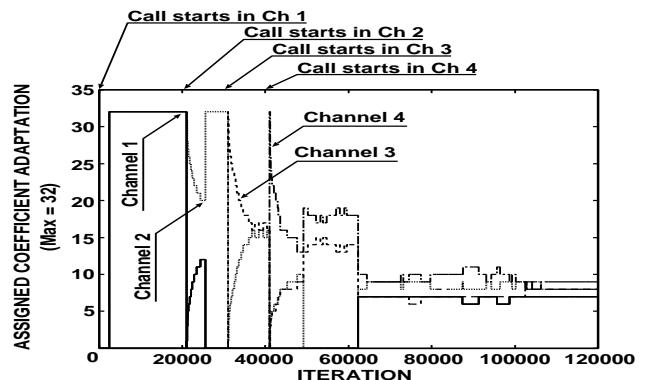


Figure 9: Distribution of Coefficient Adaptation (Speech Signals).