Real-Time Implementation Of An IIR Acoustic Echo Canceller On ADSP21020

M. Tahernezhadi and L. Liu
Department of Electrical Engineering
Northern Illinois University
DeKalb, IL 60115

ABSTRACT

In this paper, a practical IIR (pole-zero) lattice adaptive *acoustic echo canceller* (AEC) for teleconferencing application is proposed. The proposed algorithm consists in two parts: forward lattice and inverse lattice. Collectively, they are referred to as LATIN (*L*attice and *In*verse *L*attice) configuration. While the forward lattice is responsible for acoustic echo cancellation, the inverse lattice is employed in the *double-talk* (DT) mode only as to undo the distortion of the near-end speech brought about by forward lattice when suppressing the acoustic echo. Assuming M poles and M zeros for the proposed AEC, the complexity of the proposed algorithm is approximately twice the complexity of an M-tap FIR gradient lattice algorithm. Real-time experimentation conducted on ADSP21020 floating-point DSP chip attests to the stability and fast convergence of proposed IIR algorithm.

I. INTRODUCTION

Pole-zero modeling is most parsimonious in identification of systems whose impulse response has a long duration, such as acoustic echo path transfer function in teleconferencing systems. Acoustic echo is the dominant culprit that degrades the speech quality in such applications. The reverberation of a teleconferencing room results in a very long acoustic echo tail, the acoustic echo could last up to hundreds of milliseconds. For example, a filter length of 3200 is required to cancel 400ms acoustic echo when the sampling rate is 8kHz. The modeling of this kind of echo path not only requires enormous FIR adaptive filter, but also increases the number of memory locations and number of operations.

The LMS algorithm converges poorly when input signal has large eigenvalue spread and when the length of adaptive filter is large. This precludes the application of LMS-type algorithm as an AEC, especially when a large number of taps is required. These problems have motivated other approaches with the double purpose of reducing the computational complexity and of improving the convergence speed. Many studies are carried out on transversal filters due to their simplicity and stability. The RLS algorithm converges well but the computational cost is high [1]. The most mature and successful scheme is subband filtering [2]. However, even with subband techniques, the inclusion of large number of taps will be costly.

In recent years, stable and fast converging lattice pole-zero adaptive filters have been proposed [3]-[6], where the pole-zero modeling is accomplished by interpreting the problem as a two channel *auto-regressive* (AR) modeling problem. The pole-zero acoustic echo canceller proposed by Chao and Tsujii [4] employs both a forward and inverse lattice structure known as LATIN configuration. The LATIN configuration allows the distortion-free separation of near-end speech signal from the echo signal during the DT mode.

The stability requirements always come with the implementation of an IIR adaptive filter. Simple hyperstable adaptive recursive filter (SHARF) algorithm [5] and pole-zero structure [3] are proposed to increase the stability of IIR system. The IIR echo canceller proposed by Chao and Tsujii [4]-[5] uses a two-dimensional lattice structure. The stability requirement for this IIR lattice is that the eigenvalues of all the partial correlation matrices are inside the unit circle. This condition can be satisfied for a multichannel maximum entropy (ME) lattice filter, which has an enormously computational cost. A simpler lattice filter [4], which is updated by the *Levinson-Wiggins-Wittle-Robinson* (LWR) algorithm, needs additional gains to ensure the stability.

In this paper, a pole-zero lattice algorithm using LATIN configuration is proposed. The proposed algorithm combines the reference signal x(n) and the desired signal d(n) into one signal s(n), whose value at even time indices is x(n) and at odd indices is d(n). This in turn allows us to perform the pole-zero adaptive filtering via an algorithm that has a computation complexity of 20M for forward lattice and 4M for the inverse lattice (utilized only in the DT mode), where M is the number of feedforward and feedback taps. Thus, the complexity of the proposed algorithm is approximately twice the complexity of an M-tap FIR gradient lattice filter [7].

II. The PROPOSED LATTICE POLE-ZERO AEC

Consider the echo canceller structure in Fig. 1, where number of AR and *moving average* (MA) parameters assumed to be equal to M. x(n) and d(n) respectively denote the far-end signal (reference) and near-end (desired) signals. In the absence of DT, d(n) is the acoustic echo of x(n) and in the DT mode consists of the near-end signal plus the acoustic echo of x(n). From a practical point of view, one can send out $e'(n)$ as the final error during the single talk periods, thus obviating a pole-zero filtering stage [4]. During the DT mode, however, since the the near-end speech will be imbedded in the acoustic echo, the all pole filter 1/B(z) has to be supplied before $e'(n)$ is sent out as to avoid distortion of near-end speech signal.

The error $e'(n)$ given by

$$e'(n) = d(n) - \sum_{i=1}^{M} (b_i d(n-i) - a_i x(n-i)) \tag{1}$$

can be regarded as forward prediction error of d(n) for an (M,M) ARMA model. Likewise, one can define the forward prediction error of x(n) for an (M,M) ARMA model as

$$e''(n) = x(n) - \sum_{i=1}^{M} (q_i x(n-i) - p_i d(n-i)) \tag{2}$$

At this point, let us define a new sequence s(n) constructed from x(n) and d(n) as

$$s(2n) = x(n) \qquad s(2n+1) = d(n) \tag{3}$$

That is, the even and odd indices of s(n) corresponds to x(n) and d(n), respectively. The $e'(n)$ and $e''(n)$ can now be expressed in terms of s(n) as

0-7803-2431-5/95 $4.00 © 1995 IEEE

$$e'(n) = s(2n+1) - \sum_{i=1}^{2M} u_i s(2n+1-i) \qquad (4a)$$

$$e''(n) = s(2n) - \sum_{i=1}^{2M} v_i s(2n-i) \qquad (4b)$$

where $u_{2i} = b_i$, $u_{2i-1} = a_i$, $v_{2i} = q_i$, and $v_{2i-1} = p_i$. Thus, $e'(n)$ and $e''(n)$ are the linear prediction error of s(n) at time indices 2n+1 and 2n, respectively. That is, the problem has been transformed into a one dimensional linear prediction problem which can be solved using a one dimensional lattice predictor characterized by [7]

$$f_m(n) = f_{m-1}(n) - k_m^f(n) b_{m-1}(n-1) \qquad (5)$$

$$b_m(n) = b_{m-1}(n-1) - k_m^b(n) f_{m-1}(n) \qquad (6)$$

The backward and forward reflection coefficients, $k_m^f(n)$ and $k_m^b(n)$, are allowed to be different, as opposed to a conventional gradient lattice filter [3], because the stochastic components of s(n) vary from odd time indices to even time indices. The $k_m^f(n)$ and $k_m^b(n)$ are optimized according to least-square criterion. The forward reflection coefficient at odd indices, $k_m^f(2n+1)$ are chosen to minimize the cost function $E_m^f(2n+1)$ given by

$$E_m^f(2n+1) = \sum_{l=0}^{n} w^{n-l} |f_{m-1}(2l+1) - k_m^f(2n+1) b_{m-1}(2l)|^2 \qquad (7)$$

It can be readily shown that the minimization of (7) with respect to $k_m^f(2n+1)$ leads to the following recursive time updation for reflection coefficients at odd time indices

$$k_m^f(2n+1) = k_m^f(2n-1) + \frac{b_{m-1}(2n) f_m(2n+1)}{\eta_m(2n+1)} \qquad (8)$$

where

$$\eta_m(2n+1) = w\eta_m(2n-1) + |b_{m-1}(2n)|^2 \qquad (9)$$

Likewise, the above least-square based recursive time updation can be extended to determine the time updation of even index forward reflection coefficients as well as backward reflection coefficients at odd and even indices. Hence, one arrives at the complete forward lattice algorithm shown in Table I.

As mentioned before, whereas in the single talk mode, the desired output is $e'(n)$, during the DT mode the algorithm output should be e(n), which is $e'(n)$ filtered by the all pole filter 1/B(z). Since $b_i = u_{2i}$, the filter 1/B(z) can be realized by an inverse lattice structure as follows. One can easily show that

$$f_o(2n+1) = f_{2M}(2n+1) + \sum_{i=1}^{M} (b_i f_o(2n-2i+1) + a_i f_o(2n-2(i-1)) \qquad (10)$$

Since it is only required to do the 1/B(z) filtering, one needs to set all $f_o(n)$ at even indices to zero. Thus, (10) becomes

$$f_o(2n+1) = f_{2M}(2n+1) + \sum_{i=1}^{M} b_i f_o(2n-2i+1) \qquad (11)$$

Clearly, (11) can be readily realized via an inverse lattice algorithm. The complete algorithm of the inverse-lattice structure for the calculation of e(n) is shown in Table II, where $f'(n)$ and $b'(n)$ are used to represent the forward and backward prediction errors in the inverse lattice structure.

The minimum phase of the forward lattice is guaranteed if the absolute values of the reflection coefficients are less than unity. The minimum phase of the forward lattice is equivalent to the stability condition of the inverse lattice. The algorithm to update the reflection coefficients used here can always satisfy this condition. Moreover, one can use a clipper to constrain the absolute value of the reflection coefficients to prevent the possibility of divergence due to numerical error. The algorithm is observed to be stable in the real-time experimentation and in simulation.

III. The REAL-TIME IMPLEMENTATION OF THE PROPOSED AEC

The laboratory set-up for real time experimentation is shown in Fig. 2. The experiment is carried out to testify the performance of the algorithms. The LEC and HC respectively denote adaptive *line echo canceller* and adaptive howling controller, which are not discussed in this paper; however, there are required for an integrated teleconferencing system. A sampling frequency of 8kHz and a clock cycle of 25 MHz for ADSP 21020 are employed, which means the maximum number of instructions per iteration is 3125. The AEC performance is measured by the frequency response of the acoustic echo signal and the residual signal after cancellation.

The flowchart for the real-time implementation of the integrated teleconferencing system (it includes AEC, LEC and HC) is shown in Fig. 3. The corresponding flowchart for the implementation of the proposed AEC is shown in Fig. 4. The main program enters an idle loop, waiting for timer interrupts. The AEC algorithm is implemented in the timer interrupt service routine (ISR).

When an interrupt happens, the ISR will send out the calculated output of last sample "e" to the microphone out (D/A), i.e., to the hybrid. Subsequently, it reads in new sample of mic-in and far-end signal "d" and "x" through serial ports. Then it will calculate the new microphone output "e" using the IIR-lattice algorithm according to: (i) Pass "d" through the forward lattice filter, update the power estimate and reflection coefficients and calculate the output of each stag; (ii) Pass "x" through the forward lattice filter, update the power estimate and reflection coefficients and calculate the output of each stage; and (iii) Pass the calculated output through inverse lattice. The reflection coefficients are fixed (using those in forward lattice). It should be noted that in step (iii) the inverse lattice, required only for the DT mode, can be implemented only after the forward lattice has converged.

The experiments for both white noise and speech signal are carried out. The lattice IIR algorithm is implemented in real time using 30 poles and 30 zeros. The results are shown in Fig. 5 and Fig. 6 and the amount of cancellation that is measured is approximately 25dB for white noise input and actual speech input.

IV. CONCLUSIONS

In this paper, a new pole-zero AEC for teleconferencing application was proposed. The algorithm implemented using LATIN configuration has twice the complexity of an FIR gradient lattice structure in the forward lattice, and an added complexity of 1/3 of an FIR gradient lattice for the inverse lattice part used only when for the DT mode. The proposed algorithm was implemented in real-time using ADSP21020 floating-point DSP for an integrated teleconferencing system, and in excess of 25 dB cancellation of acoustic echo was obtained.

REFERENCES

[1] J. Makhoul, J., "A class of all-zero lattice digital filters: properties and applications," *IEEE Trans. Acoust., Speech, Signal Processing*, vol. ASSP-32, no. 4, pp. 791-802, 1984.

[2] Gilloire, A. and Vetterli, M., "Adaptive filtering in subbands with critical sampling: analysis, experiments, and application to acoustic echo cancellation," *IEEE Trans. Signal Processing*, vol. 40, no. 8, pp. 1862-1875. Aug. 1992

[3] Long, D., Shewed, D. and Falconer, D. D., "Study of a pole-zero adaptive echo canceler," *IEEE Trans. Circuits Syst.*, vol. CAS-34, no. 4, Aug 1982.

[4] Chao, J. and Tsujii, S., "A stable and distortion-free IIR echo and howling canceler," *IEEE Trans. Signal Processing*, vol. 39, no. 8, pp. 1812-1821, Aug. 1991.

[5] Chao, J. and Tsujii, S., "A new configuration for echo canceler adaptable during double talk periods," *IEEE Trans. Comm.*, vol. 37, no. 9, pp. 969-947, Sept. 1989.

[6] Lee, D. T. L., Friedlander, B. and Morf, M., "Recursive ladder algorithms For ARMA modeling," *IEEE Trans. Autom. Control*, vol. AC-27, no. 4, pp. 753-764, Aug. 1982.

[7] J. G. Proakis, *Digital Communications*, New York: McGraw-Hill Inc., 1983.

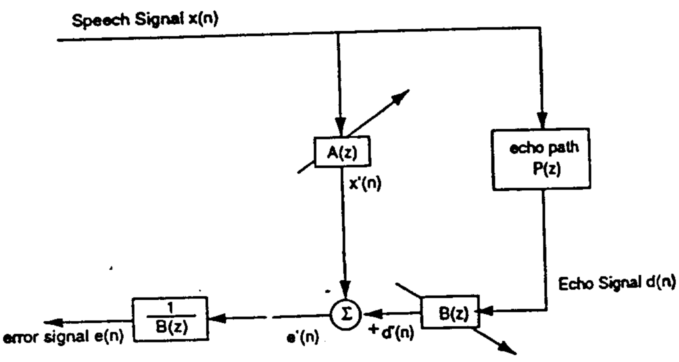

Fig. 1. The pole-zero structure AEC

Table I. The Proposed Forward Lattice Algorithm

$$f_o(2n) = b_0(2n) = x(n)$$

For Stage m=1:2M

$$\left(\begin{array}{c}
\eta_m(2n) = w\eta_m(2n-2) + |b_{m-1}(2n-1)|^2 \\
\theta_m(2n) = w\theta_m(2n-2) + |f_{m-1}(2n)|^2 \\
f_m(2n) = f_{m-1}(2n) - k^f_m(2n-2)\, b_{m-1}(2n-1) \\
b_m(2n) = b_{m-1}(2n-1) - k^b_m(2n-2) f_{m-1}(2n) \\
k^f_m(2n) = k^f_m(2n-2) + \dfrac{b_{m-1}(2n-1) f_m(2n)}{\eta_m(2n)} \\
k^b_m(2n) = k^b_m(2n-2) + \dfrac{f_{m-1}(2n)\, b_m(2n)}{\theta_m(2n)}
\end{array}\right)$$

$$f_o(2n+1) = b_0(2n+1) = d(n)$$

For Stage m=1:2M

$$\left(\begin{array}{c}
\eta_m(2n+1) = w\eta_m(2n-1) + |b_{m-1}(2n)|^2 \\
\theta_m(2n+1) = w\theta_m(2n-1) + |f_{m-1}(2n+1)|^2 \\
f_m(2n+1) = f_{m-1}(2n+1) - k^f_m(2n-1)\, b_{m-1}(2n) \\
b_m(2n+1) = b_{m-1}(2n) - k^b_m(2n-1) f_{m-1}(2n+1) \\
k^f_m(2n+1) = k^f_m(2n-1) + \dfrac{b_{m-1}(2n) f_m(2n+1)}{\eta_m(2n+1)} \\
k^b_m(2n+1) = k^b_m(2n-1) + \dfrac{f_{m-1}(2n+1)\, b_m(2n+1)}{\theta_m(2n+1)}
\end{array}\right)$$

$$e'(n) = f_{2M}(2n+1)$$

Table II. The Proposed Inverse Lattice Algorithm

$$f'_o(2n) = b'_0(2n) = 0$$

For Stage m=1:2M

$$\left(\begin{array}{c}
f'_m(2n) = f'_{m-1}(2n) - k^f_m(2n-2)\, b'_{m-1}(2n-1) \\
b'_m(2n) = b'_{m-1}(2n-1) - k^b_m(2n-2) f'_{m-1}(2n)
\end{array}\right)$$

$$f'_{2M}(2n+1) = e'(n)$$

For Stage m=2M:1

$$f'_{m-1}(2n+1) = f'_m(2n+1) + k^f_m(2n-1)\, b'_{m-1}(2n)$$

$$b'(0) = f'(0)$$

For Stage m=1:2M

$$b'_m(2n+1) = b'_{m-1}(2n) - k^b_m(2n-1) f'_{m-1}(2n+1)$$

$$e(n) = f'_o(2n+1)$$

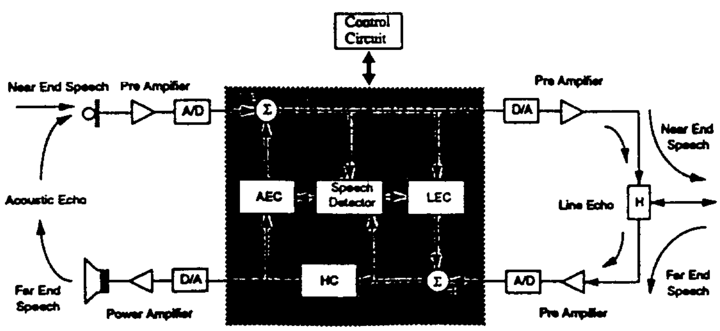

Fig. 2. An integrated teleconferencing system using ADSP21020

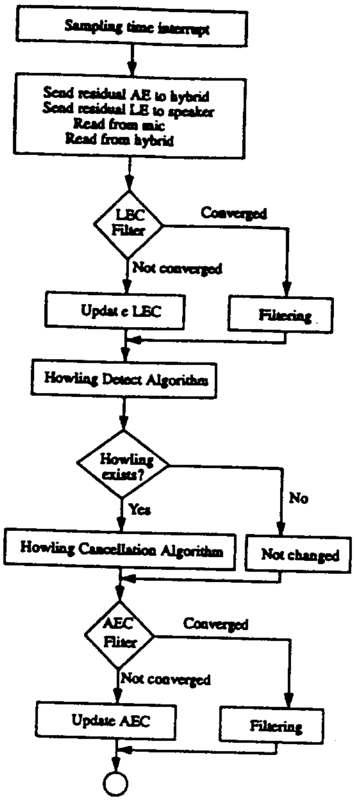

Fig. 3. Flowchart for an integrated teleconferencing system.

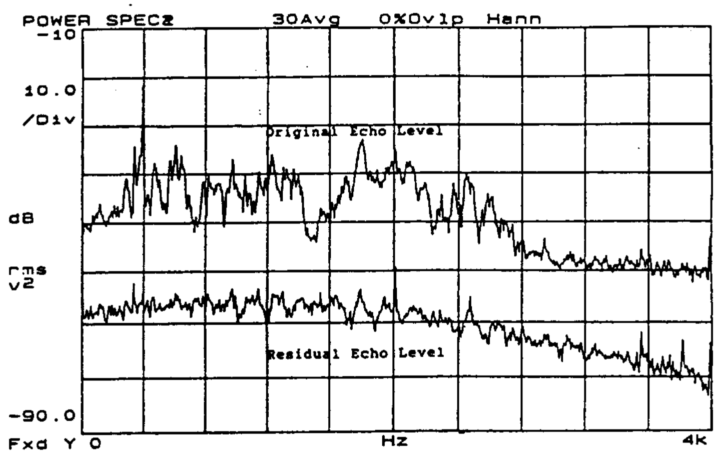

Fig. 5. Real-time results using speech input.

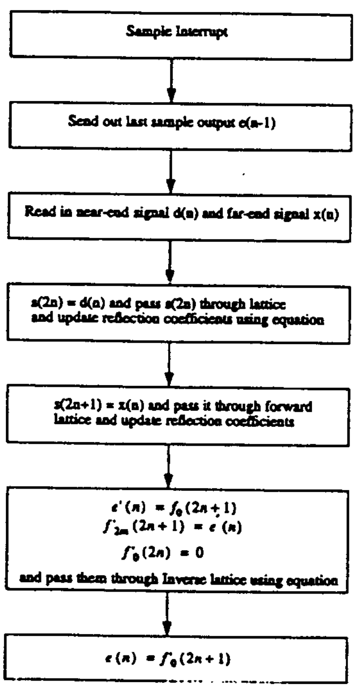

Fig. 4. Flowchart for the proposed AEC.

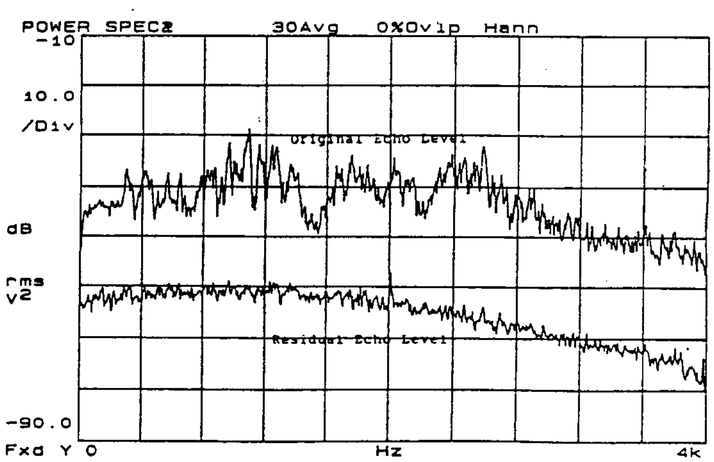

Fig. 6. Real-time results using noise input.