Acoustic Echo Cancellation for Hands-free Telephone

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Abstract : An adaptive algorithm for the acoustic echo canceller is presented. This paper proposes a modified LMS algorithm for the adaptive filter and applys the algorithm to the acoustic echo canceller. An objective of the proposed algorithm is to reduce the hardware complexity. In order to test the performances, a model of the echo path is established, and a program is described. The impulse responses of the echo path have the length of 125msec or more, and then the FIR filter with 1000 taps is required. The results from simulations show that the acoustic echo canceller adopting the proposed algorithm achieves the ERLE of 25dB or more within 1sec. If an echo canceller is implemented with this algorithm, its computation quantity is reduced to two times less than the one that is implemented with the normal LMS algorithm, without the degradation of performances.

1. Introduction

Since the LMS(Least Mean Square) algorithm[1] for the adaptive system was presented, it has been used widely in the various applications. Its reasons are because it is simple and stable. However, to improve its performance, many attempts have been done until the recent. Among the works, there are to increase the convergence speed or to reduce the computation quantity. The sign algorithm[2] as a way for reducing computation quantity lessens dramatically the hardware complexity, but slows down the convergence speed. In this paper, a new adaptive algorithm which reduces the computation quantity without the degradation of the convergence speed, is proposed. This algorithm is applied to the acoustic echo canceller and simulated.

The echo is a phenomenon that the signals radiated from a source go back to the origin after a delay time. The acoustic echo signal is a source of the noises in the communication system and disturbs the normal conversations. The acoustic echo which is generated by the acoustic coupling between the loud-speaker and the microphone, have to be compensated to communicate in full-duplex. In order to test the proposed algorithm, we use the acoustic echo canceller that the impulse response of its echo path is long. The acoustic echo signal is differently generated according to its environment such as the room size. This research is proceeded under the assumption that the typical office room has the size of about 3~4m.

This paper includes the proposed adaptive algorithm in section 2, the application of this algorithm to an acoustic echo canceller in section 3, the simulations and results in section 4. And the conclusion is described in section 5.

2. Proposed Adaptive Algorithm

The identification problem of the unknown system as showed in Figure 1 is solved repeatedly by using adaptive algorithm. The characteristics of the unknown system are adaptively evaluated from the information of the error. We use usually the LMS algorithm as the adaptive algorithm. This algorithm is simple and robust.

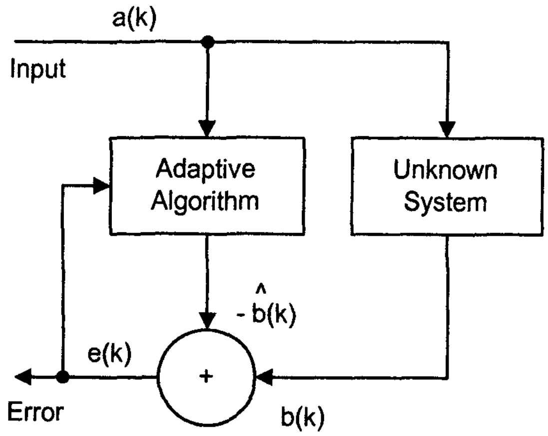


Figure 1. System identification problem

In Figure 1, the signal $a(k)$ is the input to the system and $b(k)$ is the output from the unknown system. Its estimate $\hat{b}(k)$ is generated by convoluting the inputs with the coefficients.

$$\hat{b}(k) = \sum_{i=0}^{N-1} c_i \, a(k-i) \tag{1}$$

where N is the number of taps of the filter. The error $e(k)$ is the value which subtracts the estimate signal from the output of the unknown system. The LMS algorithm for updating the coefficients of the adaptive filter is

$$c_i(k+1) = c_i(k) + 2\mu \, e(k) \, a(k-i) \tag{2}$$

where the constant □ is a step value which adjusts the coefficients.

In the above equation, the operation which multiplys the input by the error is required for all coefficients. If the number of taps is large, its computation quantity also is increased. Therefore, it is difficult that we implement the hardware for the large number of taps. To resolve this problem, we propose a modified LMS algorithm for reducing its complexity.

$$c_i(k+1) = c_i(k) + 2\mu\, e(k)sgn\{a(k-i)\}ma(k) \qquad (3)$$

where $ma(k)$ is an expected value of N inputs. Though there are the multiplicative arithmetics for the updates of the coefficients in the proposed algorithm, it is needed a multiplication for the total number of taps. That is, the operation which multiply the error by the mean of inputs is processed once for the update of all coefficients per a cycle. Since it requires only an additive arithmetic per the coefficient, we can dramatically reduce the computation quantity. Its performances after this modification are the same values as the ones of the LMS algorithm. The convergence time and the ERLE(Echo Return Loss Enhancement) for both algorithm are showed as the results of simulations in section 4.

3. Application to Acoustic Echo Canceller

The general structure of the speaker-phone system [3] is showed in Figure 2.

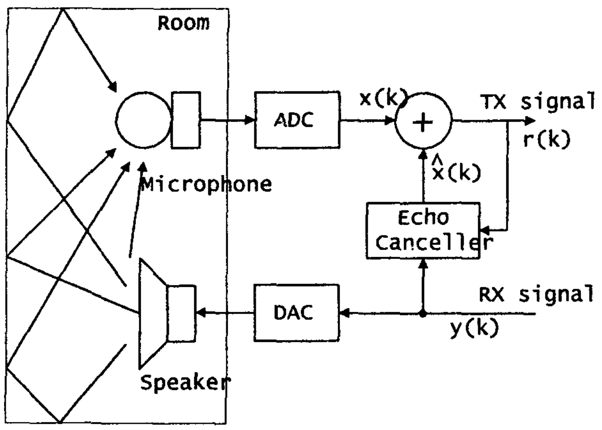


Figure 2. Speaker-phone system

The acoustic echo signal radiated from the speaker is reflected on the surfaces of several objects such as the walls of the room, and then its part is added to the transmit signal through the microphone. The transfer function of the physical path from the speaker to the microphone is evaluated and a replica of the acoustic echo signal due to the far-end talker signal is generated by the acoustic echo canceller. Therefore the acoustic echo canceller prevents the acoustic echo signal from coupling with the input signal by subtracting it from the output signal of the microphone. When the value of the coefficients of the echo canceller are equal to the impulse response of the echo path, the echo signal is canceled completely. That is a principle for compensating the acoustic echo signal.

The characteristics of the echo path are depended on the environment, that is, the size of the room, the materials of the wall, the relative position of the speaker and the microphone, the magnitude of the voice, etc. The acoustic echo signals have the unique properties, compared with the echo signals in the other applications. That is, the impulse response of the echo paths is very long and the properties of the echo path are changed rapidly. The path of the acoustic echo signal due to the far-end talker signal is modeled by a linear system of the sampled impulse responses. Figure 3 shows a general impulse response of the echo path under an assumption that we communicate by means of a hands-free telephone in the closed room of about $20m^3$. The impulse response time of the echo path is 125msec or more. And, if the input signal is sampled at a frequency of 8kHz, the FIR filter must have approximately 1000 taps or more to get the ERLE of 25dB or more.[4]

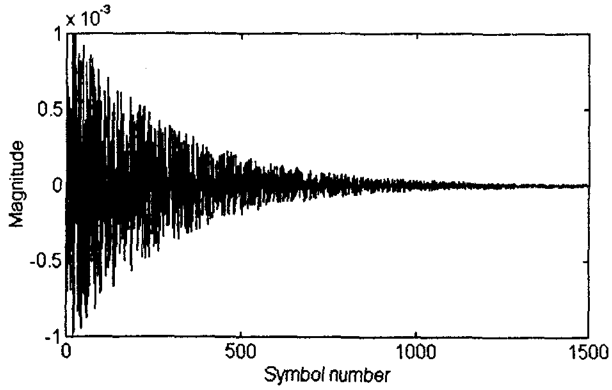


Figure 3. Impulse response of an echo path

In case that an acoustic echo canceller is implemented with 1000 taps, the required computation quantities are summarized at Table 1. It shows that the number of multiplication of the proposed algorithm, which burden the hardware with heavy computations, is two times less than one of the LMS algorithm without increasing the memory space.

Table 1. Comparison of the computation quantities

Items	LMS	Modified LMS
Multiplication	2000	1001
Addition	2000	2004
Memory[word]	2000	2002

4. Simulations and Results

To check the performances of the proposed algorithm, a C-program for the speaker-phone system is described. Generally, the impulse response of the echo path vanishes exponentially with the positive or negative value according to elapsing the time.[5] The used model of the echo path is represented by the following equation.

$$h_i = K\, r_i\,(1.00346)^{-i} \quad \mathit{for}\ i=0,1,\ldots,N-1 \tag{4}$$

where K is a constant number so that the sum of the total power of h_i keeps unvarying although the number of the tap changes. It is the worst case that the sum of the power of h_i is unity and means what the output of the speaker is introduced into the microphone without any attenuation. The input signal to the test system is generated by sampling the random voice signal with the frequencies of 300□ 3,400Hz at a frequency of 8KHz. The echo canceller is formed of the FIR filter with the coefficients of 1,000 taps. The ERLE and the convergence time are gotten through the simulations. First, the simulations in the initial tracking mode are proceeded and its convergence curves are showed in Figure 4. In this figure, the continuous curve indicates the ERLE by the normal LMS algorithm and the dotted curve indicates ERLE by the proposed algorithm.

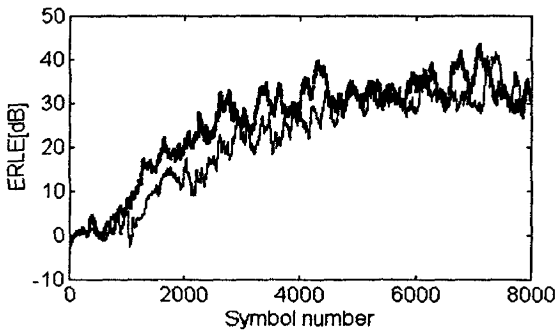
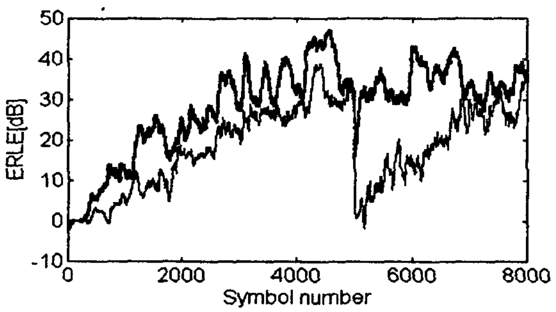


Figure 4. Initial convergence curves

As what is viewed from the above curves, the acoustic echo canceller adopting the proposed algorithm displays almost the same result as one adopting the normal LMS algorithm in terms of the convergence speed and the ERLE.

Figure 5 plots the convergence curves of the echo canceller for an impulsive noise at the steady state after the convergence. The continuous curve in the figure indicates the ERLE by the normal LMS algorithm and the dotted curve indicates ERLE by the proposed algorithm. It is turned out from these curves that the proposed algorithm also indicates the same performance as the LMS algorithm



for the reconvergence mode initiated by an impulsive noise.

Figure 5. Curves with a change of the impulse response after convergence

5. Conclusion

In this paper, an adaptive algorithm for reducing the hardware complexity is proposed. To test performances of the algorithm, a C-program for the acoustic echo canceller was written and simulated. In case that we communicate by using a speaker-phone system in the room of $20m^3$, the results of simulations show that the echo canceller gets the ERLE of 25dB or more within 1sec. And the computation quantity needed by this algorithm is almost two times less than the one needed by the normal LMS algorithm.

References

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