# AN ADAPTIVE FILTER COEFFICIENTS ADJUSTMENT ALGORITHM STABLE AGAINST REFERENCE SIGNAL POWER FLUCTUATION AVAILABLE FOR ACOUSTIC ECHO CANCELLER SYSTEMS

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**Abstract**

The ERLE (echo return loss enhancement) iterates greatly up and down, if the adaptive filter coefficients are continuously adjusted in disregard of the reference signal power fluctuation. This paper presents a method of always maintaining the specified ERLE, even when the adjustment is continued in voiceless noise terms. The method is based on the 'summational' NLMS (normalized least mean square) algorithm in which the coefficients are updated after the reference signal norm, and the product of the residual echo and the reference signal have been summed up for continuos iterations (a block). The SNLMS algorithm can keep the ERLE at the specified level, if the coefficients are updated after the summed norm has been amounted to a value which was evaluated from a given surrounding noise power.

## 1. INTRODUCTION

Acoustic echo canceller inevitably uses a speech signal as reference to estimate the coefficients of the adaptive filter, which makes a replica of the echo detected by a microphone. The speech signal power fluctuates extremely, and moreover becomes even silent. The ERLE provided by the acoustic echo canceller should be stable against such an unstable reference signal power. The ERLE in the NLMS algorithm [1], is evaluated as a function of a given step gain and the power ratio (SNR) of the reference signal and the surrounding noise [2]. The low power reference signal will quickly reduce the ERLE, if no preventation is added to the estimation algorithm.

Most practical designs employ a fixed step gain and a method that suspends an estimation of the coefficients if the reference signal power is lower than the limit calculated from a given ERLE. The suspension will naturally increase the convergence delay, and besides, the reference signal with higher power than the limit will provide more ERLE than the satisfactory level. The suspension time can be reduced by choosing a small step gain. However, such a small step gain slows down the convergence rate, and also makes the ERLE unnecessarily large for the higher reference signal power.

The above relation between the reference signal power and the ERLE seems to suggest that the control of proportioning the step gain to the reference signal power is a good solution. The NLMS algorithm is featured by the normalization which corresponds to a division by the reference signal power. The devision also includes the step gain in the dividend. Apparently, such step gain control cancels the effect of the normalization, and thus the control makes the NLMS algorithm result in the LMS algorithm. This paper presents a method of keeping the ERLE at a prescribed level in spite of the reference signal power fluctuation, without controling the step gain.

## 2. THE DIFFICULTY IN CONTROLING THE STEP GAIN

Acoustic echo canceller design requires that the ERLE should be kept at a higher level than the limit over which howling isn't generated. The convergence value of the ERLE is provided by the step gain evaluated under a fixed SNR. In acoustic echo canceller which uses actual speech as reference signal, unfortunately, the SNR changes extremely even if the surrounding noise power is constant. These changes make the ERLE go up-and-down. Most practical designs employ a method that suspends the estimation of the coefficients againt the changeful SNR. The estimation is suspended, if the reference signal power becomes lower than the limit over which the

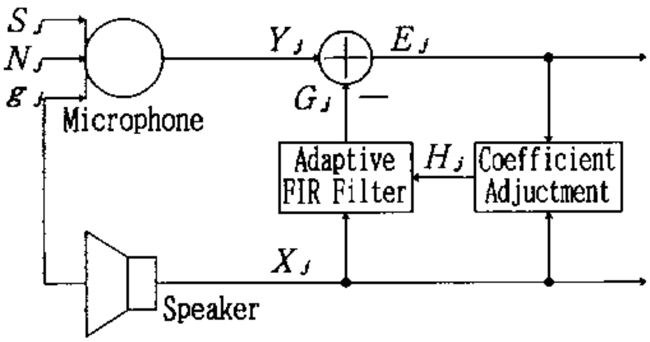


Fig. 1 An acoustic echo canceller system.

specified convergence value is obtained.

The suspension will naturally increase the convergence delay. The steadiest solution to reduce the suspension time, is to continue the estimation independently of the reference signal power fluctuation. However, to continue the estimation, by means of providing the NLMS algorithm the step gain proportional to the instantaneous reference siganl power, does not become a solution. Such step gain control restores the NLMS algorithm to the LMS algorithm.

The convergence value is also expressed as a function of the SNR. The expression suggests that another solution is derived from adjusting the SNR. The adjustment can be accomplished in a well-known version of the NLMS algorithm [3], which is called 'summational' NLMS algorithm in this paper, for convenience' sake.

## 3. 'SUMMATIONAL' NLMS ALGORITHM

In a practical system, the SNLMS algorithm is programed, by using the symbols specified in Fig. 1, as follows:

$$H_{n+1}(m)=H_n(m)+K A_n(m)/P_n \qquad (1)$$

$$A_n(m)=\sum_{j=nJ+1}^{(n+1)J} E_j X_j(m) \qquad (2)$$

$$P_n=\sum_{j=nJ+1}^{(n+1)J} \left[\sum_{j=1}^{I} E_j X_j(m)\right], \qquad (3)$$

where $H_n(m)$ is the m'th tap coefficient of the FIR (finite impulse response) filter in the n'th block whose length is J. And other symbles are defined as follows:

- K a step gain defined in the SNLMS algorithm,
- $E_j$ the residual echo,
- $X_j$ the reference (far-end talker's signal),
- I the tap length of the FIR filter.

The summation reduces the surrounding noise component, which is included in the numerator, by the average effect proportional to the block length. This reduction enhances the SNR, and as a result, it will increase the convergence value of the ERLE. Thus, the SNLMS algorithm makes us expect that the convergence value can be controlled by the adjustment of the block length in addition to the step gain.

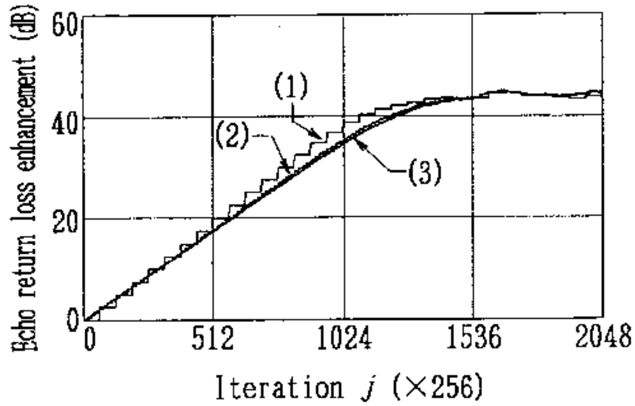


Fig. 2 An example confirming the relation between the step gains and the block length.
(1) K=128 and J=16384 (SNLMS)
(2) K=32 and J=4096 (SNLMS)
(3) $\mu$=1/128 (NLMS)

These two variables are related with a different step gain, $\mu$, which is used in the NLMS algorithm, as follows:

$$\mu=K/J. \qquad (4)$$

Figure 2, which is calculated for I=512, is an example to confirm the above relation. The three convergence properties are expressed with the estimation error vector norm. The norm is calculated by two different white noise sequences used as the reference signal and the surrounding noise, respectively. Where the power ratio is 20 dB. The example shows that the SNLMS algorithm provides almost the same convergence property as the NLMS algorithm where the relation is kept steady.

Figure 3 is also an example showing that a fixed convergence value of the ERLE can be obtained, if the block length is adjusted so as to be inversely proportional to the SNR. In Fig. 3, the three block lengths of 2, 8 and 32 give the same convergence value for the different three SNR's of 20, 14 and 8 dB. The adjustment is equivalent to shorten and lengthen the block length, so that the 'summational' normalization power $P_n$ will have a common value for the different reference

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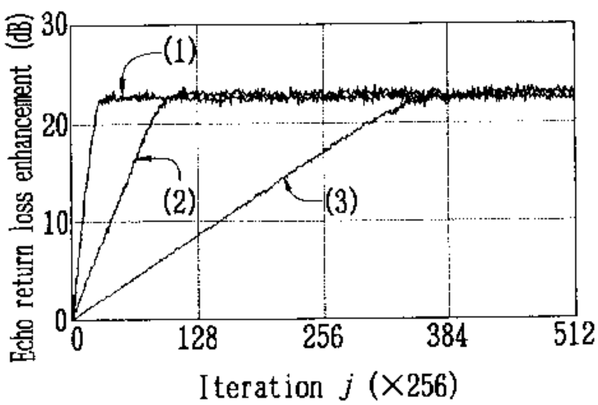


Fig. 3 convergence properties provided by the SNLMS algorithm with a function of adjusting the block length so as to be inversely proportional to the reference signal power.

(1) J=2 (SNR=20 dB)
(2) J=8 (SNR=14 dB)
(3) J=32 (SNR=8 dB)

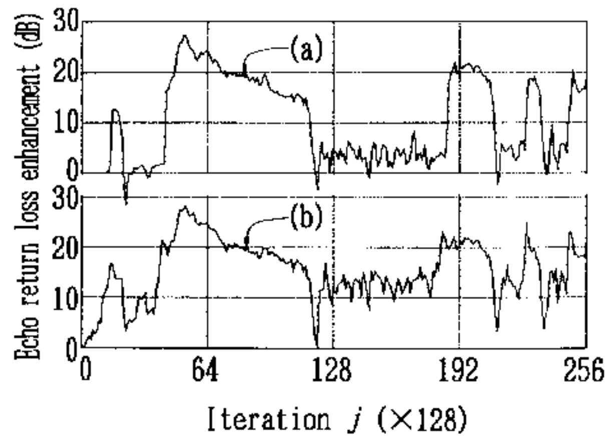


Fig. 4 The ERLE obtained by actual speech.
(a) NLMS with the suspending method
(b) SNLMS with the block length adjustment

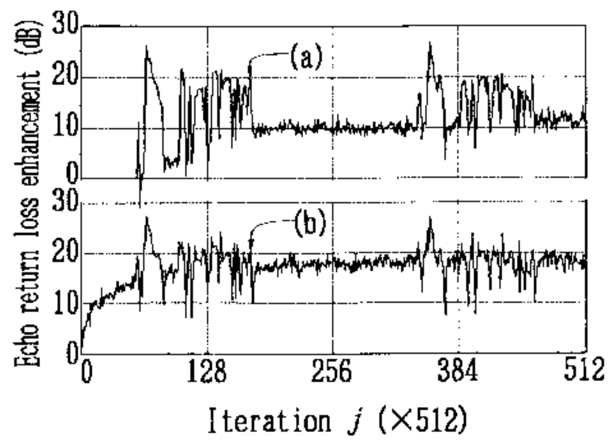


Fig. 5 ERLE obtained in voiceless noise terms.
(a) NLMS with the suspending method
(b) SNLMS with the block length adjustment

signal powers, but only the fixed surrounding noise power is fixed. Apparently, the block length adjustment for fixing the normalization power against the reference signal power fluctuation, does not waste the normalization shown in Eq.(1), unlike the step gain control.

## 4. SIMULATIONS BY USING ACTUAL SPEECH

Figure 4 is an example demonstrating that the SNLMS algorithm with the block length adjustment is applicable to the acoustic echo canceller. In this example, the adaptive filter coefficients are updated after the block has been lengthened until the normalization power $P_n$ exceeds a criterion value $P_0$ requisite for keeping the ERLE at a prescribed level. The criterion power $P_0$ is also evaluated as the value which keeps the convergence value of the ERLE at 20 dB for a SNR of about 0 dB.

The SNLMS algorithm with the block length adjustment provides faster and higher ERLE in comparison to that of the conventional NLMS algortihm ($\mu=1$). The conventional algorithm suspends the estimation when the reference signal power is lower than the limit over which the ERLE is obtained. The higher ERLE is obtained in the middle voiceless noise term of the example. This shows the advantage of the block length adjustment.

The adjustment makes it possible to continue the estimation of the coefficients even in voiceless noise terms, if the criterion power $P_0$ is evaluated so that a ERLE of 20 dB will be kept in all the terms. Figure 5 is an example of confirming the advantage of continuing the estimation.

In an actual acoustic echo canceller system, the far-end talker's voice provides various power levels. The low power talker's voice will naturally increase the suspension time for the estimation system which uses the conventional NLMS algorithm. In such a system, it is desirable to continue the estimation, even if the convergence rate decreases for low power reference signals. The SNLMS algorithm with the block length adjustment presents such a solution.

## 5. EVALUATION OF THE CRITERION POWER $P_0$

Finally, this paper presents a method of automatically evaluating the criterion power $P_o$.

The criterion power $P_o$ is provided by a prescribed convergence value of the ERLE and a given surrounding noise power. In a practical system, the noise power is not evaluated beforehand, and besides may change during its operation. The system inevitably requires a function that estimates the surrounding noise power $P_n$ and then adjusts the criterion power $P_o$.

The surrounding noise power $P_n$ is detected as the minimum power of the acoustic echo canceller output which is measurable in the actual system. The measurable signals, hoever, include the near-end talker's voice which disturb the estimation. It is necessary for the detection to maintain the minimum level independent of the disturbance. The detection is accomplished by a low pass filter with both a long rise time and a short fall time.

The next procedure of deriving the criterion power $P_o$ is to evaluate the gain of the path (from the reference signal input point, through the acoustic path, to the output of the echo canceller). The gain $C_o$, which is, for example, a level to prevent howling from being generated, is derived from the signal level diagram of hands-free communication system, which is designed beforehand.

The gain $C_o$ is correspondent to the estimation error of the adaptive filter coefficients, which is equal to the power ratio,

$$C_o = P_n / P_x \tag{5}$$

between the reference signal and the surrounding noise power, when the step gain is unity. Equation (5) is writen as follows:

$$P_x = P_n / C_o, \tag{6}$$

if the only known powers are put together on the right side. The power $P_x$ gives the reference signal power requisite for keeping the gain $C_o$ at a pescribed level against the surrounding noise power.

Thus, the criterion power $P_o$, correspondent to the reference signal norm, can be expressed as I times the power $P_x$ given by Eq.(6), that is,

$$P_o = I\, P_n / C_o. \tag{7}$$

## 6. CONCLUSION

In this paper, we presented a method of keeping the ERLE at a satisfactory level independently of the reference signal power fluctuation. The method is based on the SNLMS algorithm in which the adaptive filter coefficients are updated after the reference signal norm, and the product of the residual echo and the reference signal have been summed up for continuos iterations. The method of adjusting the block lengh so as to be inversely proportional to the reference power, makes it possible to continue the estimation even in voiceless noise terms. We also presented a method for evaluating the criterion power which keeps the ERLE at a satisfactory level.

The basic idea shown in this paper, can also be applied to the 'individually normalized' LMS algorithm [4] and the 'polarized-x' algorithm [5] suitable for fixed point processing.

### References

[1] J. Nagumo and A. Noda, "A learning Method for System Identification," IEEE Trans. Automat. Contr., vol. AC-12, no. 8, pp.282-287, June 1967.

[2] H. Itakura and Y. Nishikawa, "On Some Characteristics of an Echo Canceller using a Learning Identification Algorithm," Trans. IECE Japan, vol. J60-A, No. 11, pp.1015-1022, Nov. 1977.

[3] G. A. Clark, S. K. Mitra and S. R. Parker, "Block Implementation of Adaptive Digital Filters," IEEE Trans. Circuits Syst., vol. CAS-28, no. 6, pp.584-592, June 1981.

[4] K. Fujii and J. Ohga, "'Individually Normalized' LMS algorithm suitable for Fixed Point Processing and Its Application to Active Noise Control," Trans. IECIE Japan, vol. J78-A, No. 6, pp.669-677, June 1995.

[5] K. Fujii and J. Ohga, "Some Studies on the 'Polarized-x' Algorithm Available for the Fixed Point Processing," Trans. IECIE Japan, vol. J79-A, No. 1, pp.11-18, Jan. 1996.