An Adaptive Echo Canceller with High Convergence Rate Using Nonlinear Distortion

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1. Introduction

In telephone networks with a long round-trip delay, such as satellite circuits and TV conference systems, speech conversations would be greatly disturbed by echo. Recently, the problem of echo has arisen in mobile communications and VoIP (Voice over IP). As shown in Fig. 1, in a long-haul telephone connection including satellite communication links or submarine cables, local loops near subscribers are 2-wire and long transmission trunks between the local loops are 4-wire for the purpose of signal amplification. Hybrid coils connect the 2-wire loops to the 4-wire trunks. The hybrid coils are designed to match its impedance to that of the 2-wire loops, but complete impedance matching cannot always be expected because there is a variety of the 2-wire loops impedance.

A mismatching results in echo, so that the received signal coming to the 4-wire input terminal leaks to the 4-wire output terminal. This disturbing influence of the echo increases with an increase in round-trip delay of networks.

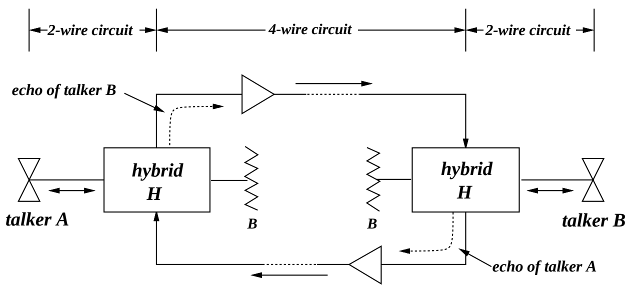

Figure 1: A typical long distance telephone circuit.

Echo cancellers have been researched and developed in many countries to reduce the echo influence. The echo canceller artificially generates a replica of the echo (called a pseudo echo), to cancel the echo by force without disconnecting the echo return path. There are off-line and on-line procedures as a method of identifying characteristics of the echo path in order to generate a replica of the echo. With the off-line type, the echo path characteristics are measured and stored just after a communication starts. Hardware for this method may be simple, but it does not always give high measurement accuracies due to channel noises, and does not yield satisfactory echo cancellation because it neglects time variations of the path characteristics. The on-line type (adaptive type), which updates estimates of the path characteristics time after time, is preferable.

Many digital algorithms have been examined for the on-line method [1]-[6]. The simplest algorithms often used are the so-called NLMS (Normalized Least Mean Square) and its variations. They are advantageous in hardware implementation and give good canceling performance for a white Gaussian signal, but usually worse canceling performance for a real voice signal. On the other hand, the echo cancellers using Kalman filter [7][8] and that using RLS (Recursive Least Square) [9] have been proposed. These have good cancellation performance independent on the statistical characteristics of the input signal. The performance of the echo cancellers using these algorithm is very good, but the amount of memory and operation required becomes fairly large. For reduction of the difficulty arising from these issues in hardware implementation and for high convergence rate, the de-correlation of input speech signal and the addtion of white Gaussian to input signal will be efficient.

In this study, we propose an NLMS echo canceller using nonlinear distortion. This method is expected to have high convergence characteristics with a simple procedure.

2. NLMS algorithm

The circuit of echo canceller using NLMS algorithm is shown in Fig. 2. It is assumed here that the echo canceller is composed entirely of digital circuits. It is also assumed that the echo path has linear and time-invariant transmission characteristics.

This work was suported by ICF (International Communications Foundation).

NLMS is represented by Eqs. (1)-(6).

$$\mathbf{h}_N(k+1) \;=\; \mathbf{h}_N(k) + \alpha \frac{\mathbf{x}_N(k)}{||\mathbf{x}_N(k)||^2} e(k) \qquad (1)$$

$$e(k) \;=\; d(k) \;-\; y(k) \;+\; n(k) \qquad (2)$$

where

$$\begin{aligned} \mathbf{x}_N(k) &= (x(k-1), x(k-2),, \cdots, x(k-N)) && (3)\\ \mathbf{h}_N(k) &= (h_1(k)\ , h_2(k),,\ \ldots,\ h_N(k)) && (4)\\ d(k) &= \mathbf{w}_N \mathbf{x}_N(k)^T && (5)\\ y(k) &= \mathbf{h}_N(k)\mathbf{x}_N(k)^T && (6) \end{aligned}$$

$x(k)$: *input signal*
$e(k)$: *error*
$d(k)$: *real echo*
$y(k)$: *pseudo echo*
$n(k)$: *noise (near-end talker's speech signal)*
α : *step gain*
$\mathbf{h}_N(k)$: *estimated impulse response*
$\mathbf{w}_N$: *impulse response of unknown system*
N : *tap length*

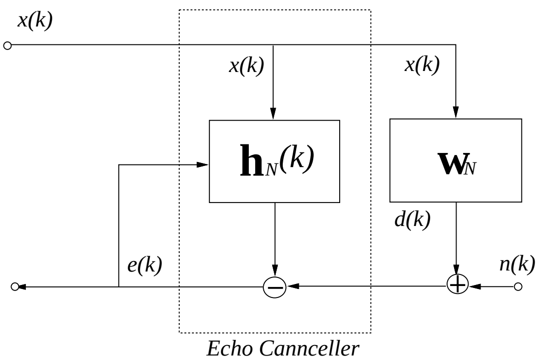

Figure 2: Circuit of echo canceller.

3. Nonlinear distortion

Generally, speech signals are colored or correlated. Therefore, the convergence rate becomes slow, when speech signals are input into the NLMS. So we use the input signal which is more whitened by adding a nonlinear distortion. Figure 3 shows a block-diagram of nonlinear distortion processor. We use the follwing three kinds of nonlinear distortion. Here, $t(k)$ and a_j are nonlinear distortion and constants, respectively.

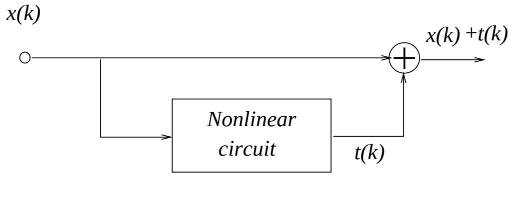

Figure 3: Nonlinear distortion processor.

(1) Distortion 1

$$t(k) = \begin{cases} a_1 & (a_1 \leq x(k)) \\ x(k) & (-a_1 < x(k) < a_1) \\ -a_1 & (-a_1 \geq x(k)) \end{cases}$$

This distortion is shown in Fig. 4.

(2) Distortion 2

$$t(k) = \begin{cases} |x(k)| & (-a_2 < x(k) < a_2) \\ a_2 & (a_2 \leq x(k)\,,\ -a_2 \geq x(k)) \end{cases}$$

This distortion is shown in Fig. 5.

(3) Distortion 3

$$t(k) = \begin{cases} a_3 & (x(k) > 0) \\ 0 & (x(k) = 0) \\ -a_3 & (x(k) < 0) \end{cases}$$

This distortion is shown in Fig. 6

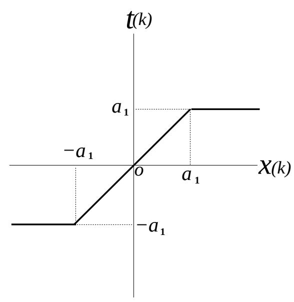

Figure 4: Distortion 1.

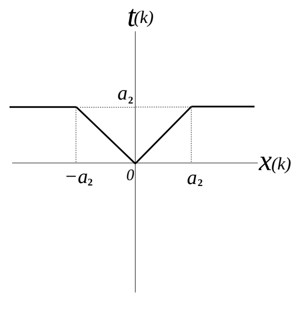

Figure 5: Distortion 2.

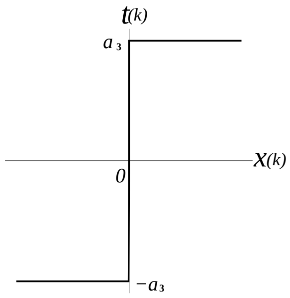

Figure 6: Distortion 3.

We carry out computer simulations using three different values of a_j for each nonlinear distortion.

4. Computer simulations

Receive-in speech signal as shown in Fig. 7 is used in the simulations. In this study, the channel noise is not added. In addition, the tap length N and the step gain α of the NLMS are set to 320 and 1, respectively. Table 1 shows values of the constants a_j and the name of each distortion. The names in Table 1 are used in Figs. 8-11.

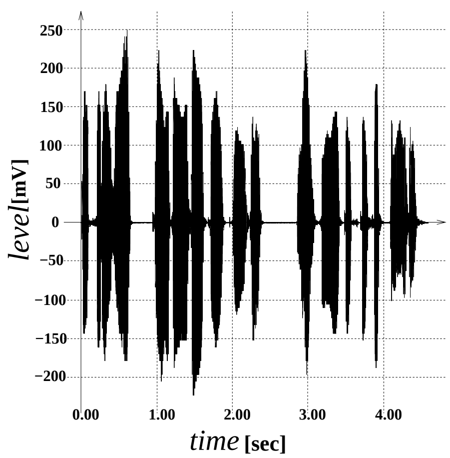

Figure 7: Receive-in speech signal.

Table 1: Nonlinear distortion

Distortion 1		Distortion 2		Distortion 3	
a_1[mV]	name	a_2[mV]	name	a_3[mV]	name
10	$1-1$	10	$2-1$	4	$2-1$
20	$1-2$	20	$2-2$	5	$3-2$
30	$1-3$	30	$2-3$	6	$3-3$

4.1. Erle and norm

In order to measure the cancellation performance and the convergence rate, *erle* and *norm* are used which are defined in Eq. (7) and Eq. (8).

$$erle = 10 \log_{10} \frac{\sum d^2(k)}{\sum (d(k) - y(k))^2} \tag{7}$$

$$norm = 10 \log_{10} \frac{\|\mathbf{w}_N - \mathbf{h}_N(k)\|^2}{\|\mathbf{w}_N\|^2} \tag{8}$$

Figures 8-10 show *erle* and *norm* for the three distortions. Compared with the normal NLMS argorithm, *erle* for all the cases are improved a little but *norm* are improved 2 to 27dB. From this result, we can say that convergence rate increases for all the cases. Moreover, Distortion 3 achieves the best improvement in *norm*.

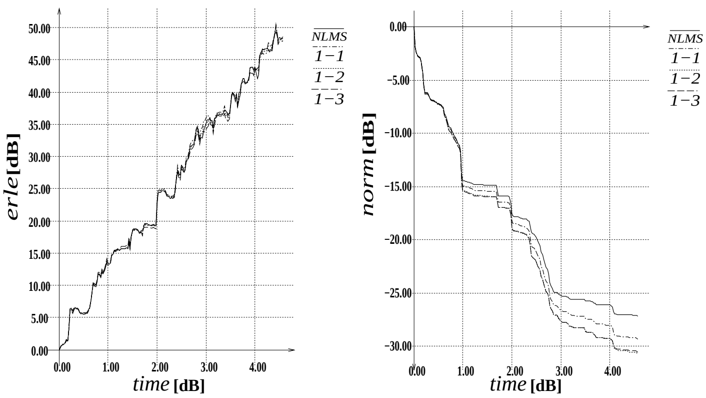

Figure 8: Estimation of Distortion 1.

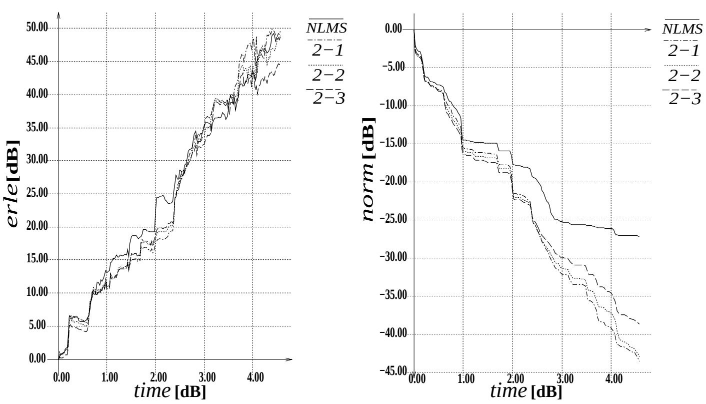

Figure 9: Estimation of Distortion 2.

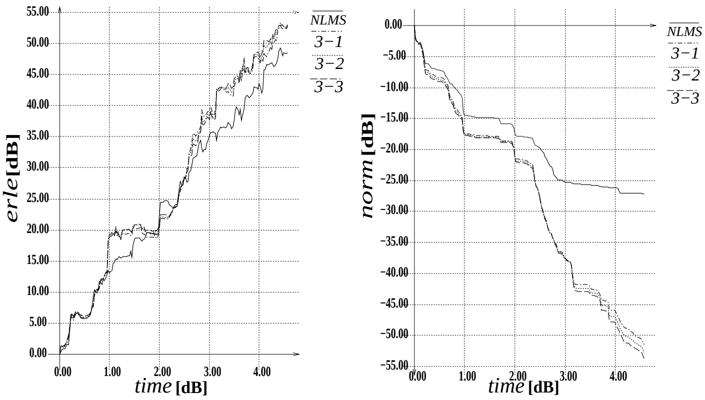

Figure 10: Estimation of Distortion 3.

4.2. Speech quality assessment

We conduct a speech quality assessment for each input signal with nonlinear distortion. Figure 11 shows MOS (Mean Opinion Score) of each input signal. MNR (Modulated Noise Reference) is recommended as a method which makes standard speech signal [10].

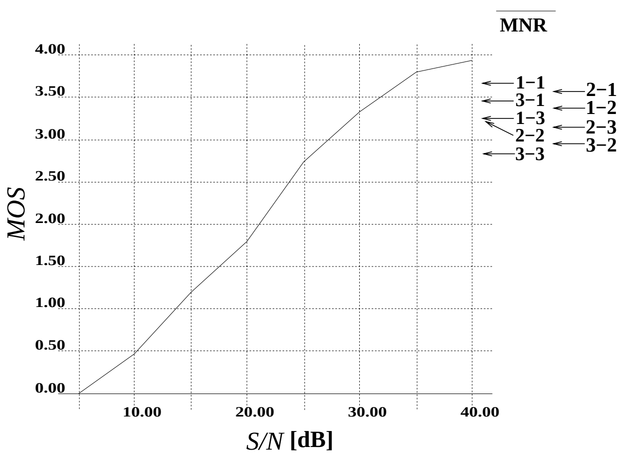

Figure 11: Comparison with MNR.

All signals with nonlinear distortions are assessed to be better than MNR-25[dB]. Their convergence rates are improved in all of the cases. Auditive deteriorations increase with larger nonlinear distortion only in Distortion 3, but *norm* in this case is better than those in the other cases. From results, it is shown that the convergence rates are improved with very small or no auditive deterioration in all of the cases, and, Distortion 3 achieves the best performance among the three kinds of nonlinear distortions.

5. Conclusions

The NLMS algorithm had a problem where convergence rate becomes slow when speech signals are input. This study has tried to improve the convergence rate by adding nonlinear distortion. We examined auditive deterioration of the input signal by adding nonlinear distortion. The results show that the convergence rate is improved without deterioration of speech quality. In addition, the proposed method provides a similar result for other speech signals.

In a future work, we need to apply the proposed method to detect the double-talking in the echo cancellers and to suppress the hawling phenomenon in TV conferences.

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