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Shun-Chieh Chang
Department of Electrical Engineering
National Taipei University of Technology
Taipei, Taiwan, ROC
t6319011@ntut.edu.tw

Shaw-Hwa Hang
Department of Electrical Engineering
National Taipei University of Technology
Taipei, Taiwan, ROC
hsf@ntut.edu.tw

Abstract

In this paper, the dynamic least mean square(DLMS) algorithm is firstly proposed to reduce the computation of LMS. In the DLMS architecture, the redundant coefficients of LMS can be grouped automatically. The coefficient number of filter and the computation are reduced. The DLMS-based acoustic echo cancellation (AEC) is employed to evaluate the performance of DLMS. The experimental results confirm the good performance in DLMS algorithm.This paragraph describes your major work corresponding to the Chinese part in your paper.

Keywords: not exceeding five.

1. Introduction

In the past, the LMS (Least Mean Square) algorithm [*] employed the FIR (Finite Impulse Response) filter and the adaptive algorithm to simulate the echo canceller. Because the characteristics of an acoustic echo path are very non-stationary, the adaptive feedback cancellation algorithm is the most effective method for the acoustic echo cancellation. However, the adaptive algorithm needs computation dramatically while the length of filter was too long. But the filter’s length is decided by the acoustic echo delay and can’t be shorter than acoustic echo delay.

In order to improve the performance of echo cancellation, the normal LMS (NLMS) was proposed to balance the different gain of signal. The recursive least square(RLS) algorithm was proposed to converge quickly. The multi-dimension frequency(MDF) was proposed to reduce the computation of echo cancellation. However, the MD algorithm is too complex. It is not easily implemented.

Generally, the LMS is a simple algorithm for implementation. Moreover, it can adaptively track the dynamic echo path. Thus many consumer ICs for communication adopt the LMS algorithm. However, the heavy computation of LMS make the computation power and cost not better for market. Nevertheless, the length of LMS filter must be longer than echo path for efficient echo cancellation. The length of echo path is dynamic and un-predictive. Thus, the default length of LMS filter must be set as a reasonable large value for unknown and un-predictive echo path. This will worse the computation issue of LMS algorithm,

In the paper, the dynamic LMS is proposed to alleviate the computation issue of LMS algorithm. In the general case, the LMS filter is long enough for the long echo delay. Many redundant coefficients of LMS filter will waste the CPU time. The DLMS will automatically group the redundant coefficient and reduce the computation of LMS. The DLMS will keep the speech quality of LMS algorithm. But the computation of LMS will be reduced dramatically.

In the following sections, the detail description of DLMS will be given in section II. The experimental results will be listed and analyzed in section III. The summary and conclusion will be given in the last section.

2. Content

The LMS algorithm is popular for echo cancellation. However, due to the dramatic computation requirement for long echo trip, the DLMS algorithm is proposed to reduce the redundant coefficient and achieve the goal of computation saving. The DLMS algorithm will group the un-influence and stable coefficients automatically. Each group associates with one coefficient only. The Fig.1 shows the filter architecture of LMS and DLMS. The filter length is 10 for both LMS and DLMS. The group number is 4 for the DLMS. Obviously, the last 5 coefficients of LMS can't track the echo path exactly. These 5 coefficient are un-useful. Thus the last 5 coefficients are grouped into one coefficient for computation issue.

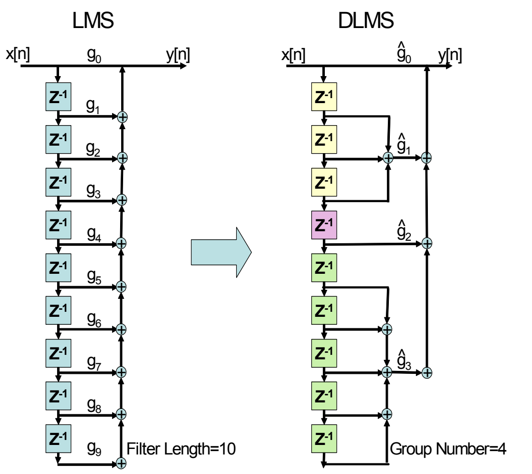

Figure.1 The filter architecture of LMS and DLMS.

In the LMS mode, if the neighbor coefficients are fall into stable state and with the same value, these coefficients will be grouped and replaced by one coefficient and the DLMS mode will be started. Moreover, in the DLMS mode, if the grouped coefficient is unstable, the LMS mode will be started and the coefficients of LMS architecture will be recovered. The detail description and flow-chat of decision between LMS and DLMS is depicted in the Fig.2.

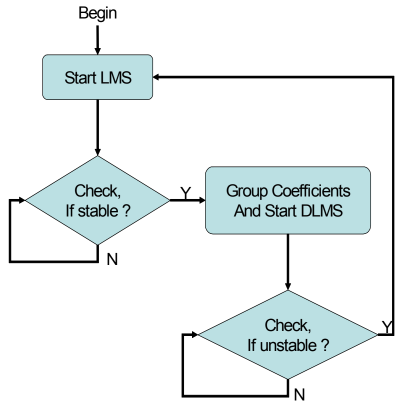

Figure 2. The flow-chat of decision between LMS and DLMS.

In the Fig.2, The LMS algorithm is started firstly and the stable state is determined by the following rule.

$$\left| g_i - c \right| < \eta \quad , \quad i \in [a, b],$$

The c is constant and the η is a small value. The default value of η is 0.001. If the above rule is keep for a fixed time-interval, the grouping method on coefficient and the DLMS algorithm will be started. The coefficients $[g_a, g_{a+1}, \ldots, g_{b-1}, g_b]$ will be grouped and replaced by one coefficient $\hat{g}_k$ only.

In the DLMS mode, the unstable state will also be checked. The decision rule of unstable state is done on the grouped gain $\hat{g}_i$. If the following rule is satisfied, the grouped gain must be divided and recovered into the LMS architecture.

$$|\hat{g}_k[n+1] - \hat{g}_k[n]| > \eta$$

In the DLMS mode, each group of coefficient is associated with one grouped coefficient $\hat{g}_i$. The output of DLMS filter is defined as below:

$$y[n] = \sum_{k=0}^{M-1} \hat{g}_k \sum_{m=-L_k/2}^{L_k/2-1} x[n - d_k - m]$$

The $\hat{g}_k$ is the grouped coefficient at *k-th* group. The M is the group number. The L_k is the coefficient number at the *k-th* group. The d_k is the average delay at the *k-th* group. It is a integer value. The following conditions are satisfied in the DLMS mode.

$$M \le N$$

$$L_0 + L_1 + ... + L_{M-1} = N$$

The adaptive equation of DLMS algorithm is derived and listed as below:

$$\hat{g}_k[n+1] = \hat{g}_k[n] + 2\mu e(n)\overline{X}_k$$
$$k = 0,1,2,...,M-1$$

The $\overline{X}_k$ is the summation at the k-th group.

$$\overline{X}_k = \sum_{m=-L_k/2}^{L_k/2} x[n - d_k - m]$$

The adaptive equation of LMS and DLMS algorithms is concluded and listed in the Table 1. The *N* is filter length of LMS and the *M* is the group number of DLMS.

Table 1. The adaptive equation of LMS and DLMS.

Algorithm	Equation
LMS Tap:N	1. $y[n] = \sum_{i=0}^{N-1} g_i x[n-i]$ 2. $e[n] = (d[n] - y[n])$ 3. $g_i[n+1] = g_i[n] + 2\mu e[n]x[n]$ i=0,1,2,…,N-1
DLMS Tap:N Group:M	1. $y[n] = \sum_{k=0}^{M-1} \hat{g}_k \sum_{m=-L_k/2}^{L_k/2-1} x[n - d_k - m]$ 2. $e[n] = (d[n] - y[n])$ 3. $g_k[n+1] = g_k[n] + 2\mu e[n]\overline{X}_k$ k=0,1,2,…,M-1

Moreover, the computation requirement for the LMS and DLMS algorithm is analyzed and listed in the Table.2.

Table.2 The computation requirement of LMS and DLMS.

Algorithm	Multiply(*)	Adder(+)
LMS Tap:N	1. N 2. 0 3. N+1 Total: 2N+1	1. N-1 2. 1 3. N Total: 2N
DLMS Tap:N Group:M	1. M 2. 0 3. M+1 Total: 2M+1	1. M-1+(N-M) 2.1 3. M Total: N+M

The computation requirement of Table.2 is associated with the adaptive equation of Table.1. There are 3 equations within the LMS and DLMS algorithm. The equation 3 of DLMS needs *M+1* multiplies and *M* adders. In the DLMS algorithm, totally *2M+1* and *N+M* are needed for multiplication and adder, respectively.

The computation requirement and percentage of computation saving for real case is listed as below. In the case of *(N,M)=(20,3)*, the percentage of computation saving is 62.5%. It is reasonable good.

Table.3 The computation requirement for some real cases.

(N,M)	LMS		
	*	+	Total
(20,3)	41	40	81
(32,3)	65	64	129
(32,5)	65	64	129
(40,3)	81	80	161
(100,5)	201	200	401
(N,M)	DLMS		
	*	+	Total
(20,3)	7	23	30(37.5%)
(32,3)	7	35	42(32.0%)
(32,5)	11	37	48(36.7%)
(40,3)	7	43	50(31.1%)
(100,5)	11	105	116(28.9%)

3. Experimental Results

In order to evaluate the performance of DLMS, the acoustic echo cancellation is employed. Three cases are studied and experimental results are listed. The first case is a single echo path. The echo equation is defined as below:

$$e[n]=0.35x[n-10], n=0,1,2,...,600$$

The $e[n]$ is the acoustic echo signal. The filter length of LMS is set as 40. The curve of each coefficient for the traditional LMS mode is depicted in Fig.3. These coefficients of LMS will converge at the time index *n=230*. These coefficients will keep in stable value after *n=230*.

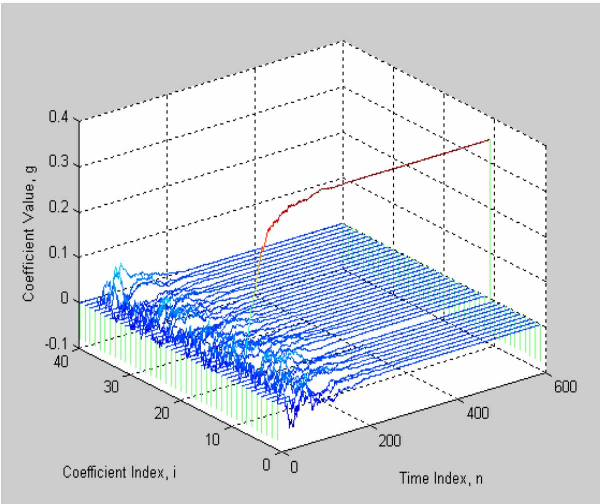

Fig.3 The curve of coefficient of LMS(Case #1).

In the Fig.4, the DLMS algorithm is employed. These coefficients after *n=230* will fall into the stable state. Thus, three groups are determined and only 3 grouped coefficients are need for the DLMS mode.

In this case, the value of *(N,M)=(40,3)* is obtained and the computation saving with *68.9%* is achieved.

In the second case, the echo path is defined as below:

$$e[n]=\begin{cases}0.25x[n-10]+0.65x[n-25], n=0\sim 500\\ 0.5x[n-15]+0.35x[n-30], n=500\sim 1000\end{cases}$$

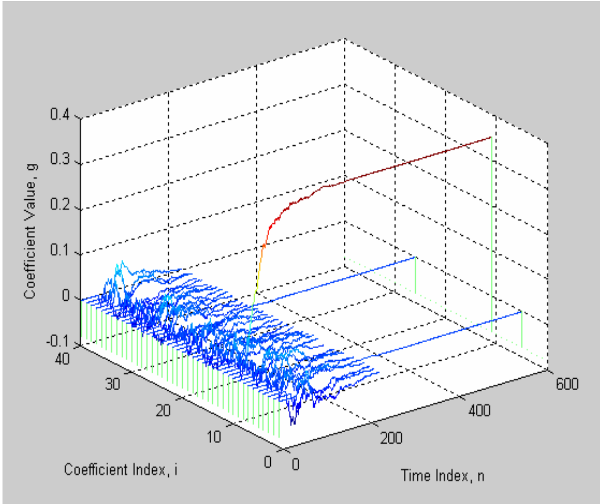

Fig.4 The curve of coefficient of the DLMS(Case #1).

The curve of coefficient LMS and DLMS are depicted in Fig.5 and Fig.6, respectively.

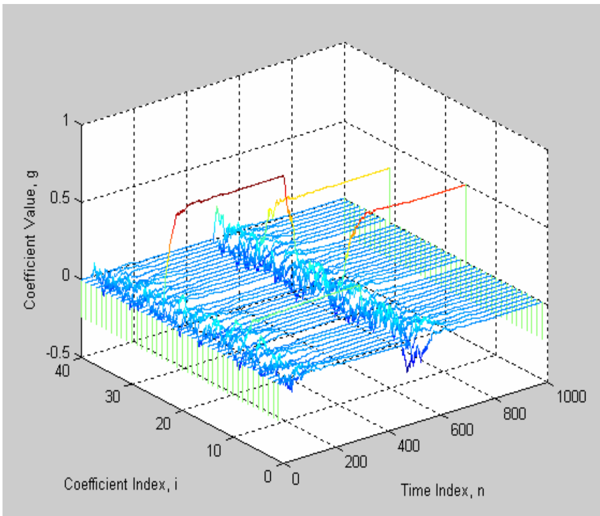

Figure.5 The curve of coefficient for LMS(Case #2)

For the case #3, the echo path is defined as below.

$$e[n]=\begin{cases}0.25x[n-10]+0.65x[n-25], n=0\sim 500\\ 0.65x[n-25]+0.35x[n-35], n=500\sim 1000\end{cases}$$

The curve of coefficient LMS and DLMS are depicted in Fig.7 and Fig.8, respectively.

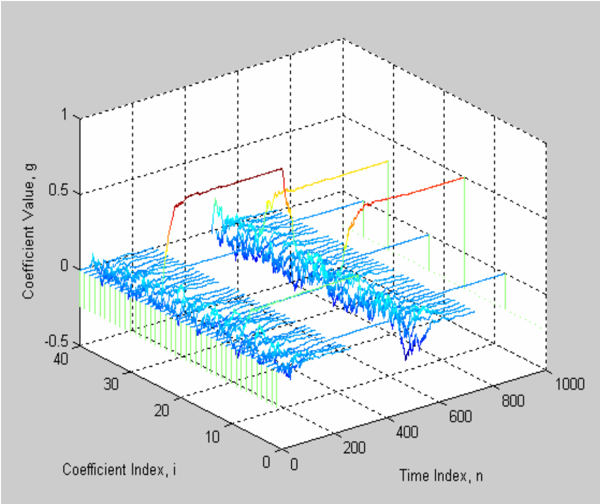

Figure.6 The curve of coefficient for DLMS(Case #2)

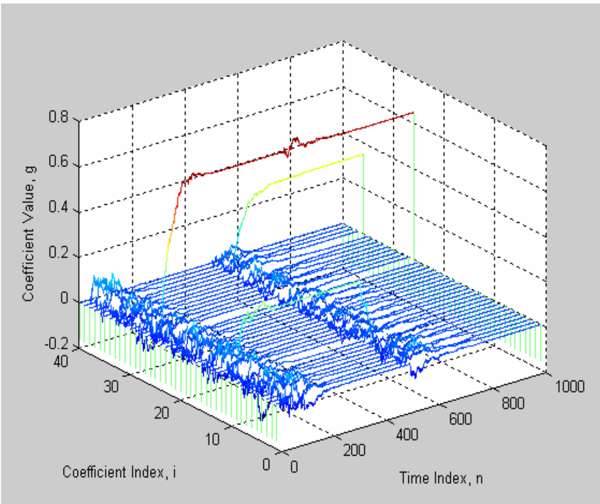

Figure.7 The curve of coefficient for LMS(Case #3)

4. Conclusion

In this paper, the DLMS is firstly proposed. The DLMS algorithm improves the serious drawback of LMS. In the DLMS algorithm, the length of LMS filter can be set as long as enough to match the long echo delay. The computation issue of LMS will be avoided in DLMS. The redundant coefficients will be grouped and computation will be reduced automatically.

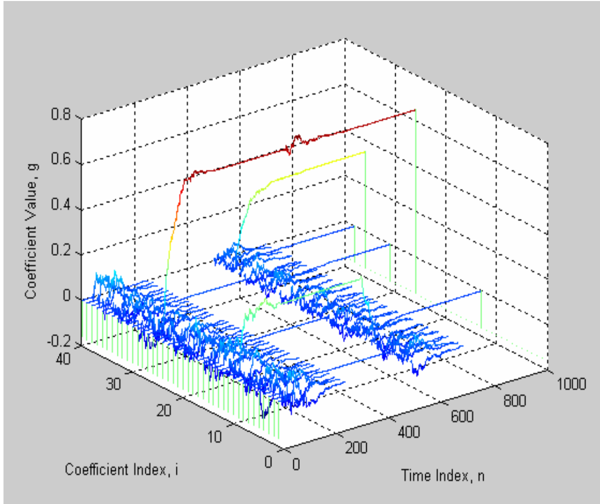

Figure.8 The curve of coefficient for LMS(Case #3)

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