An Adaptive Equalization of Amplitude Chrominance Distortion by using the Variable Step-size Technique

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Abstract: This paper presents an adaptive equalizer using finite impulse response (FIR) filter and least-mean square (LMS) algorithm. Herein, the variable step-size technique (VSLMS) for compensating the amplitude of chrominance signal is utilized. The proposed equalizer can be enhanced and compressed the chrominance signal at color subcarrier. The LMS algorithm employed in simplicity structure but gives slow convergence speed. Thus, the variable step-size is very attractive algorithm due to its computational efficiencies and the speed of convergence is improved. In addition, experimental results are carried out by using the modulated 20T sine squared test signal. It is shown here that the adaptive equalizer can be equalized the amplitude chrominance distortion in color television transmission without relative delay distortion.

Keywords: Adaptive Equalizer, Adaptive step size, LMS algorithm, Chrominance Signal

1. INTRODUCTION

In recent years, adaptive filtering has been receiving a great deal of attentions in the area of communications, sonar, radar, mechanical design, and biomedical electronics [1]. Adaptive filters are self-designing systems, which can adjust themselves to different environments, so the use of adaptive filter provides a significant performance over the use of a fix filter. Basic classes of adaptive filtering applications are recognized, namely adaptive equalizer. The linear distortions of a communication channels may be, in turn, divided into the amplitude and phase distortions parts. To compensate the later distortions, an adaptive equalizer should be employed at the receiver [2].

In this paper presents an adaptive equalizer consists of a transversal (a finite impulse response-FIR) filter and their corresponding VSLMS algorithm is also proposed for compensating the amplitude of chrominance signal. The chrominance-to-luminance gain inequality (relative chrominance level) can be separated low gain ($A<1$), $A_l=\dfrac{1-y}{1+y}$ and high gain ($A>1$), $A_h=\dfrac{1+y}{1-y}$, where y is the normalized amplitude of the signal peak. The proposed equalizer can be enhanced and compressed the chrominance signal at color subcarrier 4.43 MHz and it is not taken into account how the signal is distorted.

In addition, experimental results are carried out by using the modulated 20T sine squared signal. It is shown here that the adaptive equalizer can be equalized the distortion of the chrominance amplitude in color television transmission without relative delay distortion. The simulation result obtain from experimentation is in good agreement.

2. FIR FILTER DESIGN BASED ON VARIABLE STEP-SIZE LMS ALGORITHM

One of the most popular algorithms in adaptive signal processing is the least mean square (LMS) algorithm presented by Widrow and Hoff [3]. The LMS algorithm has always been attractive to the researchers in adaptive signal processing, because of its simplicity and its robustness to numerical error [4]. The VSLMS algorithm has initially been introduced as a technique for increasing the initial convergence of the LMS algorithm, while achieving a small steady-state misadjustment.

The most commonly used structure in the implementation of adaptive filter is the transversal structure, depicted in Fig. 1.

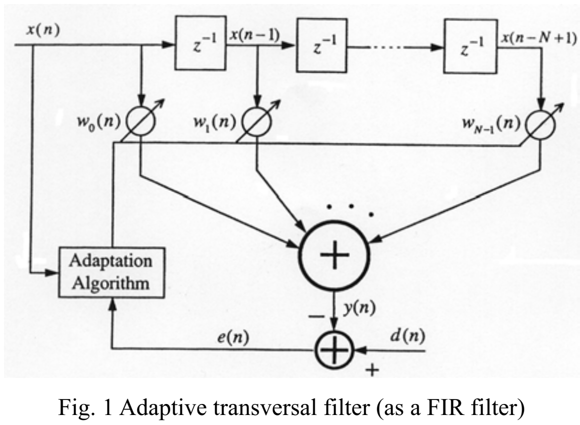

Fig. 1 Adaptive transversal filter (as a FIR filter)

The output, ($y(n)$), is generated as a linear combination of the delayed samples of the input sequence, ($x(n)$), according to the equation [5].

$$y(n)=\sum_{i=0}^{N-1} w_i(n)x(n-i) \qquad (1)$$

then the output of adaptive FIR filter can be expressed in matrix form as:

$$y(n)=\mathbf{w}^T(n)\mathbf{x}(n) \qquad (2)$$

where superscript T denotes the vector or matrix transpose, N is the filter length, $\mathbf{w}(n)$ is the tap-weight vectors and $\mathbf{x}(n)$ is input vectors, respectively, and they are defined in the column vectors form as follows:

$$\mathbf{w}(n) = [w_0(n) \quad w_1(n) \quad .... \quad w_{N-1}(n)]^T, \tag{3}$$

$$\mathbf{x}(n) = [x(n) \quad x(n-1) \quad ...... \quad x(n-N+1)]^T \tag{4}$$

we drop the time index n from the tap-weight vector $\mathbf{w}(n)$ in the following discussion. If the desired output at the instant n is $d(n)$, the error $e(n)$ is written as

$$e(n) = d(n) - y(n) \tag{5}$$

Eq. (2) can be substituted into Eq. (5), we get

$$e(n) = d(n) - \mathbf{w}^T(n)\mathbf{x}(n) \tag{6}$$

The conventional VSLMS algorithm is a stochastic implementation of the steepest descent algorithm. It simply replaces the cost function $\xi = E[e^2(n)]$ by its instantaneous coarse estimate $\hat{\xi}(n) = e^2(n)$ for ξ in the steepest descent recursion Eq. (6), and replacing the iteration index k by the time index n, we obtain

$$\mathbf{w}(k+1) = \mathbf{w}(k) - \mu\nabla_k\xi, \tag{7}$$

$$\mathbf{w}(n+1) = \mathbf{w}(n) - \mu\nabla e^2(n) \tag{8}$$

where ∇ is the gradient operator defined as the column vector

$$\nabla = \left[\frac{\partial}{\partial w_0} \quad \frac{\partial}{\partial w_1} \quad .... \quad \frac{\partial}{\partial w}\right] \tag{9}$$

we note that the ith element of the gradient vector $\nabla e^2(n)$ is

$$\frac{\partial e^2(n)}{\partial w_i} = 2e(n)\frac{\partial e(n)}{\partial w_i} \tag{10}$$

substituting Eq. (5) in the last factor on the right-hand side of Eq. (10) and noting that $d(n)$ is independent of w_i, we obtain

$$\frac{\partial e^2(n)}{\partial w_i} = -2e(n)\frac{\partial y(n)}{\partial w_i} \tag{11}$$

substituting for $y(n)$ from Eq. (1), we get

$$\frac{\partial e^2(n)}{\partial w_i} = -2e(n)x(n-i) \tag{12}$$

using Eq. (9) and Eq. (12), we obtain

$$\nabla e^2(n) = -2e(n)x(n) \tag{13}$$

substituting this result in Eq. (8), we get

$$\mathbf{w}(n+1) = \mathbf{w}(n) + 2\mu e(n)\mathbf{x}(n) \tag{14}$$

The VSLMS algorithm works on the basis of a simple heuristic that comes from the mechanism of the LMS algorithm. Each tap of the adaptive filter is given a separate time-varying step-size parameter and the LMS recursion is written as

$$w_i(n+1) = w_i(n) + 2\mu_i(n)e(n)x(n-i) \qquad \text{for } i = 0, 1, ..., \text{N-1} \tag{15}$$

where $w_i(n)$ is the ith element of the tap-weight vector $\mathbf{w}(n)$ and $\mu_i(n)$ is its associated step-size parameter at iteration n. The VSLMS algorithm step-size parameters, the $\mu_i(n)$, may be adjusted using the following recursion

$$\mu_i(n) = \mu_i(n-1) + \rho\, sign[g_i(n)]sign[g_i(n-1)] \tag{16}$$

where the gradient term $g_i(n) = e(n)x(n-i)$, and ρ is a small position step-size parameter.

The set of update Eq. (14) may be written in vector form as follows:

$$\mathbf{w}(n+1) = \mathbf{w}(n) + 2\boldsymbol{\mu}(n)e(n)\mathbf{x}(n) \tag{17}$$

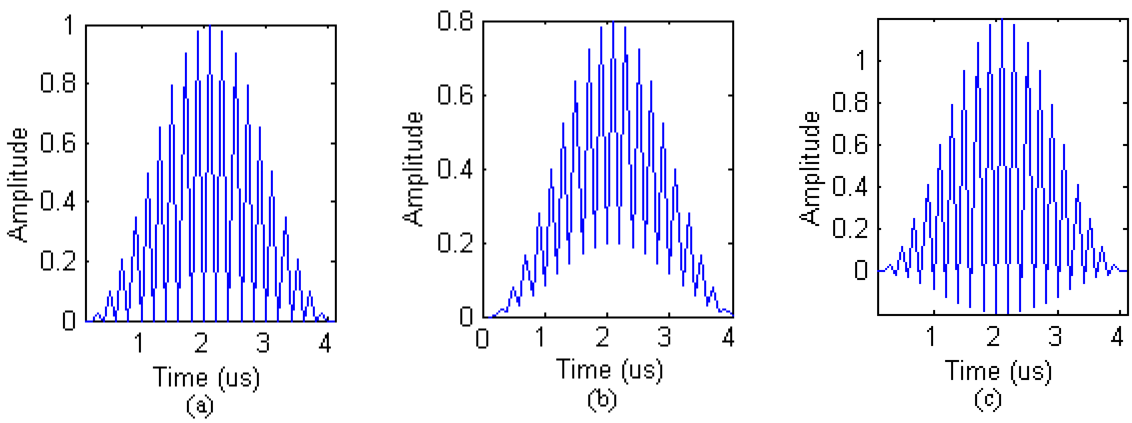

Fig. 2 shows the modulated 20T sine-square pulse signal

3. DESCRIPTION OF CHROMINANCE SIGNAL AMPLITUDE

3.1 Definition

In this paper, we use the modulated 20T sine squared pulse test signal can be obtained as follows: (see Fig. 2) [5].

$$x(n) = \begin{cases} \frac{1}{2}\sin^2\left(\frac{\pi t}{40T}\right) + \frac{A}{2}\sin^2\left(\frac{\pi t}{40T}\right)\cos\omega_c t; & |t| \le 20T \\ 0 & ; |t| > 20T \end{cases} \qquad (18)$$

where A is the linear distorted chrominance signal amplitude, T = 0.1 μS and for PAL system t is instantaneous time, $\omega_c = 2\pi f_c$, and the color subcarrier f_c = 4.43 MHz.

Fig. 2(a) shows the undistorted modulated 20T test signal with A = 1, Fig. 2(b) shows the low gain chrominance distortion with A < 1, and Fig. 2(c) shows the high gain chrominance distortion with A > 1.

3.2 Design adaptive equalizer using VSLMS algorithm

Fig. 3 shows block diagram of an adaptive equalization of chrominance signal amplitude, by using FIR filter and VSLMS algorithm. Herein, $x(n)$ is input signal, which have been occurred the amplitude chrominance distortion, $d(n)$ is the reference signal, $y(n)$ is the output signal, which is modified the amplitude chrominance, and $e(n)$ is the error signal.

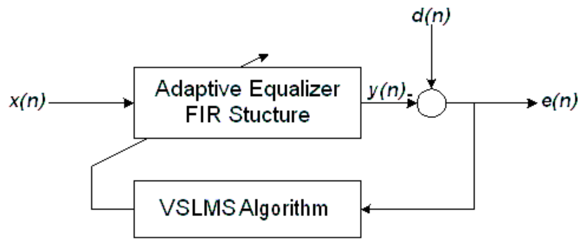

Fig. 3 shows block diagram of an adaptive equalization of chrominance signal amplitude.

In order to assure the theoretical results of the previous section, simulation results of an adaptive equalizer for compensating the chroma amplitude distortion is carried out. The procedure of an implementation of the VSLMS algorithm is shown in Table 1.

4. EXPERIMENTS AND RESULTS

4.1 Comparison of the proposed algorithm

In this section, we used the single sinusoidal signal additive white Gaussian noise as the input of FIR adaptive filter. The performance of the proposed algorithm (VSLMS algorithm) compared with LMS algorithm, assume data range N=1,500.

Fig. 4 shows the comparison of LMS algorithm and the proposed algorithm in the noisy environment. The proposed algorithm have fast convergence rate.

Table 1. Summary of an implementation of VSLMS algorithm

Input: Tap-weight vector, $\mathbf{w}(n)$

Input vector, $\mathbf{x}(n)$

Gradient terms, $g_0(n-1), g_1(n-1), \ldots, g_{N-1}(n-1)$

Desired output, $d(n)$

Output: Filter output, $y(n)$

Tap-weight vector update, $\mathbf{w}(n+1)$

Gradient terms, $g_0(n), g_1(n), \ldots, g_{N-1}(n)$

Updated step-size parameters,

$\mu_0(n), \mu_1(n), \ldots\ldots, \mu_{N-1}(n)$

Filtering: $y(n) = \mathbf{w}^T(n)\mathbf{x}(n)$

1. Error estimation: $e(n) = d(n) - y(n)$
2. Tap weights and step-size parameters adaptation:

For $i = 0,1,\ldots\ldots\ldots, N-1$

$g_i(n) = e(n)x(n-i)$

$\mu_i(n) = \mu_i(n-1) + \rho sign[g_i(n)] sign[g_i(n-1)]$

if $\mu_i(n) > \mu_{\max}, \mu_i(n) = \mu_{\max}$

if $\mu_i(n) < \mu_{\min}, \mu_i(n) = \mu_{\min}$

$w_i(n+1) = w_i(n) + 2\mu_i(n) g_i(n)$

end

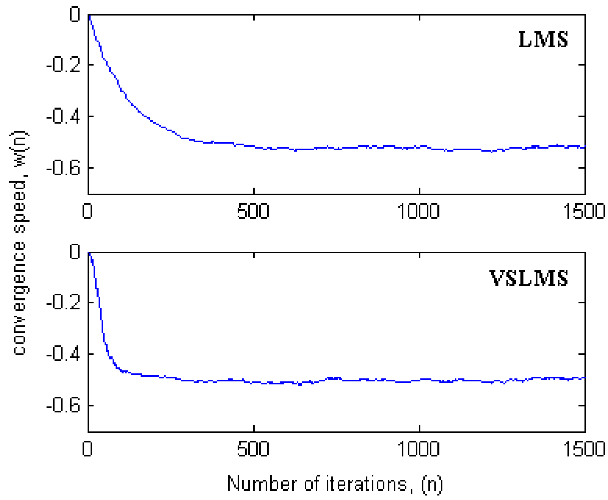

Fig. 4 shows the learning curve of coefficient $\mathbf{w}(n)$

4.2 Simulation results

The Chrominance-to-Luminance ratio or gain inequality (relative chrominance level)[7] can be separated low gain and high gain. The gain inequality of high gain and low gain are expressed as

$$A_h = \frac{1+y}{1-y}, \qquad (19)$$

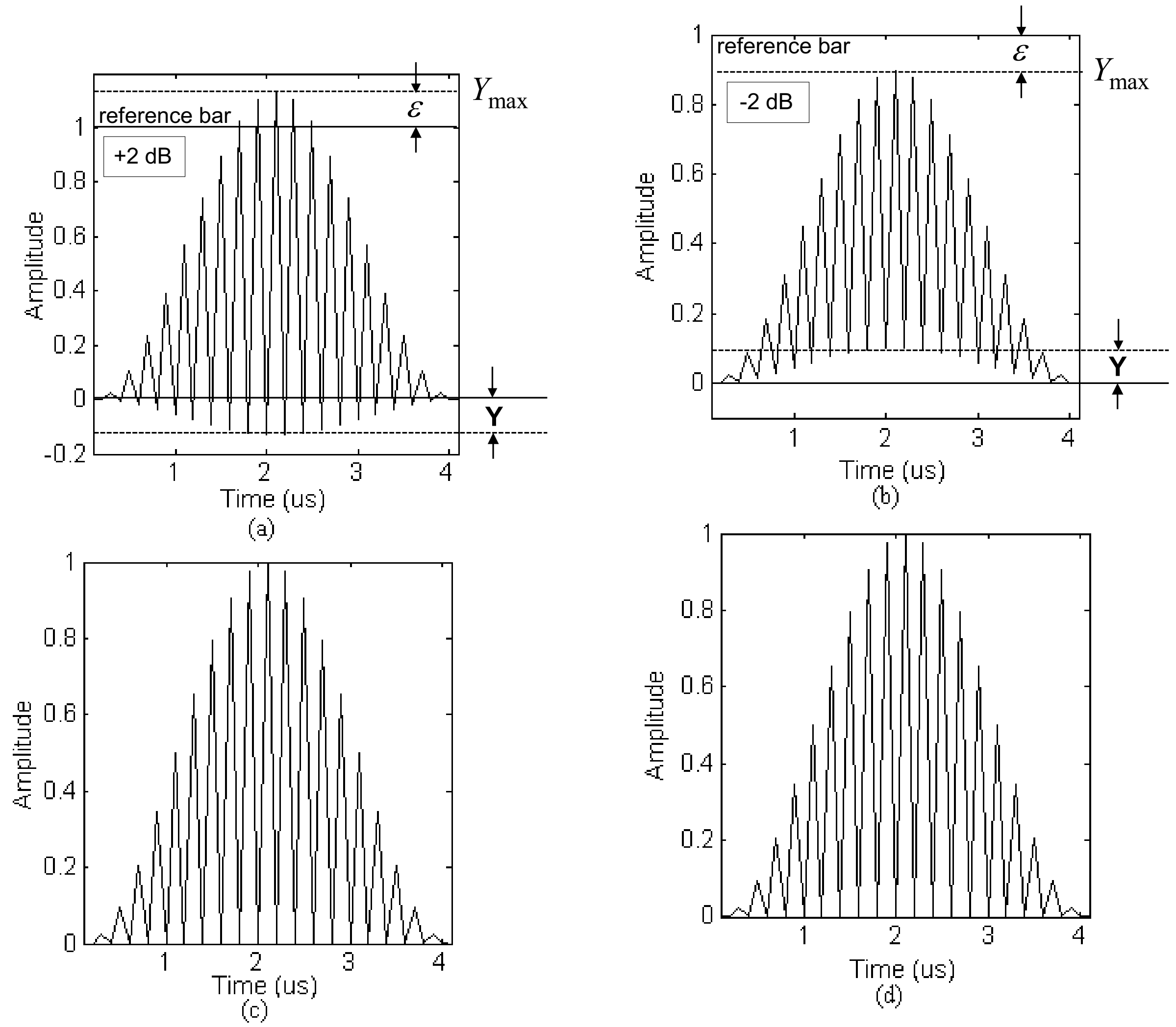

Fig. 5 shows the amplitude of chrominance signal equalized by adaptive equalization using VSLMS algorithm

$$A_l = \frac{1-y}{1+y} \tag{20}$$

where y is the normalized amplitude of the single peak. Hence

$$y = \frac{Y}{Y_{\max}} \tag{21}$$

From Fig. 5 shows the distortion of the modulated 20T sine-squared pulse that the peak pulse height $y = 0.1148$. Then the gain inequality of high gain in decibels, $A_h = +2dB$ and low gain $A_l = -2dB$ are shown in Fig. 5(a) and 5(b) respectively. Fig. 5(c) and 5(d) show the amplitude of chrominance signal is equalized by adaptive equalization using VSLMS algorithm.

It is shown that from the experimental can be enhanced and compressed the chrominance signal without impact of group delay distortion. If the presence of group delays distortion, the baseline of the modulated pulse is not flat or symmetrical positive and negative peaks after it is equalized.

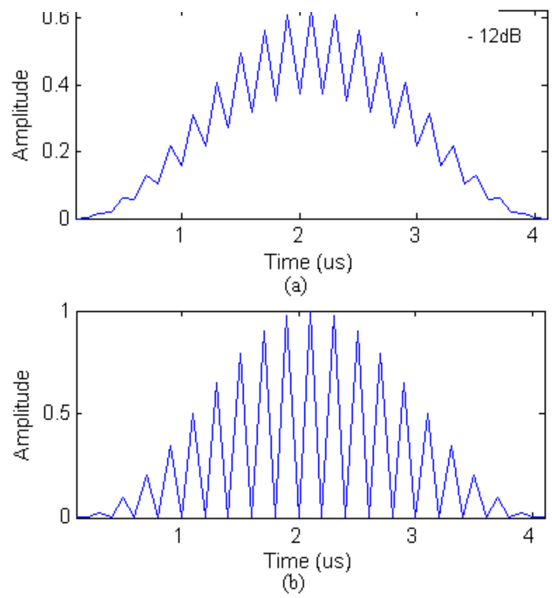

Fig. 6 shows the distorted chrominance signal amplitude of the modulated 20T sine-square pulse that low gain is –12 dB

Fig. 6 shows the distorted chrominance signal amplitude of the modulated 20T sine-squared pulse that low gain is –12dB. In brief, the adaptive equalizer can be equalized the amplitude chrominance distortion in color television transmission with relative delay distortion and the system design has complexity less than analog filter.

5. CONCLUSIONS

The design of adaptive equalizer using FIR filter and the variable step-size LMS (VSLMS) algorithm for compensating the amplitude of chrominance signal is utilized, which the performance of the proposed algorithm has fast convergence speed better than the LMS algorithm. As the result, the adaptive equalizer can be enhanced and compressed the chrominance signal without impact of group delay distortion. The advantage of this system design has lowest complexity and high flexible than analog filter, and iterative solutions can find a good optimum performance without adjustable.

In brief, the results obtained from the simulation can be equalized the amplitude chrominance distortion, low gain and high gain, in color television transmission without relative delay distortion.

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