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DSP TMS32020 IMPLEMENTATION OF AN ERROR FUNCTION ADAPTIVE RECURSIVE FILTER

Elsayed A. Soleit (Ph. D)
The Military Technical College, Cairo, Egypt.

ABSTRACT

The real time implementation of the adaptive recursive filter (ARF) is of great interest. Moreover, the error function ARF is hyperstable and its performance criterion is unimodal. In this paper a proposal to implement an error function ARF in real time is introduced using the Texas digital signal processor DSP TMS32020. The designed filter can operate at a sampling rate of 14.8 kHz which is suitable for various applications.

KEY WORDS

The adaptive filter, the error function, the adaptation algorithm, and the digital signal processor.

1- INTRODUCTION

The adaptive recursive filter (ARF) has wide spectrum of applications. It is used in system modeling, adaptive control and communication fields. However, the ARF suffers from two main drawbacks; the tendency to instability and the multimodality of the performance criterion [1,2]. However, the error function criterion is unimodal and there is a unique optimal solution for the filter coefficients. Moreover, the stability of the adaptive recursive filter is ensured either according to the bounded input bounded output (BIBO) stability theorem or according to the hyperstability theorem. Fortunately, the error function ARF is always hyperstable during the adaptation process [1-3,4]. Hence, the error function filter represents a good example for the real-time implementation.

The implementation of the BIBO adaptive recursive filter in real time using the DSP TMS32020 is illustrated in [6]. In this paper, the hyperstable error function ARF is

implemented using the digital signal processor DSP TMS32020 that is developed by Texas Instruments [5].

This paper has four sections. Section two illustrates the formulation of the error function ARF while section three explains the real-time implementation of the ARF. Conclusions of the whole paper are given in section four.

2- THE ERROR FUNCTION ADAPTIVE RECURSIVE FILTER

The filter output

The ARF is used to model an unknown echo path of a practical hybrid circuit in the telephone channels. The output signal of the echo path represents the desired response of the ARF that is represented by:

$$\mathbf{d}_k = \sum_{i=0}^{N} \mathbf{c}_{i,k} \mathbf{x}_{k-i} + \sum_{j=1}^{M} \mathbf{f}_{j,k} \mathbf{d}_{k-j} \tag{1}$$

Where $\{c_i\}$ and $\{f_j\}$ are known as the forward and the backward coefficients of the echo path model. Also, $\{xk\}$ and $\{d_k\}$ are called respectively, the input and the output observations. Assuming that the ARF has the the same order as the echo path model, then, the filter output signal can be expressed as:

$$\mathbf{y}_k = \sum_{i=0}^{N} \mathbf{a}_{i,k} \mathbf{n}_{2,k-i} + \sum_{j=1}^{M} \mathbf{b}_{j,k} \mathbf{y}_{k-j} \tag{2}$$

Where $\{a_i\}$ and $\{bj\}$ are known as the forward and the backward coefficients of the ARF.The purpose of the adaptive filter is to generate a good replica of the output of the echo path based on the minimization of the mean square error of the deviation error resulting as the filter output is subtracted from the echo path output. Then, the error signal is defined as:

$$\varepsilon_k = \mathbf{d}_k - \mathbf{y}_k \tag{3}$$

Substituting eq. (1) and eq. (2) in eq. (3) yields:

$$\varepsilon_k - \sum_{j=1}^{M} \mathbf{b}_{j,k} \varepsilon_{k-j} = \sum_{i=0}^{N} (\mathbf{c}_i - \mathbf{a}_{i,k}) \mathbf{x}_{k-i} + \sum_{j=1}^{M} (\mathbf{f}_j - \mathbf{b}_{j,k}) \mathbf{d}_{k-j} \tag{4}$$

Define the left-hand terms of eq. (4) as an error function, [1,2]. Then, the error function is defined as:

$$\rho_k = \sum_{i=0}^{N} (\mathbf{c}_i - \mathbf{a}_{i,k}) \mathbf{x}_{k-i} + \sum_{j=1}^{M} (\mathbf{f}_j - \mathbf{b}_{j,k}) \mathbf{d}_{k-j} \tag{5}$$

It is apparent that the error function is a linear function of the filter coefficients. Then, its mean square is a quadratic function and possesses a unique optimal solution of the filter coefficients when the mean square of the error function is minimized [1-4].

The error function adaptation algorithm

The filter coefficients are updated such that the mean square of the error function is minimized. The well-known error function least mean square ARF [1,2,4] is used to update the filter coefficients. Consequently, the forward and the backward coefficients are updated as:

$$\mathbf{a}_{i,k+1} = \mathbf{a}_{i,k} + 2\mu\,\rho_k\,\mathbf{x}_{k-i} \qquad (6)$$

$i=0,1,2,\ldots\ldots N$

and the backward coefficients are also updated as:

$$\mathbf{b}_{j,k+1} = \mathbf{b}_{j,k} + 2\mu\,\rho_k\,\mathbf{d}_{k-j} \qquad (7)$$

$j=1,2,\ldots\ldots M$

Where μ is called the step size of the adaptation algorithm which controls the adaptation speed and noise during the adaptation process [1-4].

3- FILTER IMPLEMENTATION USING THE DSP TMS32020

The error function ARF is implemented in real-time using the DSP TMS32020 development board [7]. Figure 1 demonstrates the system realization scheme of the echo cancellation. The hardware requirements includes a program memory, interface circuits, analog to digital and digital to analog converters and a timing circuit. The software requirements are the program instructions that include three interrupt service routines; the initialization, the convolution response and the adaptation algorithm routines.

The filter performance is evaluated in the transient and the steady states. The learning curve of the ARF during the adaptation process explains the dynamic behavior of the ARF by indicating the instantaneous mean square error versus the sampling time. This curve is measured by ensemble averaging of the Error Square of 16 separate experiments. Figure 2 illustrates the error function ARF learning curve while Figure 3 indicates the steady state error after convergence versus the sampling time. Moreover, The residual mean square error (RMSE) is measured after 2000 samples and it is averaged over a period of 1000 samples.

It is concluded that the implemented error function ARF provides RMSE=-57 dBm. Furthermore, it is found that the implemented filter with N=12 and M=4 can operate on a maximum sampling rate of 14.8 kHz when it is used for the echo cancellation. Also, it is found from the practical implementation of the error function ARF that, the total number of the processor instructions is represented by:

$$\mathbf{N_t = 14N + 15M + 109} \qquad (8)$$

Where N and M are known as the forward and the backward coefficients of the error function ARF. Moreover, the sampling frequency can be obtained in terms of the cycle time of the TMS32020 processor as:

$$f_s(\mathrm{kHz}) = \frac{1}{(14N + 15M + 109)Tc} \tag{9}$$

Where T_c is defined as the operation cycle time that is given for the DSP TMS32020 processor by [5]:

$$T_c(\mathrm{m\,sec}) = \frac{1}{5000} \tag{10}$$

4- CONCLUSIONS

The error function adaptive recursive filter can be implemented successfully in real time using the Texas Instrument DSP TMS32020. The error function ARF is almost hyperstable during the adaptation algorithm. A small high speed EPROM can accommodate the software program instructions. The realized filter can operate on a 14.8 kHz sampling rate using the DSP TMS32020 that suits a wide range of applications.

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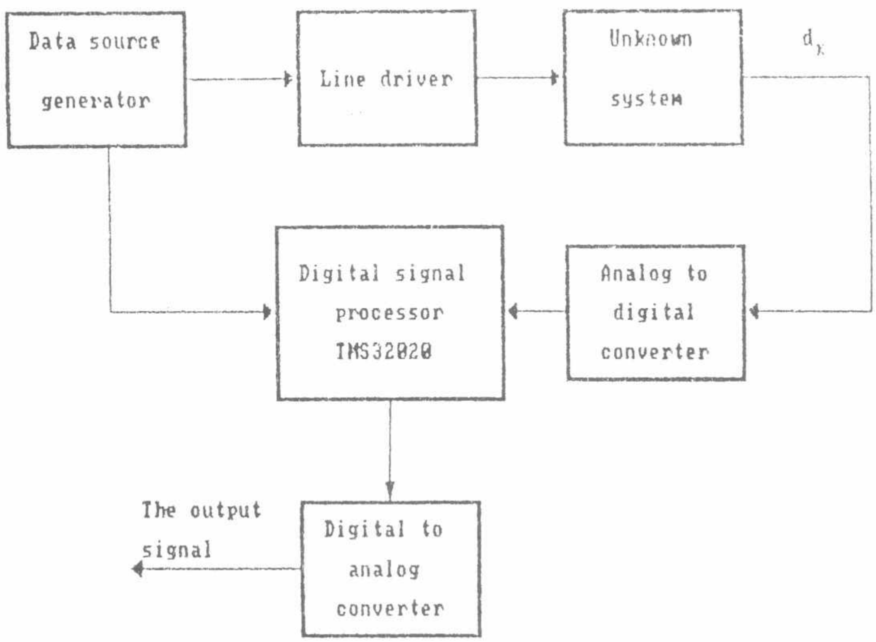

Figure 1 Implementation of the adaptive filter using the DSP TMS32020

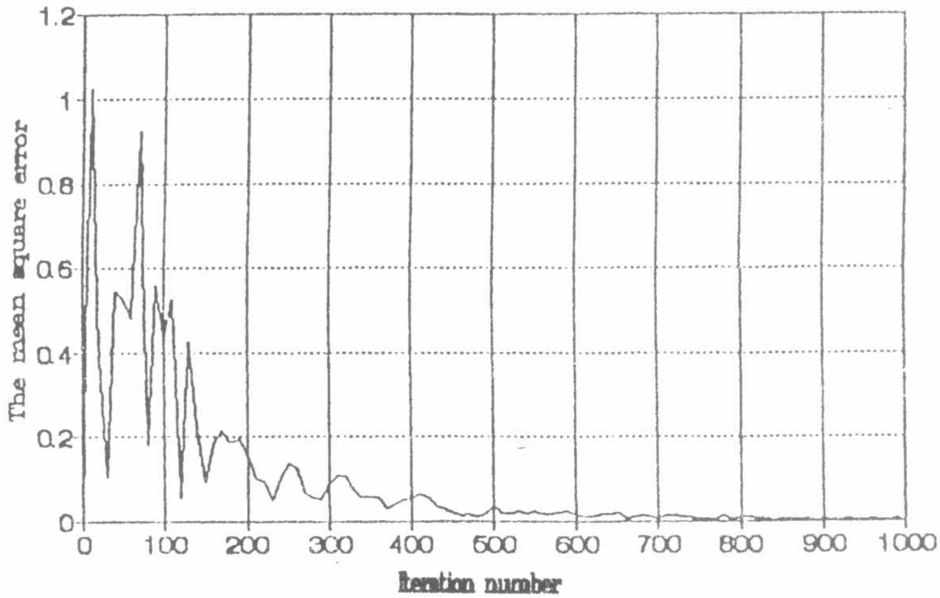

Figure 2 The learning curve of the error function ARF

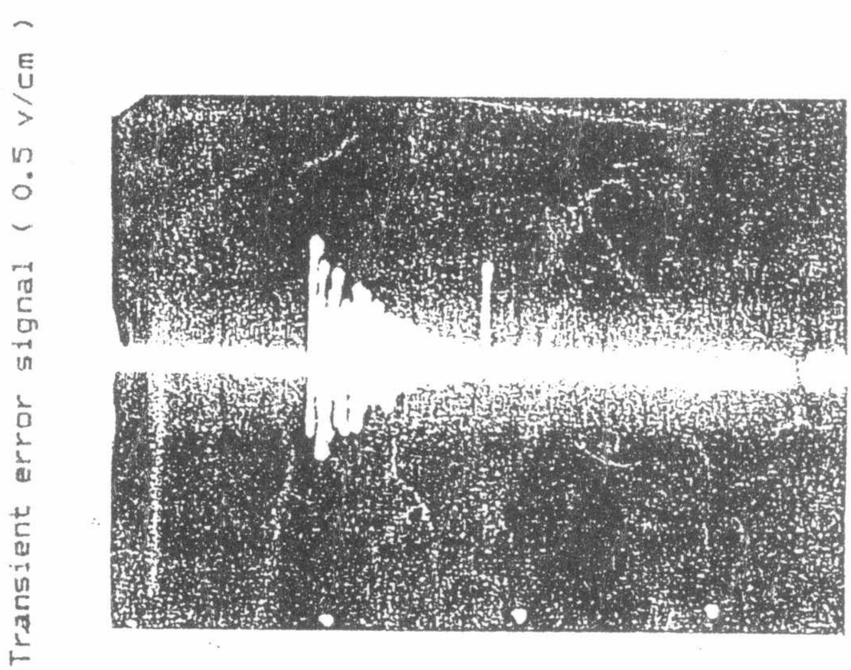

Figure 3 The steady state output error signal