Adaptive Filter Design Based On The LMS Algorithm in SVC

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Abstract

The paper proposed the adapter filter design based on the improved LMS algorithm. The paper first introduced the minimum mean square (LMS) algorithm, which is a very useful and very simple estimated gradient method. This algorithm has been widely used since the early 1960s quickly, its advantage is that the small amount of calculation, and the rapid development of the digital signal processor also allows real-time adaptive filter economy realize it possible. For the design of SVC, the adaptive filter design is outstanding performed in the operation. The paper implemented on the DSP LMS adaptive filter algorithm and related technologies for echo cancellation applications do simple research and discussion.

Keywords: LMS, Adaptive Filter, Echo Canceller

1. Introduction

The digital filter is a linear system that used widely in Digital Signal Processing, which is an important foundation of the Digital Signal Processing. Its function is essentially let a set of input sequence of numbers into another set of output sequence of numbers by computing. Since the adaptive filter can automatically adjust its own parameters, and it requires very little in the design or simply does not require any a priori statistical knowledge of signal and noise, thus the digital filter developed very soon.

Adaptive equalization algorithm has made remarkable development, is currently the communication of research in the field of hot spots. In modern communication systems, Intersymbol interference [1-2] is an important factor restricting the communication quality. In order to reduce Intersymbol interference, the channel needs to be adequate compensation to reduce the error rate, improve the quality of communication. Receivers can compensate or reduce the Compensator reception Intersymbol interference is called Equalizer. For most digital communication system using equalizer, channel characteristics are unknown, even with time-varying. Can accurately compensate for channel changes in transmission characteristics and dynamic tracking channels [3], adjust balance parameters of filter system functions in a timely manner, called smart Adaptive Equalizer features. Because computer hardware limitations, it is hoped that small amount of fast convergence Adaptive equalization algorithm, calculation, and low bit error rate of the system. On this basis, for which research is mainly concentrated in the smallest mean square (LMS) class, recursive least squares (RLS), and several other aspects.

LMS algorithm is generated based on stochastic gradient approximation of the minimum mean square error criteria of the guidelines that is simply replaced by instantaneous error value. Benefits of this approximation is saving the computation of input signal required huge amount of computation of the autocorrelation matrix, makes it very simple to tap coefficients of iteration formula. In most of the higher signal-to-noise ratio and channel for slow time-varying circumstances, as long as the convergence step within a certain range, better convergence of LMS algorithm. However, this approximate brings 2 disadvantages [4]:

1. Introduces the tap coefficient of noise. Leading to large amount of steady-state dissonance;
2. Convergence is relatively slow, poor adaptability to non-stationary signals, making compensation algorithm of channel tracking ability decreases.

In response to these two issues, people made a variety of improvements of LMS adaptive filtering algorithms, notably, transform domain LMs algorithm for variable step-size LMs algorithm.
2. Adaptive Filter Theory

Adaptive Filter is composed of two parts: the digital structure of adjustable parameters (or adaptive processor) and the adaptive algorithm. Adjustable Algorithm principle: the final mean square value of $e(n)$ is minimum. Adaptive Filter is a special Wiener Filter that able to automatically adjust the parameters itself. When design it, we have not to know the input signal and noise statistics. It can gradually learn in their own work or to estimate the required statistical properties. And pursuant to automatically adjust its parameters to achieve the best filtering the purpose. The biggest characteristics of adaptive filter are: learning and tracking. Least-Mean-Square Adaptive Algorithm minimum mean square (LMS) adaptive algorithm, its core is replaced with a square error mean square error [5]. The LMS algorithm is a useful estimated gradient method. Its prominent advantage is the small amount of calculation, and not off-line computing, you just need to know that the input signal and the reference response.

Its general idea is that by the $\hat{e}(n) \rightarrow E[\hat{e}^2(n)]$

Let a single square of the gradient of the error sequence $\hat{\nabla}(n) \rightarrow$ multiple statistical average gradient squared error sequence. $\hat{\nabla}(n) = \frac{\partial \hat{\nabla}(n)}{\partial W}$

$LMS$ algorithm's Basic relationship

In addition to the basic algorithm, there are many variations in LMS algorithm[2] , as Sign-error LMS algorithm , Sign-data LMS algorithm , the Sign- Sign LMS algorithm , etc. Here are several LMS algorithm coefficient iterative formula .

Basic LMS: $w(k, n+1) = w(k, n) + u e(n)x(n-k)$ (1)

Sign-error LMS: $w(k, n+1) = w(k, n) + u \text{sgn}[e(n)]x(n-k)$ (2)

Sign-data LMS: $w(k, n+1) = w(k, n) + u \text{sgn}[e(n)] \text{sgn}[x(n-k)]$ (3)

Sign-sign LMS: $w(k, n+1) = w(k, n) + u \text{sgn}[e(n)] \text{sgn}[e(n-k)]$ (4)

They gave a new RLS algorithm-like recursive least squares algorithm, the algorithm is directly related to the input signal processing functions instead of the processed input signal itself. Document proposes a new RLS algorithm for quantization input, with a new quantum function of the input signal is clipping, convergence speed of this new algorithm and RLS algorithm for tracking capabilities are superior to conventional. RLS algorithm of document suggested a partial update, it is of a higher order filter function is broken down into 2 simple low-order function, and then updated on the lower part of the filter coefficients to reduce computational complexity. In addition, there are many improved RLs algorithms, they retained the advantages of RLS algorithm converge faster, while significantly reducing the computational complexity (Figure 2), has been applied to many areas.

In the above formulas, $W(k, n)$ and $w(k, n+1)$ respectively stands for equation coefficient values before and after the iteration, n and n +1 stands for two moments before and after, $k = 0$ to $N -1$, $N$ is
the order of the filter. \(u\) is the convergence factor, \(e(n)\) is the error signal, \(x(n-k)\) is the input signal. \(\text{sgn}[x]\) is the sign function, when \(x \geq 0\), and its value is 1, otherwise its value is -1. Because of the strong requirements of real-time of adaptive filter, it need high-performance DSP chip to realize. The following article takes adaptive filtering of basic LMS algorithm as an example, describes the design and implementation of adaptive filter.

3. Adaptive Filter Structure and The Chip Type Selection

The adaptive filter structure can use the FIR or IIR structure. Because the impact of the echo path response was a kind of a high degree but extremely irregular shape, therefore, in order to achieve a certain quality of the identification results, the adaptive filter must provide many adjustable parameters; it also needs adaptive filter has enough good stability in the adaptive process, the design in this article uses the FIR filter (above in Figure 1) of the lateral type structure based on the two considerations.

In practical applications, the adaptive filter coefficients continually adjusted and unstable, however, it needs strong real-time requirements, so need high-performance DSP chip to achieve. What is more, it should also take into account the cost, packaging design requirements, hence the cost is relatively high fixed-point DSP chip to become the first choice. This article choose the most popular TMS320C54x fixed-point DSP chip [6] for the target platform of hardware and software implementation of the algorithm.

4. The Implementation of TMS320C54x and LMS Adaptive Filter

TMS320C54x is the TI's specifically designed to achieve low-power, high-performance fixed-point DSP chip, the series chips are smaller than others and have advanced CPU structure and the Intelligent Peripheral Interface, and the establishment of the three low-power mode, which has a high cost-effective, and has been widely used in many fields of communication. Selection of the C54x to achieve LMS adaptive filter algorithm has a great advantage in the software programming. The series of chip integration two 40 bit accumulator, they can be in one instruction cycle to operate at the same time, in the C54x instruction set, it provides some instructions, we can make full use of this property to reduce the operations required in the instruction cycle, which largely improving the efficiency of programming. For example, the variance of LMS ST||MPY and RPTB instructions [7] offer great convenience for the realization of adaptive filters. The LMS instruction in the implementation of MAC directive function, it also achieve a four to five Rounding addition at the same time. So, LMS, ST||MPY and RPTB combination can be realized in the calculation of the current adaptive filter output, at the same time, updating the filter coefficients. For every given moment \(n\), \(u(n)\) is constant. This factor is stored in a temporary register \(T\), for an update on to use of. This provides a possible for the high computational complexity and better performance algorithm.
The order of adaptive filter coefficients and the input sample values stored in the memory is shown in Figure 2. Assume that ue(n) already exists in the T register, then the implementation of the basic LMS algorithm TMS320C54x procedures are:

```assembly
ST    A,*W_COFF_P+ ; Storing the updated filter coefficient 
||MPY *XW_DATA_P+0%,A ; A = ue(n)x(n-k) 
LMS *W_COFF_P,*XW_DATA_P ; A = w(k,n)+ue(n)x(n-k) 
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; B = B+w(k)x(n-k)

**Figure 2.** Sequence diagram of coefficients and input sample values stored in memory

Circular buffer has been used in the actual programming, its main features is to open a sliding window buffer which has N elements in the data storage area, the latest N input sample values are stored in the sliding window. When each new sample is input, the new sample will overwrite the oldest data in the sliding window, and other data is not required to move. The programming using chip BK (circular buffer length) register indirect addressing the sliding window, and thus the circular buffer address is inclusive adjacent. The circular buffer does not need to move data, The circular buffer position can be set in any position of data stored [6].

Following is TMS320C54x achieve adaptive filter offset ordinary FIR filter output in Figure 3:

**Figure 3.** TMS320C54x adaptive filter structure diagram

FIR digital filters using MATLAB design. Program FIR digital filter passband edge frequency is 1500Hz, passband ripple of less than 1dB; stopband edge frequency is 2000Hz, stop-band attenuation greater than 40dB, the sampling frequency is 8000Hz low-pass filter [8]. Adopt the MATLAB window function design, the choice of the Hamming window, its procedures are: b = fir1(16,1500/8000 * 2); derived coefficients are converted into Q15 format and placed on the DSP assembler firdata.inc file.

The simulation frequency is 254Hz, 500Hz, 1000Hz, 2000Hz and 2370Hz, the synthesis of the interference signal data file of the discrete sampling points (sampling frequency is 8000Hz) through
the C language may be very easy to achieve. Then through the LMS algorithm, where in the desired frequency of the signal is extracted.

This article using TI's CCS (Code Composer Studio) integrated development environment, which uses a Windows-style interface, set edit, compile, link, software simulation and hardware debugging and real-time tracking and other functions into one, greatly facilitates the design of the DSP program and development.

CCS provides many ways to show the drawing of data generated by the program, including time domain / frequency domain waveform display, constellation diagram, eye diagram and image display.

The simulation results of this paper are the CCS frequency domain / time domain waveform display function GraphProperty [9] to observe the results.

The simulation results are as follows:

(a) Input data XW (n) in time domain

(b) Input data XW (n) in frequency domain

(c) Reference signal D (n) in time domain
We can be seen from the time domain figure of simulation lab that at first the error is larger, and smaller filter output waveform amplitude [7]. With the convergence of the filter parameters are adjusted, the error is getting smaller and smaller, the corresponding filter output gradually increase, eventually become the same with reference signal d (n) and is almost the sinusoidal signal. When the step $u = 0.01$ to 0.1, the convergence speed increases and the simulation waveforms shown as following. However, the value of $u$ cannot be unlimited increase, otherwise the output signal distortion and noise increase, as shown below ($u = 0.93$), it is necessary to consider the various factors. As convergence range $0 < u < \left[ \frac{1}{L+1} \right] \text{Pin}$ (where Pin is the input signal power), offset amount like.

5. LMS Adaptive Filter and Its Application

In recent years, the neural network there have been remarkable advances in theory and practice, is now widely used in communications, seismic prospecting, bio-medical, speech, signal processing, and image processing, pattern recognition, automatic control and other fields. Ma Ye, on the uncertainty of complex problems using neural network with adaptive and learn the advantages of proposed an Adaptive Kaman filtering algorithm for fuzzy neural network. The Adaptive fuzzy filter algorithm and combining neural networks error compensation, effectively preventing the filter divergence, reduce the estimation error, improving estimation accuracy. Dr. Johnson [8] provided an introduction to General linear base function neural networks in Adaptive Filter, in the linear Adaptive Filter introduces a non-linear hidden layer, making it both Adaptive Filter and nonlinear processing capacity. Gao Weygand proposed a paper for BP neural network training [10], which indicated the disadvantages such as slow and vulnerable to local minimum problem, proposes a network of improved algorithms.

5.1. Matlab Simulation

We using difference method to simulate the Adaptive Filter Design model using The LMS Algorithm in SVC, the simulation results can be presented directly by using Matlab.
Figure 5. The matlab wave of Input Signal

Figure 6. The curve convergence simulation in matlab
We are able to claim that the nonlinear signal processing is the information processing which learnt in the of an important research area, used neural network can is good to address nonlinear signal processing problem, will nonlinear neural network application Yu shot body track system, and image recovery, and mode recognition and the fuzzy control system., are can made more excellent or unique of performance, even can address general information processing method by cannot solution of problem, is a has value of research direction. In addition, a combination of neural network and fuzzy computation, evolutionary computation, constitute an intelligent information processing system for Adaptive Computing, neural will have a new breakthrough of intelligent information processing, is a subject worthy of research.

Adaptive filter can automatically adjust to its own parameters, when we design it, we only need a few or simply does not require any a priori statistical knowledge of signal and noise, and thus it has a very wide range of applications, such as adaptive interference cancellation - back wave canceller. Telephone line echo is due to a two / four-wire conversion hybrid coil does not match the formation of [9]. Due to the imbalance of the mixing coil cause current leakage, part of the signal energy is reflected back to the signal source. This reflex path delay, make both sides of telephone can hear themselves or each other echo during a time lag, thus forming the echo.

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7. Conclusion

Auto adapted filtering is an important base for digital signal processing and it is under rapid developing recent years, the auto adapted filtering has received a wide range of applications in many areas. Facing the practical problems, there is an urgent need to study nonlinear Adaptive Filter theory and algorithms. The adapted filtering realized by LMS algorithm is just the base one in it, however, such techniques like neural network and artificial intelligence has its own advantages. So that we are able to achieve ideal result when using the combination of auto adapted filtering, neural networks and artificial intelligence to deal with the signal processing, this is also an up-to-the-moment research interests.
8. References


