

Performance Analysis of Basic Adaptive Filter Algorithms for DSP Processor in LabVIEW

Shivani Joshi¹ Darshit Jadawala²

^{1,2} Electronics & Communication Department

^{1,2} Gujarat Technological University Ahmedabad, Gujarat, India

Abstract— DSP Processor is a very efficient and error-correcting option in today's mobile communication age. There are many algorithms to optimize the performance of such processor. Here DSP processor is implemented along with adaptive filters for noise cancellation in mobile phones. This is implemented by using LabVIEW. Algorithm used here is NLMS. The performance is further studied and better algorithm has been searched out to improve the performance.

Keywords: - RMS, LMS, NLMS, Noise Canceller, Adaptive Filters

I. INTRODUCTION

Our goal in is to survey mobile processors. More specifically we summarize the main types of processor for mobile applications, their system software, and the related system hardware components. We classify mobile processors in four main groups: computation-oriented (e.g. laptops, handhelds, PDA, and calculators), communication-oriented (e.g. wireless phones), media-oriented (e.g. MP-3 players and digital cameras) and devices for emerging applications (e.g. smart cards and sensor networks).

The importance of digital signal processors (DSPs) for communications, and in particular mobile communications, has been ever increasing. Today DSPs present a key technology for executing base band modem and lower layer protocol functions.

These days communication has become a very major aspect of everyone's life. So an effective and reliable system is must. The mobile phones are also another name of communication. So it becomes a work of importance to constantly improve the performance of mobile phones. Now processors are the heart of these mobile phones. So we shall concentrate on various processors. The selection of proper processor out various options available is very crucial. So among such processors, DSP processor has emerged out as a very good and reasonable option.

Noise removal is a very challenging problem often encountered in numerous applications concerning speech since the spectral and power characteristics of both, speech and noise, are continuously and rapidly changing over time. As the use of conventional filters would lead to unacceptable distortion of the desired speech signal, it is more suitable to employ a filter with adjustable response so that it can adapt itself automatically to the speech and noise variations. Because of its ability to successfully track statistic changes, even when the input signals specifications are unknown, adaptive filters have been used in diverse speech enhancement applications.

II. DSP PROCESSOR IN MOBILE PHONES

A basic DSP block diagram is represented here. The two main blocks are Analog to Digital Converter n vice versa. Filters are used to minimize noise signal and aliasing signal.

A DSP contains these key components:

- Program Memory: Stores the programs to process data
- Data Memory: Stores the information to be processed
- Compute Engine: Performs the math processing, accessing the program and the data
- Input/ Output: Serves a range of functions to connect to the outside world

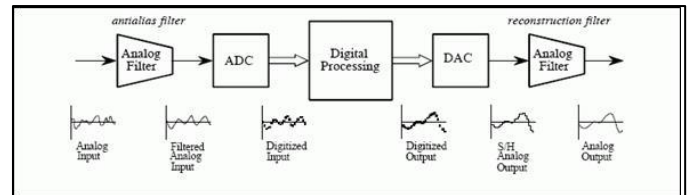


Fig. 1: Basic Key Components of DSP

DSP processors are a mandatory component in today's mobile phones. It is very instructive constructive to analyse the trends in DSPs in the last two decades. The DSP architecture has been strongly impacted by both the evolution in technology and the targeted applications.

The tremendous growth of development in the digital signal processing area has turned some of its specialized areas into fields themselves. If accurate information of the signals to be processed is available, the designer can easily choose the most appropriate algorithm to process the signal. When dealing with signals whose statistical properties are unknown, fixed algorithms do not process these signals efficiently. The solution is to use an adaptive filter that automatically changes its characteristics by optimizing the internal parameters. The adaptive filtering algorithms are essential in many statistical signal processing applications. The adaptive filter has the property that its frequency response is adjustable or modifiable automatically to improve its performance in accordance with some criterion, allowing the filter to adapt to changes in the input signal characteristics.

Adaptive Noise Cancellation is an alternative way of cancelling noise present in a corrupted signal. The principal advantage of the method is its adaptive capability, its low output noise, and its low signal distortion. Output noise and signal distortion are generally lower than can be achieved with conventional optimal filter configurations.

III. ADAPTIVE FILTERS

An adaptive filter is a filter that self adjusts its transfer function according to an optimizing algorithm. It adapts the performance based on the input signal. Such filters incorporate algorithms that allow the filter coefficients to adapt to the signal statics.

Subtracting noise from a received signal involves the risk of distorting the signal and if done improperly, it may lead to an increase in the noise level. This requires that the noise estimate n' should be an exact replica of n . If it were possible to know the relationship between n and n' , or

the characteristics of the channels transmitting noise from the noise source to the primary and reference inputs are known, it would be possible to make \hat{n} a close estimate of n by designing a fixed filter.

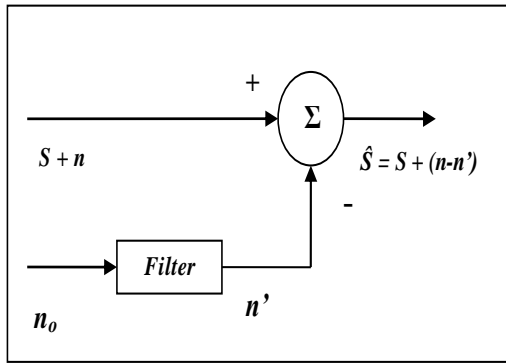


Fig. 2: Basic Adaptive Filter

The error signal to be used depends on the application. The criteria to be used may be the minimization of the mean square error, the temporal average of the least squares error etc. Different algorithms are used for each of the minimization criteria e.g. the Least Mean Squares (LMS) algorithm, the Recursive Least Squares (RLS) algorithm etc. To understand the concept of adaptive noise cancellation, we use the minimum mean-square error criterion.

The common adaptive algorithms that have found widespread application are the Least Mean Squares (LMS) and the Recursive Least Squares (RLS). Few more adaptive algorithms are discussed here.

A. LMS Algorithm

Among adaptive algorithms, LMS is most simple and computationally less expensive algorithm. Weight update equation for LMS is given by (1),

$$w(n+1) = w(n) + \mu v(n)e(n) \quad (1)$$

where $w(n)$ is weight vector for LMS adaptive filter. μ is learning rate and $0 < \mu < 1$. $e(n)$ is error signal and given by (2),

$$e(n) = [x(n) + v(n)] - y(n) \quad (2)$$

where $y(n) = \hat{v}(n)$. Learning rate for LMS algorithm can be set to 0.01 and filter order can be chosen 10.

Least mean squares (LMS) algorithms are used in adaptive filters to find the filter coefficients that relate to producing the least mean squares of the error signal (difference between the desired and the actual signal). It is a stochastic gradient descent method in that the filter is only adapted based on the error at the current time.

The idea behind LMS filters is to use the method of steepest descent to find a coefficient vector which minimizes a cost function.

B. RMS Algorithm

Root mean square (RMS) is a fundamental measurement of the magnitude of an ac signal. Its definition can be both practical and mathematical. Defined practically, the rms value assigned to an ac signal is the amount of dc required to produce an equivalent amount of heat in the same load.

RLS is relatively complex algorithm as compared to LMS and NLMS algorithm. Also performance of RLS in terms of convergence and Mean Square Error (MSE) is better than LMS and NLMS. RLS adaptation algorithm with

input signals $y(n)$ and $x(n)$ is given below. Initial values for RLS algorithm is given by (3) and (4).

$$\Phi_{yy}(n) = \delta I \quad (3)$$

$$w(0) = w_l \quad (4)$$

For $m = 1, 2, \dots$ filter gain update vector is given by (5),

$$K(n) = \lambda^{-1} \Phi_{yy}(n-1) y(n) / [1 + \lambda^{-1} y^T(n) \Phi_{yy}(n-1) y(n)] \quad (5)$$

Error signal equation is given by (6),

$$e(n) = x(n) - w^T(n-1) y(n) \quad (6)$$

Filter Coefficient adaptation is given by (7),

$$w(n) = w(n-1) - k(n)e(n) \quad (7)$$

Inverse correlation matrix update is calculated using the given equation

$$\Phi_{yy}(n) = \lambda^{-1} \Phi_{yy}(n-1) - \lambda^{-1} k(n) y^T(n) \Phi_{yy}(n-1) \quad (8)$$

The Recursive Least Squares (RLS) algorithm is based on the well-known least squares method. The least-squares method is a mathematical procedure for finding the best fitting curve to a given set of data points. This is done by minimizing the sum of the squares of the offsets of the points from the curve. The RLS algorithm recursively solves the least squares problem.

C. NLMS Algorithm

Normalized Least mean Square Algorithm (NLMS) is a variation of LMS algorithm having fast convergence as compared to LMS algorithm. In NLMS, the step size takes the form of,

$$\mu(n) = \frac{\beta}{\|x(n)\|^2} \quad (9)$$

Filter coefficient update equation for NLMS algorithm is given by (9),

$$z(n+1) = z(n) + \mu \{ v(n) / |v(n)|^2 \} e(n) \quad (10)$$

where $z(n)$ is weight vector for NLMS adaptive filter. μ is learning rate and $0 < \mu < 1$. An important limitation of adaptive algorithms is that of selection of certain value for the learning rate μ implies compromise between speed of convergence and steady state mis-adjustments.

D. SRLMS Algorithm

The signed regressor algorithm is obtained from the conventional LMS recursion by replacing the tap-input vector $x(n)$ with the vector $\text{sgn}\{x(n)\}$. Consider a signed regressor LMS based adaptive filter that processes an input signal $x(n)$ and generates the output $y(n)$ as per the following:

$$y(n) = w^T(n) x(n) \quad (11)$$

where, $w(n) = [w_0(n), w_1(n), \dots, w_{L-1}(n)]$ it is a L-th order adaptive filter.

The adaptive filter coefficients are updated by the Signed-regressor LMS algorithm as,

$$w(n+1) = w(n) + \mu \text{sgn}\{x(n)\} e(n) \quad (12)$$

Because of the replacement of $x(n)$ by its sign, implementation of this recursion may be cheaper than the conventional LMS recursion, especially in high speed applications such as biotelemetry these types of recursions may be necessary.

E. SLMS Algorithm

Sign Algorithm (SLMS) is obtained from conventional LMS recursion by replacing $e(n)$ by its sign. This leads to the following recursion,

$$w(n+1) = w(n) + \mu x(n) \text{sgn}\{e(n)\} \quad (13)$$

F. S-SLMS Algorithm

Sign – Sign Algorithm (S-SLMS) can be obtained by combining signed-regressor and sign recursions, resulting in the following recursion,

$$w(n+1) = w(n) + \mu \operatorname{sgn}\{x(n)\} \operatorname{sgn}\{e(n)\} \quad (14)$$

Where $\operatorname{sgn}\{ \cdot \}$ is well known signum function, $e(n) = d(n) - y(n)$ is the error signal. The sequence $d(n)$ is the so-called desired response available during initial training period. However the sign and sign – sign algorithms are both slower than the LMS algorithm. Their convergence behavior is also rather peculiar. They converge very slowly at the beginning, but speed up as the MSE level drops.

IV. IMPLEMENTATION AND RESULT SYNTHESIS

LabVIEW (trademark National Instruments) has been in existence since the mid 1980s and is normally associated with virtual instrumentation. Based on the graphical language ‘g’ it is a block-diagram approach to programming. It is perhaps a little harder to learn than approaches such as MATLAB but as will be shown the effort is more than worth the end results. There is still a wealth of information available on how to use LabVIEW to control a whole range of instrumentation but far less on the basics of signal processing.

This lab includes three examples showing how the LabVIEW DSP Module can be used to run DSP graphical codes directly on a DSP target board without performing any C programming. These examples correspond to the waveform generation, digital filtering, and adaptive filtering labs covered in the previous chapters. As stated earlier, to begin designing DSP systems by using the LabVIEW DSP Module, double-click on the LabVIEW Embedded Edition icon on the Windows desktop.

A. Basic Digital Filter

The basic approach to start with LabVIEW is to implement a Digital Filter in the DSP module. The filter and waveform can be modified using any of the filters. Here a most simple system with a low pass filter is implemented to start with. A low pass filter can be designed with desired cut-off frequency. Let us design a low pass filter with 2200 Hz. The specifications can be adjusted in the interactive graphical way in the configuration window. The VI file of the filter is shown here with its results.

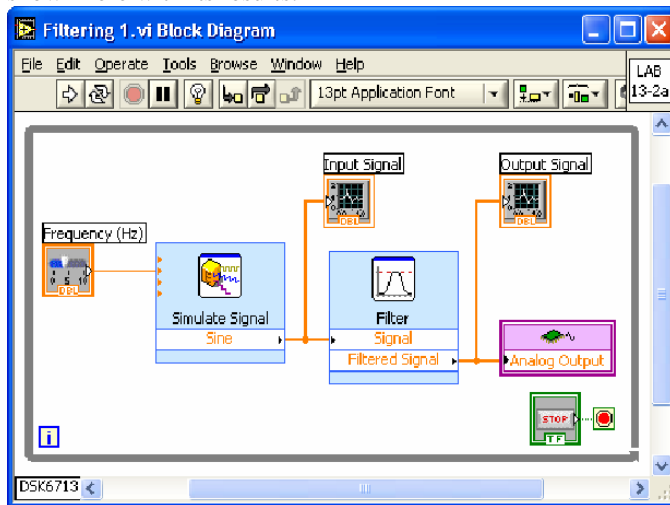
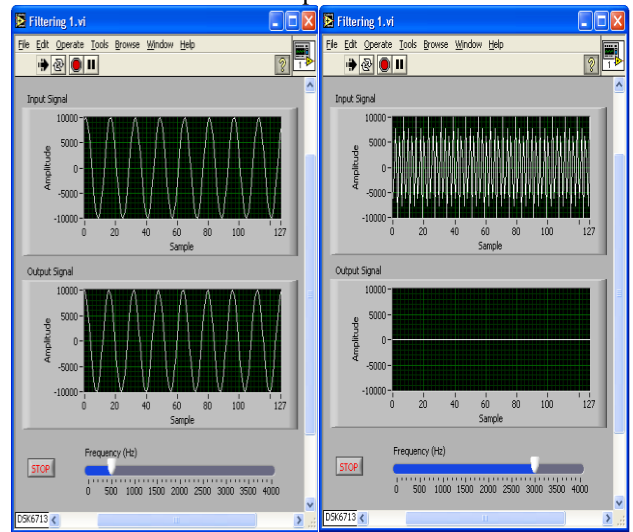


Fig. 3: Basic Digital Filtering using Filter Express VI

The results for the same are described here. Two instances of generated input and output signals are shown here. The figure 4.2 illustrates the input as well as output signals. The figure (a) shows the input signal with frequency 500 Hz with pass-band filter and its filtered output. And the figure (b) shows the input signal with 3000 Hz with stop-band filter with its filtered output.



(a) (b)
Fig. 4: Input and output Signal in (a) Pass-band (b) Stop-band

B. Basic Adaptive Filter

Adaptive techniques use algorithms, which enable the adaptive filter to adjust its parameters to produce an output that matches the output of an unknown system. This algorithm employs an individual convergence factor that is updated for each adaptive filter coefficient at each iteration.

A system is said to be adaptive when it tries to adjust its parameters with the aid of meeting some well-defined goal or target that depends upon the state of the system and its surroundings. So the system adjusts itself so as to respond to some phenomenon that is taking place in its surroundings.

The adaptive filter can prove to be a good alternative for noise cancellation in real time applications. These filters can therefore be used for various applications in mobile phones. The adaptive noise cancellation system is covered here using the LabVIEW DSP Module. The BD of the adaptive noise cancellation system is illustrated in Figure 5. Let us briefly mention the VIs and functions of this BD. The simplest algorithm has been used here. LMS algorithm block has been implemented.

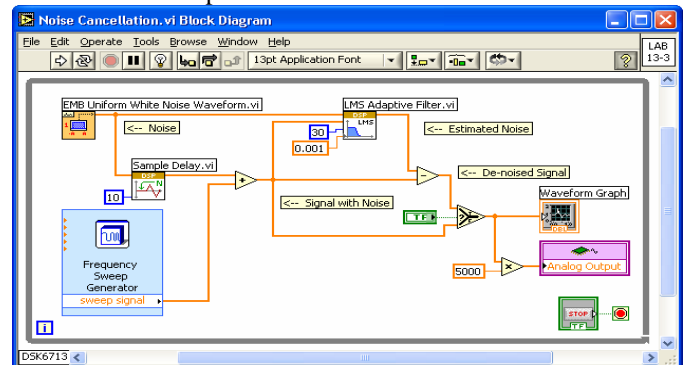


Fig. 5: Basic Adaptive Filter

The EMB Uniform White Noise Waveform VI is used to generate a white noise signal which is then added to an input signal. Before adding the noise signal, a delay is introduced by using the Sample Delay VI. This is done to simulate an ideal channel which causes a time delay with no gain or frequency change. The input signal consists of a chirp signal whose frequency sweeps between two frequencies. This is implemented by using the Frequency Sweep Generator Express VI. The configuration of this VI has to be done. This signal and the delayed noise signal are summed together to construct the signal to be processed.

Next, add the LMS Adaptive Filter VI to the BD. Notice that the output of the LMS Adaptive Filter VI corresponds to the estimated noise. Thus, the output of the LMS filter needs to be subtracted from the summed signal to obtain the de-noised signal.

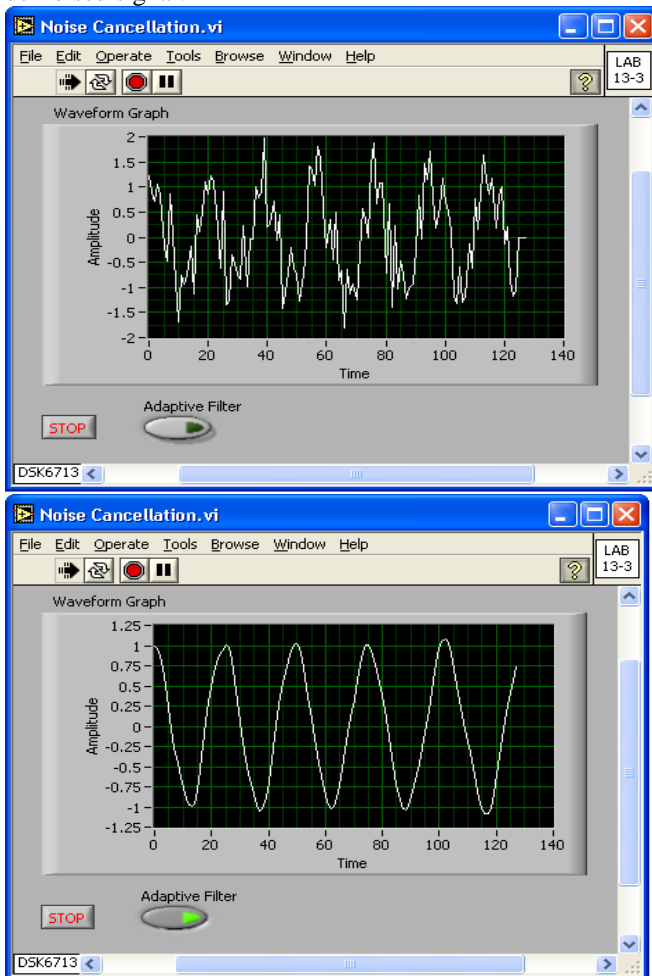


Fig. 6: Adaptive Cancellation Waveforms

Place the function Select to compare the noise corrupted input signal and the de-noised output signal. Either of these signals can be wired to the Waveform Graph and Analog Output function for sending out data through the codec output port. Figure 6 shows the noise corrupted input signal and the de-noised output signal.

V. CONCLUSIONS

We studied the LMS adaptive algorithm that is certainly the most popular among the adaptive-filtering algorithms. The attractiveness of the LMS algorithm is due to its simplicity and accessible analysis under idealized conditions. As

demonstrated in the present chapter, the noisy estimate of the gradient that is used in the LMS algorithm is the main source of loss in performance for stationary environments. Because of its simplicity, the LMS algorithm is the most popular adaptive algorithm. However, the LMS algorithm suffers from slow and data dependent convergence behavior.

Further More converging algorithm can be implemented. RMS can be used at the cost of increased complexity. Also it can be implemented in LabVIEW. For more proper results other LMS techniques like NLMS, SLMS, FxLMS, FuLMS, Feedback ANC, Hybrid ANC and others. Better prospective for adaptive filter coefficient can help to improve the results.

ACKNOWLEDGMENT

The authors are grateful to the HGCE for their encouragement and support. I am also honorably thankful to my external guide Prof. Nirav Chauhan for giving me the perfect and truly effective guidance. I am also thankful to Prof. Anand Patel and the all faculty members of E&C Department for providing me all the necessary guidance throughout the term which provided lots of help.

REFERENCES

- [1] J. A. Tavares Reyes, E. Escamilla Hernández, J. C. Sánchez García, "DSP-Based Oversampling Adaptive Noise Canceller for Background Noise Reduction for Mobile Phones", SEPI-ESIME Culhuacan, Instituto Politécnico Nacional, Mexico, IEEE - 2012.
- [2] Shivi Chaturvedi, "The Role of Digital Signal Processors (DSP) for 3G Mobile Communication Systems", ISSN : 0975-8364, International Journal on Emerging Technologies 1(1): 23-26(2010).
- [3] Sorin ZOICAN, "The Role of Programmable Digital Signal Processor (DSP) for 3G Mobile Communication Systems", Politehnica, University of Bucharest.
- [4] Adrian Mackenzie, "Intensive movement in wireless digital signal processing: from calculation to envelopment", CESAGen - Centre for Social and Economic Aspects of Genomics Lancaster University.
- [5] Farinaz Koushanfar, Vandana Prabhu, Miodrag Potkonjak, Jan M. Rabaey, "Processors for Mobile Applications", University of California, CA.
- [6] Katsuhiko Ueda, "Digital Signal Processors for Mobile Phone Terminals", Matsushita Electric Ind. Co., Ltd., 1991.
- [7] J.Gerado Avalos, Daniel Espinobarro, Jose Velazquez, Juan C. Sanchez, "Adaptive Noise Canceller using LMS Algorithm with Codified Error in DSP", National Polytechnic Institute ESIME, Culhuacan.
- [8] L.J. Ericsson, M.C.Allie, C.D.Bremigan, "Active Noise Control Using Adaptive Digital Signal Processing", Corporate Research Department, USA.
- [9] G. Nugent, A.O.Richardson, "Optimization of a Real-Time Adaptive Self-Tuning Filter Implemented on a Texas Instruments TMS320C30 Floating Point DSP", Department of Electrical/Electronic Engineering, California State University.
- [10] Jamil Chaoui, Sebastien de Gregoria, Guillaume Gallissian, "DSP Based Solution for Ambient Noise Reduction in Mobile Phones", Texas Instruments, France

- [11] M. Shell. (2002) IEEEtran homepage on CTAN. [Online]. Available: <http://www.ctan.org/tex-archive/macros/latex/contrib/supported/IEEEtran/>
- [12] FLEXChip Signal Processor (MC68175/D), Motorola, 1996.
- [13] "PDCA12-70 data sheet," Opto Speed SA, Mezzovico, Switzerland.
- [14] A. Karnik, "Performance of TCP congestion control with rate feedback: TCP/ABR and rate adaptive TCP/IP," M. Eng. thesis, Indian Institute of Science, Bangalore, India, Jan. 1999.
- [15] J. Padhye, V. Firoiu, and D. Towsley, "A stochastic model of TCP Reno congestion avoidance and control," Univ. of Massachusetts, Amherst, MA, CMPSCI Tech. Rep. 99-02, 1999.
- [16] Wireless LAN Medium Access Control (MAC) and Physical Layer (PHY) Specification, IEEE Std. 802.11, 1997.

