Design of Adaptive Filters Based on And LMS Algorithm: A Survey And Taxonomy

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Abstract- *In this paper, the concept of adaptive filtering has been proposed. It has been shown that signals often undergo variation in the noise environment encountered. Hence it becomes mandatory to design filters based on and adaptive noise cancellation. Such filters are called adaptive filters and are used for applications requiring real time noise cancellation such as speech processing, live video broadcasting etc. This paper proposes two basic approaches i.e. the Least Mean Square (LMS) algorithm and Noise shaping filter. The paper focuses on the various approaches used thus far for the purpose, citing the pros and cons of each approach.*

Keywords- Adaptive Filter Design, Noise Filtering, LMS Algorithm, Noise shaping filter, Signal to Noise Ratio, Dynamic Range, Quantization Noise

I. INTRODUCTION

In order to catch up with the present day technology advancements, sigma-delta converters are used with an aim of high level of reliability and functionality with reduced chip cost. It is applied in communication equipment, medical devices, automated production facilities, computers, weapons, navigation equipment, tools etc. Hence, if substantial analog signal processing (ASP) is performed, stochastic artifacts (noise) will accumulate, and the resulting signal may not represent the desired signal with the required significance.[l] needing adaptive filtering.

This paper focuses exclusively on the LMS algorithm and the delta-sigma modulation as chosen technique for adaptive filtering. Based on the combination of oversampling and quantization error shaping techniques Delta -sigma modulator achieve a high degree of insensitivity to analog circuit imperfections, thus making them a appropriate choice to realize embedded analog-to-digital interfaces in modern systems-on-chip (SoCs) integrated in nanometer CMOS. Oversampling is inherently implemented in most sigma-delta (X-A) ADCs with integrated digital filters, where the modulator clock rate is typically 32 to 256 times the signal bandwidth, but X- *A* ADCs are limited for applications that require fast switching between input channels.

In medical applications multiple devices face the hurdle of SNR e.g. Electromyogram (EMG), Electrocardiogram (ECG), therefore the paper focuses on medical devices that require higher SNR to improve its performance. High-performance data-acquisition signal chains used in medical equipment require wide dynamic range and high accuracy.

II. WORKING

The basic structure of an adaptive noise filter can be implemented using the noise shaping filter. A delta-sigma converter uses many samples from the modulator to produce a stream of l-bit codes. The delta- sigma ADC accomplishes this task by using an input- signal quantizer running at a high sample rate. The delta-sigma modulator takes an input and produces a stream of digital values same as other quantizers that represents the voltage of the input. The delta-sigma modulators are of two types the time and the frequency domain

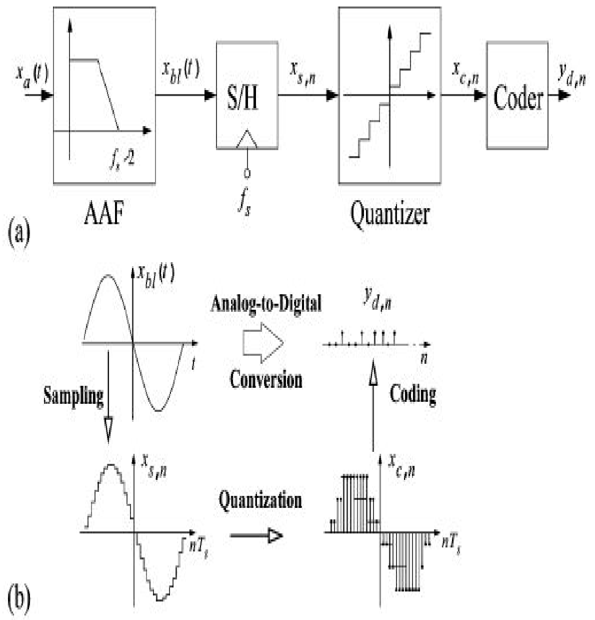

Fig.1 Basic Structure of an Adaptive Filter

An adaptive filter always has an error feedback loop often called the Noise shaping filter. The modulator in Fig. l illustrates a first order sigma-delta modulator. It comprises of an integrator, a 1-bit quantizer, and a 1-bit DAC. The integrator ramps the input signals up and down. The integrator acts as the circuit which shifts the noise from pass band to stop band. The output of the integrator is given to the comparator and then the comparator output is fed back through a 1-bit DAC to the Summing circuit. Oversampling is the process of taking more samples per second than required on the basis of the Nyquist- Shannon criterion. By changing the sampling rate the signal power and total quantization noise power is not affected. Therefore, the signal to quantization noise ratio is not changed. However, the quantization noise is spread over a larger frequency range, which reduces the spectral density of the quantization noise. The quantization noise power is reduced by 3 dB for every doubling of the oversampling ratio and the signal to quantization noise ratio is improved accordingly if the original Nyquist band is considered only. The oversampling ratio also affects the signal to noise ratio. If oversampling is increased, the signal to noise ratio is also increased exponentially.

III. PARAMETERS FOR ADAPTIVE FILTERING

(1) OVER SAMPLING RATIO:

When a significantly sampling frequency in a signal higher than the twice of bandwidth of digital samples known as Over sampling p, defined as

$$\mathbf{P = fs / 2B} \qquad (1)$$

Where fs is the sampling frequency, B is the bandwidth or highest frequency of the signal, the nyquist rate is 2B.[2]

The theoretical limit of the SNR of Associate in Nursing ADC activity is predicated on the quantisation noise owing to the quantisation error inherent within the analog-to-digital conversion method once there's no oversampling and averaging. Since the quantisation error depends on the quantity of bits of resolution of the ADC the simplest case SNR is calculated as a perform of the Effective range of Bits

$$\mathbf{SNR= (6.02*ENOB) - 1.767} \qquad (2)$$

for the Effective number of bits , using the measured SNDR

$$\mathbf{ENOB = SNDR - 1.76\ dB / 6.02dB/bit} \qquad (3)$$

Effective number of bits (ENOB) is simply the signal to noise-and-distortion ratio expressed in bits rather than decibels by solving the ideal SNR" equation [7] In the presentation of measured results, ENOB is identical to SNDR, with a change in the scaling of the vertical axis.

(2) QUANTIZATION AND QUANTIZATION ERROR

It is bound by [-A/2 to +A/2] where A represents the amplitude of the analog signal.

$Qe_{(Max)} = \Delta/2$

Here Δ represents the step size.

(3) :

The noise transfer function can be given by:

$$\mathbf{NTF\ (z) = (1 - z^{-1})\ L} \qquad (4)$$

Where L denotes the order of filter

(4) DYNAMIC RANGE:

Dynamic range is the parameter exhibiting the variation of the signal in the time domain. It is mathematically given by:

$$\mathbf{A-(-A) = 2A} \qquad (5)$$

(5) FIGURE-OF-MERIT:

The figure of merit is the inverse of the signal to noise ratio and is given by

$$\mathbf{FOM = 1/ SNR} \qquad \mathbf{(6)}$$

Comparison of the power efficiency of two AD converters that achieve identical signal conversion specifications, i.e. have the same sampling rate and realize the same SNR for every input signal, is an easy task; the one with the lowest power consumption is the best. Although the FoM of combining weight, (6) is wide used, it cannot be accustomed build honest comparisons between low resolution and high resolution AD converters. once the resolution of associate ADC is inflated, some extent is reached wherever thermal noise is limiting the SNR, so as to scale back the impact of the noise by three sound unit, capacitances have to be compelled to be doubled to extend the amount of effective bits by one, a six sound unit reduction of the noise is needed, which implies an element four increase in capacitance. Since power scales linearly with the quantity of capacitance to charge, the facility will increase with an element four. Thus, the FoM can become a minimum of an element a pair of worse once the ENOB is inflated by one.

The Error minimization gradient computed by the LMS algorithm is given by:

$$\min E[z^2] = E[s^2] + \min E[(n_0 - y)^2] \qquad (7)$$

IV. PREVIOUS WORK

Jiandong Cheng et al. (2016) have suggested a new method for approximately calculating SNR of switched capacitor Noise shaping filter. He suggested the truncation of NTF polynomial through Taylor's series with proper approximation which would provide the facility of fast calculation of SNR. For SNR calculation, the NTF is considered as rational function with the highest order of numerator and denominator should be same with equal coefficient. Through this method the sensitivity can also be measured with respect to the loop filter parameters.

Philip M. Choppet al. (2015) have proposed a frequency translating band pass Noise shaping filter that down converts the 4 MHz bandwidth signal from 225MHz to 25 MHz. The bandpass SDM uses a single path mixing within the closed feedback loop which will simplify the synchronization of Local Oscillator (LO) frequency at the receiver end. The author has used 6^{th} order loop filter, 3 bit quantizer and 100MHz as a sampling frequency. The bandpass is performed around the 25MHz results in reduction of sensitivity of coefficient variation. This paper also concluded that the feedforward feedback loop filter technology is beneficial for improving the linearity, and reducing power consumption and the sensitivity to timing errors.

HisatoFujisakaet al. (2014) have proposed sorter based arithmetic circuits for Sigma Delta Domain Signal Processing. Fabricating circuits in nanometer - scale introduces electromagnetic interference and also transient device error which can cause system failure too. To overcome the above problem Pulse Signal Processing (PSP) is used i.e. Sigma Delta Domain Signal Processing (SDSP). For designing arithmetic and functional operations on SD modulated signal, sorting networks are used with two bit manipulation and permutation and for bit reversal NOT gate is used. Author has presented arithmetic modules like adder, exponential function. By using these modules, several transcendental functions and log- domain arithmetic operators can be designed.

Jose M. de la Rosa (2013) has described different design techniques of Noise shaping filter. Fundamental of SDM contains the concept of oversampling and quantization. Noise shaping filter also known as modulator as it pushes the noise out of band of interest and the Noise Transfer Function behaves as High Pass Filter or Band stop Filter. The Signal Transfer Function behaves as Low Pass Filter or Band Pass Filter allowing the low bandwidth baseband signals. The author also illustrated the effects of increase in order of modulator, OSR and quantizer bit. The author has also emphasized on the benefits of Noise shaping filter as it provides high resolution, low power consumption, easy system - on -chip implementation in CMOS technologies, and can be used in number of application like digital receiver, Software defined Radio etc.

Philippe Benabeset al. (2011) have introduced high level system modeling for reducing the conception effort by using Fast Simulation techniques with MATLAB and VHDL AMS on Noise shaping filter. In this work, design methodology for macro models extraction and high level modeling for continuous time functions was proposed for improvement of speed. A sixth order Noise shaping filter is implemented from the extracted macro models which resulted in valid stable signal transfer function and improvement of simulation speed of about 30 times.

V. CONCLUSION

In this paper the basic approach of and noise filtering have been illustrated. Focus has been on the adaptive design parameters of filters. The critical aspects of filter design along with the relevant parameters have been explained. A summary of contemporary work in the field has also been explained. It is expected that the paper would pave the path for research in the relevant field of adaptive filter design.

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