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Low Power Adaptive Line Enhancer (ALE) Design for Noiseless Signal Processing

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Abstract- **In the case of the LMS adaptive filter, with coefficients being continuously updated, the decorrelating transform is applied to these calculated coefficients with minimal hardware or computational overhead. The correlation between filter coefficients is exploited to achieve a word length reduction from 16 bits down to 10 bits in the FIR filter block. The variable length update algorithm is based on the principle of optimizing the number of operational filter taps in the LMS adaptive filter according to operating conditions. The number of taps in operation can be increased or decreased dynamically according to the mean squared error at the output of the filter.The Adaptive Line Enhancer (ALE) is an effective learning filter for reducing Gaussian noise with a large SNR. The filter adjusts the filter weights to pass the desired input signal while reducing the noise portion of the signal with little to no filter roll-off up to the Nyquist rate (Fs/2). An adaptive filter can alter its own frequency response in order to improve the filter's performance on-the-fly. Therefore, minimizing the length of the equalizer will not result in poorer MSE performance and there is no disadvantage in having fewer taps in operation. If fewer taps are in operation then switching will not only be reduced in the arithmetic blocks but also in the memory blocks required by the LMS algorithm and FIR filter process. All design architectures are coded in Verilog hardware description language at register transfer level (RTL). Once functional specification of the design is verified, synthesis is carried out using either XILINX ISE to create a gate level net list.**

I.INTRODUCTION

Signal transmission using electronic signal processing. Transducers convert signals from other physical waveforms to electric current or voltage waveforms, which then are processed, transmitted as electromagnetic waves, received and converted by another transducer to final form. The signal on the left looks like noise, but the signal processing technique known as the Fourier transform (right) shows that it contains five well defined frequency components. Signal processing concerns the analysis, synthesis, and modification of signals, which are broadly defined as functions conveying, "information about the behavior or attributes of some phenomenon", such as sound, images, and biological measurements. For example, signal processing techniques are used to improve signal transmission fidelity, storage efficiency, and subjective quality, and to emphasize or detect components of interest in a measured signal.

II.DIGITAL SIGNAL PROCESSING

Digital signal processing is the processing of digitized discrete-time sampled signals. Processing is done by general-purpose computers or by digital circuits such as ASICs, field-programmable gate arrays or specialized digital signal processors (DSP chips). Typical arithmetical operations include fixed-point and floating-point, real-valued and complex-valued, multiplication and addition. Other typical operations supported by the hardware are circular buffers and look-up tables. Examples of algorithms are the Fast Fourier transform (FFT), finite impulse response (FIR) filter, Infinite impulse response (IIR) filter,and adaptive filters such as the Wiener and KalmanfiltersApplication fields Seismic signal processing, Audio signal processing , Digital signal processing, Speech signal processing , Image processing, Video, Wireless communication Control systems, Array processing , Process control

A.Active Noise Control

Noise is any unwanted disturbance present in a signal. Noise cancellation can beused in areas where noise can be harmful to one's hearing, such as: engine rooms or aircraft runways. In signal processing, noise is data within the wanted signal that carries no real value. Noise cancellation involves removing an unwanted noise while keeping the source sound.

There are two techniques for cancelling noise: passive noise reduction and active noise cancellation.

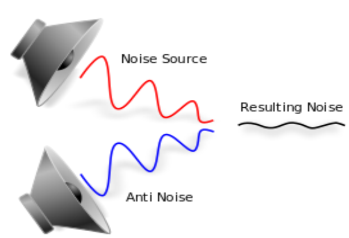

Figure 2.1 Graphical depiction of active noise reduction

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Active noise control (ANC), also known as noise cancellation, or active noise reduction (ANR), is a method for reducing unwanted noise by the addition of a second sound specifically designed to cancel the first.Active noise cancellation involves creating a supplementary signal that deconstructive interferes with the outside, ambient noise.

B.active vs. Passive noise controller

Noise control is an active or passive means of reducing sound emissions, often for personal comfort, environmental considerations or legal compliance. Active noise control is sound reduction using a power source. Passive noise control is sound reduction by noise-isolating materials such as insulation, sound-absorbing tiles, or a muffler rather than a power source.

Active noise canceling is best suited for low frequencies. For higher frequencies, the spacing requirements for free space and zone of silence techniques become prohibitive. In acoustic cavity and duct based systems, the number of nodes grows rapidly with increasing frequency, which quickly makes active noise control techniques unmanageable. Passive treatments become more effective at higher frequencies and often provide an adequate solution without the need for active control.

C.Least Mean Square Filter

The idea behind a closed loop adaptive filter is that a variable filter is adjusted until the error (the difference between the filter output and the desired signal) is minimized. The Least Mean Squares (LMS) filter and the Recursive Least Squares (RLS) filter are types of adaptive filter.

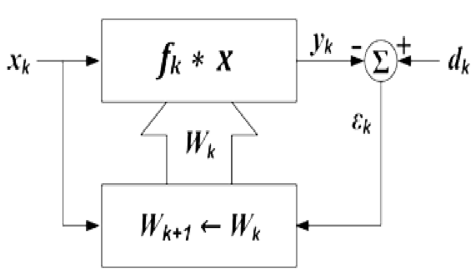

Figure 2.2 Least mean square filter

Where k = sample number, x = reference input, X = set of recent values of x, d = desired input, W = set of filter coefficients, ε = error output, f = filter impulse response, * = convolution, Σ = summation, upper box=linear filter, lower box=adaption algorithm.

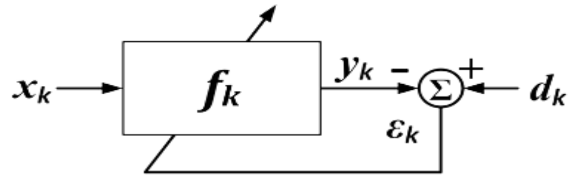

Fig 2.3 Adaptive filter in compact representation

Above is Adaptive Filter in compact representation, where k = sample number, x = reference input, d = desired input, ε = error output, f = filter impulse response, Σ = summation, box=linear filter and adaption algorithm.There are two input signals to the adaptive filter: dk and xk which are sometimes called the primary input and the reference input respectively.

III.LMS ALGORITHM

If the variable filter has a tapped delay line FIR structure, then the LMS update algorithm is especially simple. Typically, after each sample, the coefficients of the FIR filter are adjusted as follows for μ is called the convergence factor.The LMS algorithm does not require that the X values have any particular relationship; therefor it can be used to adapt a linear combiner as well as an FIR filter. In this case the update formula is written as:The effect of the LMS algorithm is at each time, k, to make a small change in each weight. The direction of the change is such that it would decrease the error if it had been applied at time k. The magnitude of the change in each weight depends on μ, the associated X value and the error at time k.The weights making the largest contribution to the output, are changed the most. If the error is zero, then there should be no change in the weights. If the associated value of X is zero, then changing the weight makes no difference, so it is not change.

IV.LEAST MEAN SQUARE (LMS) ADAPTIVE LINE ENHANCER (ALE) DESIGN

The Adaptive Line Enhancer (ALE) is an effective learning filter for reducing Gaussian noise with a large SNR. The filter adjusts the filter weights to pass the desired input signal while reducing the noise portion of the signal with little to no filter roll-off up to the Nyquist rate (Fs/2). An adaptive filter can alter its own frequency response in order to improve the filter's performance on-the-fly. The following design topics will be covered:

A. Features

- ALE filter architecture design fundamentals
- Filter stability dependant on learning coefficient, mu
- Filter scaling for more efficient implementation
- Custom VHDL component design and instantiation to include an 18-bit CODEC controller and ALE filter
- Filter characteristics and validation

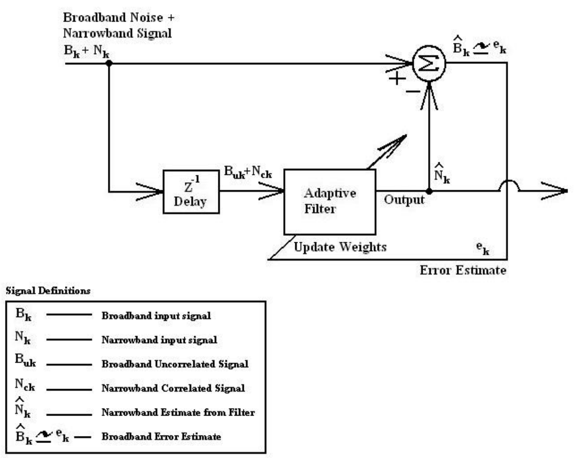

Figure 4.1 – ALE Signal Flow Diagram

The ALE filter works by initializing the filter weights to arbitrary values and adjusting them each sample period. This is done by computing the filter output N-hat-k, using equation 1 below. This narrow band signal (N-hat-k) is used to compute the error estimate (ek), see equation 2 below. The ALE implements a least-squares-error cost function to find optimal filter weights to estimate the signal output. Instead of using the expected values from the gradient vector in the parabolic mean-square-error cost function, the current input FIFO values (the instantaneous gradient) are multiplied by a learning coefficient (mu). This adjusts the weights by using a least-mean-square approximation; a step in the greatest decent is taken using equation 3 below. The decent moves down a parabolic structure in n-dimensions, the dimension is determined by the order of the filter.

When the gradient is zero the least-mean-squared error has been maximally reduced. To avoid overstepping, we take small steps determined by the learning coefficient mu. The mu value is chosen by using equation 4 below. This mu value governs how much the filter values change in a given sample period. It is handy to choose a value for mu which allows one to simply barrel shift the product of the error estimate and FIFO values, but also important to keep mu small enough to avoid divergence from the error surface of the n-dimensional parabola. A simplified representation of the Mean-Squared-Error surface with two weights can be seen below

B.Adaptive Line Enhancer Filter Architecture

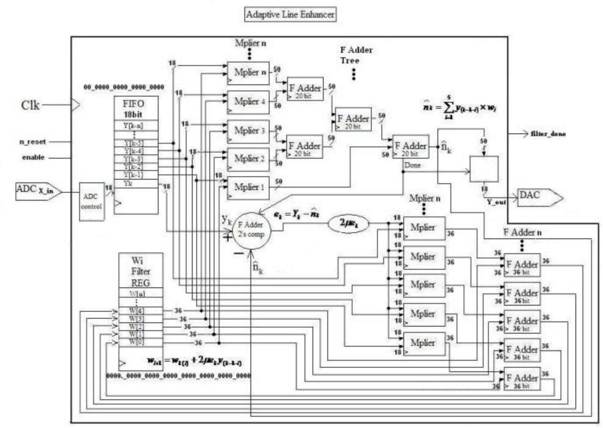

Figure 4.2 *Adaptive Line Enhancer Filter Architecture*

The good convergence and good stability of NLMS algorithm have made it preferable. Adaptive filtering comprises one of the kernel technologies in digital signal processing and are used extensively admitting system identification, adaptive equalization, adaptive noise cancellation, wireless communication and echo cancellation. As compared conventional LMS it has been proven that NLMS algorithm has good behavior. A novel pure-hardware design of LMS-based adaptive FIR filter core which is highly efficient in FPGA area/resource utilization and speed is proposed by OmidSharifi-Tehrani in 2011.Unlike HW/SW co-design and other pure-hardware methods, the required area/resource is reduced while keeping the speed in an appropriate level by taking advantage of Finite State Machine (FSM) and using internal block-rams (BRAM). This model because of being completely general (device independent), gives the ability of implementation on different FPGA brands and thus, is suitable for embedded systems, system-on-programmable-chip (SoPC) and network-on-chip (NoC) applications.

C.Advantage Of The Proposed Scheme

Adaptive filtering has been successfully applied in such diverse fields as communications, radar, sonar, and biomedical engineering. Although these applications are indeed quite different in nature, nevertheless, they have one basic common feature: an input signal and desired response are used to compute the error, which is in turn used to control the values of a set of adjustable filter coefficients. However, the main difference among the various applications of adaptive filtering arises in the manner in which the desired response is extracted.

Adaptive Noise Cancellation primarily deals with the generation of anti-noise which in theory would cancel out the ambient noise. Adaptive noise cancellation can be broadly divided into three different setups depending on the position of microphone/s, with each setup having its advantage over the other – FeedforwardNoise Cancellation, Feedback Noise Cancellation and Hybrid Noise Cancellation.LMS adaptive filters, may be implemented using Field Programmable Gate Arrays (FPGAs) due to some of their attractive advantages.

- Flexibility and programmability.
- The LMS adaptive filter enjoys a number of advantages over other adaptive algorithms, such as robust behavior when implemented in finite-precision hardware, well understood convergence behavior and computational simplicity for most situations as compared to least square methods .
- The Adaptive Line Enhancer (ALE) is an effective learning filter for reducing Gaussian noise with a large SNR.
- The filter adjusts the filter weights to pass the desired input signal while reducing the noise portion of the signal with little to no filter roll-off up to the Nyquist rate (Fs/2).
- An adaptive filter can alter its own frequency response in order to improve the filter's performance on-the-fly.

V.RESULT & PERFORMANCE ANALYSIS

A.Hardware Implementation

The module was thoroughly tested through the use of Xilinx's simulation software, Modelsim. A basic timer-driven state machine controls enable flags for synchronizing the math operations, truncating and dithering intermediate operations, and latching in the output each sample period. Each sample period advances the delay lines and starts the flag state machine.

The module was also verified in hardware as seen above in Figures 1 through 3. The purpose of the hardware implementation is to give some common platform and fair comparison between our proposed architecture and similar previous designs. The focus in this study is not targeted toward the details of the architecture implementation; instead our aim is to extract the hardware time and area parameters of the main blocks to build a fair comparison study between the designs. Therefore, our implementation exploration here is going to be limited to the level needed.

B. Xilinx ISE & ModelSim Simulator

Xilinx ISE (Integrated Software Environment) is a software tool produced by Xilinx for synthesis and analysis of HDL designs, enabling the developer to synthesize ("compile") their designs, perform timing analysis, examine RTL diagrams, simulate a design's reaction to different stimuli, and configure the target device with the programmer.Xilinx ISE provides an integrated flow with the Model Technology ModelSim simulator which enables simulation to run from the Xilinx Project Navigator graphical user interface. After ModelSim is installed and configured in your ISE session preferences, all applicable ModelSim simulation Processes and Properties are available to you in the Processes tab.

C. VHDL Design

VHDL is commonly used to write text models that describe a logic circuit. Such a model is processed by a synthesis program, only if it is part of the logic design. A simulation program is used to test the logic design using simulation models to represent the logic circuits that interface to the design. This collection of simulation models is commonly called a testbench.A VHDL simulator is typically an event-driven simulator.

This means that each transaction is added to an event queue for a specific scheduled time. E.g. if a signal assignment should occur after 1 nanosecond, the event is added to the queue for time +1ns. Zero delay is also allowed, but still needs to be scheduled: for these cases Delta delay is used, which represent an infinitely small time step. The simulation alters between two modes: statement execution, where triggered statements are evaluated, and event processing, where events in the queue are processed.

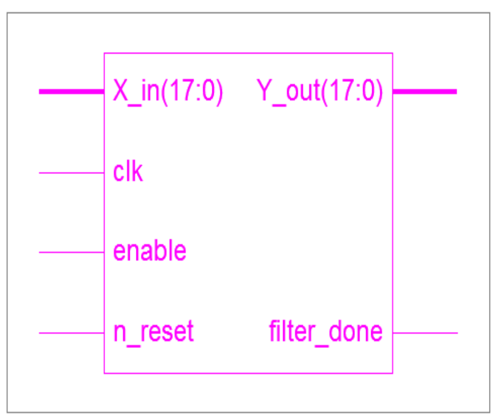

Figure 5.1 – Xilinx ISE Schematic Top Level

Table 1. Power summary

Power summary:	I(mA)	P(mW)
Total estimated power consumption:		419
Vccint 1.00V:	273	273
Vccaux 2.50V:	57	143
Vcco25 2.50V:	1	3
Clocks:	0	0
Inputs:	2	2
Logic:	33	33
Quiescent Vccint 1.00V:	238	238
Quiescent Vccaux 2.50V:	57	143
Quiescent Vcco25 2.50V:	1	3

Table 2. Device utilizations Summary

Slice Logic Utilization	Used	Available	Utilization
Number of Slice Registers	1,404	28,800	4%
Number used as Flip Flops	1,404		
Number of Slice LUTs	1,843	28,800	6%
Number used as logic	1,811	28,800	6%
Number using O6 output only	1,340		
Number using O5 output only	464		
Number using O5 and O6	7		
Number used as exclusive route-thru	32		
Number of route-thrus	503	57,600	1%
Number using O6 output only	496		
Number using O5 output only	7		
Slice Logic Distribution			
Number of occupied Slices	685	7,200	9%
Number of LUT Flip Flop pairs used	2,182		
Number with an unused Flip Flop	778	2,182	35%
Number with an unused LUT	339	2,182	15%
Number of fully used LUT-FF pairs	1,065	2,182	48%
Number of unique control sets	20		
IO Utilization			
Number of bonded IOBs	40	220	18%
Specific Feature Utilization			
Number of BUFG/BUFGCTRLs	1	32	3%
Number used as BUFGs	1		
Number of DSP48Es	48	48	100%
Total equivalent gate count for design	28,269		
Additional JTAG gate count for IOBs	1,920		

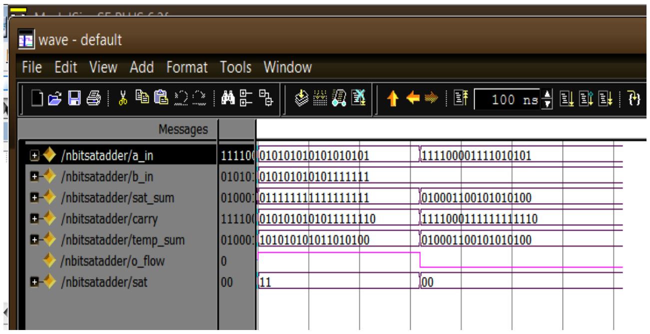

Figure 5.2 Simulation result

VI.CONCLUSION

The adaptive filter excels in filtering signals where one might experience changing conditions, spectral overlap between wide-band noise and the narrow-band signal, or in applications where the noise source is unknown or contentiously changing. This design was targeted and tested in audio applications, although the design can be scaled down for a faster filter with less resolution. The fundamentals of ALE VHDL module's filter design have also been covered.

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