International Journal of Engineering and Technical Research (IJETR)
ISSN: 2321-0869, Volume-3, Issue-4, April 2015

Design and synthesis of fixed-point IIR digital filters architecture for using active filtering

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***Abstract*— The objective of this research was to define a methodology for designing fixed-point IIR digital filters using modeling tools. A significant observation realized during the course of this research dealt with the parallel structure implementation. Designing more than two transfer functions sections introduces the problem on how to go about summing the sections together. Because of the fixed-point representation, the non-linear aspect of summing could potentially be a setback as was the case with trying to implement the data communications/imaging bandwidth. Deciding which sections to add together requires future work in terms of analysis**

Memory elements in the final filter design. Combinational logic delay is a factor a digital designer must deal with when moving to lower levels of design abstraction. Incorporating a parallel register methodology significantly reduces the effect of this problem. It reduces this timing problem by eliminating potentially timing violations at the filter output. Overall, the methodology outlined in this research is technically sound because it provides an interface between DSP design techniques and digital design. This design which is used for correction of harmonics current produced by non linear loads .the designed hardware architecture of the digital filter was coded in VHDL for FPGA implementation.

***Index Terms*— VHDL,IIR,,DSP,SOC.**

I. INTRODUCTION

The use of embedded System on Chip (SoC) solutions that modern Field Programmable Gate Arrays (FPGAs) offer. Specifically, it will investigate their use in efficiently implementing digital filtering applications, Different architectures for the digital filter will be compared with LMS algorithms implemented in the FPGA fabric only, to determine the optimal system architecture. This should, in theory and under the right circumstances, it is an attractive option for wireless communication as it can provide benefits like multiple accesses, high data transmission rate, efficient modulation schemes, low power consumption and reduced interference. This communication technology offers a solution for the bandwidth, cost effective and reduced power consumption and dimension of size requirements of next-generation consumer electronic device the Digital Filters are the discrete-time systems used mainly for filtering of arrays and sequences. The arrays or sequences are obtained by sampling the input analog signals .The digital filters perform the frequency related operations such as low pass, high pass band pass band reject and all pass etc. Also the design specification include cut-off frequency sampling frequency of input signal pass band variation stop band attenuation approximation type of filter and realization form Digital filter may be realized through hardware and software. Actually the software digital filter needs hardware for their operation Based on combining ever increasing.

Computer processing speed with higher sample rate processors, Digital Signal Processors (DSP's) continue to receive a great attention in technical literature and new product design. The following section on digital filter design reflects the importance of understanding and utilizing this technology to provide precision standalone digital or integrated analog/digital product solutions utilizing DSP's capable of sequencing and reproducing hundreds to thousands of discrete elements design models can simulate large hardware structures at relatively low cost. DSP techniques can perform functions such as. Digital filters are a very important part of DSP. In fact, their extraordinary performance is one of the key reasons that DSP has become so popular. As mentioned in the introduction, filters have two uses: signal separation and signal restoration. Signal separation is needed when a signal has been contaminated with interference, noise, or other signals. For example, imagine a device for measuring the electrical activity of a baby's heart (EKG) while still in the womb. The raw signal will likely be corrupted by the breathing and heartbeat of the mother. A filter might be used to separate these signals so that they can be individually analyzed.

Signal restoration is used when a signal has been distorted in some way. For example, an audio recording made with poor equipment may be filtered to better represent the sound as it actually occurred. Another example is the de blurring of an image acquired with an improperly focused lens, or a shaky camera. These problems can be attacked with either analog or digital filters. Which is better? Analog filters are cheap, fast, and have a large dynamic range in both amplitude and frequency. Digital filters, in comparison, are vastly superior in the level of performance that can be achieved. Digital filters can achieve thousands of times better performance than analog filters. This makes a dramatic difference in how filtering problems are approached. With analog filters, the emphasis is on handling limitations of the electronics, such as the accuracy and stability of the resistors and capacitors. In comparison, digital filters are so good that the performance of the filter is frequently ignored. The emphasis shifts to the limitations of the signals, and the theoretical issues regarding their processing.

Manuscript received April 21, 2015.
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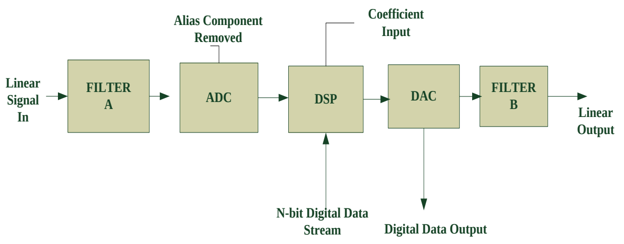


Figure-1 Typical Digital Filter Configuration

(a)Infinite Impulse Response Filter: Digital filter are linear time invariant (LTI) systems which are characterized by unit sample responses. The IIR system has infinite duration unit sample response i.e

$$H(n)=0 \quad \text{for } n<0 \tag{1}$$

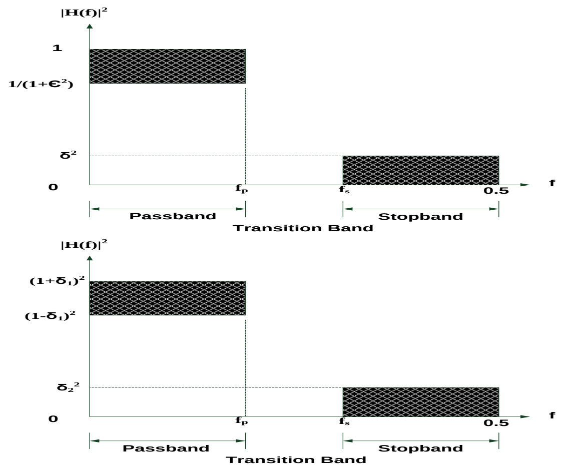
This unit samples response exists only for the duration from 0 to ∞.Therefore, this is an IIR system. There are various techniques which are available for the design of digital filter having infinite duration unit impulse response. This means that the design of an IIR filter involves design of a digital filter in the analog domain and transforming the design into digital domain.

II. Design of IIR Filters

A filter is generally designed to satisfy a frequency response specification. IIR filter design normally focused on satisfying a magnitude response specification. If the phase response is essential, it is usually satisfied by a phase compensation filter, such as an all pass filter we will adopt a magnitude specification that is normalized so that the maximum magnitude is 1. The magnitude square in the pass band must be at least 1= (1 /1+ϵ_2) and at most 1. While it must be no larger than d_2 in the stop band. The pass band edge is denoted by fp and the stop band edge by fs .Shows such a specification for a low-pass filter (LPF). The region between the pass band and the stop band is the transition band. There is no constraint on the response in the transition band.

Specifications for other types of filters (high-pass, band pass, and bands top) are similar

We can classify various IIR filter design methods into three categories: the design using analog prototype filter, the design using digital frequency transformation, and computer-aided design. In the first category, an analog filter is designed to the (analog) specification and the analog filter transfer function is transformed to digital system function using some kind of transformation. The second category assumes that a digital LPF can be designed. The desired digital filter is obtained from the digital LPF by a digital frequency transformation. The last category uses some algorithm to choose the coefficients so that the response is as close (in some sense) as possible to the desired filter. Design methods in the first two categories are simple to do, requiring only a handheld calculator. Computer-aided design requires some computer programming, but it can be used to design nonstandard filters for the duration



III MODEL OF THE PROPOSED WORK

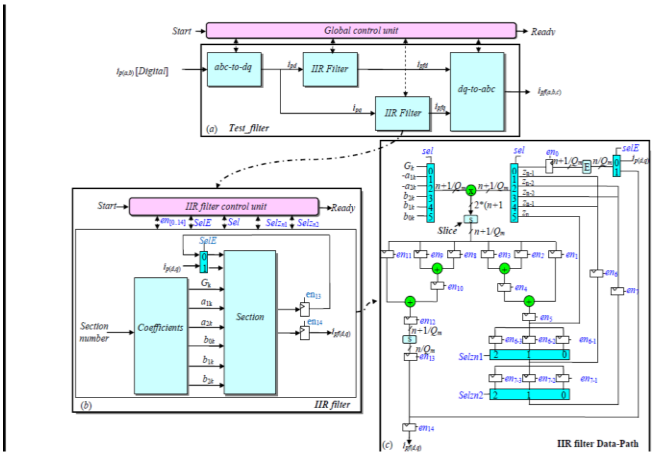


Figure 1.Overview of Designed of hardware architecture (b) IIR Module (c) IIR Filter Path

The objectives of our research work will be:

Write VHDL Code of digital Filter FPGA technology. Analysis of Active filtering application. Improvement of performance parameters which overcome multi-dimensional optimization problems for designer. To simulate the design using nanometer technology in Xilinx tool.

III. CONCLUSION

The objective of this research was to define a methodology for designing fixed-point IIR digital filters using modeling tools. A significant observation realized during the course of this research dealt with the parallel structure implementation. Designing more than two transfer functions sections introduces the problem on how to go about summing the sections together. Because of the fixed-point representation, the non-linear aspect of summing could potentially be a setback as was the case with trying to implement the data communications/imaging bandwidth. Deciding which sections to add together requires future work in terms of analysis. The cascade structure implementation provides the designer better control in terms of handling the interface from section to section

IV. SCOPE OF RESEARCH &FUTURE WORK

future work improve more and more reliable using association rules in making of digital filter for any production company .The hardware architecture of the IIR filter was designed having in mind the reduction of the hardware resources and only one multiplier was used for the whole architecture. Moreover, a very low computation time

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