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Area Efficient Interpolator Using Half-Band Symmetric Structure

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Abstract: In this paper a cost effective Interpolator has been designed and simulated. An area efficient method has been presented to implement cost effective interpolator for wireless communication systems. Interpolator is particularly useful for smoothing signals such as sinusoids or baseband I/Q waveforms. For these signals, interpolation filter is used to accurately produce new samples of the waveform without reducing signal quality. In this paper three structures for interpolator has been used namely Direct Form FIR Polyphase, Nyquist Filter and Half-Band Low pass Filter. The developed interpolator has been compared for performance and implementation cost using Mat Lab. The results show that the performance of all the designs is almost similar, but cost variation is very high. The Half-Band structure shows 52% of reduction in multipliers as compared to Direct Form FIR Polyphase structure. The Half-Band structure is further implemented in Transposed Structure and Symmetric Structure. The Half-Band symmetric structure shows 27% reduction in multipliers as compared to Direct Form FIR Polyphase, Nyquist Filter and Half-Band Low pass Filter structure to provide cost effective solution for wireless applications.

Index Terms: DSP, Filter, FIR, Interpolator

I. INTRODUCTION

The widespread use of digital representation of signals for transmission and storage has created challenges in the area of digital signal processing. The applications of digital FIR filter and up/down sampling techniques are found everywhere in modem electronic products. For every electronic product, lower circuit complexity is always an important design target since it reduces the cost. There are many applications where the sampling rate must be changed. Interpolators and decimators are utilized to increase or decrease the sampling rate. Up-sampler and down sampler are used to change the sampling rate of digital signal in multi rate DSP systems [1].This rate conversion requirement leads to production of undesired signals associated with aliasing and imaging errors. So some kind of filter should be placed to attenuate these errors. Today's consumer electronics such as cellular phones and other multimedia and wireless devices often require multirate digital signal processing (DSP) algorithms for several crucial operations in order to increase speed, reduce area and power consumption [2]. Due to a growing demand for such complex DSP applications, high performance, low-cost Soc implementations of DSP algorithms are receiving increased attention among researchers and design engineers. The recent great growth of portable digital audio products has increased the demand for an audio digital-to-analog converter (DAC) [3]. As we know, there are several key specifications for a DAC intended for portable audio systems, except for the low power dissipation, high performance and small area are also required. Recent designs mostly concern high performance, but ignore the consideration of area [4]. The number of portable digital audio products is tremendously increasing in recent days. The main concerns for these portable devices are their cost and power consumption. This has created interest among the researchers to find efficient implementation architectures. Interpolator is an integral part of digital audio DAC which is used to relax the stringent design requirements for analog filters. The recent trend in the design of audio interpolation filters is focused on the use of FIR based approaches. Digital for high quality audio DACs use a multistage cascade of comb and FIR filters[5]. Multirate filters are the digital filters that can increase or decrease the sample rate of the input sampled signal. The examples of multirate filters are the Interpolation and Decimation filters that increases or decreases the sampling rate of a signal respectively. Interpolation is the process of inserting new sample values between existing samples. Interpolation rate change by an integer factor has been used in many modern digital communication systems. This can be easily done using different kind of FIR filters like Equiripple FIR filter, Half-Band Filter [6]. Multirate systems have gained popularity since the early 1980s and they are commonly used for audio and video processing, communications systems, and transform analysis to name but a few. In most applications multirate systems are used to improve the performance, or for increased computational efficiency. Multirate digital filters and filter banks find application in communications, speech processing, image compression, antenna systems, analog voice privacy systems, and in the digital audio industry. During the last several years there has been substantial progress in multirate system research Industry [7]. During the last several years there has been substantial progress in multirate system research. In most applications multirate systems are used to improve the performance, or for increased computational efficiency. The two basic operations in a multirate system are decreasing (decimation) and increasing (interpolation) the sampling-rate of a signal. Multirate systems are sometimes used for sampling-rate conversion, which involves both decimation and interpolation [8]. The applications of digital FIR filter and up/down sampling techniques are found everywhere in modem electronic products. For every electronic product, lower circuit complexity is always an important design target since it reduces the cost, which is a basic component of all signal processing and telecommunication systems.

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The primary functions of a filter are one or more of the followings: to confine a signal into a prescribed frequency band or channel for example as in anti-aliasing filter or a radio/TV channel selector, to decompose a signal into two or more sub-band signals for sub band signal processing, for example in music coding, to modify the frequency spectrum of a signal, for example in audio graphic equalizers, and to model the input-output relation of a system such as a mobile communication channel, voice production, musical instruments, telephone line echo, and room acoustics [9]. The theory of multirate signal processing has proven itself useful in a variety of applications over the last ten years. It is in communications, however, that multirate systems have thus far had the most impact. For example, the transforms used today in state-of-the-art image and video compression algorithms are themselves multirate systems. While multirate theory has had its greatest impact on compression systems, it has also influenced other areas of the communications field. In particular, multirate systems form the basis of time-frequency scrambling methods for secure voice communications and they are now also being used to generate broadcast waveforms for code orthogonal frequency division multiple access [10].

II. MULTI RATE SIGNAL PROCESSING

Multirate systems have gained popularity since the early 1980s and they are commonly used for audio and video processing, communications systems, and transform analysis to name but a few. In most applications multirate systems are used to improve the performance, or for increased computational efficiency. Multirate signal processing refers to systems which allow sequences which arise from different sampling rates to be processed together. A digital signal processing system that uses signals with different sampling frequencies is probably performing multirate digital signal processing. Multi-rate processing and sample rate conversion, or interpolation and decimation as they are known, are a clever digital signal processing (DSP) techniques that broadband and wireless design engineers can employ during the system design process. Multi-rate processing finds use in signal processing systems where various sub-systems with differing sample or clock rates need to be interfaced together. At other times multi-rate processing is used to reduce computational overhead of a system. For example, an algorithm requires k operations to be completed per cycle. By reducing the sample rate of a signal or system by a factor of M, the arithmetic bandwidth requirements are reduced from kfs operations to kfs/M operations per second. The two basic operations in a multirate system are decreasing (decimation) and increasing (interpolation) the sampling-rate of a signal. Multirate systems are sometimes used for sampling-rate conversion, which involves both decimation and interpolation. In multirate digital signal processing the sampling rate of a signal is changed in order to increase the efficiency of various signal processing operations.

III. UP-SAMPLER

Digital filtering (interpolation) shifts spectral images to be centered around the new, interpolated sample rate. This effect has two advantages. First, because the nonzero rise time of a signal generator acts like a natural low pass filter, the high-frequency images experience slight attenuation. The second advantage of interpolation is that shifting spectral images to higher frequencies enables them to be more significantly attenuated by an analog low pass filter on the instrument. Up-sampler is basic sampling rate alteration device used to increase the sampling rate by an integer factor, an up-sampler with an up-sampling factor L, where L is a positive integer. It develops an output sequence xu[n] with a sampling rate that is L times larger than that of the input sequence x[n]. The Up-sampler is shown in figure1. Up-sampling operation is implemented by inserting equidistant zero-valued samples between two consecutive samples of x[n]. The input and output relation of up sampler can be expressed as

$$Xu[n] = \begin{cases} x[n/L], & n=0, \pm L, \pm 2L \ \ \pm 2L \\ 0, & \text{otherwise} \end{cases} \qquad (1)$$

Figure.1. Up Sampler

The zero-valued samples inserted by the up-sampler are replaced with appropriate nonzero values using some type of filtering process called interpolation. The input-output relation of an up-sampler with factor of 2 in the time-domain is given by

$$Xu[n] = \begin{cases} x[n/L], & n=0, \pm 2, \pm 4 \\ 0, & \text{otherwise} \end{cases} \qquad (2)$$

The z transform of input output relation is given by

$$Xu(z) = \sum_{n=-\infty}^{\infty} x_u[n] z^{-n} \qquad (3)$$

$$= \sum_{\substack{n=-\infty \\ n=\text{even}}}^{\infty} x[n/2]\, z^{-n} \qquad (4)$$

$$= \sum_{m=-\infty}^{\infty} x[m] z^{-2m} = X(z^2) \qquad (5)$$

In a similar manner, we can show that for a factor of L up-sampler

$$Xu(z) = X(zL) \qquad (6)$$

On the unit circle, for $z = e^{j\omega}$, the input-output relation is given by

$$Xu(e^{j\omega}) = X(e^{j\omega L}) \qquad (7)$$

A factor-of-2 sampling rate expansion leads to a compression of $X(e^{j\omega})$ by a factor of 2 and a 2-fold repetition in the baseband [0, 2π]. This process is called imaging as we get an additional "image" of the input spectrum. Similarly in the case of a factor-of-L sampling rate expansion, there will be L-1 additional images of the input spectrum in the baseband. Interpolator is used as low pass filter to remove the xu[n] images and in effect "fills in" the zero-valued samples in xu[n] with interpolated sample values.

IV. MATLAB BASED INTERPOLATOR

In this paper we have designed and compared three structures of interpolator Direct –Form FIR Polyphase, Nyquist Filter and Half-Band Low pass Filter. These three structures are designed and simulated using Mat lab and their performance in terms of area is compared.

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Firstly the Direct –Form FIR Polyphase Structure is designed and simulated with filter order 66. In the context of multirate signal processing, interpolation usually refers to band-limited interpolation. Ideal band-limited interpolation will take a digital (sampled) signal and produced an interpolated signal that will be identical to the signal that would be obtained by sampling the underlying continuous-time signal at a higher rate. Ideal band-limited interpolation can be accomplished by means of up sampling and using an ideal low pass filter. A time-domain interpretation of the ideal interpolator, naturally leads to polyphase implementations. A polyphase interpolation structure implements the filter. The resulting discrete-time signal has a sampling rate L times the original sampling rate. The term 'phase' refers to a time-delayed replica of the signal. A 'polyphase' filter simply means that several 'phases' are computed in parallel. The original implementation required hardware implementation of a filter with N taps. When the modified implementation, the hardware implementation of the filter requires only N/L taps. The original implementation required the input samples to the filter to be clocked at the higher sampling rate of Lfs. With the modified implementation, the input samples to the filter is clocked at the original sampling rate of fs, we need to have additional circuitry to dynamically load the coefficients and latch the output at the up sampled frequency Lfs. These filter sets are called polyphase filters. The Direct Form Polyphase structure is shown in figure 2.

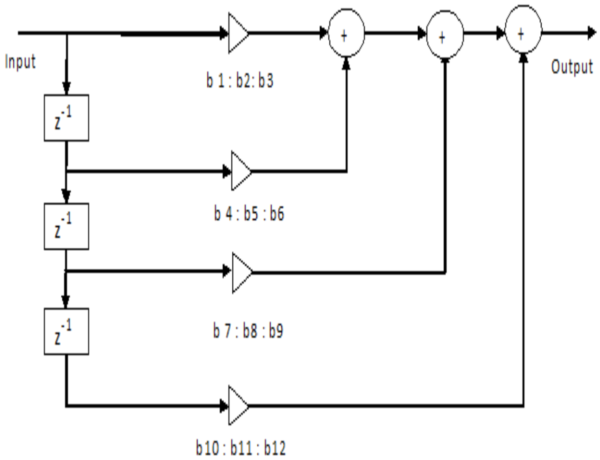

Figure.2 Direct –Form FIR Polyphase Structure

The Interpolation factor for Direct –Form FIR Polyphase Structure specifies the amount to increase the sampling rate of the input signal. It must be an integer. Here the interpolation factor is 2, with this factor, the resulting signal perfectly matches the original, but with twice as many samples, one between each original sample, as shown in the following figure3. Direct –Form FIR Polyphase Structure is further analyzed for its various output responses shown by figure 4.

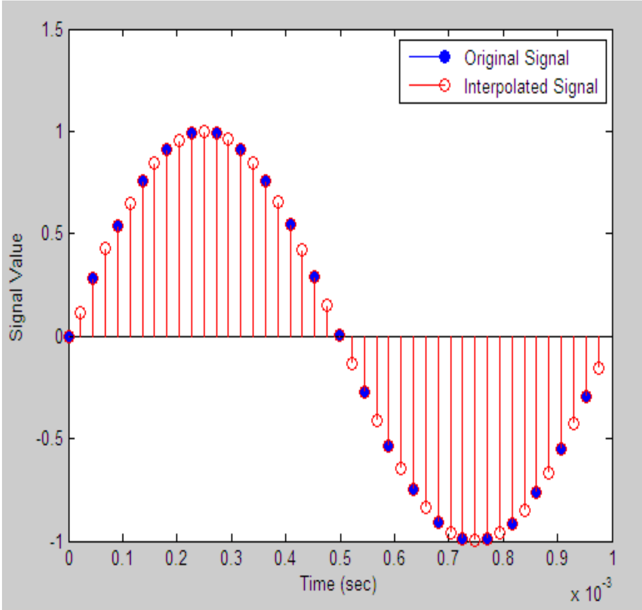

Figure. 3. Interpolator Response

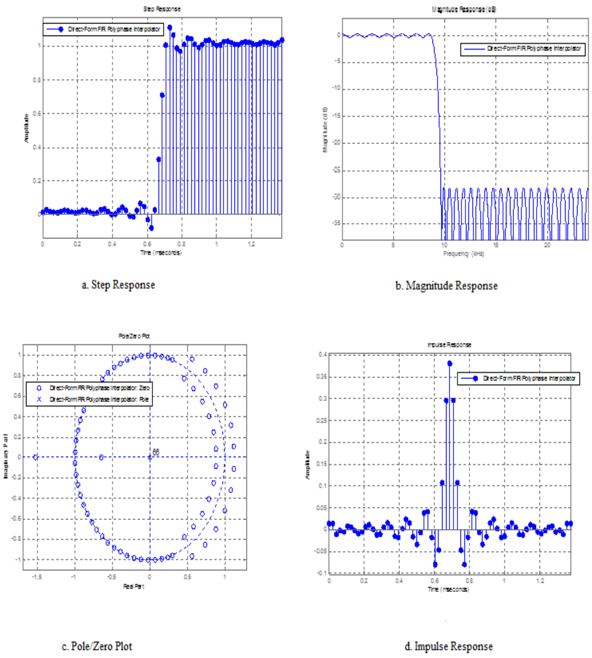

Figure.4 Output responses of Direct –Form FIR Polyphase Structure

Secondly Nyquist filter structure is designed and simulated using and Mat Lab with order 66 . Nyquist filter is one for which one of its polyphase components is a pure delay and thus leaves the input signal unchanged (except for a possible delay). When designing an interpolation filter, it is desirable for it to be a Nyquist filter since this will ensure that even a non ideal filter will allow the input samples to pass through unchanged. Nyquist filters are also called Lth-band filters because the passband of their magnitude response occupies roughly 1/L of the Nyquist interval. Nyquist Filter Structure is further analyzed for various responses such as Step, Magnitude, Pole/Zero and Impulse response shown by figure 5.

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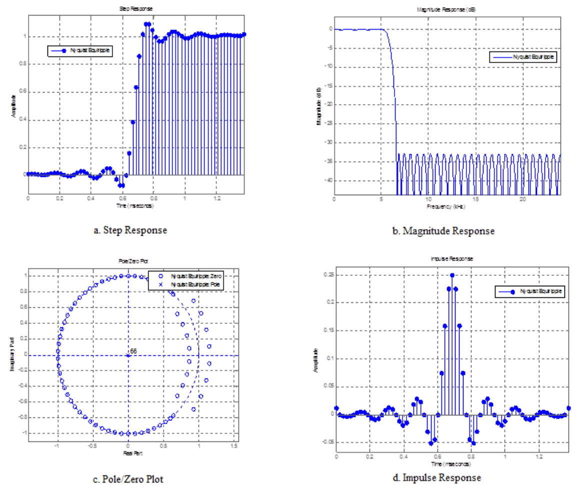

Figure.5 Output responses of Nyquist Filter Structure

Now t hird filer is Half-Band Low Pass filter which is designed and simulated using Mat Lab with same order 66. The Half-Band filter (QMF filters) are used in sub-band coding of digital video and audio signals. Due to their high efficiency, digital Half-Band filters are widely used as versatile building blocks in digital signal processing applications. A digital Half-Band filter (HBF) is, in its basic form with real-valued coefficients. Half-Band filters are commonly used when interpolating (or decimating) by a factor of 2. The cutoff frequency for a Half-Band filter is always 0.5π.Half-Band filter is also a low pass FIR filter. The bandwidth of the filter is half of the Nyquist rate or quarter of the sampling rate and it makes half of the filter coefficients zero. The name 'Half-Band' comes from the fact it is using only half of the available bandwidth. In Half-Band filters about 50% of the coefficients are zero. This reduces the computational complexity of the proposed interpolator significantly. Half-Band Filter Structure is further analyzed for various responses such as Step, Magnitude, Pole/Zero and Impulse response shown by figure 6.

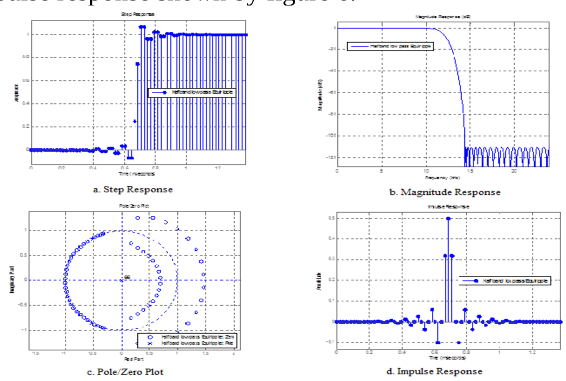

Figure.6 Output responses of Half-Band Low Pass Filter Structure

V. HARDWARE REQUIREMENT

In this paper three structures namely Direct Form FIR Polyphase, Nyquist Filter and Half-Band Low pass Filter is designed and simulated with order 66 in Mat Lab. The implementation requirements are shown in table 1. The outcomes shows that Half-Band Low Pass Filter structure is more area efficient than the Direct Form FIR Polyphase, Nyquist Filter structures. As it is using only 35 multipliers as compared to Direct Form FIR Polyphase and Nyquist Filter. Half-Band Filter structure is consuming only 52% of multipliers as compared to polyphase structures. So it shows that Half-Band Filter structures provide area effective as well as cost effective solution for wireless applications.

Table 1. Area Comparison for Direct Form FIR Polyphase, Nyquist Filter andHalf-Band Low Pass Filter structure

Area Utilization	Direct Form FIR Polyphase	Nyquist Filter	Half-Band Low Pass Filter
No.of Multipliers	67	51	35
No. of Adders	65	50	34
Multi Per input sample	67	51	35
Adder Per input sample	65	50	34

Further Half-Band Filters are implemented in two forms Transposed and Symmetric with filter 66. The direct-form structure has the disadvantage that each adder has to wait for the previous adder to finish before it can compute its results. For high-speed hardware such as FPGAs/ASICs, this introduces latency which limits how fast the filter can be clocked. A solution to this is to use transposed direct-form structure. With this structure, the delays between the adders can be used for pipelining purposes and therefore all additions and multiplications can be performed fully parallel. This allows real-time handling of data with very high sampling frequencies. Transposed structure is given in figure7.

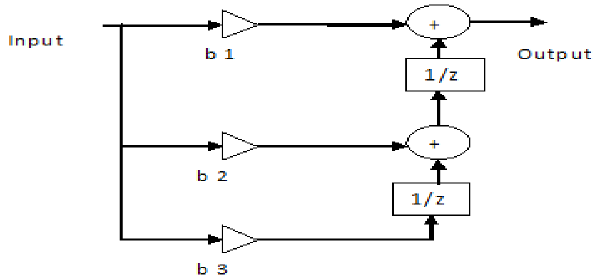

Figure.7 Direct Form Transposed Structure

In Symmetric structures filter coefficients are symmetric and phase is linear. It has the advantage to reduce number of multipliers. Quantization errors do not affect the Linear Phase characteristic of filters in this structure.

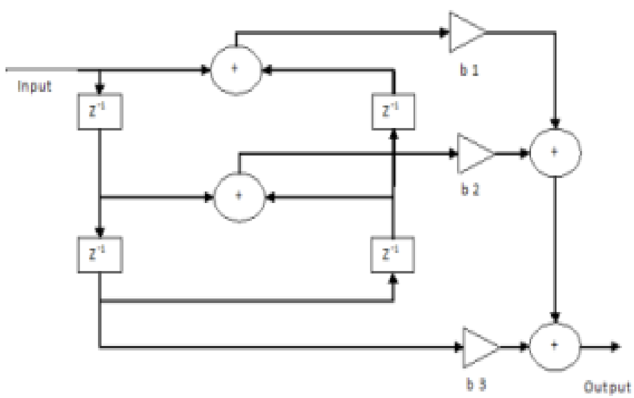

Figure 8 Direct Form Symmetric Structure

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In this structure the area optimization is much better than the others which are discussed in this Paper. It uses less no. of adders and multipliers so the required area for this structure is also very low. Direct Form Symmetric Structure is given in figure8.

Half-Band filter is implemented for two structures, transposed structure and symmetric structure. The area utilization has been shown in table 2. The outcomes show that Transposed structure is using 35 multipliers and Symmetric structure is using 18 multipliers. Symmetric filter structure is using 27% of reduction in multipliers as compared to polyphase structures and 35% of reduction in multipliers as compared to Nyquist filter structure. So Half-Band symmetric architectures are cost efficient for design an interpolator.

Table 2. Area Comparison for Half-Band Transposed Structure

Area Utilization	Transposed structure	Symmetric Structure
No. of Multipliers	35	18
No. of Adders	34	35
Mult Per input sample	35	18
Adder Per input sample	34	35

and Half-Band Symmetric Structure

VI. CONCLUSION AND FUTURE WORK

In this paper an area efficient and cost efficient interpolator has been designed and simulated with the help of Mat Lab. The interpolator is designed and simulated by three structures namely Direct Form FIR Polyphase, Nyquist Filter and Half-Band Low pass Filter. The designed Half-Band Low Pass filter structure has shown same performance as compared to Direct Form FIR Polyphase and Nyquist Filter structures. In Half-Band structures 35 multipliers have been saved for hardware implementation as compared to Direct Form FIR Polyphase and Nyquist Filter structure. The Half-Band structure was further implemented using Transposed and Symmetric structures. Half-Band symmetric structures consuming 18 multipliers as compared to other structures to provide cost effective solution for signal processing in wireless communications. In future the structures described in this paper can be implemented on hardware.

REFERENCES

1. Rajesh Mehra and Ravinder Kaur, "Reconfigureurable Area and Speed Efficient Interpolator Using DALUT Algorithm", Springer-Verlag Berlin Heidelberg 2010 , Conference in Computer Science and Information Technology 2011, CCIS 132, Page(s):117-125 .
2. Rajesh Mehra, "FPGA Design of optimised CIC Interpolator for DSP based Wireless Communication System". IMS MANTHAN Journal , Vol. 5 , December 2010, Page(s):43-46.
3. Mukhtar, A.; Jamal, H.; Farooq, U,"An Area Efficient Interpolation Filter for Digital Audio Applications". IEEE Transactions on Consumer Electronics, Vol. 55, No. 2, MAY 2009, Page(s): 768 – 772.
4. Xiaowei HUANG, Yan HAN, Lei CHEN, "The Design and FPGA Verification of a General Structure, Area-optimised Interpolation Filter Used in delta Sigma DAC", IEEE Conference on Solid State and Integrated circuit Technology (ICSICT) , October 2006, Page(s):2111-2113 .
5. Jing Li; Xiao-Bo Wu; Jin-Chen Zhao," An improved area-efficient design method for interpolation filter of Sigma-Delta audio DAC ", IEEE Conference on Solid State and Integrated Circuit Technology , Page(s):327-329, November 2010
6. M.D. Lutovac, and L.D. Milic, "Approximate linear phase Multiplier less IIR half-band filter," IEEE Signal Processing Letters, Vol.7, Page(s): 52-53, March 2000.
7. Navjot Singh, Amandeep singh, "Design and implementation of optimum interpolator filter using Farrow structure", International Journal of Engineering Science and Technology(IJEST), Vol. 3, No.5, May 2011, Page(s):4108:4113
8. Zhao De-An; Lian Xian-Guo; Yang Ping," Multi-rate Signal Processing For Software Defined Radio(SDR) and realize on FPGA", IEEE conference on Computer Science Technology and Applications , Vol. 1,Decmber 2009, Page(s):251-254.
9. Allred, D.J, Heejong Yoo, Krishnan, V, Huang, W, Anderson, D.V, "A novel high performance distributed arithmetic adaptive filter implementation on an FPGA", IEEE International Conference on Acoustics speech and signals , Vol. 5, August 2004, Page(s): V - 161-4.
10. Al-Haj, A , "An Efficient Configureurable Hardware Implementation of Fundamental Multirate Filter Banks". IEEE Conference on Systems, Signals and Devices, Page(s): 1–5, July 2008.

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