ISSN 2278 – 0211 (Online)

Design of Programmable Digital Filter: Band Pass Using Second Order All Pass Transformation and Coefficient Decimation

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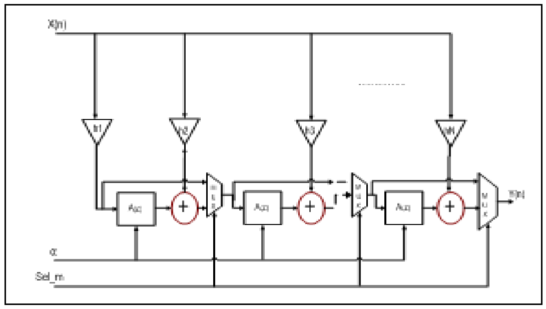
Abstract
In this paper, the design of a computationally efficient programmable digital filter (PDF) is presented. The centre frequency and bandwidth of this filter can be for the desired one without updating the filter coefficients. The warped infinite impulse response (W-IIR) filters, obtained by replacing each unit delay of an IIR filter with an all pass filter, are widely used for various audio processing applications. However, W-IIR filters fails to provide programmable bandwidth band pass responses for a given center frequency using first order all pass transformation. To overcome this drawback, the design is accomplished by combining W-IIR filter with the coefficient decimation (CDM) technique.

Keywords*— second order allpass transformation, IIR filter, Variable digital filter, Coefficient decimation.*

1. Introduction
Warped filters require first-order all-pass transformation to obtain programmable low-pass or high-pass response, and second-order all-pass transformation to obtain programmable band pass or bands top response. To overcome this drawback, the proposed method combines the warped filter with coefficient decimation technique, from where it is to vary both cutoff frequency and bandwidth to obtain programmable band pass filter.
The approach is where combining the warped filter and coefficient decimation technique with less number of gates and consumes minimum power to design a programmable digital filters, considering the design of band pass with the proposed approach, the reason for selecting band pass filter is, as it requires both cut-off frequency and band width to design a programmable band pass filter.
In many signal processing applications a clear need to change the parameters of the filters used exists. Such applications are found in telecommunications, digital audio equipment, medical electronics, radar, sonar and control systems, adaptive and tracking systems, spectrum and vibration analyses, formant speech synthesizers and in numerous laboratory instruments. The most general term for filters with changeable parameters is “variable filters”, but they are often called “tunable” (although this term is correct only when frequency-related parameters are the subject of change), “adjustable” (this term is correct only for changes in some narrow range of values of a given parameter) or “programmable” (when parameters can be reprogrammed or are controlled by a computer). In analog variable filters, tuning is often achieved by trimming some passive element and thus the term “trimmed” is also used to refer to the filter.
IIR band-pass transfer function for detection of a single band-pass signal from a broadband signal. Such a second-order band-pass/band-stop function is very effective if a band-pass signal to be enhanced or suppressed has a line spectrum, i.e. infinitely narrow-band spectrum. However, if the spectrum of a band-pass signal has some finite bandwidth, the use of a second-order function is not a good choice.

2. Existing System
In case of programmable band pass, the two programmable parameter, centre frequency and bandwidth, depend on a single controlling parameter hence, the warped filter fails to provide programmable bandwidth responses for a given center frequency using first-order all pass transformation. Also- whenever the type of responses needs to be changed the filter coefficient need to be updated. Which incurs a large number of memory, read and write operations. Though the second-order all-pass transformation provides low-pass to band pass/band top transformation, warped filters require separate first- and second order all pass structures which made the overall filter highly complex and power inefficient.

3. Proposed System



Figurer 1: Proposed System Architecture

Consider *Nth* order band pass prototype filter, $H(z)$ with center frequency, *fc0* and coefficients *h0*, *h1*,.., *hN*. The filter coefficients are fixed and hence can be hard-wired. The proposed PDF architecture, obtained by combining W-IIR filter with CDM technique, is shown in Fig. 1. The CDM[1] is implemented using the multiplexers controlled by signal *sel_m*. In the proposed PDF, there are two controlling parameters, warping coefficient, α and decimation factor, M which controls center frequency and bandwidth. The value of warping coefficient α required to obtain the desired $fc\alpha$ is calculated. For the proposed PDF, $|\alpha|<1$ and M takes positive integer values. When M=1, the proposed PDF is same as W-IIR filter. The CDM operation has an inherent disadvantage of deterioration of stop band attenuation. Therefore, based on the desired stop band attenuation specifications, the prototype filter should be designed with larger stop band attenuation keeping account of the deterioration of the stop band response.

3.1. Warped Infinite Impulse Response

Digital filters where unit delays are replaced with frequency dependent delays, such **as** first order allpass sections, are often called warped filters since they implement filter specifications on a warped non-uniform frequency scale. Warped IIR (W-IIR) filters cannot be realized directly due to delay free loops. Specific solutions have been known that make W-IIR filters realizable but no general approach has been available so far. In this paper we will explore the generation of such filters, including new filter structures. The robustness and computational efficiency of W-IIR filters are studied and most potential applications are discussed. Let the transfer function $H(z)$ of a discrete-time filter be given by:

$$\boxed{H(z) = \frac{B(z)}{A(z)} = \frac{1}{1 - az^{-1}}}$$

Governed by the parameter a real number with $0 < |\alpha| < 1$, $H(z)$ is stable and causal with a pole at a. The time-domain impulse

response can be shown to be given by:

$$\boxed{h(n) = a^n u(n)}$$

Where $u(n)$ is the unit step function. It can be seen that $h(n)$ is non-zero for all $n \geq 0$, thus an impulse response continues infinitely. WFIR filters typically take 3 to 4 instruction cycles per tap compared to 1 for ordinary FIRS. For W-IIR filters 4 to 5 instructions are needed per order compared to 2 for direct form I1 structures. Thus we may conclude than in best cases W-IIR filters may be 2 to 5 times faster than traditional IIR filters for purposes where the warping principle works ideally.

3.2. Second Order All Pass Structure

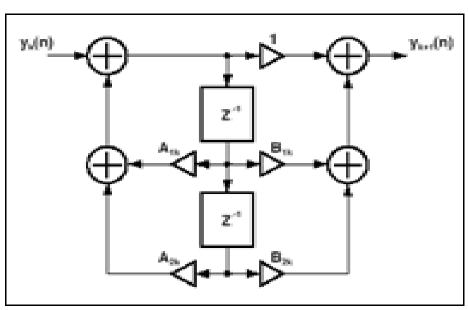


Figure 2: Second Order All Pass Structure

Fig.2 shows second order all pass structure. A **Passive Band Pass Filter** is classed as a second-order type filter because it has two reactive components within its design, the capacitors. It is made up from two single RC filter circuits that are each first-order filters themselves.

4. Implementation

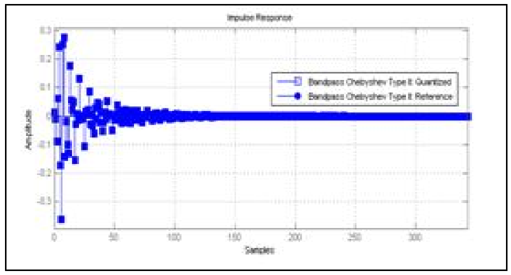


Figure 3: Impulse Response of Bandpass Chebyshev Filter

The performance of the proposed VDF is discussed in this section with the help of a suitable design example. Let the pass band and stop band ripple specifications are 1 dB and -80 dB respectively.The variable band pass responses obtained using the proposed VDF are shown in Fig. 6. By changing α and M, center frequency and bandwidth can be varied. Thus, proposed VDF provides variable bandwidth band pass responses for arbitrary center frequency. Similarly, variable band stop responses can be obtained using band stop prototype filter. Fig.4 displays magnitude response of bandpass Chebyshev filter.

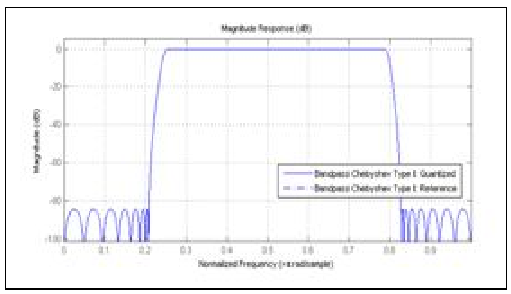


Figure 4: Magnitude Response of Bandpass Chebyshev Filter

5. Implementation Results

In this section, the implementation results is been discussed. Fig.4 illustrates the high level RTL schematic of the designed band pass filter.

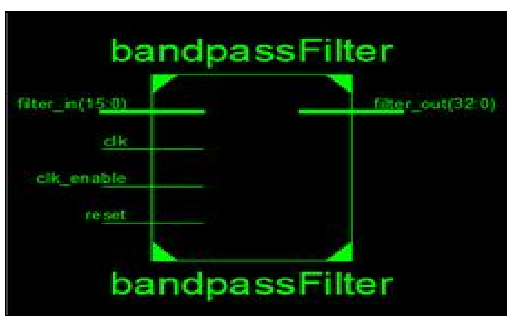


Figure 5: High Level RTL Schematic

The complexity of the system is expressed in terms of total number of gate count. A 16x16 bit multiplier, a 4:1 multiplexer, a word of memory and 32 bit adder were synthesized.Fig.5 displays the implemented design graph The complexity of the system is expressed in terms of total number of gate count. A 16x16 bit multiplier, a 4:1 multiplexer, a word of memory and 32 bit adder were synthesized.Fig.5 displays the implemented design graph which tells about how much percentage Slice flip flop, 4 input LUT, Slice, IO, MUX and multipliers have been utilised.

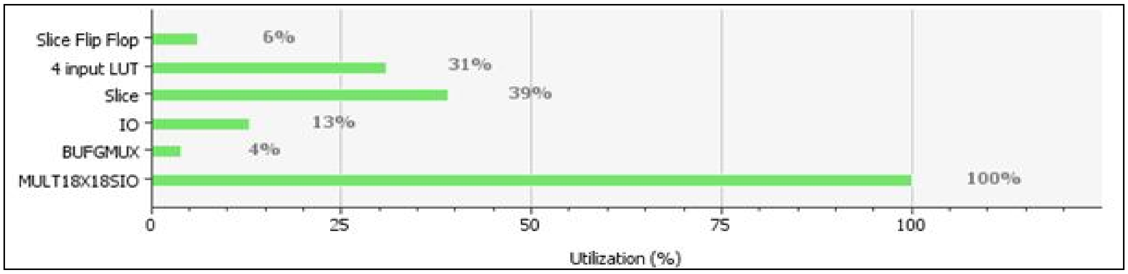


Figure 6: Implemented Estimation Graph

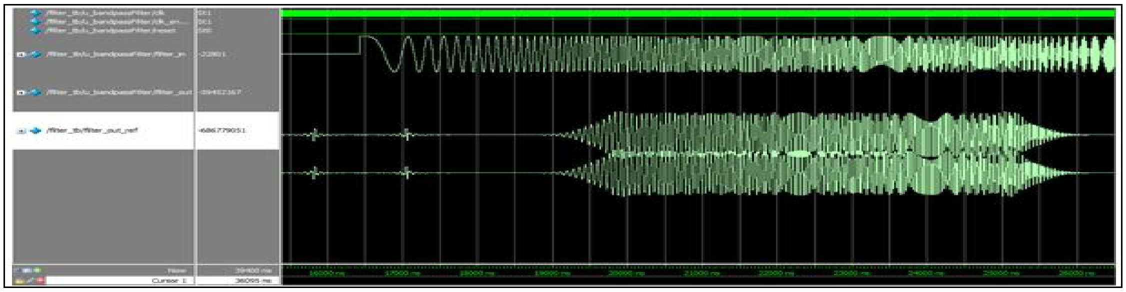


Figure 7: Waveform of the Bandpass Filter

Fig.6 demonstrates the waveform of the bandpass chebyshev type II filter obtained in ModelSim. The area in terms of multipliers, adders, multiplexers and registers is as shown in figure. Table 1.Shows the proposed VDF, the prototype filter needs to be overdesigned as discussed before.

VDFs / Number of	W-FIR filter [1]	W-FIR with 2nd order transform [3]	W-FIR with memory [4]	Previous VDF	Our Work
Multipliers	3125	1375	825	901	119
Multiplexers	0	0	0	600	1
Adders	1750	3300	1650	1800	120
Words of memory	0	0	1275	0	2001
Total gate count	**5706250**	**3080000**	**1820750**	**1957700**	**543500**

Table 1: Comparison of our work with the existing system

6. Conclusion

A computationally effective programmable band pass digital filter using warped infinite impulse response filter and coefficient decimation technique is presented. The centre frequency and bandwidth of this filter can be changed online without updating the filter coefficients. The programmable digital filter presented here is simple to design.

7. References

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