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Survey on the Filters Used in the Design of Reconfigurable Digital down Convertor for Wideband Applications

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Abstract— Nowadays the fastest growing field is the optimization. In some situations such as in real time environment the priori information about the statistics of the signal is not known completely, so in such situations filters are preferred. This paper deals with the survey of various filters used in Reconfigurable Digital down Convertor for wideband applications. The evaluation of RAM capacity, logic resources and number of non-constant coefficient multipliers are taken into account and the filter is designed with efficient combinations for wideband applications. The performance analysis is done in terms of the RAM capacity and logic resources used.

Keywords— Digital Down Convertor (DDC), resampling filter, Cascaded-Integrated Comb filter.

I INTRODUCTION

Reconfigurable digital down convertor is a vital component in wideband digital receivers, and the right reception and resampling of radio signal is difficult. Mixed signal circuit can be implemented in Very Large Scale Integration (VLSI) with the digital filter as the building blocks. The enhancement should be done for both speed and area in the design of digital down convertor. As compared with digital filters, analog filters are less expensive and quicker. Whereas in amplitude and frequency the dynamic range is found to be very high. The performance level is found to be very high in digital filters. In digital filters the quality is found to be better as compared with the digital filters [1]. If this reconfigurable FIR filter is created effectively, at that point it gives preference of rapid, low chip region and low power utilization over its conventional counterpart.

The three important subcomponents of digital down convertors are direct digital synthesizer, low pass filter, down sampler. At the intermediate frequency (IF), the DDS produces a complex sinusoid. An image is produced at the sum and difference frequency if the input signal is product of the intermediate frequency. The low pass filter provides the complex baseband representation of the input signal by the rejection of the sum frequency image [5]. It is then down sampled so that it is comfortable to many DSP algorithms.

For the most part, digital filters are used to alter or change the feature of the signal in either time domain or frequency domain. Linear Time

Invariant (LTI) filter is the familiar digital filter. It performs scientific work on a tested or discrete time flag to grow less or improve certain parts of that signal [5]. A LTI colleague with its information movement through a procedure called direct convolution. LTI advanced channels are for the most part classified as being limited motivation reaction (i.e., FIR), or very large drive reaction (i.e., IIR). FIR channel comprise of a limited number of test esteems, decreasing the above convolution whole to a limited total for every yield test brief time period. An IIR channel, nevertheless, wants that a boundless whole be performed.

Digital Down Convertor can be implemented using or application specific integrated circuits. All the operations will run at the sampling rate of the input rate in the DDS, multipliers and input stages of the low pass filters if the software implications are possible. From the output which is obtained from the analog to digital convertor the sampling of the data is done of several megahertz. Distributed arithmetic is one method to implement convolution with multiplier less unit, where the MAC operations are changed by a series of LUT access and summations [18]. Transferred Arithmetic is a substitute method for executing computerized channels. Basically all look up table is a bundle of individual piece memory cells putting away single piece esteems in each one of its cells.

The organization of this paper is given below: This Section II deals with literature survey. In this section an introduction to Half-band filter, Cascaded-integrated Comb (CIC) filter, Polyphase filter, Farrow-based variable fractional delay filter (VFD) and lookup table (LUT) based variable digital filter (VDF) is presented. A conclusion is given in section III.

II LITERATURE SURVEY

A. Half-Band Filter

In multirate signal processing applications, half band filters are widely used. It is used by interpolating or decimating by a factor of two [2]. The ripples should be same for both the pass band and stop band, the frequencies should be at equal distance from the half band frequency for both the pass band edge and stop band edge are its important characteristics. It is widely used in multirate applications because of its efficiency. Half band filter can be considered for realizing the interpolator. It provides higher computation speed at the cost of the phase nonlinearity [3].

B. Cascaded-integrated Comb Filter

By cascading the digital accumulator i.e. the cascading of the digital differentiators in equal number the CIC filter is formed. With the help of digital switch the sampling frequency of the comb signals is reduced and generally it is placed in between the various combs and the integrator. By using this kind of filter, the architecture is found to be more powerful [4]. For narrow band filter it is possible to obtain the down computational complexity in accordance with the single stage low pass FIR for decimation process. By the above process the filter which is operating at the clock rate is reduced so that it is used for very low power and high speed applications. This filter is found to be more popular since it uses the registers and adders and it requires no multiplication operation and sometimes subtractors is also used.

The filter response for this integrator comb filter is obtained and the cost is also reduced [5]. Each filter transfer function has a pole and it corresponds to the integrator whereas the comb is associated with the value of zero. Applications of this filter is that it is mainly used in processing of signals due to several advantages such as its simplicity and the reduction in noise that is its optimal behavior. The step response is found to be retained. This is found to be very worse to separate the different range of frequencies when compared with each other in frequency domain encoded signals.

To use these kind of filters for DSP operations the sampling frequency is found to be very large by using the interpolation at its ending stage and at the initiating stage the process of decimation is used because of its low cost. Even though this filter has advantage it is found to be effective if it is combined with other filters. Due to this reason expensive filters are utilized effectively. The up sampling filter is used in this filter. This architecture can be implemented for various stages of interpolation and decimation. But it is found to be effective for multiple stage filter because their responses are wide as compared with other filter. By using this filters their responses are further increased and the rate is also reduced.

C. Polyphase Filter

In polyphase filter two main operations are performed. The important operations are mixing and then followed by filtering. These filters performs the operation of mixing and filtering with a very low data rate at any one of its end. This method efficiently solves the bottleneck problem by using down conversion method [13]. The structure of polyphase filter is given

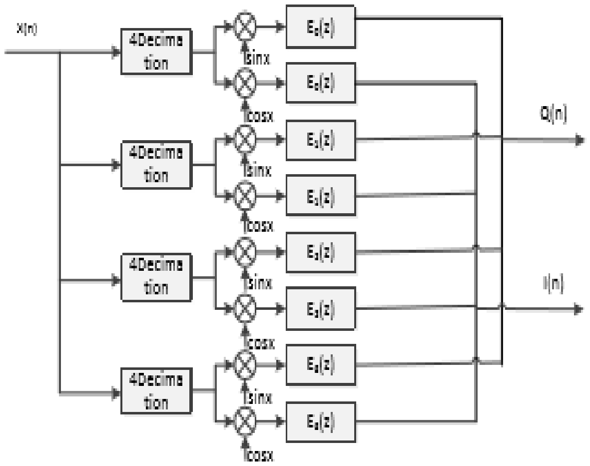

Fig. 1 Polyphase Filter Structure

Purpose of the decimation filter is to decimate the useful signal from the high speed data stream, and to reduce its sampling rate, so that subsequent DSP processes into line real time. After completion of the useful signal spectrum shifting, decimation filter after filtering can be performed data extraction. Decimation filter is to prevent aliasing after extraction.

D. Farrow based Variable fractional Delay Filter

The variable fractional delay (VFD) filter is obtained by considering the fractional delay. Fractional delay is required for many applications like the removal of echo and in many rate conversion methods at its sampling instants as compared with unit delay. With the help of this filter the original sampled signal and its associated delay can be reconstructed [9]. Once again the sampled signal can be resampled. The important point is that for the given order the sub filters are found to be fixed. The requirement of the adders and multipliers is found to be increased by its square rate if its order is found to be very high. The requirement of the number of the operators is reduced by using the modified farrow structure [6].

The usage of the number of the operators can be reduced with the usage of this modified farrow structure. In order to analyze this kind of problem Taylor structure is used and it is the efficient structure rather than other structure. The number of operators is reduced by using this Taylor structure [7]. As discussed before the requirement of the adder, multiplier is less in accordance with the above discussed filters.

E. Lookup table (LUT) based variable digital filter

Generally during the sampling process the tuning should be very fine at various instants so that the operation can be done in a particular limits. LUT based filter is used for this tuning purpose. By using various techniques these filters are designed efficiently and flexibly [18]. It is often specified in frequency domain and it utilizes various algorithms in order to provide a better toolbox and to improve its usage. The main concern should be on the transient elimination when used in time domain FD filters and its implementation also plays a vital role.

III CONCLUSION

Among all the filters which is discussed above, the CIC filter is found to be better since it can operate at very high speed. The important advantage of using this CIC filter is that the process of multiplication is not involved in this kind of filter because of the decimation factor which is available. The side lobe level for the CIC single stage filter is found to be very high as compared with the CIC multistage filter since the attenuation is found to be bad. Therefore the cascaded form of the CIC filter is utilized. In CIC filter the number of stages is less since it is based on several reasons such as flatness of the band.

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