Design of CIC Filters for Software Radio System

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ABSTRACT

Cascaded integrator comb (CIC) filters are used to realize computationally efficient decimation, interpolation or sample rate conversion. However, since CIC filters have a limited number of tuning parameters and exhibit a passband droop, their use in Software Defined Radio (SDR) systems may be limited. In this paper, we propose an alternative structure to CIC filters that, at the expenses of a higher computational effort, is suitable for SDR systems.

1. INTRODUCTION

A SDR system is a general platform of reconfigurable and reprogrammable hardware/software capable of supporting inter-communication among different communication systems. Essentially, a SDR receiver digitalizes the signals with a high-speed analog-to-digital converter (ADC) and processes them using, for example, a digital signal processor (DSP) [1, 2].

Usually, in SDR systems is necessary to accommodate different communication standards with different clock rates which can vary from many tens of MHz to around 100 kHz. This rate conversion requirement leads to large order and high- rate digital filters [2].

A filter adapted for rate conversion, decimation or interpolation, with an efficient structure and indicated to be implemented in integrated circuit is the cascaded integrator-comb (CIC) filter [3, 4]. The conventional structure of a decimator CIC filter consists of an integrator section operating at high sampling rate and a section of comb filter operating at a low rate, separated by a downsampling. Fig. 1 presents the structures of the CIC decimator and interpolator filters.

2. ANALYSIS OF THE CIC FILTER

The basic structure of a CIC filter is based on a class of FIR filter for processing multirate proposed by Hogenauer [4]. Later on, several modifications on this original structure were proposed [3, 5-7, 9].

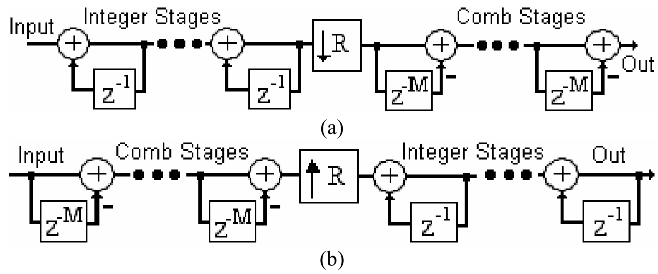

Fig. 1. Convencional CIC filter (a) decimator (b) interpolator

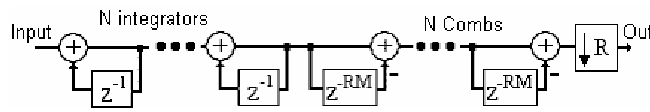

Fig. 2. Equivalent CIC structure for analysis purposes

For analysis purposes, the Noble Identity can be used to combine the integrator and comb sections in Fig. 1(a), resulting in a single transfer function, as shown in Fig. 2 [8].

In general, the factor M is made equal to either 1 or 2. The global transfer function is

$$H(z) = \left(\frac{1 - z^{-RM}}{1 - z^{-1}}\right)^N = \left(\sum_{k=0}^{RM-1} z^{-k}\right)^N , \qquad (1)$$

and the frequency response is

$$\left|H(\omega)\right| = \left[\frac{\operatorname{sen}\left(\dfrac{\omega RM}{2}\right)}{\operatorname{sen}\left(\dfrac{\omega}{2}\right)}\right]^N . \qquad (2)$$

Fig. 3 shows the frequency response of $|H(\omega)|$ for N=4, R=6 and M=1, with f_c, = 1/8. We observe that the nulls happen at multiples of $F_s/(RM)$, or, using normalized frequency, $k/(RM)$; where k is an integer value with k = 1, 2,..., $\lfloor RM/2 \rfloor$, and $\lfloor x \rfloor$ indicates the integer part of x.

Usually, the performance indices of a decimation filter consider the attenuation in the passband, the error due to spectrum aliasing and the minimum attenuation applied in the first band (between 1-f_c and 1+f_c) that suffers aliasing into the desired baseband signal.

A conventional CIC filter presents the following advantages that turn it ideal for decimation applications and interpolation in SDR systems [4, 8]:

- no multipliers are required;
- no storage is required for filters coefficients;
- the structure of CIC filters is very regular, consisting of only two simple types of building blocks;
- not much or no external control is necessary.;
- as it is a filter whose impulse response is finite (FIR filter), it is always stable.

The main inconveniences of this basic structure are:

- the register can become very large for high values of R;
- the frequency response is only determined by three factors (R, M, N), which limits the range of configuration of the filter;
- the existence of a significative attenuation in the passband that depends, mainly, on the order N and on decimation factor R [3];
- a wide transition band.

3. AN ALTERNATIVE PROPOSAL

We observe that there are many known proposals of modifying the basic structure of the CIC filter [3, 5-7, 9]. Some of them have as main goal the power saving and others better frequency response. Some of them modify the whole basic structure while others maintain the structure almost intact.

In this paper the original CIC filter [4] is modified through the inclusion of the following structures: an auxiliary IIR filter, additional delays in one of the comb filters and the structure shown in Fig. 4 [9].

With these modifications we can get a certain reduction of power consumption, through a smaller order for the final filter, besides a better frequency response.

The main modification is accomplished with the use of the structure shown in Fig. 4, whose transfer function is given by:

$$H(z)=\left(\frac{1-b.z^{-R}+z^{-2R}}{1-a.z^{-1}+z^{-2}}\right). \qquad (4)$$

With this change, zeros can be moved for $k/R \pm \alpha$, where k is a positive integer number related to R, as indicated in Section 2, and α is smaller than or has the same value of f_c. In the next section we will explain the choice of the α value and the calculation of "a" and "b".

Thus, the signals located between k-kf_c and k+kf_c (aliased bands) suffer a larger attenuation.

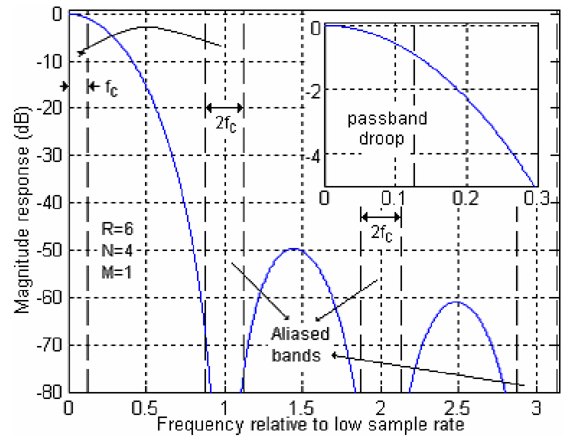

Fig. 3. Frequency response of $|H(\omega)|$ for N=4, R = 6, M = 1.

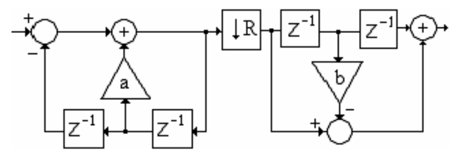

Fig. 4. Structure used to modify the nulls location in the CIC filter [9].

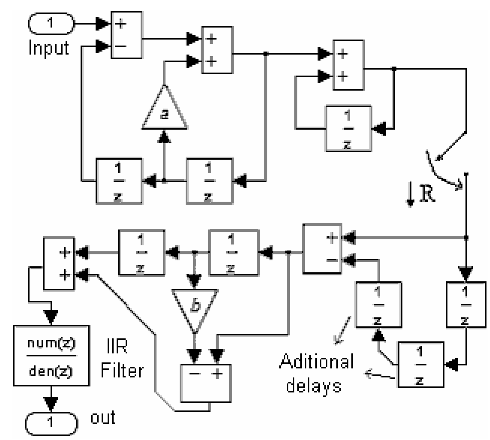

Fig. 5. Modified CIC filter, order N = 3.

The inclusion of the additional delays in the comb section increases the number of zeros in the transfer function reducing the transition band and energy of the aliasing bands.

The additional low-pass IIR filter of second order is to compensate the excessive attenuation in the passband and to reduce even more interferences in the rejection band. A modified CIC filter of order N = 3 with two additional delays is shown in Fig. 5.

4. DESIGN EXAMPLE OF A MODIFIED CIC DECIMATOR FILTER AND SIMULATIONS

To illustrate the proposed modification and to compare it with the mentioned papers, we will make a sample project as indicated in [4] using the following data:

- original sampling rate: 6 MHz;
- desired rate: 240 kHz

- band of interest: 30 kHz;
- aliasing attenuation must be better than 60 dB;
- falloff in passband of less than 3 dB.

Using these data we observe that R = 25 and the bandwidth relative to low sampling rate is f_c = 1/8. The data of Table 01 indicate the maximum attenuation in the passband and the attenuation in 1 - f_c, for the original CIC filter and for its version with additional delays. These values were calculated using (5). The first column of Table 01 indicates the delays used in the comb section, for example, M = [113] indicates two comb filters with unitary delay and one with three delays.

$$\left|H(\omega)\right| = \text{sen}\left(\frac{\omega R M_1}{2}\right)\cdots \text{sen}\left(\frac{\omega R M_N}{2}\right) \Bigg/ \left(\text{sen}\left(\frac{\omega}{2}\right)\right)^N \quad (5)$$

Observing Table 01, we see that, using the basic structure, the simple increase of delays is not capable of finding significant attenuation in the rejection band (aliasing), and degrades the passband considerably.

In [4], the chosen CIC filter was of order 4 with M_i = 1. But, with the proposed modifications in this work, the order of the filter can be reduced for N = 3, what leads to a reduction of the system power consumption, even considering the increment of the auxiliary filter and of the extra delays. This power savings are obtained because the dynamic dissipation of power in the integrated circuit CMOS is proportional to the operation frequency.

The proposed modifications also increase the attenuation in the bands that are not in the aliasing bands.

The introduction of zeros near to 1±f_c, reducing the energy in the aliasing bands is desirable. To achieve this goal without worsen the passband behavior, it is necessary to modify the integrators section and one of the comb filters, as indicated in Fig. 4. The values of "a" and "b" can be calculated as [9]:

$$a = 2\cos(2\pi\alpha / R) \quad (6a)$$

$$b = 2\cos(2\pi\alpha) \quad (6b)$$

choosing α equal to 0.85f_c we obtain: a = 1.9993 and b = 1.5706. In this project it was chosen a comb filter with three delays samples: $H(z) = 1 - z^{-3}$. Thus, the transfer function of the modified CIC is (in relation to high sampling rate):

$$H(z) = \left(\frac{1 - 1.5706 z^{-25} + z^{-50}}{1 - 1.9993 z^{-1} + z^{-2}}\right)\left(\frac{1 - z^{-75}}{1 - z^{-1}}\right) \quad (7)$$

The project of the auxiliary filter does not present difficulty: we can locate the zeros at π rad, the poles at $\pm 3 f_c \pi$ rad and a gain of 2.5 dB in f_c. After necessary calculations, the transfer function of the auxiliary filter results in:

$$H_{aux}(z) = \left(\frac{1 + 2z^{-1} + z^{-2}}{1 - 0.5511 z^{-1} + 0.5184 z^{-2}}\right) \quad (8)$$

TABLE 01

Delays	Gain in f_c, (dB)	Gain in 1-f_c, (dB)
Order N = 3		
M = [111]	-0.6721	-51.3266
M = [112]	-1.3598	-52.0143
M = [113]	-2.5591	-53.2135
Order N = 4		
M = [1111]	-0.8962	-68.4354
M = [1112]	-1.5839	-69.1231
M = [1113]	-2.7831	-70.3223

This filter presents an attenuation of approximately 2.57 dB in the passband. Using a second order auxiliary IIR filter it is possible to compensate this effect. Moreover, we could add one more zero in the transition band. A simpler second-order FIR filter could also be used, with the exclusive purpose of improving passband behavior, as done in [7].

In Fig. 6 we have an application example of the proposed filter. In this simulation, the signals of interest occupy the frequency range from 0 to 1/(8R) and they are contaminated for many interferences and noise. In this example, some of the interferences signals have superior power in comparison to the signals of the interest band, as shown in Fig. 6a. In Fig. 6b we have the signal after the modified CIC filter, and in Fig. 6c, the final effect given by the auxiliary filter is shown.

The advantage of the proposed structure is more evident when the order of the filter is large. For example, if in our project an attenuation larger than 120 dB is required in the first aliasing band, the order of the filter with conventional structure would be N = 8, as calculated by (9) [6].

$$N \geq \frac{120}{20\log_{10}\left(\frac{\text{sen}(\pi f_c)\,\text{sen}(\pi(1 - f_c)/R)}{\text{sen}(\pi f_c / R)\,\text{sen}(\pi(1 - f_c))}\right)} = 7{,}11 \quad (9)$$

With the use of the proposed structure, the order is reduced to N = 6, which leads to a significant economy in the consumption of power by the system.

When $N \geq 3$, we can use the following steps to find the order of the proposed CIC filter:

$$N_O = \left\lceil \frac{dB\min}{20\log_{10}\left(\frac{\text{sen}(\pi f_c)\,\text{sen}(\pi(1 - f_c)/R)}{\text{sen}(\pi f_c / R)\,\text{sen}(\pi(1 - f_c))}\right)} \right\rceil - 1 \quad (10)$$

$$g0 = \frac{2-b}{2-a} \quad (11)$$

$$\omega_x = 1 - \left(\frac{N_O - 1}{N_O}\right)\alpha . f_c \quad (12)$$

$$g1 = \left|\frac{1 - b\exp(-j\omega_x R) + \exp(-j2\omega_x R)}{1 - a\exp(-j\omega_x) + \exp(-j2\omega_x)}\right| \quad (13)$$

$$g2 = \left|\frac{\mathrm{sen}(3\omega_x/2)}{\mathrm{sen}(\omega_x/2R)}\right| \quad (14)$$

$$N_m = \left\lceil \frac{dB\min - 20\log_{10}(g1/g0)}{20(\log_{10}(g2) - \log_{10}(MR))} + 2 \right\rceil \quad (15)$$

where dBmin is the minimum attenuation in the aliasing band and N_m the order of the modified CIC filter. In Fig. 7 we can observe the reduction of the modified filter order in relation to the original filter, considering $f_c = 1/8$ and a wide attenuation in the first folding band.

The main disadvantages of the proposed structure are the use of two multipliers that are not power of 2 and the addition of an auxiliary filter, in relation to the conventional CIC filter. This structure is indicated for filters with order $N \geq 3$, which does not represent a serious inconvenience.

5. CONCLUSIONS

In this paper, we proposed an alternative architecture for the CIC filters, which reduces power dissipation and presents a better filter response in terms of both passband attenuation and aliasing band. We described the modified filter through a design example. Such structure is applicable to mobile communication systems and SDR.

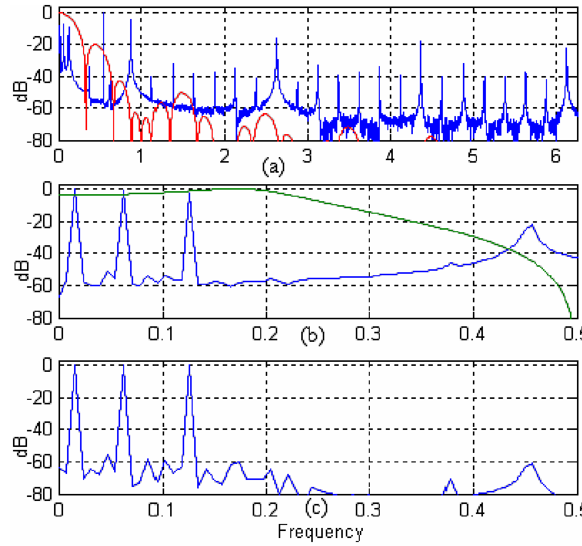

Fig. 6 (a) signal contaminated by noise and interferences and the frequency response of the modified CIC filter, (b) signal of interest after the modified CIC filter and magnitude response of the auxiliary filter, (c) signal output of the auxiliary filter.

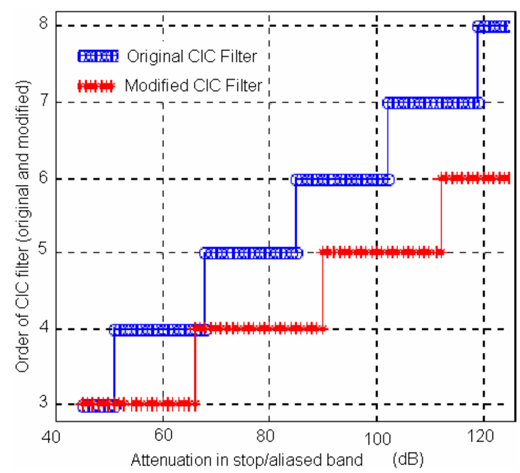

Fig 7. Reduction of the order of the CIC filter modified.

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