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CIC Filter:- Review & Analysis

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Abstract: *This review paper represents the performance and analysis of sharpened CIC filters design. In this paper we discuss the improving method of designing CIC filters. For discussing the improving method of CIC filters we consider the some standard paper which is based on designing of CIC filter.First we discuss about CIC filter. In CIC filter there is problem in passband droop and stopband aliasing. For improving the performance of CIC filter we can use many techniques, such as compensation filter cascaded with CIC filter, Sharpening technique proposed by Kaiser and Hamming, polyphase decimation FIR filter to achieve wide broadband compensation of the CIC filter.After discussing the techniques we give the review analysis on improving method of CIC filter design.*

I. 1.Introduction

As the data converters become faster and faster, the application of narrow band extraction from wideband sources, and narrow band construction of wideband signals is becoming more important [1]. These functions require two basic signal processing procedures: decimation and interpolation. Decimation is used to reduce the sampling rate by passing a signal through low pass or band pass filter. Interpolation is used to increase the sampling rate.

E.B. Hogenauer [2] proposed a filter with an efficient way of performing decimation and interpolation. Hogenauer devised a flexible, multiplier-free filter suitable for hardware implementation that can also handle arbitrary and large rate changes. These are known as cascaded integrator comb filters(CIC filters) because their structure consists of a integrator section operating at high sampling rate and a comb section operating at low sampling rate.

The two basic building of a CIC filter are an integrator and a comb. An integrator is simply a single-pole IIR filter with a unity feedback coefficient:

$$y[n] = y[n-1] + x[\mathrm{n}] \quad (1.1)$$

This system is also known as an accumulator. The integrator section of CIC filters consists of N digital integrator stages operating at high sampling rate, f_s. Each stage is implemented as a one pole filter with a unity feedback coefficient. The transfer function for a single integrator on the z-plane is

$$H_I(\mathrm{z}) = \frac{1}{1-z^{-1}} \quad (1.2)$$

The basic integrator figure as shown in below:

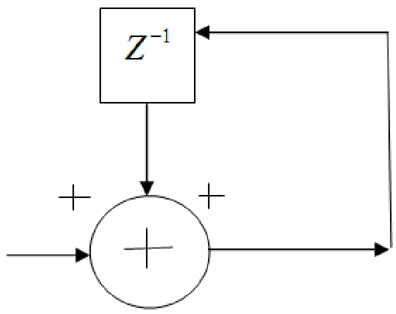

Figure 1.1:- Basic Integrator

A comb filter is described by:

$$y[n] = x[n] = x[n-RM] \quad (1.3)$$

Comb filter section operates at the low sampling rate, f_s / R, where R is the integer rate change factor. This section consists of N comb stages with a differential delay of M samples per stage. The differential delay is a filter design parameter used to control the filter's frequency response. In practice, the differential delay is usually held to M=1 or 2. The system function for a single comb stage referenced to the high sampling rate is

$H_C(z) = 1 - z^{-RM}$ (1.4)

The basic comb filter figure as shown in below:

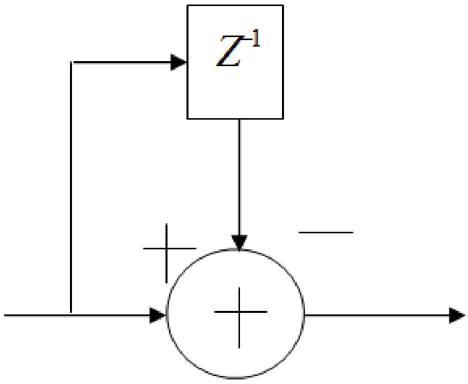

Figure 1.2:- Basic Comb Filter

Between the two filters sections is a rate change switch. CIC decimation would have N cascaded integrator stages clocked at f_s, followed by a rate change by a factor R, fol

lowed by N cascaded comb stages running at f_s / R as shown in figure 1.3.

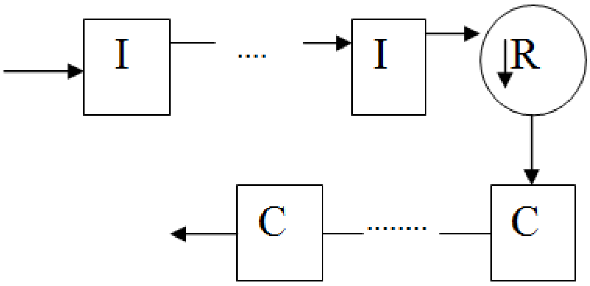

Figure 1.3:- CIC Decimation Filter

CIC interpolator would be N cascaded comb stages running at f_s / R, followed be a zero-suffer, followed by N cascaded integrator stages running at f_s as shown in figure 1.4.

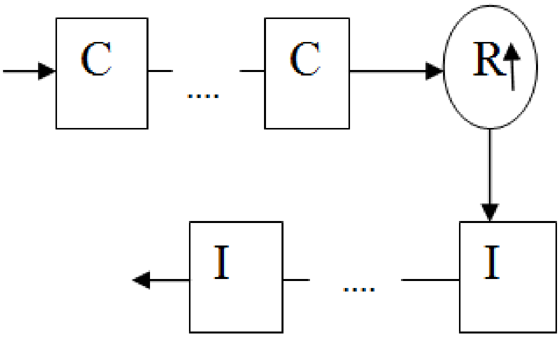

Figure 1.4:- CIC Interpolator Filter

The transfer function for a CIC filter at f_s is

$$H_{CIC}(z) = H_I^N(z) H_C^N = \frac{(1-z^{-RM})^N}{(1-z^{-1})^N} = \left[\sum_{k=0}^{RM-1} z^{-k} \right]^N \quad (1.5)$$

This equation shows that even though a CIC has integrators in it, which by themselves has an infinite impulse response, a CIC filter is equivalent to N FIR filters, each having a rectangular impulse response. Since all of the coefficients of these FIR filters are unity, and therefore symmetric, a CIC filter has a linear phase response and a constant group delay.

Filter sharpening can be used to improve the response of a CIC filter. This technique applies the same filter several times to an input to improve both passband and stopband characteristics. More we will be discuss in the literature survey.

There are some advantages of the CIC filers:-

- ✓ No multipliers are required
- ✓ No storage is required for filter coefficients
- ✓ Intermediate storage is reduced by integrating at high sampling rate and comb filtering at low sampling rate
- ✓ Little external control or complicated local timing is required
- ✓ The same filter design can be used for a wide range of rate change factor R with the addition of a scaling circuit and minimal change to the filter timing

Some problems encountered with the CIC filters include the following:

- ✓ Register widths can become large for large rate change factors
- ✓ The frequency response is fully determined by only three integer parameters (R, M and N) resulting in a limited range of filter characteristics.

The application for CIC filters seems to be in areas where high sampling rates make multipliers an uneconomical choice and areas where large rate change factors would require large amount of coefficient storage or fast impulse response generation.

II. Literature Survvey

Filter sharpening can be used to improve the response of a CIC filter. In most applications it is required to have a flat passband, otherwise the original signal may be destroyed. Unfortunately, the CIC filter alone suffers with a passband droop, which in many cases, cannot be accepted. The big droop is due the sinc-like characteristic of the filter. Hence, it is of a great interest to get a flat passband using a compensation filter. The compensation filter will take the form of the inverse of the CIC filter frequency response in the passband, and attenuate as much as possible in the stopband. This is the one way to improve the response of CIC filter.

Another way is proposed by Kaiser and Hamming [3]. If H(z) is a symmetric FIR filter , then a sharpened version, $H_s(\mathrm{z})$, can be expressed as $\mathrm{H}_s(\mathrm{z}) = \mathrm{H}^2(\mathrm{z})[3-2\,\mathrm{H}(\mathrm{z})]$ (2.1)

The magnitude version of sharpened CIC filter would then be

$$|\,\mathrm{H(f)}\,| = \left| 3\left(\frac{\sin \pi Mf}{\sin \dfrac{\pi f}{R}}\right)^{2N} - \left(2\frac{\sin \pi Mf}{\sin \dfrac{\pi f}{R}}\right)^{3N} \right| \quad (2.2)$$

2.1 IMPROVING METHOD OF CIC FILTER RESPONSE

There are many methods to improve the filter response of CIC filter. We will discuss some papers on sharpening technique of filter.

Paper 1

Zhang et al [4] present a paper to improve the performance of CIC filter. In this paper they used the compensation filter which has the inverse magnitude response to CIC filter was proposed, which improved the passband and transition band features of the CIC filter and improved the performance of CIC filter.

CIC filter is equivalent to a number of rectangular windows cascaded recursive filter form, and therefore has a more significant performance limitations. CIC filter has low attenuation and a droop in the desired passband that is dependent upon the decimation factor R and the number N of section in cascade.

In this paper, the magnitude response of CIC filter has been improved by introducing a compensation filter. To improve the magnitude drop, some compensation filters that have a magnitude response such as the sine compensator and cosine filters have been proposed.

The magnitude characteristic compensates the passband droop of the CIC filter. The transfer function of the proposed filter is given as

$$H_{\sin}(\mathrm{e}^{jw}) = \mathrm{e}^{-jwM/2}(1+\sin^2(Mw/4)) \quad (2.3)$$

The corresponding magnitude characteristic is of the form

$$|\,H_{\sin}(\mathrm{e}^{jw})\,| = |\,1+\sin^2(Mw/4)\,| \quad (2.4)$$

The cosine filter transfer function is given as

$$\mathrm{H}_{\cos}(\mathrm{z}) = (1+\mathrm{z}^{-L})/2 \quad (2.5)$$

It has the magnitude characteristic in cosine form

$$|\,\mathrm{H}_{\cos}(\mathrm{e}^{iw})\,| = \cos(\mathrm{wL}/2) \quad (2.6)$$

They analyzed that with the help of sine and cosine compensator filter passband details have been improved but this improvement is not satisfactory. So they proposed the new compensator filters. This is the inverse of the CIC filter, which can be efficient to improve the magnitude response of the CIC filter.
To achieve a flat passband, the compensation FIR filter should have a magnitude response that is the inverse of the equation (1.5) as show following

$$|\mathrm{H}_{comp}(\mathrm{f})| = \left| \frac{MR\sin(\pi f_p / R)}{\sin(\pi M f_p)} \right|^N \quad (2.7)$$

In equation (2.7), f_p is bandwidth of the passband of the single. It determines the compensation bandwidth. If R is large, the response of the compensation is approximated by the inverse sinc function, so their filter is named "inverse sinc compensation filter".
The magnitude response of the compensation filter has been proposed, so, make use of cascade of these filter, as show

$$H(\mathrm{z}) = \mathrm{H}_{CIC}(\mathrm{z})\,\mathrm{H}_{comp}(\mathrm{z}) \quad (2.8)$$

Through the compensated of the inverse sinc compensation filter, the total filter passaband magnitude response has been improved becomes flat. Transition zone is improved to be very sharp and stopband improved to very good stopband rejection.
In this paper they introduced a new design parameter F_0. F_0 is a normalized cutoff frequency, it must be meet condition : $0 < F_0 \le 0.5/M$. So, when R is large enough, that it cannot meet the above condition. The design will to be wrong. They noticed that the choice of value of F_0 is according to design parameter.
We concluded that with the help of inverse sinc compensation filter improved and became flat passband, transition zone of it's became sharp, as a result the performance of the filter has been improved.

Paper 2
Dolecek et al [5]present a paper new two-stage CIC based decimation filter for input signals occupying ¾ of the digital band. They assumed that the decimation factor M to be an even number. The decimation factor of the first stage is M/2, whereas, that of the second stage is 2. They introduced a sine based compensator filter to decrease the passband droop of the CIC filter and a cosine filter to improve the overall stopband characteristics. As a result the signal-to-noise (SNR) is improved. They used the polyphase decomposition method of the comb filter. The polyphase components of the comb filters are moved to the lower rate which is M/2 less than the input rate. Consequently there is no filtering at the high input rate. The proposed filter performs decimation efficiently using only additions/subtractions making it attractive for software radio applications.
They introduced sine-based compensator to improve the passband characteristics of the CIC filter. The magnitude characteristics compensate the passband drop of the CIC filter. The transfer function of the proposed filter is given as $H_{\sin}(\mathrm{e}^{jw}) = \mathrm{e}^{-jwM/2}(1+\sin^2(Mw/4))$ (2.9)
The corresponding magnitude characteristic is of the form

$$|H_{\sin}(\mathrm{e}^{jw})| = |1+\sin^2(Mw/4)| \quad (2.10)$$

In z-transform the transfer function is

$$H_{\sin}(\mathrm{z}) = \frac{1}{4}\left[-1+6z^{-M/2}-z^{-M}\right] \quad (2.11)$$

M must be even in order to avoid fractional delay. The compensated CIC filter is given as

$$H_{CCIC}(\mathrm{z}) = H_{CIC}(\mathrm{z})\,\mathrm{H}_{\sin}(\mathrm{z}) \quad (2.12)$$

They noticed that as a result of the compensated CIC filter, the passband droop at the end of the passband is decreased about 50%. As a consequence the side lobes at the stopband are increased. To solve this problem they proposed to use cosine filters.

$$\mathrm{H}_{\cos}(\mathrm{z}) = (1+\mathrm{z}^{-L})/2 \quad (2.13)$$

It has the magnitude characteristic in cosine form

$$|\mathrm{H}_{\cos}(\mathrm{e}^{iw})| = \cos(\mathrm{wL}/2) \quad (2.14)$$

The transfer function of the compensated filter as

$$H_{2CCIC} = H_{CCIC}(\mathrm{z})\,\mathrm{H}_{\cos}(\mathrm{z}) \quad (2.15)$$

They notice that the choice of L is the trade off between the improvement of passband and the stopband of the proposed filter. They proposed to make use of the cascade of the filters of equation (2.15) resulting in

$$H(z) = H_{CIC}^{k_1}(z)\, H_{\cos}^{k_2}(z)\, H_{\sin}^{k_3}(z) \quad (2.16)$$

They noticed that as increase in the values of parameters, k_1 and k_2 improves the stopband response while an increasing the value of k_3 improves the passband.

For efficient structure they split the decimation in two stages. The decimation factor at the first stage is M/2 an in the second one is 2. From the equation (1.5) they rewrite as

$$H_{CIC}(z) = \left[\frac{1}{2}\frac{1-z^{-M}}{1-z^{-M/2}}\right]^{k_1}\left[\frac{1}{M/2}\frac{1-z^{-M/2}}{1-z^{-1}}\right]^{k_1} (2.17)$$

Using equations (2.11), (2.13) and (2.17), from equation (2.16) they get

$$H_{pr}(z) = AH_0(z)H_1(z^{M/2})H_2(z^M) \quad (2.18)$$

Where

$$A = 2^{-2k_3-k_2}M^{-k_1} \quad (2.20)$$

$$H_1(z) = \left[\frac{1}{1-z^{-M/2}}\right]^{k_1}\left[-1+6z^{-M/2}-z^{-M}\right]^{k_3} \quad (2.21) \qquad H_2(z) = \left[1-z^{-M}\right]^{k_1} \quad (2.22)$$

Using the multirate identity the filter of equation (2.21) can be moved to the lower rate which is M/2 times less than the high input rate. Similarly the filter of equation (2.23) can also be moved to lower rate which is M times less than the high input rate. The final structure is shown in figure.

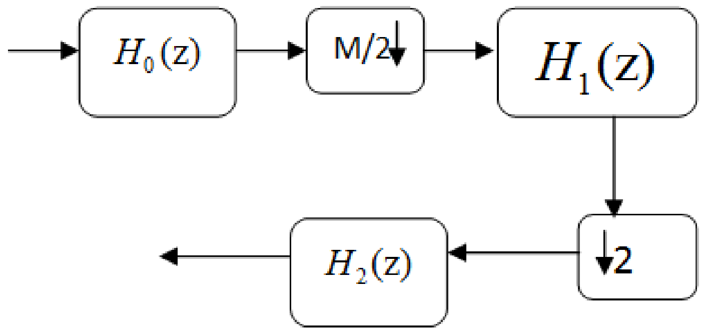

Figure 2.1:- Proposed Structure of CIC Filter

Applying the polyphase decomposition to the filter of equation (2.20) the polyphase components can also be moved to lower rate which is M/2 time less than the high input rate. Consequently the resulting structure has no filtering at high input rate.

The proposed filter performs decimation using only additions/subtractions and exhibits higher SNR compared to that of the corresponding CIC filter, as the expense of slight increase of complexity.

Paper 3

Mitral et al [6]present a paper on "Efficient Sharpening of CIC Decimation Filter". In this paper, they proposed an efficient sharpening of a CIC decimation filter for an even decimation factor. The proposed structure consists of two main sections: a section composed of a cascade of the first order moving average filters, and a sharpening filter section. The proposed decomposition scheme allows a sharpening section to operate at half of the input rate. In addition, the sharpened CIC filter is of length that is half of that of the original CIC filter. With the aid of the polyphase decomposition, the polyphase sub filters of the first section can also be operated at the half of the input rate.

Frequency response of the CIC filter exhibits a linear phase, lowpass $\sin Mx / \sin x$ characteristics with a droop in the desired frequency range in the passband that is dependent upon the decimation factor M. The signal components aliased into the passband, which is caused by the down-sampling, appear on both sides of the nulls. The worst case aliasing occurs in the passband near the first null at 1/M.

There are many schemes to sharpen the frequency response of the CIC filter but they used the simplest case of the sharpened filter. The transfer function of this case is given by

$$H_{sh}(z) = 3H_{CIC}^2(z) - 2H_{CIC}^3(z) \quad (2.23)$$

The resulting sharpened filter has a significantly improves the frequency response, i.e. the reduced passband droop and improved alias rejection. The main drawback of this structure is that the filtering is performed at the high input rate and for a given decimation factor M, the complexity increase with the number of the stages K.

For overcome these problems they implemented the filter in two stages and allowing filtering at a rate lower than the high input rate while reducing the length of the sharpened CIC filter to one half of that of the original CIC filter.

They considered the case where M=2N. In this case they rewrote the equation (1.5) as

$$H_{CIC}(z) = \left[H_1(z^2)\,H_2(z)\right]^K \quad (2.24)$$

Where

$$H_1(z^2) = \frac{1}{N}\frac{1-z^{-2N}}{1-z^{-2}} \qquad (2.25)$$

$$H_2(z) = \frac{1}{2}(1+z^{-1}) \qquad (2.26)$$

They observed that $H_1(z^2)$ has zeros at the same location as $H_{CIC}(z)$ except $z=-1$, which is provided by $H_2(z)$. Therefore they applied the sharpening to $H_1(z^2)$ instead of $H_{CIC}(z)$ and use $H_2(z)$ to achieve the necessary attenuation of the last side lobe. This suggests that the decimation filter can be constructed using different number of stages for two sections as indicated below:

$$H_n(z) = [H_1(z^2)]^K [H_2(z)]^L \qquad (2.27)$$

Applying the sharpening to $[H_1(z^2)]^K$ and the transfer function of the proposed decimation filter as

$$H_{sh,n}(z) = \{[3H_1(z^2)]^{2K} - [2H_1(z^2)]^{3K}\}[H_2(z)]^L \quad (2.28)$$

The computationally efficient realization of the proposed structure obtained using the cascade equivalence is shown in figure 2.2:

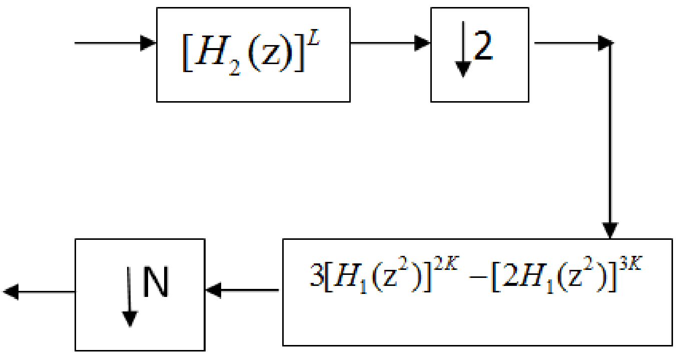

Figure 2.2:- Proposed CIC Filter

Note that the sharpening in the new structure operates at half of the input rate and that the sharpened filter $H_1(z)$ has the half the number of the coefficients of the original CIC filter $H_{CIC}(z)$. The first section consists of the cascaded of L first order moving average filters. Using the polyphase decomposition the first section can also be moved to a lower rate and implemented efficiently by using dedicated shift and add multipliers and eliminating redundant operations using the data broadcast structure.

Paper 4

Li et al [7]present a paper on "Compensation Method for CIC Filter in Digital Down Converter". In this paper, a method with the structure of polyphase decimation FIR filter is proposed to achieve wide broadband compensation of the CIC filter and sampling rate conversion. The paper focused on the usage of FIR filters to compensate CIC filters with the coefficients obtained by simulation, which extend the bandwidth and improves the flexibility of the compensation filters.

There are many methods to compensate CIC filter, i.e. based on sharpening technique, the interpolation binomial compensation method and inverse sinc function compensation. But these methods cannot guarantee the flatness of passaband when the bandwidth is wide. Because of advantage of FIR filters, i.e. adaptability, flexibility, easily adjusted coefficients and the structure of polyphase filters can be used to FIR filters, most of the commercial down-conversion chip still uses the FIR filters to compensate CIC filters.

It can be seen that CIC filter is a rectangular window. The highly symmetric structure allows efficient implementation in hardware. The disadvantage of CIC filter is that its passband is not flat, which is undesirable in wide application. Single stage CIC filter is difficult to meet the practical requirements for its poor stopband droop. We can use multistage CIC filter to increase the stopband attenuation to meet practical demands. However, with the number of the cascaded CIC filter increasing, CIC filter's passband performance will significantly become worse. Generally, the order is limited to 5 orders.
The method, using the inverse of the magnitude response of CIC filter, is proposed to design a compensation filter. The frequency response can be expressed as

$$|H(\mathrm{e}^{j\omega})| = \left|\frac{M\sin(\omega/2)}{\sin(M\omega/2)}\right| \qquad (2.29)$$

The order of the compensation filter must be minimum without any increase in complexity of the compensation filter. To obtain the filter coefficient with MATLAB they used many considerations. Due to the limitation of the filter order, it cannot get a good compensation result by using the inverse of magnitude response of CIC filter. This requires adjustment of parameters, and the compensation function is defined as

$$H_c(\omega) = \begin{cases} |H(\mathrm{e}^{j\omega})| = \left|\dfrac{M\sin(\omega/2)}{\sin((M-a)\omega/2)}\right|^{N+b} & 0 \le \omega \le \omega_p \\ 1 - \dfrac{\omega - \omega_p}{\omega_s - \omega_p} & \omega_p \le \omega \le \omega_s \\ 0 & \mathit{others} \end{cases} \qquad (2.30)$$

Where a and b are parameters which are needed to be adjusted, ω_p is the passband cut off frequency and ω_s is the stopband start frequency.

They used the mean and square error can be used to measure the performance of the filter after compensation. If the order of the filter is fixed, in order to get the best compensation effect, the square error should be minimum. They observed that if the order of the compensation filter is at the range of 12 to 24, the compensation effect is better. After getting the FIR coefficient with the best compensation effect, in order to reduce the computational speed and save hardware resources, they used polyphase decimation FIR filter to complete data decimation. They observed that after FIR compensation filter significantly reduces passband drop and improves stop band rejection.

From this paper we conclude that sharpening techniques and the interpolation binomial compensation method cannot meet broadband application. FIR filter, based on polyphase structure, is used to compensate CIC filter in view of small decimation. It enlarges the compensation bandwidth and improves the flexibility of the compensation. Filter coefficient is properly quantified, it can ensure that the flatness of the belt and stopband attenuation both obtain high computational efficiency.

Paper 5
Stephen et al [8] present a paper on "High Speed Sharpening of Decimation CIC Filter". In this paper they present a modification of an existing technique which compensates for passband droop in the frequency response of CIC filter for power of two decimation factor. The new techniques delivers passband droop correction comparable to existing techniques, while allowing faster operation than for existing structures by moving integrator away from the filter input.
They used a sharpened technique developed by Kaiser and Hamming which is discussed in equation (2.1). This technique allows the programmable FIR filter to replaced with a constant coefficient FIR filter, which, despite resulting in increased hardware consumption by the CIC portion of the circuit, leads to a more hardware efficient solution overall.
The hardware implementations of both CIC and sharpened filters result in a series of integrators running at the input (high) sample rate. As these sections are recursive, they cannot be pipelined, thus limiting the critical path to one adder operation. Non-recursive structures can be pipelined (at bit-level for increased speed) thus reducing the critical path to a fraction of an arithmetic operation and allowing the circuit to be clocked at a faster rate.
They proposed a modification by factorising the CIC filter transfer function and partially non-recursive structure is obtained. Equations (2.31)-(2.33) show factorisation by 2^p for a decimation factor of $R = 2^p F$:
$H_{CIC}(\mathrm{z}) = [\mathrm{H}_1(\mathrm{z}^{2p})]^N [H_2(\mathrm{z})]^s$ (2.31)
Where

$$H_1(z^{2p}) = \frac{1}{F}\frac{1-z^{-2p}F}{1-z^{-2p}} \quad (2.32)$$

$$H_2(z) = \frac{1}{2p}\sum_{i=0}^{2p-1} z^{-i} \quad (2.33)$$

N refers to the numbers of sections in the recursive portion of the filter, while s presents the number of cascaded of the non-recursive portion of the filter. Equation (2.1) can be implemented as p stage separated non recursive sections, each decimating by 2. Sharpening is the applied to equation (2.32), yielding a transfer function of

$$H_{PCIC}(z) = [H_2(z)]^s[3H_1^{2N}(z^{2p}) - 2H_1^{3N}(z^{2p})] \quad (2.34)$$

For the partially sharpened CIC (PCIC) filter. The z^{2^p} terms become z terms when down sampling by 2^p is moved prior to the sharpening of H_1. The integrator sections in the PCIC filter run at a speed 2^p times slower than those in the sharpened filter while, at the filter input, the non recursive sections can be implemented using the polyohase decomposition, a parallel processing technique which enables high speed implementation. In addition, N and s may assume different values; for N=2s, acceptable alias rejection may be maintained while allowing simpler polyphase implementation of the non-recursive portion of the filter and improved passband droop correction. By multiplexing the cascades, the filter can programmed for different power of two decimation ratios from 4 upward.

They observed that for N=s, the sharpened CIC filter still provides better passband droop correction and, in most cases, improved alias rejection. However, for N=2s, the PCIC provides better passband droop performance than the sharpened CIC filter at the expense of less alias suppression. For decimation ratio of 32 and above, the results for the two PCIC and sharpened CIC filter converge, providing near identical values for alias rejection and passband droop. Therefore, across a wide range of decimation ratios the PCIC with N=2s provides a passband droop correction very similar to that of sharpened CIC filter. The PCIC would be suitable for use in circuits in which operating speed is of greater concern than a very high degree of alias rejection.

2.2 DISCUSSION

In **paper 1,** they cascaded the compensation filter which has inversed magnitude response to CIC filter, which improved the passband and the transition band features of the CIC filter and improved the performance of CIC filter. Value of normalized frequency F_o change according to design requirements, which is depends upon the value of R. This is the gap of the paper.

In **paper 2,** they cascaded the sine and cosine filter with CIC filter and using the polyphase decomposition of the comb filters, the polyphase components of the comb filters are moved to the lower rate. Sine filter decrease the passband droop of CIC filter and cosine filter improve the overall stopband characteristic. As a result the SNR is improved. For getting high SNR of CIC filter expense of slight increase of complexity and the choice of the design parameters are not more systematic. This is the gap of the paper.

In **paper 3,** they used the sharpening technique proposed by Kaiser and Hamming with polyphase decomposition technique. The sharpening is moved to lower section, which operates at the half of the high input rate. But this CIC filter is good for only even decimation factor.

In **paper 4,** they used a method with the structure of polyphase decimation FIR filter to achieve wide broadband compensation of the CIC filter and sampling rate conversion. FIR compensation filter reduces passband droop and improves the stopband rejection. But to get better compensation effect, order of filter and design parameter values are in the range.

In **paper 5,** they used the power of two decimation factors technique, which delivers passband droop correction comparable to existing techniques nad allowing faster operation than for existing structure by moving integrator sections away from the filter input. But if N=2s then the PCIC provides better passband droop performance than the sharpened CIC filter at the expense of less alias suppression.

Comparison table is shown below:-

Paper	Technique	Positive Points	Gaps
Paper 1	Cascaded the compensation filter which has inversed magnitude response to CIC filter	Improved the passband and the transition band features of the CIC filter and improved the performance of CIC filter	Value of normalized frequency F_o change according to design requirements
Paper 2	Cascaded the sine and cosine filter with CIC filter and using the polyphase decomposition of the comb filters	Sine filter decrease the passband droop of CIC filter and cosine filter improve the overall stopband characteristic. As a result the SNR is improved	Increase of complexity and the choice of the design parameters are not more systematic
Paper 3	used the sharpening technique proposed by Kaiser and Hamming with polyphase decomposition technique	Sharpening is moved to lower section, which operates at the half of the high input rate	Only even decimation factor
Paper 4	Polyphase decimation FIR filter	Reduces passband droop and improves the stopband rejection	order of filter, design parameter values are in range.
Paper 5	Used the power of two decimation factors technique	Delivers passband droop correction and allowing faster operation	if N=2s then the PCIC provides better passband droop performance

Table 2.1:- Comparison Table of Improved CIC Filter

III. Conclusion

E.B. Hogenauer proposed a CIC filter with an efficient way of performing decimation and interpolation. It is flexible, multiplier-free filter suitable for hardware implementation that can also handle arbitrary and large rate changes. Because their structure consists of a integrator section operating at high sampling rate and a comb section operating at low sampling rate. CIC filter is equivalent to N FIR filters, each having a rectangular impulse response. Since all of the coefficients of these FIR filters are unity, and therefore symmetric, a CIC filter has a linear phase response and a constant group delay.

Filter sharpening can be used to improve the response of a CIC filter. In most applications it is required to have a flat passband, otherwise the original signal may be destroyed. Unfortunately, the CIC filter alone suffers with a passband droop, which in many cases, cannot be accepted.

For improving the passband and the transition band features of the CIC filter and improving the performance of CIC filter we can use many techniques, such as compensation filter cascaded with CIC filter, Sharpening technique proposed by Kaiser and Hamming, polyphase decimation FIR filter to achieve wide broadband compensation of the CIC filter.

All techniques are good for improving CIC filter design but in each technique has some gaps. But polyphase decimation FIR filter to achieve wide broadband compensation of the CIC filter is better technique to other techniques.

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