ISSN: 2278-0181
Vol. 2 Issue 7, July - 2013

Analysis And Implementation Of Multi-Rate Cascaded Integrator Comb (CIC) Interpolation & Decimation Filter With Compensation Filter For Software Defined Radio.

Prof. Madki M. R. Ms Pathak A. A.

Department of Electronics,
Walchand Institute of Technology,
Solapur.413006

I Abstract

Software defined radio (SDR) is a radio in which the properties of carrier frequency, signal bandwidth, modulation ,and several other characteristics are defined by software. Normal FIR architectures and its variants fail to work at such high frequencies. Cascaded Integrator comb (CIC) decimation filter is useful to reduce the data sampling rate such high bandwidth applications

The implementation of decimator & interpolator using MATLAB as standard FIR and cascaded integrator comb filter by using the multistage design techniques. The hardware saving can be achieved by using multistage Nyquist decimator design. It Reduces computational work load, lower filter order, lower coefficient sensitivity and noise and less stringent memory requirements. This paper also presents a simple design of compensation FIR filter for CIC decimation filter which will correct the passband droop. Compensation filter design consists of polynomial based design of FIR filter which are used in cascade with CIC filter. The resulting structure is multiplier less and exhibits small passband droop in comparison to CIC filter. It modifies the frequency response of CIC decimation filter while maintaining the linear phase

Keywords: CIC Filters, FIR Filters, Sample Rate Conversion (SRC), Decimator, Interpolators, MATLAB

II Introduction

With the advance in digital signal processing and digital communication techniques, software defined radio (SDR) becomes a reality. For an SDR with multi-protocol and/or multi-band capabilities, sample rate conversion (SRC) is an essential element in the software architecture of the SDR.

The most popular and computationally efficient approach for SRC is to use the cascaded integrator-comb (CIC) filter which allows for a direct digital-digital SRC, i.e the conversion is performed digitally with digital input and digital output but without an analog stage. CIC decimation filters require less computation but large passband droop occurs in the frequency response. This paper presents a simple design of compensation FIR filter for CIC decimation filter which will correct the passband droop. Compensation filter design consists of polynomial based design of FIR filter which are used in cascade with CIC filter.

III Software Defined Radio

Software-Defined Radio (SDR) refers to the technology where software modules running on a generic hardware platform consisting of DSP's and general purpose microprocessors are used to implement radio functions such as generation of transmitted signal (modulation) at transmitter and tuning/detection of received radio signal (demodulation) at receiver.[1]

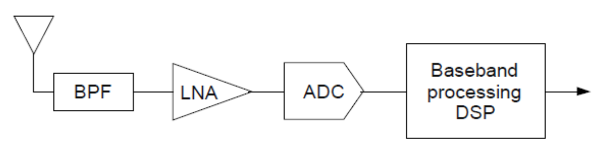

Fig 1 Software Defined Radio

This technology is based on software defined wireless communication protocols instead of hardware implementations. In other words, frequency band, air interface protocol and functionality can be upgraded with software download and update instead of a complete hardware replacement. SDR provides an efficient and secure solution to the problem of building multi-mode, multi-band and multifunctional wireless communication devices. An SDR is capable of being re-programmed or reconfigured to

ISSN: 2278-0181
Vol. 2 Issue 7, July - 2013

operate with different waveforms and protocols through dynamic loading of new waveforms and protocols. These waveforms and protocols can contain a number of different parts, including modulation techniques, security and performance characteristics defined in software as part of the waveform itself [2].

GSM covers wide range of frequency band in corporate modern applications. It is difficult to accommodate this entire frequency band in a single hardware, hence to cover all applications one has to change hardware. The Software Defined Radio gives liberty to use same hardware with selectable software .The software filters plays vital role in SDR. Presently used Cascaded Integrator Comb filters shows poor performance at the low frequencies.

To enhance the performance of Cascaded Integrator Comb filter we propose Cascaded Integrator Comb filter in addition with the compensation filter at appropriate locations. This addition will be benefit for Software Defined Radio at the lower frequency. This compensator helps to avoids passband droops with maintaining the linear phase & by reducing the delays between the samples we make our system more efficient for fast operations in Software Defined Radio.

IV Basic filtering process

The Digital filter can be used to remove undesired signal, such as noise and hence to extract valuable information of the signal. Basic Fourier transform theory states that the linear convolution of two sequences in the time domain is the same as multiplication of two corresponding spectral sequences in the frequency domain.

Filtering is in essence the multiplication of the signal spectrum by the frequency domain impulse response of the filter. For an ideal lowpass filter, the pass band part of the signal spectrum is multiplied by one and the stopband part of the signal by zero. A digital filter takes a digital input, gives a digital output, and consists of digital components. In a typical digital filtering application, software running on a digital signal processor (DSP)[3].

Fig 2 Basic Filter Structure

A Cascaded Integrator Comb (CIC) filter is a special class of linear phase, finite impulse response (FIR) filter. CIC filters do not require multipliers and use a limited amount of storage. Therefore, CIC filters are more efficient than conventional FIR filters, especially in fixed-point applications [3].CIC filters were invented by Eugene B. Hogenauer, and are a class of FIR filters used in multi-rate processing. The CIC filter finds applications in interpolation and decimation. Unlike most FIR filters, it has a decimator or interpolator built into the architecture[4].The system function for the composite CIC filter referenced to the high sampling rate, f_s is:

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

Where:
R = decimation or interpolation ratio
M = number of samples per stage (usually 1 but sometimes 2)
N = number of stages in filter

Sample rate conversion (SRC) with rational factors can be realized by interpolation followed by decimation. A time-variant implementation of CIC-filters is presented which circumvents the high intermediate sample rate. This time-variant implementation results in a linear periodically time-variant system (LPTV) which is completely equivalent to its original linear time-invariant system (LTI) consisting of the interpolator and the decimator. Thus well-known methods of system analysis can be used by analyzing the LTI system, while implementing the system as an LPTV system, avoiding the high intermediate sample rates of the LTI system[5] . The advantage of CIC-filters not having stored the coefficients of the impulse response but rather the description of the impulse response, enabling an implementation which is independent of the interpolation- as well as the decimation-factor, is preserved with the LPTV system. In contrast to Lagrange interpolators cancelling only the image components of the interpolated signal, time-variant CIC-filters also cancel the aliasing components, which is important in applications, where anti aliasing is more important than anti-imaging[6] .

A cascade of interpolator and decimator requires an intermediate clock rate which is higher than input sample rate.[7] Since only the integrator section is operating at this high intermediate sampling rate it should be merged with the up- and down-sampler to a time-variant unit which is clocked only at input sampling rate *fin*. Two facts are used on the one hand up-sampling means just filling in L - 1 zero pads between each pair of input samples and on the other hand down-sampling means picking up every M^{th} value of the signal at the output of the filter. While calculating only the necessary intermediate steps, up-sampling will be done virtually.[10]

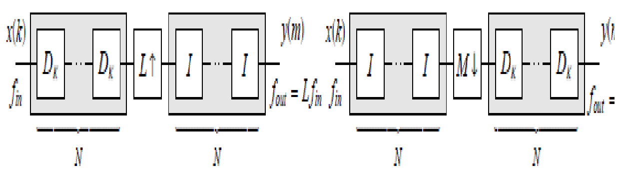

Fig 3 Structure of Interpolator & Decimator

V Methodology

1. Implementation of CIC Interpolation Filters for Receivers.

Digital interpolation filters are often referred to as filters with adjustable fractional delay and are widely used in symbol timing recovery block. This is deals with the efficient implementation of conventional Farrow-type polynomial interpolator. This structure avoids the use of two sub-filters without any performance degradation, leading to a very efficient interpolator. [8]

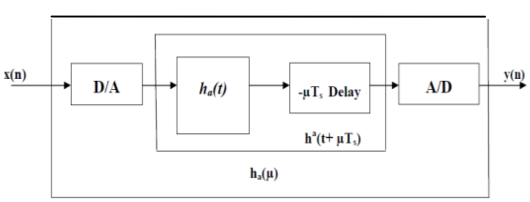

Fig 4 CIC Interpolation Model

For a given fractional delay value μ and input sequence $x(n)$, digital Interpolation filter which produces an output $y(n)$delayed by $-\mu Ts$, where Ts is the input sampling time, can be modeled as depicted in Fig. Input sequence $x(n)$ is converted to a discrete time signal through an ideal D/A converter, and is approximated to its original analog signal by filtering it with analog interpolation filter $ha(t)$.The output delayed by $-\mu Ts$ is transformed again to digital signal $y(n)$via an ideal A/D converter.

Let impulse response of equivalent discrete interpolator be $h_n(\mu)$ then it can be represented as

$$h_n(\mu) = h^a(\text{t})_{|\text{t}\leftarrow \text{n}+\mu}$$

Now assume that h^a (t) satisfies the following properties:

[1] h^a(-t)= h^a(t)

[2] h^a(t) is non zero only in the interval

The recursive structure, proposed in [5], also known as the *cascaded integrator-comb* (CIC) *filter*, consists of two main sections: an integrator section and a differentiator section separated by a down-sampler. The non-recursive structure is realized by cascading identical filters each down-sampled by a factor of 2 [6]–[7]. A new efficient structure for a sharpened comb factor-of decimation filter is proposed. The structure consists of two main sections: a cascade of comb filters followed by down-sampling with a factor M1 , and a sharpened comb filter followed by downsampling with a factor ,M2 where M= M1M2 .Over sampled filter banks are used to minimize aliasing by reducing the number of significant aliasing terms [8], [9]. This paper presented an approach for designing low-delay nonuniform filter banks. The low delay is achieved by relaxing the linear phase constraints of traditional pseudo-QMF banks.

2. Implementation of Decimator using CIC Filter

Down sampler is basic sampling rate alteration device used to decrease the sampling rate by an integer factor .In this, decimator is implemented and analyzed using different designs. So multistage Nyquist decimators are the best to perform down sampling and provide cost effective solutions for DSP based wireless communication applications.

A new efficient structure for a sharpened comb factor-of decimation filter is proposed. The structure consists of two main sections: a cascade of comb filters followed by down-sampling with a factor M1 , and a sharpened comb filter followed by downsampling with a factor ,M2 where M= M1M2 .Over sampled filter banks are used to minimize aliasing by reducing the number of significant aliasing terms [8], [9]. This paper presented an approach for designing low-delay non uniform filter banks. The low delay is achieved by relaxing the linear phase constraints of traditional pseudo-QMF banks. When decimating, the bandwidth of a signal is reduced to an appropriate value so that minimum aliasing occurs when reducing the sampling rate. An acceptable transition width needs to be incorporated into the design of the low pass filter used for decimation along with pass band ripple and finite stop band attenuation[10]

3. Implementation of Compensation Filter

In typical decimation/interpolation filtering applications we want reasonably flat passband and narrow transition-region filter performance. These desirable properties are not provided by CIC filters alone, with their drooping passband gains and wide transition regions. We alleviate this problem, in decimation for example, by following the CIC filter with a compensation non-recursive FIR filter, to narrow the output bandwidth and flatten the passband gain. The compensation FIR filter's frequency magnitude response is ideally an inverted version of the CIC filter passband response. With the dotted curve representing the uncompensated passband droop of a 1st-order R = 8 CIC filter, the solid curve represents the compensated response of the cascaded filters. If either the passband bandwidth or CIC filter order increases the correction becomes greater, requiring more compensation FIR filter taps.

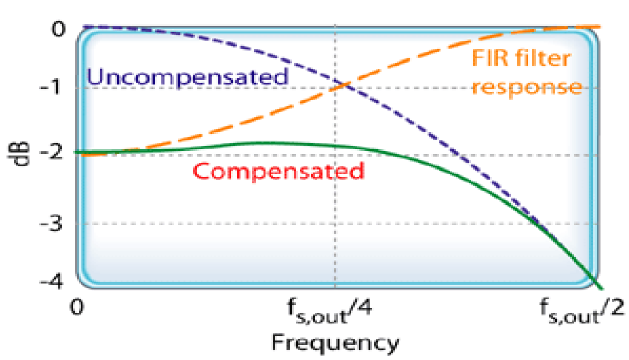

Fig 5 Compensation FIR filter responses; with decimation CIC filter

Those dashed curves in **Fig 5** represent the frequency magnitude responses of compensating FIR filters within which no sample-rate change takes place. (The FIR filters' input and output sample rates are equal to the fs,out output rate of the decimating CIC filter.)

The structures presented here are cascades of CIC decimation and simple polynomial-based filter finite-impulse response (FIR) filter [12].The key point is to design the FIR filter so that the frequency response droop remain minimum and filtering performance of the overall structure remain at the same level [8].FIR filter design essentially consists mainly of two parts approximation problem and realization problem. In this method a desired or ideal response is chosen, usually in the frequency domain, then an allowed class of filters is chosen (e.g. the length N for a FIR filters). In this design we will present an approach to realize compensation FIR filters using polynomials but here, a special type of transformation is used along with polynomial approach to design compensation FIR filter. This method is very simple in terms of computation and approach. Also this method has lot of variations so that we can define more than one type of FIR filter from one set of specification.

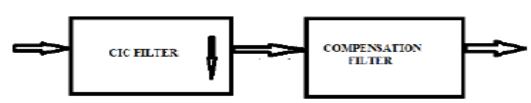

Fig 6 CIC filter with compensation filter

Compensation filter is designed using proposed method [8]:

1. First transform the required filter characteristics to a function, which we call as object function, by using a special transformation.
2. This object function is thereafter realized by using a previously defined set of polynomials.
3. The realized object function is then converted back to filter characteristics using inverse of the transform.

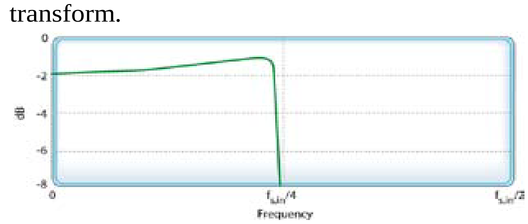

Fig 7 Frequency magnitude response of a decimate-by-2 compensation filter

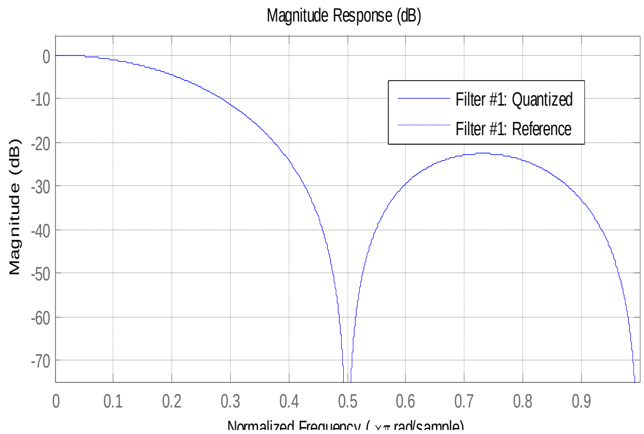

Fig 8 Magnitude response for the frequencies in GHZ by CIC Decimation filter

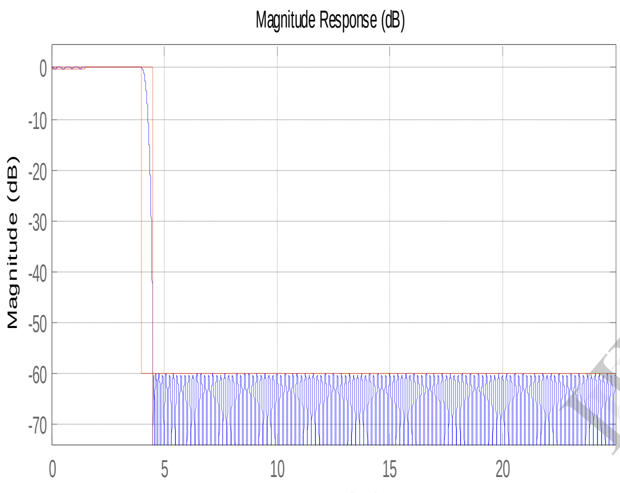

Fig 9 Magnitude response for the frequencies in GHZ by CIC compensation filter.

Fig 8 shows the response of CIC decimator at input frequency in GHZ & Fig 9 shows the responce of CIC Compensation circuit which minimizes the passband droop & gives flat passband. In case of large stop band attenuation the number of stages also increases as a result CIC filter frequency response does not have a wide, flat passband. To achieve the frequency response correction a FIR filter that has a magnitude response, inverse of the CIC filter can be applied. Such filters are called compensation filters [11]. In other words, the compensation filter always operates at the lower rate in a rate conversion design. One benefit of running the compensation filter at the low rate is to achieve a more efficient hardware solution, that is, more time sharing in the compensation FIR filter.

VI Conclusion

This method will be useful to improve the Interpolation & Decimation performance in the limited hardware, or reduce the hardware complexity in software defined radio. Digital interpolation filters have been widely employed in Consumer areas to simplify the analog front end, and to improve the overall receiver performances.

This method deals with the design and implementation of a decimation filter to be used in wideband radio-frequency wireless systems. A decimation filter cascade structure will design to meet the GSM and DECT standards specifications. In this design less computation is required to design cascade of CIC with compensation filter now. On using proper polynomial passband droop can be reduced at maximum level and without much increase in complexity. In case of more passband droop we can use the polynomial of higher order. Proper selection of compensation filter also help to define frequency response more close to ideal values Also multistage filtering can be done in order to reduce the sampling rate at first stage .

VII References

1) "Software Defined Radio (SDR) Forum, Technical Definitions, "http://www.sdrforum.org.
2) J. Mitola, "Software radios-survey, critical evaluation and future directions," in Proceedings of IEEE National Telesystems Conference (NTC '92), vol. 13, pp. 15–23, Washington, DC ,USA, May 1992.
3) Donadio, Matthew (2000) CIC Filter Introduction "Hogenauer introduced an important class of digital filters called 'Cascaded Integrator-Comb', or 'CIC' for short (also sometimes called 'Hogenauer filters').
4) C. W. Farrow, "A continuously variable digital delay element," in Proc. IEEE Int. Symp. Circuits & Syst., Espoo, Finland, June 1988, pp. 2641-2645.
5) F. Harris, "Performance and design of Farrow filter used for arbitrary resampling," in Proc. 13th Int. Conf. on Digital Signal Processing, Santorini, Greece, July 1997, pp. 595- 599.
6) S. Chu and C. S. Burrus, "Multirate filter designs using comb filters," IEEE Trans. Circuits Syst., vol. CAS-31, no. 11, pp. 913–924, Nov. 1984.
7) Y. Jang and S. Yang, "Non-recursive cascaded integrator-comb decimation filters with integer

ISSN: 2278-0181
Vol. 2 Issue 7, July - 2013

multiple factors," in Proc. IEEE 2001 Midwest Symp. Circuits and Systems, vol. 1, 2001, pp. 130–133.

8) Z. Cvetkovic and J. Johnston, "Nonuniform oversampled filler banks for audio signal processing," IEEE Trans. Speech Audio Process., vol.11, no. 5, pp. 393–399, Sep. 2003.
9) Daeyoung Kim, Madihally J. Narashima, "Design of Optimal Interpolation Filter for Symbol Timing Recovery," IEEE Trans. Comm., vol. 45, No. 7, pp. 877-884, July 1997.
10) F. Harris, "Performance and design of Farrow filter used for arbitrary resampling," in Proc. 13th Int. Conf. on Digital Signal Processing, Santorini, Greece, July 1997, pp. 595- 599.
11) S. Chu and C. S. Burrus, "Multirate filter designs using comb filters," IEEE Trans. Circuits Syst., vol. CAS-31, no. 11, pp. 913–924, Nov. 1984.