International Journal of Engineering Research and General Science Volume 2, Issue 5, August-September, 2014
ISSN 2091-2730

Implementation of Decimation Filter for Hearing Aid Application

Prof. Suraj R. Gaikwad, Er. Shruti S. Kshirsagar and Dr. Sagar R. Gaikwad
Electronics Engineering Department, D.M.I.E.T.R. Wardha
email: surajrgaikwad@gmail.com
Mob. No: +91982389441

Abstract - A hearing aid is a small electronic device that one wears in or behind his/her ear who have hearing loss. A hearing aid can help to the people to hear more in both quiet and noisy situations. It makes sounds louder so that a person with hearing loss can listen, communicate, and participate better in daily activities. In this paper, we implemented a digital filter which is used for hearing aid application. The implemented filter is based on the multirate approach in which high sampling rate signal is decimated to low sampling rate signal respectively. This proposed decimated filter is designed and implemented using the Xilinx System Generator and Matlab Simulink.

Keywords - Digital Filter, CIC filter, FIR filter, Half band filter and Oversampling Concept.

INTRODUCTION

Filters are a basic component of all signal processing and telecommunication systems. The primary functions of a filter are one or more of the followings: (a) to confine a signal into a prescribed frequency band or channel (b) to decompose a signal into two or more sub-band signals for sub-band signal processing (c) to modify the frequency spectrum of a signal (d) to model the input-output relation of a system voice production, musical instruments, telephone line echo, and room acoustics [2].

Hearing aids are primarily meant for improving hearing and speech comprehensions. Digital hearing aids score over their analog counterparts. This happens as digital hearing aids provide flexible gain besides facilitating feedback reduction and noise elimination. Recent advances in DSP and Microelectronics have led to the development of superior digital hearing aids [6]. Many researchers have investigated several algorithms suitable for hearing aid application that demands low noise, feedback cancellation, echo cancellation, etc., however the toughest challenge is the implementation [8].

DIGITAL FILTER

A digital filter uses a digital processor to perform numerical calculations on sampled values of the signal. The processor may be a general-purpose computer such as a PC, or a specialized DSP (Digital Signal Processor) chip [3]. The analog input signal must first be sampled and digitized using an ADC (analog to digital converter). The resulting binary numbers, representing successive sampled values of the input signal, are transferred to the processor, which carries out numerical calculations on them. These calculations typically involve multiplying the input values by constants and adding the products together [7]. If necessary, the results of these calculations, which now represent sampled values of the filtered signal, are output through a DAC (digital to analog converter) to convert the signal back to analog form. In a digital filter, the signal is represented by a sequence of numbers, rather than a voltage or current. The figure1: shows the basic setup of such a system.

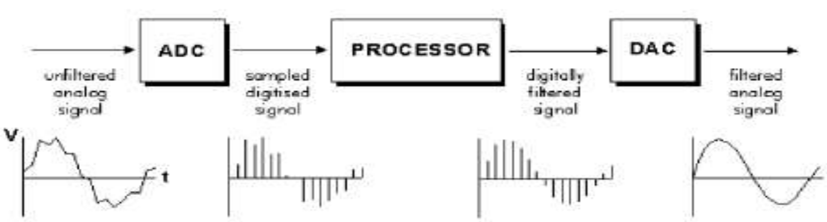

Figure1: Basic set-up of a digital filter

CIC FILTER

In 1981, E. B. Hogenauer introduced 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) [14].

The simplest CIC filter is composed of a comb stage and an integrator stage. The block diagram of three-stage CIC filter is shown in figure 2.

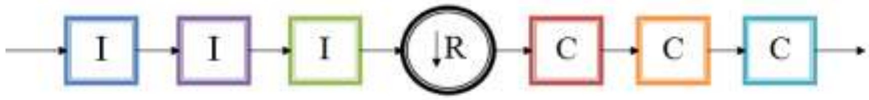

Fig: 2 (a) Three Stage Decimating CIC Filter

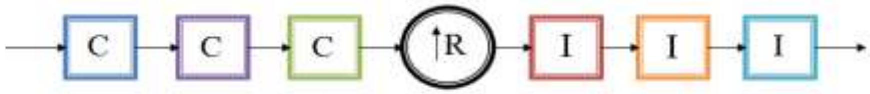

Fig: 2 (b) three Stage Interpolating CIC Filter

FIR FILTER

In signal processing, a finite impulse response (FIR) filter is a filter whose impulse response (or response to any finite length input) is of *finite* duration, because it settles to zero in finite time.This is in contrast to infinite impulse response (IIR) filters, which may have internal feedback and may continue to respond indefinitely (usually decaying) [12]. The impulse response of an Nth-order discrete-time FIR filter (i.e., with a Kronecker delta impulse input) lasts for $N + 1$ samples, and then settles to zero. The non-recursive nature of FIR filter offers the opportunity to create implementation schemes which significantly improve the overall efficiency of the decimator.

We have designed and implemented a conventional comb-FIR-FIR decimation filter. FIR filters offer great control over filter shaping and linear phase performance with waveform retention over the pass band.

OVERSAMPLING CONCEPT

In signal processing, oversampling is the process of sampling a signal with a sampling frequency significantly higher than the Nyquist frequency. Theoretically a bandwidth-limited signal can be perfectly reconstructed if sampled at or above the Nyquist frequency. Oversampling improves resolution, reduces noise and helps avoid aliasing and phase distortion by relaxing anti-aliasing filter performance requirements [3].

IMPLEMENTATION OF CIC-FIR-FIR DECIMATION FILTER STRUCTURE

The incoming oversampled signal at the rate of 1.28 MHz has to be down-sampled at the rate of 20 KHz. We have chosen passband frequency of 4 KHz because the human ear is sensitive to all the sounds within the range of 4 KHz. Figure 3 shows that the proposed decimation filter structure using CIC-FIR-FIR filter.

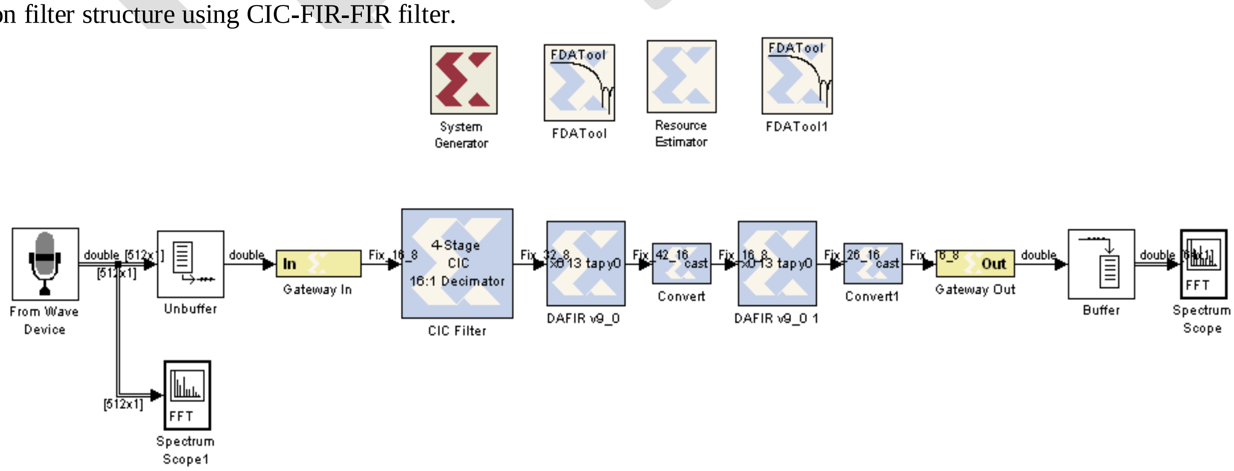

Fig. 3: Simulink model of CIC-FIR-FIR Decimation filter

This Simulink model of CIC-FIR-FIR Decimation filter is designed using Matlab Simulink and Xilinx System Generator. In this design, the incoming sampling rate is 1.28 MHz which is first down sampled by using Xilinx CIC filter and then two Xilinx DAFIR filters. These FIR filters are based on the 'Distributed Arithmetic' principle, which results in less hardware and less power consumption compared to other decimation filters.

The overall frequency specification of CIC filter is given in table 1.

No. of Stages (N)	4
Sampling Frequency (Fs)	1.28 MHz
Decimation Factor (R)	16
Bit gain (G)	65536
No. of output bits (Bout)	32
Filter Gain (Gf)	1
Scale Factor (S)	1

Table 1.Frequency specification of CIC filter

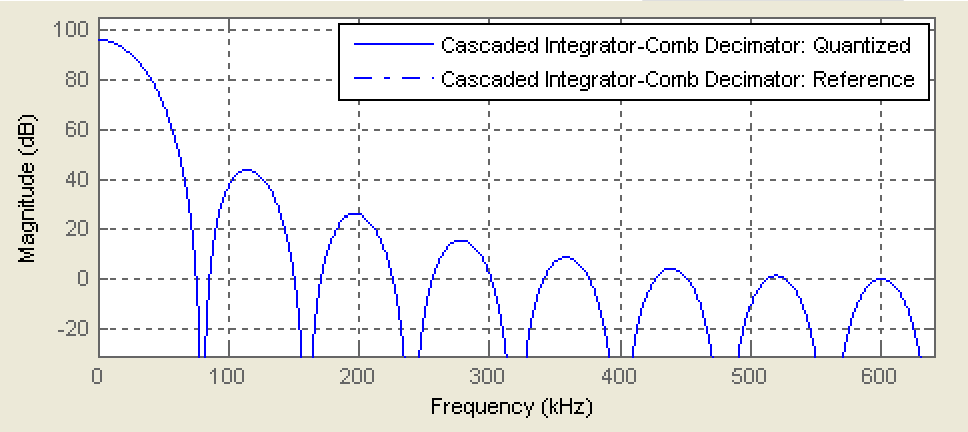

Fig. 4: Magnitude response of 4 stage CIC filter

Above figure shows that the magnitude response of 4 stage CIC filter in which the attenuation is obtained is about 48 dB. This magnitude response is plotted with N = 4, R = 16 and M = 1.

FIRST FIR FILTER DESIGN

By considering the application requirements, FIR filter and IIR filter structures can be used to meet the design specifications. FIR filters offer great control over the filter shaping and linear phase performance with the waveform retention over the pass band. Due to its all-zero structure, the FIR filter has a linear phase response necessary for audio application, but at the expense of the high filter order. IIR filter can be designed with much smaller orders than the FIR filters at the expense of the nonlinear phase. It is very difficult to design a linear phase IIR filter. Thus, we have designed FIR filter as a compensation filter. The filter specification of this FIR filter is given in table 2.

Sampling Frequency (Fs)	80 KHz
Passband Frequency (Fpass)	20 KHz
Stopband Frequency (Fstop)	35 KHz
Transition width (Δf)	0.1875
Passband Attenuation (Apass)	1 dB
Stopband Attenuation (Astop)	85 dB
Filter Length (N)	12

Table 2: Filter specification of first FIR filter

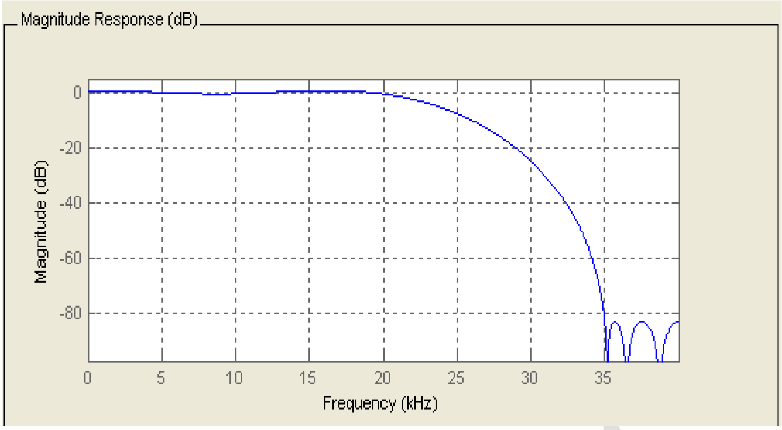

Fig: 5: Magnitude response of FIR filter

Above figure shows that the magnitude response of first FIR filter in which the stop band attenuation is obtained is about 85 dB. This magnitude response is plotted Fpass = 20 KHz and Fstop = 35 KHz.

SECOND FIR FILTER DESIGN

An additional FIR filter is designed to push out of band undesired signals. The FIR filter is used in the last stage instead of a shaping filter for less power consumption because a shaping filter has more taps than an FIR filter. Second FIR filter is used as corrector filter that having passband of 4 KHz because the human ear is sensitive to all the sounds within the range of 4 KHz. From the frequency response of second FIR filter it can be seen that stop band attenuation of more than 100 dB is obtained which is suitable for this corrector filter. Filter specification of second filter is given in the table 3.

Sampling Frequency (Fs)	40 KHz
Passband Frequency (Fpass)	4 KHz
Stopband Frequency (Fstop)	15 KHz
Transition width (Δf)	0.275
Passband Attenuation (Apass)	1 dB
Stopband Attenuation (Astop)	100 dB
Filter Length (N)	8

Table 3: Filter specification of second FIR filter

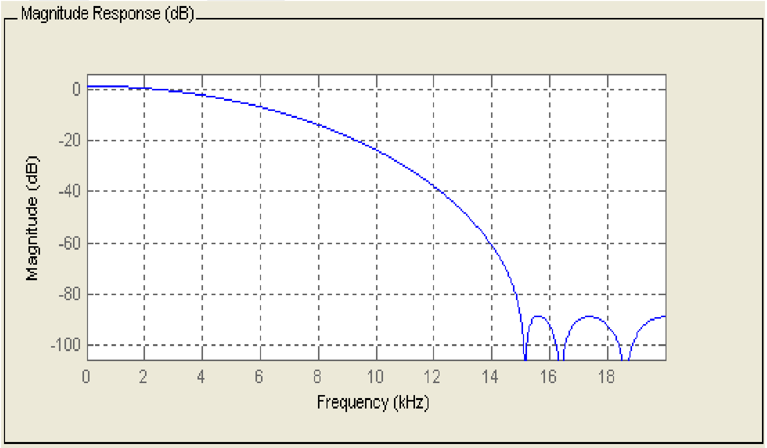

Fig: 6: Magnitude response of second FIR filter

Above figure shows that the magnitude response of second FIR filter in which the stop band attenuation is obtained is more than 100 dB. This magnitude response is plotted Fpass = 4 KHz and Fstop = 15 KHz.

CIC-HALF BAND FIR-FIR DECIMATION FILTER STRUCTURE

This decimation filter is implemented by using CIC-Half band FIR-FIR filter and the block diagram of this filter is shown in fig. 6.8. The operation of this filter is very similar to the CIC-FIR-FIR filter. The incoming oversampled signal at the rate of 1.28 MHz has to be down-sampled at the rate of 20 KHz. We have chosen passband frequency of 4 KHz because the human ear is sensitive to all the sounds within the range of 4 KHz.

A half-band IIR filter can have fewer multipliers than the FIR filter for the same sharp cutoff specification. An IIR elliptic half-band filter when implemented as a parallel connection of two all-pass branches is an efficient solution. The main disadvantage of elliptic IIR filters is their very nonlinear phase response [9]. To overcome the phase distortion one can use optimization to design an IIR filter with an approximate linear phase response, or one can apply the double filtering with the block processing technique for real-time processing. For the appropriate usage of digital filter design software in half-band filter design, it is necessary to calculate the exact relations between the filter design parameters in advance and accurate method can be found in the FIR half-band filter.

We have designed a CIC-Half band FIR-FIR decimation filter using Matlab Simulink model and Xilinx system Generator for the same specification of CIC-FIR-FIR decimation filter and the designed Simulink model of CIC-Half band FIR-FIR filter shown in figure 7.

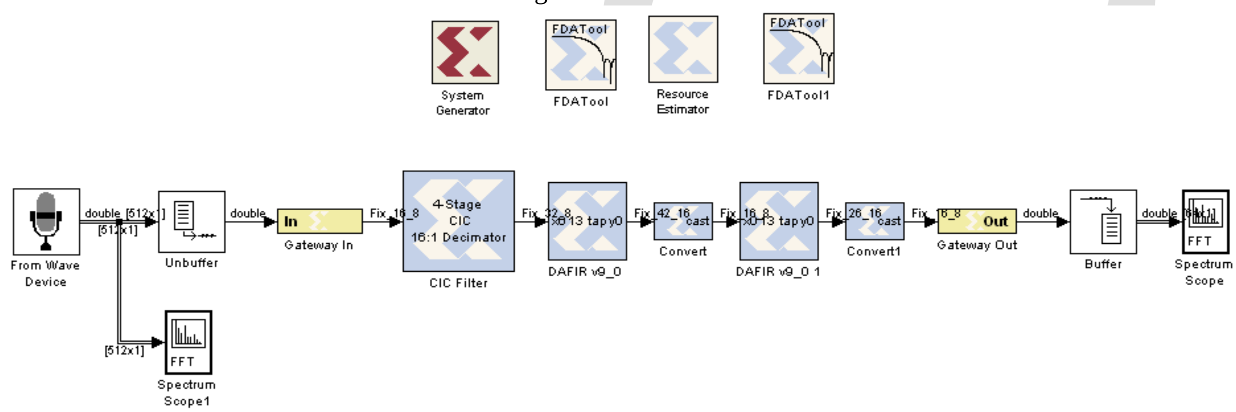

Fig. 7: Simulink model of CIC-Half band FIR-FIR Decimation filter

This Simulink model of CIC-Half band FIR-FIR Decimation filter is designed using Matlab Simulink and Xilinx System Generator. In this design, the incoming sampling rate is 1.28 MHz which is first down sampled by using Xilinx CIC filter and then two Xilinx DAFIR filters. In this case, first DAFIR filter is set as a half band FIR filter. These FIR filters are based on the 'Distributed Arithmetic' principle, which results in less hardware and less power consumption compared to other decimation filters.

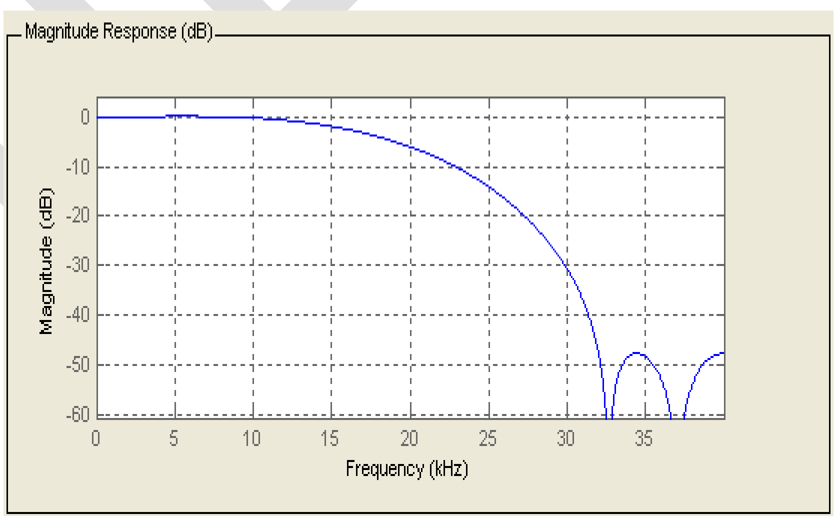

Fig: 8: Magnitude response of Half-band FIR filter

Above figure shows that the magnitude response of half band FIR filter in which the stop band attenuation is obtained is more than 50 dB.

RESULT

Decimation Filter Architecture	Number of Taps	Number of Slices	Number of Flip-Flops	LUT	IOB
CIC-FIR-FIR Filter	20	2644	4769	3561	32
CIC-Half band FIR-FIR Filter	14	2548	4729	3394	32

Table 4: Comparison between Decimation filter architectures

The table 4 shows that the cell used for CIC-FIR-FIR and CIC-half band FIR-FIR filter design. The CIC-Half band FIR-FIR filter required less number of taps due to half-band filter and also it uses less number of slices, flip-flops and LUTs as compared to CIC-FIR-FIR filter. Thus the area used and power consumption is less using the CIC-Half band FIR-FIR filter design compared to the CIC-FIR-FIR filter design. Hence we have concluded that the designed CIC-Half band FIR-FIR decimation filter is a hardware saving structure.

CONCLUSION

The decimation filter is designed using oversampling sampling rate for audio application. CIC-FIR-FIR filter and CIC-Half band FIR-FIR filter are designed and compared in terms of storage requirement, area used and power consumption for same specifications. It is observed that the CIC-Half band FIR-FIR filter required less storage for filter coefficients, less area and less power consumption than the CIC-FIR-FIR filter. Hence, CIC-Half band FIR-FIR filter is highly efficient than CIC-FIR-FIR filter.

REFERENCES:

[1] L.C Loong and N.C Kyun, “Design and Development of a Multirate Filters in Software Defined Radio Environment”, International Journal of Engineering and Technology, Vol. 5, No. 2, 2008.

[2] Suraj R. Gaikwad and Gopal S. Gawande, “ Implementation of Efficient Multirate Filter Structure for Decimation”, International Journal of Current Engineering and Technology, Vol.4, No.2 , April 2014.

[3] Fredric J. Harris and Michael Rice, “Multirate Digital Filters for Symbol Timing Synchronization in Software Defined Radios”, IEEE Journal vol. 19, no. 12, December 2001.

[4] Ronald E. Crochiere and Lawrence R. Rabiner, “Further Considerations in the Design of Decimators and Interpolators”, IEEE Transactions on Acoustics, Speech, and Signal Processing, Vol. ASSP-24, no. 4, August 1976.

[5] Suraj R. Gaikwad and Gopal S. Gawande, “Design and Implementation of Efficient FIR Filter Structures using Xilinx System Generator”, International Journal of scientific research and management, volume 2 issue 3 March 2014.

[6] University of Newcastle upon Tyne, “Multirate Signal Processing”, EEE305, EEE801 Part A.

[7] Ljiljana Milic, Tapio Saramaki and Robert Bregovic, “Multirate Filters: An Overview”, IEEE Journal, 1-4244-0387, 2006.

[8] L. D. Milic and M.D. Lutovac, “Design multirate filtering”, Idea Group Publishing, pp. 105-142, 2002.

[9] S.K. Mitra, “Digital Signal Processing: A Computer based approach”, The McGrow-Hill Companies, 2005.

[10] Yonghao Wang and Joshua Reiss, “Time domain performance of decimation filter architectures for high resolution sigma delta analogue to digital conversion”, Audio Engineering Society ConventionPaper 8648 Presented at the 132nd Convention, April 2012.

[11] Kester, “Mixed-signal and DSP Design Techniques. Norwood, MA: Analog Devices, Ch.3, pp. 16-17. 2000.

[12] Ljiljana D. Milić, “Efficient Multirate Filtering”, Software & Systems Design, 2009.

[13] Damjanović, S., Milić, L. & Saramäki, T., “Frequency transformations in two-band wavelet IIR filter banks”, Proceedings of the IEEE Region 8 International Conference on“Computer as a Tool”, EUROCON 2005.

[14] Hogenauer, E. “An economical class of digital filters for decimation and interpolation”, IEEE Transactions on Acoustics, Speech and Signal Processing,. Vol. 29, No. 2, pp. 155-162. 1981.

[15] N. J. Fliege, Multirate digital signal processing, New York: John Wiley & Sons, 1994.

[16] P.P. Vaidyanathan, Multirate systems and filter banks. Englewood Cliffs, NJ: Prentice Hall, 1993.

[17] A.I. Russel, "Efficient rational sampling rate alteration using IIR filters," IEEE Signal processing Letters, vol. 7, pp. 6-7, Jan. 2000.

[18] M. D. Lutovac, and L. D. Milic, "Approximate linear phase multiplierless IIR half-band filter," IEEE Signal Processing Letters, vol.7, pp. 52-53, March 2000.

[19] Matthew P. Donadio, “CIC Filter Introduction”, m.p.donadio@ieee.org ,18 July 2000.

[20] Fredric J. Harris and Michael Rice, “Multirate Digital Filters for Symbol Timing Synchronization in Software Defined Radios”, IEEE Journal vol. 19, no. 12, December 2001