iJRASET
International Journal For Research in
Applied Science and Engineering Technology

INTERNATIONAL JOURNAL FOR RESEARCH

IN APPLIED SCIENCE & ENGINEERING TECHNOLOGY

Volume: 6 **Issue:** II **Month of publication:** February 2018

DOI: http://doi.org/10.22214/ijraset.2018.2025

International Journal for Research in Applied Science & Engineering Technology (IJRASET)
ISSN: 2321-9653; IC Value: 45.98; SJ Impact Factor: 6.887
Volume 6 Issue II, February 2018- Available at www.ijraset.com

A Fast method for Denoising the signal based on CIC filter

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Abstract: This paper presents use of cascaded integrated comb filter and is mainly used to denoise the signal easily .It keeps the continuous record of the filters and noise can be removed from the signal and then performing the cascaded process on the signal. Recently non uniform filter bank technique used. The input signal can be reconstructed same as the output signal .The main drawback in this method is there is no difference between the input and output signal and also there is no possibility to calculate any metrics. So, to overcome the above problem we prefer the cascaded integrated comb filter (CIC) .Acquiring the input signal and applying CIC in order to denoise the signal. ,In this CIC the process carried out in seven phases as the output of first phase connected to the input of second phase and the process continues up to seven phases then we obtain the denoised signal. The metric calculation can also be done using MAE (mean absolute error) In the extension, we use cascaded integrated comb filter and is mainly used to denoise the signal easily .It keeps the continuous record of the filters and noise can be removed from the signal and then performing the cascaded process on the signal.
Keywords: Cascaded Integrated Comb Filter (CIC),Mean Absolute Error(MAE), Denoised Signal, Cascaded process.

I. INTRODUCTION

As 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. These functions require two basic signal processing procedures: decimation and interpolation. And while digital hardware is becoming faster, there is still the need for efficient solutions. Techniques found in work very well in practice, but large rate changes require very narrow band filters. Large rate changes require fast multipliers and very long filters. This can end up being the largest bottleneck in a DSP system. In an efficient way of performing decimation and interpolation was introduced. 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, or CIC filters for short. This paper summarizes the findings. An extension of CIC filters has been published, and is briefly mentioned here. When in doubt, the reader should refer to these sources

A. Building Blocks

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

$$y[n] = y[n - 1] + x[n] \quad (1)$$

This system is also known as an accumulator. The transfer function for an integrator on the z-plane is

$$HI (z) = 1\ 1 - z -1 \quad (2)$$

When we build a CIC filter, we cascade, or chain output to input, N integrator sections together with N comb sections. This filter would be fine, but we can simplify it by combining it with the rate changer. Using a technique for multi rate analysis of LTI systems from [CR83], we can "push" the comb sections through the rate changer, and have them become $y[n] = x[n] - x[n - M]$ 2 at the slower sampling rate fs R . We accomplish three things here. First, we have slowed down half of the filter and therefore increased efficiency. Second, we have reduced the number of delay elements needed in the comb sections. Third, and most important, the integrator and comb structure are now independent of the rate change. This means we can design a CIC filter with a programmable rate change and keep the same filtering structure.

Because of the pass band droop, and therefore narrow usable pass band, many CIC designs utilize an additional FIR filter at the low sampling rate. This filter will equalize the pass band droop and perform a low rate change, usually by a factor of two to eight. In many CIC designs, the rate change R is programmable. Since the bit growth is a function of the rate change, the filter must be designed to handle both the largest and smallest rate changes. The largest rate change will dictate the total bit width of the stages, and the smallest rate change will determine how many bits need to be kept in the final stage. In many designs, the output stage is followed by a shift register that selects the proper bits for transfer to the final output register. A system designer can use the equation for Bout for a decimator for an interpolator to calculate proper shift values. This lies in the interval.

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 pass band and stop band Since their inception, CIC filters have become an important building block for DSP systems. They have found a particular niche in digital transmitters and receivers. They are currently used in highly integrated chips from Intersil, Gray chip, Analog Devices, as well as other manufacturers and custom designs. This paper has attempted to summarize key points found and provide some insight into designs. While many journal submissions are of limited value to an engineer, this paper was written for designers. As such, the reader should try to locate as the definitive reference for CIC filters.

II. EXISTING METHOD

The non-uniform filter bank with integer decimation and linear phase is realized with the help of tree structured techniques, which is based on building the filter bank using a two-channel filter bank as basic building blocks. The generalized structure of M channel filter banks based on tree structure approach is depicte. For M-channel NUFB having decimation M0, M1, M2, . . ., MM−1 ,for each band, then decimation factors are such that [1,2]$\sum_{k=0}^{m} \frac{1}{m_k}=1$

Total number of two-channel systems required is 2P−1. On the other side, the process of reconstructing the original input signal can be seen as mirror image of the decomposition process at analysis side as and its equivalent parallel structure . After resolving the tree structured non uniform filter bank into its parallel forms, the following relations can be deduced.

The Interpolation Identity of together with the equivalence of enables us to convert any CTFB to an equivalent DTFB preceded by Nyquist rate sampling and followed by low pass filtering, and vice versa. Thus, any perfect reconstruction (PR) filter bank (i.e. a DT analysis-synthesis filter bank for which the input and output are equal) can be converted to a CTFB, which can then be interpreted in terms of sampling and reconstruction. As an example consider the PR filter bank of. The theory of PR filter banks is well established and closed form solutions for the synthesis filters given the analysis filters are known. We can convert the analysis part of the filter bank to a sampling strategy by applying the equivalence . This results in the sampling strategy depicted where the signal is filtered by three CT filters with frequency responses , , and the outputs are sampled at the corresponding rates. The reconstruction is obtained by applying the Interpolation Identity to the synthesis part of the filter bank followed by impulse modulation and low-pass filtering, resulting in the reconstruction depicted The sampling procedure together with the reconstruction and constitute a generalization to Papoulis' well known generalized sampling expansion Papoulis showed that a band limited signal is uniquely determined by the samples of the responses of LTI systems with input , sampled at one of the Nyquist rate. By converting a PR filter bank with unequal decimation factors to a sampling and reconstruction scheme, we allow for different sampling rates of the filters outputs, thus generalizing Papoulis' theorem. Papoulis does not derive necessary and sufficient conditions on the filters such that the signal can be reconstructed from the generalized samples. However, such conditions can be derived to convert the CT filters to DT filters comprising a DT filter bank. Given the analysis filters of a DTFB, we can determine

III. PROPOSED METHOD

A. Enhancement Work- Cascaded INTEGRATED Comb filter

In digital signal processing, a cascaded integrator–comb (CIC) is an optimized class of finite impulse response (FIR) filter combined with an interpolator or decimator. A CIC filter consists of one or more integrator and comb filter pairs. In the case of a decimating CIC, the input signal is fed through one or more cascaded integrators, then a down-sampler, followed by one or more comb sections (equal in number to the number of integrators). An interpolating CIC is simply the reverse of this architecture, with the down-sampler replaced with a zero-stuffer (up-sampler). CIC filters were invented by Eugene B. Hogenauer, and are a class of FIR filters used in multi-rate digital signal processing. The CIC filter finds applications in interpolation and decimation. Unlike most FIR filters, it has a decimator or interpolator built into the architecture. The figure at the right shows the Hogenauer architecture for a CIC interpolator.

B. Characteristics of CIC Filters

1) Linear phase response
2) Utilize only delay and addition and subtraction that is, it requires no multiplication operations

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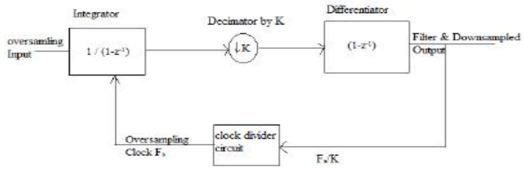
Cascaded integrator-comb (CIC) digital filters are computationally efficient implementations of narrowband low pass filters and are often embedded in hardware implementations of decimation and interpolation in modern communications systems. CIC filters were introduced to the signal-processing community, by Eugene Hogenauer, more than two decades ago, but their application possibilities have grown in recent years. Improvements in chip technology, the increased use of polyphase filtering techniques, advances in delta-sigma converter implementations, and the significant growth in wireless communications have all spurred much interest in CIC filters. While the behavior and implementation of these filters isn't complicated, their coverage has been scarce in the literature of embedded systems. The basic block for cascaded integrated comb filter is as follows



The impulse response of CIC is given as $h(n) = \frac{1}{m}; 0 \leq n \leq m-1$

The decimation factor for CIC is given as $h(z) = \frac{1}{m}\sum_{j=0}^{m-1} z^{-j}$

Which can be modified to obtained transfer function of CIC

$$H(z) = \frac{1}{m}\frac{(1-z^{-m})}{1-z^{-1}}$$

The transfer function of the resulting decimation filter, also known as a RRS (recursive running sum) or comb filter is given by

$$H_{comb}(z) = \left[\frac{1}{m}\left(\frac{1-z^{-m}}{1-z^{-1}}\right)\right]$$

Where *M* is the decimation factor, and *K* is the number of the stages. The transfer function in will be also referred to as the comb filter. The integrator section works at the higher input data rate thereby resulting in higher chip area and higher power dissipation.

$$H(z) = \left[\frac{1}{m}\right][1 + z^{-1} + z^{-2} + \cdots z^{-(m-1)}]$$

The magnitude characteristic of the comb decimator must satisfy two requirements:
To have a low droop in the frequency band defined by the pass band frequency in
Order to preserve the signal after decimation.
To have a high attenuations in so called folding bands, i. e. the bands around of the
Zeros of the comb filter.

$$\frac{2\pi i}{m} - \omega_p ; \frac{2\pi i}{m} + \omega_p ;$$

C. Input signal

Here considering input as sinusoidal signal and total processing is based on that signal.

D. Filtering the signal using Filter bank

In signal processing, a filter bank is an array of band-pass filters that separates the input signal into multiple components, each one carrying a single frequency sub-band of the original signal. One application of a filter bank is a graphic equalizer, which can attenuate the components differently and recombine them into a modified version of the original signal. The process of decomposition performed by the filter bank is called analysis (meaning analysis of the signal in terms of its components in each sub-band) the output of analysis is referred to as a sub bands signal with as many sub bands as there are filters in the filter bank. The reconstruction process is called synthesis, meaning reconstitution of a complete signal resulting from the filtering process.

E. Cascaded Process

In cascaded process total the signal passes through seven stages. Here the output of first stage is taken as input to the second stage .likewise this process continues up to seven stages .then the efficient and enhanced signal obtained at the fifth and seventh stage.

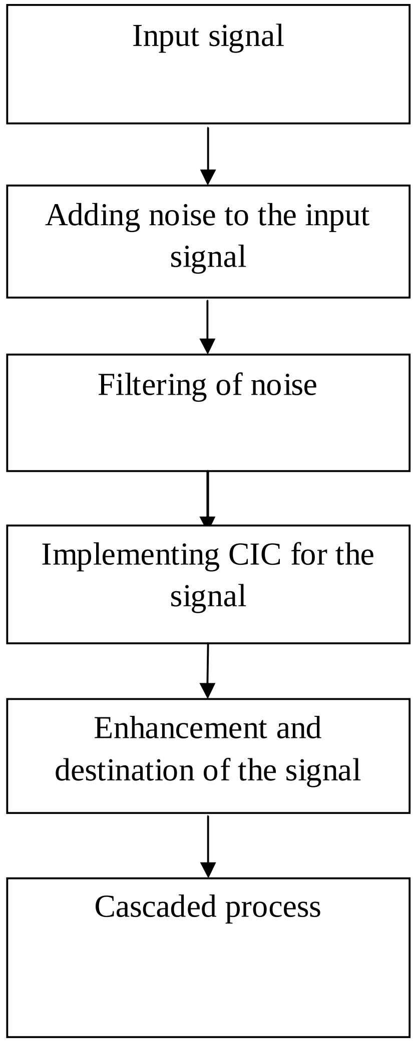


Figure : Block Diagram of Proposed Method

F. Filtering of Noise

Noise reduction is the process of removing noise from a signal. All recording devices, both analog and digital, have traits that make them susceptible to noise. Noise can be random or white noise with no coherence, or coherent noise introduced by the device's mechanism or processing algorithms.

The transfer function of filter is given as $H(z) = \frac{B(z)}{A(z)} = \frac{b_o + b_1 z^{-1} + \cdots . b_N Z^{-N}}{1 + a_1 z^{-1} + a_2 z^{-2} + \cdots a_m z^{-m}}$

Impulse response of a filter is given by$y_n = \sum_{k=0}^{N} h_k x_{n-k}$

In statistics, mean absolute error (MAE) is a measure of difference between two continuous variables. Assume X and Y are variables of paired observations that express the same phenomenon. Examples of Y versus X include comparisons of predicted versus observed, subsequent time versus initial time, and one technique of measurement versus an alternative technique of measurement.

The mean absolute error is given by $MAE = \frac{\sum_{i=1}^{n} |y_i - x_i|}{n} = \frac{\sum_{i=1}^{n} |e_i|}{n}$

Where mean error is given as $MAE = \frac{\sum_{i=1}^{n} |y_i - x_i|}{n}$

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IV. RESULTS

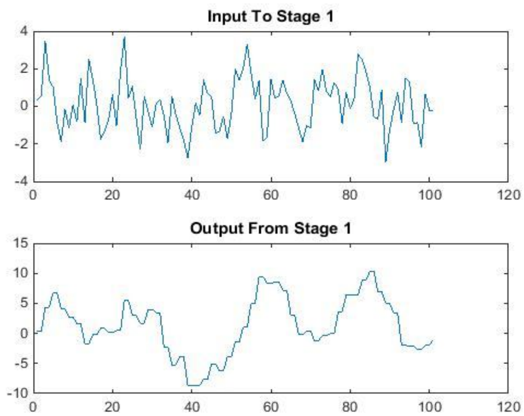

Figure 1 : Input and Output of Stage 1

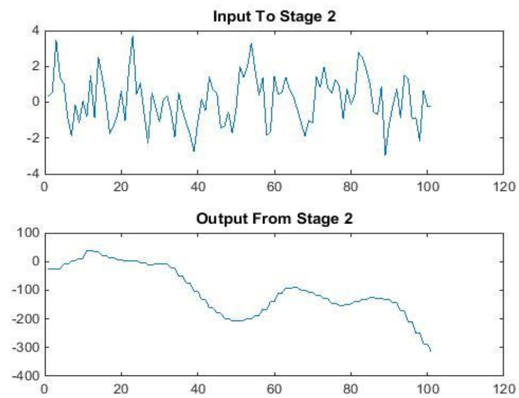

Figure 2 : Input and Output of Stage 2

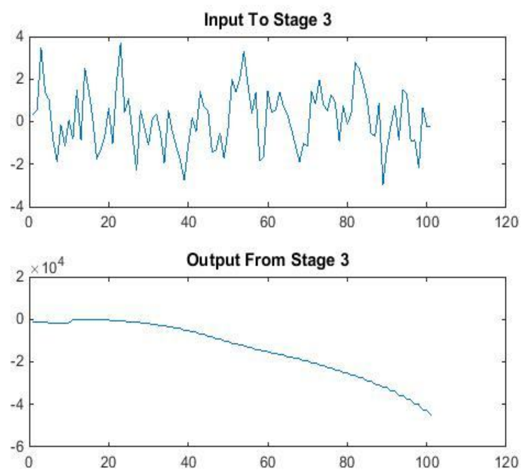

Figure 3: Input and Output of Stage 3

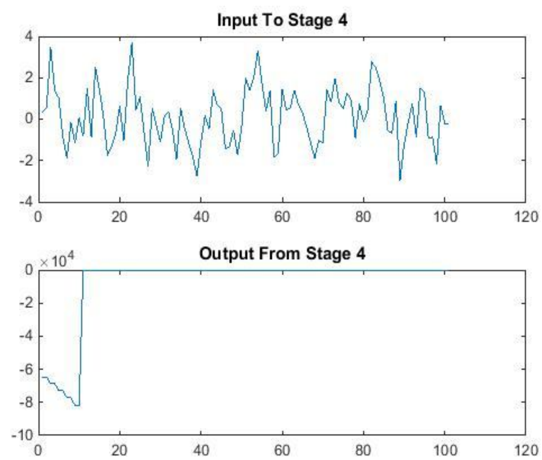

Figure 4 : Input and Output of Sta

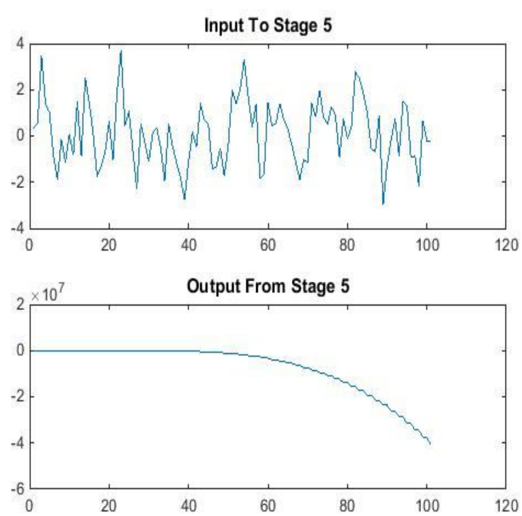

Figure 5 : Input and Output of Stage 5

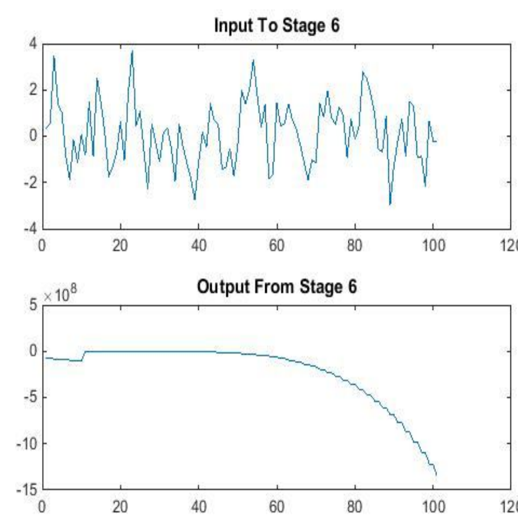

Figure 6 : Input and Output of Stage 6

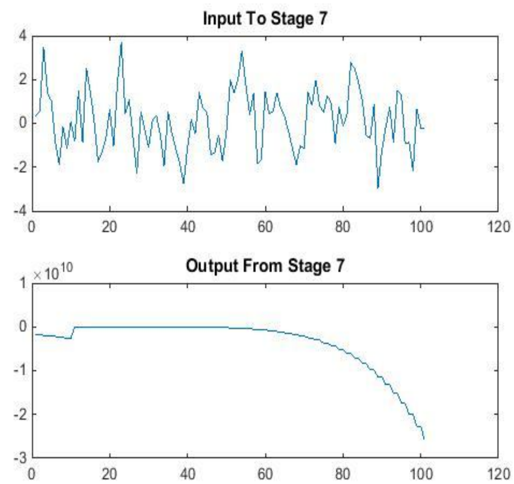

Figure 7 : Input and Output of Stage 7

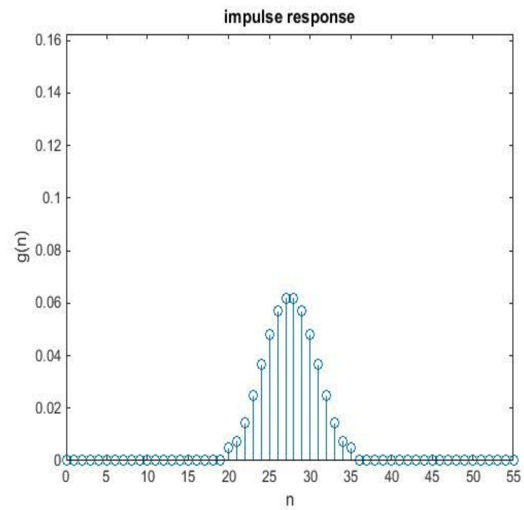

Figure 8 : Impulse Response

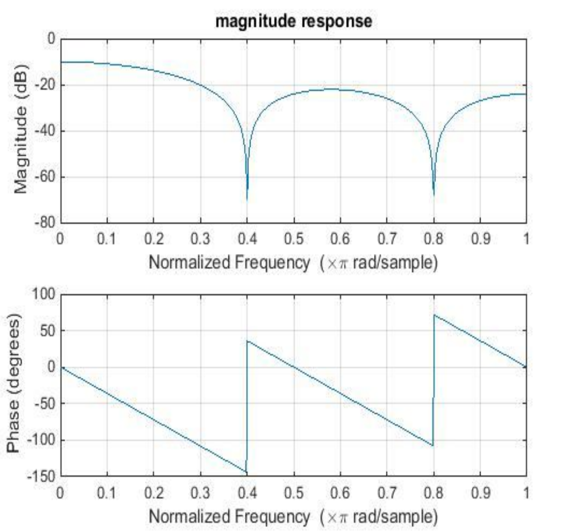

Figure 9 : Magnitude And Frequency Response

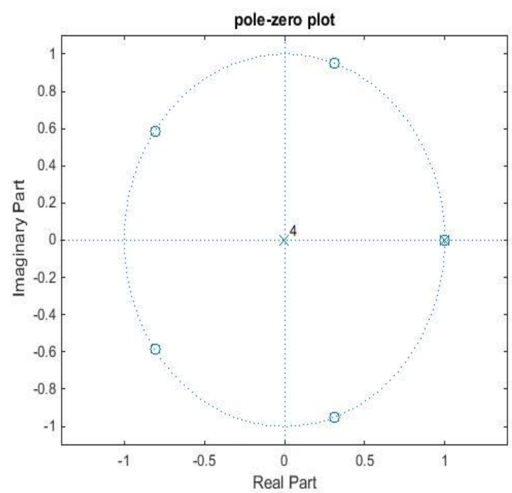

Figure 10 : Pole-Zero Plot

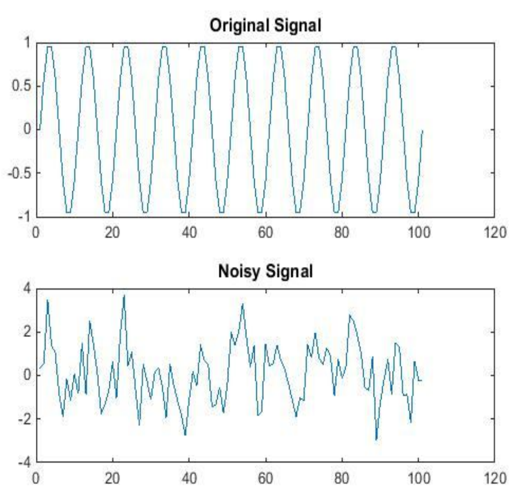

Figure 11 : original and noisy signal

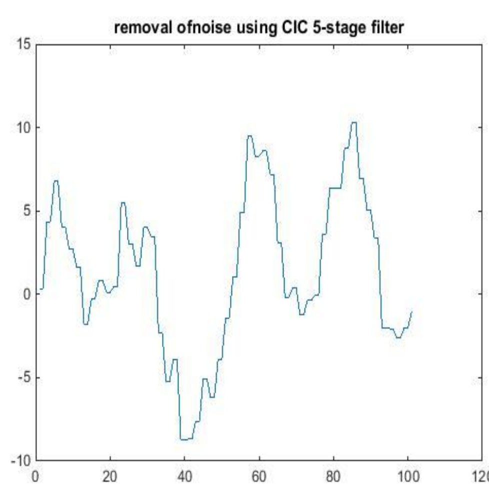

Figure 12 : removal of noise using CIC 5 stage filter

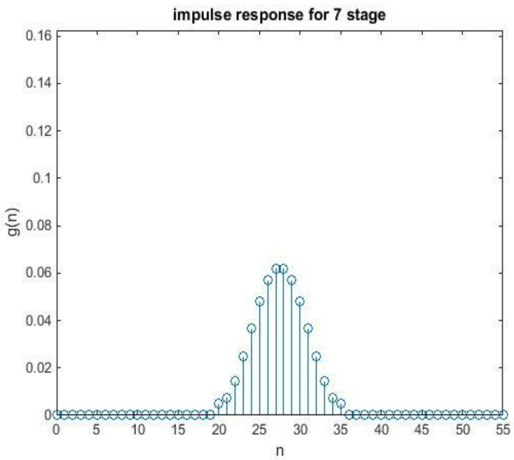

Figure 13 : Impulse Response for 7 stage

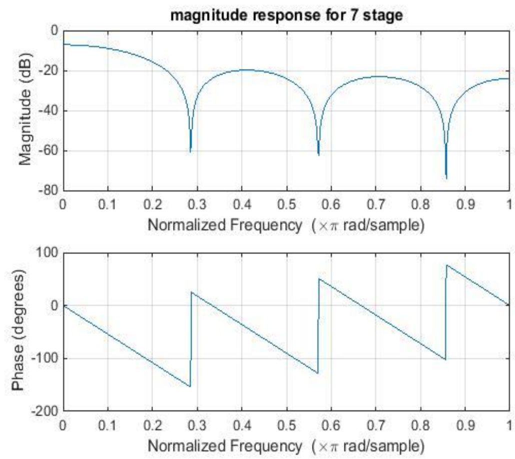

Figure 14 : magnitude and frequency response for 7 stage

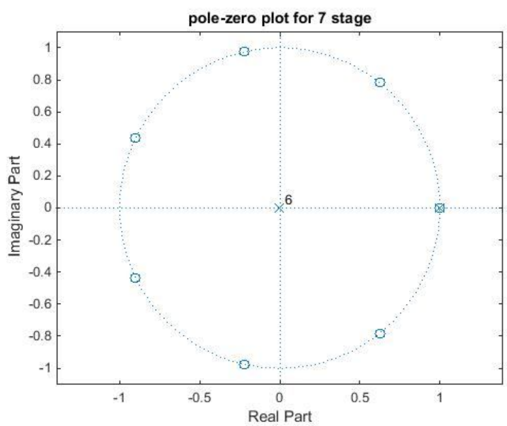

Figure 15 : Pole-zero Plot For 7 Stage

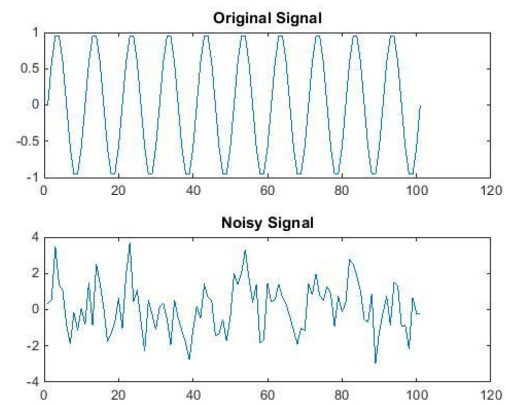

Figure 16 : Original and the Noisy Signal

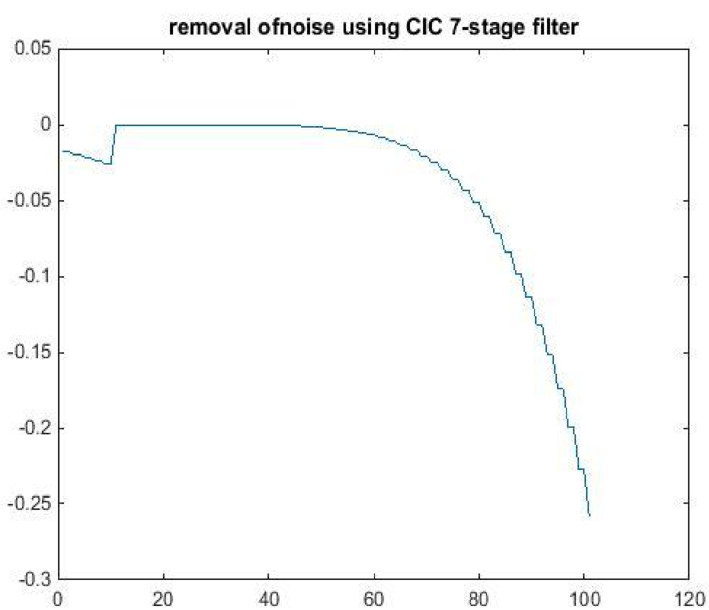

Figure 17 : Removal of noise using CIC 7 stage Filter

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Volume 6 Issue II, February 2018- Available at www.ijraset.com

VI. CONCLUSION

In cascaded integrated comb filter(CIC) totally it contains 7 phases first level output is connected to second level input and the process continues up to seventh phase .in first level integrated part converted to complexion is divided into destination and differentiation .In CIC for the input signal the noise gets added to the signal ,again filtering the signal and performing the denoising on the signal in order to make the signal a noise free signal .The CIC filter gives better results and better performance than any other filters .It is the best method of denoising a signal.

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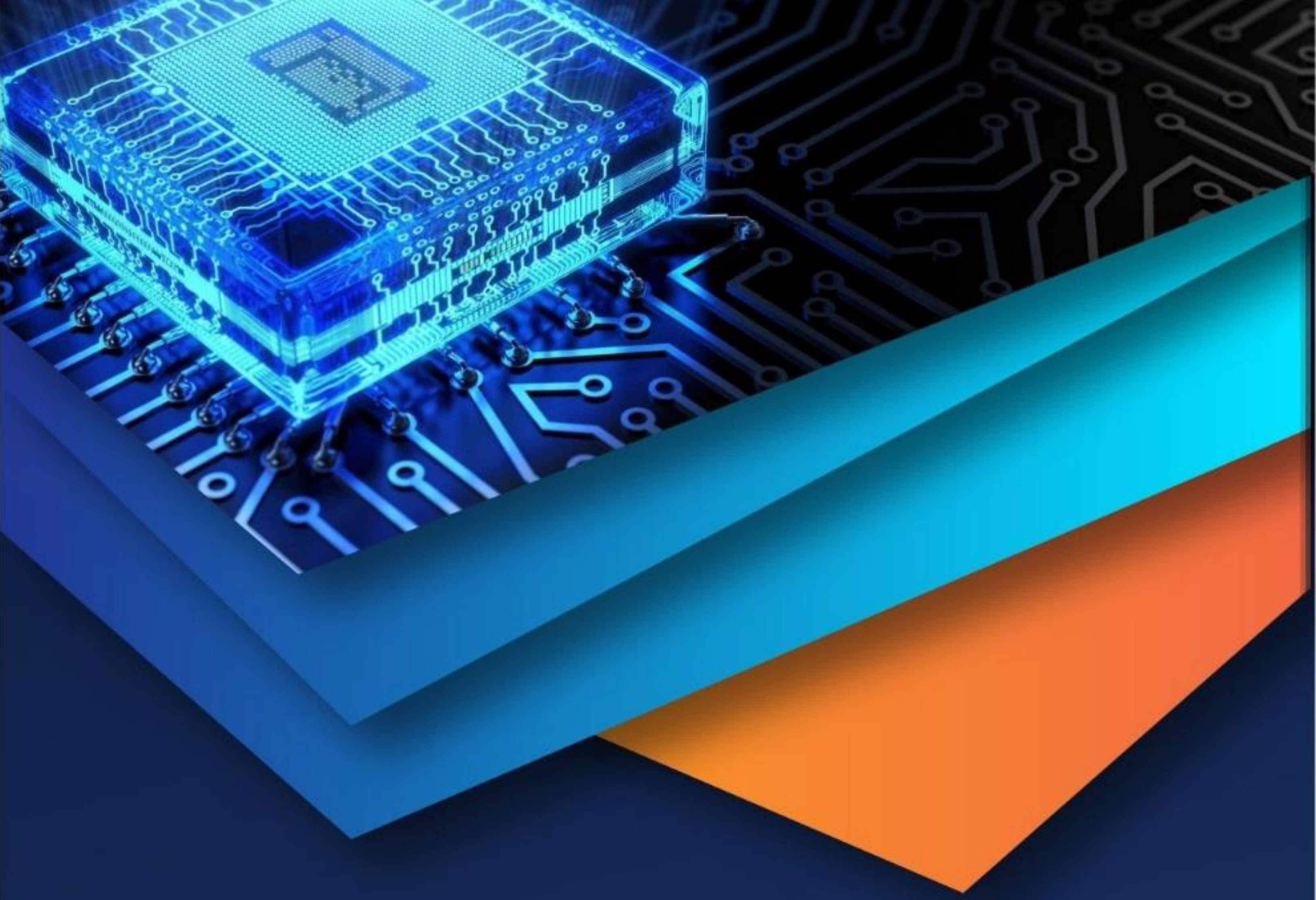

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