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Prediction of Round-off Noise Power Spectrum and Magnitude Response of Direct Form Second Order filter by Least Pth-norm IIR Filter Design Method

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Abstract: **In this research paper, we have designed least Pth – norm IIR filter by using arbitrary magnitude response type having density factor=18 and pth norm =120. The ability is to create filters that have arbitrary shape frequency response curves, and filters that meet performance constraints, such as band pass width and transition region width, is well beyond that of analog .Quantization is a natural outgrowth of digital filtering and digital signal processing development. hpsd = noise psd (H,L) is the relation that computes the power spectral density (PSD) at the output of the object or filter System objects H, occurring because of round off noise. This noise is produced by quantization errors within the filter. L is the number of trials used to compute the average. The PSD is computed from the average over the L trials. The more trials you specify, the better the estimate, but at the expense of longer computation time Unfortunately, many in the electronics field dare uncomfortable with the subject, whether due to a lack of a molarity with it, or a reluctance to grapple with the mathematics involved in a complex filter design This application note is intended to serve as a very basic introduction to some of the fundamental concepts and terms associated with filters. It will not turn a novice into a filter designer, but it can serve as a starting point for those wishing to learn more about filter design.**

***Keywords*- Arbitrary Magnitude Response, Density Factor, Filter Order, Filter approximation function.**

1. INTRODUCTION

Analog systems with memory may be further classified as lumped or distributed. The difference can be explained by considering the meaning of memory in a system. Future output of a system with memory depends on future input and a number of state variables, such as values of the input or output at various times in the past. If the number of state variables necessary to describe future output is finite, the system is lumped; if it is infinite, the system is distributed. Designers use filters because of following uses; some of them have been discussed below

(1) To separate signals that has been combined, such as the noise added during the recording process.
(2) To separate signals into their constituent frequencies.
(3) To demodulate signals
(4) To restore signals that has been degraded by some process, known or unknown.

Digital filters have some advantages like they offer greater flexibility and accuracy in comparison with analog filters. digital signal processing (DSP) depends in large measure on digital filtering to meet the needs of its users .if we look Analog filters are very cheaper, faster, and have greater dynamic range; digital filters outstrip their analog cousins in flexibility. Also, there is a growing need for fixed-point filters that meet power, cost, and size restrictions. When we will try to convert a filter from floating-point to fixed-point, then we use quantization to perform the conversion. When filter designers try to use digital filters in applications where power limitations and size constraints drove the filter design, they moved from double-precision, floating-point filters towards fixed-point filters. Whenever we have enough power to run a floating-point digital signal processor, such as on your desktop PC or in our car or any vehicle, we found fixed-point processing and filtering are unnecessary to use. But, when we needs to run filters in a cellular phone or any kind of a hearing aid for hours instead of seconds, fixed-point processing can be essential ensure long battery life and small size and long durability .also in order to offer tools for analyzing the effects of quantization on filter performance we need to study filters in detail. The input of the digital signal processor contains a finite-length time interval in which the true Gaussian signal is contaminated by Gaussian noise. We analytically derive the round off noise-to-signal ratio in the measurement of the signal power. We also present computer simulations which validate the analytical results. These results can be used in tradeoff studies of hardware design such as number of bits required at each processing stage.

2. LEAST-PTH NORM IIR FILTERS DESIGN

Similar to equi-ripple or least-square designs, we can weigh the optimization criteria to alter the design. However, unlike equiripple, we have the extra flexibility of providing different weights for each frequency point instead of for each frequency band this method is used in our research paper for designing IIR filter .In this type of IIR filter we use feedback, because of this reason we can say that this filter is treated as recursive filter. Many digital filters use both input values and previous output values from the filter to calculate the current output value.IIR filters can be implemented with feedback, although this is unusual. The numerator of the transfer function. When this expression falls to zero, the value of the transfer function is zero as well that is called as zero of the function and the denominator of the transfer function. When this expression goes to zero (division by zero), the value of the transfer function tends to infinity; called pole of the function or filter. In this kind of filter first we made analog filter and then we convert into discrete form to purpose of simplicity. IIR filters that design optimal solutions to yourfilter requirements. With these new filter functions; you can design filters to meet your specifications that you could not design using the IIR filter design functions in Signal Processing .Least-pth norm unconstrained optimization algorithm is to design IIR filters that have arbitrary shape magnitude response curves. For basic information about the least-pth algorithms used in the IIRfilter design functions, refer to Digital Filters. We may notice that the IIR design functions use the same syntax, input, and output arguments. Because the design functions use very similar algorithms, common input and output arguments apply. Different kinds of this kind of filters are possible like impulse invariance method, bilinear transformation method but we found least Pth norm method of IIR filter is suitable to use .Arguments are used in the same way, and carry the same defaults and restrictions. Roughly speaking, the optimal design is achieved by minimizing the error between the actual designed filter and an ideal filter in the Pth-norm sense. Different values of the norm result in different designs. When specifying the P-th norm, we actually specify two values, 'InitNorm' and 'Norm' where 'InitNorm' is the initial value of the norm used by the algorithm and 'Norm' is the final value for which the design is optimized. Starting the optimization with a smaller initial value aids in the convergence of the algorithm. Here we discuss filter design methodology with various steps .First step is to select the response and then we go to specification block and then we select the algorithm for particular method and later we design the filter and after the designing we will verify the result.

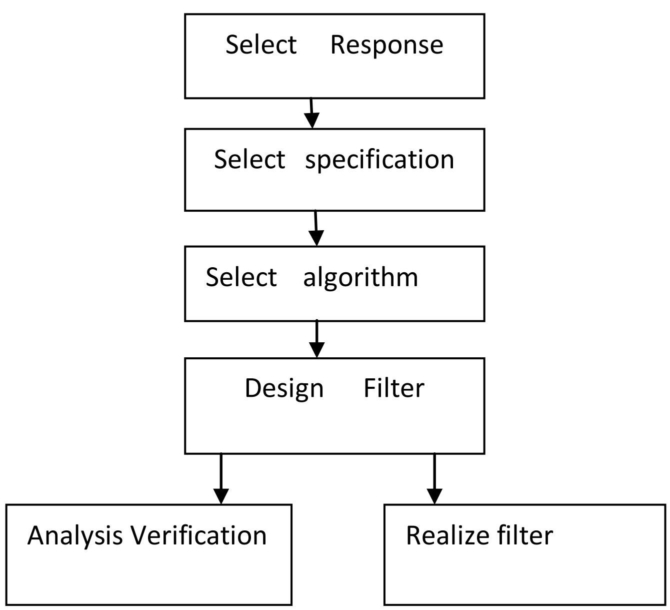

Figure1. Block diagram of Filter Design Methodology

Because of their feedback structure it is very easy for us to recognize filter designs. Now in order to know attributes we need to study all the filter parameters in detail. Many digital filters take advantages of input values and previous output values from the filter in order to calculate the current output value. In case of IIR filters, the transfer function is a ratio of polynomials: The numerator of the transfer function. When this expression falls to zero, the value of the transfer function falls to zero which is so called as zero of the function and denominator of the transfer function. When this expression goes to zero (division by zero), the value of the transfer function tends to become infinity; called a pole of the function or filter. The design functions found very useful to read similar algorithms, where we use common input and output arguments. Arguments are used in the same way, and carry the same defaults and restrictions. If the IIR filter's input suddenly becomes a sequence of all zeros, the filter's output falls to nonzero forever. Particular peculiar attributes of IIR filters comes into play while realization of filters like the feedback structure of their delay units, multipliers, and adders.

3. ANALYSIS OF FILTER DESIGN METHODOLOGY

This methodology differs from the traditional approaches since no prior knowledge of the noise statistics is mandatory, instead the noise signals are only assumed to have large amount of energy. Since the estimation criterion used for the design of this kind of filter is to reduce the worst possible amplification of the estimation error signal in terms of modeling errors and additive noise. This method is highly reliable and important in practical synthesis of signals. This paper presents a least Pth approach to the optimal design of IIR digital filter banks in the mini max sense for doing analysis and synthesis of signals of different kind. Here in this section we will be trying to put concentration on developing impulse response. For this kind of normalized low pass transfer function we use the pass band edge radian frequency 1 rad/sec. But this seems a rather unusual frequency, since seldom would a low pass filter be required to have such allow frequency. Impulse response is easy to recognize .Before we begin the development of the approximation functions for analog filters, it may be helpful to go over the general approach taken in these sections. In each case, the general characteristics of the approximation method will be discussed, including its relative advantages and disadvantages. Next, a description of the transfer function for each approximation will be given. Now in order to know the normalization transfer function we have to study basic parameters of filters also. There will be no attempt to give an exhaustive derivation of each approximation method in this text; there are more than enough sources of theoretical developments already available. The Frequency Response of filter for doing analysis is given in the equation below with

$$\begin{aligned} &H(W) = \exp(-jw\alpha), \left| w \le w_c \right| \\ &0 = otherwise \end{aligned} \quad \textbf{(1)}$$

$let x_i(n) = A_i \exp(jw_i n)$ Be a frequency component of signal then the corresponding output signal will be written in this form

$$Y_i(N) = A_i \exp(jw_i n) H(W_i) \quad \textbf{(2)}$$

If $\left[W_i\right] \le W_{C,}$,then above equation can be written in mathematical form below describe by third equation that is represented by form

$$Y_i(n) = A_i \exp\lfloor jwi(n-\alpha \rfloor \quad \textbf{(3)}$$

i.e. $x_i(n)$ is passed with constant delay, if $\left|w_i\right| \ge w_{c,} then$, $Y_i\left(n\right)=0$ i.e. $X_i(n)$ is stopped completely then filter has best frequency selectivity. The impulse response of filter will be obtained by equation below.

$$h(n) = \frac{\sin\left[w_c(n-\alpha)\right]}{\pi(n-\alpha)} \quad \textbf{(4)}$$

This design method is concerned with filter bank structure and a method for filtering and separating an information signal into different bands. Later desired signal is reconstructed from the independent components representing every band. Now direct form second order Pth norm IIR design method is shown in the diagram2 below. In figure 3 we use table that represents all the current filter information with different filter parameters and these are the parameter that we use in our research paper to get better results those found easy to understand and figure 4 is used to show round-off noise power spectrum which is the concerned part of of research .After getting waveform for round-off noise power spectrum we have also shown waveform for magnitude response of IIR least Pth norm filter. However, sequential processing, can be accommodated when delays at the beginning of a processing segment can be tolerated new algorithm for minimizing a sum of the pth power of nonlinear functions can be derived This algorithm requires neither derivative calculation nor linear search. However, filters do not exclusively act in the frequency domain; especially in the field of image processing many other targets for filtering exist. However, the technique actually allows the filter designer considerable latitude for designing this kind of filters for getting better results and in this research paper we found all the results those shows better accuracy also.

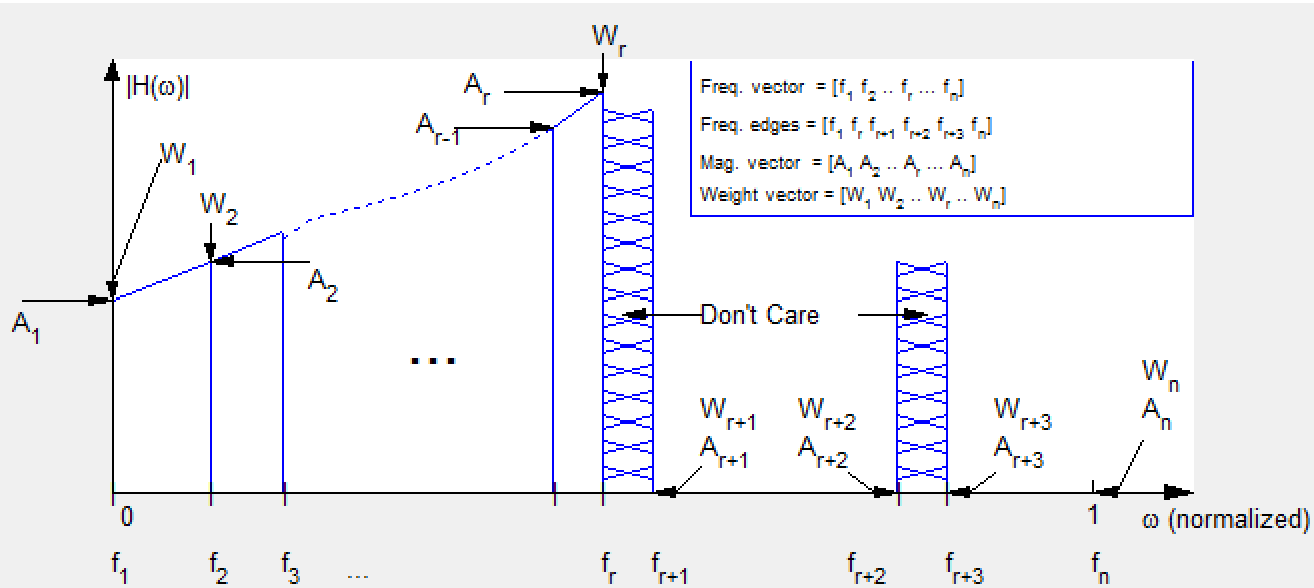

Figure2. Direct Form Second Order Section Filter by least Pth norm IIR Design Method

Table used for current filter information

S.No	Technical Specifications	Current Filter Information
1	Response Type	Arbitrary Method
2	Design Method	Least P_{TH} IIR Norm Method
3	Filter Order	Numerator Order =7, Denominator Order =7,
4	Density Factor	18
5	P_{TH} Norm	120
6	Frequency vector	0.005:0.175:0.02:0.215:0.251

Figure3.Design Filter Information

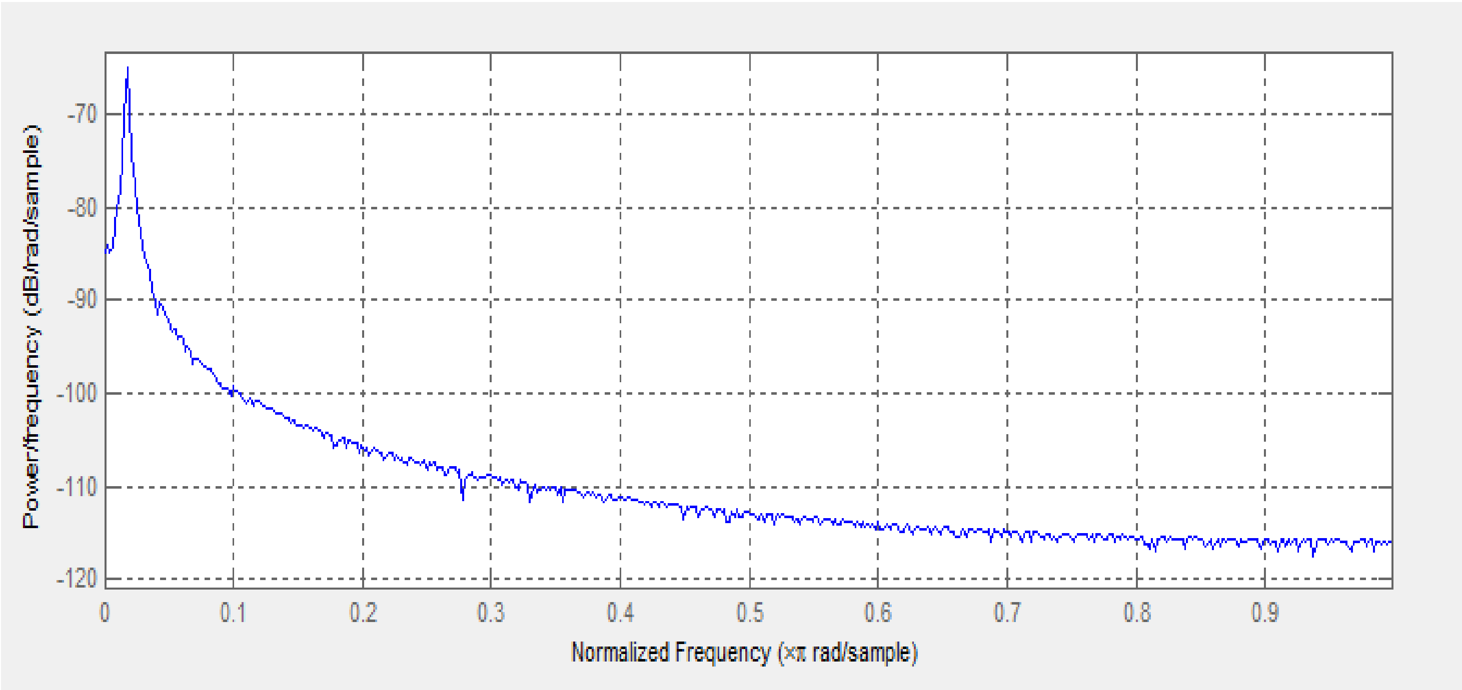

Figure4. Round off noise power spectrum

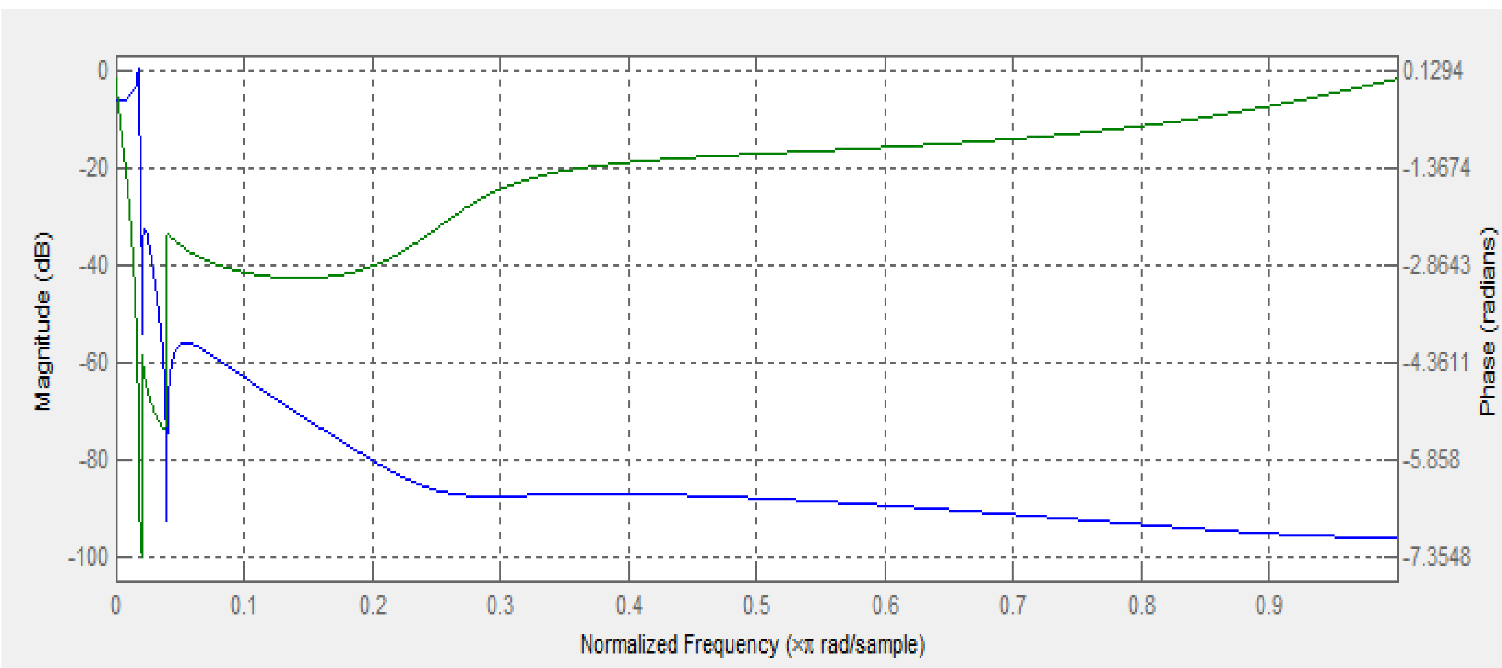

Figure5.Magnitude response

4. CONCLUSION

Here we come to notice that least Pth norm IIR filter design method provide better result than other types. We analyzed round off noise power spectrum for least pth norm IIR filter by using above discussed current filter information parameter like response type of filter is arbitrary magnitude and density factor=18 as shown in the table that we use in this research paper . This results in a smooth degradation of texture quality as distance from the viewer increases, rather than a series of sudden drops. Many digital filters use both input values and previous output values from the filter to calculate the current output value. Non-linear phase as well as for multi-objective designs. Next future of this research work can be extended for finding least squares error design orthogonally using Orthogonal Least Square algorithm to design linear and Non-linear digital filters. Filtering is a class of signal processing, the defining feature of filters being the complete or partial suppression of some aspect of the signal. Most often, this means removing some frequencies and not others in order to suppress interfering signals and reduce background noise.

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