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Minimize Group Delay of IIR Filter Using Complex Cepstrum

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Abstract—*The present paper proposes a new technique towards minimizing the group delay of IIR filter to get the desired magnitude response .For designing the IIR filter the approach used is complex cepstrum and all pass filter .The results in this paper has been achieved with the help of MATLAB and SIMULINK Filter design Toolbox.*

Keywords: *Complex Cepstrum,All pass Filter,IIR Filter,MATLAB and Simulink Filter design Toolbox.*

1. INTRODUCTION

DSP(Digital Signal Processing) is an area of science and engineering that leads day by day to an advance development in digital computer technology and fabrication of Integrated circuits. This rapid development in integrated circuits technology leads to the development of powerful, smaller, cheaper, faster digital computers, which have made it possible to construct and perform complex digital signal processing functions which were usually very difficult and expensive to be performed by analog circuitry. Hence many signal processing tasks are now performed by less expensive and reliable digital hardware which were conventionally performed by analog means. The digital hardware and associated software provides a greater degree of flexibility in system design, as hardware allows programmable operations and software modifies the signal processing functions to be performed by hardware. Signal processing concerns with representations, transformation, manipulation and extracting the information of signal. *Filters*-Filters are frequency selective circuits that allows a certain band of frequency to pass while attenuating the other frequencies .Filters are classified as Analog Filter and Digital Filter. The digital filter is one of the most powerful tools of DSP which consist of software and hardware. The input and output signals in the digital filter are digital or discrete time variant. The digital filters are further classified as FIR(Finite impulse response) and IIR(Infinite Impulse Response). FIR filters have a useful property that they exhibits linear phase shifts for all frequency but IIR filters have much better frequency response than FIR filter.IIR Filters have feedbacks and require less resources for computing than FIR.

Cepstrum- Cepstrum is defined as taking the inverse Fourier transform of logarithmic of spectrum of signal. There are many types of cepstrum available like complex cepstrum, real cepstrum ,power cepstrum, phase cepstrum. The present paper uses the approach of complex cepstrum which is defined as inverse Fourier transform of logarithmic of complex cepstrum. The complex cepstrum is used for reconstruction of signal as it contains both magnitude information and phase information.

Group Delay-Group Delay is a measurement of time or in other words it is that amount of time required for signal to propagate through device. In this paper the concerned is not only with the filter's delay but however also with each frequency component of the signal that experience the same delay so that there phase or shape with each other is maintained.

2. PROBLEM FORMULATION

In this paper the consideration is mainly on the frequency response error specially on group delay response of IIR Filter case. The overall delay remains an important consideration in practical applications, thus to minimize the group delay and get the desired frequency response is the main objective of this paper and exploring the results upto approximation of ideal response.

Results formulation

1.The magnitude, phase response, group delay formulas and assumptions have been derived from [1].

The magnitude as

$$log|H(e^{jw})| = \begin{matrix} \widehat{he[0]} + \\ 2\sum_{k=1}^{N} \widehat{he[k]} \end{matrix} cos(wk)$$

phase response as

$$\angle H(w) = -2\sum_{k=1}^{N} \widehat{h_o[k]}\, sin(wk)$$

Group delay response as

$$\angle H(w) = -\frac{d\angle H(w)}{dw} = 2\sum_{k=1}^{N} k.\widehat{h_o[k]}\, cos(wk)$$

*2.Calculating the coefficients of complex cepstrum:*With the help of MATLAB and SIMULINK Signal processing toolbox the coefficients along with there responses can be calculated as :

1.Determine the order of filter and delay d.

2.By taking the fast Fourier transform(fft) of h(n) where h(n) in the impulse response of filter and shifting in fft,by using the freqz commands of Matlab we plot the frequency response of complex cepstrum.

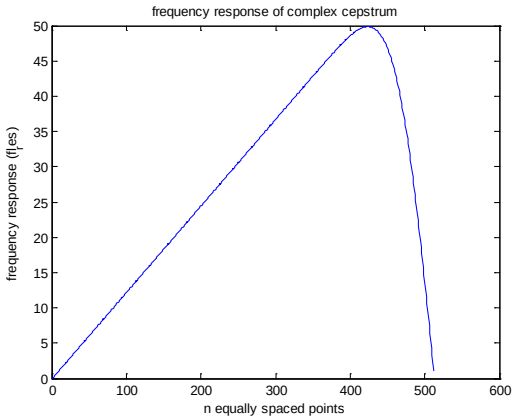

Fig. 1

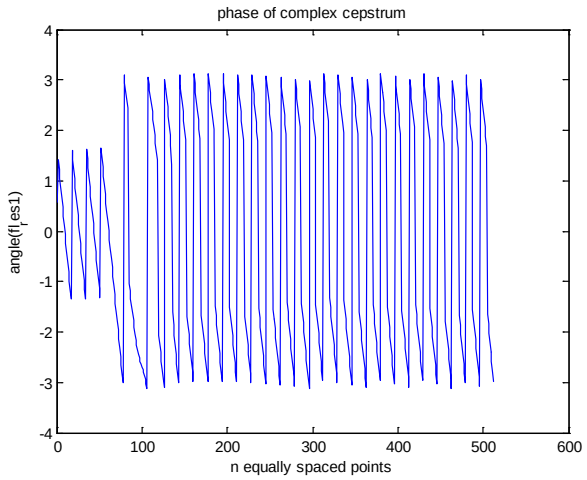

Fig. 2

3. Farrow structure

In the synthesized structure one main problem arises is the involvement of large phase delay. Although as the subsystem M increases, the phase delay also increases which also leads increment in group delay as in turn it will reinforces the flatness of frequency response in useful band but the practical applications demands smaller phase delay .Thus farrow structure came into light by modifying the structure into one with smaller phase delay.

With the help of Matlab and Simulink Signal processing toolbox the delay of IIR filter can be calculated:

1.Determine the N(order of filter); Rp(passband ripple); Rs(stopband ripple);Wp(passband frequency)

2.Using the elliptic filter command calculation of the zeros,poles,gain.[z,p,k] and converting them to sos(second order matrix) form,the desired group delay is calculated as Gd=max(g)-g.

3.using the allpass filter the delay has been minimized .

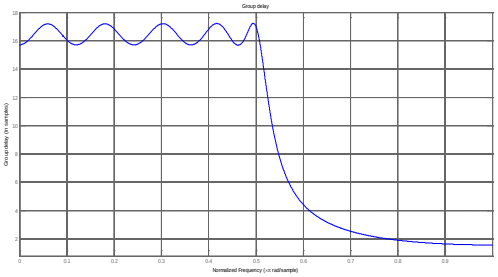

Fig. 3

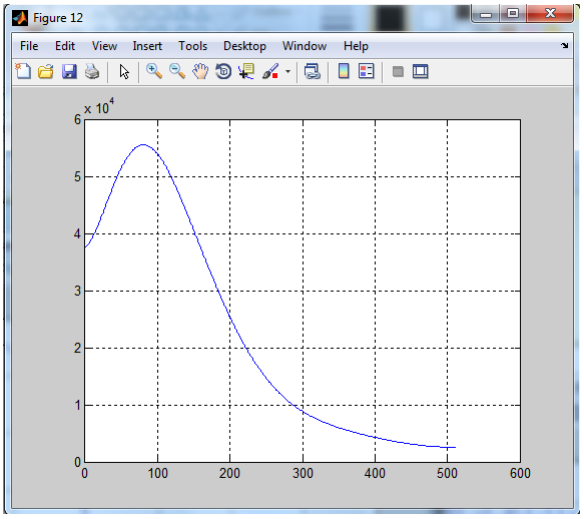

Fig. 4: Group delay

Results :-The employment of IIR all pass filters brings the facility that there is no need to advert the constraints on magnitude since It is in unity everywhere in the frequency domain. Furthermore, since the denominator's coefficients treated as a minimum-phase sequence by making its complex cepstrum a causal sequence, the resultant IIR filter is always stable.

3. CONCLUSIONS

A cepstrum-based approach is proposed to design infinite-impulse-response (IIR) group delay filters. Under a fixed filter order, the set of normalized complex cepstra needs to be computed once and stored, and the specific set for an arbitrary group delay is obtained by simply multiplying the stored set with the delay value. The closed-form solution of the complex cepstrum is directly proportional to the value of group delay. By using the allpass filter the delay has been minimized upto the approximation value.which facilitates updating the filter to match the desired delay. The proposed IIR designs are ensured to be stable since we can make the denominator in the transfer function to have a minimum phase by arranging a causal complex cepstrum for it.

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