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IIR Filter Design Techniques – A Critical Review

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ABSTRACT: The neural network draws the inspiration from the existing human nervous system. The neural network has been trained to perform complex functions in various fields of application including pattern recognition, identification, classification speed vision and control system. In recent years, the interconnections of all pass filters have found numerous applications in many practical filtering problems such as low-sensitivity filter structures, complementary filter banks, multi-rate filtering. In this paper,we have thoroughly investigated research for designing the IIR filters with the cooperation of a different algorithm. Various properties and application are also introduced in this research work.

KEYWORDS: Neural network ; IIR filter ; Control system ; Complex function.

I. INTRODUCTION

Digital filters are capable of performing various operations that the world, at best, is highly difficult to pursue with analogue filters. Filters are used to eliminate the unwanted frequencies from applied signal which has both wanted and noise signals. The types of digital filters are IIR filter and FIR filter. In designing the IIR filter the main hindrance is to set the lowest order for the purpose of the magnitude response and linear phase response. Digital all-pass filters have received much attention in many signal applications such as digital communications, notch filtering, phase equalisation, multi-rate filtering system, construction of a wavelet, image processing.

II. LITERATURE REVIEW

Goran Stancic and Sasa Nikolic have designed linear phase IIR notch filter. The digital notch filter remove non-desired frequency which is suitable for the signal. The notch bandwidth is very small because it requires less computation to design IIR notch filter. There are several methods to design filter by direct transforming analog notch filter into digital notch filter, pole –zero placement in the Z-plane and the third approach is notch filter using all-pass filters. The notch filter is designed by using the parallel connection. In this method, the authors have used two IIR filters which are connected in parallel. At both the branches, they contributed phase jump of 2π radian at different frequencies with notch frequency. Phase characteristic of the filter is equiripple and nearly equal to ideal linear phase response. The notch filter magnitude and phase difference are dependent on each other. By using this condition, the authors have calculated maximally allowed phase error for given maximal attenuation. When parallel branches are used than in the worst case the maximal error is twice higher than known method and delay line is used in the first branch. They can say that in order to control maximum loss in a given range then select next values for maximal phase errors.[1]

Yue-Dar Jour et al. have used neural network-based architecture for the weighted least squares to design IIR all-pass filter. The difference between the desired phase response and phase of the design all pass filter is to reflect the error and formulated as a Lyapunov error criterion. The neural network achieves convergence to get the filter parameter by using the corresponding dynamic function. Further,weighted updating algorithm is introduced to achieve good performance with the minimum and maximum solution. Weighted least square method is more used in research field because of the flexible use of the method to design any type of filter.[2]

Peng-Hua wang et al. have designed the maximally flat all pass fractional delay filter and fractional Hilbert transformers by using cepstrum based approach. The cepstral coefficient is used to get the maximal fatness condition in the phase delay response. It is found that these coefficient are controlled by design parameter because of this simple

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non-complex methodology can be developed for the coefficient controlling phase shift delay. There is only need of one set and another group can be obtained by simple mathematical operation on the previous matrix of the coefficient.[3]

S.C. Chan et al. have designed a causal-stable digital all-pass filter with the help of minimum & maximum design criteria by using second-order cone programming. SeDuMi toolbox is used to solving the problem of second-order cone programming. The total filter design issue can be explained by series of linear programming subproblem and bisection search algorithm. They have used another method which is imposing stability constraints and they have focused on nonlinear constraints such as pole radius constraints of the filter which can be solved by using Rouche's theorem. Pole radius constraints allow an excess trade off between estimate error and stability margin. When pole radius is satisfied then required solution is obtained for a given ripple.[4]

Valeri M. Mladenov and Nikos E. Mastorakis have designed a 2-D filter to decrease the constraints minimization problem and the solution is obtained by the convergence of an exact neural network. There are two type of algorithm to design 2-D filters which are based on the exact transformation of 1-D and other is based on proper optimisation technique. The stability parameter is important for their practical implementation. The optimisation algorithm is used in trial-and-error approach and there is not guarantee of the stability of the filter. By solving this problem they used the continuous-time neural network for optimisation procedure.[5]

Xi Zhang et al. have assumed the complex Chebyshev approximation problem of IIR filter and introduced a new method to reach the magnitude and phase response in the complex Chebyshev sense. By using Remez multiple exchange algorithms,they solved the generalised eigenvalue problem. The filter coefficient is obtained by solving eigenvalue to find the exact minimum eigenvalue,then Chebyshev approximation is reached through iteration starting from the initial value. This method is efficient because it retains the speed inherent in the Remez exchange algorithm.[6]

Hiroshi Iwakura and Hiroshi Iwakura have designed all pass filter with a specified phase response in the Chebyshev sense, but it is not in use because there is no solution in simplest form. There are also another methods such as minimum p-error criteria,approximation generalised exchange method ,linear programming algorithm and weighted least square algorithm. This approach is not easy to use for design IIR all-pass filter.So they have designed a new method based on eigenvalue problem for IIR digital all-pass filter with an equiripple phase response by using Remez exchange algorithm. The solution of rational interpolation problem can easily get by solving the eigenvalue problem. This method is already proposed by Werner. There is more than one eigenvalue in the eigenvalue problem but they have found one eigenvalue which gives the accurate solution of the rational interpolation problem. But Werner did not give this rule for eigenvalue problem. So new selection rule is introduced in which rational interpolation applied only when the real maximum eigenvalue is selected. When they find the equiripple phase response then they used iteration procedure for finding the coefficient filter. This method is used to retain the speed inherent in the Remez exchange algorithm, but also simplified the interpolation step because the calculation is reduced of the real maximum eigenvalue.[7]

Sunder S. kidambi has studied that the difference between the desired phase response and practical phase response in a quadratic equation is formulated which reflected the weighted error. They have solved the system linear equation which includes teoplitz & Hankel matrix to find the coefficient filter & minimise the error. They have introduced a method to design all pass filter that has the least square or an equiripple phase error response . By using frequency dependent weighted function, they have to design all pass filter which included the least square or equiripple phase error variation . In the system of linear equation of the error, the result can be minimization by involving the teoplitz & Hankel matrix . The all-pass filter has many applications which involve notch filtering, complementary filter banks, multi-rate filtering and group delay equalisation.[8]

Ju-Hong Lee and Charng-Kann Che have designed all pass filters.FIR and IIR technique is intermittently used. The Weighted least squares is a systematic technique for adjusting the least square error weighted function in frequency domain. Weighted least squares (WLS) algorithm is used to resolve the problem.further, the solution is formulated using reducing it to the lowest possible values. The minimax algorithm is used to arrive at the final results. The salient feature of the algorithm is to find out Chebyshev approximation for a function under consideration in linear subspace and the nonlinear minimax algorithm can also solve by linear Chebyshev approximation. The authors have demonstrated the effective use of WLS technique with the help of different illustrations.[9]

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Markus Lang and Timo I. Laakso study the least square error criterion and they have approximate the design of all pass filter for a given phase function. Authors have assumed that the desired phase response is fixed at the discrete set of frequency point.They have used a weighted function to formulated the least square equation error solution.This method is used for both high pass and band pass equalisation. With bandpass filters, the results can be further enhanced by introducing an adjustable phase offset in the design equations. By introducing the adjustable phase offset in the optimisation to reduce the total phase error. In addition to the iterative weighting of the equation error,the small value of the desired group delay was also adjusted iteratively such that the total phase error measure in equaliser application can be minimised. This new feature essentially eliminates the difficult choice of the nominal group delay in the equaliser design which is known to have a significant effect not only on the total approximation error but also on the stability of the designed all pass filter. The stability of the all pass equaliser is not guaranteed ,but in numerous experiments,a stable all pass filter could be designed except for very narrowband IIR filters.[10]
Truong Q. Nguyen et al. have designed all pass filter which is based on an eigenfilter algorithm for a given phase response. This approach is based on a formula which is in a quadratic form and obtains the desired filter as an eigenvector which is real,symmetric,positive definite matrix. In the case of phase estimation,quadratic form error is not present because of the non-linear trigonometric function are involved. However,estimate least square phase error solutions can be designed which gives an eigenfilter formula.The main reason for the use of estimate linear phase IIR filter is that their phase response is quite linear. Which is compared with the nonlinear phase response of an elliptic filter , overall calculation, and filtering delay is less then FIR filter.[11]

III. CONCLUSION

In this paper, we discussed all the technique that help to design of the IIR filter. When they design the filter they should satisfy the properties so that they gives a better result. The mathematical equation of the system with proper assumptions can increase the accuracy of the algorithm. We also discussed the multi-objective problem and find that the simplest way to use multi-objective problem is to convert the multi-objective problem into the single objective problem by providing the weight to every objective according to its importance. Given the immense range of naturally occurring filters in the domain of music, it is reasonable to expect that filter theory will continue to provide valuable tools for the analysis, synthesis, and manipulation of sound.

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