International Research Journal of Engineering and Technology (IRJET) e-ISSN: 2395-0056
Volume: 08 Issue: 02 | Feb 2021 www.irjet.net p-ISSN: 2395-0072

DESIGN OF OPTIMIZING DIGITAL FILTERS AND ITS APPLICATIONS USING ALGORITHMS

Hari Prasanna A[1], Suresh T P[2]

[1]*PG Scholar, Applied Electronics, Department of ECE, PMC Tech, Hosur, Tamil Nadu, India*
[2]*Assistant Professor, Department of ECE, PMC Tech, Hosur, Tamil Nadu, India*

---***---

Abstract: *In this project analyzing of Voiced, Unvoiced and silence regions of speech from their time domain and frequency domain representation. The deconvolution Cepstral homomorphism processing is frequently used. It relates to the flow of air that comes from lungs during speech generation. For unvoiced sounds like in "s" as well as "f", vocal coeds are relaxed, and the glottis is opened. For voiced sounds like, "a", "e". The vocal cord actually vibrates which in turn gives the pitch value. The shaping of the spectrum is done by using filter. This helps in creating different sounds and relates towards vocal tract organs. A speech recognition system tries to reduce the influence on the source (the system must provide same "answer" to get a high pitch woman's voice and for to secure a low pitch male voice) and specify the filter. The problem in source e(n) along with filter impulse answer h(n) are convoluted. Mel-scale is used for auditory modeling. Two famous tests that generated the bark and Mel scale machines were taken from the literature and its outcomes are reported here to characterize human auditory system. The specific LPC is approximately such as sufficient statistics of haphazard samples throughout statics. The LPC coefficients are actually the very least squares estimators from the regression coefficient if the minimum variance linear estimators from the regression coefficients. In LP research the redundancy within the speech signal is used.*

Key Words: Optimization, nonlinear optimization, linear programming, digital signal processing, filter banks, digital filters, coefficient quantization, fractional delay filters, linear-phase recursive filters, multiplier-less design, lattice wave digital filters, VLSI implementation.

1.INTRODUCTION

During the last two decades, the role of digital signal processing (DSP) has changed drastically. Twenty years ago, DSP was mainly a branch of applied mathematics. At that time, the scientists were aware of how to replace continuous-time signal processing algorithms by their discrete-time counterparts providing many attractive properties. These include, among others, a higher accuracy, a higher reliability, a higher flexibility, and, most importantly, a lower cost and the ability to duplicate the product with the same performance.

Thanks to dramatic advances in very large scale integrated (VLSI) circuit technology as well as the development in signal processors, these benefits are now seen in the reality. More and more complicated algorithms can be implemented faster and faster in a smaller and smaller silicon area and with a lower and lower power consumption. Due to this fact, the role of DSP has changed from theory to a "tool". Nowadays, the development of products requiring a small silicon area as well as a low power consumption in the case of integrated circuits is desired. The third important measure of the "goodness" of the DSP algorithm is the maximal sampling rate that can be used. In the case of signal processors, the code length is a crucial factor when evaluating the effectiveness of the DSP algorithm. These facts imply that the algorithms generated twenty years ago must be re-optimized by considering the implementation constraints to generate optimized products.

1.1 Desired Form for the Optimization Problem

It is desired that the optimization problem under consideration in converted into the following form: Find the adjustable parameters included in the vector ϕ to minimize.

ρ (ϕ) = max 1_i_I fi (ϕ) . . ., (3.1.1) subject to constraints

gl (ϕ) ≤ 0 for l = 1, 2, . . ., L (3.1.2)

and

hm (ϕ) = 0 for m = 1, 2, . . ., M. (3.1.3)

Here consider three alternative effective techniques for solving problems of the above type. The convergence to the global optimum implies that a good start-up vector ϕ can be generated using a simple systematic design scheme. This scheme depends on the problem at hand.

1.2 Constrained Problems Under Consideration

There exist several problems where one frequency response of a filter or filter bank is desired to be optimized in the minimax or least mean- square sense subject to the given constraints. Furthermore, this contribution considers problems where a quantity depending on the unknowns is optimized subject to the given constraints. In the sequel, we use the angular frequency ω, that is related to the "real frequency" f and the sampling frequency Fs through ω = 2πf/Fs, as the basic frequency variable. We concentrate on solving the following three problems:

Problem I: Find ϕ containing the adjustable parameters of a filter or filter bank to minimize

€A = max!2XA |EA(ϕ, ω)|, (1)

where

EA(ϕ, ω) = WA(ω)[A(ϕ, ω) − DA(ω)], (2)

subject to the constraints to be given in the following subsection.

Problem II: Find ϕ to minimize

€A = Z!2*e*XA [EA(ϕ,ω)] dω, (3)

where EA (ϕ, ω) is given by Eq. (3.1.5), subject to the constraints to be given in the following subsection.

Problem III: Find ϕ to minimize

€A = Ψ(ϕ), (4) 10

where _ (ϕ) is a quantity depending on the unknowns included in ϕ, subject to the constraints to be given in the following subsection.

For Problems I and II, XA is a compact subset of [0,π], A(ϕ, ω) is one of the frequency responses of the filter or filter bank under consideration, DA(ω) is the desired function being continuous on XA, and WA(ω) is the weight function being positive on XA. For Problems I and II, the overall weighted error function EA(ϕ,ω), as given by Eq. (2), is desired to be optimized in the minimax sense and in the least-mean-square sense, respectively.

1.3 Nearly Perfect-Reconstruction Cosine Modulated Filter Banks:

During the past fifteen years, the sub-band coding by M-channel critically sampled FIR filter banks have received a widespread attention. Such a system is shown in Fig.1. In the analysis bank consisting of M parallel bandpass filters Hk(z) for k = 0, 1, . . . ,M − 1 (H0(z) and HM−1(z) are lowpass and high-pass filters, respectively), the input signal is filtered by these filters into separate sub-band signals. These signals are individually decimated by M, quantized, and encoded for transmission to the synthesis bank consisting also of M parallel filters Fk(z) for k = 0, 1, . . ., M−1. In the synthesis bank, the coded symbols are converted to their appropriate digital quantities, interpolated by a factor of M followed by filtering by the corresponding filters Fk(z). Finally, the outputs are added to produce.

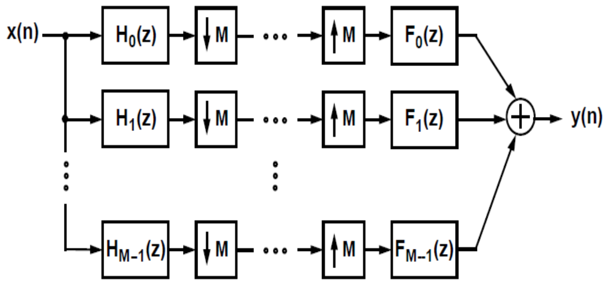


Fig 1 M-channel maximally decimated filter bank

The smaller aliasing error levels are desired to be 18 achieved, then additional constraints must be imposed on the prototype filter. In the case of the perfect reconstruction, the additional constraints are so strict that they dramatically reduce the number of adjustable parameters of the prototype filter.

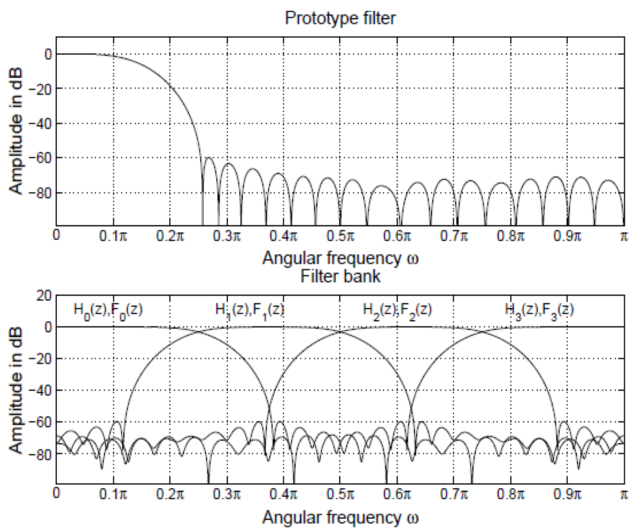


Fig. 2 Example amplitude responses for the prototype filter and for the resulting filters in the analysis and synthesis banks for M = 4, N = 63, and ρ = 1.

2.GENERAL OPTIMIZATION PROBLEMS FOR THE PROTOTYPE FILTER:

This section states two optimization problems for designing the prototype filter in such a way that the overall filter bank possesses a nearly perfect-reconstruction property. Efficient algorithms are then described for solving these problems.

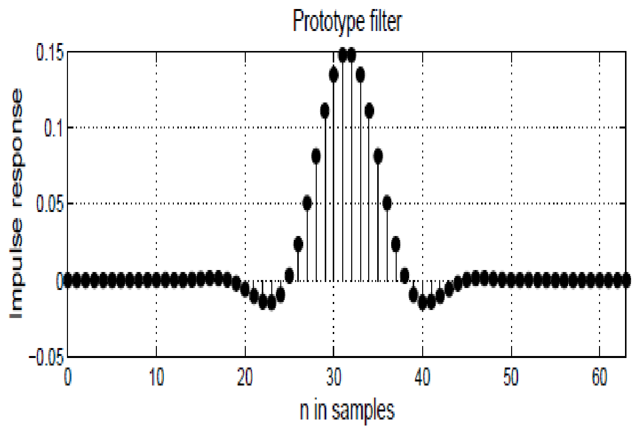
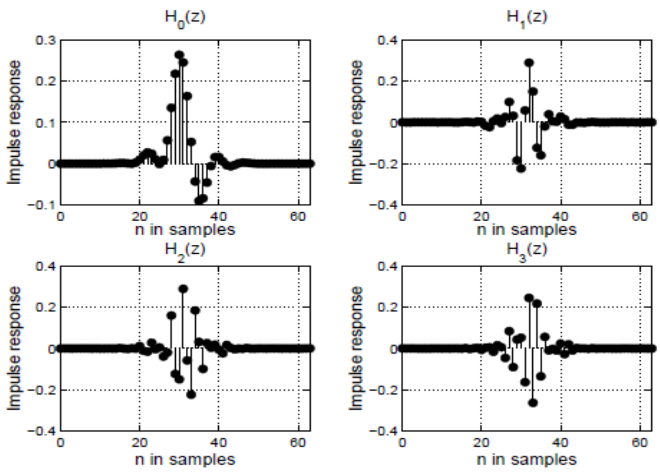


Fig. 3 Impulse response of prototype in the case of Fig. 2



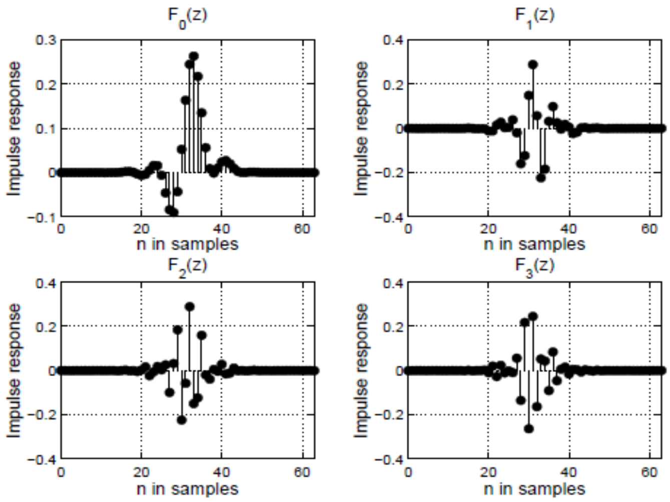


Fig. 4 Impulse response of the filter banks in the case of fig. 2

2.1 Comparisons of Perfect Reconstruction Filter Bank and Prototype Filters:

For comparison purposes, several filter banks have been optimized for ρ = 1 and M = 32, that is, the number of filters in the analysis and synthesis banks is 32. The stopband edge of the prototype filter is thus located at ωs = π/32. The results are summarized in Table I. In all the cases under consideration, the order of the prototype filter is K · 2M − 1, where K is an integer and the stopband response are optimized in either the minimax or least-mean-square sense. δ1 shows the maximum deviation of the amplitude response of the reconstruction error T0(z) from unity, whereas δ2 is the maximum amplitude value of the worst-case aliasing transfer function Tl(z). The boldface numbers indicate that these parameters have been fixed in the optimization. E1 and E2 give the maximum stopband amplitude value of the prototype filter and the stopband energy, respectively. The first two banks in Table I are perfect-reconstruction filter banks where the stopband performance has been optimized in the least-mean-square sense and in the minimax sense. The third and fourth designs are the corresponding nearly perfect reconstruction banks designed in such a way that the reconstruction error is restricted to be less than or equal to 0.0001. For these designs as well as for the fifth and sixth design in Table I, no constraints on the levels of the aliasing errors have been imposed.

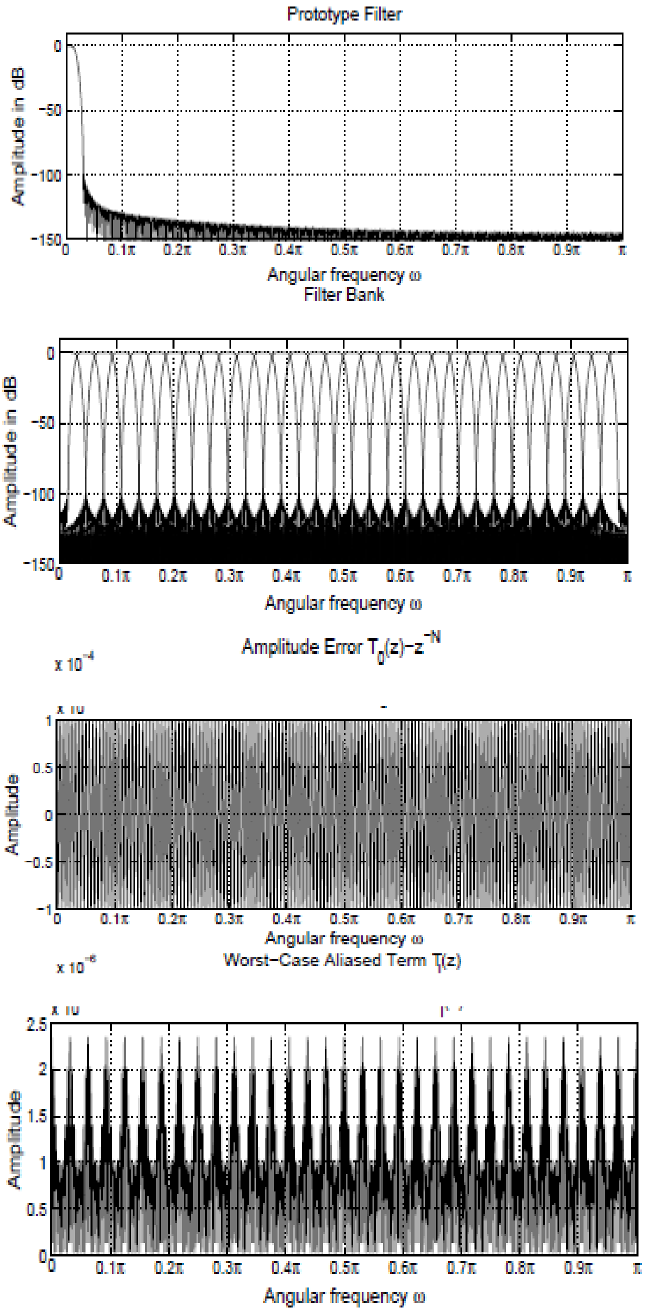


Fig. 5 One example of Filter bank of M = 32 filters of length N + 1 = 512 for ρ = 1 and δ1 = 0.0001. The least-mean-square error design has been used.

2.2 Dutta-Vidyasagar Algorithm:

A very elegant algorithm for solving the constrained optimization problem stated in the second algorithm of

Dutta and Vidyasagar. This algorithm is an iterative method which generates a sequence of approximate solutions that converges at least to a local optimal solution.

$$P(\Phi,\xi) = \sum_{i|f_i(\Phi)>\xi} [f_i(\Phi)-\xi]^2 + \sum_{l|g_l(\Phi)>0} w_l[g_l(\Phi)]^2 + \sum_{m=1}^{M} v_m[h_m(\Phi)]^2.$$

In above equation, the first summation contains only those fi(ϕ)'s that are larger than similarly, the second summation contains only those gl(ϕ)'s that are larger than zero. The wl's and vm's are the weights given by the user. Usually, they are selected to be equal. Their values have some effect on the convergence rate of the algorithm. If ξ is very large, then ϕ can be found to make P(ϕ, ξ) zero or practically zero. ϕ The key idea is to find the minimum of ξ for which there exists _ such that P(ϕ, ξ) becomes zero or practically zero. In this case, ρ(ϕ) ≈ ξ, where ρ(ϕ) is the quantity to be minimized.

2.3 Sequential Quadratic Programming Methods:

Some implementations of the SQP method can directly minimize the maximum of the multiple objective functions subject to constraints that is, these implementations can be directly used for solving the optimization problem formulated in the Subsection 4.1. A feasible sequential quadratic programming (FSQP) algorithm solves the optimization problem stated in Subsection 4.1 using a two-phase SQP algorithm. This algorithm can handle both the linear and nonlinear constraints. Also, the optimization toolbox from MathWorks Inc. [10] provides a function f mini-max which uses a SQP method for minimizing the maximum value of a set of multivariable functions subject to linear and nonlinear constraints.

3. CONCLUSIONS:

Based on the above methods and algorithms we are calculating the models of optimised digital filter and banks. Its overcome after error controls, measure efficiency, Low power consumption and so on. Majorly we focused about algorithms to implement the modules which has more transfer function and less aliasing while generation. Few more features have been observed while modelling the proposed method such as,

- While generating filter banks in prototype filters using Cosine-Modulation Techniques we need to analyse, and synthesis banks of the overall system become approximately peak scaled.

- The most straightforward approach to arrive at a recursive filter having simultaneously a selective amplitude response and an approximately linear phase response in the passband region is to generate the filter in two steps. First, a filter with the desired amplitude response is designed. Then, the phase response of this filter is made approximately linear in the passband by cascading it with an all-pass phase equalizer.

- Algorithms for finding initial filters such as Cascade form filters, Parallel form filters and sub-algorithms.
- The Dutta-Vidyasagar algorithm can be applied in a straightforward manner to further reducing the phase error of the initial filter.
- Modified Farrow structure with adjustable fractional delay.

- Optimization of Pipelined Digital filters using MATLAB

- The solution to the stated optimization problem can be found in two steps. In the first step, a filter with infinite-precision coefficients is determined in such a way that it exceeds the given amplitude criteria to provide some tolerance for the coefficient quantization. The second step involves finding a filter meeting the given criteria with the simplest coefficient representation forms.

REFERENCES:

1. L. Davis and M. Streenstrup, "Genetic algorithms and simulated annealing: An overview," in *Algorithms and Simulated Annealing*, L. Davis, Ed., chapter 1, pp. 1–11. Morgan Kaufmann,1987.

2. S. R. K. Dutta and M. Vidyasagar, "New algorithms for constrained minimax optimization," *Math. Programming*, vol.13, pp. 140–155, 1977.

3. N. J. Fliege, *Multirate Digital Signal Processing*. JohnWiley & Sons, Chichester, 1994.

4. H. S. Malvar, *Signal Processing with Lapped Transforms*. Artech House, Boston, MA, USA, 1992.

5. H. S. Malvar, "Modulated QMF filter banks with perfect reconstruction," *Electron. Lett.*, vol. 26, pp. 906–907, June 1990.

6. P. P. Vaidyanathan, *Mult-irate Systems and Filter Banks*. Prentice Hall, Englewood Cliffs, NJ, 1992.

7. T. A. Ramstad and J. P. Tanem, "Cosine-modulated analysis synthesis filter bank with critical sampling and perfect reconstruction," in *Proc. IEEE Int. Conf. Acoustics, Speech, and Signal Processing*, Toronto, Canada, May 1991, pp. 1789–1792.

8. R. D. Koilpillai and P. P. Vaidyanathan, "New results on cosine-modulated FIR filter banks satisfying perfect reconstruction," in *Proc. IEEE Int. Conf. Acoustics, Speech, and Signal Processing*, Toronto, Canada, May 1991, pp. 1793–1796.

9. R. D. Koilpillai and P. P. Vaidyanathan, "Cosine-modulated FIR filter banks satisfying perfect reconstruction," *IEEE Trans. Signal Processing*, vol. 40, no. 4, pp. 770–783, Apr.1992.

10. T. Saram¨aki, "Designing prototype filters for perfect-reconstruction cosine-modulated filter banks," in *Proc. IEEE Int. Symp. Circuits Syst.*, San Diego, CA, May 1992, pp. 1605–1608.

11. J. Vesma and T. Saram¨aki, "Interpolation filters with arbitrary frequency response for all-digital receivers," in *Proc. IEEE Int. Symp. Circuits Syst.*, Atlanta, Georgia, May 1996, pp. 568–571.

12. J. Vesma and T. Saram¨aki, "Design and properties of polynomial-based fractional delay filters," in *Proc. IEEE Int. Symp. Circuits Syst.*, Geneva, Switzerland, May 2000, vol. 1, pp. 104–107.