

Design of Band-Pass Filter using Artificial Neural Network

Shushank Dogra, Narinder Sharma
Department of Electronics and communication,
Amritsar college of Engineering and technology,
Amritsar, India,

Abstract - For the design of Band pass FIR filters complex calculations are required. Mathematically, by replacing the values of pass-band ripple, stop band attenuation, pass-band frequency F1, pass-band frequency F2, sampling frequency in any of the methods from window method, frequency tasting method or optimal method we can get the values of filter coefficients $h(n)$. Here, window method is used in which Kaiser window method has been chosen preferably because of the presence of ripple factor (β). Here, I have design Band pass FIR filter using artificial neural network which gives optimum result i.e. the difference between the actual and desired output is minimum.

1. INTRODUCTION

The basic function of digital filter is to eliminate the noise and to extract the signal of interest from other signals. A digital filter filter is a basic device used in digital signal processing. There are several techniques available to design the digital filters. But generally while designing a digital filter, first an analog filter is designed and then it is converted into the corresponding digital filter. With the technological development, great advances have been made on design techniques for various digital filters. A filter is essentially a system or network that selectively changes the wave shape amplitude – frequency and or phase – frequency characteristics of a signal in a desired manner . A digital filter is a mathematical algorithm implemented in hardware and/or software that operates on a digital input signal to produce a digital output signal for the purpose of achieving a filtering objective.

1.1 Analog Filters[8]

This is necessary because generally digital filters are designed using analog filters. Some parameters related to analog filters:

- Pass band: It passes certain range of frequencies. In the pass band, attenuation is 0.
- Stop band: It suppresses certain range of frequencies. In the stop band, attenuation is infinity.
- Cut-off frequency: This is frequency which separates pass band and stop band.

Types of analog filters:

The different types of analog filters are as follows:

- Low pass filter (L.P.F): It passes the frequency from 0 upto some designated frequency, called as cut-off frequency. After cut-off frequency, it will not allow any signal to pass through it.

- High pass filter (H.P.F): It passes the frequency above some designated frequency called as cut-off frequency. If input signal frequency is less than the cut-off frequency, then this signal is not allowed to pass through it.
- Band Pass Filter (B.P.F): It allows the frequencies between two designated cut-off frequencies.
- Band Stop Filter (B.S.F): It attenuates all frequencies between two designated cut-off frequencies, while it passes all other frequencies.
- All Pass Filter: It passes all the frequencies. But by using this filter the phase of input signal can be modified.

2. FIR FILTER DESIGN [1]

The design of a digital filter needs five steps:

A. Filter Specification

This may include stating the case of filter, for example low pass filter, the desired amplitude and/or phase response and the tolerances (if any) we are prepared to accept, the sampling frequency and the wordlength of the input data.

B. Coefficient Calculation

At this step, we determine the coefficients of transfer function, $H(z)$, which will satisfy the specifications given in (1). Our choice of coefficient calculation method will be influenced by various factors, the most crucial of which are the critical requirement in step (1).

C. Realization

This involves converting the transfer function obtained in (2) into suitable filter network or structure.

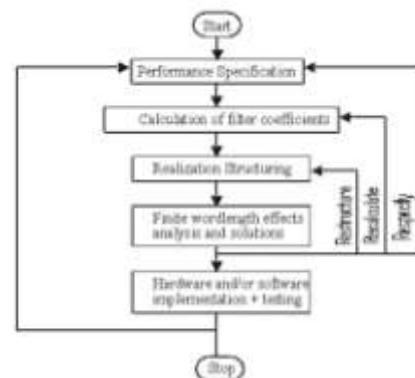


Fig.1.Design of digital filter

D. Analysis of Finite Wordlength Effects

Here, we analyze the effects of quantizing the filters coefficients and input data as well as the effect of carrying

out the filtering operation using fixed wordlengths on the filter performance.

E.Implementation

This demand developing the software code or hardware and performing the actual filtering.

The criteria is a linear phase response in frequency domain called phase response (Jin et al., 2006) as shown in Fig.. Finally, because there is a tradeoff between filter complexity and implementation feasibility, complexity is a performance criteria. Ideal filter characteristics are practically unrealizable.

We have many methods to design FIR filter that are:-

- Fourier series method
- Frequency Sampling method
- Window method

The most of these design techniques suffer from some kinds of drawback, Some of them could not give optimal design in any sense, some is lacking of generality, some need long computing time, and so on (Bagachi and Mitra, 1996).

Kaiser window method has been used because of the presence of ripple parameter beta.

2.1 Various Window Functions

There are many windows proposed that approximate the desired characteristics. The basic window functions are listed below:

Rectangular Window

$$w_r(n) = \begin{cases} 1, & \text{for } -(M-1)/2 \leq n \leq (M-1)/2 \\ 0, & \text{otherwise} \end{cases}$$

Bartlett Window

$$W_T(n) = \begin{cases} 2n/(M-1) & \text{for } 0 \leq n \leq (M-1)/2 \\ 2-2n/(M-1) & \text{for } (M-1)/2 < n \leq (M-1) \\ 0, & \text{otherwise} \end{cases}$$

This window is also called Barlett window.

KAISER WINDOW

Kaiser fixed empirically that the value of β need to achieve a specified value of A is given by

$$\beta = \begin{cases} 0.1102(A-8.7) & \text{for } A > 50 \\ 0.5842(A-21)^{0.4} + 0.07886(A-21) & \text{for } 21 \leq A \leq 50 \\ 0.0 & \text{for } A < 21 \end{cases}$$

Recall that the case $\beta = 0$ is the rectangular window for which $A = 21$. Further more, to achieve prescribed values of A and df , M must satisfy equations

$$M = \begin{cases} \frac{A-8}{14.36df} + 1 & \text{for } A > 21 \\ (0.922/df) + 1 & \text{for } A \leq 21 \end{cases}$$

Finite Impulse response filters (Öner and paper., 1999) are preferred for their stable and linear phase feature. But due to long impulse response of FIR filters there will be more hardware complexity.

Linear-phase form: When an FIR filter has a linear-phase response, its impulse response exhibits certain symmetry conditions. In this form exploit these symmetry relations to reduce multiplications by about half.

2.2 Artificial Neural Network

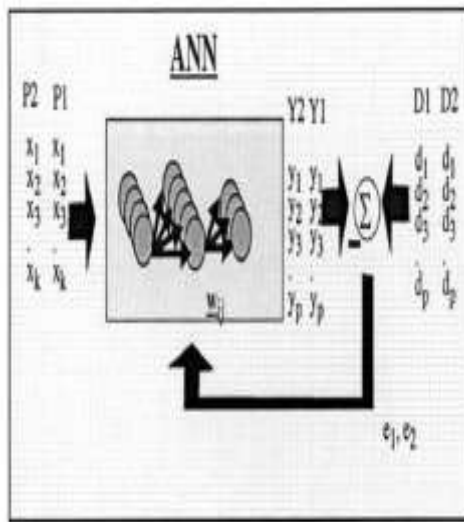
An Artificial Neural Network is an information processing paradigm inspired by the way the densely inter-connected, parallel structure of the mammalian brain process information. ANN have successfully applied to a number of problems including the identification and control of dynamical systems, communications networks, coordination of robotics handeye movements. It is also referred to as a neuromorphic system, follows connectionist architecture, and parallel distributed processing. Artificial Neural Networks are collections of mathematical models that emulate some of the observed properties of biological nervous system. The key element of the ANN is the novel structure of the information processing systems.

Some other advantages of ANN are as under:

1. Adaptive learning
2. Self-Organisation
3. Real Time Operation

3. Main title

The main title (on the first page) should begin 1-3/8 inches (3.49 cm) from the top border of the page, focused, and in Times 14-point, bold face case. Capitalize the beginning missive of nouns, pronouns, verbs, adjectives, and adverbs; do not take advantage articles, align conjunctions, or prepositions (unless the title begins with such a word). Allow for two 12-point blank lines later the title.



The style of neural computation.

Fig.2 General structure showing performance of the artificial neural network

2.3 Bandpass Filter

A band-pass filter is a device that passes frequencies within a certain range and rejects (attenuates) frequencies outdoor that range. Optical band-pass filters are of mutual usage. An example of an analogue electronic band-pass filter is an RLC circuit (a resistor-inductor-capacitor circuit). These filters can too be made by combining a low-pass filter with a high-pass filter.

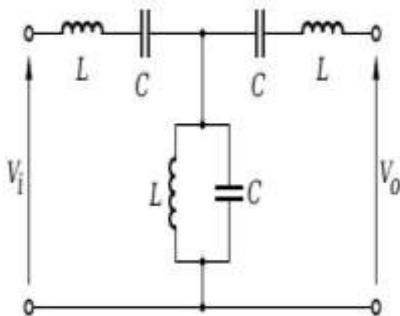


Fig.3 Bandpass filter

2.4 Multilayer Perceptron Networks

Multilayer Perceptron Networks form a class of feed forward neural networks. They are not a single layer network but consist of an input layer, arbitrary number of hidden layers and an output layer as shown in figure 1.5. Here input is fed to each of the input layer neurons. The outputs of the input layer feed into each of next layer neurons and so on, forming a layered structure having one input layer, one output layer and L-2 hidden layers in an L layer network.

3. Formulation of Problem

The design of digital filter means basically finding the values of filter coefficients so that given filter specification are achieved the window based design method are exclusively used for calculating there coefficient. We have used Kaiser window for this purpose. The Kaiser window function goes somehow in overcoming the incorporating a ripple control parameter, ANNs have been used for the design of digital filter with Passband ripple, stop band attenuation, passband frequency F1, passband frequency F2, sampling frequency as input parameters. In this thesis ANN have been used to design the band pass FIR filter coefficient that are matching with coefficient given by Kaiser with here the multi layer perceptron feed forward network has been used for the design because this method is efficient, accurate, less complex and easily implemented. The network has been trained in such a manner so that the error comes minimum, means there may be very less difference in the results comes from actual calculations that has been come from matlab and the output comes from trained artificial neural network.

3.1 Objective:

The objective of the present work are divided into the following sections.

- (1) To prepare the data sheet using different values of filter parameter achieve the filter coefficient.
- (2) Choosing ANN a Band-pass FIR filter has been designed such that its coefficient match with coefficients obtained with window method.

4. EXPERIMENTATION

The code has been implemented in MATLAB. The name Matlab stands for matrix laboratory MATLAB is an intercalate system whose basic data element is an array that does not require marking. This allows solving many technical computing, in a fraction of the time for the simulation and the corresponding analyses of the given application the following framework and distributed design situations have been taken case of and then implemented feel typical set of settings.

Preparation of data sheet with following that are

- (1) Passband Ripple (Ap)
- (2) Stopband Attenuation (As)
- (3) Pass Band Frequency (F1)
- (4) Passband Frequency (F2)
- (5) Sampling Frequency (Fs)

Filter coefficient are calculated and in this topic works is carried out using approximately 25 such values of all the above parameters to calculate the filter coefficients. The range of different parameters has been taken which are:

- (a) Ap 0.7 - 1.3 dB
- (b) As 40-55 dB
- (c) F1 7000 - 11000 hz
- (d) F2 17000-21000 hz
- (e) Fs 47000-52000 hz

Using this data set the Artificial Neural Network has been trained and can be use to calculate filter coefficients for input parameters in this range. Now, ANN is use to design

the Band pass FIR filter. There is very no difference in the Ann results and the calculated results.

5. RESULT

The ANN is designed for maximum value of $N = 17$ to $N = 21$ so it has outputs as shown in Fig.4. The MATLAB Software has been used for this work. The network has been trained using Multilayer Perceptron in which Error Back Propagation Algorithm has been used to design BAND PASS FIR filter. using “Levenberg –Marquardt” (trainlm) in the neural network feedforward (newff) the goal meet condition has been achieved as shown in fig

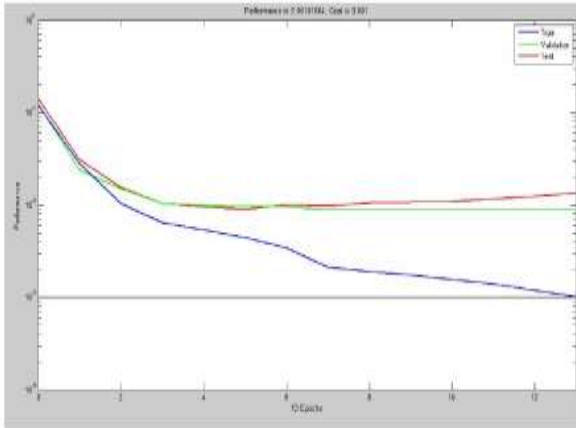


Fig.4: Training of Artificial Neural Network

5.1 Kaiser Vs Artificial Neural Network

Input parameters:- $A_p = 0.729$, $A_s = 41.502$, $F_1 = 9815$, $F_2 = 18234$, $SF = 49661$, Filter Length=17

h(n)	Kaiser Window Method	Artificial Neural Network	Error Values
h(0)	0.0001	-0.2214	0.0015
h(1)	0.0047	0.0058	-0.0011
h(2)	0.0109	0.0100	0.0009
h(3)	-0.0204	-0.0209	0.0005
h(4)	-0.0108	-0.0101	-0.0007
h(5)	0.0329	0.0331	-0.0002
h(6)	-0.0019	-0.0015	-0.0004
h(7)	-0.0306	-0.0306	-0.0000
h(8)	0.0076	0.0075	0.0001
h(9)	0.0001	0.0003	-0.0002
h(10)	0.0188	0.0185	0.0003
h(11)	0.0286	0.0284	0.0002
h(12)	-0.0906	-0.0895	-0.0011
h(13)	-0.0166	-0.0175	0.0009
h(14)	0.1686	0.1695	-0.0009
h(15)	-0.0560	-0.0547	-0.0013
h(16)	-0.1980	-0.1982	0.0002
h(17)	0.1482	0.1483	-0.0001

Table no.1

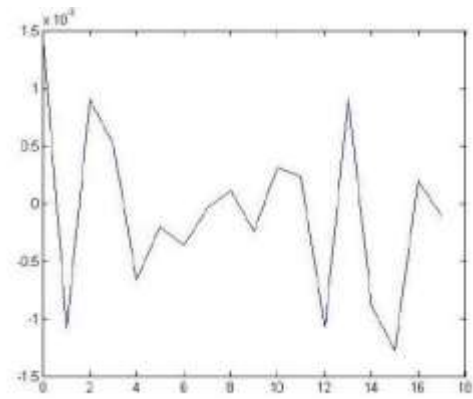


Fig.5 Fault graph

Input parameters:- $A_p = 0.894$, $A_s = 40.2$, $F_1 = 10000$, $F_2 = 18000$, $SF = 50000$, Filter Length=18

h(n)	Kaiser Window Method	Artificial Neural Network	Error Values
h(0)	-0.0015	0.0436	-0.0451
h(1)	-0.0003	-0.0422	0.0419
h(2)	0.0189	0.0226	-0.0037
h(3)	-0.0083	0.0099	-0.0182
h(4)	-0.0276	-0.0529	0.0253
h(5)	0.0214	-0.0271	0.0485
h(6)	0.0219	0.0837	-0.0618
h(7)	-0.0283	-0.0334	0.0051
h(8)	-0.0072	-0.0523	0.0451
h(9)	0.0032	0.1160	-0.1128
h(10)	0.0006	-0.0018	0.0024
h(11)	0.0470	0.0563	-0.0093
h(12)	-0.0357	0.0286	-0.0643
h(13)	-0.0958	-0.1307	0.0349
h(14)	0.1102	-0.0213	0.1315
h(15)	0.0972	0.1712	-0.0740
h(16)	-0.1924	-0.0318	-0.1606
h(17)	-0.0420	-0.2228	0.1808
h(18)	0.2265	0.1049	0.1216

Table no.2

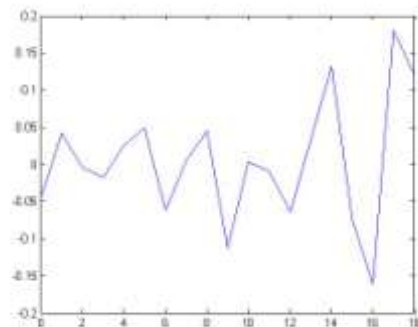


Fig.6 Fault graph

Input parameters:- $A_p = 0.894$, $A_s = 42.3$, $F_1 = 10000$, $F_2 = 18000$, $SF = 50000$, Filter Length=19

h(n)	Kaiser Window Method	Artificial Neural Network	Error Values
h(0)	0.0004	0.0004	-0.0000
h(1)	-0.0033	-0.0029	-0.0004
h(2)	-0.0001	0.0004	-0.0005
h(3)	0.0208	0.0203	0.0005
h(4)	-0.0077	-0.0082	0.0005
h(5)	-0.0271	-0.0261	-0.0010
h(6)	0.0225	0.0228	-0.0003
h(7)	0.0227	0.0218	0.0009
h(8)	-0.0269	-0.0270	0.0001
h(9)	-0.0061	-0.0056	-0.0005
h(10)	0.0024	0.0023	0.0001
h(11)	0.0013	0.0024	-0.0011
h(12)	0.0489	0.0481	0.0008
h(13)	-0.0356	-0.0374	0.0018
h(14)	0.0956	-0.0925	-0.0031
h(15)	0.1107	0.1127	-0.0020
h(16)	0.0973	0.0922	0.0050
h(17)	0.1907	-0.1908	0.0001
h(18)	0.0412	-0.0354	-0.0058
h(19)	0.2253	0.2227	0.0026

Table no. 3

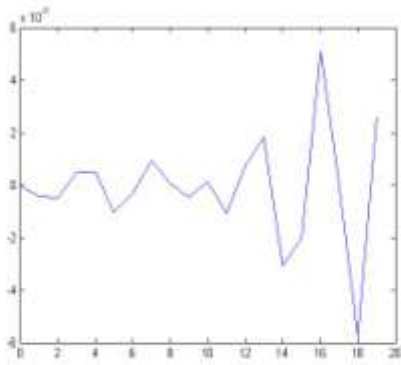


Fig.7 Fault graph

Input parameters:- $A_p = 0.799$, $A_s = 47.9$, $F1 = 9891$, $F2 = 18012$, $SF = 48481$, Filter Length=20

h(n)	Kaiser Window Method	Artificial Neural Network	Error Values
h(0)	0.0008	0.0329	-0.0321
h(1)	-0.0024	-0.0393	0.0369
h(2)	0.0015	0.0483	-0.0468
h(3)	-0.0070	-0.0051	-0.0019
h(4)	-0.0019	0.0320	-0.0339
h(5)	0.0197	0.1932	-0.1735
h(6)	-0.0105	0.0029	-0.0134
h(7)	-0.0263	-0.1372	0.1109
h(8)	0.0288	0.2251	0.1963
h(9)	0.0146	-0.0031	0.0177
h(10)	-0.0293	-0.0975	0.0682
h(11)	0.0014	0.0169	-0.0155
h(12)	-0.0033	-0.0140	0.0107
h(13)	0.0153	0.2033	-0.1880
h(14)	0.0442	-0.2205	0.2647
h(15)	-0.0840	-0.3883	0.3043
h(16)	-0.0339	1.0523	-1.0862
h(17)	0.1687	-0.1109	0.2796
h(18)	-0.0360	-1.6279	1.5919
h(19)	-0.1998	1.3188	-1.5186
h(20)	0.1421	1.2667	-1.1246

Table no. 4

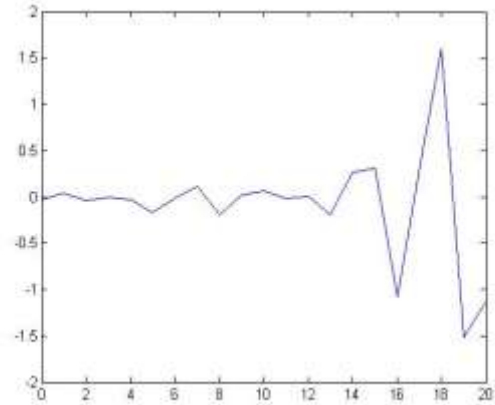


Fig.8 Fault graph

Input parameters:- $A_p = 0.7$, $A_s = 45$, $F1 = 10000$, $F2 = 18000$, $SF = 50000$, Filter Length=21

h(n)	Kaiser Window Method	Artificial Neural Network	Error Values
h(0)	0.0020	1.0404	-10384
h(1)	-0.0036	-0.8495	0.8459
h(2)	-0.0006	1.5396	-1.5402
h(3)	-0.0061	0.3459	-0.3520
h(4)	-0.0010	0.8101	-0.8111
h(5)	0.0177	-0.5527	0.5704
h(6)	-0.0078	-1.5426	1.5384
h(7)	-0.0276	0.0803	-0.1079
h(8)	0.0226	-0.6122	0.6348
h(9)	0.0230	-0.8297	0.8527
h(10)	-0.0275	0.1117	-0.1392
h(11)	-0.0064	0.6550	-0.6614
h(12)	0.0032	0.3480	-0.3448
h(13)	0.0013	0.5629	-0.5616
h(14)	0.0484	-1.3082	1.3566
h(15)	-0.0354	-0.2396	0.2042
h(16)	0.0956	0.7822	-0.8778
h(17)	0.1108	-0.1199	0.2307
h(18)	0.0977	1.3670	-1.2693
h(19)	-0.1913	0.5530	-0.7443
h(20)	-0.0414	0.3605	-0.4019
h(21)	0.2261	0.0093	0.2168

Table no.5

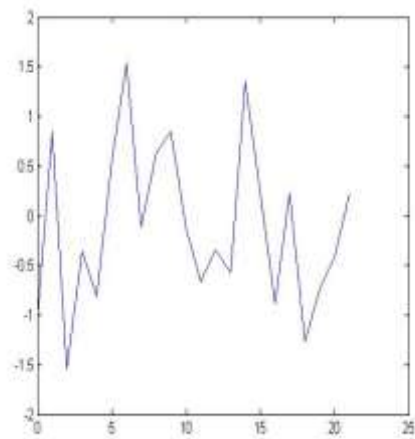


Fig.9: Fault graph

CONCLUSION:-

Artificial Neural Network is better and easy method of design of Band Pass FIR Filter. Also, using Fourier series, Frequency sampling or Window methods the filter can be design but for each unknown parameter the filter coefficients have to calculated. In comparison with ANN, the trained network can calculate the filter Coefficient for unknown parameter in that specified range. Using ANN if error graphs are drawn between ANN output and Kaiser Window method is almost zero if we use back propagation method.

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