Comparative Study About the Effect of Order on Different Analog filters and Adaptive Filters(RLS Algorithm) using MATLAB

Naveen Kumar S ,Vamshi Bharadwaj M V, Thondamala Sai Kiran, Bharath Kumar M

[1] *Student,Dept. of Electronics and Communication,M S Ramaiah University Of Applied Sciences, Karnataka, India*

[2] *Student,Dept. of Electronics and Communication,M S Ramaiah University Of Applied Sciences, Karnataka, India*

[3] *Student,Dept. of Electronics and Communication,M S Ramaiah University Of Applied Sciences, Karnataka, India*

[4] *Asst.Professor,Dept. of Electronics and Communication,M S Ramaiah University Of Applied Sciences, Karnataka, India*

ABSTRACT

Filters mainly represent a niche in processing a signal. Analog and digital sections have their own pros and cons. The paper gives an overview of the effect, that the order of the filter imposes on the response of the analogue filters such as: Butterworth, Chebyshev, Elliptic(Cauer)and Bessel filter. The paper also includes the experimental analysis of the noise cancellation using adaptive filters. The implemented example is used to obtain the effect of, the order of the Recursive least square algorithm used for filtering, and the results are demonstrated in frequency domain.

1. INTRODUCTION

A filter is device that eliminates the undesired frequencies and allows all the desired ones. The filtering in general has the mentioned definition associated with it. Filtering is mainly divided into two domains namely: Analog and digital filters; which are furthers divided with the advancement based on the needs.The paper is about the analog filters and their properties.The types of analog filters are:

<table>
<tr>
<td>The types of analog filters are:
1. Butterworth filter
2. Chebyshev filter
3. Elliptic (Cauer) filter
4. Bessel filter
5. Gaussian filter
6. Linkwitz–Riley filter
7. Optimum "L" (Legendre) filter</td>
<td>Note: Since for the experimentation MATLAB tool is used, only (1), (2), (3), (4) can be used for the comparison.These filters have their own field of applications and are further divided into (low pass, high pass, band pass, band stop) filters.Then the paper tries to draw a direct comparison with analogue adaptive filtering. Subsection(1,6)</td>
</tr>
</table>

1.1 Butterworth filter:

It’s a type of filter in which the pass band is maximally flat as mathematically possible ie ; it is ripple free in the passband. Hence it is also called as flat magnitude filter. The quality factor of this type of filter is only 0.707. This filter achieves flatness at an expense of relatively wide transition region from pass band to stop band with average transient characteristics. The normalized poles of Butterworth filter lie in the unit circle of s- plane.

The magnitude of this filter response is given by

$$|H(j\omega)| = \frac{1}{\sqrt{1+\left(\frac{\omega}{\omega c}\right)^{2n}}}$$

where ; $H(j\omega)$ is transfer function at angular frequency ω

ω is Angular frequency which is equal to $2\pi f$

ω_c is cutoff frequency which is equal to $2\pi fc$

Applications of Butterworth filter include anti-aliasing filter in data convertor applications, motion analysis and also in radar target track display.

1.2 Chebyshev type 1 and type 2 filter:

Type 1:

This filter is also called as passband ripple filter. This filters are all pole filters and they have monotonic characteristics in the stop band. The magnitude response of this filter is shown in fig.1.2a also the ripples can be observed in the passband.

The magnitude of this filter response is given by

$$|H(j\Omega)|^2 = \frac{1}{1+\epsilon^2\, C_N^2\left(\frac{\Omega}{\Omega c}\right)}$$

where $C_N\left(\frac{\Omega}{\Omega c}\right)$ is Chebyshev polynomial of order N and ϵ is ripple parameter in passband

Type 2:

This filter is also called as stopband ripple filter. These filters contains zeros as well as poles in the s-plane. This filter shows equiripple behaviour in stopband and monotonic behaviour in the passband. The main drawback is this filter doesn't roll-off.

Basically Chebyshev filters are used in RF application which require wide transition between passband and stopband to remove intermodulation of harmonics. They are also used in medical applications such as removal power line interference and also removal of baseline wander.

1.3 Elliptic filter:

This filter is also known as Cauer or Zolotarev Filter. These filters can achieve the smallest filter order for same specifications or narrowest transition region for the same filter order. This type of filter produces faster transition from passband to stopband but also exhibits Gain ripple in both stopband as well as in stop band.

These filters are used in various RF applications due to its fast transition between passband and stopband.

The magnitude of this filter response is given by

$$|H(j\omega)| = \frac{1}{\sqrt{1+\epsilon^2\, Rn^2\left(\frac{\omega}{\omega s}\right)}}$$

$H(j\omega)$ is transfer function at angular frequency ω, ω is Angular frequency which is equal to $2\pi f$ and ω_s is the scaling frequency

1.4 Bessel filter:

It's a type of filter with maximally flat group or phase delay and also maintains the wave shape of filtered signals in the passband as per the original signal . It has the maximum linear phase response compared to other filters. Bessel crossover has the most linear phase among Butterworth, Chebyshev and Elliptic filter along with fairly good magnitude flatness and minimal lobing for the even orders.

Bessel filters finds application in audio cross over systems.

The transfer function of lowpass filter is gien by

$$H(s) = \frac{\theta_n(0)}{\theta_n(s/\omega_0)}$$

Where $\theta n(s)$ is the reverse Bessel Polynomial and ωo is the frequency chosen to get desired cut off frequency.

- **Magnitude Response Of Different Filters**

Butterworth Filter

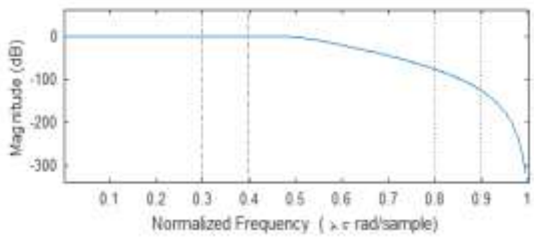


fig.1.1a

Chebyshev Type I

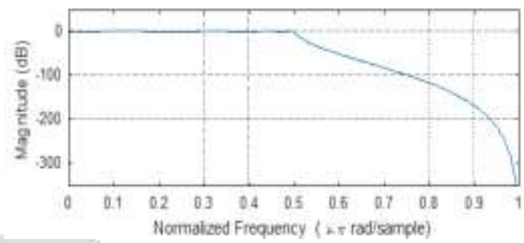


fig.1.2a

Chebyshev Type II

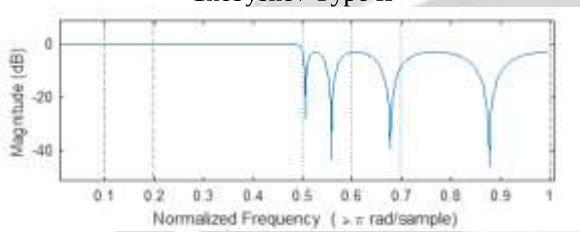


fig.1.2b

Elliptic Filter

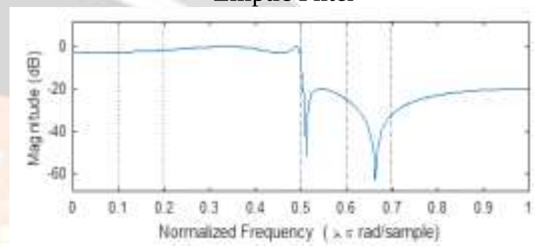


fig.1.3a

Bessel Filter

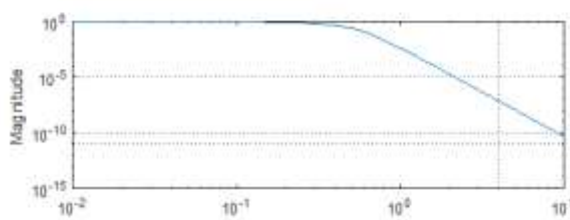


fig1.4a

1.5 Adaptive filters

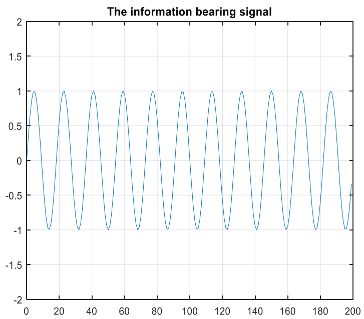
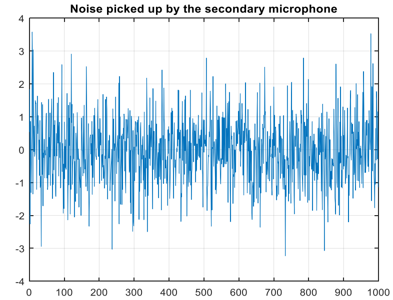
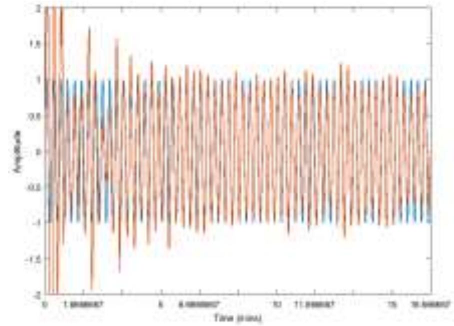
Adaptive filters are the linear filters with their parameters adjusting themselves, based on the error obtained during comparison of the signal. i.e. the desired using some of the optimization algorithms.

Analogue and digital adaptive filtering techniques have already replaced many applications of simple filters as these give higher accuracy and efficiency. These filters use optimization algorithms for obtaining the feedback. There are adaptive lattice algorithms, the recursive least squares algorithm and algorithms based on hyper stability simulated annealing, genetic optimisation and random or linear searches. All these are of digital type. For analog filters we have mainly two: Least mean square and heuristic algorithms are used.

Generally, at lower speeds with low complexity and high efficient applications digital filters hold the upper hand over analog. But when it comes to high speed and low power consumption analog adaptive filters aces the race.

Majority of the filters in the present industry digital. Analog adaptive filters have very limited fields or applications left with due to the digital logics which have both speed and is cheaper comparatively.

Since in Least mean square algorithm has no direct implementation in a MATLAB more over the noise added is of White Gaussian. So LMS fails to filter out the original signal, hence RLS (recursive least squares algorithm) is implemented as an example.

This figure is the representation of the input signal.  fig.1.5a	This figure is the white Gaussian noise added to the input signal. 	This figure is the input signal to that of filtered output signal obtained. 

Block diagram that describes the adaptive filtering:

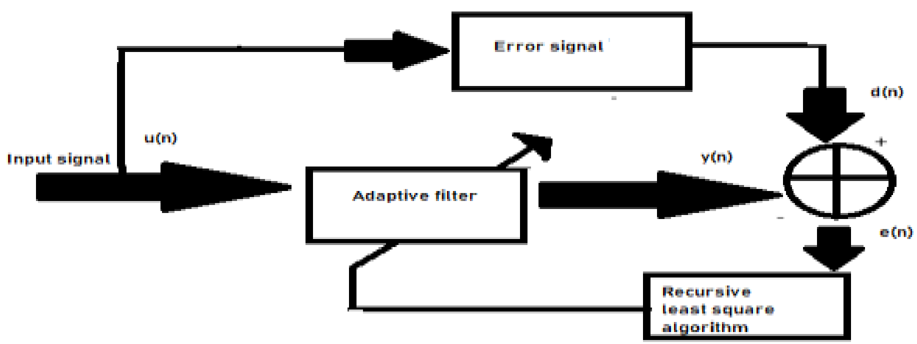


fig.1.5a

2. Comparison:

- **Butterworth filter:**

The response of the filter is obtained in the frequency domain. As a traditional and most popular filter in analog and digital domain this filter gives no ripples in the passband and they are present in the stop band.
So a sharp cutoff is very difficult to obtain in at the cutoff frequency. From fig2.1a, as the order of the filter increases, the transition band is reduced.

- **Chebyshev filter:**

The major difference between Butterworth and Chebyshev filter is that, the poles of Butterworth filter lie on the circle of s-plane whereas for Chebyshev, the poles lie on the ellipse. As mentioned in the section (1), chebyshev filter provides two variations in its results using and they are termed as type 1 and type 2. Though type 1 have recognizable ripples in its pass band, stop band is free from ripples and has a sharp cutoff compared to Butterworth, even at the lower order. The transition band comparatively takes longer time to settle. In the type 2 of chebyshev filter, the pass band is flat and has a sharp cutoff frequency. The only hitch in this filter is the stop band ripples. These ripples vary from 0-40 dB and hence there might be some noise even after the filter is applied to the input signal. And from the fig2.2a and fig2.2b: with the increase in order of the filter the stop band ripples comes closer to each other.

- **Elliptic filter:**

The elliptic filter gives ripple in each band, which can be independently adjustable. Transition in gain is achieved faster than all other filters for the given ripples. Even if the independent adjustability of the pass and ripple bands are not used practically, the insensitive behavior for component variation in the circuit, gives an upper hand over all other filters.From the figure 2.3a, passband has few small ripples and stopband has the attenuation in the form of larger ripples. The stop band attenuation varies in larger magnitudes. As mentioned in section (1.3), this filter is combination of both Butterworth and chebyshev filters.

- **Bessel filter:**

The response obtained by the Bessel filter is very ideal compared to all the filters that are mentioned in the section (1). It gives a maximally flat pass band. From the experiment performed and the resultant graphs(fig2.4a), it can be stated that as the order of the filter increases, the filter characteristics are decreasing. The stop band attenuation of the low pass filter designed decreases as the order of the Bessel is increased.

➢ **Magnitude Response of Different Filters with varying filter order(n):**

Butterworth Filter

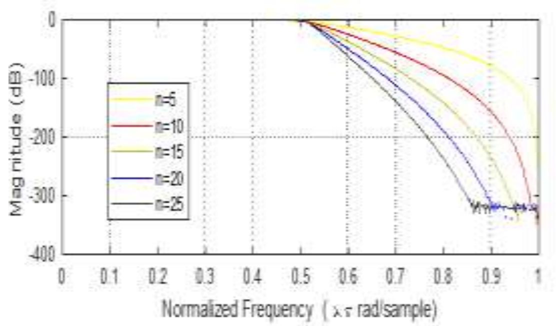


fig.2.1a

Chebyshev Type I

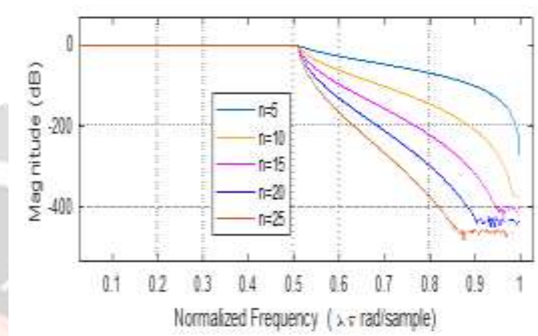


fig.2.2a

Chebyshev Type II

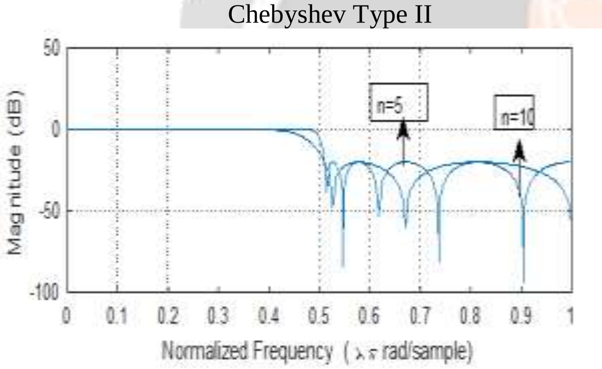


fig.2.2b

Elliptic Filter

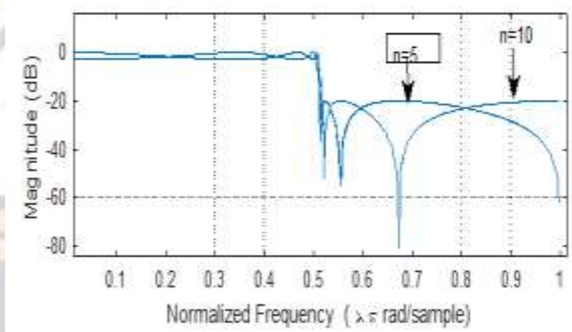


fig.2.3a

Bessel Filter

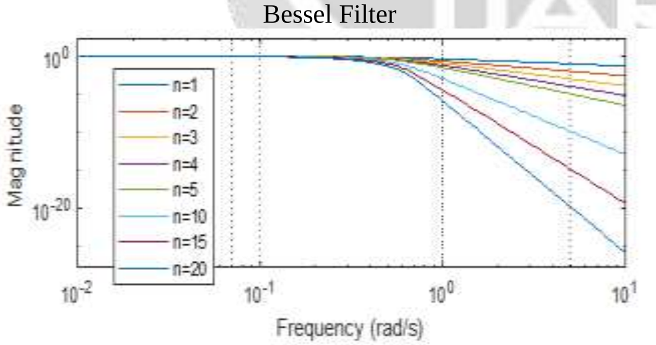


Fig2.4a

- **Adaptive filters**

Adaptive filters are practically demonstrated using a noise cancellation technique using the recursive least squares algorithm. On the account of results obtained, the filter is nearly ideal. In the fig2.5a the message signal is generated of sinusoidal form, random noise generated and is added to the message signal which is displayed in fig2.5b and fig 2.5c is the output that is obtained after filtering out the noise.

From the graphs that are mentioned in the table below, it is evident that a system stabilizes to a maximum extent as the order of the transfer function of the adaptive filter increases. But the variation with respect to the mean value is maximum at both ultra-higher and lower orders of the mean frequencies. This phenomenon can be observed in the graph where M=100,200 and M=20,25 respectively. The median order is 32 which was obtained experimentally through the scope.

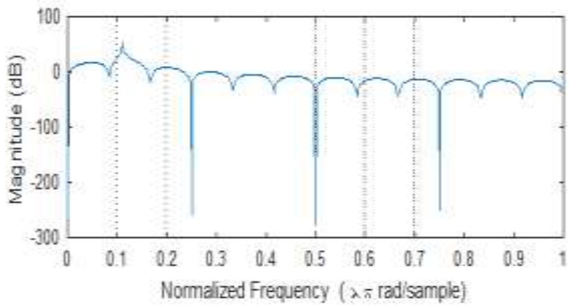


fig2.5a

M=20

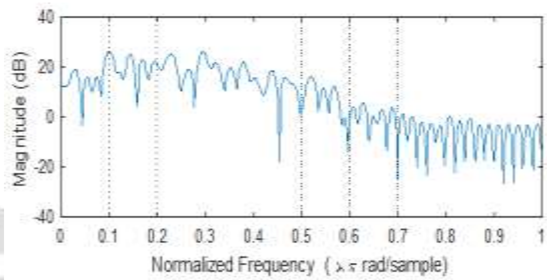


fig2.5b

At M=25

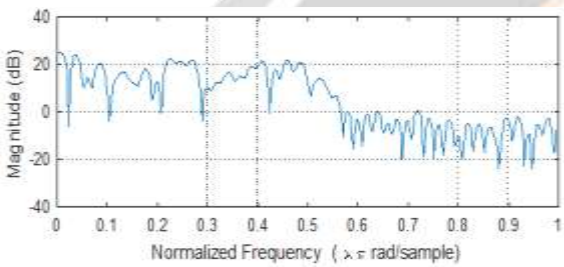


fig2.5c

M=32:

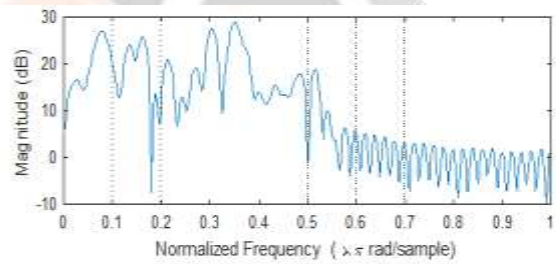


fig2.5d

At M = 100

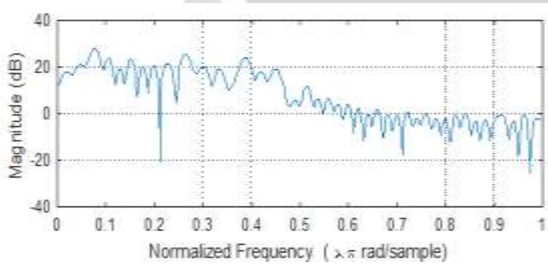


fig.2.5e

At M=200

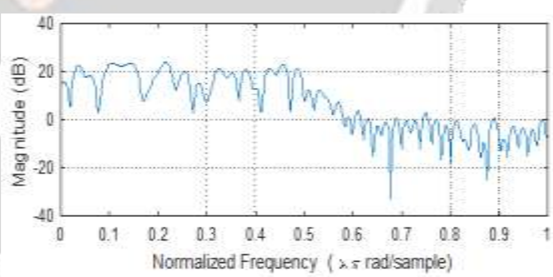


fig.2.5f

3. CONCLUSIONS

From the results that are obtained from the previous section following analysis can be obtained.

- In case of Butterworth filter the increase in order stabilizes the output filter response to that of required size. The cutoff frequency decreases, accuracy increase. No ripples are observed in either cases.
- In chebyshev filter type 1 the increase in order decreases the number of ripples in the passband and makes the cutoff steeper by reducing the transition band.
- In chebyshev filter type 2 as order increases, stop band ripple shortens their width and becomes closer to one another.
- In case of ellipse as the order increases the ripple in the pass band stabilizes but the stop band ripple comes close to each other by giving a sharp cutoff frequency.

- Bessel filter has a unique characteristics compared to all other filters mentioned above. As mentioned in the section (1) is the filter with flatness at the passband. From obtaining the graphs at different orders it is evident that, as the order increases the stopband attenuation decreases. The characteristics of the designed filter is lost. Though at any cases passband or stopband ripple doesn't exist in a recognizable manner.
- Adaptive filters are said to give more efficient results at the cost of complexity while designing the circuit, it was mentioned that the LSM is not used because of the implementation of the white Gaussian noise. But even in the RLS an offset voltage was observed initially. this can be reduced using a differential amplifier to eliminate the excess voltage.
- Output of the adaptive filters were obtained in the frequency domain for simplified comparison. And frequency response was not obtained during the course of results. From the graphs, as the order decreases below a threshold, the output is distorted and the order above an upper boundary makes the filter to give distorted output.
- Hence, in adaptive filtering there is a clear boundary to maintain a good frequency response.

As the order of the transfer function corresponds the increase in the complexities, it is not universal that the efficiency or the accuracy must increase. In many cases,
it is either ways. Since the determination of order, while designing a filters very important, it was the purpose of this paper to determine the real consequences of scaling the order during any designing work.

4.Future work:

- Obtaining the properties of Gaussian, Linkwitz–Riley filter, Optimum "L" (Legendre) filter for different orders of transfer function.
- Determining boundary for the order of transfer function for the adaptive filters used.
- Obtaining a better algorithm to reduce the dc offset in the output of the analog adaptive filters.

5. REFERENCES

[1] Nosakani and Van , "**Design theory and data for electrical filters**", 1997..

[2] Kaufman and Seidman, "**Handbook of electronic calculations**", 1979.

[3] A.Carusone and D.A.Johns "**Analogue adaptive filters: past and present**" IEE Proc.-Circuits Devices Syst.. Vol. 147, No I , February 2000

[4] J. Gerardo Avalos, Daniel Espinobarro, Jose Velazquez, Juan C. Sanchez, "**Adaptive noise canceller using LMS algorithm with codified error in a DSP**", *Circuits and Systems 2009. MWSCAS '09. 52nd IEEE International Midwest Symposium on*, pp. 657-662, 2009.

[5] "Comparative study of different types of an active filter " Abbas.S. Hassan

[6] WIDROW, B., and STEARNS, S.D.: 'Adaptive signal processing'(Prentice Hall, New Jersey, 1985)

[7] HAYKIN, **S.:** '**Adaptive filter theory**' (Prentice Hall, New Jersey,5th edition.)

[8] "Analog Filters Using Matlab 1st Edition" by **Lars wanhammar.**

[9] R .A. Baki, M.N. El-Gamal, "**A low-power 5-70-MHz seventh-order log-domain filter with programmable boost group delay and gain for hard disk drive applications**", *Solid-State Circuits IEEE Journal of*, vol. 38, no. 2, pp. 205-215, 2003.

[10] "**Frequency domain approach to fault diagnosis of analog filters"** Dražen Jurišić , Neven Mijat and Vladimir Čosić - Proceedings of Third International Conference on Electronics, Circuits, and Systems, 1996