



# Designing of Multiband FIR Digital Filter for SR Signal

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**ABSTRACT:** This paper deals the designing as well as application of the constrained least squares multiband FIR digital filter on the electromagnetic Schumann resonance (SR) signal, and design methods based on MATLAB simulation. FDATool in MATLAB is used to determine low order filter coefficients and simulate FIR multiband pass filters to find necessary first four modes of SR signal, 8, 14, 20 and 26 Hertz, with the daily variation of about +/- 0.5 Hertz, by employing a set of 3-component search coil magnetometer at tropical station Agra, India (Geograph.lat.27.21N, long. 781E). I also use IIR notch filter to remove a big sharp peak noise due to local interference (like train noise) in the SR signal. The results prove that the performance of the designed low order multiband FIR filter reaches the appointed requirement. The result shows before filtering and after filtering the SR signal, which clearly indicates the measurement of continuous SR signal frequencies and corresponding intensities are accurate for each channel.

**KEYWORDS:** FIR, IIR,SR signal, PSD,MATLAB

## I. INTRODUCTION

Digital filters can be categorized in infinite impulse response (IIR) digital filter and finite impulse response (FIR) digital filter. FIR filters have the various primary advantages as they are always stable and can have exactly linear phase. The design methods are generally linear. They can be realized efficiently in hardware also. An FIR system has a lot of useful properties, such as system stability, only zeros, fast operating speed, design flexibility and linear phase characteristics, so that FIR has been widely used in image processing, data transmission, digital audio, and other areas. FIR filter has a variety of ways to achieve, with the processing of latest electronic technology.

In this paper, I present a FIR constrained least-squares Multiband digital filter design and compare this with other possible multiband filters as FIR Least Square, FIR Equiripple, and FIR Complex Equiripple type methods based on FDATool available in MATLAB simulation to find necessary first four modes of Schumann resonance (SR) signal, 8, 14, 20 and 26 Hertz, with the daily variation of about +/- 0.5 Hertz. The Schumann resonance (SR) is one of the most exciting electromagnetic phenomena occurring in the earth ionosphere cavity. It shows standing waves at ELF frequencies of 8, 14, 20, 26...Hz as a result of resonance between direct and round-the-world ELF waves radiated from lightning discharges. Though, the phenomena of SR was predicted long ago in the year 1952[1] and detected experimentally in 1960 [2], This is a useful analysis which provide very important global information as global thermometer, determining thunderstorm Centre's, ionosphere studies, and forecasting monsoon activity, etc. The output of filter will be used to analyze Schumann resonance data, which are taken from continuous monitoring of ULF/ELF magnetic field emissions employing a set of 3-component search coil magnetometer as shown in Fig.(1) that has been imported from Lviv center of Institute of Space Research, Ukraine[3].

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Vol. 6, Issue 7, July 2018



Fig. 1. 3-Component search coil magnetometer installed at Bichpuri, Agra (India),

The three sensors of the magnetometer are buried 1m underground in orthogonal directions such that X-component is oriented in north-south, Y-component in east-west, and Z-component in vertical directions. The location is chosen in Agriculture field of the Bichpuri campus of R. B. S. College, in rural area, about 12 km west of Agra city where electric and electromagnetic disturbances are low[4]. I also design and use IIR notch filter to remove a big sharp peak noise due to (like train noise) in the SR signal.

## II. FIR DIGITAL FILTER

### A. The basic concept of FIR digital filter

FIR filters are digital filters with finite impulse response. They are also known as non-recursive digital filters as they do not have the feedback (a recursive part of a filter), even though recursive algorithms can be used for FIR filter realization. For impulse response  $h(n)$  of FIR filter has been limited, the difference equation of N order of the recursive digital filters can be expressed as by equation (1)

$$y(n) = \sum_{k=0}^{N-1} h(k)x(n-k) = \sum_{k=0}^{N-1} (b_k x(n-k)) \quad (1)$$

Where,  $y(n)$  is the output signal,  $x(n-k)$  is the input sample sequence on the  $n$ th times,  $h(n)$  is the filter tap coefficient and N is the number of the filter tap.

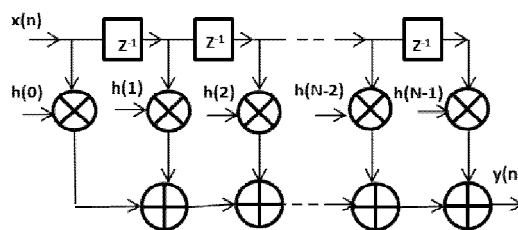


Fig. 2. N-order FIR digital filter block diagram

The basic structure of N- order FIR digital filter is shown in Figure 2. I can express the output signal in frequency domain by convolution of the input signal  $x(n)$  and the impulse response  $h(n)$ .

$$y(n)=x(n)*h(n) \quad (2)$$

The output signal is determined as,

$$y(n) = x(0) * h(n) + x(1) * h(n-1) + x(2) * h(n-2) + \dots$$

$$\dots + x(n) * h(0) \quad (3)$$



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Vol. 6, Issue 7, July 2018

The coefficient  $b_k$  in equation (1) equals to the successive value  $h(n)$  of unit-sample response. The system function  $H(z)$  can be expressed in the following form:

$$H(Z) = \sum_{K=0}^{N-1} (b_K Z^{-1}) \quad (4)$$

$H(z)$  is a polynomial of  $z^{-1}$ . This means that all poles are only plotted at the origin of the  $Z$ -plane.

## B. The methods of multiband FIR digital filter

There are total six different methods for multiband FIR digital filter design based on MATLAB simulation FDATool.

- *Least Square Multiband FIR Digital Filter*: It designs a linear-phase FIR filter that minimizes the weighted, integrated squared error between an ideal piecewise linear function and the magnitude response of the filter over a set of desired frequency bands.
- *Equiripple Multiband FIR Digital Filter*: It designs a linear-phase FIR filter using the Parks-McClellan algorithm [5]. The Parks-McClellan algorithm uses the Remez exchange algorithm and Chebyshev approximation theory to design filters with an optimal fit between the desired and actual frequency responses.
- *Constrained least Squares Multiband FIR Digital Filter*: The Constrained Least Squares (CLS) FIR filter design functions implement a technique that enables us to design FIR filters without explicitly defining the transition bands for the magnitude response. The ability to omit the specification of transition bands is useful in several situations. For example, it may not be clear where a rigidly defined transition band should appear if noise and signal information appear together in the same frequency band. Similarly, it may make sense to omit the specification of transition bands if they appear only to control the results of Gibbs phenomena that appear in the filter's response. See Selesnick, Lang, and Burrus [6] for discussion of this method. The key feature of the CLS method is that it enables us to define upper and lower thresholds that contain the maximum allowable ripple in the magnitude response. Given this constraint, the technique applies the least square error minimization technique over the frequency range of the filter's response, instead of over specific bands. The error minimization includes any areas of discontinuity in the ideal, "brick wall" response. An additional benefit is that the technique enables us to specify arbitrarily small peaks resulting from Gibbs' phenomena.
- *Constrained band Equiripple Multiband FIR Digital Filter*: It designs equiripple FIR filters while constraining one or more bands of the filter to have a maximum ripple value.
- *Complex Equiripple Multiband FIR Digital Filter*: It is Complex and nonlinear phase equiripple FIR filter design. It allows arbitrary frequency-domain constraints to be specified for the design of a possibly complex FIR filter. The Chebyshev (or minimax) filter error is optimized, producing equiripple FIR filter designs.
- *Generalized Equiripple Multiband FIR Digital Filter*: It is a mini max filter design algorithm that can be used to design of real FIR filters: types 1-4 linear phase, minimum phase, maximum phase, minimum order (even or odd), extra ripple, maximal ripple, constrained ripple, single-point band (notching and peaking), forced gain and arbitrarily shaped filters

## C. FIR filter Design

The frequency spectrum is divided into bands and channels. Specifically, the frequency band is 0-30 Hz wide which contain several channels. A channel is used to refer to the frequency range specified by each Schumann resonance mode.

1) *Multiband FIR Filter Design using FDATool*: The Filter Design and Analysis Tool (FDATool) is a effective user interface for designing and analysing filters quickly. FDATool enables us to design digital FIR or IIR filters by setting filter specifications, by importing filters from a MATLAB workspace, or by adding, moving or deleting poles and zeros. FDATool also provides various window functions, designing functions or filter and realizable functions of filter. Multiband option is the most commonly used method to design only the FIR filter. FDATool has very important function in the design of multiband FIR filters. It design linear phase filters only. The coefficients or 'taps' of such filters obey either an even or odd symmetry relation. Depending on this symmetry, and on whether the order  $n$  of the filter is even or odd, a linear phase filter (stored in length  $n+1$  vector  $b$ ) has certain inherent restrictions on its frequency response. The phase delay and group delay of such linear phase FIR filters are equal and constant over the frequency band. For an order  $n$  linear phase FIR filter, the group delay is  $n/2$ , and the filtered signal is simply delayed by  $n/2$  time's steps. This property preserves the wave shape of signals in the passband because there is no phase distortion. The performance of multiband digital filter design can be improved, if the proper multiband FIR function is chosen [7].

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Vol. 6, Issue 7, July 2018

In this paper, I design and compare Equiripple type, Least square type, Constrained least square type and Complex equiripple type of multiband FIR filters for frequency band 0 – 30 Hz wide which contained four channels of each 4 Hz passband corresponding to first four Schumann resonance (SR) signal modes with a cut-off frequency of 8, 14, 20 and 26 Hertz. In the filter design, we set filter specifications as low filter order equal to 32 for preserves the wave shape of SR signal in passband, normalize the frequency vector in 0-1 Hz by dividing 30 Hz corresponding to vector of band edges and specify the maximum amount of ripple for each band corresponding to SR signal modes. We start design of multiband FIR digital filter with lowest feasible filter order, suitable vector of frequency, vector of amplitude and vector of weights for each type of multiband FIR digital filter. We implement our each type of multiband FIR digital filter in Schumann resonance raw data analysis software and measure the Schumann resonance frequencies and corresponding intensities of each mode at same instant, repeat the same designing process and measurement process after increasing the filter order by one at a time and continue till last as feasible. In this paper, we vary the filter order from 25 to 55, and measure the Schumann resonance frequencies and corresponding intensities of each mode at same instant. We plot the first mode of SR frequencies and SR intensities with corresponding filter order for each type of multiband FIR digital filter and compared our measurement with true values of first mode SR frequency and SR intensity, which measure by other verified method.

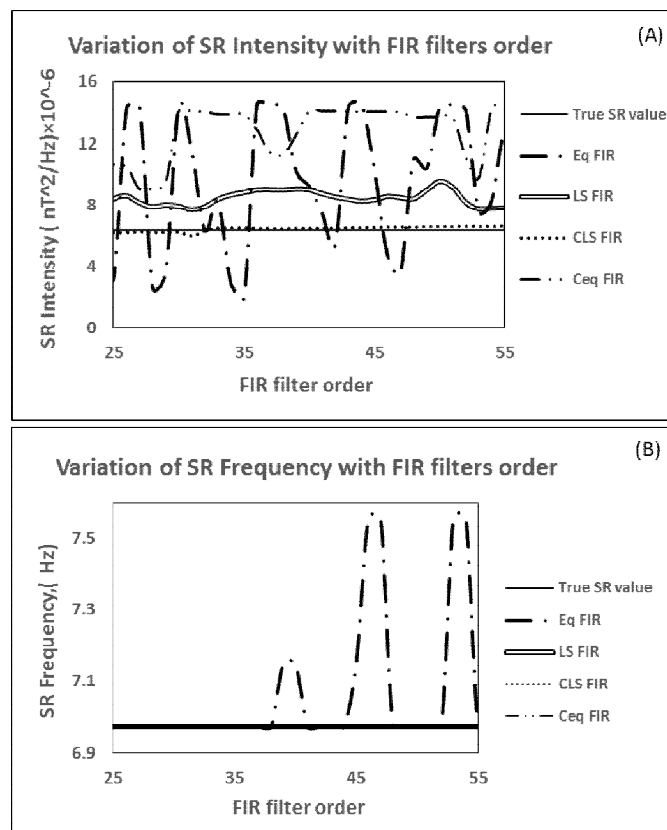


Fig. 3. Variation of first mode SR values measured by multiband FIR digital filters vs filter order (A) SR intensities (B) SR frequencies

From the Fig. (3A), it is clear that measurement of first mode SR intensities are nearly equal to true value only for constrained least squares multiband FIR digital filter (dotted line) and measurement error is maximum for equiripple multiband FIR digital filter (dark thick line-single dot). From the Fig. (3B), it is clear that measurement of first mode SR frequencies are equal to true value for constrained least squares multiband FIR digital filter (dotted line) and measurement error is maximum for equiripple multiband FIR digital filter (dark thick line-single dot). From Fig. (4), it is very clear that measurement error (difference between true value and measure value) of first mode SR

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Vol. 6, Issue 7, July 2018

intensities is minimum for constrained least squares FIR digital filter with filter order equal to 32. Hence it is clear from Fig. (3 & 4) that constrained least squares multiband FIR digital filter with filter order equal to 32 is the best choice to design for our Schumann resonance raw data analysis. The output characteristic of constrained least squares FIR digital filter

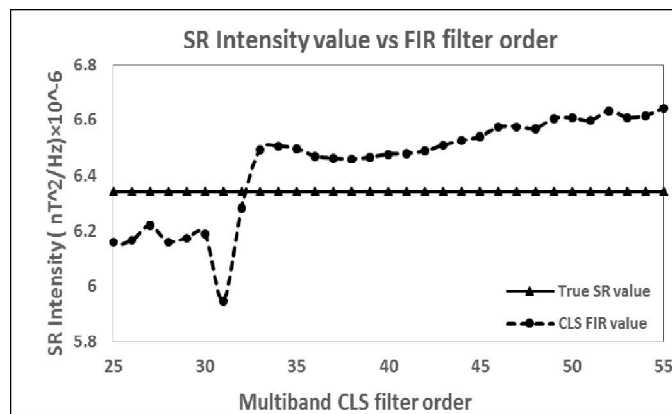


Fig. 4. Variation of first mode SR intensity values measured by multiband constrained least squares FIR digital filters vs filter order

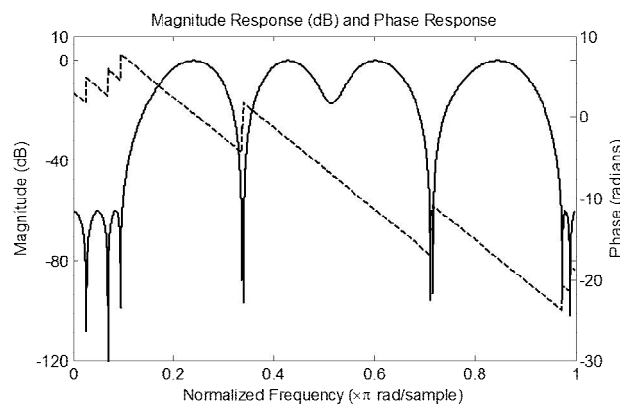


Fig. 5. Magnitude response (dB) and Phase response (radian) of multiband constrained least squares FIR digital filter order of 32

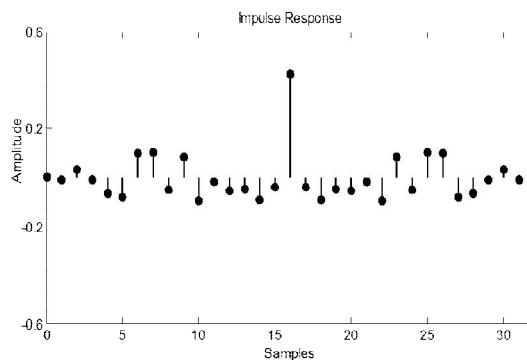


Fig. 6. Impulse response of multiband constrained least squares FIR digital filter order of 32

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Vol. 6, Issue 7, July 2018

design can be understood by performance plots as magnitude plot, phase plot, impulse response plot, step response plot and pole-zero plot (Figure 5, 6, 7 and 8). We use FDATool to produce all performance plots. Notice that the phase response is linear within the multiband. The group delay is defined as the gradient of the phase with respect to frequency. Constant group delay of value 16 (half of filter order) is remain constant. Fig. (8) shown the z-plane pole-

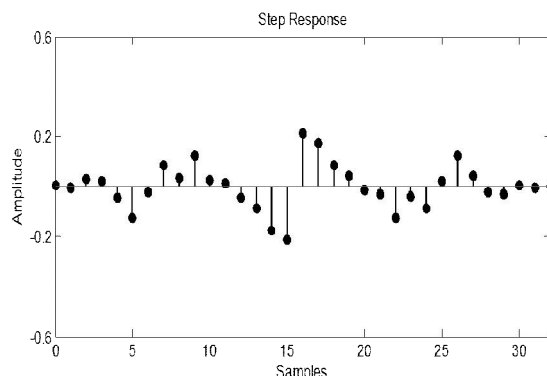


Fig. 7. Step response of multiband constrained least squares FIR digital filter order of 32

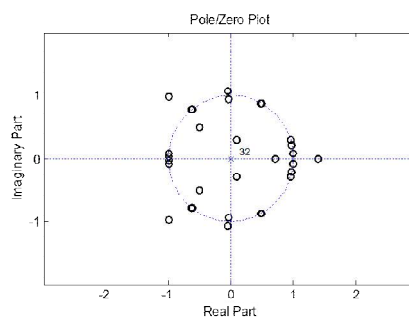


Fig. 8. Pole Zero plot of multiband constrained least squares FIR digital filter order of 32

zero map of the filters. Notice that MATLAB has placed 32 poles at the origin ( $p=0$ ). These poles are infinitely fast ( $s = -\infty \rightarrow z = s^j = 0$ ) and are therefore essentially ineffective.

2) *Constrained Least Squares Multiband FIR Filter Design using FDATool*: In this paper, we design finally constrained least squares multiband FIR filter using FDATool. Digital filter design problem involves the determination of a set of filter coefficients to meet a set design specifications. These specifications typically consist of the width of the pass band and the corresponding gain, the width of the stop band(s) and the attenuation therein; the band edge frequencies (which give an indication of the transition band) and the peak ripple tolerable in the pass band and stop band(s). There are many techniques for selecting coefficients. We can use a spreadsheet like Microsoft Excel, or there are many design packages which can do the job. I have used MATLAB for this purpose as it is the most advanced tool for DSP applications. I have designed FIR constrained least-squares Multiband filter using FDATool of M-file generated by MATLAB (R2013a) and the signal processing Toolbox 6.2. FDATool provides tools for analysing filters, such as magnitude and phase response, impulse response, step

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Vol. 6, Issue 7, July 2018

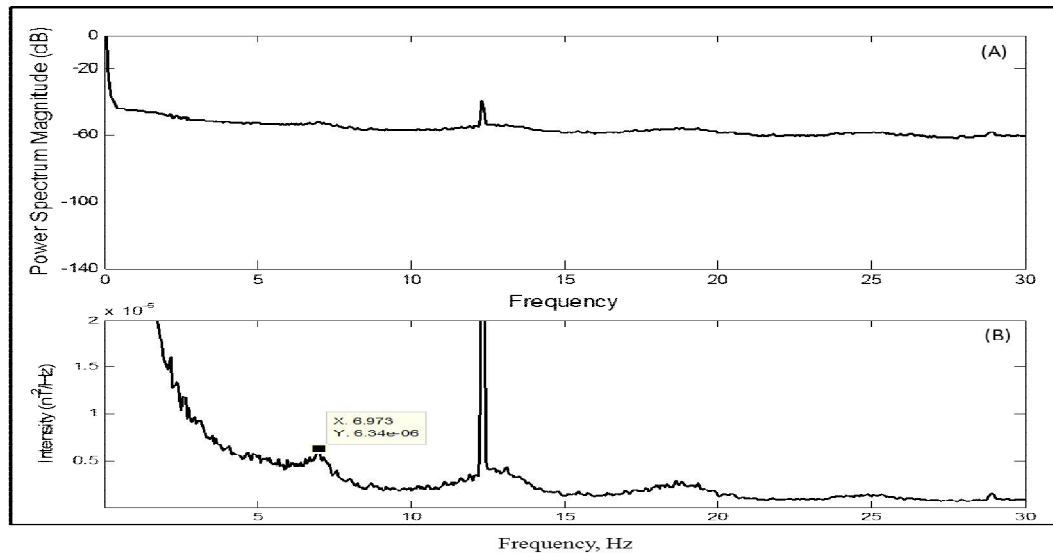


Fig. 9. Original SR signal before filtering (A) Power spectrum magnitude (dB) , (B) Intensity ( $nT^2/Hz$ ) plot vs frequency

response and pole-zero plots. During design process, first we open FDATool in command window, then select multiband type and and insert values of frequency vector, amplitude vector, upper vector, lower vector and filter order. The combination of the length of the filter (number of taps) and the values of the coefficients determine the filter's operation. Designing the filter is just a case of deciding how many taps and choosing the coefficients [8]. We design a direct form FIR stable filter with impulse response order 32 to meet our output, other filter design information's are given in table1. We implement our filter on the SR signal. During design process, we compare the first few filter coefficients ( $b_{25}$ ,  $b_{26}$ ,  $b_{27}$ , . . .) with the last ones (. . .  $b_{54}$ ,  $b_{55}$ ) by export our coefficients to the MATLAB workspace.

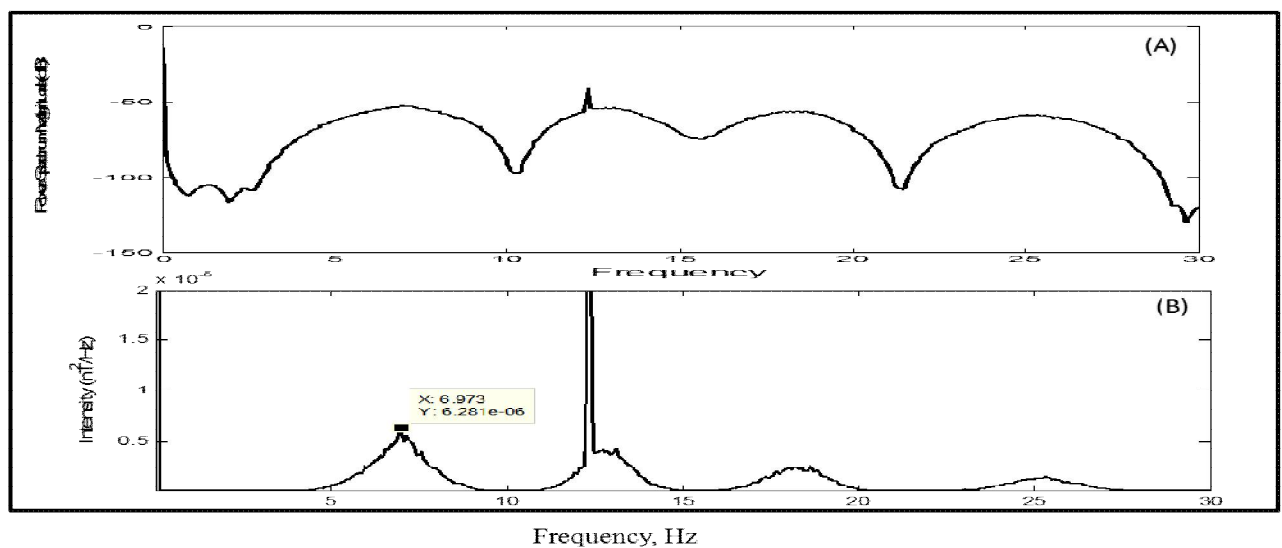


Fig. 10. SR signal after filtering by multiband constrained least squares FIR digital filter (A) Power spectrum magnitude (dB) , (B) Intensity ( $nT^2/Hz$ ) plot corresponding PSD vs frequency (Hz)

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Vol. 6, Issue 7, July 2018

TABLE 1 FIR filter information

Discrete- Time FIR Filter (real)	
Filter Structure	Direct-Form FIR
Filter Length	33
Stable	Yes
Linear Phase	Yes (Type 1)
Number of Multipliers	33
Number of Adders	32
Number of States	32
Multiplications per Input Sample	33
Additions per Input Sample	32

## D. Simulation Result and Discussion

Figure(9A) shows the original power spectrum magnitude (dB) output of Schumann resonance (SR) signal before filter at a particular instant and figure (9B) shows the intensity ( $\text{nT}^2/\text{Hz}$ ) plot corresponding to figure (9A) to reproduce each modes of Schumann resonance. Figure (10A) shows the power spectrum magnitude (dB) output of Schumann resonance (SR) signal corresponding to multiband FIR digital filtered by constrained least square type in MATLAB. Figure (10B) shows the intensity ( $\text{nT}^2/\text{Hz}$ ) plot corresponding to figure (10A) to reproduce each modes of Schumann resonance after filtering. From figures (9B & 10B), it is clear that measurement of first mode SR frequency by multiband constrained least squares FIR filter method is same equal to 6.973 Hz as measured without filter method, and the measurement of first mode SR intensity by multiband constrained least squares FIR filter method is  $6.281 \times 10^{-06} \text{ nT}^2/\text{Hz}$  which is approximately equal to as measured without filter method ( $6.34 \times 10^{-06} \text{ nT}^2/\text{Hz}$ ). Figure (10B) shows that the wave shapes of output SR signal of each channel due to constrained least square multiband FIR filter is exactly same as input wave shape of SR signal (figure 9B) due to negligible distortion. Low order FIR method for analysis the characteristic of Schumann resonance is also recommended by author Satori and Zieger (page no. 29667) [9]. During Schumann resonance (SR) analysis original wave shape is essential to give accurate information of frequency and intensity corresponding to each mode of Schumann resonance. In figure (9, 10 and 11A), there is a big sharp noise at 12.15 Hz due to local interference(like train noise) in second pass band corresponding to second mode of SR signal.

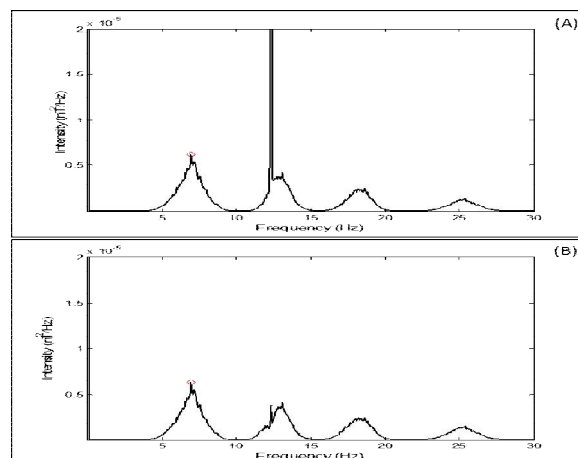


Fig. 11. SR intensity ( $\text{nT}^2/\text{Hz}$ ) plots vs frequency (A) before filtering with big sharp noise at 12.19 Hz (B) after filtering by IIR notch filter

I also design and implement the IIR notch filter to remove such big sharp peak noise in the SR signal. Figure (11B) shows filtered output signal of IIR notch filter. Hence we can remove such type of sharp noise from SR signal.





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Vol. 6, Issue 7, July 2018

## E. Conclusion

the noise present in Schumann resonance (SR) signal lead to improper analysis so the multiband FIR digital filter can be used to remove these noise. In this paper, we compare our result output for each as Equiripple type, Least square type, Constrained least square type and Complex equiripple type of multiband FIR filter using same FDATool in MATLAB. The FIR filter has the following advantages:

- FIR filter are always stable as they have no recursive structure.
- They gave the exactly linear phase.
- Efficiently realizable in hardware.
- Filter response is of finite duration.

Thus noise removal using FIR digital filter is better option in MATLAB. Thus it is clear from the results that constrained least squares multiband FIR digital filter is the best option as compared to other multiband FIR digital filters due to sufficient noise reduction with minimum distortion of wave shape with minimum order, linear phase within the multiband pass as well as their unconditioned stability and robustness. Moreover, it is observed that there is sufficient scope for more work on complexity reduction by further reducing the order of the filter.

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