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Design and Implementation of Fir Filter Using Window techniques for Suppressed of Power Line Interferences

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Abstract: In this paper we introduce the design of FIR filter using a Different Window technique which helps to increase the Stability and reduce the requirement of component than other designing technique. Using a window greatly reduces the ringing. This improvement is at the expense of transition width (the windowed version takes longer to ramp from passband to stopband) and optimality (the windowed version does not minimize the integrated squared error). The simulation is done in Matlab 7 and it can be observed that the minimization of the sidelobes increase the efficiency of the filtering process as well as decreasing the power consumption.

Key words: Digital filters, FIR filters, Different Window Technique, Matlab

I. Introduction

A signal in the communication system is the information containing part which is to be process, but during the processing of the signal some noise is added in the signal and signal becomes noisy. This is now mandatory to eliminate this noise from the signal to get information from the signal. a wavelet filter based on Butterworth IIR filter and Kaiser Window FIR filter is designed for the signal analysis[1]. the design of maximally flat linear phase finite impulse response (FIR) filters is handled with totally two different approaches. The first one is completely deterministic numerical approach where the problem is formulated as a Linear Complimentarily Problem (LCP). The other one is based on a combination of Markov Random Fields (MRF's) approach with messy genetic algorithm (MGA) [2].several classes of maximally flat filters are unified under a single formula. The generating function of the filters is also derived. This enables us to develop multiplier less cellular array structures for exact realization of a subset of the filters [3]. In Digital filters, a novel method is proposed for designing maximally flat IIR filters with flat group delay responses in the pass band. First, systems of linear equations are derived from the flatness conditions of IIR filters given in the pass band and stop band, respectively. Then, a set of filter coefficients can be easily obtained by simply solving this system of linear equations [4]. Author presents some new explicit expressions differentiating Hilbert transformers. The proposed closed-form design is based on the full band least-squares differentiator and relations between differentiator and Hilbert transformer. The obtained simple formulas give an efficient way to determine tap-coefficients of designed Hilbert transformers even with a hand calculator [5]. Gibb's phenomenon is reduced in the filter implementation thus giving a flat pass band and good stop band attenuation. A closed form expression for impulse response coefficients is obtained. The filter design is easily tunable and allows for variation in transition bandwidth of each band. A speech processing scheme is implemented using a pair of the proposed sharp transition multiband FIR filters to split the speech spectrum into complementary short time spectral bands [6]. Author filter bank having good frequency response, though it is based on linear equation. It can lead a filter bank without multiplier, if we want such a bank [7]. Blackman window function is used to design an FIR filter for efficient value of α, this window function provide higher side lobe attenuation comparisons to hamming and hanning window and main lobe width of this window function is slightly greater than hamming window [8]. In FIR filter make transfer function of HTs, DDs and FDs by expanding some suitable function in power series .fir HTs based on expanding the signum function into power series MFs FIR DDs are designed by expanding some inverse triangular function into power series. FIR FDs are designed by expanding the ideal transfer function into Binomial series [9]. Optimization technique are used reduce the error which cause by frequency sampling technique at the non sampled frequency point [10]. Fir filter are also design either by software VLSI technique or minimum hardware solution to improve the linear phase response [11]. FIR filter can be realized either recursively or non recursively .Recursively realization of frequency sampling design are simple to program and can be very efficient .Non recursive realization include direct convolution and fast convolution [12]. Algorithm independent lower bound on the achievable approximation error and then present an Approximation method involve the solution of a fixed number of all pass extension problem so is called the Nehari shuffle [13]. Author use the simplex algorithm for linear programming to find the linear phase filter of minimum length which limit on the frequency response and maximize the distance from the constraints [14]. Design of Analog and Digital filter using Matlab special importance is the pole placer function that is used to synthesis filter with ER and MF pass band and arbitrary piece wire constant stop band [15]. This paper is organized as follows. In Section 2, we propose a design of the FIR based on Window technique such as Rectangular window ,Hamming Window ,Hanning Window and Triangular window .In section 3 FIR filter are design in Matlab and represent characteristics of Low pass filter ,High pass filter ,Band stop filter , and Band pass filter using different window technique. A final conclusion discussed in Section4.

II. Window Technique

2.1 FIR filter approximation with window functions

A straight forward way to overcome this limitation is to define a finite-length auxiliary sequence h'(n), yielding a filter of order M, as

$$h'(n) = \begin{cases} h(n), & for\ |n| \leq \frac{M}{2} \\ 0, & for\ |n| > \frac{M}{2} \end{cases} \quad [1]$$

assuming that M is even. The resulting transfer function is written as

$$H'(z) = h(0) + \sum_{n=1}^{\frac{M}{2}} \left(h(-n)z^{n} + h(n)z^{-n}{}_{n}\right) \quad [2]$$

This is still a non causal function which we can make causal by multiplying it by$Z^{\frac{-M}{2}}$, without either distorting the filter magnitude response or destroying the linear-phase property. The example below highlights some of the impacts that the truncation of the impulse response in equations (1) and (2) has on the filter frequency response. The ripple seen in Figure 1.close to the band edges is due to the slow convergence of the Fourier series h(n) when approximating functions presenting discontinuities, such as the ideal responses seen in Figure This implies that large amplitude ripples in the magnitude response appear close to the edges whenever an infinite-length h(n) is truncated to generate a finite-length filter. These ripples are commonly referred to as Gibbs' oscillations. It can be shown that Gibbs' oscillations possess the property that their amplitudes do not decrease even when the filter order M is increased dramatically. This severely limits the practical usefulness of equations (1) and (2) in FIR design, because the maximum deviation from the ideal magnitude response cannot be minimized by increasing the filter length. Although we can not remove the ripples introduced by the poor convergence of the Fourier series, we can

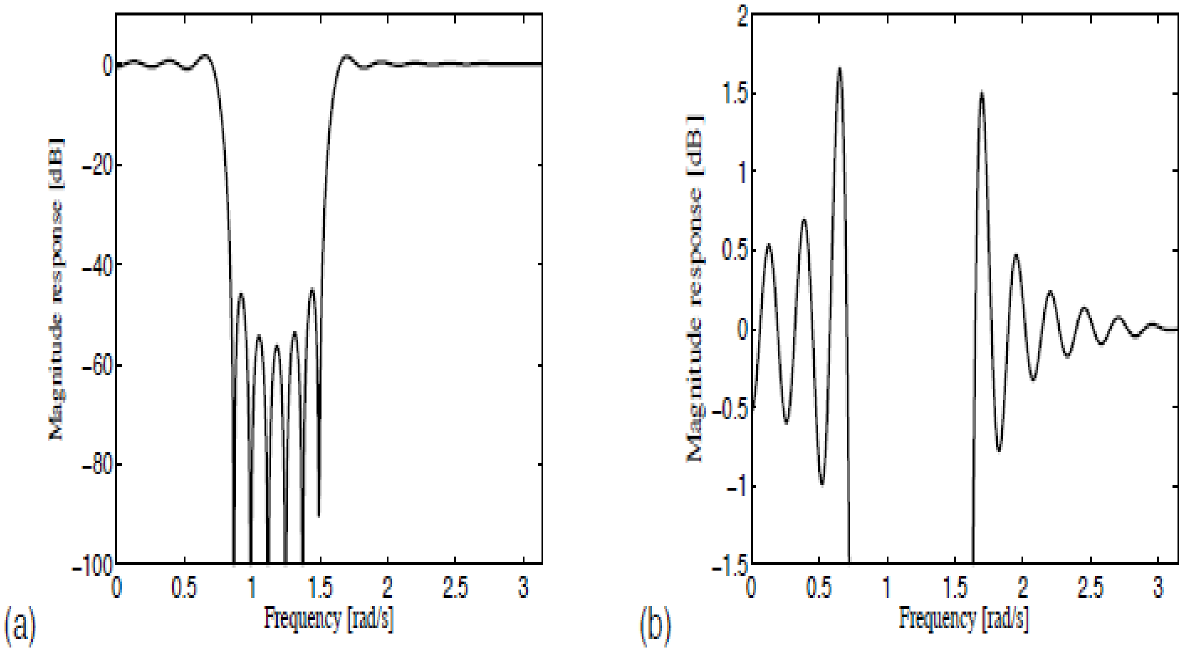


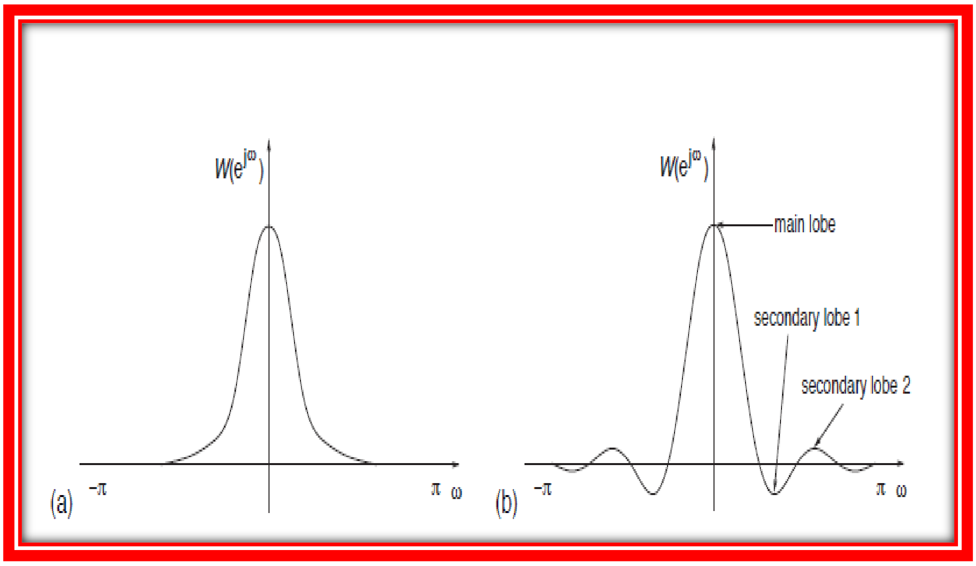
Figure 1;Band stop filter : (a) magnitude response ; (b) pass band detail.

still attempt to control their amplitude by multiplying the impulse response h(n) by a window function w(n). The window w(n) must be designed such that it introduces minimum deviation from the ideal frequency response. The coefficients of the resulting impulse response h'(n) become

$$h'(n) = h(n)w(n)$$

In the frequency domain, such a multiplication corresponds to a periodic convolution operation between the frequency responses of the ideal filter, H ($e^{j\omega}$), and of the window function, W ($e^{j\omega}$), that is

$$H\left(e^{j\omega}\right) = \frac{1}{2\pi}\int_{-\pi}^{\pi} H\left(e^{jw'}\right) W\left(e^{j(w-w')}\right) dw'$$



Fiqure 2: Magnitude response of a window function : (a) ideal case; (b) practical case.

2.1 Rectangular window

A simple truncation of the impulse response as described in equation (1) can be Interpreted as the product between the ideal h (n) and a window given by

$$W_r(n) = \begin{cases} 1, & for\ |n| \le \frac{M}{2} \\ 0, & for\ |n| > \frac{M}{2} \end{cases} \qquad (3)$$

Note that if we want to truncate the impulse responses using the above equation and still keep the linear-phase property, the resulting truncated sequences would have to be either symmetric or antisymmetric around n = 0.
For the case of M odd, the solution would be to shift h(n) so that it is causal and apply a window different from zero from n = 0 to n = M− 1. This solution, however, is not commonly used in practice from equation (3); the frequency response of the rectangular window is given by

$$\begin{aligned} W_r(e^{jw}) &= \sum_{n=-\frac{M}{2}}^{\frac{M}{2}} e^{-jwn} \\ &= \frac{e^{jw\frac{M}{2}} - e^{-jw\frac{M}{2}} e^{-jw}}{1 - e^{-jw}} \\ &= e^{-j\frac{w}{2}} \frac{e^{jw\frac{M+1}{2}} - e^{-jw\frac{M+1}{2}} e^{-jw}}{1 - e^{-jw}} \\ &= \frac{\sin\left[\omega\left(\frac{M+1}{2}\right)\right]}{sin\left(\frac{\omega}{2}\right)} \end{aligned}$$

….. [4]

2.2 Triangular windows

The main problem associated with the rectangular window is the presence of ripples near the band edges of the resulting filter, which are caused by the existence of side lobes in the frequency response of the window. Such a problem is due to the inherent discontinuity of the rectangular window in the time domain. One way to reduce such a discontinuity is to employ a triangular-shaped window, which will present only small discontinuities near its edges.
The standard triangular window is defined as

$$W_t(n) = \begin{cases} \frac{2|n|}{M+2} + 1, & for\ |n| \le \frac{M}{2} \\ 0, & for\ |n| > \frac{M}{2} \end{cases} \qquad [5]$$

2.3 Hamming and Hanning windows

The generalized Hamming window is defined as

$$W_H(n) = \begin{cases} \alpha + (1-\alpha)\cos\left(\frac{2\pi}{M}\right), & for\ |n| \le \frac{M}{2} \\ 0, & for\ |n| > \frac{M}{2} \end{cases} \qquad [6]$$

With $0 \le \alpha \le 1$.

This generalized window is referred to as the Hamming window when α = 0.54, and for α = 0.5, it is known as the Hanning .The frequency response for the general Hamming window can be expressed based on the frequency response of the rectangular window. We first write equation (6) as

$$W_H(n) \;=\; W_r(n)\ \left[\alpha + (1-\alpha)\cos\left(\frac{2\pi}{M}\right)\right] \qquad (7)$$

By transforming the above equation to the frequency domain, clearly the frequency response of the generalized Hamming window results from the periodic convolution between $W_r(e^{jw})$ and three impulse functions as

$$W_H(e^{jw}) \;=\; W_r(e^{jw}) * \left[\alpha\delta(\omega) \;+\; \left(\frac{1-\alpha}{2}\right)\delta\left(\omega - \frac{2\pi}{M}\right) + \left(\frac{1-\alpha}{2}\right)\delta\left(\omega + \frac{2\pi}{M}\right)\right]$$

And then

$$W_H(e^{jw}) \;= \alpha\, W_r(e^{jw}) \;+\; \left(\frac{1-\alpha}{2}\right) W_r(e^{j\left(\omega-\frac{2\pi}{M}\right)}) \left(\frac{1-\alpha}{2}\right) W_r(e^{j\left(\omega+\frac{2\pi}{M}\right)})$$

From this equation, one notices that $W_H(e^{jw})$ is composed of three versions of the rectangular spectrum $W_r(e^{jw})$ the main component, α $W_r(e^{jw})$, centered at ω = 0, and two additional ones with smaller amplitudes, centered at ω = ±2π/M, that reduce the secondary lobe of the main component.

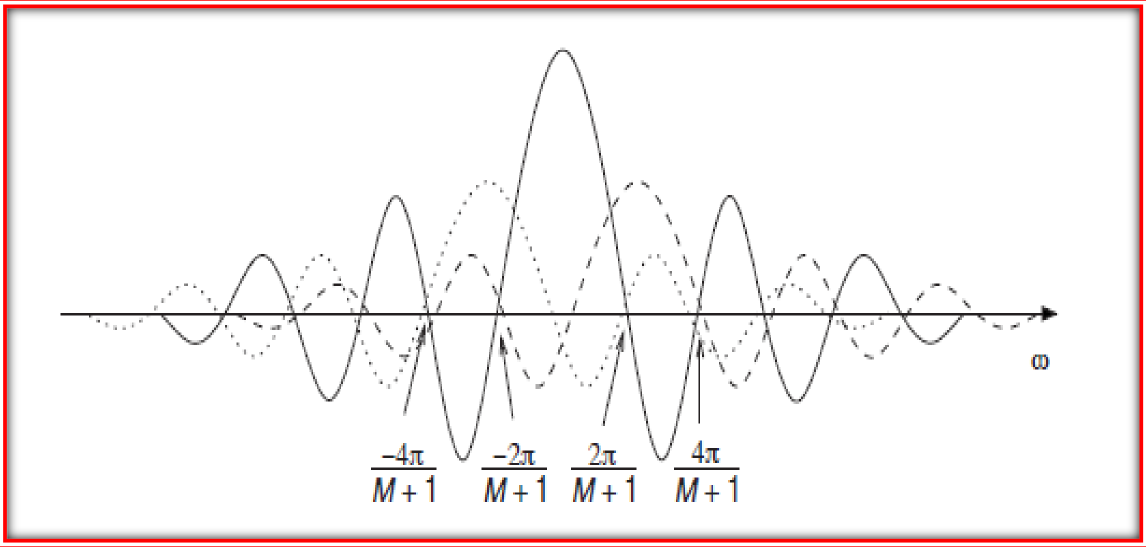


Figure 3: The three components of the generalized Hamming window combine to reduce the resulting secondary lobes. (Solid line – $\alpha\, W_r(e^{jw})$); dashed line –

$$\frac{1-\alpha}{2} W_r\left(e^{j\left(\omega-\frac{\pi}{M}\right)}\right);\ \text{dotted line}\ -\frac{1-\alpha}{2} W_r\left(e^{j\left(\omega+\frac{\pi}{M}\right)}\right) \qquad [8]$$

The main characteristics of the generalized Hamming window are:

(i) All three $W_r(e^{jw})$ components have zeros close to ω = $\pm\frac{4\pi}{M+1}$. Hence, the main-lobe total width is $\frac{8\pi}{M+1}$.
(ii) When α = 0.54, the main-lobe total energy is approximately 99.96% of the window total energy.
(iii) The transition band of the Hamming window is larger than the transition band of the rectangular window, due to its wider main lobe.
(iv) The ratio between the amplitudes of the main and secondary lobes of the Hamming window is much larger than for the rectangular window.
(v) The stop band attenuation for the Hamming window is larger than the attenuation for the rectangular window.

III. Matlab Work

Designing of FIR filter based on different window technique:

```
% LPF stands for Low pass filter
% HPF stands for High pass filter
% BPF stands for Band pass filter
% BRF stands for Band reject filter
% Rectangular, Hanning and hamming are different types of Windowing
% techniques
clear all
clc;
w=0:pi/20:pi;
q=1:1:7

%plottting different types of windows
%rectangular window
for j=1:1:7
   rect(j)=1;
end
plot(q,rect,'r*')
xlabel('time (seconds)')
ylabel('Magnitue')
title('\it{Plotting rectangular window}',' )
text(2.5,1.5,'rect',' )

%Hanning window
for j=1:1:7
   hanning(j)=(1/2)*(1-cos((pi/3)*j));
end
0plot(q,hanning,'g')
xlabel('time (seconds)')
ylabel('Magnitue')
title('\it{Plotting hanning window}',' )
text(5,0.8,'hanning',' )

%Hamming window
for j=1:1:7
   hamming(j)=0.54-0.46*cos((pi/3)*j);
end
plot(q,hamming,'b')
xlabel('time (seconds)')
ylabel('Magnitue')
title('\it{Plotting hamming window}',' )
text(5,0.8,'hamming',' )

%plotting LPF using different windows
rectangular
for i=0:pi/20:pi
lpf_rect((i)*(20/pi)+1)=0.318+2*(0.268*cos(i)+0.145*cos(2*i)+0.015*cos(3*i));
end
plot(w,lpf_rect,'r*')      % rectangular window
xlabel('Frequency (Hz)')
ylabel('Gain')
title('\it{Plotting LPF using rectangular window}',' )
text(2,1,'lpfrect',' )


%plotting LPF hanning windows
for i=0:pi/20:pi
lpf_hanning((i)*(20/pi)+1)=0.318+2*(0.201*cos(i)+0.036*cos(2*i));
end
```



```
plot(w,lpf_hanning,'g*')      % Hanning window
xlabel('Frequency (Hz)')
ylabel('Gain')
title('\it{Plotting LPF using hanning window}',' )
text(2,0.7,'lpfhanning',' )
```

%plotting LPF hamming windows

```
for i=0:pi/20:pi
lpf_hamming((i)*(20/pi)+1)=0.318+2*(0.206*cos(i)+0.045*cos(2*i)+0.012*cos(3*i));
end
plot(w,lpf_hamming,'b*')        % Hamming window
xlabel('Frequency (Hz)')
ylabel('Gain')
title('\it{Plotting LPF using hamming window}',' )
text(2,0.75,'lpfhamming',' )
```

%plotting HPF using different windows
Rectangular window

```
for i=0:pi/20:pi
  hpf_rect((i)*(20/pi)+1)=0.682-2*(0.268*cos(i)+0.145*cos(2*i)+0.015*cos(3*i));
end
plot(w,hpf_rect,'r*')      % rectangular window
xlabel('Frequency (Hz)')
ylabel('Gain')
title('\it{Plotting HPF using rectangular window}',' )
text(0.3,1.1,'hpfrect',' )
```

hanning window

```
for i=0:pi/20:pi
hpf_hanning((i)*(20/pi)+1)=0.682-2*(0.201*cos(i)+0.036*cos(2*i));
end
plot(w,hpf_hanning,'g*')      % Hanning window
xlabel('Frequency (Hz)')
ylabel('Gain')
title('\it{Plotting HPF using hanning window}',' )
text(0.5,1,' hpfhanning',' )
```

hamming window

```
for i=0:pi/20:pi
hpf_hamming((i)*(20/pi)+1)=0.682-2*(0.206*cos(i)+0.045*cos(2*i)+0.012*cos(3*i));
end
plot(w, hpf_hamming,'b*')        % Hamming window
xlabel('Frequency (Hz)')
ylabel('Gain')
title('\it{Plotting HPF using hamming window}',' )
text(0.5,1,'hpfhamming',' )
```

%plotting BPF using different windows
Rectangular window

```
for i=0:pi/20:pi
  bpf_rect((i)*(20/pi)+1)=0.318-2*(0.022*cos(i)+0.265*cos(2*i)+0.045*cos(3*i));
end
plot(w,bpf_rect,'r*')      % rectangular window
xlabel('Frequency (Hz)')
ylabel('Gain')
title('\it{Plotting BPF using rectangular window}',' )
text(2,0.8,'bpfrect',' )
```

hanning window

```
for i=0:pi/20:pi
bpf_hanning((i)*(20/pi)+1)=0.318-2*(0.0165*cos(i)+0.066*cos(2*i));
end
plot(w,bpf_hanning,'g*')      % Hanning window
```



```
xlabel('Frequency (Hz)')
ylabel('Gain')
title('\it{Plotting BPF using hanning window}',' )
text(2.1,0.48,'bpfhanning',' )
```

hamming window

```
for i=0:pi/20:pi
bpf_hamming((i)*(20/pi)+1)=0.318-2*(0.0169*cos(i)+0.082*cos(2*i)+0.0036*cos(3*i));
end
plot(w,bpf_hamming,'b*')        % Hamming window
xlabel('Frequency (Hz)')
ylabel('Gain')
title('\it{Plotting BPF using hamming window}',' )
text(2.1,0.48,'bpfhamming',' )
```

%plotting BRF using different awindows

rectangular

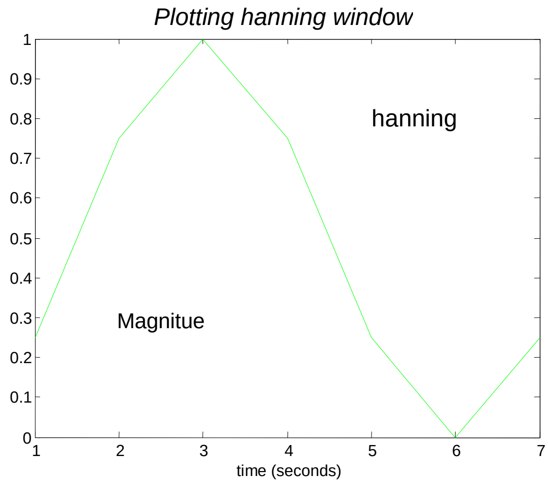
```
for i=0:pi/20:pi
brf_rect((i)*(20/pi)+1)=0.318+2*(0.022*cos(i)+0.265*cos(2*i)+0.045*cos(3*i));
end
plot(w,brf_rect,'r*')       % rectangular window
xlabel('Frequency (Hz)')
ylabel('Gain')
title('\it{Plotting BRF using rectangular window}',' )
text(2,0.8,'brfrect',' )
```

hanning window

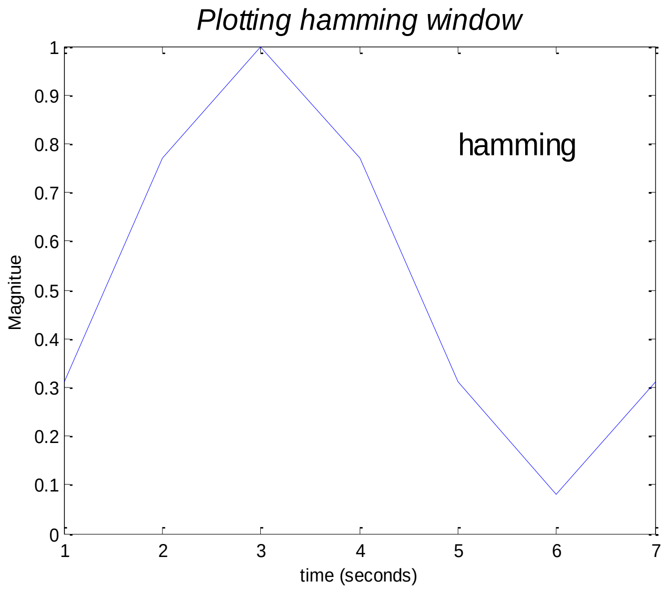
```
for i=0:pi/20:pi
brf_hanning((i)*(20/pi)+1)=0.318+2*(0.0165*cos(i)+0.066*cos(2*i));
end
plot(w,brf_hanning,'g*')      % Hanning window
xlabel('Frequency (Hz)')
ylabel('Gain')
title('\it{Plotting BRF using hanning window}',' )
text(2.1,0.48,'brfhanning',' )
```

hamming window

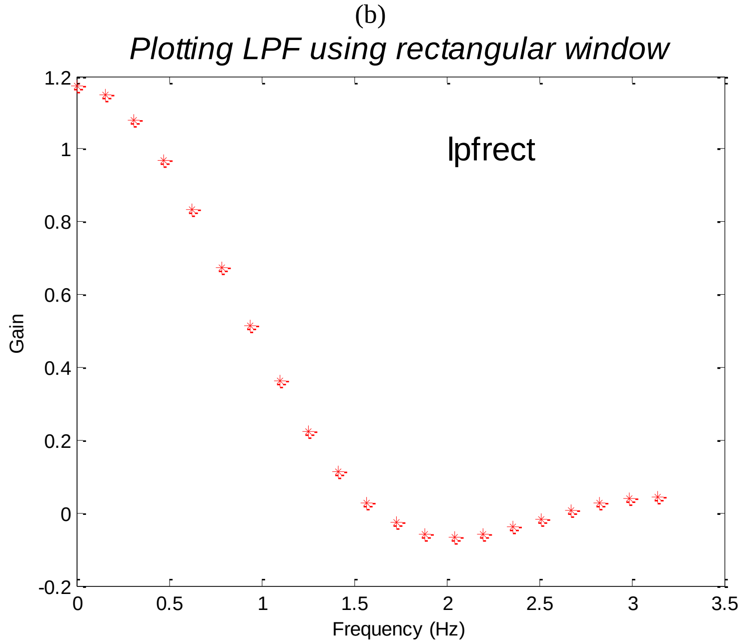
```
for i=0:pi/20:pi
brf_hamming((i)*(20/pi)+1)=0.318+2*(0.0169*cos(i)+0.082*cos(2*i)+0.0036*cos(3*i));
end
plot(w,brf_hamming,'b*')        % Hamming window
xlabel('Frequency (Hz)')
ylabel('Gain')
title('\it{Plotting BRF using hamming window}',' )
text(2.1,0.48,' brfhamming',' )
```



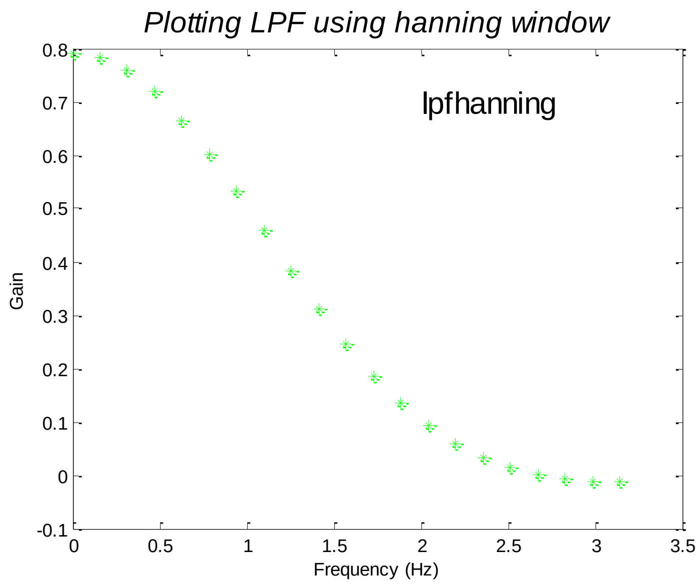
(a)


Plotting hamming window
hamming
Magnitue
time (seconds)

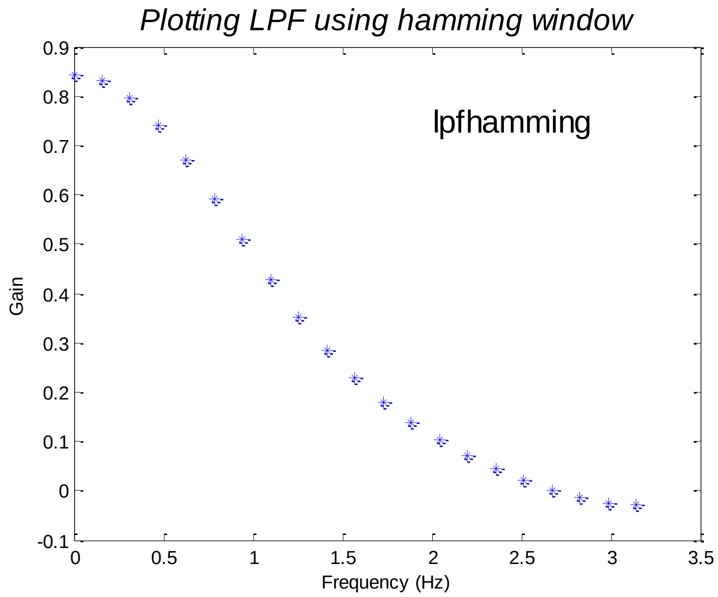
(b)


Plotting LPF using rectangular window
lpfrect
Gain
Frequency (Hz)

(c)


Plotting LPF using hanning window
lpfhanning
Gain
Frequency (Hz)

(d)


Plotting LPF using hamming window
lpfhamming
Gain
Frequency (Hz)
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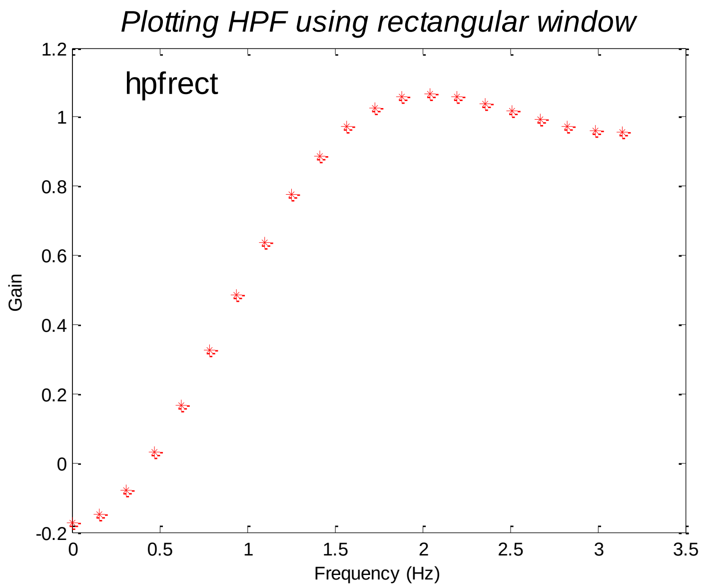
0.8
0.7
0.6
0.5
0.4
0.3
0.2
0.1
0
-0.1
0
0.5
1
1.5
2
2.5
3
3.5

(e)


Plotting HPF using rectangular window
hpfrect
Gain
Frequency (Hz)
1.2
1
0.8
0.6
0.4
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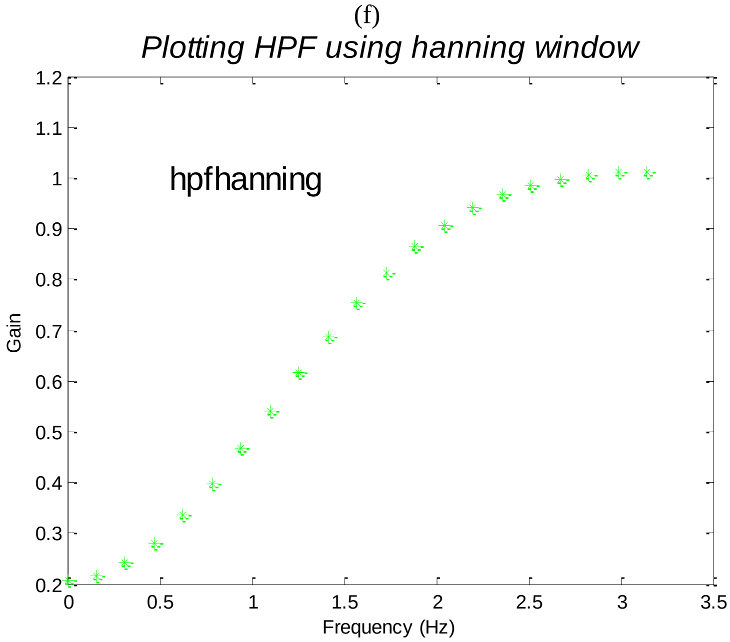
0
0.5
1
1.5
2
2.5
3
3.5

(f)


Plotting HPF using hanning window
hpfhanning
Gain
Frequency (Hz)
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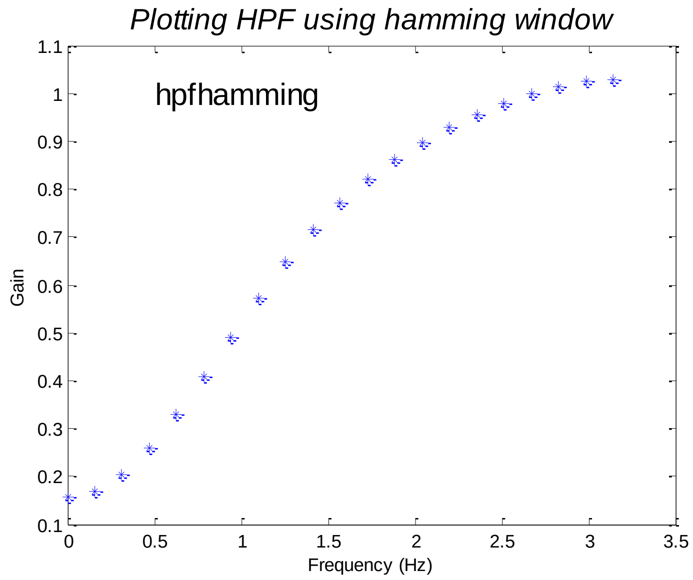
1.1
1
0.9
0.8
0.7
0.6
0.5
0.4
0.3
0.2
0
0.5
1
1.5
2
2.5
3
3.5

(g)


Plotting HPF using hamming window
hpfhamming
Gain
Frequency (Hz)
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1
0.9
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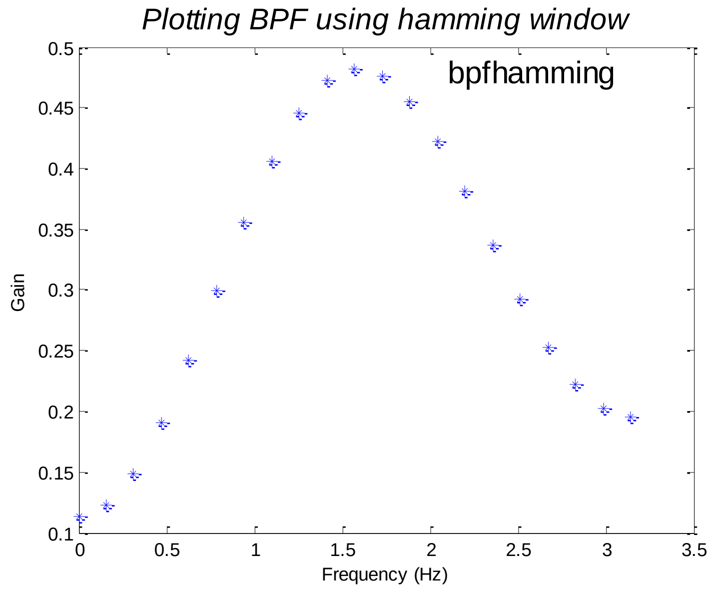
0.7
0.6
0.5
0.4
0.3
0.2
0.1
0
0.5
1
1.5
2
2.5
3
3.5

(h)


Plotting BPF using hamming window
bpfhamming
Gain
Frequency (Hz)
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0.15
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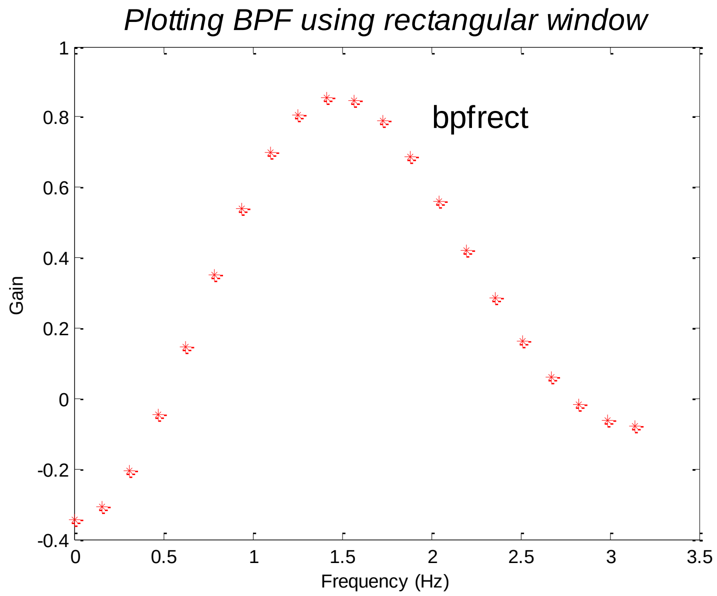
0
0.5
1
1.5
2
2.5
3
3.5

(I)


Plotting BPF using rectangular window
bpfrect
Gain
Frequency (Hz)
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0.8
0.6
0.4
0.2
0
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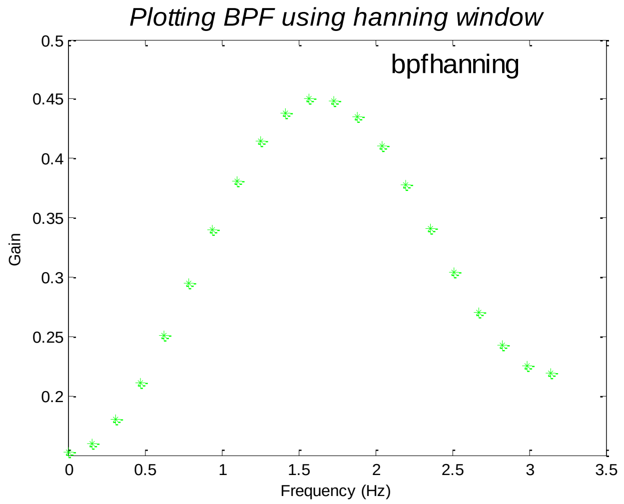
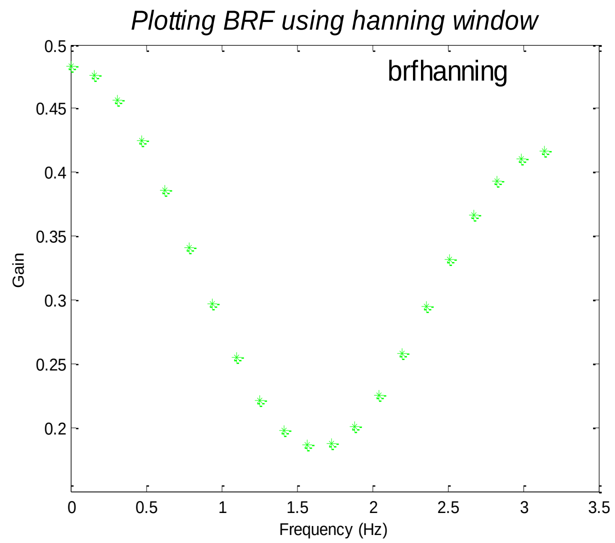
-0.4
0
0.5
1
1.5
2
2.5
3
3.5

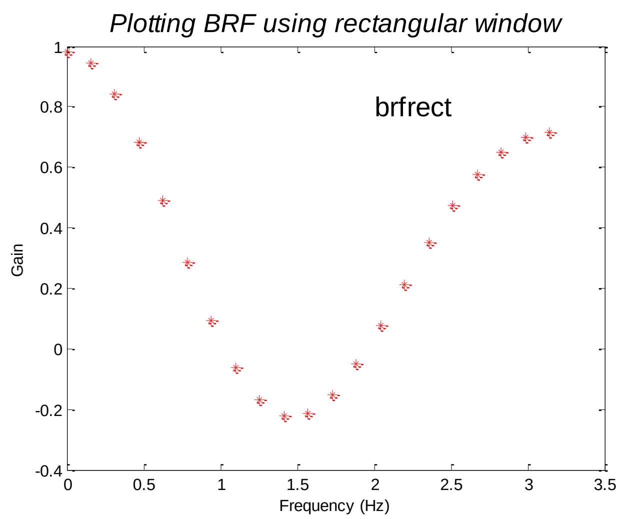
(j)



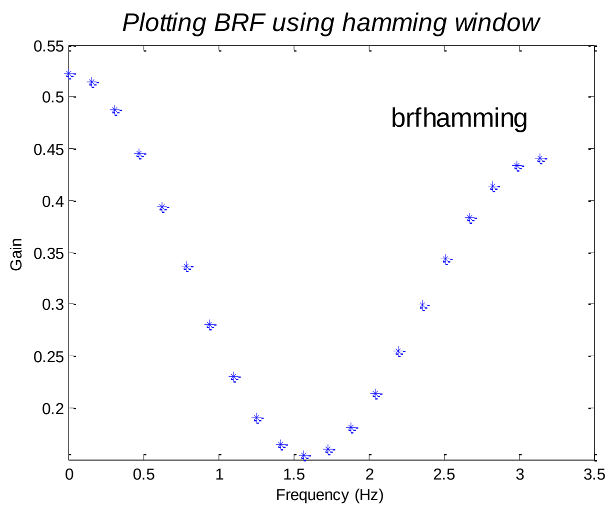
(k)



(l)



(m)



(n)

IV. **Conclusion**

We have presented a fast and robust procedure for the design of FIR filters. The proposed procedure is suitable for increase Stability and reduces component requirement and therefore a high computational cost to achieve the specifications desired. There are many ways of addressing this. Using a window greatly reduces the ringing. This improvement is at the expense of transition width (the windowed version takes longer to ramp from passband to stopband) and optimality (the windowed version does not minimize the integrated squared error). We will be looking at all these approaches in the following paper. We will then look into implementation of filters and discuss issues that arise when implementing a filter using fixed-point arithmetic

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