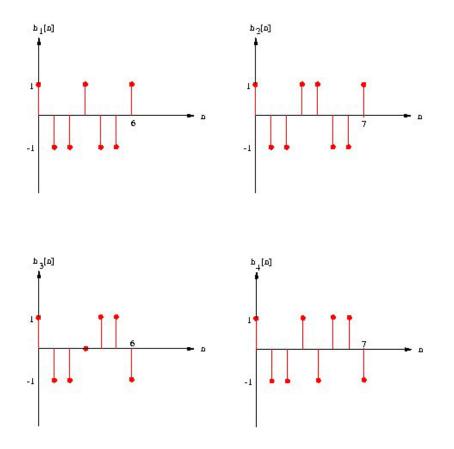
# J-DSP Lab 4: FIR and IIR Filter Design

## Introduction

Lab 4 concentrates on FIR and IIR filter design.

## Problem 4-1: FIR Linear Phase Systems

Consider the following four impulse responses:



a) For each impulse response, find the transfer function. Use J-DSP to plot the frequency response (magnitude and phase) of each system.

(Hint: these are FIR filters - see if they have symmetries)

b) For each system, describe the symmetries of the zeros of H(z).

(Hint: use J-DSP to find the roots).

c) Determine the group delay of each system. Use the tabulated values in the output dialog box to derive the exact group delay.

(Hint: plot the phase response and measure its slope)

d) Use J-DSP to obtain pole-zero plots and note symmetries in the z plane.

#### Problem 4-2: FIR Design by Windowing

Let

$$h(n) = 0.2 sinc(\frac{\pi n}{5})$$

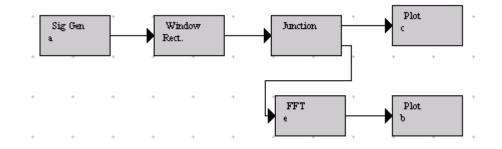
be the ideal impulse response of a low-pass filter with a cutoff at  $0.2\pi$ . To construct a truncated version of this impulse response we will generate a sequence as follows:

- Signal type: Sinc
- Amplitude: 0.2
- Pulse width: 120 samples
- Periodic: No
- Period T: 10 samples
- Time Shift: 30 samples

These settings provide you with a truncated, shifted, and causal version of the impulse response. Use a *Window* block and check the frequency characteristics for each window in the following order:

- a) Rectangular (default)
- b) Bartlett (Triangular)
- c) Hamming

The J-DSP flow-gram should look like this:



- a) Check the magnitude and phase response for each impulse response.
- b) Observe that all designs are linear phase. (why?)
- c) Observe that with rectangular window truncation you get the narrowest transition but the worst ripple effect.
- d) Note that tapered windows have better behaved side lobes and hence better-behaved ripple effect relative to the rectangular window.

## Problem 4-3: FIR Design using the Kaiser Window

Design a high-pass filter with generalized linear phase using the Kaiser window method.

Use the following specifications:

$$\left| H(e^{j\Omega}) \right| \le 0.05 \text{ for } 0 \le \Omega \le 0.375\pi$$
$$0.95 \le \left| H(e^{j\Omega}) \right| \le 1.05 \text{ for } 0.425 \le \Omega \le \pi$$

Use the *Kaiser* block for the filter coefficients and plot the frequency response of the filter. Use the following J-DSP flow-gram. Double click on the *Kaiser* block and enter appropriate parameters. Convert tolerances in dB.

🔉 Kaiser FIR Filter Design		
a a a	Kaiser Filter Parameter	S
Freq-Resp	Name: a	
	Filter type: Stop-Band  Coel -0.06 0.0	
Filter	Cut-off Frequencies: -0.13 0.23	
	Wp1:         0.25         Ws1:         0.5         0.18           Wp2:         1.0         Ws2:         0.75         0.18	64 Beta: 1.33 91 64
Kaiser	Ripple(dB): -0.13	328
â	PB: 20.0 SB: 25.0 -0.00	61
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a a a	Close Update Help	
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#### Problem 4-4: IIR Filter Design

In this part, you will design an IIR filter with J-DSP. The filter will be designed using four different IIR methods (Butterworth, Chebychev I, Chebychev II and Elliptic) so that results of the 4 different methods can be compared. The specifications for the filter are shown below.

- Filter Type = Low-pass
- Cutoff frequencies:  $\omega_{p1} = 0.4\pi$  and  $\omega_{s1} = 0.6\pi$
- Tolerance in pass-band = 1.0dB
- Tolerance (rejection) in stop-band = -45.0dB

The design can be done using the *IIR* block under the filter blocks menu in J-DSP. This block will automatically calculate the filter coefficients based on the filter specifications provided, using a bilinear transformation for any of the four design methods mentioned above. Attach the output of the *IIR* block to a *Freq-Resp* block to get a plot of the filter's frequency response or to a *PZ-Plot* block to see a plot of its poles and zeros, or both through a junction block. Note that the *IIR* block will calculate filters with a maximum of 10 filter coefficients. Enter the cutoff frequencies into the *IIR* block as fractions of the sampling frequency. For  $\omega_{p1}=0.4\pi$  simply enter 0.4.

For each of the four filter designs, do the following:

- 1. Plot the filter's frequency response.
- 2. Create a pole-zero plot of the filter.
- 3. Note the order of the filter.
- 4. Examine the filter's frequency response in the pass-band and in the stop-band.
- 5. Observe the phase response of each design.

Design the filter using each of the following four methods.

- 1. Design an IIR Butterworth filter according to the given specifications.
- 2. Redesign the filter using the Chebyshev I method.
- 3. Repeat the process using a Chebyshev II filter. Plot the frequency response on a dB scale.
- 4. Finally, design using an Elliptic filter. Observe its frequency response in the passband on a linear scale and its response in the stop-band on a dB scale.

Try to provide answers to the following questions:

- Which filter requires the highest order to meet the specifications?
- Which filter requires the lowest order to meet the specifications?
- Do any of the four filters have linear phase?
- Where does the greatest deviation from constant group delay occur?
- Which of the four filters is equi-ripple in the pass-band and monotonic in the stop-band?
- Which of the four filters is monotonic in the pass-band and equi-ripple in the stop-band?
- Which of the four filters is equi-ripple in both the stop-band and the pass-band?
- Which of the four filters is monotonic in both the stop-band and the pass-band?
- For the filters, which are monotonic in the pass-band, where are all the zeros located?

If you have time, try the above exercise using a high-pass filter and a band-pass filter of your choice.