EXPERIMENT #5
DIGITAL FILTER DESIGN AND FINITE-LENGTH EFFECTS
DUE: Apr. 5, 1995

Objective
The purpose of this experiment is to explore various aspects of discrete-time filter design in terms of the advantages and disadvantages of various approaches and the effect of finite-length arithmetic on the frequency response.

Prelab Preparation
Before going to the lab you are expected to do the following:

1. Familiarize yourself with the MONARCH filter design procedures (FIR, IIR) and the DSP board implementation of filters.

2. Do the pre-lab exercises.

3. Predict the results of the experimental steps described below.

SECTION 1: IIR FILTER DESIGN

Procedure

1. In all the designs in this part of the experiment, assume that the sampling frequency is 10 kHz. The filters designed here will be useful in speech processing where such a sampling frequency is used.

2. Using the IIR filter design component, design a Butterworth lowpass filter with the following specifications: passband edge frequency 2.5 kHz, stopband edge frequency 3.9 kHz, passband attenuation of 3 dB and order 2. Choose the largest value possible for the stopband attenuation that insures the order of 2. Plot the magnitude and phase frequency responses of the resulting filter as well as the location of its poles and zeros. Comment about the phase of the filter; how close to the desired linear phase is it? Without any additional computation determine the half-power frequency and bandwidth of the designed filter. Change the frequency scale to radians and give the half-power and bandwidth in radians.

3. Use CHEB I to design a Chebyshev lowpass filter that has as passband attenuation 3 dB, passband edge frequency of 2.25 kHz, stop-band attenuation of 20 dB, and is second order. Choose the smallest possible stopband edge frequency that permits the filter to be second order. Plot magnitude and phase frequency responses, location of poles/zeros of this filter. How does this filter compare with the Butterworth designed in 1? When would you use this filter instead of the one in 1? Explain.

4. Design a Butterworth bandpass filter that satisfies the following specifications: the stopband edge frequencies are 1.5 kHz and 3.5 kHz; the passband edge frequencies are 2 kHz and 3 kHz; the passband attenuation is 3 dB and the
stop-band attenuation is 20 dB. Plot the magnitude frequency response of the
designed filter using a log scale and verify that the specifications are met by the
designed filter. Give the order of the filter, location of its poles and zeros, and
its bandwidth in radians.

5. Suppose you wanted to improve the bandpass filter designed in step 3, by increasing
the stopband attenuation to 33 dB and decreasing the passband attenuation
to 0.3 dB, but you were interested in keeping the same order that was obtained
in 3. Try to achieve these specifications using the Butterworth design. Can you
do it? If not, use the elliptic filter design and plot the magnitude in log scale:
(to determine that the specifications are satisfied), determine the bandwidth (in
radians) of the elliptic filter and plot its poles/zeros location.

6. Design a Chebyshev highpass filter (use CHEB I) that satisfies the following
specifications: passband attenuation of 3 dB, stopband attenuation of 25 dB,
stopband edge frequency of 1.5 kHz and passband edge frequency of 2 kHz. What
is the minimum order of the filter that satisfies these specifications? What is the
half-power frequency (in radians) of the designed filter? Plot the magnitude fre-
quency response in log scale, and the pole/zero location of the elliptic filter. How
does the elliptic filter differ with respect to the Butterworth and the Chebyshev
filters designed before?

7. Keep the specifications of the highpass filter in 6. If you use the elliptic filter
design, determine how much you can increase the stopband attenuation and still
have an order identical to that of the filter designed in 5. Plot the magnitude
frequency response of the designed elliptic filter in log scale.

Discussion

• Comment on the advantages and disadvantages of the Butterworth, Chebyshev
and elliptic filter design methods. When would you prefer to use one rather than
the other?

SECTION 2: FIR FILTER DESIGN

Procedure

1. Design using the DIRECT procedure a lowpass FIR filter with an order of 21
and normalized edge frequencies \( f/f_s \) of 0.2 and 0.3. Plot the impulse response
and magnitude frequency response in linear scale. What kind of symmetry do
you notice in the impulse response? Should the phase of this filter be linear? In
SIGLAB, read the impulse response file and plot the phase of the filter, is the
obtained phase linear? Is the phase linear in the stopband? The window used
in this design is the rectangular, generate a rectangular window signal used and
plot its magnitude frequency response using a 256- point FFT.
2. In SIGLAB, read the impulse response of the FIR designed above and multiply it by a Hamming window of length 15 and shifted to the right by 3 samples. Plot the magnitude and phase frequency responses of the designed filter. Compare this filter with the one designed in 1 above, how do they differ? Is the phase of the designed filter linear? Compute the 256-point FFT of the Hamming window used and plot its magnitude frequency response.

3. Repeat 2, using again the impulse response of the FIR designed in 1 and a Kaiser window of length 15 and the beta parameter of 3. Shift the window by 3 samples to the right before multiplying the impulse response. Compare this filter (Kaiser window design) with the one designed in 1 (rectangular window design), and the one in 2 (Hamming window design). Which of the three is a better design and why? Plot the magnitude frequency response of the Kaiser window using a 256-point FFT.

Discussion

- The three FIR designs in this part of the experiment were done using the window technique. This procedure depends on the type of window used, what would be the desired characteristics of an ideal window? Explain.
Procedure/Discussion

1. Design a first order FIR differentiator using the DIFFERENTIATOR design procedure. Let the filter order be 6; the grid density be 64; the first band have normalized edge frequencies of 0 and 0.48 and unity weight and gain; the second band have 0.49 and 0.5 as lower and upper edge normalized frequencies, a weight of 0.1 and a gain of 0. Plot the impulse response and magnitude frequency response of the designed filter. What type of symmetry does the impulse response have? Why is this called a differentiator?

2. In SIGLAB, read the impulse response file of the differentiator designed above and filter a sinusoid. Did you obtain the derivative of the sinusoid as output? Plot the input and output signals. Plot the magnitude frequency response of the filter along with the magnitude of the 256-point FFTs of the input and output signals.

3. Implement the differentiator on the DSP board. Use a signal generator to input various signals into the filter and use the scope to look at the input and the output signals. Do you get the expected results? What happens if you disconnect the input (i.e. if the input is noise)?

4. Round up the coefficients of the differentiator so as to get integers (i.e. assume the DSP board is fixed-point rather than floating-point) and repeat 3. Does this have any significant effect on the filtering? Explain.