University of Pennsylvania
Department of Electrical and System Engineering
Digital Signal Processing

ESE531, Spring 2020 — HW7: Filter Design — Sunday, Mar. 29

Due: Sunday, April 5, 11:59PM

- **Recommended Problems for Practice:** From the book: 5.51, 7.5, 7.24, 7.28abc
- **Homework Problems:** All problems must be turned in and are not optional for full credit

  1. Homework problems from the book: 5.46, 7.16 (see section 7.6 for help), 7.27
  2. Matlab problem 1: Design an FIR filter using Truncation
     Ideally, a low-pass filter with cutoff frequency ω_c should have a frequency response of

     $$H_{ideal}(e^{j\omega}) = \begin{cases} 1, & |\omega| \leq \omega_c \\ 0, & \omega_c < \omega \leq \pi \end{cases}$$

     and a corresponding impulse response of

     $$h_{ideal}[n] = \frac{\omega_c}{\pi} sinc\left(\frac{\omega_c n}{\pi}\right) \text{ for } -\infty < n < \infty$$

     However, no real filter can have this frequency response because $h_{ideal}[n]$ is infinite in duration. One method for creating a realizable approximation to an ideal filter is to truncate this impulse response. A truncated impulse response is of finite duration, yet the filter is still noncausal. In order to make the FIR filter causal, it must be shifted to the right by M units.

     To examine the effect of filter size on the frequency characteristics of the filter, write a Matlab function `LPFtrunc(N)` that computes the truncated and shifted impulse response of size N for a low pass filter with a cutoff frequency of $\omega_c = 2.0$. For each of the following filter sizes, plot the magnitude of the filters DTFT in decibels. Hints: The magnitude of the response in decibels is given by $|H_{dB}(e^{j\omega})| = 20log_{10}|H(e^{j\omega})|$. Note that the `log` command in Matlab computes the natural logarithm. Therefore, use the `log10` command to compute decibels. To get an accurate representation of the DTFT make sure that you compute at least 512 sample points using the command `[X,w]=DTFT(filterResponse,512)`.

     Design two filters with size N=21 and N=101.

     - Submit the plots of the magnitude response for the two filters (not in decibels). On each of the plots, mark the passband, the transition band and the stopband.

  - Submit the plots of the magnitude response in decibels for the two filters.
  - Explain how the filter size effects the stopband ripple. Why does it have this effect?

Now download the noisy speech signal nspeech2.mat , and load it into the Matlab workspace. Apply the two filters with the above sizes to this signal. Since these are FIR filters, you can simply convolve them with the audio signal. Listen carefully to the unfiltered and filtered signals, and note the result. Can you hear a difference between the two filtered signals? In order to hear the filtered signals better, you may want to multiply each of them by 2 or 3 before using sound.

  - Comment on the quality of the filtered signals. Does the filter size have a noticeable effect on the audio quality?

3. Matlab problem 2: Design an FIR filter using Windowing
   Design a length-23 linear-phase FIR low-pass filter with a band edge of $\omega_0 = 0.3\pi$ using the following windows:
   - Rectangular (`rectwin`)
   - Triangular or Bartlett (`bartlett`)
   - Hann (`hann`)
   - Hamming (`hamming`)
   - Blackman (`blackman`)

   Plot the impulse response, magnitude response, and phase response or the 5 filters. Compare the characteristics of the magnitude response of the 5 filters. Do this by comparing the squared error and transition bandwidth. The discrete squared error is defined as

$$\epsilon = \frac{1}{N} \sum_{k=0}^{N-1} |H_d(e^{j\omega_k}) - H(e^{j\omega_k})|^2$$

4. Matlab problem 3: Design an FIR filter using Parks-McClellan Algorithm

   Design a symmetric low-pass FIR filter using `firpmord` and `firpm` in Matlab to meet the design specifications given below:

$$\begin{aligned} \omega_p &= 1.8 \\ \omega_s &= 2.2 \\ \delta_p &= 0.05 \\ \delta_s &= 0.005 \end{aligned}$$

   Compute the DTFT of the filters response for at least 512 points, and use this result to compute the passband and stopband ripple of the filter that was designed. Adjust the filter length until the minimum order which meets the design constraints is found. Plot the magnitude of the DTFT in dB of the final filter design.

Submit the final measured values of filter length, passband ripple, and stopband ripple. How accurate was the filter order computation using Matlabs firpmord? Submit the plot of the filters DTFT.