HACETTEPE UNIVERSITY
Department of Electrical and Electronics Engineering
ELE 409: Digital Signal Processing Laboratory

EXPERIMENT 4 - IIR FILTERS

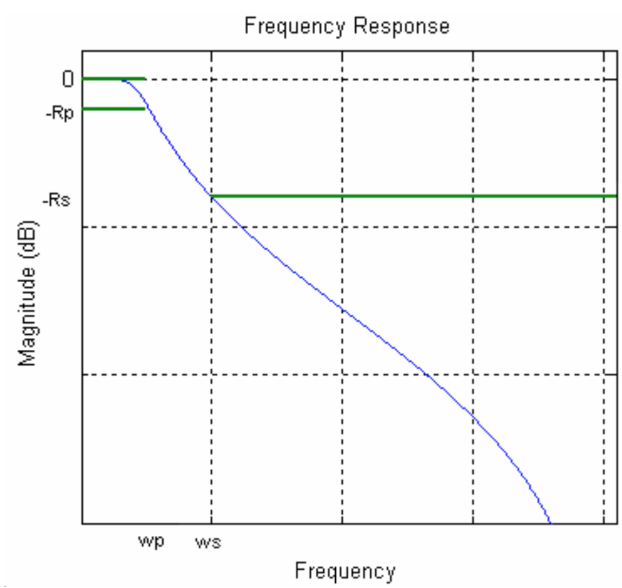
I. PURPOSE

The purpose of this experiment is to design of Infinite Impulse Response (IIR) filters. Design of discrete-time filters with Butterworth and Chebyshev approximations are investigated with using Matlab's sptool application.

- Learn how to use the following built-in functions and the reason why do we use:
  audioread, sound, freqz, sptool.
- Frequency range should be $[-\frac{f_s}{2}, \frac{f_s}{2}]$ in your frequency domain figures.

II. PRELIMINARY WORK

1. Load the sound file sound4.wav. Name the data as x_input, plot the waveform and its magnitude spectrum. Determine the peak frequencies.
2. Load signal processing tool by writing sptool to command window. SPTOOL opens the SPTool window which allows you to import, analyze, and manipulate signals, filters, and spectra. You will design and test your IIR filters with help of sptool.
3. Assume that you are asked to preserve the first dominant peak and filter the higher harmonics of input signal. In order to perform this task you will design a lowpass filter. For example, in the following figure, signal loses no more than Rp dB in the passband and has at least Rs dB of attenuation in the stopband.



   Determine necessary Rp, wp, Rs and ws values to perform this task

4. With sptool filter design application implement your lowpass filters using Chebyshev type I, Chebyshev type II and Butterworth filter types.
5. Filter the input signal with your filters. Plot the waveform and spectra of output signals. Compare the results.
6. Assume that you are asked to preserve the second dominant peak and filter the lower and higher harmonics of input signal. In order to perform this task, what kind of filter do you need? Determine the parameters of this filter.

7. With sptool filter design window implement your filter (which is designed in question 6) using Chebyshev type I, Chebyshev type II and Butterworth filter types.

8. Filter the input signal with your filters. Plot the waveform and spectra of output signals. Compare the results. What is the frequency of resulting signal?

9. What can you say about Chebyshev type I, Chebyshev type II and Butterworth type filters considering the results of this experiment.

10. Load the sound file bana.wav. Filter this signal with a lowpass filter of any type with cut off frequency of 2kHz. Listen the filtered signal and its original form. An interesting property of speech signals is that they are robust to filtering. You can understand the information whether it is lowpass, highpass or bandpass filtered.