Armstrong Atlantic State University
Engineering Studies
MATLAB Marina – Sound Processing Exercises

1. Write a MATLAB program that will:
   - Create a signal that is a single tone of 500 Hz with a duration of two seconds and an amplitude of one. Use a sampling frequency of $f_s = 10k$ Hz.
   - Plot two periods of the signal. The period of a signal is the inverse of the frequency, $T = \frac{1}{f}$.
   - If you have a set of headphones play and listen to the signal. Is 500 Hz a low, middle, or high audio frequency?
2. Write a MATLAB program that will:
   - Create a signal that is a sum of three sinusoids (100 Hz, 500 Hz, and 2000 Hz) with duration of two seconds. Each sinusoid should have an amplitude of one. Use a sampling frequency of $f_s = 10k$ Hz.
   - Plot two periods of the signal. The period of a signal that is a sum of sinusoids is the least common multiple of the individual periods.
   - If you have a set of headphones play and listen to the signal.
3. Write a MATLAB program that will:
   - Create a signal that is a sum of three sinusoids (100 Hz, 500 Hz, and 2000 Hz) with a duration of two seconds. Each sinusoid should have an amplitude of one. Use a sampling frequency of $f_s = 10k$ Hz.
   - Use MATLAB's `filter` function to filter the signal with a five point averaging filter. The filter coefficients of five point averaging filter are: b = [0.2, 0.2, 0.2, 0.2, 0.2] and a = [1].
   - In separate vertically tiled axes of the same figure window (subplots), plot the signal and the filtered signal. What does the filter do to each of the sinusoidal components in the signal?
   - Experiment with averaging filters of different lengths. How does the length of the averaging filter affect the filtered signal?
4. Write a MATLAB program that will:
   - Create a signal that is a sum of three sinusoids (100 Hz, 500 Hz, and 2000 Hz) with a duration of two seconds. Each sinusoid should have an amplitude of one. Use a sampling frequency of $f_s = 10k$ Hz.
   - Use MATLAB's `fft` function to determine the spectrum of the signal.
   - In separate vertically tiled axes of the same figure window (subplots), plot the magnitude and phase of the spectrum of the signal.
5. Write a MATLAB program that will:
   - Load the audio signal in the wave file `frog.wav`.
   - Recreate the time vector corresponding to the frog signal read from the `frog.wav` file. You will need the sampling frequency to generate the time vector.

- In separate vertically tiled axes of the same figure window (subplots), plot the time-domain frog signal and the first second of the time-domain frog signal.
- Using the MATLAB `fft` function, determine the spectrum of the frog signal.
- Plot the spectrum of the frog signal.

6. Write two versions of a MATLAB function to amplify (scale) a signal. The functions should take a signal and a constant (amplification factor) and return the amplified signal. One version should use loops to amplify each sample of the signal; the other version should use array operations to amplify all the samples in one operation. Test your two functions by amplifying the signal created in Exercise 2.
7. Write a MATLAB function named `myFirFilter` that will filter a signal that is a 1d array using an FIR filter. Filtering a discrete-time signal x[n] with a FIR filter with filter coefficients bk is performed via $y[n]=\sum_{k=0}^{M} b_k x[n-k]$. Use the function header of Figure 1.

```
function yy = myFirFilter(bb, xx)
% -----------------------------------------------------
% myFirFilter.m
% -----------------------------------------------------
% myFirFilter: filters a signal using a FIR filter
% -----------------------------------------------------
% usage: yy = myFirFilter (bb, xx)
%   bb = FIR filter coefficients
%   xx = discrete-time signal to be filtered
%   yy = filtered signal
% -----------------------------------------------------
```

Figure 1, myFirFilter function header

Last modified Thursday, October 31, 2013