EECS 360 Signal and System Analysis

Audio Filtering

1. Record a sample of your own voice. You can do this, for example, with the Sound Recorder in Windows→Programs→Accessories. Save your voice as a Microsoft WAVE (.wav) sound file with a sample rate of 44100 samples/second (it doesn't matter if it is mono or stereo). Load the sound file into MATLAB with *wavread*, plot out the data to see the waveform of your voice, and play the audio file by *sound*:

   ```
   [x,sr] = wavread('filename.wav');
   plot(x(1:100:end));
   sound(x,sr);
   ```

   The commands above load a WAVE file specified by filename, returning the sampled data in x, and returning the sample rate (sr) in Hertz in sr. The filename input is a string enclosed in single quotes. Because x will most likely contain a huge number of samples, it is sufficient to plot only every 100^{th} sample. Knowing the number of data samples (length of x) and the sample rate, how many seconds long is your sound file?

2. Download the recordings of acoustic impulse responses from the Downloads in our class webpage. Read it into MATLAB, plot it to see the waveform and play it. How many seconds long are these sound files?

3. Use the impulse response to filter your voice:

   ```
   y = filter(h,1,x);
   ```

   filter returns the convolution of h (h is the sample data of impulse response) and x, in order to play the output by *sound*, we need to normalize it.

   ```
   y = y/max(abs(y(:)));
   ```

   then plot out the output and play it with *sound*.