EE 40471: Project 1

Due 22 February, 2017

Poor Dr. Sauer suffers from high-frequency hearing loss (because he didn't protect his hearing when he was young); his auditory response begins to roll off at $\omega = \pi/2$. Your assignment here is to write the Matlab code for a high-frequency emphasis filter to put into his hearing aid. He is not willing to pay more than \$29.95 for the device, so you can't afford to use anything more sophisticated than a third-order filter (up to three poles and three zeros). The goal is to maintain a response of near 1.0 for $0 < \omega < \pi/2$ and ramp the magnitude of the response upward approximately linearly to near 10.0 over the range $\pi/2 < \omega < \pi$. The filter must be stable, and near-linearity of phase is desired. Make sure the response at zero frequency is 1.0.

Implement the filter as a Matlab function accepting a vector of arbitrary length, realizing a third-order difference equation for the input/output relation, and returning the enhanced audio vector. For initial conditions, you may assume the signal is zero for $n < 0$. You may hard-wire the filter coefficients if you like. For the design process, use your intuition, imagination, and *fdatool()*'s pole/zero editor, but not any complete design functions. Start with a first-order, highpass, IIR filter in $fdatool()$, then add and relocate poles and zeros. In $fdatool()$, you could select filter types, assign specs, and get a design for free, but please refrain from using these powers. Your main job is to effectively implement the difference equation and get a reasonable filter response, so you won't get lots of extra credit for the beauty of your frequency response curve - that's for later when we actually talk about filter design details. Write the function for a single, mono audio track. We'll process stereo audio, but I'd like to preserve the option of filtering just one side. The calling function can contain the logic on stereo/mono processing.

For testing your function, you may want to read in sample audio files via, for example $[x, fs] = audioread(filename.wav)$.

The material you submit for evaluation must include:

1. The pole/zero plot of your design
2. Plot of the frequency response (magnitude and phase) with easily-read labeling of axes
3. Plot of the impulse response of your filter
4. A brief explanation, with equations, of your conversion of the pole/zero design into the difference equation for your filter's implementation. You may use Matlab to assist with any tedious calculations, but make sure you show that you know how it's done if you'd have to do this manually.
5. Your Matlab function implementing the filter in such a way that it accepts an arbitrary audio vector (y = *yourfunction(x)*), and returns the filtered signal. Write the filter yourself as a loop in which you multiply by the coefficients of your difference equation.

Items 1-4 may be submitted in a formatted file (Word, PDF, etc) via email to Dr. Sauer, or turned in as hard copy in class. Item 4 may be written by hand. The Matlab file should be emailed. Including your calling function is optional.