AET 5420 Quiz 2

Name : ______________________________

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1 SPECTRAL PROCESSING

One of the most important types of processors for an audio engineer is the spectral processor. These types of processors can be used to change the relative amplitude of different frequencies in a signal. There are many kinds of spectral processors: low-pass filter, high-pass filter, band-pass filter, shelving filter, and peaking filter. The combination of these types of filters can be used to create an equalizer and are also used in other types of complex spectral processors.

1.1 TILT EQUALIZER

One less common type of spectral processor is the **tilt equalizer**. The basic concept of this type of equalizer is to process the entire spectrum of a signal either by simultaneously increasing the amplitude of high frequencies and reducing the amplitude of low frequencies or vice versa. Essentially, the spectrum is tilted to emphasize the high frequencies or low frequencies. An example of this type of spectral processor is the Tonelux Tilt.

1.2 PROBLEM

The purpose of this problem is to create a MATLAB **function** that will perform tilt equalization. There should be two input variables to the function. The first variable should be the input signal represented by an array of samples. The second variable should be a value between -6 and 6 representing the tilt amount. There should be one output variable, the processed signal.

- Name the function - tiltEQ.m

- There should be two (2) input variables
  - input : column vector of a sound file to be processed
  - tilt : represents the tilt amount (-6 to 6 dB)
    * When the tilt amount is negative, the equalizer should decrease the amplitude of high frequencies and increase the amplitude of low frequencies.
    * When the tilt amount is positive, the equalizer should increase the amplitude of high frequencies and decrease the amplitude of low frequencies.
    * The tilt amount is the maximum change in amplitude at 0 Hz and at the Nyquist frequency.
    * Between 0 Hz and the Nyquist frequency, the change in amplitude has a constant slope across the entire spectrum.
- The output variable of the function should be a column vector representing the processed audio file.
- At the start of the function, transform the tilt amount from a decibel value to the linear scale and assign it to a new variable.
  - It will be necessary to calculate one tilt amount for low frequencies and a different tilt amount for high frequencies.
- Create another new variable called **order**
  - This variable represents the order or length of the filter.
  - Experiment with different values for the length of the filter and observe its influence on the outcome.
- Next, use the MATLAB function, **`fir2`**, to create the filter.
- Finally, convolve the input signal with the filter and assign the result to the output variable

While you are working on your function, it may be helpful to analyze the filter by using : **freqz(h)**. Remember to add comments to your code to explain what each command is accomplishing.

1.3 Submission

When completed, put the script and sound files in a compressed zip folder. Name the zip file: XX_AET5420_Quiz2.zip, where XX is your last name. Then email the zip file to: eric.tarr@belmont.edu.