Alexandria University
Faculty of Engineering
Computer and Systems Engineering
Department

CS332: Digital Signal Processing
Assigned: Sun 17th Jan, 2016
Due Date: Tue 26th Jan, 2016

Digital signal processing project

It is required to develop audio equalizer using MATLAB. One common Audio equalizer is the Winamp program as shown in figure. The equalizer function is to vary the gain of each specific band as the user prefers (i.e. if the user likes base he will increase the gain of low frequencies).

Project procedure:

Inputs to program:

User should input the following:

(1) The wave file name.
(2) Gain of each of the frequency bands in dB. (9 bands)
(3) Type of filters used (FIR-IIR) and the type of FIR.
(4) Output sample rate.

You may make use of the following functions: `input - menu - uigetfile`

Method:

(1) Develop the frequency band filters in the following bands
   a. (0 - 170Hz) – (170 - 310Hz) – (310 - 600Hz) –
      (600 - 1000Hz) – (1 - 3KHz)- (3 - 6KHz) –
      (6 - 12KHz) – (12 - 14KHz)- (14 - 16 KHz).
(2) Analyze and export these filters (Gain, phase, impulse and frequency responses, order, error and poles/zeros). You can use the command “fdatool” to design and plot all the necessary filters.
   http://www.mathworks.com/help/signal/ug/opening-fdatool.html
   http://www.mathworks.com/help/signal/ug/iir-filter-design.html
   http://www.mathworks.com/help/signal/ug/fir-filter-design.html
(3) Filter the wave file using the filters developed in step 1.
(4) Draw the output signals in Time and frequency domains
(5) Amplify the output signals using the user-defined gain
(6) Add the amplified-output signals in time domain to form composite signal.
(7) Draw and compare the composite signal with the original signal (in time and frequency).
(8) Play and save the output wave signal. (you can use `wavwrite-sound)`

Outputs/Requirements of project:

(1) User interface to input the data to M-file even if as command window lines (GUI is a plus).
(2) The original and composite signals (.wav files)
(3) All figures of signals in time and frequency domain.
  a. The input signal in time and frequency domain.
  b. For each filter (impulse response, frequency response (Gain-Phase), Group delay and pole/zero plot).
  c. For each output signal before the gain "signal from each filter before multiplying the gain" (time and frequency domain).
  d. The composite signal in time and frequency domain.
(4) Well commented code
(5) Filter analysis results. (Description of how you designed the filter and its specifications including investigations on the delay, maximum error and average error of the filter).
(6) Different sample runs of code including the following cases
  a. If design is using FIR filters (include all FIR filters discussed during the course).
  b. If design using IIR filters
  c. Output signal in case if doubling output sample rate or decreasing it to half
  d. Output signal if base is required to be dominant (what is the gains you used).

Submission policy:

(1) Submission will be via following mail: omar.salaheldine@gmail.com
with subject: [DSP Project]
make sure to include your full names and IDs in the mail.
(2) Submit the following:
  a. Your original signal (sound file preferably of small size and contains multiple freq. bands)
  b. The m-file(s) that contain(s) the project.
  c. A report describing your approach, specially in the design of filters (By making a well commented code, you can simply use the MATLAB: (file → publish M-file) and it will make full report with sample runs that you can simply add the additional data regarding error analys (maximum error and average error) as well as filter specifications.
  c. The 4 sample runs in the description (include all plots and output sound files).
(3) The project will be done in groups of up to **three.**
(4) Due date for mail delivery is: Tuesday 26/1/2016 @11:59 pm
(5) Discussions will take place on 27/1 & 28/1 according to a spreadsheet that will be made later.

Good Luck