

Digital signal processing

Digital Filter- Assignment

It is required to develop audio equalizer using MATLAB. One common Audio equalizer is the winamp program as shown in figure 1. 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].



Assignment Procedure:

Inputs to program:

User should input the following:

- 1) The wave file name.
- 2) Gain of each of the frequency bands in dB.
- 3) Type of filters used (FIR-IIR)

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- 170 Hz) - (170- 310 Hz) - (310- 600 Hz) -(600- 1000 Hz) -(1- 3 KHz) - (3- 6 KHz) - (6- 12 KHz) - (12-14 KHz) - (14-16 KHz)

2) Analyze and export these outputs (magnitude, phase, impulse and step response, order and gain/poles/zeros).

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 the output wave signal. (you can use wavwrite-sound)

Outputs/Requirements of project:

You should submit ONE PDF file contains the followings:

1) Copy of your code, the code must be well commented

2) Different sample runs of code including the following cases:

a. If design is using FIR filters.

b. If design is using IIR filters.

c. Output signal in case if doubling output sample rate or decreasing it to half

3) the analysis of each filter and the exported outputs

4) All figures of signals in time and frequency domain.