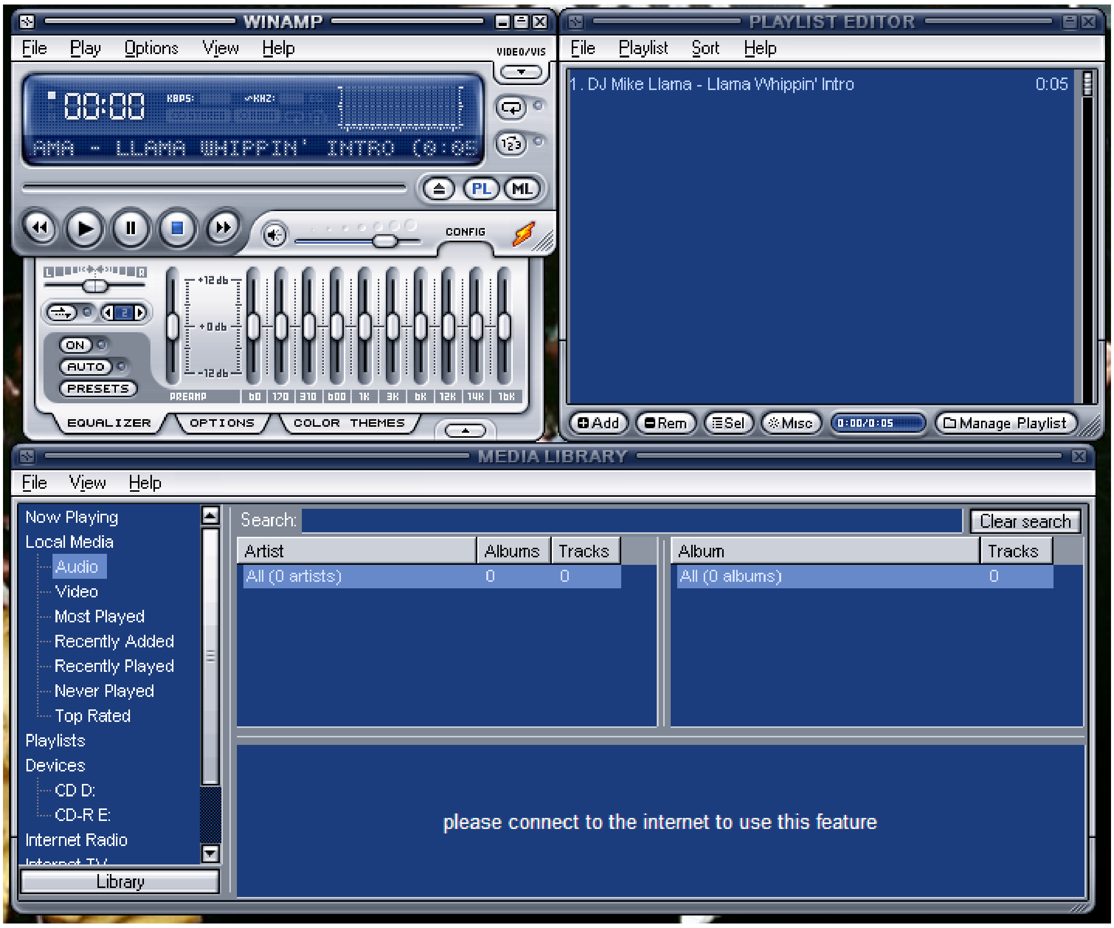
Digital signal processing
Digital Filter- Assignment

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

.The wave file name (1

.Gain of each of the frequency bands in dB (2

(Type of filters used (FIR-IIR (3

.Output sample rate (4

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

<u>:Method</u>

Develop the frequency band filters in the following bands (1

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

Analyze and export these files (Gain, phase, impulse and step response, order and (2 .(poles/zeros

.Filter the wave file using the filters developed in step 1 (3

.Draw the output signals in Time and frequency domains (4

.Amplify the output signals using the user defined gain (5

.Add the amplified - output signals in time domain to form composite signal (6

(Draw and compare the composite signal with the original signal. (in time and frequency (7

(Play and save the output wave signal. (you can use wavwrite-sound (8

<u>:Outputs/Requirements of project</u>

(User interface to input the data to M- file even if as command window lines (GUI is a plus (1

(The original and composite (.wav files (2

.All figures of signals in time and frequency domain (3

.Well commented code (4

.Filter analysis results (5

:Different sample runs of code including the following cases (6

.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