Abstract
An Audio Signal Processor analyze sound signals, process them and explore the signal properties. This paper is a part of the work carried out by the authors for designing the Audio Processor in MATLAB software providing basic functions to read, play and write the sound signal along with signal processing toolbox, which includes numerous functions that work on signals. A unique feature of the paper is development of Graphical user interfacing (GUI) helping users to understand the functioning and parameters of the system project and work on it in an easy way. The paper focuses on implementation of latest audio features like jazz, pop,
rock, treble, bass and myfav using filters and includes the various curves like amplitude vs frequency, amplitude v/s time etc giving information about the variation of amplitude with frequency and time of signal. The paper illustrates practical aspects of FIR filter design as well, using the Filter Design Toolbox and the Signal Processing Toolbox for this purpose. The emphasis is mostly on lowpass filters, but many of the results apply to other filter types as well. The paper concludes with time domain and frequency domain analysis of the signals.

Reference

16. B. L. Sturm, SSUM: Signal and Systems Using MATLAB; Creating an Effective Application for Teaching Media Signal Processing to Artists and Engineers, 2004, (M.S. Project) University of California, Santa Barbara, Graduate Program in Media Arts and Technology, USA.
<table>
<thead>
<tr>
<th>Key words</th>
<th>Index Terms</th>
<th>Component</th>
</tr>
</thead>
<tbody>
<tr>
<td>styling</td>
<td>formatting</td>
<td>style</td>
</tr>
<tr>
<td>insert (key words)</td>
<td></td>
<td></td>
</tr>
</tbody>
</table>