Implementation of Graphical Equalizer using LabVIEW for DSP Kit DSK C6713

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Abstract— Digital signal processors plays an important role in fine tuning the sound system using Equalizers to overcome the acoustic problems. In digital communication, Graphical equalization techniques strengthen the successful provision of high speed and reliable data transmission over dispersive channels. The mentioned paper describes the development of equalizer for the digital signal processing (DSP) kit DSK C6713. Programming was done in 'C' language and compilation using code composer studio (CCS) integrated development environment using LabVIEW (Laboratory virtual Instrumentation Engineering Workbench) as front end. Different Sliders are designed for different frequency bands, the performance of the Equalizer is observed by altering the gain of the frequencies in the specific bands while playing an audio signal.

Index Term— Graphical Equalizer, LabVIEW, Code Compose Studio, DSK C6713.

1 INTRODUCTION

In the present telecommunication field, the communication system designers face ever increasing challenges in utilizing available bandwidth efficiently. The transmission of high speed data through a channel is limited by Inter-symbol interference (ISI) caused by distortion in the transmission channel. This leads to reducing the quality of the received signal as measured Bit Error Rate (BER). ISI is caused by many different phenomena such as filtering effects from hardware or frequency selective folding, from non-linearity's and from charging effects. Very few systems are immune from it and ISI is nearly always present in digital communication.

High speed data transmission through channels with severe distortion can be achieved in several ways. One of the way is to design transmit and receive filters so that the combination of filters and channel results in an acceptable error from the combination of ISI and noise. And the other way is by designing an equalizer in the receiver that counteracts the ISI and channel distortion.

In selection criteria between two types of equalizer techniques, we mainly deal with the Graphic Equalizer which is also termed as the Q Graphic Equalizer. The other technique in discussion is parametric equalization. Generally these Equalizers have identical set of amplifiers for each channel in an audio system [2]. The usage of six band equalizers helps us to control the gain (volume) which is in proportion with the audio signal presented. A Graphical Equalizer application heads mainly in an audio controller in stereophonic system that helps the user to control various frequency bands individually.

LabVIEW as front end that displays the Graphical Equalizers for the user in order to control the frequencies focus is done on development of the audio equalizer with a digital signal processor. Testing is carried out using digital signal processor starter kit (DSK) with TMS320C6713, floating point processor. Compilation includes development of C program which generates an assembly program that helps DSK C6713 to generate the desired signals with desired frequency.

2 PRE STUDY

The equalizer is implemented by studying many methods for filters to design and what kind of equalizer to design and ways to implement using digital signal processor.

How to develop an equalizer and How to implement?

2.1 Equalizer: Methods and Types

Equalizer is defined as the system that is capable to attenuate and boost the frequencies as per as the desired frequency. The two types of the equalizers are:

Manuscript received April 27th, 2012; accepted May 31st, 2012; Date of Publication June 10th, 2012 The Editor coordinating the review process for this paper was Dr M Taylor.

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- Graphical equalizer.
- Parametric or Variable Parameter Equalizer.

In brief, there are fixed number of filters with each filter having fixed gain and bandwidth and centre frequency are the components associated with the parametric equalizer [1]. Unwanted variations in frequency characteristics associated with loudspeakers are determined. A Graphic Equalizer is a popular and the simplest type of equalizer which divides the frequency band of the given audio signal into sub-bands and varies the characteristics of the signal. The sliders can be graphically controlled by the user. A Graphical Equalizer is comprised of sliders that perform boosting and cutting of sound frequency of different ranges. The Graphical Equalizer exhibits high fidelity and enables the user to view graphically and adjust individually several distinct frequency bands in a stereophonic system. A typical design of a Equalizer incorporates a number of audio amplifiers and filters, each attuned to a specific frequency range as shown in Fig. 1. A majority of Graphical Equalizer comprise two identical sets of amplifiers and filters, one for each channel in the stereophonic system.

The aim of this project is to design Graphic Equalizer with six different sliders where the first filter was designed using the low pass FIR hamming window having 48000Hz sampling frequency, 250Hz cut off frequency with 64 order . The second, third, fourth and fifth filters were designed using a Band pass filter with increasing cut off frequencies, the sixth slider was designed using high pass FIR filter with 16000 Hz cut-off frequency. The implementation of a Graphical Equalizer is differing to the tone controls.

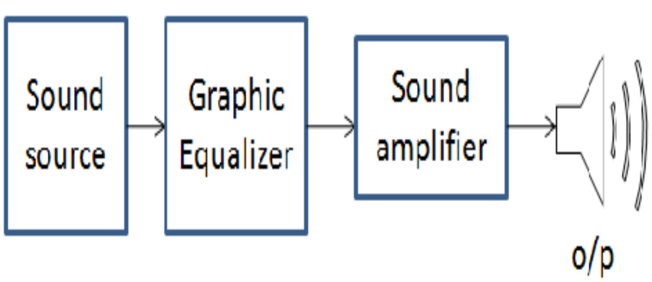

Figure 1 Graphical Equalizer

2.2 DSK-TMS320C6713 Processor

The TMS320C6713 DSP Starter Kit (DSK) developed jointly with Spectrum Digital is a low-cost integrated device designed to improve the performance speed in high precision applications [2]. Broad range of applications in the fields of communication and speech processing extends the hand to develop a low cost software and hardware support. TMS320C6713 processor is very-long-instruction-word architecture (VLIW). The board uses a sigma-delta technology for analog and digital conversation and digital to analog conversation takes places. It is connected to 12-Mhz system clock, the on board kit is having includes SDRAM and flash memory and four connectors MIC IN, LINE IN, LINE OUT and HEADPHONE OUT.

The components include:

a) C6713 floating-point digital signal processor.
b) A 32-bit stereo codec TLV320AIC23 (AIC23).

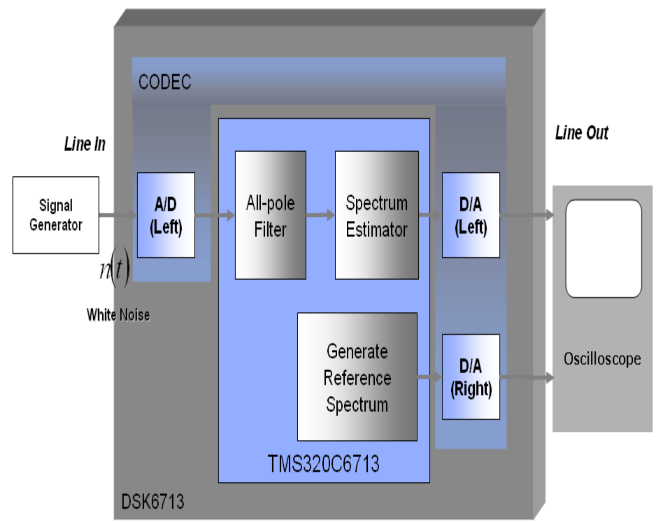

Figure 2 Block diagram of DSK 6713

The internal program memory is structured so that a total of eight instructions can be fetched for every cycle. Memory unit consists of 256KB internal and 256KB of cache memory [1]. The block diagram is shown in the below Fig. 2 and Fig. 3.

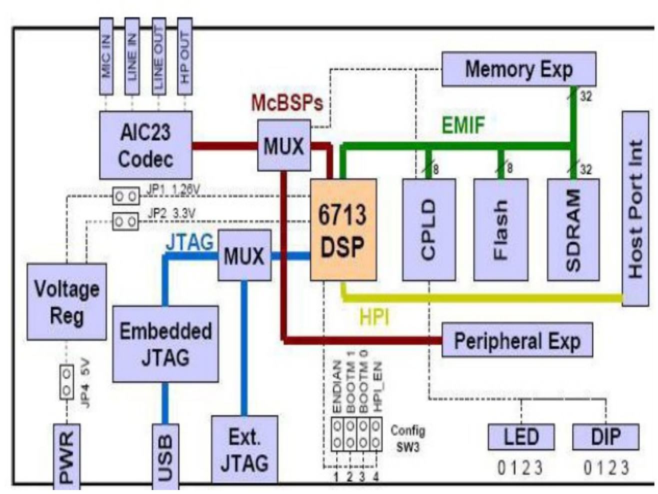

Figure 3 Block diagram of TMS320C6713-based DSK board.

2.3 LabVIEW

Its interactive platforms enable the user to develop application graphically and executed in parallel. The application is to interface the external data acquisition to signal processing devices and mandatory tool for all signal processing applications [4]. The toolkit in LabVIEW enables the user to explore wide range functionality of software and hardware analysis related with signal processing as in Fig. 4.

The appearance and the operations of the LabVIEW programming, termed as virtual instruments, is a mirror image of physical instruments such as function generator, Oscilloscopes etc. LabVIEW contains a complete set of tools for acquiring, analyzing, displaying, and storing data, as well as tools for trouble-shooting the source code [4].

In our project we used LabVIEW as a front end for development of the equalizer. The front panel with gray background and the block diagram window with white. The front panel design includes controllers, Indicators, which are interactive with user for input and output terminals. Controls includes the knobs, push buttons and input simulate the instrument mechanisms and supplies data to the block diagram developed. The indicators include graphs. LED's and display simulates the output of the instruments used.

Figure 4 LabVIEW

3 MAIN CONTRIBUTION

3.1 Setup for 6-Band Graphic Equalizer

The setup for the 4-band Graphic Equalizer consists of the hardware and software where we need TMS320C6713 DSP kit , Personal Computer (PC) with CCS, LabVIEW, a universal synchronous bus which connects the DSK and the PC, a 5V power supply to connect to DSK, Oscillator, Signal Generator and Speaker [5] as shown in the below Fig. 5.

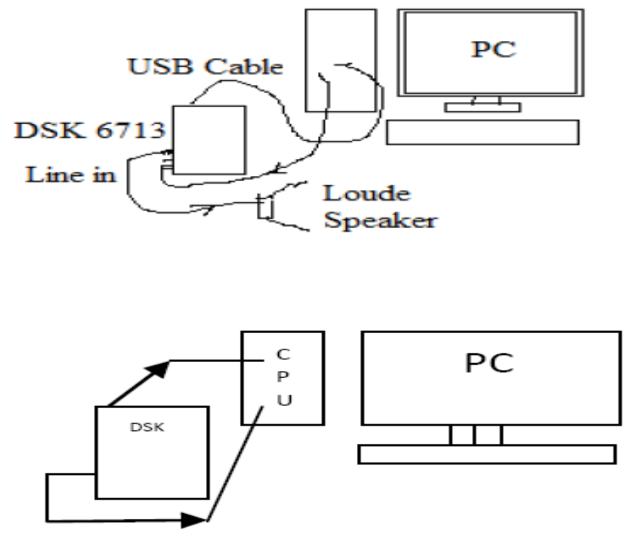

Figure 5 Project Setup for Graphical Equalizer

a. Template Design in LabVIEW

The front panel of the task is developed on LabVIEW as in Fig 7. The block diagram description "the new template is opened by selecting the option from windows menu". In the project three stacked sequence structured block diagram where each stack contains one or more sub diagrams that executes sequentially as shown in Fig 6. The necessary functions were added in the stacks to create the objects in the front panel. The LabVIEW block diagram designed for the Graphic Equalizer.

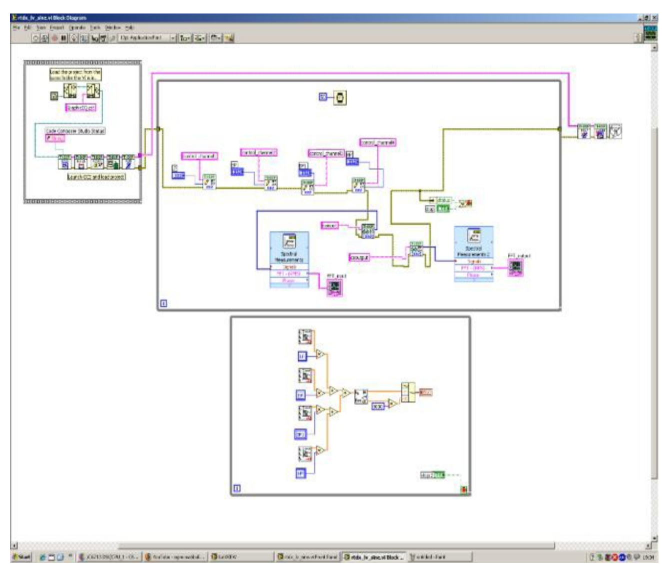

Figure 6 LabVIEW Back End Template of Graphic Equalizer.

The first stack consists of three functions current VI path function, Strip path function and Build path function. The CCS open project, CCS download code icons are connected using wiring tool to return the file to current VI in order to launch the project. The second stack consists of 6 CCS RTDX Write icons, 2 CCS RTDX Read icons, 2 Spectral measurements express VI, 2 graph function icons and one OR function connected in a particular order. The on board joint test action group (JTAG) helps in data exchange between the host and the processor while the processor is running [3] the main part is played by RTDX real time data exchange that contains both host and target components. The CCS RTDX writes numeric data to the corresponding RTDX channels [1]. The FFT based spectral measurements are performed. The two structures were connected sequentially to CCS, HALT, CCS close and Simple Error Handler. To halt the output running on the development kit and also used to indicate the error. The third stacked sequence structure consists of FIR Windowed Coefficients VI, Build Waveform function along with several multiply functions connected in parallel. This structure was designed to plot the filter characteristics. The frequency characteristics of low pass filter, 4 band pass filters and high pass filter were linked together and were plotted in a single graph. The structure was terminated using a loop condition.

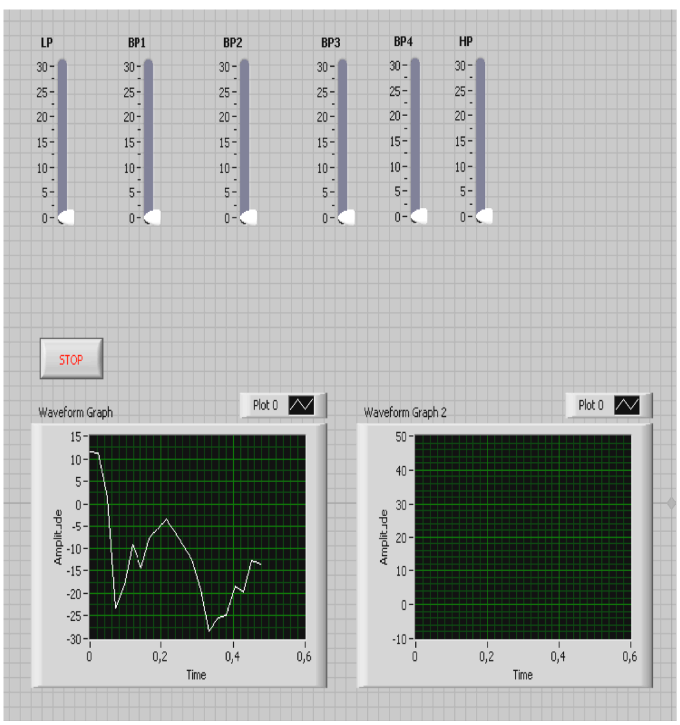

Figure 7 Graphic Equalizer Front End Template

3.2 Filter Design using MATLAB

The development of six filters contributes the major part of the task. The design of filter coefficients is carried away with MATLAB Functional Data Analysis (FDA) tool. Six filters include:

1. One Low pass filter: 250 Hz cut-off frequency.
2. Four Band pass filters: 250-1000 Hz, 1000-2000 Hz, 2000-4000 Hz, 4000-16000Hz cut off frequencies.
3. One High pass filter: 16000 Hz cut-off frequency.

Windowing technique used Hamming window.
Sampling frequency 48000Hz

A total of 64 filters co-efficient were generated for each filter which corresponds to each slider of the equalizer. Centre frequencies coefficients are calculated using signal processing tool in MATLAB consists of a Graphical User Interface (GUI) tool called FDA tool where the coefficients are generated with the header file the centre frequencies are of ISO standards for Graphic Equalizer are low pass, high pass and four band pass. The filter bandwidth for all filters is measured by taking the difference $f_2 - f_1 > 0$ where f_2 and f_1 correspond to the frequencies where the gain is down *3 dB* from the maximum gain (*12 dB*) at the centre frequency. The centre frequency F_c corresponds to the geometric mean of f_1 and f_2, i.e. $F_c = (f_1 + f_2)^{1}/_{2}$.

The centre frequency gain for each band pass filter should vary between *+12 dB* and *-12 dB*. Use a slider gain block to adjust the gain between these two settings. FIR filter is implemented using Kaiser Window in recent years. It has variable parameters to change the size of the side lobe with respect to main lobe. The implementation of the FIR filter is by taking the coefficients obtained from FDA tool as mentioned in the above filter design; here we take the input sample from the audio source which is given as line in to the DSP kit. The algorithm used for the interface between LabVIEW and the DSK kit [6] is shown as:

```
RTDX_enableInput(&control_channel0);

   if(!RTDX_channelBusy(&control_channel0 ))

   { RTDX_readNB(&control_channel0,

&bass_gain, sizeof(bass_gain)); }
```

In this way, the design and implementation of 6 filters for each centre frequency is done and at last outputs of these filters is added and given to the output sample where the equalized controllable output is obtained from the speaker. To control the gain of the band pass filters, a program is written in .gel file. Gel object is a graphic user interface to vary the gain of the equalizer for all 6 bands.

3.3 Implementation of the Graphic Equalizer

a) Compilation and Rebuilding in CCS

The DSP kit is tested and connected to PC through CCS. The project file with *.pjt* extension is loaded and opened in the code composer studio. The build operations are modified by selecting C6710 in the basic category selecting the path of *DSK6713.h*. Using *C* program the source code is executed on LabVIEW.

b) To run project in LabVIEW

The audio signal is from a PC as an input signal and the equalizer by connecting the input to the connector to LINE IN on the DSP board. The command "RUN" is used to load and run the program. The positions of the sliders are adjusted in the equalizer and change in gain is observed from the speakers which are connected to LINE OUT of the DSP board. The command STOP will stop the programming.

3.4 Testing

The *C* program is developed and its filter co efficient explains the performance and tested with the input music signal with *.MP3* extension is loaded and tested by varying the positions of slides in the equalizer. The FFT input, output and the filter characteristics were observed on the template. The Fig. 8 shows the tuned Graphic Equalizer template with the performance plots.

4 Conclusion

Implementing the Graphic Equalizer explains how to implement the digital signal processing concept on the real time system C6713 DSK. Hardware Programming skills are developed and proper usage of the available resources like FIR filters. This project involves the design of 6-Band Graphic equalizer using FIR Band pass filter the corresponding algorithms of the filters are presented and implemented using code composer studio and tested with the TMS320C6713 digital signal processor. The Graphic Equalizer improves the sound quality and adjusts frequency and gain. The future work can be extended by developing codes to control heavy systems, like robots with the controls, in the field of Robotics.

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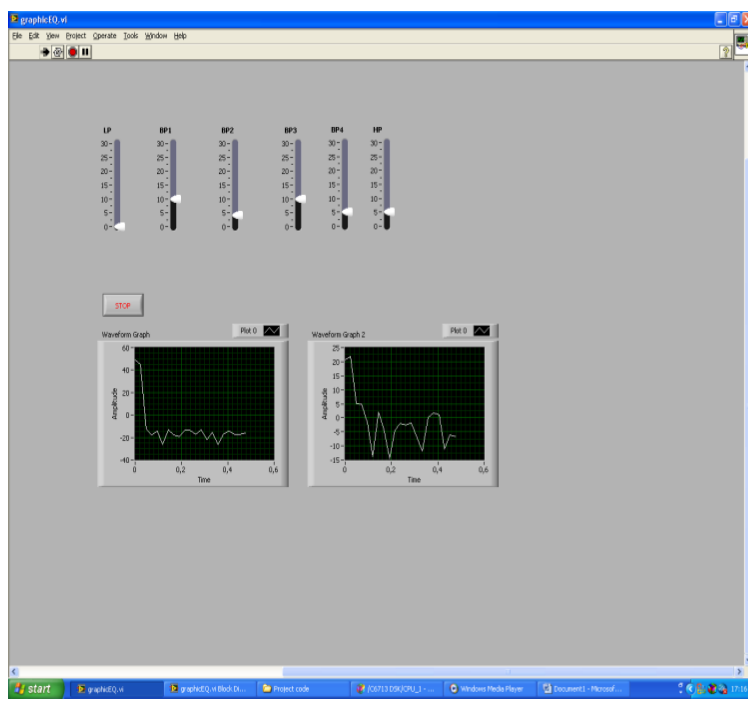

Figure 8 Template showing tuned equalizer with FFT input, FFT output and Filter characteristics plot

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