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Design and Implementation of Digital IIR & FIR Filters on DSK6713 using CC STUDIOV6.1 Platform

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Abstract: A digital filter plays a very important role in communication over analog filter. Digital filters are easy to design, less complex, flexible and portable compared to analog filters. In this paper, the procedure to implement any real-time projects on DSK6713 is explained. DSK6713 is widely used in the field of industry and educational institutions. This paper represents the basic introduction of DSK6713 processor and quick procedural steps for code composer studiov6.1. The CCStudiov6.1 IDE tool is used for coding and implementing code on DSK6713. This paper explains the detail procedural steps used for design and implementation of IIR and FIR digital band-pass filters.

Keywords* — *CCStudiov6.1, DSK6713, TMS320C6713, IIR filters, FIR filters, Band pass filter.

I. INTRODUCTION

The digital filters are highly accurate, linear phase response (FIR Filter), adaptive filtering possible, easy to simulate and design, ADC and DAC conversions can be done [1]. The analog filters are less accurate, nonlinear phase, difficult to design and simulate, no ADC and DAC conversions, adaptive filtering is a difficult task. A digital filter can be implemented using a DSP processor. An analog filter adds component based thermal noise. Programmable coefficients are possible in digital filter and not in analog filters. Higher order digital filters can be feasibly designed compared to analog filters.

A Finite Impulse Response (FIR) filter is characterized by an impulse response which is finite in time domain. The FIR filter response is computed by the present and the previous vales of the input signals. This type of filters is called non-recursive filers. FIR filter transfer function contains only zeros and hence there is a less probability of the filter becoming unstable. An Infinite Impulse Response (IIR) filter is characterized by an impulse response which is infinite in time domain. The IIR filter response is computed by the present and the previous vales of the input and output signals. This type of filters is called recursive filers. IIR filter transfer function contains both zeros and poles, hence there is a more probability of the filter becoming unstable if not designed properly. FIR filter has many advantages as compared to IIR filters such as: finite impulse response, easy optimization, linear phase, better stability etc. Different fields where FIR filters are applicable are Image processing, audio signal processing etc. Any filter characteristics can be easily determined by frequency response, which in turn depends on filter coefficients. If the coefficients of the equation are properly selected then frequency selective filters can be designed that will pass particular frequency components and rest all will be attenuated [2].

The floating-point DSP processor TMS320C6713 is present onboard a Digital Starter Kit called DSK6713, which permits high speed debugging of code with Code Composer Studiov6.1. DSK6713 is a low-cost development platform that allows the students and researchers to evaluate and develop DSP applications for TIC67xx family. For a TMS320C6713 DSP processor, DSK6713 is the hardware reference design. To reduce time to market and to make hardware development easier DSK contains many logic equations, schematics and application notes. DSK6713 contains onboard USB Interface, SDRAM, ROM, analog interface circuit (AIC CODEC) for ADC and DAC, embedded JTAG emulation support. It also includes stereo codec AIC23, which is software controlled via TI McBSP –Compatible Multiprotocol serial post. Audio input/output signal is via McBSP compatible AI [2].

DSP industry's first fully developed IDE tool is Code Composer Studio, which has DSP specific libraries and functions that make the debugging and execution faster. With a user friendly environment CCS enables the users to create a project, write C/ assembly code, build, compile, debug and run the project. Many other additional features include injection of I/O from the file, graphical user interface, debugging with multi-processor, testing and customization through a C scripting language etc.

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CC Studio provides many chip support library files for linking the basic operations of on-chip CODEC, DIP switches, LED lights and many other components of DSK6713.

II. TMS320C6713 DIGITAL STARTER KIT

Fig 1 shows the block diagram of 6713 Digital Starter kit. This contains 32-bit EMIF for the SDRAM and daughter card expansion interface. The AIC23 CODEC is used for transmitting and receiving signals. This audio input/output can be done through a 3.5mm audio jack that connects to microphone in, line in, line out and headphone output. McBSP0 is used for codec control interface and McBSP1 is used for data transmission [9].

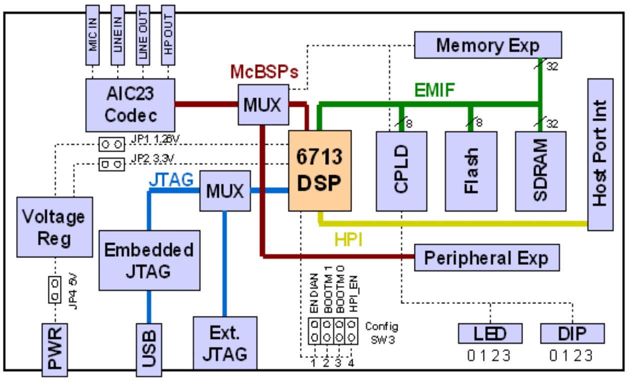

Fig 1: Block diagram of DSK6713

Either microphone-in or line-in can be connected to codec for input signal and either headphone-out or line-out can be connected for the purpose of output. The line-out is a fixed gain and the headphone is an adjustable gain connector. To provide a user with interactive feedback the DSK provides with 4 DIP switches and 4 LEDs. The board is powered on by an external 5V power supply. The voltage regulator onboard provides 1.26V as DSP core voltage, 3.3V analog and 3.3V digital voltages [3]. The DSK can be programmed and communicated by CC studio through an embedded JTAG emulator with a USB host interface. Fig 2 shown is a physical layout of DSK6713. Spectrum Digital designs and manufactures the board, where as the C6713, which performs core DSP related operations is from Texas Instruments [8].

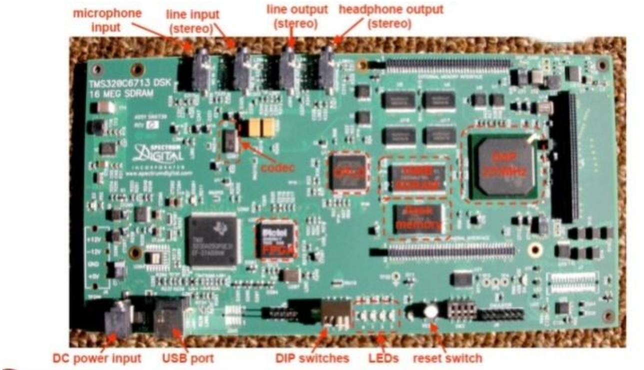

Fig 2: DSK6713 Physical layout

In DSP operations the most extensively used operation is MAC (Multiply and Accumulate) [4]. This operation is inbuilt and can be performed as a single instruction using 6713 DSK. The AIC23 codec chip is usually a 2-channel audio input output device.

A. *Features of DSP TMS320C6713*

- is a floating point DSP with high performance characteristic.
- In one cycle eight 32-bit instructions are executed.
- Data Word is of size 32/64 bit
- Best suited for audio applications and rich peripheral set.
- Very long and advanced instruction word
- C/C++ compiler with highest optimization.
- All functional units are independent:
- 2 fixed point ALUs
- 4 floating and fixed point ALUs
- 2 floating- and fixed-point multipliers
- 16-bit HPI
- 2 McBSPs (Audio serial ports)
- Flexible Phase-Locked-Loop (PLL) Based Clock Generator Module.
- IEEE-1149.1

III. CODE COMPOSER STUDIOV6.1

The procedural steps for working with Code Composer Studiov6.1 are explained in this section. These steps can be used for both non-real time and real time programming applications [6].

Step1: In the current workspace, create a new project File -> New ->New CC S project.

Step2: Give the project name as required. As it is a filter design any filter name can be given. Fill the other details as shown in the Fig 3.
Then click -> Finish. A new CCS project will be created with the given name in the current workspace.

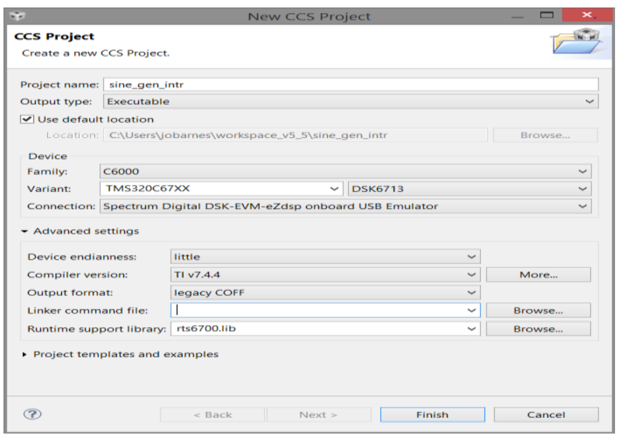

Fig 3: Create a new CCS project

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Step3: Select project properties window. Click on the project name. - > Project -> Properties.

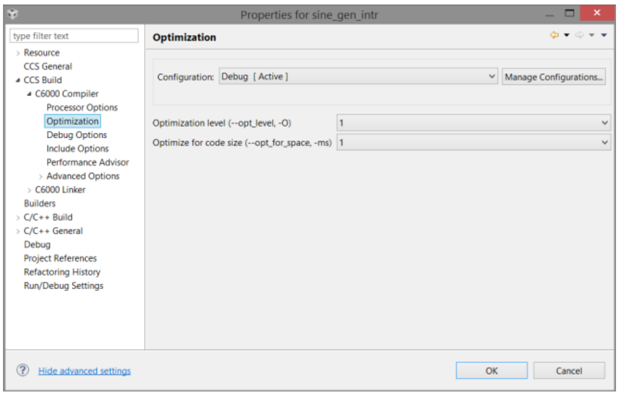

Fig 4: CCS Project properties window

Step4: Select the Target processor version. Go to Build and C6000 Compiler and Processor option. Enter 6700 in the blank space.

Step5: Setting optimization level=1: Go to Build and C6000 Compiler and Optimization and set the value to 1.

Step6: To select Include Options: Go to Build and C6000 Compiler and Include Options [5]. The window shows two blank spaces, to which the proper paths need to be added. The first path is added by the CCS, which points to the core library functions. The second path points to the board and chip support libraries.

These include files and chip support library files are as shown in Fig 5.

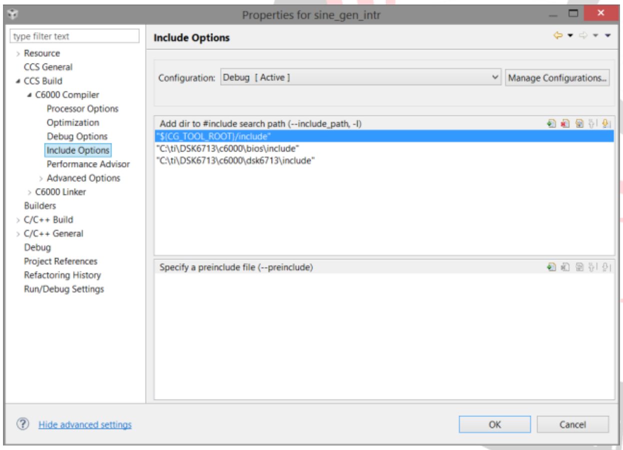

Fig 5: CCS Include settings

Step7: Select the predefined symbols. Go to Build - C6000 Compiler - advanced options - Predefined Symbols. Enter CHIP_6713 to the pre-defined name blank space.

Step8: Linking the paths: Go to Properties – Build - C6000 Linker - File Search Path. Add “libc.a”, “csl6713.lib”, “dsk6713bsl.lib” to the top panel.

Step9: Now type the program in the source file and save it.

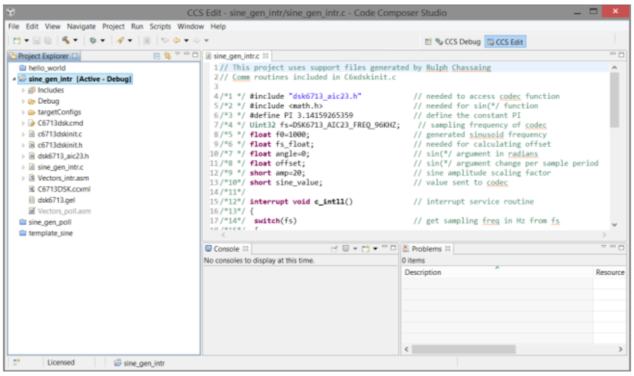

Fig 6: CCS window

Step10: Go to Project →Build project. (after build project, ***Build Finished*** message will come in the console window.

Step11: Run →Debug.

Step12: After the program is loaded successfully, click Run→Terminate the program

Step13: After terminating the program hear the output from MICOUT (Headphone or speakers or line out is connected)

IV. DESIGNING THE FILTER

A. IIR filter design specifications:

Sampling frequency = 48 KHz
Stopband frequency1 = 500 Hz
Passband frequency1 = 1 KHz
Passband frequency2 = 3 KHz
Stopband frequency2 = 3.8 KHz

The coefficients of IIR filter are designed from fdatool of MATLAB which generates the filter coefficients. The coefficients obtained from fdatool is stored in a file for eg “bpiirfilter.cof”. Fig 7 shows the fdatool window of MATLAB used to design IIR filter for given specifications.

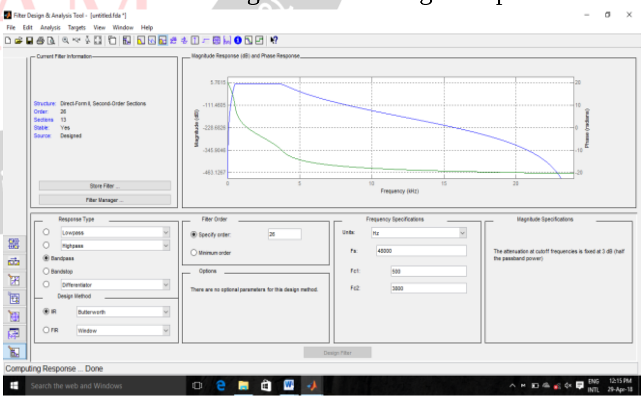

Fig 7: IIR filter magnitude and phase response

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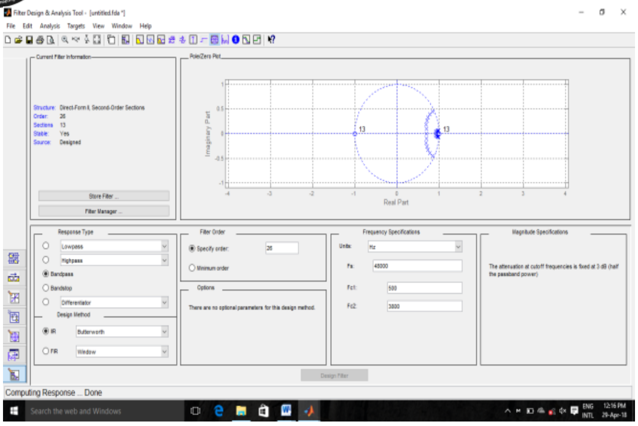

Fig 8: IIR filter pole-zero plot

B. FIR filter design specifications:

Sampling frequency = 48 KHz
Stopband frequency1 = 500 Hz
Passband frequency1 = 1 KHz
Passband frequency2 = 3 KHz
Stopband frequency2 = 3.8 KHz

The coefficients of FIR filter for these above specifications obtained from fdatool is stored in the file for eg "bpfirfilter.cof". Fig 9 and 10 shows the fdatool screen for designing of FIR filter along with frequency response and its pole-zero plot.

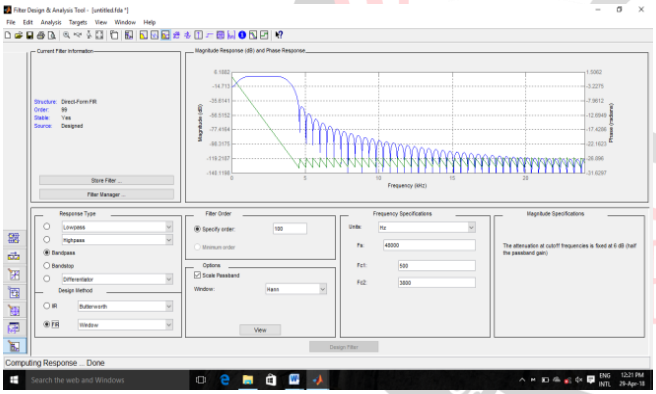

Fig 9: FIR filter magnitude and phase response

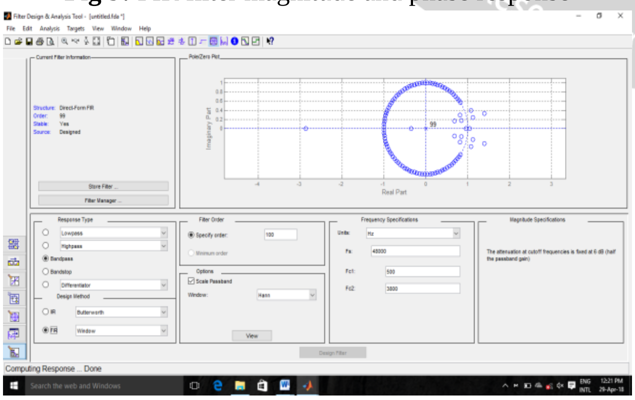

Fig 10: FIR filter pole-zero plot

V. Implementation

The designed filter coefficients are used in C code which is loaded on the DSK6713 processor along with all supporting files. Fig 11 shows the real-time implementation of the work.

Fig 11: Real time implementation picture

The audio input is given to LINE_IN and filtered output is taken from LINE_OUT. Either a sine wave from the signal generator or an audio input can be given as input signal and the output can be seen on a CRO for observing the amplitude levels in passband and stopband regions of the filter by varying the input signal frequencies or it can be heard through a pair of speakers.

VI. Applications

As DSK6713 can also be used for direct programming applications from Simulink platform using embedded C code compiler, it gives added advantages of effortless writing of codes on MATLAB and debugging these codes on DSP.

Digital filters have wide area of applications in communication field for removing signal harmonics, noise, interferences etc. DSK6713 is the commonly found DSP kit in many educational institutions for teaching-learning process. This kit is also found important in audio echo, filtering, speaker recognition and many other fields.

CC Studio v6.1 is the modern and updated IDE platform which can be used in academics and industrial organizations for the development of any real time and non-real time DSP applications.

VII. Experimental Results

For experimental purpose, an input sine wave signal with 1Vpp and variable frequency range is set. The results or filter outputs are shown in following figures for the different cases.

As the FIR band-pass filter is designed for 1KHz to 3Khz passband frequencies, the simulation results show that the output of the filter is attenuated and rejected for the input signal frequency 100Hz as shown in Fig 12. The output signal of the filter is distortion-less clean signal for the input signal frequency 2KHz, which falls in the passband region of the filter as shown in Fig 13. The stopband region operation is shown in Fig 14.

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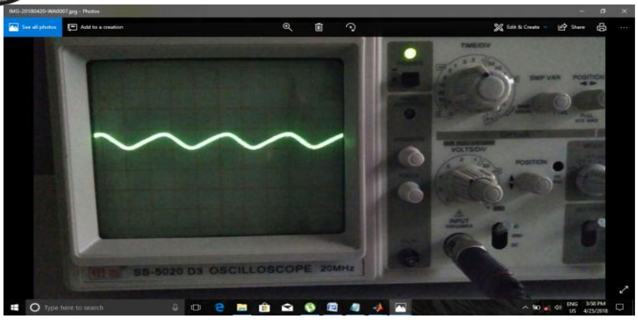

Fig 12: FIR filter output for input signal frequency=100Hz

Fig 13: FIR filter output for input signal frequency=2000Hz

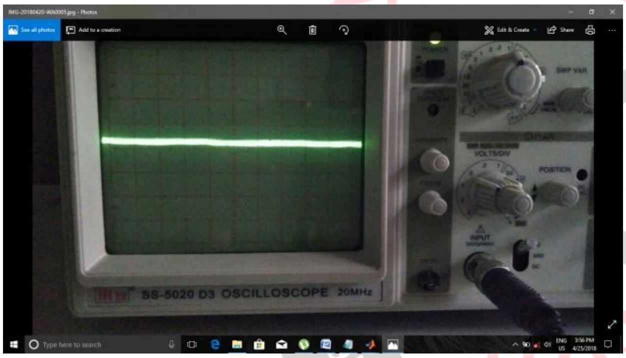

Fig 14: FIR filter output for input signal frequency=4000Hz

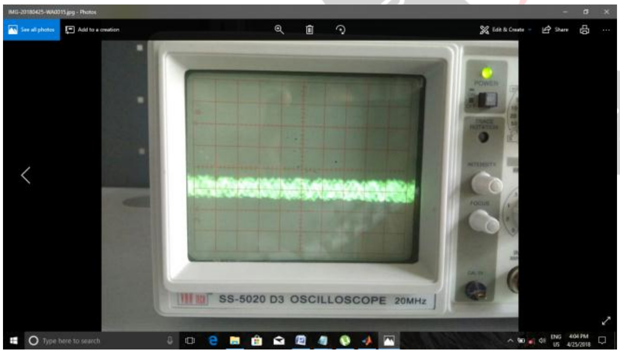

Fig 15: IIR filter output for input signal frequency= 500Hz

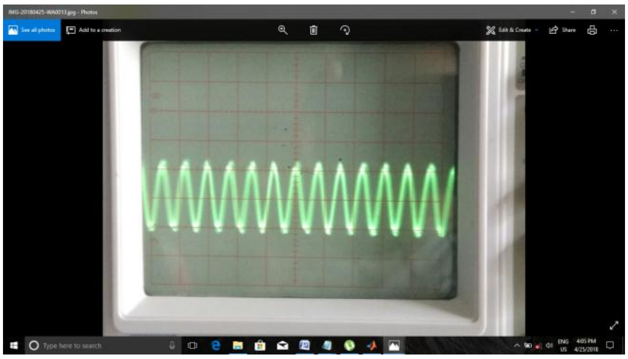

Fig 16: IIR filter output for input signal frequency= 2KHz

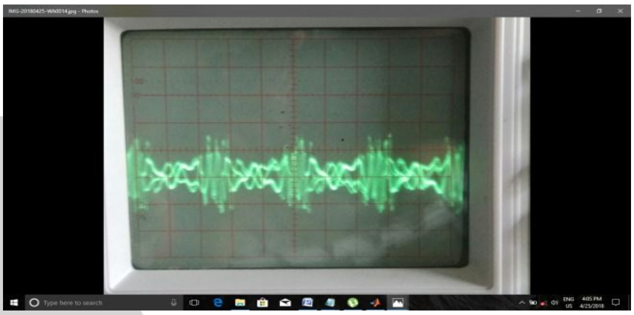

Fig 17: IIR filter output for input signal frequency= 4KHz

Fig 15, 16 and 17 shows the stopband and passband operations of the IIR bandpass filter. With respect to distortion and attenuation levels of the output signal, FIR filter gives good performance when compared to IIR filter.

VIII. CONCLUSION

In this paper, the basic introduction to DSK6713 processor, procedural steps for working with CCSv6.1 and real time implementations of IIR and FIR filters are discussed and executed. From the experimental results, it is clear that any digital filter with given specifications can be feasibly designed and implemented with less power consumption, less time, more portability and flexibility etc. From the experimental results, it is clear that IIR filter offers less attenuation level and FIR filter offers high attenuation levels in the stopband region. Also because of pole locations, stability is a major problem in IIR filter and FIR filters are almost and always stable and provides distortion-less output signal in its passband region.

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