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PERFORMANCE ANALYSIS OF DIGITAL FILTER ON USING TMS320C6713 DSP STARTER KIT

Vimarsh Bhatt[1], Prof. Preet Jain[2]
M. Tech. Scholar[1], Asst. Prof.[2] & HOD
Department of Electronics and Communication Engineering
Shri Vaishnav Institute of Technology and Science, Indore (M.P.) India
vimarshbhatt@gmail.com[1]

ABSTRACT

Digital Signal Processing is one of the leading technologies used in firstly growing application areas like wireless communication, audio signals and video signal processing and many industrial control equipments. The variety of products, where some form of Digital Signal Processing needed, grown largely in last few years. Now days Digital Signal Processing has become a key component. In many of the medical, communication, consumer and industrial products implements DSP using various aspects like microprocessors, ICs, Field Programmable Gate Array (FPGA) etc. Because of increase in popularity of mentioned application area, DSP capable processors expanded vastly in varieties. DSPs are microcomputers or processors whose software, hardware and instruction sets are optimized for high speed numeric processing applications, which is essential aspect for processing digital data. Increasing popularity of DSP processors is because of various advantageous features such as reprogramming ability at the field, cost effectiveness, efficiency, speed, energy etc.

The design process of digital filter is long and tedious if done by hand. With the aid of computer programs performing filter design algorithms, designing and optimizing filters can be done relatively quickly. This report discusses the use of Code Compose Studio, a software package, to design, manipulate and analyze digital filters.

A variety of filter design algorithms are available for both FIR and IIR filters. Different options are available today for conversion of analog filters to digital filters of Low Pass, High Pass, Band Pass and Band Stop filters. This project mainly discusses the of Low Pass FIR filter with Kaiser Window Technique on TMS320C6713 DSK.

***Keyword**:* Finite Impulse Response (FIR), Infinite Impulse Response (IIR), filters, TMS320C6713 DSK kit, Code Compose Studio (CCS), Window Technique, Digital Signal Processing (DSP).

INTRODUCTION

DSP techniques have been very successful because of the development of low-cost software and hardware support. For example, modems and speech recognition can be less expensive using DSP techniques. DSP processors are concerned primarily with real-time signal processing. Real-time processing requires the processing to keep pace with some external event, whereas non-real-time processing has no such timing constraint [1]. The external event to keep pace with is usually the analog input. Whereas analog-based systems with discrete electronic components such as resistors can be more sensitive to temperature changes, DSP-based systems are less affected by environmental conditions. DSP processors enjoy the advantages of microprocessors. They are easy to use, flexible, and economical.

Digital signal processors such as the TMS320C6x (C6x) family of processors are like fast Special-purpose microprocessors with a specialized type of architecture and an instruction set appropriate for signal processing. The C6x notation is used to designate a member of Texas Instruments' (TI) TMS320C6000 family of digital signal processors [2]. The architecture of the C6x digital signal processor is very well suited for numerically intensive calculations. Based on a very-long-instruction-word (VLIW) architecture, the C6x is considered to be TI's most powerful processor. Digital signal processors are used for a wide range of applications, from communications and controls to speech and image processing. The general-purpose digital signal processor is dominated by applications in communications (cellular). Applications embedded digital signal processors are dominated by consumer products. They are found in cellular phones, fax/modems, disk drives, radio, printers, hearing aids, MP3 players, high-definition television (HDTV), digital cameras, and so on. These processors have become the products of choice for a number of consumer applications, since they have become very cost-effective. They can handle different tasks, since they can be reprogrammed readily for a different application.

TMS320C6713 DSP STARTER KIT (DSK)

The TMS320C6713 DSP Starter Kit (DSK) developed jointly with Spectrum Digital is a low-cost development platform designed to speed the development of high precision applications based on TI´s TMS320C6000 floating point DSP generation.

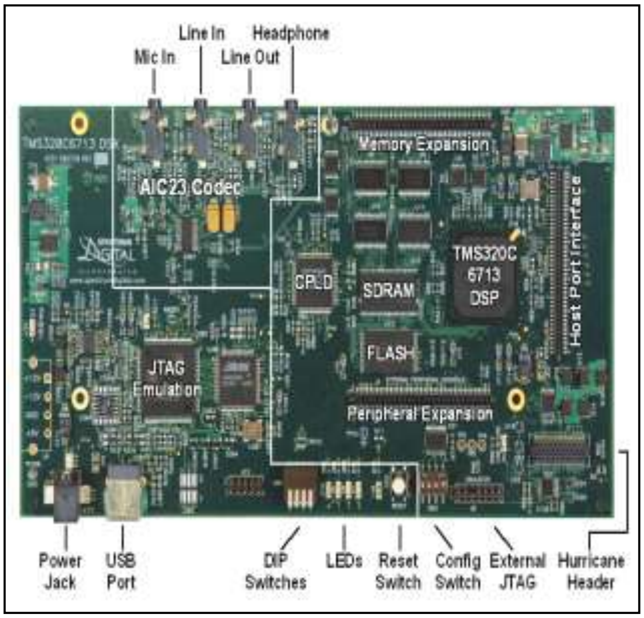

Figure 1: Layout DSK C6713 [2]

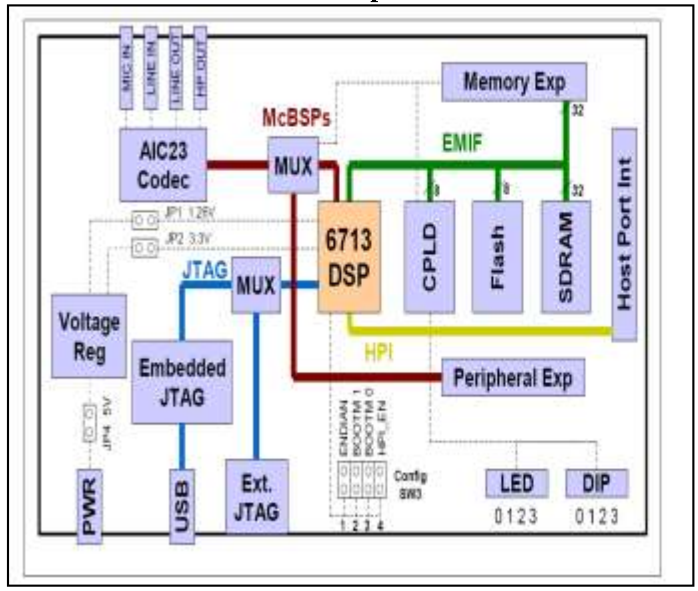

Figure 2: Functional Block Diagram of the DSK C6713 [2]

The kit uses USB communications for true plug-and-play functionality. Both experienced and novice designers can get started immediately with innovative product designs with the DSK full featured Code Composer Studio™ IDE and eXpressDSP™ Software which includes DSP/BIOS and Reference Frameworks.

The TMS320C6713 DSP Starter Kit is the newer version of the TMS320C6711 DSP Starter Kit. This DSK with up to 1800 MIPS of processing power allows the developing of algorithm in fields like networking, communications, imaging and other applications. Important for the project was the support of USB and enough processing power.

A. Features TMS320C6713 DSP Starter Kit

The DSK comes with a full complement of on-board devices that suit a wide variety of application environments. Key features include:

- A Texas Instruments TMS320C6713 DSP operating at 225 MHz
- An AIC23 stereo codec
- 16 Mbytes of synchronous DRAM
- 512 Kbytes of non-volatile Flash memory (256 Kbytes usable in default configuration)
- 4 user accessible LEDs and DIP switches
- Software board configuration through registers implemented in CPLD
- Standard expansion connectors for daughter card use
- JTAG emulation through on-board JTAG emulator with USB host
- Interface or external emulator
- Single voltage power supply (+5V)

The CPU is working with very-long instruction words (VLIW) (256 bits wide). The DSP 6713 interfaces on-board peripherals through a 32-bit wide EMIF bus (External Memory Interface). The SDRAM, Flash and CPLD (Complex Programmable Logic Device) are all connected to this bus, see Figure 2. Third parties use this expansion of the EMIF bus for video support, memory extension, other sound codec, etc. Analogue audio signals are accessed via an on-board AIC23 codec and four 3.5-mm audio jacks (microphone input, line input, line output and headphone output). The analogue input can be microphone (fixed gain) or line (boost), the output line-out (fixed gain) or headphone (adjustable gain).

The CPLD is a programmable logic device used to tie board components together and has a register-based interface to configure the DSK. The DSK has 4 LEDs and DIP switches to allow user to work interactive with the board. To use this interactive method the CPLD register gets read and written. Code Composer Studio communicates with the DSK via the integrated JTAG emulator on-Board. They are connected with a USB interface. Programs can be downloaded to the board into the SDRAM or Flash. The advantage of the flash memory is that it will keep the program after a restart of the board.

DIGITAL FILTER

In signal processing, the function of a filter is to remove unwanted parts of the signal, such as random noise, or to extract useful parts of the signal, such as the components lying within a certain frequency range. The filtering process alters the frequency content of a signal. For example, the bass control on a stereo system alters the low-frequency content of a signal, while the treble control alters the high-frequency content. Two common filtering applications are removing noise and decimation. Decimation consists of low pass filtering and reducing the sample rate. There are two main kinds of filter, analog and digital.

1. Analog Filters
2. Digital Filters.

A. *Types of FIR Filters*

As the terminology suggests, impulse response of FIR filter extends up to some finite period of time, because they are non-recursive. That means no feedback of previous response is provided in this structure. By varying the weight of the coefficient and the number of filter taps, virtually any frequency response characteristic response can be realized with an FIR filter. As has been shown, FIR filters can achieve performance levels which are not possible with analog filter techniques (such as perfect linear phase response). However, high performance FIR filters generally requires a large number of multiply-accumulates and therefore requires fast and efficient DSPs.

B. *Design Technique for FIR Filters Window Technique [5]*

- Rectangular Window Function

$$w_R(n) = 1, \quad for\ |n| \le \frac{M-1}{2}$$
$$= 0, \quad \text{Otherwise} \tag{1}$$

- Hamming Window Function

$$w_H(n) = 0.54 - 0.46\cos\frac{2\pi n}{M-1}, \quad for\ 0 \le |n| \le M-1$$
$$= 0, \quad \text{Otherwise} \tag{2}$$

- Hanning Window Function

$$w_{Hamn}(n) = 0.5 - 0.5\cos\frac{2\pi n}{M-1}, \quad for\ 0 \le |n| \le M-1$$
$$= 0, \quad \text{Otherwise} \tag{3}$$

- Blackman Window Function

$$w_b(n) = 0.4 - 0.5\cos + 0.08\cos\frac{4\pi n}{M-1}, for\ 0 \le |n| \le M-1$$
$$=0, \quad \text{Otherwise} \tag{4}$$

- Barlett Window Function

$$w_B(n) = 1 + n, \quad for -\frac{M-1}{2} < n < 1$$
$$=1 - n, \quad 1 < n < \frac{M-1}{2} \tag{5}$$

- Kaiser Window:

$$w_K(n) = \frac{I_0(\beta)}{I_0(\alpha)}, \quad for\ |n| \le \frac{M-1}{2}$$
$$=0, \quad \text{Otherwise} \tag{6}$$

- Low Pass FIR Filter

$$h_d(n) = \begin{cases} \left(\frac{2f_c}{F}\right)\frac{\sin\frac{2\pi n f_c}{F}}{\frac{2\pi n f_c}{F}}, & for\ n > 0 \\ 1 - \left(\frac{2f_c}{F}\right), & for\ n = 0 \end{cases} \tag{7}$$

- High Pass FIR Filter

$$h_d(n) = \begin{cases} -\left(\frac{2f_c}{F}\right)\frac{\sin\frac{2\pi n f_c}{F}}{\frac{2\pi n f_c}{F}}, & for\ n > 0 \\ 1 - \left(\frac{2f_c}{F}\right), & for\ n = 0 \end{cases} \tag{8}$$

- Bandpass Filter

$$h_d(n) = \begin{cases} \frac{1}{n\pi}\left[\sin\left({}^{2\pi n f_{c2}}/_{F}\right) - \sin\left({}^{2\pi n f_{c1}}/_{F}\right)\right], & for\ n > 0 \\ \frac{2}{F}(f_{c2} - f_{c1}), & for\ n = 0 \end{cases} \tag{9}$$

- Bandpass Filter

$$h_d(n) = \begin{cases} \frac{1}{n\pi}\left[\sin\left({}^{2\pi n f_{c1}}/_{F}\right) - \sin\left({}^{2\pi n f_{c2}}/_{F}\right)\right], & for\ n > 0 \\ \frac{2}{F}(f_{c1} - f_{c2}) + 1, & for\ n = 0 \end{cases} \tag{10}$$

SIMULATION RESULTS

The design of FIR filter using hamming window function for different values of ripple and frequency are shown in the figure below. A Finite Impulse Response (FIR) filter is a discrete linear time-invariant system whose output is based on the weighted summation of a finite number of past inputs. An FIR transversal filter structure can be obtained directly from the equation for discrete-time convolution. Required Equipments: Operating System – Windows XP, Constructor -Simulator , Software - CCStudio 3-TMS320C6713 DSK Board.

A. *Graph Property Dialog*

Table 5.1 Graph Property Dialog

Display Types	Single Time
Graph Title	FIR_TRIANG
Start Address	Wt
Acquisitions Buffer Size	64
Index Increment	1
Display Data Size	64
DSP Data Type	32-bit floating point
Sampling Rate (Hz)	1
Plot Data Form	Left to Right
Left-shifted Data Display	Yes
Auto scale	On
DC Value	0
Time Display Unit	s

Status Bar Display	On
Magnitude Display Scale	Linear
Data Plot Style	Bar
Grid Style	Zero Line
Cursor Mode	Data Cursor

B. Bartlett window

A Bartlett window is a triangular shaped window function. The Bartlett window has higher side lobe attenuation than the rectangular window. The Bartlett window is defined:

$$w(n) = 1 - \frac{\left|n - \frac{M-1}{2}\right|}{\frac{M-1}{2}}$$

(11)

The frequency response for Bartlett window is shown in figure 3 and figure 4 shows the frequency response of a lowpass FIR filter designed using Bartlett window.

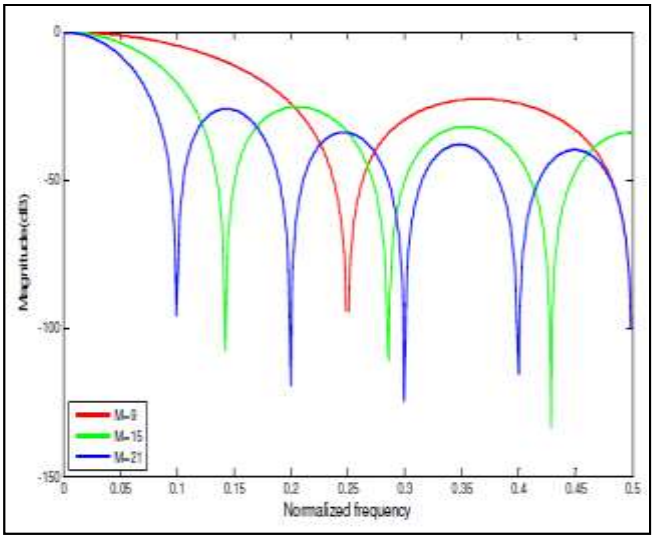

Figure 3: Frequency response for Bartlett window

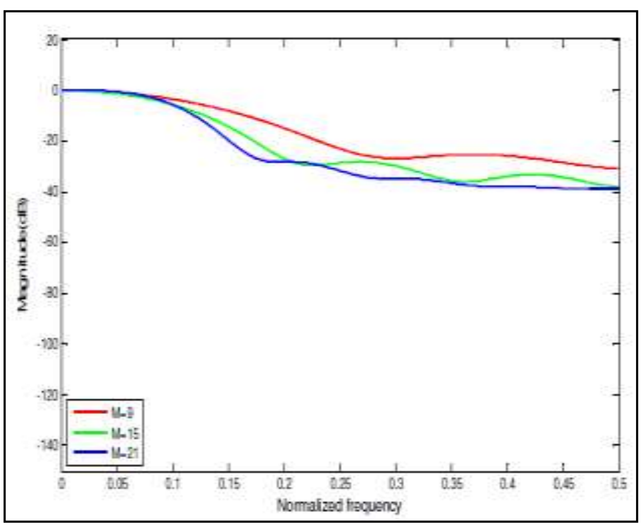

Figure 4: Lowpass FIR filter designed with Bartlett Window

C. Hamming window

The Hamming window is, like the Hanning window, also a raised cosine window. The Hamming window exhibits similar characteristics to the Hanning window but further suppress the first side lobe. The Hamming window is defined as in equation 12

$$w(n) = 0.54 - 0.46Cos\frac{2\pi n}{M-1} \quad , for\ 0 \le |n| \le M - 1 \quad (12)$$

The frequency response for Hamming window is shown in figure 5 and figure 6 shows the frequency response of a lowpass FIR filter designed using Hamming window.

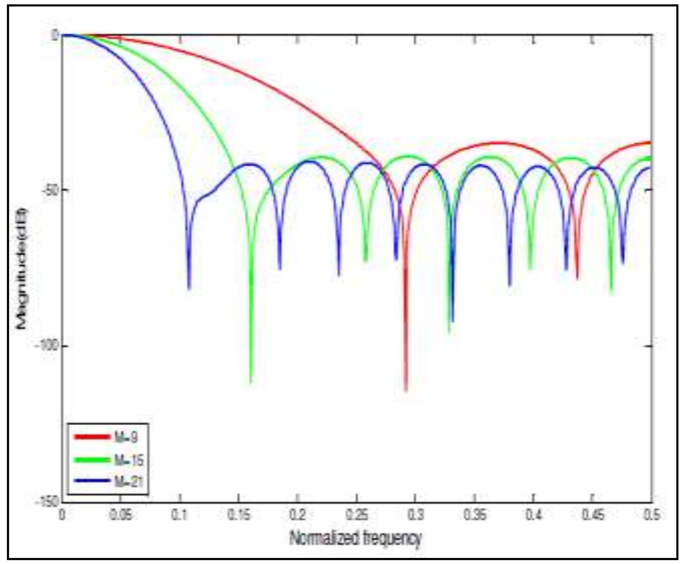

Figure 5: Frequency response for Hamming window

D. Blackman window

The Blackman window is similar to the Hanning and the Hamming windows. An advantage with the Blackman window over other windows is that it has better stopband attenuation and with less passband ripple. The Blackman window is defined as in equation 13.

$$w(n) = 0.42 - 0.5cos\frac{2\pi n}{M-1} + 0.08cos\frac{4\pi n}{M-1}$$

(13)

The frequency response for Blackman window is shown in figure 7 and figure 8 shows the frequency response of a lowpass FIR filter designed using Blackman window.

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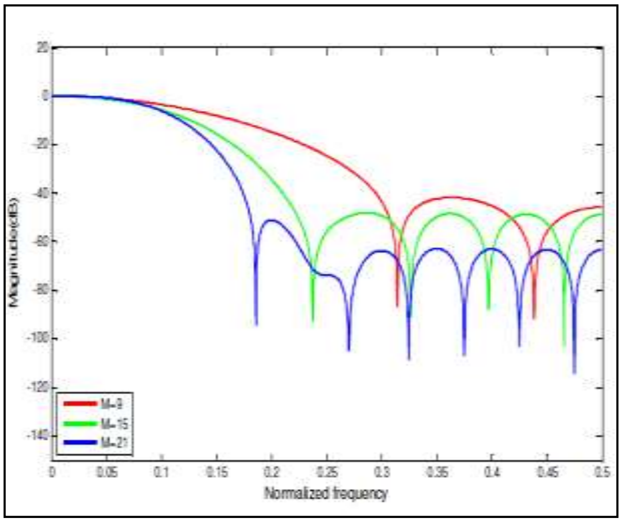

Figure 6: Lowpass FIR filter designed with Hamming Window

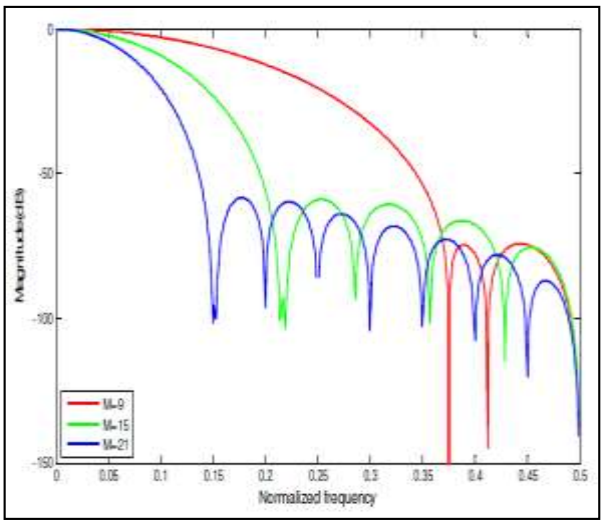

Figure 7: Frequency response for Blackman window

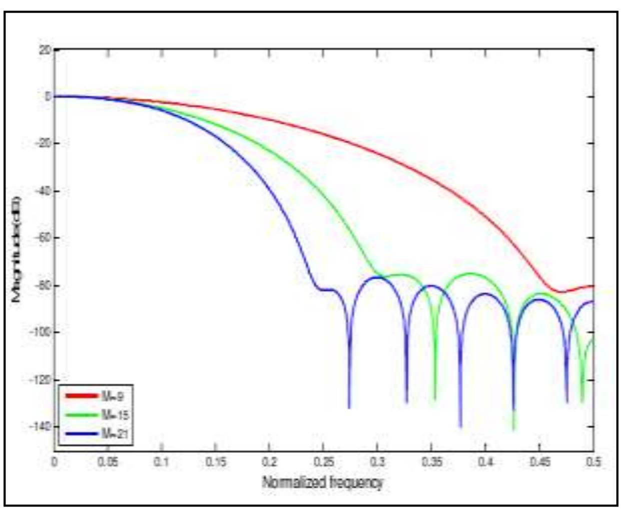

Figure 8: Lowpass FIR filter designed with Blackman Window

CONCLUSION

Digital Signal Processors (DSPs) are special-purpose microprocessors designed with specialized architectures very suitable for different type of signal processing applications. The flexibility though reprogramming and the power efficiency provided by the DSP made it very suitable for most embedded applications. The TMS320C6713 DSP Starter Kit (DSK) developed jointly with Spectrum Digital is a low-cost development platform designed to speed the development of high precision applications based on TI´s TMS320C6000 floating point DSP generation.

We can see that the response of Blackman window is more smooth and perfect than that of the Hamming window. So, the Black man window is more perfect and advantageous than that of Hamming window. For FIR lowpass filter design, there are four sidelobes exists in case of Hamming window but for same specification and filter order, in Blackman window, there exists only two sidelobes, that the efficiency is increased losing less power. So, it is clear that Blackman window technique is more powerful and perfect than the FIR filter designed with a Hamming window.

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