FACULTY OF ENGINEERING, UNIVERSITY OF JAFFNA

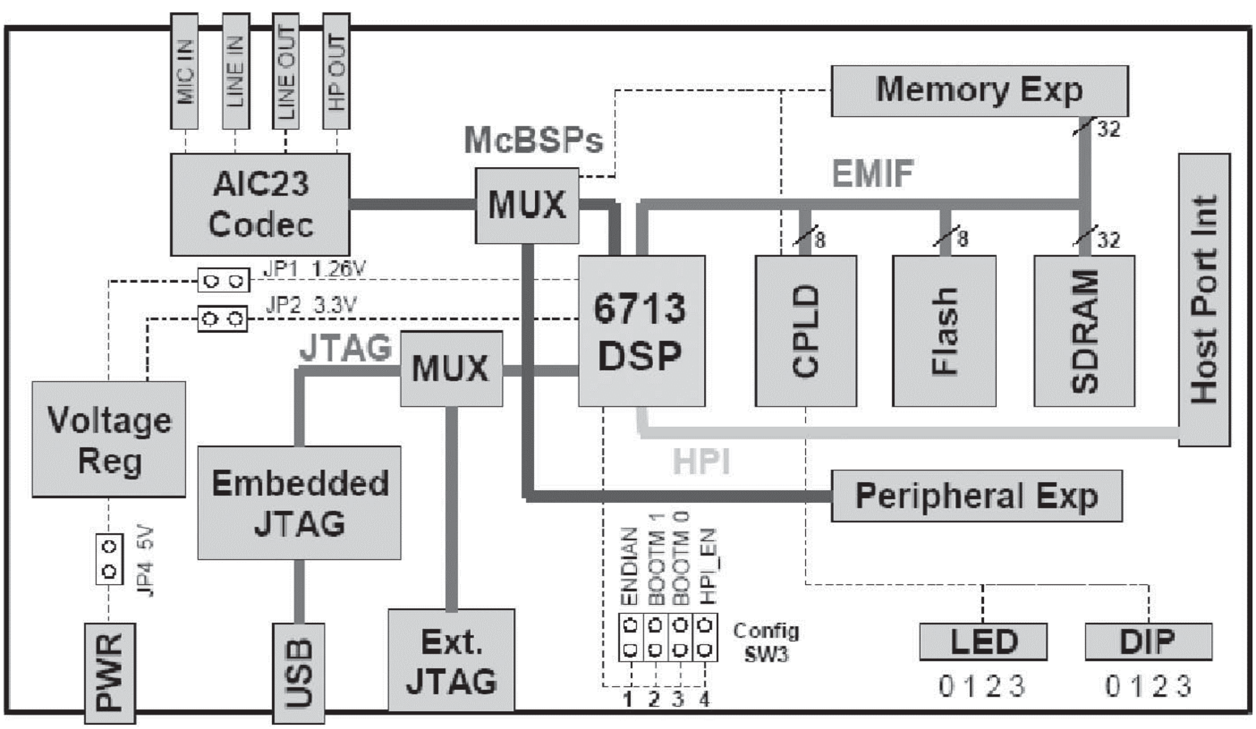
DIGITAL SIGNAL PROCESSING – EC5010

LABORATORY SESSION 4

DIGITAL SIGNAL PROCESSING using TMS320c6713 DSK

Introduction

The Texas Instrument TMS320C6713 Digital Signal Processing Starter Kits are development platforms for real - time digital signal and Image processing applications. It comprises a small circuit board containing aTMS320C6713 floating - point digital signal processor and a TLV320AIC23 analog interface circuit (codec) and connect to a host PC via a USB port. PC software in the form of Code Composer language to be compiled and/or assembled, linked, and downloaded to run on the DSK.



Part 1 - Familiarizing the interfaces in DSK

There are four interfaces inTMS320C6713 DSK. They are Codec, Dip switch, LED and flash memory. These are here to communicate with real world.

1) Implement the following task.

If you press dip switch 0 ,the codec should takes input from any audio source and output to a headphone an blink led number 0.

freq ID	Value	Frequency
DSK6713_AIC23_FREQ_8KHZ	0x06	8000 Hz
DSK6713_AIC23_FREQ_16KHZ	0x2c	16000 Hz
DSK6713_AIC23_FREQ_24KHZ	0x20	24000 Hz
DSK6713_AIC23_FREQ_32KHZ	0x0c	32000 Hz
DSK6713_AIC23_FREQ_44KHZ	0x11	44100 Hz
DSK6713_AIC23_FREQ_48KHZ	0x00	48000 Hz
DSK6713_AIC23_FREQ_96KHZ	0x0e	96000 Hz

Part 2 - Generating different waveforms.

Create a sine lookup table and generate a sinewave. Eight points are enough to create the wave. First you have to find lookup table values.

Signal frequency=1KHz

Sampling frequency-8KHz

Now change the sampling frequency and observe the waveform.

How can you create a different frequency of a sinewave for a particular sampling frequency.

Generate sine wave using inbuilt sin function.

Observe the wave using headphone or oscilloscope.

Compare results between above two methods.

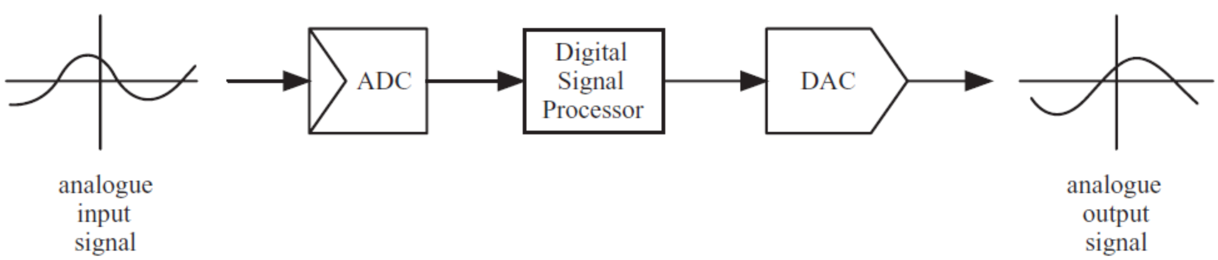
Part 3 Implementing convolution algorithm

Create x[n] and plot graph

Create h[n] and plot graph

Find y[n] using convolution and plot the output

Part 4 Designing digital filter.(FIR)



Open filter Designer tool in matlab.(>>filterDesigner-type in command)

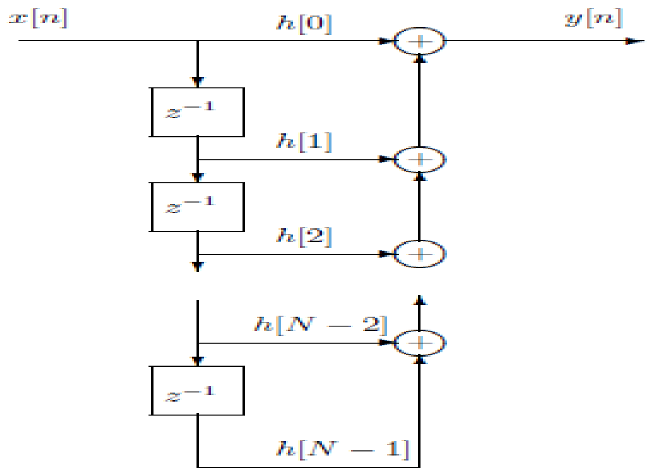
Design a lowpass FIR filter with following characteristics.

Sampling frequency-48000Hz

Fpass-3000Hz

Fstop-5000Hz

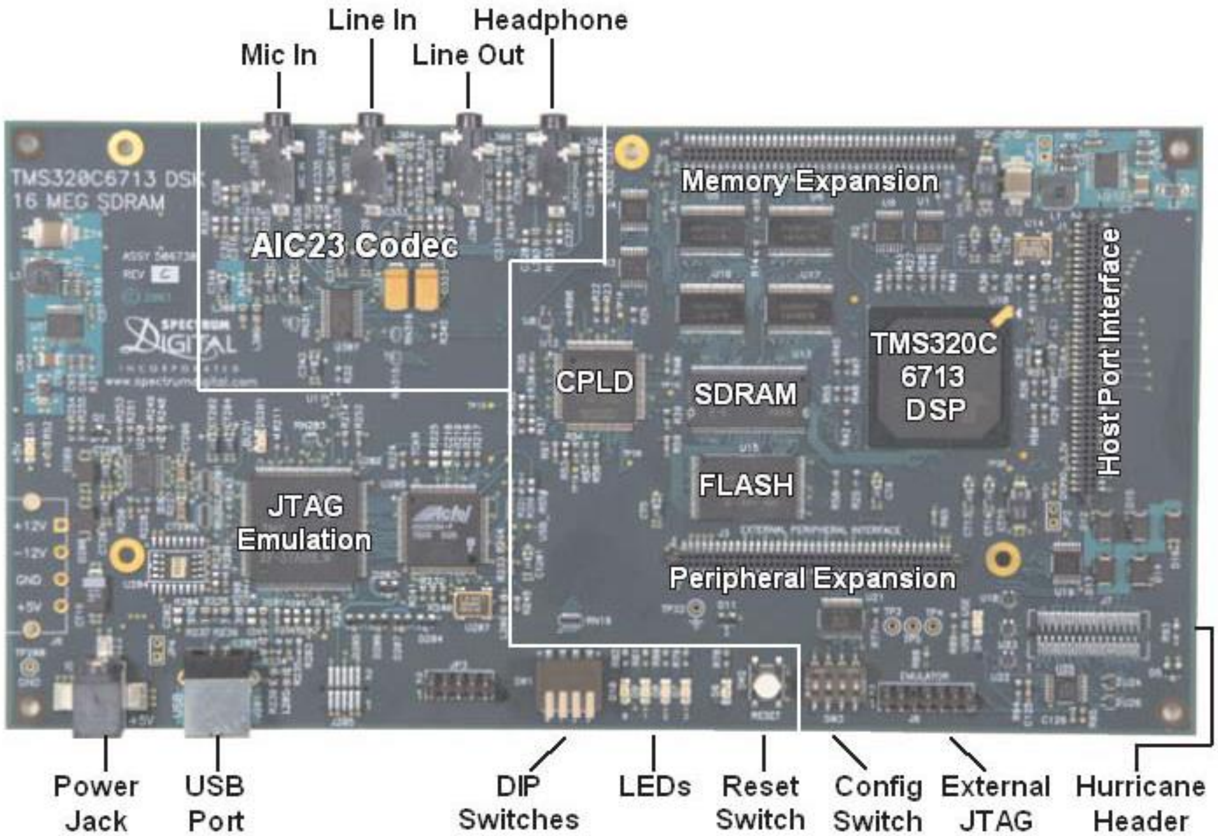
Generate the coefficient header file in filter Designer.(go to target files>)



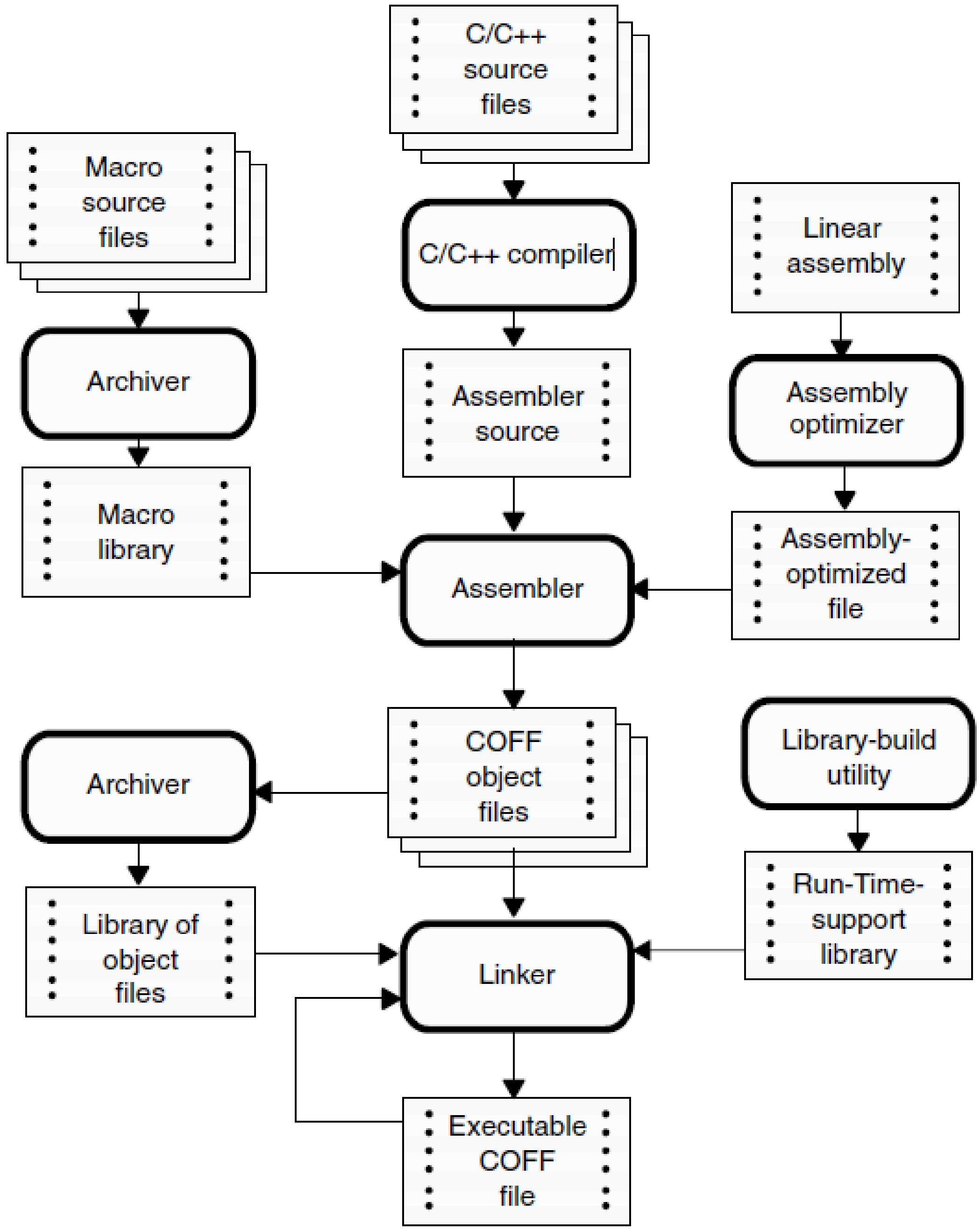
Using the filter coefficients build a FIR filter in CCS.

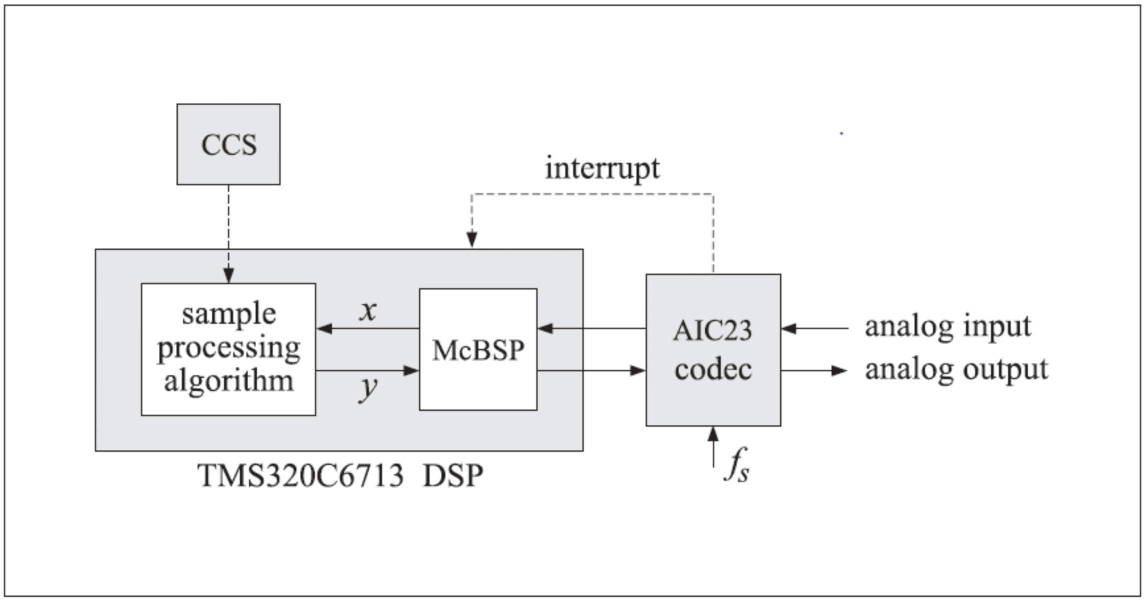
Load the board and observe.

Appendix



Mic In
Line In
Line Out
Headphone
Memory Expansion
AIC23 Codec
TMS320C
6713
DSP
CPLD
SDRAM
FLASH
Host Port Interface
JTAG
Emulation
Peripheral Expansion
TMS320C6713 DSK
16 MEG SDRAM
Power
Jack
USB
Port
DIP
Switches
LEDs
Reset
Switch
Config
Switch
External
JTAG
Hurricane
Header


C/C++ source files
Macro source files
C/C++ compiler
Linear assembly
Archiver
Assembler source
Assembly optimizer
Macro library
Assembler
Assembly-optimized file
COFF object files
Library-build utility
Archiver
Run-Time-support library
Library of object files
Linker
Executable COFF file



Shortcuts

compile & load: F7,   Project -> Build

run program: F5,   Debug -> Run

halt program: Shift+F5,   Debug -> Halt

Key features

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

Configurable boot options

Standard expansion connectors for daughter card use

JTAG emulation through on-board JTAG emulator with USB host

Note

All processing operations during the execution of the ISR must be completed in the time nterval between samples, that is, T = 1/fs. For example, if fs = 44.1 kHz, then, T = 1/fs = 22.68 μsec. With an instruction cycle time of Tc = 4.44 nsec, this allows T/Tc = 5108 cycles to be executed during each sampling instant, or, up to 8×5108 = 40864 instructions, or half of that per channel.