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Design of Data Acquisition System (DAS) With A Digital Notch Filter Based on Digital Signal Processing (DSP)

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ABSTRACT: *This research is aimed at using Data Acquisition System (DAS) in designing a Notch Digital Filter based on Digital Signal Processing (DSP). In order to successfully implement the research, the following characteristics were put to use: Th Central Frequency, F0 is 50Hz; the Bandwidth of Digital Filter, dF is +10 Hz; the Sample Frequency, Fs is 10 KHz; The work sets out to compare two Digital Filter types: Finite Impulse Response (FIR) filter type and Infinite Impulse Response (IIR) filter type. The former being when Impulse response of the filter falls to zero after a finite period of time, and the latter being when the impulse response exists indefinitely. The research also sets out to Estimate Accuracy, Complexity and Quality of Digital Filters. Consequently, a program is written for the filters using the main element DSP of family ADSP-218x.The power supply is or range +5.5Volts…+7Volts and the DAS must be connected to the Personal Computer (PC) through Universal Serial Bus (USB) standard RS-232 interface. MATLAB software is used for the design and Analysis of the Digital Filter Characteristics.*

KEYWORDS*- high-pass, low-pass, high-band, notch & digital filters*

I. INTRODUCTION

Data Acquisition System is one of the first step in analysing data in Digital Signal Processing (DSP). [5]. Data Acquisition Systems as the name implies, are the products and/or processes used in collecting information for documentation and analysis of some phenomenon [5]. Technically logging the temperature of an object with any instrument is a form of data acquisition. Technology has however, brought some level of sophistication in making the process of data acquisition easier, simpler, efficient, more accurate, versatile and reliable through electronic equipment or devices. Such devices include but not limited to sensors can monitor flow, level, temperature and pressure. Few terminologies in data acquisition are: Analog-to-Digital Converter (ADC); Digital-to-Analog Converters (DAC); Digital Input and Output (DIO); Differential Input; General Purpose Interface Bus; Resolution; RS-232/485; Sample Rate and Single-Ended input (SE). Filtering is the process of altering the frequency content of a signal [8]. It is probably the most widely used signal processing operation. Filters are ubiquitous in the world around us, with the most common examples being in our home stereo systems. The various types of data acquisition system are based on characteristics such as distance it supports, the devices to be connected, DAS sample rate, the power source and the cost of implementation [1].
Few Data Acquisition Systems (DAS) are:

- Serial Communication Data Acquisition System
- USB Data Acquisition System
- Data Acquisition Plug-in Boards
- Parallel Port Data Acquisition

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There are two main classes of digital filters namely: Finite Impulse Response Filters and Infinite Impulse Response filters [9]. As such, the processes involved in designing them also depends on which Filter one is designing.

In subsequent chapters, there is an appreciation of the practical considerations such as phase distortion, and particular implementations of the filters under discussion.

CLASSIFICATION OF FILTERS

Filters, both in continuous (Analogue) and discrete-time (Digital), can be grouped loosely into one of four categories:

- Low-Pass,
- High-Pass
- Band-Pass and
- Notch

This research work centres on the Notch category of Filters. However, the different types of filters are very intuitive.

Low Pass Filters:

Low-Pass filters allow the passage of low frequencies, but stop high frequencies.

High Pass Filters:

A bandpass filter passes a band of frequencies, but stops other frequencies

II. DIGITAL (DISCRETE TIME) FILTERS

The digital computation of filter transfer functions has always been an important area of digital signal processing. Apart from the obvious advantages of virtually eliminating errors in the filter associated with voltage and temperature drift, component aging, and EMI-induced power supply noise, digital filters are capable of performance specifications that would, at best, be extremely difficult to achieve with an analogg implementation. Digital filters are able to realize sharp cutoff characteristics, tight passband and stopband specifications, exactly linear phase responses, and even arbitrary magnitude responses. Many of the routines in this chapter make use of circular buffers for storing data and coefficients. To implement circular addressing, the length register (Ln) that corresponds to the circular buffer pointer register (In) must be set to the buffer length.

TYPES OF DIGITAL FILTERS

Fir Digital Filters

The digital computation of filter transfer functions has always been an important area of digital signal processing. Apart from the obvious advantages of virtually eliminating errors in the filter associated with voltage and temperature drift, component aging, and EMI-induced power supply noise, digital filters are capable of performance specifications that would, at best, be extremely difficult to achieve with an analog implementation. Digital filters are able to realize sharp cutoff characteristics, tight passband and stopband specifications, exactly linear phase responses, and even arbitrary magnitude responses. Many of the routines in this chapter make use of circular buffers for storing data and coefficients. To implement circular addressing, the length register (Ln) that corresponds to the circular buffer pointer register (In) must be set to the buffer length.

INVESTIGATING THE FIR DIGITAL FILTER

To investigate this filter type, we shall implement our task according to this filter type and then make some observations.

Using the given task parameters:
Central Frequency: F_0=50 Hz,
Bandwidth of Digital Filter: dF=$\pm$10 Hz.
Sample Frequency: F_s=10 KHz.

III. RESULTS

MATLAB DESIGN AND ANALYSIS OF FIR FILTER:
Fpass1 = 40; % First Passband Frequency
Fstop1 = 49.9; % First Stopband Frequency
Fstop2 = 50.1; % Second Stopband Frequency
Fpass2 = 60; % Second Passband Frequency
Apass1 = 1; % First Passband Ripple (dB)
Astop = 60; % Stopband Attenuation (dB)
Apass2 = 1; % Second Passband Ripple (dB)
Fs = 10000; % Sampling Frequency
h = fdesign.bandstop('fp1,fst1,fst2,fp2,ap1,ast,ap2', Fpass1, Fstop1, ... Fstop2, Fpass2, Apass1, Astop, Apass2, Fs);
Hd = design(h, 'equiripple', ...
'FilterStructure', 'dfsymfir');

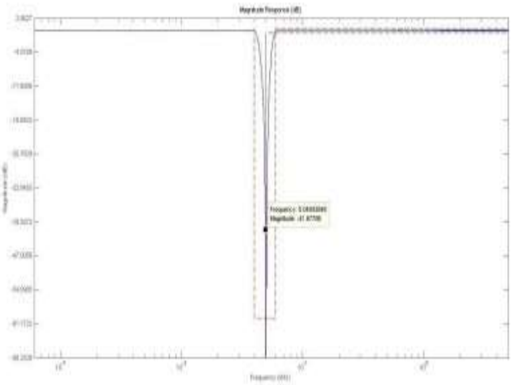


Fig 1: FIR type magnitude response

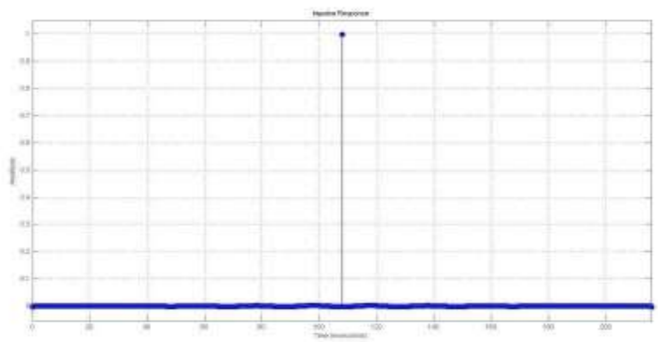


Fig 2: FIR type impulse response

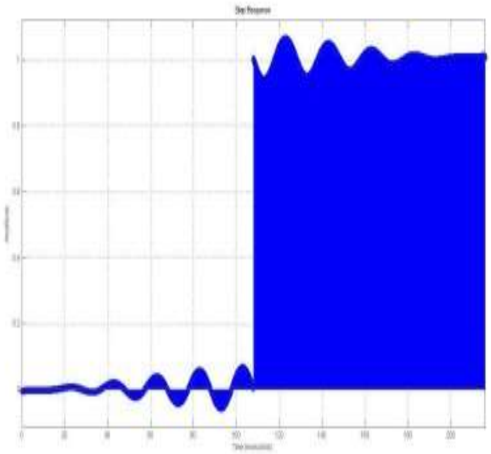
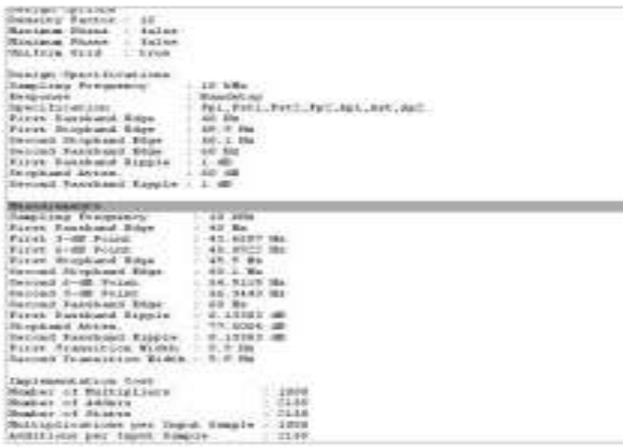


Fig 3: FIR type Step response



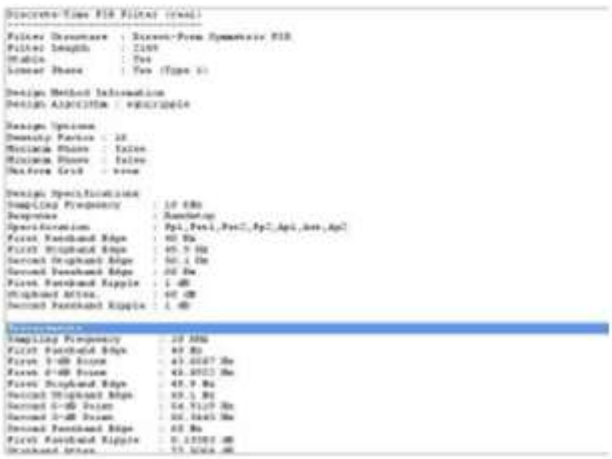


Fig 4 &5 FIR filter information

CORRESPONDING ASSEMBLY PROGRAM PROTOTYPE FOR THE ADSPMODULE fir_sub;{
FIR Transversal Filter Subroutine Calling Parameters
I0 —> Oldest input data value in delay line L0 = Filter length (N)
I4 —> Beginning of filter coefficient table L4 = Filter length (N)
M1, M5 = 1
CNTR = Filter length - 1 (N-1) Return Values
MR1 = Sum of products (rounded and saturated)
I0 —> Oldest input data value in delay line I4 —> Beginning of filter coefficient table Altered Registers
MX0, MY0, MR

Computation Time
N - 1 + 5 + 2 cycles
All coefficients and data values are assumed to be 1.15format.
}
. ENTRY fir;
fir: MR=0, MX0=DM (I0, M1), MY0=PM (I4, M5); DO sop UNTIL CE;
sop: MR=MR+MX0*MY0(SS), MX0=DM (I0, M1), MY0=PM(I4, M5); MR=MR+MX0*MY0(RND);
IF MV SAT MR;
RTS;
. ENDMOD;

Code For Double Precision FIR implementation on the ADSP:
. MODULE dfir_sub;
{
Double-Precision Transversal Filter Subroutine

Calling Parameters
I0 —> Oldest input data value in delay line L0 = 2 × Filter length (N)
I4 —> 2nd location (LSW of 1st value) of filter coefficient tab L4 = 2 × Filter length (N)
I4 —> 2nd location (LSW of 1st value) of filter coefficient tab L4 = 2 × Filter length(N)
M0,M4 = 1
M1,M5 = 2
M2,M6 = 3
AX0=Filter length - 2 (N-2) CNTR = Filter length - 2 (N-2)
Return Values
MR1,MR0 = sum of products
(conditionally saturated to 32 bits)
I0 —> Oldest input data value in delay line
I4 —> 2nd location (LSW of 1st value) of filter coefficient tab Altered Registers
MX0,MY0,MR
Computation Time
3 × (N - 2) + 16 + 9
All coefficients and data values are assumed to be in 1.15 format.
}
.ENTRY dfir;

dfir: MR=0, MX0=DM(I0,M1), MY0=PM(I4,M5);
DO hlloop UNTIL CE;
hlloop: MR=MR+MX0*MY0(SU), MX0=DM(I0,M1), MY0=PM(I4,M5);

```
MR=MR+MX0*MY0(SU), MX0=DM(I0,M2), MY0=PM(I4,M4);
 MR=MR+MX0*MY0(SU), MX0=DM(I0,M1),
MY0=PM(I4,M5);
 CNTR=AX0;
DO lhloop UNTIL CE;
lhloop: MR=MR+MX0*MY0(US), MX0=DM(I0,M1), MY0=PM(I4,M5); MR=MR+MX0*MY0(US), MX0=DM(I0,M0),
MY0=PM(I4,M5);
MR=MR+MX0*MY0(US), MX0=DM(I0,M1), MY0=PM(I4,M5);
MR0=MR1; {downshift 16 places}
MR1=MR2;
CNTR=AX0;
DO hhloop UNTIL CE;
hhloop: MR=MR+MX0*MY0(SS), MX0=DM(I0,M1), MY0=PM(I4,M5); MR=MR+MX0*MY0(SS),
MX0=DM(I0,M1), MY0=PM(I4,M6);
MR=MR+MX0*MY0(SS);
IF MV SAT MR;
RTS;
ENDMOD
```

COMPARISON OF IIR AND FIR FILTERS

After the design analysis of both IIR and FIR digital filters with the same parameters, the following observations and comparisons can be made:

From the filter information sheets above, we see that the implementation of the FIR type requires the following:

1080 multipliers, 2159 adders, And it does 1080 multiplications and 2159 additions per input sample. The IIR type requires 27 multipliers, 20 adders, And it does 27 multiplications and 20 additions per input sample.

IV. CONCLUSION

The FIR filters non-recursions and no computational loops allow for easier implementation on the ADSP. However, the design implementation requires more adders and multipliers and does more calculations per input cycle.

The IIR filters have recursions and feedback (computational) loops. This allows for fewer calculation steps per input cycle by the ADSP.

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