Design and Study of Digital FIR Filters

Pesha D. Petrova[1] , Boyan D. Karapenev[2]

***Abstract* - An algorithm for the design of digital FIR filters by using DSP TMS320C5x and MATLAB program package has been developed. A program code for DSP control has been created. Low-pass filter, High-pass filter, Band-pass filter and Band-stop filter of various orders at various sample frequencies have been designed. Amplitude frequency responses from the study and simulation of the designed variants have been presented. The results allow the correct register adjustment of the analog interface of DSP. They indisputably support the correctness of the algorithm proposed.**

***Keywords* – FIR filters, DSP, amplitude frequency response, MATLAB**

I. Introduction

It is known [1], [2] that the output signal *y(n)* of FIR filters at a certain moment *n* is a function of the input signal *x(n)* at this moment and of its previous values *x(n-i)*. A mathematical representation in direct form of a FIR filter is given as

$$y(n) = F\big((x(n), x(n-1),..., x(n-M)\big), \qquad (1)$$

where M determines the filter order and *n =0, 1, …., N-1 (N* - the number of samples).

For constant parameter linear filters the functional relation is set by a linear combination of input samples *x(n-i)* by coefficients of proportionality b_i , $i = 0,....,M$. Where at

$$y(n) = b_0 x(n) + b_1 x(n-1) + ... + b_M (x-M) = \sum_{i=0}^{M} b_i x(n-i) \qquad (2)$$

is obtained.

This equation is the result from the vector inner product between the filter coefficient vector *b* and the order-reversed input data vector *x*.

The same equation is presented graphically in Fig.1.

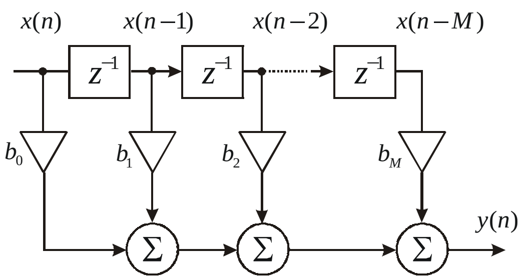

Fig. 1. Direct-form FIR.

[1]Pesha D. Petrova is with the Department of Communication Equipment and Technologies, Technical University of Gabrovo, 5300 Gabrovo, 4, Hadji Dimitar St.,Bulgaria, E-mail: daneva@tugab.bg

[2]Boyan D. Karapenev is with the Department of Communication Equipment and Technologies, Technical University of Gabrovo, 5300 Gabrovo, 4, Hadji Dimitar St.,Bulgaria, E-mail: bkarapenev@tugab.bg

The design and study of FIR filters is reduced to determining b_i coefficients by defined parameters and filter type [3], block diagram synthesis, visualization and analysis of functional characteristics and parameters.

These problems can be solved successfully by suitable software and digital signal processors (DSP).

II. Design and Study of FIR Filters

The MATLAB [4] – DSP medium is particularly suitable for the design and study of FIR filters. The principal stages in the design-study process of FIR filters by MATLAB and DSP TMS320C5x [5] are represented in the block diagram in Fig.2.

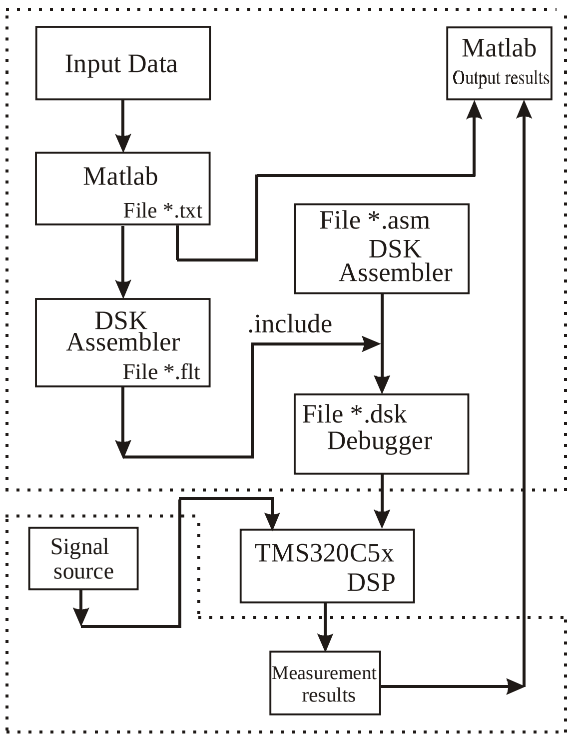

Fig. 2. Block diagram of FIR filters design-study process.

The procedure of filter synthesis and analysis is reduced to:

1. Calculating the filter coefficients

The coefficients of the filter designed are calculated, scaled and recorded in a text file *.txt by MATLAB file which contains the following input data:

- filter type: low-pass filter, high-pass filter, band-pass filter, band-stop filter;
- filter order;
- sampling frequency;
- cut-off frequency.

2. Creating a file *.flt

The coefficients from *.txt file are recorded in *.flt file by means of the directives .set and .word. Since delay circuits are also needed for the design of FIR filter, then information about them is entered in *.flt file by the directive .word.

3. Creating a file *.asm

This file contains the major source code of the digital filtration program. The adjustments of registers of the analog integrated circuit (AIC) are contents of great importance while creating the file. They can be set by the DSP program in order to adjust the sampling frequency of the AIC as well as the cut-off frequencies in several built – in filters. TA and TB are necessary for transmitting data to the AIC (D/A converting) whilst RA and RB are used for receiving data (A/D converting). Thus, there can have different sampling frequencies for either direction. Each mentioned register works in conjunction with a counter, which is loaded with the value hold in the corresponding register. The counters are decremented each clock cycle and generate an interrupt if zero is reached. They are then reloaded with the register value again. There are two counters for each direction because two frequencies have to be set – the sampling frequency and the switched capacitor frequency which controls the several band-pass and low-pass filters. The input band-pass filters can be switched off by the software and are not used by default. The output filter however is always active, thus, has to be taken in consideration. It is a low-pass filter that cuts - off frequencies above a special threshold. The A registers (TA or RA) define this threshold. The switched capacitor frequency f_{SCF} can be expressed by

$$f_{SCF} = \frac{f_{MCLK}}{2counter_A} \tag{3}$$

where f_{MCLK} is the frequency produced by the DSP timer, i.e. Master Clock Frequency. It is usually 10 MHz and $counter_A$ is defined by TA or RA.

The cut-off frequency of output filter can be expressed by

$$f_{cut-off} = \frac{f_{SCF}\,3700}{288000}\,. \tag{4}$$

To adjust the sampling rate it has to set the B registers accordingly. The sample rate should be double the cut-off frequency if frequencies in this range are expected. So

$$f_s = \frac{f_{SCE}}{counter_B} = \frac{f_{MCLK}}{2counter_A.counter_B} \tag{5}$$

where $counter_B$ is defined by TB or RB.

RA and TA (and RB and TB) should have equal values in order to get predictable results because TA/TB control the output sampling frequency but RA/RB however clock the algorithm which generates the desired output waveform. Thus, they have a direct relationship.

4. Creating a file *.dsk

By the command

dsk5a [filename] [options]

a file *.dsk is created by means of which DSP is controlled and the desired digital filter is realized. The files *.asm and *.flt are linked by the directive .include.

5. Experimenting the designed FIR filters

Before starting the loaded *.dsk file, a constant amplitude signal is sent from a source signal to DSP. When the program code is correct the type of filter, fixed by the code, is designed. The output signal amplitude for various frequencies is measured and it is useful to display its wave, which is a criterion of the correctness of the scaling coefficients. The latter are corrected if required.

6. Processing of results

In the procedure of design and study of filters it has been made possible the theoretical and experimental results to be processed by MATLAB. To this purpose a data file is formed about the measured amplitude-frequency response and the coefficients recorded in *.txt file by which the filter amplitude-frequency response is simulated. The real and simulated characteristics are visualized in a general system of coordinates. The summarized file from Fig.3 performs the overall data processing.

```
%***********************************************
%Amplitude Frequency Responses of FIR Filters
%***********************************************
%Calculate the FIR filter coefficients
b = fir1(M, wn, ‘ftype’);
c = round(b*2^k);  %k is the scaling factor
%***********************************************
%Create a formatted data file
fid = fopen(filename.txt, ‘w’);
fprint(fid, ‘%d\n’, c);
fclose(fid);
%***********************************************
%Simulate the FIR filter amplitude-frequency response
[h,w] = freqz(b, 1, n); m = abs(h);
%***********************************************
%Generate vectors containing measured data
f = [];  %frequencies
m1 = [];  %amplitudes
%***********************************************
%Visualization of real and simulated frequency responses
plot(f, m1,’k--‘, m,’k’);
%***********************************************
```

Fig.3. Summarized MATLAB file for simulation and visualization of real and simulated frequency response of FIR filters.

In order to estimate the results promptly and make a proper choice of parameters and adjustment when designing digital filters with defined parameters, in addition to MATLAB, we recommend processing of the experimental data by EXCEL as well.

III. Experimental and Simulated Results

To estimate the presented approach, low-pass, high-pass, band-pass and band-stop FIR filters of various order for various sampling frequencies have been designed and studied. The design is based on DSP TMS320C5x and a suitable program code created. In compliance with Eqs. (3), (4) and (5) and the recommendations in [6], [7] Table I shows variants of parameter values necessary for the development of the filters.

TABLE I
VALUES OF TA/RA, TB/RB, FS AND SNR

TA/RA	TB/RB	F_S, kHz	SNR, dB
10	50	10.000	80
16	31	10.081	60
5	50	20.000	67
8	31	20.161	78
4	26	48.077	60

The information from Table I is represented graphically in Fig.4.

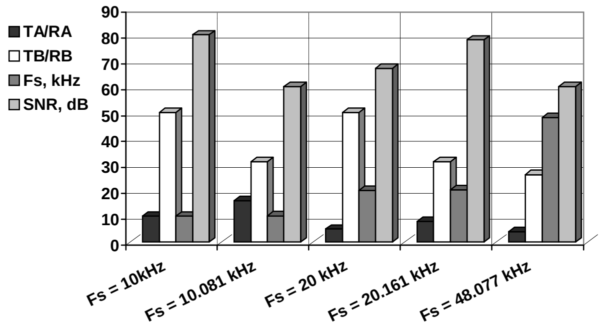

Fig.4. Relations between TA/RA, TB/RB, Fs and SNR.

The graphic representation shows indisputably that the values of registers TA/RA and TB/RB, which determine the sampling frequency, depend substantially on the signal/noise ratio. This fact has to be taken into account when adjusting the registers in the process of design of digital filters.

The simulation and experimental results of designing FIR filters of 85^{th} order are given below.

For each of the 4 types of filters (low-pass, high-pass, band-pass and band-stop) the measurement data are processed by MATLAB and by EXCEL.

Fig. 5, Fig. 6, Fig. 7 and Fig. 8 show the experimentally obtained and simulated by MATLAB amplitude frequency response, respectively for low-pass filter, high-pass filter, band-pass filter and band-stop filter at sampling frequency of Fs=10.081 kHz.

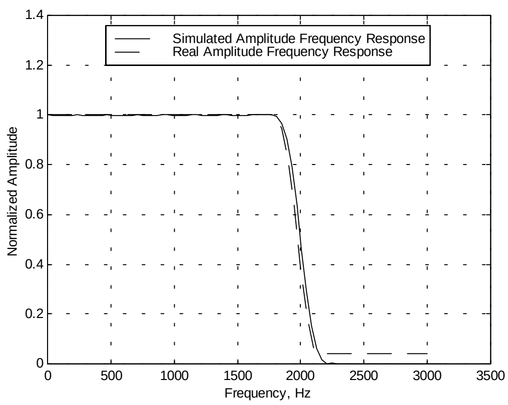

Fig. 5. Measured and simulated amplitude frequency responses for low-pass filter.

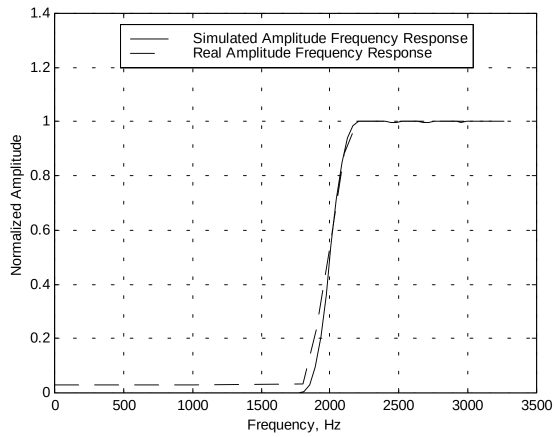

Fig. 6. Measured and simulated amplitude frequency responses for high-pass filter.

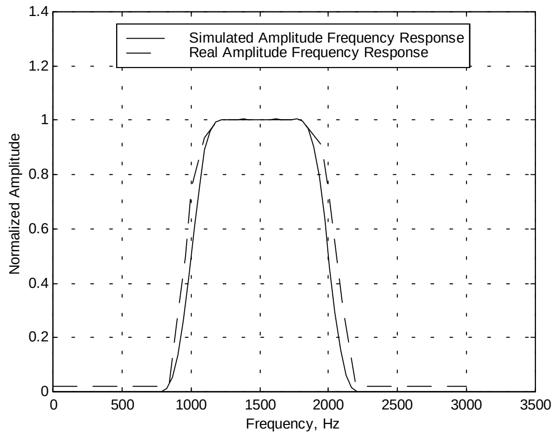

Fig. 7. Measured and simulated amplitude frequency responses for

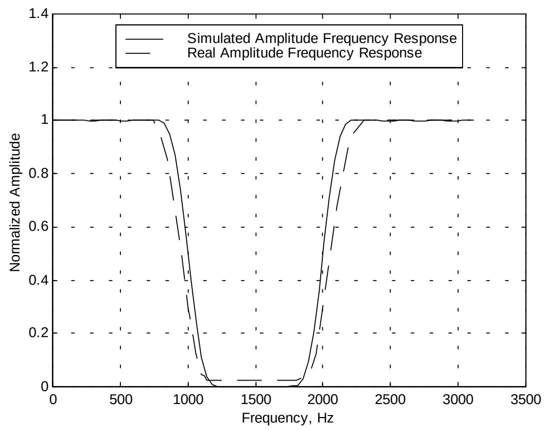

Fig. 8. Measured and simulated amplitude frequency responses for band-stop filter.

As it is seen in the figures, the simulated and real characteristics are very close. From the obtained amplitude frequency response of the designed filters at various sampling frequencies, the cut-off frequencies are determined and the summarized results from the various types of filters are illustrated in Fig.9 (low-pass filter and high-pass filter), Fig.10 (band-pass filter) and Fig.11 (band-stop filter). From the diagrams in Fig.9 it follows that with increasing the sampling frequency for low-pass filter and high-pass filter the cut-off frequency increases, i.e. the slope of amplitude frequency response decreases. For band-pass filter (Fig.10) and band-stop filter (Fig.11) when the sampling frequency increases, besides the increase in cut-off frequencies, the pass-band width of the respective filter varies and this variation is proportional to the sampling frequency.

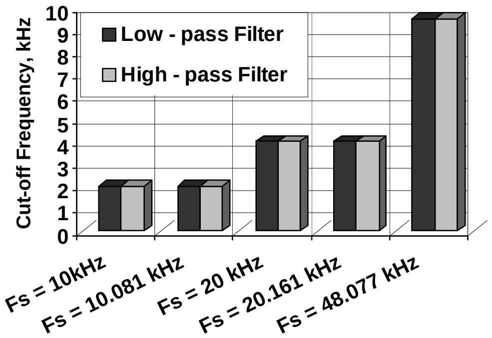

Fig. 9. Dependence of cut-off frequencies of low-pass filter and high-pass filter on sampling frequency.

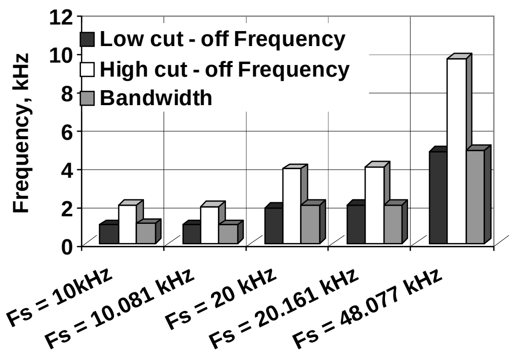

Fig. 10. Dependence of cut-off frequencies and bandwidth of band-pass filter on sampling frequency.

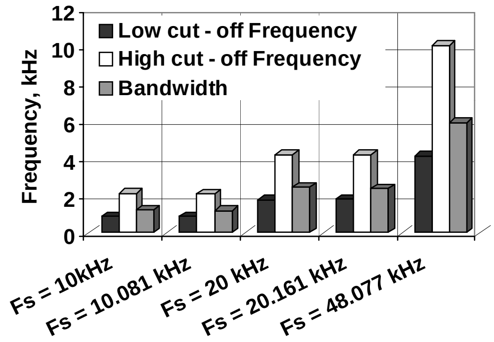

Fig. 11. Dependence of cut-off frequencies and bandwidth of band-stop filter on sampling frequency.

IV. CONCLUSION

An algorithm for the design and study of digital FIR filters by using DSP TMS320C5x and MATLAB and EXCEL software has been proposed. A program code for DSP control in designing various types of filters has been created. Frequency characteristics and parameters for low-pass filter, high-pass filter, band-pass filter and band-stop filter at various requirements to them, have been obtained by experiments and simulations. The experimental and simulated outcomes support the efficiency and correctness of the proposed approach to design and study of digital FIR filters.

REFERENCES

[1] J. Smith, “Introduction to Digital Filters”, Stanford University, California, 2003.
[2] R. Hamming, “Digital Filters”, Dover Publications, Inc, Canada, 1989.
[3] D. Schlichtharle, “Digital Filters: Basics and Design”, Springer Verlag, 2000, ISBN 3540668411.
[4] MATLAB User’s Guide, MathWorks Inc., 2002.
[5] TMS320C5x User’s Guide, Texas Instruments Inc., 1997.
[6] S. Smith, “Digital Signal Processing: A Practical Guide for Engineers and Scientists”, Publisher: Newnes, 2002, ISBN 075067444X.
[7] T. Heinrich, “DSP Starter Kit and Filter programming”, 2002.