SIMULATION AND PROGRAM REALIZATION OF RECURSIVE DIGITAL FILTERS

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In this paper, a technique for design, simulation, and DSP based program realization of recursive digital filters is proposed. The Matlab Filter Design Toolbox is used to obtain the coefficients, frequency and time domain responses with finite word length and second-order section realization of bandpass recursive digital filter. With the Real Time Workshop the previously simulated Simulink digital filter model has been automatically translated into DSP source code for the Texas Instruments DSP C5000 series. The TI Code Compose Studio is used to create a project that has been compiled by Visual Linker and has been loaded on the TMS320C5416 DSK board.

Keywords: IIR digital filter, Simulink, DSP, TI CC Studio

1. INTRODUCTION

There are numerous techniques developed for design, simulation, and implementation of digital filter based on MathWoks PC MATLAB Toolboxes [1], [2] and [3]. In resent years Digital Signal Processor (DSP) are getting to be used more and more to realize the digital signal processing system replacing the more conventional hardware elements like discrete digital components or a general purpose microprocessor based systems.

In present work, a technique for design, simulation, and DSP based program realization recursive or IIR digital filters is proposed. The main idea of this approach is reduced to the following steps: 1) Obtaining the coefficients of bandpass recursive digital filter, frequency and time domain responses, and transfer function second-order section realization with finite word length using Matlab Filter Design Toolbox; 2) Simulation and examination the behavior of Simulink digital filter model when appropriate harmonic excitation signals are applied to the digital filter input. 3) Translation the Simulink digital filter model into DSP C source code for the Texas Instruments DSP TMS320C5416 by means of Simulink Real Time Workshop (RTW). 4) Creation the project in the TI Code Compose Studio, compilation by Visual Linker and loading on the TMS320C5416 DSK board.

2. FINITE WORDLENGHT DESIGN OF RECURSIVE DIGITAL FILTERS

Filter Design Toolbox is a collection of tools of the PC MATLAB that provides advanced techniques for designing, simulating, and analyzing FIR and IIR digital filters. It offers fixed-point implementation of single-rate and multirate FIR and IIR filters, including second-order sections (SOS) with section scaling and reordering. It

integrates the filter design process in MATLAB with system-level simulation in Simulink [4].

PC MATLAB Filter Design Toolbox is used to design a recursive digital filter with 16 bits fixed-point finite wordlength. The applied quantization method is "round" and "saturate" is chosen as a method to overcome the overflow effects. The quantized filter parameters are optimized and appropriate scaling values are introduced to normalize the outputs of the filter sections. The digital filter transfer function is realized as a cascade of the second-order sections in the form:

$$H(z) = H_0 \prod_{\kappa=1}^{L} \frac{b_{0k} + b_{1k}z^{-1} + b_{2k}z^{-2}}{1 + a_{1k}z^{-1} + a_{2k}z^{-2}} , \qquad (1)$$

where H_0 is a constant, b_{0k}, b_{1k}, b_{2k}, a_{1k}, a_{2k} , k=1…L, are the coefficients of the nomination and denominator and L is a number of second order sections.

The results from Filter Design Toolbox are saved in a <>.fda file, with specific format and the digital filter fixed-point SOS structure is written in a <>.mdl file, which is an executive file for Simulink Toolbox.

3. SIMULATION OF RECURSIVE DIGITAL FILTERS

The PC MAtlab Simulink Toolbox is a block-diagram modelling environment for simulating dynamic systems, evaluating performance, and refining control, DSP, and communications system designs. Add-on products extend the Simulink environment with tools for specific modeling and design tasks and for code generation, algorithm implementation, test, and verification [4].

3.1. Simulation model of the recursive digital filters

Using PC MAtlab Simulink Toolbox the recursive digital filter simulation model is developed as shown in Fig. 1. The structure of this simulation model is based on some blocks with the following functions: "Filter" block, included in <>.mdl file; "FromWaveFile" block is the way to apply a set of harmonic excitations with different frequencies, generated in wave format, to the filter input and "ToWaveFile" block, used to save the filter responses in wave files, which can be displayed with specialized software.
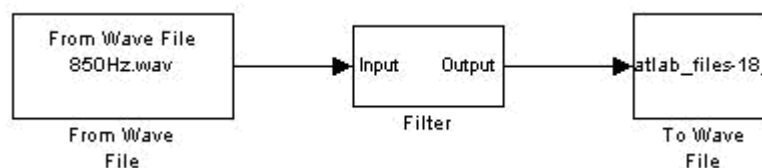

Fig.1. Block structure of the IIR digital filter simulation model

The full structure of the simulation model is built using Filter Design Toolbox as a cascade of SOS that are realized in direct form II transposed, shown in Fig. 2.

3.2. Generation of harmonic input signals

Cool Edit Pro is a digital audio editor computer program from Adobe Systems featuring both a multitrack, non-destructive mix/edit environment and a destructive-approach waveform editing view of digital audio.

The program Cool Edit Pro is used to generate the appropriate harmonic excitations that are necessary to simulate the behavior of the recursive digital filters. For the purposes of the examination of the filter simulation model are generated the following harmonic signals: input signal with frequency in passband region of the designed IIR filter; input signal with frequency in transition region; input signal with frequency in stopband region. The complete observation of the filter behavior can be obtained when a sweep of input frequencies is applied to the filter input. This sweep of input harmonic signals is generated in the frequency range from 1 Hz to fs/2 Hz.
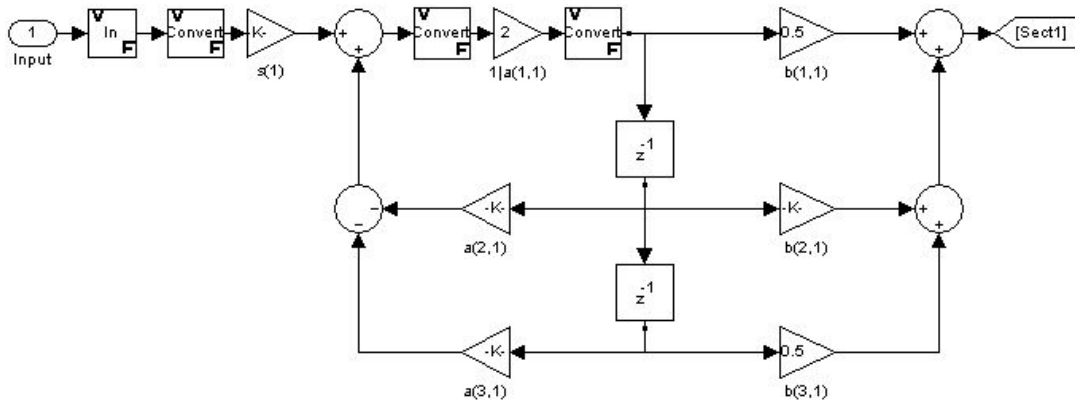

Fig. 2. Structure of the second-order sections realized in direct form II transposed

4. CODE DEVELOPMENT FROM SIMULINK MODEL

Real-Time Workshop generates and executes stand-alone C code for developing and testing algorithms modeled in Simulink. The resulting code can be used for many real-time and non-real-time applications, including simulation acceleration, rapid prototyping, and hardware-in-the-loop testing. [4].

The Real-Time Workshop of the PC MATLAB Simulink Toolbox is used to generate an ANSI C code for filter simulation model, designed for 16 bit fixed-point arithmetic and Texas Instruments DSP TMS320VC5416.

5. PROGRAM REALIZATION OF RECURSIVE DIGITAL FILTERS

5.1. TMS320C5416 DSP Starter Kit

The TMS320VC5416 fixed-point [5], digital signal processor is based on an advanced modified Harvard architecture that has one program memory bus and three data memory buses. This DSP provides a 40-Bit arithmetic logic unit with a high degree of parallelism, application-specific hardware logic, on-chip memory, and additional on-chip peripherals. Separate program and data spaces allow simultaneous access to program instructions and data, providing a high degree of parallelism. The program realization of the recursive digital filter is developed on the base of Texas Instruments TMS320C5416 DSK (DSP Starter Kit) [5] that is shown in Fig. 3.

5.2. Code Composer Studio Integrated Development Environment

Code Composer Studio (CCStudio) IDE is an integrated development tool that is a key component of the "eXpressDSP Software and Development Tools" strategy of Texas Instruments. Code Composer Studio includes the TMS320C54x code generation tools along with the APIs and plug-ins for both DSP/BIOS and RTDX [6].
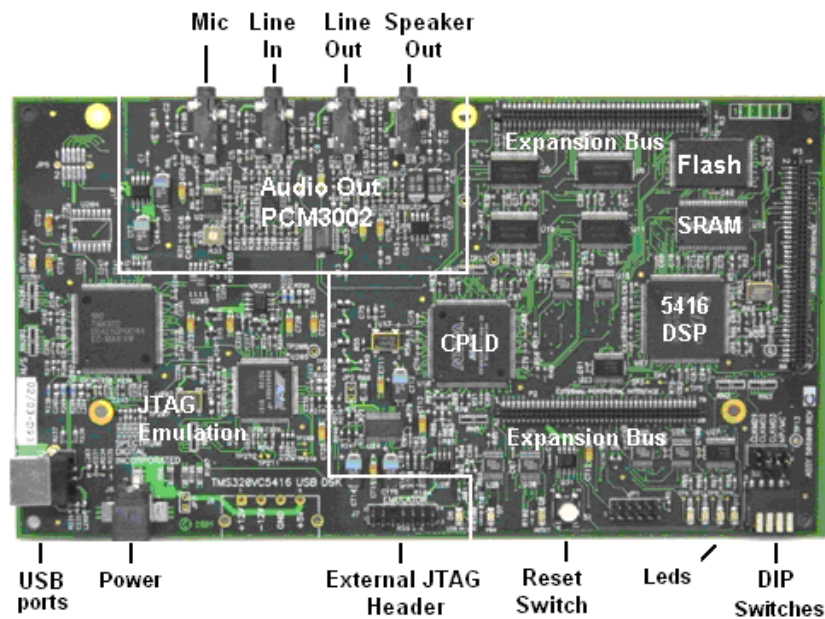
Fig. 3. TMS320C5416 DSP Starter Kit

The project created with CCStudio must include the files, generated with RTW and the following header files: rtmodel.h, rtlibsrc.h, rt_mxclassid.h, rt_logging.h, dspeph_rt.h, dsp_rt.h, dsp_iso_math_rt.h, dsp_iso_math_rt.h, oper_32b.h, basic_op.h. It is necessary to change the length of the variables in the file limits.h to 16-bit, because RTW generates this variables for 8-bit processor. The creation of new project and configuration of a Visual Linker of the CCStudio is illustrated in Fig. 4.

5.3. Algorithm for realization of IIRdigital filter based on RTW and CCStudio

The idea for program realization of IIR filters based on DSP is reduced to the following steps: generation of ANSI C code of the IIR filter structure with PC Matlab Real-Time Workshop; defining parameters and format of the code, which is included in TI CCStudio; load the generated <>.c and <>.h files in CCStudio; add the files which control the ADC and DAC for the DSK TMS 320C5416; put the folders containing the code, variables, stack and constants to the different memory regions; configure the Visual Linker; creation of <>.rcp configuration file with Visual Linker; compilation of the project and generation of <>.out file.

6. RESULTS

The proposed approach is applied to design, simulate, and realize a bandpass recursive digital filter specified by the following requirements: bandpass cutoff frequencies $F_{p1} = 650$ Hz and $F_{p2} = 1500$Hz; lower bandstop range from 0 to $F_{st1} = 450$ Hz; upper bandstop range from $F_{st2} = 2000$ Hz to $F_s/2 = 3000$ Hz; sampling frequency - $F_s = 6000$ Hz ; maximum loss in the bandpass $A_p = 2$dB and minimum loss in the bandstop frequency ranges A_{s1} $A_{s2} - 40$dB.

Using Filter Design Toolbox the 8-th order elliptic recursive digital filter with 16 bits fixed-point arithmetic has been designed and saved in file Elliptic.fda. The digital filter frequency responses are shown in Fig. 5 – amplitude response and Fig. 6 – loss

response. The digital filter responses to the input harmonic signals with frequencies in the passband 850 Hz and in the stopband 2000 Hz are respectively shown in Fig. 6 and Fig. 7. The sweep of input harmonic signals generated in frequency range (20Hz-3kHz) and the digital filter reaction are shown respectively in Fig. 8 a) and Fig. 8 b).
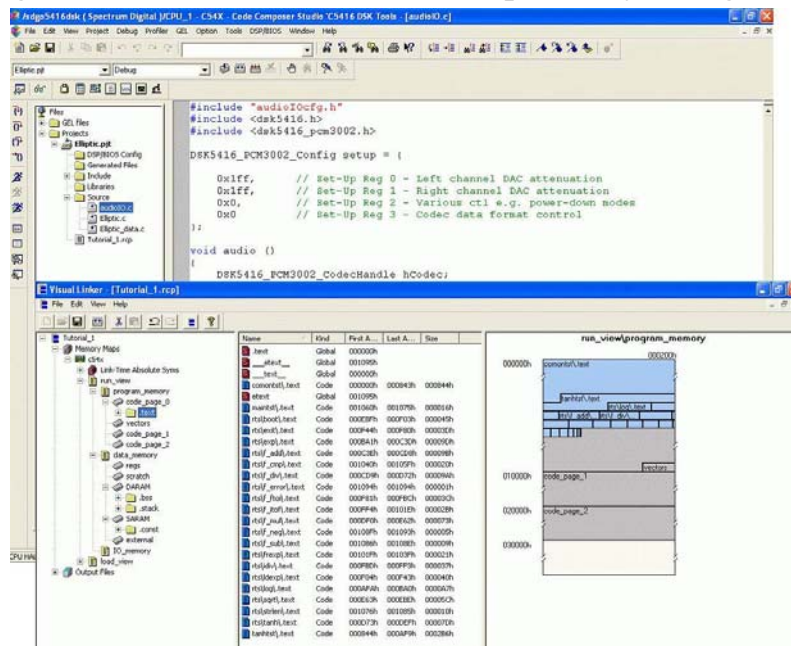

Fig. 4. Creation of new project and configuration of Visual Linker of CCStudio

The following files have been generated with Real-Time Workshop: Elliptic.c, Elliptic.h, Elliptic_data.c, Elliptic_private.h, Elliptic_types.h, limits.h, tmwtypes.h, simstruc_types.h,. It is necessary to add additional files: audioIO.c, audioIOcfg.h, audioIOcfg_c.c, dsk5416.h, dsk5416_pcm3002.h, to the created in CCStudio project EllipCCS.pjt, which control the ADC and DAC. The Visual Linker of CCS is used to compile the project for the specified target board TMS320C5416. To organize the memory map is necessary to put the folders containing the code, variables, stack and constants to the specialized memory regions respectively: program memory - code_page_0, data memory - DRAM, data memory- DRAM, data memory -> SRAM.
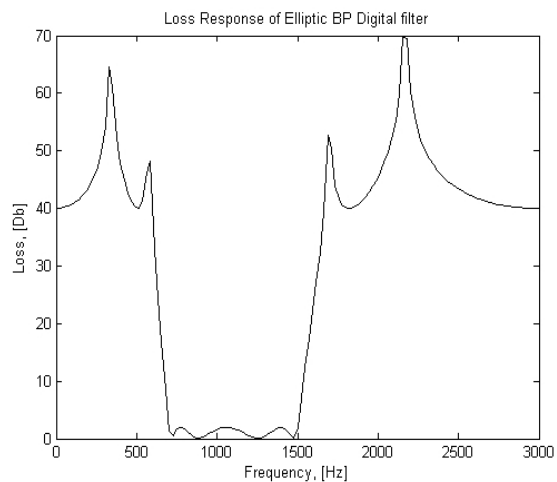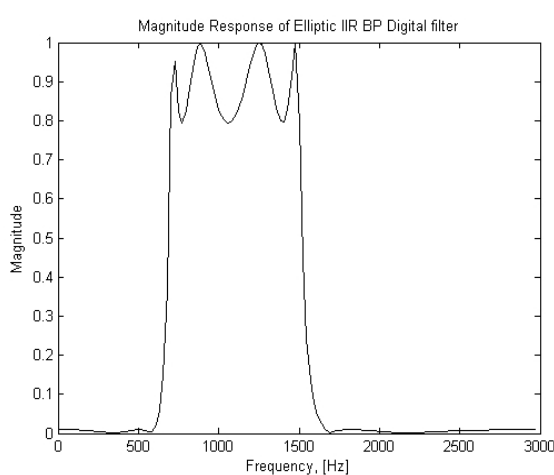

Fig. 5. Amplitude response of the recursive digital filter   Fig.6. Loss response of the recursive digital filter
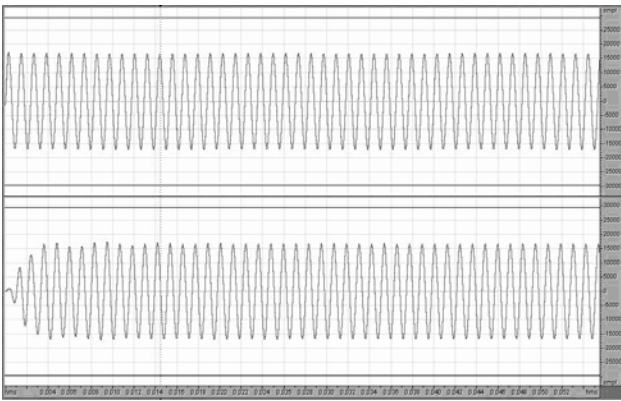
Fig. 6. Input signal with frequency 850Hz
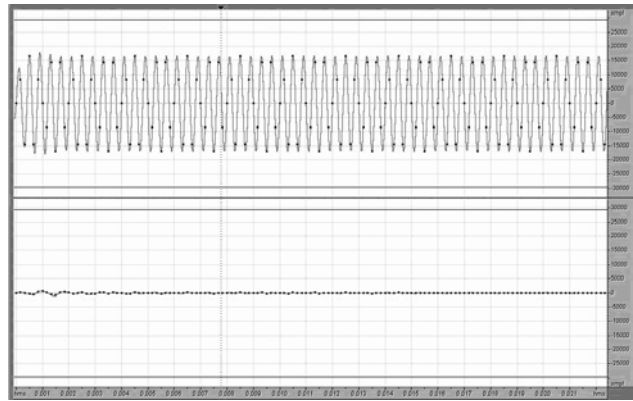and recursive digital filter response

Fig. 7. Input signal with frequency 2500Hz
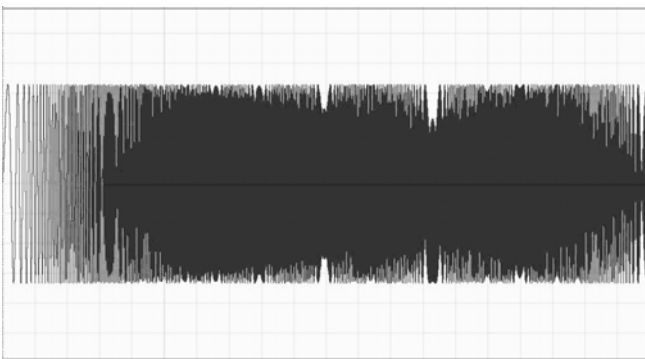and recursive digital filter response

Fig. 8a). Input sweep signal in the frequency range
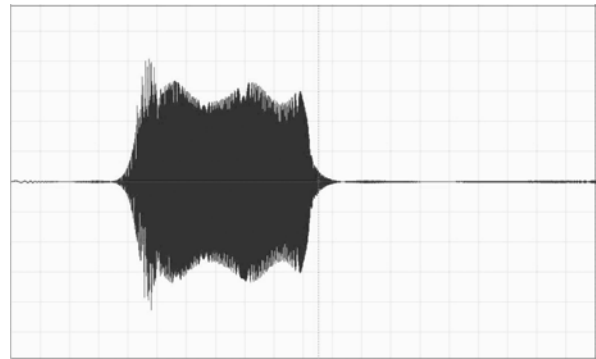[20Hz - 3000Hz]

Fig. 8b). IIR digital filter response of the sweep
input signal

7. CONCLUSIONS

In this paper, a technique for design, simulation, and DSP based program realization of recursive digital filters is proposed. Using the Matlab Filter Design Toolbox, Simulink Toolbox, and Real Time Workshop have been obtained the coefficients, frequency and time domain responses with finite word length, SOS transfer function realization, has been simulated the Simulink digital filter model and has been generated DSP source code for the Texas Instruments DSP TMS320C5416. Using Texas Instruments Code Compose Studio the project has been create, compiled by Visual Linker, and has been loaded on the TMS320C5416 DSK board. The effectiveness of the proposed approach has been demonstrated with designing of a bandpass recursive digital filter.

8. REFERENCES

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