Teaching Digital Signal Processing with MatLab and DSP Kits

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Abstract — *With the arising of new digital technologies in the global market of telecommunications, multimedia, medical equipment, automatic control, and others, the study of Digital Signal Processing has become a most important subject in electrical engineering courses. A methodology, based on progressive steps, has been developed, so that the students be prepared to design and implement typical industrial projects, such as digital filters, voice processing algorithms, and others, and also be able to correlate this knowledge with other disciplines. They start with an analog system, described by a differential equation, from which a generic discrete equation is generated. The systems that are studied for implementation on the DSP Kit, are always based on a fundamental equation of differences, representing a digital filter: this is very important for the study of digital processing concepts, such as system stability, system order, computational complexity, and so on. The projects are initially simulated with software tools , and they are then implemented with digital signal processing kits . The simulation helps the students to understand a system digitalization process. The results obtained with the students in the course show the efficiency of this methodology. A large part of the success must be credited to DSP tools, which stimulate the students, and enable them to implement digital systems they are interested in.*

Index Terms — *DSP, filters, MatLab, signal generators.*

INTRODUCTION

With the arising of new emergent digital technologies in the worldwide market of telecommunications, multimedia, medicine, control, etc, the study of digital signal processing has become an important part in electrical engineering. Students have shown great interest in being able to carry through and to implement, in this area, real projects and not only didactic ones. They are apprehensive to put up with the difficulties that this demands, and do not have yet the capacity to properly imagine a complete scene, including modeling, analysis and project synthesis. Based on this it was decided to convey to their interest and to adopt a methodology, with practical aspects, for the project and implementation of systems based on DSP's.

This work presents a teaching methodology that is based on the study of cases, from modeling to synthesis, using the MatLab simulation tool and Texas Instruments DSP Kits, with TMS320C31 and TMS320C54 devices. The TEXAS floating-point Start Kit DSP was used due to the easiness offered for programming, in relation to fixed point DSPs, since numerical data can be represented the with the necessary precision in its original form, without having to apply any scaling to the fractional part.

METHODOLOGY

The main objective is to help the students to learn DSP's and carry out a project in stages, through gradual steps, in subjects related with signal processing, which can be found usually in industry, such as digital filters, echo cancellers, audio equalizers, modems, voice processing algorithms, imaging, etc. They should also be able to relate their knowledge to others disciplines of the engineering course. This didactic procedure is being used in Electronic and Telecommunications Engineering laboratory classes, to complement theoretical studies. The project is carried through in four different stages, as seen in figure 1.

In the first stage signal processing theory is studied, with numerical examples and case study, such as filters, controllers, signal synthesizers and others, after what the professor chooses a project to be implemented by the students. Initially there is a theoretical exposition on continuous systems, where the student already presents reasonable knowledge, and then a case study is made, where the discretization of a continuous system is explained. In the Telecommunications Engineering course the study is based on a FIR (finite impulsive response) digital filter, very common in the telecommunications industry, which is based on the Fourier series, and impulse invariance methods. These methods are implemented by the students and later simulated with the MatLab tool.

The subjects studied by the students are based on the classic equation of differences, of the type:

*a(1)y(n) = c(1)*x(n) + c(2)*x(n-1) + ... +c(nb+1)*x(n-nb)   -a(2)*y(n-1) - ... - a(na+1)*y(n-na)*

for which it is necessary to determine *c* and the coefficient values of the equation. The basic equation of differences corresponds to the discrete generic equation that always represents a digital filter that is obtained from a differential equation of the system under study. The equation of differences that represents a digital filter is very useful for the study of the concepts of digital signal processing, such as system stability, system feasibility, system order, computational complexity and others [1,2].

In the following phase the simulation of the project is carried through, with the MatLab software [3], that performs the calculations and simulation, based on the information of the theoretical study made in the previous phase. The students feed the MatLab, through a program, with the equations to be simulated, and interact with the tool until the simulations get to the expected results.

In the third stage, after the project has run correctly in the Simulator, the students study the architecture and applications of a Digital Signal Processor [4], to be able to make the project implementation. It is studied the didactic Texas Start Kit DSP TMS320C31 [5], that is composed of a card with the DSP, the assembler DSKA, and the debugger and recording code program DSK3D. The students learn to elaborate a program in machine language and how to run it in the kit. Modifications had to be made to the kit, adding an auxiliary card (figure2), to limit input voltage, so as to prevent damages to the DSP kit and easily be used in a classroom.

In the fourth phase the implementation of the project, which had already been simulated in the MatLab, is finally carried through.

CASE STUDY I: FILTER PROJECT

Four types of filters are presented, *Lowpass, Highpass, Bandpass and Bandstop*, as shown in figure 3, for implementation and choice, with frequencies up to 10khz .The Fourier series and impulse invariance are the methods usually adopted, since they are very common in signal processing.

The filter pass band, as well as the cut off frequency, is defined to the students. The example is interesting, since the advantage in using DSP's stands out: in low frequencies, as this is the case, a filter with a 4 Hz cut off frequency, when implemented with discrete analog components, would be comparatively be too large, due to the size of the components. On the contrary, when using DSP, the device is much smaller. A complete system is shown in figure 4.

Use of Fourier series for FIR filters

Here is an example of the Fourier series [6,7] given to the students: symmetrical lowpass filter, 4Hz cut off frequency, broadband of 4Hz, 3dB attenuation in the cut off frequency, with 11 coefficients. The FIR Filter (figure 5) project using the Fourier series method is such that its transfer function H (z) is close to the filter frequency response requested. The digitalized transfer function is:

$$\mathrm{Hd}(w) = \sum_{n=-\infty}^{\infty} C_n e^{jnwt} \qquad |n| < \infty$$

Where Cn is the Fourier series coefficient .

(a) Lowpass; (b) Highpass, (c) Bandpass; (d) Bandstop

Each filter is calculated through the formula:

(a) Lowpass : $C_0=v_1$

$$C_n = \int_0^{v_1} H_d(v)\cdot\cos(n\pi v)\,dv = \frac{\sin(n\pi v_1)}{n\pi}$$

(b) Highpass : C_0=1-v_1

$$C_n = \sum_{v_1} H_d(v)\cos(n\pi v)\cdot dv = -\frac{\sin(n\pi v_1)}{n\pi}$$

(c) Bandpass $C_0 = v_2 - v_1$

$$C_n = \int_{v_1}^{v_2} H_d(v)\cdot\cos(n\pi v)\,dv = \frac{\sin(n\pi v_2) - \sin(n\pi v_1)}{n\pi}$$

(d) Bandstop C_0=1-(v_2-v_1)

$$C_n = \int_0^{v_1} H_d(v)\cdot\cos(n\pi v)\,dv + \int_{v_2}^{1} H_d(v)\cdot\cos(n\pi v)\,dv = \frac{\sin(n\pi v_1) - \sin(n\pi v_2)}{n\pi}$$

Where v1 and v2 are the normalized cut off frequency, and Hd is the digitalized transfer function.

Use of Fourier series for the implementation of FIR Filters

The first approach to the project is through a theoretical study and successive verifications with the MatLab software that is used in the implementation of these equations, allowing that the students concentrate mainly on the project. The data obtained in the MATLAB, are analyzed and written down for future comparison with the implementation in the DSP. The students generate input signals and simulate filter interaction with MatLab. In figure 6 a 1Hz, 1Vpp, signal, generated in the MatLab, applied at the filter input, can be seen. This signal does not suffer significant attenuation, because it is within the limits of the established 4Hz filter passband. On the other hand, as it shown figure 7, with a frequency signal higher than 4Hz, the output signal is attenuated in 3dB, showing that the filter is correct.

In the following project phase, having already been calculated the values of the coefficients, and the simulated filter, it is generated the code for the filter implementation in the kit. A partially source program, which contains the DSP initialization, A/D and D/A converters, serial communication channel, is supplied to the students, at the beginning of the course, that must be completed to incorporate the necessary code for the Digital Filter implementation. The program is then debugged, before it may be loaded and executed in the DSP. Figure 8 shows the contents of the DSP memory, recorders, and line code to be executed, all the processor status. With the program debugged and loaded onto the DSP, the students verify, with a signal generator applied at the DSP TMS320C31 analogical input, the wave forms, which are then compared to those obtained with in the MatLab for simulation. These signals, shown in figures 9 and 10, were obtained with a 2-channel oscilloscope. The amplitude corresponds to the 4 Hz cut off frequency, with 3dB attenuation, as simulated. The project is then concluded by the students, who compare the waveforms obtained in the MatLab, with the corresponding ones obtained in the implemented in the DSP. Finally the students must elaborate a report, which will be part of the evaluation criterion.

CASE STUDY II: IMPLEMENTATION OF A SIGNAL GENERATOR FUNCTION

Another experiment is the sinthetization of signals, using MatLab and DSP Kits, that can be used in the generation of sounds for a digital piano. This application has shown to be very attractive to the students, who generally spend many hours working on the project, due to the expectation of the results. The synthesizer block diagram [8] used to generate sounds is shown in figure 11. The signals can be generated with frequencies up to 10KHz.

The block Z-1 represents a delay cycle, and A1, where A1= 2*cos (2*pi*F/Fs), can be altered to change signal amplitude and frequency. F is the frequency of the signal to be generated, and Fs is the sampling frequency, which is chosen to be 10 times the frequency to be generated, so, for F=1000Hz, then Fs=10000Hz. The process is similar, as to simulation and implementation in kit DSP, to the previous ones.

In the course of Electronic Engineering the project adopted is a PID controller, and the procedures used are similar to the other projects. The basic differential equation is obtained from the system to be studied. Initially the students raise the empirical model of a servomechanism with experimental data. Matlab is used for modeling, and the controller is designed and discretized. The student, after having followed all the procedures, work on the program to be used in the positioning control servomechanism. The students are also stimulated to develop new algorithms, besides working on the recommended ones. With this a significant change in the students' attitude towards studies has been perceived, for they concentrate on the problem more easily, without being distracted with other operational aspects, which are solved with the use of proper tools.

CONCLUSION

The first evaluations point out that the students have shown much more interest, freedom and autonomy in developing systems that can be implemented with DSP's. As a direct result the number of graduation projects that use these resources and techniques has increased. The students gradually overcome the difficulties at each stage, and this contributes to increase their the self-confidence. The results have shown to be efficient as to the learning, and the students reported that they had more facility in understanding related disciplines, in telecommunications, microprocessors and control. A part of the success must be credited to the feasibility in the accomplishment of interesting and complex projects that would otherwise be impracticable with conventional electronics available in a school lab: this also increased their self-esteem and sense of professional accomplishment.

Recently the students were invited, for their great satisfaction, to participate as expositors in Texas Instruments stand, at Telexpo 2002 in São Paulo, which is the largest telecommunications event carried through in Brazil, where they could present their own projects to an international public.

REFERENCES

[1] Sorensen, H. V., and Chen., Jianping, *"Digital Signal Processing Laboratory Using the TMS320C30"*, Prentice Hall, 1997

[2] Chassaing, R.,"Laboratory Experiments Using C and the TMS320C31 DSK", *Digital Signal Processing;* John Wiley & Sons, Inc. 1999.

[3] Hwang, K., *"Advanced Computer Architecture: Parallelism, Scalability, Programmability"*, McGraw-Hill, Inc. 1993

[4] Oppenheim, A. V. ; Schafer, R.,*"Discrete-Time Signal Processing"*, Prentice-Hall, Englewood Cliffs, NJ, 1989.

[5] Antoniou, A., "*Digital Filters: Analysis, Design, and Applications",* McGraw-Hill, New York, 1993.

[6] Mitra, S. K.,*"Digital Signal Processing A Computer-Based Approach"*, McGraw-Hill, New York, 1998.

[7] MatLab, The Math Works Inc., MA, 1997.

[8] Texas Instruments Inc.,*"TMS320C3x General-Purpose Applications User´s Guide"*, Dallas, TX, 1998.

FIGURES AND TABLES

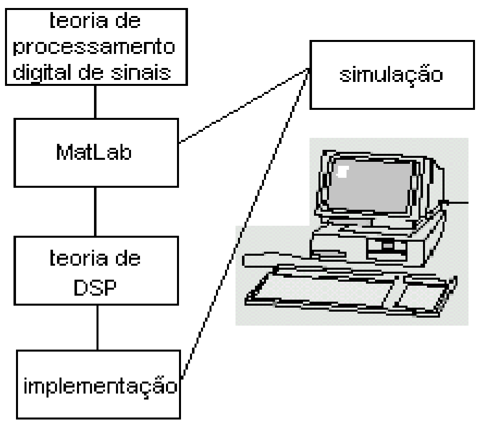

FIGURE.1 PROCEDURES USED WITH MATLAB E DSP.

FIGURE. 2 DSP TMS320C31 MODIFIED CARD.

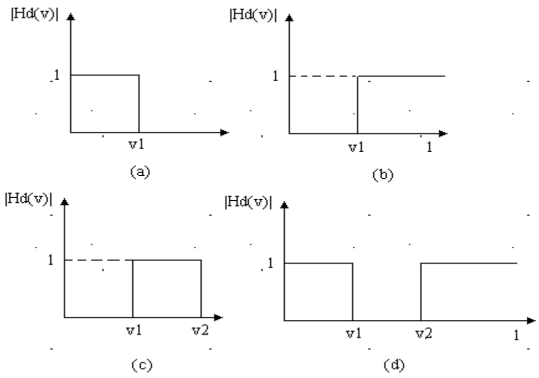

FIGURE. 3 FILTER TYPES :(A)LOWPASS; (B) HIGHPASS; (C) BANDPASS;
(d) BANDSTOP

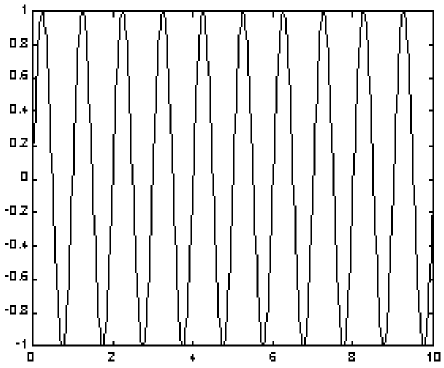

FIGURE. 6 INPUT SIGNAL SIMULATED WITH MATLAB, 1HZ, 1VPP AMPLITUDE .

FIGURE. 4 BASIC SYSTEM WITH DSP TMS320C31, MICROCOMPUTADOR, GERADOR DE SINAIS E OSCILOSCÓPIO

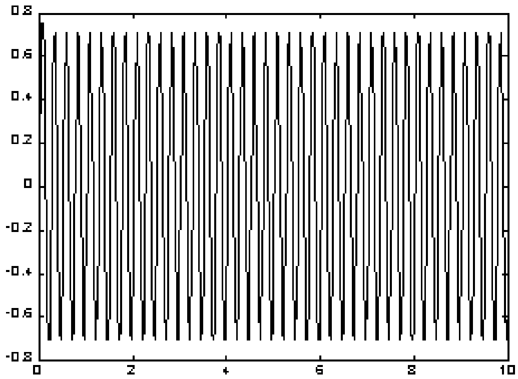

FIGURE 7 OUTPUT SIGNAL SIMULATED WITH MATLAB, 4HZ CUT OFF FREQUENCY, 3DB AMPLITUDE ATTENUATION .

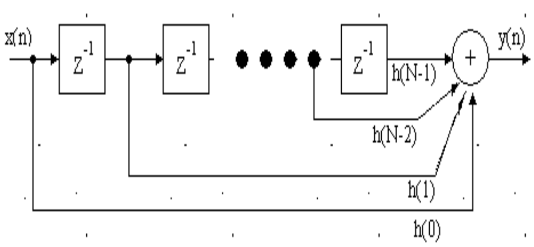

FIGURE. 5 FIR FILTER STRUCTURE.

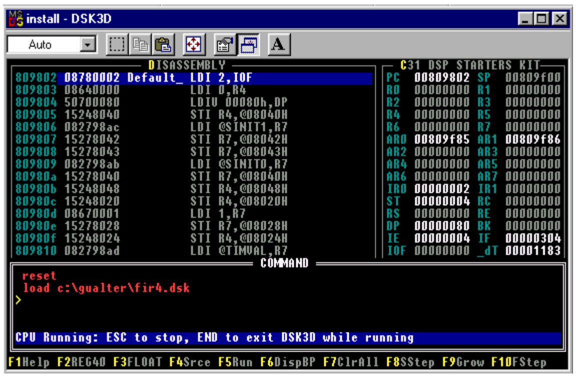

FIGURE. 8 DEBUGGER DSK3D.

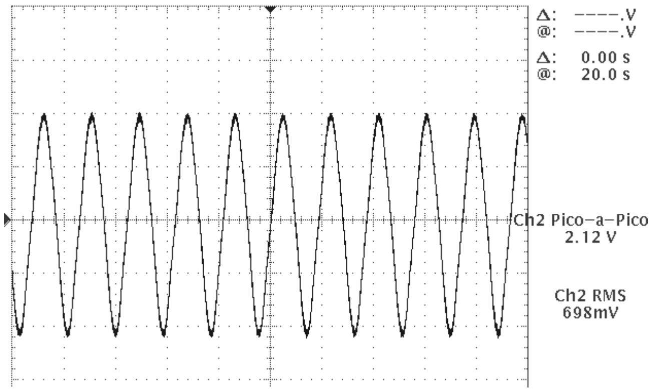

FIGURE .9 OUTPUT SIGNAL ON DSP

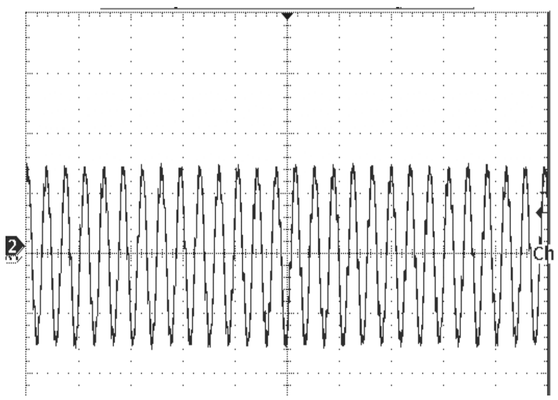

FIGURE 10 OUTPUT SIGNAL SHOWN BY MATLAB.

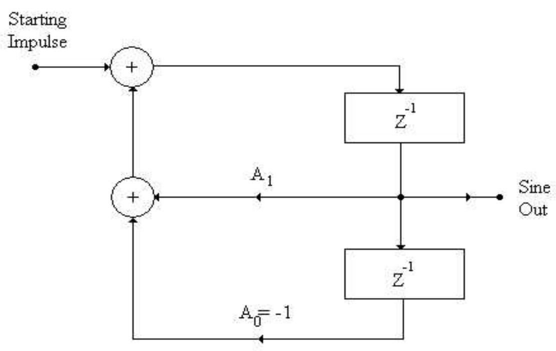

FIGURE 11 SYNTHETIZER BLOCK DIAGRAM