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THE DSP EXPERIMETS FOR UNDER GRADUATE STUDENTS

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ABSTRACT

This paper describes digital signal processing (DSP) microprocessor experiments designed for university juniors majoring in electric & electronic (E & E) engineering. At the time of enrollment, most students have only studied the curiculum for analog signal processing which is taken in the prior semester. The proposed DSP microprocessor experiments are included along with analog signal processing. This early-on introduction to DSP technology allows students to realize that, in comparison with analog circuits, DSP microprocessors can process the same signals in real-time with broader flexibility. Such an understanding is considered important to instill strong incentive for students to become interested in the field of DSP.

1. INTRODUCTION

Digital signal processing (DSP) microprocessors possess the capability to allow signal processing in real time, being a unique characteristic that has led to an ever increasing number of DSP applications in many electronic-based industries involved with acoustics, communications, and control systems, for example. In Japan, most engineers participating in the DSP field acquire the necessary skills either after they graduate or during graduate studies. Although a number of studies have been directed at topics related to DSP education [1] [2] [3], and a variety of educational techniques do exist using several tools/media[4] [5] [6] [7] , only limited attention has been focused on developing DSP educational programs targeted for undergraduate students studying electrical/electronic engineering. Accordingly, in one of our experiment-based courses for juniors, a DSP microprocessor experiment is included along with analog signal processing experiments. Students participating in the course have only studied the curriculum for analog signal processing at this point. However, an introduction to DSP technology allows them to realize early on that, in comparison with analog circuits, DSP microprocessors can process the same signals in real time with broader flexibility. Such an understanding is considered important to instill strong incentive for students to become interested in the DSP field.

The experiment is conducted over two weeks, i.e., two 3-h experiment classes each week. Other subjects experimentally studied during the semester include amplifiers, oscillators, active filters using OP-amps, amplitude and frequency modulation, servo mechanisms, circuit design with CAD, microprocessors, and DSP. Such a program allows students to easily evaluate analog versus digital filters. Each topic within the DSP experiment is written in assembly language. To allow students to grasp the material within a limited time, detailed comments are included within the main part of each program, and they must modify the data sections of the programs in conformity with each topic.

2. EXPERIMENT CONTENTS

A Texas Instruments TMS320C3x DSP Starter Kit (DSK) and IBM PC/AT compatible personal computer (PC) is utilized as experiment equipment. The DSP experiment includes six topics discussed next.

2.1. I/O analog signals

In this topic, analog signals are input into an A/D converter without processing output signals using a DSP microprocessor; a fundamental experiment, although changing the input signal frequency shows the performance of the A/D or D/A converter and aliasing in the frequency domain. Students can accordingly grasp sampling theory which is a basic concept of digital signals. They are shown that placing signal processing programs between input and output signals enables various processing capability, while simultaneously providing an understanding of the effects of the programs and how the timing of hardware interrupts A/D conversion.

2.2. DSP microprocessor as signal generator

This topic involves computing waveforms inside a DSP microprocessor, and outputting the waveform through a D/A converter. Initially, value “A” is set into the DSP CPU register, and a smaller value is subtracted from the register value

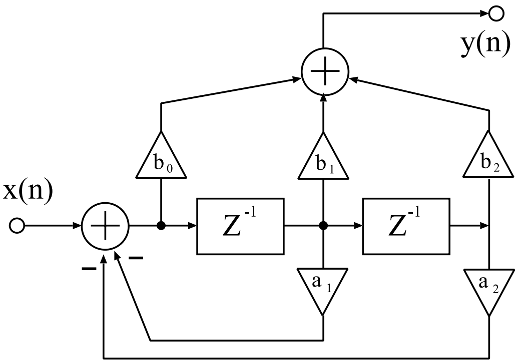

Figure 1. A second-order IIR subsystem

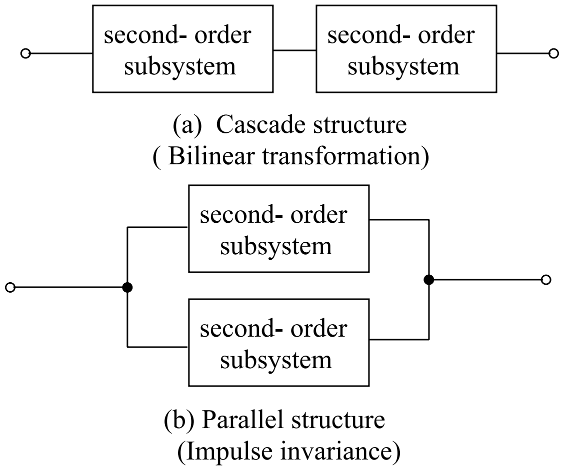

Figure 2. Cascade and parallel structure for forth-order system.

every time frame. When the register value is negative, “A” is set again. This computation is output to D/A converter and the output waveform is a sawtooth wave. By modifying the program, a triangular wave with a center value of 0 is easily computed. Then, by applying a sinusoid development formula to instantaneous values of the triangular wave, instantaneous values of a sinusoid can be computed and output to the D/A converter such that a sinusoid wave is synthesized. This topic is associated with many applications involving waveform generation performed by synthesizers for example.

2.3. Finite Impulse Response(FIR) digital filters

This topic involves designing FIR digital filters. Namely, coefficients of FIR digital filters are used to demonstrate their impulse responses; hence making it easy to comprehend FIR filters with regard to the time domain. Although FIR filters can be designed in a number of ways, the simplest method is called the window method [8] . Initially, rectangular windows are used, i.e., we adopt 8th- and 31st-order rectangular windows in which students are able to understand characteristics of a rectangular window, cutoff characteristics, cutoff frequency based on window length, size of side lobe to main lobe, and other features. The simple theory involved here allows experimental results to be theoretically investigated such that students gain an understanding of time and frequency Fourier transform pair. Next another FIR digital filter is designed using a 31st-order Hamming window, and its characteristics are compared to those of the FIR filter designed using a rectangular window. In addition, the linear-phase characteristic of an FIR filter is observed. Overall, students come away with a feeling that they can design FIR digital filters with a variety of frequency characteristics by adjusting the limited length coefficients of FIR filters.

2.4. Infinite Impulse Response (IIR) digital filters

The design IIR digital filters is carried out in this topic, as they can be designed based on analog filter design theory. Using digital filters to realize the same characteristics as analog filters allows students to grasp the usefulness of digital filters. During a previously performed experiment, students learn about analog active filters using OP-amps; thus here the same characteristics are given to an IIR digital filter. In other words, by observing frequency characteristics of both filters, students are able to practically confirm that digital filters can realize characteristics of analog filters. Specifically studied is the basic construction element of a 2nd-order IIR digital filter expressed as

$$H(z) = \frac{b_0 + b_1 z^{-1} + b_2 z^{-2}}{1 + a_1 z^{-1} + a_2 z^{-2}}. \tag{1}$$

The characteristics of the analog filter are similarly expressed. Figure 1 shows a block diagram of this 2nd-order IIR digital filter subsequently used to design a 4th-order Butterworth low-pass filter with a 1.25-kHz cutoff frequency. Students must first calculate the transfer function of the equivalent analog filter, then design the IIR digital filter based on its coefficients. Bilinear transformation and impulse invariance design [8] are adopted in this design, and two 2nd-order IIR

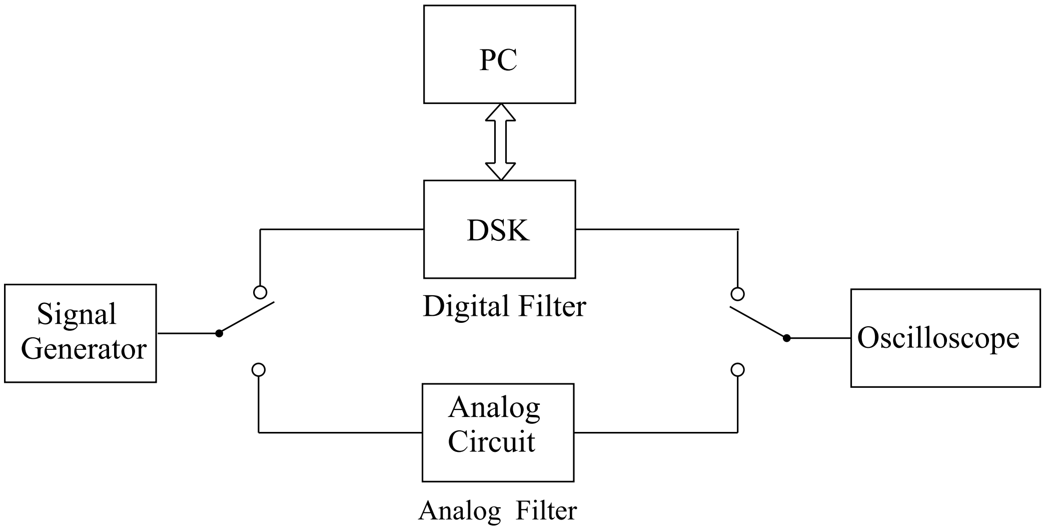

Figure 3. Measurements for frequency characteristics of digital filter and analog filter.

filters are connected in series or parallel as well (Fig. 2).

The series connection is for bilinear transformation and the parallel connection for impulse invariance design. Programs are given with missing coefficients such that students must calculate them so as to construct both filters by completing the program. Next, impulse and frequency responses of both filters are observed by inputting short-duration pulses. Such frequency responses can not only be theoretically investigated, but the frequency response of the equivalent analog active filter can also be observed in the analog filter experiment (Fig. 3). Overall, students complete the experiment with a solid understanding of how digital filters can be constructed using coefficient values designed for analog filters.

2.5. Delaying signals

Synthesis of a reflected waveform is considered here using a simple example to demonstrate that DSP microprocessors can be used to alter input signals. Basically, input signals are delayed for a given period of time, multiplied by a reflecting coefficient, and then output. Students can vary the delay time and reflecting coefficient. A short-duration pulse is first input to confirm the action of the program, after which students input their own voice using a microphone and listen to the output on a loudspeaker. The processed voice is commonly called an "echoed voice," whose characteristics are distinctly heard such that students can actually feel the effect of signal processing and witness the wide range of possibilities involving applications incorporating DSP microprocessors.

2.6. Fast Fourier Transform (FFT)

The objective of this topic is to observe an amplitude spectrum. That is, input signals are subjected to short-duration FFT performed by a DSP microprocessor, with the resultant FFT-produced short-duration amplitude spectrum being input into a PC. The continuous display is similar to that seen on the display monitor of a spectrum analyzer. As a DSK attachment demo program is used, this prevents modifying the program. Students can, however, input various sinusoid frequencies, their voice, and other signals and observe the respective amplitude spectrum. In this manner, students gain a strong appreciation regarding the usefulness of applying a DSP microprocessor for effective digital signal processing.

3. RESULT

Integration of DSP training into a junior curriculum containing no prior theoretical coursework makes it impractical to assume that students can thoroughly understand this complex subject. That is why the introductory experiments commence with basic concepts. I/O analog signals (topic 1) simply involves only an analog signal input and output program. It does, however, require some time for students to familiarize themselves with the equipment, though many students can easy handle a PC and quickly learn to operate the DSK assembler and debugger. Providing detailed comments in the program allows them to understand the fundamental operation of the DSP microprocessor. And, by observing frequency characteristics, they can grasp the concepts of sampling frequency, aliasing, and operation of an internal analog filter. Use of a DSP microprocessor as a signal generator (topic 2) involves inserting a counter program using a register in the topic 1 program, i.e., the slope of a sawtooth waveform is shown to be dependent on a subtraction value. This task enables students to confirm the effect of changing the value. In addition, a subroutine is used to compute a sine function for sinusoidal wave generation, which gives a feel for the flow of the calculating process. The ability gained thus far by operating the DSP microprocessor is useful for studying about an FIR digital filter (topic 3), which assists students in understanding the relationship between time domain and frequency domain. Similarly, an IIR digital filter (topic 4) is derived from an analog filter design. The filter is a 4th-order Butterworth low-pass filter used as in an analog active-filter experiment during the same semester. Two IIR filter design methods are introduced: impulse invariance design and bilinear transformation. In this design, students are given the formula for converting coefficients from analog to digital, which is a fairly arduous task for most. By observing characteristics of the filters, good approximation between analog and digital filters is easy to confirm. In the following signal delay experiment (topic 5), students could by this time understand the program and therefore expect the process of the program. By inputting a human voice through a microphone, the speaker output provided a real feeling of the effect of real-time signal processing; hence raising student interest in field of DSP. Finally, the FFT experiment (topic 6) allows students to verify that a DSP microprocessor is able to process a very complex task in real-time.

4. CONCLUSION

Most university juniors are familiar with a wide variety of computer software, yet have almost no knowledge about DSP microprocessors. We accordingly introduced a DSP experiment course at this level to give the students a familiarity to DSP. The first experiment topic involves use of a simple program, written in assembly language with detailed comments, that aids in gradually making students aware of the relation between program instructions and the actions produced by actual hardware. Use of a DSK debugger that can run programs step-by-step enables them to understand the functions of DSP. Study of two types of DSP filters subsequently allows comparing the characteristics of analog and digital filters such that the flexibility and effectiveness of digital filters can be grasped. Enjoyable experiments using sounds, on the other hand, provide a good opportunity for creating interest in real-time digital signal processing and motivation for further study as well. Before the course begins, students are asked in a survey if they know anything about a DSP processor, and most answer no. Upon completion of the six topics studied, however, 88% of students answer that they feel they understand a DSP processor, and that the experiments were a key aspect in gaining this understanding, which is an important step in generating future interest in the ever-growing, highly complex, field of DSP.

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