Teaching Speech and Audio Processing Implementations Using LabView Program and DAQ Boards

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Abstract

We present our new pedagogy for teaching speech and audio processing implementations using LabView program and DAQ (Data Acquisition) board. In the Electrical and Computer Engineering Technology (ECET) curriculum, the LabView program and DAQ board data acquisition have been used as a popular platform for teaching a real-time DSP (digital signal processing) course in the junior year. This course is the second signal processing course which is offered in electrical and computer engineering technology (ECET) program according to the current DSP industry trend and student interests in their career development. The course has a significant component on real-time filtering applications. The pre-requisite includes student working knowledge and skills of Laplace transform, Fourier series, Fourier transform, and different types of analog active filter design. After completing the course, students not only become familiar with MATLAB and LabView software development tools, but also gain the real-time signal processing experience. They are able to design the low-pass and band-pass filters and then program them using MATLAB and LabView software, and apply the software and hardware interface for real speech applications. Comparing with the traditional DSP course which mainly focus on heavy mathematical development in sampling and recovering, spectrum analysis, FIR or IIR filter design with the limited computer simulations, our real-time speech project allow students easily to understand how his/her own voice could be enhanced (by an audio amplifier), digitized and collected (by a PCI-1200 DAQ board), displayed(by LabView and spectrum analyzer), processed(by MATLAB and LabView software), and recovered (by an active 2nd-order lowpass filter). Therefore, in this paper, we first proposed a complete DSP platform, which consists of an LM386 audio amplifier, an analog lowpass filter, a PCI-1200 DAQ board interface circuit, and a DSP software to perform the real-time data processing. Using the proposed DSP platform, we present our instructional techniques for digital filter implementations in which an FIR lowpass filter and an IIR bandpass filter were designed and tested for real time processing. The student survey results show that the adoption of LabView and DAQ boards to teach DSP and digital filter design is a learning effective tool.

I. Introduction

Digital signal processing (DSP) technology and its advancements have continuously impacted the disciplines of electrical, computer, and biomedical engineering technology programs. This is due to the fact that DSP technology plays a key role in many current applications of electronics, which include digital telephones, cellular phones, digital satellites, digital TV’s, ECG analyzers, digital X-rays, and medical image systems in the areas of communications, instrumentation, and biomedical signal processing. There are many DSP related products such as digital voice recorders, CD/DVD players, MP3 players, digital cameras, internet audios, and images and videos. A quick review of the current jobs advertised in technical magazines and on the internet web sites further reveals a demand for individuals with a refined DSP knowledge. Hence, the

qualified engineering technologists capable of operating, maintaining, repairing, evaluating, and helping to specify and design DSP-based products have significant competency for their employment.
To prepare engineering technology students for such an industrial trend, many undergraduate programs in engineering technology not only offer a course to cover the fundamentals of signal processing, but also provide a second DSP course in which real-time applications and corresponding advanced topics such as speech signal processing, adaptive filtering, and digital image[1,2,3,5] are introduced.

In our engineering technology program, the second signal processing course is designed for junior students starting to experience the real-time signal processing applications and get interested in DSP real applications, such as the innovative real-time speech processing project covered in this paper. Our complete real speech processing system including design and testing greatly motivates students and allows students to understand how his/her own voice could be enhanced (by an audio amplifier), digitized and collected (by a PCI-1200 DAQ board), displayed(by LabView and spectrum analyzer), processed(by MATLAB and LabView software), and recovered (by an active 2nd-order lowpass filter in the last stage), while the traditional DSP teaching only focus on heavy mathematical development in signal sampling and recovering, spectrum analysis, FIR or IIR filter design with the limited computer simulations. The course prerequisite assumes that the students have already acquired working skills of the Laplace transform, the Fourier analysis, and analog filters from the first signal processing course. The technology students in this course will continue to explore advanced techniques such as real-time digital filter implementations and adaptive filtering. While offering a broad coverage of topics and the real-time DSP system application, this course could be beneficial to all electrical technology students.

Since teaching advanced DSP topics within the engineering technology program has the requirement of being at a hands-on and engineering technology level, adopting the traditional teaching approaches and using textbooks dealing with complicated mathematics and theories used in the four-year engineering program may not be appropriate. Hence, in this paper, we will present our innovative pedagogies and experiences from teaching the subjects of advanced DSP in the engineering technology curricula.

The paper is organized as follows. We will explain the course prerequisites and describe our class content first, and then we will introduce real-time signal processing hands-on project using a DAQ (Data Acquisition) board and simulation tools such as MATLAB and MultiSIM. We will also present the course assessment and outcome, which include how the students apply their gained DSP knowledge to their capstone senior projects. Finally, we will address possible improvement of the course content and associated laboratories.

II. Course Prerequisite Requirements

In this section, we will explain the course pre-requisites, which can be divided into three categories, as described below.

A. Digital Signal Processing Course Requirement

The first signal processing course covering the key topics of analog signal processing, such as common analog functions, Laplace transform, Fourier series, Fourier transform, and different types of analog active filter design. Students in the DSP course apply these established skills for designing, implementing, and verifying various applications such as the digital crossover audio systems, speech signal processing, and so on. Specifically, the skills of the digital FIR filter design and signal spectral analysis are necessary for low-pass and band-pass filter designs and verifications in the area of speech processing. Furthermore, a grasp of concepts and principles of the first signal processing course indicates the gained knowledge of the analog signal processing (ASP) course, in which three major topics, the Laplace transform, Fourier analysis, and analog filters, are covered. For example, the Laplace transform and Laplace transfer function serve for the analog filter design and digital IIR filter design, while the Fourier analysis supports for spectral analysis and digital FIR filter design with window functions. Hence, the prerequisite of the first signal processing course implies the ASP course.

B. Math Requirement

While satisfying the prerequisite of the signal processing course, students are gaining maturity in the comprehension and application of math including basic calculus, and proficiency in using algebra. A firm grasp of calculus concepts is also beneficial in understanding the advanced course materials such as the employment of the derivative operation or difference equations. Since the calculus course is a prerequisite for the first DSP course or the combined ASP and DSP course, it is not necessary that we list it as an additional prerequisite.

C. Software Requirement

To design, analyze, and simulate the DSP algorithms, MATLAB programming is required; this requirement was enforced in the previous signal processing course. In addition, MultiSIM will be used to verify different filter design.

As a summary, the DSP course needs the prerequisites as listed below:

1. Analog signal processing
2. MATLAB programming and MultiSIM simulation.

III. Course Content and the Associated Real-Time Project

We have divided the course content into two portions. First, the DSP fundamentals were covered, such as the sampling theorem, the z-transform and z-transfer functions, the discrete Fourier Transform, FFT algorithm, signal spectrum analysis, filter frequency responses, and filter implementations using the direct-form I and direct-form II, and so on. The second portion introduced FIR design and IIR design with an emphasis on real-time digital filter implementation and applications. We focus on the real-time DSP speech project in this paper.

The course was taught in 16 weeks with 3 lecture hours and 3 laboratory hours per week. The textbook selected was "Digital Signal Processing: Fundamentals and Applications." published by Elsevier, 2007 [1]. The textbook presents course materials at an appropriate math level, uses an ample amount of simplified and clearly worked examples, adopts MATLAB programs to

demonstrate simulations, and provides application examples to motivate students. Simplification of real-time DSP implementations to the engineering technology level is a plus.
To minimize the time for learning different simulation tools, we simply selected MATLAB, which was familiarized by students when they took the first signal processing course as a major simulation and design tool. However, other simulation tools were also welcomed when time was permitted.

The PCI-1200 DAQ board and LabView were chosen as a platform for teaching real-time signal processing. Students had gained their working knowledge after completing their project. We requested the students to design an FIR low-pass filter and an IIR band-pass filter using MATLAB and then applied the results to LabView programs to process his/her own audio signal sampled from a microphone. Meanwhile, the original speech spectrum and the processed speech spectrum were obtained and displayed by LabView for comparison.

Real-Time Digital System Implementations

A. We first introduce a tutorial to verify the data acquisition in the system to establish the students' working knowledge.

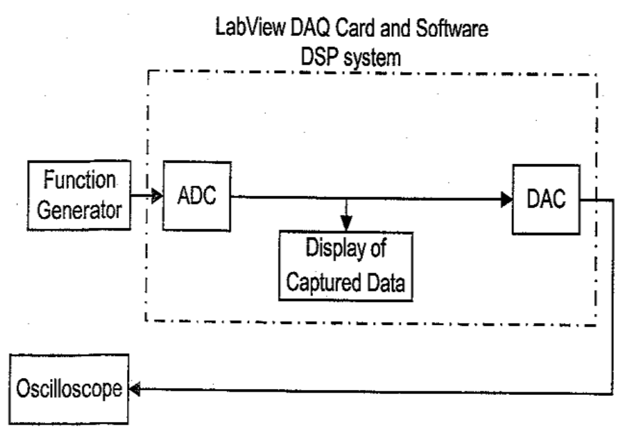


Figure 1 Real-time DSP Laboratory Setup.

The laboratory setup is shown in Figure 1 so that the students can verify the sampling and recovering process, and gain control of the analog-to-digital conversion (ADC) input and digital-to-analog conversion (DAC) output via DAQ card.

B. Design and construct a signal conditioning circuit (an audio amplifier using LM386 with the gain of 200) and put it into the audio system using a microphone as the input, as shown in Figure 2.

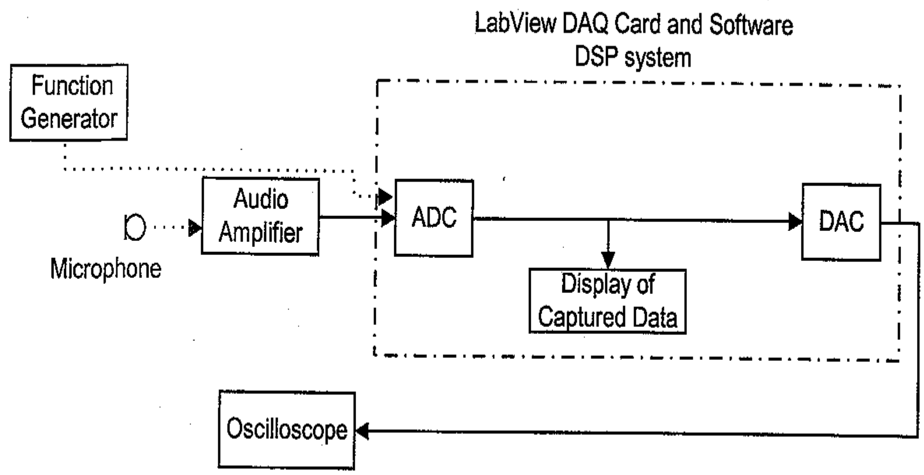


Figure 2 Test of Audio Amplifier.

C. Design and construct a second-order lowpass Sallen and Key Filter with a cut-off frequency of 3,200 Hz and connect it into the audio system, as shown in Figure 3.

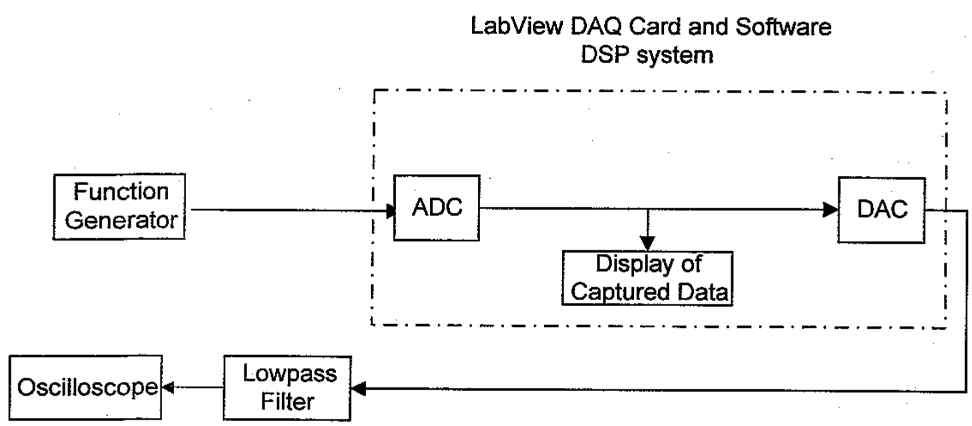


Figure 3 Test of the Low-pass Filter.

D. FIR filter design and implementation

First, design a lowpass FIR filter using the Hamming window design method by MATLAB. Then apply the filter coefficients from MATLAB for a digital filter created in LabView. The filter has 15 taps, a cut-off frequency of 800 Hz, and a sampling rate of 8,000 Hz. This lowpass FIR filter can effectively remove the high-frequency background noise.

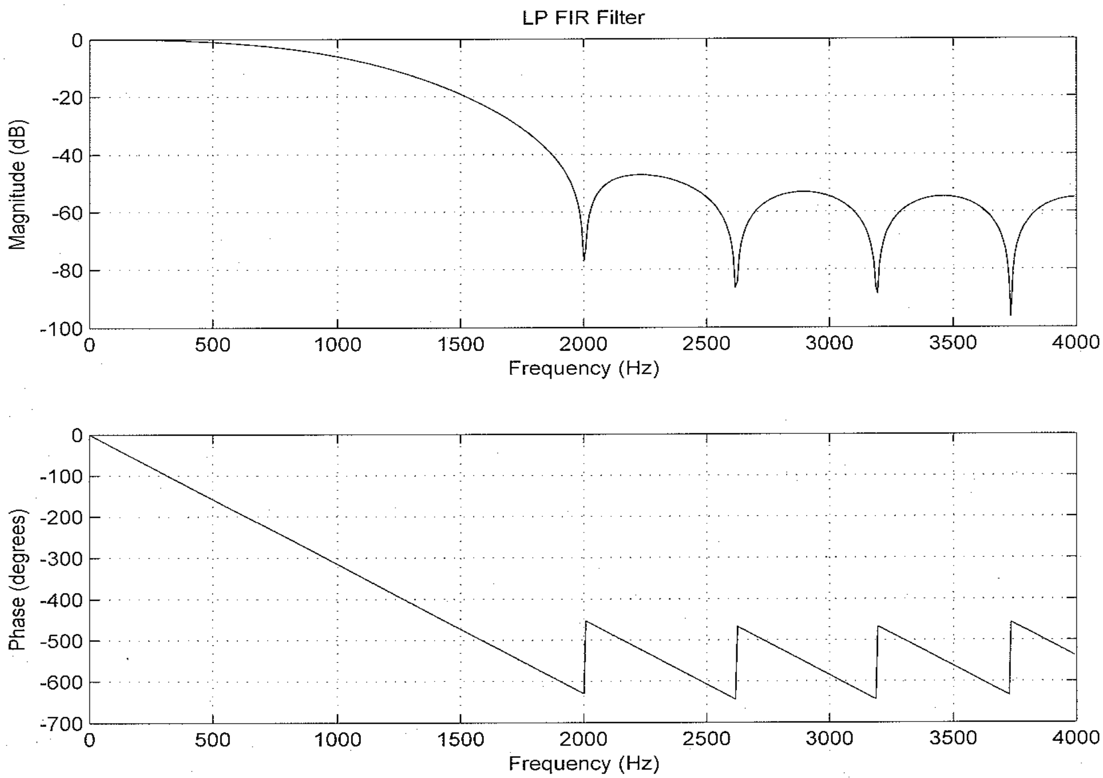
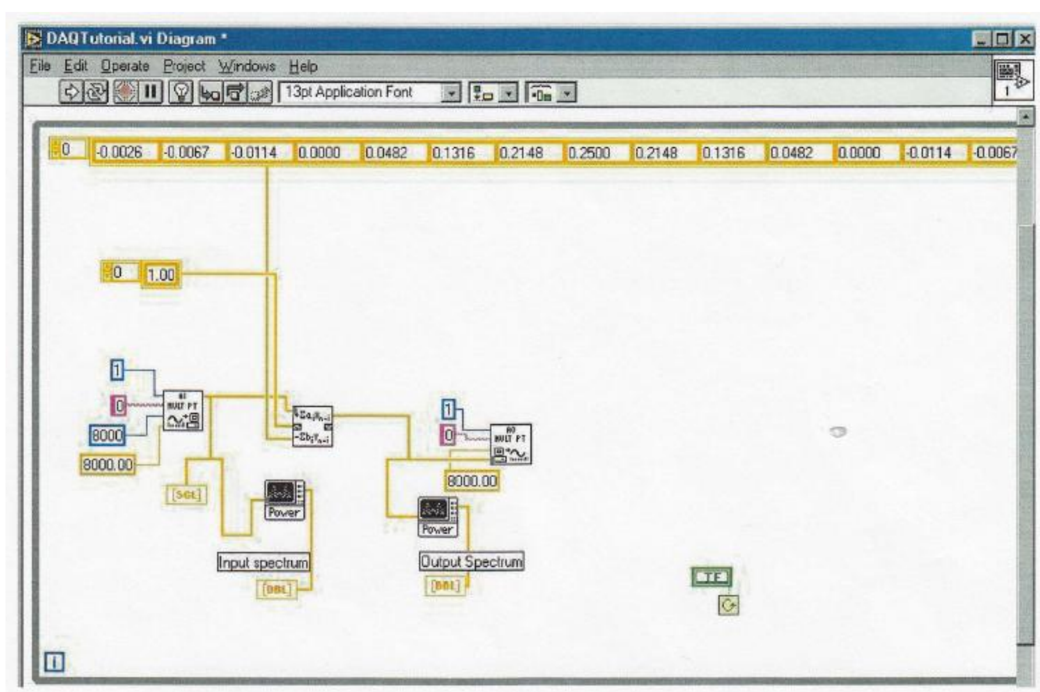


Figure 4a Frequency Responses of the Lowpass FIR Filter by MATLAB.



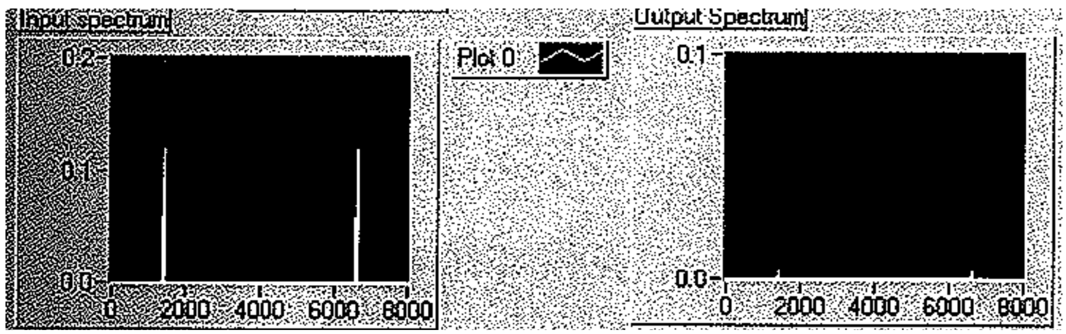


Figure 4b. Lowpass Filter Implementation by LabView.

E. IIR filter design and implementation

First, create a band pass IIR filter using the bilinear-transform method. This is a second order Butterworth filter that passes frequencies between 1,200 Hz and 1,600 Hz. A sampling rate of 8,000 Hz is used.
Then apply the filter coefficients from MATLAB for a digital filter created in LabView.

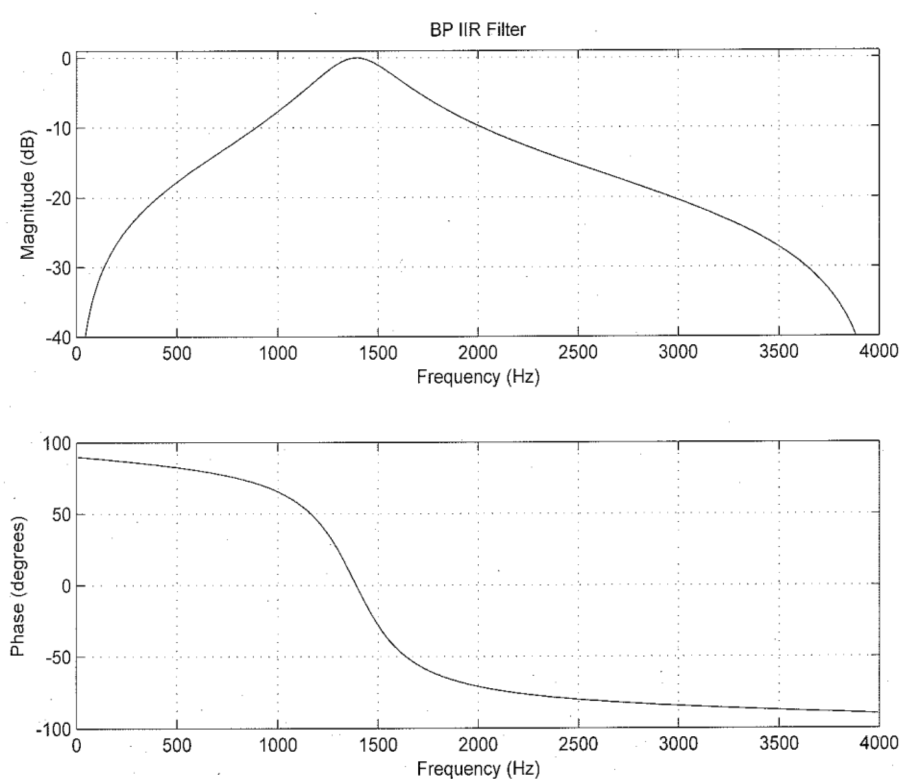


Figure 5a. Frequency Responses of the Band-pass Filter by MATLAB.

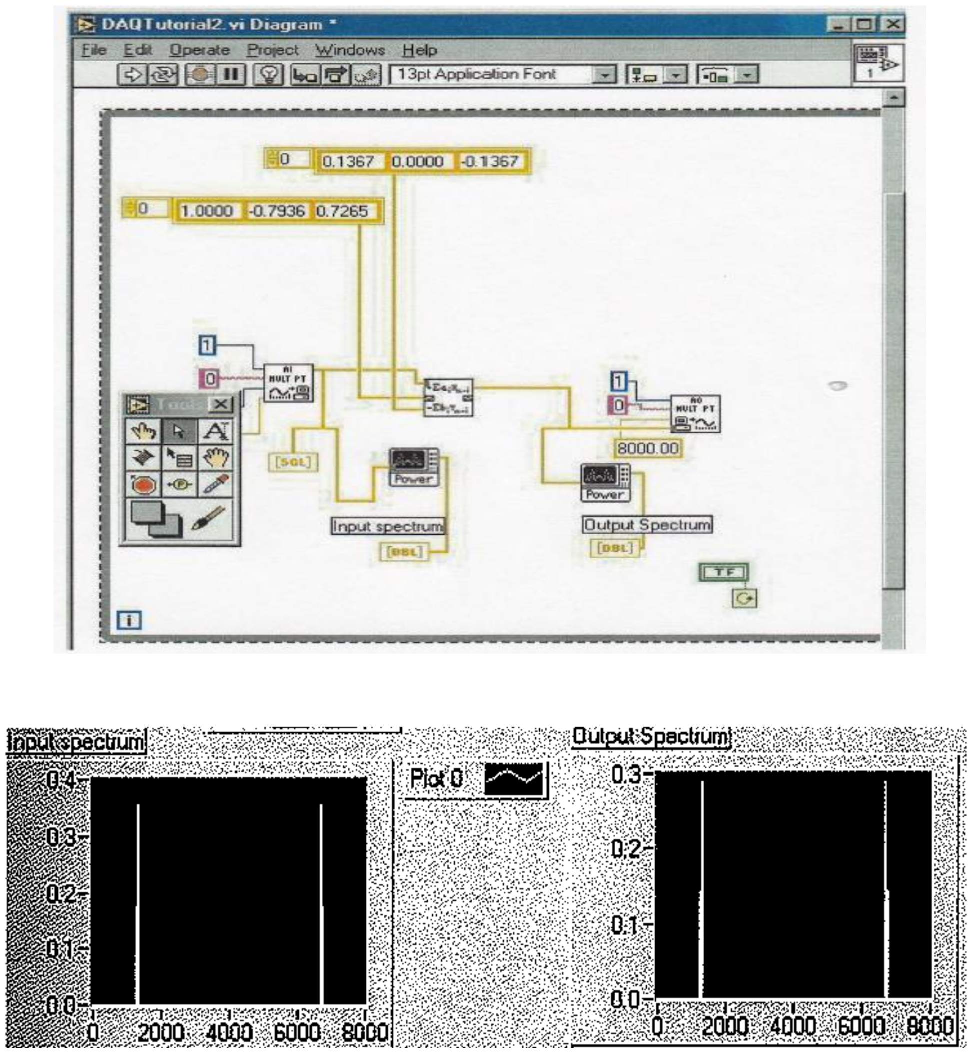


Figure 5b. Band-pass Filter Implementation by LabView.

F. The complete system

Finally, a complete audio system shown in Figure 6 is established and the snapshot of its hardware realization is shown in Figure 7.

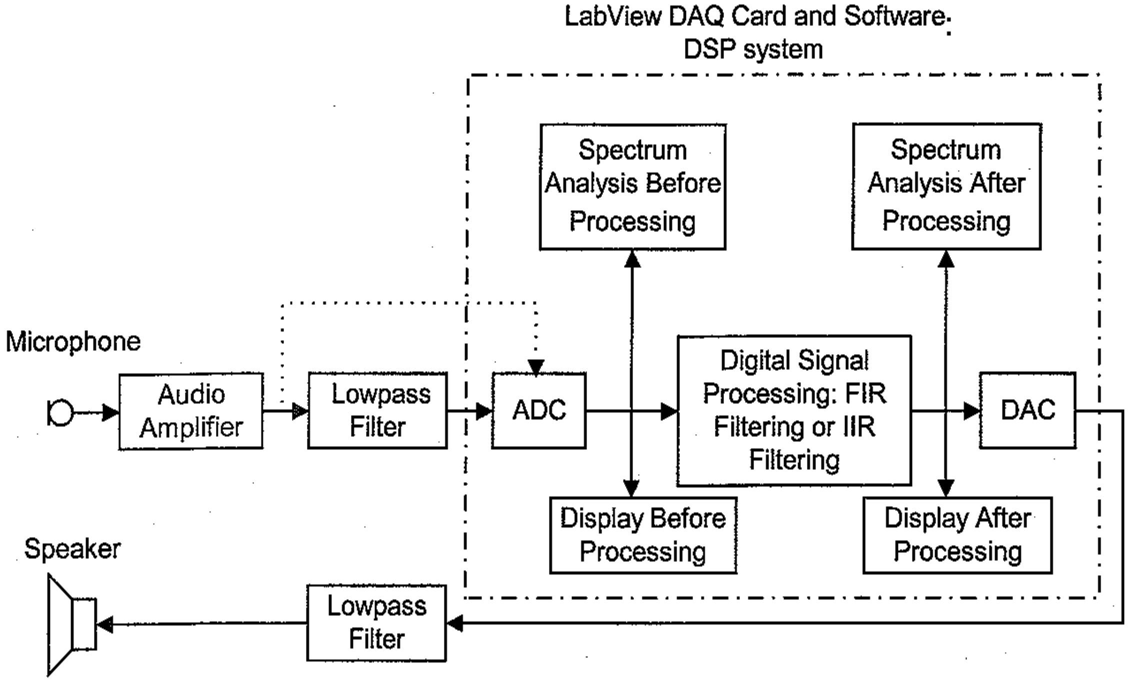


Figure 6. The Overall Speech Processing System.

To determine the functionality of the project, allow the speech input from a microphone and output to a speaker. Then test the program by inputting signals at various frequencies. Next, try speech through each filter: the low-pass filter cut off the high-end frequencies, making voices sound muffled; while the band-pass filter made speech samples sound watered down by only passing a small range of frequencies.

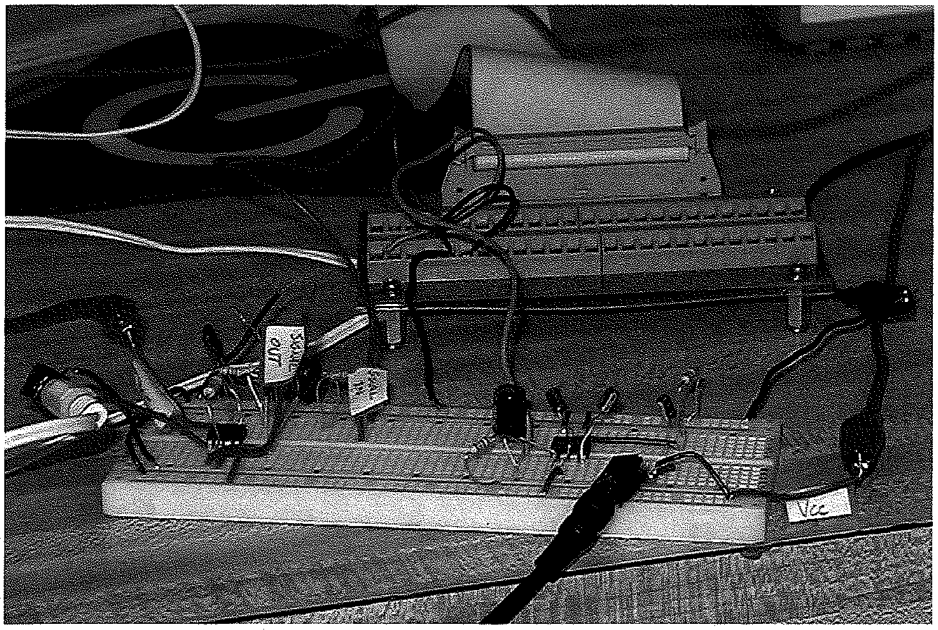


Figure 7. Snapshot of the Hardware Realization.

In summary, students were asked used the MATLAB tool to perform filter design, digital filtering, and spectral analysis of the input and output signals to examine the filtering effects. Students can also compare the calculated spectrum of the original speech to that of the filtered speech. Meanwhile, the students examined the filtered speech by listening to the processed sound and comparing it to his/her original voice.

IV. Course Outcome and Assessment

Upon the completion of the course, a survey was conducted to ask each student to evaluate his or her achievement. Table 1 indicates survey results collected from the previous three semesters for a total number of 40 students. Note that the rating scale in Table 1 was based on the percentage of the overall students.

Table I. Student Survey for Achievements.

Rating Scale	Level of understanding	Excitements	Textbook
4 - excellent	80%	85%	90%
3 - good	15%	15%	10%
2 - fair	5%	0%	0%
1 -unsatisfactory	0%	0%	0%

Most of the students remained excited about the course, since the hands-on real-time laboratories had motivated them. The textbook also helped a great deal to develop concepts using the worked numerical examples and MATLAB simulation examples. Table 2 summarized the DSP project evaluation.

Table II. Evaluations of DSP Project.

Rating Scale	Project Performance
4 - excellent	75%
3 - good	25%
2 - fair	0.0%
1 - unsatisfactory	0.0%

As shown in Table 2, 75% of students gave an “excellent” rating during the evaluation while the remaining percentage obtained a “good” rating. The evaluation data shows promising results in which students continue to apply their gained DSP knowledge to their career development. It is very encouraging to teach the advanced DSP course in the engineering technology program.

After learning DSP courses, the technology students were able to apply their newly gained knowledge and skills to their senior projects and our real-time DSP labs served as good preparation and practice for senior projects. In our campus, senior students are required to present and demonstrate their senior projects in the senior project fair, in which those projects were evaluated by the engineering technology faculty members and other senior students.

V. Future Improvement

Based on our experiences from teaching DSP courses, we felt that in Portion 1, all the lectures containing well-established topics including the digital spectrum, the FIR and IIR filter implementations and developed laboratories are suitable. Even though the topics of DFT, FFT, bilinear transform method and optimum design seemed challenging to our technology students due to the demand of their math proficiency to understand certain subjects, we still have successfully delivered the course materials with an emphasis on principles and hands-on applications instead of theoretical development. On the other side, based on the DSP industrial trend, we could improve the course by introducing additional topics such as adaptive filtering, subband coding and wavelet coding (as well as its applications). To improve our lab, we should make use of the lab equipment fund to adopt more advanced DSP platforms with multi-channel ADCs and DACs, so that many practical real-time DSP laboratory projects can be developed.

VI. Conclusions

It has been a continuous demand in the industry for engineering technology students to possess a working knowledge of the advanced and real-time DSP techniques. The traditional treatment of teaching those subjects using the profound mathematics is not appropriate. However, with the mathematical simplification equipped with numerical examples, MATLAB simulations, and real-time laboratories, the technology students are able to grasp concepts effectively and apply their gained DSP knowledge to their careers and future technical practice.

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