AC 2010-1594: A GRADUATE LEVEL COURSE: AUDIO PROCESSING LABORATORY

Buket Barkana, University of Bridgeport

A Graduate Level Course: Audio Processing Laboratory

Abstract

Audio signal processing is a part of the digital signal processing (DSP) field in science and engineering that has developed rapidly over the past years. Expertise in audio signal processing - including speech signal processing- is becoming increasingly important for working engineers from diverse disciplines. Audio signals are stored, processed, and transmitted using digital techniques. Creating these technologies requires engineers that understand real-time application issues in the related areas. With this motivation, we designed a graduate level laboratory course which is Audio Processing Laboratory in the electrical engineering department in our school two years ago. This paper presents the status of the audio processing laboratory course in our school. We report the challenges that we have faced during the development of this course in our school and discuss the instructor's and students' assessments and recommendations in this real-time signal-processing laboratory course.

1. Introduction

Many DSP laboratory courses are developed for undergraduate and graduate level engineering education. These courses mostly focus on the general signal processing techniques such as quantization, filter design, Fast Fourier Transformation (FFT), and spectral analysis [1-3].

Because of the increasing popularity of Web-based education and the advancements in streaming media applications, several web-based DSP Laboratory courses have been designed for distance education [4-8]. An internet-based signal processing laboratory that provides hands-on learning experiences in distributed learning environments has been developed by Spanias et.al[6] . This laboratory is based on an object-oriented Java-tool called Java Digital Signal Processing (JDSP). It facilitates interactive on-line simulations of modern statistical signal and spectral analysis algorithms, filter design tools, QMF banks, and state-of-the-art vocoders.

Although most of the universities and colleges offer DSP laboratory courses, traditional DSP laboratory courses do not provide the needed hands-on experience with real-time audio processing technologies. During the past years in our school, signal processing courses such as DSP laboratory, speech signal processing, speech coding, and multimedia signal processing have attracted a steady group of graduate students, mostly without practical knowledge in these fields.

Knowledge in the audio processing field is essential to the understanding of the function of current and future digital audio processing systems and to forming a strong foundation for the learning of newly developed digital devices/systems with applications to audio signals. The main contribution of the proposed course to the education of engineers will be (i) the understanding, through practical applications, of the DSP theory, (ii) student involvement with state-of-art technology, (iii) the development of real-time DSP hardware and software experiences in the music and speech processing fields, (iv) student familiarization of industry development processes, (v) the processing and editing of engineering documentation, and (vi) the development of design and research skills in the audio processing field.

Computer musicians use digital filters in the music they create. Without digital reverberation, it is difficult to get rich, full-bodied sound from the computer. However, reverberation is only the start of the capabilities of digital filters. A digital filter can arbitrarily shape the spectrum of a sound signal. Yet very few musicians are prepared to design the filter they need, even when they know exactly what they want in the way of a spectral modification [9]. At this point, it is very important to teach filter design in audio processing. By doing this, the proposed course can be adapted to the needs of different schools, such as Speech and Audio Technology centers, Departments of Music, Linguistics Departments, communications companies, computer science and engineering schools, and forensic audio engineering.

The following sections take a closer look at some of the tools, hardware, and experimental setups used in conducting this lab course. Section 2 outlines the course content that includes the possible audio and voice/speech projects using real-time DSK. Section 3 discusses the evaluation criteria of this course using the assessments and feedback of the students and the instructor.

2. ELEG 459: Audio Processing Laboratory

Lab Facilities
Hardware: Eleven PC stations. Each station is equipped with multimedia hardware capabilities including a real-time DSP board DSK5510. In addition, headphones, microphones, function generators, and oscilloscopes are available for every workstation in the laboratory.

Software: Each computer is equipped with general software tools for developing labs and projects including MATLAB R2007a, Code Composer Studio (CCS), Microsoft Office 2003, and Goldwave Digital Audio Editor. Goldwave Digital Audio Editor is a tool to play, edit, mix, and analyze audio signals for pod casting or telephone systems and more.
There are two reasons to choose fixed-point DSK for the proposed course. The first reason is that the fixed-point DSP is chosen most often for superior audio because of the following advantages:
1. Less dynamic range
2. Double-precision capable 48-bit processing
3. Programming flexibility that can guarantee proper behavior under the adverse conditions presented by audio signals
4. Lower-power consumption (floating point hardware is more complicated than fixed point; more transistors require more watts)

The second reason for choosing the fixed-point DSP is the recommended textbook for the course. The textbook, *Real-Time Digital Signal Processing Implementations and Applications* by Kuo, includes many hands-on experiments using C55x intrinsics with CC Studio version 2.0 based on DSP [10]. Students do not have time to design real-time DSP hardware and software from scratch in the class hours since the proposed course is a 3-cr course. The laboratory exercises are directly drawn from those pertinent to the student's material (the proposed textbook), and students are involved with the development of the target-oriented code.

More information on the C55x processor can be found at reference 11. The TMS320C5510™ DSP which is given in Fig.1. is supported by the CCS application which provides an integrated environment with an editor, debugger, project manager, profiler, C/C++ compiler, assembly optimizer and linker (code generation tools), simulator, real-time operating system (DSP/BIOS™), real-time data exchange between the host and target (RTDX™), real-time analysis, and visualization.

Fig.1. The fixed-point TMS320C5510 DSP Starter Kit (DSK) from Texas Instruments (TI).

Course design and content

Goal and Objectives

Our goal is to provide a laboratory course for audio signal processing courses that meets both the hands-on experience and educational needs of graduate engineering students. The objectives of the planned laboratory course are:

- To develop knowledge and understanding about the practical and real world applications of audio (voice, speech, music) processing
- To become familiar with audio processing hardware and software
- To develop knowledge and understanding about the fundamentals in DSP, using a combination of the various media of audio analysis
- To become familiar with real-time implementations of signal processing
- To develop an audio signal processing laboratory that can be adopted by other institutions.

Major Learning Outcomes

After successfully completing the course the student will be able to:

- Demonstrate knowledge and understanding about audio signal processing and its real world applications
- Manage audio software to generate and process audio signals

- Demonstrate skills and varied experiences in real-time DSP hardware

Organization of the course
A 3 credit, one semester lab course has students organized into groups of two students. At the beginning of each class, each group gives a written report to the instructor, and this report includes the results of the previous week's laboratory exercise. Each report is corrected, graded, and returned to the students the next week. Students have an early feedback of their performance, and this is very helpful in subsequent laboratory work.

Each group prepares a hands-on project as a final project. Students start preparing their final project at the middle of the semester. Each student is expected to spend approximately three hours per week in the lab, in addition to the scheduled lab times. Since students have access to the laboratory outside of scheduled lab times, there is plenty of time to finish their final project on time.
Prerequisite:
DSP is the only prerequisite course for the proposed course.
Recommended Textbooks:
Sen M. Kuo, Bob H. Lee, and Wenshun Tian, Real-Time Digital Signal Processing Implementations and Applications, Second Edition, John Wiley and Sons Inc., ISBN 0-470-01495-4, 2006
Course Syllabi
Students are introduced to the DSP hardware and software development tools during the first two weeks of the semester. In addition, lectures will cover background material pertinent to lab in the following areas: the acoustics and acoustic analysis of audio (music and speech), the physiology of speech production, and the perception of audio.

Lab experiments
Most of the laboratory experiments are adopted from the textbook by Kuo et al. [18]. Since students do not have a strong knowledge on C/C++ programming, they are asked to be involved with the development of target-oriented codes from the textbook. A brief description of the experiments is given below.

Lab1: *Introduction to Real Time Digital Signal Processing.*

Lab2: *Introduction to TMS320C55x Digital Signal Processor (Hardware Introduction).*

Lab3: *Sampling and quantization of audio signals*
This experiment is used to demonstrate quantization effects, overflow and saturation arithmetic, and to determine the proper fixed-point representations using speech or music signals. Students find the mean and variance of quantization noise and evaluate the quantization effects of speech and music signals using the DSK for real-time experimentation with different word lengths. The Goldwave audio editor is used to play the quantized signals.

Lab4: *Design and Implementation of Finite Impulse Response (FIR)*
This experiment aims to overcome filtering and filter design issues using audio signals. Students are asked to remove unwanted frequency components on speech and music signals with different

order FIR filters. In addition, the audio signal sampling rate is converted from 32 to 24 kHz. Students first interpolate the signal sampled at 32 kHz to 96 kHz, and then decimate it to 24 kHz.

Lab5: *Design and Implementation of Infinite Impulse Response (IIR) Filters*
Students focus on the design, realization, implementation, and application of digital IIR filters.

Lab6: *Adaptive Filter Design*
Modern mobile communications technology requires knowledge of adaptive filter design. Audio signals and background noise are often random in nature. In this experiment, the properties of the random processes are reviewed. Students implement the filter design in Figure 2. Students are asked to adjust the step size of the LMS algorithm and the length of the filter to minimize the error using the TMS320C5510. The Goldwave audio player and MATLAB are used to watch the convergence/divergence of the output signal.

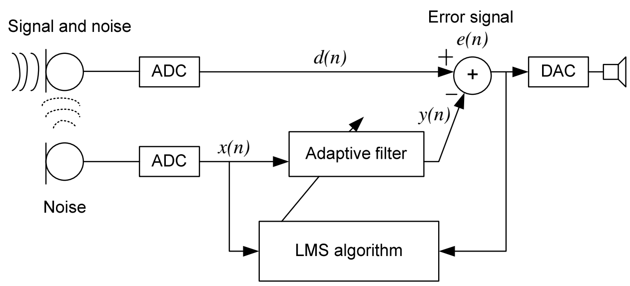

Fig.2. Noise canceler using a one-tap adaptive filter [19]

Lab7: *Echo Cancellation*
This experiment focuses on adaptive voice echo cancellers for long-distance networks. Cancelling the voice echo in long distance signal transmission and acoustic echo in hands-free speakerphones is very important.

Lab8: *Audio Enhancement Techniques*
Excessive noise degrades the performance of audio signal processing techniques. Speech enhancement (noise reduction) algorithms become increasingly important to improve voice quality in noisy environments for hands-free applications.
Students are given speech signals that are recorded in a noisy environment and are asked to use voice activity detection algorithm and various filtering ideas to reduce the background noise using DSK5510. This experiment is another example of the combination of the usage of MATLAB, the Goldwave Audio Editor, and DSK5510.

Lab9: *Modified Discrete Cosine Transform*
Speech coding is a process that leads to the representation of analog waveforms by sequences of binary digits. Although high-bandwidth channels and networks are becoming more viable, speech coding for bit rate reduction has retained its importance. Here, a basic audio coding algorithm in the frequency domain, the Modified Discrete Cosine transform (MDCT), is introduced. Students gain knowledge and experience (i) on the use of filter banks, (ii) on converting time-domain signals to frequency domain coefficients, (iii) on developing and using psychoacoustic models,

(iv) on quantization/dequantization, and (v) on aliasing cancellation in the time domain and frequency domain.

Lab10-11: *Final Project*
Students are asked to design and solve open-ended problems as part of the laboratory final project.

3. Evaluation and Assessment

Formal and informal evaluations were conducted. For formal evaluation, a questionnaire was developed for distribution at the end of the corresponding semester. The questionnaire results obtained with Audio Lab, students considered their experience with the lab to be very good. The questionnaire is given at Table 1.

Table 1. Student Evaluations, the questionnaire.

Sample size: 57 Response rate:29 Questions	% SA Strongl Agree	% A Agree	%N Neutral	%D Disagre	%SD Strongly Disagre
1. The laboratory experiments provided me with a better understanding of DSP concepts for audio signals.	%78	%11	%11		
2. The lab experiments were clearly outlined and objectives are well explained.	%67	%16.5	%16.5		
3. Booklet (it includes experiments and their theory) was useful and informative.	%87.5		%12.5		
4. I believe that this real-time DSP experience is valuable to my professional future.	%100				
5. The final project gave me the opportunity to demonstrate individual initiative and creativity.	%87.5	%12.5			
6. I recommend this laboratory course to other students.	%67	%22	%11		
7. The teaching assistant was very helpful in the laboratory.	%100				
8. Lab should be available to students more often.	%56	%33	%11		

All the students think that the real-time DSP experience is helpful to their professional future. In addition, all the students found the instructor very helpful in the lab. A high percentage considers the Audio Lab useful in helping them to understand the concepts of DSP in general and of music and speech signals in particular. Almost 88 percent of the students strongly agreed the final project as an opportunity to demonstrate individual initiative and creativity. Most of the students showed great interest in the final project. Some of the projects were presented at national conferences as student papers or posters [12,13]. Again, almost 88 percent of the students strongly agreed that instructional course materials were useful and informative. 56 percent of the students strongly agreed that lab should be available to them more often.

Informal discussions of perceptions of laboratory work were held with individuals and groups of students for two years. Although these were not audio taped, notes were written up immediately afterwards. Based on these notes and teaching experience from the 2007-2009 academic years, most of the students enjoyed the Audio laboratory experience and appreciated the skills they gained.

Challenges faced by students and instructor

Software skills: Some students found the course very challenging with their initial weak background in software. Most of the engineering graduate students are from overseas, coming to America and to the University of Bridgeport. Although they have a good mathematical background, they usually do not have any experience with software tools such as MATLAB. This issue causes a big challenge to students and the instructor of this course. To overcome this challenge, mini MATLAB courses are offered at the beginning of the semester. The lab assistant teaches the Digital Signal Processing Toolbox in MATLAB. The students who have a software background perform much better than the others who do not have any software background.

Team work: During the past semester, students were organized into groups of three. The instructor noticed that one student was always left out. For this reason, the capacity of the course is reduced by half. The students may elect to work alone or with a lab partner.

Time: The students have three hours per week to complete the listed experiments. We noticed that most of the students needed more time to finish each lab. With the lab assistance's guidance, they were allowed to work in the laboratory using necessary hardware and software tools outside of the normal lab/lecture hours.

4. Conclusion

An audio processing laboratory has been presented to enhance the material and concepts covered in DSP, speech, and multimedia signal processing courses. This course is integrated in the EE curriculum at the School of Engineering, University of Bridgeport, USA. Hands-on real-time experiments are based on the TMS320C5510 DSK and MATLAB. Students learn industry-style design with the TMS320C5510 DSK. This experience helps the students when they work as a professional engineer in audio processing. Students have gained experience on these topics: research experience, industrially relevant experience, MATLAB programming, real-time DSP, designing and implementing projects, writing and presentation skills, all of which they will need later when working in industry.

Our experience with the laboratory has been very positive. The students like the experiments, and they feel that they are useful in solidifying concepts covered in lecture. They have a better sense of the power of the algorithms used in the signal processing of music and speech. To overcome their weak software background, students receive a set of pre-laboratory assignments in many cases, and these provide theoretical and practical background in conducting experiments.

References

1. I. M. Abdel-Qader, B. J. Bazuin, Real-Time Digital Signal Processing in the Undergraduate Curriculum, *IEEE Transactions on Education*, vol.46, no.1, February, 2003
2. D. Jacoby and R. Saint-Nom, Nice experiences teaching SP in Argentina, Acoustics, Speech, and Signal Processing, *IEEE International Conference on acoustics, Speech, and Signal Processing*, 2689-2692 vol.5, 2001
3. J. P. Frantz, H. Choi, R. Baraniuk, DSP Education at Rice University,
4. S. Gallardo, F. Barrero, S. L. Toral, M. J. Durán, eDSP*lab*: A remote-accessed instrumentation laboratory for digital signal processors training based on the Internet, IEEE Industrial Electronics, IECON 2006 - 32nd Annual Conference on, 6-10 Nov. 2006
5. A. Spanias, and V. Atti, Interactive Online Undergraduate Laboratories Using J-DSP, IEEE Transactions on Education, vol.48, no.4, November 2005
6.A. Spanias, S. Urban, A. Constantinou, M. Tampi, A. Clausen,X. Zhang, J. Foutz and G. Styliano, Development and Evaluation of a Web-Based Signal and Speech Processing Laboratory for Distance Learning, Computers in Education Journal, April-June 2000
7.Z. Dvir, Web-Based Remote Digital Signal Processing (DSP) Laboratory Using The Integrated Learning Methodology (ILM), 2006 International Conference on Information Technology: Research and Education , 2006
8.A. Kalantzopoulos, D. Karageorgopoulos, E. Zigouris, A LabVIEW based Remote DSP Laboratory, International Journal of Online Engineering (iJOE), vol.4, 2008
9. Smith, J.O., “Introduction to Digital Filters”, in Introduction to Digital Filters with Audio Applications,W3K Publishing, http://books.w3k.org/,2007, ISBN 978-0-9745607-1-7.
10. S.M. Kuo, B.H. Lee, and W. Tian, Real-Time Digital Signal Processing Implementations and Applications, pp.49-50 England: Wiley, 2007
11. *TMS32055x Code Composer Studio: Tutorial*, Texas Instruments, http://www.ti.com/
12. Bachu R., Adapa B.K., Kopparthi S., Barkana B.D., “Separation of Voiced and Unvoiced Speech Signals using Energy and Zero Crossing Rate”, ASEE Regional Conference, West Point, March 2008. (Best Student Paper Award, Second Place)
13. Stark B., “Dual Architecture Implementation of Acoustic Echo Cancellation”, Advisor: Barkana B.D., Proceedings of the 2009 ASEE NE American Society for Engineering Education Conference, April 3-4, 2009, (Best Student Paper Honorable Mention Award)