# AC 2009-124: TEACHING ADVANCED DIGITAL SIGNAL PROCESSING WITH MULTIMEDIA APPLICATIONS IN ENGINEERING TECHNOLOGY PROGRAMS

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# Teaching Advanced Digital Signal Processing with Multimedia Applications in Engineering Technology Programs

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## Abstract

In this paper, we present our pedagogies and experiences from teaching advanced digital signal processing (DSP) within the engineering technology curricula, which include electrical, biomedical, and computer engineering technologies. The course is an elective for senior students and is designated as the second DSP course in engineering technology programs with a focus on real-time processing and multimedia applications. The course prerequisite assumes that the students have acquired working skills of the Laplace transform, the Fourier analysis, the z-transform, the discrete Fourier transform, and analog and digital filters from the first DSP course. In this course, the technology students will continue to explore advanced techniques such as real-time digital filter implementations, adaptive filtering, multi-rate signal processing, and digital image processing, and will further examine DSP applications in the areas of telecommunications, biomedical engineering, and multimedia systems. By offering a broad coverage of topics and case studies, the course could possibly be beneficial to all electrical, biomedical, and computer engineering technology students.

Since teaching advanced DSP topics throughout the engineering technology program has the requirement of being at a hands-on and engineering technology level, adopting traditional teaching approaches and using textbooks dealing with complicated mathematics and theories used in a four-year engineering program may not be appropriate. In this paper, we will explain course prerequisites and will describe our teaching methods, which include real-time signal processing laboratories using low-cost DSP processors, and hands-on projects. We will also present a course assessment and outcome, which will include how the students apply their gained DSP knowledge to their capstone senior projects. Finally, we will address the possible improvement of the course content and associated laboratories.

## 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 DSP, but also provide a second elective DSP course in which real-time applications and corresponding advanced topics such as multi-rate signal processing, adaptive filtering, and digital image and video processing [1, 9, 10] are introduced.

In our engineering technology program, the second DSP elective course is designed for senior students with a focus on real-time signal processing and multimedia applications. The course prerequisite assumes that the students have already acquired working skills of the Laplace transform, the Fourier analysis, the z-transform, the discrete Fourier transform, and analog and digital filters from the first DSP course. The technology students in this course will continue to explore advanced techniques such as real-time digital filter implementations, adaptive filtering, multi-rate signal processing, and digital image processing, and will further examine DSP applications in the areas of telecommunications, biomedical signal processing, and multimedia systems. While offering a broad coverage of topics and case studies, this course could be beneficial to all 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 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 will describe our designed course content first, and then we will introduce real-time signal processing laboratories using low-cost DSP processors such as the TI TMS320C671x DSP board and simulation tools such as MATLAB, and hands-on projects. 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. Signal Processing Course Requirement

The first DSP course covering the key topics of the sampling theorem, digital spectrum, z-transform, properties of the digital filters, and design of the finite impulse response (FIR) and infinite impulse response (IIR) filters is a prerequisite. Students in the advanced DSP course apply these established skills for designing, implementing, and verifying various applications such as the digital crossover audio systems, dual tone multi-frequency (DTMF) tone generation and detection, and so on. Specifically, the skills of the digital FIR filter design and signal spectral analysis obtained from the first DSP course are necessary for the decimator and interpolator filter design and verifications in the area of sampling rate conversions. Furthermore, a grasp of concepts and principles of the first DSP 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 the window functions. Hence, the prerequisite of the first DSP course implies the ASP course. On the other hand, some technology program may possibly offer a combined ASP and DSP course, which could be an alternative for the pre-requisite.

### 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 beneficial in understanding the advanced course materials such as the employment of the derivative operation in order to develop the least mean square (LMS) algorithm for adaptive filters. 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. On the other hand, to gain real-time signal processing experience using a DSP board, students are required to code DSP algorithms using the C language. Similarly, since the students at the senior level have already acquired their C program knowledge early on in the C programming course, which has been used in the microcontroller embedded systems course, this requirement can easily be satisfied.

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

1. Digital signal processing (or analog and digital signal processing)
2. MATLAB programming and basic C programming skills

## III. Course Content and Associated Laboratories

We have divided the course content into three portions. First, the DSP fundamentals were reviewed with an emphasis on real-time digital filter implementation and applications. The second portion introduced advanced DSP topics, which include adaptive filtering, waveform coding, and multi-rate signal processing, such as sampling rate conversions. Finally, in the third portion, we taught students the basic techniques of digital image processing.

The course was taught for 15 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 book provides most of the topics required by the course, specifically, the topics of adaptive filters, multi-rate signal processing, multimedia applications such as waveform coding, image processing basics, and video signals. The text 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. The book also covers materials taught in the first DSP course (Chapter 1 through Chapter 8 [1]) and can serve as a comprehensive reference.

To minimize the time for learning different simulation tools, we simply selected MATLAB, which was familiarized by students when they took the first DSP course, as a major simulation and design tool. However, other simulation tools were also welcomed when time was permitted.

The TI TMS320C671X DSP board was chosen as a platform for teaching real-time signal processing. Students had gained their working knowledge of the TI Code Composer Studio (CCS) and had experienced real-time DSP coding by building the CCS project, compiling and linking it, loading the executable code onto the C671X DSP board, and running the DSP programs. We requested the students to implement DSP algorithms using a float-point format due to its simplicity (no signal scaling and coefficient quantization are necessary) in order to ensure that they can focus on various applications.

**A. Real-Time Digital Filter Implementations**

We first briefly reviewed several key topics from the first DSP course: the sampling theorem, DFT and signal spectrum, filter frequency responses, and filter implementations using the direct form I and direct form II. These selected topics were covered in Chapters 2, 4, 6 [1]. Again, we also concisely reviewed the digital filter design in Chapters 7 and 8 dealing with the FIR and IIR filters. As shown in Table 1, Experiment 1 introduces a tutorial on TI CCS with the TMS320C671X DSP board [3-8] to establish the students' working knowledge. Experiment 2 is a comprehensive MATLAB experiment which reviews the signal spectral analysis using the fast Fourier transform (FFT), and the design of digital FIR and IIR filters.

Table 1 List of Experiments in Part 1.

| | |
|---|---|
| Experiment 1 | TI Code Composer Studio tutorial, sampling and gain control for input and output using TMS320C671X DSP |
| Experiment 2 | Review of DFT/FFT, signal spectrum, and FIR and IIR filter design using MATLAB |
| Experiment 3 | FIR filter implementations using TMS320C671X DSP |
| Experiment 4 | IIR filter implementations and waveform generation using TMS320C671X DSP |
| Experiment 5 | Implementation of the DTMF tone generator using digital IIR filters and DTMF detector using the Goertzel algorithm |

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

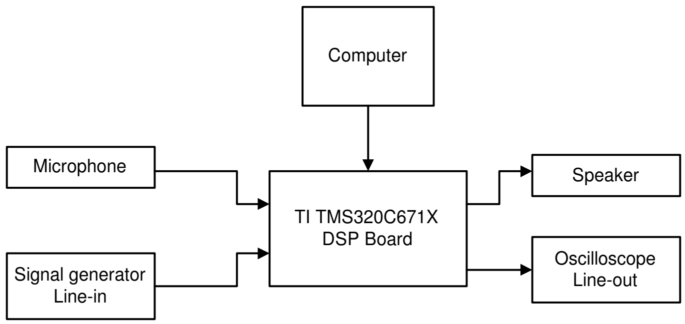



Figure 1 Real time DSP laboratory setup.

In Experiment 2, a review lab, students used the MATLAB tool to perform filter design, digital filtering, and spectral analysis of the input and output signals to examine the filtering effects. Figures 2a to 2c show an example of the filtering effect of a speech signal conducted in Experiment 2. Figure 2a shows the frequency responses of the designed bandpass filter while Figure 2b depicts the original speech (top plot) and the bandpass filtered speech (bottom plot). As illustrated in Figure 2c, the students can compare the calculated spectrum of the original speech to that of the bandpass filtered speech. Meanwhile, the students examined the filtered speech by listening to it and comparing it to the original sound. The lowpass and highpass filters were examined accordingly.

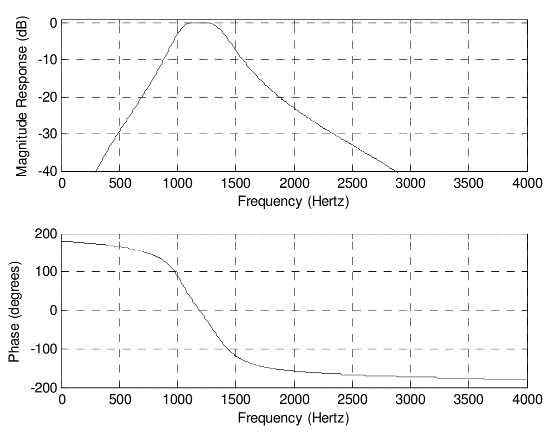



Figure 2a Frequency responses in Experiment 2.

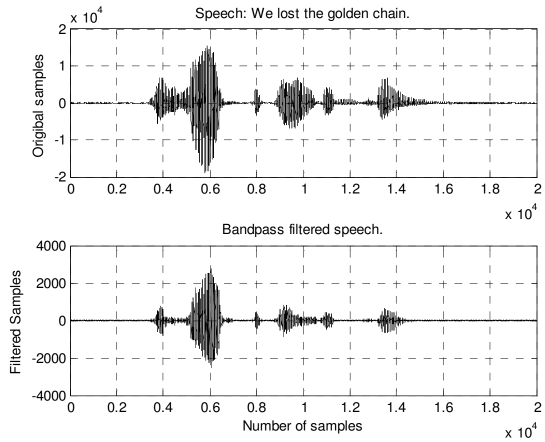



Figure 2b Original speech and filtered speech in Experiment 2.

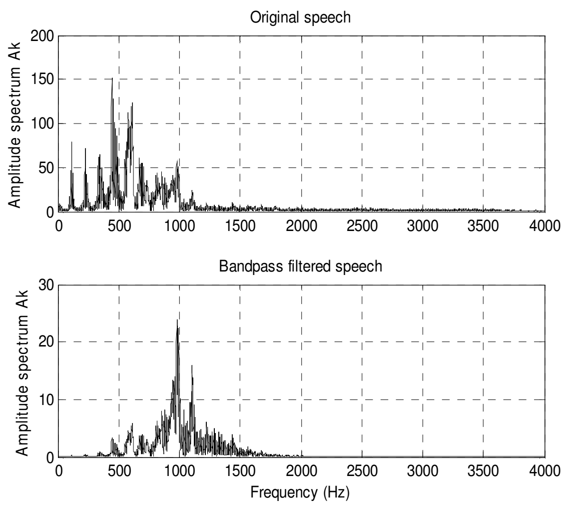



Figure 2c Spectra of the original speech and filtered speech in Experiment 2.

Next, the students implemented their designed FIR and IIR filters in Experiment 2 using the direct forms I and II, respectively, and verified the filter performances via the listening procedure described in Experiment 2. The adopted laboratory experiments 3-4 are listed in Table 1. Sample code examples in the direct form I for FIR and IIR filters are given in Figure 3 and Figure 4, respectively. In addition, DSP applications such as digital oscillators and waveform generators (according to the Fourier series expansion) were conducted in Experiment 4. The students firstly learned how to implement a digital oscillator and then later applied it to generate the fundamental frequency oscillator and harmonic frequency oscillators based on the Fourier series expansion of a periodic waveform (such as the square waveform) within the Nyquist bandwidth limit set in the DSP board. The periodic waveform was eventually produced by adding these digital oscillator outputs with their designated amplitudes.

```
volatile int sample;
float x[3]={0.0, 0.0, 0.0};
float b[3]={0.5, 0.2,  0.5};
float y[1]={0.0};
 interrupt void AtoD()
 {
        int i;
      sample=mcbsp0_read();  /* ADC */
      for(i=2; i>0; i--)    /* Update the input buffer x[3] */
      {   x[i]=x[i-1]; }
      x[0]= (float) sample;
      y[0]=0;
      for(i=0; i<3; i++)
      {  y[0]=y[0]+b[i]*x[i]; }
      sample= (int)  y[0];  /* the processed sample will be sent to DAC */
 }
```

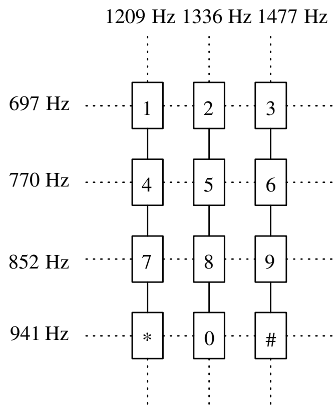
Figure 3 Example of FIR filtering using the direct form I.

```
volatile int sample;
float b[3]={0.5, 0.7, -0.5};
float a[3]={1, 0.4, -0.6};
float x[3]={0.0, 0.0, 0.0};
float y[3]={0.0, 0.0, 0.0};
interrupt void AtoD()
{
       int  i;
      sample=mcbsp0_read();  /* ADC */
      for(i=2; i>0; i--)          /* Update the input buffer */
      {   x[i]=x[i-1];   }
          x[0]= (float) sample;
      for (i=2;i>0;i--)      /* Update the output buffer */
      {    y[i]=y[i-1];  }
      y[0]=b[0]*x[0]+b[1]*x[1]+b[2]*x[2]-a[1]*y[1]-a[2]*y[2];
       sample= (int)  y[0];  /*  the processed sample will be sent to DAC */
}
```

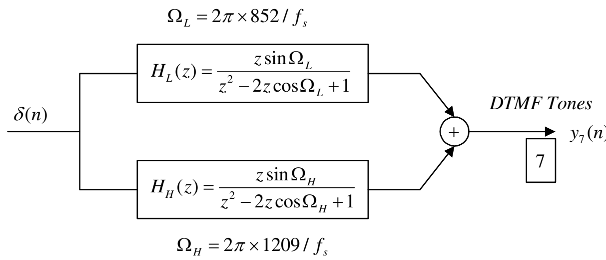
Figure 4 Example of IIR filtering using the direct form I.

In order to demonstrate filter applications, the DTMF tone generation and detection based on the Goertzel algorithm were introduced in the lecture (see Chapter 8 [1]). Then the first part of the course ended with Experiment 5 as listed in Table 1, in which students performed both the MATLAB simulation and real time implementation.

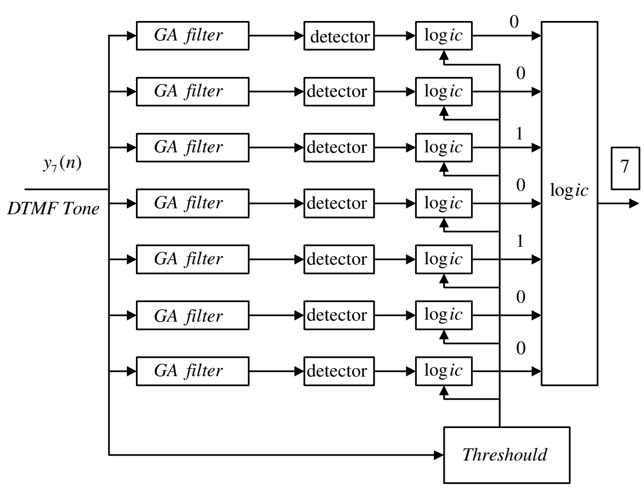
Figure 5 describes a block diagram of the DTMF tone generation and detection for the digit “7”. Notice that the sampling rate is set to 8 kHz. The students first generated the DTMF tone according to the dual frequency specifications listed in Figure 5a using a combination of two digital oscillators (developed in Experiment 4), and then they verified each DTMF tone by comparing it to the corresponding tone produced from the standard telephone keypad. Next, seven (7) Goertzel filters were developed and implemented, each of which is responsible of detecting its designated frequency as shown in Figure 5a. Based on all the Goertzel filter outputs, a binary codeword (after comparing to the threshold) was generated to identify the dialed digit as depicted in Figure 5c for the case of a digit 7. This experiment had demonstrated a practical DSP application in the telephone industry.




a. DTMF tone specifications




b. DTMF tone generator for the digit “7”




c. DTMF tone detection for the digit “7”

Figure 5 DTMF tone generation and detection.

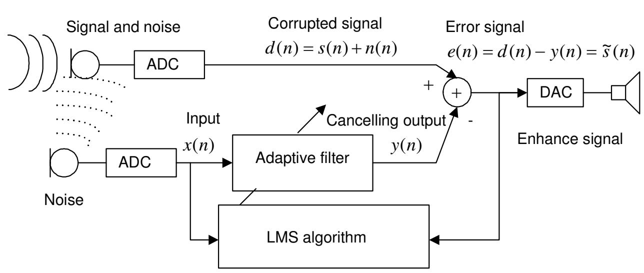
**B. Adaptive Filtering, Waveform Coding, and Multi-rate Signal Processing**

In the second portion of the course, the advanced DSP topics, which include adaptive filtering, speech and waveform coding, and multi-rate signal processing, were introduced. Regarding adaptive filtering, we taught the LMS type FIR filters with a focus on their applications such as system modeling, noise cancellation, echo cancellation, and line enhancement. Simplified numerical examples and MATLAB simulations for case studies were illustrated instead of teaching the adaptive signal processing theory. As shown in Table 2, Experiment 6 only covered system modeling and noise cancellation, leaving the other applications from which the students could select for their final course projects.

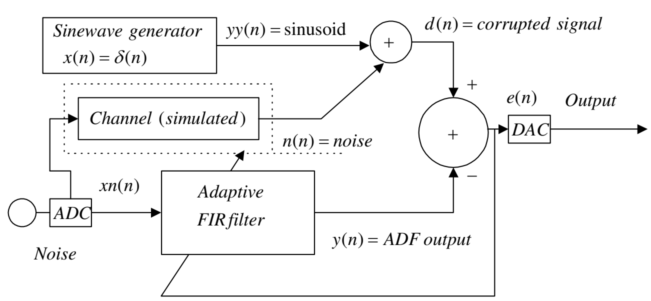
Table 2 List of Experiments in Part 2.

| | |
|---|---|
| Experiment 6 | Adaptive filtering: system modeling and noise cancellation |
| Experiment 7 | PCM, mu-law PCM, and ADPCM coding with applications to speech compression |
| Experiment 8 | Sampling rate conversions and polyphase implementations for decimation and interpolation filters |

Real-time implementation of noise cancellation requires two ADC units (one for sensing the primary signal and the other for sensing the reference signal) and one DAC unit for outputting the enhanced signal. This is challenging in our real-time implementation, since the TMS320C671X DSP board only accommodates one ADC unit and one DAC unit. However, we could adopt a modified laboratory setup for noise cancellation as depicted in Figure 6, where the primary signal is a generated sinusoid corrupted internally by a disturbance, which is produced from the simulated channel using the sensed reference noise obtained from the ADC channel on the DSP board. The adaptive LMS FIR filter uses the sensed reference noise, internal error signal, and the LMS algorithm to produce the canceling output. This canceling output is subtracted from the corrupted signal to remove the correlated noise. The resultant enhanced signal is then sent to the DAC channel, which produces a clean signal when the adaptive filter converges and continues to operate.




a. Principle of noise cancellation




b. Laboratory setup for noise cancellation

Figure 6 Laboratory setup for noise cancellation using the adaptive LMS filter.

Waveform coding, such as pulse code modulation (PCM), mu-law PCM, and adaptive differential PCM (ADPCM), was able to be covered in lectures due to a significant amount of help from MATLAB simulations. Due to a time constraint, the students mainly focused on simulations using MATLAB (see Experiment 7 in Table 2), in which the students compressed the speech data samples using the given MATLAB programs for PCM, mu-law PCM, and ADPCM algorithms, and recovered the speech waveform using the compressed codes. Furthermore, the students verified the compression ratio for each codec and evaluated the sound quality by listening to each recovered speech and then comparing it to the original one. The real-time implementation was not required due to a lack of the communication channel in our current DSP board, but a real-time simulation could possibly be used as in Figure 7, in which all coding and quantization were done internally so that there was no need for a transmission channel. The students could therefore spend sufficient time to verify the speech/audio coding quality by listening and comparing the recovered speech to the original speech.

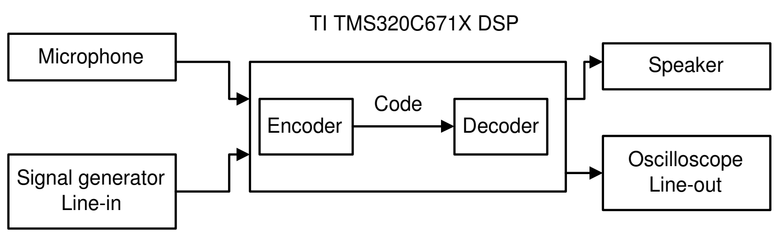
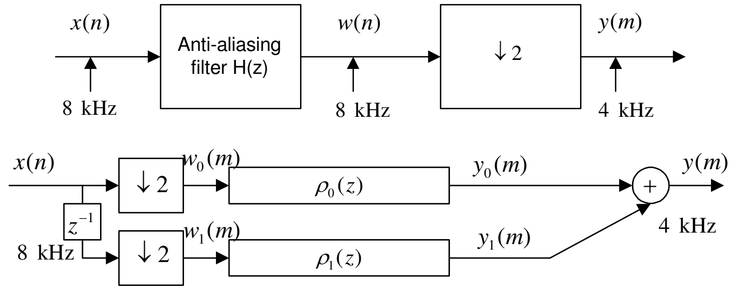


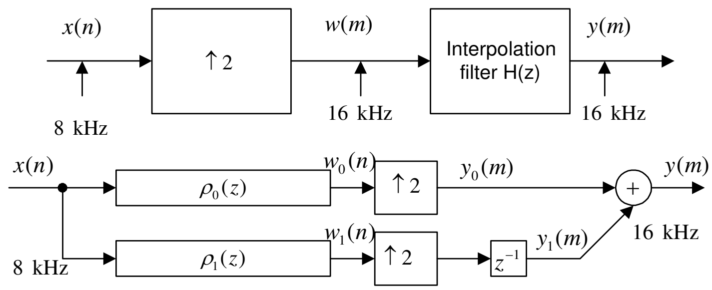

Figure 7 Real time simulation for waveform coding.

The course continued to cover state-of-the-art coding techniques, such as discrete-cosine transform (DCT), windowed DCT (WDCT), and MPEG audio. Our lecture examined these coding techniques via the MATLAB simulations listed in Chapter 11 in the textbook [1].

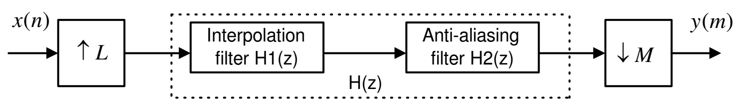
Regarding multi-rate signal processing, our lectures focused on designing digital anti-aliasing and anti-image (interpolation) FIR filters for sampling rate conversions. In addition, we also covered polyphase structures for implementing anti-aliasing and anti-image filters. Figure 8a illustrates the decimator along with its polyphase filter implementation, in which an original sampling rate of 8 kHz is down converted to 4 kHz, while Figure 8b depicts an interpolation process and its polyphase filter implementation to convert a sampling rate of 8 kHz to a higher sampling rate of 16 kHz. The sampling rate conversion with a non-integer factor is illustrated in Figure 8c, in which a cascade of the interpolation filter and anti-aliasing filter is employed. The students first conducted the MATLAB simulations (see Experiment 8 listed in Table 2) as described in Figure 8, and then examined the converted speech in the frequency domain and verified the converted sound by listening to it and comparing it to the original speech.




a. Decimation




b. Interpolation




c. Changing the sampling rate with the non-integer factor

Figure 8 Sampling rate conversions in Experiment 8.

For the real-time implementation of the sampling rate conversion, it was not feasible to set different sampling rates for both ADC and DAC. To overcome this barrier, we designed a lab in which the students were required to convert the pre-stored sound data in the DSP memory to the designated 8 kHz sampling rate. Then, the converted data samples were sent to the DAC channel on the DSP board for verification. The laboratory design shown in Figure 9 is detailed in [2].

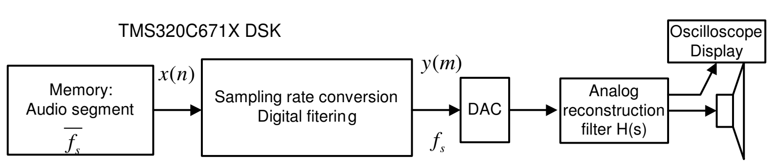



Figure 9 Laboratory setup for sample rate conversions in real time.

## C. Image Processing Basics

The third portion of the course introduced image processing techniques. Topics regarding image processing are especially beneficial to technology students in biomedical engineering. Instead of developing algorithms, we focused on teaching image processing concepts and the usage of software for basic processing techniques such as image histogram equalizations, image segmentations, image noise filtering, pseudo color generation, and JPEG image compression. Experiments 9-10, which are listed in Table 3, have an emphasis on using the MATLAB tool instead of the algorithm design.

Table 3 List of Experiments in Part 3.

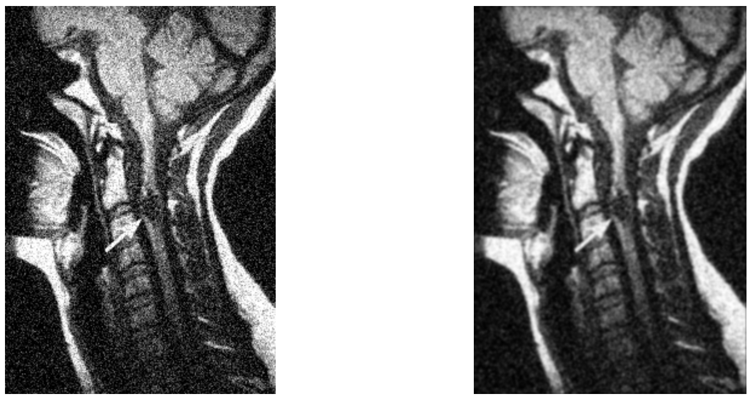
| | |
|---|---|
| Experiment 9 | Image conversion and enhancement using MATLAB |
| Experiment 10 | Image edge detection and pseudo color generation using MATLAB |
| Experiment 11 | Quality effects on JPEG image compression using MATLAB |

Experiment 9 first required students to familiarize themselves with the various formats of color image representation and their conversions, and then to perform the image enhancement. The lectures were based upon developing concepts and illustrations via simplified numerical examples followed by the MATLAB simulations (see Chapter 13 [1]). An example of image enhancement using histogram equalization is shown in Figure 10. As shown Figure 10a, the original RGB color outdoor scene was taken under light exposure. To enhance the image to have a greater contrast, the students first converted the RGB (red, green, and blue components) color image to the YIQ (luminance channel and chrominance channels) format, and then equalized the luminance component Y (which contains 93% of signal energy) only. Notice that the color components I and Q were unchanged and left as they were. Next, the students repacked the Y channel back to the YIQ format, and converted the equalized YIQ format back to its RGB color format as displayed in Figure 10b. The indexed-color image equalization and enhancement of the color image by equalizing each RGB color channel were conducted, respectively, and their corresponding equalized effects were verified.

a. Original RGB color image b. Equalized RGB color image

Figure 10 Image histogram equalization.

Figure 11 shows another result from Experiment 9 in which a noisy image was enhanced using a Gaussian lowpass filter. The students obtained the filter kernel from the MATLAB image processing toolbox which included the average kernels, and performed image filtering with various parameters and kernel sizes. Figure 11b shows the enhanced image, which appears less noisy as compared to the original one. Furthermore, the students also applied the median filter for removing the “pepper and salt” noise (impulse noise) in the corrupted image and verified the enhanced images based on the selected median filter kernel.



a. Noisy image of a human neck                    b. Enhanced image

Figure 11 Noise filtering using a Gaussian lowpass filter.

Figure 12 shows an example from Experiment 10 listed in Table 3 for image edge detection using the Laplacian of Gaussian filter, whose kernel could be obtained from the MATLAB image processing toolbox. Image edge detection is often a requirement for applications in pattern recognition, fingerprints, and iris biometric identifications. Besides using the Laplacian of Gaussian filter, the students also experimented and contrasted the other filter convolution kernels such as the horizontal Sobel, the vertical Sobel, and the Laplacian edge detectors. To increase the accuracy of object detection, the students were required to demonstrate the pseudo color generation of a grayscale image in which they chose the sine functions for the RGB color transformations and changed the phase and period of each sine function in order to produce the specified color for the object (red, for example) and the specified color for the background color (blue). Figure 13 displays the enhanced pseudo color image of a human neck, in which there is a red object (pseudo color) indicated by an arrow symbol. The detailed MATLAB sample programs for Experiments 9 and 10 can be found in Chapter 13 [1].


a. Original image b. Segmented image

Figure 12 Image edge detection.

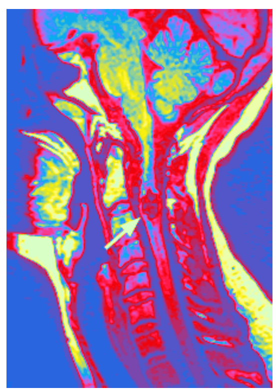
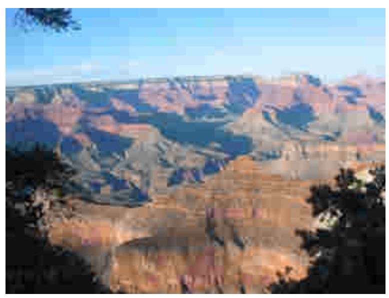


Figure 13 The pseudo-color enhanced image.

Lastly, in Experiment 11, the students manipulated the JPEG image compression used in their daily lives. Our lectures explained the JPEG compression procedure which included the RGB to the YIQ transformation, the two-dimensional discrete-cosine transform (2D-DCT), the quantization of DCT coefficients, zig-zag ordering, differential PCM (DPCM), run-length coding, entropy coding, and in addition, the JPEG decompression process. The students then performed image compression, and examined the recovered image quality using the given MATLAB program [1] with selected quality factors. Figure 14 shows typical JPEG image compression results.



Original image Quality factor =1 (best quality)





Quality factor =4 (quality reduced by 4) Quality factor=8 (quality reduced by 8)

Figure 14 JPEG image compression controlled by the quality factor.

As shown in Figure 14, increasing the quality factor indicates that a lesser number of bits is allocated for coding each DCT coefficient in the JPEG compression scheme, and hence, less data storage will be required. However, the images with a lower quality are produced due to the lessened accuracy of the recovered DCT coefficients.

**D. Hands-on Course Projects**

After completing the second portion of the course, the students were divided into pairs to develop their selected DSP projects and generate their design reports. During the developing stage, we helped the students to select their project topics and also offered them the necessary advisement. Finally, each group demonstrated their completed projects to the entire class. The successfully implemented projects are as listed below:

1. Real-time speech enhancement for hearing aids
2. Real-time digital speech and audio equalizer
3. Real-time 60 Hz Hum eliminator with harmonics
4. Digital audio crossover

5. Real-time line enhancement using adaptive filtering
6. DTMF tone generator and DTMF tone decoder
7. Multi-rate signal processing of audio signal
8. Medical image enhancement

## 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 4 indicates survey results collected from the previous three semesters from a total number of 42 students. Note that the rating scale in Table 4 was based on the percentage of the overall students.

Table 4 Student Survey for achievements.

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

Most of the students remained excited about the course, since the hands-on real-time laboratories had motivated them. Some students felt that the CCS tool required more effort for which to become familiarized. The textbook also helped a great deal to develop concepts using the worked numerical examples and MATLAB simulation examples.

After learning advanced DSP, the technology students had applied their newly gained knowledge and skills to their senior capstone 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. Some DSP related senior projects are included below. Notice that there are 3 EET (electrical engineering technology), 2 CET (computer engineering technology), and 3 BMET (biomedical engineering technology) projects.

1. Voice controlled digital audio equalizer (EET)
2. Voice controlled wireless music player (EET)
3. Smart house with speech recognition system (EET)
4. Robot controlled using DTMF tones (CET)
5. Security system using fingerprint identification (CET)
6. Health monitoring system with Electrocardiagraphy (ECG) and Electroencephalogram (EEG) (BMET))
7. EEG signal processing and its applications (BMET)
8. Robot arm control using Electromyography (EMG ) signal processing (BMET)

We summarized the results of senior project evaluations in Table 2, in which the rating scale is in terms of the percentage each of these eight projects received.

Table 5 Evaluations of DSP related senior projects.

| Rating scale | Project performance |
|---|---|
| 4 - excellent | 75% |
| 3 - good | 25% |
| 2 - fair | 0.0% |
| 1 - unsatisfactory | 0.0% |

As shown in Table 5, 75% of the DSP related projects achieved 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.

## V. Future Improvement

Based on our experiences from teaching advanced DSP, 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 adaptive filtering, waveform coding and multi-rate signal processing in Portion 2 seemed challenging to our technology students due to the demand of their math proficiency to understand certain subjects, however, we have successfully delivered the course materials with an emphasis on principles and hands-on applications instead of theoretical development. In this regard, and based on the DSP industrial trend, we could improve the course by introducing additional topics such as subband coding and wavelet coding (as well as its applications). Portion 3 serves as a good introduction to digital image processing, in which the students can explore more applications and develop more enthusiasm in this field. As a result, the students could be ready to take the image, video, and multimedia processing courses if they are determined to pursue a career in this field. 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 the engineering technology students to possess a working knowledge of the advanced 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 well-designed 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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