Hands on Experiments to Instill a Desire to Learn and Appreciate Digital Signal Processing

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Introduction

The field of digital signal processing (DSP) is rapidly changing as new development software along with advancements in hardware technologies has enabled the adaptation of these processors in diverse fields. Today DSP processors, specialized single board computers that are designed to efficiently handle computationally intensive signal processing algorithms, can be found in almost every consumer electronic gadgets. Thus, one can argue that the field of DSP is now a mainstream field within the Electrical Engineering discipline, and thus, it can be argued that all undergraduate electrical engineering students should be exposed to this field to gain a solid understanding of the fundamental issues.

Currently our curriculum lacks this exposure. The only DSP course that we offer is not a required course and is tailored to senior/graduate level students. In order to alleviate this shortcoming a set of simple yet interesting and challenging experiments have been developed for the junior level laboratory course, Intermediate Laboratory, which is a required one-semester hour laboratory course in our department. The experiments are designed such that they will not require *a priori* knowledge of the mathematical foundations of digital signal processing. These experiments are intended to generate interest in the mathematical aspects of sampling, filtering, and reconstruction, which is covered in depth in our senior/graduate level elective course, Digital Signal Processing.

These four laboratory experiments have been made available to select undergraduate students during the development stages. The experiments will be fully integrated with our junior level Laboratory course in the next academic year. For the sake of brevity the highlights of the four laboratory experiments are provided and a more detailed account of these hands on experiments will be provided during the presentation using visual aides.

Introductory Lab

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The objective of the first introductory lab is to provide a broad overview of Digital Signal Processing (DSP). Some of the important aspects of DSP are first presented through a 1.5 hour lecture. A short historical overview of the field is provided to emphasize to the students that DSP is a branch of electrical engineering that is concerned with the use of discrete time signals. A brief overview of signal classifications is also presented to highlight the differences between analog and digital signals. The students are then introduced to the process of Analog-to-Digital (A/D) and Digital-to-Analog (D/A) conversions. The lecture thus emphasizes on the fact that most real world signals are analog in nature and the proper conversion of these to digital signals is essential for successfully manipulating the signals in a desired manner. An overview of DSP algorithms for signal manipulation is then provided to conclude the lecture aspect of this introductory lab.

The remaining 1.5 hour of the lab time is devoted to providing an overview of the hardware as well as software packages that will be used in addition to walking the student through a simple experiment of capturing a signal and playing it back. The hardware setup consists of a personal computer connected to a Texas Instrument's DSP Starter Kit (DSK), namely the TMS320C6711 DSK [1], in addition to speakers and a microphone. Texas Instrument provides an integrated software development package, named Code Composer Studio, for this hardware platform to write, debug, and download the code, normally written in assembly (or C). The students are not required to develop the code but rather become familiarized with this hardware/software platform, which is used in the next three follow-up experiments.

Spectrum Analysis experiment

At the beginning of this lab period a brief lecture is given to impress upon the students the importance of sinusoidal signals in DSP. It is shown that all periodic signals can be constructed by summing up several harmonically related sinusoids with appropriate amplitudes. These amplitudes are said to be the weighting that is given to the particular frequency of that portion of the signal, which is also called the frequency content. A useful way to determine the frequency content is to perform a Fourier transform on the signal [2]. The use of MATLAB [3] software to determine frequency content of given signal is then demonstrated to the students and followed by the interpretation of the resulting plots in MATLAB workspace (a MATLAB M file is provided to student for their future use). The students are then to perform a spectrum analysis of a voice signal as follows.

Using the setup of the first laboratory experiment, students are required to capture (record, sample) the voice signal corresponding to the sentence "The quick brown fox jumped over the three lazy dogs". This sentence has all the letters in the English language contained within it, and provides variety for analysis of voice data. The next task involves processing of the digital data via MATLAB. The digitally recorded signal is first saved in the PC hard drive for post processing. The students are then to undertake the following tasks and explore the outcomes:

1-Play the recorded digital signal back with MATLAB and make observation as how the playback sound compared to the non-recorded sentence.

2-Using MATLAB perform a Fourier analysis on the signal (that is run the m-file provided in lecture) and analyze the graph of frequency content that is produced to determine the frequencies found in the signal.
3-Which components are dominant, and which are less dominant?

Filtering

For this laboratory experiment a brief lecture at the beginning of the lab period exposes the students to different types of filters (digital/analog, FIR/IIR, lowpass, highpass, bandpass, and bandreject). The concept of digital filters is emphasized as defined by J.F. Kaiser [4] " … computational process or algorithm by which a sampled signal or sequence of numbers (acting as the input) is transformed into a second sequence of numbers termed the output signal. The computational process may be that of lowpass filtering (smoothing), bandpass filtering, interpolation, the generation of derivatives, etc." MATLAB is used to show how one can design a digital filter and a specific notch filter is designed which is used for the subsequent implementation by students.

After the lecture students are asked to implement a notch filter to explore the effect of noise on a voice signal. First they are to record a simple sentence with the DSK using the techniques that they learned in the prior lab. They are then instructed to corrupt the signal by superimposing a pre-recorded 2-kHz cosine tone. They are then to replay the uncorrupted and corrupted signals and compare them. The next task requires the students to implement the notch filter on the hardware platform to clean out the corrupted signal. This essentially entails the implementation of a difference equation with the coefficients as those obtained via MATLAB. The students are asked to listen to the filtered signal and try to compare it with the original signal, as well as the corrupted one. They are also to determine and compare the frequency contents of these three signals, which should be included in their lab report.

Sampling and Reconstruction

The final laboratory experiment deals with the concept of sampling and reconstruction. Although in the previous experiments students actually sampled and reconstructed signals, they had not been exposed to underlying foundations. Thus, the lecture portion of this lab introduces the student to the concept of Nyquist Sampling Criterion. The knowledge that they have gained in determining the frequency content of a signal is tied to the choice of sampling frequency; sampling frequency, f_s, must be greater than twice the maximum frequency content, $f_{\max}$, of the signal. This will ensure that the signal can be reconstructed without the occurrence of aliasing. Aliasing is a distortion in a signal that is caused when higher frequency components of the signal overlap with the lower frequency components due to undersampling. When the signal is reconstructed, the higher frequencies are subsumed into the lower ones [5]. The typical voice frequency band ranges from about 300 Hz to 3.3 kHz. Thus a reasonable estimate on the upper limit on the frequency is 4 kHz. The effect of sampling frequency is then to be experimentally observed as follows.

The effect of undersampling is explored by first recording a simple sentence using the technique learned in previous labs. The students are asked to use a sufficiently high sampling frequency (10 kHz) to ensure that the signal is appropriately sampled since this signal is used as the benchmark for subsequent comparisons. Next the students are required to use the knowledge gained in the previous three labs to determine the frequency content of the signal with the aim of determining the maximum frequency content of the signal. They are then to determine the lower limit of sampling frequency. Now the exercise of sampling and reconstruction of the signal is to be carried out using sampling frequencies which are twice that of $f_{\max}$, 1.5 time $f_{\max}$, and 10 times $f_{\max}$. They are finally required to compare the reconstructed signals with the benchmark signal and make observation to the quality of reconstructed signals and compare that with predicted results based on Nyquist Sampling Criterion.

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Biography

Jay Adams is a graduate student in the Department of Electrical & Computer Engineering at Youngstown State University pursuing a Master of Science in Electrical Engineering. He is interested in the area of Control Systems and is planning to pursue a PhD study with a control system concentration upon graduation from Youngstown State University.

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