SIGNAL PROCESSING EDUCATION AT QUEENSLAND UNIVERSITY OF TECHNOLOGY AUSTRALIA

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ABSTRACT

The School of Electrical and Electronic Systems Engineering at Queensland University of Technology (QUT) has 3 Bachelor's degree programs in Electrical and Computer Engineering. In all these programs there is a strong emphasis on the area of Signal Processing. The school also conducts a MEngSc coursework program in Signal Processing. In addition, challenging research programs in the area of Signal Processing leading to Masters and PhD research degrees are also offered. The teaching of Signal Processing at QUT is strongly influenced by the presence of a University centre within the School called the Signal Processing Research Centre. The centre has 10 academics with a PhD in the area of Signal Processing. This paper describes the unique nature of the undergraduate and postgraduate courses in Signal Processing, the laboratories that have been developed to support these courses and the future directions of Signal Processing education at QUT.

1. INTRODUCTION

The School of Electrical & Electronic Systems Engineering of Queensland University of Technology (like many other universities around the world) has recognised the importance of Signal Processing education in the undergraduate curricula. The 3 Bachelor's programs in Electrical and Computer Engineering offered at the school all have a strong Signal Processing content. In addition there is a Masters by Coursework program in Signal Processing. A university centre established within the School called the Signal Processing Research Centre strongly influences the undergraduate and postgraduate signal processing education in the school. The Centre has 10 academic staff members, all of whom hold a PhD degree in Signal Processing and is headed by a Professor in Signal Processing. The mission of the centre is to bring to the community the benefits of research, teaching and scholarship in the area of Signal Processing.

2. SIGNAL PROCESSING IN THE POSTGRADUATE COURSES :-

Postgraduate studies in the Signal Processing area can be undertaken as a coursework program or a research program. The research program consists of a Masters program leading to a MEng degree in the signal processing area or a doctoral program leading to a PhD degree in the area. Usually students take 2 years for the Masters and 4 years to complete the PhD.

The entry requirements of postgraduate studies is a grade point average of 5.8 out of 7 or higher. Students who don't meet this entry requirement but are otherwise motivated to pursue research studies have an upgrade program available to them. They do this by enrolling in a MEngSc coursework program first. On completing this program with a score of 5 out of 7 or higher, they become eligible to enter the research degrees. Overseas applicants are assessed on a case by case basis.

The MEng and PhD programs are supervised by the 10 academics in the Signal Processing Research Centre. Currently there are 26 PhD students and 6 MEng students conducting research in the centre. The centre has graduated 5 PhD students and 1 Masters degree student during the last 3 years.

The postgraduate studies are conducted in three major research laboratories in the centre. These are the :

1. Signal Theory Laboratory
2. Speech Research laboratory
3. Image Research Laboratory

Each laboratory is equipped with state of the art research facilities to undertake world class research in the respective areas. Following are titles of some of the major postgraduate research programs currently undertaken by the students in these three laboratories:

1. **SIGNAL THEORY LABORATORY**

* *Time frequency analysis*
* *Nonlinear system identification*
* *Knock detection in spark ignition engines*
* *Signal detection and parameter estimation in high frequency radar*
* *Biomedical signal processing*
* *Time-varying higher-order spectral analysis*
* *multiscale signal analysis*
* *The bootstrap and its applications*
* *Applications of advanced time-frequency analysis to speech*
* *Time-frequency peak filtering*
* *Signal processing in mobile communication*

2. **SPEECH RESEARCH LABORATORY**

* *Speech enhancement (single microphone)*
* *Speech enhancement (multi-microphone)*
* *Speaker identification*
* *Speech recording authentication*
* *Robust speaker recognition in adverse environments*
* *Speaker verification*
* *Variable bit-rate speech coding*
* *Audio compression*

3. **IMAGE RESEARCH LABORATORY**

* *Object recognition using wavelet transforms*
* *Use of superquadrics in image modelling and object recognition*
* *Computer vision*
* *Scale space object modelling*

The Signal Processing Research Centre has imposed several requirements on PhD candidature. These include the following:

1. Each student must undertake a subject called "Advanced Information Retrieval Skills" which enables the student to get familiar with the facilities available in the library for literature search.

2. Each student must undertake 3 subjects at postgraduate level which are relevant to his area of research.

3. Each student must give at least one formal seminar each year in the centre, in addition to conference presentations and other informal seminars.

4. The student should aim to publish at least three international conference papers and one refereed international journal paper during the candidature.

5. At 18 months after commencement, the student is required to present a confirmation seminar, after which time the committee decides whether they continue the PhD program, discontinue their studies or transfer to a Masters program.

3. SIGNAL PROCESSING IN UNDERGRADUATE COURSES :-

In the undergraduate courses the students study 4 core subjects in Signal Processing:-

Signals and Linear Systems - 2nd year - 1st semester.
Signal Processing - 3rd year - 2nd semester.
Digital Signal Processing - 4th year - 1st semester
Signal Computing and Real Time DSP - 4th year - 2nd semester

In addition, there is one elective in the area:

Signal Filtering and Estimation - 4th year - 2nd semester

The first subject, Signals and Linear Systems is introduced in the second year. This is followed by a subject called Signal Processing in the third year. The aim of these subjects is to introduce random signal theory as well as lay the foundation for a treatment of digital signal processing. The third subject "digital signal processing" covers in depth techniques such digital filtering and fast fourier transforms.

These first three subjects have a strong mathematical content providing plenty of intuition about the mathematical properties of algorithms but little about applications. The fourth subject in the above list has been added recently. The aims of this subject are to complement the theory with computer based experiments and demonstrate the concepts of signal processing through applications. The subject has been named Signal Computing and Real Time Digital Signal Processing. The subject is intensively laboratory based. A new laboratory has been specifically developed for this subject to provide a "hands-on" approach to the teaching of signal processing techniques. The motivation for the development of this laboratory was the cliche "What I hear I remember but what I do I understand". The laboratory provides practical training to approximately 150 final year undergraduate students each year. More will be said about this subject and the undergraduate teaching laboratory later.

The elective subject Signal Filtering and Estimation has been specially designed to attract high calibre undergraduate students to research. The subject has a unique blend of theory and applications of signal processing.

The mathematical foundation for undertaking signal processing subjects is provided to the students by 6 maths subjects run by the School of Mathematics. These subjects which run every semester for the first 3 years of the course are prerequisites for the Signal Processing subjects

4. THE UNDERGRADUATE SIGNAL PROCESSING LABORATORY :-

The aim of the undergraduate signal processing laboratory is to complement the teaching of signal processing with computer based experiments. The laboratory also serves to provide "hands-on" approach to the teaching of signal processing techniques. One of the unique features of this laboratory is that the basic Digital Signal Processing techniques are taught using real world signals such as speech and images. This makes the experiments interesting to the students. The laboratories form a part of the subject "Signal Computing and Real Time DSP. This subject is divided into 4 modules and each module has a number of experiments which are performed in the laboratory. For each module, five lectures of 90 minutes duration are given to introduce the module as well as fill in concepts that were not covered in the previous three signal processing subjects.

The laboratory consists of a number of PC486 based work stations. The 4 modules taught in the laboratory consist of the following :-

SIGNAL THEORY MODULE :- This module is aimed at developing a deep insight into the concepts of signal processing. Understanding of the concepts of convolution, correlation, IIR and FIR digital filters, Fast Fourier transforms etc. are introduced using two PC based packages: Matlab and Hypersignal.

SPEECH PROCESSING MODULE :- Experiments in this module were done using Hypersignal and Matlab with speech acquisition and play back facilities. The students learn basic speech processing techniques and its applications to speech coding, speech recognition, speaker verification and identification.

IMAGE PROCESSING MODULE :- Experiments in this module are done using Hypersignal Windows and Matlab with the image library. The experiments include two dimensional filtering, edge detection, image enhancement and coding. The hardware used includes a camera, frame grabber and colour printer.

REAL TIME DSP MODULE :- This module is based on the TMS320C30 EVM board on a PC equipped with C compiler, assembler, linker, and simulator. The students learn how to implement real time DSP systems.

SIGNAL PROCESSING SOFTWARE :- Two world standard signal processing software have been chosen for the experiments in the laboratory. These are Hypersignal Block diagram and Matlab.

Hypersignal Block Diagram is a window-based visually-programmed object-oriented simulation package. By arranging the icons, signal processing algorithms can be designed and tested. The icons represent input functions, processing functions, output functions and display functions. The most interesting aspect of Hypersignal which makes it an ideal teaching aid is that students can create their own functions using a C-compiler. The block diagram approach enables students to quickly build signal processing systems and test them using input signals and also interactively change and analyse and graphically observe the results. Hypersignal provides a number of standard library blocks such as FFT, convolution, etc.

As a part of their assignments the students have built additional blocks to extend this library. Several of the new blocks can be used by other students to build various signal processing tasks. The blocks that have been designed by students include: (1) Adaptive filter block using Widrow's Least Means Squares algorithm, (2) Discrete Cosine Transform Block, and (3) Discrete Hartley Transform Block. Feedback from the students indicates that the block oriented approach of implementing and analysing signal processing functions improves their understanding of basic signal processing concepts and helps them to remember the relevant signal processing techniques.

The second software package that is used in the laboratory is Matlab which is already in widespread use in both academia and industry. The main way in which we have used Matlab is as a problem solving tool. Matlab provides an effective as well as an efficient tool for the students to develop problem solving skills. The students are provided with a number of problems in linear systems theory and are then asked to solve these using Matlab. Many of these problems can be solved with pencil and paper but the advantages of Matlab are its high speed mathematical calculations, high speed interactive graphics and simple programmability.

5. EXAMPLE OF EXPERIMENTS :-

SIGNAL THEORY EXPERIMENTS :-

One of the assignments which the students performed was to use FFT techniques to implement digital filters. The relevant concepts such as linear convolution, circular convolution, zero-padding and overlap and add techniques are all taught in the Signal Processing theory subject. The assignment gives the student the opportunity to recall the theory and test it with real signals.

Another example is the study of adaptive filtering. In this assignment the students use an adaptive filtering block which has been designed using Widrow's algorithm for tap update. The parameters which can be changed are the step size mu and the number of taps in the filter. The students found the "hands-on" experience with adaptive filters extremely useful. It is possible to apply a wide variety of synthetic and real signals to the filter and study the effect on convergence rate. The laboratory also helps to establish the connection between eigenvalue spread and peakedness of the spectrum and on the whole provides the students with a clear and more intuitive understanding of the adaptive filtering concepts.

SPEECH EXPERIMENTS :-

The experiments in the speech module are designed to illustrate the applications of signal processing in the area of speech technology. Linear predictive analysis and cepstral analysis are introduced in the lectures. A overview is given of major speech technology areas such as speech coding, speech recognition, speaker identification etc. The students then perform experiments gain a more intuitive understanding of speech processing techniques.

An example of an experiment is the study of deconvolution using homomorphic processing. This is demonstrated using cepstral analysis of speech. The real cepstrum defined as the inverse Fourier Transform of the Log of the magnitude of the Fourier transform of a signal, can be implemented using Hypersignal Block diagram. By applying speech as the input signal the students estimate the pitch and formant frequencies of speech. The experiment enables the students to understand the concepts of devolution and cepstral processing. They also gain a basic understanding of the speech production mechanism. We noticed that the students were excited about seeing a real world application to the concept of deconvolution using cepstral techniques.

IMAGE PROCESSING EXPERIMENTS :-

The image processing module uses the Hypersignal Block Diagram, Matlab Image processing tool box and Image Libraries for student assignments. This library includes blocks for some common image processing operations such as image file read/write operations, unary and binary logic/arithmetic operations, histogram computation and plotting, linear and nonlinear filters, edge detectors, and some morphological operations. Additional blocks can be developed such as custom blocks for fast 2D-DCT computation. The students were given a 90 minute lecture for each week for five weeks and this was followed by a laboratory session during which they "taught themselves" the basics and the details of various image processing concepts by "doing" experiments with the image library blocks.

The assignment comprised of three sections :- histograms and image enhancement,filtering, and edge detection. A fourth section on image coding was made optional and the students were asked to experimentally investigate the relationship between image compression, histograms and coding redundancy. The first three sections comprised of several small tasks, each designed to emphasise a certain concept or group of concepts.

For example, in order to learn about a concept such as contrast, the students had to examine an image and its histogram simultaneously both before and after multiplying each pixel value by a constant. The value of this constant was varied to shrink and stretch the image in terms of contrast. For learning about edge detectors they would compare the output from various edge detection blocks and check for direction sensitivity, location of edges, etc.

This technique was appreciated by the students who enjoyed the laboratory sessions and their laboratory reports demonstrated that they grasped the concepts quite well because they could "do" these operations while they were being made to think about the underlying concepts, rather than observing them in a textbook or during a lecture.

REAL TIME DSP EXPERIMENTS :-

This module is introduced to enable students to gain "hands on" experience with modern DSP devices. The Texas Instruments TMS320C30 was chosen as it represents the state of the art in floating point DSP technology.

The laboratory work in this module began with familiarising the students with the TMS320C30 EVM hardware and software they would be using throughout the semester. This was accomplished with a series of five practicals of increasing difficulty that the students would complete in their own time. The C30 code for each practical was given to the students. Assessment of each practical involved a brief demonstration and interview and completion of all practicals was made compulsory to ensure they were all attempted. A brief description of each practical is given below.

Introduction to the TMS320C30 Tools. Setting up directories and environment variables. Running the C compiler, assembler, and simulator using a matrix multiplication program as an example. Further use of the EVM board was demonstrated by designing an FIR filter using Hypersignal and running the generated code in real time.

Host Communication and AIC Access. Demonstrates the use of the communications channel between the PC and the EVM. Example programs involve using the EVM as a co-processor to the PC as well as a keyboard controlled sine wave generator.

Real Time FFT Examples. Real time FFT programs of size 8 and 512 are provided for the students to compile and run. The output spectrum is viewed on an oscilloscope. The programs demonstrate the use of double buffering and interrupt driven sampling.

Adaptive Filtering using the TMS320C30. Adaptive filter C programs are provided for the students to compile and run on the Simulator. Techniques such as noise removal, filter order effects, learning speed, and prediction are demonstrated. The students are asked to provide example time and frequency plots for each stage of the practical to include with their notes.

Real Time Adaptive Filtering. Adaptive modelling and equalisation are demonstrated. The students are asked to adaptively equalise and model a band pass filter channel that was provided and convolve the resultant impulse responses to gauge the performance. The software allows the student to interactively alter mu, filter taps, block delay, and the filter being modelled and see the results instantly.

The students were then asked to try their hand at writing their own real time programs. They were split into groups of two or three and given the choice of a number of different programming assignments. A selection of these is shown below.

Cepstral Processing. In this assignment the students are asked to implement a realtime version of the cepstrum. Difficult algorithms such as the logarithm and the FFT code are supplied to the students pre-written in assembly language so that they may spend more time investigating the DSP algorithm rather than the intricacies of assembly language programming. The assignment exposes the students to important real-time programming techniques such as circular double buffering and interrupt driven data acquisition. The students tested their programs by speaking into a microphone and displaying the calculated cepstrum on an oscilloscope.

Speech Time Warping. The goal of this assignment is to produce an artificial change in speech frequency by a programmed amount without changing the sample rate. The method is useful in recovering speech from a slowed down or sped up recording after digitising. The speech is stored in a large circular buffer referenced by two buffer pointers for the input and the output. Input speech is placed at the input pointer and the output pointer is moved through the buffer at the desired output rate, interpolating or decimating data as necessary.

DCT Speech Scrambler. Scrambled speech is produced by calculating the DCT of the input speech, randomly transposing the frequency components, and outputting the result after calculating the inverse DCT. The students are asked to experiment with different block sizes and transposition algorithms.

Digital Cro/Spectrum Analyser. The object of this assignment is to turn the PC into a digital oscilloscope with provision for a spectrum display. This will require two programs : one running on the C30 board sampling and buffering and/or calculating the FFT of the data, and one running on the PC accepting frames of data for graphics display.

Isolated Word Speech Recogniser. A very simple algorithm for isolated word speech recognition is the frequency bin average method. In this, the C30 is set to continuously sample into a circular buffer when waiting for a command word. When a word is detected in the buffer (using energy threshold), the beginning and end of the utterance is detected. The word is then filtered into a number of frequency bands (using FFT or FIR bandpass filters) and cut up into a number of segments. The energy in each frequency band is then averaged over the segments to form a pattern for the word.

Vocal Tract Area Function. The students write a program to display a cross section of the human vocal tract on the PC in real time. The vocal tract area function is calculated by first generating the Linear Predictive Coefficients of the input speech and converting these to area functions using an iterative function. The area function coefficients are then passed to a plotting program running on the PC.

The response to these assignments was very encouraging. The students enjoyed seeing their programs in action - a usually dynamic end result of their labours. A great deal of lateral and inventive approaches were exhibited in the solutions to the problems, exploring both the algorithmic techniques as well as real time programming techniques.

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