DEPARTMENT OF ELECTRICAL AND COMPUTER ENGINEERING

The 1995 Mississippi State University Conference on

Digital Signal Processing

What:	EE 4773/6773 Project Presentations
Where:	Simrall Auditorium, Mississippi State University
When:	December 4, 1995 — 1:00 to 4:00 PM

SUMMARY

The Department of Electrical and Computer Engineering invites you to attend a mini-conference on Digital Signal Processing, being given by students in EE 6773 — Introduction to Digital Signal Processing. Papers will be presented on a wide range of topics including speech and image processing, parallel processing, and acoustic echo cancellation.

Students will present their semester-long projects at this conference. Each group will give a 10 minute presentation, followed by 5 minutes of discussion. After the talks, each group will be available for a live-input real-time demonstration of their project. These projects account for 50% of their course grade, so critical evaluations of the projects are welcome.

Session Overview

1:00 PM — 1:10 PM: J. Picone, Introduction

1:15 PM — 1:30 PM: **N. Doss** and T. McMahon, "An Integrated Khoros and MPI System for the Development of Portable Parallel DSP Applications"

This project combines two public-domain paradigms to create a parallel software environment for DSP programming. MPI (Message-Passing Interface), a message passing system, is an evolving standard for parallel computing. Khoros is an integrated software environment for DSP. This purpose of this project is to describe and demonstrate a software design that exploits Khoros and MPI parallel libraries for the deployment of parallel DSP. The resulting system enables parallel DSP using the Khoros system for development and the MPI system for performance portability. Specifically, the new system provides MPI-based toolboxes containing data parallel modules and utilizes MPI as a means of communication between modules. Evaluation is based upon performance and portability characteristics of the integrated Khoros and MPI system.

1:30 PM — 1:45 PM: Y. Chen, L. Wang, A.M. Yusuf, and **H. Zhang**, "Noise Reduction in Laser Induced Breakdown Spectroscopy"

Noise reduction of Laser Induced Breakdown Spectroscopy system (LIBS) is presented. The noise characteristics will be investigated and a proper digital filter will be chosen to reduce the noise. The result of this scheme will be compared with that of the analog method. The result will show an improvement of the ratio of the signal over noise and reduction of the data collection time. The improvement will enhance the capability of LIBS system to perform *in situ* measurement.

1:45 PM — 2:00 PM: **X. Du**, W. Couvillion Jr., and M.H. Kiu, "Correction of Scan-Line Shifts of Digitized Video Images"

The United States Forestry Service (USFS) has several images taken from a moving airplane using a conventional camcorder, and then digitized. Either from motion of the camcorder, the digitization process, or both, some of the scan-lines in the digital image became shifted several pixels.

Several algorithms exist for enhancing digitized images. These algorithms can be applied to test images distorted in a manner similar to the USFS images. The enhanced test images can be compared to the original test images to objectively measure the amount of distortion remaining. The best algorithm for correcting the test images can be applied to the distorted forestry images.

A good algorithm for enhancing aerial images recorded and digitized using inexpensive equipment would have widespread applications in the areas of forestry, agriculture, and crowd estimation.

2:00 PM — 2:15 PM: J. Beard, **S. Given**, and B.Y. Young, "DTMF Detection Using Goertzel's Algorithm"

The goal of this project is to develop DSP software that will facilitate Dual Tone Multifrequency (DTMF) detection. The method that the group will use for DTMF detection is Goertzel's algorithm, which calculates Discrete Fourier Transforms (DFT's). The implementation of this algorithm will be written in C++. The testing of this algorithm will be done using a standard DTMF tape from MITEL and one from Bell Communications Research. Some testing may also be done using data taken by the DTMF group.

2:15 PM — 2:30 PM: C.R. Jones, R. Seelam, M.E. Weber, **S. Wilson**, "Physical Modeling of a String Instrument"

The outcome of this project will be a high fidelity physical model of a stringed instrument using flexible digital signal processing algorithms, giving the artist complete control over the spectral content of the instrument.

A model will be developed for each functional part of the instrument including the string, the resonating cavity, and the transducer which transmits the energy of the string to the cavity. Live data will be captured and used to develop these models through analysis of harmonic content and dynamics.

2:30 PM — 2:45 PM: K. Bush, A. Ganapath, **J. Trimble**, and L. Webster, "Real-Time Speech Endpoint Detection"

This proposal describes a plan to design and implement a real-time endpoint detection software package using the C++ language. Accurate endpoint detection is vital for robust speech recognition systems. In fact, it has been shown that speech recognition performance is directly related to endpoint accuracy [1]. C++ virtual functions will be used to allow run-time modification of the detection algorithms. Real-time energy-based, zero crossing, least square periodicity estimate (LSPE), and spectral information detection methods will be implemented and their performance evaluated. Performance will be measured against hand-marked data and public-domain endpoint detection software using an objective scoring algorithm.

2:45 PM — 3:00 PM: V. Allen, **M.E. Henderson**, E.S. Wheeler, and J. Williams, "Active Noise Cancellation in a Highly Reverberant Chamber"

In an increasingly noisy society methods of reducing noise are becoming more important. Noise, or unwanted sound, may be reduced in an environment by two basic means: passive noise control and active noise control (ANC). Active noise control is a method of reducing noise by canceling a sound wave with an inverted copy of itself. This process works best in a simple environment: one in which the wavelength of the noise is long in relation to the dimensions of the space. ANC has been most successful in reducing noise in ducts and headphones (essentially one dimensional problems). This project will center around applying ANC techniques to reducing the reverberation of low frequency noise in the stairwells of Simrall.

3:00 PM — 4:00 PM: Demonstrations in 414 and 434 Simrall