Thirteen Quarterly Progress Report

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Open Architecture Research Interface for Cochlear Implants

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1. Introduction

The main aim of this project is to develop a research interface platform which can be used by researchers interested in exploring new ideas to improve cochlear implant devices. This research platform includes a stimulator unit which can be used for electrical stimulation in animal studies, a recording unit for collecting evoked potentials from human subjects and a portable processor for implementing and evaluating novel speech processing algorithms after long-term use. The research platform chosen for this project is the personal digital assistant (PDA).

2. Summary of activities for the quarter

Work in this quarter focused on the refinement of the bench stimulator for animal studies. These refinements involved the inclusion of optical couplers to electronic isolate the output current outputs from other recording devices. Further optimization of the ACE implementation was also investigated based on the use of wavelet transforms. These optimizations are aimed at reducing the processing time and thus provide to researchers more "room" to develop sophisticated signal-processing algorithms in real-time. Ongoing work is focused on preparing an IDE application for our PDA speech processor.

2.1 Hardware refinements of the bench stimulator for animal studies

Hardware refinements of our first generation bipolar stimulation (BiSTM) test board focused on addressing the need for electronically isolated output current channels. This is necessary in situations where animals are simultaneously stimulated (electrically) while recording neuron activity and responses. In such scenarios, a common ground shared by both the BiSTM test board and the neural recording system may be present. The common reference creates a path for current to flow from the output of the current stimulator to the ground electrode present at the input of the neural recording system. The induced current, in turn, may be sufficiently large enough to cause muscle twitching in animals.

Modifications to the first generation BiSTM test board have been made to eliminate the common ground problem. These changes include the addition of a DC-to-DC converter with isolated inputs and outputs and numerous optical couplers (optocouplers) that effectively disconnect the reference of the current output channels from the reference of the input power supply, thereby breaking all connections between the BiSTM board and any other devices that may be connected to the animal.

Additionally, analog switches have also been placed at the outputs of the current channels. Because the bipolar current source chip (BiSTM chip) has at its outputs a compliance voltage of 5V at all times, whether active while outputting a current signal or at rest when inactive, analog switches have been added to the outputs to remove the 5V from the electrode array implant when the device is at rest. The analog switches may optionally be controlled to remain closed or to reapply the 5V while at rest if a particular application requires so.

The next generation BiSTM board, denoted as the BT-BiSTM board, where the BT designation stands for bench top, is shown in Figure 1. Amongst the various highlighted components is the 68 pin control signal connector located at the bottom-most edge of the board. The BT-BiSTM board is designed to be controlled by a desktop PC with a National Instruments PCI-6534 high speed digital output card plugged into the PCI slot. The PCI-6534 card is equipped with 32 digital signals needed to control the various input signals found on the BiSTM chip which specify parameters of the output signals such as the current output amplitude for each of eight channels as well as the direction of current flow in order to generate either a positive or negative voltage drop across electrode array elements. The timing between output pulses is also specified by use of these input signals.

Under development is a LabVIEW graphical user interface (GUI) that will allow users to specify a desired set of output waveforms with relative ease and flexibility. The GUI translates the input waveform specifications into the necessary series of digital output control signals that are generated by the NI-6534 card and then forwarded to the BT-BiSTM board. The combined BT-BiSTM board, NI-6534 digital output card, and LabVIEW GUI comprise the complete bipolar stimulator system.

Under development is the next revision of the SDIO board, denoted as the SDIO-BiSTM board, designed for chronic animal studies. The SDIO-BiSTM retains the SDIO interface and FPGA circuitry of the SDIO board, but replaces the Cochlear Ltd. HS-8 headstage interface with an 8 channel bipolar stimulator (BiSTM) chip.

The BiSTM chip, once integrated into the new board, will be controlled by the on board FPGA. Much like the BT-BiSTM board is controlled by the PCI-6534 digital output card, the BiSTM chip will be controlled by the digital output signals of the FPGA. Just as the LabVIEW GUI is used to specify the desired output waveform of the BT-BiSTM board, a similar GUI loaded on the PDA will translate the desired waveform specifications into the appropriate series of digital control signals to be generated by the FPGA.

Additional circuitry will also be included into the design for voltage translation between the FPGA digital outputs and the BiSTM digital inputs. Because the FPGA only operates at 3.3V while the BiSTM operates at 5V, additional circuitry is needed to convert the 3.3V FPGA output to the 5V necessary to control the BiSTM chip. Further details for the design of the SDIO-BiSTM board will be provided in the next quarterly report.

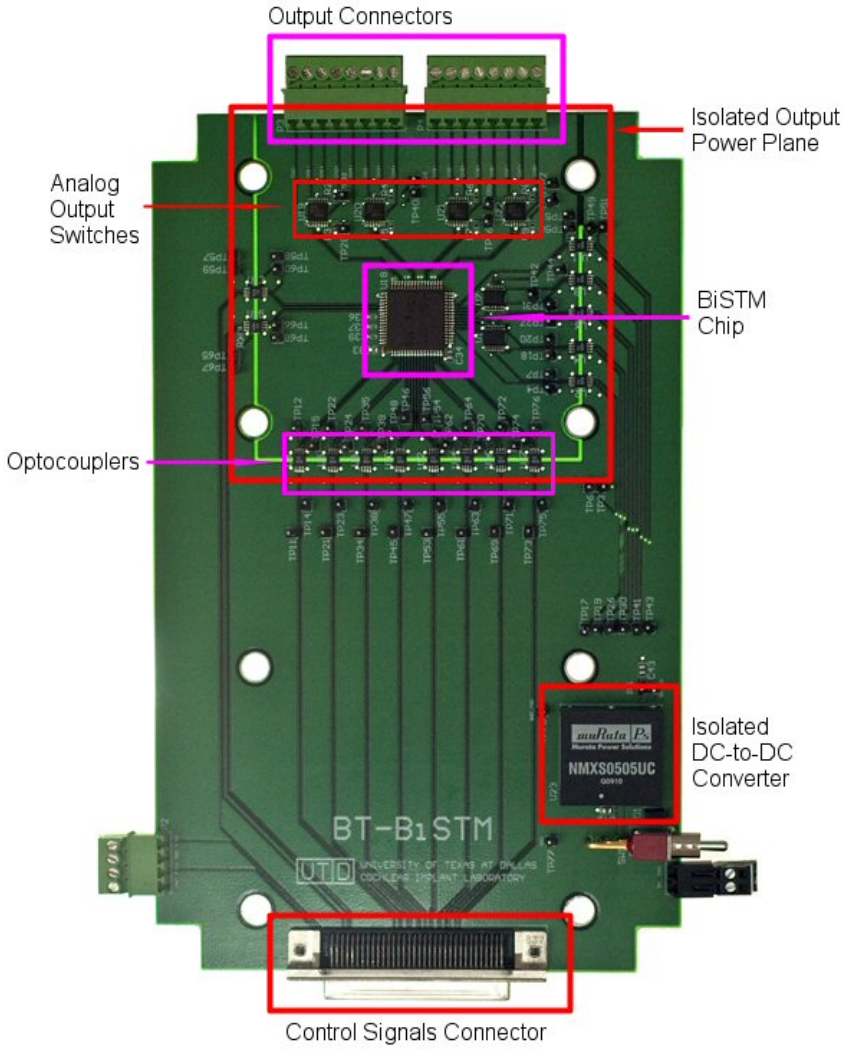

Figure 1. The BT-BiSTM stimulator board for animal studies.

2.2 Further optimization of the ACE implementation using wavelets

Emphasis was also placed in this quarter on the development of a wavelet transform based signal processing analysis method as an alternative to the filterbank or the FFT methods commonly used in the CIS and ACE strategies. The wavelet transform based analysis was implemented on the PDA platform in fixed-point using 16 and 32 bit word sizes. The method developed was analyzed using 3 different wavelet packet tree structures (Krimi *et al*., 2005). Out of the three tree structures, the binary tree structures shown in Figure 2 exhibited a channel frequency spacing similar to the ACE strategy. The developed program has an option to change the filter order. The 29^{th} order filter generated the best performance in terms of producing the lowest spectral leakage.

Performance of the above wavelet based method was compared to that of our previously developed recursive discrete Fourier transform (RDFT) (Gopalakrishna *et al*., 2009) and the conventional filterbank methods in terms of computational complexity and processing time on the PDA platform.

Table 1 shows the time required by the three methods to process a frame of 46.4 ms at a 22,050 kHz sampling frequency. Table 1 only shows the time required to compute the 22 channel outputs needed in the ACE implementation. The time required for envelope compression and maximum selection is not included as it is common to all three methods. The main advantage of using the wavelet packet for decomposition can be seen in terms of reduction of processing time (Table 1). The price paid in order to achieve this high processing speed is the reduction in the sampling rate for stimulation. Due to the downsampling done at every stage in the wavelet packet transform, the number of computations is reduced considerably in comparison with the other two methods. Table 2 shows the computational complexity of the three methods. In order to have a stimulation rate in the range of 800 to 2000 pulses per channel per second, the analysis frames need to be interpolated. The timings listed in the table also include the interpolation time.

Figures 3 and 4 show comparison of channel envelope amplitude (channels 1 and 20, respectively) computed using the RDFT and the wavelet packet tree methods for a test sentence.

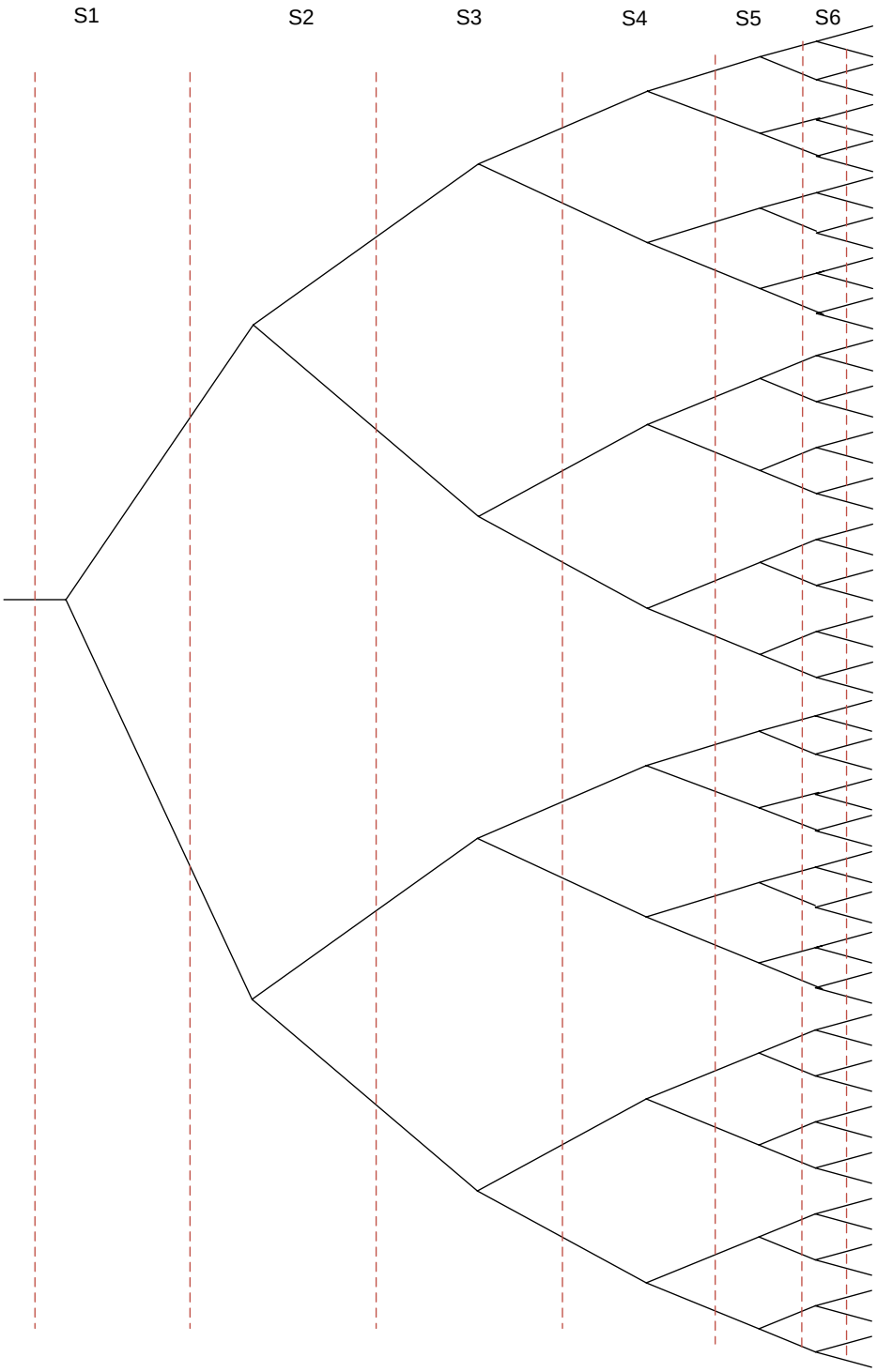

Figure2: Binary wavelet packet tree structure.

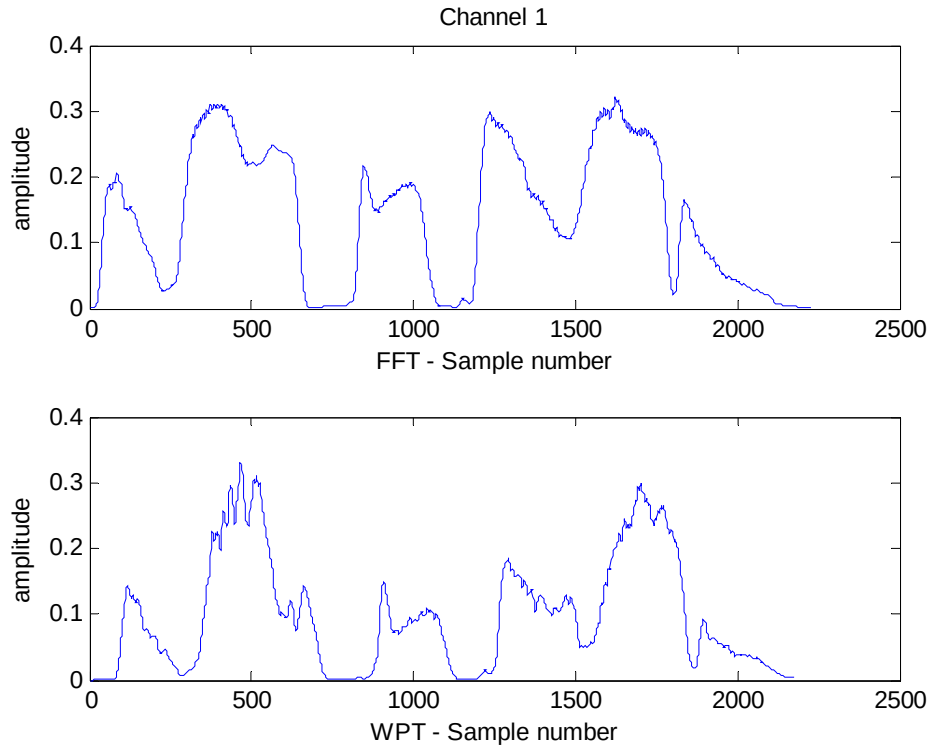

Figure 3: Channel 1 envelope output of the RDFT (top panel) and the wavelet packet tree (bottom panel) methods.

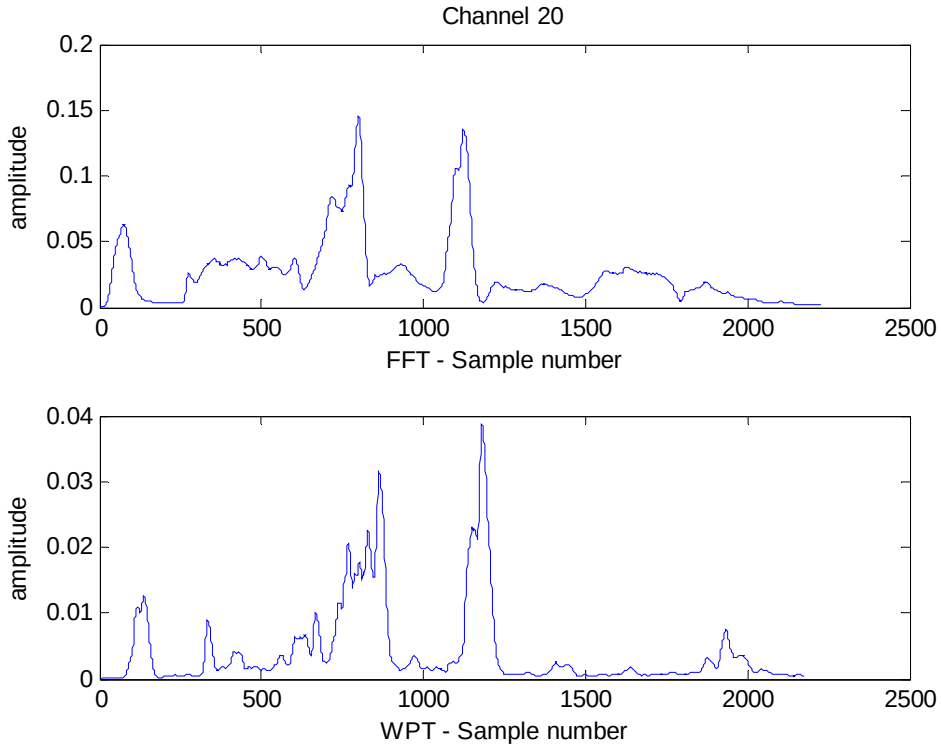

Figure 4. Channel 20 envelope output of the RDFT (top panel) and the wavelet packet tree (bottom panel) methods.

Table 1: PDA processing time using the filterbank, RDFT and Wavelet Packet methods.

Specifications: 22 channels Frame Length 1024 samples = 46.4 ms Filterbank: 6^{th} & 4^{th} order BPF & LPF FFT: 128-point Wavelet Packet 29^{th} order filter Time in ms (rounded to nearest integer)	Filterbank		RDFT Rectangular window		Wavelet Packet Daubechies/Symmlets	
	Non interactive	Interactive				
	Using C + IPP+ Assembly	Using LabVIEW + C	Using LabVIEW + IPP + C		Using LabVIEW + C	
			16 bit	32 bit	16 bit	32 bit
	6	19	8	10	4	5

Table 2: Computational complexity of the filterbank, FFT, RDFT and Wavelet Packet methods.

Number of real multiplications for 1024 sample frames, 128-point FFT, 22 channels , 11^{th} 23^{rd} and 29^{th} order WP filters	Filterbank	FFT (N-point)	Recursive DFT update	Wavelet Packet Symmlet/ Daubechies		
	No. of channels * (3 stage cascaded 2^{nd} order BPF)	$2N(\log_2 N) + (N/2)*2$	$(N/2)*4 + (N/2)*2$	FL*(12)		
	≈338K	≈2M	≈400K	≈73K (11^{th} order)	≈147K (23^{rd} order)	≈184K (29^{th} order)

References

Krimi .S, Ouni .K, and Ellouze . N. (2005). " Realization of a Psychoacoustic Model for MPEG 1 using Gammachirp Wavelet transform," *Proc. of 13th European Signal Processing Conference*.

Gopalakrishna, V., N. Kehtarnavaz, and P. Loizou (2009). "Real-time PDA-based recursive Fourier transform implementation for cochlear implant applications," *IEEE Conference on Acoustics, Speech and Signal Processing (ICASSP)*, 1333 – 1336.