A DSK BASED SIMPLIFIED SPEECH PROCESSING MODULE FOR COCHLEAR IMPLANT RESEARCH

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ABSTRACT

A simple yet flexible research module for laboratory experiments in evaluating performance of different speech processing strategies of Cochlear Implant is developed. It enables algorithm development and evaluation through a series of three sub-modules. Software Evaluation Module provides a graphical environment to the researchers to develop new strategies as well as experiment with parametric variation of these strategies. Evaluation is done by means of Waveform Analyzer and audible reconstruction of the synthesized sound. Hardware Evaluation Module evaluates real-time operational performance of the algorithms in DSP environment. It highlights the practical bottlenecks of any algorithm when transferred from software to hardware domain. Finally, Real-time Patient Evaluation Module allows performance evaluation of the developed strategy on actual Cochlear Implant patients. Simple and easy to use GUIs enable clinicians and researchers to experiment with speech processing strategies with great ease and flexibility without requiring advanced programming skills.

Index Terms **–** Biomedical applications**,** Cochlear Implant, Continuous Interleaved Sampling (CIS), Cochlear Implant Signal Processing

1. INTRODUCTION

A Cochlear Implant is an electronic prosthetic device surgically implanted into the inner ear that restores partial hearing to the profoundly deaf [1]. Unlike commercial hearing aids which benefit patients with conductive hearing loss, Cochlear Implant, on the other hand, benefits patients with sensory-neural hearing impairment. It bypasses the normal hearing mechanism and directly stimulates the inner ear sensory cells of the auditory nerve by delivering electrical signals to an electrode array implanted inside the cochlea. These electrical signals are derived from the external sound acquired from a microphone. Sound signals are first manipulated by an external speech processor and then transmitted via transcutaneous link in the form of electromagnetic waves to the inner ear where they are finally converted into electrical pulses. With high success rates and increasing demand of Implants worldwide, a substantial growth & progress is seen in the Cochlear Implant research in the last two decades.

A speech processing strategy or speech coding strategy is one of the key features which affect the overall performance of the device [1]. Speech Processing may be regarded as a functional space composed of a series of nonlinear functions which maps acoustic signals to the electric al domain such that the final electrical impulses mimic the actual impulses delivered by a healthy cochlea. Depending upon the speech processing strategy, the speech processor extracts various parameters from the acoustic signals and converts them into electrical signals. Various speech processing strategies have been developed and reported in literature [2,3,4] over time for cochlear prosthesis which include Continuous Interleaved Sampling (CIS), Spectral Peak (SPEAK), Advanced Combination Encoder (ACE), Spectral Maxima Sound Processor (SMSP), Simultaneous Analog Strategy (SAS), Paired Pulsatile Sampler (PPS), Quadruple Pulsatile Sampler (QPS) and various Formant based Strategies. Numerous algorithms based on Wavelet Transform [5], Wavelet Packets [6], Bionic Wavelet Transform [7] & Auditory Models [8] are also found in literature. In contrast to traditional approaches, various algorithms especially for tonal languages have been developed, which emphasize on the extraction of maximum tonal & pitch information from speech [9]. It should, however, be noted that performance of these strategies varies from individual to individual.

Although significant speech recognition can be achieved with commercially available multichannel CI systems [2], researchers around the world are developing new and better algorithms for enhanced speech recognition. For easy assessment & real time evaluation of these algorithms, speech processing research platform needs to be simple and flexible; qualities absent in most commercial research platforms [10]. The important factor that hinders their use for speech processing research is that a skilled programmer is required to implement the algorithm in high-level or low-level language.

This paper addresses the need for the development of a low-cost, simplified speech processing module for laboratory research in the development and evaluation of strategies & algorithms pertinent to Cochlear Implant. The proposed Research platform is based on a commercially available floating point DSP Starter kit (DSK TMS320C6713). DSK was used for the research module because it provides an efficient and stable DSP development environment and it is a robust, low-cost and easily available DSK in both universities and industry. The developed system is highly flexible and it can be used by clinicians & researchers for the investigation of novel algorithms without requiring advanced programming skills.

The proposed speech processing module consists of a series of three sub-modules: I) Software Evaluation Module, II) Hardware Evaluation Module and III) Real-time Patient Evaluation Module (Figure 1). *Software Evaluation Module* is used for the implementation & assessment of any algorithm and to analyze the effect of different parameters on the net performance. *Hardware Evaluation Module,* on the other hand, evaluates real-time speech recognition ability along with real-time operational performance of the algorithm in DSP environment. Finally, *Real-Time Patient Evaluation Module* is used for laboratory testing of the developed and optimized algorithm with Cochlear Implant patients. Paper is organized in a similar fashion. Each of the three sub-modules is discussed in detail in Section 2 followed by conclusion and recommendations.

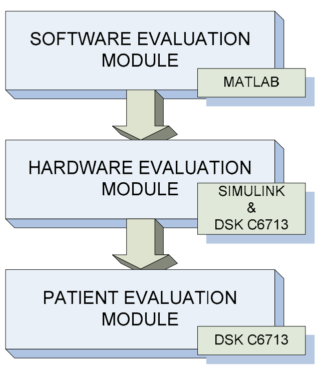

Figure 1: Speech Processing Module

2. SPEECH PROCESSING RESEARCH MODULE

2.1. Software Evaluation Module

Software Evaluation Module is a graphical environment developed in MATLAB for the algorithm development, assessment and to study the effect of different parameters on overall performance of the algorithm including speech quality of the reconstructed signal. This rich yet simple Graphical User Interface (GUI) utility enables independent parametric analysis of different speech processing strategies through high resolution waveform analyzer. Waveform analyzer enables visual representation of waveforms at different channels. Moreover, it has capability to reconstruct multichannel signals in the form of synthesized sound for the evaluation of speech quality. The developed software

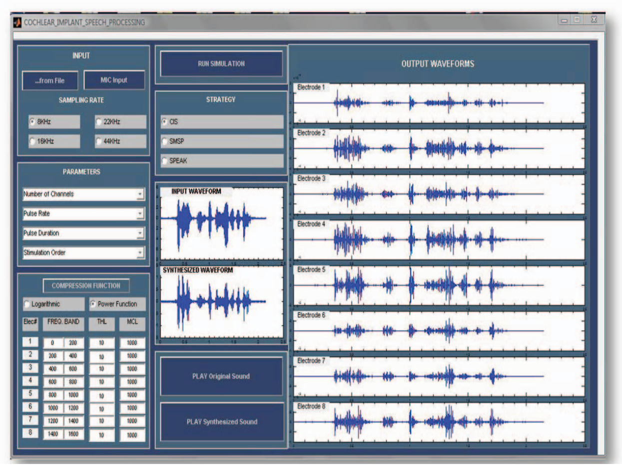

Figure 2: Snapshot of GUI – Software Evaluation Module

module is highly flexible and can easily be extended to incorporate any speech processing strategy without the need of astringent programming. Figure 2 shows a snapshot of the GUI for an eight channel CIS algorithm.

In order to demonstrate the true potential of the Software Evaluation Module, we present CIS strategy as a case study throughout this paper. A brief overview of CIS is presented first. Digitized audio signal is divided into n frequency bands through n band-pass filters. Envelope of the filtered waveforms from n channels is computed though rectification followed by a low-pass filter. For the adaptation to dynamic range of patient's Threshold level (THL) & Maximum Comfortable level (MCL), nonlinear amplitude mapping is done by a suitable compression function. Finally, each output of the n channel compression function is modulated with time multiplexed biphasic pulses. Flow diagram of the complete algorithm is shown in Figure 3.There are a number of parameters associated with the basic CIS algorithm that can be varied to optimize the speech recognition. However, it should be noted that there are no optimized standard values for these parameters as performance of any strategy varies for different parameters depending upon the patient & the Cochlear Implant system

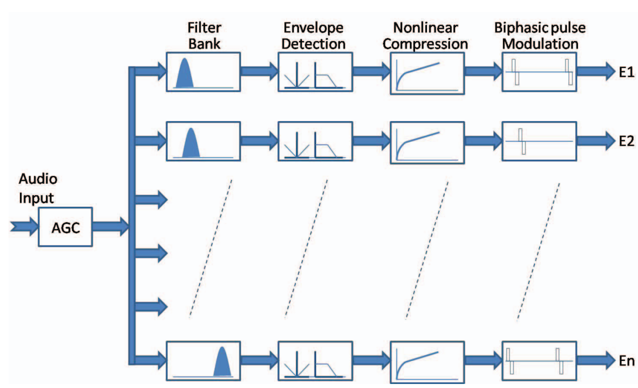

Figure 3: CIS Algorithm

under test [1, 11]. These parameters include number of channels, sequence of channel stimulation, order of bandpass & lowpass filters, cutoff frequencies of filters, type and parameters of the compression function, pulse width & Pulse duration of modulating pulses and stimulation rate. A detailed study of the effects of these parameters on the overall performance can be found in [12].

Software Evaluation Module enables full control of these variable parameters and allows us to maneuver through the Hilbert Space formed by the variability of these parameters easily and effectively in order to assess and optimize these parameters according to the requirement. Waveforms of original signal and output of different channels at different stages of the algorithm as well as synthesized signal can all be analyzed with great ease. Moreover, original & reconstructed speech signals can be listened for performance evaluation. Beside CIS, SMSP & ACE have also been implemented and other speech processing strategies can easily be incorporated into the Software Module for evaluation.

2.2. Hardware Evaluation Module

Hardware Evaluation Module is a software utility to test the algorithm's performance when it is realized in the hardware domain. Hardware Evaluation Module evaluates the real time operational performance of the algorithm in DSP environment as well as the final speech recognition ability given by any speech processing strategy in real-time. This is an important step when transferring algorithm from software to hardware domain as it reveals the actual hardware issues pertinent to the design such as computation cost of an algorithm, data rates and real-time issues which are bottleneck performance parameters of any implantable medical device. Hardware Evaluation Module was developed using SIMULINK & TMS320C6713 DSK. It utilizes MATLAB toolboxes for Filter design, Real time Workshop (RTW), MATLAB Target for TI C6000 and Code Composer Studio (CCS) applications to realize the software module on DSP.

Hardware Evaluation Module is directly linked with its parent Software Evaluation Module which provides the parametric values to the SIMULINK model/design of any strategy. SIMULINK model is then converted to Binary code through RTW and CCS. Finally the binary code is downloaded to DSK C6713 which enables real-time input & synthesized output to be listened to through line in & line out port of the DSK. Process flow of this module is illustrated in Figure (4) and the code generation routine from the SIMULINK is shown in Figure (5).

Using SIMULINK models to generate DSP code is a simple routine before actually hand coding algorithms directly in any high level or low level language as it enables quick code generation for analysis and optimization in hardware.

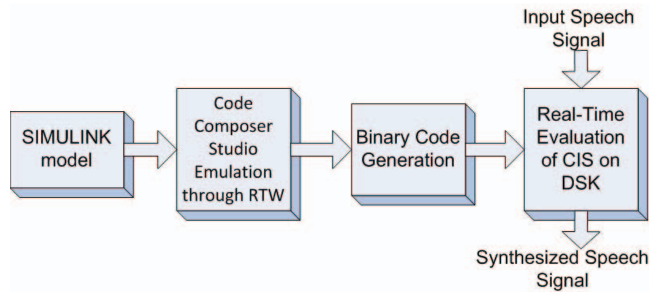

Figure 4: Process Flow - Hardware Evaluation Module

Hardware Evaluation Module helps in the real time assessment of the algorithm and to optimize the parameters that result in maximum speech recognition & functional performance in real-time DSP environment. For example: Increasing the filter order of the bandpass filters gives better band resolution, on the contrary, it adds to the computation cost and thus adversely effects the signal processing in real time. Therefore, a tradeoff needs to be developed to optimize the design. Evaluation of these issues could not be achieved through software evaluation alone, therefore Hardware Evaluation helps in identifying the practical bottlenecks of any design. Other possibilities can include issues like Channel Interaction, which can be modeled in the SIMULINK design to analyze its effect on the net performance of any strategy.

2.3. Real-time Patient Evaluation Module

Real-time Patient Evaluation Module is essentially a Digital Signal Processing Board directly linked with Software Evaluation and Hardware Evaluation Modules. Software routines of speech processing strategies are hand-coded in C. Its purpose is to finally evaluate the optimized speech processing strategy on actual Cochlear Implant patients.

Hardware routine of the module is illustrated in Figure 7. Sound is captured through Microphone Input Port of DSK. The DSK has an AIC23 audio codec connected to audio input and output ports of DSK. Multichannel serial buffered ports (MCBSP) are used to control/configure the codec and to transfer data with codec. MCBSP1 is used to

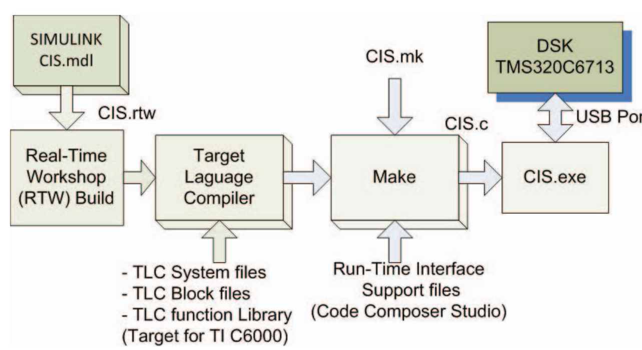

Figure 5: DSP Code generation routine from SIMULINK

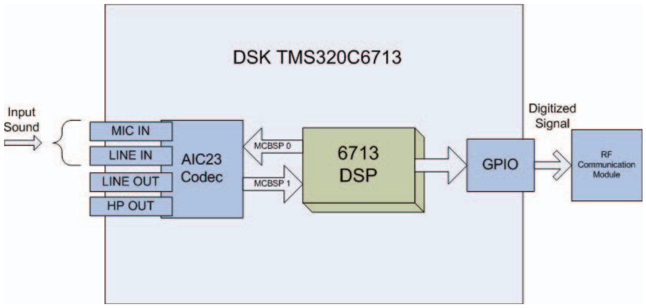

Figure 6: Hardware Routine - Patient Evaluation Module

configure & control the codec and MCBSP2 is used to for the flow of audio data from the microphone through the codec. The EDMA is configured to take audio samples in 16 bit signed integer form and store them in a buffer. Output from the *n* channels is finally converted into binary form, encoded and framed in the DSK before it is actually send to the transmitter of the Cochlear Implant system. GPIO module of the DSK is used for the digital output of framed data from DSK.

Real-time Patient Evaluation Module enables testing of developed strategies on actual Cochlear Implant Patients. It enables final evaluation of patient's response to any particular speech processing strategy as well his/her response to parametric variations of the strategy. Unlike the tuning procedure of Cochlear Implant device, it enables testing of novel algorithms easily yet effectively.

3. CONCLUSION

A DSK based Speech Processing and Analysis Module is developed for laboratory experiments with Cochlear Implant's Speech Processor. It enables easy to develop and modify different speech processing strategies and evaluate their operational performance in software & hardware as well as testing on actual cochlear implant patients. Although a few commercial platforms are available for research but they demand excellent programming skills. The developed system is not only flexible in terms of experimentation of new algorithms but also helps in identifying bottlenecks of any strategy before they are actually hand-coded. This research module would provide much ease to the algorithm developers who would be able to concentrate on better and efficient algorithm development rather than handling complex programming tasks.

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