CCi-MOBILE: A Portable Real Time Speech Processing Platform for Cochlear Implant and Hearing Research

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***Abstract*—Experimental hardware-research interfaces form a crucial role during developmental stages of any medical, signal-monitoring system as it allows researchers to test and optimize output results before perfecting the design for the actual FDA approved medical device and large-scale production. These testing platforms, intake raw signals through which performance of novel algorithms can be analyzed and modified to generate the desired data points for an optimized output, allowing the advancement of the medical device. With cochlear implants (CIs) and hearing aids (HAs) becoming a more common solution for varying degrees of hearing impairment, having modern signal processing strategies tested for such speech sensitive systems is a necessity. But the rigid design requirements of commercial CI and HA processors make it difficult to explore novel algorithms for research investigations and conducting longitudinal studies. This study presents the design, development, clinical evaluation, and applications of CCi-MOBILE, a computationally powerful signal processing testing platform built for researchers in the hearing-impaired field. The custom-made, portable research platform allows researchers to design and perform complex speech processing algorithm assessment offline and in real-time. It can be operated through user-friendly, open-source software and is compatible with implants manufactured by Cochlear Corporation. The FPGA design and hardware processing pipeline for CI stimulation is discussed followed by results from an acute study with implant users' speech intelligibility in quiet and noisy conditions. The results show a consistent level of performance compared with CI users' clinical processor, thus confirming the viability of the platform in chronic CI based studies.**

***Index Terms*—Cochlear implant (CI), ACE strategy, electrical stimulation, FPGA processing, firmware, bilateral, bi-modal.**

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I. Introduction

HARDWARE and software research interfaces have been a prominent resource in the process of biomedical research advancements as they allow the flexibility of analyzing the validity and efficiency of new ideas, be it in the form of output simulations or hardware test signals [1]. Noise suppressing, filtering, and digitization of analog speech signals for hearing based electrical systems, be it medical devices or voice recognition modules, always require a platform to inspect the signal while varying multiple parameters and modifying requirements. With HAs and CIs gaining more popularity among the hearing impaired in the late 90's and the early 2000's, it has been important to address the limitations and accelerate the progress made to these medical systems. Testing a new signal processing algorithm with refined features on a stand-alone research platform is much more feasible than evaluating it on a delicate Digital Signal processing (DSP) chip of the final product. And since commercial DSP chips on CIs/HAs are constrained by manufacturer design/production restrictions, using research platforms becomes a necessity for researchers to advance algorithms and conduct investigational studies with CI/HA subjects.

A CI is a medical device for the deaf or individuals with profound hearing loss which consists of an intracochlear, implanted electrode array, an RF transmitter/receiver, and a behind-the-ear (BTE) speech processor fitted with microphones [2], [3]. The electrode array consisting of 12-24 electrodes, is surgically implanted in the cochlea (inner most part of the ear) to mimic the functionality of the healthy hair cells in normal-hearing individuals. The array converts incoming analog sounds into electrical signals that are sent to the auditory nerve for speech perception [4], [5]. HAs for individuals with gradual hearing loss [6], use a mechanism consisting of directional microphones and a speech processor that comprehends dynamic range compression, amplification of incoming sound, speech enhancement, and binaural processing.

The initial fittings for CI/HA users known as the subject's MAP, done by audiologists, involves mapping parameters like sound level, number of channels, stimulation mode, gain, amplification, etc. These settings are performed in a clinical sound booth ensuring quiet conditions for the CI/HA user hearing audio at comfortable levels. While changes made in the sound booth give audiologists an indication of success for the user in real-life settings, these mapping sessions do not replicate the CI/HA user's day-to- day environments nor are customized to their lifestyle. Only the psychophysical mapping parameters [7] are tailor-made for each user, while multiple aspects of the speech coding algorithms and associated parameters (e.g., compression function, filter spacing) are kept default and are not optimized in any way. Moreover, for bimodal (i.e., HA in one ear and CI in the other) or bilateral users (i.e., two CIs) adjustments of sound parameters are done separately for each implant, rather than jointly. The above issue effects the ability to utilize the full capabilities of CI/ HA devices as they are not customized for different listening environments.

These limitations have therefore, served as the motivation behind the design and development of a portable research speech processor created at UT- Dallas referred to as the Costakis Cochlear Implant Mobile (CCi-MOBILE) research interface [8]. This platform allows researchers to a) change MAP parameters in real time, b) customize MAPs for different listening situations to provide maximum support/benefit, c) collect subjective feedback, d) support new algorithm development for both laboratory and field evaluations, e) support research for unilateral, bilateral, and bimodal hearing cases. The proposed platform currently works with Cochlear Corporation's CI24 implant system and Nucleus[1] series.

Already available experimental research interfaces provided by implant manufacturers and other institutions will be discussed in the following section. These research tools lacked flexibility for evaluations in real-world environments or had internal hardware design restrictions, which limited the potential impact of those systems [9], [10]. Hence CCi-MOBILE aims to overcome the limitations mentioned and contribute towards benefiting the hearing-impaired community.

II. Previous Work

Although research interfaces developed by CI manufacturers and other research institutions demonstrate rigid functionality, they provided core motivation in the development of the CCi-MOBILE. Currently, there are three major cochlear implants manufacturers- Cochlear Corporation, Advanced Bionics and Med-El. Each manufacturer has its own proprietary implant design, peripheral components, and sound coding strategies [9].

The Clarion Research Interface (CRI 1 and 2) is a research tool developed by Advanced Bionics to explore sound processing algorithms and diagnostic tools that benefit the HiResolution Bionic Ear cochlear implant users. It allows direct programmable, transcutaneous communication to the implanted electronics of the cochlear implant [11], [12]. Bionic Ear Programming System (BEPS+) is the software running on the CRI. It allows the researcher to manipulate, modify and analyze results in real-time by varying pulse width, pulse shape, grounding configurations, electrode grouping, and adjustable filter coefficients. The major drawback of the Clarion Research interface is its need to be tethered to a PC for acute laboratory experiments while testing novel speech algorithms developed, rather than allowing subjects to be evaluated in naturalistic listening conditions outside the lab [13].

[1]Nucleus series of implants is an implant system developed by Cochlear Corporation which is the largest manufacturer of CIs.

MED-El, collaborated with the Univ. of Innsbruck to create the Research Interface Box 2 (RIB2), which is compatible with MED-EL's CI models released since the Combi 40H [14]. RIB2 comprises of MED-EL's programming interface - MAX and Dynamic Link Library (DLL) linked with the graphic user interface (GUI) on a personal computer (PC). High-level programming of algorithms written in either MATLAB or Python are required to interface with the Windows DLL. Although it allows real-time testing of speech algorithms on subjects, offline streaming of data is preferred as getting the system set up for real-time is inconvenient and requires double buffering implementation. Like CRI, the RIB2 only supports lab/benchtop testing.

An interface was developed by the research group Experimental Oto-rhino-laryngology (ExpORL) at the Catholic Univ. of Leuven (KU Leuven) in 2000, called APEX, which was an acronym for Application for Psycho-Electric eXperiments [15]. The APEX system was compatible with the LAURA Cochlear Implant which was a registered trademark of Philips Hearing Implants (formerly Antwerp Bionic Systems), by the Univ. of Antwerp, in the late 1980's [16]. Experiments were controlled by a run time program Apex.exe on a Windows 95 operating system. It could only perform offline psycho-electric experiments like analyzing threshold levels, loudness growth and frequency discrimination. The LAURA CI was discontinued in the late 2000's and is no longer available commercially and as the APEX system was not compatible with any other CIs, it is included here as more of a design reference for future platform development.

To support the research with Cochlear Corporation, five research interfaces have been developed, which are discussed chronologically in accordance with the time frame of their development. First is the House Ear Institute Nucleus Research interface (HEINRI) in the early 90's [17]. House Ear Institute developed HEINRI to provide a user-friendly communication means to researchers so that they could control the tissue simulating prosthesis in a desired manner. The platform was customized to support Cochlear Corporation's Nucleus 22 series. It could deliver synchronous bilateral stimulation using two HEINRIs setup in a master and slave configuration, but as externally synchronizing the clocks of two HEINRIs was a tough task, this setup was avoided. Also, HEINRI only was compatible with the Nucleus 22, one of the oldest implants designed by Cochlear Corporation which were discontinued.

Next, Cochlear Corporation developed the Clinical Programming System (CPS) and Nucleus Implant Communicator (NIC) library, which was patented in 2004 [18], [19]. The software library runs on a PC/ tablet with a GUI that sends information to the testing interface- CPS also called Portable Programming System (PPS). Parameters such as RF cycle timing, current amplitudes, prosthesis powering requirements could now be monitored and adjusted by researchers to perform subject testing for optimizing cochlear implant algorithms programmed in universal languages like C or MATLAB [20]. Some limitations of the NIC system were that it (i) was not portable, (ii) was not suitable for real-time processing due to restricted buffer size causing higher latency when the stimulus length is greater than 100ms. (iii) required two separate CPS/PPS to be set up as master

(a) (b)

Fig. 1. **(a)** ci-PDA with 3D enclosures. **(b)** Transitioning to the CCi-MOBILE.

and slave while performing bilateral experiments, which is hard to synchronize.

SPEAR3 research sound processor, created in 2008 by the HEARing Cooperative Research Center (Hearworks Pty Ltd.) was compatible with Cochlear Corporation's Nucleus CI22 and CI24 implants [21]. The SPEAR3 provided the ability to upload new complex strategies and allowed real-time data processing unlike the previously available research platforms. As SPEAR3 was portable, it also allowed CI users to try new sound processing strategies in take home experiments and not remain confined to lab settings. In addition to that SPEAR3 could drive two Nucleus implants at the same time for bilateral stimulation using just one platform [22].

While the SPEAR3 was the ideal research platform for its time, the lack of its user-friendly parameters prevented the platform from being accepted as the optimal research tool across the research community. Although researchers had access to the digital signal processing (DSP) files that contained the implant strategies, algorithms had to be coded in assembly language, which is not universally known by speech scientists or signal processing engineers in general. This meant that researchers had to rely on hardware engineers proficient in assembly language, to translate the algorithms and then load the new assembly file for testing. This process seemed to be tedious for researchers, but the concepts implemented with the SPEAR3 contributed immensely to the development of the CCi-MOBILE for mass-acceptance.

In the early 2000s, when Personal Digital Assistants (PDAs) were considered state of the art technology for mobile computing, the CRSS-CILab at UT-Dallas designed the ci-PDA (Fig. 1(a)). The ci-PDA was built with a goal of overcoming drawbacks from previous research platforms. The ci-PDA had benefits of being (i) portable and wearable due of its light weight, (ii) flexible to be programmed in C, C++, MATLAB, JAVA, (iii) capable of performing real-time processing of incoming data in naturalistic environment, (iv) supportive of CIs, HAs and easy to implement bi-lateral stimulations [23].

The platform had a 625 MHz ARM processor which supported a multithreaded programming environment and DLL libraries, providing smooth operations under Windows mobile environment. Several research studies and clinical studies have been carried out using the ci-PDA [24]–[28]. However, soon the PDA became obsolete technology due to the advent of smartphones and flexibility offered by Android, the new mobile operating system, made it imperative to re-design the PDA based research platform to establish the CCi-MOBILE.

III. Innovations

CCi-MOBILE is a novel research tool, developed to effectively generate electrical stimulation, facilitate perception of acoustical sound, and support the CI research community towards advancing the CI/ HA technology. Analyzing all the previous research platforms, it was concluded that for an implant-algorithm testing unit to be widely acceptable within the research community, three major pre-requisites needed to be implemented.

First, the research platform should be compact, portable and perform real-time speech processing as the speech stimulation strategies may require weeks of listening experience for implant users to adapt and benefit from the process due to neural plasticity. A portable device can be employed for take-home chronic assessments in naturalistic environments as laboratory settings, do not accurately reflect real-life complex multi-tiered acoustic environments (e.g., multi-source noise, reverberation, music, speaker diversity including emotion, stress, etc.).

Second, it should have support for all forms of hearing impairment- unilateral, bilateral, and bimodal [29] users, such that the entire community can benefit from novel speech processing strategies. Subjects with unilateral hearing impairment have complete hearing loss in one ear and normal hearing in the other. Bilateral hearing impairment involves profound hearing loss in both ears, while Bimodal hearing impairment comprises of severe hearing loss in one ear and gradual hearing loss in the other (i.e., CI plus HA scenarios).

Third, the research platform should be programmed in high level languages requiring minimum computational power. Most clinicians and speech scientists find it challenging to program in low-level complex programming environment like C or assembly language. To ensure user-friendly customization of sound processing and algorithm development by a wide range of researchers, high level programming languages like MATLAB and JAVA need to be employed. With the aforementioned prerequisites and community feedback implemented during developmental stages, CCi-MOBILE stands out by being portable, wearable, and enabling real-time on-the-go signal processing adjustment support for researchers.

IV. CCi-Mobile Research Platform

The migration from SDIO-based connection interface of ci-PDA to a faster USB/Wi-Fi based communication platform for the CCi-MOBILE (Fig. 1(b)) led to a complete hardware modification of the system. The most noticeable improvements include redesign of the interface board, upgrading the on-board FPGA and android based mobile application for sound processing. The current solution addresses shortcomings of previous platforms, supports all features provided by ci-PDA and has been designed to perform a variety of experiments with the latest CIs manufactured by Cochlear Corporation.

A. Platform Overview

There are two main external parts of a cochlear implant: (i) a DSP integrated circuit (IC) with a BTE microphone, that captures audio and evaluates which implanted electrodes to

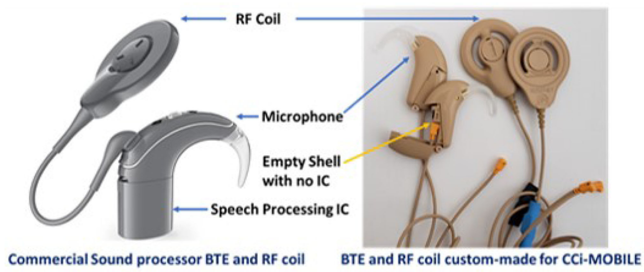

Fig. 2. Difference between the external components of a commercial Cochlear implant and the CCi-MOBILE.

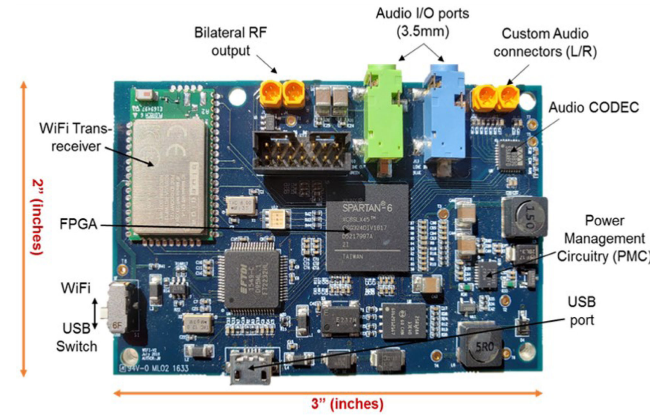

Fig. 3. The CCi-MOBILE custom-made PCB with SPARTAN-6 FPGA.

be stimulated in the ear and, (ii) a radiofrequency (RF) coil which wirelessly transmits the evaluated results to the implanted electrode array. To aid requirements of optimizing algorithms running on the DSP chip, CCi-MOBILE mimics the external system of a CI, by replacing the industry-built DSP speech processor with a custom-made Printed Circuit Board (PCB) and a reconfigurable Spartan-6 FPGA that can run any novel algorithm developed by researchers. BTE microphones and RF coils of the clinical processor are also replaced with custom-made microphone shells and RF coils compatible with the system. Researchers can perform subject tests using the CCi-MOBILE system to optimize algorithms prior to implementing a very costly FPGA to DSP IC conversion to fit the algorithm in a BTE processor.

B. Input and Output Configurations

CCi-MOBILE supports the Cochlear Corporation's Nucleus 5-7 series, which is one of the most widely used implant designs. The system consists of 22 channels with an array of 22-electrodes implanted in the cochlea. Electrode 1 corresponds to the most apical electrode and electrode 22 corresponds to most basal electrode [30], [31]. The lower electrodes capture low frequency sounds while higher electrodes are mapped to higher frequencies of the sound spectrum. BTEs and RF coils were custom-made to match the specifications of the clinical components.

Commercial BTEs (Cochlear Corporation) consist of a microphone connected to a proprietary sound processor, however the custom-made BTEs replace the processor with an empty shell having only the microphone circuit and no DSP chip because CCi-MOBILE acts as the reconfigurable DSP chip (Fig. 2). The left and right BTE microphones are connected directly to input ports of CCi-MOBILE, which transmits the incoming sound captured by the microphones to an on-board audio codec IC for analog-to-digital (A/D) conversion of incoming sound signals. Microphone array inputs are currently not supported by the platform, as CCi-MOBILE requires a single channel input from both the left and right microphones. The previous ci-PDA had an analog circuitry that included separate components of pre-amplifier, anti-aliasing filter, and A/D converter, which had to be connected with closely placed wire traces for digitalization of incoming audio. Replacing the pre-amplifier and A/D converter with a commercial audio codec in the hardware design improved the signal quality of audio streams for subsequent processing on the CCi-MOBILE. In order to make input configurations similar to the clinical processor, the audio sampling rate for each left/right side is maintained at 16 kHz, and frame duration for the acoustic buffer is 8ms to support real-time implementation of the signal processing strategies.

C. Major Components of the Hardware

The CCi-MOBILE interface board has a high quality 4-channel A/D codec, Xilinx Spartan 6 FPGA, USB interface, and a fully integrated Wi-Fi transceiver for wireless data transmission. It is a light-weight PCB with dimensions of 3"x2". The Codec and the FPGA perform all major tasks of translating the input communication signal from the BTE microphones to the computing interface (PC/ mobile). Fig. 3 highlights the major functional units of the CCi-MOBILE.

1) Spartan 6 FPGA: The central processor of the interface board (Fig. 4) is a Spartan 6 Field Programmable Gate Array (FPGA) from Xilinx (XC6SLX45). The FPGA controls the data flow within the system. It is responsible for synchronized capture of audio frames from the codec, two-way data transfer, encoding of RF data, and synchronized delivery of stimulation pulses to the RF coils. The codec, Wi-Fi and USB transmission units communicate with the core FPGA using the UART standard protocol. In addition to the RF encoding, Proprietary firmware for the FPGA verifies that all electrical stimulation received meets implant stimulation requirements set by International Electrotechnical Commission (IEC) standards. The FPGA is programmed in Verilog, which is locked, and not editable or accessible to researchers/users.

2) Audio Codec: The audio codec from Wolfson Microelectronics (WM8983) is a highly integrated input/output device designed for mobile computing and communications (see Fig. 4). The input from the microphones is directly streamed to the codec for A/D conversion, then the digitized signal is sent to the FPGA. The WM8983 has a sampling rate of 16 Khz, 4-channel ADC with an over sample rate of 128, and an equalizer gain of 0db with a cut-off at 80 Hz. A programmable high-pass filter in the ADC path provides for wind noise reduction and an

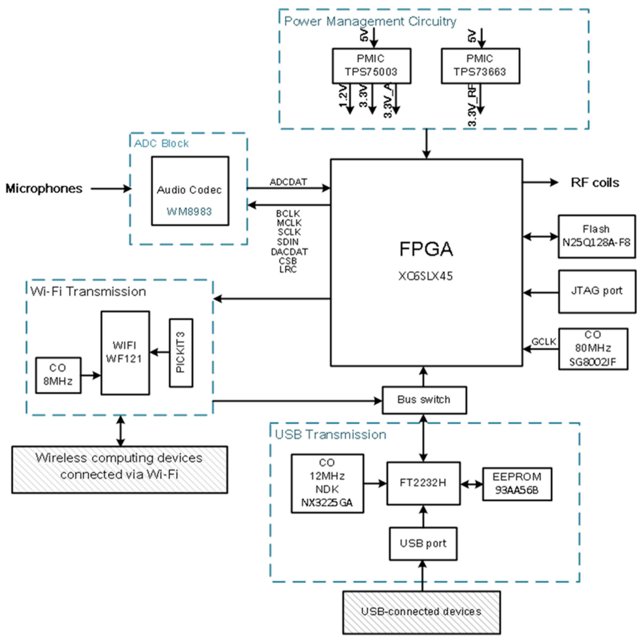

Fig. 4. Block diagram showing design flow of each Hardware component.

IIR with programmable coefficients is used as a notch filter to suppress fixed-frequency noise.

3) Wi-Fi Transceiver: The Wi-Fi transceiver from Bluegiga (WF121) is a stand-alone Wi-Fi module that provides fully integrated 2.4 GHz 802.11 b/g/n radio, TCP/IP stack along with a 32-bit micro controller (MCU) platform for simple, low-cost and low-power wireless IP connectivity. The interface of preference of the WF121 with the CCi-MOBILE is the 20 Mbps UART, USB on-the-go, which supports wireless communication to and from the computing platform. The main reason for having a Wi-Fi module on the platform is to ensure portability of the system, unlike most previously available research interfaces.

4) USB Interface: CCi-MOBILE has both wireless and wired options for data transfer, managed by an on-board physical switch. The wired interface through FT2232H is also portable when connected to a smartphone or tablet. CCi-MOBILE is not portable only when researchers wish to perform experiments through a personal computer. The FT2232H is a USB 2.0 Hi-Speed (480Mb/s) to UART/FIFO IC used for direct connection to PC/laptop/tablet with CCi-MOBILE via a micro-USB connector.

5) Computing Platform: The UART interface transfers all the digitized incoming data to any computing platform that can run the novel test CI/HA speech processing algorithm programmed either in MATLAB or JAVA Android. For portability purposes, this solution urges using a tablet for testing the algorithms coded in MATLAB and any stock Android smartphone (e.g.,-Google Pixel) for algorithms programmed into an app in Java. The algorithm performs the final processing of generating output desired by a CI/HA user with the help of the computing platform's resources and sends it back to the CCi-MOBILE through the UART to be directed to the implant of the user.

D. Power Management

The circuit board runs either through a 5-V ROMOSS battery-pack which connects to the micro-USB port of the board when the CCi-MOBILE is switched to wireless mode or draws direct current from the computing platform when connected in the wired USB mode. The power management circuitry comprises of two main components: TPS75003 and TPS73663. The TPS75003 is an integrated triple supply power management IC (PMIC) for the Xilinx Spartan FPGA and regulates voltage and current levels of the digital part of the circuit board. The TPS73663 is a low-dropout (LDO) regulator and is used to provide stable power to the RF analog circuitry. The use of two separate LDOs ensures lower cross-interference between the digital and analog circuitries. The CCi-MOBILE can drain a stand-alone smart-phone battery in about 10-12 hours in wired mode, depending upon the type of smart-phone being used. This study used Google pixel for all evaluations.

E. Modes of Application

1) Stimulation Modes: Three different CI/HA configurations exist:

- Unilateral: CI in one ear and normal hearing for the other ear, requiring 1 output channel.
- Bilateral: CI in both ears which requires 2 outputs.
- Bimodal: CI in one ear and a HA on the other ear, also requiring 2 output channels.

The stimulation modes here correspond to three ways in which the input needs to be processed and directed to the output. The unilateral stimulation mode requires one stream of output which either can be left or right. Bilateral stimulation requires the 2 output channels to synchronize with each other for both the left and the right ear to maintain localization of incoming sound. Bimodal stimulation has 2 output channels synchronized with each other generating electric and acoustic stimulations (EAS) simultaneously one for a CI and the other a HA.

Each category requires specific stimulation strategies, which are different from each other. CCi-MOBILE provides complete support for the unilateral/bilateral and bimodal CI stimulation. The electric-only stimulation for unilateral and bilateral CIs can be switched within the software. The synchronization of the left and right output channels has been matched using an oscilloscope. The platform supports unilateral and bilateral acoustic stimulation. Although, bimodal operation with CI/HA is also supported, further research advancements continue. Specifically, if electric(CI) and acoustic(HA) stimulation is needed for left/right (L/R) ears, the current firmware effectively delivers this output. However, based on community researcher feedback, many technology and scientific investigations require that acoustic signal arrival timing from input to output be preserved. This is critical for investigations in sound localization and speech/speaker perception based on direction-of-arrival (DOA). Therefore, further built-in timing checks and advanced simulations will be available in future open-source releases of the firmware to better support directionality and sound localization features.

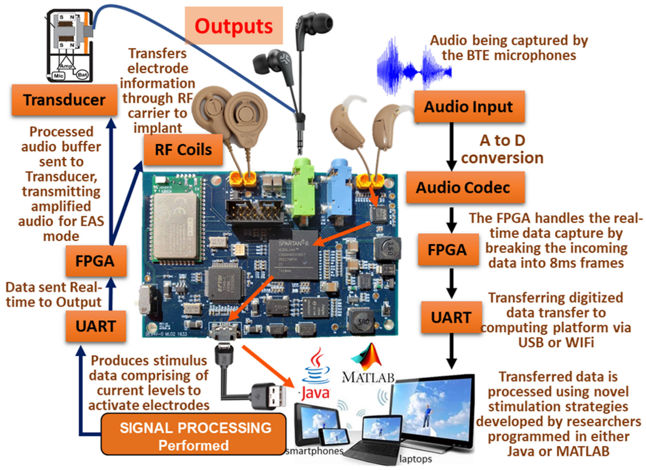

Fig. 5. Data flow for real-time parallel firmware processing.

2) Data Communication Modes: As previously noted, CCi-MOBILE communicates with the computing platform (PC/tablet/smartphone) through both wired and wireless data transfer. Early versions of CCi-MOBILE were designed to be wireless with Wi-Fi being the only communication mode with the computing platform. While wireless Wi-Fi support was operational, based on user algorithm, complexity, and real-time operations, some packets could be dropped to maintain real-time performance. Hence, in addition to wireless mode, design modifications were made to add a USB based wired data transmission with a toggle switch to change between the modes.

V. Firmware and Operational Behavior

CCi-MOBILE was programmed to provide both real-time as well as and off-line processing of incoming data, to support a wide range of experiments. Firmware at system level is programmed in Verilog and is not available to researchers for use. The firmware for the software, running on the computing platform is programmed in MATLAB (for PC/tablets) and JAVA (for smartphones. The software is open source and available for modification by researchers.

Fig. 5 describes the real-time functioning: (1) 8ms data frame is captured from A/D, (2) data frame is transmitted to the processing module, (3) signal undergoes sound processing using novel algorithms on the PC/smartphone/tablet, (4) data is transmitted back to the interface board, (5) received data is encoded using the RF communication protocols, and finally (6) the encoded data is streamed to the RF coil. This six-step process is carried out in real-time repeatedly on each 8ms data frame. To maintain integrity of real-time processing/streaming, several methods were employed simultaneously [32]. These include:

a) Parallel processing firmware implementation to control data transmission and data routing;
b) High speed data transmission to and from the interface board with shorter PCB trace paths;
c) Optimization of sound processing libraries, best programming practices for real-time applications, and use of applied DSP (e.g., fixed point programming).

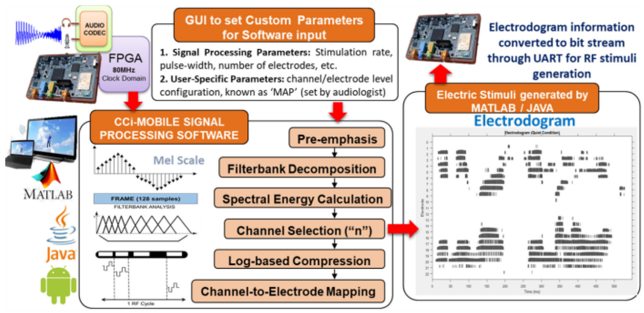

Fig. 6. Signal processing flow in the software of computing platform.

A. Firmware at Software Level (PC/smartphone)

The internal signal flow processing from the CCi-MOBILE to a PC/smartphone programmed in MATLAB/JAVA (android) is shown in Fig. 6. For converting natural audio to electrical stimuli compatible with cochlear implants, a clinical 'n-of-m' processing strategy entitled Advanced Combination Encoding (ACE) [33] is implemented. ACE is one of the most widely used sound coding algorithms in Cochlear Corporation's Nucleus series of CI devices. This strategy selects a subset of 'n' electrodes out of 'm' total electrodes based on frame energy of the corresponding frequency channels.

Here, ACE is implemented in both MATLAB and Android Java for the CCi-MOBILE research platform. Researchers developing novel speech enhancement, noise cancellation, or localization techniques, can insert their new algorithm within the current ACE strategy framework and analyze the developments. CCi-MOBILE MATLAB and JAVA ACE code implementation are made open-source to support community researchers. As discussed before, the platform acquires 8ms (128 samples) frames from the A/D converter at 16 kHz per left or right channel and transfers them to the computing PC/smartphone. The incoming frames are first passed through a 1st order pre-emphasis filter. The signal is then buffered into the overlapping Block-shift windows (each 16 samples) with a 128-point Hann window applied.

The Hann window $w(m)$ is defined as:

$$w\left(m\right) = 0.5 \left[1.0 - \cos\left(\frac{2m\pi}{L-1}\right)\right] \qquad m = 0, \ldots, L-1 \tag{1}$$

A 128-point FFT is then performed, which yields bin center frequencies (f_c) that are linearly spaced at multiples of 125 Hz. Because the input signal is real, the output is Hermitian symmetric. Therefore, only bins 1 to 64 are used to calculate the magnitude squared spectrum as:

$$X(k)^2 = X_R^2\left(k\right) + X_I^2\left(k\right) \qquad k = 1, \ldots, L/2 \tag{2}$$

The squared magnitude response is then appropriately scaled by a weight matrix which determines the frequency boundaries of each filter channel. The nth filter channel envelope is:

$$Y\left(n\right) = \sqrt{\sum_k a\left(n,k\right) X^2\left(k\right)} \qquad n = 1, \ldots, N \tag{3}$$

TABLE I
OPERATING RANGES OF USER-DEFINED VARIABLES

User Defined User Parameters			
Parameters	Min	Max	Default
Number of Electrodes	1	22	8
Total Stimulation Rate (Hz)	125	14400	1000
Pulse Width	25	400	25
Data Bytes Per Frame	1	115	64
Overlap/Block-shift window (samples)	2	128	16
Sensitivity (dB)	0.0	10.0	5.5
Gain (dB)	0.0	50.0	25.0
Fixed parameters			
Sampling Frequency (Fs)(kHz)	16000		
Data Frame Length (seconds)	0.008		
Samples per Frame (Fs*$Frame\ Length$)	128		
Inter-pulse gap (seconds)	0.000008		

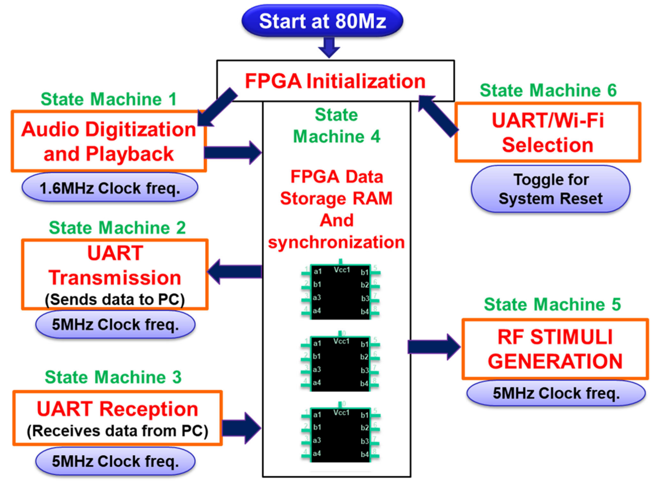

Fig. 7. High level data synchronization between the state machines.

To allow shape adjustments of the frequency response, a vector of variable filter channel gains g(n) is also defined. The final filter bank output vector is given as:

$$Z(n) = G(n)\,Y(n) \qquad n = 1, \ldots, N \tag{4}$$

Each input block analysis produces one vector of N filter bank envelope samples. The output is sorted and *Nmaxima* outputs are selected, which is compressed logarithmically. Finally, the output is mapped to necessary current levels using the subject's MAP parameters (maximum comfort loudness, dynamic range, and threshold values) in the form of an electrodogram for visual analysis. An electrodogram illustrates the frequency channel dependent electrical stimulation information as a function of time. The electrodogram is more like an accuracy evaluator on the software side to help researchers in debugging if anything seems abnormal. This mapped output information of current levels, number electrodes to stimulate and amplitude values are embedded in a data stream back to the FPGA. The CI only accepts input in form of biphasic pulses due to restrictions set by the FDA. Hence, a special state machine on the FPGA firmware starts generation of biphasic pulses embedding the mapped output in the stream of pulses, such that the implant can comprehend data transferred through the RF coils. Researchers can then compare the final output with their desired results and perform parameter/system debugging, as necessary.

B. Operational Behavior Based on User Defined Parameters

With the software being open source and researchers having the flexibility to add their own algorithms to investigate possibilities of advancement, it is also important to understand the safety thresholds for making modifications to the parameter configurations [34], [35]. The max and min limits on these configurations are hard set both in software and hardware as shown in Table I. Any stimulation or input audio beyond the authorized stimulation range, initiates an error message on the software GUI and blocks transmission to CCi-MOBILE. This ensures safe execution of both, researchers selected CI/HA parameters and also protects CI/HA user if potentially harmful transmission occur.

There are three main specifications that cannot be altered and will remain grayed out in every GUI as they are hard wired in the FPGA firmware- audio sampling frequency (16 KHz), frame-size for data transfer (8 ms) and the inter pulse gap (8μs). The following parameters - stimulation rate, number of electrode excitations (*Nmaxima*) and current levels, require re-running the assessment program every time changes are made due to variation in backend processing and requires monitoring by the researcher. The user specified choices of desired gain, sensitivity and loudness can be changed even by subjects during subject tests, on-the-go in any naturalistic field environment.

C. Firmware at System Level (FPGA): Parallel Processing

Fig. 7 demonstrates the Pipeline for CCi-MOBILE to Run in real-time. Six State-Machines (SM) Run Simultaneously Using the FPGA's Inherent Parallelism feature. These SMs Perform 6 Major tasks: (1) Audio Data streaming, A/D Conversion and Playback for Both Left and Right Audio channels, (2) Audio Data Transmission Through the UART, (3) UART reception, (4) Synchronization of Data Across Different Clock domains, (5) RF Stimuli Transmission to Both Left and Right coils, (6) State System Changes to Support the USB/Wi-Fi Toggle switch. Parallelism in Spartan 6 FPGA Is Implemented By These 6 SMs Programmed in Verilog That Run Concurrently Within the Top module, Synchronized By the System clock. The Function of Each SM Is Explained Below Followed By Their Implementation Within the Real-Time pipeline.

1) State Machine 1: Audio Digitization and Playback: Incoming naturalistic analog audio captured by the BTE microphone is directly streamed to the WM8983 codec for A/D conversion at a sampling frequency of 16 kHz, per clinical processor requirements. Each sample is a 16-bit signed integer number. The FPGA acts as a master and controls the functionality of the codec, which is configured using the Verilog code. It supports two audio input channels, one from the microphone on the left ear and the other input channel from the right microphone. For now, the codec is not programmed to support microphone array inputs, although there are sufficient resources to support and expand the research platform's capabilities towards more advanced front ends for both HAs and CIs.

Once CCi-MOBILE is powered and FPGA initialized, it activates the codec by sending a chip select, and register data signals through the bit clock. The digital audio interface is driven by the Bit Clock (BCLK) from the codec to the FPGA. The Left Right Clock (LRC) is the frame clock which signals the start of each audio frame of 8ms. The LRC is always equal to the sampling rate. To maintain a fixed sampling rate of 16Khz, the BCLK must be running at 1.6Mhz, resulting in the FPGA to use 100 BCLK transitions for every LRC transition. Sampling is then achieved at 1.6M/100 = 16KHz.

The first frame output from the codec is stored in the FPGA's first RAM buffer register which enables the flag for UART transmission through the USB/Wi-Fi to the computing platform. The first sample byte of the first frame corresponds to the left channel of digitized audio, that is stored in the even RAM address buffer locations. Right channel bytes are stored in the odd RAM address locations. Once the first 8ms frame is sent to the UART channel, the buffer is overwritten by the incoming next frame and this process continues until real-time audio is being captured. In bimodal configuration, the codec also performs D-to-A conversion (DAC), by generating an amplified audio output sent to the hearing aid transducer port in real-time.

2) State Machine 2 (SM-2): UART Transmission: Although UART transmission and reception are performed concurrently in real-time, tasks carried out during each phase involve different parameters and are considered two alternate SMs implemented simultaneously in a bidirectional manner. CCi-MOBILE receives power through the UART, either by pulling the power from the computing platform when using the USB or from an externally connected battery for Wi-Fi mode. When a manual toggle switch is set to USB and the platform is connected via micro-USB port, the on-board FTDI chip will immediately receive power and a send driver installation information through the micro-USB cable to the computing platform (PC/table/smartphone). Once the device is recognized by the computing platform and serial com port is assigned, the FPGA is initialized.

The next UART transmission task allows the computing platform to transmit a Ready-to-Send (RTS) signal to the FPGA after the UART flag is enabled by the audio codec State machine. Data transmission of 8ms frames start from the FPGA by sending a handshake Clear-to-Send (CTS) signal and transmission continues at a 5Mhz clock rate, in between start, and stop header for asynchronous handshaking.

3) State Machine 3 (SM-3): UART Reception: A challenging design task is how should data be processed from the algorithm output through COM ports of the computing platform. The data format streamed back to the CCi-MOBILE includes: (i) encrypted current level values, (ii) CI electrode information for left and right output channels, requiring individual FPGA decryption for RF stimuli generation. Once the transmission handshake is successfully established, a reception handshake starts simultaneously sending dummy frames to the computing platform via two FPGA UART COM ports to initiate real-time processing. To maintain UART data integrity, two separate RAM buffers are used for left and right stimuli generation, (i.e., two COM ports for the processed data reception). Also, a start and stop header check is performed for sequence validation of processed data.

The two 8ms frames returned from the computing platform to the RAM buffer contains 258 bytes of left and right stimuli information i.e., 516 bytes in total. An additional 516 bytes of acoustic data is streamed back to the CCi-MOBILE in real-time assuming bimodal firmware is running on the FPGA. For bimodal firmware operation to be activated, the right stimuli 515$^{\text{th}}$ byte remains unoccupied in electric only stimulation mode and is set to **'bb'** to enable bimodal features of receiving electric and acoustic output simultaneously (still in test phase).

4) State Machine 4 (SM-4): Memory Allocation and Data Sync.: An 8-bit wide RAM memory buffer of 2048 bytes is used to store, process, and release real-time data. Incoming audio data is attended to, simultaneously on a frame-by-frame basis. Frame duration is chosen based on the most efficient and benchmarked seek time for UART communication, which is 8 ms (i.e., the buffer sizes are maintained to accommodate 8 ms of real-time data, for both the Codec's acoustic buffer and RAM buffers). The samples per frame ($Samples_{Frame}$) is based on audio sampling frequency (Fs) at 16 KHz, and data frame rate at 8ms:

$$Samples_{Frame} = (Fs) * (Frame_{length}) \tag{5}$$

$$Samples_{Frame} = 16000 * 0.008 = 128 samples/frames \tag{6}$$

Each frame comprises of 128 samples, each representing 2 bytes, for a total of 512 bytes in the RAM buffer at any given point of time. A total of 125 frames, each of 8ms duration are processed per second. Additionally, start and stop bytes are added to ensure proper asynchronous UART data transmission, making the total frame length 516 bytes. The frame of 516 bytes is stored in the first 516 memory locations of the 2048-byte long RAM buffer, leaving 1532 RAM buffer memory locations unoccupied. The preceding frame of 516 bytes is simultaneously processed and received back from the computing platform in a separate buffer.

To achieve real-time processing with minimum throughput delay, the data management unit of CCi-MOBILE uses a ping-pong buffer criterion to leverage two buffers which increase overall throughput of the device and help prevent bottlenecks. The first buffer is called Ping, while the second is called Pong, so when the Ping buffer stores data, Pong buffer processes the data, moving memory back and forth so that different parts of the system can use it without interference (see Fig. 8). The ping-pong buffers reduce overall system latency varying up to 30ms. The time taken for a normal human ear to capture and transport audio to the auditory nerve represents a latency of ∼166ms [36], [37]. Hence, an initial latency up to 30ms, ensures that CCi- MOBILE can operate in real-time, maintain processing and natural perception.

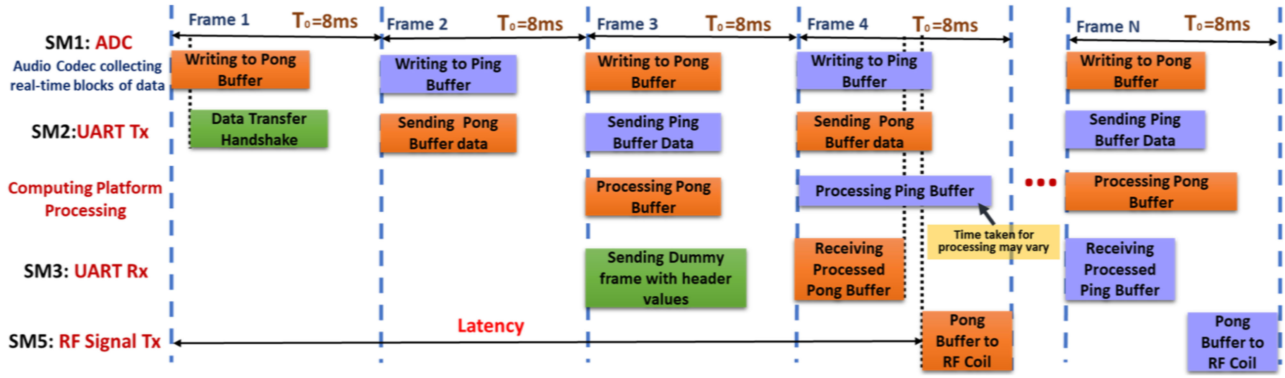

Fig. 8. Showing real-time data storing, transfer and memory allocation by Data management State Machine (SM4) and synchronization with state machines (SM1, SM2, SM3, SM5). The two RAM buffers selected as ping and pong work simultaneously processing the incoming real time data.

5) State Machine 5 (SM-5): RF Stimuli Generation: In general, RF stimuli generation can be referred to as a black box mechanism specific to each cochlear implant manufacturer [38]. For Nucleus series implants by Cochlear Corp., a burst modulated 5MHz RF carrier is used to transfer wireless information from the output RF coil of the FPGA to the RF receiver at a data rate of 5Mbps. Required information to generate the stimuli include: – (i) the number of electrodes, (ii) current level at each electrode, (ii) stimulation rate (timing) and (iv) stimulation mode. This information is received from the computing platform in the form of a digital bit stream.

The prime function of SM-5 is to encode and embed the generated digital bits into symmetric square biphasic pulses [39], such that information can be transmitted over the carrier to the implant device wirelessly to excite the desired 22 channel electrodes for efficient hearing of the implant user. Information encoded in biphasic pulses are decoded by the implanted IC of the cochlear implant, which stimulates the selected set of electrodes at a given point in time. A train of symmetric square biphasic pulses are used to stimulate the auditory nerve via the intracochlear electrode array. In the event of current spreading or current leakage, a neutralization process ensures each biphasic peak will have the same intensity as the peak prior to maintain a safe overall stimulation criterion (Fig. 9). SM-5 is also responsible for powering the cochlear implant through wireless RF power transfer.

During data transmission, RF stimuli is generated through 8ms frames. Each biphasic pulse comprises of two data blocks: Data Block 1 and Data Block 2 (see Fig. 9). Each block contains a start token (ST), an Error Token (ET), a Phase Extender (PE) and 3 Data Tokens (DT) with each data token consisting of 3 bits. One biphasic pulse contains information for any one electrode (5 bits, E1-E5), its current amplitude (8 bits, A1-A8) and timing information (5 bits, M1-M5). A single biphasic pulse holds 24 bits, 18 bits for the electrode stimulation information and 6 bits for the start and stop tokens. Researchers can freely select the number of electrodes to stimulate simultaneously at a given point of time.

Stimulation rate, the timing information of biphasic pulse trains, is an important factor and is defined as the number of biphasic pulses per second. The maximum FDA approved

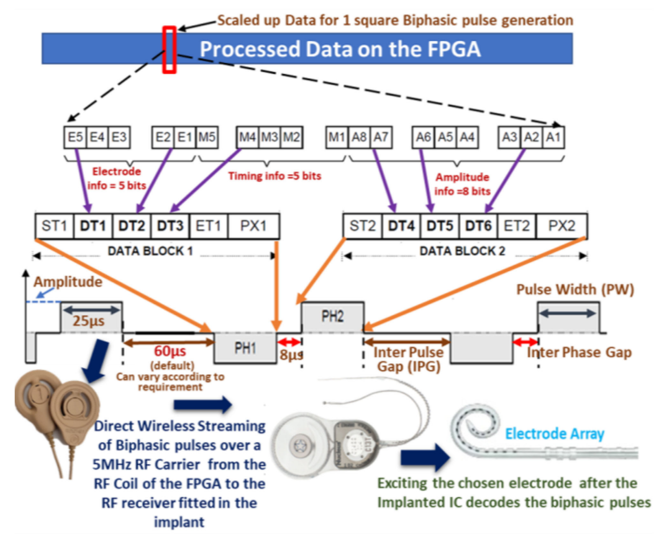

Fig. 9. Generation of biphasic pulses and data encoding for RF stimuli excitation from processed data received from the computing platform.

stimulation rate for the Cochlear Corporation implants is 14,400 Hz or pulses per seconds(pps). As the number of stimulated electrodes per frame is increased, the individual stimulation rate available per channel decreases. For example, if 8 channel electrodes are specified by the researcher per frame (this is referred to as 'n-maxima': N_{maxima}) the maximum stimulation rate (SR_{max}) per channel reduces to 1800 pps/channel (see Eq. 7), (i.e., a researcher cannot set a stimulation rate ($StimRate_{ch}$) higher than 1800Hz). N_{maxima}, originally set by audiologists (for the user's clinical CI configuration) can be defined by the researcher as any number of channel electrodes between 1 to 22 which constrains the stimulation rate.

$$SR_{max}/ch = (SR_{max})\,/N_{maxima} \tag{7}$$

$$SR_{max}/channel = (14400)\,/8 = \ 1800pps/ch \tag{8}$$

Therefore, processing and transmission of 8ms frames for biphasic pulse generation varies with varying stimulation rates. The total number of biphasic pulses generated in each 8ms frame

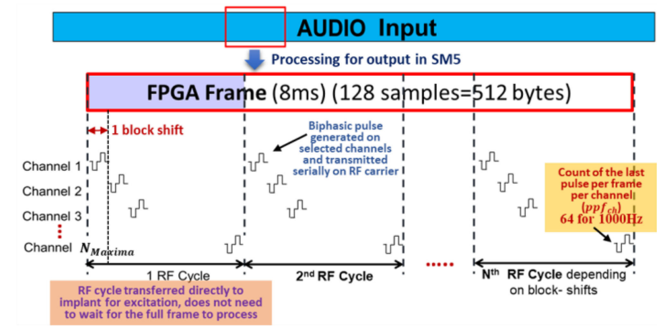

Fig. 10. Biphasic pulse stream generation through RF stimuli state machine. While one RF cycle is processed at a particular instant in a ping buffer, another RF cycle is sent to the RF coils through pong buffer, due to the real-time parallel processing of incoming data.

of real-time RF stimuli generation can be calculated as follows:

$$Pulses_{Frame} = \left(\frac{Frame_{length} * Fs}{Fs/StimRate_{ch}}\right) * (N_{Maxima}) \quad (9)$$

Here, $Frame_{length}$ as indicated before, is hard-wired at 8 ms or 0.008 s and Fs is also set to 16 KHz. Researcher defined factors impacting rate of RF stimuli generation are $StimRate_{ch}$ and N_{Maxima}. For values of N_{Maxima}= 8, and $StimRate_{ch}$ = 1000Hz, the $Pulses_{Frame}$ is calculated as-

$$Pulses_{Frame} = \left(\frac{0.008 * 16000}{16000/1000}\right) * 8 = 64 pulses/frame \quad (10)$$

For the 64 pulses per frame (ppf), 8 biphasic pulses are generated at any given instant, and the remaining pulses in the frame are also generated as a packet of 8 biphasic pulses. A packet of 8 biphasic pulses stimulating 8 electrodes at a time is termed an RF cycle. Therefore, 8 RF cycles (each cycle with 8 pulses) are needed to generate 64 biphasic pulses. To generate these packets in real-time, the 8ms frame needs to be further divided into block-shifts to accommodate and process each RF cycle. This is achieved by implementing a 'block-shift' for each RF cycle so as to send back final results faster with simultaneously processing. The number of samples processed in one block-shift is determined by dividing the incoming constant Sampling frequency (Fs = 16000Hz) with the chosen $StimRate_{ch}$, which for our example is 1000Hz. This gives 16000/1000 = 16 samples per block shift. Fig. 10 shows the corresponding biphasic pulse stream formation within SM-5.

6) State Machine 6: USB/Wi-Fi Toggle Reset: To enable both wired and wireless data transmission CCi-MOBILE supports USB and Wi-Fi modes. With a UART data transfer rate of 5Mbps, it facilitates USB and Wi-Fi modes through a single pole double throw switch (SPDT) mounted on the PCB. The state machine is programmed such that whenever the switch toggles and changes state from 0 to 1 or vice versa, the FPGA reboots and the system is re-initialized. The toggle switch should be set before starting an experiment, since switch toggling in the middle of data processing will result in errors.

Fig. 11. Android app for real-time ACE sound processing application, environment classification, and adjusting User-defined parameters.

VI. Key Technical Design Advancements

In order to provide the most flexible CI/HA research platform for the research community, a number of novel and flexible hardware/firmware advancements were integrated into the CCi-MOBILE. These advancements provide scientists and researchers in CI/HA field improved opportunities to explore new concepts and theories for the hearing-impaired community.

A. Bringing More Convenience to the GUIs

Both MATLAB and Android GUI were developed and made open source for research and lab testing purposes. Through the MATLAB GUI, the PC/tablet can be configured to perform signal processing tasks in real-time mode as well as off-line, giving a visual simulation of the impact the parameters made on the implant electrodes excitation. The android GUI specifically has been custom-made for testing and analyzing responses of CI/HA users in multiple outdoor location and natural environments. The Java Android version works best with a stock android phone, preferably a Google Pixel (Fig. 11) allowing improved performance. Both the MATLAB/JAVA GUI and the backend software are programmed and tested for thread assigned priority, making sure that the software or the algorithm running within the system will not get interrupted by other higher priority events like phone calls or sudden system triggered sound/pop-ups.

B. Pulse Structure Control With Stimulation Rate

An early developed firmware for the ci-PDA [25] only allowed a constant stimulation rate and pulse width combination for both left and right channels supporting bilateral stimulation. For the new research platform formulation, the firmware supports researcher-based specification of independent stimulation rate and pulse-width combinations for both left and right ears. This feature was developed to enhance accuracy for CI research focused on localization experiments.

Currently, the algorithm architecture of user-specified parameter monitoring, ensures that minimum and maximum pulse-width are met for each 8ms frame. To compensate for parameters outside the operating region, the system automatically changes the entered user-specific stimulation rate to the max allowable number, to ensure integer value of samples for each frame. This adjustment is accomplished during system initialization process.

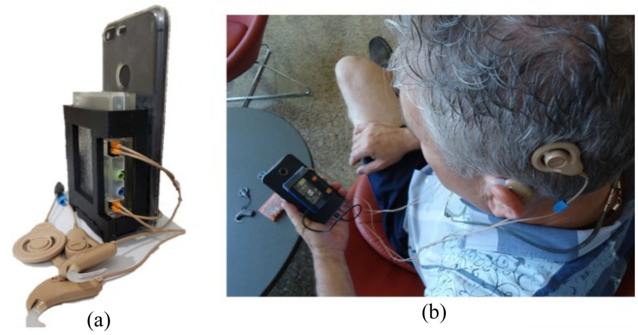

Fig. 12. (a) 3-D printed casing for portable/wearable CCi-MOBILE with a smartphone (b) A subject using CCi-MOBILE with the compact casing.

C. Real-time Resource Utilization and Debugging

The functionality of the FPGA design is verified through RTL simulation at the hardware level. Debugging was accomplished with a JTAG port present on the PCB, ensuring integrity of the data exchange between the computing platform and hardware interface board through a Xilinx ISE Design Suite 14.7 and ChipScope pro analyzer. The design summary at run-time recorded total FPGA resource utilization to be an average of 30% of the available hardware resources. This means, even with multiple processes running in parallel, the system has sufficient processing power and room for future advancements.

D. 3-D Case Design and Upgrades

To ensure the CCi-MOBILE supports use in laboratory/benchtop tests as well as take-home field testing, an effective housing for hardware and connectivity for processor (PC/ Android) system, CI BTE mics and RF coil connectors was needed. For the CI user to use CCi-MOBILE in a daily wearable and safe mode, a well-covered enclosure is designed. The enclosure comprises of a phone case and connecting slots for a smartphone, making the system easy to carry around. SketupPro 3D printing software was used to design the casing for CCi-MOBILE and the phones (Fig. 12).

VII. Experimental Analysis and Validation

The CCi-MOBILE has been cleared as an Investigation Device Exemption (IDE) by the FDA and is registered under a series of IRBs set by the University of Texas at Dallas. Three stages of experiments were performed with CCi-MOBILE to evaluate its functionality, accuracy, safety, and durability. The current output levels at the electrode array were verified extensively using 1) an implant emulator, and 2) a decoder and implant emulator toolbox, from Cochlear Corporation termed as the DIET box. Electrodograms produced by each processor were compared with oscilloscope results to ensure consistency and accuracy for the same input. First, the accuracy of the device was compared against the standard clinical processor for sentence intelligibility. Next, to check for sound and parameter setting safety thresholds, a wide range of field acoustic signals and MAP parameter combinations were tested with the CCi-MOBILE.

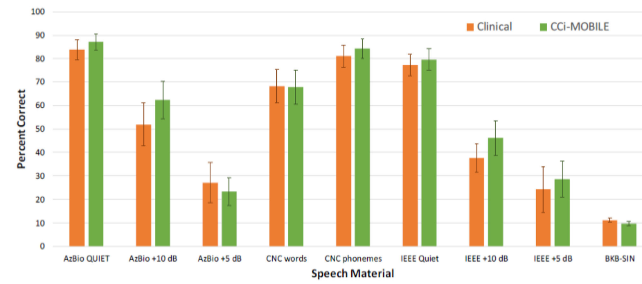

Fig. 13. Percentage correct mean speech recognitions scores with clinical processor and CCi-MOBILE research platform. N = 8.

Lastly, the durability, battery-life and performance integrity were quantified dependent on user input parameters.

A. Functional Accuracy Testing

In order to assess the efficacy of the research interface, human subject evaluations were carried out with 8 CI users. The aim of the study was to evaluate human speech recognition performance of CI users with the CCi-MOBILE research interface to ensure that performance will be consistent with their clinical processor. Eight post-lingually deafened adult CI users participated in this study. The assessment of speech recognition was accomplished with AzBio [40] and IEEE [41] sentences presented in quiet, 10dB, and 5dB signal-to-noise ratios as well as with CNC words/phonemes. Study participants were tested in a naturalistic environment, both with their clinical processor and the CCi-MOBILE research platform.

Both devices were programmed with a standard ACE sound coding strategy. Across all measures of test material, the proposed CCi-MOBILE custom-built mobile research interface produced equivalent performance levels ($\mu = 54.348 \pm 6.163$) to each individual's clinical processors ($\mu = 52.276 \pm 6.318$). Repeated Measures Analysis of Variance (ANOVA) revealed no statistically significant difference between the two processor types. The results from this study (see Fig. 13) indicate that performance levels with the research platform are comparable to their clinical processors, and therefore able to accurately duplicate the user's existing clinical configuration [42]. This result suggests great potential for conducting reliable speech assessments in future studies with CCi-MOBILE.

B. Acoustic Safety Threshold Analysis

This experiment involves connecting the RF-Coils to the Implant Emulator (DIET Box) as shown in Fig. 14, instead of a human subject's implant during the test. The DIET box has the same in-built technology as the surgically implanted CI receiver and electrode array, hence acting as an effective replacement for researchers who do not have enough access to hearing impaired subjects. For this Burn-in test, the desired computing platform presents the CCi-MOBILE platform with a total of 480 hours of input stimuli (hours of acoustic sounds at various acoustic levels) [43] which were assembled for probing the entire human acoustic space. These include:

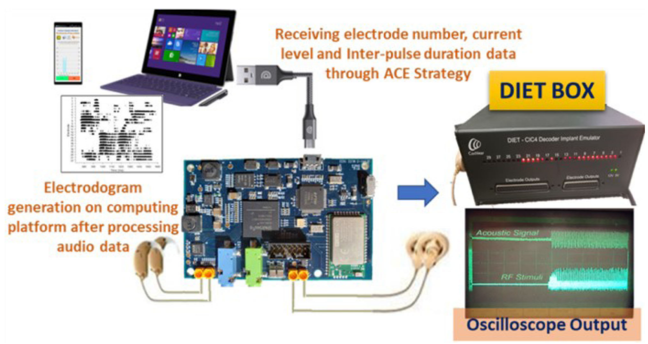

Fig. 14. Validation of CCi-MOBILE using the DIET box and oscilloscope.

- Speech signals (IEEE, TIMIT, AzBio databases) at various SNR and intensity levels (30dB, 40dB, 50dB, 60dB, 70dB, 80dB, 90dB and 100dB SPL)
- Noise corpora (Noizeus, Noisex etc.) consisting of everyday realistic environments (e.g.,- babble, speech-shaped train, car, gunshot, impulse and synthetic noises)
- Artificial created sounds (tones, chirps, random mixtures) at various SPL levels.

MATLAB and bash scripts were written to monitor project any abnormalities into output scatter plots as well as output electrodogram pulse patterns, which are easier to compare against a CI user's clinical system.

C. User Defined Specification Testing

Although CCi-MOBILE gives researchers the freedom to make on-the-go modifications to speech processing parameters, as noted before in order to ensure that the parameters chosen are safe, it was necessary to have a well-defined set of tests, ensuring that the user-defined settings remain within operational limits of the platform. The main user-defined parameters are quantified based on three input parameters: (a) stimulation rate, (b) pulse width, and (c) number of activated electrodes. Each of the three input parameter settings were iteratively adjusted and tested with integer-step size within respective operational ranges (a) stimulation rate (125-14400 pulses per second), (b) pulse width (25-400μs), (c) number of electrodes (1-22). Approximately 70% of the user defined MAP operational combinations can produce clinically feasible electrical output. Eq. 5 through 7 decide, if clinical levels per electrodes exceed the maximum clinical level post processing before sending the stimulation to the board. If they do, the user would be notified of the compliance issue and the clinical levels are decremented to the maximum allowable parameter value.

VIII. Conclusion

This study has presented a portable and wearable signal processing algorithm testing platform for research focused on CI advancements. This allows researchers an opportunity to realize complex algorithmic and scientific inquiry to advance speech technology for the hearing impaired. CCi-MOBILE is aimed at developing a "stream-lined" approach to accelerate lab-to-world algorithm advancement to provide the hearing-impaired a better user-experience. The flexibility of the device enables researchers to advance their theories and algorithms to create new technologies with less programming skills.

It was shown that the research platform allows both off-line benchtop/lab testing as well as naturalistic field testing for real-time "on-the-go" CI user studies. The real-time mode operates like the clinical processor and allows practical assessment of new algorithms in naturalistic environments which provides spontaneous feedback from CI users. To ensure CCi-MOBILE is universally accepted by both speech processing engineers and scientists alike, MATLAB and JAVA frameworks have been used for programming the firmware. The potential to control individual stimulation parameters, timing and stimulation pattern of each electrode provides researchers with a broad spectrum of experimental avenues to explore. Acoustic stimulation for bimodal processing using CCi-MOBILE is currently under development and will be further evaluated in future studies. The platform has also contributed towards modern day IoT and smart room-based technologies, which seems to be a promising future [44].

Currently, 20 CCi-MOBILE units have been shared through subscription with various research labs worldwide with additional 25 units available. The software suite is open-source, published on the CRSS-CILab website and the GitHub site provides support for any changes, updates, and additional resources to be accessed by sites using the platform. More details can be found at- https://crss.utdallas.edu/CILab/.

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