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Assessing Speech Processing HA/CI Advancements for Naturalistic Field Testing: Advancements with CCi-MOBILE Research Platform

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ABSTRACT

Advancements for hearing assistive technologies for hearing aids (HA) and cochlear implants (CI) have advanced significantly in the past two decades, with new emerging machine learning (ML) approaches used in speech technology applications (e.g., speech enhancement, recognition, diarization, etc.). However, advanced computing resources and real-time operational needs can limit realizing viable solutions in real-world settings for CI/HA based processors. In this study, we present a high-level overview of the design, development, clinical evaluation, and applications of the CCi-MOBILE [1], a powerful signal processing platform built for researchers in the CI/HA hearing impaired field. We consider a range of past and current speech enhancement methods proposed for front-end noise and reverberation suppression, and address trade-offs for real-world naturalistic field testing for CI/HA scenarios. Here, recent work on Bilateral and BiModal based processing for HA/CI systems is explored. Experimental needs for extended day-long field testing with CCi-MOBILE research platform is also considered. This effort aims to create new research opportunities for scientists and researchers to investigate viable real-world human subject testing of emerging speech enhancement solutions that leverage ML concepts in complex acoustic scenarios.

Keywords: Speech enhancement, Speech Processing, Cochlear Implants, Hearing Aids, Machine Learning for Hearing Impaired Speech Communications.

1. INTRODUCTION

Sensorineural hearing loss, characterized by reduced or damaged functionality of the hearing system, can restore auditory sensation with the use of a cochlear implant (CI). According to the World Health Organization, nearly 1.5 billion people suffer from hearing loss, and the number is projected to grow to 2.5 billion by 2050. A CI consists of an intracochlear electrode array, an RF receiver (internally implanted) and transmitter, and a clinical processer with multiple microphones. The captured/recorded acoustic signal is processed and translated into an electric signal consisting of biphasic electrical stimulation pulses that encode time, frequency, and amplitude information and are used to directly stimulate the cochlear in an effort to restore hearing function. Signal processing strategies have evolved from simple, feature-extraction based approaches over the last 30 years due to the performance of CI listeners, increased number of intracochlear electrodes, improved understanding of the tradeoffs in spectral-temporal information, and the recent rise of machine learning (ML) capabilities. Historically, CI research has focused on addressing scientific and technology questions which employ benchtop/lab-based CI testing in controlled settings using pre-recorded/processed audio data. When theories or algorithmic advancements prove to be promising, they generally transition to CI manufacturers for potential implementation in commercial CI processors. This paradigm has some challenges since (i) there is a large time gap between academic research studies to follow-on industry investigation, (ii) not all promising basic research advancements show promise once implemented in clinical/commercial CI processers, (iii) results from time limited controlled lab experiments generally do not reflect the diversity of acoustic environments or conversational communication challenges CI users experience in the field, and (iv) benchtop CI listener experiments can only assess short-term acute testing performance, and do not allow for more long-term adaptation of the CI user's hearing function due to neural plasticity. Finally, to better understand the benefits of potential ML and smart speech technology advancements, allowing for "in-the-field" user feedback via EMA (ecological momentary assessment), an "on-the- go" data collection method for instantaneous feedback from CI subjects.

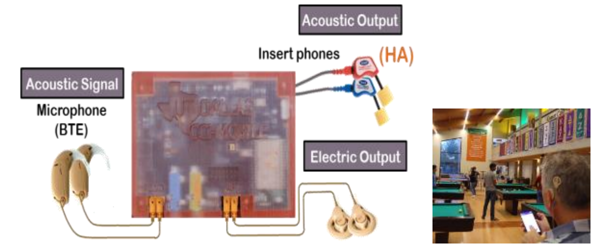

Fig.1: *CCi-MOBILE Research Platform – Cochlear Implant & Hearing Aid research for benchtop/lab & naturalistic take-home field testing.*

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2. BACKGROUND

A cochlear implant (CI) is a surgically implanted medical device that directly stimulates the auditory nerve with an electrically processed signal and is beneficial for the deaf or individuals who have completely lost their capability to interpret incoming audio[2]. Hearing aids (HAs) on the other hand are externally fitted in the outer ear and assist individuals who experience gradual loss of hearing. As the rate of hearing loss is different on both ears for any individual, having a HA in the better-ear and a CI in the ear with profound residual hearing, provides better hearing support to Bimodal CI users as compared to when the CI or HA was used alone [3], [4]. After CI surgery and HA installation, obtaining equal loudness amplification for both ears play an important role in providing the CI/HA user with a smooth user experience. Bimodal settings can also be beneficial for adult CI users eligible for sequential bilateral implants, who are awaiting for their second implantation after their first. For the time between the first and second CI surgeries, it is common to keep using a HA in the non-implanted ear. Various experiments, to determine speech recognition and localization abilities of bilateral and bimodal CI users in comparison to normal hearing (NH) subjects, have been conducted by research labs to help audiologists make the best fitting solution possible for each user, but performance of subjects have been widely varying with no set pattern of improvement as discussed in [5], [6].

3. CCI-MOBILE RESEARCH PLATFORM

In this study, we present recent advancements for the CCi-MOBILE research platform, which provides the CI/HA research community with an open-source, flexible, easy-to-use, software-mediated, computing research interface to conduct a wide variety of listening experiments (Fig.1). It supports CIs and hearing aids (HAs) independently, as well as bimodal CI/HA hearing. It is ideally suited to address hearing research for: both quiet and naturalistic noisy conditions, sound localization, and lateralization. The platform uses commercially available smartphone/tablet devices as connected portable sound processors and can provide bilateral electric and acoustic stimulation. CCi-MOBILE provides a range of research opportunities to explore new CI sound encoding and stimulation paradigms, including researcher control over an extensive number of parameters to create output RF stimulation pulses on a per channel frequency basis. A critical advancement for CCi-MOBILE is that it is fully portable and supports take-home/in-the-field testing, with operational battery support of up to 4hrs. Recent advancements include CCi-CLOUD support for data sharing, creation of EMA user feedback capabilities to capture both user initiated feedback from CI listener or conversational engagement in the field. Currently, 32x CCi-MOBILE research platforms have been adopted by 21 University/Research labs worldwide. There are a number of system innovations that include hardware components, firmware, and the open-source software suite to demonstrate safety for the speech scientist/engineer and CI/HA user, highlight user-specificity, and outline various applications of the platform for research. We will also emphasize new field operational capabilities that would support extended (day-long) CI testing in the field for take-home evaluation by CI users.

4. MULTI-DIMENSIONAL NATURALISTIC HA/CI TEST ENVIRONMENTS

It is important to differentiate traditional HA and CI testing paradigms vs. what HA/CI Users experienced in real-world environments. Fig. 2 highlights the multi-dimensional factors that impact the acoustic listening conditions where HA/CI users need to accomplish communications. A goal for future HA/CI research innovation requires moving into extended field evaluations where natural real-world multi-dimensional sources of distortion/interference is present. Past research studies have explored complex acoustic distortion content based on "Environmental Sniffing"[11-13], to characterize the complex sometimes random occurrences of individual noise/distortion events which impact speech quality and intelligibility for HA/CI users. More recent work has also explored machine learning based CNNs for non-linguistic sound classification and enhancement in normal hearing, hearing aid, and cochlear implant conditions (Chandra-Shekar & Hansen, [14-15]). This suggests great efforts are needed to integrate the acoustic sound field structure in preparation for both HA/CI based speech enhancement processing in future advancements needed in clinical deployed systems.

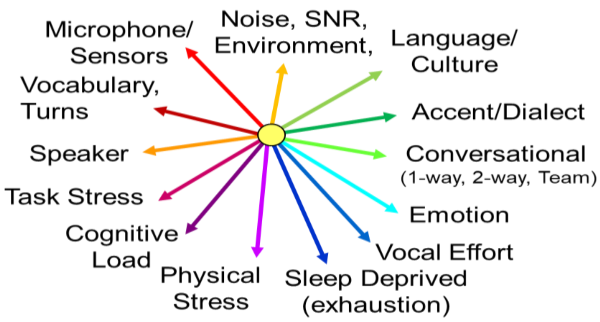

Fig. 2: Factors that impact the resulting speech audio content necessary for speech processing algorithms to address for HA/CI users in real-world naturalistic settings. Three broad dimensions include: (1) SPEAKER: speaker physiology, speaker style (task stress, cognitive load, physical stress, emotion, vocal effort, etc.), language, accent/dialect; (2) CONVERSATIONAL: monologue, 2-person conversation, multi-person group/team discussion; (3) ENVIRONMENT: environmental noise, reverberation, SNR levels, room acoustics (T60, RIR), competing noise/sound sources.

Traditional evaluation paradigms for speech, audio, acoustic processing algorithms for Hearing Aids (HA) and Cochlear Implants (CI) historically rely on pre-recorded read sentences with testing in controlled settings such as sound-booths or laboratory spaces. To fully realize the effectiveness of speech processing algorithms for HA/CI users, and to determine the viability of transitioning research based advancements towards clinical based solutions for use by subjects employing clinical processors, it is necessary to perform real-world testing in naturalistic field scenarios. Fig. 2 highlights a range of factors relating to the: (a) speaker {language, accent, speaking style/state, etc.}, (b) conversational context {1-way, 2-way, team/group discussion}, and (c) environment {noise, reverberation, public space, home, etc.}. These three dimensions contribute to the complex realistic scenarios HA/CI users experience on a daily basis in achieving effective human communications with the normal hearing (NH) community.

5. CCI-MOBILE SUPPORT FOR HA/CI REAL-TIME ENHANCEMENT PROCESSING

In cases where Bimodal HA/CI systems are needed, it is important to understand differences in HA vs. CI processing pipelines, which introduce different processing delays in the Left-vs-Right ears in Bimodal operation. Here, we explore these issues to ensure proper direction of arrival (DOA) information in terms of ITDS/ILDs. A major obstruction to accurate source localization for CI/HA users is the distortion of interaural time and level difference cues (ITD and ILD), and limited ITD sensitivity. It is necessary to develop and test algorithms that provide better localization and sound source identification cues. Bimodal processing adds further challenges to the left and right channel synchronization when compared to bilateral CIs, due to difference in output channel characteristics even after processing is complete. One output channel is an acoustic analog wave, feeding the contralateral HA, while the other is an electric pulse transmitted straight to the CI implant. The acoustic output being a travelling wave leads to additional latencies while propagating from the base to the apex of the cochlea (Fig. 3(a)), which is not the case for the electric output [1]. To keep this unavoidable temporal delay between the implanted ear and the HA ear as low as possible (preferably less than 10ms) additional measured forced delay and buffers are added to the CCi-MOBILE data management chain which is discussed more in [7]. These naturally induced temporal delays allow HA to support the peripheral auditory system extending ITDs, making sound localization, and sound appreciation more manageable for Bimodal or HA users. CCi-MOBILE supports the flexibility to program and process CI coding strategies and HA algorithms simultaneously. It also allows researchers to vary parameters and choose the best possible fittings in natural environments, during bimodal testing. This work discusses the functionality, design flow and performance verification of CCi-MOBILE when processing bimodal type CI algorithms.

5.1: EAS ALGORITHM PROCESSING WITH CCI-MOBILE

It is noted here that acoustic processing is intended to support HA and CI systems, which we consider as Bimodal (HA+CI) or Bilateral (assuming CI+CI) in the Left+Right ears. Previous research highlights bimodal or EAS processing as having two variants of its own [6], software algorithms supported by CCi-MOBILE, have also been made compatible for both variations. The first variant is where an individual has CI on one ear and HA on the other (Fig. 3(a)). The other is a type of partial insertion EAS implant, which combines CI with an acoustic HA, allowing simultaneous transmission of low frequencies through the HA by conventional acoustic amplification, while the electric stimulation supports the higher frequencies (e.g., this is more of a "split" type combination of HA+CI together). The CCi-MOBILE firmware has been programmed to support both EAS type partial and separate bimodal implants.

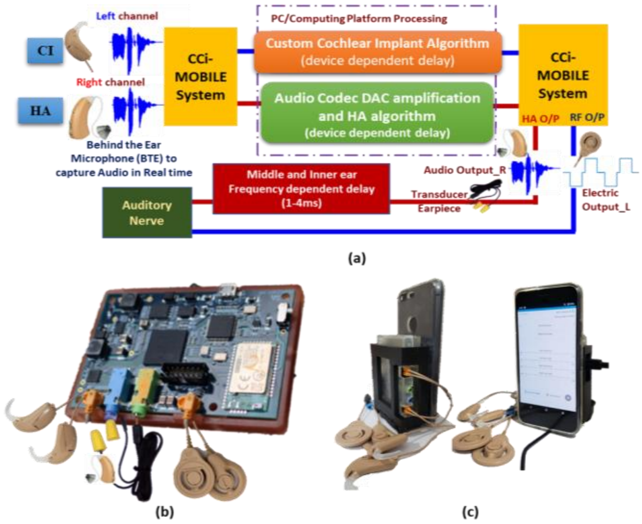

Fig. 3(a). Addedl Temporal delay due to audio wave transmission for the HA, while the CI directly stimulates the auditory nerve. Device dependent delay is fine-tuned accordingly to bimodal user's ear specification. **(b)** CCi-MOBILE, custom-made PCB for CI/HA algorithm processing. **(c)** CCi-MOBILE is portable via smartphone.

5.2: FIRMWARE & BIMODAL ALG. IMPLEMENTATION

A successful bimodal algorithm supporting benefits of acoustic and electric features would include features like amplification of the incoming sound, dynamic range compression, speech enhancement, noise reduction [7] etc. The sensitivity modulation along with loudness scaling has also been an important factor for optimized fitting for bimodal users. CCi-MOBILE in bimodal mode, processes the electric output in the same manner as the bilateral firmware as discussed in [7]. For this work, primary focus of the bimodal acoustic processing is based on controlled amplification and sensitivity suiting the loudness levels of the user's MAP parameters along with single microphone background noise suppression. The algorithm processing is initiated on the computing platform (computer/tablet) through MATLAB, while on-board AUDIO codec executes firmware specified amplification levels and the design flow is discussed in Fig. 4(b). The sensitivity of the output can be controlled both through the MATLAB GUI and an external headphone amplifier device. The noise suppression methods discussed by Kates [8] are placed in our MATLAB framework to further expand the acoustic processing power of CCi-MOBILE. Researchers can place their own algorithm modules in the acoustic processing framework set for CCi-MOBILE and efficiently test for bimodal synchronization and fitting analysis with subject tests.

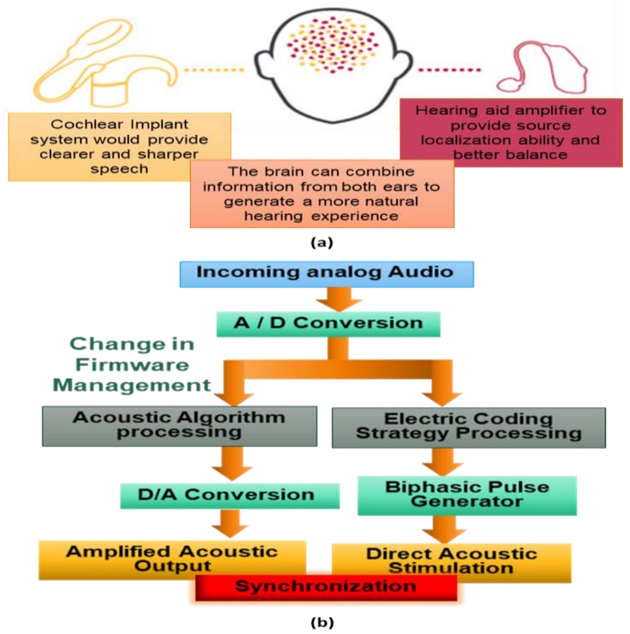

Fig. 4. (a)Bimodal hearing that is combined through the brain, and (b) Algorithm Design Flow for Bimodal Hearing using the research platform CCi-MOBILE for researchers and field testing.

5.3: REAL-TIME PROCESSING OF INCOMING AUDIO

To support real-time audio amplification and simultaneous electric stimulation for bimodal processing, CCi-MOBILE samples the incoming analog signal at a rate of 16KHz and breaks it down into 8ms data frames. The system initialization latency is about 10ms, with a dummy frame activating the entire buffering process and initializing the streaming of data frames. With four RAM buffers (2048x16 bits) transmitting and receiving data in ping-pong style, synchronized parallel processing is maintained between both the left and right ear channel outputs. Both channels process audio independent of each other and depending upon the direction of arrival (DoA) of an audio signal, a certain delay gets introduced between the channels, marking them right leading or left leading. If the left channel is leading, this means audio is being perceived from the left direction and vice versa for the right channel. MATLAB is employed for the user-interface controls for the processed signal in, where parameters like sensitivity, loudness and gain can be changed on the go by the user or researcher to match their hearing needs. The custom-developed HA or CI algorithm (Continued Interleaved Strategy (CIS) [9] algorithm was used for this work) would run in the background, with researchers being able to place and test their self-programmed algorithms in the open-source framework created for CCi-MOBILE. Similarly, an Android GUI was created with options to select type of environment and stimulation strategy of the cochlear implant at any given point in time. A Wide Dynamic Range Compression (WDRC) algorithm programmed in Java for the HA and the Advanced Combination encoder (ACE) coding strategy for the CI output channel was used for the Android modules. If an algorithm requires heavy processing, CCi-MOBILE also allows offline streaming and testing of such an algorithm, using a separate set of user-interfaces and code framework.

More information about the software framework and resources to obtain the testing platform can be found at the UT-Dallas CI Lab website: https://crss.utdallas.edu/CILab/.

5.4: EXPERIMENTAL SETUP

The Left and Right channel synchronization for bimodal CI processing was analyzed through a set of extensive simulations based on objective tests. Validating the output synchronization between both channels ensures proper analysis of hearing-impaired algorithms and authenticates that CCi-MOBILE is capable of testing complex as well as psychoacoustic based algorithms. The experimental test setup is divided into **Phase-#1**: Objectively analyzing real-time synchronization between left and right channels for various DoA of incoming audio through an oscilloscope; and **Phase-#2:** monitoring processing time of each channel on a large-scale simulation to ensure that even though processing for each L&R channel is independent of each other, the output is streamed simultaneously.

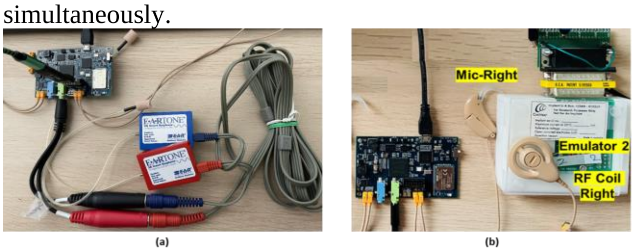

Fig. 5: CCi-MOBILE setup for bimodal firmware experimental validation: (a) EAR-Tone transducer for HA (b) CI implant emulator

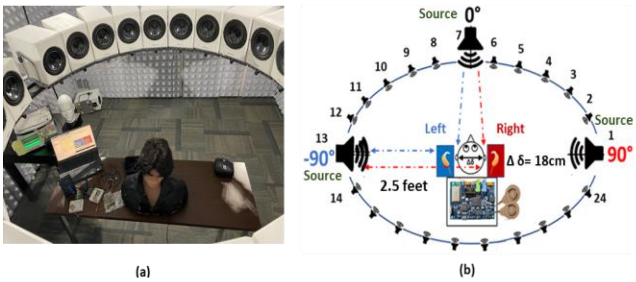

Fig. 6. (a) 3D Audio Sound Booth at UT-Dallas Callier Center; (b) Test setup in Sound Booth for real-time bimodal sound localization.

To analyze the acoustic output the EAR-Tone Research HA transducer was used which is plugged into the Line-OUT (green) port of the CCi-MOBILE as shown in Fig. 5(a). The electric output channel from the RF coil was probed on the oscilloscope using a CI emulator that was received from Cochlear Corporation® (Fig 5(b)). Source localization test was performed with oscilloscope captures in the 3D Audio Lab (Fig. 6(a)) at UT-Dallas Callier Center, to analyze if any streaming delay exists between the channels. Sound was played from -90°, +90°, 0° speakers of the circular mic array (5ft diameter) with CCi-MOBILE BTEs placed on a human head dummy at the array center (Fig. 6(b)). Left & right channel output performance were monitored on the oscilloscope, to see if right channel leads when sound is played from +90° speaker, and left channel output leads when sound is played from -90° speaker. When sound played from 0° speaker, which is equidistant from both

BTEs, then both channel outputs should be synchronized, giving a sense of source localization.

6. RESULTS

Channel synchronization analysis was performed in Audacity, and then confirmed on oscilloscope after final versions of programming the FPGA. Both channels were first tested in the 3D Audio Sound booth settings, for bilateral HA combination ensuring output synchronization with the noise suppression algorithm from Kates[8]. Fig. 7(a) presents oscilloscope capture for situation when audio is played at the left BTE microphone, making its amplitude greater than the right channel output, proving that pipeline processing on both channels is independent of each other. Fig. 7(b) presents synchronized output where the source is almost equidistant from both BTE microphones. Both the Left and Right HA output channel processes incoming audio independent of each other and are programmed to have a gain ability of 35 to 55 decibels, translating into between 85 to 142 decibels of volume sound.

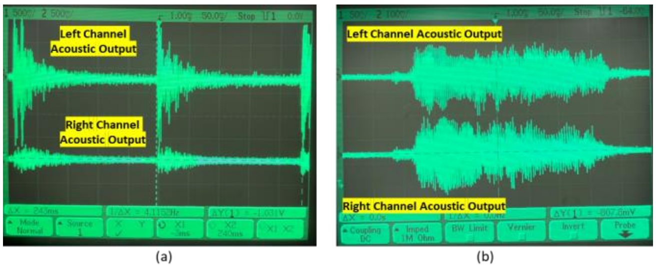

Fig. 7. Real-time Synchronized Bilateral HA CCi-MOBILE processing with (a) tonal signal played close to left BTE microphone (b) a sentence type audio played equidistant from both ears.

6.1. Processing-Time Simulation Based Results

Simulations were performed using +1000 .wav files, comparing output buffer streaming of the electric channel output for the CI versus the acoustic output buffer streaming for the HA channel. The main goal was to observe: (i) if any drastic processing delay mismatch is observed between two alternate left+right algorithms running on each channel. Fig. 8 shows both electric and acoustic outputs do coincide over the diagonal with only a few outliers creating a mismatch, confirming a negligible processing mismatch delay that exists between channels after streaming through the output buffer. The output buffer of the CCi-MOBILE ensures that the faster processing channel is "stalled" until enough data from the slower channel has been captured and processed. It is noted that input-output Probability Distribution Function (PDF) distribution comparison was also performed to validate CCi-MOBILE's capability to capture and retain initial DoA parameters and present that information at the output even after processing.

6.2. Objective-based Bimodal Source Localization

Fig. 9(a) shows a case where negligible delay occurs between electric and acoustic outputs, when the source and noise are both equidistant from Left and Right BTE microphones. The oscilloscope capture is for the word 'ape' and the electric pulse train is generated at a rate of 1000pps. With parameters kept the same, a second oscilloscope output (Fig. 8(b)) showed a slightly longer delay of 1.40ms, when sound was played closer to the CI (left; -90° speaker) and a 400Hz frequency tonal noise was played from the side of the HA (right; +90° speaker). This test was performed in iterations of 3 for word files, tonal signals and sentences being played in real-time from speakers in our 3D-Audio sound booth. All iterations gave similar results. This confirms CCi-MOBILE is capable of retaining any form of DoA information captured by BTE mics and presenting as output to the CI/HA user, assisting in source localization.

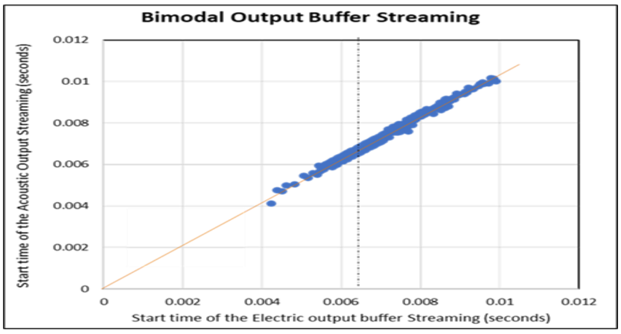

Fig. 8. Electric vs Acoustic output streaming synchronization with CCi-MOBILE. Input vs output pdfs with external delay added at input were also verified in this evaluation test.

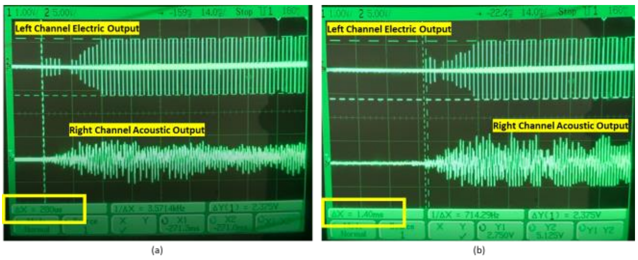

Fig. 9. (a) Output signal streamed from electric & acoustic channels with negligible delay of 280µs; (b) output with delay of 1.40ms.

4. CONCLUSIONS

CCi-MOBILE, developed as a research platform to support the CI/HA community is being made open-source, and has the capability to support hearing research labs which are unable to access industry based proprietary test devices, allowing them to implement novel ideas and explore perceptual hearing science for the hearing-impaired community. Most available research platforms only support either CI or HA, and do not support bimodal processing of CI and HA algorithms. Also, as research in the field of advanced hearing solutions including machine learning (ML) based algorithm development for HA/CI advance, it is critical to be able to transition lab based testing to real-world naturalistic field testing. CCi-MOBILE allows for take-home/extended out of the lab testing for either scientific hypothesis testing in hearing research as well as algorithm field evaluations in naturalistic spaces. The algorithm processing pipeline is different for CI vs HA needs, and only limited research has been done to support audiologists who need fitting options for bimodal CI users, or Bilateral HA users with transition to one side being CI. CCi-MOBILE has been designed to address this void by assisting research for bimodal algorithms, validating efficiency, and promoting usage of bimodal HA/CIs in the community. This work shows that while left and right channels on CCi-MOBILE are processed independently, synchronization of input L+R audio streams is maintained at the output, demonstrating a reliable device for any valid set of user configurations. The platform also is capable of transmitting DoA information of the source, making it ideal for sound localization type experiments. Finally, future advancements of CCi-MOBILE will continue to support those in the scientific research field exploring hearing aid (HA) technology advancements, as well as those bridging the gap between HA and cochlear implant (CI) research, both for bimodal operation to support HA+CI processing separately in each ear, and ultimately bilateral for dual CI processing in both ears.

7. REFERENCES

[1] J.H.L. Hansen, H. Ali, J.N. Saba, R. Charan-Shakar, N. Mamun, R. Ghosh, A. Brueggeman, "CCi-MOBILE: Design and Evaluation of a Cochlear Implant and Hearing Aid Research Platform for Speech Scientists and Engineers," IEEE EMBS BHI-19/BSN-19: Conf. Wearable and Implantable Body Sensor Networks, Chicago, IL, May 19-22, 2019.

[2] G. Clark, “Cochlear Implants,” in Speech Processing in the Auditory System, Springer-Verlag, 2004. doi: 10.1007/0-387-21575-1_8.

[3] M. Luntz, T. Shpak, and H. Weiss, “Binaural–bimodal hearing: Concomitant use of a unilateral cochlear implant and a contralateral hearing aid,” Acta Oto-Laryngologica, vol. 125, no. 8, Jan. 2005. (DOI: 10.1080/00016480510035395).

[4] A.C. Neuman, M.A. Svirsky, “Effect of Hearing Aid Bandwidth on Speech Recognition Performance of Listeners Using a Cochlear Implant and Contralateral Hearing Aid (Bimodal Hearing),” Ear & Hearing, vol.34, Sept. 2013 (DOI: 10.1097/AUD.0b013e31828e86e8)

[5] T. Francart and H.J. McDermott, “Psychophysics, Fitting, and Signal Processing for Combined Hearing Aid and Cochlear Implant Stimulation,” *Ear & Hearing*, vol. 34, no. 6, Nov. 2013. (DOI: 10.1097/AUD.0b013e31829d14cb).

[6] R.H. Gifford, “Bimodal Hearing: How to Optimize Bimodal Fitting,” *The Hearing Journal*, vol. 72, no. 2, pp. 10–13, Feb. 2019.

[7] R. Ghosh, H. Ali, and J.H.L. Hansen, “CCi-MOBILE: A Portable Real Time Speech Processing Platform for Cochlear Implant and Hearing Research,” IEEE Transactions on Biomedical Engineering, vol. 69, pp. 1251-1263, Mar. 2022. (DOI: 10.1109/TBME.2021.3123241).

[8] J.M. Kates, “Modeling the effects of single-microphone noise-suppression,” Speech Communication, vol. 90, Jun. 2017. (DOI: 10.1016/j.specom.2017.04.004).

[9] K.L. Plant, L.A. Whitford, C.E. Psarros, and A.E. Vandali, “Parameter selection and programming recommendations for the ACE and CIS speech-processing strategies in the Nucleus 24 cochlear implant system,” Cochlear Implants International, vol. 3, no. 2, Sep. 2002. (DOI: 10.1179/cim.2002.3.2.104).

[10] R. Ghosh, J.H.L. Hansen, "Bilateral Cochlear Implant Processing of Coding Strategies with CCi-MOBILE, an Open-Source Research Platform," IEEE Trans. Audio, Speech, and Language Processing, vol, 31, pp. 1839-1850, Apr. 2023. (DOI: 10.1109/TASLP.2023.3267608)

[11] Y. Chung, J.H.L. Hansen, "Compensation of SNR and noise type mismatch using an environmental sniffing based speech recognition solution," EURASIP Journal Audio, Speech, and Music Processing, Vol. 2013, pp.1-14, June 2013. (DOI: 10.1186/1687-4722-2013-12)

[12] M. Akbacak, J.H.L. Hansen, "Environmental Sniffing: Noise Knowledge Estimation for Robust Speech Systems," IEEE Trans. Audio, Speech and Language Processing, vol. 15, no. 2, pp. 465-477, Feb. 2007 (DOI: 10.1109/TASL.2006.881694)

[13] M. Akbacak, J.H.L. Hansen, "Environmental Sniffing: Noise Knowledge Estimation for Robust Speech Systems," IEEE ICASSP-2003: Inter. Conf. Acoust. Speech & Signal Proc., vol. 2, pp. 113-116, Hong Kong, China, April 2003.

[14] R.C.M. Chandra Shekar, J.H.L. Hansen, "A CNN-based framework for analysis and assessment of non-linguistic sound classification and enhancement for normal hearing and cochlear implant listeners," Journal of Acoustical Society of America, vol, 152(5), pp. 2720-2734, Nov. 2022 (DOI: 10.1121/10.0014955).

[15] R. Chandra Shekar, J.H.L. Hansen, “CNN-based Comparative Framework for Non-Linguistic Sound Classification and Enhancement in Normal Hearing, Hearing Aid and Cochlear Implant conditions,” IHCON-2022: Inter. Hearing Aid Research Conference, Paper DP-110, Lake Tahoe, CA, Aug. 10-14, 2022