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CLABOX. Audiological Evaluation Tool for Remote Assessment of Cochlear Implants

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Abstract

Cochlear implants (CI) are now widely accepted as an effective treatment for profound deafness. Several commercial devices are currently available and technological resources are constantly updated. However, we still face difficulties in the standardization of evaluation of CI users, such as large variation in the acoustics of test environments, use of different types of speech recognition materials, inconsistent use of noise in speech tests, and absence of normality criteria. The CLABOX was developed in an attempt to standardize audiological tests and allow comparisons between cochlear implant user outcomes among centers and countries in Latin-America. The CLABOX is an audiological evaluation tool written in Matlab designed to provide a direct connection between the computer and the cochlear implant via a commercial audio interface (Presonus AudioBox iOne). This feature allows an evaluation condition independent of local acoustic conditions or the availability of good sound stimuli, as the CI external microphone can be turned off during tests and all audio files used for testing are pre-recorded. CLABOX offers the following sound stimuli: (1) pure tones to verify the CI calibration; (2) Ling sounds, which are language independent and can be used to evaluate detection and identification in both adults and children; and (3) sentences with or without additive noise to mimic HINT conditions (fixed or adaptive), which can be used to evaluate speech recognition skills. The CLABOX environment was compared to measurements in a sound booth. Fifty individuals with cochlear implants (CIs), 33 adults and 17 children, were included. For speech recognition in noise, averages were better in the CLABOX; for pure tone threshold, they were higher in the CLABOX; and for Ling sounds, there was no significant change in the categories between sound booth and CLABOX, suggesting that the CLABOX is an effective tool to assess pure tone and speech recognition tests.

Keywords: cochlear implant, remote assessment, audiometry, speech recognition.

PACS: 43.64.Me; 43.66.Sr; 43.71.-k.



1. INTRODUCTION

Cochlear implant is an outstanding device that offers the opportunity to hear for those who cannot benefit from conventional hearing aids. The number of adult cochlear implant (CI) users is increasing annually. This growth is speeding up because the population of CI candidates increases due to changing regulations, expanding candidacy criteria, and technical improvements in CIs [1].

However, despite the technological advances in implant technology and the benefits they provide, CI users still find it challenging to understand speech in noisy environments [2].

Speech recognition tests are the most important component to assess and monitor the outcomes of patients who have received cochlear implants (CI). The speech recognition test requires high quality free field loudspeakers and should be administered in a room with as little background noise as possible to reduce external influences in the test results [3, 4].

However, many clinics around the world do not have specialized equipment for assessment. This limits locations where these services can be provided, which can introduce barriers for provision of care. This is also the case in Brazil, with states that still do not perform CI surgery and with centers that do not yet have an acoustic booth and/or an acoustically treated room for pre- and postoperative CI assessments.

Providing speech test stimuli directly to the CI via a direct audio input (DAI) may offer a potential solution to overcome the negative effects of background noise and reverberation associated with test sites that lack an acoustic booth [1]. DAI deactivates the processor microphones and introduces the stimuli directly to the recipient, thereby eliminating reverberation and background noise effects from the external environment.

Based on the need for remote audiological evaluation, to help audiologists evaluate psychoacoustic data to guide programming and validation, Cochlear Corporation developed a new CI assessment tool called CLABOX. It consists of a presentation software installed in a regular per-

sonal computer (PC) and a commercial audio interface. The presentation software was written in MATLAB and is made of three evaluation modules: pure tone threshold audiometry, Ling sounds recognition test, and speech recognition tests in silence and in noise. The audio interface is used for direct connection with the CI via the DAI, therefore, the system can only be used on individuals with branded CI.

In this manuscript we first present the developed CLABOX system and further evaluate whether the results of pure tone audiometry and speech in noise recognition test conducted with the proposed system are equivalent to results obtained with the conventional method.

2. CLABOX

The CLABOX test suite consists of a commercial audio interface and a presentation software with accompanying graphical user interface (GUI). This section details the implementation and functioning of the CLABOX test suite.

2.1 Hardware

The audio interface selected for the CLABOX system was the Presonus AudioBox iOne. The external module connects with the PC via USB 2.0 audio interface. It provides 24 bit digital audio resolution at a sampling frequency of 44.1, 48, 88.2, and 96 kHz, being 44.1 kHz the sampling rate used within the CLABOX environment.

The stereo headphone output was used to connect the interface with the CI via DAI, as shown in Figure 1. This analog output has a flat frequency response (± 1.0 dB) in the range of 20 Hz to 30 kHz, with an output impedance of 10 Ω . The independent control for headphone level was mechanically fixed at maximum gain to avoid alterations during use. Furthermore, a hard case was fabricated to enclose the audio interface and avoid unwanted alterations in the adjusted settings, as depicted in Figure 1.

2.2 Software

The CLABOX presentation software was written in MATLAB with an accompanying graphical



Figure 1: Example of how to connect the CLABOX to the CI via DAI.

user interface (GUI), written with MATLAB's AppDesigner. The GUI has five tabs; a tab for the examiner to input the patient's data, a tab for the pure tone audiometry, a tab for the Ling test, a tab for the speech recognition test, and a tab for the examiner to adjust high level preferences.

The software was compiled using the MATLAB compiler as a standalone application. All the required audio files were bundled in the installation package that also included MATLAB runtime.

2.2.1 Personal information

After initializing the CLABOX standalone application, the GUI opened at a tab named "Patient", see Figure 2. In this tab the examiner must input the patient's name, age, and information about the cochlear implant. The GUI will not change to any other tab unless this information is completed.

In this tab the examiner has the option to save the current section, i.e., to generate a report with all the current information and also has the option to clear all recorded information and start a new blank section.

2.2.2 Pure tone audiometry

The following tab of the presentation software is named "Tone", as shown in Figure 3, and allows the examiner to perform a pure tone audiometry. The examiner can choose between the frequencies of 250 Hz, 500 Hz, 1 kHz, 2 kHz, 3 kHz, 4 kHz and 6 kHz with sound intensity, measured in decibels relative to hearing threshold (dB_{HL}), varying from 0 dB_{HL} to 90 dB_{HL} in steps of 5 dB.

Each presentation consists of a sequence of four tone bursts, with duration of 500 ms, including a fade in window of 50 ms and a fade out window of 50 ms. Between two consecutive tone bursts there was 200 ms of silence.

To obtain correct hearing levels, the digital levels used in the software, measured in decibels relative to full scale (dB_{FS}) levels had to be properly converted to decibels relative to hearing threshold. To do so, we verified that the Presonus iOne produces an electric output level of 5.5 dB_{V} when excited with a tone of 0 dB_{FS} and the headphone output level was set at its maximum. Moreover, we were informed by Cochlear Corporation that the processor CP900 produces a sensation equivalent to 78 dB_{SPL} when excited with a pure tone of 1 kHz and -48 dB_{V} . Therefore, considering the whole system to be linear, the loudest 1 kHz pure tone generated by the computer will result in a perception equivalent to that generated by a 127.5 dB_{SPL} acoustic tone inciding at the CI's microphone. The conversion from sound pressure level (dB_{SPL}) to hearing level (dB_{HL}) was made following ANSI/ASA S3.6-2010 specification for tones in sound field.

If the patient is a bilateral CI user, the examiner can select between left and right presentation. After selecting the desired frequency and level, the examiner has to click the "Stimulate" button to present the stimulus. The last level presented for each ear and each frequency is considered to be the tonal hearing threshold. It is plotted at the GUI and recorded for inclusion in the test report.

2.2.3 Ling test

The next tab, named "Ling", is shown in Figure 4 and provides an interface for the examiner to conduct the Ling Sounds Test [5], which presents familiar speech sounds that broadly represent the speech spectrum from roughly 250 to 8000 Hz, testing approximately the same spectral range as a conventional audiometry. The phonemes for the Ling Six Sound Test are [m], [a], [i], [u], [f] and [s], selected to target low, middle and high frequency sounds. Sound files used in the CLABOX suite were provided by Cochlear Latam.

The examiner can select the level the signals



Figure 2: Example of the “Patient” tab, where the examiner must input the patient’s name, age, and information about the cochlear implant.

should be presented in A-weighted decibels relative to sound pressure level (dB_A). Standard value is 65 dB_A , in the loudness range of a normal conversation.

For each ear and each of the Ling sounds, the examiner can check two boxes: “Detection” if the patient was able to hear that sound, and “Identification” if the patient was able to recognize that sound. The examiner can also annotate remarks about that patient in a text box.

2.2.4 Speech recognition tests

The next tab, named “Logo” and depicted at Figure 5, provides an interface for the examiner to conduct the different types of assessment with verbal stimuli, namely: Speech Recognition Index (SRI), Speech Recognition Threshold (SRT), or Adaptive Signal-to-Noise Ratio (SNR).

The GUI allows the examiner to select between the different test modes, and enter the necessary test parameters. Only the parameters required for each type of test are available for selection in

the GUI.

It is also possible to choose between Portuguese and Spanish as the language of the test sentences. The software will randomly choose a list of sentences in the chosen language. If required, the examiner can also select a specific list of sentences from the database. It is also possible to select if the sentences within each list will be presented in random or sequential order. Sound files in both languages were provided by Cochlear Latam.

After selecting the test type and defining all parameters, the examiner must “Start” the test. The excitation signal will be presented to the patient every time the examiner presses “Stimulate”. It is the examiner’s responsibility to count the number of correct words the patient was able to repeat and press either “Correct” or “Wrong”. To help with this task, the GUI shows the transcription of the current sentence and its total number of words. The answer is added to the plot and is used to determine the next presentation level and, when sufficient input is available, also to cal-

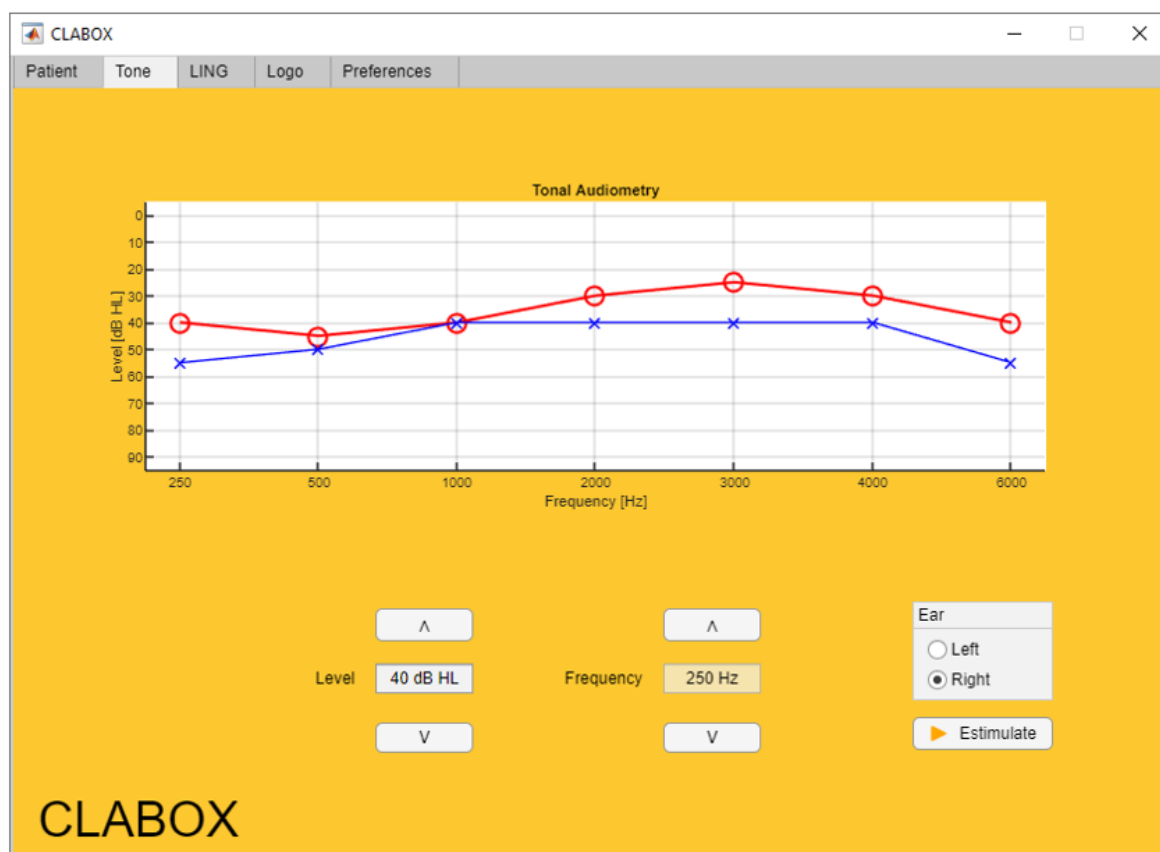


Figure 3: Example of the “Tone” tab, where the examiner can present pure tones at six different frequencies at different hearing levels to verify the patients listening threshold.

culate the SRI, SRT or SNR. The trial can be terminated at any moment by pressing “Stop”.

2.2.4.1 SRI

The SRI test assesses speech discrimination at a fixed level defined by the examiner (standard value is 65 dB_A). It corresponds to the Word Recognition Score (WRS) commonly performed at the audiological clinic. The result obtained will be a percentage of correct answers out of the total number of answers.

The SRI can be administered with or without additive noise. In case additive noise is selected, the examiner needs also to select the signal-to-noise ratio, thus indirectly choosing the noise level, which remains constant throughout the assessment. The signal used as noise is a pre-generated speech shaped Gaussian white noise, limited to 11 kHz. The noise signal starts 50 ms before the target signal and continues by another 50 ms after the target signal ends.

2.2.4.2 SRT

The SRT, as the name says, determines the speech recognition threshold, i.e., the lowest level at which the patient is able to still recognize the words in a sentence. As the SRI, the SRT is also routinely performed at the audiological clinic.

The SRT test will start presenting the sentences at an examiner-defined level (standard value is 65 dB_A). The level of the subsequent sentence is adjusted adaptively depending on the listener’s response to the previous sentence. If the patient is able to correctly repeat a number of words greater than a predefined percentage (usually 50 % correct words), the examiner scores the sentence as correct, and the amplitude of the next sentence is reduced. On the other hand, if the patient is unable to meet the previous criteria, the examiner scores the sentence as wrong, and the amplitude of the next sentence is increased. The levels are adjusted in a *Large Step* (standard value is 4 dB) until the first reversal in performance is recorded and in a *Small Step* (standard



Figure 4: Example of the “Ling” tab, where the examiner can present the six Ling sounds and mark if the patient detected or identified the sound.

value is 2 dB) thereafter.

The independent midpoint is calculated as the average of the level of a positive-going reversal and the level of the succeeding negative-going reversal. The result obtained will be the average of the independent midpoints, presented in A-weighted decibels (dB_A).

The examiner can choose to administer the SRT with or without additive noise. In the case additive noise is selected, the examiner needs to determine the initial SNR. The noise signal is the same as in the SRI mode and it remains constant throughout the assessment.

2.2.4.3 Adaptive SNR

The Adaptive SNR test assesses what is the signal-to-noise ratio at which the patient is able to understand a predefined percentage (usually 50 % correct words) of the presented stimuli. Thus, the presence of noise is necessary. In this mode the examiner can select to vary the speech level while keeping the noise level constant, or

to vary the noise level while keeping the speech level constant. Note that when choosing to vary the speech level, this exam type is the same as the SRT mode with additive noise.

However, other than the SRT test, in the Adaptive SNR mode the examiner needs to define the level of the stimulus that will remain constant, be it speech or noise (standard value is 65 dB_A). The examiner will also need to define the initial SNR (standard value is 10 dB), the large step (standard value is 4 dB), and the small step (standard value is 2 dB). Therefore, if the test is programmed to speech, with a noise level of 60 dB_A and an initial SNR of 10 dB, then the initial speech level will be coming out at 70 dB_A.

The independent midpoints are calculated in the same way as in the SRT mode, so as their average. As a result, the Adaptive SNR test establishes the SNR value in dB in which the patient is able to understand a predefined percentage of the words in presented sentences.



Figure 5: Example of the “Logo” tab, where the examiner can choose between several speech recognition test, adjust their parameters, and test the subject.

2.2.5 Preferences

The last available tab is named “Preferences” and shown in Figure 6. It allows the examiner to choose the folder where to save the reports, which are generated automatically every time the interface is closed or the examiner presses “Save” or “New” in the “Patient” tab. The examiner can also force the program to generate a report by pressing “Generate Report”.

The application connects to the audio interface through the playrec utility [6], which buffers all audio samples so Matlab can continue with other processing tasks. The connection with the Presonus iOne is established via the ASIO host API. This is a very stable way of presenting audio with Matlab and very seldomly does it malfunction. However, if that was the case, the operator can reset the connection with the audio interface by pressing “Reset Audio”.

3. CLABOX EVALUATION

To evaluate the CLABOX suite, we compared the results of auditory assessment and speech recognition conducted with CLABOX to conventional clinical assessment in a sound booth. The study was analyzed and approved by the Research Ethics Committee of the Faculty of Health Sciences, University of Brasília, under protocol 4,327,050, and all participants and/or guardians consented to participate in the research.

3.1 Participants

Participation in the study included 50 individuals, 33 adults and 17 children, aged from 8 years old, 36 females and 14 males, 38 with prelingual and 12 postlingual hearing loss, 15 bilateral CI users and 35 with unilateral, total of 63 evaluated ears, with a minimum of 6 months of CI use, all Cochlear Corporation brand users.

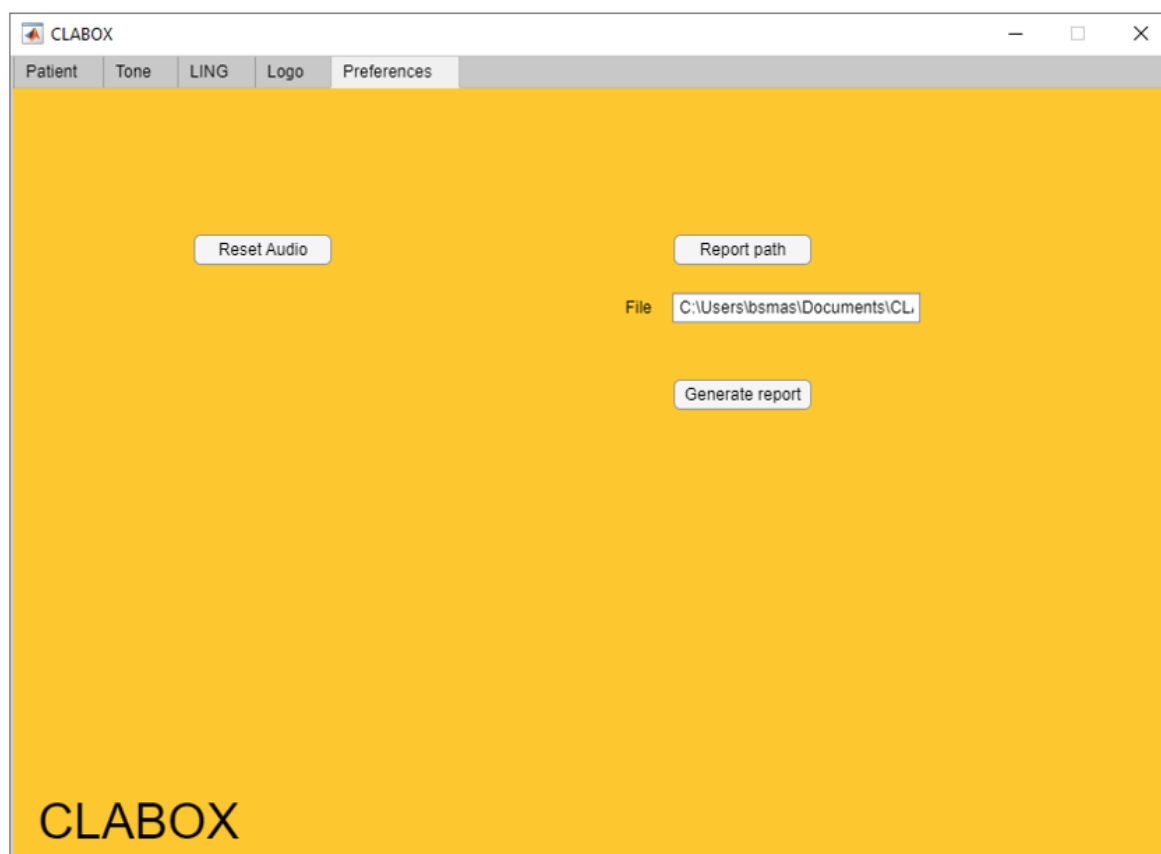


Figure 6: Example of the “Preferences” tab, where the examiner can choose the folder where to save the report, force the program to generate a report and reset the connection with the audio interface.

3.2 Design and Procedure

Speech recognition tests with additive noise and audiometric thresholds were evaluated in the conventional condition (sound booth) and in the CLABOX condition (with the DAI connection).

In both scenarios, audiometric thresholds were investigated at frequencies of 250, 500, 1000, 2000, 3000, 4000 and 6000 Hz with pulsatile tone (see Section 2.2.2). For speech recognition evaluation, two conditions were tested: SRI with additive noise and SRT with additive noise. For the SRI, the noise level was kept fixed at 55 dB_A and the speech level was kept fixed at 65 dB_A, thus with a constant SNR of noise level was kept fixed at 10 dB. For the SRT, the noise level was kept fixed at 55 dB_A and the speech level was adaptively adjusted (large step of 4 dB and small steps of 2 dB), with an initial value of 65 dB_A.

To investigate the degree of difficulty associated with the two systems, after performing each of the tests, the participants were asked to indicate, on an increasing scale from 0 to 10, how difficult

it was to perform the test. This was done with the aid of a visual analogue scale (VAS).

3.2.1 Settings for sound booth

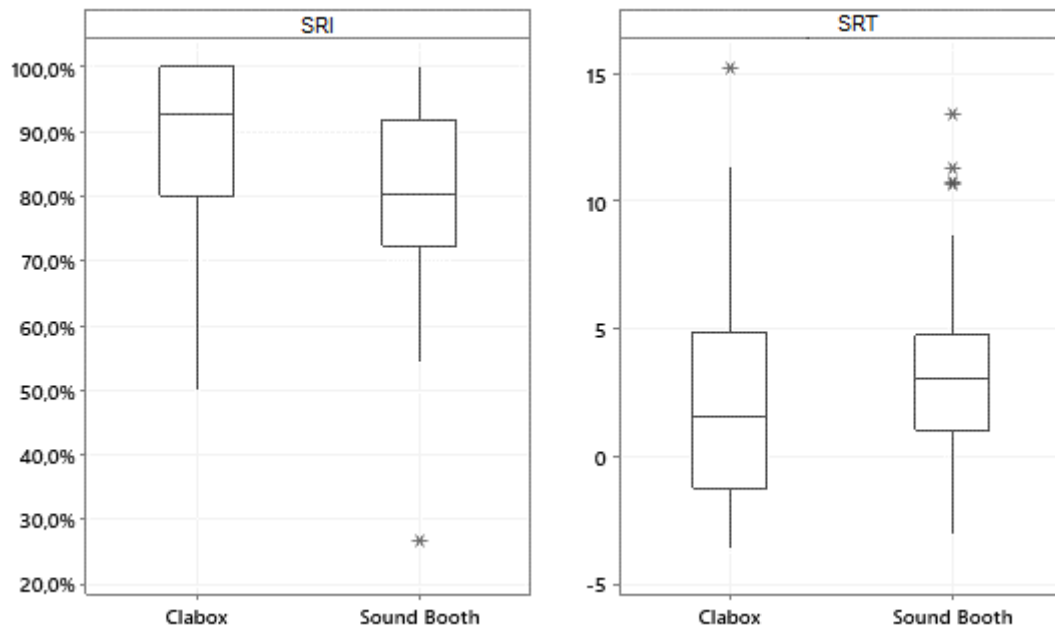
A MADSEN-Itera II audiometer was used for the evaluation in the sound booth. Furthermore, the speech processing algorithm of the CI was deactivated, replicating natural hearing. All tests were performed at 0° azimuth, one metre from the loudspeaker.

3.2.2 Settings for CLABOX

The CLABOX suite was installed on a conventional computer, as described in Section 2. All participants were tested with a plain CP910 sound processor, with all specific programs and features turned off. The DAI connection was set, so the participant heard only the test signals and no ambient sounds [1, 7].

Table 1: Comparison of speech perception evaluation results in the sound booth and CLABOX.

		Average	Median	Standard deviation	Q1	Q3	N	CI	P-value
SRI	Sound booth	78,9%	80,3%	15,3%	72,5%	91,8%	40	4,7%	<0,001
	CLABOX	88,3%	92,5%	13,4%	80,0%	100%	40	4,2%	
SRT	Sound booth	3,42	3,03	3,60	1,04	4,65	40	1,12	0,007
	CLABOX	2,14	1,57	4,21	-1,00	4,46	40	1,31	


Figure 7: Boxplot with the results of speech recognition evaluation results in the sound booth and CLABOX.

4. RESULTS

The comparison of the systems in regards to speech recognition is shown in Table 1 and Figure 7. Differences between the systems were observed in both test modes: SRI and SRT. The average results were better in CLABOX than in the acoustic booth. For the SRI with additive noise condition, the number of correct words in CLABOX was significantly ($p\text{-value} < 0.001$) higher than scored in the sound booth, 88.3% versus 78.9%, respectively. In SRT with additive noise condition, the average SNR that still allows speech recognition was significantly ($p\text{-value} = 0.007$) lower in the CLABOX than in the sound booth, 2.14 dB versus 3.42 dB, respectively.

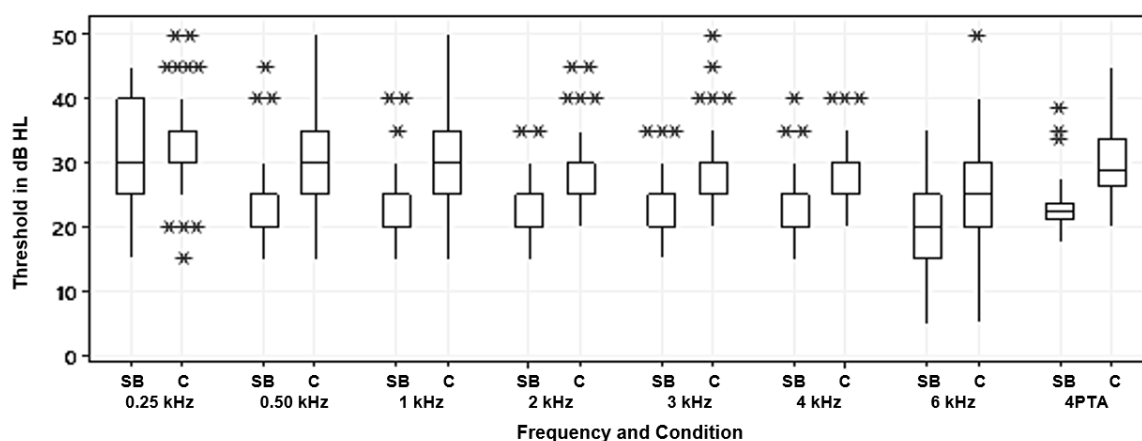
The systems were also compared using the results of the tonal audiometry in each of the tested frequencies and with the 4PTA (average of 0.50, 1, 2, and 4 kHz). The results are presented in Table 2 and Figure 8. There was a significant

difference ($p\text{-value} < 0.001$) between both systems at all frequencies, except for the 250 Hz, as shown in Table 2. For all tested frequencies the average tonal threshold of hearing was higher (worse) with the CLABOX than with the acoustic booth. The difference between hearing threshold measured with CLABOX and sound booth was, on average, 6 dB.

The perceived difficulty when performing the test, both for speech recognition as well as for the tonal audiometry, when using the acoustic booth and the CLABOX were compared with the VAS. The results are presented in Table 3 and Figure 9. Although CLABOX presented the highest average VAS, that is, participants refer greater effort when completing the text in this presentation mode, there was no difference in VAS for the two systems.

**Table 2:** Comparison of tonal audiometry results in the sound booth and CLABOX.

		Average	Median	Standard deviation	Q1	Q3	N	CI	P-value
250 Hz	Sound booth	30,8	30,0	7,9	25,0	37,5	63	1,9	0,188
	CLABOX	32,0	30,0	7,0	30,0	35,0	63	1,7	
500 Hz	Sound booth	24,7	25,0	5,9	20,0	25,0	63	1,5	<0,001
	CLABOX	32,2	30,0	8,2	25,0	35,0	63	2,0	
1 kHz	Sound booth	24,0	25,0	5,1	20,0	25,0	63	1,3	<0,001
	CLABOX	30,9	30,0	7,1	25,0	35,0	63	1,8	
2 kHz	Sound booth	21,7	20,0	4,5	20,0	25,0	63	1,1	<0,001
	CLABOX	28,9	30,0	5,9	25,0	30,0	63	1,5	
3 kHz	Sound booth	22,2	20,0	4,8	20,0	25,0	63	1,2	<0,001
	CLABOX	28,4	25,0	6,1	25,0	30,0	63	1,5	
4 kHz	Sound booth	21,7	20,0	5,0	20,0	25,0	63	1,2	<0,001
	CLABOX	27,1	25,0	4,8	25,0	30,0	63	1,2	
6 kHz	Sound booth	20,3	20,0	5,5	15,0	25,0	63	1,3	<0,001
	CLABOX	24,9	25,0	6,3	20,0	27,5	63	1,6	
Four-Tone average	Sound booth	23,0	22,5	3,9	21,3	23,8	63	1,0	<0,001
	CLABOX	29,8	28,8	5,7	26,3	33,8	63	1,4	

**Figure 8:** Boxplot with the results of tonal audiometry in the sound booth and CLABOX.

5. DISCUSSION

In this study we compared the performance of pure tone audiometry and speech recognition tests conducted in free field condition (using an acoustic booth) to the novel CLABOX test suit developed for use with the direct audio input (DAI) present in Cochlear's CP910 audio processor.

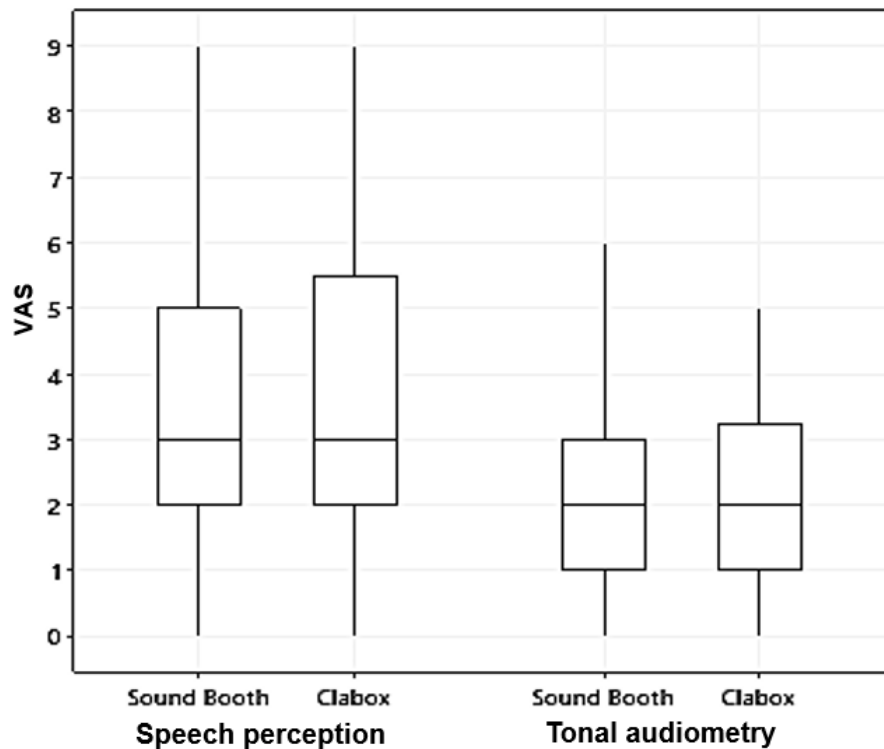
Accessing hearing health services in remote or resource-limited settings is a pervasive global challenge. Innovative hearing assessment models are necessary for offering better quality hearing monitoring services [8]. The literature indicates that auditory assessment with the DAI connec-

tion alleviates the technical requirements present for the free field condition, such as quality of loudspeakers, low level of background noise and low reverberation time, which are commonly not met in domestic environments, and can interfere with the results of audiological tests [1, 4].

More recently, Sevier et. al. compared speech tests conducted remotely (telepractice) with the aid of DAI versus the traditional method in an acoustic booth in person. Their results showed no difference in scores between remote and face-to-face conditions, thus suggesting that the tests with the DAI system can be used as a valid assessment alternative [7]. However, the authors alert that for this technology to arrive at clinical

Table 3: Comparison of perceived degree of difficulty associated with test in the sound booth and CLABOX.

		Average	Median	Standard deviation	Q1	Q3	N	CI	P-value
Speech perception	Sound booth	3,83	3,00	2,13	2,00	5,00	81	0,46	0,773
	CLABOX	3,84	3,00	2,27	2,00	5,00	81	0,49	
Tonal audiometry	Sound booth	2,26	2,00	1,61	1,00	3,00	38	0,51	0,372
	CLABOX	2,50	2,00	1,35	1,25	3,00	38	0,43	


Figure 9: Boxplot with the perceived degree of difficulty associated with test in the sound booth and CLABOX.

implementation, there is still a need for commercial development of calibration tools for stimuli delivered via DAI.

In contrast to the results in [7], all speech recognition tests conducted in our study presented a significant difference between CLABOX and sound booth.

Furthermore, in our study, participants consistently performed better when tested in DAI than in free field condition, partly contradicting the results of Sevier et al. with Cochlear devices. In this study, performance was generally worse in the DAI condition than in free field for Cochlear brand users, while the performance was generally better in the DAI condition than in person for participants using Advanced Bionics devices [7].

In the tonal audiometry test there was a signif-

icant difference between the systems in practically all frequencies evaluated, except for 250 Hz. For all tested frequencies, the CLABOX presented a higher level for the measured threshold of hearing when compared to acoustic booth. In the range from 500 to 6 kHz, the observed difference was, on average, of 6.2 dB.

This difference could have been caused by an inaccuracy in the calibration procedure of the CLABOX. Another reason for that could be the difference in frequency response that Cochlear processors exhibit between DAI connector versus microphone input [1, 7]. It may be that some people with CIs are more sensitive to this difference in frequency responses. The spectrum seen at the sound processor for a given sound delivered from speakers in the sound field to the processor microphones will differ to a certain extent



compared with the same sound delivered to the processor via DAI. In the processor design, the microphone response and the pre-emphasis filter were carefully calibrated for maximum hearing performance, but the DAI utilizes a basic first-order filter in the cable to approximate this frequency shaping without additional filtering.

The use of the VAS was chosen to quantify the subjective auditory difficulty of the participants for each of the two evaluated systems. The VAS has been previously used in CI research and it has been reported that it is a useful and sensitive method to quickly assess the listening effort perceived by hearing device users [9]. The results showed there was no difference in auditory perception between both systems. It is worth mentioning that, even though we did find a difference in the results of auditory thresholds and speech recognition, the participants did not report any difference in the listening effort required to conduct these tests in DAI or sound booth conditions.

6. CONCLUSIONS

In this study we presented a new tool for evaluating CI patients over the direct audio input. The presented system is easy to handle, light and small, can be used on a table, and could be used in centres that do not have an acoustic booth and free field for evaluation system, a common situation in a continental and developing country such as Brazil. The CLABOX can support telepractice, allowing the practitioner to remotely collect data on the CI user, evaluate its performance in pure tone audiometry and speech perception tests, and possibly offer subsidies for improvement.

The results showed that the CLABOX is an effective tool to evaluate the pure tone and speech recognition tests when compared to the conventional evaluation with an acoustic booth in the CI user population. Some adjustments to the software are necessary in order to guarantee equal performance in comparison with the standard clinical test and, thus, become a valid assessment tool in clinical routine.

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