

Design and Implementation of Software Based Audiometer System

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Abstract—This paper proposes design and implementation of software based audiometer using LabVIEW. In this study, basic definition and test procedure of audiometer are explained. The implementation of audiometer is realized by LabVIEW. The most preferred clinical audiometer method (Modified Hughson-Westlake) is selected for study. The test procedure and audiometer implementation are explained by structure of binary case system. The ambient noise level is determined by decibel meter used in calibration. Besides, audiometer calibration process is performed for successful and realistic results. The calibration process is implemented by polynomial fitting. Furthermore, configurable white noise is used for minimizing interference between active and passive sound channel on duration of audiometer test. As a result, this paper achieves implementing a pure tone audiometer based LabVIEW with channel masking.

Index Terms—audiometer, LabVIEW, audio signal processing, biomedical equipment, medical signal processing

I. INTRODUCTION

Audiometer system determines hear loss for specific frequency and amplitude values. Ability of hearing on humans can be change by the time. The usage of earphone causes reducing hearing ability. Audiometer system determines this hearing loss ratio [1]-[3]. Frequency hearing ability on a healthy human can change nearly between 16 Hz to 20 kHz [4]. Human can barely sense sound signal lower than 16 Hz. However, human never sense sound signal higher than 20 kHz. This range represents to human hearing range [5]. Some of the experimental or clinical works are out of this hearing range. Audiometer system is designed for 0 dB to 80 dB and 250 Hz to 8000 Hz [6]. Hearing range can change by environment, age, some kind of hearing diseases such as bacteria, genetic etc. [7]. Audiometer test procedure should be implemented in very quiet room which is isolated from environment noises. The process should begin from the lowest sound frequency and amplitude value which can be increased step by step (5-10 dB). Result of traditional audiometer test is shown in Fig. 1.

In literature, LabVIEW software has been used for medical technologies [8], [9], however there are fewer studies on audiometer systems. Tan *et al.* designed an audiometer prototype to test speech signal and pure tone hearing [10]. Vencovsky *et al.* designed a Graphical User

Interface (GUI) for pure tone audiometer via MATLAB. In their study, the white noise was not used to minimize interference effect on passive sound channel [11]. Voon *et al.* developed a hearing loss calibration system by using LabVIEW [12]. The aim of their study was to calibrate audiometer system.

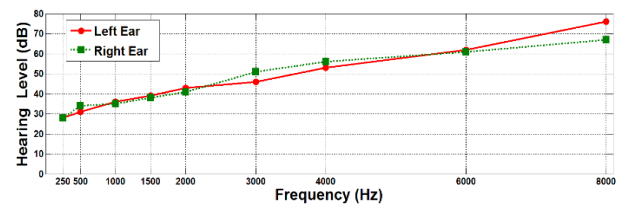


Figure 1. Example result of audiometer for two channels

In this study, LabVIEW software is used to develop a GUI for audiometer test. It is added the white noise to de-active sound channel for hearing nothing. In the developed GUI, the user can configure the amplitude of white noise, frequency, and amplitude of pure tone sound signal. The generated specifications of sound are between 0 dB-80 dB and 250 Hz-8000 Hz as in the literature. The audiometer test results can be seen in GUI screen.

II. SOFTWARE BASED AUDIOMETER DESIGN

Software based audiometer design consists of many steps. At first step, audiometer system is developed by using LabVIEW and then, pure tone sound signal is created. The user can change the amplitude and frequency values via slider button on LabVIEW interface. Sample frequency of pure tone is 44100 Hz for this experiment [13]. Furthermore, the user sets time of test process with slider button. This process can be changed between 0 and 2 second. Initial condition before test process is started at minimum sound amplitude level. In this study, test process starts from 250 Hz for minimum frequency value and 50 dB for amplitude level. The decreasing value of sound pressure is 5 dB. The last point of the generated amplitude of sound hearing by patient gives amplitude threshold value for specific frequency level. The different frequency values for test process are used in the developed interface [14], [15]. These frequency steps are selected as commercial audiometer devices.

III. IMPLEMENTATION OF SOFTWARE BASED AUDIOMETER

Modified Hughson-Westlake method is used in this study. There are several methods for audiometer test

procedure. Modified Hughson-Westlake is one of the most useful and successful methods on this area [16]. 125 Hz for audiometer test is not preferred by some clinicians unless there is a specific reason. The most clinicians use 1000 Hz for initial test frequency and change sound amplitude values with a 5 dB step. The modified Hughson-Westlake method starts from 60 dB and 1000 Hz values. However, the opinion of some scientists is that 60 dB values disturb some of the healthy people [17]. Thus, they suggest starting the test from 30 dB value. At the end of audiometer test procedure, the user can see own result of hearing ability. In this paper, audiometer design is started from 250 Hz and 30 dB for initial frequency value and amplitude as in [7]. In the first step, the user configures time interval of the generated sound signal. A channel desired by the user is selected to test. After selection of channel, the user can add white noise for passive sound channel and configure amplitude of white noise. Then, the test procedure is started and the results obtained from developed algorithm are saved and plotted in the GUI screen.

IV. LABVIEW BASED AUDIOMETER

LabVIEW which is a system or design platform develops environment for a visual programming language [18]. LabVIEW is generally utilized for data acquisition, instrument control, signal process, and industrial automation on a variety of platforms including Microsoft Windows, various versions of UNIX, Linux, and Mac OS X. In LABVIEW, two different interfaces are used. The first interface is Front Panel, second one is Block Diagram. The Front Panel is a GUI which related with screen for users. The user can modify data with block diagram. Furthermore, the user can select sound channel and addition white noise via case structure blocks. The case structure block performs different situations related with input. In this study, eight different cases are defined for audiometer design. These cases are shown at Table I.

TABLE I. CASE STRUCTURE

| Cases | Left | Right | Masking | Result (Output) |
|-------|------|-------|---------|--|
| 0 | 0 | 0 | 0 | No sound |
| 1 | 0 | 0 | 1 | Left and right channel are masking |
| 2 | 0 | 1 | 0 | Right channel is active |
| 3 | 0 | 1 | 1 | Right channel is active Left channel is masking |
| 4 | 1 | 0 | 0 | Left channel is active |
| 5 | 1 | 0 | 1 | Left channel is active Right channel is masking |
| 6 | 1 | 1 | 0 | Left channel is active Right channel is active |
| 7 | 1 | 1 | 1 | Left and right channel are masking |

For example, in case the user selects left channel and masking button, the system gives pure sound signal for left channel and at the same time it gives white noise signal to right sound channel. These structures are coded in base of binary. The case 6 and 7 are not meaningful as

shown in Table I since masking of left and right channels are pointless simultaneously. The audiometer system has to save previous results for output graphs. The users can determine input of process or observe input or output variables instantly in this panel. In the developed algorithm, sound signal frequency and amplitude, test process time, selection of sound channel, and hear check button for specific sound channel are set by front panel. Feedback node is used for saving previous results and it is related with sound channel check button. Once the user hears a voice and pushes the check button, feedback node becomes active and last result is saved to array. The user can save frequency and amplitude results via this structure and plot these values related with each other. This graph shows the audiometer results. The user can also see own result dynamically on front panel.

V. ENVIRONMENT CALIBRATION OF AUDIOMETER

Basic description of sound is a pressure magnitude changing based on the time. Sound pressure unit is PASCAL (Pa). Sound level L_p is represented by SPL. This is a basic term in acoustic science. Calculation of this term is shown in (1). P_{ref} is the reference sound pressure and P_{rms} is the root mean square of sound pressure being measured [19].

$$L_p = 10 \log_{10} \left(\frac{p_{rms}^2}{p_{ref}^2} \right) = 20 \log_{10} \left(\frac{p_{rms}}{p_{ref}} \right) \quad (1)$$

Audiometer test process should be performed in a very quiet room. Average noise levels are shown in Table II for different frequency values. These noise levels are convenient for audiometer test process [20].

TABLE II. FREQUENCIES AND AMBIENT NOISE LEVELS

| Frequency (Hz) | SPL (dB) |
|----------------|----------|
| 250 | 35.16 |
| 500 | 36.10 |
| 1000 | 40.24 |
| 1500 | 37.82 |
| 2000 | 42.77 |
| 3000 | 44.61 |
| 4000 | 48.55 |
| 6000 | 47.22 |
| 8000 | 44.69 |

There may be a difference between created sound signal and expected sound signal amplitude. These amplitudes must be synchronized with each other's. The decibel meter is used for this synchronization in this work. The audiometer system has to minimize this difference for accuracy of test. Polynomial function is used for minimizing error between measured amplitude and reference amplitude [21].

The selection of first order polynomial function is suitable for reducing error [22]. This polynomial equation is shown in (2) for our test room.

$$p(x) = 0.9884x + 0.1423 \quad (2)$$

The expected sound level and the measured sound levels are shown in Fig. 2. There are minor differences between each other. The mean squared error is 0.1958 dB for 16 measurement steps [23].

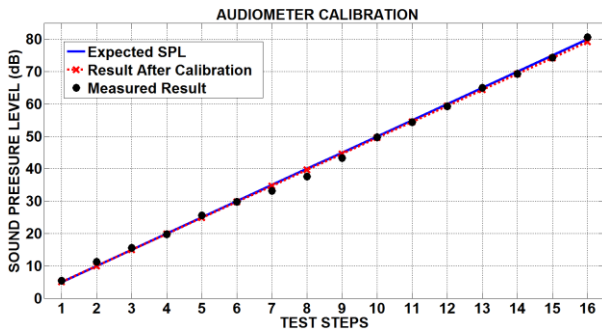


Figure 2. Expected SPL and measured SPL

VI. COMPUTER BASED AUDIOMETER USER INTERFACE

Audiometer test process of user interface is illustrated in Fig. 3. The aim of this user interface design is to provide implementation process of audiometer system for clinical works [24], [25]. The contents of software based audiometer are given as follows

- Introduction manual
- File path for saving result
- Timer value (0 sec-2 sec)
- Select sound channel (left or right)
- Add a white noise for deactivate channel and configure amplitude
- Left channel test sequence
- Right channel test sequence
- Test states
- Test result of left channel
- Test result of right channel
- Output signal including left, right channel and white noise signals.



Figure 3. User interface

The flowchart for computer based audiometer is shown in Fig. 4.

The user interface about white noise for inactive sound channel is shown in Fig. 5 and left sound channel test interface is shown in Fig. 6.

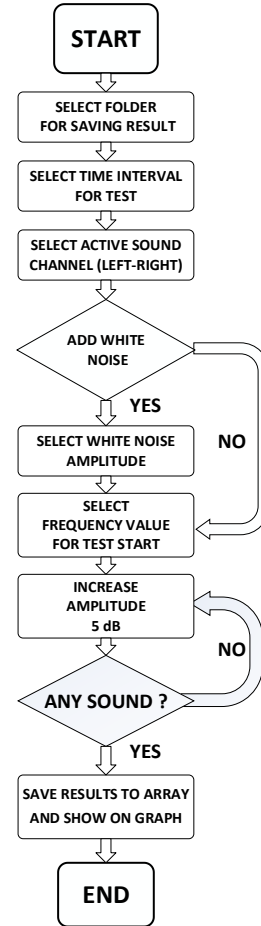


Figure 4. Computer based audiometer flowchart

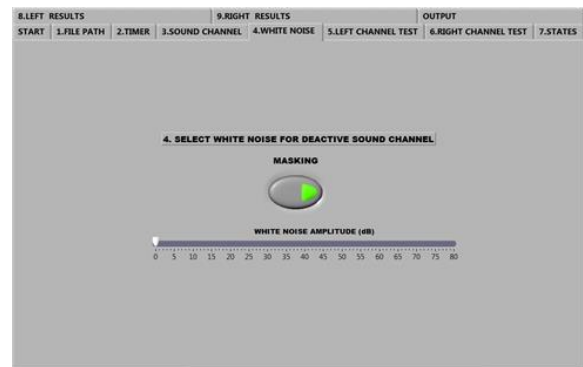


Figure 5. Interface of white noise

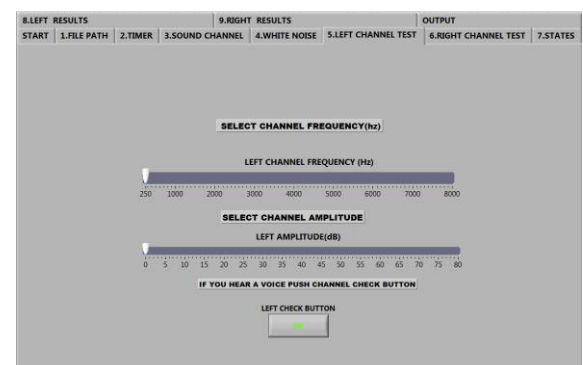


Figure 6. Left channel test interface

The software based audiometer experimental result is shown in Fig. 7. This audiometer has 5 dB resolutions for this application. Frequency resolution is easily configured by front panel for different values.

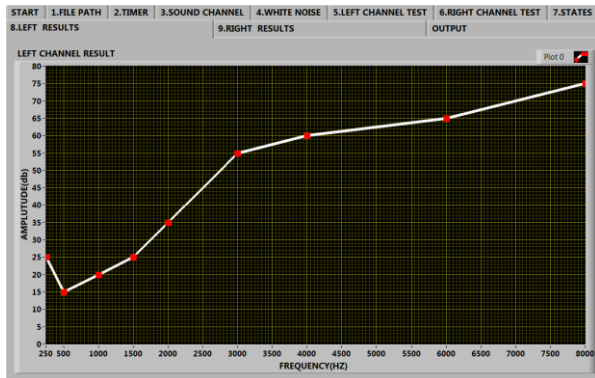


Figure 7. Left channel result

The block diagram of sound recording and playing is shown in Fig. 8.

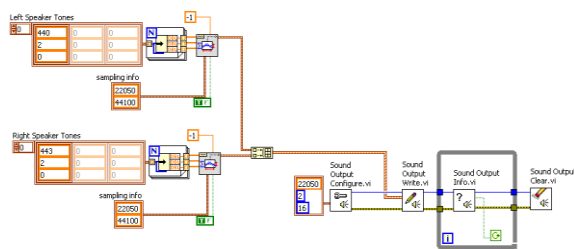


Figure 8. Sound playing and recording block diagrams



Figure 9. Output signal waveform

Output of signal is given in Fig. 9. In this figure, there are two signals. The first one is white noise which has 30 dB on right channel. The second is a sound signal on left channel and it has 30 dB amplitude value and 500 Hz frequency value.

VII. CONCLUSION AND FUTURE WORK

In this paper, a computer based audiometer system is designed on LabVIEW software. A sound level meter has been used for calibration process and nearly similar relationship between expected amplitude values and measured amplitude values has been obtained by polynomial curve fitting method. The first degree

polynomial is adequate for minimizing error between measured values and expected values in this study. 0.1958 dB mean squared error has been received for 16 steps. The software based audiometer system has been designed for 9 different frequency values like using in audiometer test sequences commonly. Modified Hughson-Westlake which is the most successful method in the literature has been used for test process. Besides, a white noise has been added to passive sound channel in order to hear nothing in the passive sound channel. In the future study, dynamic closed-loop calibration system can be performed by sound level meter. Audiometer calibration process can be implemented automatically. This can give an idea about environment noise level and audiometer can be minimized the effect of interference on the measurement signals. Speech sound signal can be also added for detailed hearing test.

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