Android Automated Studio Monitors

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Abstract: Android Automated Studio Monitors is a prototype developed for people with partial hearing loss. It can even be used by normal people who wants a exact reference from the monitor speakers. This Idea enables the sound engineers to mix properly with a perfect flat frequency response. The first module is developed in Android platform which detects the hearing loss in a particular frequency and the second module is connected with the studio monitors which calibrates according to the data from the first module. By using these two modules the frequency is manipulated according to the hearing of the sound engineers. The data transfer is done using GSM (Global system for mobile).

Keywords: Android, Studio monitors, GSM, automatic audio calibration, hearing loss analysis.

I. INTRODUCTION

Normally Sound Engineers get fed up with the wrong reference and muddy mix they make. This is because of Bad studio monitors or wrong reference due to aging and continuous exposure to sound. This Novel idea is for the people who are facing difficulty. It gives a right reference to the user and makes the mix Good.

II. RELATIVE STUDY

The reference paper [1] is taken as the base paper for the overall Prototype, the same working is followed. In the reference paper [2] & [3], The Equalizer part is studied and implemented in the second module. The Digital signal processing of Hearing Aid in reference paper [4] is also implemented in the second module. A study with sound engineers with hearing loss is done by taking the reference paper [5]. The reference paper [6] gives a in depth knowledge in volume control in smart mobile phones which helped in developing the first module for hearing loss analysis. Digital Filter bank is developed as per the reference paper [7]. Security system is added to the Android application as per the reference paper [8] and size of the Android application is maintained to be within the limit as per the reference paper [9]. The Digital filter can can either Graphical or Parametric according to the reference paper [10] and [11].

III. BASIC DESIGN OF THE SYSTEM



Fig 1 : Block diagram of the overall system

Vol. 4, Issue 4, pp: (74-78), Month: October - December 2016, Available at: www.researchpublish.com

The overall system consists of two modules:

- 1. Android based testing system.
- 2. Addon for Studio Monitors.

The first module communicates with the second one via GSM. Both the modules are elaborated below.

IV. ANDROID BASED TESTING SYSTEM

Android based hearing loss testing system is developed in android domain. The figure 2 depicts the hearing loss testing mechanism. It consists of user, sinewave generator, equalizer, DAC (Digital to analog converter), GSM module.



Fig 2 : Block diagram of Android based testing system

1.User:

A person with problems in hearing is considered as the user. The Android automated Studio Monitors is designed in such a way that any user can operate the system by themselves or with very little assistance from any other person. The user is required to wear headphones in which the sound generated by a sinewave is heard. When the user hears the sound of the sinewave the user is required to activate the store activate button in the Android application which is shown as switch in figure 2. The user further has to enter the collected sinewave values manually in the older android operating system (OS) applications but for newer versions of the OS, the data is automatically stored in the device.



Fig 3 : Block diagram of Sinewave generator.

2. Sinewave generator:

As shown in fig 3 sinewave generator is virtually built. The audio samples of sinewave are stored in the memory card. According to the amplitude and frequency input the file which is to be played is selected. The mp3 decoder decodes the mp3 file to hear it in the headphones. The sinewave generator produces sinewave of different frequency and amplitude.

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The frequencies used for hearing loss analysis are 125Hz, 250Hz, 500Hz, 1000Hz, 2000Hz, 4000Hz, 8000Hz. The frequency is kept constant and the amplitude is varied.

3. Equalizer:

As in fig 4, the equalizer is a 7 band equalizer used to calibrate the headphones used in the system. The headphones will have its own frequency response. In order to maintain the same amplitude in all the frequencies a volume equalizer is used.



Fig 4 : Block diagram of Equalizer.

In the android domain the inbuilt equalizer is overwritten as per the frequency response of the individual headphones. The values of the seven bandpass filters are 63-188Hz, 125-375Hz, 250-750Hz, 500-1500Hz, 1000-3000Hz, 2000-6000Hz, 4000-12000Hz.

4. DAC (Digital to analog converter):

The inbuilt digital to analog converter of the android device is used in the design of the Android automated Studio Monitors. The system requires only 16,000Hz sample rate for converting into analog but 44,100Hz is used as it is the default sample rate of the android device.

5. GSM Module:

The GSM module is inbuilt is most of the android devices. The GSM module consists of a SIM (Subscriber identity module) card with a valid number. In the Android automated Studio Monitors, the GSM module is utilized to send the collected audiogram values to the hearing aid through SMS service.



Fig 5 : Flowchart of Android based system

Vol. 4, Issue 4, pp: (74-78), Month: October - December 2016, Available at: www.researchpublish.com

The figure 5 shows the data processing of Android automated Studio Monitors. After the sinewave generation an if condition is used to select left or right. Then the frequency input is read and they are set by the user. Same like the frequency, amplitude is read and set by the user. Finally as mentioned equalizer is used to flatten the frequency response of the headphones used in the system.

V. ADD-ON FOR STUDIO MONITORS

Add-on for studio monitors can be developed in any processor which is capable of Handling audio signals and its should also have GSM facility



Fig 6 : Block diagram of Add-on for studio monitors.

1. Inputs:

The Inputs given to module is unbalanced input as the pre-amplification stage is done in the studio monitors.

2. ADC (Analog to digital converter):

In most of the boards like beagle bone analog to digital converters are inbuilt. The analog signals are digitized for digital processing. Without an ADC, signals cannot be processed digitally.

3. Equalizer:

As in the android based testing system, equalizer is made up of seven bandpass filters. In order to flatten the frequencies of the signal from the microphone, an equalizer is used.

4. Bandpass filters:

Seven bandpass filters are used to divide the signals into seven parts. The values of the seven bandpass filters are 63-188Hz, 125-375Hz, 250-750Hz, 500-1500Hz, 1000-3000Hz, 2000-6000Hz, 4000-12000Hz as in the android based testing system.

5. Automatic Amplitude control:

Seven automatic amplitude controllers are given to the outputs of the seven bandpass filters. It controls the amplitude by the values received via GSM. It uses gate and limiter.

6. DAC (Digital to analog converters):

In the DAC, the digital output of the ADC is converted to analog output. It works the vise versa of analog to digital converters. Most of the boards like beagleboard, arduino due, etc., have inbuilt DAC. A normal headphone accepts only analog signals. In order to hear the processed signal in the headphones DAC is used.

7. GSM module:

GSM module (or) GSM modem is a device which receives and transmits the SMS. Some of the GSM modules are SIM300, SIM900A. the analyzed values are received by using AT(Attention) commands.

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VI. RESULTS

The system is tested and there is a ratio of 5 to 10% change in the system when its compared with manual calculation which is not mostly considerable.

VII. CONCLUSION

Android automatic Studio monitors will create a Revival within Sound Engineers. It will spread drastically since its based on the latest technology and trends used. It can be even used in the normal home audio or Public address systems for getting a perfect frequency response and a decent sound. This system can even be used in the live sound calibration with a small crowd and even by People with hearing disability. When they can't wear the Hearing Aid this can be used is a static place while watching movies or any other similar applications.

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