An Audiometer Using PC and Wirelessly Interfaced Audiometric Module

A dissertation

submitted in partial fulfillment of

the requirements for the degree of

Masters of Technology

in

Electrical Engineering

by

Prachir Puri

(14307R026)

under the supervision of

Prof. P. C. Pandey

Department of Electrical Engineering

Indian Institute of Technology Bombay

June 2017

Page intentionally left blank

Indian Institute of Technology Bombay
Department of Electrical Engineering

M. Tech. Dissertation Approval

This dissertation entitled **"An Audiometer Using PC and Wirelessly Interfaced Audiometric Module"** by **Prachir Puri** (Roll No. **14307R026**) is approved, after the successful completion of *viva voice* examination, for the award of the degree of **Master of Technology** in **Electrical Engineering** with specialization in **Electronic Systems.**

Supervisor:	30/06/2017	**(Prof. P. C. Pandey)**
Examiners:		**(Prof. V. Rajbabu)**
		(Dr. Arun Pande)
Chairman:		**(Prof. V. Rajbabu)**

Date: 30 June 2017
Place: Mumbai

Page intentionally left blank

DECLARATION

I declare that this written submission represents my ideas in my own words and where ideas or words are taken from others, I have adequately cited and referenced the original sources. I also declare that I have adhered to all principles of academic honesty and integrity and I have not misrepresented or fabricated or falsified any idea/data/fact/source in my submission. I understand that any violation of the above will be cause for disciplinary action by the Institute and can also evoke penal action from the sources which have thus not been properly cited or from whom proper permission has not been taken when needed.

Prachir Puri

(Roll No. **14307R026**)

Date: 30 June 2017

Place: Mumbai

Page intentionally left blank

Prachir Puri / Prof. P. C. Pandey (Supervisor), “An Audiometer Using PC and Wirelessly Interfaced Audiometric Module,” *M. Tech. Dissertation*, Department of Electrical Engineering, Indian Institute of Technology Bombay, June 2017.

Abstract

An audiometer is an electro-acoustic instrument for quantifying the hearing loss and for identifying its causes. It involves presentation of sounds through calibrated transducers and assessing the responses. A two-channel advanced diagnostic audiometer using a PC and an external audiometric module is developed for performing subject-response audiometry and psychoacoustic tests for clinical and research applications. A PC-based application is used to generate the audio waveforms and the controls for the audiometric tests. The audiometry module presents the audio signals with precise attenuation in accordance with the controls received from the PC and collects the subject response. The PC and the audiometric module are wirelessly interfaced through Bluetooth for transfer of signals, controls, and responses. The system can be used for performing different audiometric tests including the pure-tone test in manual and automated mode, SISI test, tone decay test, ABLB test, speech test using live speech or recorded speech, and psychoacoustic tests. The audiometry module is designed as a compact battery-powered device including features like maintaining required signal-to-noise ratio over the full audiometric range of sound levels, monitoring the output voltage level of the transducer terminals to detect the connection status, automatic calibration using a sound level meter's feedback, acquiring and acknowledging patient response using a response switch with an indicator, and two microphone inputs which can be used for talk-back and ambient noise monitoring.

Page intentionally left blank

CONTENTS

Abstract		**i**
Contents		**iii**
List of Abbreviations		**v**
List of Figures		**vii**
List of Tables		**xi**
Chapter 1	**INTRODUCTION**	**1**
1.1	Background	1
1.2	Project Objective	2
1.3	Dissertation Outline	3
Chapter 2	**AUDIOMETRY BASICS**	**5**
2.1	Auditory Physiology	5
2.2	Hearing Impairment	6
2.3	Audiometry Techniques	8
2.4	Pure Tone Audiometry	9
2.5	Tone Decay Test	10
2.6	Short Time Sensitivity Index	10
2.7	Loudness Balance Test	10
2.8	Bekesy Audiometry	11
2.9	Speech Audiometry	11
2.10	Notched-Noise Test for Auditory Filter Shape	12
Chapter 3	**SYSTEM DESIGN FOR A VERSATILE LOW-COST AUDIOMETER**	**15**
3.1	Introduction	15
3.2	Earlier Developments	16
3.2	Design Approach	16
Chapter 4	**WIRELESSLY INTERFACED AUDIOMETRY MODULE**	**21**
4.1	Introduction	21
4.2	Bluetooth Transceiver	25
4.3	Pre-amplifier	26
4.4	Attenuator	28
4.5	Power Amplifier	30
4.6	Transducer Selection and Output Attenuator Switching	31

4.7	Circuit for Output Level Monitoring	32
4.8	Response and Acknowledgment Circuit	35
4.9	Touch Switch	36
4.10	Microcontroller	38
4.11	Power Management Circuit	38
4.12	Microcontroller Program	44
4.13	PCB Design	56
Chapter 5	**PC-BASED APPLICATION FOR AUDIOMETRY**	**59**
5.1	Introduction	59
5.2	Stimuli and Maskers Generation	59
5.3	Software Description	63
Chapter 6	**TEST RESULTS**	**79**
Chapter 7	**SUMMARY AND CONCLUSIONS**	**83**
Appendix A	**Hearing Threshold Determination Methods**	**85**
Appendix B	**Audiometer Specifications**	**89**
Appendix C	**Transducer Characteristics**	**93**
Appendix D	**First Version of the Audiometric Module**	**99**
Appendix E	**Balanced Current Source**	**127**
Appendix F	**Component List**	**131**
Appendix G	**Schematic Diagrams and PCB Layouts**	**133**
Appendix H	**Images of the Prototype**	**141**
References		**143**
Acknowledgement		**147**
Author's Resume		**149**

LIST OF ABBREVIATIONS

Abbreviation	Term
A2DP	Advanced Audio Distribution Profile
ABLB	Alternate binaural loudness balance
ADC	Analog-to-digital converter
ANSI	American National Standard Institute
ASHA	American Speech-Language-Hearing Association
BTL	Bridge-tied load
CCI	Communication class interface
CDC	Communication device class
CPU	Central processing unit
CTMU	Charge time measurement unit
DAC	Digital-to-analog converter
DCI	Data class interface
DPDT	Double pole double throw
DSO	Digital storage oscilloscope
EEPROM	Electrically erasable programmable read-only memory
GPIO	General purpose input output
GUI	Graphical user interface
HFP	Hands-Free Profile
Hi-Z	High impedance
HL	Hearing level
HLVD	High low voltage detect
HSP	Headset Profile
HTL	Hearing threshold level
I2S	Inter IC sound
IC	Integrated circuit
IFFT	Inverse Fourier transform
IIR	Infinite impulse response
I/O	Input-output
IRQ	Interrupt request
ISR	Interrupt service routine
IS	Indian Standard

LCD	Liquid crystal display
LED	Light emitting diode
Li-Ion	Lithium ion
LSB	Least significant bit
NMOS	N-type metal-oxide-semiconductor
PID	Product identifier
PMOS	P-type metal-oxide-semiconductor
PC	Personal computer
PCB	Printed circuit board
PLL	Phase locked loop
PTH	Plated through-hole
RETSPL	Reference equivalent threshold sound pressure level
rms	Root mean square
SCK	Serial clock
SD	Secure digital
SDO	Serial data out
SDS	Speech discrimination score
SDT	Speech discrimination test
SISI	Short time sensitivity index
SL	Sensation level
SLM	Sound level meter
SMD	Surface mount device
SNR	Signal-to-noise ratio
S/PDIF	Sony/Philips Digital Interface Format
SPI	Serial peripheral interface
SPL	Sound pressure level
SPP	Serial port profile
SRT	Speech reception threshold
THD	Total harmonic distortion
UART	Universal asynchronous receiver transmitter
USB	Universal Serial Bus
VID	Vendor identifier
WIAM	Wirelessly interfaced audiometry module

LIST OF FIGURES

Figure	Caption	Page
2.1	The peripheral hearing system consisting of the outer ear (pinna and ear canal), the middle ear (ossicular chain) and the inner ear (cochlea) (Adapted from [2]).	6
2.2	Notched-noise method for determining the shape of the auditory filter. The shaded are as corresponds to the amount of noise passing through the auditory filter centered at the signal frequency (Adapted from [14]).	13
3.1	Design approach of the system.	18
3.2	Design approach of the first version using USB and PC sound card output.	18
3.3	Design approach of the second version using wireless interfacing for signals and commands.	18
4.1	Block diagram of the external audiometric module.	22
4.2	Detailed block diagram of the WIAM.	23
4.3	Circuit diagram of the Bluetooth transceiver.	25
4.4	Pre-amplifier circuit for left and right audio channels along with the bias voltage generators.	27
4.5	Attenuator circuit for the left and right channels.	29
4.6	Timing diagram of the serial interface of the logarithmic attenuator LM1971 [21].	30
4.7	Power amplifier circuits for left and right channel.	31
4.8	The L-pad 40-dB attenuator circuit.	32
4.9	Circuit for transducer selection and output attenuator switching.	33
4.10	Circuit for output level monitoring along with the bias voltage generator.	35
4.11	Circuit for response and acknowledgement.	36
4.12	Layout of the capacitive touch switch.	36
4.13	Capacitance model of the touch pad with and without a finger near it.	37
4.14	The microcontroller circuit.	39
4.15	Block diagram of the power management circuit of the WIAM.	41
4.16	Battery charging and configuring circuit.	42
4.17	The powering circuit.	44

4.18	The main program of the microcontroller.	46
4.19	Flowchart of the initialization block of the main program.	47
4.20	ISR of Timer 2 interrupt.	50
4.21	A subroutine for touch press decoding and slow update of baseline.	51
4.22	ISR of UART receiver interrupt.	54
4.23	ISR of HLVD and Timer 0 interrupts.	55
4.24	ISR of PORTB interrupt.	56
5.1	Waveform of 125 Hz warble tone.	60
5.2	Waveform of 125 Hz pulsed tone.	61
5.3	Waveform of pure tone of 125 Hz for SISI test.	62
5.4	Filter responses for narrowband noise filtering at audiometric test frequencies.	62
5.5	Notch-noise with notch placed symmetrically around the signal frequency.	63
5.6	A screenshot of the home screen of the GUI.	64
5.7	A screenshot of the pure-tone test module.	65
5.8	The automatic pure-tone audiometry procedure for threshold determination.	67
5.9	Procedure for the automated tone decay test.	68
5.10	Procedure for the determination of the SISI score.	70
5.11	A screenshot of the ABLB test module.	71
5.12	A screenshot of the speech audiometry test module.	72
5.13	A screenshot of the notch-noise test module.	73
5.14	A screenshot of the audiogram module showing the air and bone conduction plots of left and right ear. The plot indicates that the person has a sensorineural hearing loss in the left ear and normal hearing in the right ear.	75
5.15	A screenshot of the calibration module.	76
5.16	Flowchart of the procedure followed in automatic calibration of the transducer.	78
A.1	Hearing threshold determination method as recommended by ASHA [3].	87
C.1	Plot of dB HL values at audiometric frequencies for various supra-aural headphones at 1 V_{rms} input. AC – Air conduction.	95

C.2	Plot of dB HL values at audiometric frequencies for three insert earphones at 1 V_{rms} input. AC – Air conduction.	96
C.3	Plot of dB HL values at audiometric frequencies for two bone vibrators at 1 V_{rms} input. BC – Bone conduction.	97
D.1	The block diagram of the first version of the audiometry module.	100
D.2	Detailed block diagram of the first version of the audiometry module.	101
D.3	Pre-amplifier circuit for left and right channels of audio input.	103
D.4	Attenuator circuit for the left and right channels.	104
D.5	Timing diagram of the serial interface of the logarithmic attenuator LM1971 [22].	105
D.6	Power amplifier circuits for left and right channel.	106
D.7	The L-pad 40-dB attenuator circuit.	107
D.8	Relay-controlled circuit for transducer selection and output attenuator switching.	108
D.9	Circuit for output level monitoring.	110
D.10	Circuit for response and acknowledgement.	110
D.11	The microcontroller circuit.	111
D.12	Power circuit.	113
D.13	Bias voltage generator.	114
D.14	Flowchart of the microcontroller program.	115
D.15	A screenshot of the home-screen of the GUI.	119
D.16	Automatic procedure for calibrating the PC-sound card volume level.	120
D.17	Schematic sheet-1: The pre-amplifier and the attenuator circuit of left and right channels.	122
D.18	Schematic sheet-2: The power amplifier and the relay controlled attenuator and transducer selection circuit for left and right channels.	123
D.19	Schematic sheet-3: The output level monitoring circuit, microcontroller circuit, response button circuit, and power circuit.	124
D.20	PCB layouts: (i) Top overlay, (ii) bottom overlay, (iii) top layer, and (iv) bottom layer of the PCB design.	125
D.21	(i) Top view, and (ii) bottom view of the assembled PCB of the first version of the audiometry module.	126

E.1	An op amp based balanced current source circuit.	127
E.2	Balanced current drive circuit using two single-ended output op amps.	128
E.3	A balanced current source using an audio amplifier.	129
G.1	Schematic sheet-1: The battery charging and configuring circuit.	133
G.2	Schematic sheet-2: The powering circuit.	134
G.3	Schematic sheet-3: The Bluetooth transceiver circuit.	135
G.4	Schematic sheet-4: The pre-amplifier and the attenuator circuit of left and right channels.	136
G.5	Schematic sheet-5: The power amplifier and the analog audio switch controlled transducer selection and output 40-dB attenuator switching circuit for left and right channels.	137
G.6	Schematic sheet-6: The level monitoring, microcontroller, and response button circuit.	138
G.7	PCB layouts: (i) Top overlay, (ii) bottom overlay, (iii) top layer, (iv) second layer, (v) third layer, and (vi) bottom layer of the 4-layer PCB design.	139
G.8	PCB layouts: (i) Top overlay, (ii) bottom overlay, (iii) top layer, and (iv) bottom layers of the 2-layer PCB design.	140
H.1	(i) Top view of the 4-layer assembled PCB, (ii) bottom view of the 4-layer assembled PCB, (iii) top view of the 2-layer assembled PCB, (iv) bottom view of the 2-layer assembled PCB, and (v) isometric view of the of the audiometry module after stacking the 4-layer and the 2-layer PCBs.	141

LIST OF TABLES

Table No.	Title	Page
2.1	dB SPL for 0 dB HL [2].	6
2.2	Pure tone average threshold (average of hearing thresholds at 0.5, 1, 2, and 3 kHz) for different grades of hearing loss [3].	7
2.3	Stimulus and masking noise presentation for different tests.	9
3.1	Comparison of the some audiometers developed at IIT Bombay.	17
4.1	A comparison of three digitally-controlled logarithmic attenuators.	28
4.2	Controls for selecting transducers for the two audio channels. First three combinations are preferred as they consume less relay activation currents.	34
4.3	Controls for switching the 40-dB attenuators.	34
4.4	Function assigned to the I/O port pins of the microcontroller U11.	40
4.5	Estimate of current requirement when the module is turned on and the module is operating on batteries.	44
4.6	Functions performed by the microcontroller based on the commands received through Bluetooth transceiver.	53
5.1	Audiometric symbols used when plotting air and bone conduction hearing thresholds (adapted from [33]). R – Response, NR – No Response.	75
6.1	Measured maximum voltage level across a TDH39P transducer at audiometric frequencies.	79
6.2	Measured frequency accuracy and total harmonic distortion at audiometric frequencies for maximum output level without load.	80
6.3	Measured maximum achievable sound level in dB SPL for pure-tones at audiometric frequencies using TDH39P headphone. dB HL = dB SPL − RETSPL.	81
6.4	Ambient noise level at frequencies close to audiometric frequencies in sound proof room (measured using 1/3rd octave filters).	81
B.1	Minimum required dB HL_{MAX} for various categories of pure tone audiometer at different audiometric frequencies as per ANSI S3.6, 1996 [12]. The minimum sound level of these audiometers at all the audiometric frequencies is −10 dB HL.	90
B.2	Maximum level (dB HL_{MAX}) achievable at the audiometric frequencies in some of the commercially available Type 1 and Type 2 audiometers. Minimum level achievable at all the frequencies is −10 dB HL. AC: Air conduction using TDH39P transducer, BC: Bone conduction using	90

	B71 bone vibrator on mastoid.	
B.3	Tolerance requirement for various types of audiometers [12].	91
C.1	Sound output in dB SPL and dB HL at audiometric frequencies for some supra-aural headphones at 1 V_{rms} input. Z_0 = impedance at 1 kHz. dB HL = dB SPL − RETSPL (as given in Table C.2).	94
C.2	Reference equivalent threshold sound pressure level (RETSPL) for various supra-aural headphones based on their datasheets. All the values are in dB SPL.	95
C.3	Sound output in dB HL at audiometric frequencies for commonly used insert earphones at 1 V_{rms} input. Z_0 is the nominal impedance measured at 1 kHz.	96
C.4	Sound output in dB HL at audiometric frequencies for B71 and B81 bone vibrators at 1 V_{rms} input. Z_0= impedance at 1 kHz.	97
D.1	Controls for selecting transducers for the two audio channels. First three combinations are preferred as they consume less relay activation currents.	109
D.2	Controls for switching the 40-dB attenuators.	109
D.3	Function assigned to the I/O port pins of the microcontroller U12.	112
D.4	Estimate of current requirement.	113
D.5	Functions performed by the microcontroller based on the commands received through USB.	117

Chapter 1

INTRODUCTION

1.1 Background

Audiometry is a technique for the identification and quantification of hearing loss. It is widely performed in audiometry clinics for prescribing a suitable medical treatment, hearing aid fitting, or prescribing further diagnostic tests. It is used for the regular hearing assessment of industrial workers with significant noise exposure and is also needed in hearing research. Audiometric tests can be broadly classified into two categories: the tests based on physiological response which do not require active involvement of the subject, and the tests based on subject-response which require voluntary response from the subject [1], [2], [3]. In physiological or objective tests, the involuntary physiological response to the stimulus is sensed and recorded by the instrument. Some of the commonly used physiological response based audiometric tests are otoacoustic emission audiometry, brain stem evoked response audiometry, and tympanometry. Each of these tests requires a special instrument for presenting the stimuli and recording the response. A subject-response based test requires active participation of the subject during the test by responding to the sound stimulus through a key press or a hand raise. Pure tone audiometry is the commonly performed subject-response audiometry in which the hearing sensitivities to pure tones at standard audiometric frequencies (a set of frequencies in the 125 – 8000 Hz range) are determined. Some of the other commonly used subject-response audiometric tests are tone decay test, SISI test, loudness balance tests, and speech test. In all these tests, a stimulus at a precisely controlled level is presented through calibrated transducers and the responses are assessed. The objective tests are generally used in the situations where subject-response audiometry cannot be administered, like for babies.

The term "audiometer" is commonly used for an instrument used for performing subject-response audiometry. Simplest type of audiometer generally consists of a oscillator with selectable frequency, controller, attenuator with selectable setting, and power amplifiers for generation and presentation of the signals, response switch for patient's response acquisition, and hardware for the user interface. It may also have a provision for generation and control of a masking noise. A single-channel audiometer presents sound through only one channel and is usually inexpensive whereas a two-channel audiometer can present sounds through both left and right channels and permits their independent selection and level control.

For testing full range of the auditory thresholds, the sound stimulus needs to have dynamic range of 130 dB [2], [3]. The main design challenge is to produce signals with low distortion at high output level and low noise at low output level. Giannini et al. [4] proposed an audiometer design using a DSP chip, 16-bit stereo codec (DAC), and a power amplifier for stimulus generation. In this design, low level sounds cannot be produced with high fidelity due to relatively large quantization noise. Several PC-based audiometers have been developed to reduce the cost of specialized hardware, provide flexible user interface, enabling remote administration of audiometric tests, and exploiting the large storage capacity of PC for storing and retrieving the test results and patients history [5], [6], [7], [8]. Wu et al. [5] reported an audiometer design of a PC-based audiometer using a direct digital synthesizer as signal generator, a microcontroller, log attenuators, power amplifiers, and RS232 communication link with the PC. In this design, the PC is used for performing the audiometry by controlling the peripherals in the hardware. Dille and Elingson [9] reported a PC sound card based audiometric test system in which the dynamic range of the output sound is maximized by manipulating the amplitudes of the synthesized sound and PC sound card attenuator settings. Inherent noise floor at the output of the PC sound card limits use of this technique for generating low level sounds

1.2 Project Objective

The project objective is to develop a system for use as a two-channel advanced diagnostic audiometer that is capable of performing different types of subject-response audiometry and psychoacoustic tests so that it can be used for clinical and research applications. The system designed consists of a PC with an audiometric application and an audiometry module as a wirelessly interfaced compact battery-powered external hardware. The sounds and the controls for the audiometry tests are generated from the PC and are transmitted to the external hardware module over Bluetooth. The external hardware is used for presenting the audio signals to both the ears at calibrated levels over the full range of audiometric levels for assessing hearing of the test ear. A wirelessly interfaced audiometric module (WIAM) is developed for extending the dynamic range of the sound stimuli for performing audiometry with a variety of transducers. It is designed as a battery-powered device with an output dynamic range of 164 dB, three transducer outputs (v*iz.* left headphone, right headphone, and bone vibrator), level detector circuit for monitoring the output voltage level across the transducer terminals, an input for automatic calibration of the transducer using a sound level meter's feedback, an input for acquiring the patient response, two microphone inputs for talk-back feature and ambient noise detection. An application software on the PC is developed for performing various subject-response audiometry like pure-tone test in manual and automated modes, SISI test, tone decay test, and speech test using live or pre-recorded speech. A psychoacoustic test called the notch-noise test, which involves determining the auditory filter shape and critical bandwidth using the principle of spectral masking, has also been implemented. The audio waveforms for all the tests are synthesized and stored as files on the PC. The sounds are presented in accordance with the test selected in the software, and the control signals for controlling its level are sent to the audiometry module over Bluetooth. The PC-based audiometry application can store the calibration data, test results, and can also plot the audiogram of the test results. The calibration of a transducer can be performed in an automated way using the dc output from the sound level meter as a feedback input to the audiometry module.

1.3 Dissertation Outline

The second chapter provides an overview of the physiology of the auditory system and types of hearing impairment followed by a brief overview of the commonly used subject-response audiometry tests. The third chapter gives an overview of the specifications of an audiometer and the earlier audiometers developed at IIT Bombay followed by the design approach for a versatile low-cost audiometer. The fourth chapter describes the hardware design of the audiometry module and the fifth chapter describes the user interface of the PC-based audiometry application. The sixth chapter presents the test results and the last chapter provides a summary of the completed work along with some suggestions for future development.

Page intentionally left blank

Chapter 2

AUDIOMETRY BASICS

2.1 Auditory Physiology

Auditory perception involves peripheral auditory system comprising outer ear, middle ear, and inner ear, as shown in Figure 2.1, and the central auditory processing located in the brain [1], [2], [10]. The outer ear comprises pinna and ear canal. The pinna collects the sound and funnels it to the ear canal. The sound from the outer ear is coupled to the middle ear through the vibrations of the eardrum, also known as the tympanic membrane. The middle ear is an air-filled cavity containing a chain of three small bones (the ossicles: malleus, incus, and stapes), which transmits the vibrations of the tympanic membrane to the stapes footplate on the oval window of the fluid filled cochlea. The ratio of the areas of the eardrum and oval window and the lever action of the chain of middle ear bones provide impedance matching for coupling of the acoustic vibrations in the air to those in the cochlea. The cochlea has sensory hair cells which convert the acoustic vibrations into electrical signal in the form of neuron firings which are transmitted through auditory nerve to the auditory processes, and finally to the auditory cortex (i.e. temporal lobe of the brain) where the sensation of sound is analyzed and interpreted.

A normal ear is considered to respond to sound pressure ranging from 20 μPa to 20 Pa, i.e. a dynamic range of 10^6. The sound pressure is generally expressed as sound pressure level (SPL) in dB by converting the rms sound pressure P using the relation

$$\text{dB SPL} = 20\log(P/P_{ref}) \tag{2.1}$$

with reference P_{ref} = 20 μPa. The sound pressure in dB SPL is measured using an instrument known as the sound level meter. The minimum level at which a tone remains audible varies with frequency. For audiometry, the auditory threshold is generally represented in dB HL (hearing level) which is dB relative to the minimum audible sound level at the test frequency as measured for a large number of otologically normal young listeners. This representation of sound level makes it convenient to visualize the degree of hearing loss when the thresholds at different frequencies are plotted. A 0 dB HL at a particular frequency corresponds to minimum audible sound level as measured for large number of otologically young listeners. Hence an increase from a 0 dB HL baseline in the plot indicates the degree of hearing loss.

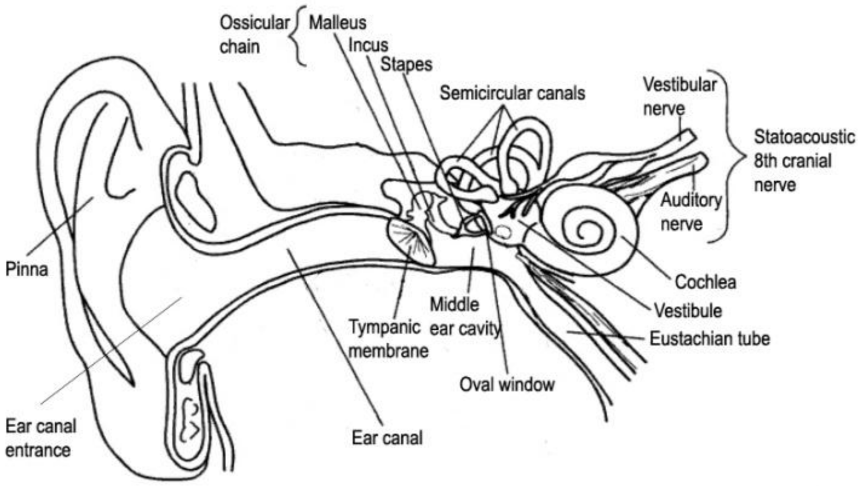


Figure 2.1: The peripheral hearing system consisting of the outer ear (pinna and ear canal), the middle ear (ossicular chain) and the inner ear (cochlea) (Adapted from [2]).

Table 2.1: dB SPL for 0 dB HL [2].

Frequency (Hz)	125	250	500	750	1 k	1.5 k	2 k	3 k	4 k	6 k	8 k
dB SPL for 0 dB HL	45.0	25.5	11.5	8.0	7.0	6.5	9.0	10.0	9.5	15.5	13.0

The sound is measured in dB SPL using a sound level meter and the measured value can be converted to dB HL using the relation

$$\mathrm{dB\,HL} = \mathrm{dB\,SPL} - (\mathrm{dB\,SPL\ for\ 0\ dB\,HL}) \tag{2.2}$$

The values for dB SPL corresponding to 0 dB HL are given in Table 2.1.

2.2 Hearing Impairment

Hearing impairment results in the loss of hearing sensitivity and in abnormal auditory perception. Based on the increase in hearing thresholds, hearing loss is graded as normal, mild, moderate, severe, very severe, and profound [3] as given in Table 2.2. Based on the site of pathology, the hearing impairment is categorized as conductive, sensorineural, mixed, central, and functional loss [1], [2], [10], [11].

Conductive hearing loss: It occurs when the transmission of sound to the inner ear is impaired due to problems either in the outer or middle ear. Some of its causes are accumulated ear wax or foreign body in the ear canal, perforated eardrum, accumulation of fluid in the middle ear, calcification of the joints of the middle ear ossicles (otosclerosis), or a physical trauma caused by hard blow. Conductive loss generally results in an attenuation of the incoming sound. Therefore, the hearing sensitivity decreases across the entire range of

Table 2.2: Pure tone average threshold (average of hearing thresholds at 0.5, 1, 2, and 3 kHz) for different grades of hearing loss [3].

Hearing threshold (dB HL)	0 – 25	26 – 40	41 – 55	56 – 70	71 – 90	> 90
Grade of impairment	Normal loss	Mild loss	Moderate loss	Severe loss	Very severe loss	Profound loss

frequencies. This type of loss can generally be corrected by medical treatment or by use of a hearing aid with linear amplification.

Sensorineural hearing loss: It is caused by pathology in the cochlea or in the auditory pathway from the inner ear to the brainstem. The loss specifically due to cochlear disorders is termed as sensory loss, while the loss due to disorders in the auditory pathway is termed as retrocochlear loss. Sensory loss occurs due to damage of sensory hair cells in the cochlea which may result in reduction in the hearing sensitivity for a certain range of frequencies, loss of clarity of speech especially in the presence of background noise, or discomfort with the loud sounds due to loudness recruitment. Loudness recruitment is an abnormal rapid increase in the perceived loudness with the increasing sound level. Aging is the common cause for sensorineural loss as the sensory hair cells in cochlea degenerate with age. Patients with age-related hearing loss (presbycusis) often have elevated hearing threshold and often find difficulty in understanding a conversation in the presence of the background noise. Exposure to loud noise, congenital defects (due to hereditary factors), and use of ototoxic drugs are some of the other causes for damage of the sensory hair cells. Sensorineural hearing loss is irreparable and cannot be treated through medication. Effects of such type of loss can be alleviated by using hearing aids with frequency-dependent amplification along with dynamic range compression and signal processing to suppress background noise and reduce the effects of intra-speech masking. In severe cases, cochlear implants can be used to bypass the dead sensory hair cells.

Mixed hearing loss: It involves a combination of conductive hearing loss and sensorineural hearing loss.

Central and functional loss: Sometimes the hearing loss occurs due to lesion in the central auditory pathway beyond the eight cranial nerve because of the damage of the brain cortex due to cerebral meningitis, skull trauma, or congenital defects. Such hearing loss is termed as central loss and it may lead to reduced ability to interpret, or understand speech. In addition to the organic types of hearing loss, there is also a possibility of functional or nonorganic hearing loss, caused by psychological or emotional factors.

2.3 Audiometry Techniques

Audiometry involves clinical tests to quantify the extent of hearing impairment and to identify its causes. The most commonly used audiometry test involves determining the hearing thresholds of an ear for different types of sound stimuli. Hearing threshold is defined as the lowest level of presentation at which the subject responds at least 50% of the time. The threshold to a stimulus can be determined using either air conduction or bone conduction. In air conduction, the stimulus is presented to the test ear using an external headphone or insert earphone and the stimulus reaches the cochlea through the outer and the middle ear. The stimulus presented in the test-ear through air conduction can pass to the non-test ear through skull bones with inter-aural attenuation of about 40 – 80 dB [3]. This leakage of the sound can stimulate the cochlea of the non-test ear and is known as cross hearing. If one cochlea is better than the other, the response during testing of worse ear may be due to the stimulus reaching the better cochlea. In bone conduction test, the stimulus is applied with a bone vibrator placed over the mastoid bone or forehead and the stimulus reaches the cochlea of both the test and the non-test ears directly through the skull bones with no inter-aural attenuation. To ensure that the subject's response is only due to the stimulation of the cochlea of the test ear, the non-test ear needs to be masked by presenting a broadband noise or a narrowband noise centered at the stimulus frequency to the non-test ear by air conduction. Table 2.3 shows a stimulus and masking noise presentation schemes for carrying tests in both the ears. The level of the masking noise in the non-test ear should be enough to mask the stimulus due to cross hearing, but not so high as to influence the sensitivity of the test ear. Air conduction measures total loss, while bone conduction measures sensorineural loss. Test using both air and bone conduction are needed to separately quantify the conduction and sensorineural loss.

Audiometric tests can be broadly grouped as being based on subject response and physiological response. In subject response tests, the patient has to actively respond on hearing the presented stimulus. The commonly used subject-response tests are pure-tone test, tone decay test, SISI test, loudness balance test, Bekesy audiometry, and speech audiometry [1], [2], [3]. In physiological-response tests, the involuntary response to the stimuli is automatically recorded by the instrument. Some of commonly used physiological-response tests are otoacoustic emission audiometry, brain evoked response audiometry, and tympanometry [1], [2], [3]. A combination of different audiometric tests is used to diagnose the nature of hearing impairment and localizing the sites of the disorders. For quantifying

Table 2.3: Stimulus and masking noise presentation for different tests.

Test	Left headphone	Right headphone	Bone vibrator
RAC: Right ear with air conduction	Masker	Stimulus	-------
LAC: Left ear with air conduction	Stimulus	Masker	-------
RBC: Right ear with bone conduction	Masker	------	Stimulus
LBC: Left ear bone conduction	-------	Masker	Stimulus

hearing characteristic other than increase in hearing thresholds, several specialized psychoacoustic tests have been developed. Only some of these are used in clinical practice.

Some of the commonly used subject-response audiometric tests and a psychoacoustic test for measuring auditory filter bandwidth are described in the subsequent sections.

2.4 Pure Tone Audiometry

Pure tone audiometry measures the hearing sensitivity of an ear to pure-tones at different frequencies. A pure tone at a test frequency is presented through a calibrated transducer and the level is varied in small steps (generally 5 dB). The minimum level (in dB HL) to which consistent responses are obtained is recorded as the hearing threshold level for the test frequency. The hearing threshold determination is generally carried out by the Hughson-Westlake technique. The patient is familiarized with the test sound by presenting the pure-tone at a comfortable level. Then the level is decreased in 10 dB decrements until the patient no longer responds. The level is increased in 5 dB increments until the patient responds again. The procedure is repeated, i.e. the level is decreased by 10 dB whenever the patient responds and increased by 5 dB whenever the patient stops responding. The level at which the listener responds two out of three times is recorded as the threshold. The pure-tone test is carried out with tones of 125, 250, 500, 750, 1000, 1500, 2000, 3000, 4000, 6000, and 8000 Hz. The manner in which the tones of different frequencies are presented to determine the thresholds is further described in Appendix A.

In addition to the continuous tone, warble tone (frequency modulated), or pulsed tone (amplitude modulated) may be used as stimulus for improving the consistency of the test. The test may be conducted using air or bone conduction with a masker. A comparison of the plot of the hearing thresholds obtained from air and bone conduction can be used to determine the degree and type of loss.

2.5 Tone Decay Test

Tone decay test is used to determine the minimum level of a pure tone which a patient can hear continuously for about 60 s. The test is performed after determining the pure-tone threshold of the ear using pure-tone audiometry. The most commonly used method is the Carhart's method [3]. In this method, a pure-tone stimulus is presented 10 dB below the hearing threshold level and the sound level is raised in 5 dB steps until the patient responds. As soon as the patient responds, a stopwatch is started and the level of the tone is maintained. The patient is instructed to indicate whenever the tone disappears. If the patient is able to hear the tone for full one minute then the test is terminated and the level is recorded as the threshold. If the patient stops hearing before the end of one minute then the tone level is raised by 5 dB and test is conducted again. Determination of differences between the tone decay thresholds and the pure-tone thresholds at different frequencies is helpful in diagnosing if the hearing impairment is due to retrocochlear lesion. The differences of 10 – 15 dB, 20 – 25 dB, and greater than 30 dB are indicative of mild, moderate, and severe retrocochlear lesion, respectively.

2.6 Short Increment Sensitivity Index (SISI)

SISI test is used to check the ability of a patient to detect small changes in the level of sound. It is carried out after finding the hearing thresholds using pure tone audiometry. The test determines the capacity of a patient to detect a brief 1-dB increment in a 20 dB supra-threshold tone (called carrier tone) at various frequencies. The 1-dB increment is repeated every 5 s for 200 ms, with rise and fall times of 50 ms each, and the patient is instructed to indicate whenever an increase in the tone level is heard. Twenty such 1-dB increments are presented and the total number of responses is recorded and multiplied by 5 for the percentage SISI score. In case of significant sensorineural loss, the scores of this test are helpful in differentiating cochlear from retrocochlear pathology. Test score above 70% indicates cochlear pathology and the score below 20% indicates retrocochlear pathology.

2.7 Loudness Balance Test

This test is used to determine the loudness recruitment in a patient with unilateral hearing loss. It is performed if there is a significant difference (at least 25 dB) in the pure tone thresholds of the two ears. Sound level in dB with reference to the hearing threshold of the test ear is known as sensation level (SL). Tone at test frequency is presented alternately to the two ears. The worse ear is presented with 20 dB SL and the better ear is presented with 0 dB SL. The patient is instructed to indicate the ear in which the sound appears to be louder and when the sounds appear to be of same loudness in both the ears. If the tone in the worse ear appears to be louder, then the tone level in the better ear is raised by 5 dB. If the tone in the better ear appears to be louder, then the tone in the better ear is decreased by 5 dB. The procedure is repeated till the loudness balance is achieved. When the loudness in both ears is felt equal, then the level of tone in the better ear is recorded as equal in loudness to 20 dB SL in the worse ear. This process is repeated for every 10 dB SL increment in the worse ear level i.e. for 30 dB SL, 40 dB SL and so on until the audiometers limit. The same process can also be followed by keeping the level in the better ear fixed, and varying the level in the worse ear. The results from the test are plotted in the form of a ladder, also called as laddergram, by using one axis for better ear and the other one for worse ear. The hearing level of the better ear and the worse ear which are equal in loudness (for every 10 dB SL increment in worse ear) are connected by straight lines. The laddergram helps in characterizing recruitment in the worse ear.

2.8 Bekesy Audiometry

This is a self recording form of pure-tone audiometry in which the frequency of the tone is continuously swept in forward/backward direction and the level of the tone is controlled by the patient's response switch. The patient is instructed to press the switch as soon as the sound is heard and to release it when the sound is not heard. On pressing the switch, the instrument automatically starts decreasing the sound level and on releasing the switch, it starts increasing the sound level. A graphical representation of the patient's hearing threshold across entire frequency range is obtained which contains successive crossing and re-crossing of the hearing threshold in the form of a jagged line. The procedure is performed using continuous tones and pulsed tones. The traces obtained from both of them are superimposed and analyzed. If the two traces are in an interweaving pattern and there is not much gap between them, then the test ear may be normal ear or may be having a conductive loss. If the difference between the two traces is 15 – 20 dB at higher frequencies, then a cochlear lesion is suspected. If the two traces have large difference across all the frequencies, then a retrocochlear lesion is suspected.

2.9 Speech Audiometry

It assesses the hearing sensitivity for the speech signal. There are two types of speech audiometry: speech reception threshold (SRT) test and speech discrimination test (SDT).

The SRT, often also referred to as speech understanding or word understanding test, is a measure of an individual's ability to recognize familiar words from a closed set of words.

The SRT testing is performed using spondee words i.e. words with two syllables having equal emphasis on each syllable, e.g. baseball, airplane, and playground. The spondee words are presented at 25 dB above the average pure-tone threshold and the level is decreased to find the lowest level at which 50% of a list of spondee words are correctly identified by the subject. In case of retrocochlear pathology, the SRT is significantly higher than the pure-tone average threshold.

The SDT, often also referred to as word recognition test, assesses the patient's ability to understand monosyllabic words presented at 35 dB above the SRT level.The percentage of the total number of monosyllabic words which the subject is able to identify correctly gives the speech discrimination score (SDS). Low SDS can be due to high frequency loss which reduces the perception of many consonant sounds.

2.10 Notched-Noise Test for Auditory Filter Shape

The analysis of sound in the frequency domain is conceptualized as a bank of overlapping bandpass filters, known as the auditory filter bank. The frequency selectivity in hearing depends on the shape and the bandwidth of these filters. Widening of the auditory filters increases the spectral masking, thereby causing difficulty in speech perception, particularly in the presence of background noise [13]. Psychoacoustic tests are used to estimate the shape and bandwidth of the auditory filter.

A method known as notched-noise test may be used for estimating the shape of the auditory filters [13]. In this method, a tone is presented at a test frequency as the signal in the presence of bandpass noise with a spectral notch placed around the test frequency as shown in Fig 2.2. The amount of noise passing through the auditory filter centered at the test frequency is proportional to the shaded areas in the figure. The level of the notched-noise is maintained and the level of the stimulus is varied. The minimum level of the stimulus at which it becomes just audible in the presence of this notched-noise is recorded as the masked threshold. The masked threshold is found for different notch-widths for a particular test frequency. A relation between the masked thresholds and the notch-width for a particular test frequency is established and used for estimating the auditory filter shape at the test frequency.

The masked threshold is considered as equal to the total noise power within the auditory filter as shown by the shaded area in Figure 2.2 and given as the following equation:

$$P_s(\Delta f) = KN_o \left[\int_0^{f_c - \Delta f} \mathrm{W}(f)\mathrm{d}f + \int_{f_c + \Delta f}^{\infty} \mathrm{W}(f)\mathrm{d}f \right] \tag{2.3}$$

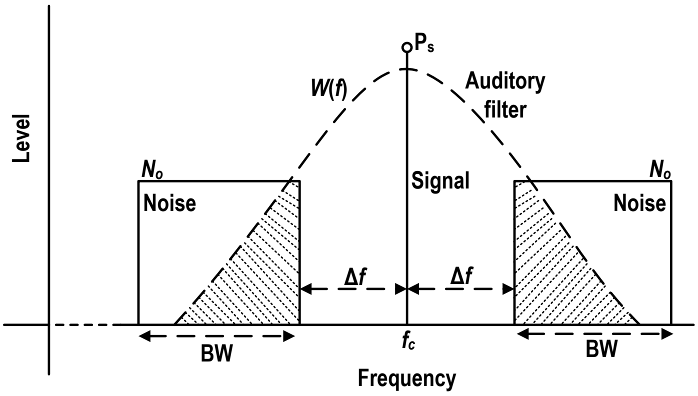


Figure 2.2: Notched-noise method for determining the shape of the auditory filter. The shaded area corresponds to the amount of noise passing through the auditory filter centered at the signal frequency (Adapted from [13]).

where $P_s(\Delta f)$ = power of the stimulus tone, K = constant related to the efficiency of the detection mechanism, N_o = power spectral density of the noise in its passband, f_c = center frequency, Δf = half of the notch width, W(f) = function representing the shape of auditory filter centered at f_c. For moderate noise levels, the auditory filter is almost symmetrical on a linear frequency scale and therefore (2.3) can be simplified as

$$P_s(\Delta f) = 2KN_o \int_0^{f_c-\Delta f} \mathrm{W}(f)\mathrm{d}f \tag{2.4}$$

For a selected noise power spectral density N_o, power of the stimulus P_s is plotted as a function of half notch bandwidth Δf. The value of W(f) at deviation Δf from the center frequency f_c is calculated by finding the slope of the masked threshold function at the half notch-width Δf.

$$\mathrm{W}(f_c - \Delta f) = -\frac{1}{2KN_o}\frac{\partial P_s(\Delta f)}{\partial \Delta f} \tag{2.5}$$

The auditory filter shape function can then be estimated by fitting a polynomial function over the calculated $W(f)$ values.

Page intentionally left blank

Chapter 3

SYSTEM DESIGN FOR A VERSATILE LOW-COST AUDIOMETER

3.1 Introduction

The electro-acoustic instrument used for performing audiometry is known as audiometer. Specialized types of audiometers are used for physiological response audiometry. The term "audiometer" is generally used for subject-response audiometer, which is used for administering subject-response audiometric tests by presenting stimulus and masker through calibrated transducers and controlling their level to determine the hearing sensitivity of the patient. The stimulus can be pure tone, warble tone, and pulsed tone at audiometric frequencies in the 125 – 8000 Hz range, and the masker can be broadband and narrowband noise. The dynamic range of the output sound for air conduction test and bone conduction test are 130 dB and 80 dB, respectively. For speech audiometry, recorded speech or live speech through a microphone is used as stimulus. An audiometer can also have extended high frequency capabilities up to 20 kHz for screening the hearing loss from ototoxicity. The levels of the stimulus and masker are controlled independently in 5-dB steps or smaller throughout the full range of the instrument. The stimulus can be presented either using air conduction or bone conduction. Transducers for air conduction tests are supra-aural headphone, circum-aural headphone, and insert earphone and transducer for bone conduction is a bone vibrator. The audiometer is calibrated for a specific transducer before using it for audiometry.

The technical specifications of an audiometer are defined in the standards like ANSI S3.6, 1996 [12] and IS 10565:1999 [14]. The specification includes the discrete frequencies for pure-tone tests, frequency accuracy for the pure-tones, accuracy of presentation levels, minimum and maximum sound levels, permissible THD, bandwidth of the masker sounds, and description for waveforms for pulsed tone, warble tone, and SISI tone. ANSI S3.6, 1996 [12] classifies audiometers into four categories *viz.* advanced diagnostic audiometer, diagnostic audiometer, simple diagnostic audiometer, and screening audiometer. Each category has specifications for the maximum achievable output level at audiometric frequencies, permissible harmonic distortion, frequency accuracy for the tones, accuracy for the output levels, types of conduction, and the types of audiometric tests that can be performed like speech audiometry and sound field audiometry. The specifications of these four types of audiometers and various commercially available audiometers are given in Appendix B.

The project objective is to develop a two-channel advanced diagnostic audiometer which can be used for clinical and research applications. It should be a low-cost portable instrument with the features of an advanced diagnostic audiometer that is capable of performing different types of subject-response based audiometric tests. It should also useable for conducting tests in both manual and automated manner and it should be convenient to calibrate. Some of the earlier developments are summarized in the next section, followed by the description of the design approach in the third section.

3.2 Earlier Developments

Several microcontroller based audiometers have been developed at IIT Bombay for carrying out subject-response audiometry. The specifications of some of them are compared in Table 3.1. The most recent audiometer design developed by Thirupathi [17] has two controllers, one microcontroller for controlling the peripherals and handling the user interface and another digital signal controller for generating stimulus and masker, with the waveforms pre-synthesized and stored in an SD card. A log attenuator with a dynamic range of 127 dB and steps size of 0.5 dB is used in each channel to control the level of the stimulus and masker independently. Three separate power amplifiers are used for driving the left headphone, right headphone and the bone vibrator. Attenuators are used before the power amplifier stage. A 4 × 4 keypad and a 128 × 64 graphical LCD is provided for user interface. The calibration data of various transducers are stored in the EEPROM of microcontroller. A response button is used for taking the response from the patient. A serial link is provided to transfer the test result to the PC. The audiometer can be operated as a stand-alone unit with its own keypad and display, or it can be fully controlled through the PC using a graphical user interface.

3.3 Design Approach

Audiometry involves outputting stimulus and masker at selected levels to determine the thresholds. The audiometer has three transducers for presenting the sounds: the left headphone, the right headphone, and the bone vibrator. The stimulus may be applied to any one of the three transducers, while the masker is applied to one of the two headphones. For an advanced diagnostic audiometer, the dynamic range of the output tones must be −10 to 120 dB HL for mid frequencies like 500, 750, 1000, 1500, 2000, 3000, and 4000 in air conduction test. For a particular transducer, the sound level in dB HL for a pure tone for a given input

Table 3.1: Comparison of the some audiometers developed at IIT Bombay.

Specifications	Audiometer design		
	Patel (2002) [15]	**Mossa (2010)** [16]	**Thirupathi (2011)** [17]
Frequency range	250 – 8000 Hz	125 – 8000 Hz	125 – 8000 Hz
Type	Simple diagnostic	Simple diagnostic	Diagnostic
Attenuator steps	0.375 dB	0.5 dB	0.5 dB
Signal selection	Continuous, warble, pulse	Continuous, warble, pulse	Continuous, warble, pulse
Masking noise	Broadband, narrowband	Broadband, narrowband	Broadband, narrowband
Tests	Pure-tone, SISI, tone decay, and speech.	Pure-tone, SISI, tone decay, and speech.	Pure-tone, SISI, tone decay, and speech.
Operation modes for pure-tone test	Manual/Automatic	Manual/Automatic	Manual/Automatic
Supply voltage	±5V	±8 V	±9 V
User interface	16× 2 LCD, 4 × 4 keypad	128 × 64 LCD, 4 × 4 keypad	128 × 64 LCD, 4 × 4 keypad
Microcontroller	8-bit AT89C52	8-bit P89V51RD2	8-bit PIC16F1939
Tone generation scheme	Switch capacitor filter (IC LMF100)	Direct digital synthesizer (IC AD9833)	Digital Signal Controller (IC DSPIC33F)
Noise generation scheme	Shift registers + LPF	Digital Signal Controller (IC DSPIC33F)	Above chip is also used for noise generation

voltage at a test frequency depends on the frequency characteristic and the frequency response of the transducer and the frequency-dependent dB SPL-to-dB HL relation. For example, if a pure tone of 4 kHz with 1 V_{rms} produces 120 dB HL for a transducer, the sound produced by a more sensitive transducer for same input may be 140 dB HL. To perform air conduction test we need output from −10 to 120 dB HL and to perform bone conduction test we need output from −10 to 70 dB HL. The characteristics of various commercially available audiometric transducers are given in Appendix C. Based on the characteristics of various transducers, it can be inferred that the required dynamic range for conducting air conduction test with different transducers is 150 dB and for bone conduct test is 100 dB.

In case of digitally synthesized waveforms, attenuation can be realized by scaling down the samples. However, scaling of the digitally synthesized waveform before outputting from the digital-to-analog converter reduces the number of bits of the signal samples and hence increases the quantization noise-to-signal ratio. Therefore, the signal attenuation for the audiometer has to be carried out using attenuators after digital-to-analog conversion.

In earlier designs [15] – [17], an audiometer was developed with its own keypad and LCD for user interface, signal generation, response recording, audiometry in manual and

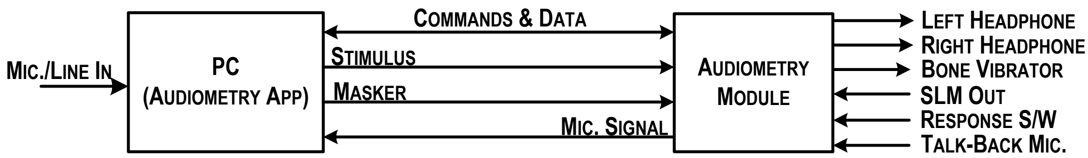


Figure 3.1: Design approach of the system.

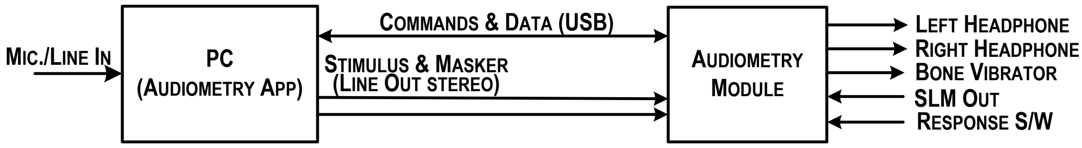


Figure 3.2: Design approach of the first version using USB and PC sound card output.

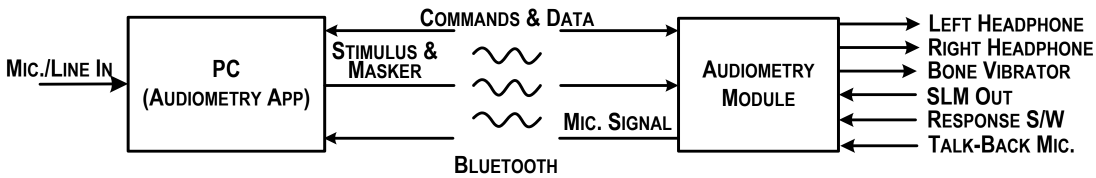


Figure 3.3: Design approach of the second version using wireless interfacing for signals and commands.

automated modes, and serial interface for transferring the test results to PC. This approach posed restrictions in modifying the user interface and in developing different audiometric test procedures. Using a software on a desktop, notebook, or tablet PC for user interfacing and signal generation can remove these restrictions. Hence the project objective is to develop an external hardware unit to serve as an audiometry module connected to a PC along with application software on PC for performing audiometry in a flexible manner.

A block diagram of the design approach is shown in Figure 3.1. It consists of a PC with application software for user interface and signal generation, and an audiometry module for signal presentation through transducers, acquiring subject response, microphone input, and acquiring sound level meter's dc output for calibration purposes. The commands for controlling the peripherals, and the sounds required for various audiometric procedures are transmitted to the audiometry module for signal attenuation and presentation. The microphone input or the line-in of the PC can be used to perform live speech audiometry or for giving instructions to the patient sitting in the soundproof room. The stimulus and masker can be presented through left headphone, right headphone, or bone vibrator in any combination. Also, the stimulus and the masker can be presented through the same transducer as required for some of the psychoacoustic tests. The microphone input to the audiometry module can be used for communicating with the patient sitting in a soundproof room. Since the audiometry module does not synthesize the test tones by itself, this approach offers advantages over earlier designs in terms of cost and complexity.

Two versions of the audiometry module have been developed and tested. The first version uses wired communication between the audiometry module and the PC. Its block diagram is shown in Figure 3.2. The line-out from the PC sound card is used for outputting the stimulus and masker sounds as required for audiometry and the USB is used for transmitting commands and powering the audiometry module. The microcontroller-based audiometry module is used to extend the dynamic range of the audio signal. Dynamic range of an audiometer must be at least 150 dB, as explained in Appendix C, so as to perform audiometry with a wide range of transducers. It has been found that large dynamic range as required for audiometry is difficult to realize using programmable logarithmic attenuators due to ground noise in the power amplifier output. Therefore, a dynamic range of 124 dB with 1 dB resolution is provided using programmable logarithmic attenuators and a switchable 40-dB passive attenuator is used after the power amplifier to boost the dynamic range without adversely affecting the SNR of the sound. For switching the passive attenuators, DPDT relays are used. The audiometry module has three outputs for connecting to left headphone, right headphone, and bone vibrator. The design uses two power amplifiers where the outputs are connected to the appropriate transducer using relays. Sound level meter's dc output is used for calibration for the automatic calibration of the transducers. A detailed description of this version is given in Appendix D.

After assembly and detailed testing of first version of the system, the design was revised as the second version to provide all the communication links between the audiometry module and the PC are wireless through Bluetooth for communicating commands and data, stimulus and masker, and microphone signal. By making the hardware completely wireless, the cabling that is required from an audiometer to the soundproof room is eliminated and the waveforms are not affected by noise in the PC circuit. The block diagram of the design approach is shown in Figure 3.3. Apart from making the audiometry module wireless, the DPDT relays that were used in the first version, for selecting the transducer and switching the 40-dB passive attenuator, are replaced by analog audio switches to reduce the power consumption. Also a battery charging and powering circuit is incorporated to make the module a compact battery-powered instrument. A detailed description of this version is presented in the next chapter.

Page intentionally left blank

Chapter 4

WIRELESSLY INTERFACED AUDIOMETRY MODULE

4.1 Introduction

A subject-response audiometer is an instrument for generating audio signals and presenting them through transducers at calibrated levels over a wide dynamic range. The audio signals serve as the stimulus and masker as required for carrying an audiometric procedure. The designed system consists of a PC with an application software for user interface and signal generation and an external audiometric module for digitally controlled attenuation, signal presentation, subject response acquisition, and automatic calibration. The signals for stimulus and masker are synthesized on the PC, for providing flexibility in generating the test stimuli and maskers for various audiometric tests without involving the cost of developing dedicated hardware for signal generation and user interface. Use of digitally controlled attenuator post signal generation in the external audiometric module permits generation of low-distortion stimuli over large dynamic range as needed for audiometry.

A compact battery-powered wirelessly interfaced audiometry module (WIAM) is designed to perform audiometry by communicating with the PC over Bluetooth. The block diagram of the WIAM is shown in Figure 4.1. It consists of a microcontroller for controlling the peripherals, a Bluetooth transceiver for receiving the audio signal and exchanging controls and data with the PC, a 2-channel pre-amplifier for amplifying and converting the differential audio signal to single-ended signal, a 2-channel digital attenuator for controlling the audio levels in each channel, a 2-channel power amplifier for driving transducers, audio switches for switching a 40-dB attenuator that is present after the power amplifier and for selecting transducers for the stimulus and masker presentation, a dual 4-channel analog multiplexer and a level detector for selecting and monitoring the output voltage levels across the transducers or selecting the output of the sound level meter for automatic calibration of a transducer, power management system for charging the batteries and powering the ICs, and a touch switch for switching on/off the module.

The detailed block diagram of the WIAM is shown in Figure 4.2. Its operation is controlled using PC through Bluetooth. The Bluetooth transceiver selected for this application supports serial port profile (SPP) and various audio profiles like advanced audio distribution

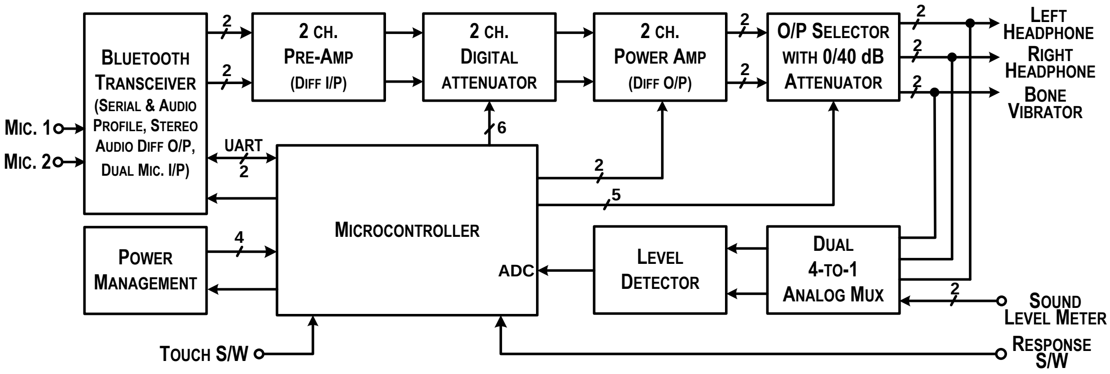


Figure 4.1: Block diagram of the external audiometric module.

profile (A2DP) and headset/hands-free profile (HSP/HFP) for outputting the audio stimulus and masker waveforms and receiving the microphone signals. The stereo audio signal is sent from the PC over Bluetooth using A2DP Bluetooth profile and the microphone signal from the Bluetooth transceiver to the PC is sent using HSP/HFP profile. The Bluetooth transceiver module has a 2-channel 16-bit DAC and an integrated amplifier with differential output for outputting the audio signal. The communication of the commands and control data between the PC and the Bluetooth module is done using SPP profile. The module is capable of connecting in audio profile (A2DP or HSP/HFP) and serial port profile (SPP) simultaneously i.e. the control data and the audio signals can be sent together. The bidirectional communication of data between the Bluetooth transceiver and the microcontroller is through UART serial interface.

The differential audio output of each channel of the Bluetooth transceiver is connected to the pre-amplifier. The pre-amplifier provides single-ended output with voltage gain of 2.2 and the dc bias VBIAS to ensure the signals are well within the permitted signal swing. The pre-amplifier output of each channel is fed to a digitally programmable logarithmic attenuator with the dynamic range of 124 dB and resolution of 1 dB. The attenuation settings are loaded in the attenuator by microcontroller using SPI. The attenuator output is fed to the power amplifier with differential input, bridge-tied load (BTL) output, gain of 2, and a dc bias. Use of BTL configuration doubles the output voltage swing. It does not require dc blocking capacitors and hence the load can be directly coupled resulting in flat low-frequency response. The power amplifier has a shutdown pin which is controlled by the microcontroller to save power when the amplifier is not in use.

The dynamic range of attenuator must be at least 150 dB so as to perform audiometry with various transducers in air conduction tests. The dynamic range provided by the

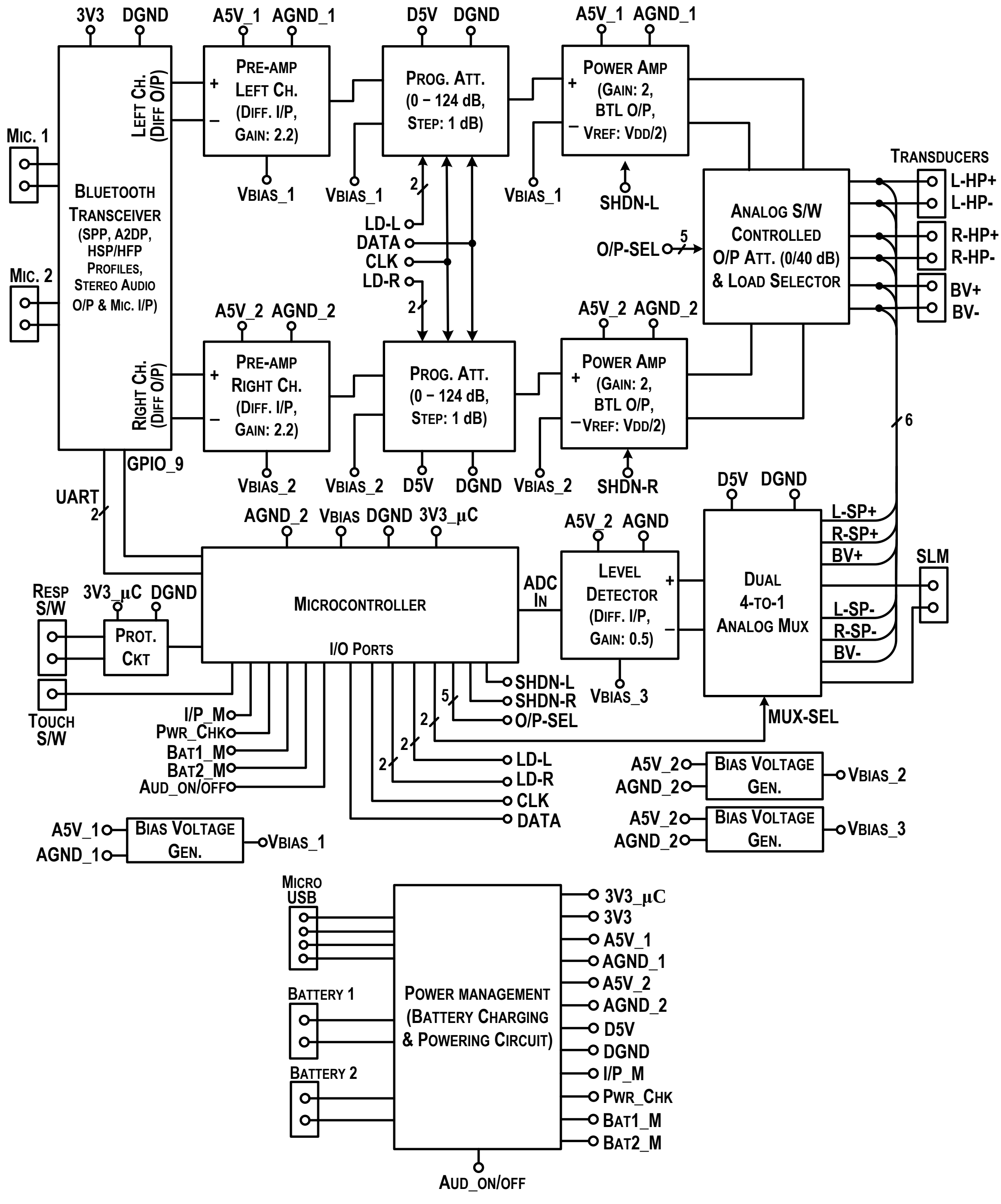


Figure 4.2: Detailed block diagram of the WIAM.

programmable attenuator is 124 dB which is not sufficient for audiometry. It can be extended by cascading another programmable attenuator. However, the SNR at low level output gets limited by the noise of the power amplifier. Use of an attenuator after the power amplifier does not adversely affect the SNR. Hence a switchable attenuator of 40-dB is used after the power amplifier. The output of the audiometry module can be connected to left headphone, right headphone, or bone vibrator in any combination. For switching the post-amplification attenuators and selecting the transducer at the output, analog audio switches controlled by the microcontroller port pins are used.

A multiplexer and level detector is used to detect the transducer and monitor the output signal level. For this purpose, a dual-channel 4-to-1 multiplexer is used to select the differential voltage at the transducer terminals of left headphone, right headphone, or bone vibrator. The fourth channel is used for connecting analog dc output of a sound level meter for automated calibration of the instrument. The multiplexer output is fed to a level detector which is a differential input full-wave rectifier followed by a low-pass filter. The dc signal from the level detector is applied to the ADC input of the microcontroller. A response button is interfaced through a protection circuit for acquiring the patient response and providing acknowledgement on acceptance of the response. For a simpler assembly and durability of the instrument, no switches, buttons, or keys are used in the design. A capacitive touch switch is interfaced with the microcontroller for serving as on/off button. In "off" mode, the peripherals are powered down and the microcontroller enters a "power save" mode.

The commands for controlling the peripherals like attenuators, analog switches, power amplifiers, response button, and multiplexer is sent over SPP profile of the Bluetooth from the PC. The received data from the Bluetooth is communicated to the microcontroller through UART serial interface.

A separate regulator for powering the digital chips is used to reduce the coupling of digital noise to analog signals. The audiometry module is designed as a battery-powered device. A power management circuit has been designed to provide two 3.3 V regulated supplies (3.3_μC, 3V3) and three 5 V regulated supplies (A5V_1, A5V_2, D5V) from two Li-Ion batteries. A separate 5 V regulator for each channel is used to power the analog ICs (op amps and power amplifiers) to reduce the crosstalk between the power rails. Separate bias voltages VBIAS_1 and VBIAS_1 are locally generated for each channel using the power supply of that channel. VBIAS_3 is a locally generated as the reference voltage for the level detector circuit. A separate 5 V regulator for powering the digital chips is used to reduce the coupling of digital noise to analog signals. A separate 3.3 V regulator is used for powering the microcontroller and the Bluetooth module to avoid the digital noise affecting the DAC output of the Bluetooth module.

A micro USB input is provided to charge the Li-Ion batteries. When charging the batteries, the power management circuit configures the batteries terminals to charge them individually. When the charger is removed, the power management circuit connects the

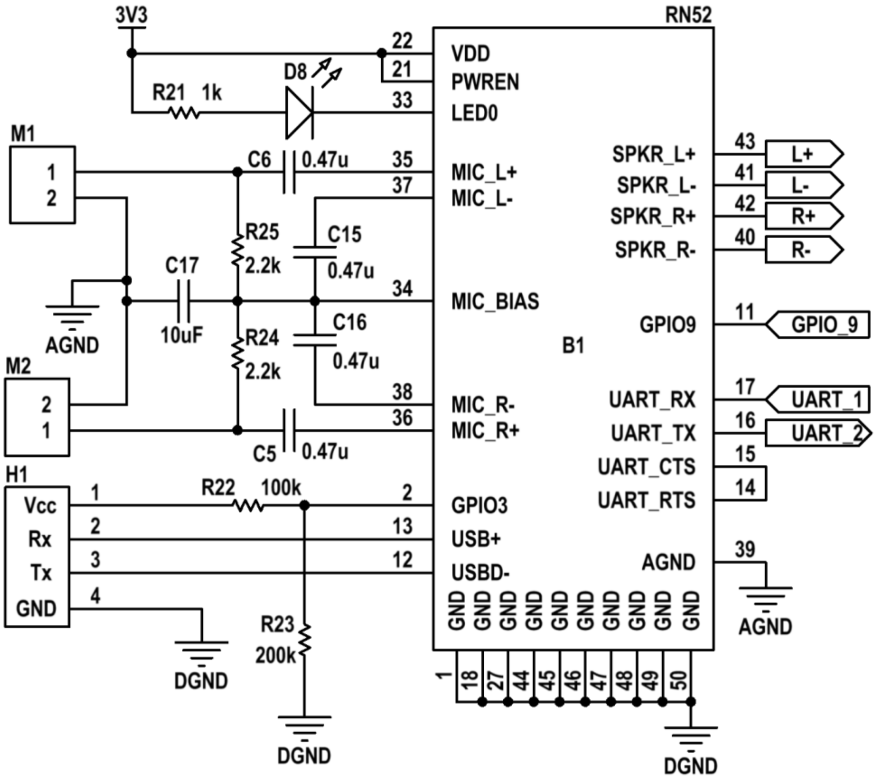


Figure 4.3: Circuit diagram of the Bluetooth transceiver.

batteries in series to power the audiometry circuit. The individual blocks of WIAM are described in the following sections, followed by a description of the microcontroller program for handling the communication with the PC over Bluetooth.

4.2 Bluetooth Transceiver

The Bluetooth transceiver is used for the wireless connectivity with devices like laptop and mobile phones, for exchanging signals and commands. The Bluetooth transceiver RN52 [18] (from Roving Networks) was selected for our purpose. Its circuit diagram is shown in Figure 4.3. It is a Bluetooth 3.0 module with the support profiles like advanced audio distribution profile (A2DP), headset/hands-free profile (HSP/HFP), serial port profile (SPP). It comes in a 50-pin SMD package with an integrated trace antenna. It has 10 m range, generally sufficient for indoor application. It works on supply voltage of 3.3 V with the operating current of 30 mA and standby current (i.e. when disconnected) of less than 0.5 mA. Its A2DP audio profile is used for receiving audio samples of up to 24-bit depth and 44.1 kHz sampling rate. The received audio samples can be routed as digital output through I2S or S/PDIF, or as analog output through an in-built 16-bit stereo DAC. It also has an integrated amplifier after the DAC with the programmable gain of −21 to 0 dB in 3-dB steps. Each audio channel has a differential output for low susceptibility to external noises and doubling the signal amplitude. The maximum output swing of the differential audio output is 1.7 V_{pp} with an SNR of 95 dB.

The Bluetooth transceiver module has support for two microphone inputs M1 and M2. One microphone is installed on the WIAM itself for monitoring the ambient noise and the second microphone input is used as talk-back microphone for receiving verbal response from the patient. The bias voltage for the microphones is generated by the Bluetooth transceiver. The Bluetooth transceiver has a differential input pre-amplifier for each microphone input with a switchable gain of 24 dB and a programmable gain of 18 dB in 3-dB steps. The switchable gain of 24 dB allows using a microphone or a line-in and the programmable gain can be used to optimize gain for different microphones. The input impedance of the pre-amplifier is 6 kΩ. The values C15, C16 are selected to keep the cutoff frequency of the high-pass filter $1 / (2\pi C_{15} \times 6000)$ greater 50 Hz. With $C_{15} = C_{16} = 470$ nF, the high-pass cutoff frequency is 56 Hz. The HSP/HFP Bluetooth profile is used for sending the microphone signal from the module to the PC.

The SPP profile of the Bluetooth transceiver is used for exchanging the data wirelessly with the PC and the UART interface is used for exchanging the data with the microcontroller. The Bluetooth transceiver communicates through UART in two modes *viz.* data and command. In the data mode, the module relays the data it receives over Bluetooth to the UART and vice versa. The Bluetooth transceiver module enters into command mode when its GPIO9 pin is pulled low. In command mode, the Bluetooth transceiver receives the data through UART for setting its configuration parameters like device name, security pin, automatic shutdown time, GPIO port pins, connection mask, Bluetooth profiles, microphone gain levels, speaker gain level etc. The connector H1 is used for connecting to the USB port of the PC for updating the firmware of the Bluetooth transceiver. The Bluetooth transceiver enters the firmware upgrade mode when its GPIO3 pin is pulled high using the supply from the USB port. The LED D8 conveys the status of the module i.e. whether it is in configuration mode, discoverable mode, or firmware update mode.

4.3 Pre-amplifier

The RN52 Bluetooth provides two differential audio outputs: the pins 'L+' and 'L−' for the left channel and the pins ‘R+’ and 'R−' for the right channel. The circuit, shown in Figure 4.4, uses op amp U1A and U2A from two quad op amp ICs MCP604 [19] (from

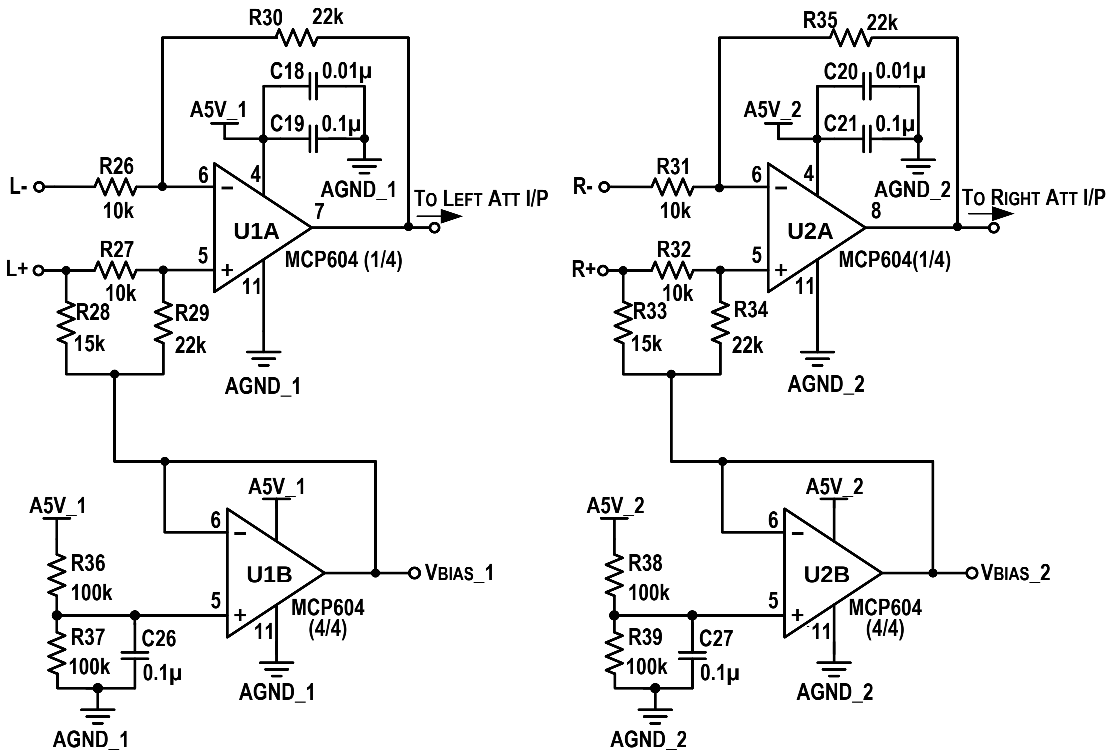


Figure 4.4: Pre-amplifier circuit for left and right audio channels along with the bias voltage generators.

Microchip). This IC has low input bias current (<10 pA) and rail-to-rail output voltage. The pre-amplifier circuits for the left and right channels are identical.

Separate ICs are used for each channel to avoid signal crosstalk and separate supply voltages are used to avoid signal crosstalk through the power supply rails. Two bias voltages, each of 2.5 V are generated using their respective supply voltage, for each audio channel i.e. VBIAS_1 for left and VBIAS_2 for right. for each channel the differential voltage from the Bluetooth transceiver is converted to single-ended voltage superimposed on the bias voltage. The gains and the input resistances of the left and right channel pre-amplifier are similar. The gains and input resistances at the two input terminals of the left channel pre-amplifier are given as

$$A_{v(L-)} = -R_{30} / R_{26} \tag{4.1}$$

$$A_{v(L+)} = [R_{29} / (R_{27} + R_{29})][1 + R_{30} / R_{26}] \tag{4.2}$$

$$R_{in(L-)} = R_{26} \tag{4.3}$$

$$R_{in(L+)} = R_{28} \,||\, (R_{27} + R_{29}) \tag{4.4}$$

Table 4.1: A comparison of three digitally-controlled logarithmic attenuators.

Specification	**Attenuator IC**		
	PGA2311 [20]	**AD7111** [21]	**LM1971** [22]
Supply voltage	±5 V	5 V	4.5 to 12V
Communication	3-wire serial	8-bit parallel	3-wire serial
No. of channel	2	1	1
Dynamic range	120.0 dB	88.5 dB	62.0 dB
Resolution	0.5 dB	0.375 dB	1 dB
THD	0.0004%	0.0028%	0.0008%
Voltage reference pin	No	No	Yes
Zero-crossing detector	Yes	No	No

The resistor values are selected as $R_{26} = R_{27} = 10$ kΩ, $R_{28} = 15$ kΩ, and $R_{29} = R_{30} = 22$ kΩ, to get

$$A_{V(L+)} = A_{V(L-)} = 2.2$$

$$R_{in(L+)} = R_{in(L-)} \approx 10 \text{ k}\Omega$$

The circuit and the resistor values for the right channel are the same as for the left channel, resulting in the same gain and input resistance. The circuit for generating the bias voltage for each channel is also shown in Figure 4.4. A potential divider comprising two equal valued resistors is used for generating half of the supply voltage. The op amp U1B, one of the four op amps in IC MCP604, is used as a unity gain buffer. The capacitors C26 and C27 with values of 0.01 µF and 0.1 µF are used for noise filtering.

4.4 Attenuator

For each of the two channels, a digitally-controlled attenuator is required with dynamic range of approximately 150 dB and a resolution of 1 dB. Further the circuit has to operate with single 5 V supply with signal superimposed on a dc bias. Several logarithmic attenuator ICs were considered for this application and a brief comparison of three of them is shown in Table 4.1. Although all have sufficient resolution, cascading of the chips is needed for getting the required dynamic range. The IC LM1971 [22] (from Texas Instruments) with single 5 V supply operation, provision for setting the reference voltage, and 3-wire serial interface was selected. It has a dynamic range of 62 dB, step size of 1 dB, and total harmonic distortion of less than 0.0008%. The attenuation step of 1 dB facilitates calibration for

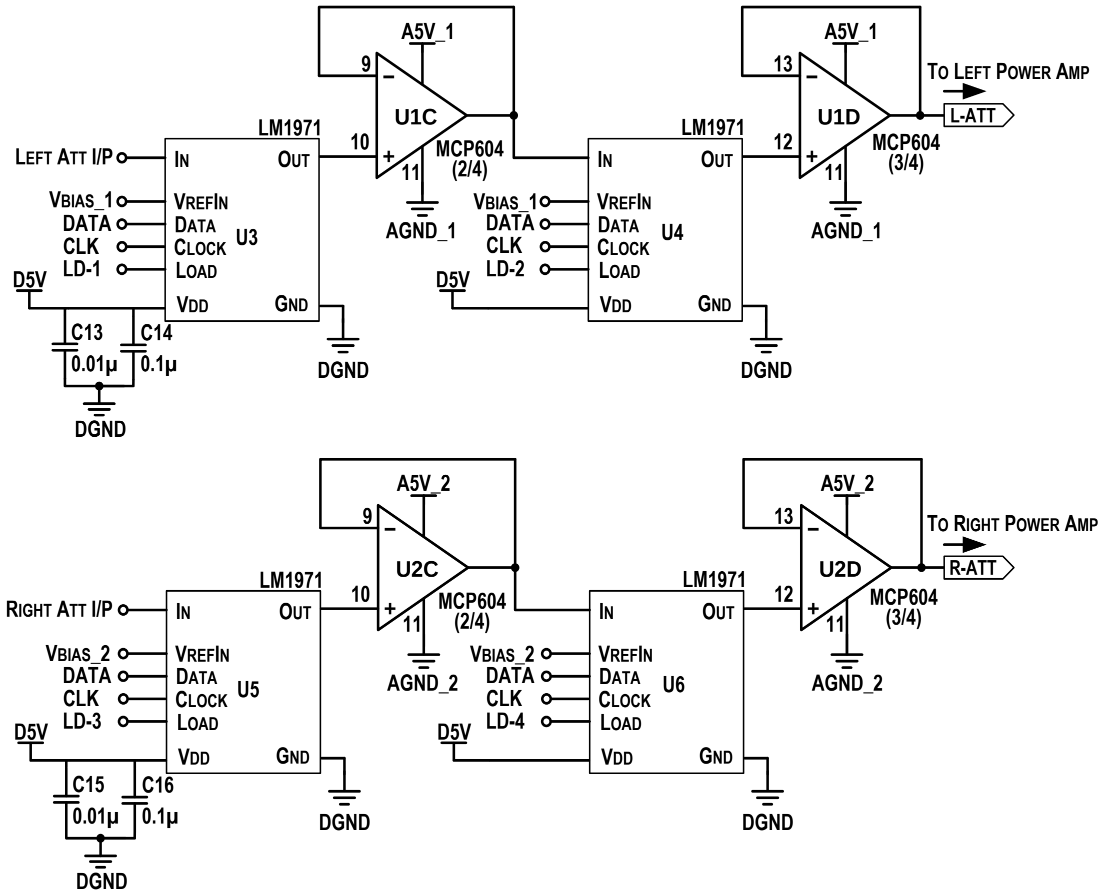


Figure 4.5: Attenuator circuit for the left and right channels.

different transducers. Zero-crossing detection in the attenuator chips is used to delay the new attenuation setting until the next zero crossing in order to avoid audible clicks associated with abrupt changes in the output voltage. The selected chip does not have this feature, but it is not a drawback for audiometric application because the attenuation setting can be changed when audio signal is zero. Two chips are cascaded in each channel in order to extend the dynamic range to 124 dB. The output impedance of the attenuator IC varies non-linearly between 25 – 35 kΩ with the attenuation steps and hence it requires a buffer with low input bias current to avoid clicks at switching due to dc offsets. An additional switchable 40-dB attenuator, described later in Section 4.6, is provided after the power amplifier so as to extend the range of attenuation.

The attenuator circuit for the two audio channels is shown in Figure 4.5. The left channel has attenuators U3 and U4, with buffering by op amps U1C and U1D, respectively. The similar functions for the right channel are performed by U5, U6, U2C, U2D. The attenuator IC has 3-wire synchronous interface consisting of DATA, CLOCK, and LOAD pins. As the attenuator chip does not have serial out, the chips cannot be interfaced using

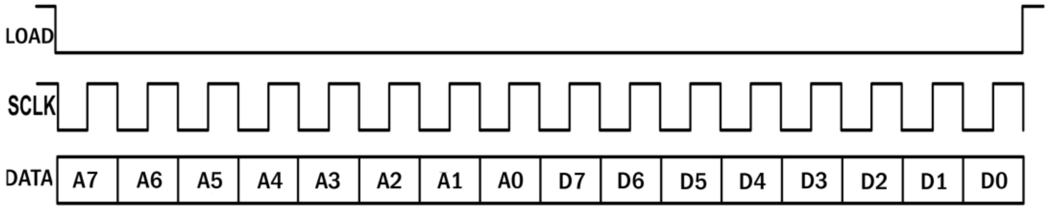


Figure 4.6: Timing diagram of the serial interface of the logarithmic attenuator LM1971 [22].

daisy chaining and have to be interfaced using separate load pins. The four attenuator ICs share the DATA and CLOCK pins, while the LOAD pin of each one is connected to a separate pin of the microcontroller. The DATA and CLOCK pins are connected to RD4 and RD5 of the microcontroller. These pins are the remappable pins of the microcontroller and they are mapped to SDO and SCK of an SPI module in the software. The timing diagram for the 3-wire serial interface is shown in Figure 4.6. An active low on the LOAD pin enables the data input register of the attenuator IC. The DATA pin receives serial data corresponding to the value of the attenuation on each rising edge of the CLOCK. The serial data are composed of 8-bit address and 8-bit attenuation setting, with A0 as the address LSB and D0 as the data LSB. The device 8-bit address for the attenuator is 0x00h. The specific attenuator chip is selected by active low on corresponding LOAD pin (LD-1, LD-2, LD-3, or LD-4). The attenuation setting corresponds to the attenuation in dB with a maximum of 0x3Eh for 62 dB.

4.5 Power Amplifier

The power amplifiers for the left and right channels are realized using ICs U7 and U8. For this purpose, we have selected the IC TPA6211A1 [23] (from Texas Instruments). It is a 3.1-W amplifier with differential input and bridge-tied load (BTL) output for driving load greater than 3 Ω. The BTL outputs are 180^0 out of phase, thereby doubling the output voltage swing across the load as compared to the single ended output configuration for the same supply voltage. As the BTL output is differential, a coupling capacitor is not required at the output to block the dc to the headphone coil and it results in a flat low-frequency response.

The power amplifier circuits for left and right channels are shown in Figure 4.7. The power amplifier IC has on-chip 40 kΩ resistance in the feedback path, and hence R21 and R22 of 20 kΩ are used to provide a gain of 2. Capacitor C22 of 1 uF is used for filtering the internally generated reference voltage. The shutdown pin is connected to save power when signal presentation is not required. Each pin of the BTL output is connected to two 10 Ω resistors in parallel to provide net source resistance of 10 Ω. Although it reduces the output voltage swing, it flattens the frequency response of the headphone by reducing the change in

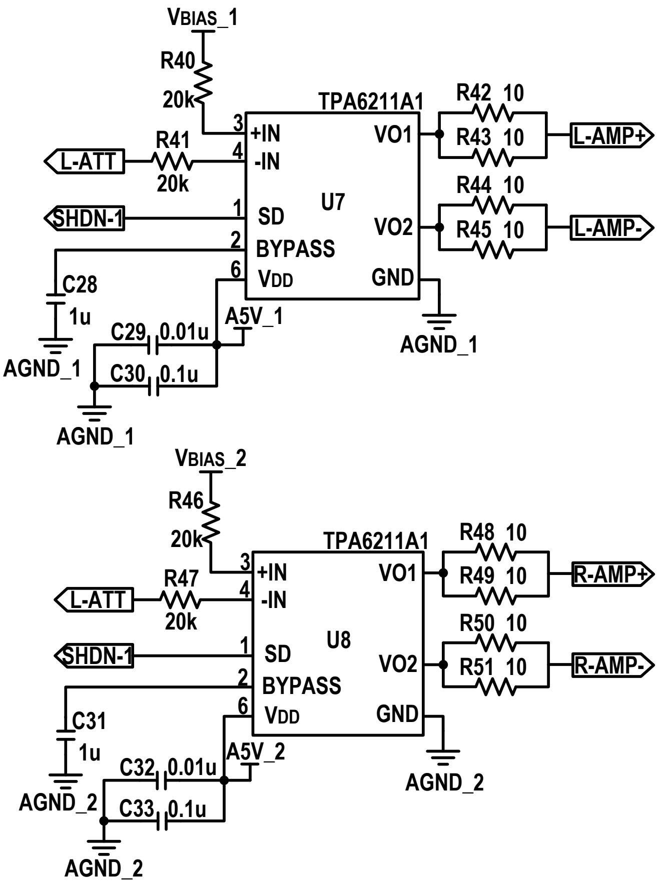


Figure 4.7: Power amplifier circuits for left and right channel.

impedance with frequency and acts as a current limiter at accidental short of the output terminals. The source resistance is also used to detect whether a transducer is connected by measuring the voltage across the output.

4.6 Transducer Selection and Output Attenuator Switching

The audiometry module is designed for connecting to three transducers, *viz.* the left headphone, the right headphone and the bone vibrator. These can be connected in different combinations to the left and the right audio channels for different types of tests. For air conduction test of the left ear, the stimulus is applied to the left audio channel and presented through the left headphone, while the masker is applied to the right audio channel and presented through the right headphone. For air conduction test of the right ear, the stimulus is applied to the right audio channel and presented through the right headphone, while the masker is applied to the left audio channel and presented through the left headphone. For bone conduction test of the left ear, the stimulus is applied to the left audio channel and presented through the bone vibrator, while the masker is applied to the right audio channel

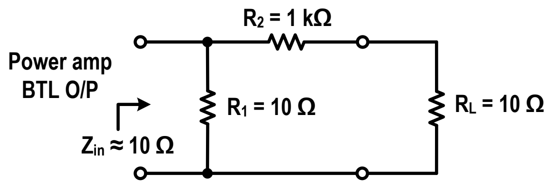


Figure 4.8: The L-pad 40-dB attenuator circuit.

and presented through the right headphone. For bone conduction test of the right ear, the stimulus is applied to the right audio channel and presented through the bone vibrator, while the masker is applied to the left audio channel and presented through the left headphone.

In order to extend the dynamic range of the audio signal to the headphones, a switchable 40-dB attenuator is provided after the power amplifier stage. As it is present after the power amplifier, it does not alter the signal-to-noise ratio. The bone vibrator is connected without the switchable attenuator as it requires lower output dynamic range as compared to air conduction transducers (explained in Appendix C). An L-pad attenuator circuit is designed for the BTL output from the power amplifier.The circuit is shown in Figure 4.8. The resistance values for this output attenuator are selected such that its input impedance for a load of 10 Ω is 10 Ω. The circuit for transducer selection and output 40-dB attenuator switching is shown in Figure 4.9. In this circuit, DPDT analog audio switches A1, A2, A3, A4, and A5 are used for the selection of transducers and switching the output attenuator. The DPDT analog audio switche used for this purpose is MAX4993 [24] (from Maxim Integrated). It has low on-resistance of 300 mΩ with the flatness of 1 mΩ over the full supply range, and 0.004% THD + N. Also, it has slow turn-on time to reduce click-and-pop noise caused by abrupt changes in the voltage across a transducer. These voltage change occurs when a single supply audio amplifier with a dc bias is turned on, causing a spike of current. It has an off-isolation of −120 dB and crosstalk of −130 dB at 1 kHz. The switches are controlled using O/P-S0, O/P-S1, O/P-S2, O/P-S3, and O/P-S4 from microcontroller port pins. Tables 4.2 and 4.3 show the microcontroller port pin outputs for selecting the transducers and switching the attenuators, respectively. It may be noted that the 40-dB post-amplfier attenuator assumes the transducer to be a resistive load of 10 Ω. Any deviation in the transducer impedance from 10 Ω resistance, will change the value of the attenuation and needs to be compensated for during calibration.

4.7 Circuit for Output Level Monitoring

The output level monitoring circuit, shown in Figure 4.10, is used for monitoring the voltage output levels to the transducers. A dual 4-to-1 analog multiplexer U9 is used for

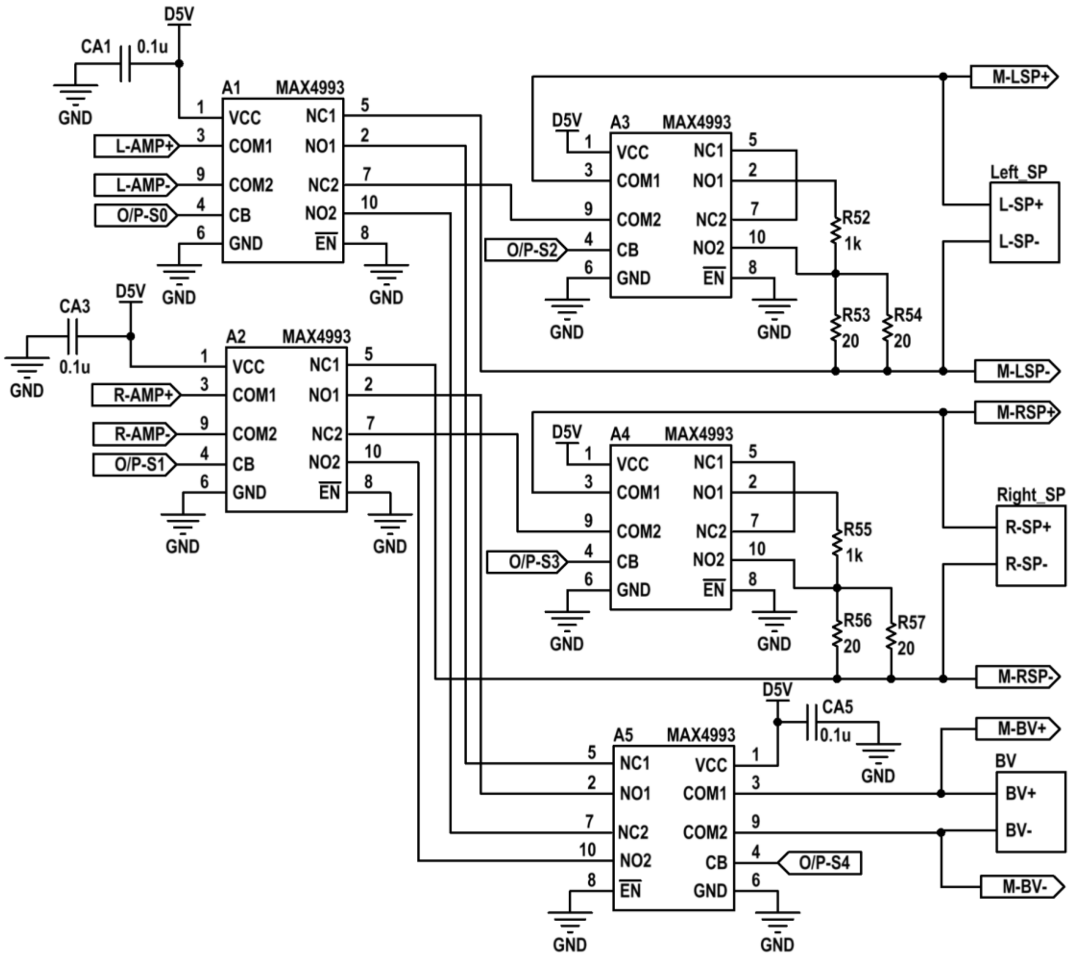


Figure 4.9: Circuit for transducer selection and 40-dB attenuator switching.

selecting the transducer (left headphone, right headphone, or bone vibrator) for monitoring the output being applied to it. The IC CD4052 [25] (from Texas Instruments) is used for this purpose. The selected voltage is applied to the difference amplifier to generate single-ended output and fed to the full-wave rectifier and low-pass filter to get the average value $2V_P/\pi$, where V_P is the peak value. The op amp used for realizing this circuit is MCP604 [19] (From Microchip), which is a quad op amp IC. This voltage is fed to the ADC of the microcontroller. The values of the resistors R58, R59, R60, R61, and R62 are selected so that the input resistances at the two input terminals of the difference amplifier are equal and the gain is 0.5. As the BTL output of the power amplifier is ±5 V, the gain of the difference amplifier is set to avoid clipping of the signal after biasing the input signal to VBIAS_3. With R_{58}, R_{59} = 100 kΩ, R_{60} = 300 kΩ and R_{61}, R_{62} = 50 kΩ, the input resistance is 100 kΩ and the gain is 0.5. The values of C35 and R67 forming the low-pass filter are selected to keep the cutoff frequency $1/(2\pi C_{35}R_{67})$ much lower than the lowest audiometric frequency of 125 Hz.

Table 4.2: Controls for selecting transducers for the two audio channels. First three combinations are preferred as they consume less relay activation currents.

Control pins			Transducer connected to the left audio channel	Transducer connected to the right audio channel
O/P-S0	**O/P-S1**	**O/P-S4**		
0	0	0	Left headphone	Right headphone
1	0	0	Bone vibrator	Right headphone
0	1	1	Left headphone	Bone vibrator
0	0	1	Left headphone	Right headphone
0	1	0	Left headphone	None
1	0	1	None	Right headphone
1	1	0	Bone vibrator	None
1	1	1	None	Bone vibrator

Table 4.3: Controls for switching the 40-dB attenuators.

Control pins		Attenuation	
O/P-S2	**O/P-S3**	**Left headphone**	**Right headphone**
0	0	0 dB	0 dB
0	1	0 dB	40 dB
1	0	40 dB	0 dB
1	1	40 dB	40 dB

With R_{67} = 200 kΩ and C_{35} = 220 nF, the low-pass cutoff frequency is 3.6 Hz. The bias voltage VBIAS_3 for this circuit is locally generated by utilizing an op amp from the quad op amp IC U10 as shown in Figure 4.10.

The circuit in Figure 4.10 is also used for automatic calibration of the instrument for various transducers. It can be carried out by placing the transducers on the artificial ear in case of headphone and on the artificial mastoid in case of bone vibrator and using the sound level meter to obtain dc voltage corresponding to the sound pressure level of the sound from the transducer. The dc output from the sound level meter, corresponding to the rms value of the signal received by its microphone, is selected by the analog multiplexer U9. The ADC value corresponding to the level is transferred by the microcontroller to the PC for updating the calibration table.

Monitoring of the voltage across the output terminals can be used to check whether the transducer is connected at the output. The output voltage level is approximately halved if the transducer is connected because of the series resistance of 10 Ω. The output voltage is zero in case of a short at the output.

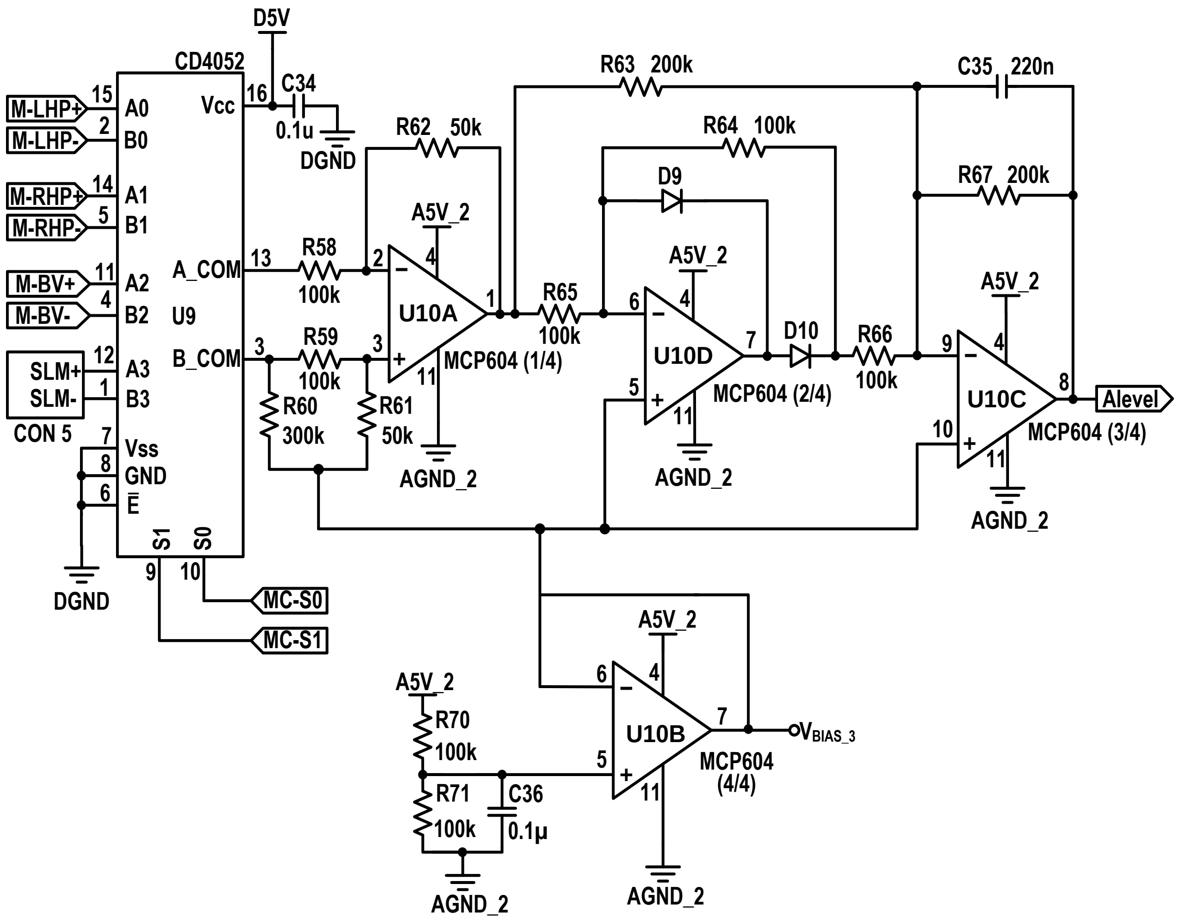


Figure 4.10: Circuit for output level monitoring along with the bias voltage generator.

4.8 Response and Acknowledgement Circuit

For receiving the response from the patient, a single-pole single-throw switch, in the form of a key or button, is interfaced to pin RE0 of the microcontroller U11. The response key K is connected to the audiometer circuit using a 2-wire cable as shown in Figure 4.11. The LED L1 is connected in parallel to the switch K to provide an acknowledgement for the key press. The Resp_S/W port of the microcontroller is configured as the input port when scanning for input. The LED L1 is normally off due to high value of R68. If the response is recorded, RE0 is set in output mode with logic high for few seconds. After the key is released, the LED L1 gets turned on because of logic high at RE0. Thus the circuit is used for receiving the subject response as well as providing acknowledgement without using an additional wire. The key K and LED L1 are in the subject response box while the rest of the components are in the WIAM. The combination of D11, D12, and C43 is used to suppress any spikes which may get picked up in the cable to the subject response key.

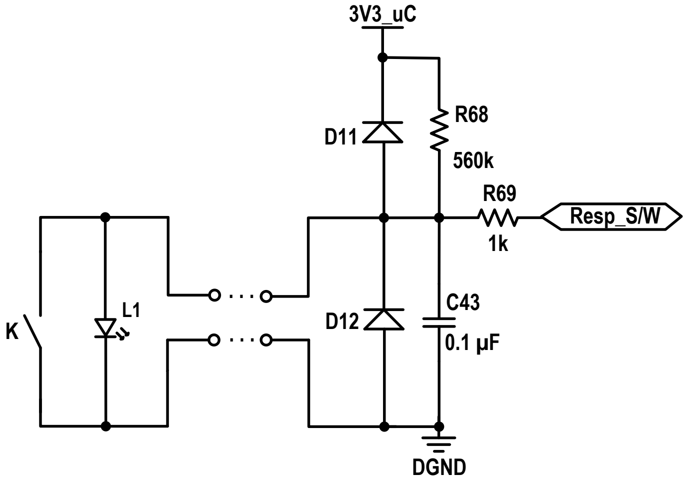


Figure 4.11: Circuit for response and acknowledgement

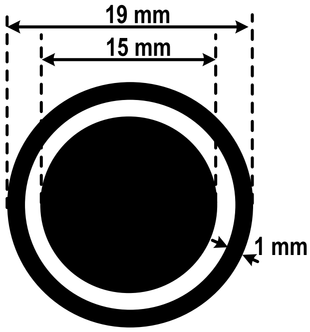


Figure 4.12: Layout of the capacitive touch switch.

4.9 Touch Switch

A capacitive touch switch is interfaced with the microcontroller to serve as an on/off button for switching the power to the peripherals. The switch consists of a circular touch pad of diameter 15 mm (average person's finger press size), surrounded by a concentric ring of width 1 mm and separation 1 mm. The PCB layout of the capacitive touch switch is shown in Figure 4.12. The outer concentric ring of the touch switch is connected to the ground.

The capacitance models of the touch pad without and with a finger near it are shown in Figure 4.13. Let the net capacitance between the touch pad and the circuit ground be C_{o1} without the finger near it and C_{o2} with the finger near it.

$$C_{o1} = C_{PR}$$

$$C_{o2} = C_{PR} + C_{PF}\text{-series-}C_{FR}$$

where C_{PR} = capacitance between the touch pad and the outer ring, C_{PF} = capacitance between the touch pad and the finger, and C_{FR} = capacitance between the finger and the outer ring. The

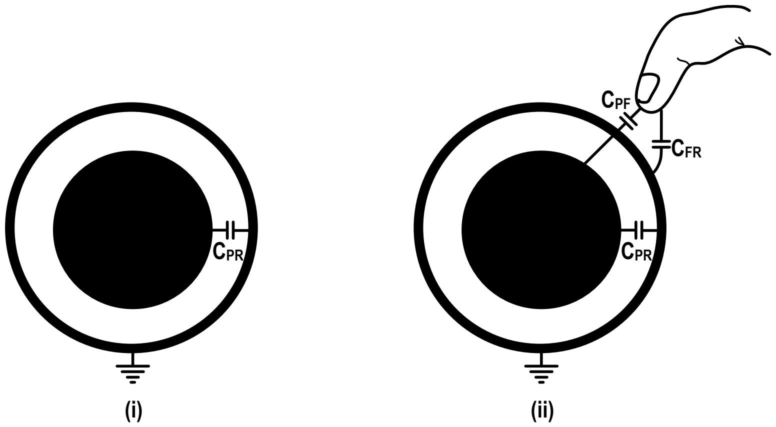


Figure 4.13: Capacitance model of the touch pad (i) without a finger near it, and (ii) with a finger near it.

shift in the capacitance due to the finger is $\Delta C = C_{o2} - C_{o1}$, or as $\Delta C = C_{PF}\, C_{FR} / (C_{PF} + C_{FR})$. The relative shift in the capacitance is given as

$$\frac{\Delta C}{C_{o1}} = \frac{1}{C_{PR}\left(\dfrac{1}{C_{FR}} + \dfrac{1}{C_{PF}}\right)} \tag{4.5}$$

Therefore, to increase the sensitivity, the capacitance between the fingers and the ring should increase and the capacitance between the pad and the ring should decrease. This can be achieved by making the outer ring wider and increasing the separation between the pad and the ring, but that will require a larger area. Thus, the dimension of the pad and the ring as shown in Figure 4.12 was selected to provide stability at the cost of reduced sensitivity. It also helps in reducing the possibility of false touches.

The microcontroller has a module known as charge time measurement unit (CTMU) which consists of a constant current source. The shift in the capacitance is detected by periodically charging the touch pad using the constant current source I for a fixed time interval t, and reading the resulting voltage V_t using ADC. The equation that relates capacitance C, current I, time interval t, and the voltage level V is given as

$$V_t = \frac{I}{C} t \tag{4.8}$$

When the touch pad is touched, the net capacitance is increased and hence the voltage reached after charging the touch pad with current I for a duration t decreases. The acquired ADC value is filtered and compared with threshold values to decide if the touch has occurred. The description of the filtering and decision making process is given in the microcontroller program section i.e. Section 4.12.

4.10 Microcontroller

The operations of the Bluetooth transceiver, attenuators, analog switches, power amplifiers, multiplexer, and touch switch are controlled by the microcontroller IC U11. A 3.3 V microcontroller IC PIC18F46J50 [26] (from Microchip) was chosen for this purpose. It has an SPI module for communication with the digital attenuators, UART module for communicating with the Bluetooth transceiver, 10-bit ADC for acquiring the output of the level detector circuit, charge time measurement unit (CTMU) for capacitive touch sensing, and sufficient number of I/O port pins for generating the controls to the analog audio switches, analog multiplexer, monitoring the battery charging status, monitoring battery voltage level, and interfacing the patient response switch. The circuit diagram of the microcontroller port connections is shown in Figure 4.14. Its internal RC oscillator of 8 MHz is used for providing clock to its CPU and its peripherals. The microcontroller has reconfigurable pins for mapping its peripherals like SPI and UART module to a wide range of I/O port pins. This is useful in optimizing the PCB layout. The output and input levels of this 3.3 V microcontroller are compatible with the logic levels of the digital attenuator IC and the analog audio switch IC that are operating on 5 V supply. The microcontroller has a high-low voltage detect (HLVD) module that compares the external voltage with an internally generated bandgap reference voltage source. This HLVD module is used to detect the battery voltage to avoid it from over discharging. The VREF+ is connected to VBIAS_3 and the VREF- is connected to AGND_2 for the proper operation of the ADC. The connector DEBUG is a 5-pin connector provided for inline programming and debugging of the microcontroller using its PGD, PGC, and MCLR pins. To filter the output of the on-chip voltage regulator, capacitor C41 (10 uF) is connected between VDDCORE and DGND. The functions assigned to various port pins of the microcontroller are listed in Table 4.4.

4.11 Power Management Circuit

A power management circuit is designed to charge the Li-Ion batteries and to provide separate regulated supplies for the analog and digital ICs. The block diagram of the power management circuit is shown in Figure 4.15. The circuit consists of a micro USB port for using external 5 V supply for charging batteries, two independent battery charger ICs, two switches for configuring the battery terminals, input power selector for selecting either battery or external 5 V supply for powering the microcontroller voltage regulator, a switch for turning on/off the battery supply to the voltage regulators, voltage attenuators for providing attenuated input of 5 V charger supply and the battery voltage level to the microcontroller for monitoring purpose, and five voltage regulators for providing separate supplies to digital and

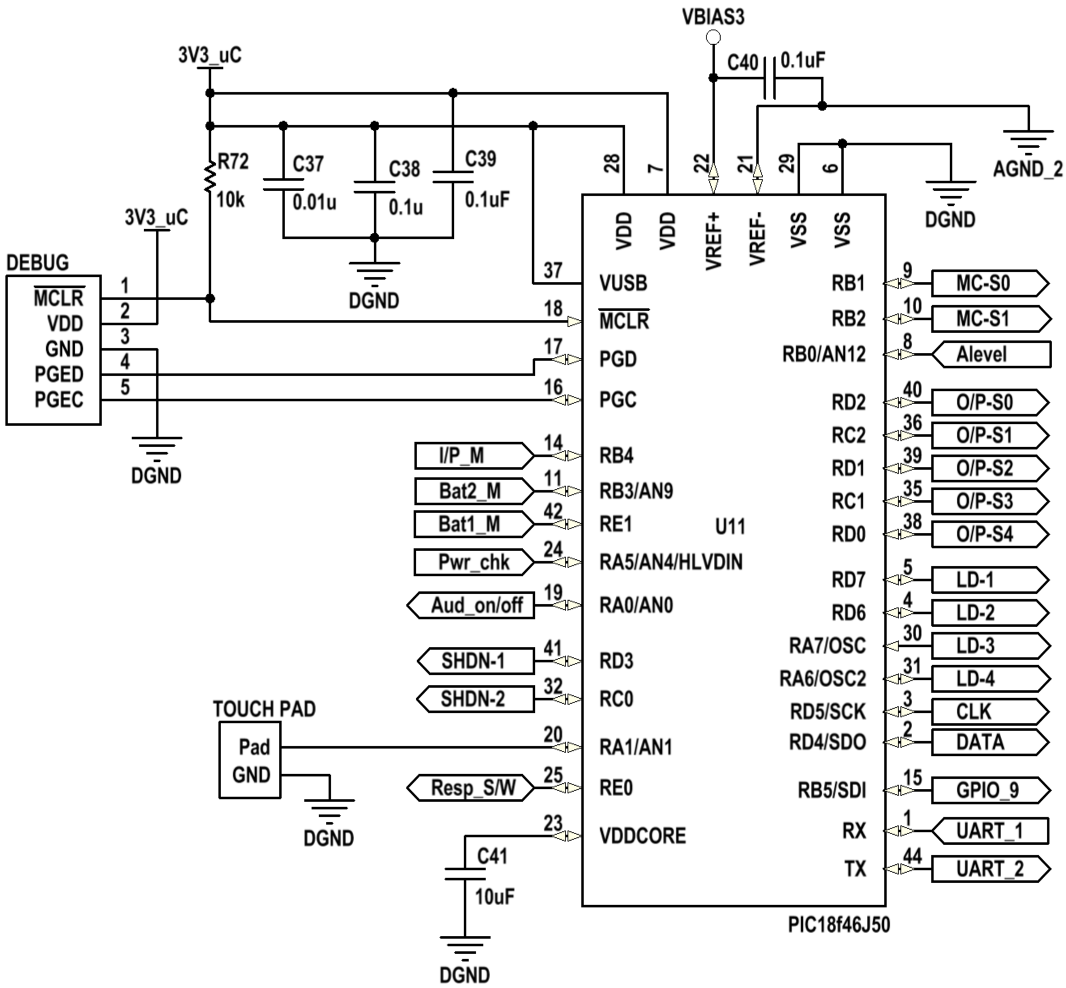


Figure 4.14: The microcontroller circuit.

analog ICs in the WIAM. The description of the power management circuit is divided in to a battery charging circuit and a powering circuit and is described subsequently.

The battery charging circuit is designed to charge two Li-Ion batteries using 5 V input from either USB port of a PC or from a 5 V charger. The connector used for charging is a micro USB connector which is commonly used for charging portable devices. The nominal voltage of a Li-Ion battery is 3.7 V. Since the hardware needs a supply of 5 V, for powering the module two Li-Ion batteries is connected in series and a linear regulator is used thereafter. There can be two approaches to charge two Li-Ion batteries from a 5 V supply. The first approach is to use a dc-dc converter to boost the input voltage and charge the batteries in series. The batteries needs to be balanced while charging them in series to avoid overcharging of one of the two batteries. In the second approach, the batteries can be charged separately and can be connected in series when powering the module. The drawback of this approach is that the module cannot be powered while charging the batteries. In our design, the second approach

Table 4.4: Function assigned to the I/O port pins of the microcontroller U11.

Port pins	In /Out	Function assigned
RB0	In	Alevel of the level detector circuit.
RD5/SCK	Out	CLOCK of the attenuator ICs U4, U5, U6, U7 (LM1971).
RD4/SDO	Out	DATA of the attenuator ICs U4, U5, U6, U7 (LM1971).
RD7	Out	LOAD of attenuator IC U3 (LM1971).
RD6	Out	LOAD of attenuator IC U4 (LM1971).
RA7	Out	LOAD of attenuator IC U5 (LM1971).
RA6	Out	LOAD of attenuator IC U6 (LM1971).
RC6/Tx	Out	UART Tx for serial interface with Bluetooth transceiver B1 (RN52).
RC7/Rx	In	UART Rx for serial interface with Bluetooth transceiver B1 (RN52).
RB6	In	PGC of programmer/debugger module (PICkit 3).
RB7	In	PGD of programmer/debugger module (PICkit 3).
RE3	In	MCLR of the programmer/debugger module (PICkit 3).
RE0	In/Out	Resp_button from response switch circuit.
RA2/VREF-	In	AGND_2 for ADC negative reference input
RA3/VREF+	In	VBIAS_3 for ADC positive reference input.
RB4	In	I/P_M for monitoring the charging source.
RB3	In	Bat1_M for battery 1 charging status monitoring.
RC4	In	Bat2_M for battery 2 charging status monitoring.
RA5	In	Pwr_chk for battery level monitoring.
RA0	Out	Aud_on/off for switching the power to the peripherals.
RA1	In/Out	Touch pad.
RD5	Out	GPIO9 of the RN52 Bluetooth transceiver module B1 (RN52).
RD2	Out	CB control input of analog switch A1 (MAX4993).
RC2	Out	CB control input of analog switch A2 (MAX4993).
RD1	Out	CB control input of analog switch A3 (MAX4993).
RC1	Out	CB control input of analog switch A4 (MAX4993).
RD0	Out	CB control input of analog switch A5 (MAX4993).
RB1	Out	S1 select line of multiplexer U10 (CD4052).
RB2	Out	S0 select line of multiplexer U10 (CD4052).
RD3	Out	$\overline{SD}$ shutdown pin of power amplifier IC U8 (TPA6211A1).
RC0	Out	$\overline{SD}$ shutdown pin of power amplifier IC U9 (TPA6211A1).

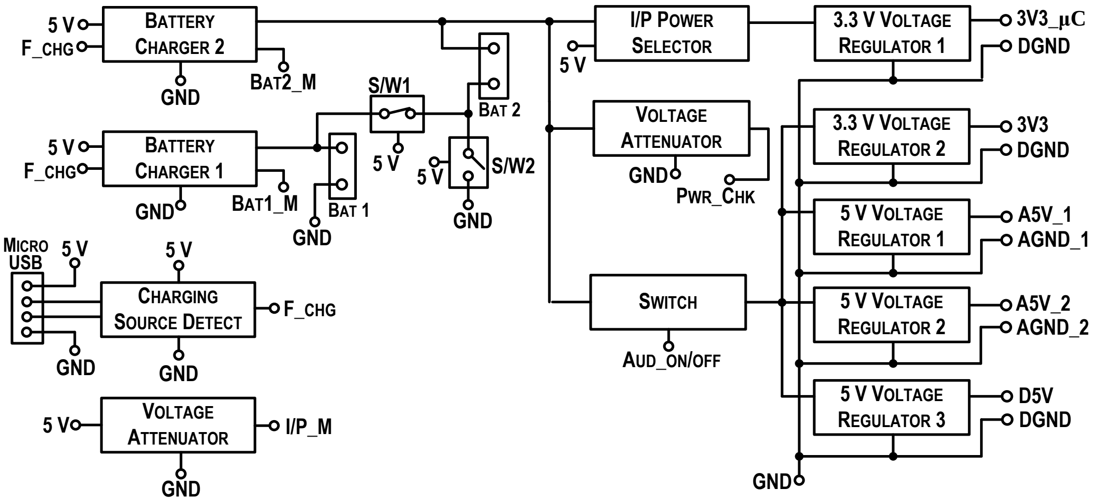


Figure 4.15: Block diagram of the power management circuit of the WIAM.

is used i.e. the batteries are charged individually using two separate battery charger ICs CH1 and CH2, and are connected in series using a switching arrangement when powering the module.

The circuit of the battery charger and the switching arrangement is shown in Figure 4.16. The battery charger IC used is BQ21040 [27] (from Texas Instruments). It is a linear charger with 1% charge voltage accuracy and its charging current during the constant current phase of the charging process can be programmed using an external resistor. It has an open drain status pin $\overline{\text{CHG}}$ which is low while charging and Hi-Z when charging is complete or not charging. An LED for each battery is used to indicate when the battery is charging and the corresponding $\overline{\text{CHG}}$ port is also monitored using the microcontroller. When an input charging device is connected, the gate of NMOS Q4 and PMOS Q3 is pulled high. This turns on the NMOS Q4 and turns off the PMOS Q4. The diode D5 gets forward biased while the diode D6 gets reverse biased because the voltage at Bat_Supply port is less than 5 V input voltage. Therefore, the power to the microcontroller is drawn from the input and not from the battery while charging. The voltage regulator IC VR1 used for supplying 3.3 V to the microcontroller is MCP1802 [28] (from Microchip) which has 25 μA quiescent current. A Schottky diode D5 with low forward voltage drop is used for the proper operation of the voltage regulator VR1. The diode used is DFLS130L [29] (from Diodes Inc.) with a forward voltage drop of 0.3 V for 1 A current.

A USB 2.0 port of a PC can deliver up to 500 mA of current. Since we are charging two batteries independently, the charging rate from the USB port of PC will be slow. The batteries can be charged faster when a charging device is used. To enable or disable the fast

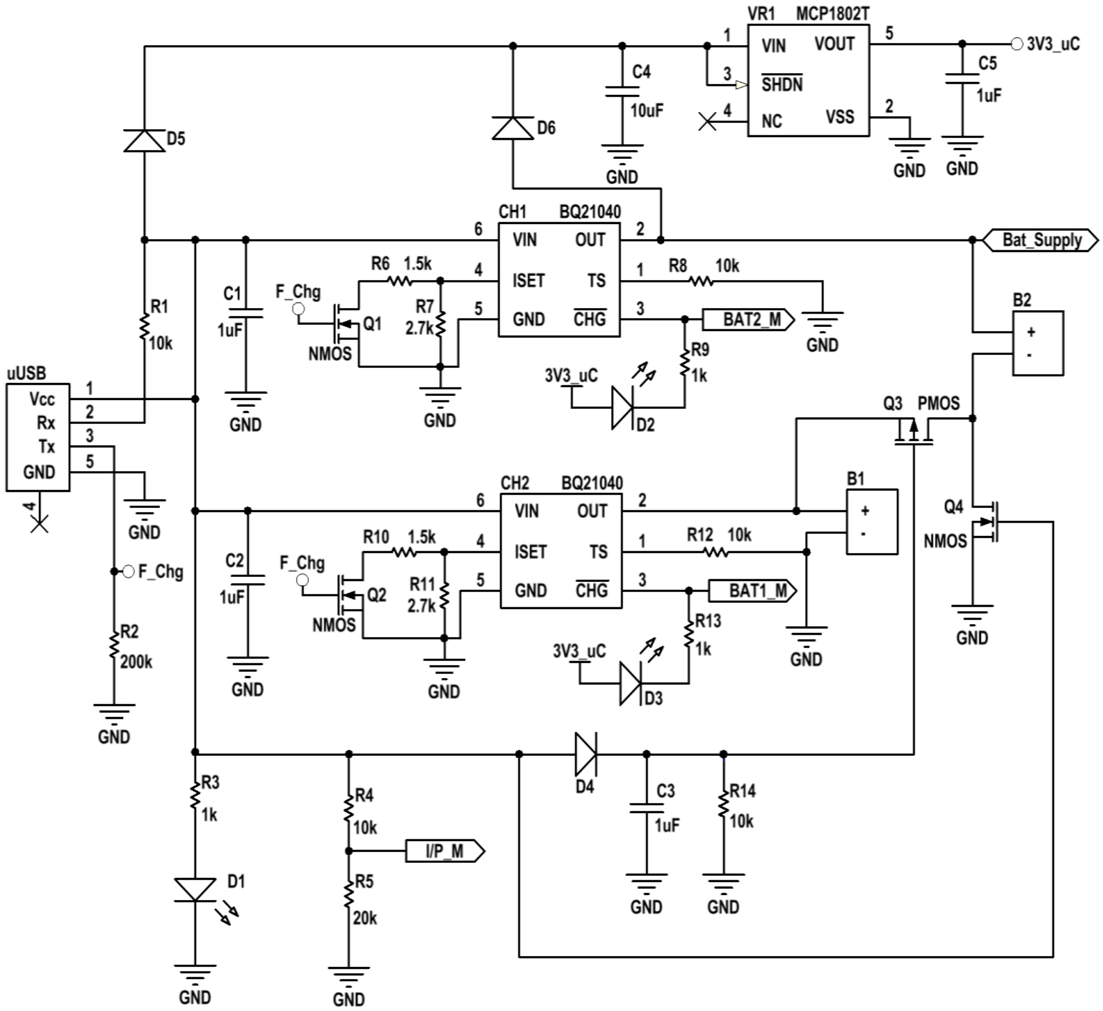


Figure 4.16: Battery charging and configuring circuit.

charging of the batteries, a detector is needed. In a USB port, the data lines D+ and D- are grounded through a 15 kΩ resistance while in a standard 5 V charger, the D+ and D- lines are either shorted or connected with a 200 Ω resistance. Therefore, a potential divider comprising of R1 and R2 is used to detect the input charging device. The F_Chg is high in case a standard 5 V charger is connected else it is low in case a USB port is connected. The F_Chg is then used to turn on the NMOS Q1 and Q2 to reduce the overall external resistance required to set the charging current of the battery charger. The NMOS used for this purpose is IRLML2402 [30] (from International Rectifier). The external resistor R6, R7, R10, and R11 are chosen to set the charging current of 200 mA for each battery when connected to USB port and 560 mA for each battery when connected to a charger. The charger used should have the capability of delivering up to 1150 mA of current. It may be noted that since the data lines of USB port are pulled up and pulled down permanently, when the module is connected to the PC USB port for charging, a notification "USB device not recognized" may appears due to lack of handshaking.

When no input charging device is connected, the batteries are automatically configured in series using PMOS Q3 and NMOS Q4. The NMOS used for this purpose is IRLML2402 [30] and the PMOS is IRLML6401 [31] (from International Rectifier). The maximum allowed continuous drain current in the PMOS and NMOS is 4.3 A and 1.2 A respectively. When no input is connected, the gate of both PMOS Q3 and NMOS Q4 are pulled down. A parallel combination of resistor R14 and capacitor C3 ensures that the PMOS Q3 is turned on after NMOS Q4 is turned off to avoid short circuit of battery B2. The resistor R14 and the capacitor C3 are selected as 10 kΩ and 1 μF to keep the RC time constant for turning on the PMOS as 10 ms. The LED L1 is used to serve two purposes: to provide an indication that input charging device is connected and to reduce the RC time constant for turning off the NMOS Q4 by pulling its gate down immediately. A Schottky diode D4 (DFLS130L) with a low forward voltage drop is used to ensure that the PMOS is turned off while charging i.e. $V_{bat} - V_G < |V_{th}|$. In the above circuit, there is an NMOS Q4 in the path while charging the battery B1, which results in asymmetric charging rate of the two batteries due to series resistance of the NMOS ($R_{DS(on)}$= 0.25 Ω [30]).

The output voltage of the series connected batteries is monitored from Pwr_Chk port using high-low voltage detect (HLVD) port pin of the microcontroller. This port pin compares the external voltage with an internally generated reference voltage and interrupts the microcontroller whenever the voltage goes above/below it. The internally generated bandgap reference voltage in the microcontroller is approximately 1.28 V. Therefore, the R15 and R16 are selected so that the HLVD module trips when the voltage of the series connected batteries falls below 6.4 V (i.e. 3.2 V each). This can be used to avoid over discharge of the battery by turning off all the peripherals and putting the microcontroller into deep sleep mode.

The powering circuit for the analog and digital ICs of the module is shown in Figure 4.17. The input power to the voltage regulators is controlled using a PMOS Q5 (IRLML6401) in order to save the power when not operating and to cut-off the supply while the batteries are charging. When the Aud_on/off port is pulled high, the gate of Q5 is pulled down using an NMOS Q6 (IRLML2402). The resistors R17 and R18 are used to keep V_{GS} above −8 V, which is the maximum acceptable limit as per the datasheet of PMOS Q5. The Bluetooth transceiver module is powered using a 3.3 V linear voltage regulator MCP1802 [28] (from Microchip). Analog ICs in each audio channel is separately powered using a 5 V linear regulator LM1117 [32] (from Texas Instruments). For powering digital ICs, a separate

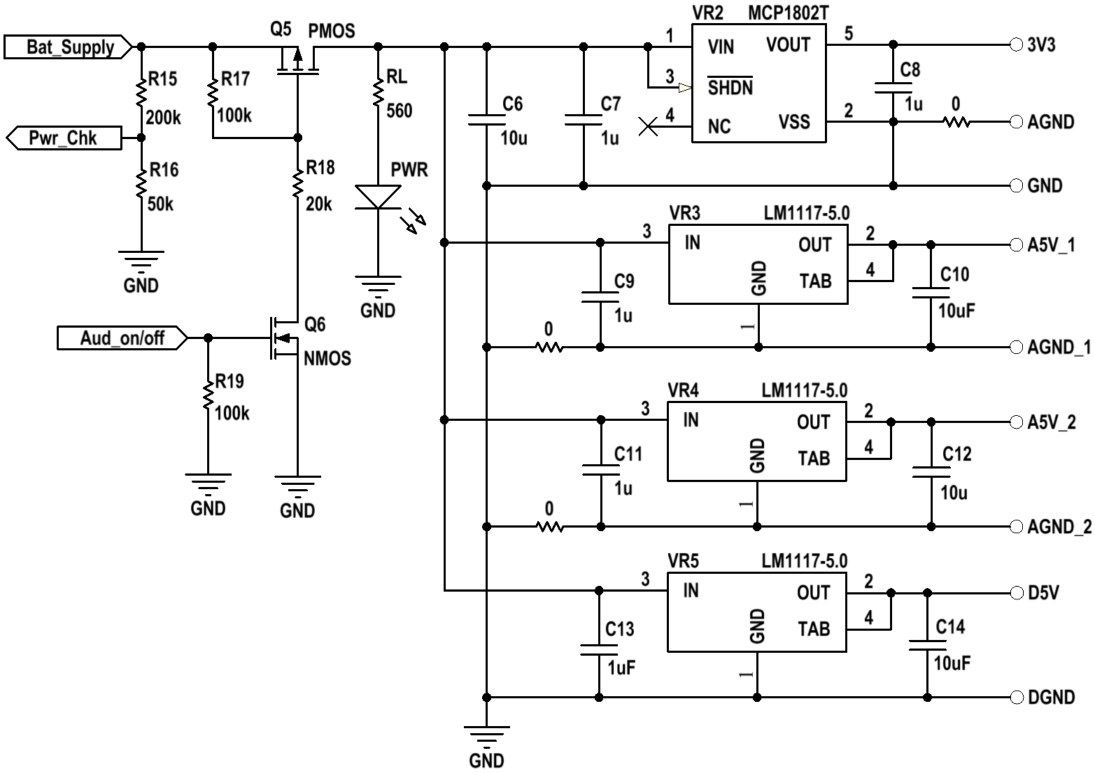


Figure 4.17: The powering circuit.

Table 4.5: Estimate of current requirement when the module is turned on and the module is operating on batteries.

Component	Current drain (mA)
VR3–VR5: LM1117	30.0
B1: RN52	30.0
U4–U7: LM1971	12.0
U8, U9: TPA6211A1	390.0
U1–U3: MCP604	3.3
U10: CD4052	0.2
U12: PIC18F46J50	2.6
D8, L1, PWR: LED	10.0
Total (Approx)	478 mA

LM1117 regulator is used to prevent the digital switching noise affecting the analog signals. The maximum currents drawn by the ICs are given in Table 4.5. The estimated peak current drawn by the power amplifier is for 10 Ω transducers with full output swing i.e. 7.6 V_{pp}. The total maximum current, estimated as the sum of current drains of individual ICs, is 478 mA.

4.12 Microcontroller Program

The microcontroller is programmed to perform the following tasks:

(i) Set the Bluetooth transceiver into data or command mode;

(ii) Communicate the commands and data from the Bluetooth transceiver through UART;

(iii) Set the attenuation settings of the digital attenuators using SPI;

(iv) Switch the 40-dB attenuators;

(v) Select the output transducers for left and right audio channels;

(vi) Enable or disable the power amplifiers;

(vii) Select the input for the analog multiplexer;

(viii) Input the average voltage level from the level detector to its ADC;

(ix) Scan the response key;

(x) Scan the touch switch periodically;

(xi) Turn on/off the power to the peripheral ICs;

(xii) Check if the charging source is connected or not;

(xiii) Monitor the battery level and indicate when the battery level is low; and

(xiv) Monitor the charging status of the batteries.

To carry out the above operations, the program is written in Embedded C using 'Microchip MPLAB X IDE' and loaded on the microcontroller on-chip program memory using the programmer/debugger Microchip PICkit 3 (from Microchip). The code has a main program, a subroutine for executing the commands received through UART, a subroutine for decoding the touch pad press, and four interrupt service routines. The five interrupt service routines are: (i) Timer 2 ISR for scanning the touch switch; (ii) UART Rx ISR for receiving the data from Bluetooth transceiver through UART; (iii) HLVD ISR for monitoring the battery voltage level; (iv) Timer 0 ISR for blinking the battery status LEDs when the battery voltage falls below a critical level; and (v) PORTB ISR to detect the charging source. The main program, subroutines, and the four ISRs are described subsequently.

4.12.1 Main Program

The flowchart of the main program is shown in Figure 4.18. In the first step, the CTMU, PORTB, HLVD, Timer 2, Timer 0, and the I/O ports are initialized.

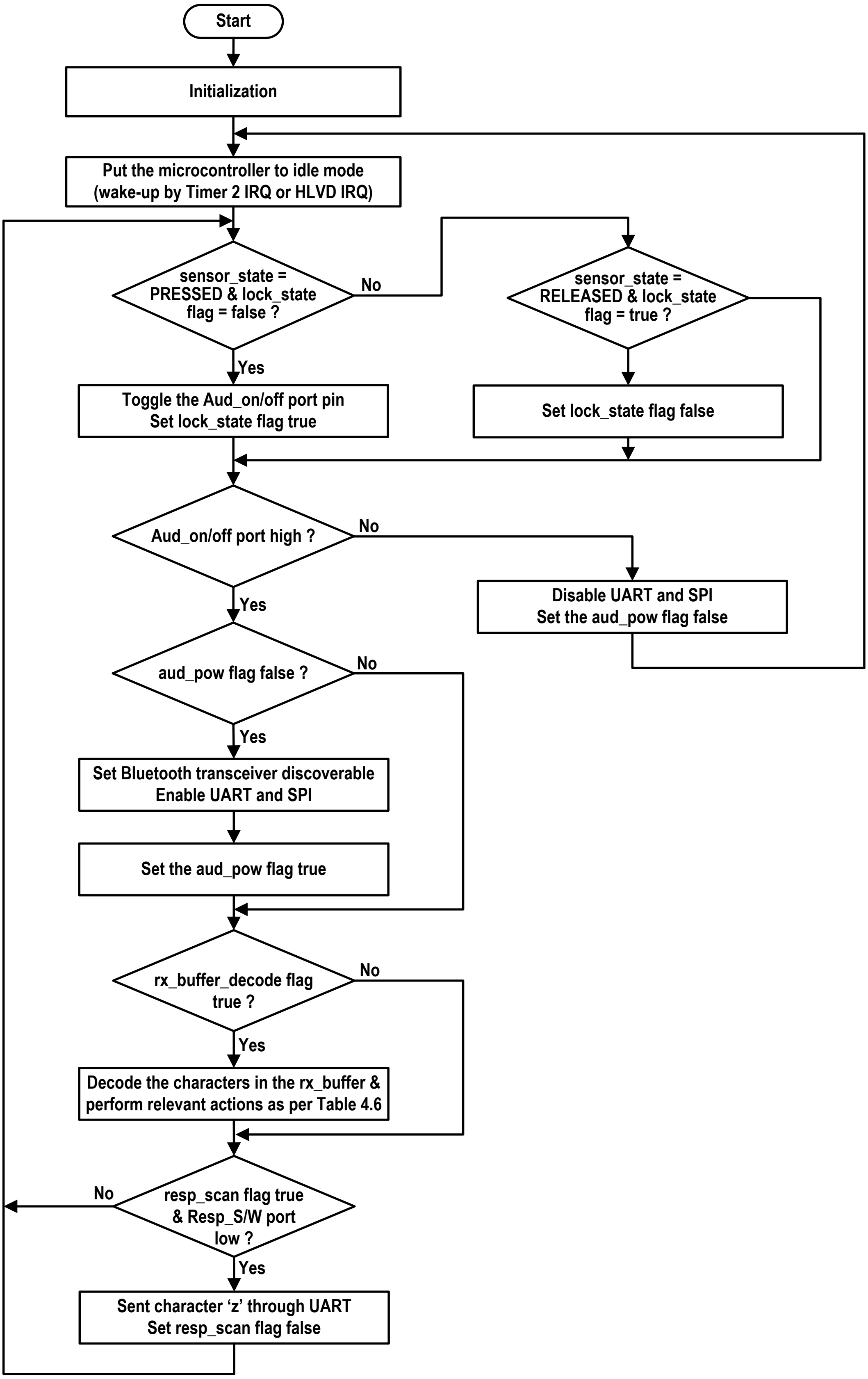


Figure 4.18: The main program of the microcontroller.

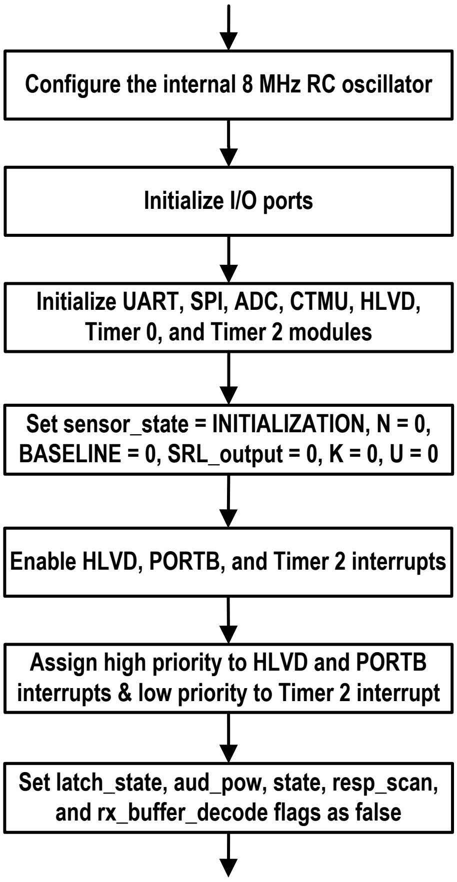


Figure 4.19: Flowchart of the initialization block of the main program.

The flowchart of the initialization block is shown in Figure 4.19. The microcontroller is programmed to use its internal 8 MHz RC oscillator to provide clock to its CPU and peripherals. The I/O port pins interfaced with the power amplifiers (RD3, RC0), digital attenuators (RD7, RD6, RA7, RA6), analog multiplexer (RB1, RB2), and relay driver (RD2, RC2, RD1, RC1, RD0) are configured as output. The unused port pins are set as output and driven to logic low to avoid external pickups. The UART port *viz.* UART_1 and UART_2, are configured to transmit and receive 8-bit data at a baud rate of 115200 bps. The SDO and SCK of the SPI port are mapped to the RD5 and RD6 port pins of the microcontroller. The SPI port *viz.* CLK and DATA, are configured to transmit data at 125 kHz with transmission occurring at the leading edge of the clock. The HLVD unit in the microcontroller is configured to monitor the Pwr_chk (RA5) port and generate an interrupt when the voltage level falls below the internally generated bandgap reference voltage. The CTMU of the microcontroller is configured to inject a current of 209 nA while scanning the touch pad. The ADC is initially configured to monitor the touch pad with its reference voltages selected as the supply voltages of the microcontroller. The conversion time per bit (T_{AD}) of the ADC is set as 1 μs and the acquisition time is set as 6 μs. The Timer 2 is configured to generate an interrupt and scan the touch switch every 2 ms. The variables used in Timer 2 ISR for touch pad scanning *viz.* sensor_state, N, BASELINE, SRL_output, B, and U, are initialized.

The sensor_state variable can hold three values *viz.* INITIALIZATION, PRESSED, and RELEASED. Initially the sensor_state is initialized as INITIALIZATION. The Timer 0 is configured to generate an interrupt every 500 ms to blink the BAT1 and BAT2 LEDs when the battery voltage falls below a critical level. The PORTB interrupt-on-change feature of the microcontroller is enabled to detect if a charger is connected to the audiometry module. Initially, the HLVD, PORTB and Timer 2 interrupts are enabled while the UART receiver interrupt is disabled. The microcontroller supports two levels of interrupt priority. The HLVD and the PORTB interrupts are assigned a high priority and the Timer 2 is assigned a low priority. The flags used in the main program are (i) latch_state flag to prevent frequent toggling of the Aud_on/off port pin state when the touch pad is kept pressed, (ii) aud_pow flag to run some steps only once when the audiometry module is powered on, and (iii) resp_scan flag to enable and disable the scanning of the response key. All these flags are initialized as false in the initialization step.

After initialization step, the microcontroller is put to "idle" mode to reduce the power consumption. In "idle" mode, the clock to the CPU is disabled while its peripherals like CTMU, HLVD, Timers, and hardware interrupts continue to operate. Timer 2 is set to wake the microcontroller every 2 ms to scan the touch pad and subsequently run other blocks of the main program. The microcontroller can also wake up from HLVD interrupt when the battery voltage falls below a critical level. In the HLVD ISR, the microcontroller is put to "sleep" mode after giving an indication on the BAT1 and BAT2 LEDs. The description of the Timer 2 and the HLVD ISR is given later.

After waking up from the Timer 2 or HLVD interrupt and completing their respective ISR, the main program consists of a loop. The first step of the loop checks the sensor_state and the latch_state flag. The sensor_state is set in the Timer 2 ISR. If the sensor_state is PRESSED and the latch_state flag is false, the Aud_on/off (RA0) port pin output is toggled and the latch_state flag is set as true. If the sensor_state is RELEASED and the latch_state flag is true, the latch_state flag is set as false. The Aud_on/off port output is then checked and if it is high, the following operations occurs (i) if the aud_pow flag is false, the Bluetooth transceiver is set to discoverable mode and the UART and SPI ports are enabled, (ii) if the rx_buffer_decode flag is true, the commands in the UART receiver buffer are executed, as per Table 4.6, (iii) if the resp_scan flag is true and the Resp_S/W port pin is low, the resp_scan flag is set as false and a character 'z' is transmitted through UART. The scanning of the response key is started and stopped by sending a command from the PC to set the resp_scan flag. If the Aud_on/off port is low, the UART and the SPI modules are disabled and the microcontroller is put to "idle" mode.

4.12.2 Timer 2 interrupt service routine

The variables used in Timer 2 ISR are sensor_state, N, BASELINE, SRL_output, B, and U. The sensor_state is use to hold the state of the touch pad *viz.* INITIALIZATION, PRESSED, and RELEASED and initially it was set as INITIALIZATION in the main program. The variable N is used to count the number of acquired ADC samples, the variable BASELINE holds the baseline value of the touch pad, the variable SRL_output is the output value of the slew-rate limiter filter, the variable B counts the number of times the baseline has been updated, and the variable U counts the number of times the touch pad remains in the same state. All these variables were initialized to zero in the main program.

The flowchart of the Timer 2 ISR for scanning the touch pad is shown in Figure 4.20. The steps involved in scanning the touch pad consists of (i) discharging the touch pad for 100 μs, (ii) charging the touch pad using current source of 209 nA for 5 μs, (iii) inputting the voltage level of the touch pad using ADC, (iv) filtering the acquired samples, (v) updating the baseline, and (v) decoding the samples for touch press. The value of the current source and the time interval for charging is empirically selected such that the touch pad (with the dimensions and configuration on the PCB) charges to 3 V when the touch pad is not touched. After acquiring the 10-bit ADC value, its complement is taken. The resulting value is input to slew rate limiter filter which functions by incrementing or decrementing its previous output by 1. If the ADC value is greater than the previous output, the output is incremented by 1, else it is decremented by 1. This is filtering rejects impulsive noise and smoothes the signal. Before starting the decoding process or updating the baseline, 60 such outputs of the filter are summed up and stored in a variable SUM. It ensures enough time for switch debounce and reduces influence of a single sample on the decision process. When the sensor_state is INITIALIZATION, the SUM is used for updating the BASELINE which is used as a reference point for decoding. The baseline is updated 10 times by replacing the old values with the new values. This ensures that the slew-rate limiter filter output reaches the actual baseline level. During this time, the touch pad should not be pressed.

After establishing the baseline, the sensor_state is set as RELEASED. When the sensor_state is either RELEASED or PRESSED, the difference of the SUM and BASELINE i.e. Δ is used for decoding the touch press. A flowchart for decoding the touch press and slowly updating the BASELINE is shown in Figure 4.21. If the sensor_state is RELEASED,

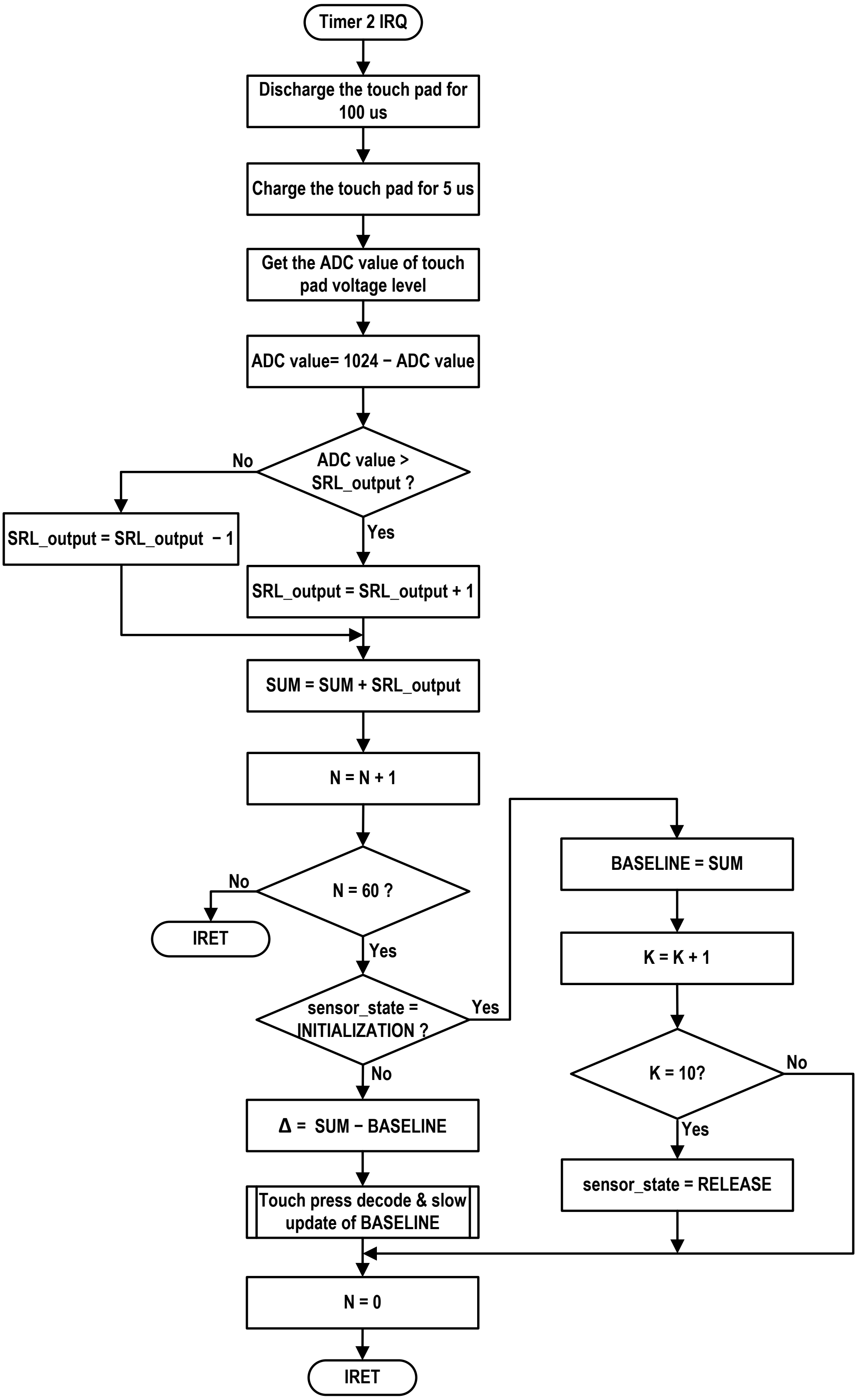


Figure 4.20: ISR of Timer 2 interrupt.

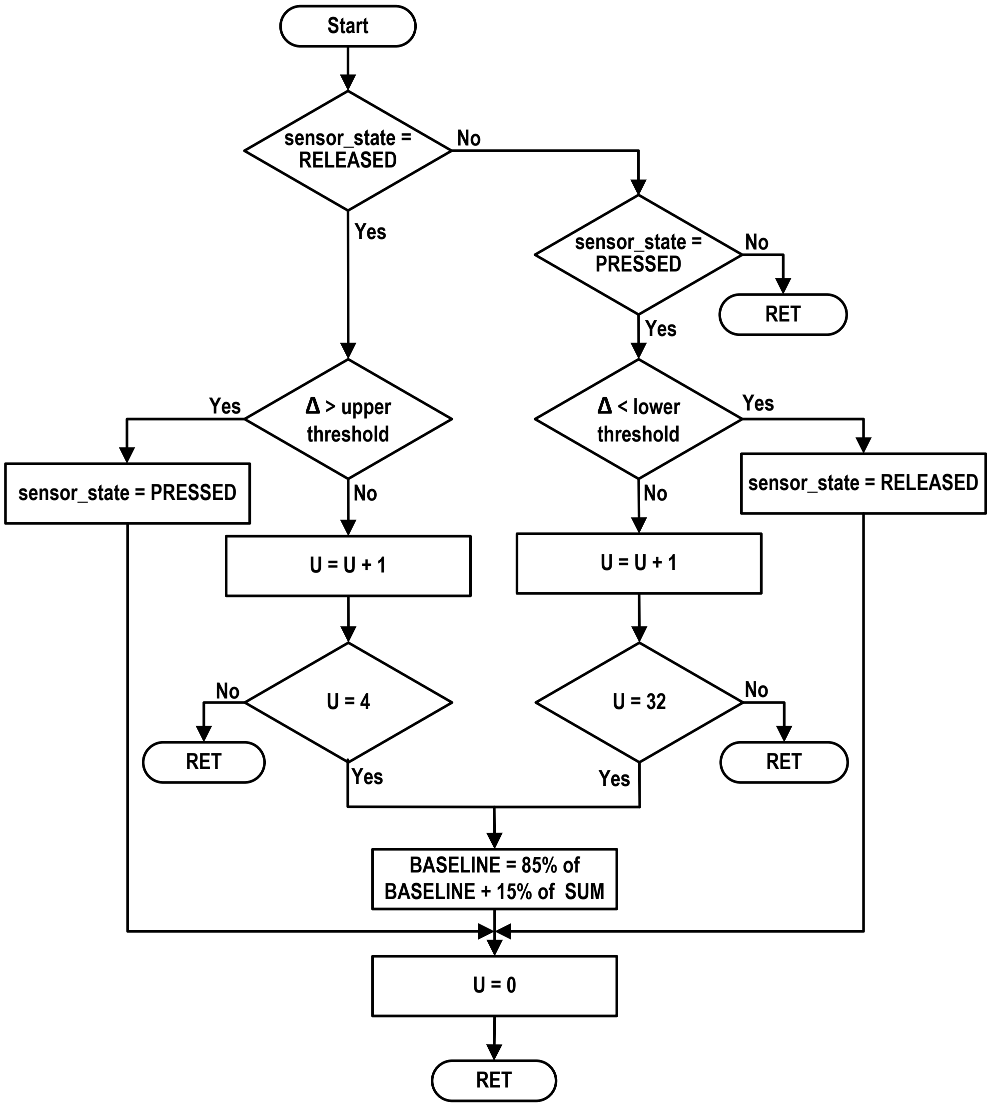


Figure 4.21: A subroutine for touch press decode and slow update of baseline.

then Δ is compared with the upper threshold value. If Δ is above the upper threshold, the sensor_state is set as PRESSED. If it is lower, then a variable U is incremented. If Δ is below the upper threshold for 4 continuous times, the BASELINE is updated by taking a weighted average of current BASELINE and the SUM. If the sensor_state is PRESSED, then Δ is compared with the lower threshold value. If Δ is below the lower threshold, the sensor_state is set as RELEASED. If it is above, then a variable U is incremented. If Δ is above the lower threshold for 32 continuous times, the BASELINE is updated by taking a weighted average of current BASELINE and the SUM. The difference between the upper and lower threshold is set to provide hysteresis and avoid frequent switching. The baseline is continuously updated at a slower rate to track environmental changes.

4.12.3 UART receiver interrupt service routine

The commands received by the microcontroller through UART are characters of variable length. The Table 4.6 lists the commands and the actions performed by the microcontroller on their reception. The flowchart of the ISR of the UART receiver interrupt is shown in Figure 4.22. On receiving a character, a UART interrupt occurs and the character is stored in the rx_buffer. When a terminating character '*' is received, a copy of the receiver buffer is made and the rx_buffer_decode flag is set. Also, the rx_buffer is reset to continue receiving commands in the receiver buffer while executing the previous commands. The rx_buffer_decode flag is polled continuously in the main program loop when the audiometry module is powered on, and when the flag is true, a subroutine for executing the commands received in the rx_buffer is called.

4.12.4 Command execution subroutine

The attenuation settings for the digital attenuators in left and right channels are sent from PC as whole number values following a special character *viz.* '$' and '#' for left channel attenuators and '%' and '&' for right channel attenuators. For example, if we want to set the attenuation of first digital attenuator of left channel as 60 dB, then '$60' is sent and similarly if we want to set the attenuation of second digital attenuator of left channel as 60 dB, then '#60' is sent. The sequence of operation for setting the attenuation level of a digital attenuator IC consists of (i) pulling its LOAD pin low with all other attenuators LOAD pins kept high, (ii) writing the address byte 0x00h to the SPI buffer, (iii) polling the flag of buffer full status to ensure that the transmission of data in SPI buffer to the SDO pin is complete, and (iv) writing the attenuation setting to the SPI buffer.

The commands 's' and 'x' are used to get the ADC values corresponding to the voltage level at Alevel port and Pwr_chk port, respectively. When the ADC of the microcontroller is used for sampling the Alevel port, its reference voltages are set equal to the external voltage references VREF+ and VREF- and when it is used for sampling the Pwr_chk port and touch switch, its reference voltages are set equal to the supply voltage of the microcontroller. The other commands from the PC are used to set the I/O ports of the microcontroller and configure the Bluetooth transceiver. When configuring the Bluetooth module, the GPIO_9 port is pulled down so that the module enters into command mode.

4.12.5 HLVD and Timer 0 interrupt service routines

The HLVD unit of the microcontroller is configured to generate an interrupt when the voltage level at Pwr_chk falls below the bandgap reference voltage of 1.28 V (This value was measured using ADC while testing the audiometry module). When Pwr_chk port voltage

Table 4.6: Functions performed by the microcontroller based on the commands received through Bluetooth transceiver.

Commands	Function
$	The next two consecutive characters following this character are converted into decimal value and are used to set the attenuation level of the first digital attenuator in the left channel.
#	The next three consecutive characters following this character are converted into decimal value and are used to set the attenuation level of the second digital attenuator in the left channel.
%	The next three consecutive characters following this character are converted into decimal value and are used to set the attenuation level of the first digital attenuator in the right channel.
&	The next three consecutive bytes following this character are converted into decimal value and are used to set the attenuation level of the second digital attenuator in the right channel.
l0	Disable the left power amplifier.
l1	Enable the left power amplifier.
r0	Disable the right power amplifier.
r1	Enable the right power amplifier.
e0	Disable the output attenuator of left channel.
e1	Enable the output attenuator of left channel.
f0	Disable the output attenuator of right channel.
f1	Enable the output attenuator of right channel.
a0	Select headphone as the output transducer in left channel.
a1	Select bone vibrator as the output transducer in left channel.
b0	Select headphone as the output transducer in right channel.
b1	Select bone vibrator as the output transducer in right channel.
s	Get ADC value corresponding to the voltage level at Alevel port.
t	Set the port pin interfaced with response key as output and set it high to glow the LED.
u	Set the port pin interfaced with the response key as input and start scanning for response.
m0	Select left channel transducer terminals as input to the level detector circuit.
m1	Select right channel transducer terminals as input to the level detector circuit.
m2	Select the bone vibrator terminals as input to the level detector circuit.
m3	Select the SLM dc output as input to the level detector circuit.
x	Get ADC value corresponding to the voltage level at Pwr_chk port.
*	Make a copy of the receiver buffer and set the rx_buffer_decode flag.
B	Set the UART baud rate to either 9600, 19200, 115200 bps based on the next character received.
C	Send an acknowledgment signal required for connecting HSP Bluetooth profile
K	Kills a Bluetooth profile viz. SPP, A2DP, HSP/HFP based on the next character received as 1, 2, 3 or 4 respectively.
O	Set the bit width and the sampling rate of the audio as 24-bit, 44.1 kHz or 24-bit, 48 kHz or 32-bit, 44.1 kHz or 32-bit, 48 kHz.

P	Set the Pairing time out in seconds.
Q	Reconnect the previously connected profile.
R	Reboot the Bluetooth transceiver module.
U	Clear the entire previously paired device list from Bluetooth transceiver memory.
X	Change the left and right audio channel gain level in the Bluetooth transceiver based on the next received character.
Z	Change the gain of both the microphone inputs based on the next received character.
!	Accepts/Rejects the pairing key.
@	Turn the Bluetooth into discoverable mode or not

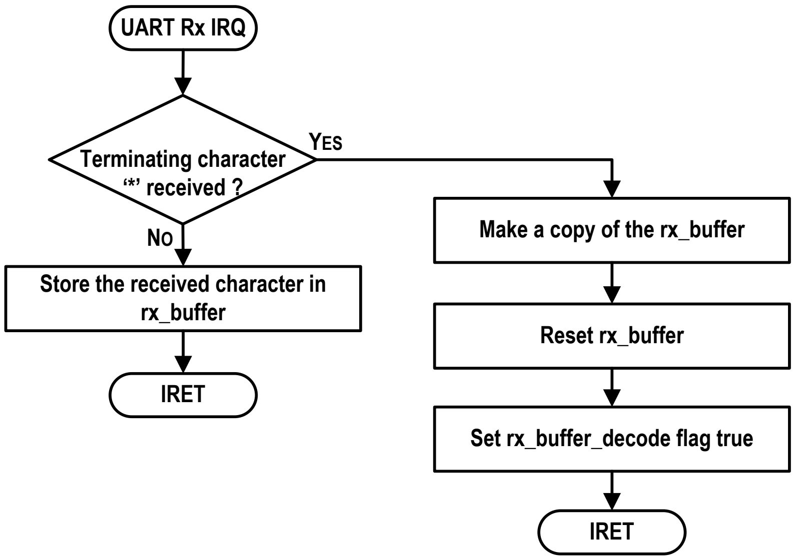


Figure 4.22: ISR of UART receiver interrupt.

level is 1.28 V, the individual battery voltage level is 3.2 V. The HLVD interrupt was enabled and given a high priority in the initialization step of the main program. The flowchart of the HLVD ISR is shown in Figure 4.23. The sequence of operation in the HLVD ISR consists of (i) disabling the HLVD module to stop generating interrupt, (ii) enabling the Timer 0 to generate an interrupt with high priority after every 500 ms, and (iii) setting the BAT1_M and BAT2_M port as output for indication. The flowchart of the Timer 0 ISR is also shown in Figure 4.23. The Timer 0 interrupt is generated after every 500 ms to toggle the BAT1 and BAT2 LEDs by toggling the output state of BAT1_M and BAT2_M ports and monitor the voltage level at Pwr_chk port using the ADC. The reference voltages of the ADC are set equal to the supply voltage of the microcontroller. If the ADC value falls below a threshold, the microcontroller is put to sleep mode and the blinking of LEDs is stopped to reduce the

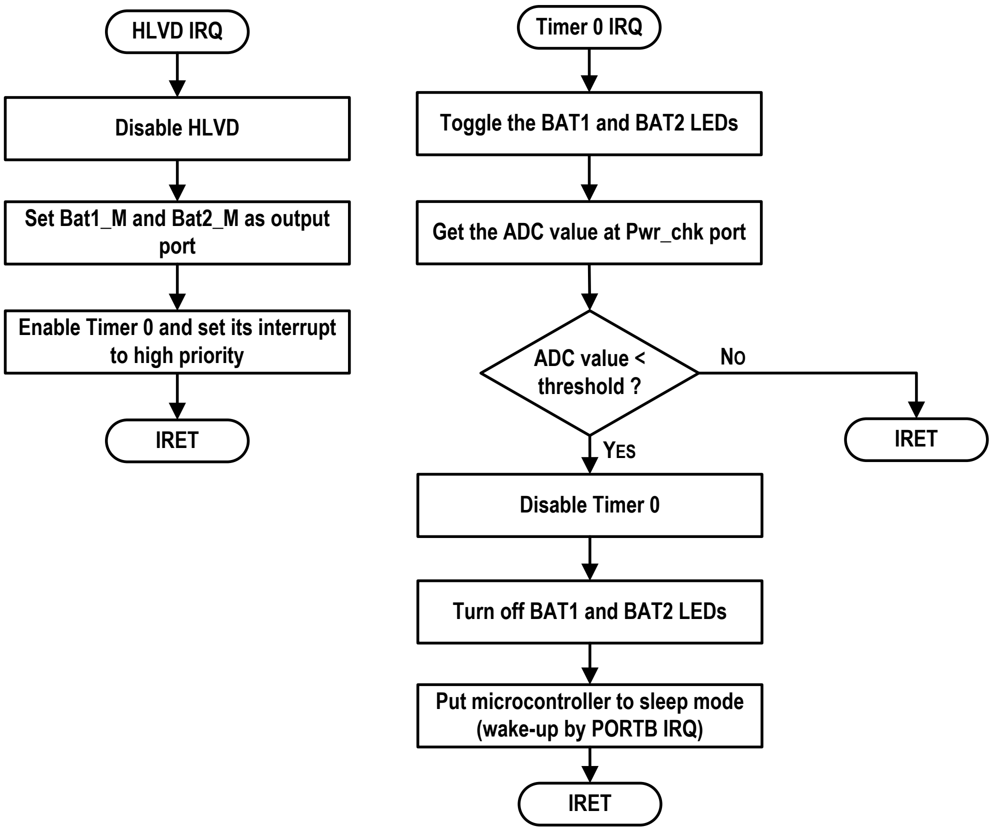


Figure 4.23: ISR of HLVD and Timer 0 interrupts.

power consumption. The threshold value of ADC was set corresponding to 100 mV below the HLVD trip level i.e. 1.28 V.

4.12.6 PORTB interrupt service routine

The PORTB interrupt-on-change feature is used to detect if a charger is connected to the audiometry module. In the initialization step of the main program, this interrupt was given a high priority. When a charger is connected to the audiometry module, the I/P_M port (RB4) changes from low to high which interrupts the microcontroller. The flowchart of the ISR of the PORTB interrupt is shown in Figure 4.24. When a charger is connected to the audiometry module, the I/P port changes from low to high which generates an interrupt. The sequence of operation in the ISR, if the I/P port is high, consists of (i) disabling the HLVD module to stop monitoring the voltage level while charging the batteries, (ii) turning off the power to the peripheral ICs by pulling down the Aud_on/off port, (iii) putting the microcontroller to sleep mode to reduce the power consumption. In sleep mode the clock to the CPU and the peripherals are disabled but the state of the registers are retained. The scanning of the touch switch is also stopped since the clock to the Timer 2 is disabled. When the charger is plugged

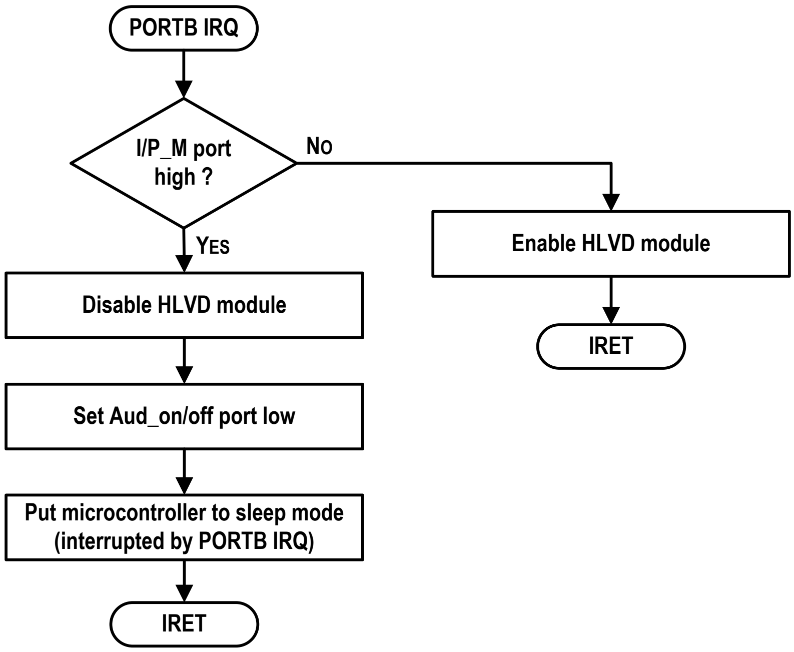


Figure 4.24: ISR of PORTB interrupt.

out, the I/P_M port changes from high to low which again generates an interrupt to wake up the microcontroller from sleep mode and enable the HLVD module.

4.13 PCB Design

The schematics of the circuits described in the earlier sections are given in Figure F.1, F.2, F.3, F.4, F.5, and F.6 of Appendix F. These were tested using Multisim (from National Instruments) and subsequently, a PCB was designed using Altium Designer. The PCB is a four-layer board of 50 mm × 45 mm size in order to reduce the noise pickups and make the hardware compact. Floor planning was done to reduce the noise by keeping the ground plane below every signal track. Since the circuit blocks on the PCB have mixed signals (i.e. analog and digital), special care was taken in the layout design to avoid coupling of digital ground to the analog circuit. Decoupling capacitors of 0.1 µF and 0.01 µF have been used for analog IC close to its power supply pin to filter out noise over wide band and decoupling capacitors of 0.1 µF have been used for digital IC close to its power supply pin to filter out voltage spikes. The port pins of the microcontroller were assigned based on the placement of the peripheral ICs. The differential outputs of the power amplifiers were routed close and parallel to each other. A small copper plane was placed under the power amplifier ICs to serve as its heat sink. The analog grounds (AGND_1, AGND_2) and digital ground (DGND) were routed separately throughout the board and are shorted at the input power supply. Digital 3.3 V (3V3_uV) and analog +5 V (A5V_1, A5V_2) copper planes have been provided on the top layer and third layer of the board while the digital ground (DGND) and analog ground (AGND_1, AGND_2) copper planes have been provided on the second and bottom layer of the board. Since blind vias and buried vias makes the PCB fabrication more expensive, only plated through-hole (PTH) vias have been used. The PTH vias are of 1 mm diameter with 0.5 mm size. The minimum distance between the two signal tracks is 0.127 mm.

A two-layer PCB was designed for the circuit consisting of the touch switch, indicator LEDs, and the connectors. The touch switch is a filled circular copper pad of 15 mm diameter. It is surrounded by circular ring of 1 mm width and separated from it by 1 mm. This circular ring is connected to ground to absorb any conducted or radiated noise and to increase the robustness of the touch sensor.

Mono 3.5 mm audio jacks are used for left headphone, right headphone, bone vibrator, response key, sound level meter input, and microphone input. The PCB layouts of top, second, third, and bottom layer of the four-layer PCB are given in Figure F.7, F.8, F.9, F.10 and the top and bottom layer of the two-layer PCB are given in Figure F.11, F12 in Appendix F. Images of the assembled PCBs are given in Appendix G.

Page intentionally left blank

Chapter 5

PC-BASED APPLICATION FOR AUDIOMETRY

5.1 Introduction

The PC-based application software for performing subject-response audiometric tests and a psychoacoustic test has been developed on Qt creator IDE in C++ language, with the stimuli and masker waveforms synthesized and stored using Matlab (ver. R2010 MathWorks). The application performs pure-tone test in manual and automatic mode, SISI test, tone decay test, ABLB test, speech test, and notched-noise test. The Bekesy audiometry has not been implemented in this version of the application software. The application provides user interface for selecting the test ear, the stimulus type, and the output transducer, and it handles outputting of the synthesized stimulus and masker sounds, controlling the signal level for each audio channel, receiving and acknowledging the patient response, and storing the test result. It also provides facility for storing the patient's information, plotting the audiogram, and calibrating the transducers. These operations are described in the subsequent sections.

5.2 Stimulus and Masker Generation

The stimuli and masker waveforms for the pure-tone, SISI, tone decay, ABLB, and psychoacoustic tests were synthesized at the sampling rate of 44.1 kHz with 24-bit depth and stored as "*.wav"* files. For speech test, live speech input from microphone is used as stimulus in either of the channel. Facility for using recorded speech may be incorporated later. All the stored sound files are stereo sounds with the stimuli or masker in one channel and silence in the other channel. These files are used to present different combinations of sounds in the two audio channels by playing different sound files simultaneously. In each channel, the waveforms from the two files are added and outputted. For example, for presenting a tone in the left channel and masker in the right channel, the sound file with the tone in the left channel and silence in the right channel and the sound file with silence in the left channel and masker in the right channel are played together. For producing a long-duration sound, the corresponding short duration sound waveform is looped (presented repeatedly). All the periodic waveforms have integer number of cycles to avoid discontinuities in looping.

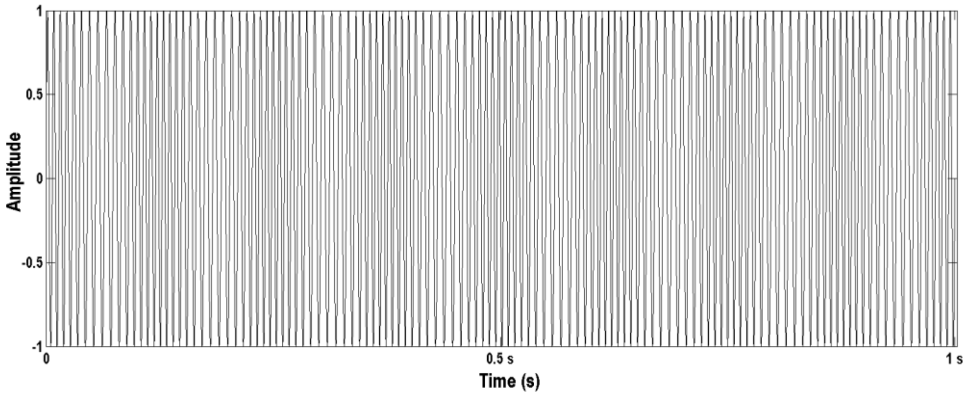


Figure 5.1: Waveform of 125 Hz warble tone.

The pure-tones for frequencies 125, 250, 500, 750, 1000, 1500, 2000, 3000, 4000, 6000, 8000 Hz were generated for 40 ms so as to get integer number of cycles for all the frequencies. The equation used for generating the pure tone is

$$x(n) = \sin(2\pi n f_o / F_s), \quad 0 \le n < N-1 \tag{5.1}$$

where f_o= test frequency, F_s = sampling frequency, and n = sample index, and N = total number of samples.

A warble tone is a frequency modulated tone whose frequency varies periodically over a small range about the test frequency f_o. The equation used for generating the warble tone is

$$x(n) = \sin(2\pi n f_o / F_s + (\alpha f_o / f_m)\sin(2\pi n f_m / F_s)), \quad 0 \le n < N-1 \tag{5.2}$$

where α = relative frequency deviation, f_m = modulation frequency, F_s = sampling frequency, n = sample index, and N = total number of samples. The instantaneous frequency of the synthesized waveform is $f(n) = f_o + \alpha f_o \cos(2\pi n f_m / F_S)$. The waveforms of 1 s duration were synthesized with α = 0.1 and f_m = 8 Hz, corresponding to 10% frequency deviation from the center frequency and 8 Hz modulating frequency, as per IS 10565:1999 [14]. The number of samples in 1 s ensured integer number of cycles of the modulating signal as well as the carrier signal. The waveform of the 125 Hz warble tone is shown in Figure 5.1. It may be noted that the number of samples for the waveform synthesized using the equation (5.1) and (5.2) have to be limited to avoid overflow in computation.

A pulsed tone is an amplitude modulated tone in which the carrier frequency is the test frequency f_o and the modulating signal is a square wave. The square wave modulating signal of 2 Hz with 300 ms on time and 50 ms rise and fall time each, as per IS 10565:1999 [14],

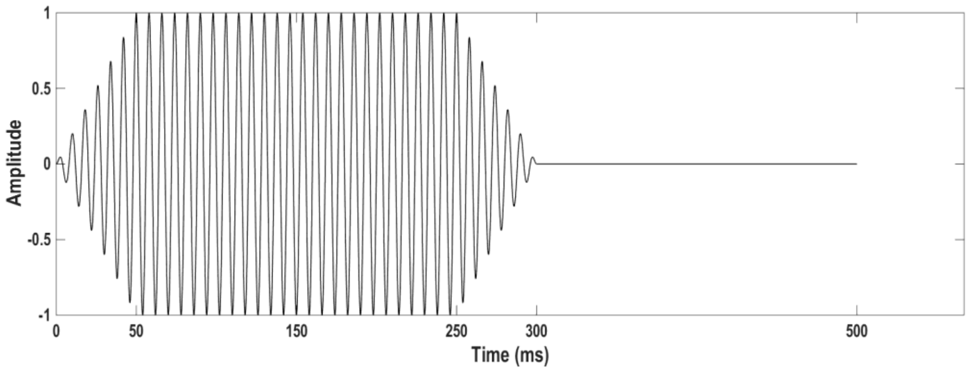


Figure 5.2: Waveform of 125 Hz pulsed tone.

was chosen with the pure-tone as carrier signal. The pulsed tone is generated using the equation

$$x(n) = a(n)\sin(2\pi n f_o / F_s), \quad 0 \le n < N_4 \tag{5.3}$$

where $a(n)$ is modulating square wave given as

$$a(n) = \begin{cases} n / N_1 & , \quad 0 \le n < N_1 \\ 1 & , \quad N_1 \le n < N_2 \\ (N_3 - n)/N_1 & , \quad N_2 \le n < N_3 \\ 0 & , \quad N_3 \le n < N_4 \end{cases} \tag{5.4}$$

where $N_1 = 0.05F_S$, $N_2 = 0.25F_S$, $N_3 = 0.3F_S$, and $N_4 = 0.5F_S - 1$, which are selected for rise times of 50 ms, steady-state segment of 200 ms, and silence of 200 ms. The waveform of the 125 Hz pulsed tone is shown in Figure 5.2. The pulsed waveforms of 500 ms are generated and stored. The length is selected to get integer number of cycles for all test frequency and the modulating signal.

The waveforms for the SISI test were generated as amplitude modulated tone with a two-level modulating signal. The SISI tone is generated using the equation

$$x(n) = a(n)\sin(2\pi n f_o / F_s), \quad 0 \le n < N_4 - 1 \tag{5.5}$$

where $a(n)$ is sample of the modulating signal given as

$$a(n) = \begin{cases} A & , \quad 0 \le n < N_1 \\ A + \dfrac{(1-A)}{(N_1 - N_2)}(n - N_1) & , \quad N_1 \le n < N_2 \\ 1 & , \quad N_2 \le n < N_3 \\ 1 - \dfrac{(1-A)}{(N_4 - N_3)}(n - N_3) & , \quad N_3 \le n < N_4 \end{cases} \tag{5.6}$$

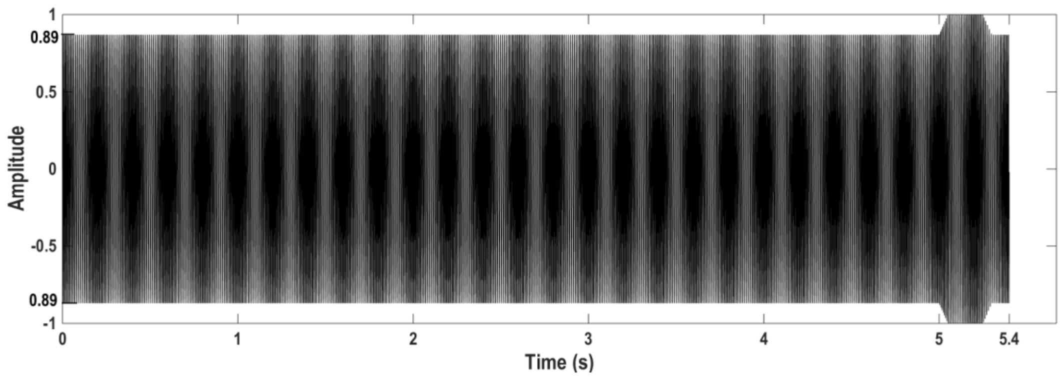


Figure 5.3: Waveform of pure tone of 125 Hz for SISI test.

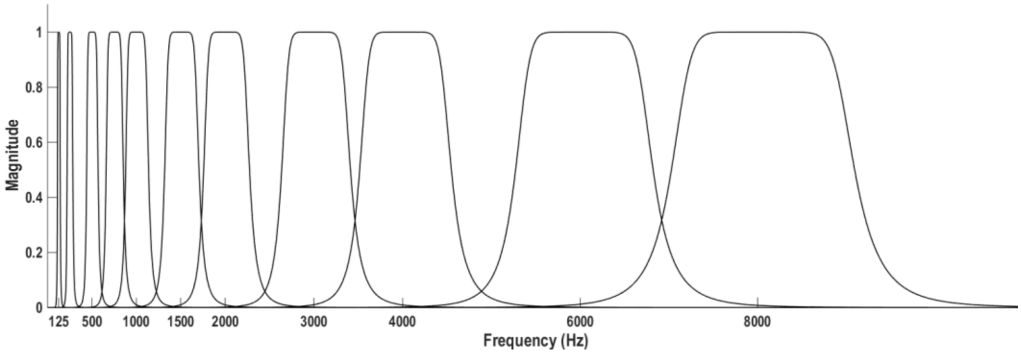


Figure 5.4: Filter responses for narrowband noise filtering at audiometric test frequencies.

where $N_1 = 5F_S$, $N_2 = 5.05F_S$, $N_3 = 5.25F_S$, and $N_4 = 5.3F_S - 1$, which are selected for a segment of 5 s, a rise time of 50 ms, increased level segment of 200 ms, and fall time of 50 ms. The value of A is selected as 0.89 which corresponds to 1 dB less from the amplitude of 1 (i.e. A = $10^{-1/20}$ = 0.89). The SISI test is conducted with pure tones. To get integer number of cycles, the waveforms for all the audiometric frequencies were generated for 5.3 s, except for 125 Hz which was generated for 5.4 s. A waveform of the 125 Hz pure-tone for SISI test is shown in Figure 5.3.

The broadband noise is generated by generating white Gaussian noise followed by lowpass filtering using an IIR filter with cut-off frequency of 8 kHz. The noise has a nearly flat spectrum over 0 – 6 kHz, as per IS 10565:1999 [14]. The length of the stored broadband noise waveform is 1 s. The narrowband noise waveforms are generated by passing the broadband noise to the 10th order Butterworth IIR filter with one-third octave bandwidth centered at the test frequencies, as per IS 10565:1999 [14]. The frequency responses of the filters are shown in Figure 5.4.

For the notch-noise test for auditory filter shape estimation, bandpass noises with a notch around the test frequency were generated. The synthesis of bandpass noise was done in

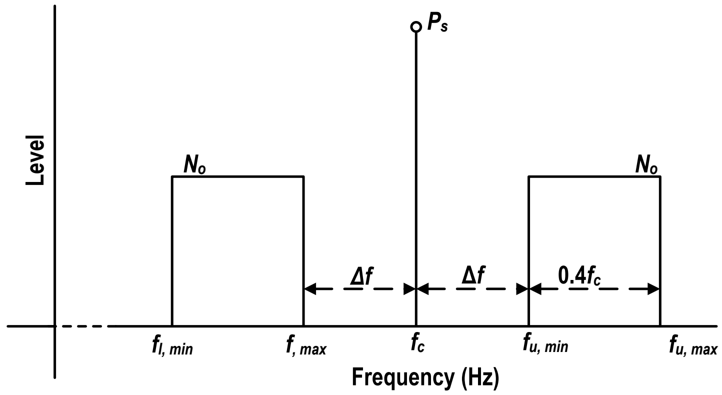


Figure 5.5: Notch-noise with notch placed symmetrically around the signal frequency.

frequency domain and the IFFT was used to get the waveform in the time domain. The frequency spectrum of the notched bandpass noise is shown in Figure 5.5. In frequency domain, the magnitudes of samples in the bands of the noise, $f_{l,\,min} - f_{l,\,max}$ and $f_{u,\,min} - f_{u,\,max}$, were set equal to a non-zero constant and their phases were randomized between 0 to 2π. The magnitudes of samples beyond the boundaries of the bands of noise were set to zero. The bandwidth of the noise bands were set as $0.4f_c$ around test frequency f_c [33]. The notch-noise at each test frequency were generated for seven notch-width *viz.* $0.0f_c$, $0.1f_c$, $0.2f_c$, $0.3f_c$, $0.4f_c$, $0.5f_c$, and $0.6f_c$ [33]. The waveforms in time domain were generated for 4 s.

5.3 Software Description

A screenshot of the home-screen of the GUI is shown in Figure 5.6. Various operations are described in the following subsections.

5.3.1 Target Connection Establishment

The Bluetooth transceiver used in the audiometry module supports SPP, A2DP and HSP/HFP profiles. The SPP profile of the Bluetooth is used for emulating a serial interface between two Bluetooth devices. The A2DP profile is used for playing stereo audio at high sampling rate i.e up to 44.1 kHz. The HSP/HFP profile is used for playing mono audio and sending the microphone signal at sampling rate of 8 kHz. When the Bluetooth of the audiometry module is paired with the PC, a COM port is assigned to the Bluetooth SPP profile for serial communication and a playback device is assigned for transmitting audio signal over A2DP Bluetooth profile. The Bluetooth connection between the PC and the audiometry module is established by selecting the COM port and pressing the "Connect" button on the toolbar of the GUI. After this, the line coding parameters for the serial

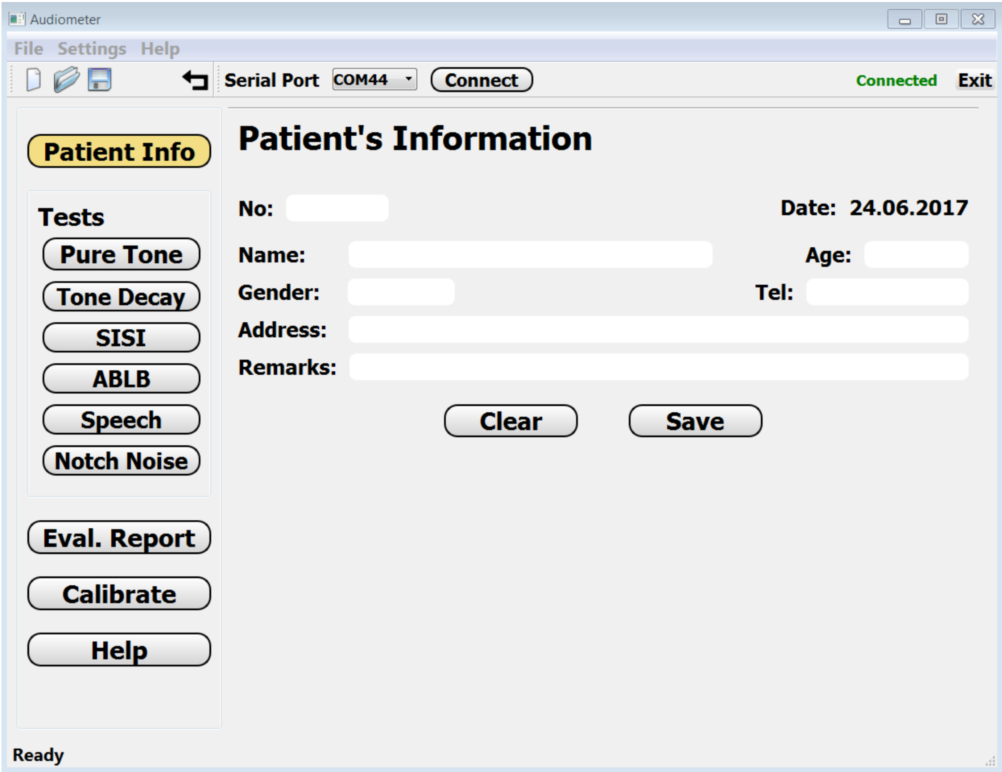


Figure 5.6: A screenshot of the home screen of the GUI.

communication are set as 115200 bps baud rate, 8-data bits, and 1-stop bit, zero parity bit. After establishing the serial connection, the volume of the playback device is set to maximum.

5.3.2 *Stimulus and Masker Level Control*

The attenuation level in dB for presenting a pure-tone at a particular dB HL level is calculated using the calibration values and the RETSPL values of the transducer. The calibration values of a transducer are the attenuator levels (L_{att}) for which the transducer produces M dB SPL at the audiometric frequencies. The RETSPL, reference equivalent threshold sound pressure level, is transducer-specific and frequency-dependent dB SPL value that corresponds to 0 dB HL. Typical RETSPL values of some of the commonly used audiometric transducers are shown in Fig C.1 of Appendix C. It may be noted that the actual value has to be obtained during the calibration with the device. If the pure-tone is to be presented at N dB HL, the attenuation in dB is calculated as

$$\text{Attenuation (dB)} = L_{att}\,(f) + (M - N - \text{RETSPL}(f)) \tag{5.7}$$

When performing air-conduction test, the sound level is controlled by setting the attenuation level of the digital attenuator before the power amplifier and by switching the 40-

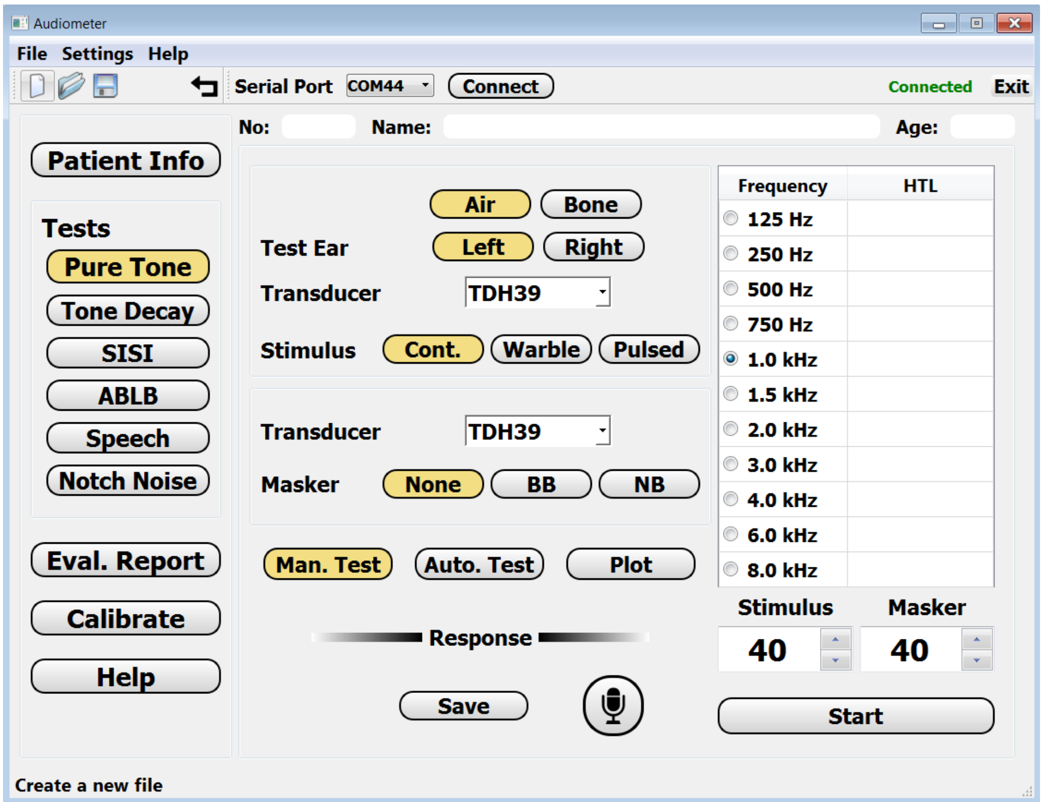


Figure 5.7: A screenshot of the pure-tone test module.

dB post power amplifier attenuator. Since each channel has two cascaded digital attenuators, the second digital attenuator in the cascade is used first to reduce any noise added in the previous stages. The switchable 40-dB attenuator is used whenever the required attenuation level exceeds its value. The actual attenuation value of the 40-dB attenuator depends on the transducer impedance and hence needs to be found during calibration. In case of bone-conduction test, there is no attenuator after the power amplifier, and therefore the digital attenuators are used to control the output level.

5.3.3 Pure-Tone Test

Pure-tone audiometry is used for finding the hearing threshold levels (HTL) for a stimulus at the audiometric test frequencies. The stimulus in the test ear can be pure tone (continuous), pulsed tone (amplitude modulated), or warble tone (frequency modulated) and the masker in the non-test ear can be broadband, narrowband noise (centered at test frequency), or none. The stimulus is presented to the patient sitting in a soundproof room through a transducer (headphones/bone vibrator). The patient is instructed to press a response key whenever he/she hears a tone in the test ear. A screenshot of the GUI of the pure-tone test module is shown in Figure 5.7. The test ear (left/right), the type of conduction (air/bone), the test and non-test ear transducers, stimulus (pure/pulsed/warble), masker (none/broadband/ narrowband), stimulus frequency, and the test mode (manual/automatic) are selected manually in the GUI. The stimulus and the masker levels are manually controlled using up/down buttons of the stimulus spin-box. The tone presentation is started by pressing the "Start" button. The response key scanning is also started by sending a command to the audiometry module. When the response key is pressed, the scanning of the response key is stopped and the response bar in the GUI begins to flash. The tone is presented for 2 s and then stopped automatically. Also, the tone can be prematurely stopped by pressing the "Stop" button. The tone level is tapered in the starting and ending to avoid audible clicks when starting and stopping the audio. In case of manual mode when the start button is pressed, all the buttons except stimulus level set spin-box, and "Save" and "Stop" buttons are disabled. The masker starts playing in the non-test ear when the noise button, "BB" or "NB", is pressed. To stop the noise, "None" button must be pressed.

The automatic procedure for threshold determination for a test frequency is shown in Figure 5.8. In the automatic mode, the software controls the stimulus level by monitoring the patient's response from the response key. Initially the pure-tone of the selected frequency is presented at 40 dB HL and if the response to the tone occurs, the level is decreased in 10 dB decrements until the patient no longer responds. After this, the level is increased by 5 dB. If the response to the tone occurs, the level is again decreased by 10 dB, else the level is again increased by 5 dB. This is known as the “up-5 down-10” technique. The minimum level at which the patient responds 2 out of 3 times is recorded as the hearing threshold. During the procedure, all the buttons, except the "Stop" button, are disabled. After the HTL is determined, the level is saved and shown in the frequency-HTL table. In both manual and automatic mode of pure-tone test, the stimulus level and the test conditions (the test ear, transducers, stimulus type, noise type, noise level) are also saved when the "Save" button is pressed.

5.3.4 Tone Decay Test

Tone decay test is used for determining the continuous listening capability of the stimulus for about 60 s. The GUI of this test is the same as that of pure-tone test. The procedure of the automated tone decay test is shown in Figure 5.9. The test is performed after determining the pure-tone threshold of a patient. The stimulus level should be set 10 dB below the hearing threshold for the selected tone frequency. The test ear (left/right), the type of conduction (air/bone), the masking type and level, the stimulus frequency, and its level are manually selected. The patient is instructed to press the response button when a tone is heard

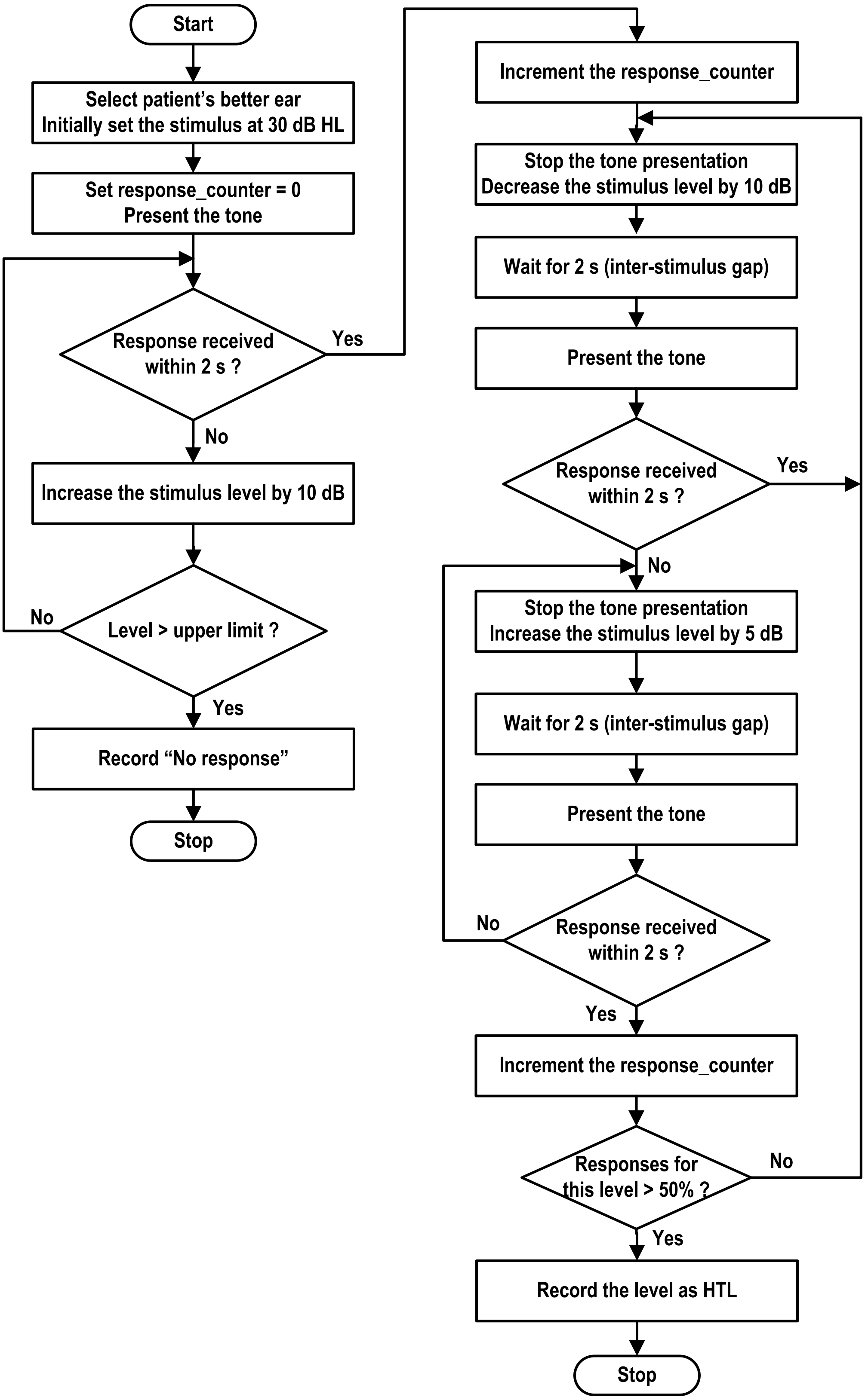


Figure 5.8: The automatic pure-tone audiometry procedure for threshold determination.

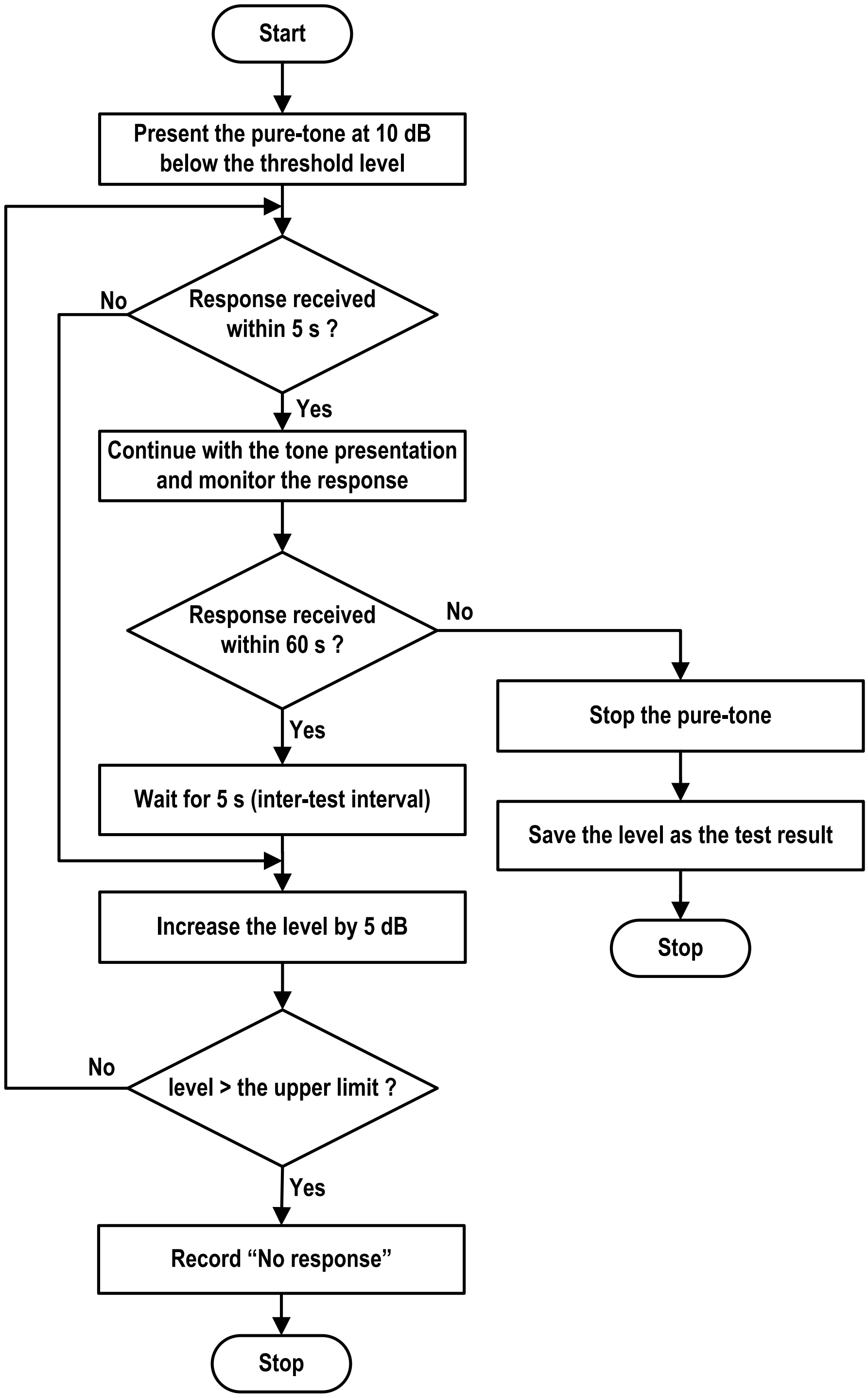


Figure 5.9: Procedure for the automated tone decay test.

for the first time in the test ear and then again when the tone disappears. The tone presentation starts after the “Start” button is pressed. At intervals of 5 s, the stimulus level is increased in steps of 5 dB untill the patient response is received. When the response is received at some level, the presentation is continued for 60 s at the same level and the response button is monitored for the patient response. If the patient response is received before the lapse of 60 s, the test is repeated by increasing the level by 5 dB, else if the patient response is not received within 60 s, the tone is terminated and the level is recorded and displayed. During the test, the status of presentation level is updated on display. The final level along with the test parameters (conduction type, noise type and its level, and transducers) are saved automatically as the result for the selected frequency and for the selected test ear. The test can be terminated by pressing the “Stop” button.

5.3.5 SISI Test

Short increment sensitivity index (SISI) test is used for determining the ability to detect a brief 1-dB increment in a 20 dB supra-threshold tone at an audiometric frequency. The GUI of the SISI test is same as that of pure-tone test with a small modification in the frequency-HTL table to show the SISI score. The test ear (left/right), the type of conduction (air/bone), the masking type and level, and the stimulus frequency and its level are manually selected. The tone presentation starts when "Start" button is pressed. The sequence of steps that is followed to find the SISI score is shown in Figure 5.10. The stored SISI test waveform is looped 20 times. The patient is instructed to press the response button whenever there is a short duration increase in the level of the tone in the test ear. The total number of responses are recorded and multiplied by 5 to give the SISI score out of 100. The score along with the test parameters (stimulus level, noise type, noise level) are automatically saved for the selected frequency and for the selected test ear, conduction and transducers. The test may be terminated prematurely by pressing the "Stop" button.

5.3.6 ABLB Test

Alternate binaural loudness balance (ABLB) test is used to determine the loudness recruitment in a patient with unilateral hearing loss. In this the stimulus at test frequency is alternately presented in two ears. The screenshot of the ABLB test module GUI is shown in Figure 5.11. The type of conduction (air/bone), the type of stimulus (pure-tone, warble tone or pulsed tone), and the stimulus frequency and its level in left and right ears are manually selected. When the "Start" button is pressed, the stimulus is first presented in the left ear for 2 s and then in the right ear for 2 s. The patient is instructed to press the response button whenever the sound appears to be of same loudness in both the ears. On receiving the response, the stimulus level in the left and right ear are saved and displayed on the table. The test can be terminated at any time by pressing the “Stop” button.

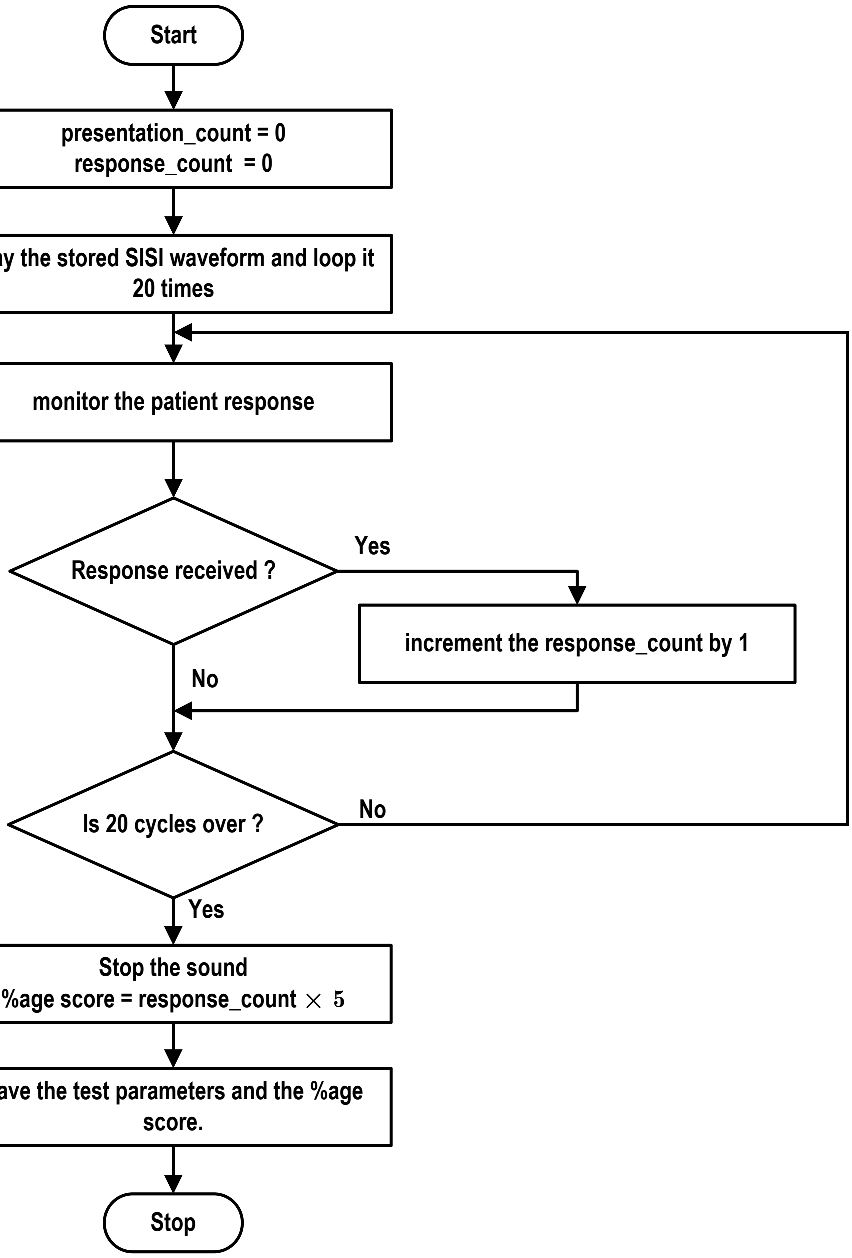


Figure 5.10: Procedure for the determination of the SISI score.

5.3.7 Speech Audiometry

Speech audiometry is used for finding the speech reception threshold (SRT) for a set of speech sounds consisting of standardized word lists in different languages. It is started with the level set as 25 dB above the hearing threshold level (HTL), as determined earlier by pure-tone audiometry. The patient is instructed to repeat the word as heard in the test ear. The screenshot of the speech audiometry test module GUI is shown in Figure 5.12. The test ear (left/right), the type of conduction (air/bone), the masking level, and the speech sound level

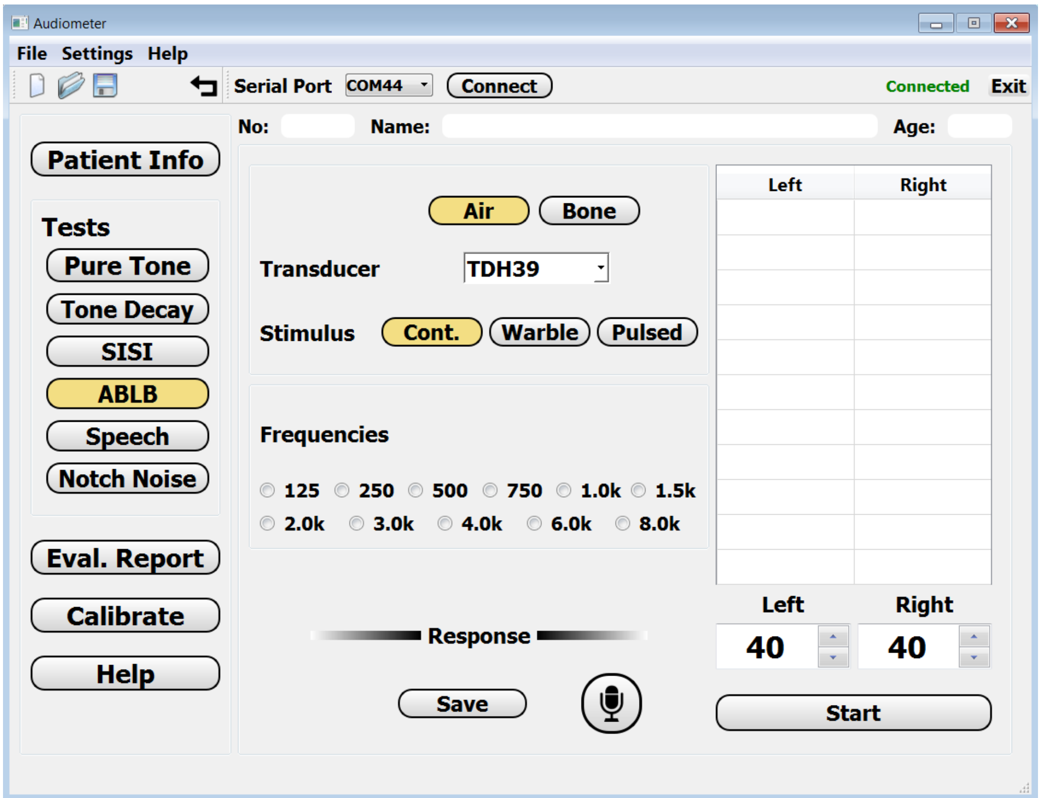


Figure 5.11: A screenshot of the ABLB test module.

are manually selected. The speech source i.e. microphone or CD player, has to be connected to the line-in of the PC. When the "Start" button is pressed, the input from the line-in of PC is sampled at 10 kHz and presented to the test ear. The stimuli from the microphone are a series of test words. The stimulus level is then manually controlled in 5 dB steps until a level is reached at which the patient identifies 50 % of the words correctly. The test can be terminated at any time by pressing the “Stop” button.

5.3.8 Notch-Noise Test

A notch-noise test is implemented for estimating the auditory filter shapes at the test frequencies. A pure-tone is presented in the presence of noise which has a spectral notch placed around the test frequency. The screenshot of the notch-noise test module GUI is shown in Figure 5.13. The test ear, test ear transducer, notch-ratio, notch-noise level, non-test ear transducer, masker, and the test frequency are manually selected.

When the "Start" button is pressed, the pure-tone and the notch-noise sounds are outputted together in the test ear. The notched-noise level is kept fixed and the stimulus level is changed in 1 dB up/down steps until the patient hears the tone at test frequency. The level at which the tone becomes just audible in the presence of the notched-noise is recorded as the

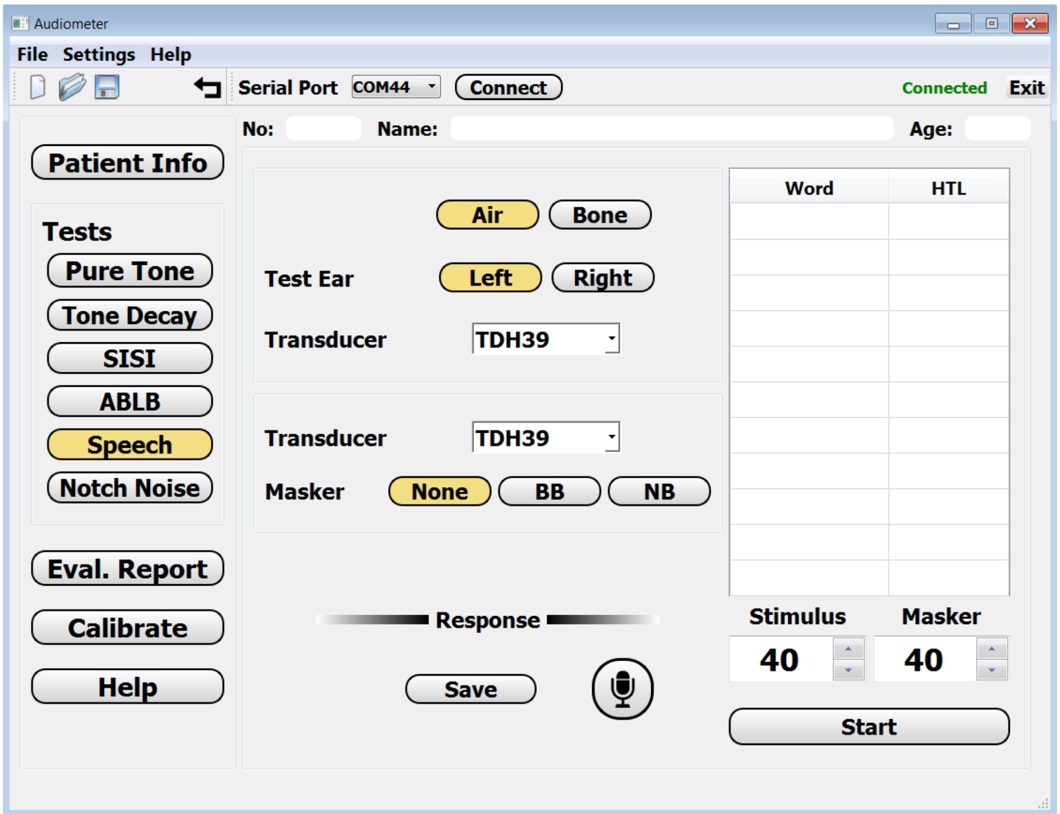


Figure 5.12: A screenshot of the speech audiometry test module.

masked threshold. The masked thresholds are found for different notch ratios at a test frequency. These are then used for estimating the auditory filter shape around the test frequency.

Since the tone and the notched-noise are in the same channel, it has to be ensured that the addition of two sounds does not exceed the maximum permissible value. For example, a 24-bit sample can have a maximum value of 2^{24}. The two sounds are scaled in such a way so that their addition utilizes the full range. When the stimulus and notched-noise are presented together, the stimulus level needs to be changed while the notched-noise level needs to be maintained. Since the attenuator in the audiometry module will attenuate the tone and notched-noise sounds equally, the ratio of their levels is set by generating the samples as weighted sum of the tone and notched-noise samples as

$$\text{output} = \alpha(\text{stimulus}) + \beta(\text{notched-noise}) \tag{5.8}$$

where α and β are the scaling factors for tone and the notched-noise samples, respectively. The scaling factors are calculated based on the difference in the levels of the tone and the notched-noise. If the difference between the notched-noise level and the tone level is $(S - N)$

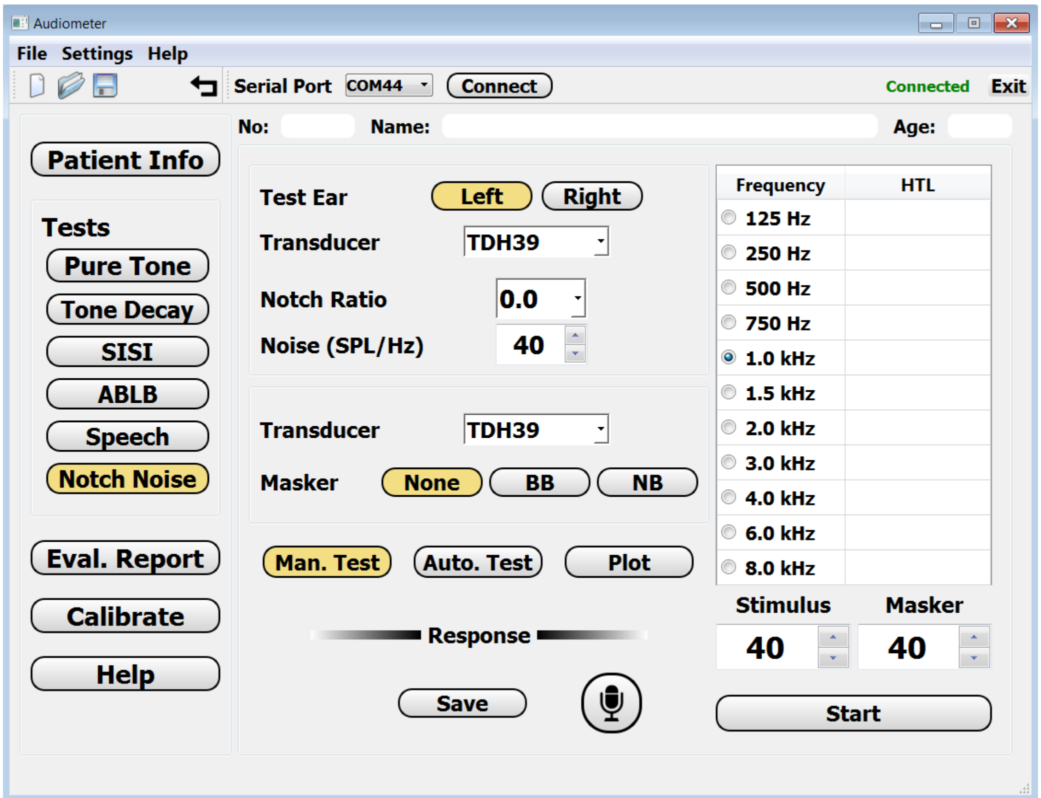


Figure 5.13: A screenshot of the notch-noise test module.

dB and the maximum achievable SPL levels for the tone and notched-noise sounds when presented alone are S_{max} dB and N_{max} dB, then the scaling factors are found by solving the equations

$$\frac{\alpha}{\beta} = 10^{\frac{((S_{max}-S)-(N_{max}-N))}{20}} \tag{5.9}$$

$$\alpha + \beta = 1 \tag{5.10}$$

The attenuation of the external attenuators of the test ear channel in the audiometry module is set equal to $(S - 20\log(\alpha/2))$dB. The masked threshold of the tone for a fixed notch-noise level is found using the "up-5 down-10" method of the stimulus level as described in Appendix A. After finding the masked threshold for one notch-ratio, the test is repeated for different notch-ratio for the same test frequency.

The test result is in the form of tone thresholds as a function of notch bandwidth for a test frequency. These values can be used for finding the auditory filter shape at the test frequency as described earlier in Section 2.10.

5.3.9 Audiogram Plotting

The threshold values recorded in the frequency-HTL table can be plotted by pressing the "Plot" button in various test modules. The symbols used for plotting the audiogram for air and bone conduction, as recommended by ASHA [34], are shown in Table 5.1. A screenshot of the audiogram showing the air and bone conduction plots simultaneously is shown in Figure 5.14. The buttons provided on the top of audiogram can be used to view the air conduction and bone conduction plots individually or simultaneously. The left ear plots are plotted in blue color while the right ear plots are plotted in red color. A comparison of the audiograms of air and bone conduction can be used to diagnose the type of hearing loss.

5.3.4 Talk-Over and Talk-Back Microphone

A button for verbal communication is provided for giving instructions to the patient in the sound-proof room. A microphone must be connected to the line-in of the PC and to the microphone input of the audiometric module before using it. For transmitting and receiving the microphone signals between the PC and the audiometry module, the HFP profile of the Bluetooth is used. The inputs from the microphones are sampled at 8 kHz and a gain factor is multiplied to each sample before transmitting. The gain factor is set for the comfortable listening from the settings menu in the menu bar of the application.

5.3.5 Calibration

Each transducer device needs to be individually calibrated to compensate for its sensitivity and frequency response. It involves finding the attenuation values of the digital attenuators in the audiometry module to generate a desired acoustic output in dB SPL from the transducer at each of the audiometric frequencies. The calibration setup for a headphone consists of an acoustic coupler along with a calibrated microphone and a sound level meter (SLM). The commonly used acoustic coupler for supra-aural headphones is B&K 4153 artificial ear along with a calibrated microphone for the measurement of sound pressure level. It is designed to simulate the acoustic impedance of a human ear. The headphone is kept on the artificial ear, with a force of 4.5 N applied over it to reduce acoustic leakage, and the sound pressure level is measured using the SLM.

The screenshot of the calibration module GUI is shown in Figure 5.15. It consists of a drop-down list for choosing the transducer to be calibrated, serial number of the selected transducer, date at which the transducer was last calibrated, calibration table for displaying the left and right calibration values, spin-boxes for setting the SPL level and the calibration

Table 5.1: Audiometric symbols used when plotting air and bone conduction hearing thresholds (adapted from [34]). R – Response, NR – No Response.

Mode	Left Ear		Right Ear	
	R	NR	R	NR
Air-Conduction –				
Unmasked	×	×↘	O	↙O
Masked	□	□↘	△	↙△
Bone Conduction –				
Unmasked	>	>↘	<	↙<
Masked	]	]↘	[	↙[

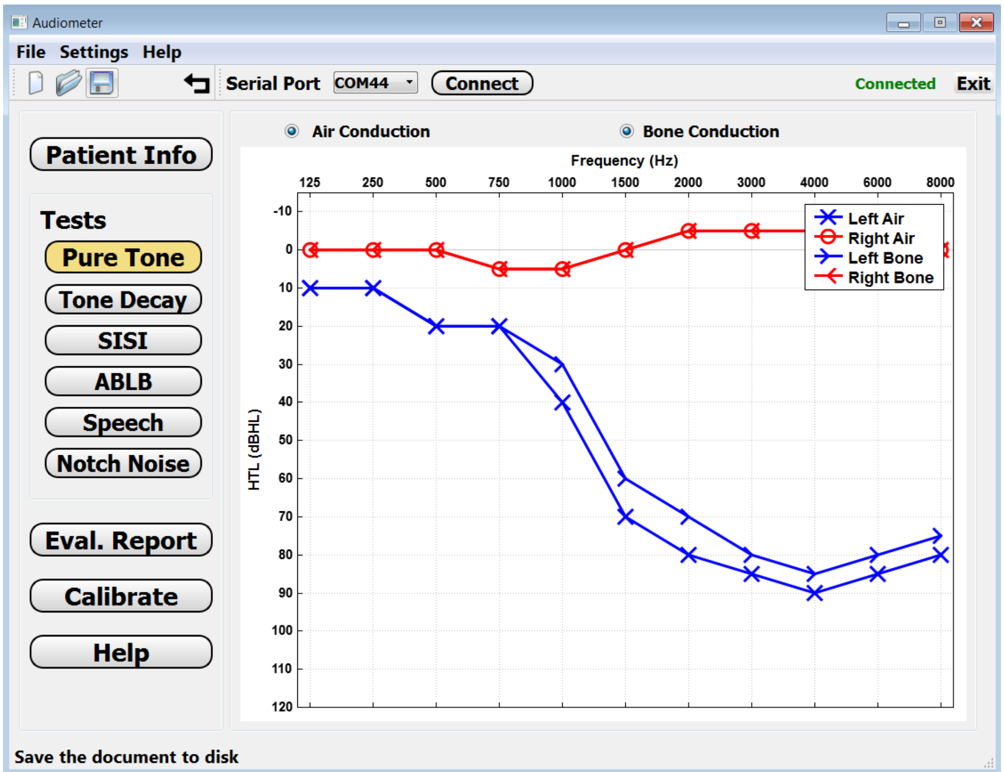


Figure 5.14: A screenshot of the audiogram module showing the air and bone conduction plots of left and right ear. The plot indicates that the person has a sensorineural hearing loss in the left ear and normal hearing in the right ear.

level, and buttons for manual /automatic and 40-dB output attenuator's calibration. The audiometry module can be calibrated either manual mode or automatic mode. In manual mode, the digital attenuator level of the audiometry module are changed manually in 1 dB

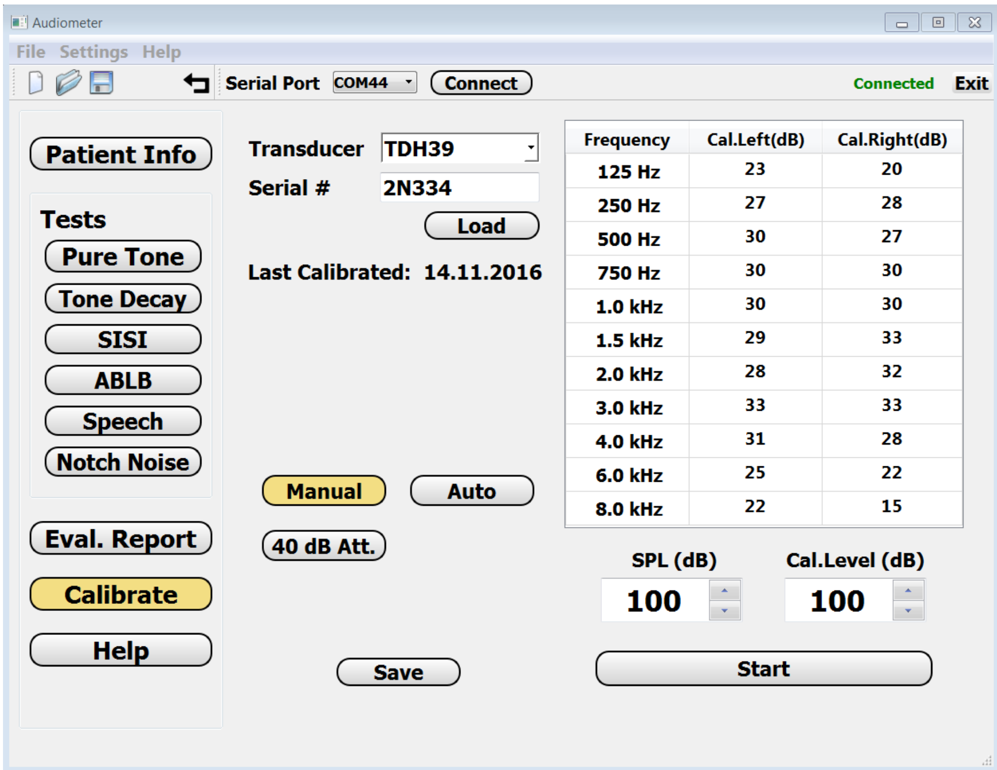


Figure 5.15: A screenshot of the calibration module.

steps using "Cal. Level" spinbox until the sound pressure level as set in "SPL" spinbox is obtained in the sound level meter. In automatic calibration, a dc feedback from the sound level meter is used to find the "Cal. Level" value for the SPL level set on the GUI. The dc output from the SLM corresponds to the sound pressure level detected by it. The dc value of the SLM is acquired using the level detector and subsequently the ADC of the microcontroller. The "Cal. Level" value is found by changing the digital attenuator of the audiometry module in a binary search method until the ADC value from the microcontroller, corresponding to the dc output from the SLM, falls in a particular range. This range corresponds to the ADC value of the set "SPL" level on the GUI. The digital attenuators can have a maximum attenuation of 124 dB. At each attenuation level, average of two consecutive ADC samples at an interval of 500 ms is used to reduce the influence of noise. For example, if the set "SPL" is 100 dB, and the sound level meter has sensitivity of 50 mV/dB for the input range 58 – 138 dB, then the SLM dc output for 100 dB SPL is 2.1 V. This dc output gets scaled to 1.45 V by the level detector circuit. The 10-bit ADC value of the microcontroller corresponding to 1.45 V level is 594. Hence the range for automatic calibration for 100 dB SPL with 1 dB tolerance and is set as ADC_{min} (= 570) and ADC_{max} (= 620) respectively. The time taken for calibration at single frequency is up to 7 s (i.e.

$\log_2 100$). It is to be noted that the digital attenuator referred to here is the attenuator present before the power amplifier in the audiometry module. The procedure followed for the automatic calibration at a test frequency is shown in Figure 5.16.

The actual attenuation of the switchable 40-dB post power-amplifier attenuator also needs to be determined since its value depends on the impedance of the transducer device. It is carried out by clicking the “40 dB Att.” button and selecting the cell in the table corresponding to 1 kHz for left/right channel. When the "Start" button is pressed, the corresponding left/right 40-dB post power amplifier attenuator is switched on and the digital attenuator level is manually set to obtain the desired SPL in the SLM. The actual attenuation value of this 40-dB attenuator is then calculated by comparing the calibration levels when was switched off and when it was switched on.

$$\text{O/P attenuator value} = (\text{dB SPL})_{\text{OFF}} - (\text{dB SPL})_{\text{ON}} + (\text{Cal. level})_{\text{OFF}} - (\text{Cal. level})_{\text{ON}} \quad (5.11)$$

Where $(\text{Cal. level})_{\text{OFF}}$ is the digital attenuator level for achieving the sound pressure level $(\text{dB SPL})_{\text{OFF}}$ when the transducer is calibrated without the 40-dB attenuator and the $(\text{Cal. level})_{\text{ON}}$ is the digital attenuator level for achieving the sound pressure level $(\text{dB SPL})_{\text{ON}}$ when the transducer is calibrated with the 40-dB attenuator.

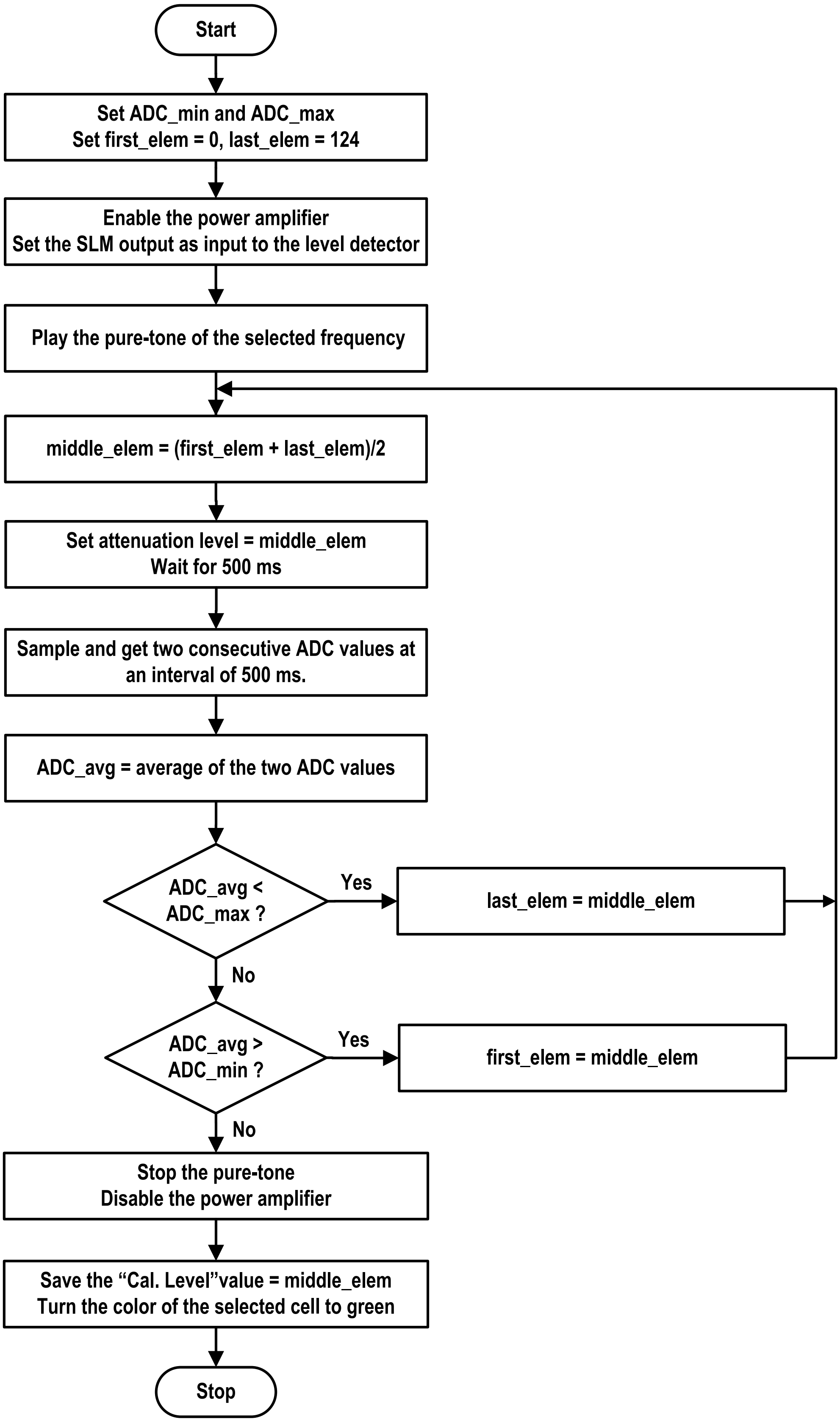


Figure 5.16: Flowchart of the procedure followed in automatic calibration of the transducer.

Chapter 6

TEST RESULTS

The implemented software modules like pure-tone test (manual/automatic mode), SISI test, tone decay test, speech test, notched-noise test, calibration (manual/automatic mode), and audiogram plotting were tested for satisfactory operation. The waveforms of the stimulus and maskers sounds of various tests were checked on the DSO. The maximum voltage measured across a 10-**Ω** TDH39P headphone at different frequencies is shown in Table 6.1. For acoustic measurement, artificial ear B&K 4153, microphone B&K 4176, and sound level meter B&K 2238 were used along with TDH39P headphones in a soundproof room.

The frequency accuracy and the harmonic distortion of the generated pure tones at audiometric frequencies are given in Table 6.2. The harmonic distortion of the pure tones at audiometric frequencies was found for maximum output level. For lower frequencies of 125 – 4000 Hz, the first 20 harmonics were considered and for the higher frequencies of 6000 – 8000 Hz, the first 11 harmonics were considered (due to sampling frequency limitation of the measuring instrument) for finding the harmonic distortion. The spectrum of the synthesized broadband noise and the narrowband noise were analyzed on the DSO.

The maximum achievable sound pressure level for pure tones at audiometric frequencies is shown in Table 6.3. Due the ambient noise level in the soundproof room,

Table 6.1: Measured maximum voltage level across a TDH39P transducer at audiometric frequencies.

Frequency (Hz)	Maximum voltage level (V_{pp})
125	3.52
250	3.52
500	3.52
750	3.60
1000	3.60
1500	3.60
2000	3.68
3000	3.76
4000	3.76
6000	3.84
8000	3.84

Table 6.2: Measured frequency accuracy and total harmonic distortion at audiometric frequencies for maximum output level without load.

Frequency(Hz)	**Frequency accuracy**	**Total harmonic distortion (in %)**
125	±0.16%	0.07%
250	±0.08%	0.08%
500	±0.08%	0.10%
750	±0.06%	0.12%
1000	±0.02%	0.14%
1500	±0.13%	0.22%
2000	±0.15%	0.27%
3000	±0.10%	0.42%
4000	±0.15%	0.62%
6000	±0.15%	0.70%
8000	±0.12%	0.70%

the minimum achievable sound level could not be found. The ambient noise level at each audiometric frequency was measured using the 1/3rd octave filter bands of the sound level meter and is shown in Table 6.4. The maximum achievable broadband noise level was 120.5 dB SPL. The maximum achievable narrowband noise level was 84 dB SPL for 125 Hz center frequency, and between 115 – 125 dB SPL for center frequencies above 250 Hz. The level accuracy of the sound level was found after calibrating the audiometer for TDH39P headphone, and comparing various sound levels (in dB HL) as shown in the GUI with that in the sound level meter. At all the audiometric frequencies, the level accuracy was found to be within ± 1dB. The SPL level accuracy, harmonic distortion, and frequency accuracy met the specification of advanced diagnostic audiometer as given in Table B.3. The talk-over and the talk-back microphones were also tested for satisfactory operation.

Table 6.3: Measured maximum achievable sound level in dB SPL for pure-tones at audiometric frequencies using TDH39P headphone. dB HL = dB SPL − RETSPL.

Frequency (Hz)	RETSPL (dB SPL)	Maximum sound level in dB SPL	Maximum sound level in dB HL
125	45.0	129.0	84
250	27.0	131.0	104
500	13.5	131.5	118
750	9.0	131.0	122
1000	7.5	130.5	123
1500	7.5	128.5	121
2000	9.0	129.0	120
3000	11.5	132.5	121
4000	12.0	134.0	122
6000	16	127.0	111
8000	15.5	119.5	104

Table 6.4: Ambient noise level at frequencies close to audiometric frequencies in sound proof room (measured using 1/3rd octave filters).

Frequency (Hz)	Ambient noise level in dB SPL
125	16
250	5.4
500	2.5
750	3
1000	3
1500	4
2000	4.7
3150	5.7
4000	6
6300	6.8
8000	6.5

Page intentionally left blank

CHAPTER 7

SUMMARY AND CONCLUSIONS

The objective of this project was to develop a two-channel advanced diagnostic audiometer using a PC and an external audiometric module for conducting subject-response audiometry and psychoacoustic tests. The audiometry module is designed as a compact battery powered wireless device to receive the audio signals and commands through Bluetooth. The audiometry module controls the level of the audio based on the commands received, and presents it at a calibrated level through a transducer. A PC-based application is designed to perform audiometric tests by sending commands and audio waveforms to the audiometry module. The design has decreased hardware complexity since the control panel is now on software on which all the controls and test results are displayed. By using wireless interfacing between PC and the audiometry module, the audio waveforms are not affected by noise in the PC circuit. Cabling between the audiometer and the soundproof room can be avoided by placing the audiometry module inside the soundproof room itself and controlling the module from outside through Bluetooth. A switchable 40-dB attenuator after the power amplifier is used for producing low level sounds with high SNR. For the automatic calibration of the transducers, a novel approach involving a feedback from the sound level meter dc output is used. Two microphone inputs for talk-back and ambient noise monitoring were also provided on the audiometry module. Among the tests which have been implemented as a part of the PC-based application are pure-tone test in manual and automated mode, SISI test in automated mode, tone decay test in automated mode, ABLB test, speech test, and notch-noise test.

Further improvement needs to be carried out to meet the specifications of an advanced diagnostic audiometer. The resistance in the path of the power amplifier reduced the output level by 6 dB for a 10 Ω load. This may be removed to achieve larger output sound level from various audiometric transducers. Use of current-drive also needs to be examined as it may lead to better frequency response of the transducers, although it has not been reported earlier. A novel circuit of a balanced current-drive is proposed in Appendix E. In PC application software, additional audiometric tests like Bekesy audiometry and psychoacoustic tests may be implemented. The audiometry module is designed to provide independent control of the signal levels over its two-channels. Therefore, any type audiometric tests can be implemented on the PC application software. The GUI may be improved after getting feedback from audiologists.

Appendix A

HEARING THRESHOLD DETERMINATION METHODS

Audiometry is performed after calibrating the test equipment along with the transducers. It is ensured that the test environment noise level is under the acceptable limits as per ANSI S3.1, 1999. The determination of the threshold level for pure-tones starts with air conduction test followed by the bone conduction test. There are three main types of air conduction transducers: supra-aural, circum-aural, and insert earphones. A supra-aural or circum-aural headphone may cause the ear canal to collapse. An insert earphone prevents the ear canal from collapsing and provides larger inter-aural attenuation, thereby reducing the need for masking of the non-test ear.

At the beginning of the test, the patient is informed about the task and familiarized with the stimulus to be used. An example of instruction for pure-tone audiometry using air-conduction [35] is:

"This test is used to find the softest sounds you can hear. Pay attention to the tone in your (left/right) ear. You may hear a noise in the other ear which should be ignored. Press the response button, whenever you hear the tone no matter how soft it is. As you release the button, a light will indicate that your response has been recorded".

There are many techniques to determine the hearing threshold level. The commonly used techniques are Hughson-Westlake technique and the technique recommended by ASHA [3].

Hughson-Westlake technique: The better ear is tested first, but if there is no difference between the two ears, then the right ear is the default starting ear. The hearing threshold is first determined at 1000 Hz. The thresholds are subsequently determined at 2000, 4000, 8000 Hz, followed by a retest of threshold at 1000 Hz. Then the thresholds are determined at 500, 250, 125 Hz. If the difference in the thresholds for these frequencies is more than 20 dB, then thresholds at intermediate frequencies i.e. 750, 1500, 3000, and 6000 Hz are determined. For testing an ear at a particular frequency, the patient is first familiarized with the tone by presenting the sound at a presumed supra-threshold level. If the response to the tone occurs, the level is decreased in 10 dB decrements until the patient no longer responds. The level is then increased by 5 dB. If the response to the tone occurs, the level is decreased by 10 dB, else the level is increased by 5 dB. This is known as the “up-5 down-10” technique. The minimum level at which the patient responds 3 out of 5 times is recorded as the hearing threshold.

Technique recommended by ASHA: The sequence of the presentation of various test frequencies is same as that of Hughson-Westlake procedure and "up-5 down-10" technique is used. The difference is at the start. A flowchart of the procedure is shown in Figure A.1. For testing the ear at a particular frequency, the sound is first presented at 30 dB HL. If the no response occurs, the tone is presented at 50 dB HL and at successive additional increments of 10 dB until a response is obtained or audiometer maximum limit is reached. If the response occurs, the hearing threshold is then determined using the "up- 5 down-10" technique i.e. the it is increased by 5 dB if no response occurs and level is decreased by 10 dB if response occurs. The minimum level at which the patient responds 3 out of 5 times is recorded as the hearing threshold.

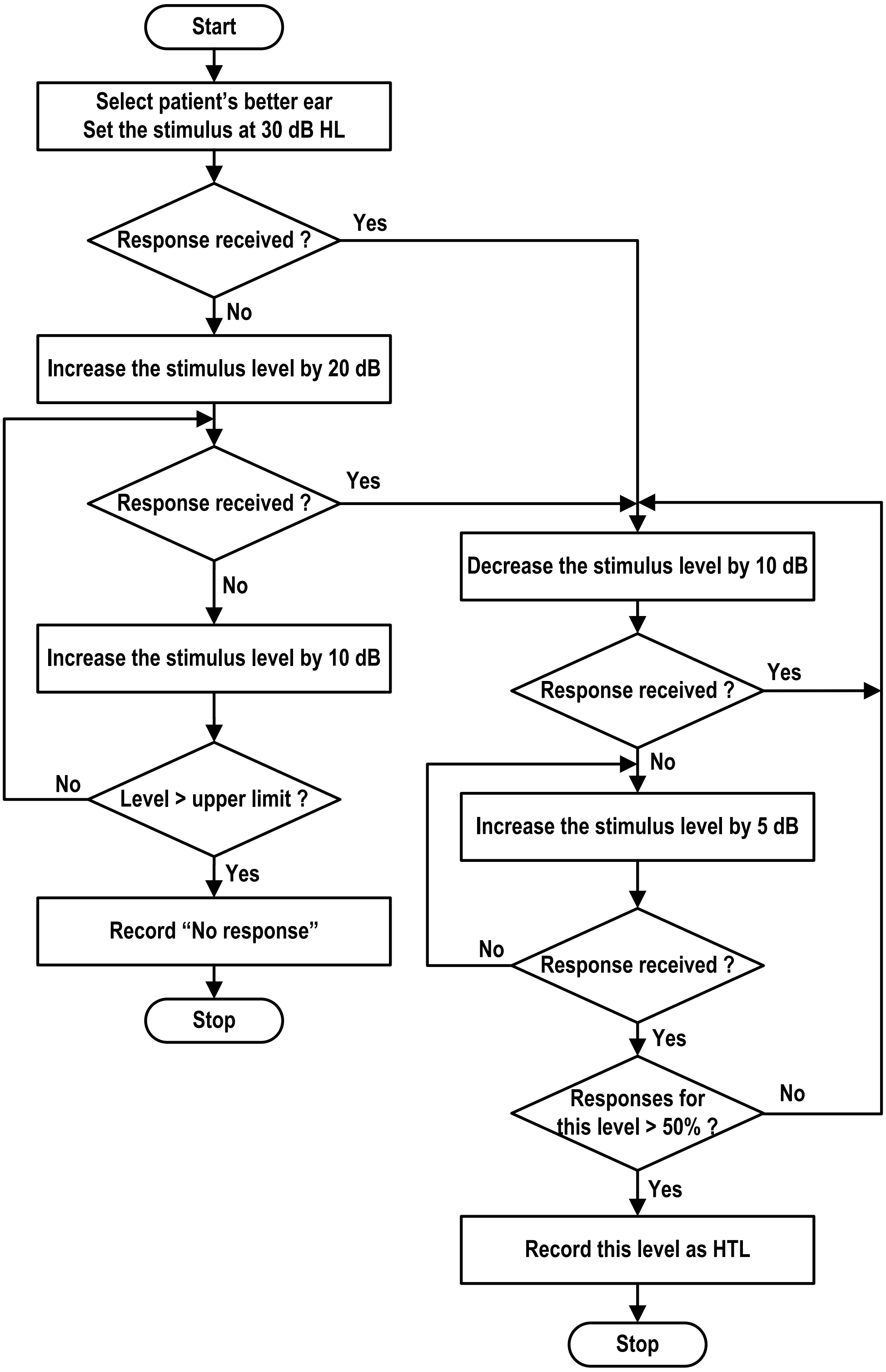


Figure A.1: Hearing threshold determination method as recommended by ASHA [3].

Page intentionally left blank

Appendix B

AUDIOMETER SPECIFICATIONS

The ANSI S3.6, 1996 [12] has classified audiometers, based on various parameters like the maximum achievable sound level (dB HL_{MAX}) at audiometric frequencies, permissible total harmonic distortion, frequency accuracy of the tones, tolerance for the sound pressure levels, and features like support for bone conduction test, high frequency pure-tone test, speech audiometry, sound field audiometry, and types of masking noise under the following four categories.

Type 1: Advanced Diagnostic Audiometer

Type 2: Diagnostic Audiometer

Type 3: Simple Diagnostic Audiometer

Type 4: Screening Audiometer

The required specification becomes less stringent from Type 1 to Type 4. Table B.1 shows the minimum required dB HL_{MAX} for various types at audiometric frequencies. The minimum sound level for all these types of audiometers at all the audiometric frequencies is −10 dB HL. Table B.2 shows the dB HL_{MAX} level in some of the commercially available audiometers. Table B.3 shows the tolerance requirements for various types of audiometers. The complete specifications for a Type 1 advanced diagnostic audiometer are as follows.

Frequency range

Air conduction (AC): 125 Hz, 250 Hz, 500 Hz, 750 Hz, 1 kHz, 2 kHz, 4 kHz, 6 kHz, and 8 kHz.

Bone conduction (BC): 250 Hz, 500 Hz, 750 Hz, 1 kHz, 2 kHz, 4 kHz, 6 kHz.

Output level

Air conduction (AC): −10 to 120 dB HL for pure tones (somewhat lower for frequencies outside 500 to 4000 Hz as shown in Table B.1), and −10 to 100 dB HL for speech.

Bone conduction (BC): −10 to 70 dB HL for pure tones (somewhat lower for frequencies outside 1000 to 4000 Hz as shown in Table B.1), and −10 to 55 dB HL for speech.

Channels

Two independent output channels for stimulus and masking noise. Microphone input for microphone signal for live speech output, and line-in for recorded sounds from tape/CD.

Table B.1: Minimum required dB HL_{MAX} for various categories of pure tone audiometer at different audiometric frequencies as per ANSI S3.6, 1996 [12]. The minimum sound level of these audiometers at all the audiometric frequencies is −10 dB HL.

Frequency (Hz)	**Type 1**		**Type 2**		**Type 3**		**Type 4**
	AC	BC	AC	BC	AC	BC	AC
125	70	-	60	-	-	-	-
250	90	45	80	45	70	35	-
500	120	60	110	60	100	50	70
750	120	60	-	-	-	-	-
1000	120	70	110	70	100	60	70
1500	120	70	110	70	-	-	-
2000	120	70	110	70	100	60	70
3000	120	70	110	70	100	60	70
4000	120	60	110	60	100	50	70
6000	110	50	100	-	90	-	70
8000	100	-	90	-	80	-	-

Table B.2: Maximum level (dB HL_{MAX}) achievable at the audiometric frequencies in some of the commercially available Type 1 and Type 2 audiometers. Minimum level achievable at all the frequencies is −10 dB HL. AC: Air conduction using TDH39P transducer, BC: Bone conduction using B71 bone vibrator on mastoid.

Frequency (Hz)	**AD226** [36]		**Sibelsound-400** [37]		**Inventis Piano** [38]		**Auditus A-1** [39]		**Oscilla USB350B** [40]	
	AC	**BC**	**AC**	**BC**	**AC**	**BC**	**AC**	**BC**	**AC**	**BC**
125	90	–	80	–	80	–	90	–	70	5
250	110	45	100	50	100	45	110	50	90	35
500	120	65	120	60	120	65	120	65	110	60
750	120	70	120	60	120	70	120	70	110	60
1000	120	70	120	70	120	75	120	75	110	60
1500	120	70	120	70	120	80	120	80	110	60
2000	120	75	120	70	120	80	120	80	–	–
3000	120	80	120	70	120	75	120	80	110	60
4000	120	80	120	70	120	75	120	80	110	60
6000	120	55	110	55	110	55	120	55	100	35
8000	110	50	110	–	100	50	110	50	90	35

Table B.3: Tolerance requirement for various types of audiometers [12].

Tolerances	**Type 1**	**Type 2**	**Type 3**	**Type 4**
Frequency accuracy	±1%	±2%	±3%	±3%
THD	2.5%	2.5%	2.5%	2.5%
SPL accuracy				
125–5000 Hz	±3 dB	±3 dB	±3 dB	±3 dB
6000–8000 Hz	±5 dB	±5 dB	±5 dB	±5 dB

Attenuation resolution

5 dB or less.

Masking noise

Broadband: −10 to 80 dB (effective masking level)

Narrowband (one-half or one-third octave bandwidth): −10 to 80 dB (effective masking level)

Speech-spectrum: −10 to 80 dB (effective masking level)

Signal selection

Continuous tone or pulsed tone (300 ms tone with 50 ms rise and fall time) or warble tone (8 Hz modulating frequency with 10% modulation depth)

Test Types

Pure tone test in manual/automatic mode, speech test, Bekesy test, loudness balancing test, tone decay test, masking level difference test, SISI, Stenger, and sound field test.

Transducers

Circum-aural headphones, supra-aural headphones, bone vibrator, insert earphones, talk back and talk forward microphones, and speakers with built in amplifier for sound field testing.

Page intentionally left blank

Appendix C
TRANSDUCER CHARACTERISTICS

Frequency characteristics of some of the commonly used audiometric supra-aural headphones are presented in Table C.1. The output SPL for all these transducers, except TDH39P, was calculated for 1 V_{rms} input using the frequency characteristics information provided in their respective datasheets. For TDH39P headphone, the frequency characteristics was measured using artificial ear B&K 4153, microphone B&K 4176, and sound level meter B&K 2238 in a soundproof room. An artificial ear is an acoustic coupler for the measurement of sound pressure in a cavity simulating the acoustic impedance of the ear canal. The contact force of the TDH39P headphone on the artificial ear was 4.5± 0.5 N. The dB SPL values were then measured at audiometric frequencies by maintaining the voltage across the headphone terminal at 1 V_{rms}.

The reference equivalent threshold sound pressure level (RETSPL) at a specific frequency is an acoustic coupler and a transducer specific dB SPL value that corresponds to 0 dB HL. Table C.2 shows the RETSPL values of commonly used audiometric supra-aural headphones (as provided in their datasheets).The dB HL value at a specific frequency for a transducer is obtained by subtracting the measured dB SPL with the corresponding RETSPL value.

$$\text{dB HL} = \text{dB SPL}_{meas} - \text{RETSPL} \tag{C.1}$$

The frequency characteristics of some of the commonly used insert earphones and bone-vibrators are presented in Table C.3 and Table C.4. Figure C.1, C.2, C.3 show the plots of the achieved sound level in dB HL in various transducers for 1 V_{rms} input and the minimum required dB HL_{MAX} for a Type 1 audiometer at audiometric frequencies (given in Table B.1). From plot in the figures, we observe that the maximum difference between the dB HL values at 1 V_{rms} and the required dB HL_{MAX} is 9 dB for Sennheiser 280 at 6000 Hz. Therefore to achieve the Type 1 audiometer minimum dB HL_{MAX} requirement, the maximum input voltage required for different transducers is different.

For the supra-aural headphones frequency characteristics are shown in Table C.1, the maximum achieved sound level achieved is 126 dB HL at 1 kHz for Sennheiser 280. Therefore for achieving −10 dB HL using this transducer, the required attenuation is 136 dB. Among the insert earphones frequency characteristics that is shown in Table C.3 for 1 V_{rms}, the maximum achieved sound level is 129 dB HL at 4 kHz for RadioEar IP30. Therefore for achieving −10 dB HL using this transducer, the required attenuation is

Table C.1: Sound output in dB SPL and dB HL at audiometric frequencies for some supra-aural headphones at 1 V_{rms} input. Z_0 = impedance at 1 kHz. dB HL = dB SPL − RETSPL (as given in Table C.2).

Frequency (Hz)	**TDH39P (Measured for SN. B51045) Z_0 =10 Ω**		**RadioEar DD45 (Based on datasheet [42]) Z_0 = 10 Ω**		**Beyerdynamics DT 48A (Based on datasheet [43]) Z_0 = 5 Ω**		**Telex 1470 Based on datasheet [44]) Z_0 = 5 Ω**		**Sennheiser HDA 280 (Based on datasheet [45]) Z_0 = 37 Ω**	
	dB SPL	**dB HL**	**dB SPL**	**dB HL**	**dB SPL**	**dB HL**	**dB SPL**	**dB HL**	**dB SPL**	**dB HL**
125	127	82	124	79	135	88	130	85	131	93
250	129	102	126	99	134	106	127	102	132	107
500	129	116	128	114	133	119	127	117	132	119
750	129	120	127	118	132	123	126	120	132	123
1000	129	122	126	119	130	122	125	122	133	126
1500	128	121	124	117	130	123	123	118	133	124
2000	128	119	125	116	131	123	121	117	126	118
3000	132	120	133	122	129	123	127	122	121	115
4000	129	117	127	115	123	118	121	115	125	116
6000	123	107	124	108	119	111	125	118	120	101
8000	122	106	119	103	119	105	121	112	110	92

139 dB. Among the bone vibrators frequency characteristics that is shown in Table C.4 for 1 V_{rms}, the maximum achieved sound level is 81 dB HL at 1.5 kHz for RadioEar B81. Therefore for achieving −10 dB HL using this transducer, the required attenuation is 91 dB. Hence we require a larger dynamic range for performing the air conduction test and relatively lower dynamic range for the bone conduction test.

Table C.2: Reference equivalent threshold sound pressure level (RETSPL) for various supra-aural headphones based on their datasheets. All the values are in dB SPL.

Frequency (Hz)	**TDH39P and RadioEar DD45** [41] [42]	**Beyerdynamics DT 48A** [43]	**Telex 1470** [44]	**Sennheiser 280** [45]
125	45.0	47.5	45.0	38.5
250	27.0	28.5	25.0	25.0
500	13.5	14.5	10.0	13.0
750	9.0	11.0	6.5	9.0
1000	7.5	8.0	3.0	7.5
1500	7.5	7.5	5.0	9.5
2000	9.0	8.0	4.0	8.0
3000	11.5	6.0	5.0	6.5
4000	12.0	5.5	6.0	9.5
6000	16.0	8.0	7.5	19.0
8000	15.5	14.5	9.0	18.0

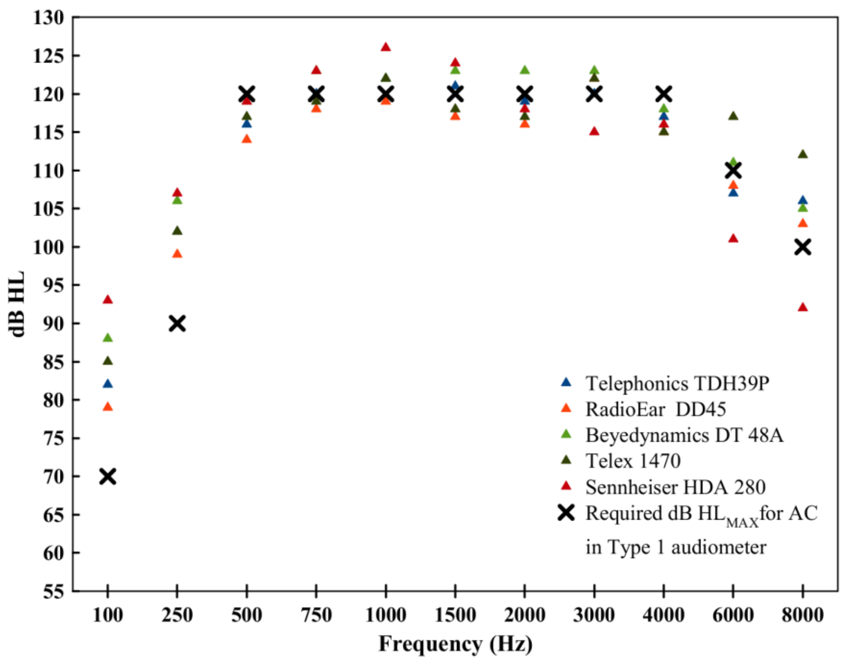


Figure C.1: Plot of dB HL values at audiometric frequencies for various supra-aural headphones at 1 V_{rms} input. AC – Air conduction.

Table C.3: Sound output in dB HL at audiometric frequencies for commonly used insert earphones at 1 V_{rms} input. Z_0 is the nominal impedance measured at 1 kHz.

Frequency (Hz)	**RadioEar IP30 (Z_0 =10 Ω)** [48] **dB HL**	**Etymotic ER-3C (Z_0 =10 Ω)** [49] **dB HL**	**Etymotic ER-3A (Z_0 =10 Ω)** [50] **dB HL**
125	99	99	96
250	107	109	106
500	117	116	118
750	121	124	119
1000	124	123	123
1500	123	122	124
2000	121	121	121
3000	124	122	120
4000	129	126	124
6000	105	101	105
8000	91	93	93

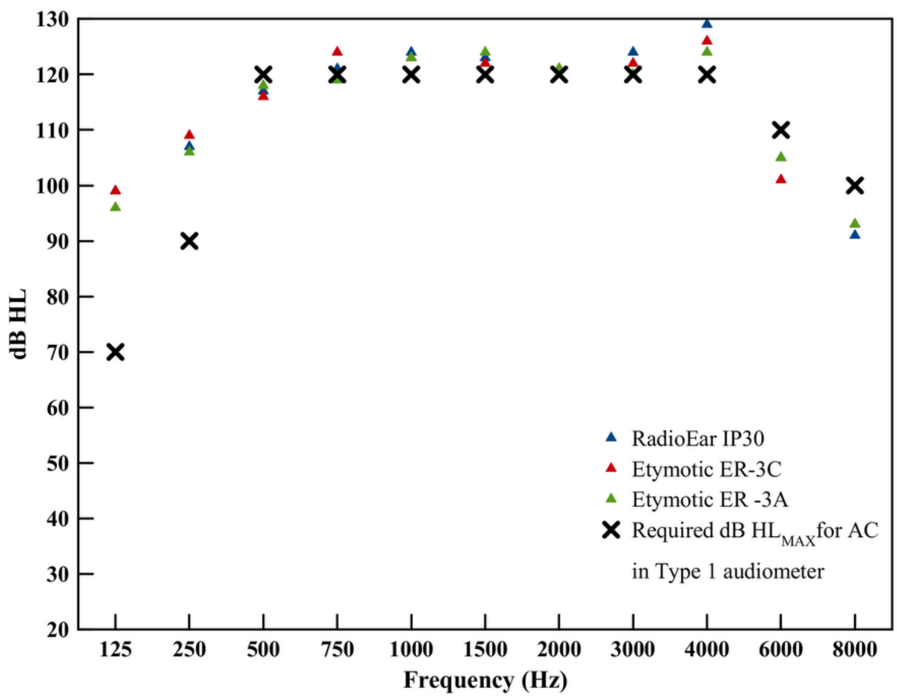


Figure C.2: Plot of dB HL values at audiometric frequencies for three insert earphones at 1 V_{rms} input. AC – Air conduction.

Table C.4: Sound output in dB HL at audiometric frequencies for B71 and B81 bone vibrators at 1 V_{rms} input. Z_0= impedance at 1 kHz.

Frequency (Hz)	**RadioEar B71 (Z_0 = 3.3 Ω) [46] dB HL**	**RadioEar B81 (Z_0 = 3 Ω) [47] dB HL**
250	45*	45
500	60	65
750	65	70
1000	71	76
1500	79	81
2000	77	77
3000	70	71
4000	69	70
6000	48	50
8000	43	41

*The THD at 250 Hz for RadioEar B71 bone vibrator is very high for high sound level. For 45 dB HL sound, the THD exceeds 6%.

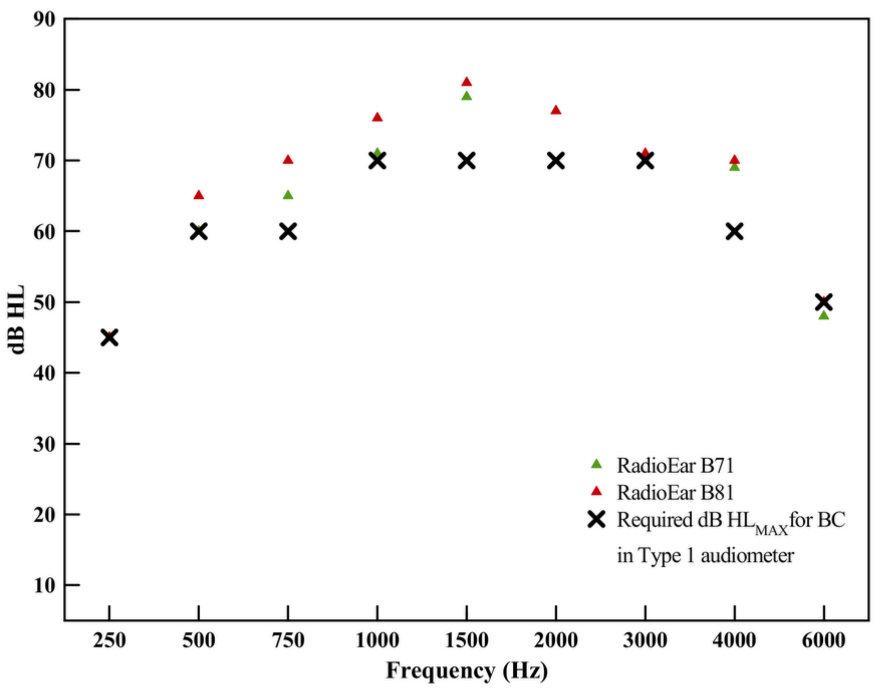


Figure C.3: Plot of dB HL values at audiometric frequencies for two bone vibrators at 1 V_{rms} input. BC – Bone conduction.

Page intentionally left blank

Appendix D

FIRST VERSION OF THE AUDIOMETRIC MODULE

D.1 Introduction

The first version of the audiometry module was designed as a USB-controlled microcontroller-based external hardware module. A PC-based application was developed to perform audiometric tests by sending commands through USB and audio signals through line-out to the external audiometric module. The audiometry module was used to precisely attenuate, amplify, and select a transducer to present the sounds at calibrated levels over the full range of audiometric levels for assessing the hearing of the test ear. The audiometry module was designed as a USB-powered device including features like maintaining the output signal-to-noise ratio over full dynamic range of sound, monitoring the output voltage level of the transducer terminals, automatic calibration using a sound level meter's feedback, talk-over microphone for giving verbal instructions to the patient sitting in soundproof room, and acquiring and acknowledging patient response using a response switch with an indicator. This design was realized as a PCB and assembled and fully tested. The design was revised to make the audiometry module wirelessly interfaced and to replace relays, which was being used in this design for transducer selection and switching the 40-dB attenuator, by analog audio switches to make the board more compact and to reduce the power consumption. The revised design has been presented in chapter 4.

D.2 Design Approach

The block diagram of the design is shown in Figure D.1. It consists of microcontroller, pre-amplifier, attenuator, power amplifier, output selector, multiplexer, level detector, and response switch. The audio signal path has four stages. The left and right channels of the audio signals from the PC line-out are applied as the inputs to the first stage which consists of a 2-channel pre-amplifier circuit to provide voltage gain and dc bias to the bipolar audio signals. The second stage consists of a 2-channel digitally-controlled programmable attenuator to provide attenuation to the audio signal in dB steps. The third stage has a 2-channel power amplifier to drive the transducers. The fourth stage uses relays as digitally-controlled switches for connecting the amplifier outputs to the transducers. It also has a switchable 40-dB attenuator after the power amplifier stage for extending the dynamic range of attenuation without adversely affecting the SNR. The switching of the 40-dB attenuator is done using relays. The output signal levels at the transducer terminals are

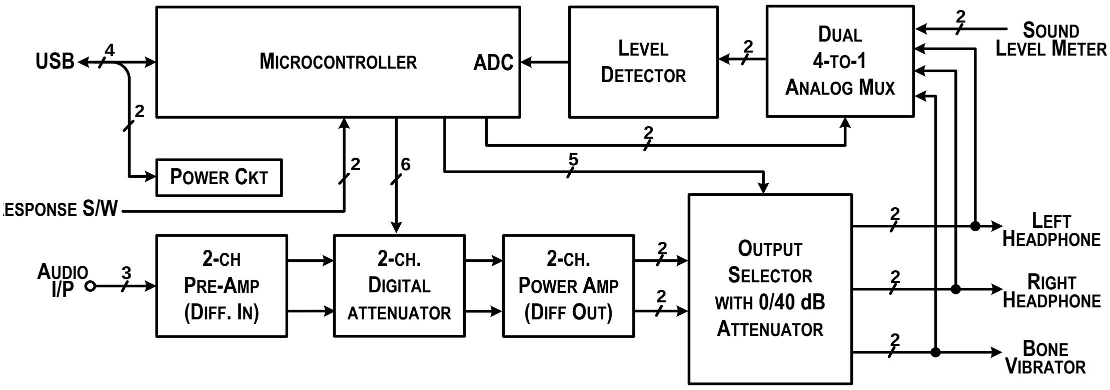


Figure D.1: The block diagram of the first version of audiometry module.

monitored using an analog multiplexer and level detector. The analog multiplexer also has an input for dc output from a sound level meter for calibration purpose. The microcontroller is used for communicating with PC through USB port, for generating the controls to the attenuators, relays, and analog multiplexer, and for interfacing the patient response switch. The module is designed to work on a single 5 V regulated supply from the USB port or from an external supply.

A detailed block diagram of the audiometry module is shown in Figure D.2. The module is controlled through USB port of a PC and its sound card is used to output the audio signals. The line-out of the PC sound card, consisting of two single ended left and right channels is connected to the inputs L-IN and R-IN of the 2-channel pre-amplifier. In each channel, the single-ended input signal is ac coupled to the differential input of the pre-amplifier. The amplifier provides single ended output with voltage gain of 2 and dc bias V_{BIAS}. The pre-amplifier output of each channel is fed to a digitally programmable logarithmic attenuator each with the dynamic range of 124 dB and resolution of 1 dB. The attenuation settings are loaded in the attenuator by microcontroller using SPI. The attenuator output is fed to the power amplifier with bridge-tied load (BTL) output, gain of 2 and dc bias VBIAS. Use of BTL configuration doubles the output voltage swing. It does not require dc blocking capacitors and hence the load can be directly coupled resulting in flat low-frequency response. The power amplifier has shutdown pin which is used to save power when the amplifier is not in use. The dynamic range of 124 dB as provided by the programmable attenuator is not sufficient for the audiometry, considering the frequency response of the transducers and the human ear. Dynamic range of attenuator must be at least 150 dB, as explained in Appendix C, so as to perform audiometry with various transducers. The range can be extended by cascading another programmable attenuator. However, the SNR at low level output gets limited by the noise of the power amplifier. Use of passive attenuator after

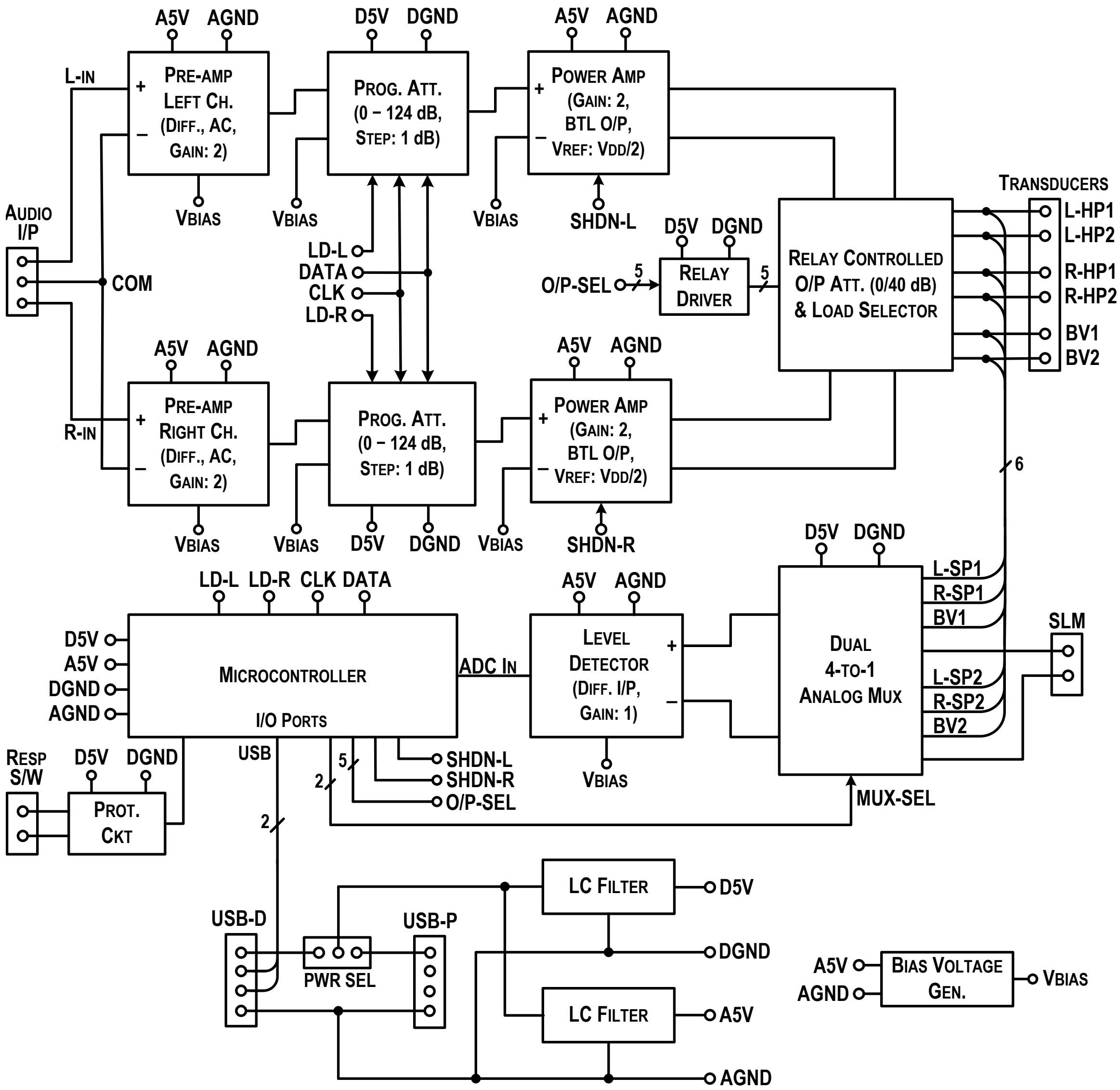


Figure D.2: Detailed block diagram of the first version of audiometry module.

the power amplifier does not adversely affect the SNR. Hence a switchable passive attenuator of 40 dB is used after the power amplifier. For switching the passive attenuators and selecting the load at the output, DPDT relays are used. A multiplexer and level detector is used to monitor the output signal level. A dual-channel 4-to-1 multiplexer is used to select the differential voltage at the transducers terminals of left headphone, right headphone, or bone vibrator. The fourth channel is used for connecting analog output of a sound level meter for automated calibration of the instrument. The multiplexer output is fed to a level detector which is a differential input full-wave rectifier followed by a low-pass filter. The dc signal from the level detector is applied to the ADC input of the microcontroller. A response button is interfaced through the protection circuit for acquiring the patient response.

The communication between the PC and the audiometry module, to control the attenuation levels of both the channels, switching the relays, controlling the power amplifiers, sending an acknowledgement of the response button, selecting the input to the multiplexer, and getting the ADC values corresponding to the voltage levels at the transducer terminals or from the sound level meter, is through the USB port. All communication and control operations are handled by the microcontroller.

The module is designed to work on a single 5 V regulated supply either from USB port through the connector USB-D or from an external supply through the connector USB-P. Two separate LC filters are used to obtain the analog and digital power supplies. The ICs involving analog signals (op amps and power amplifiers) are powered by A5V and AGND and those involving digital signals (attenuators, microcontroller and multiplexer) are powered by D5V and DGND. The reference V_{BIAS} is generated from the analog supply for biasing the analog signals and for setting the reference voltage for ADC.

The individual blocks of audiometry module are described in the following sections, followed by a description of the microcontroller program for handling the communication with the PC over USB and the description of the user interface.

D.3 Pre-amplifier

The line-out of the PC sound card consists of two single ended outputs and it is connected to the left input 'L-IN', right input 'R-IN', and common 'COM' terminals of the audio input port 'Stereo In'. In order to reduce ground noise pickup, the pre-amplifiers are designed with differential inputs. The circuit, shown in Figure D.3, uses op amps U1B and U2C from two quad op amps ICs MCP604 [19] (from Microchip). Separate ICs are used for each channel to avoid crosstalk. This IC has low input bias current (<10 pA) and rail-to-rail output voltage. The pre-amplifier circuits for the left and right channels are identical.

The L-IN and COM terminals are ac coupled to the differential input of the left channel pre-amplifier. The input coupling capacitors are selected to keep the 3-dB cut-off frequency much lower than the lowest audiometric frequency. The dc bias V_{BIAS} is added to convert the bipolar audio signal into unipolar. The gains and input resistances at the two input terminals are given as

$$A_{v(L-IN)} = R_5 / R_3 \quad \text{(D.1)}$$

$$A_{v(COM)} = [R_7 / (R_4 + R_7)][1 + R_5 / R_3] \quad \text{(D.2)}$$

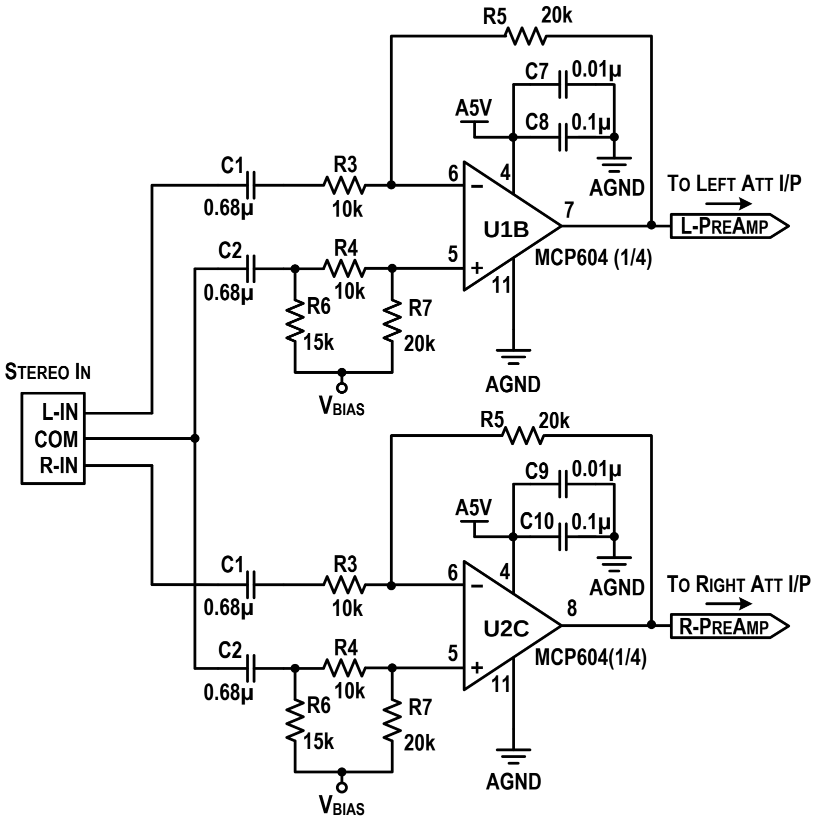


Figure D.3: Pre-amplifier circuit for left and right channels of audio input.

$$R_{in(L\text{-}IN)} = R_3 \tag{D.3}$$

$$R_{in(COM)} = R_6 \parallel (R_4 + R_7) \tag{D.4}$$

The resistor values are selected as $R_3 = R_4 = 10$ kΩ, $R_6 = 15$ kΩ, and $R_5 = R_7 = 20$ kΩ, to get

$$A_{v(L\text{-}IN)} = A_{v(COM)} = 2$$

$$R_{in(L\text{-}IN)} = R_{in(COM)} = 10 \text{ k}\Omega$$

The high-pass cut-off frequency is given as

$$f_c = \frac{1}{2\pi R_3 C_1} \tag{D.5}$$

With C_1 = 0.68 uF and R_3 = 10 kΩ, f_c = 23.4 Hz which is much lower than the lowest audiometric frequency of 125 Hz. For symmetry we select $C_2 = C_1$.

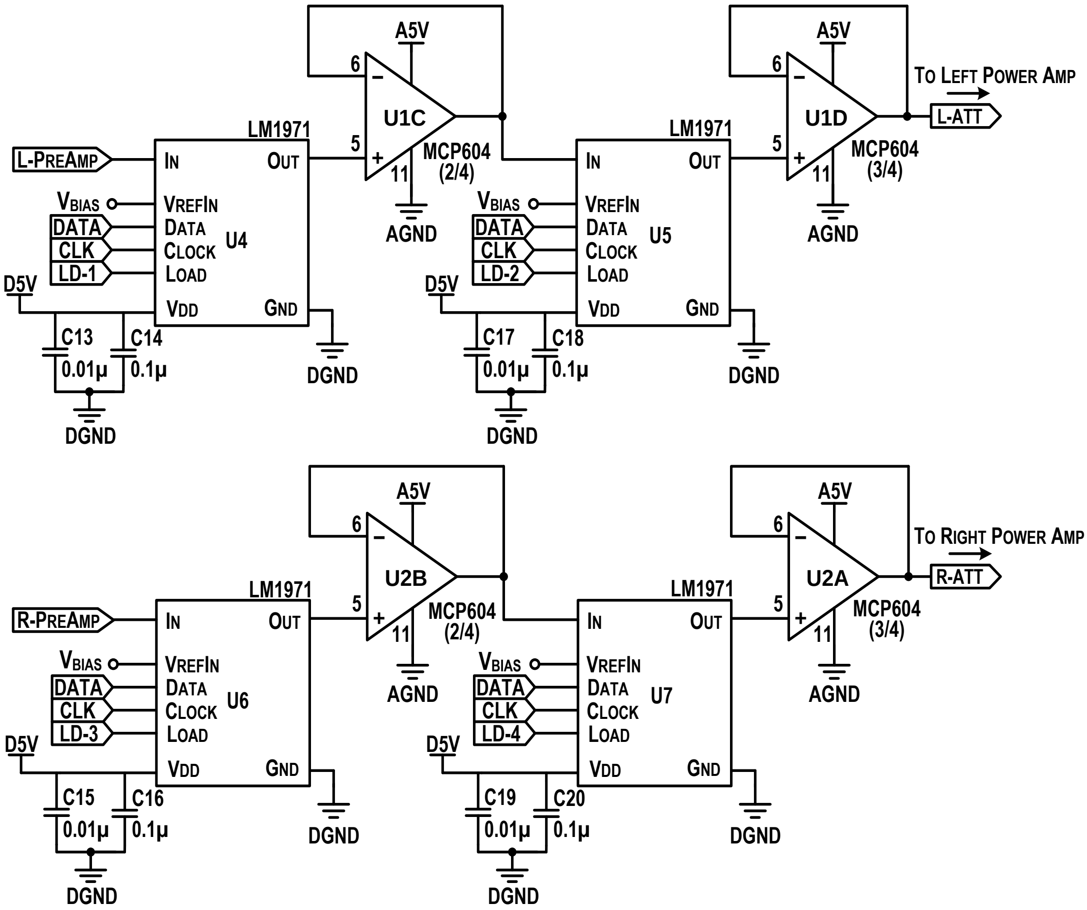


Figure D.4: Attenuator circuit for the left and right channels.

D.4 Attenuator

For each of the two channels, we need a digitally-controlled attenuator with dynamic range of approximately 140 dB and a resolution of 1 dB. Further the circuit has to operate with single 5 V supply with signal superimposed on a dc bias. For our purpose, the IC LM1971 [22] (from Texas Instruments) with single 5 V supply operation, provision for setting the reference voltage, and 3-wire serial interface was selected. It has a dynamic range of 62 dB, step size of 1 dB, and total harmonic distortion of less than 0.0008%. The attenuation step of 1 dB facilitates calibration for different transducers. Two chips are cascaded in each channel in order to extend the dynamic range to 124 dB. The output impedance of the attenuator IC varies non-linearly between 25 kΩ and 35 kΩ with the attenuation steps and hence it requires a buffer with low input bias current to avoid clicks at switching due to dc offsets. An additional switchable 40 dB passive attenuator, described later in Section D.4, is provided after the power amplifier so as to extend the range of attenuation.

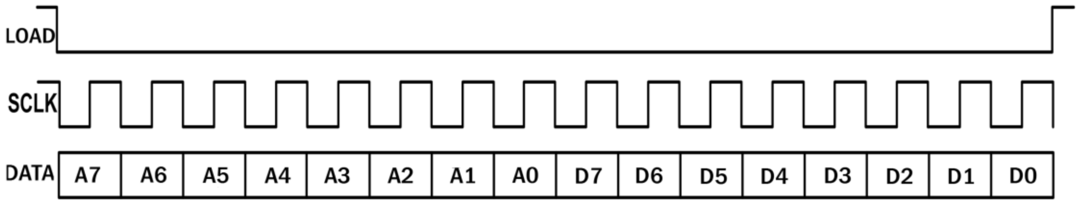


Figure D.5: Timing diagram of the serial interface of the logarithmic attenuator LM1971 [22].

The attenuator circuit for the two audio channels is shown in Figure D.4. The left channel has attenuators U4 and U5, with buffering by op amps U1C and U1D, respectively. The similar functions for the right channel are performed by U6, U7, U2C, U2D. The attenuator IC has 3-wire digital interface consisting of DATA, CLOCK, and LOAD pins. As the attenuator chip does not have serial out, the chips cannot be interfaced using daisy chaining and have to be interfaced using separate load pins. The four attenuator ICs share the CLOCK and DATA pins, while the LOAD pin of each one is connected to a separate pin of the microcontroller. The CLOCK and DATA pins are connected to SCK (RB1) and SDO (RC7) pins, respectively, of the SPI module of microcontroller. The timing diagram for the 3-wire serial interface is shown in Figure D.5. An active low on the LOAD pin enables the data input register of the attenuator IC. The DATA pin receives serial data corresponding to the value of the attenuation on each rising edge of the CLOCK. The serial data are composed of 8-bit address and 8-bit attenuation setting, with A0 as the address LSB and D0 as the data LSB. The device 8-bit address for the attenuator is 0x00h. The specific attenuator chip is selected by active low on corresponding LOAD pin (LD-1, LD-2, LD-3, or LD-4). The attenuation setting corresponds to the attenuation in dB with a maximum of 0x3Eh for 62 dB.

D.5 Power Amplifier

The power amplifiers for the left and right channels are realized using ICs U8 and U9. For this purpose, we have selected the IC TPA6211A1 [23] (from Texas Instruments). It is a 3.1-W amplifier with differential input and bridge-tied load (BTL) output for driving load greater than 3 Ω. The BTL outputs are 180^0 out of phase, thereby doubling the output voltage swing across the load as compared to the single ended output configuration for the same supply voltage. As the BTL output is differential, a coupling capacitor is not required at the output to block the dc to the speaker coil. Avoiding a coupling capacitor at the output improves the low-frequency response.

The power amplifier circuits for left and right channels are shown in Figure D.6. The power amplifier IC has on-chip 40 kΩ resistance in the feedback path, and hence R21 and R22 of 20 kΩ are used to provide a gain of 2. Capacitor C22 of 0.1 uF is used for filtering the

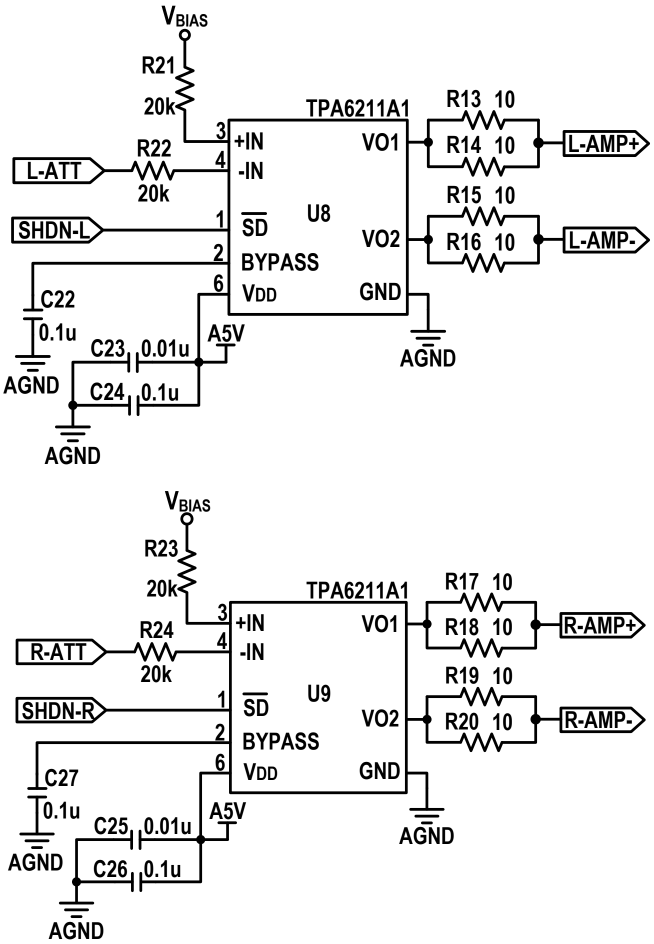


Figure D.6: Power amplifier circuits for left and right channel.

internally generated reference voltage. The shutdown pin is connected to save power when signal presentation is not required. Each pin of the BTL output is connected to two 10 Ω resistors in parallel to provide net source resistance of 10 Ω. Although it reduces the output voltage swing, it flattens the frequency response of the speaker by reducing the change in impedance with frequency and acts as a current limiter at accidental short of the output terminals. The source resistance is also used to detect whether a transducer is connected by measuring the voltage across the output.

D.6 Transducer Selection and Output Attenuator Switching

The circuit is designed for connecting three transducers, *viz.* the left speaker, the right speaker and the bone vibrator. These can be connected in different combinations to the left and the right audio channels for different types of tests. For air conduction test of the left ear, the stimulus is applied to the left audio channel and presented through the left speaker,

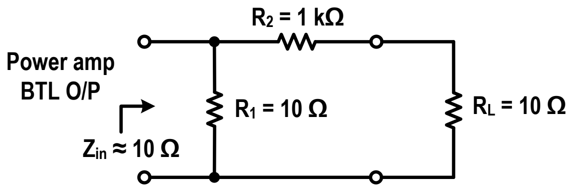


Figure D.7: The L-pad 40-dB attenuator circuit.

while the masker is applied to the right audio channel and presented through the right speaker. For air conduction test of the right ear, the stimulus is applied to the right audio channel and presented through the right speaker, while the masker is applied to the left audio channel and presented through the left speaker. For bone conduction test of the left ear, the stimulus is applied to the left audio channel and presented through the bone vibrator, while the masker is applied to the right audio channel and presented through the right speaker. For bone conduction test of the right ear, the stimulus is applied to the right audio channel and presented through the bone vibrator, while the masker is applied to the left audio channel and presented through the left speaker.

In order to extend the dynamic range of the audio signal to the speakers, a switchable 40-dB attenuator is provided after the power amplifier stage. As it is present after the power amplifier, it does not alter the signal-to-noise ratio. The bone vibrator is connected without the switchable attenuator as it requires lower output dynamic range as compared to air conduction transducers (explained in Appendix C). An L-pad attenuator circuit is designed for the BTL output from the power amplifier. The circuit is shown in Figure D.7. The resistance values for this output attenuator are selected such that its input impedance for a load of 10 Ω is 10 Ω. The circuit for transducer selection and output 40-dB attenuator switching is shown in Figure D.8. In this circuit, DPDT reed relays K1, K2, K3, K4, and K5 are used for the selection of transducers and switching the output attenuator. For this purpose, COTO2342 [51] (from Coto Technolgy), each with activation current of 25 mA, are used. The relays are controlled by microcontroller port pins using the relay driver U11. For this purpose, ULN2003 [52] (from Texas Instruments), is used. It is an array of 7 Darlington-pair transistors, each with driving capacity of 500 mA. Tables D.1 and D.2 show the microcontroller port pin outputs for selecting the transducers and switching the attenuators, respectively.

D.7 Circuit for Output Level Monitoring

The output level monitoring circuit, shown in Figure D.9, is used for monitoring the voltage output levels to the transducers. The dual 4-to-1 analog multiplexer U10 is used for

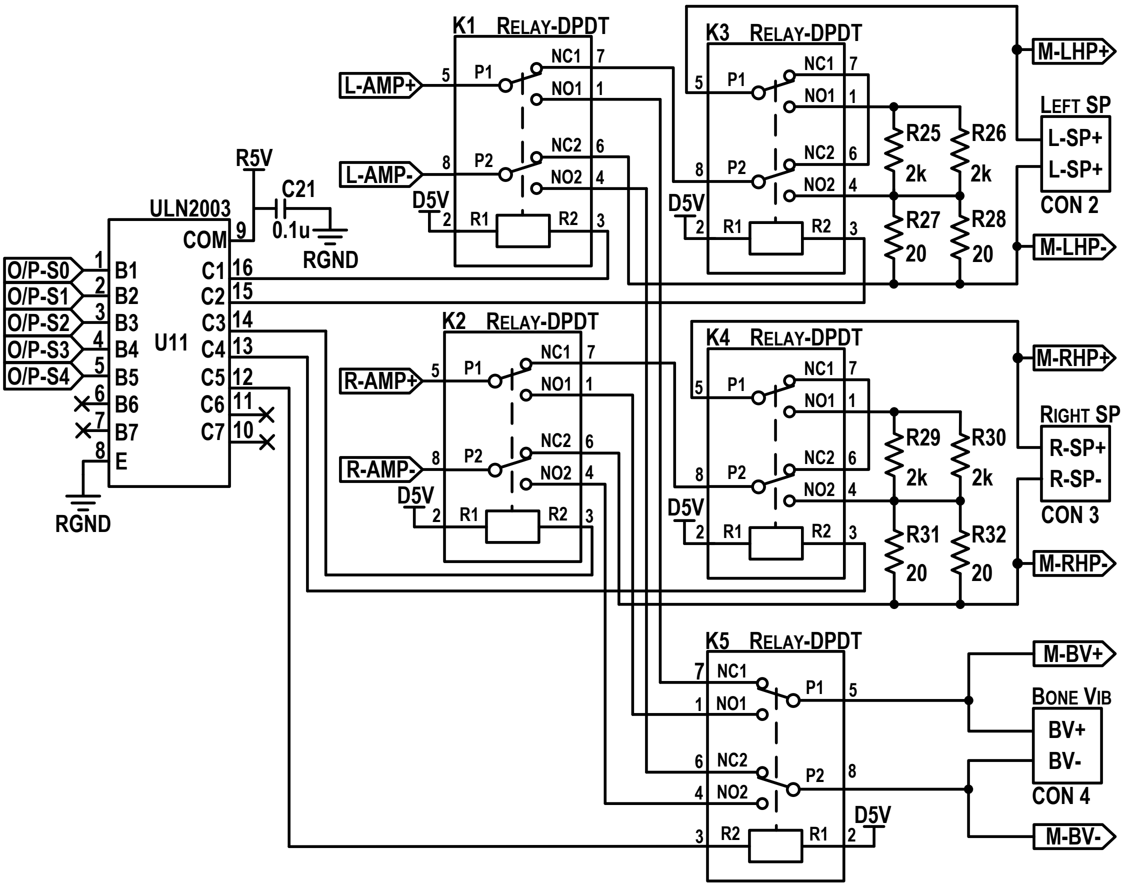


Figure D.8: Relay-controlled circuit for transducer selection and output attenuator switching.

selecting the transducer (left speaker, right speaker, or bone vibrator) for monitoring the output being applied to it. The IC CD4052 [25] (from Texas Instruments) is used for this purpose. The selected voltage is applied to the difference amplifier to generate single-ended output and fed to the full-wave rectifier and low-pass filter to get the average value $2V_P/\pi$, where V_P is the peak value. This voltage is fed to the ADC of the microcontroller. The values of the resistors R33, R34, R35, R36, and R37 are selected so that the input resistances at the two input terminals of the difference amplifier are equal and the gain is 0.5. As the BTL output of the power amplifier is ±5 V, the gain of the difference amplifier is set to avoid clipping of the signal after biasing the input signal to V_{BIAS}. With R_{33}, R_{34} = 100 kΩ, R_{35} = 300 kΩ and R_{36}, R_{37} = 50 kΩ, the input resistance is 100 kΩ and the gain is 0.5. The values of C28 and R42 forming the low-pass filter are selected to keep the cutoff frequency $1/(2\pi C_{28} R_{42})$ much lower than the lowest audiometric frequency of 125 Hz. With R_{42} = 200 kΩ and C_{28} = 220 nF, the low-pass cutoff frequency is 3.6 Hz. This circuit is also used for automatic calibration of the instrument for various transducers. This is done by placing the transducers on the artificial ear in case of headphone and on the artificial mastoid in case of bone vibrator

Table D.1: Controls for selecting transducers for the two audio channels. First three combinations are preferred as they consume less relay activation currents.

Control pins			Transducer connected to the left audio channel	Transducer connected to the right audio channel
O/P-S0	O/P-S2	O/P-S4		
0	0	0	Left speaker	Right speaker
1	0	1	Bone vibrator	Right speaker
0	1	1	Left speaker	Bone vibrator
0	0	1	Left speaker	Right speaker
0	1	0	Left speaker	None
1	0	0	None	Right speaker
1	1	0	Bone vibrator	None
1	1	1	None	Bone vibrator

Table D.2: Controls for switching the 40-dB attenuators.

Control pins		Attenuation	
O/P-S1	O/P-S3	Left speaker	Right speaker
0	0	0 dB	0 dB
0	1	0 dB	40 dB
1	0	40 dB	0 dB
1	1	40 dB	40 dB

and using the sound level meter to obtain dc voltage corresponding to the sound pressure level of the sound from the transducer. The dc output from the sound level meter, corresponding to the rms value of the signal received by its microphone, is selected by the analog multiplexer U10. The ADC value corresponding to the level is transferred by the microcontroller to the PC for updating the calibration table.

Monitoring of the voltage across the output terminals can also be used to check whether the transducer is connected at the output. The output voltage level is reduced if the transducer is connected because of the series resistance of 10 Ω.

D.8 Response and Acknowledgement Circuit

For receiving the response from the patient, a single-pole single-throw key (or push button) is interfaced to pin RC0 of the microcontroller U12. The response key K6 is connected to the audiometer circuit using a 2-wire cable as shown in Figure D.10. The LED L1 is connected in parallel to the switch K6 to provide an acknowledgement for the key press. The RC0 port is configured as the input port with a weak internal pull up. The LED L1 is

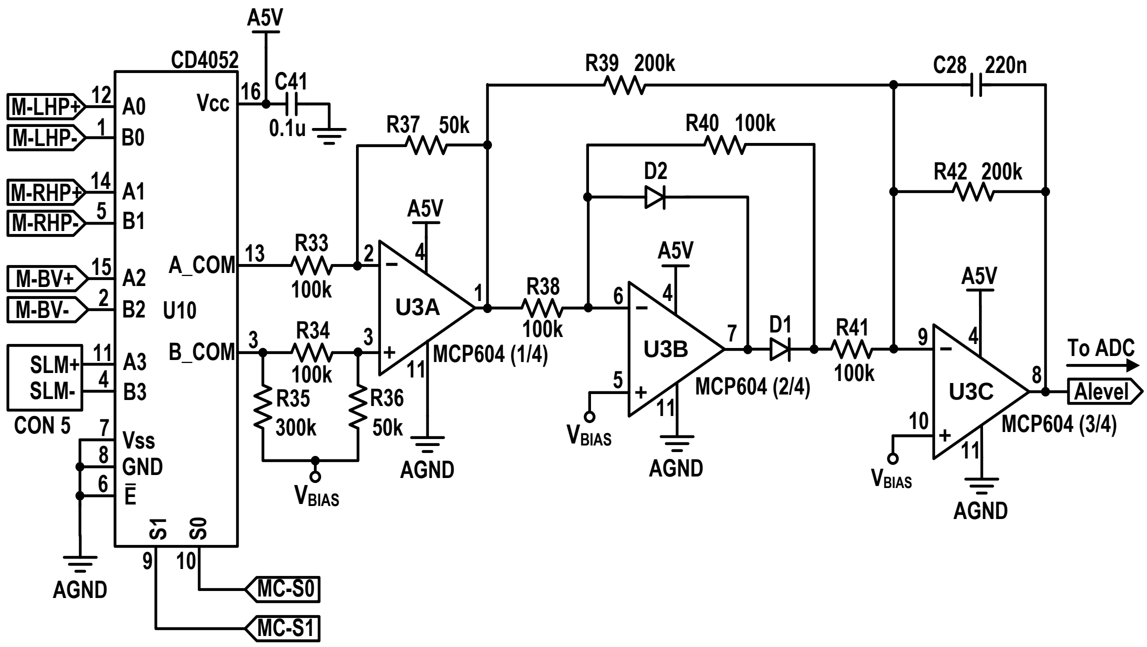


Figure D.9: Circuit for output level monitoring.

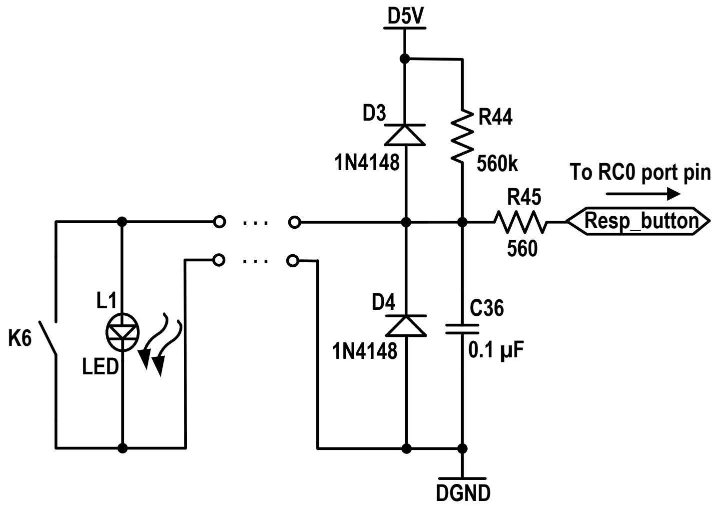


Figure D.10: Circuit for response and acknowledgement

normally off due to high value of R44. If the response is recorded, RC0 is set in output mode with logic high for few seconds. After the key is released, the LED is turned on by pulling RC0 high. Thus the circuit is used for receiving the subject response as well as providing acknowledgement without using an additional wire. The key K6 and LED L1 are in the subject response box while the rest of the components in the audiometry module. The combination of R44, D3, D4, and C36 is used to suppress any spikes which may get picked up in the cable to the subject response key.

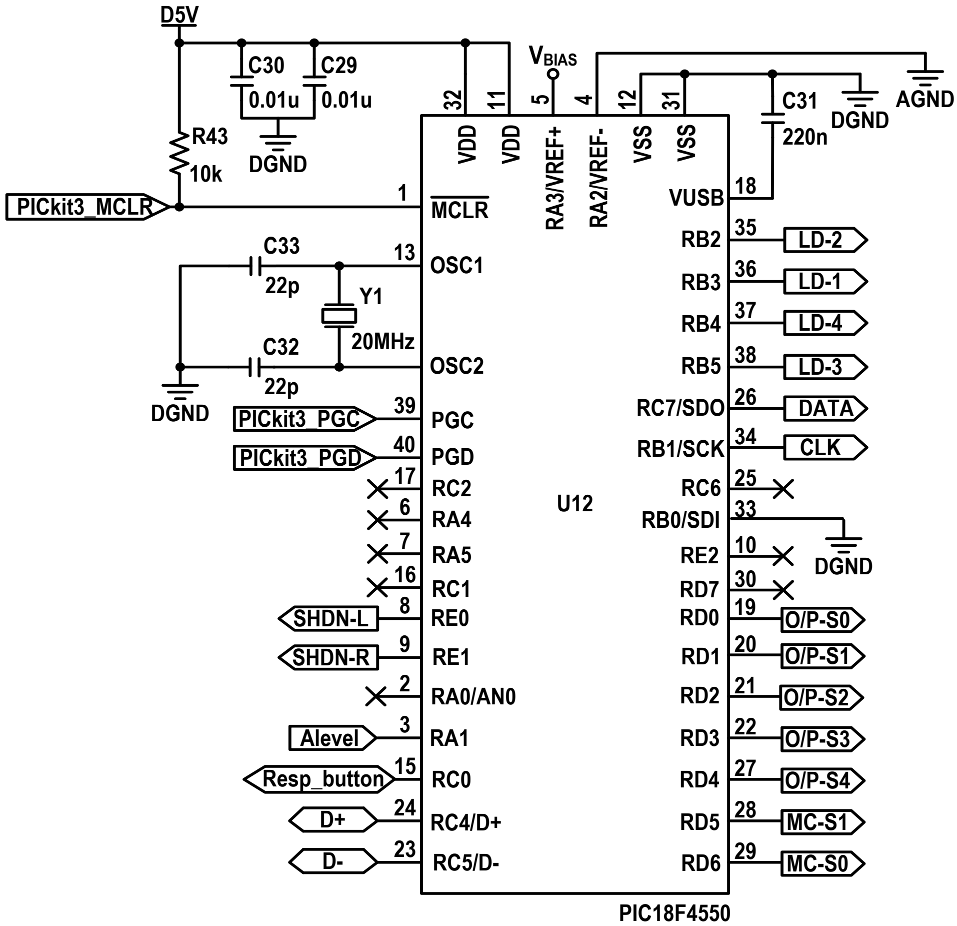


Figure D.11: The microcontroller circuit.

D.9 Microcontroller

The control and communication with attenuators, relays, power amplifiers, and multiplexer is done using a microcontroller IC U12. The microcontroller circuit is shown in Figure D.11. IC PIC18F4550 [53] (from Microchip) was chosen for this purpose since it has an USB module for communication with PC, SPI module for communication with the digital attenuators, 10-bit ADC for capturing the dc level from the level detector circuit, and sufficient I/O port pins for generating the controls to the relays driver, and analog multiplexer, and for interfacing the patient response switch. This microcontroller can work on 5 V supply, hence a voltage regulator is not required and the bus supply from the USB port can be directly used. The internal RC oscillator of the microcontroller is not stable; hence an external crystal oscillator is required to provide a stable clock for USB communication. An external crystal oscillator of 20 MHz along with the PLL of microcontroller is used to generate 48 MHz for full speed USB operation. The SDI pin of the microcontroller is shorted to ground

Table D.3: Function assigned to the I/O port pins of the microcontroller U12.

Port pins	In /Out	Function assigned
RA1	In	Level detector circuit output (Alevel) as ADC input.
RB1	Out	CLOCK of all the attenuator ICs U4, U5, U6, U7 (LM1971).
RB2	Out	LOAD of attenuator IC U5 (LM1971).
RB3	Out	LOAD of attenuator IC U4 (LM1971).
RB4	Out	LOAD of attenuator IC U7 (LM1971).
RB5	Out	LOAD of attenuator IC U6 (LM1971).
RB6	In	PGC of programmer/debugger module (PICkit 3).
RB7	In	PGD of programmer/debugger module (PICkit 3).
RC0	In/Out	Resp_button from response switch circuit.
RA2/V_{REF-}	In	AGND for ADC negative reference input
RA3/V_{REF+}	In	V_{BIAS} for ADC positive reference input.
RC4/D+	In/Out	USB (D+).
RC5/D-	In/Out	USB (D-).
RC7/SDO	Out	DATA of all the attenuator ICs U4, U5, U6, U7 (LM1971).
RB0/SDI	In	Grounded.
RD0	Out	B1 of relay driver IC U11 (ULN2003).
RD1	Out	B2 of relay driver IC U11 (ULN2003).
RD2	Out	B3 of relay driver IC U11 (ULN2003).
RD3	Out	B4 of relay driver IC U11 (ULN2003).
RD4	Out	B5 of relay driver IC U11 (ULN2003).
RD5	Out	S1 select line of multiplexer U10 (CD4052).
RD6	Out	S0 select line of multiplexer U10 (CD4052).
RE0	Out	$\overline{SD}$ of power amplifier IC U8 (TPA6211A1).
RE1	Out	$\overline{SD}$ of power amplifier IC U9 (TPA6211A1).
RE3	In	MCLR of the programmer/debugger module (PICkit 3).

since there is no serial out pin in the digital attenuator. The functions assigned to various port pins of the microcontroller are listed in Table D.3.

D.10 Power Circuit

The circuit of power supply is shown in Figure D.12. The audiometry module is provided with two mini USB connectors. It has an option of being powered either by USB or external 5 V dc supply by shorting either of the two adjacent pins of the jumper CON9. The

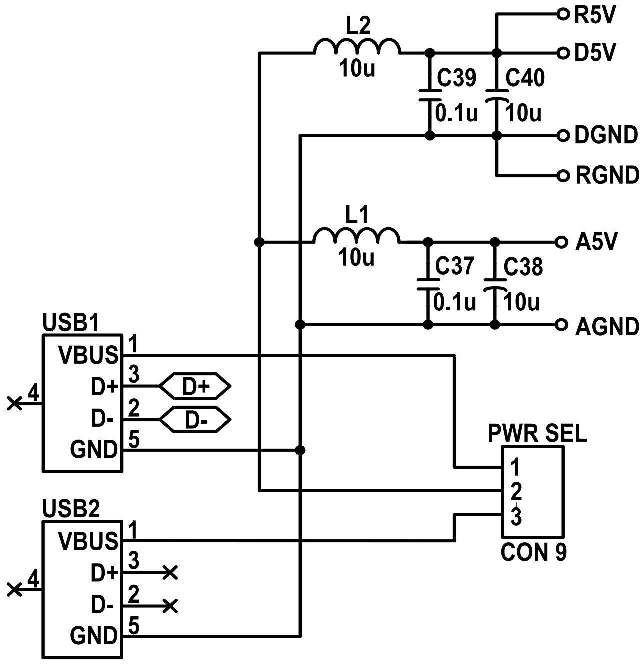


Figure D.12: Power circuit.

Table D.4: Estimate of current requirement

Component	Current drain (mA)
U4–U7: LM1971	12.0
U8, U9: TPA6211A1	380.0
U1–U3: MCP604	3.3
U10: CD4052	0.2
K1–K5: COTO2342 relays	85.7
U12: PIC18F4550	2.6
Total (Approx)	485 mA

ICs involving analog signals (op amps and power amplifiers) are powered by the tracks A5V and AGND. The ICs involving digital signals (attenuators, microcontroller and multiplexer) are powered by the tracks D5V and DGND. A separate LC filter is used for filtering the analog and digital power supply and their tracks are kept separated after the filters, to prevent the digital noise from corrupting the analog signals. A separate set of supply tracks, R5V and RGND, are taken from the beginning of the digital supply tracks for powering the relay driver to avoid the switching noise generated by the relays from affecting the other components. The maximum currents drawn by the ICs are given in Table D.4. The total maximum current, estimated as the sum of current drains of individual ICs, is 485 mA.

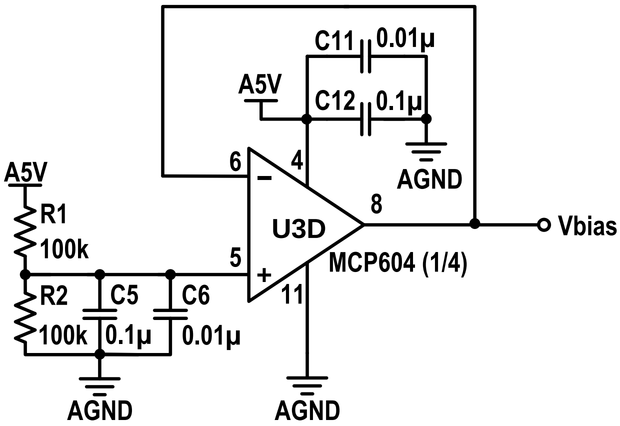


Figure D.13: Bias voltage generator.

As the circuit is designed to work on a single +5 V supply, a bias voltage of +2.5 V (i.e. half supply voltage) is generated to add a dc bias to the bipolar audio signal and make it unipolar. Figure D.13 shows the circuit used for generating the bias voltage. A potential divider comprising two equal valued resistors R1 and R2 is used for generating half of the supply voltage. The op amp U3D, one of the four op amps in IC MCP604, is used as a unity gain buffer. The capacitors C5 (0.1 µF) and C6 (0.01 µF) are used for noise filtering over a wide band.

D.11 Microcontroller Program

The microcontroller is programmed to perform the following tasks: (i) Communicate with PC through USB to get the attenuation settings and controls for the I/O ports; (ii) Set the attenuation level for digital attenuators through SPI module; (iii) Set the I/O port to switch the 40-dB post power amplifier attenuator, select a transducer (left/right headphones or bone-vibrator) at the output, enable/disable the power amplifiers, select an input to the analog multiplexer, scan the response key; (iv) Use its ADC to get the digital value corresponding to the average voltage level at the transducers terminals.

For USB communication, the Communication Class Device Abstract Control Model (CDC-ACM) is used as the USB class model. This model uses two interfaces for communication, *viz.* Communication Class Interface (CCI) and Data Class Interface (DCI). The CCI uses one endpoint for device management (control endpoint 0) and one for events notification (interrupt IN endpoint). The DCI uses two data endpoints, bulk IN and bulk OUT for data transfer to/from the PC respectively.

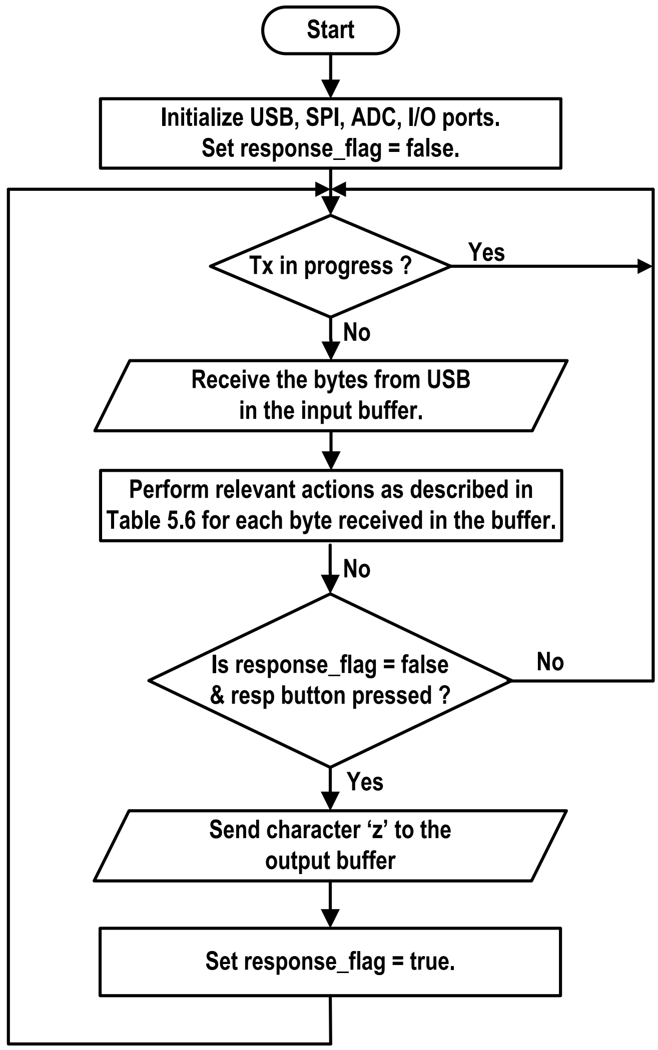


Figure D.14: Flowchart of the microcontroller program.

A flowchart of the microcontroller program is shown in Figure. D.14. In the initialization step, the USB module, SPI module, ADC module, and various I/O ports of the microcontroller are initialized. Initialization of the USB module includes configuring the USB descriptors, ACM interfaces, functional descriptors, CCI and DCI endpoints descriptors, string descriptors, interrupts flags, error flags, ping-pong buffers, line coding (data bits, baud rate, parity bits, start bits, stop bits), flushing any pending transactions and getting ready for first packet, and polling the VBUS pin to check whether the USB port is connected to the PC. Initialization of the SPI module includes configuring the SDO, SCK, and SDI pins as serial pins and setting the clock rate to 12 MHz and clock polarity and clock phase such that the transmission of data occurs at the leading edge of the clock. Initialization of ADC module includes enabling the ADC module, selecting the channel 1 as input to the ADC, setting the external voltage references (VREF+ and VREF-), and setting the acquisition time and the

A/D conversion clock period. The I/O port pins interfaced with the power amplifiers (RE0, RE1), digital attenuators (RB2 – RB5), analog multiplexer (RD5, RD6), and relay driver (RD0 – RD4) are configured as output. The power amplifiers and the relays are disabled and the digital attenuators are loaded with 0 dB attenuation value using SPI. The I/O port pin RC0, interfaced with the response key, is initially configured as input for scanning the response. In order to avoid multiple responses being recorded when the response key is pressed a flag, labeled as response_flag, and initially set as false, is used.

The program operation after initialization consists of a loop. The first step of the loop checks for any pending bytes yet to be transmitted from the 64-byte output buffer to the PC. If there are no pending transmissions, then the 64-byte input buffer is checked for new bytes received from the PC through USB. The bytes are read one by one. For each byte received, different action is performed as described in Table D.5. Next the response key status is checked. If the key is pressed and the response_flag is set false, the character ‘z’ is written to the output buffer and sent to the PC and the response_flag is set true. The program stops scanning the response switch if the response_flag is set true. The PC, after receiving the 'z' character, sends the character 't' as an acknowledgment. After receipt of this acknowledgement, the LED L1 is controlled to be on for 2 s. If another response needs to be taken, the character 'u' is sent to the microcontroller to set the response_flag as false, and start scanning the response key.

The attenuation settings for each digital attenuator in both the channels are sent from the PC as whole number values. The sequence of operation for setting the attenuation level of a digital attenuator IC consists of (i) pulling its LOAD pin low with all other attenuators LOAD pins kept high, (ii) writing the address byte 0x00h to the SPI buffer, (iii) polling the flag of buffer full status to ensure that the transmission of data in SPI buffer to the SDO pin is complete, and (iv) writing the attenuation setting to the SPI buffer. The left and the right output attenuators are enabled or disabled using RD1 and RD3 port pins of the microcontroller.

D.12 PCB Design

After completing the schematic of the circuit, as given later in Figure D.17, D.18, D.19, and simulating its modules on Multisim (ver. 11 from National Instruments), a PCB was designed using Altium Designer. The PCB is a double-sided, plated-through-hole (PTH) board of 104 mm × 75 mm size. Since the circuit blocks on the PCB have mixed signals (i.e. analog and digital), special care has been taken in the layout design to avoid coupling of digital ground to the analog circuit. The supply entry points are decoupled by parallel

Table D.5: Functions performed by the microcontroller based on the commands received through USB.

Commands	Function
$	The next three consecutive bytes following this character are converted into decimal value and are used to set the attenuation level of the first digital attenuator in the left channel.
#	The next three consecutive bytes following this character are converted into decimal value and are used to set the attenuation level of the second digital attenuator in the left channel.
%	The next three consecutive bytes following this character are converted into decimal value and are used to set the attenuation level of the first digital attenuator in the right channel.
&	The next three consecutive bytes following this character are converted into decimal value and are used to set the attenuation level of the second digital attenuator in the right channel.
a	Disable the left power amplifier.
b	Enable the left power amplifier.
c	Disable the right power amplifier.
d	Enable the right power amplifier.
e	Enable the output attenuator of left channel.
f	Disable the output attenuator of left channel.
g	Enable the output attenuator of right channel.
h	Disable the output attenuator of right channel.
I	Select bone vibrator as the output transducer in left channel.
J	Select bone vibrator as the output transducer in right channel.
K	Select headphones as the output transducer in left channel.
L	Select headphones as the output transducer in right channel.
s	Write the ADC value, corresponding to the current dc level from the level detector circuit output to the output buffer and send it to the PC.
t	Set the port pin interfaced with response key as output and set it high to glow the LED.
u	Set the port pin interfaced with the response key as input and start scanning for response.
v	Select left channel transducer terminals as input to the level detector circuit.
w	Select right channel transducer terminals as input to the level detector circuit.
x	Select the bone vibrator terminals as input to the level detector circuit.
y	Select the SLM dc output as input to the level detector circuit.

combination of 10 μF and 0.1 μF ceramic capacitors to filter out noise over wide band. Decoupling capacitors of 0.1 μF and 0.01 μF have been used for each IC close to its power supply pin to filter out voltage spikes. The differential data paths of the USB (D+ and D-) and the differential outputs of the power amplifiers are routed close and parallel to each other. The analog ground and digital ground are routed separately throughout the board and are shorted at the input power supply. Digital +5 V (D5V) and analog +5 V (A5V) copper planes are provided on the top side of the board and digital ground (DGND) and analog ground (AGND) copper planes are provided on the bottom side of the board. The lines R5V (+5 V) and RGND (ground) to power the relay driver IC to avoid the noise from switching of relays affecting other parts of the circuit through supply line. The width of the signal tracks are 0.381 mm and the width of supply tracks are 1.0 mm. Wherever needed, the top and bottom sides track are connected by 0.7 mm diameter PTH via. A small copper plane is placed under the power amplifier ICs to serve as its heat sink. Male Berg strip connectors are used for audio input, response key, transducers, input from sound level meter, and debugger. The minimum distance between the two signal tracks, two copper planes, and between signal tracks and copper planes is 0.381 mm. The PCB layouts of top layer, top overlay, bottom layer, and bottom overlay are given in Figure D.20. The top view and the bottom view of the assembled PCB is shown in Figure D.21.

D.13 User Interface Description

The stimuli and masker sound waveforms for audiometric tests were generated and stored in the PC memory. The descriptions of the waveforms are described in the chapter 5. The user interface for the first version and the second version was almost similar with few differences. In the first version the user interface had a button for USB connection establishment and for tuning the PC sound card attenuator level automatically. The description of the difference between the user interface of the first version and the second version is given here and the rest of the features of the user interface, that is similar to second version, are described in chapter 5.

A screenshot of the home-screen of the GUI is shown in Figure D.15. The USB connection between the audiometry module and the PC is established by pressing the "USB" button on the toolbar. For the PC to recognize the USB device, a driver provided by Microchip for USB to serial conversion (CDC RS232 Emulation Demo) has to be installed on the PC. When the "USB" button is pressed, all the available COM ports in the PC are scanned and the connection is established with the one having product identifier (PID) = 10 and vendor identifier (VID) = 1240. If a COM port with the above PID and VID is not found, a

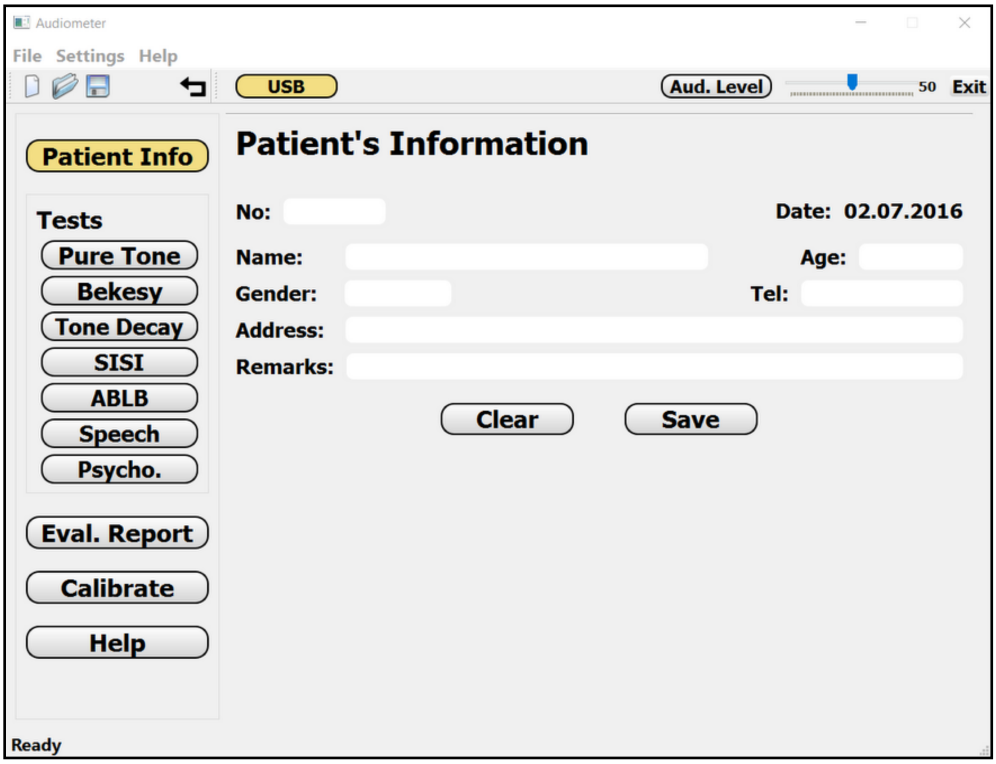


Figure D.15: A screenshot of the home-screen of the GUI.

message box saying "Error in connecting COM port" is displayed on the screen. After successfully connecting with the COM port, the line coding parameters for the serial communication are set (9600 bps baud rate, 8-data bits, and 1-stop bit, zero parity bit).

Before starting the audiometric test procedures, the PC-sound card attenuator level must be set to an appropriate level so that it is high enough for achieving the maximum output from the power amplifier in the audiometry module, but not as high to saturate them. To set the PC-sound card level, "Aud. Level" button is pressed after establishing the USB connection and connecting the line-out of the PC to the audio input of the audiometry module. When the button is pressed, the application automatically sets the PC-sound card attenuator level using feedback from the audiometry module. The feedback from the audiometry module is the ADC value corresponding to the average voltage level at the output of the power amplifier. A flowchart of the algorithm for automatically setting the PC sound card attenuator level is shown in Fig. 4.8. In this procedure, a pure-tone of 1 kHz is played and the average voltage level at one of the power amplifiers output in the audiometry module which is obtained from the level detector circuit, is converted into digital value by the microcontroller ADC and sent to the PC over USB. The procedure should be performed with the transducers disconnected at the output of audiometry module, since the voltage at the output depends on

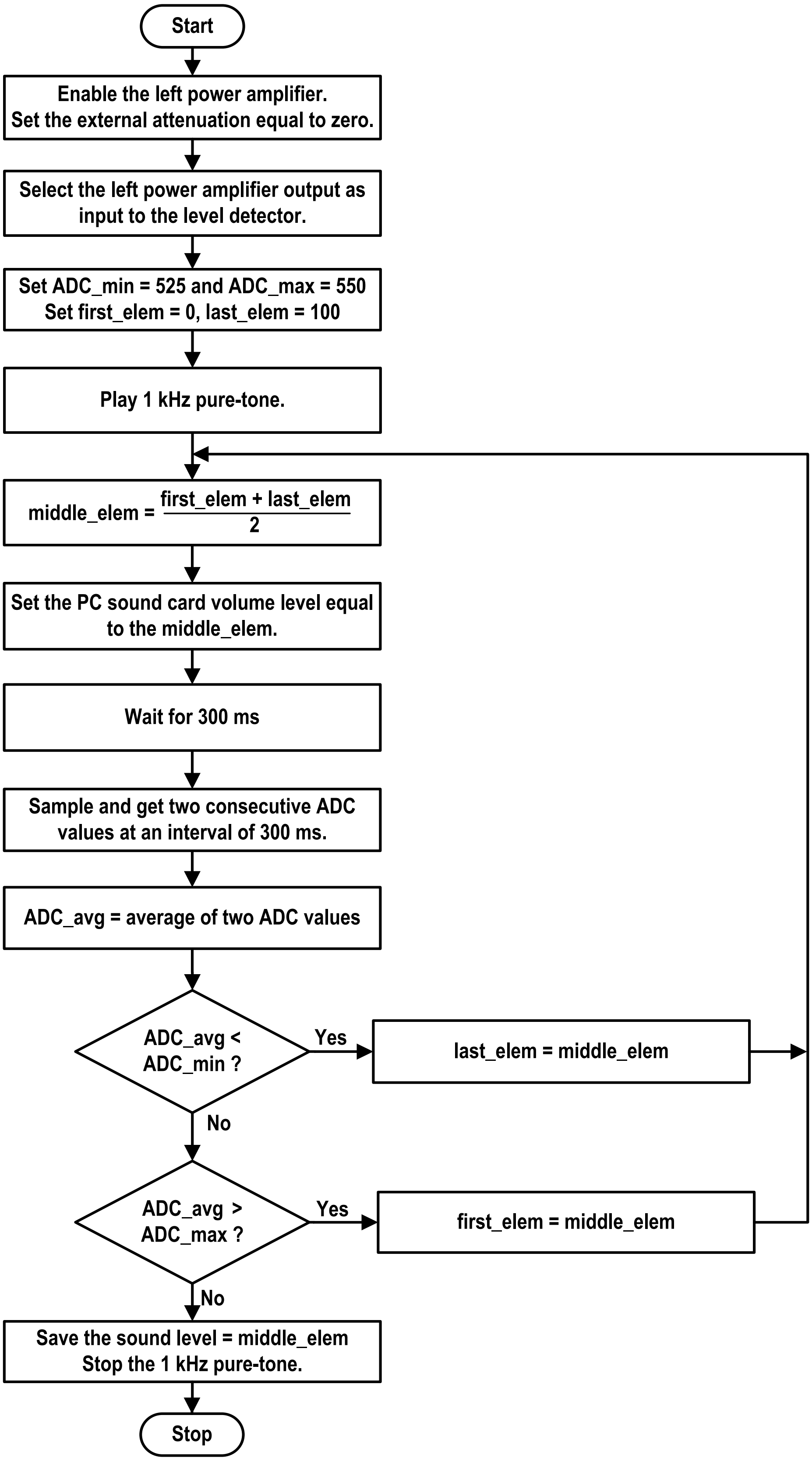


Figure D.16: Automatic procedure for calibrating the PC-sound card volume level.

the impedance of the transducers because of the 10 Ω resistance connected in the path of power amplifier output. The maximum output of the power amplifier for a sinusoidal input is 7.6 V_{pp} and it is converted by the level detector circuit which consists of a negative full-wave rectifier with a gain of 0.5 into 1.29 V dc (i.e. $\left(2.5 - \frac{V_{pp}}{2\pi}\right)$). This dc level corresponds to ADC value of 530 for the 10-bit ADC of the microcontroller in the audiometry module. The PC-sound card level is searched using binary search method until the ADC value falls within a range of ±1 dB output level tolerance, i.e. ADC_min = 525 and ADC_max = 550. When the PC-sound card level is changed, the dc signal at the output of the level detector is sampled after a delay of 300 ms to ensure settling of the audio signal and detector output. At each PC-sound card attenuator level, average of two consecutive ADC values are taken at an interval of 300 ms to average out any noise. The procedure takes up to 4 s (i.e. $600 \times \log_2 100$ ms) to reach the final value. The PC-sound card attenuator level should not be changed manually during the audiometry session once it has been set.

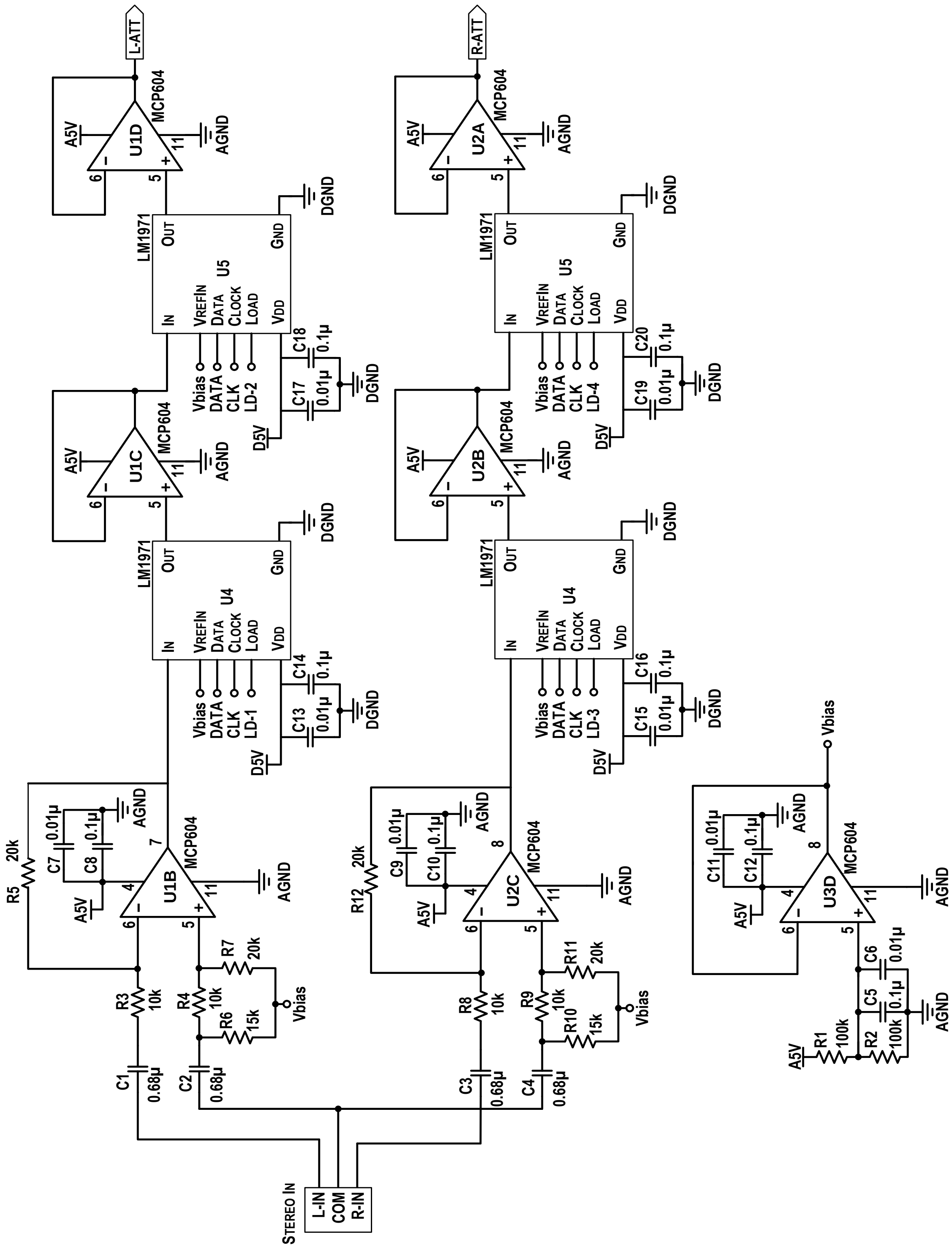


Figure D.17: Schematic sheet-1: The pre-amplifier and the attenuator circuit of left and right channels.

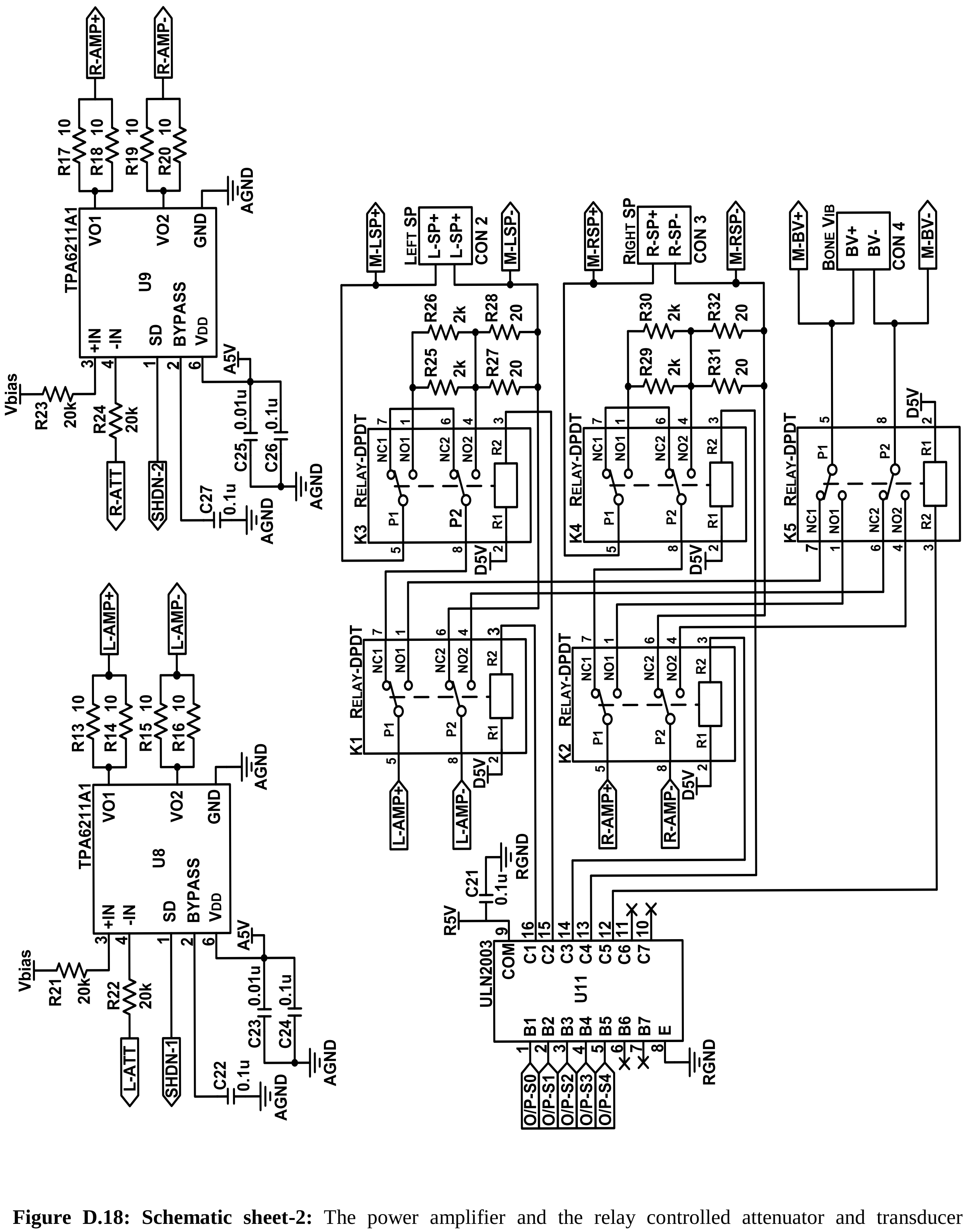


Figure D.18: Schematic sheet-2: The power amplifier and the relay controlled attenuator and transducer selection circuit for left and right channels.

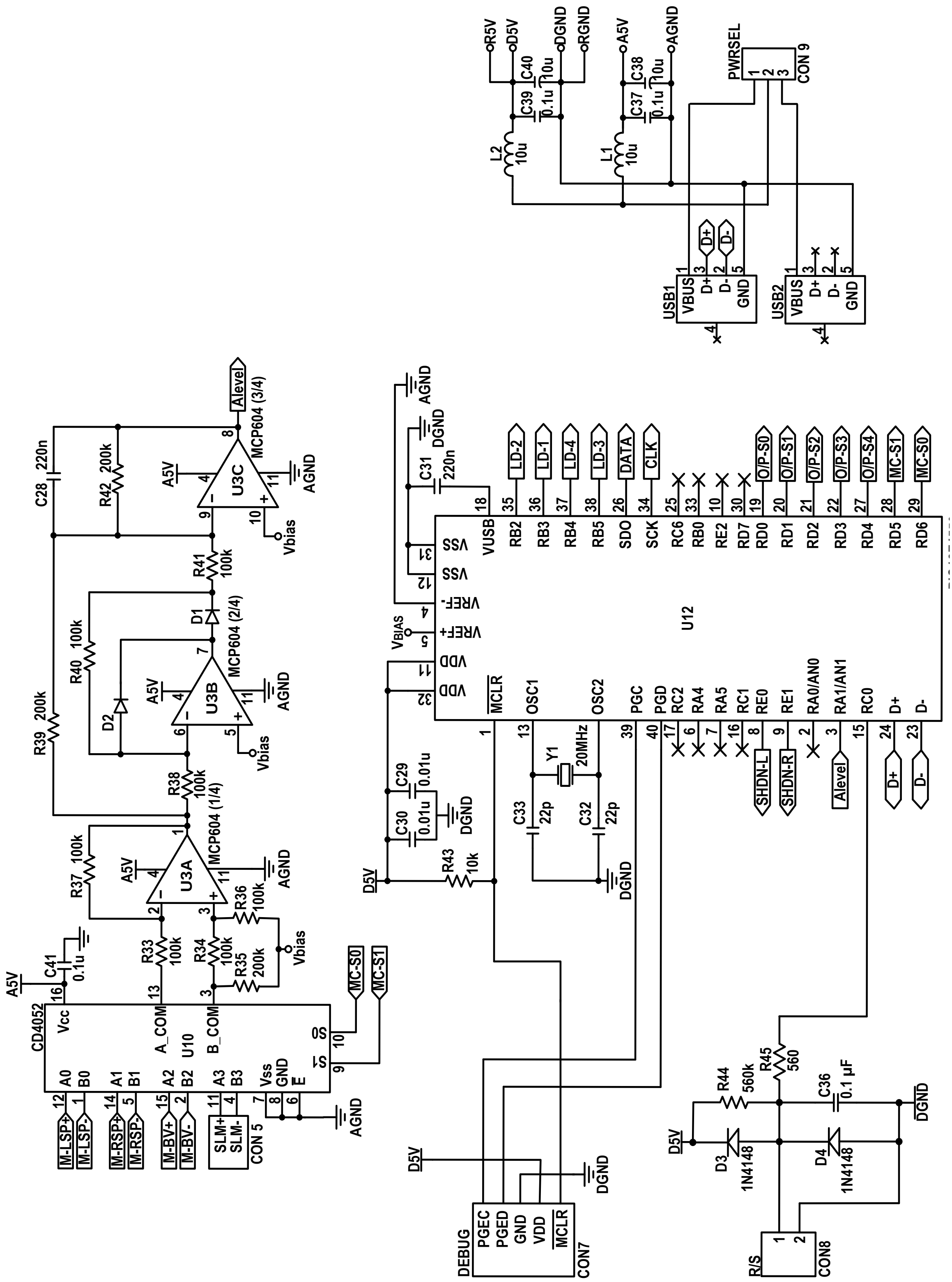


Figure D.19: Schematic sheet-3: The output level monitoring circuit, microcontroller circuit, response button circuit, and power circuit.

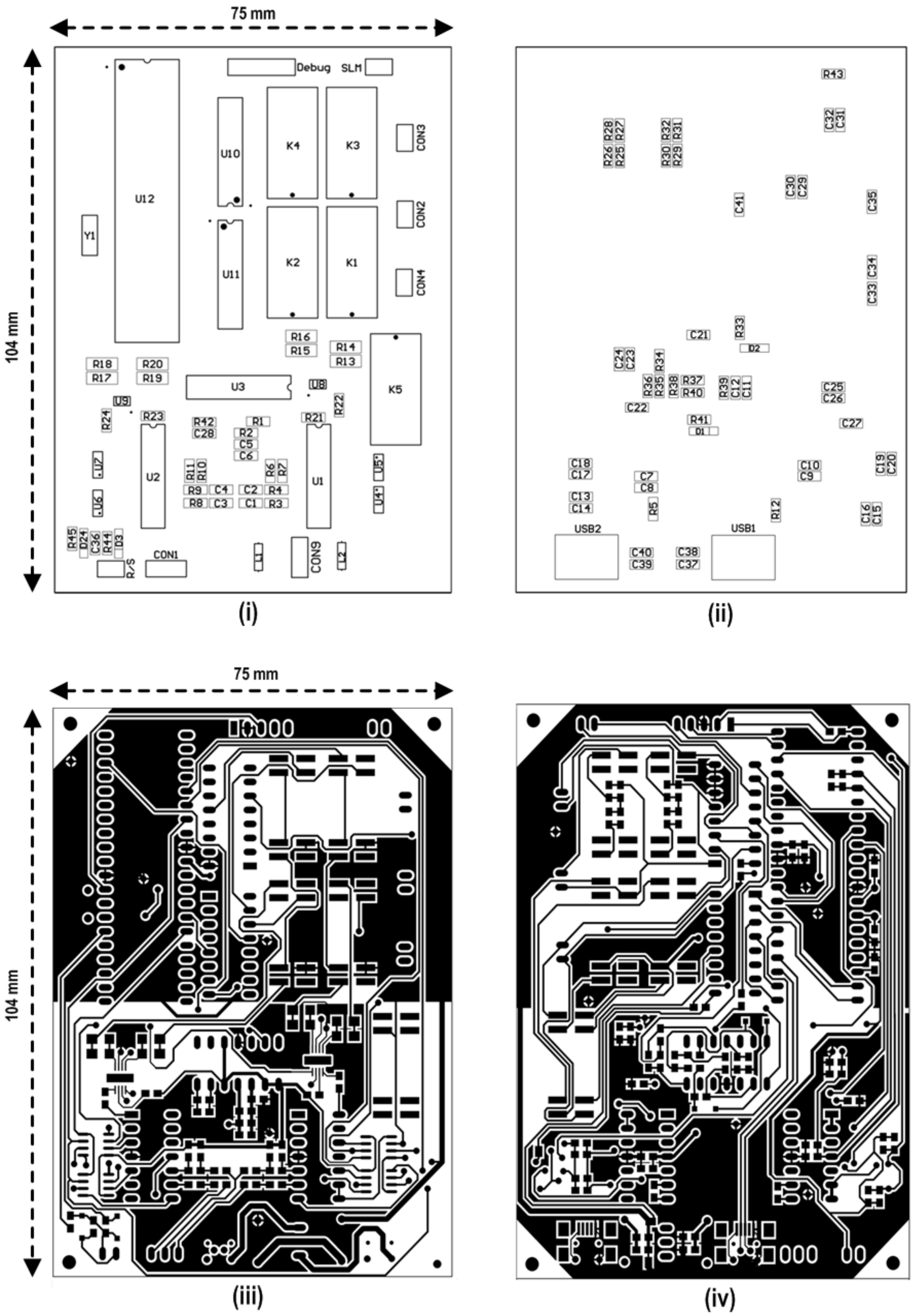
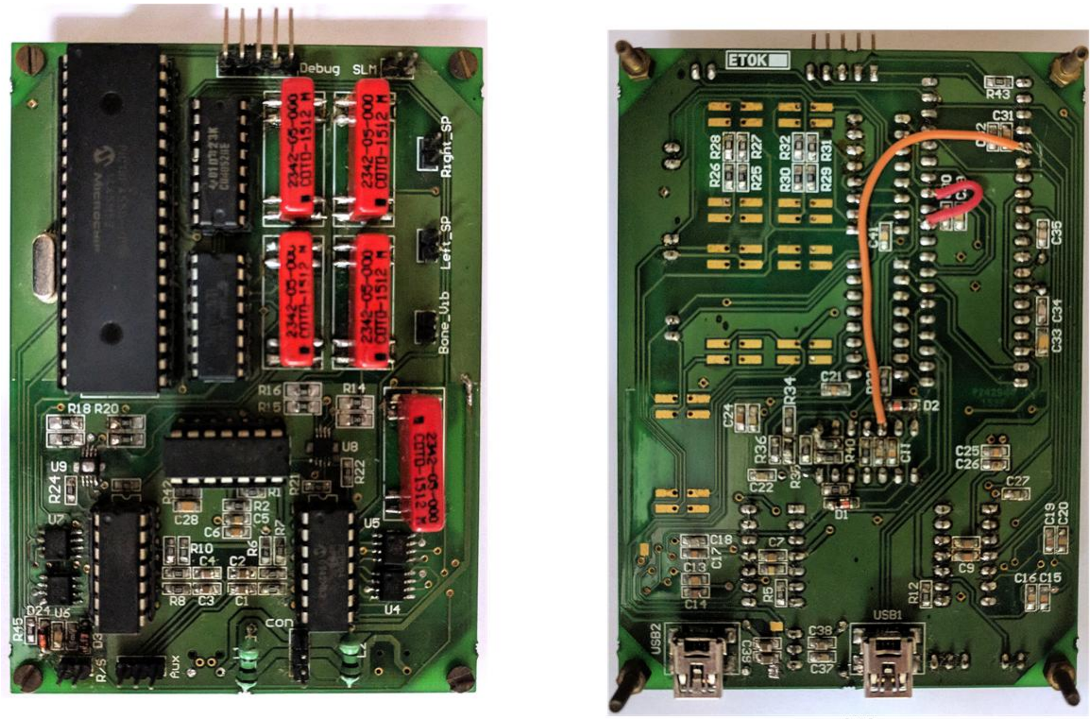


Figure D.20: PCB layouts: (i) Top overlay, (ii) bottom overlay, (iii) top layer, and (iv) bottom layer of the PCB design.



(i) (ii)

Figure D.21: (i) Top view, and (ii) bottom view of the assembled PCB of the first version of the audiometry module.

Appendix E
BALANCED CURRENT SOURCE

E.1 An op amp based balanced current source

The force on the diaphragm of a speaker is directly proportional to the current in the speaker coil and the output sound pressure from a speaker using a current drive is expected to have less frequency dependence. Hence a current drive is expected to be superior to voltage drive. A novel voltage-controlled balanced current source design is proposed for driving the transducers. It is based on the improved Howland current source. As shown in Figure E.1, it uses an op amp with balanced differential outputs.

For an ideal current source, the current is independent of the load resistance. The values of α, β, and γ are selected for the current in the load i_L to be independent of the load resistance and for the high output impedance. The nodal equations of the circuit are given below.

$$V_- = \frac{\alpha R \parallel \beta R}{\alpha R \parallel \beta R + R} V_2 + \frac{R \parallel \beta R}{R \parallel \beta R + \alpha R} V_{o1} + \frac{R \parallel \alpha R}{R \parallel \alpha R + \beta R} V_{L2} \tag{E.1}$$

$$V_+ = \frac{\alpha R \parallel \beta R}{\alpha R \parallel \beta R + R} V_1 + \frac{R \parallel \beta R}{R \parallel \beta R + \alpha R} V_{o2} + \frac{R \parallel \alpha R}{R \parallel \alpha R + \beta R} V_{L2} \tag{E.2}$$

For linear range of operation, $V_- = V_+$, and for symmetrical differential output, $V_{o1} = -V_{o2}$

$$V_{o1} = -V_{o2} = \frac{\alpha}{2}(V_{L1} - V_{L2}) + \frac{\alpha\beta}{2}(V_1 - V_2) \tag{E.3}$$

$$i_L R_L = V_{L1} - V_{L2} \tag{E.4}$$

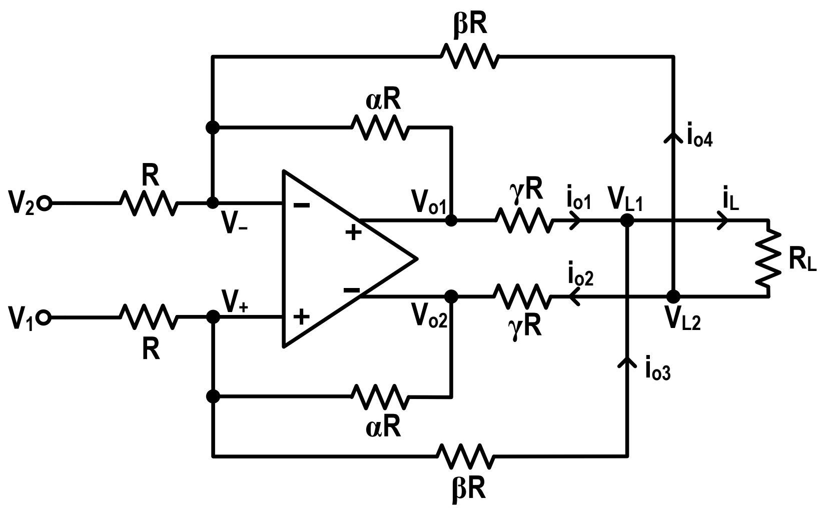


Figure E.1: An op amp based balanced current source circuit.

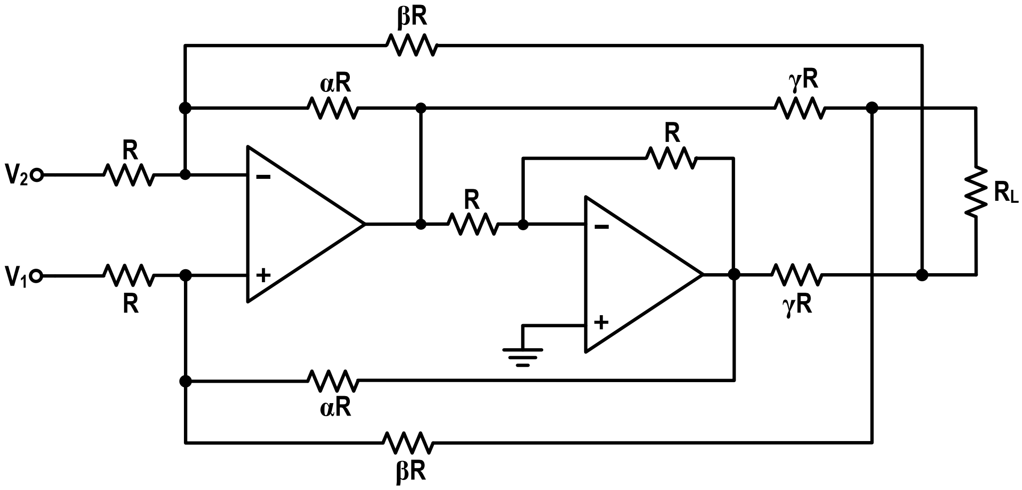


Figure E.2: Balanced current drive circuit using two single-ended output op amps.

Since $i_L = i_{o1} + i_{o3}$, we get

$$i_L = \frac{1}{\gamma R}(V_{o1} - V_{L1}) + \frac{1}{\beta R}(V_+ - V_{L1}) \tag{E.4}$$

From Equation (E.4), (E.3) and for $\alpha = \beta + \gamma$, we get

$$i_L = \frac{\alpha}{2\gamma R}(V_1 - V_2) \tag{E.5}$$

The current through the load i_L is independent of the load resistance if the condition $\alpha = \beta + \gamma$ is satisfied. Therefore, the circuit requires tight matching of the resistors for it to be a current source with high output impedance.

The circuit can be realized using two single-ended output op amps, as shown Figure E.2.

The current drive has less electrical damping due to large output impedance of the current drive. Its effect on loudspeaker response needs to be examined.

E.2 A balanced current source using an audio amplifier

A 3.1-W amplifier with differential input and bridge-tied load (BTL) output was used for implementing the balanced current drive circuit. The IC chosen was TPA6211A1 [24] (from Texas Instruments). It is a single supply IC with internal reference voltage set as Vdd/2 and with a 40 kΩ on-chip resistance in the feedback paths. The IC is a high-gain differential input power amplifier with balanced differential output. The circuit as shown in Figure E.3 is a modification of the circuit in Figure E.1. The resistances R1, R2, R5, and R6 are selected to

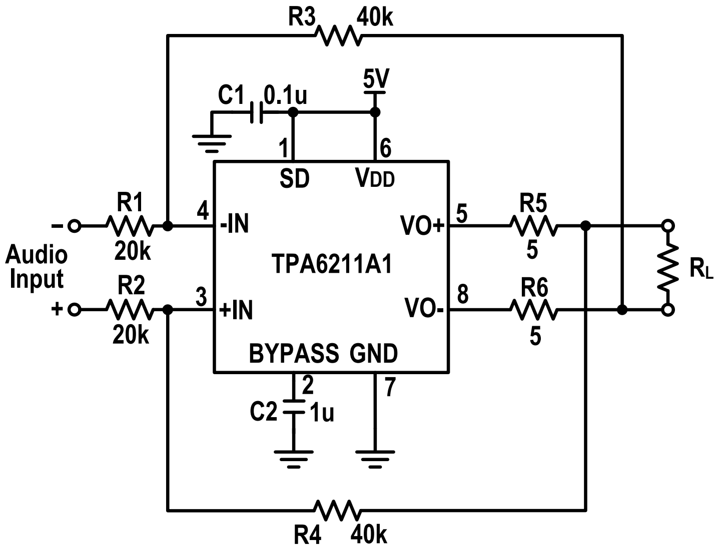


Figure E.3: A balanced current source using a audio amplifier.

keep the current gain equal to 0.2. The R5, R6 resistances is kept low to achieve large output voltage swing across the load R_L. The circuit was implemented and tested using audio signals and was found to be working satisfactorily. Its superiority over voltage drive needs to be examined.

Page intentionally left blank

Appendix F

COMPONENT LIST

Designator	Part no./value	Component description	Quantity
U1, U2, U10	MCP604	Op amp (SOIC)	3
U3, U4, U5, U6	LM1971	Attenuator (SOIC)	4
U7, U8	TPA6211A1	Power Amplifier (MSOP)	2
U9	CD4052	Dual ch. 4-to-1 Mux (SOIC)	1
U11	PIC18F46J50	Microcontroller (TQFP)	1
B1	RN52	Bluetooth transceiver (SMD)	1
A1, A2, A3, A4, A5	MAX4993	Analog audio switch (UTQFN)	5
Q3, Q5	IRLML6401	SOT-23	2
Q1, Q2, Q4, Q6	IRLML2402	SOT-23	4
CH1, CH2	BQ21040	SOT-23	2
VR1, VR2	MCP1802T	SOT-23	2
VR3, VR4, VR5	LM1117-5.0	SOT-223	3
D4, D5	DFLD130L	Diode (SMD)	2
D6, D9, D10, D11, D12	1N4148	Diode (SMD)	5
R42, R43, R44, R45, R48, R49, R50, R51	10 Ω, ½ W	Resistor (SMD)	8
R53, R54, R56, R57	20 Ω, ⅛ W	Resistor (SMD)	4
RL	560 Ω, ⅛ W	Resistor (SMD)	1
R3, R9, R13, R21, R52, R55, R69	1 kΩ, ⅛ W	Resistor (SMD)	7
R6, R10	1.5 kΩ, ⅛ W	Resistor (SMD)	2
R24, R25	2.2 kΩ, ⅛ W	Resistor (SMD)	2

R7, R11	2.7 kΩ, ⅛ W	Resistor (SMD)	2
R1, R4, R8, R12, R14, R26, R27, R31, R32, R72	10 kΩ, ⅛ W	Resistor (SMD)	10
R28, R33	15 kΩ, ⅛ W	Resistor (SMD)	2
R5, R18, R40, R41, R46, R47	20 kΩ, ⅛ W	Resistor (SMD)	6
R17, R19, R22, R36, R37, R38, R39, R58, R59, R64, R65, R66, R70, R71	100 kΩ, ⅛ W	Resistor (SMD)	14
R2, R15, R23, R63, R67	200 kΩ, ⅛ W	Resistor (SMD)	5
R68	560 kΩ, ⅛ W	Resistor (SMD)	1
C18, C20, C22, C24, C29, C32, C37	0.01 µF	Capacitor (Ceramic, SMD)	7
C19, C21, C23, C25, C26, C27 C30, C33, C34, C36, C38, C39, CA1, CA3, CA5, C40, C43, C45	0.1 µF	Capacitor (Ceramic, SMD)	18
C35	220 nF	Capacitor (Ceramic, SMD)	1
C5, C6, C15, C16	470 nF	Capacitor (Ceramic, SMD)	4
C1, C2, C3, C5, C7, C8, C9, C11, C13, C28, C31	1 uF		11
C4, C6, C10, C12, C14, C17, C41	10 µF	Capacitor (Ceramic, SMD)	7
uUSB		Micro-USB connector	1
Left_HP, Right_HP, BV, SLM, M1, Resp_S/W	Berg connector & mono audio connector	2-pin berg connector for the four layer PCB and mono audio connector for the two layer PCB	6
DEBUG	Berg connector	5-pin through hole	1
B1, B2	Berg connector	2-pin through hole	2
CHG, D8, L1	LED	Red LED (SMD)	2
BAT1, BAT2	LED	Yellow LED (SMD)	2
PWR	LED	Blue LED (SMD)	1

Appendix G

SCHEMATIC DIAGRAMS AND PCB LAYOUTS

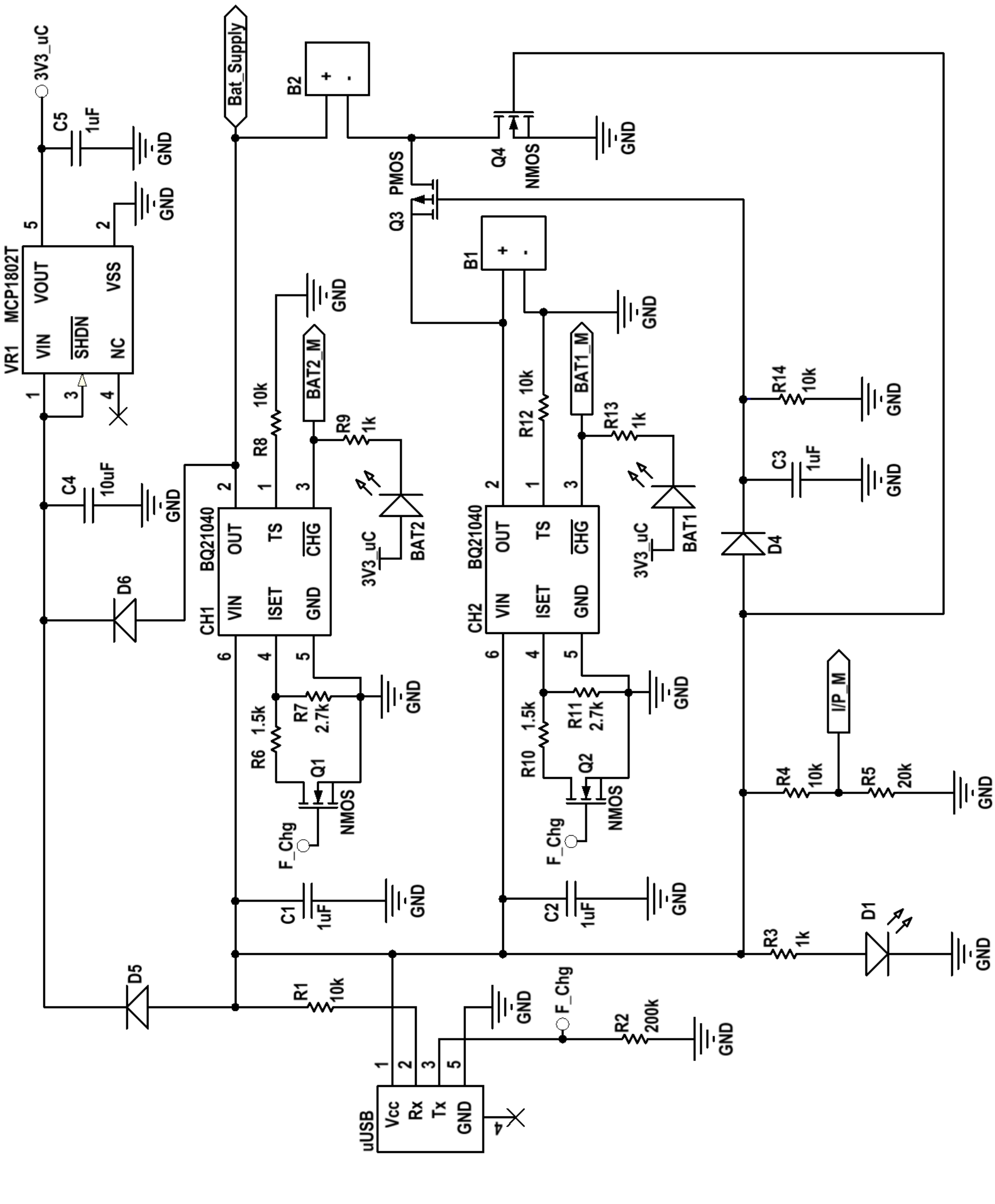


Figure G.1: Schematic sheet-1: The battery charging and configuring circuit.

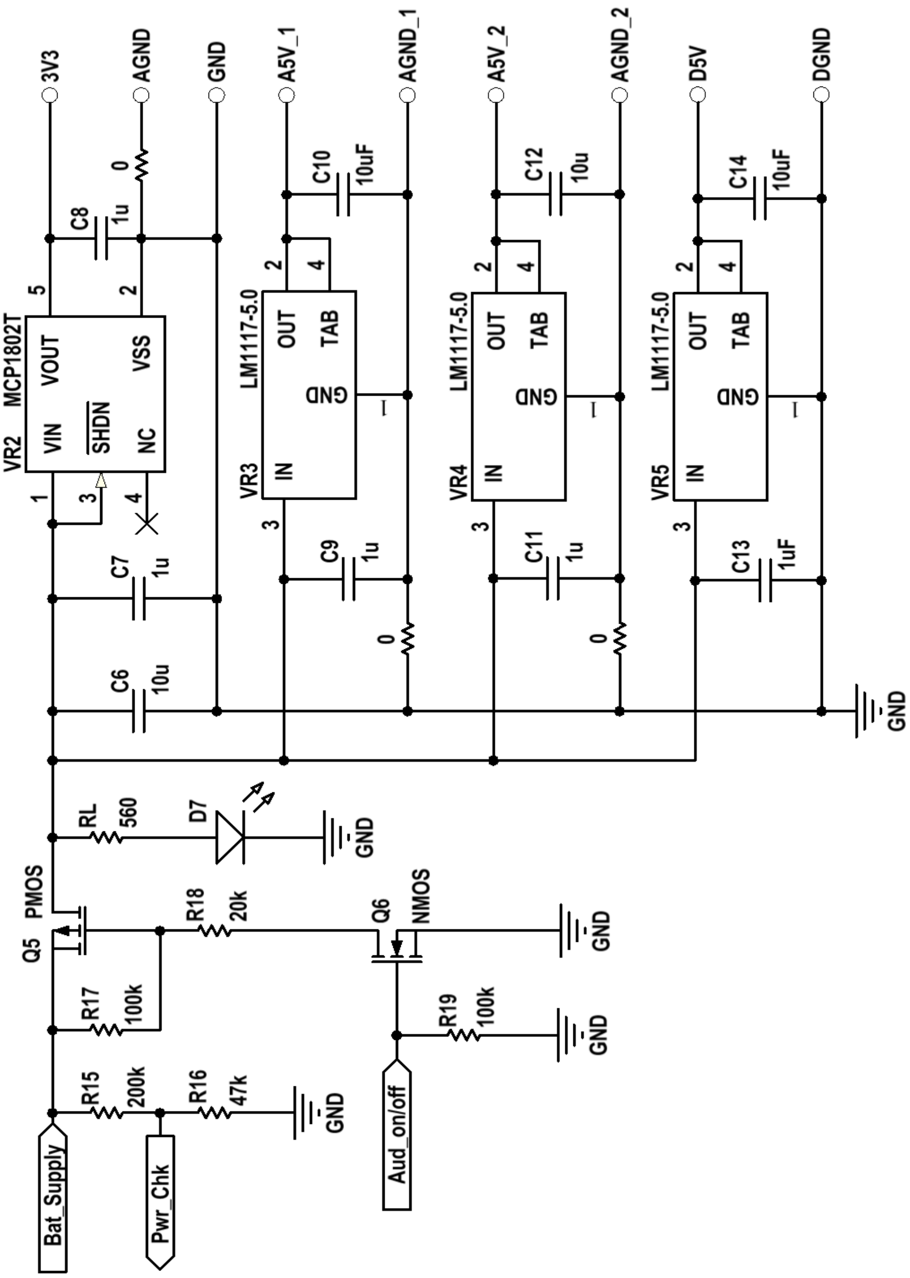


Figure G.2: Schematic sheet-2: The powering circuit.

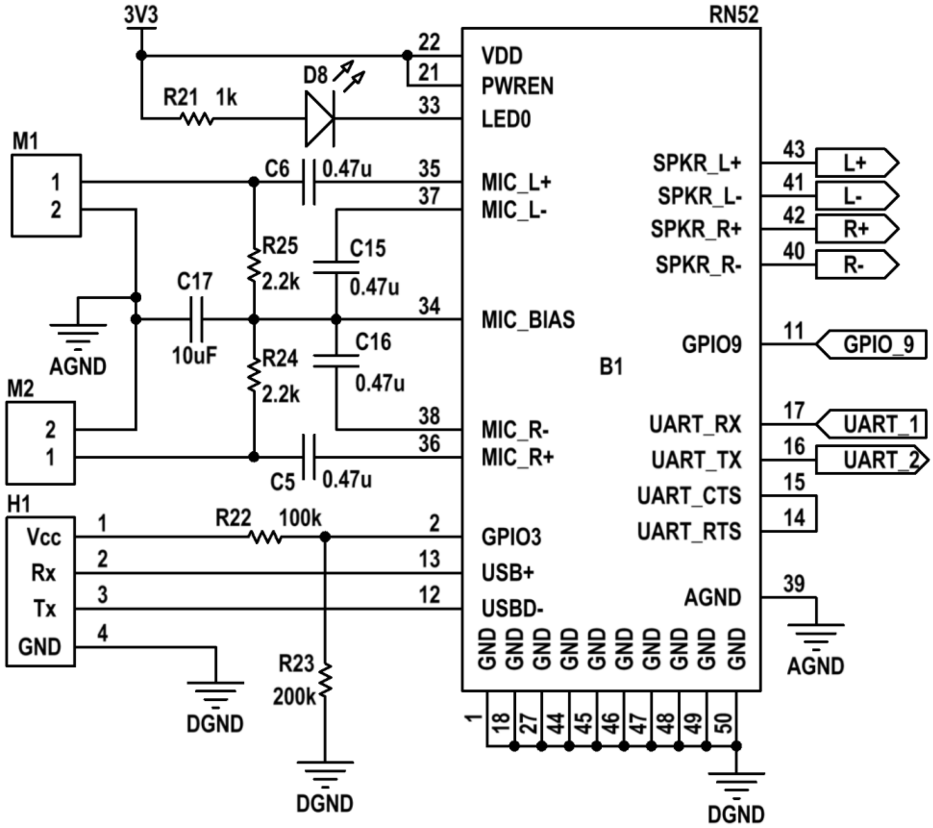


Figure G.3: Schematic sheet-3: The Bluetooth transceiver circuit.

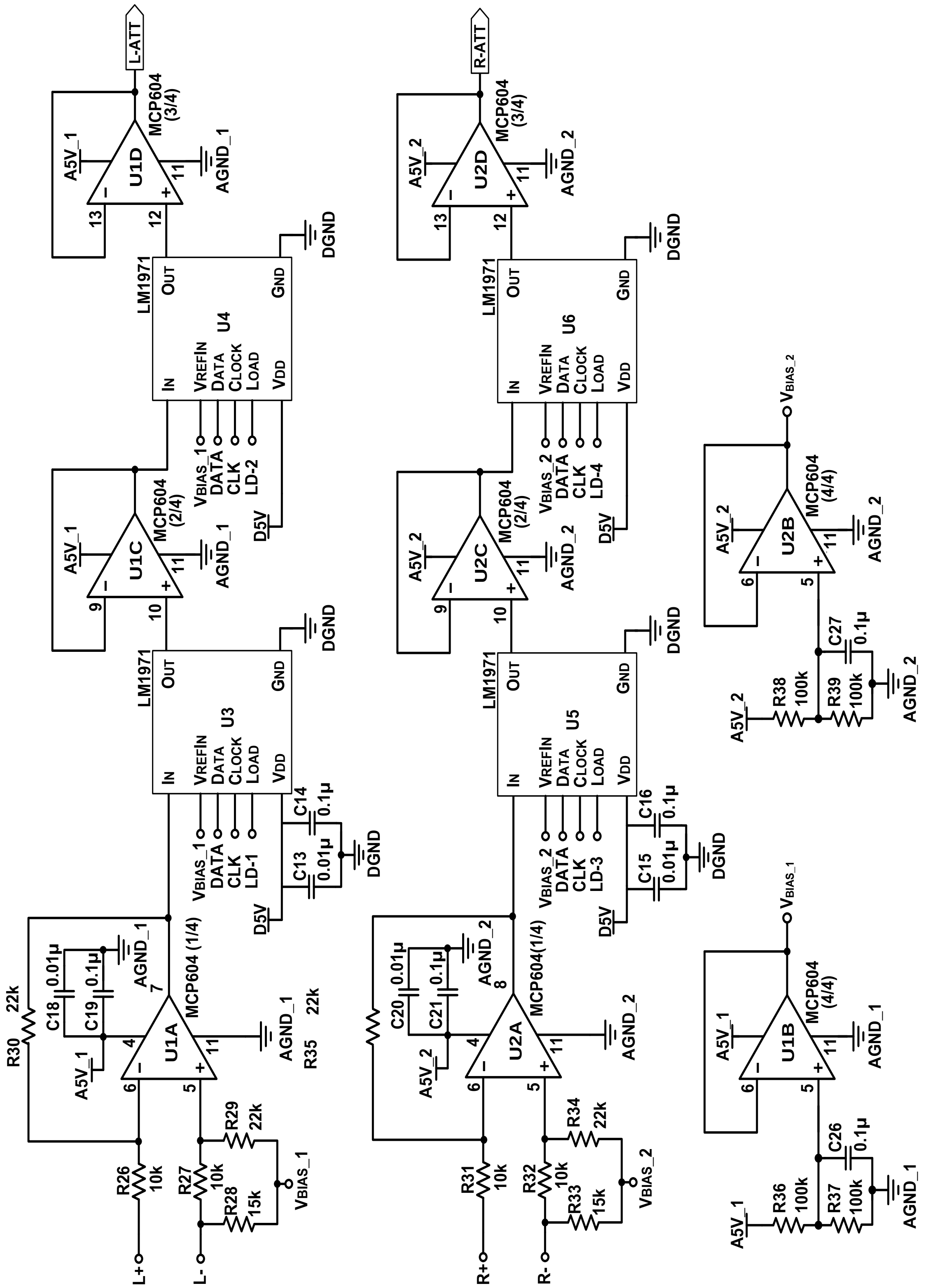


Figure G.4: Schematic sheet-4: The pre-amplifier and the attenuator circuit of left and right channels.

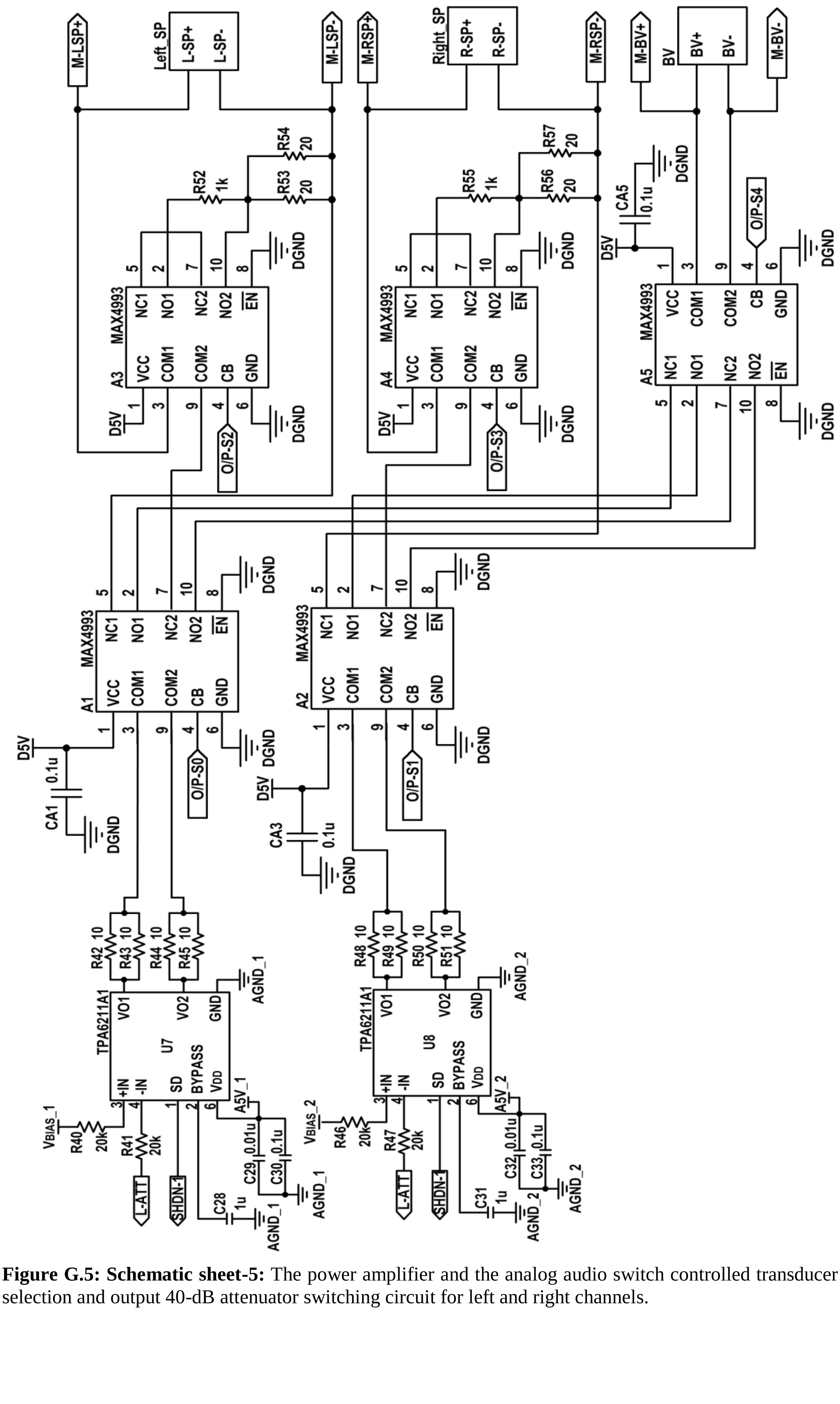


Figure G.5: Schematic sheet-5: The power amplifier and the analog audio switch controlled transducer selection and output 40-dB attenuator switching circuit for left and right channels.

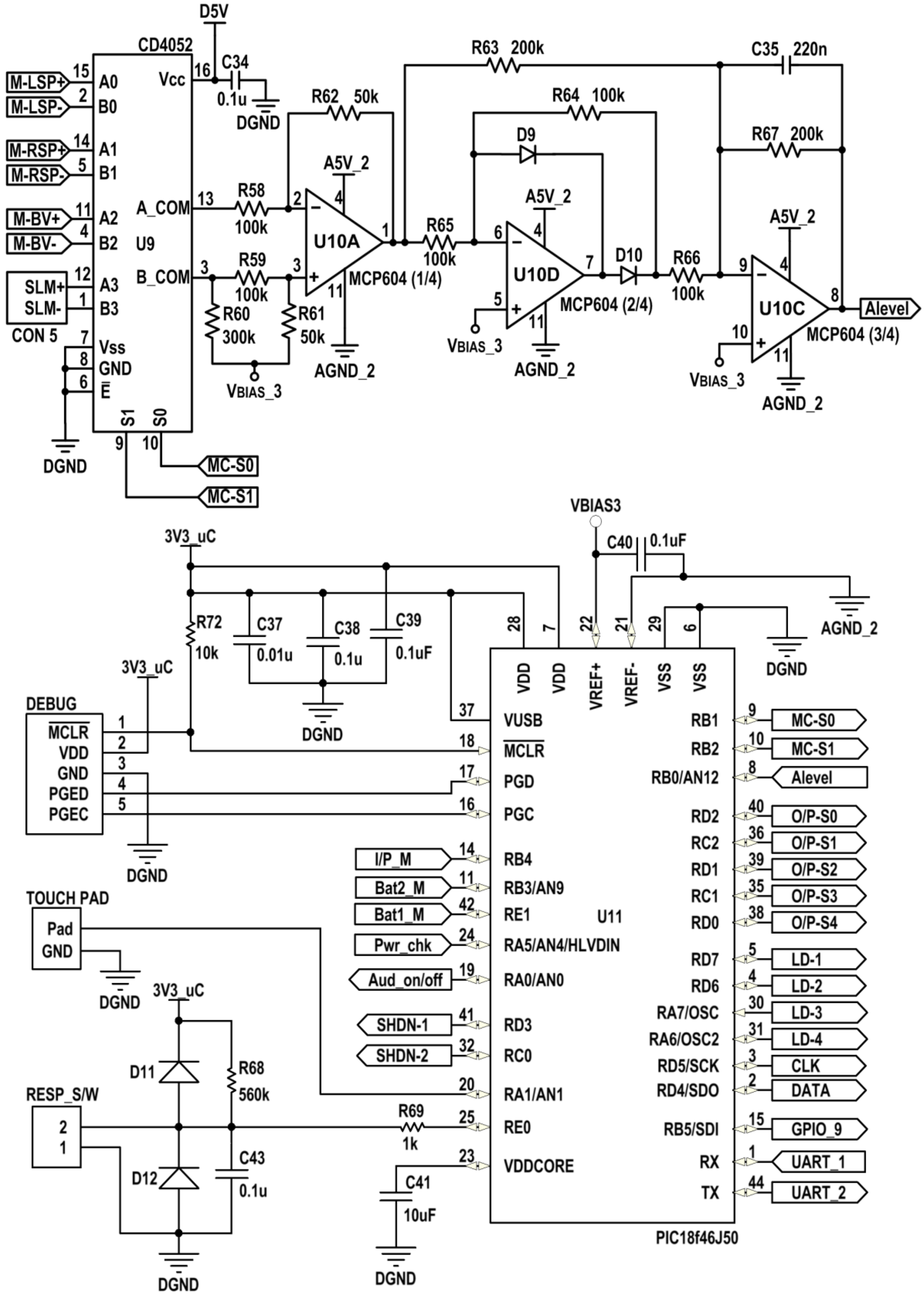


Figure G.6: Schematic sheet-6: The level monitoring, microcontroller, and response button circuit.

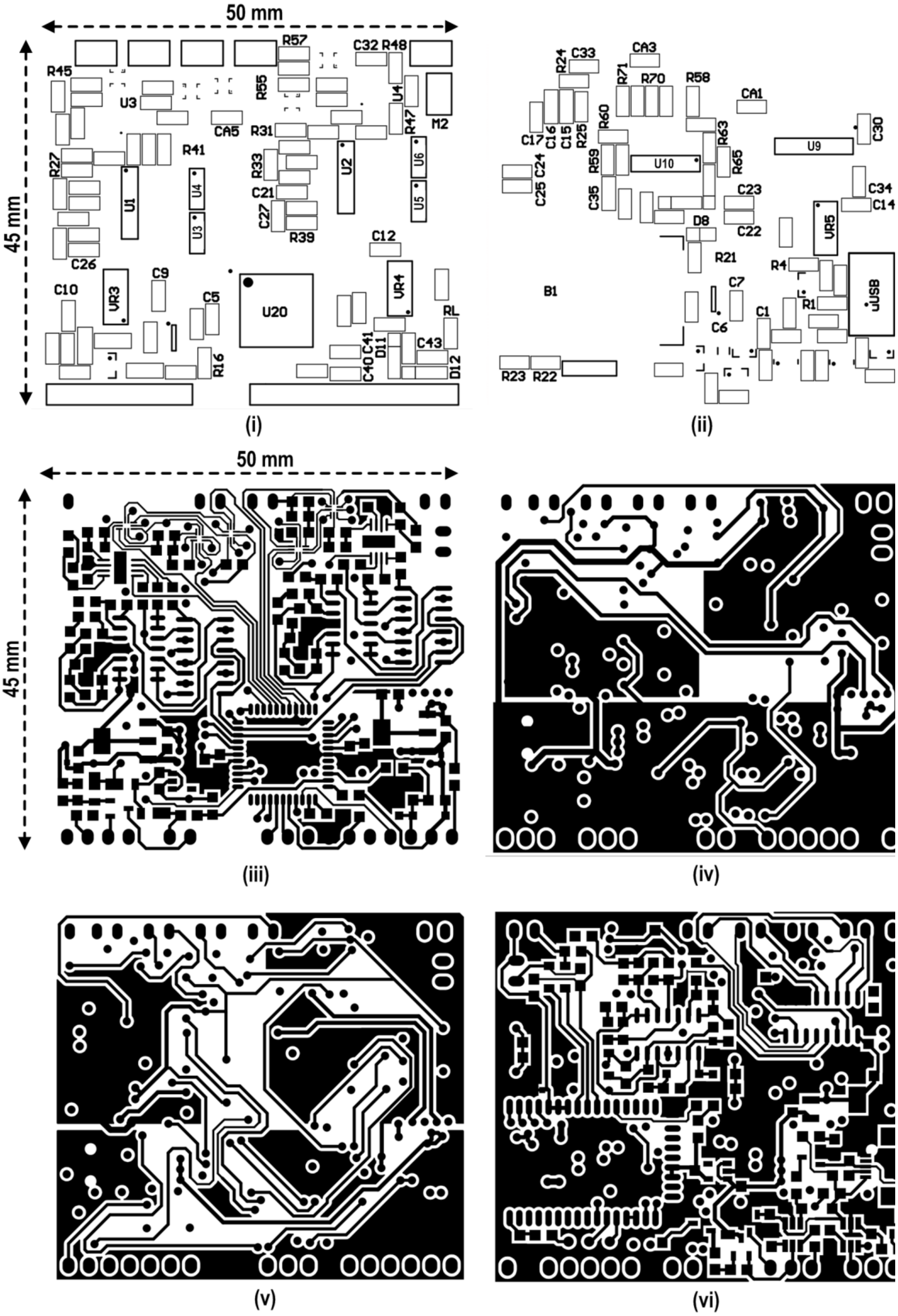


Figure G.7: PCB layouts: (i) Top overlay, (ii) bottom overlay, (iii) top layer, (iv) second layer, (v) third layer, and (vi) bottom layer of the 4-layer PCB design.

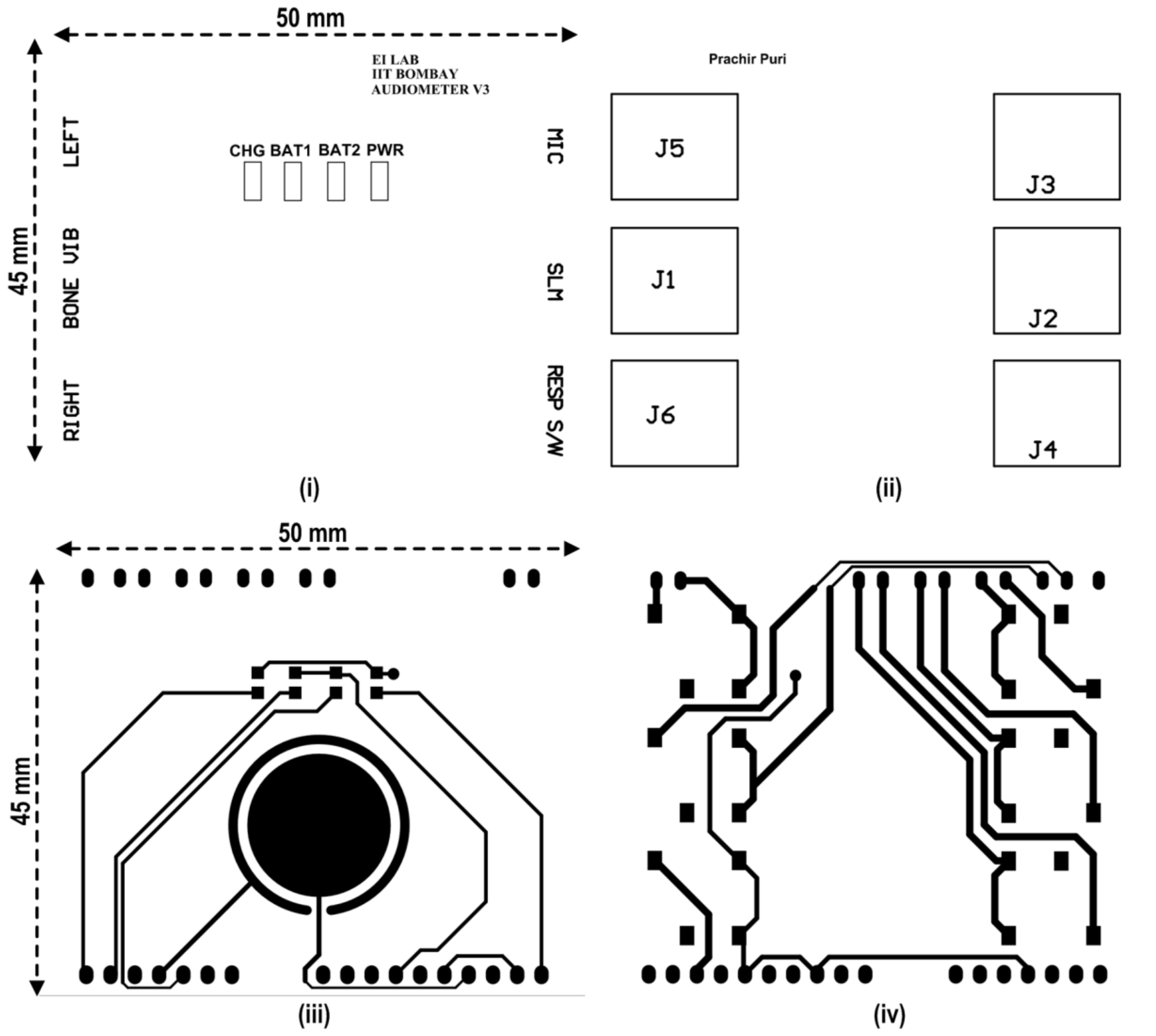


Figure G.8: PCB layouts: (i) Top overlay, (ii) bottom overlay, (iii) top layer, and (iv) bottom layers of the 2-layer PCB design.

Appendix H

IMAGES OF THE PROTOTYPE

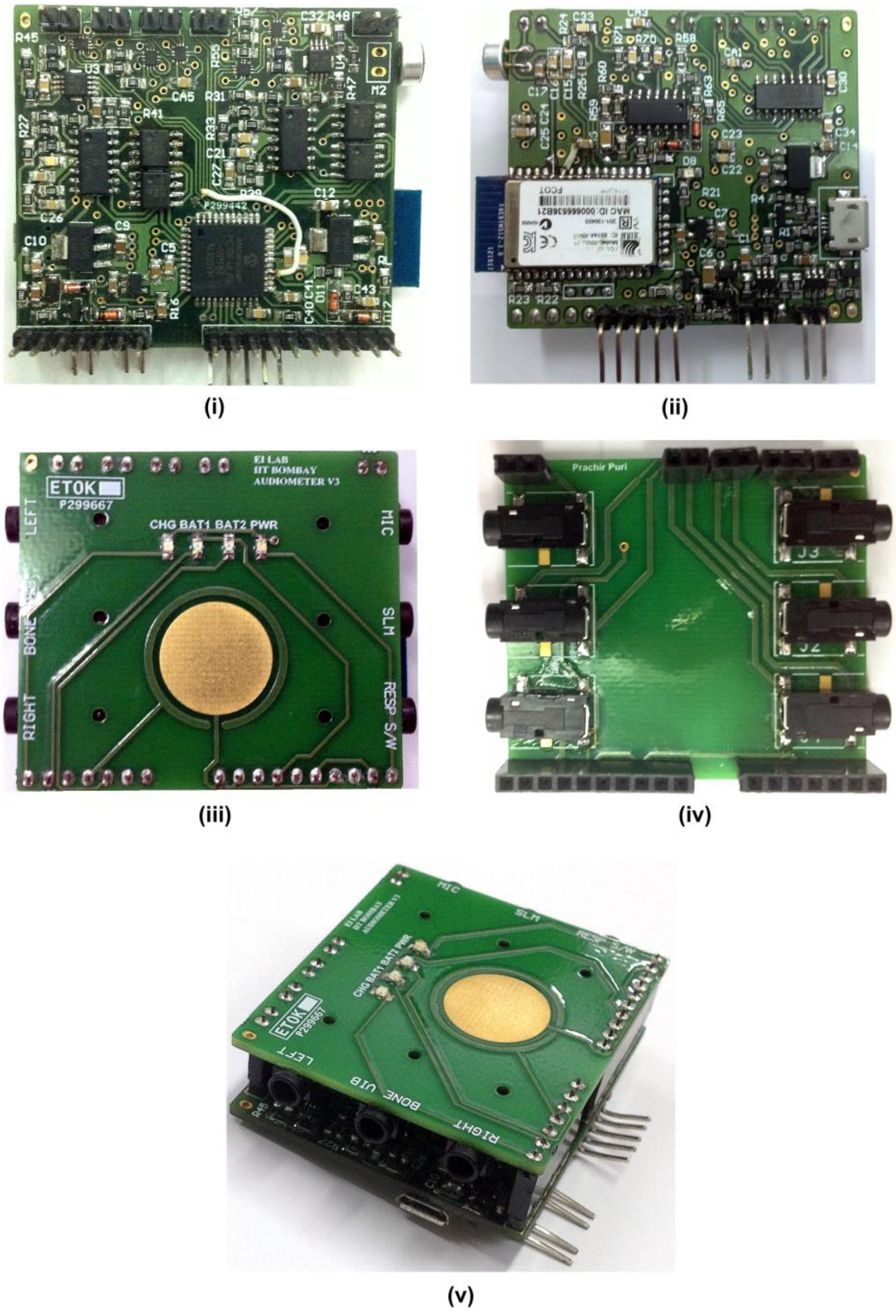


Figure H.1: (i) Top view of the 4-layer assembled PCB, (ii) bottom view of the 4-layer assembled PCB, (iii) top view of the 2-layer assembled PCB, (iv) bottom view of the 2-layer assembled PCB, and (v) isometric view of the audiometry module after stacking the 4-layer and the 2-layer PCBs.

Page intentionally left blank

References

[1] F. H. Bess and L. E. Humes, *Audiology: The Fundamentals*, 4th ed., Philadelphia: Wolters Kluwer, 2008.

[2] S. A. Gelfand, *Essentials of Audiology*, 3rd ed., New York: Thieme, 2009.

[3] A. Biswas, *Clinical Audio-Vestibularmetry for Otologist and Neurologist*, 2nd ed., Mumbai: Bhalani, 1995.

[4] R. Giannini, S.Tankovic, and V. Kozina, "A DSP based method for audio stimuli generation," in *Proc. 19th Int. Conf. IEEE, Eng. Med. Biol. Soc. (EMBS 1997), vol. 3*, Chicago, 1997, pp. 1163 – 1166.

[5] H. C. Wu, L. P. Yang, C. H. Chen, S. T. Young, and T. S. Kuo, "A low cost architecture of audiometer based on direct digital synthesizer," in *Proc. 1st Joint Conf. of Biomed. Eng. Soc. (BMES) and Eng. Med. Biol. Soc. (EMBS), vol. 2*, Atlanta, Georgia, 1999, pp. 894.

[6] D. M. Davis, "Portable audiometer enclosed within a patient response mechanism housing," US Patent 7370533 B2, May 13, 2008.

[7] J. Akita, "Audiometer," US Patent 20030065276 A1, April 3, 2003.

[8] L. Braun and J. Foreman, "Multimedia feature for diagnostic instrumentation," US Patent 6644120 B1, Nov. 11, 2003.

[9] M. L. Dille and R. M. Ellingson, "Dynamic range considerations when designing PC sound card based audiometric systems to test human hearing," in *Proc. 32nd Ann. Int. Conf. IEEE, Eng. Med. Biol. Soc. (EMBS 2010)*, Buenos Aires, Argentina, 2010, pp. 5907 – 5910.

[10] F. H. Silverman, *Speech, Language, and Hearing Disorders,* Boston: Allyn and Bacon, 1995.

[11] H. Levitt, J. M. Pickett, and R. A. Houde, *Sensory Aids for Hearing Impairment,* New york: IEEE Press, 1980.

[12] *American National Standard Specification for Audiometers*, ANSI S3.6, 1996.

[13] B. C. J. Moore, *An Introduction to Psychology of Hearing*, 6th ed., London: Academia, 1997.

[14] *Indian Standard Specification for Diagnostic Audiometers*, IS 10565:1999, 1999.

[15] D. M. Patel, "A microcontroller based audiometer," M.Tech. dissertation, Dept. Elect. Eng., IIT Bombay, Mumbai, 2002.

[16] S. F. Mossa, "A microcontroller based audiometer," M.Tech. dissertation, Dept. Elect.

Eng., IIT Bombay, Mumbai, 2010.

[17] K. Thirupathi, "A microcontroller based audiometer," M.Tech. dissertation, Dept. Elect. Eng., IIT Bombay, Mumbai, 2011.

[18] Roving Networks, "RN52 Bluetooth Audiom Module," Available online:ww1.microchip. com/downloads/en/DeviceDoc/rn-52-ds-1.0r.pdf, Acessed: May 1, 2017.

[19] Microchip, "MCP601/1R/2/3/4 2.7 V – 6.0 V, single supply CMOS op amps," Available online: ww1.microchip.com/downloads/en/DeviceDoc/21314g.pdf, Accessed: Sept. 30, 2016.

[20] Texas Instruments, "PGA2311 stereo audio volume control," Available online: www.ti. com/lit/ds/symlink/pga2311.pdf, Accessed: Sept. 30, 2016.

[21] Analog Devices, "AD7111 LC2 MOS LOGDAC logarithmic D/A converter," Available online: www.analog.com/media/en/technical-documentation/datasheets/AD7111_7111A. pdf, Accessed: Sept. 30, 2016.

[22] Texas Instruments, "LM1971 overture audio attenuator series digitally controlled 62 dB audio attenuator with mute," Available online: www.ti.com/lit/ds/snas104b/snas104b.pdf, Accessed: Sept.30, 2016.

[23] Texas Instruments, "3.1-W mono fully differential audio power amplifier," Available online: www.ti.com/lit/ds/symlink/tpa6211a1.pdf, Accessed: Sept. 30, 2016.

[24] Maxim Technologies, " Low RON, Dual-SPDT/Single-DPDT Analog Switches with Slow Turn-On Time," Available online: https://datasheets.maximintegrated.com/en/ds/MAX 4991-MAX4994.pdf, Acessed: May 1, 2017.

[25] Texas Instruments, “CD405xB CMOS single 8-channel analog multiplexer/demultiplexer with logic-level conversion," Available online: www.ti.com/lit/ds/symlink/cd4051b.pdf, Accessed: Sept. 30, 2016.

[26] Microchip, "PIC18F46J50 Family Datasheet" Available online: ww1.microchip.com /downloads/en/DeviceDoc/39931b.pdf, Acessed: May 1, 2017.

[27] Texas Instruments, “bq21040 0.8-A, Single-Input, Single Cell Li-Ion and Li-Pol Battery charger," Available online: www.ti.com/lit/ds/symlink/bq21040.pdf, Accessed: May 1, 2017.

[28] Microchip, "MCP1802 300 mA, High PSRR, Low Quiescent Current LDO," Available online: ww1.microchip.com/downloads/en/DeviceDoc/22053b.pdf, Acessed: May 1, 2017.

[29] Diodes Incorporated, "DFLS130L 1.0 A Surface Mount Schottky Barrier Rectifier,"

Available online: www.diodes.com/assets/Datasheets/ds30492.pdf, Acessed: May 1, 2017.

[30] International Rectifier, "IRLML2402 HEXFET Power MOSFET, " Available online: www.farnell.com/datasheets/13202.pdf, Accessed: May 1, 2017.

[31] International Rectifier, “IRLML6401PbF HEXFET Power MOSFET," Available online: www.infineon.com/dgdl/irlml6401.pdf, Accessed: May 1, 2017.

[32] Texas Instruments, "LM1117 LM1117 800-mA Low-Dropout Linear Regulator," Available online: www.ti.com/lit/ds/symlink/lm1117.pdf, Accessed: May 1, 2017.

[33] M. Unoki, K. Ito, Y. Ishimoto, C. T. Tan, "Estimate of auditory filter shape using notched-noise masking for various signal frequencies," *Acoust. Sci & Tech*, vol. 27, no. 1, pp. 1-11, Jan 2006.

[34] American Speech LanguageHearing Association, *Guidelines for audiometric symbols*, vol. 2, pp. 25 – 30, 1990.

[35] British Society of Audiology, "Recommended procedure pure-tone air-conduction and bone-conduction threshold audiometry with and without masking," Available online: www.thebsa.org.uk/wp-content/uploads/2014/04/BSA_RP_PTA_FINAL_24Sept11_MinorAmend06Feb12.pdf, Accessed: Sept. 30, 2016.

[36] AD226, "Service manual of diagnostic audiometer AD226," Available online: www.moxa.ru/forum/uploads/monthly_06_2010/post-1013-1275916634.ipb, Accessed: Sept. 30, 2016.

[37] Sibelmed, "Audiometer Sibelmed 400," Available online: www.zonemedical.com.au/c.728341/site/PDF/Sibelsound/Sibelsound-400UserManual.pdf, Accessed: Sept. 30, 2016.

[38] Inventis, "Piano clinical audiometer," Available online: www.inventis.it/sites/default/files/products/tech-files/technical-sheet-PIANO.pdf, Accessed: Sept. 30, 2016.

[39] Auditus, "Audiometer Auditus A-1 specifications," Available online: www.auditus.ua/eng/audiomete-auditus-a, Accessed: Sept. 30, 2016.

[40] Oscilla, "User manual USB350-B," Available online: www.oscillahearing.com/images/USER_MANUALS/USB_Models_and_SM450FF_User_Manual_UK.pdf, Accessed: Sept. 30, 2016.

[41] *Indian Standard Reference Zero for Calibration of Pure Tone Air Conduction Audiometers*, IS 4755:1987, 1987.

[42] Radio ear, "DD45 Audiometric headphones," Available online: www.radioear.us/pdfs/DD45_DataSheet.pdf, Accessed: Sept. 30, 2016.

[43] Beyerdynamic, "DT48 A.00 Dynamic audiometric headphone," Available online: www.europe.beyerdynamic.com/shop/media/datenblaetter/DAT_DT48A0_EN_A4.pdf, Accessed: Sept. 30, 2016.

[44] L. B. Beck and G. D. Causey, "Psychoacoustic calibration of the Telex 1470 earphone," *Acoust. Soc. of Amer.*, vol. 55, no. 5, pp. 1088-1089, May 1974.

[45] Sennheiser, "Sennheiser HDA280," Available online: www.enin.sennheiser.com/global-downloads/file/722/BDA_HDA280.pdf, Accessed: Sept. 30, 2016.

[46] Radio ear, "B71W Bone transducers," Available online: www.radioear.us/pdfs/RE_B71W_ R2016.pdf, Accessed: Sept. 30, 2016.

[47] Radio ear, "B-81 High output bone transducer," Available online: www.radioear.us /pdfs/RadioEarB81.pdf, Accessed: Sept. 30, 2016.

[48] Radio ear, "IP30 Insert earphone," Available online: www.radioear.us/pdfs/IP30% 202014%20X.pdf, Accessed: Sept. 30, 2016.

[49] Etymonic Research, "ER-3C insert earphones for audiometry," Available online: www.etymotic.com/downloads/dl/file/id/634/product/317/er_3c_insert_earphones_for_aud iometry_data_sheet.pdf, Accessed: Sept. 30, 2016.

[50] Etymotic Research, "ER-3A insert earphones for audiometry," Available online: www.msrwest.net/Etymotic/er-3a.pdf, Accessed: Sept. 30, 2016.

[51] Coto Technology, "2300 series multi-pole reed relays," Available online: www.farnell.com/ datasheets/1853228.pdf, Accessed: Sept. 30, 2016.

[52] Texas Instruments, "ULN200x, ULQ200x high-voltage, high-current darlington transistor arrays," Available online: www.ti.com/lit/ds/symlink/uln2003a.pdf, Accessed: Sept. 30, 2016.

[53] Microchip, "PIC18F2455/2550/4455/4550 Datasheet," Available online: ww1.microchip .com/downloads/en/devicedoc/39632c.pdf, Accessed: Sept. 30, 2016.

Acknowledgement

I would like to express my sincere gratitude towards my supervisor Prof. P. C. Pandey for his invaluable time, guidance and support. I am thankful to Mr. Nandakumar Pai for all the valuable time, help, suggestions and support he gave me for this project. I am thankful to Mr. Vidhyadar Kamble for helping me in procuring various electronic components. I want to convey my gratitude towards my lab mates Nataraj, Nitya, Yogesh, Hirak, Saketh, Shibam, and Vikas in Signal Processing and Instrumentation Lab for their help and support.

Page intentionally left blank

Author's Resume

Prachir Puri: The author received B. E. degree in electronics and communication engineering from University Institute of Engineering and Technology, Panjab University, Chandigarh in 2014. Presently he is pursuing the M.Tech. degree in electrical Engineering in Indian Institute of Technology Bombay. His research interests include electronics and embedded system design.