A MICROCONTROLLER BASED AUDIOMETER

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by

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ABSTRACT

An audiometer is an electro-acoustic instrument for quantifying hearing impairment and for diagnosing its causes. The project objective is to develop a compact portable instrument that can be used as a screening audiometer and can also be used for clinical and research applications. A microcontroller based audiometer "IITB-2K11" is developed for conducting tone audiometry, short increment sensitivity index test, tone decay test, and speech audiometry, individually on the two ears by presenting a sequence of acoustic stimuli and receiving the patient's responses. An 8-bit microcontroller is used for all the interfacing and control operations. A combination of 16-bit DSP chip with on-chip ADC and DAC and an SD card is used for generating stimuli and masking noise. Pure tone, warble tone, and pulsed tones over a frequency range from 125 Hz to 8000 Hz, and speech are available as the test stimuli. Wide-band noise, narrow-band noise, and speech-spectrum shaped noise can be used as the masker. A chip with two programmable log attenuators is used for controlling the presentation levels of the stimulus and the masker. A 4x4 keypad and a 128x64 pixel graphics display are used for user interface. The tests can be administered in manual or automated modes. For a given set of transducers, the instrument can be calibrated through its user interface with the calibration data saved in the internal nonvolatile memory. The test results can be transferred using RS232 for preparation of test report.

CONTENTS

LIST OF ABBREVIATIONS

Abbreviation	Term
AM	amplitude modulation
dB	decibel
dB HL	sound pressure level in dB with pressure reference as average hearing threshold for the young adult population for the test frequency
dB SPL	sound pressure level in dB with pressure reference of 20 µPa
D/A	digital to analog
DAC	digital-to-analog convertor
DSP	digital signal processing
FM	frequency modulation
HTL	hearing threshold level
IC	integrated circuit
LCD	liquid crystal display
NB	narrow-band
OAE	otoacoustic emission
PC	personal computer
PGA	programmable gain amplifier
RAM	random access memory
ROM	read only memory
SCF	switched-capacitor based filter
SD	secure digital
SDS	speech discrimination score
SDT	speech discrimination test
SG	signal generating
SISI	short increment sensitivity index
SPI	serial peripheral interface
SRT	speech reception threshold
TA	test administrating
TD	tone decay
WB	wide-band

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Chapter 1
INTRODUCTION

1.1 Background

Hearing impairment or deafness may be caused by various disorders in different parts of the auditory system. Audiometry involves a set of tests for measuring the hearing sensitivity for assessing the nature, degree, and probable cause of the hearing impairment, so that the best-suited medical treatment or hearing aid or some other assistive devices can be prescribed. Hearing test is conducted individually on the two ears by presenting a sequence of sound stimuli and receiving the patient's responses to these stimuli. In the most commonly administered test, known as pure tone audiometry, pure tones with varying levels are presented to the test ear. The minimum level at which consistent responses are obtained is taken as the hearing threshold. A plot of hearing threshold level vs. frequency is known as the pure tone audiogram. Other audiometric tests involve different sets of stimuli and response methods and these include loudness balancing tests, short increment sensitivity index (SISI) test, tone decay test, speech audiometry, tympanometry, auditory evoked brain-stem response (ABR), and otoacoustic emissions based audiometry [1].

For audiometric tests, the auditory stimulus is presented through calibrated headphones or speakers for ear conduction and through a bone vibrator for bone conduction. The acoustic vibrations presented to the test ear may reach the cochlea of the non-test ear through the skull bone and it may result in cross-hearing. To prevent the non-test ear from detecting the stimulus presented to the test ear, a masking noise may need to be presented to the non-test ear. A pure-tone audiometer consists of an oscillator for generating pure tone sounds with various frequencies, source of masking noise, two attenuators for controlling stimulus intensity level, a multiplexer for switching the stimulus to different transducers, power amplifiers, and a response switch.

1.2 Project objective

The objective of this project is to develop a diagnostic audiometer for conducting patient response based audiometry. After studying several audiometers, and particularly the earlier prototypes developed at IIT Bombay, a microcontroller based audiometer "IITB-2K11" is developed. The pure tone, warble tone, pulsed tone, and masking noise are generated using a DSP chip with on-chip DAC. An SD card is interfaced for storing the stimulus waveforms, particularly for speech audiometry. A 2-channel log attenuator chip with 120 dB dynamic range is used for tone and noise level control. The audiometer can be used for conducting tone audiometry, SISI test, tone decay test, and speech audiometry, using air and bone conduction. It can be used for outputting the selected types of stimuli and masking noises at the selected levels (in dB HL) and monitoring patient's responses for conducting the hearing tests in the manual and automated modes. The interface for conducting the tests is through a 4x4 keypad and a 128x64 graphical LCD. The patient's response is received through a response button which has an indicator light for feedback. The saved test results can be seen in the results field of the display and they can be transferred to a PC for preparing test reports. For a given set of transducers, the instrument can be calibrated through its user interface, and the calibration data are saved in the internal nonvolatile memory.

1.3 Report outline

The second chapter provides an overview of the audiometric techniques. Chapter 3 presents our design approach. The instrument hardware is described in the fourth chapter. Chapter 5 describes the software. The system assembly and testing of the instrument are presented in the sixth chapter. The last chapter provides a summary of the work completed and suggestions for future development.

Chapter 2
AUDIOMETRY BASICS

2.1 Physiology of the auditory system

The structures of the peripheral auditory system, through which an acoustic signal is transduced into neural action potentials and subsequently processed by the nervous system are shown in Figure 2.1. It consists of the outer, middle, and inner ear [1], [2]. The outer ear includes the ear lobe (also called as the pinna or the auricle) and the ear canal (also called as the auditory meatus). The middle ear is composed of the tympanic membrane (also called as eardrum), which serves as a divider between the outer ear and the middle ear, and the middle ear cavity containing the ossicular chain consisting of the three bones: the malleus, incus, and stapes [1], [2]. The malleus is attached to the tympanic membrane. The footplate of the stapes inserts into the oval window of the inner ear. The incus is between the malleus and the stapes. The inner ear consists of the cochlea which is composed of three fluid-filled chambers. The third fluid filled chamber (called as cochlear duct) contains the basilar membrane. It consists of finger-like projections that are arranged in rows, referred to as hair cells. The hair cells are connected to fibers of auditory nerve [1], [2].

The air borne sound waves set the tympanic membrane into vibration and the ossicular chain transfers these vibrations to the stapes which is attached to the oval window. Movement of the oval window creates motion in the cochlear fluid, which excites the sensory cells on the basilar membrane, which in turn stimulate a series of nerve endings with the initiation of the nerve impulses [2]. These impulses are relayed through the auditory nerve to the brain.

2.2 Hearing impairment

Primary function of the auditory meatus or the middle ear is the conduction of sound to the inner ear. Any deafness due to disorders in these parts is called conductive deafness. The common causes of disorder in the external auditory meatus are collection of either wax, fungal debris, or foreign bodies [3]. The hearing impairment due to blockage of the external ear is usually not very significant and can be corrected by cleaning. The common diseases affecting the middle ear are perforation in the eardrum, stiffness or damage to the chain of bones in the ear, and collection of fluid in the middle ear cavity. A perforation can usually be diagnosed by visual inspection of the eardrum; however the other middle ear disorders require

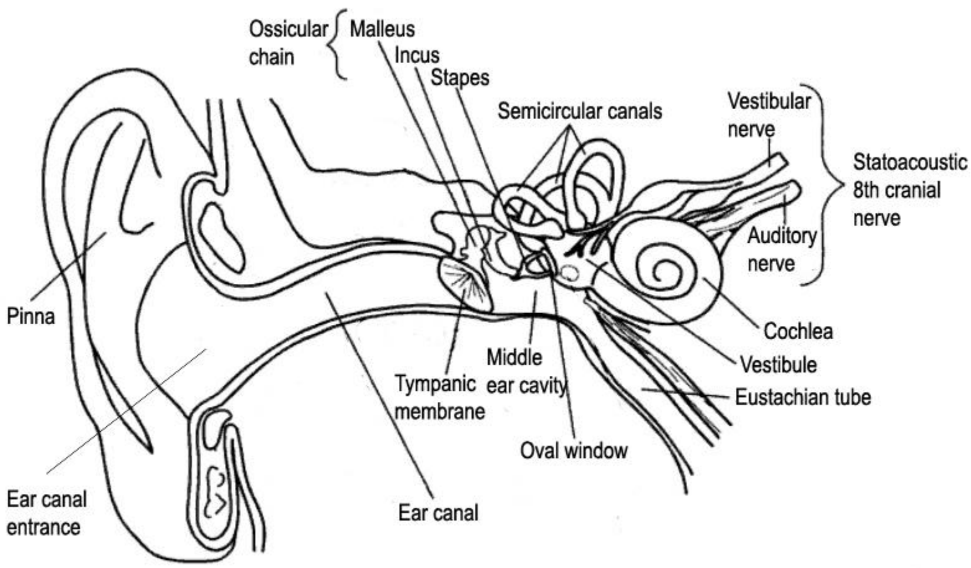


Figure 2.1 The anatomical structures involved in the auditory system. Adapted from [2]

special investigations for confirmation. Collection of fluid in the middle ear is usually treatable by medicines and perforation in the ear drum or stiffness of the ossicular chain is treatable by surgery [4].

The disorders in the inner ear result in a perceptive deafness known as the sensorineural loss and a sensation of buzzing sounds known as the tinnitus [3], [4]. Deafness due to disorders of the inner ear is generally not treatable by surgical methods and hearing aids may be the only prescribing option. An untreatable and life threatening disorder is acoustic neuroma which is a tumor of the auditory nerve [4]. The common symptoms of disorders in the auditory system are (i) difficulty in hearing normal conversation from a distance of 8 feet or a whisper from a distance of 3 feet, (ii) being able to hear people talk but having difficulty in understanding what they say i.e. spoken word appear jumbled up, (iii) hearing whistling/buzzing sounds in the ear or in the head when actually such sounds are not there, and (iv) a sensation of blockage / heaviness in one or both ears, etc [4].

2.3 Audiometric techniques

Different audiological investigations help in diagnosing the nature of hearing impairment and localizing the sites of the disorders. The pure tone audiometry measures the hearing threshold. Tympanometry is another audiological investigation which assesses the structural integrity of the middle ear and helps in diagnosing the nature of the disorder in the middle ear in cases of conductive or mixed deafness. This method also reveals whether there is any stiffness of the ossicular chain, whether the ossicular chain is broken, whether there is collection of fluid in the middle ear, or whether the eardrum has become immobile due to adhesions in the middle ear [4]. The auditory evoked brain-stem response audiometry is used for objectively assessing the hearing acuity, particularly in the case of children who can not respond properly [3].

The audiometric techniques can be broadly grouped as subject-response based audiometry and physiological-response based audiometry. In subject-response based audiometry, the patient is asked to respond to the presented stimulus i.e. whether he/she heard the sound. Physiological or objective tests are the tests in which the response depends on the involuntary body response rather than the voluntary responses from the patient [3], [5]. These tests reveal whether the peripheral auditory system is working properly or not. The main objective tests are otoacoustic emission, auditory evoked brain-stem response, and tympanometry. The types of stimuli and response methods vary in different types of audiometry. The commonly used stimuli are pure tone, warble tone, pulsed tone and speech signals. If these stimuli are presented to the subject's ear, then it is called air conduction test. In the bone conduction test, the stimulus is presented through a bone vibrator which couples the vibration through the skull bones directly to the cochlea bypassing the outer and inner ear. Combination of these tests are very helpful in quantifying the extent of conduction and sensorineural loss in cases of mixed loss [5]. Different types of tests are described further in the following subsections.

2.3.1 Tone audiometry

The patient is instructed to press a response button or raise his/her hand whenever he/she hears a tone in the test ear. In this test, pure tone (continuous), warble tone (frequency modulated, FM), or pulsed tone (amplitude modulated, AM) is applied as a stimulus with varying levels to the subject's test ear. The minimum stimulus level (in dB HL) to which consistent responses are obtained is taken as the hearing threshold level (HTL). The test is carried out with tones of 125, 250, 500, 750, 1 k, 1.5 k, 2 k, 3 k, 4 k, 6 k, and 8 kHz. The HTL plotted on a function of the frequency is known as pure-tone audiogram. The loss in hearing sensitivity may be due to a conduction disorder (in the external auditory meatus or the middle ear or both) or due to a sensorineural disorder which can be due to a disorder either in the inner ear or in the auditory nerve or both. In mixed loss, the disorders exist in both the conductive apparatus as well as the inner ear [4].

2.3.2 Tone decay test

Tone decay test is used for determining the continuous listening capability for about 1 min. In this test, pure tone at a particular frequency and intensity is presented continuously to the test ear. The subject is instructed to press a response button or raise his / her hand whenever he / she hears a tone for the first time and then again whenever the tone disappears [5]. If the patient is not able to hear the tone continuously for more than one minute, its intensity is increased by 5 dB and test is conducted again. The lowest intensity for which patient is able to hear the tone for about 1 min. is considered as the test result at that frequency [5]. A plot of tone decay threshold vs frequency is helpful in diagnosing the sensorineural loss. The diagnosis is < 5 dB HL: normal, 10-15 dB HL: mild loss, 20-25 dB HL: moderate loss, >30 dB HL: severe loss [1]. A severe loss indicates retrocochlear lesion and warrants a detailed neuro-otological examination.

2.3.3 Short increment sensitivity index (SISI)

A pure tone at a particular audiometric frequency and intensity is presented with brief increase of 1-dB above the carrier tone every 5 s [5]. The 1 dB increment is for an interval of 300 ms, out of which the rise time and fall times are 50 ms each. The subject is instructed to press a response button or raise his/her hand whenever he/she observes an increase in the level of the tone for a short duration in the test ear. The number of responses out of twenty increments is converted to percentage as the test score. The scores are plotted vs the frequency. The test helps in detecting the cochlear or retrocochlear lesions [5].

2.3.4 Speech audiometry

Speech audiometry assesses the integrity of the entire auditory system by assessing the ability to understand speech. Its main application is in the identification of neural types of hearing loss, in which both the reception as well as the discrimination of speech is impaired more markedly than in cochlear or conductive hearing loss. There are two types of speech audiometric tests; speech reception threshold (SRT) test and speech discrimination test (SDT). Sensorineural loss affects both the reception as well as the discrimination of speech while conductive loss affects only the reception.

Speech material is input to the audiometer from a microphone, tape, or CD player. SRT measures the sound levels at which speech is detected and individual words are distinguished from one another. The SRT is the lowest hearing level in dB HL at which 50 % of a list of equally stressed words is correctly identified by the subject [5].

SDT assesses the patient’s ability to understand mono-syllable words presented at 30 dB above the SRT level [5]. A list of distinct monosyllabic words are presented and the subject is asked to repeat the word after each presentation. The percentage of the total number of words which the subject is able to identify correctly gives the speech discrimination score (SDS). A high score is associated with normal hearing or conductive hearing loss and low score is associated with sensorineural loss [3], [5].

2.3.5 Otoacoustic emissions based audiometry

The sounds produced from the cochlea are known as otoacoustic emissions. The inner hair cells in the cochlea bend due to wave motion in the cochlear fluids. This bending creates an effect that triggers neural impulses that are sent to the brain for interpretation. Outer hair cells work to make inner hair cells more sensitive to sounds by expanding and contracting. These cells act as an active source of energy and influence the basilar membrane motion; causing sounds that travel back through the middle ear and out into the ear canal. These sounds can be picked up by a probe containing a miniature microphone inserted into the ear canal [6]. The microphone output is amplified, digitized, and processed for diagnosis.

These otoacoustic emissions (OAE) are of two types, the spontaneous otoacoustic emission and the stimulated also called as the evoked otoacoustic emission (EOE) [6]. The spontaneous otoacoustic emissions are generated spontaneously and they do not require any external stimulation [6]. These can be picked up by a microphone. These sounds are very weak and are not viewed as a clinical indicator of the mechanical activity of the cochlea. Absence of these signals does not indicate a loss of hearing [6]. The evoked emission occurs when a stimulus is presented to the ear. The transient evoked otoacoustic emissions (TEOAE) are produced by presenting a click sound at a level of 80 dB SPL for about 2 ms in the outer ear canal through a probe. The response emerges shortly afterwards, starting at about 3 ms and lasting till about 20 ms and is picked up by a microphone. Other commonly used stimuli are single sinusoids, half sinusoids, white noise pulse, rectangular or gaussian shaped clicks, or tone bursts [6]. The duration of the stimulus for recording TEOAEs may be from a few milli seconds to several hundreds of seconds called as short and long TEOAEs. The distortion product otoacoustic emissions (DPOAE) are obtained by presenting two pure tones simultaneously with frequencies F_1 and F_2 to the test ear through two speakers. These tones evoke a cochlear response which consists of a number of pure tones at the inter modulation frequencies i.e. $m F_1 - n F_2$, where m and n are integers [6]. The inter-modulation tones are named as distortion product OAEs because they appear to be the result of nonlinear distortion of the stimulus tones. These signals are picked up by a miniature microphone in the probe along with the two speakers. The most prominent signal among these inter modulation is frequency $2 F_1 - F_2$. For best results F_2 is chosen as $F_2 = 1.2 F_1$. The level of $2 F_1 - F_2$ signal is taken for diagnosing the hearing [6].

The presence of measurable evoked OAEs suggests normal functioning of outer hair cells and hearing that is probably normal or nearly normal. As most hearing loss involves loss of the outer hair cells; the lack of an emission is a good predictor of sensory loss. It is commonly used to screen the hearing of newborn babies.

2.3.6 Auditory brain stem response

In this technique, small surface electrodes are attached on the vertex of the scalp [1], [3]. An earphone is placed in the ear. Small electrical currents are generated by the normal auditory system when excited by some sounds through an earphone. These electrical responses are recorded across the electrodes, amplified and processed. If hearing is normal, these responses should be recorded for low level stimuli, otherwise the threshold of these responses will be elevated. It is commonly used to screen the hearing of newborn babies [1], [3]. Because the response is recorded across surface electrodes and doesn't have to travel back out through the middle and external ear, these signals are less dependent on the status of the middle ear as compared to the OAE tests.

2.4 Types of conduction

Diagnosing of an ear needs stimulation of its cochlea. This can be done by two ways; via air conduction and bone conduction. For testing using air conduction, the stimulus is presented to the test ear using a headphone and it reaches the cochlea through the outer and the middle ear. For testing with bone conduction, the stimulus is applied with a bone vibrator which is generally placed over the mastoid bone or over the forehead. The stimulus is transmitted to the cochlea of the both the ears through the skull bones [5].

2.5 Masking in audiometry

In air conduction audiometry, the sounds presented for stimulating the test-ear cochlea may also produce vibrations of the skull bones. These vibrations may stimulate the cochlea of the non-test ear, which is undesirable. If one cochlea is better than the other, the subject response may be due to the stimulus reaching the better cochlea, even if the worse ear is the test ear. Leakage of the sound to the non-test cochlea is known as cross hearing. In case of bone conduction audiometry, the vibrations from the vibrator applied either over the mastoid or frontal bone reach the cochlea of both the test and the non-test ears. To ensure that the subject's response is only due to the stimulus of the cochlea of the test ear, we should mask the functionality of the non-test ear. This can be done by presenting a noise to the non-test ear so that the non-test ear cannot participate in the hearing test. The masking noise should not be loud enough to pass over to the test ear and mask its sensitivity. Thus the masking noise level should be above a minimum masking level and below a maximum masking level [5]. In air conduction, there is typically 40 dB attenuation from the test ear to the non-test ear. Let A_t and A_m the air-conduction thresholds of the test and non-test ear, respectively, and B_t and B_m be the corresponding bone conduction thresholds. The limits for masking noise are given as [5],

Maximum masking level for both air and bone conduction = $B_t + 40$

Minimum masking level for air conduction = $A_m - B_m + A_t - 40$

Minimum masking level for bone conduction = $A_m - B_m + B_t$

For tone audiometry or SISI, or tone decay, the masking noise can be either wide-band (also called as broad-band or white noise), or narrow-band noise. The wide band noise has uniform power density spectrum over the audiometric frequency range. However the masking effect is actually contributed by frequency components centered on the test tone frequency, over a bandwidth of about 1/3 to 1/2 octave, which is known as critical band. Broad-band noise band-pass filtered with a band approximately corresponding to the critical band is known as the narrow-band noise. Compared to the wide-band noise, it gives the same masking effect at a lower sound pressure level.

2.6 Pure tone audiometer

A pure tone audiometer is used for finding the hearing threshold level at the audiometric frequencies. It consists mainly of tone generator, level controller, and power amplifier. It also has a facility for air and bone conduction tests with masking noise to the non-test ear. Figure 2.2 shows a block diagram of a pure tone audiometer. It consists of an oscillator for generating pure tone, a masking noise generator (wide-band noise), a band-pass filter for filtering the broad-band noise to get narrow-band noise, two attenuators for changing the level of stimulus and masking noise, two selector switches for selecting the type of stimulus and type of noise, three power amplifiers, a response switch for receiving the response from patient, and a control unit. The control unit controls the selection of stimulus, the frequency of the tone, the type of masking noise, the centre frequency of the band-pass filter, the stimulus and/or noise attenuating level and the response button.

The pure tone generator generates pure tones of different frequencies, ranging from 125 Hz to 8 kHz and each frequency should be within 3 % of the indicated frequency. The outputs of pure-tone generator as well as noise generator are fed to attenuators to have corresponding attenuation in dB HL. The attenuator should be capable of controlling the acoustic output over a desired range in steps of 5 dB. Three multiplexer units are used to divert the stimulus signal and masking noise to the amplifiers for presenting to the left or right ear or to the bone vibrator. The power amplifier should provide the output power required by the maximum sound pressure level from the headphone and the bone vibrator. The power amplifier must have low distortion and a good signal-to-noise ratio. It may be noted that the output level is calibrated in dB HL, and the voltage output for a given level is different for different frequencies. This is because of a frequency dependent relationship between dB HL and dB SPL as given in Table 2.1, and because of the frequency response (dB SPL vs frequency) of the transducer being used. This can be achieved either by varying the oscillator output level as a function of frequency or by varying the attenuation level as a function of frequency.

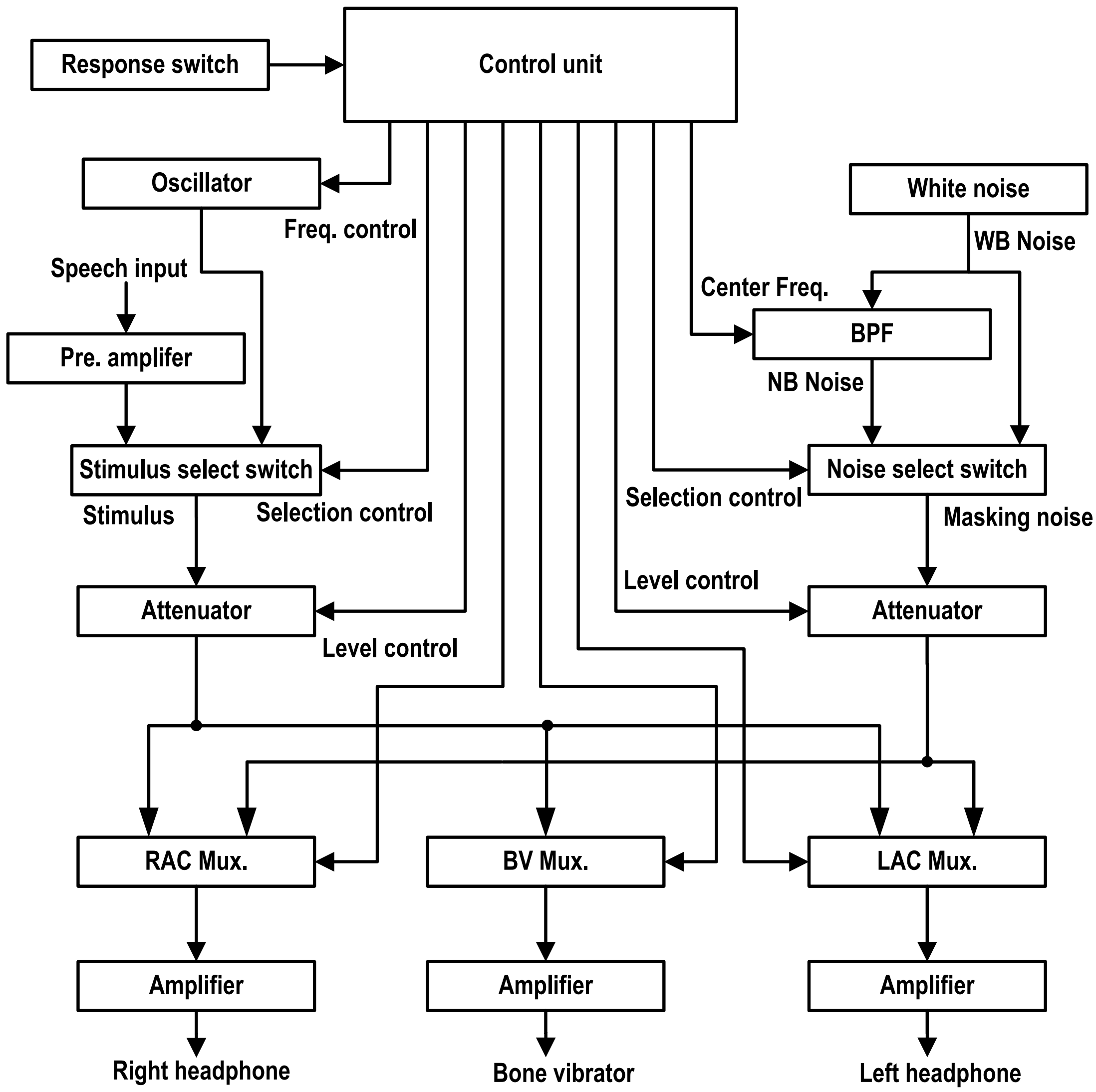


Figure 2.2 Block diagram of a pure tone audiometer as reported in [7].

2.7 Earlier schemes used in developing audiometers at IIT Bombay

In the audiometer circuits reported in [7], [8], [9], levels of the stimulus and the masker are controlled using programmable log D/A converters with parallel interface to the microcontroller. Tones are generated using a programmable oscillator realized using a SC filter circuit, with the frequency of the sinusoidal oscillation being proportional to the programmable digital clock from the microcontroller. Warble tone is generated by introducing frequency modulation in the clock frequency. Pulsed tone is generated by amplitude modulation of the sinusoidal tone by varying the attenuation in discrete steps at the zero-crossings of the tone. The noise is obtained from a shift register based pseudo-random binary sequence generator and SC based tunable band-pass filter.

Mossa [7] used a direct digital synthesizer (DDS) for generating both pure tone and warble tones. In this circuit the output frequency of the DDS is varied by sending control signals from the microcontroller. The clock frequency of the DDS is taken from the crystal clock of the microcontroller. A look-up table is used for sending control word and data to generate each frequency using the DDS. A log attenuator, PGA 2310 with a dynamic range of 127 dB and serial interface was used as level controlling [7]. The masking noise is generated using DSC dSPIC33FJ128GP804, while Philips P89V512 is used for user interface and control of the various peripherals. There designs have remained short of realizing all the features needed in a diagnostic audiometer.

Table 2.1 Conversion between dB HL and dB SPL [1]

Freq. (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	47	26	13	8	7	11	9	10	13	5	13

Chapter 3
DESIGN APPROACH

3.1 Audiometer specifications

The project objective is to develop an audiometer which can be used as a screening audiometer and can also be used for clinical and research applications. It should be easy and convenient to operate, without the need for regular calibration. It should be possible to carry out its calibration without opening the instrument. Test stimuli should include pure tone (continuous), pulsed tone (amplitude modulated), and warble tone (frequency modulated) for all the audiometric frequencies. It should facilitate conducting pure-tone threshold testing, speech test, tone decay test, and SISI test. It should provide wide-band masking noise with flat spectrum from 100 Hz to 10 kHz, narrow-band masking noise with centre frequency equal to the test tone frequency with bandwidth of 1/3 or 1/2 octaves, and speech-spectrum shaped noise. It should permit an easy selection of the test ear, type of conduction, type of audiometry, stimulus type, and masker type, and levels of the stimulus and the masker. It should have a provision to save the test results and facility for displaying the earlier results in a test. It should have a provision for presenting speech from microphone or an internal storage device for speech test. It should give an output level from -10 to 160 dB SPL for air conduction and from 10 to 90 dB SPL for bone conduction. It should have a patient response button. It should have the facility for transferring the test results to a PC and should permit the possibility of operating the instrument through a graphic user interface on the PC. Further it will be useful to have the facility of conducting the test in an automated mode, whenever permitted by the test procedure.

3.2 Design approach

A block diagram of our microcontroller based audiometer is shown in Figure 3.1. To facilitate modular program development without having the overheads of an operating system, two microcontrollers have been used. One is for test administration and it is called as the test administrating microcontroller (TA). It handles user interface and controls the test parameters. The second microcontroller, called as the signal generating microcontroller (SG) generates the stimulus and the masking noise. For a compact design, the TA microcontroller should have sufficient programmable ROM, data RAM, parallel I/O ports, UART, and a programmable timer / counter for handling all the operations without requiring additional chips. We also need nonvolatile memory for storing the calibration data. For this application

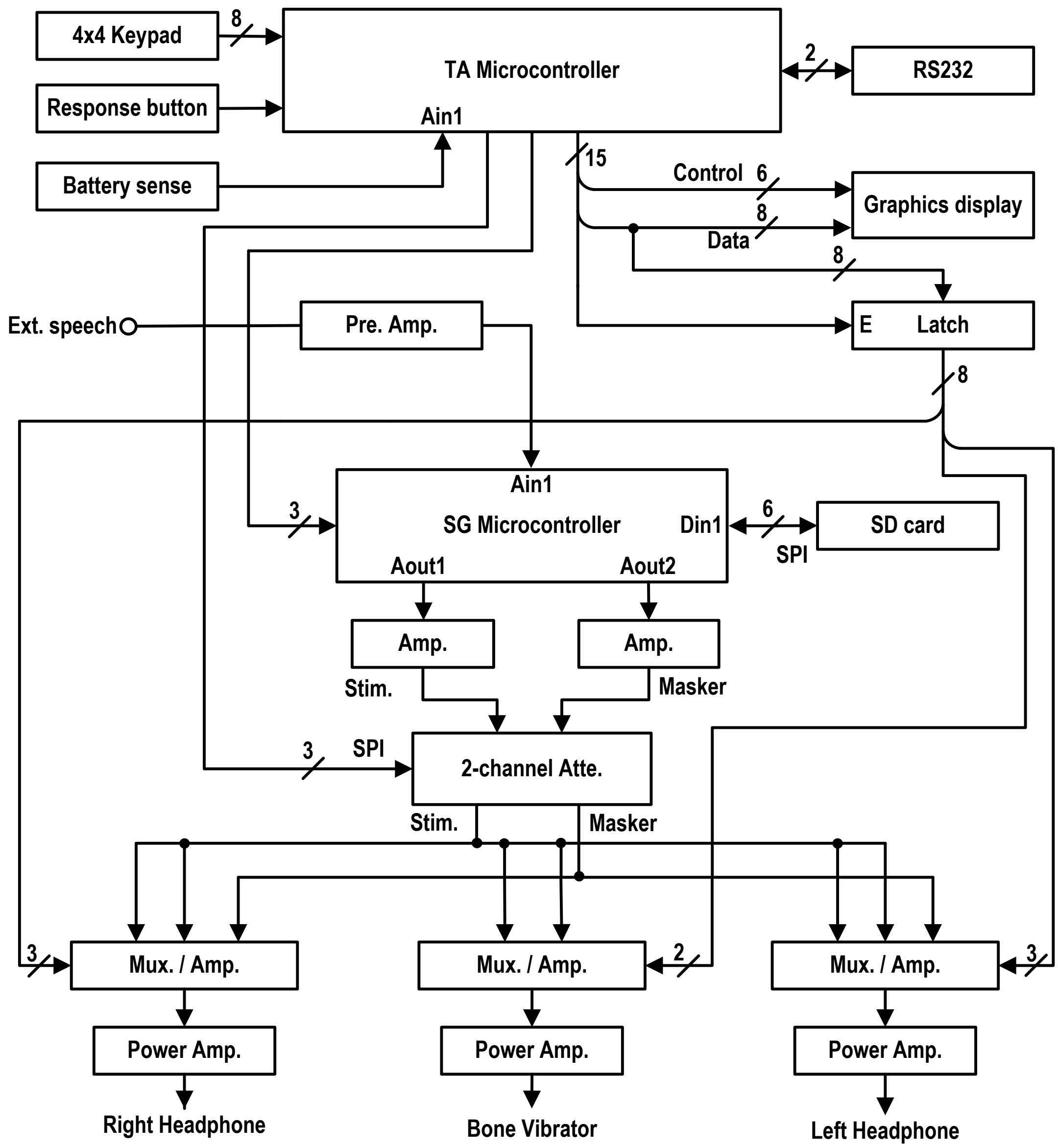


Figure 3.1 A block diagram of the audiometer using two microcontrollers

it was decided to use PIC16F1939, a 40-pin microcontroller from Microchip's 8-bit microcontroller family. Its main features include; 32 MHz clock, 5 V supply, 35 I/O lines, 3 timers / counters, 23 interrupt sources, 16 K bytes of program memory, 1 K byte data memory, 256 bytes of EEPROM for used as nonvolatile data storage, precision internal oscillator, power saving sleep mode, master synchronous serial port (MSSP) with SPI, and in-circuit serial programming [10]. A 4x4 keypad is interfaced to this microcontroller for user interface. All the information on selection is provided on 128x64 graphical display, LGM12641BS1R unit with on-board KS0108B controller. It requires 8 data lines and 6 control lines, for interfacing to the TA microcontroller.

Audiometric stimulus tones can be generated digitally by using a microcontroller with sufficient internal memory for storing the waveform samples and an on-chip DAC. For getting the low distortion waveform, we need to use an appropriate sampling rate, resolution (number of quantization bits). Masking noise can be generated digitally by a pseudo-random binary sequence which can be obtained by using a long shift register, and filtering using a digital filter [11]. As the waveform generation operation is time-critical, the task of signal generation is separated from the test administration. For this application it was decided to use dSPIC33FJ128GP802, a 28-pin microcontroller from Microchip's 16-bit digital signal controller family. Its important features include; up to 40 MIPS, on-chip 128 Kbytes of flash program memory, 16 Kbytes of data SRAM, two SPI modules supporting 8-bit and 16-bit data, 12-bit ADC module and 16-bit dual channel DAC module [12]. Right channel DAC is used for stimulus and left channel is used for masking noise. It was estimated that generation of warble tones can neither be carried out by real-time calculation of the samples, nor all the samples can be stored in the internal program memory. Hence an SD card is interfaced and used as an external memory for storing the stimulus waveforms. It also permits storage of speech stimuli. The waveform samples are transferred from the SD card to the SRAM of the microcontroller before outputting them through the DAC. For conducting speech audiometry, external speech input is given to ADC and then output through its right DAC. Alternatively, the speech output can be generated from the data samples on the SD card. This permits use of a common hardware for tone and speech audiometry. For each channel, the DAC gives two complementary outputs with a fixed DC offset. A differential amplifier is used for removing the offset and make the signal symmetrical about ground as subsequently needed for the programmable attenuator. The differential amplifier gain is selected to make the signal swing compatible with the input range of the attenuator and amplifier. The user specified test tone and masking noise specifications are sent from the TA microcontroller to the SG microcontroller.

Audiometry involves outputting the selected stimulus and noise to be output at the selected levels. The dynamic range of 16-bit DAC does not permit software scaling of the stimulus and noise waveforms for controlling the presentation levels. Hence external attenuators are needed. The attenuator should be able to control the output level in steps of 5 dB. For this purpose, PGA2310, from Texas Instruments is used. It is a digitally controlled stereo audio volume control. The chip has two programmable log attenuators, each consisting of a resistor network, an analog switch array, and a bipolar op-amp stage. The switches are used to select taps in the resistor network which determine the gain. It can be operated with ± 4.5 V to ± 15.5 V supply. The internal op amp stages provide low-noise and distortion-free drive for 600 Ω loads. The attenuation range is from -95.5 dB to +31.5 dB; which gives dynamic range of 127 dB in steps of 0.5 dB [13]. It also has mute facility. The gains for both the attenuators are set using a common SPI serial interface. The attenuator chip is interfaced using SPI to the TA microcontroller.

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. In addition to the level control provided by the attenuator chip, an additional level control of 20 dB is needed in the stimulus path for the headphones. This can be realized using the multiplexer itself. Hence we need a 3-to-1 multiplexer for each headphone (stimulus with 20 dB gain, stimulus with 0 dB gain, masker with 0 dB gain) and a 2-to-1 multiplexer for the bone vibrator (stimulus with 0 dB gain, stimulus with 20 dB gain). Each of these multiplexers is realized using a 3-channel 2-to-1 multiplexer chip and an op-amp. The digital controls, for multiplexing are given by the TA microcontroller. Due to the lack of availability of pins on the TA microcontroller, the 8 pins used for providing the data to the LCD are multiplexed using a latch to provide 8 control outputs for the analog multiplexers. The output from each multiplexer is fed to the individual power amplifier for driving the three power amplifiers.

A response button is connected to one of the port pins of the TA microcontroller for taking the responses from the patient. The same pin is also used for giving a feedback to the patient by turning on a light in the button enclosure. A serial interface is provided to communicate with PC for transferring test results along with the test conditions. This interface can also be used for controlling the audiometer by using a PC based graphical user interface. An attenuator based voltage sense unit is connected to the TA microcontroller for monitoring the status of input supply voltage. No additional hardware is needed for calibration. It is totally software controlled and the calibration data are saved in the EEPROM of the TA microcontroller.

Chapter 4
HARDWARE

4.1 Introduction

This chapter gives a description of the hardware blocks used in designing the instrument. As shown earlier in Figure 3.1, most of the blocks are interfaced to the two microcontrollers: the test administrating (TA) and signal generating (SG) microcontroller. The TA microcontroller is interfaced with the keypad, the graphical display, attenuator, latch, multiplexers, RS232 serial interface, and the response button. The SG microcontroller is interfaced with the external speech input, signal amplifier, and the SD card. User interface is achieved by interfacing a 4x4 keypad and a 128x64 LGM12641BS1R graphical display to an 8-bit, PIC16F1939 microcontroller (U1) which is used as the TA microcontroller. The 16-bit, dSPIC33FJ128GP802 microcontroller (U2) with on-chip ADC and DAC is used as the SG microcontroller. The masking noise is synthesized and output using one of its two DAC outputs. Stimulus tones are also generated by U2 by interfacing an SD card with it and using its second DAC output. Control for generation of stimulus signals and masking noise is achieved by transferring a command byte via a protocol from U1 to U2, using an asynchronous handshake mediated protocol. The circuit uses +5 V, -5 V, and +3.3 V supplies, which are generated from input ±9 V power supply by using linear regulators. Analog and digital grounds are separated throughout the hardware blocks.

4.2 Keypad

A 4x4 keypad is interfaced to the TA microcontroller U1 (PIC16F1939) as shown in Figure 4.1. Row-column matrix scanning along with software debouncing is used for inputting the key presses. It is interfaced with port A (pins RA0 to RA7) of U1. Out of the 8 pins, pins RA4 to RA7 are used for outputting the data to the keypad column lines and remaining pins are used for reading the data from the keypad row lines. Because the key press involves shorting of two port pins, the pins used for keypad cannot be used for any other purpose. Since port A does not have internal pull-ups, an external resistor pack is used to provide pull-up. Although port B has internal pull-ups, it cannot be used for keypad interfacing, because pins RB6 and RB7 are multiplexed with ICDCLK and ICDDAT pins, respectively, and are part of the port used for program loading and debugging using in-circuit debugger “PICkit^{TM}3” from Microchip.

4.3 Display

For displaying all the test states, inputs / commands, and results, a 128x64 graphical LCD LGM12641BS1R has been interfaced to the TA microcontroller U1, as shown in Figure 4.2. This display has an on-board controller KS0108 with display RAM, 64-bit data latch, decoder, and 64-bit drivers [14]. The display RAM stores the display data received from the microcontroller U1. It generates the drive for LCD pixels according to the stored data. The display has 8192 pixels arranged in 64 rows and 128 columns. These 64 rows are logically grouped into 8 pages. The 128 columns are driven by two chips, each chip having 64 columns of pixels [14], [15]. The two chips have to be addressed independently. It is powered by + 5 V supply. It has 20 pins; 8 parallel data lines (D0-D7), 2 chip select pins (CS1 and CS2), reset pin (RST), IC enable pin (E), data/command receiver pin (D/L), read/write signal pin (R/W), two back-light pins (A and K), two contrast change pins (Vo and Vee) and two supply pins (VDD and DGND). Connection of the display to U1 is shown in Figure 4.2. Port D of U1 is connected to the 8 data lines. The handshake lines E, CS1, CS2, RST, and RS of the LCD are connected to the pins RC1, RC2, RB1, RB2, and RC0 of U1. Sending a positive pulse to enable pin E of the LCD controller makes it active. As the pins D0 to D7 are not always needed for command / data transfer to the LCD, these can be multiplexed for other usage. CS1 is used to write onto first set of 64 column pixels and CS2 is used for the second set of 64 column pixels. Logic 0 on R/W pin puts the LCD in write mode, while logic 1 on this pin puts the LCD in read mode. Since in our application, reading of the data from LCD is not needed, the R/W pin of LCD is permanently connected to ground. The LCD is reset by sending logic level 0 to RST pin. Before outputting the command / data on the parallel data pins, the D/L pin should be appropriately set. A p-n-p transistor (BC557) Q1 is used to turn on / off the backlight using RE2 of U1. Contrast of the backlight is set by a resistor divider formed by R46 and R47.

4.4 Communication between the TA and SG microcontrollers

The waveform for the stimulus and masker are generated by the SG microcontroller U2 (dSPIC33FJ128GP802). For sending the parameters of these waveforms to the SG microcontroller from the TA microcontroller U1, an asynchronous serial data transfer protocol using a handshake is used. It involves interrupting and sending bits of encoded data sequentially from U1 to U2. For this communication, pin RB4 of U1 is used for interrupting U2 via its pin RB7 pin. Pin RC4 of U1 is connected to pin RB5 of U2 for sending the data bits sequentially. After reading the data bit on its pin RB5, U2 gives an acknowledgement on its pin RA4 to the pin RB0 of U1. The next bit is sent again from U1 by interrupting. This sequence will be repeated and continued till the last bit of the data is transferred from U1 to U2. The serial communication interface is shown in Figure 4.3. This interface does not require any clock generation and the protocol sets no limit on the number of bits to be transmitted.

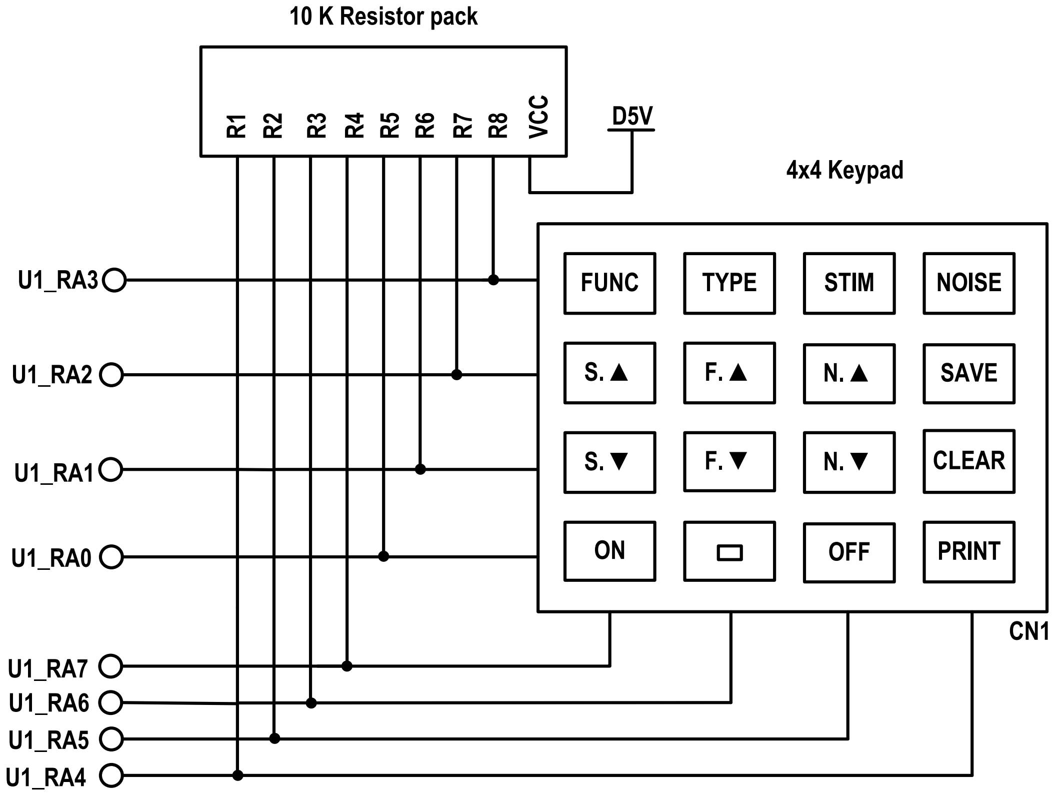


Figure 4.1 The interfacing of 4x4 keypad to the TA microcontroller U1.

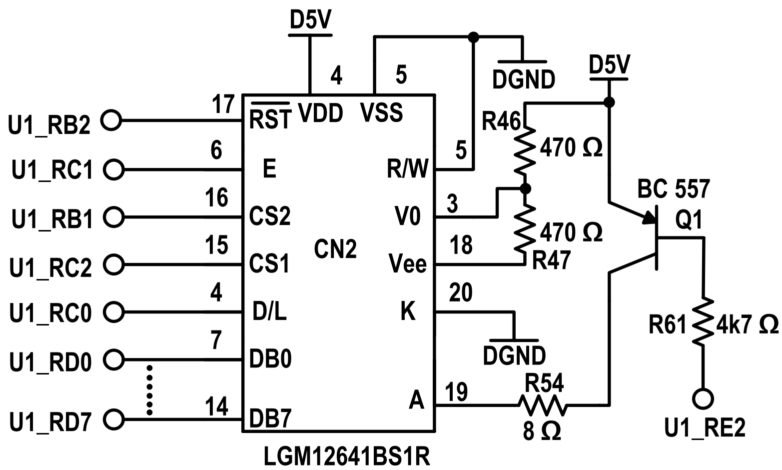


Figure 4.2 Interfacing of graphical LCD with the TA microcontroller

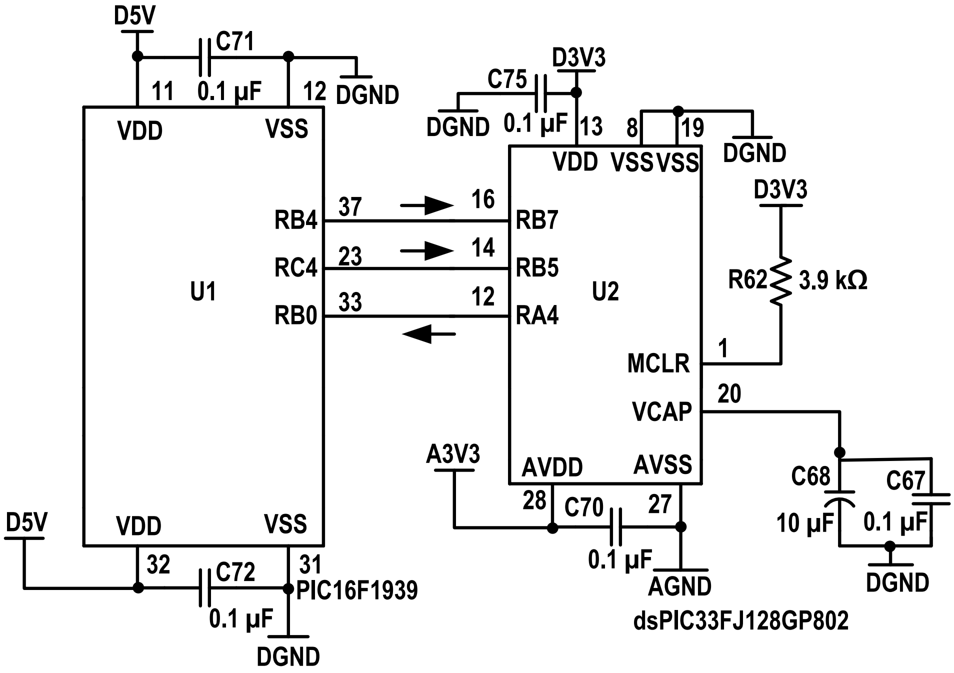


Figure 4.3 Data transfer from the TA microcontroller (U1, PIC 16F1939) to the SG microcontroller (U2, dSPIC33FJ128GP802)

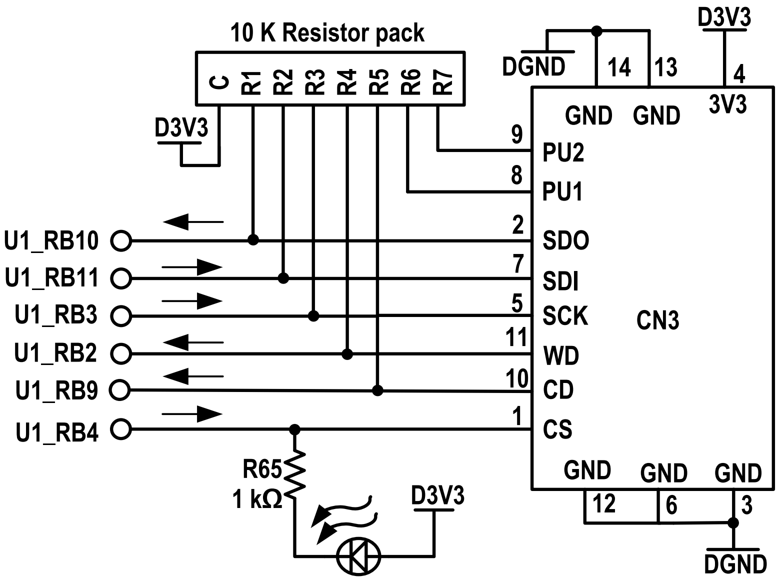


Figure 4.4 Interfacing of SD card connector (CN3) to SG microcontroller

4.5 Interfacing of SD card to the SG microcontroller

Due to difficulty in storing the samples of one cycle of each of the tone stimuli (pure tone, warble tone, and pulsed tone, at all the audiometric frequencies) in the program memory of the SG microcontroller, an SD card is interfaced for storing the waveform samples. The SD card and the SG microcontroller are powered by the same 3.3 V supply, The SD card is interfaced using the SPI protocol. The SPI module of U2 is enabled and its SPI signals are mapped to the port pins: SCK to pin RB3, SDI to pin RB10, and SDO to pin RB11. The SCK, SDI, and SDO pins of U2 are connected to the SCK, SDO, and SDI pins of the SD card, respectively. It is necessary to enable the SD card for the data transfer. For this, pin RB4 is connected to chip select CS of the SD card. The card detect (CD) and the write detect (WD) pins of the card connector are connected to pin RB9 and pin RB2 pins of U2, respectively. It is necessary to keep the write protection slide switch of the SD card in a proper position before inserting it into an SD card connector for allowing U1 to write the data into it.

4.6 Interfacing of the external speech input to the SG microcontroller

To avoid having to use a multiplexer for selecting between external speech input and the tone stimulus generated by the SG microcontroller, the speech input is applied to the ADC input of the microcontroller. For speech audiometry, the ADC input is output through the DAC. For tone audiometry, the internally generated tone stimulus is output through the same DAC. The speech signal is given to pin RA1, an analog input pin of U2 via a microphone and pre amplifier circuit as shown in Figure 4.5. Since the output of the microphone is a low signal (maximum of 200 mVp-p), a pre amplifier is used to amplify the analog speech signal. It is realized using op-amp U3. Low power op-amp MCP601 from microchip has been selected for this purpose. It can be operated with a single supply voltage of 2.7 V to 5.5 V. It is a CMOS op-amp with rail-to-rail output swing [16]. The op-amp is powered by 5 V. The gain is set by R_{49}/R_{53} as 10. It is an ac coupled circuit with the output biased with an offset of 1.3 V set as Vcc $R_{48}/(R_{48}+R_{44})$.The amplifier output is connected to the ADC module. The reference analog ground for the analog signal is given through pin RA1 of U2.

4.7 Buffering of the DAC outputs from the SG microcontroller

The SG microcontroller U2 has two DAC outputs marked as left and right channels, for stereo audio application. Each channel gives positive and negative outputs, biased at 1.65 V [12]. In our application the right DAC is used for outputting the stimulus waveform and the left DAC is used for outputting the masker waveform. A differential amplifier is used to remove the dc bias, centering the waveform at 0 V. The circuit is shown in Figure 4.6. Each amplifier is realized using op-amp MCP602. The gain is set as $R_{50}/R_{60} = R_{56}/R_{55}$ as 0.1, to make the signal swing compatible with the input range of the programmable amplifier / attenuator. The op-amps are powered by ±5 V dual supply. Generating audio signals with bipolar swing about 0 V avoids the need for ensures that we do not need to use ac coupling in the later stages all the way up to connecting to the transducers, thus avoiding the use of electrolytic capacitors and associated variation in the frequency response.

4.8 Attenuator

The intensity levels of the stimulus and masker need to be controlled in dB steps over the full audiometric range. To maintain the calibration in dB HL, the levels should be changed, with the stimulus frequency. This can be achieved by using a digitally controlled logarithmic attenuator. For this purpose, PGA2310 (from Texas Instruments) has been used, as shown in Figure 4.7. It has two independent audio channels, with a common serial interface for controlling the attenuation in dB steps [13]. Its right channel is used for the stimulus and the left channel is used for the masker. It is powered by ± 5 V supply. The gain control becomes effective at the zero crossing of the analog signal and this feature minimizes distortion in the

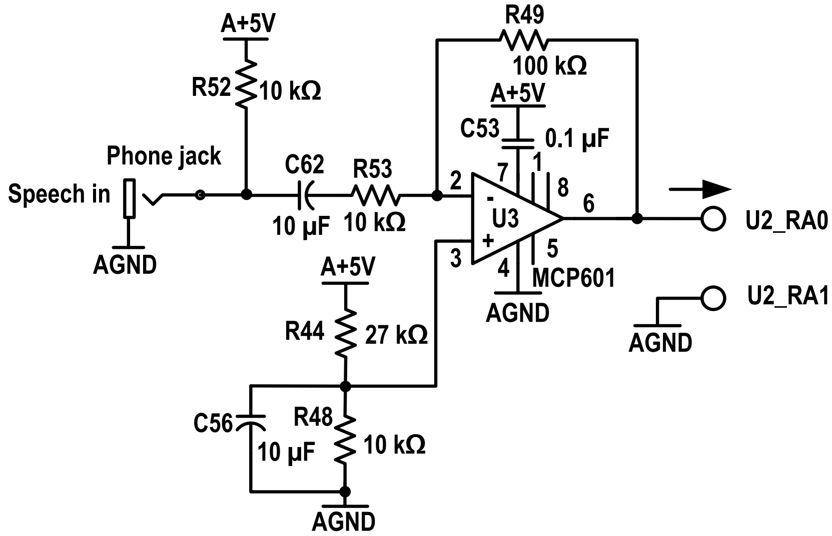


Figure 4.5 Speech pre amplifier circuit

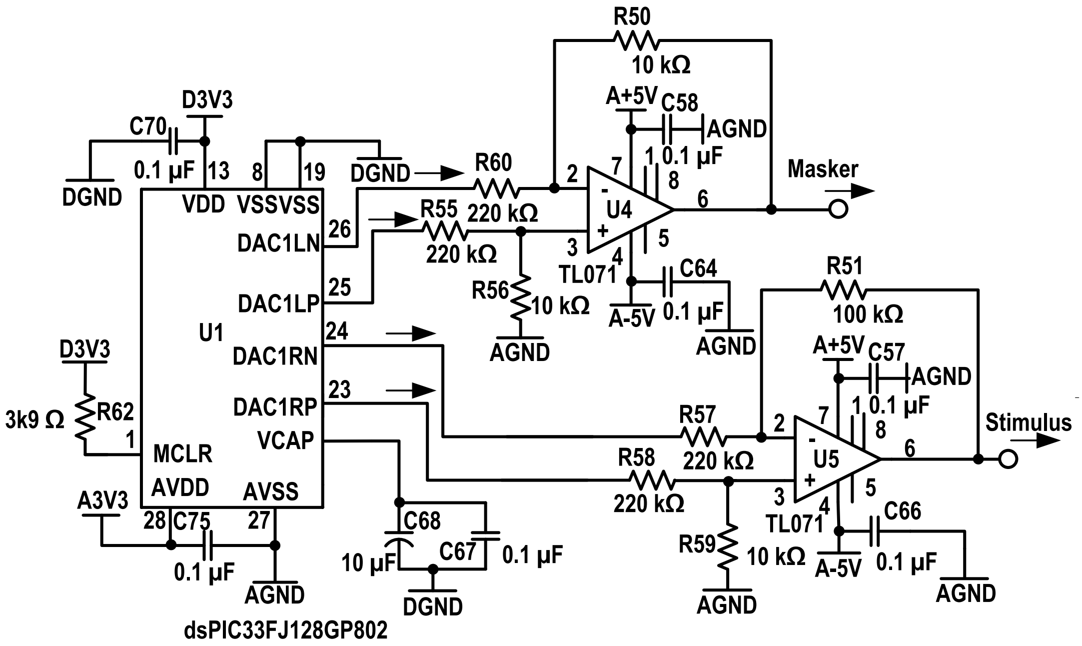


Figure 4.6 Buffering of the stimulus and masker waveform outputs from the SG microcontroller

output due to discrete control of the output level. Each channel has 127 dB dynamic range from -95.5 dB to +31.5 dB in 0.5 dB steps with a driving capacity of 600 Ω without buffering. The attenuator uses a resistor network, an analog switch array, and a bipolar op-amp stage to attenuate the input signal [13]. On power up, all internal flip-flops are reset. The gain byte value for both the left and right channels are set to 0x00H, or mute condition. The serial control port includes 4 pins. Those are active-low chip select input CS (pin 2), serial data input SDI (pin 3), serial clock input SCLK (pin 6), and serial data output SDO (pin 7). Data can be written to it after CS is made low. Control data are provided as a 16-bit word at the SDI pin, which provide 8 bits each for the left and right channel gain settings. The waveforms for the serial control port, as shown in Figure 4.8, have following features.

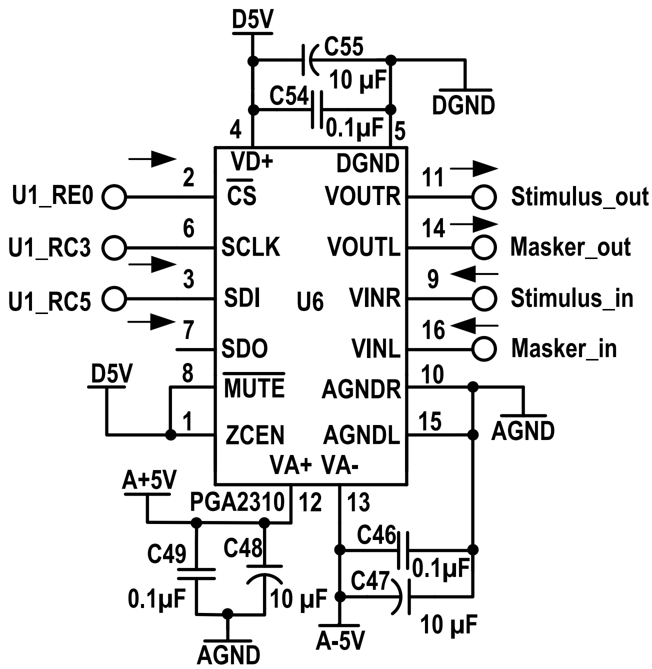


Figure 4.7 Interfacing of attenuator to the TA microcontroller

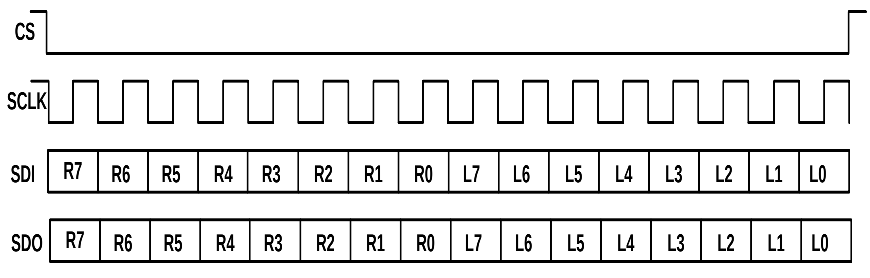


Figure 4.8 Serial interface protocol of the attenuator, PGA2310 [13].

- SDI is latched on the rising edge of SCLK.
- SDO transitions on the falling edge of SCLK.
- Gain byte format is MSB first, straight binary.
- R0 is the least significant bit of the right (stimulus) channel gain byte.

- R7 is the most significant bit of the right (stimulus) channel gain byte.
- L0 is the least significant bit of the left (masking noise) channel.
- L7 is the most significant bit of the left (masking noise) channel gain.

From 16-bit data, each channel gain is set by its corresponding 8-bit code. The gain code data are in unsigned binary format. Let the decimal equivalent of the control byte be N, N = 0 results in the mute condition, the input multiplexer is connected to the analog ground (AGNDR or AGNDL). For N in the range of 1 to 255, the gain in dB is given as 31.5 – (0.5 $(255 - N)$) [13]. This results in a gain range with a maximum of +31.5 dB (with N = 255) to −95.5 dB (with N = 1), in steps of 0.5 dB [13]. If zero crossing pin (ZCEN) is enabled, the gain changes will be applicable with zero-crossing detection of input signal. It helps in minimizing audible glitches due to digital control of the gain [13]. The new gain setting will not be latched until either two zero crossings are detected on the corresponding channel, or a timeout period of 16 ms has elapsed without detecting two zero crossings. The ZCEN pin is connected to the digital 5 V supply to permanently enable this zero-crossing feature. The mute function can be activated by either pulling the MUTE input (pin 8) to low, or by setting the gain byte value for one or both channels to 0x00H. SDOUT can be used for cascading of the attenuators, and it is not used in our application. The chip select, serial clock, and serial data input of the attenuator are connected to pin RE0, pin RC3, and pin RC5 of the microcontroller U1, respectively.

4.9 Serial interface

After completing a test, the instrument saves the test results along with the test parameters in its internal memory. For this, 540 bytes of RAM space is used. Pressing of the PRINT key on the keypad, sends the 540 bytes of information to the PC via serial interface at 2400 baud rate over the RS 232 interface, as shown in Figure 4.9. A RS 232 interface is provided for connecting the TA microcontroller to a PC. The voltage levels are made compatible to RS 232 levels (± 12 V) by using U7 (MAX 232 from Maxim) [17]. The Tx (pin RC6) and Rx (pin RC7) of the TA microcontroller U1 are used for serial interfacing

4.10 Output selection and extension of the attenuation range

Digitally controlled analog switches are used to couple the stimulus or the masker to the right headphone, left headphone, or the bone vibrator. For this purpose, CD4053 analog multiplexer, triple single-pole double-throw (3x two-to-one) analog multiplexer has been used [18]. A separate analog multiplexer IC is used for coupling the stimulus and the masker to each of the three transistors. The multiplexer chip is powered by single ±5 V supply. It can

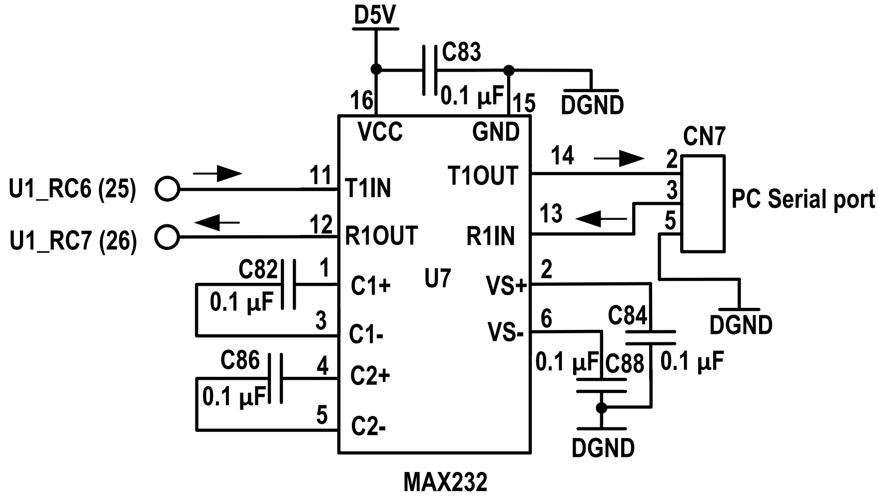


Figure 4.9 Circuit for RS232 interfacing

handle rail-to-rail signals [18]. We use current steering ensuring 0 V at the analog input and output of the multiplexer. In addition to selecting between the stimulus and the masker, this stage is also used for selecting the stimulus gain as 0 or 20 dB, thus extending the total range of the gain control.

A schematic of the signal selector is shown in Figure 4.10. Three switches 1, 2, and 3, having input resistors R_a, R_b, and R_c are controlled by controls *A*, *B*, and *C*, respectively. The output side of one of the poles of each of the switches are connected to the inverting terminal of the op-amp having feedback resistance R_d. The second pole of each of the switch is grounded. The switches 1 and 2 get analog input V_S and switch 3 gets analog input V_M. According to the logic levels at digital controls A, B, and C, the output of the op-amp will be

$$Vo = -V_s\left(\frac{R_d}{R_a}\right)A - V_s\left(\frac{R_d}{R_b}\right)B - V_M\left(\frac{R_d}{R_c}\right)C \quad (4.1)$$

R_d / R_a =10 results in gain of 20 dB, while R_d / R_b = R_d / R_c = 1 results in gain of 0 dB. The circuit for the multiplexer / amplifier for the three transducers is shown in Figure 4.11. Each transducer multiplexer / amplifier uses one CD4053. The resistors are selected for gain of 20 dB and 0 dB for the stimulus and 0 dB for the masker. Since the MCP602 IC has 2 op amps, only 2 ICs (U11 and U13) are used for extra attenuation [16]. One of the op amps in U11 along with the multiplexer U9 is used for getting 0 dB stimulus, 20 dB stimulus, or 0 dB masker for the right headphone. The second op-amp in U11 along with the multiplexer U10 is used for getting 0 dB stimulus or 20 dB stimulus for the bone vibrator. One of the op amps in U13 along with the multiplexer U12 is used for getting 0 dB stimulus, 20 dB stimulus, or 0 dB masker for the left headphone. Each of the summing amplifiers has a 3 dB low-pass cutoff frequency of 159 kHz for suppressing high frequency noise and it has no effect in the audiometric frequency range.

The digital controls, used in selecting the stimulus or the masker noise are given from the TA microcontroller. As the TA microcontroller does not have sufficient number of I/O pins for controlling all the peripherals interfaced in this application, a latch U8 (74HC373) is used to multiplex the data pins (RD0-RD7) for controlling the attenuation. This can be easily accomplished because display has an internal latch. Data to the LCD are written by first disabling the latch and then enabling the display chip select. Similarly, the data are sent to the latch by disabling the display chip select followed by enabling the latch [19]. The 8 control pins of the three multiplexers are connected to the microcontroller U1 via the latch as shown in Table 4.1. Interfacing of the latch U8 to the microcontroller U1 is shown in Figure 4.12.The latch outputs are given as selector inputs to the three multiplexers to select the input among three input signals i.e. the two stimuli and the masker.

4.11 Power amplifier

The output signals at the output of the three multiplexers need to be amplified to drive the headphones and the bone vibrator. In audiometric application, we normally need only two amplifiers, one for the stimulus and one for the masker. However, in order to avoid switching of the amplified signals using relays, we have opted for using three power amplifiers, one for each transducer. LM1877 was selected for this purpose. It is a monolithic 2-channel amplifier. It can deliver 2 W per channel continuously into 8 Ω load. It has wide supply range, 6 V-24 V, low cross-over distortion, AC short circuit protection and -65 dB channel separation [20]. Only one channel of the IC is used, in order to avoid the possibility of cross talk between the stimulus and the masker. Power amplifier circuits for the right headphone, the left headphone and the bone vibrator are shown in Figure 4.13, Figure 4.14, and Figure 4.15, respectively. The circuit is operated using dual $\pm$ 5 V supply. Each LM1877 is operated in inverting unity gain amplifier mode. The parallel resistive network formed by R6, R9, R11 and R13 (From figure 4.13) gives an output impedance of 8 Ω and it also serves as a current limiter against accidental short circuit in the output cables.

4.12 Response button

For receiving the response from the patient, a response button is interfaced to pin RB3 of U1. The subject response button is connected to the audiometer circuit using a 2-wire cable as shown in Figure 4.16. The combination of R66, D1, D2, and C87 is used to suppress any possible spikes which may get picked up in the switch cable. LED 2 is connected in parallel to the switch to acknowledge the button press. It is turned on for 2 s after a valid pressing of the response button. Normally pin U1_RB3 is in the input mode and passively pulled up. As R66 is high, the LED is not on. When the subject response is expected, the status of pin RB3 is read every 50 ms. If the response is received, RB3 is set in output mode with logic high. As soon as the subject releases the button, the LED becomes on. After 2 s, the port pin is reconfigured as digital input and LED is turned off. By using this arrangement, we are able to receive the subject response and provide acknowledgement without using an additional wire or using any circuit on the response button side.

4.13 Power supply

The audiometer board is powered by external ± 9 V dc supply through a 3-pin connector with 9 V, ground, and -9 V. From +9 V, the digital +5 V is generated using U17 (LM7805), and the analog +5 V is generated using U18. From -9 V, the analog -5 V is generated using U19 (LM7905). The analog and digital grounds are separated throughout the PCB and shorted only at the entry point of ± 9 V supply. For powering the SD card, and the digital part of the SG microcontroller, +3.3 V is generated by using U20 (LM1117) with the digital +5 V as the

Table 4.1 Connection of pins RD0 to RD7 of TA microcontroller U1 and the multiplexer control inputs

U1_Port D pin	Latch		Multiplexer control pin
	I/P	O/P	
RD0	D1	Q1	L_S 20 dB
RD1	D2	Q2	L_S 0 dB
RD2	D3	Q3	R_S 20 dB
RD3	D4	Q4	R_S 0 dB
RD4	D5	Q5	L_M 0 dB
RD5	D6	Q6	B_S 20 dB
RD6	D7	Q7	R_M 0 dB
RD7	D8	Q8	B_S 0 dB

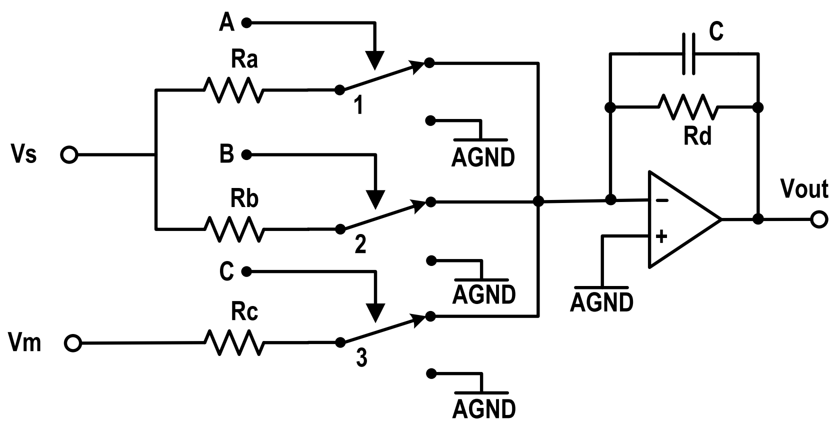


Figure 4.10 Schematic of the multiplexer / amplifier

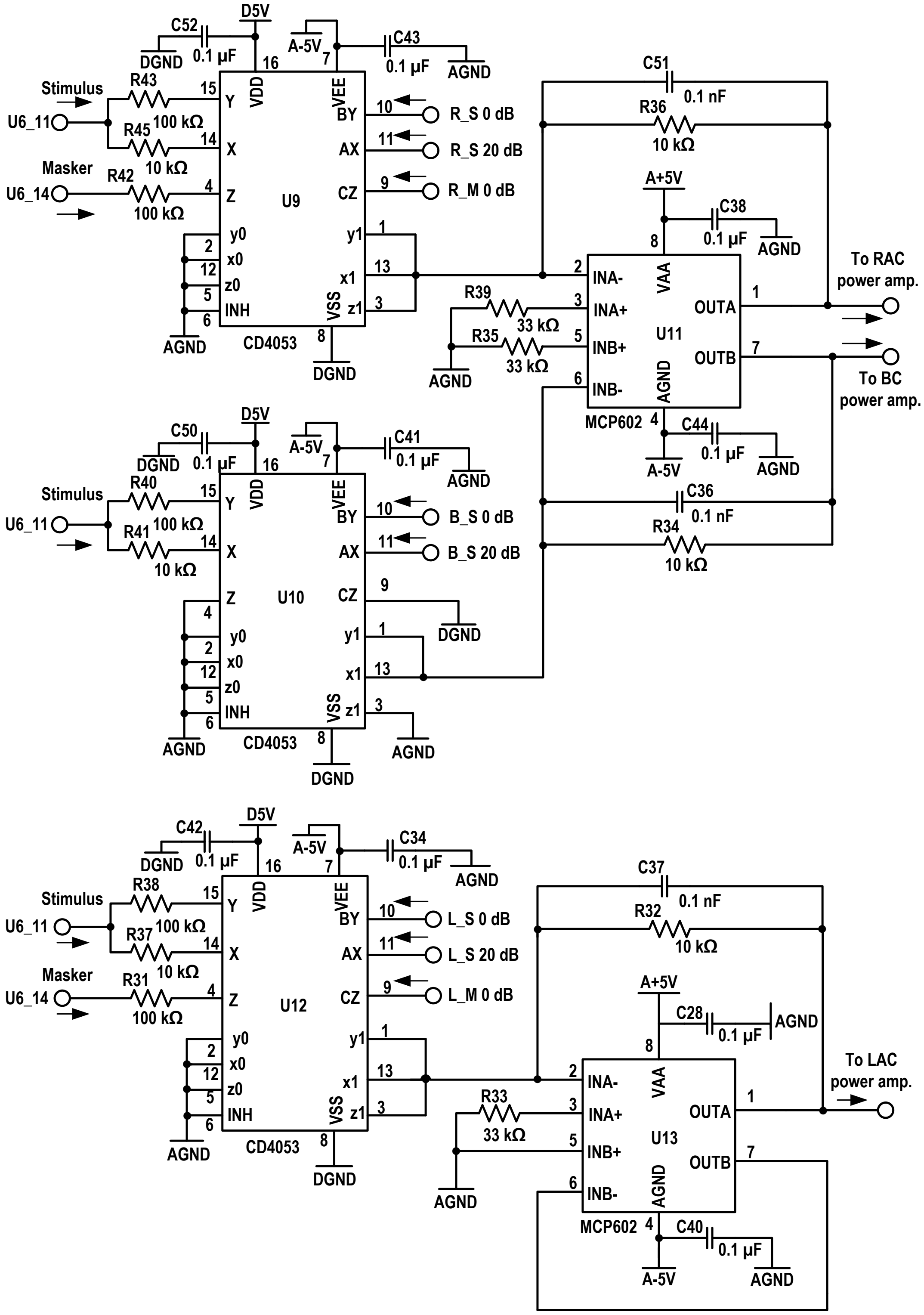


Figure 4.11 Circuit for multiplexing and summing amplifier

input. Similarly, for powering the analog section of the SG microcontroller, 3.3 V is obtained from analog +5 V as input to U21 (LM1117). An external dc/dc converter can be used for generating -9 V from +9 V supply. Status of the input +9 V is monitored by connecting the output of the attenuator formed by R63, and R64 as an ADC input to the ADC pin RB5 of the TA microcontroller U1.

4.14 Assignment of port pins of the microcontrollers

In the previous sections, interfacing of the various devices to the two microcontrollers, the TA microcontroller U1 and the SG microcontroller U2 has been described. The assignment of the two port pins of the two microcontrollers are listed in Table 4.2 and Table 4.3, respectively. Full schematic of the audiometer board is given in Appendix E.

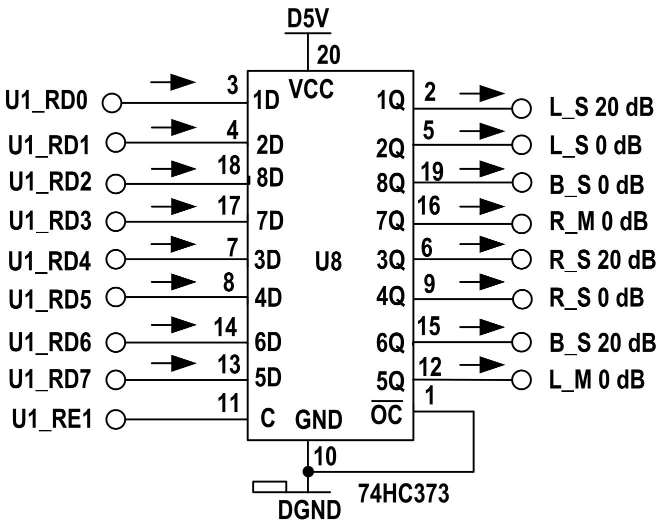


Figure 4.12 Interfacing of latch with the TA microcontroller

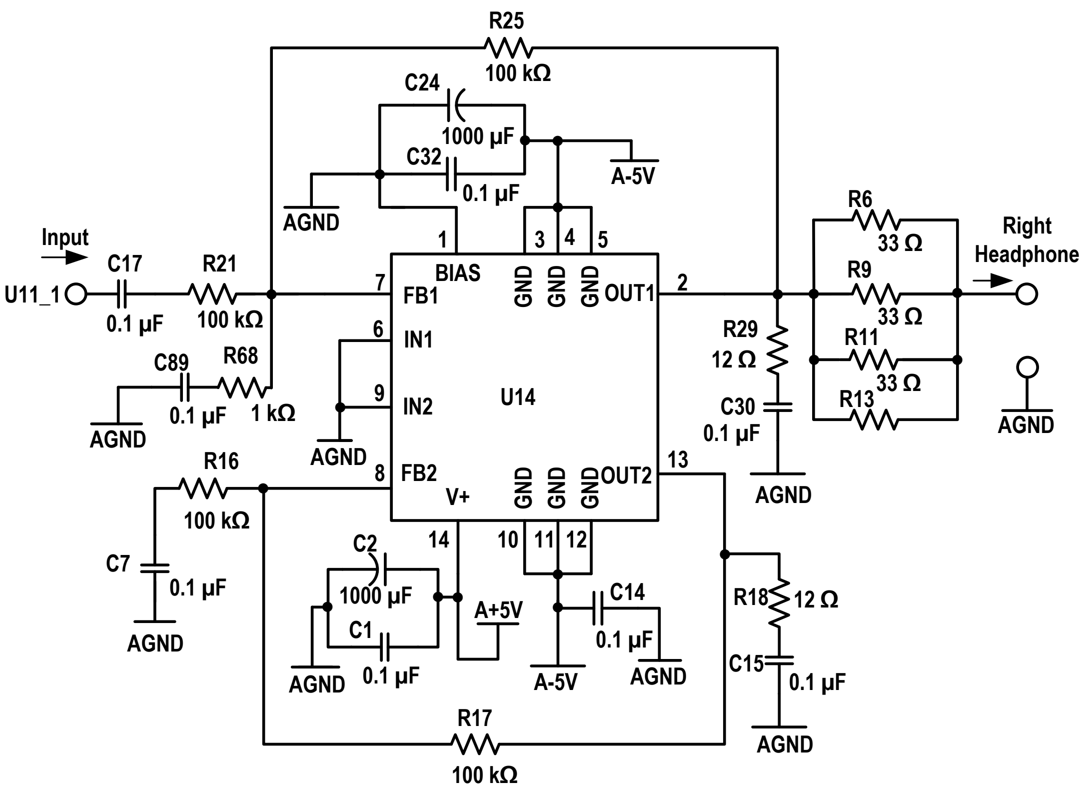


Figure 4.13 Power amplifier circuit for the right headphone

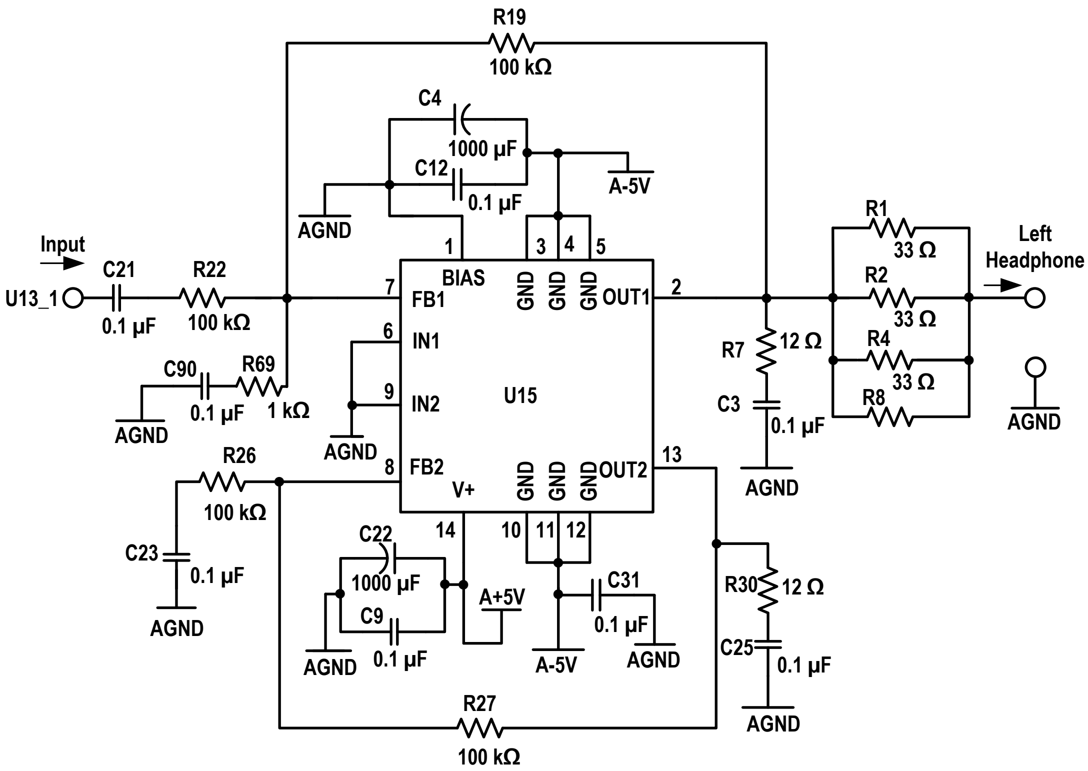


Figure 4.14 Power amplifier circuit for the left headphone

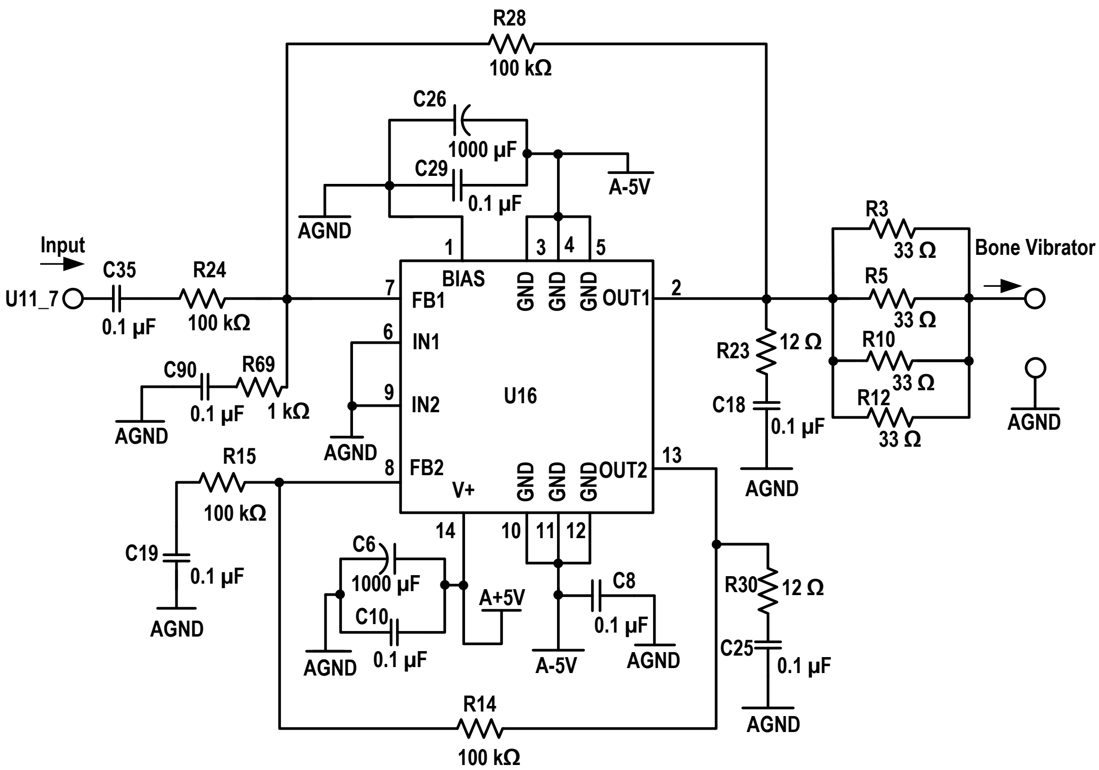


Figure 4.15 Power amplifier circuit for the bone vibrator (BC)

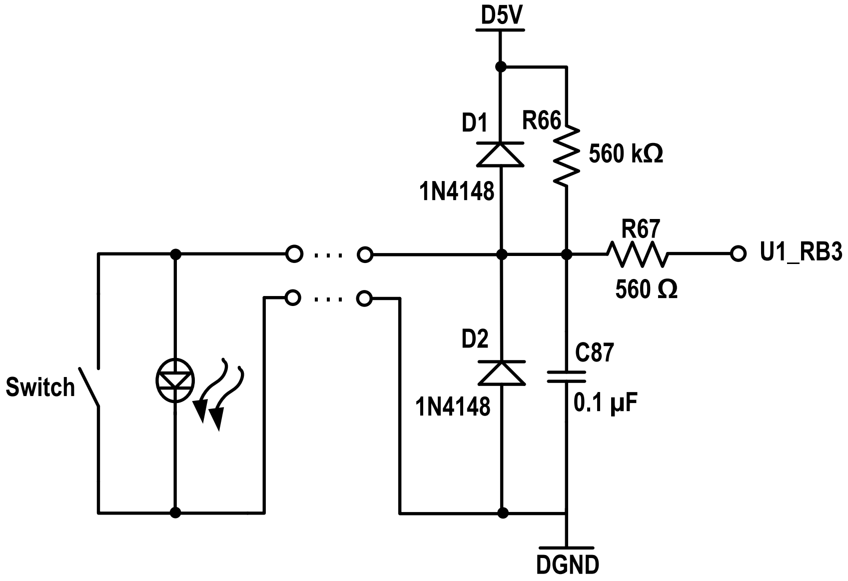


Figure 4.16 Interfacing of the response button and light indicator to the TA microcontroller

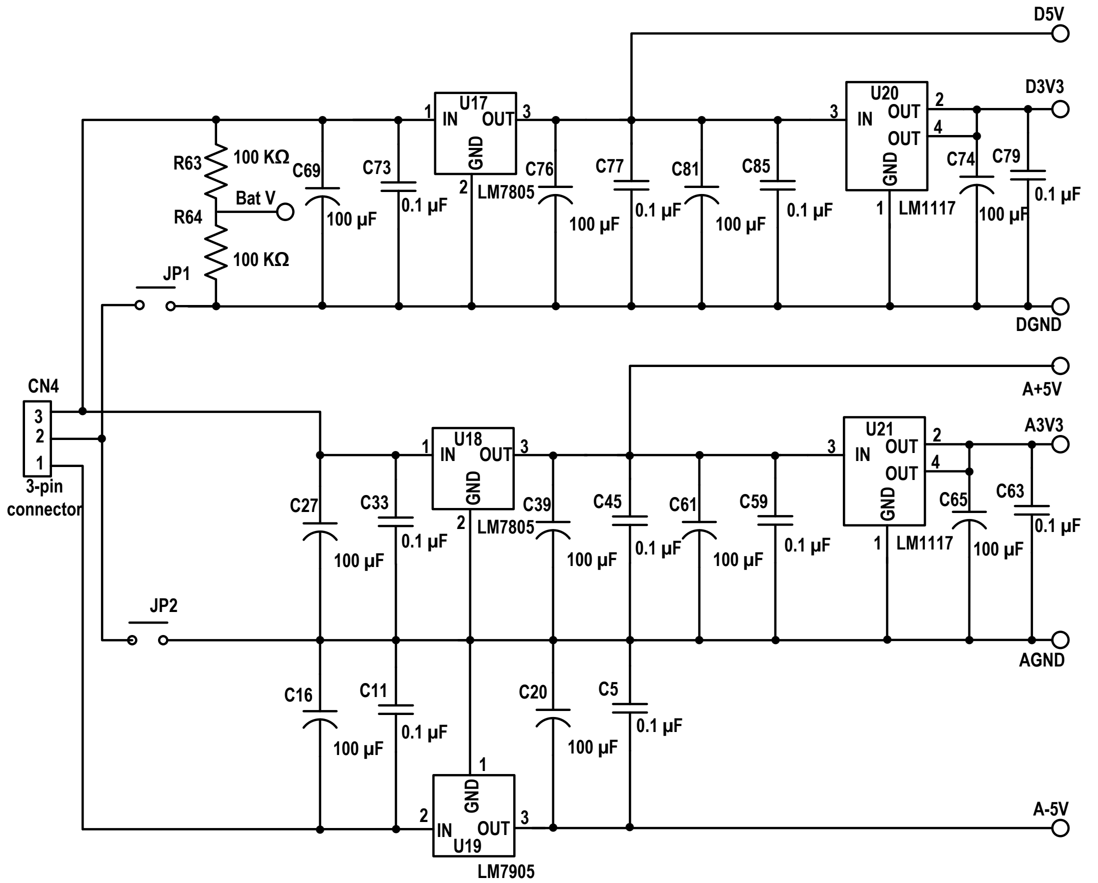


Figure 4.17 Power supply circuitry

Table 4.2 Function assigned to the I/O port pins of the TA microcontroller U1, (PIC16F1939)

I/O Port pins	In or Out	Function assigned
RD0-7	Out	Data bus for display and latch inputs
RC0	Out	Data/command selection pin of the display
RC1	Out	Enable pin of the display
RC2	Out	Chip select 1 pin of the display
RB1	Out	Chip select 2 pin of the display
RB2	Out	Reset pin of the display
RA0-3	In	Keypad row pins
RA4-7	Out	Keypad column pins
RB6	In	In circuit debugger clock pin
RB7	In	In circuit debugger data pin
RE3	In	MCLR pin of the debugger pikit3
RE0	Out	Chip select pin of U6 (PGA2310)
RC3	Out	Clock input pin of U6 (PGA2310)
RC5	Out	Serial data input pin of U6 (PGA2310)
RC4	Out	Serial comm. data pin of U2 (dSPIC33fJ128GP802)
RB0	In	Receiving interrupt from U2 (dSPIC33fJ128GP802)
RB4	Out	Sending interrupt to U2 (dSPIC33fJ128GP802)
RB5	In	Battery sense voltage
RC7	In	Rx of MAX232
RC6	Out	Tx of MAX232
RE2	Out	Back-light enable/disable pin of the display
RE1	Out	Chip select pin of the latch 74HC373
RB3	In and Out	Response button pin

Table 4.3 Function assigned to the I/O port pins of the SG microcontroller, (dSPIC33FJ128GP802)

I/O Port pins	In or Out	Function assigned
RB4	Out	Chip select pin of SD card
RB9	In	Card detect pin of SD card
RB2	In	Write detect pin of SD card
RB3	Out	Serial clock pin of SD card
RB11	Out	Serial data input pin of SD card
RB10	In	Serial data output pin of SD card
RA4	Out	Sending interrupt to U1 (PIC16F1939)
RB5	In	Serial comm. data pin of U1 (PIC16F1939)
RB7	In	Receive interrupt from U1 (PIC16F1939)
RA0	In	Pre amplified speech input pin (ADC in)
RA1	In	Analog reference ground pin for speech input
RB1	In	In circuit debugger clock pin
RB0	In	In circuit debugger data pin
RB15	Out	Left DAC negative output
RB14	Out	Left DAC positive output
RB13	Out	Right DAC negative output
RB12	Out	Right DAC positive output

Chapter 5
SOFTWARE

5.1 Introduction

This chapter gives a description of the two microcontroller programs controlling the peripherals for conducting the audiometric tests and calibration of the instrument. The TA microcontroller is programmed to perform the following tasks: (i) scan the keys, (ii) update the display, (iii) encode and send the user specified stimulus and noise parameters to the SG microcontroller, (iv) control the levels of stimulus and masking noise, receive the response from the response button, (v) send the feedback to response button LED, (vi) monitor the input power supply, (vii) monitor the stimulus level according to the test status, (viii) change the calibration settings, (ix) retrieve back the calibrated settings, (x) store the test result along with test parameters, (xi) communicate serially over RS232. The SG microcontroller is programmed to perform the following tasks: (i) get the encoded command byte from TA microcontroller, (ii) decode the byte, (iii) generate masking noise according to the decoded data, (iv) get the samples of stimulus tone from SD card, (v) output the samples of stimulus tone and masking noise word through DAC, if speech audiometry is selected, input the external speech through its ADC and output it on its DAC.

The two microcontroller programs are written in C. Each code is divided into two parts namely the main program and the interrupt service routine (ISR). The TA microcontroller's main program starts by setting the system clock, initializing the I/O port pins, retrieving back the calibration settings, initializing the default test parameters, and sending command code to the SG microcontroller, displaying default parameters and their default values on the graphical display, setting the default attenuator, and setting output selector. Subsequently it periodically scans the input keys, updates the variables, sends the command byte to the SG microcontroller, changes the level of attenuation, selects the transducer, and updates the display. The SG microcontroller's main program starts by setting the system clock, initializing the I/O port pins, initializing the default test parameters, retrieving back the sample bytes of default stimulus tone from the SD card, generating random noise word, outputting them sequentially and periodically through DAC. Subsequently it periodically scans for new encoded byte for stimulus tone and masking noise specifications, decodes the byte, updates the test parameters, gets the sample bytes from the SD card and outputs through DAC for new tone, and generates the masking noise word and outputs through DAC. The ISR program updates the values of running variable so that subsequent operation will be performed in the main program's "while" loop. The various modules in the two programs are described in the subsection of the next section. This is followed by a description of the audiometric test algorithms, operation, and calibration.

5.2 Software modules

5.2.1 Keypad

User interface is achieved by interfacing a keypad. The keypad module enables the following operation: (i) the selection of test ear, conduction type, test type, stimulus type, masker type, and tone frequency, (ii) setting of the tone and noise level, (iii) turning off the test, (iv) saving, clearing, and sending the test results (v) toggling display back-light (vi) calibration simultaneous pressing of OFF key and response button.

Row-column scanning algorithm, shown in Figure 5.1, is implemented for scanning the keypad. It scans the keypad every 10 ms using its TIMER2 interrupt. RA0-RA3 pins (which are connected to column pins of the keypad) are configured as output and PORT A.4-7 (which are connected to row pins of the keypad) are configured as input to U1. In this algorithm, U1 writes low on of the column pins and high on the rest of the three column pins and reads the input pins. The row pins are sequentially checked and the first row pin found low indicates that the corresponding row-column key is pressed. After a key press on a column is detected, the remaining columns are not scanned. Hence if multiple keys get pressed, the key on the lowest column number with the lowest row number will be accepted. For debouncing, a key press is accepted if it is detected in two consecutive scans. After being accepted, it is ignored in the subsequent scans, until a no key press or a new key press is detected.

Various key function codes are shown in Table 5.1. The functions assigned to the keys are given below.

- **FUNC key:** To reduce the number of overall keys, the selection of the test ear (left / right) and the conduction (air / bone) has been combined as selection of the audio function using a single key. This can be used to select the combination of the test ear and the conduction: the right ear with air conduction (RAC), left ear with air conduction (LAC), right ear with bone conduction (RBC), and left ear with bone conduction (LBC). The desired audio function can be selected by repeated pressing of the key. This key is active only when no test is running.
- **TYPE key:** This key is used to select the type of test: manual tone audiometry (TONE-M), automated tone audiometry (TONE-A), short increment sensitivity index test (SISI), tone decay test (TDECAY), and speech audiometry (SPEECH). The selection is in a cyclic manner. This key is active when no test is running.

- **STIM key:** This key (active when no test running) is for selecting of the stimulus type: pure tone (PURE), warble tone (WRBL), pulsed tone (PULS) and speech (SPCH). It can be used for changing the stimulus from among the available alternatives for the selected test type.
- **NOISE key:** This key (active when no test running) can be used for selecting the type of the masking noise: wide-band noise (WB), narrow-band noise (NB), speech spectrum shaped noise (SN), and none (XX).
- **S.▲ key:** This key can be used to increase the stimulus intensity by 5 dB in the operation mode and by 0.5 dB in the calibration mode, unless it has reached the maximum level. This key is disabled during automated test.
- **S.▼ key:** This key can be used to decrease the stimulus intensity by 5 dB in the operation mode and by 0.5 dB in the calibration mode, unless it has reached the minimum level. This key is disabled during automated test.
- **F.▲ key:** This key (active when no test running) is for changing the current frequency to the next higher audiometric frequency unless it has reached the maximum frequency.
- **F.▼ key:** This key (active when no test running) is for changing the frequency to the next lower audiometric frequency, unless it has reached the minimum frequency.
- **N.▲ key:** This key (active when no test running) can be used to increase the current noise level by 5 dB unless it has reached maximum level.
- **N.▼ key:** This key (active when no test running) can be used to decrease the current noise level by 5dB unless it has reached the minimum level.
- **SAVE key:** This key (active when no test running) is for saving the current test result (hearing threshold level for tone audiometry, SISI score for SISI test, HL dB value for tone decay test, hearing threshold level for speech audiometry) along with the test parameters (stimulus type, noise type and noise level) as an array. It is inactive when a test is on.
- **CLEAR key:** This key (active when no test running) is for erasing the test results and to reset the device.
- **OFF key:** When a test is on, this key turns off the test. If a test is off, simultaneously pressing this key and the response button starts the calibration mode.
- **ON key:** This key (active when no test running) starts the selected audiometric test and disables FUNC, TYPE, STIM, NOISE, S.▲, S.▼, F.▲, F.▼, N.▲, N.▼, SAVE, CLEAR, PRINT keys. The keys S.▲ and S.▼ remain active in the case of manual tone audiometry.

- **PRINT key:** This key (active when no test running) sends the audiometric test results to PC via RS232.
- □ **key:** This key (always active) toggles the back-light of the display. This feature is introduced to save the battery power if the back-light is not needed.

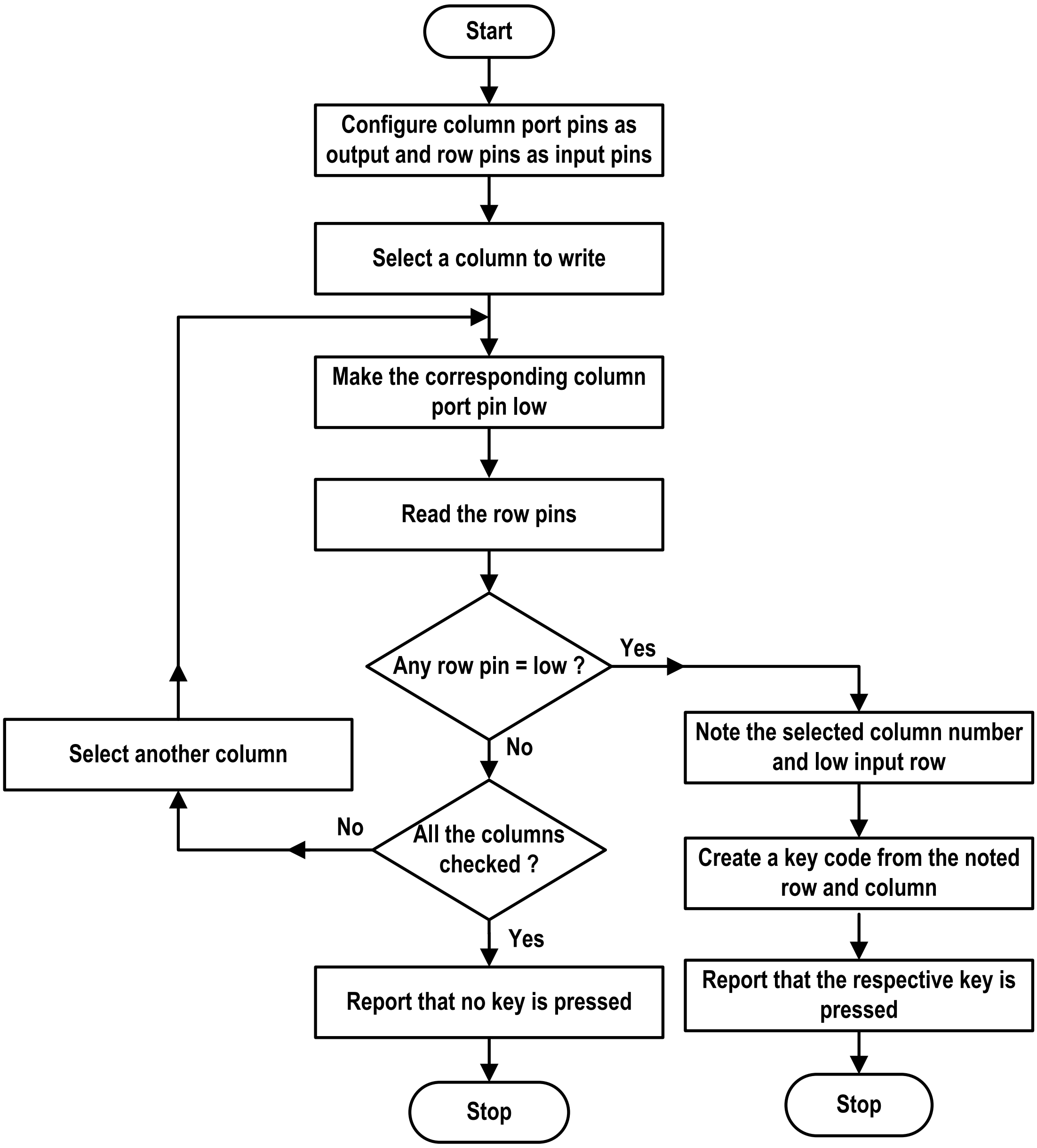


Figure 5.1 Flow chart for scanning the key press

Table 5.1 Data read for key presses

Key	Key code	Key	Key code	Key	Key code	Key	Key code
FUNC	0x38	TYPE	0x28	STIM	0x18	NOISE	0x08
S.▲	0x34	F.▼	0x24	**N.▲**	0x14	SAVE	0x04
S.▼	0x32	F.▼	0x22	**N.▼**	0x12	CLEAR	0x02
ON	0x31	□	0x21	OFF	0x10	PRINT	0x01

5.2.2 Display

The TA microcontroller program displays the user specified parameters, test status and saved tests and its results. The status of the battery power is also shown on the display. The level at the D/L pin of the LCD decides whether the incoming data through the data bus are data byte or command byte. If it is logic 0, the receiving byte will be a command byte. If it is logic 1, the byte will be a data byte. Thus D/L level should be correctly set before sending command and/or data bytes. Writing to the graphics LCD involves sequential operations of initialization, addressing the location, and writing set of graphics data bytes, shown as a flowchart in Figure 5.2. Initialization of LCD includes resetting, turning on and indicating display start line for the two chips. As mentioned in the hardware chapter, the LCD's pixels are logically grouped into 8 pages, each containing 8 rows and 128 columns. A column in a page contains 8 pixels and a single data byte can turn on/off one of these pixels. Addressing of each column in a page is by writing its location codes into the X and Y registers of the LCD controller. X register addresses the page number and Y addresses the column number [15]. For example, writing 0x00H into both X page and Y page registers will select the first 8 pixels of the first column, while writing 0x02H into X page register and 0x24H into Y register selects 17th to 25th pixels of 37th column. After setting the address location, sequence of graphics data bytes are written. The LCD controller selects the addressed location and the respective pixels are set according to the received data byte. The LCD controller automatically increments the column location after receiving the data byte for avoiding the need for sending the addresses during the sequential writing of columns in a page. LCD backlight is turned on by sending logic 0 and turned off by sending logic 1 on pin RE2.

A 5x7 matrix code is used for displaying each ASCII characters, in a 6x8 pixel area with each character taking 5 bytes. For example, character "B" is shown Figure 5.3. The bytes for all the ASCII characters are stored sequentially in the U1 microcontroller's program memory. A character is displayed by first retrieving the corresponding five bytes from the program memory and then sending them sequentially to the LCD controller. The status of battery voltage is indicated as a bar, as shown in Figure 5.4.

The display software is written for displaying 5 modes of display: (i) power-up display as shown in Figure 5.5, (ii) test mode display as shown in Figure 5.6, (iii) prompt for display to clear test results as shown in Figure 5.7, (iv) prompt for calibration mode as shown in Figure 5.8, and (v) calibration mode display as shown in Figure 5.9. It may be noted that Figure 5.6 and Figure 5.9 show examples of test and calibration mode respectively.

The display for the test mode is divided functionally into 3 fields: selection, status, and results. The selection field covers page 0 to page 3 of all the pixels of the display. In this field, some characters / words are permanent and some characters / words change. In this field all the test parameters i.e. the audio function, type of the audiometry, stimulus type, noise type, tone frequency, stimulus and noise intensity levels are displayed. The status of the test parameters while the test is running are shown using the lower right half pixels of the display i.e. status field. In tone audiometry, the tone level, number of presentations and responses for presented levels are displayed as status values. In the SISI test, the tone level, number of responses, and the total number of 1-dB increments are displayed. In tone decay test, the tone level is displayed. In speech audiometry, the present speech level is displayed as test status. After saving the test result and its parameters, the word "Saved" is also displayed in the status. Similarly "Cleared", "Sending" and "Sent" words are also displayed after clearing, after initiation of the sending of the test results to PC, and after completion of sending the test results, respectively. The saved tests for selected audio function and test type are displayed in the ascending order of the audiometric frequency (125 Hz to 8000 Hz) in results field, in the lower left half of the display. If a test is completed and saved for selected audio function, test type and/or frequency, the test result along with the test parameters by which the test is performed is also shown in this field. In the calibration mode, the selected audio function, tone frequency, and the attenuation number is displayed.

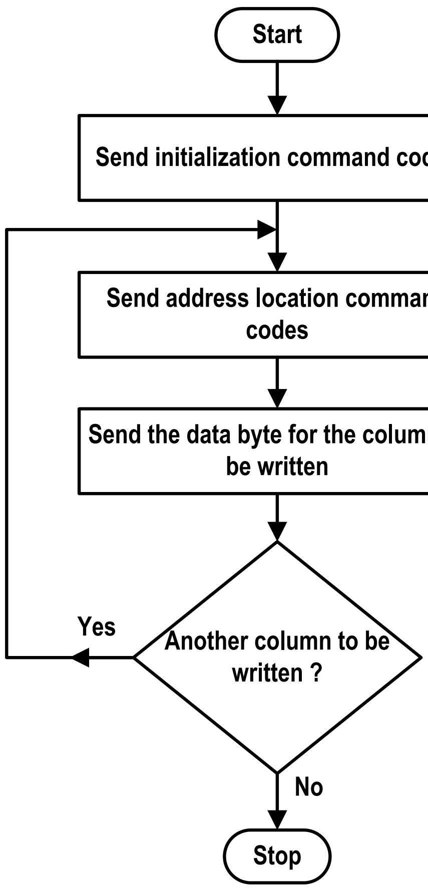


Figure 5.2 Flowchart for sequence of operations in writing to the LCD controller

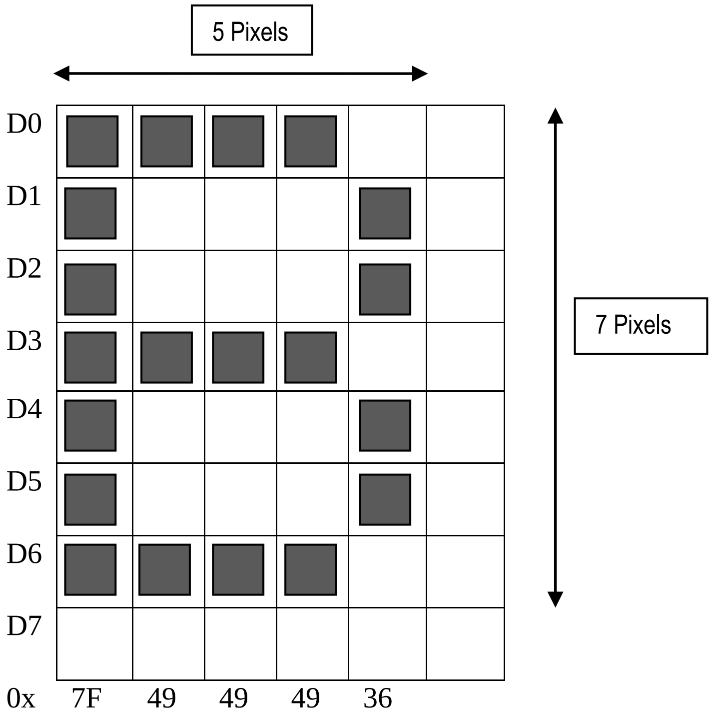


Figure 5.3 Graphical representation of character “B” (5x7 pixels in 6x8 pixel area)

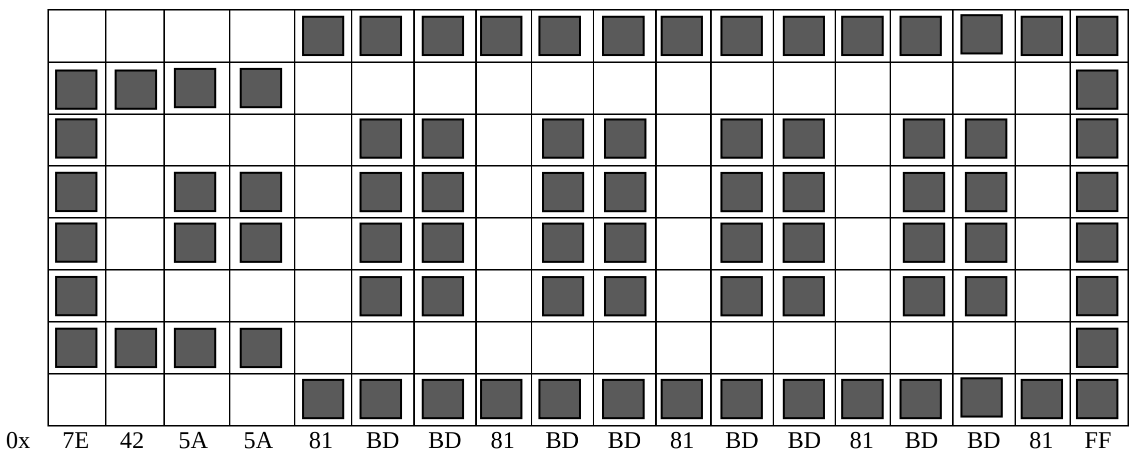


Figure 5.4 Graphical representation of the battery voltage

5.2.3 Inter-processor communication

A serial data transfer protocol is used for sending the user specified tone specifications from the TA microcontroller to the SG microcontroller. These specifications include stimulus type (pure tone, warble tone, pulsed tone, and external speech), tone frequency, noise type (wide-band noise, narrow-band noise, and speech spectrum shaped noise). These values are stored in three different 8-bit registers. To reduce the time taken in data transfer, the specifications are encoded into a single byte for sending to the SG microcontroller. The transfer uses asynchronous handshake mediated protocol. On the TA microcontroller side, it involves the cyclic sequence of write data bit, output data ready interrupt, and receive acknowledgement interrupt until all the bits have been sent. The SG microcontroller follows a complementary sequence of interrupt controlled operations, while still carrying out its routine operations of waveform generation. Both the interrupts are edge triggered at low to high transition.

U1 writes the first bit of controlling word on its pin RC4 and generates an interrupt by setting RB4 pin. After receiving the interrupt, U2 resets its acknowledgement pin, reads the data bit and stores it in a temporary register, and then sets its acknowledgement pin as an interrupt to U1 indicating that it is ready for the next bit. Upon receiving the acknowledgement, U1 resets RB4 and starts the cycle for writing the next bit.

```
     IIT  BOMBAY

AUDIOMETER  IITB-2K11

PRESS ANY KEY TO

     CONTINUE...
```

Figure 5.5 Power-up display

```
TYPE:TONE-M  FUNC: RAC
STIM:PURE    NOISE:XX
FREQ:1000
S-DB:30      N-DB:10


---- -------
.... .......
```

Figure 5.6 Test parameter selection with default values

```
  CLEAR TEST RESULTS

    PROCEED(ON)?

    CANCEL(OFF)?
```

Figure 5.7 Prompt display for clearing test results

```
    CALIBRATION MODE

     PROCEED(ON)?
      EXIT(OFF)?
```

Figure 5.8 Prompt display for calibration mode

```
    CALIBRATION MODE
FUNC:  RAC
FREQ:  125
ATT.:  XXX
```

Figure 5.9 Calibration mode display

The process is repeated by U1 until all the 8 bits have been written. This sequence of steps for transferring one byte of information from U1 to U2 is shown as flow charts in Figure 5.10, 5.11, and 5.12. The timing diagram is shown in Figure 5.13. With machine cycle frequency of 4 MHz and 40 MHz for the TA and SG microcontrollers, respectively, the minimum transition timings in the timing diagram are T_1min = 250 ns, T_2min = 75 ns, T_3min = 25 ns, T_4min = 25 ns, T_5min = 750 ns, T_6min = 250 ns.

5.2.4 Signal generation

Audiometric stimulus tones can be generated digitally by using a microcontroller having sufficient internal memory and a DAC. The samples of one cycle of the waveform are stored and the stored samples are output sequentially through the on-chip DAC at the required sampling rate. The continuous waveform is generated by looping this output operation. For getting waveform with low distortion, we need to use a sufficiently high sampling rate, number of quantization bits in the samples and the DAC.

Noise can be generated digitally by synthesizing a pseudo-random binary sequence which can be realized using a shift register of *N* bits, with serial input generated by a certain combinational operation on certain bit positions [9]. The resulting sequence of 1's and 0's after low-pass filtering generates noise with white spectrum from f_c / (2^N-1) to 0.12 f_c, where f_c

is the clock used for shifting the register [9]. The pseudo random binary sequence gets repeated after 2^N-1 bits. Use of 16-bit register requires XOR operation on 3rd, 12th, 14^{th},

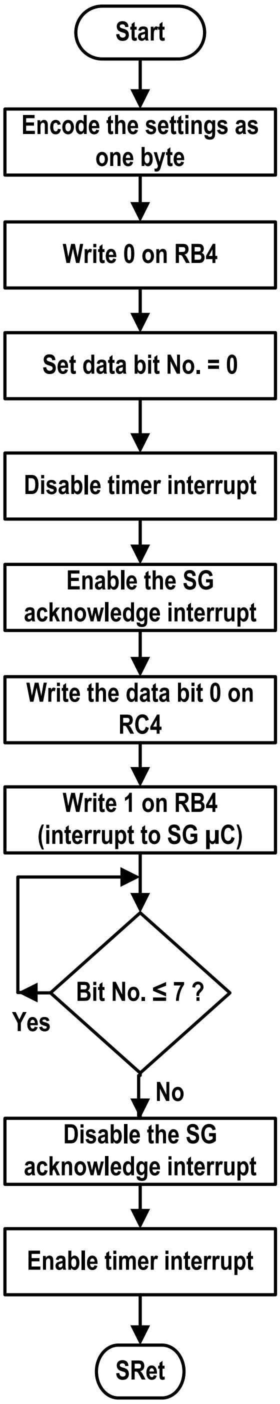


Figure 5.10 Flowchart of TA µC subroutine for serial communication

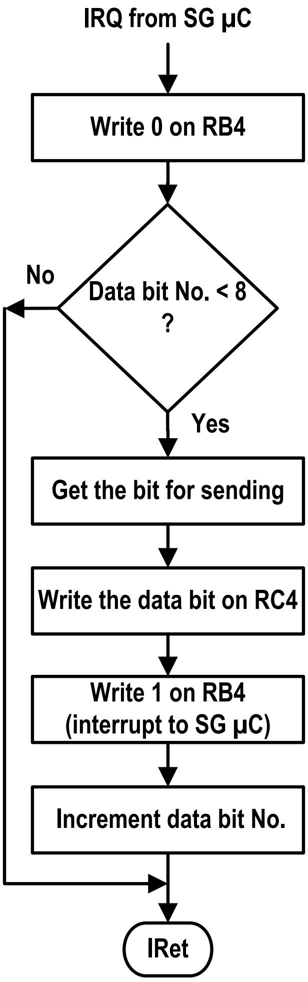


Figure 5.11 Flowchart of TA µC ISR for serial communication

and 15th bits as input to the shift register [9]. The register bits together as taken as a 16-bit random number for digital filtering and the filtered samples are output through the DAC.

In our hardware, Microchip's dSPIC33FJ128GP802 microcontroller is used for generating the audiometric stimulus tones and masking noise. It has the following features [12].

- Operating range is up to 40 MIPS.
- 16-bit wide data path, high performance DSC CPU.
- On-chip 128 Kbytes of flash program memory and 16 Kbytes of data SRAM.
- Fully integrated phase-locked loop (PLL)
- Two SPI modules which support 8-bit and 16-bit data
- 16-bit dual channel, audio digital-to-analog converter (DAC).

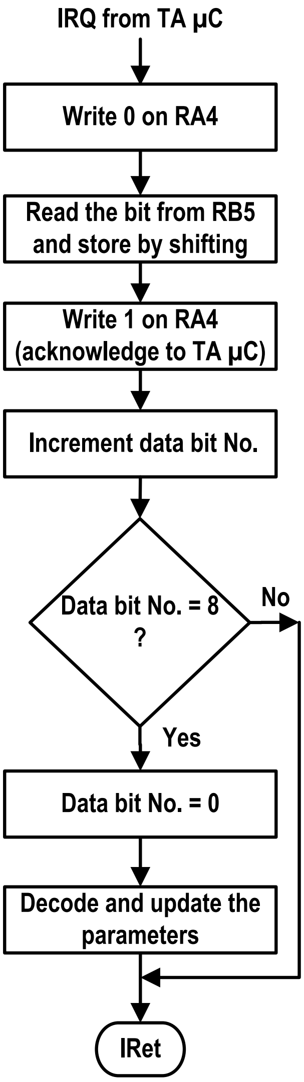


Figure 5.12 Flowchart of SG µC ISR for serial communication

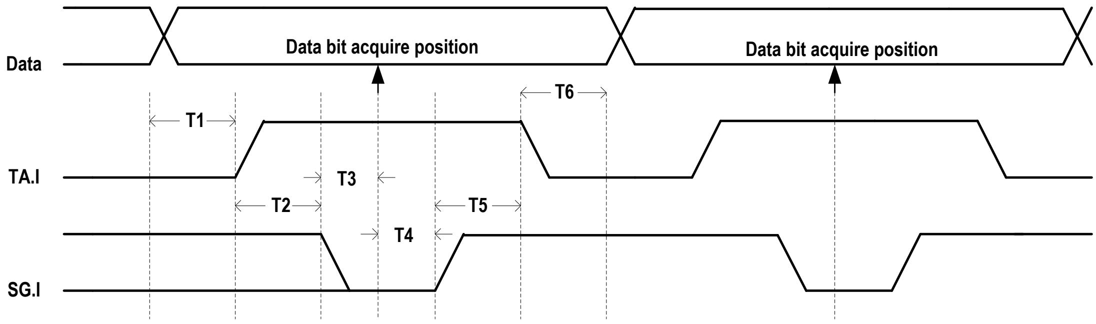


Figure 5.13 Timing diagram of asynchronous serial communication

The DAC module of the SG microcontroller has two output channels, left and right to support stereo applications. The stimulus is output through the right channel DAC and the masker is output through the left channel DAC.

Since the stimulus waveforms are periodic, the waveform can be generated by concatenating a single cycle of stored waveform. Samples of single cycle pure tone, warble tone and pulsed tones for all the audiometric frequencies were generated using MATLAB. For pure tones, 16-bit samples are used, while 8-bit samples are used for warble and pulsed tones. The pure tone samples are generated as

$$x(i) = \text{round}\left(32767.5(1 + \cos(2\,\pi\, i\, f_0/F_s))\right) \tag{5.1}$$

where f_0 = frequency, F_s = sampling frequency, and i = sample number in a cycle of the waveform.

The warble tone is a tone whose frequency varies periodically over a small range about a center frequency f_0. The equation used for generating the samples is

$$x(i) = \text{round}\left(127.5\left(1 + \cos\left(2\,\pi\, i\,(1 + \alpha \sin(2\,\pi\, i\;\, f_m/F_s)\, f_0/F_s)\right)\right)\right) \tag{5.2}$$

where αf_0 is the frequency deviation and f_m is the modulation frequency. The waveforms are synthesized using $\alpha = 0.05$ and f_m = 5 Hz.

The pulsed tone is obtained by amplitude modulating the carrier signal of pure tone with message signal of a square wave whose frequency is 4 Hz with rise and fall time of 50 ms each. The pulsed tone samples are generated as

$$x(i) = \text{round}\ (127.5(1 + a(i)\sin(2\pi i\ f_0/F_s))) \tag{5.3}$$

Where $a(i)$ is the amplitude envelope and is given as the following

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

where $N_1 = 0.05F_S$, $N_2 = 0.25F_S$, $N_3 = 0.3F_S$, and $N_4 = 0.5F_S$ -1, for a rise and fall time of 50 ms. A single cycle of the AM wave is shown in Figure 5.14. To avoid an apparent amplitude modulation, the pure tone generated by repeatedly concatenating a cycle, the cycle should have a multiple of 4 samples, i.e. F_S should be selected such that F_S / f_0 is a multiple of 4. Also the DAC output should have a smoothening filter with a cutoff lower than F_s/2. We are using the response of the power amplifier and the headphone as the smoothening filter. Since the headphones generally have a low-pass response with cutoff of about 12 kHz, we cannot use sampling frequency less than 24 kHz. Choice of F_S as used for pure, warble, and pulsed tones of different frequencies and the number of samples is given in Table 5.2.

As it was not possible to store the samples of one cycle of all the stimuli in the program memory of the SG microcontroller, the samples are stored on the SD card. In case of pure tone, one full cycle of the tone is stored on the SD card. At the time of presentation, the stimulus is read from the SD card into the RAM of SG microcontroller and it is presented by repeatedly concatenating the cycle. In case of pulsed tone, the RAM is not sufficient for storing all the samples in the ON period of one cycle. Hence half the samples are stored and the remaining samples are generated as the mirror image. This is followed by the required number of zero valued samples to complete one cycle of the pulsed tone.

File I/O functions for the SD card are implemented using Microchip's memory disk drive (MDD) file system library in accordance with the ISO/IEC 9293 specifications, for writing and reading the data from the SD card [21]. The read and write operations are through SPI protocol, using four basic pins: serial data in (SDI), serial data out (SDO), serial clock (SCK), and chip select (CS). Additionally, the SD card has card detect, and write detect pins which can be used to determine if the card is physically inserted and / or write protected [21]. All the samples for each stimulus are stored as a text file in the SD card. These samples are retrieved back into the microcontroller's SRAM and output cyclically through its right channel DAC. For speech audiometry, the ADC is used to acquire the external speech input with a sampling rate of 20 kHz and output through the right channel DAC.

The pseudo-random sequence is generated using 16-bit shift register. Analog wide-band noise is generated by outputting the random sequence through left channel DAC at the same sampling rate as the stimulus. If the selected masking noise is narrow-band noise, the pseudo random sequence is band-pass filtered before outputting. The digital band-pass filter coefficients are pre-calculated for all audiometric frequencies and stored in the SG microcontroller's program memory. For the tone frequency of 125 Hz, we use a cascade of two second order Butterworth low-pass filters such that the resultant 3-dB cutoff frequency is $2^{.0.25}\times 125$ Hz. For all the other tone frequencies, we use a cascade of two second-order Butterworth band-pass filters such that the resultant lower and upper 3-dB cutoff frequencies are $f_c/2^{0.25}$ and $2^{.0.25}$ f_c where f_c is the tone frequency. The difference equations used in generating narrow-band noise $y(n)$ from input white noise $x(n)$, using filter coefficients a_1, a_2, b_0, b_1, b_2 are as the following

$$v(n) = [b_0 x(n) + b_1 x(n-1) + b_2 x(n-2) - a_1 v(n-1) - a_2 v(n-2)] / 2^{14} \tag{5.5}$$

$$y(n) = [b_0 v(n) + b_1 v(n-1) + b_2 v(n-2) - a_1 y(n-1) - a_2 y(n-2)] / 2^{14} \tag{5.6}$$

The cutoff frequencies and the filter coefficients for all the audiometric frequencies are given in Table 5.3. It may be noted that the coefficients given in the table used have been scaled by multiplying by 2^{14} for fixed-point processing and hence the filter outputs are scaled back by the same factor. The frequency responses of the filters are shown in Figure 5.15.

5.3 Test algorithms

Manual tone audiometry, automated tone audiometry, SISI test, tone decay test, and speech audiometry have been implemented, as separate subroutines for each test in test administration microcontroller. In manual mode, the level and presentation of the test stimulus is manually controlled. In automated mode, the instrument automatically controls the level and presentation of the test stimulus by monitoring the patient's response through the response button.

Table 5.2 Sampling frequency and number of samples for different tones

f_0 (Hz)	F_s (kHz)	No of samples		
		Pure (one cycle)	Warble (one cycle)	Pulsed (half on-period)
125	64	512	12800	9600
250	64	256	12800	9600
500	64	128	12800	9600
750	48	64	9600	7200
1 k	64	64	12800	9600
1.5 k	48	32	9600	7200
2 k	64	32	12800	9600
3 k	48	16	9600	7200
4 k	64	16	12800	9600
6 k	48	8	9600	7200
8 k	64	8	12800	9600

Table 5.3 Narrow-band noise filter coefficients

Tone Freq. (Hz)	F_1 (Hz)	F_2 (Hz)	b_0	b_1	b_2	a_1	a_2
125	--	149	1	2	1	-32414	16034
250	210	297	108	0	-108	-32542	16168
500	420	594	215	0	-215	-32299	15954
750	631	892	424	0	-424	-31766	15536
1 k	841	1189	424	0	-424	-31766	15536
1.5 k	1261	1784	827	0	-827	-30515	14729
2 k	1682	2378	827	0	-827	-30515	14729
3 k	2523	3568	1579	0	-1579	-27347	13226
4 k	3364	4757	1579	0	-1579	-27347	13226
6 k	5045	7131	2908	0	-2908	-18938	10568
8 k	6728	9514	2908	0	-2908	-18938	10568

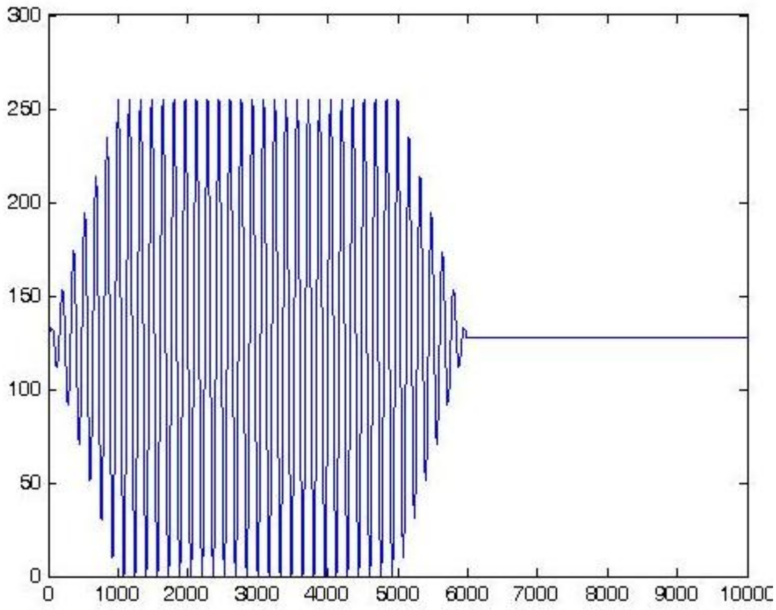


Figure 5.14 Single cycle of pulsed tone

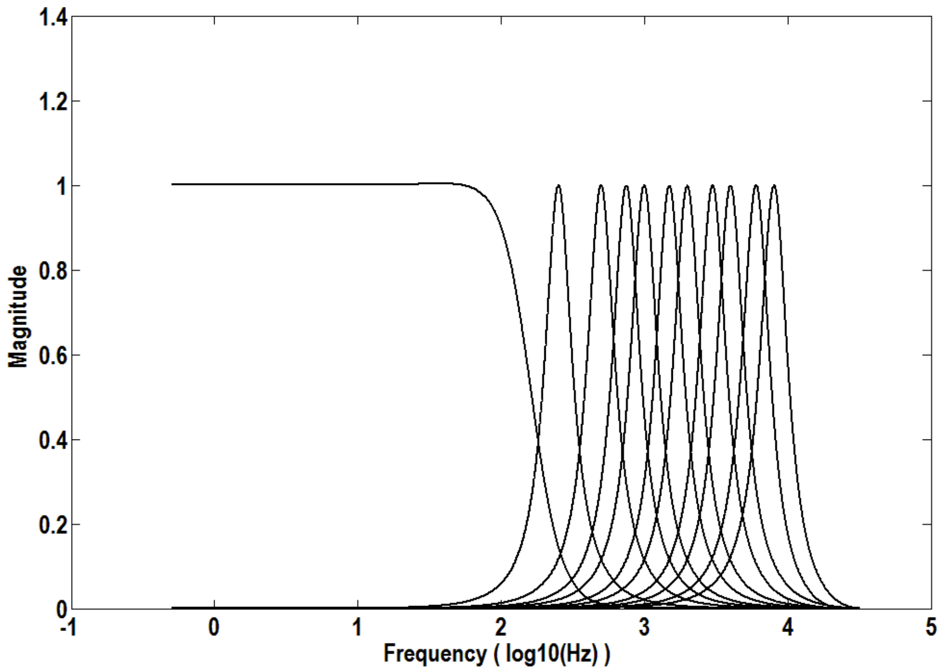


Figure 5.15 Magnitude responses of the filters for narrow band noise

5.3.1 Manual tone audiometry

Tone audiometry is used for finding the hearing threshold level (HTL) for tones at the audiometric test frequencies. The instrument permits selection of the test stimulus as pure tone (continuous), pulsed tone (amplitude modulated, AM), and warble tone (frequency modulated, FM). The patient is instructed to press the response button whenever he/she hears a tone in the test ear. The test ear (left / right), the type of conduction (air / bone), the masking

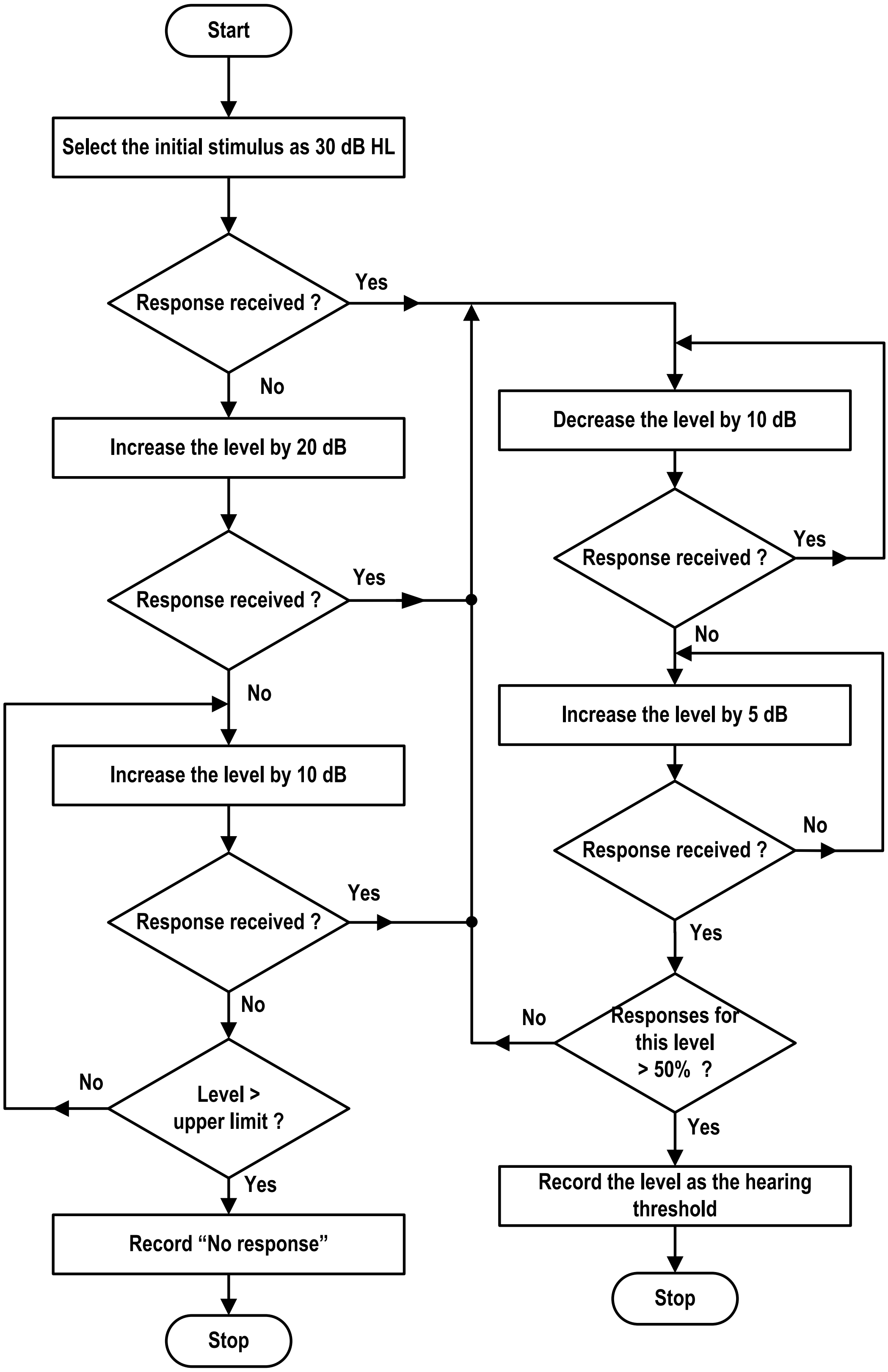


Figure 5.16 Flow chart for manual tone audiometry

type and level, and the stimulus frequency, and its type (pure/pulsed/warble), and the initial level are manually selected. In this mode, the tone is presented by pressing the ON key. It gets terminated if the patient response button is pressed or if the OFF key is pressed. The test level is manually changed by pressing the stimulus level up/down keys. The lowest level resulting in 50% response is taken as the HTL. Figure 5.16 shows the steps for finding the threshold at the selected frequency (using 5 dB up, 10 dB down method).

5.3.2 Automated tone audiometry

In this mode, once the ON key is pressed, the sequence of tone presentation and level is controlled by the instrument by monitoring the patient response button until the threshold level gets determined. The test progress (presentation level and patient response) is updated on the display, and the test can be terminated by pressing the OFF key. The subroutine for conducting this test is shown in Figure 5.17 as a flowchart.

5.3.3 SISI test (automated)

Short increment sensitivity index (SISI) test is used for determining the capacity to detect a brief 1-dB increment in a 20 dB supra-threshold tone. It is conducted with the pure tone as the stimulus. The stimulus level should be 20 dB supra-threshold for the selected tone frequency. The patient is instructed to press the response button whenever he/she hears a short duration increase in the level of the tone in the test ear. 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 after the ON key is pressed. A sequence of steps is followed automatically to find the SISI score, as shown in Figure 5.18 as a flow chart. During the test, the multiple presses between successive increments are ignored. The test is continued for 20 such increments and the percentage of the responses is displayed as the SISI score. 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 and conduction. The test may be terminated prematurely by pressing the OFF key.

5.3.4 Tone decay test (automated)

Tone decay test is used for determining the continuous listening capability for about 1 min. It is conducted with the pure tone as the stimulus. The stimulus level should be 10 dB below the hearing threshold for the selected tone frequency. The patient is instructed to press the response button whenever he/she hears a tone for the first time in the test ear and then again whenever the tone disappears. 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

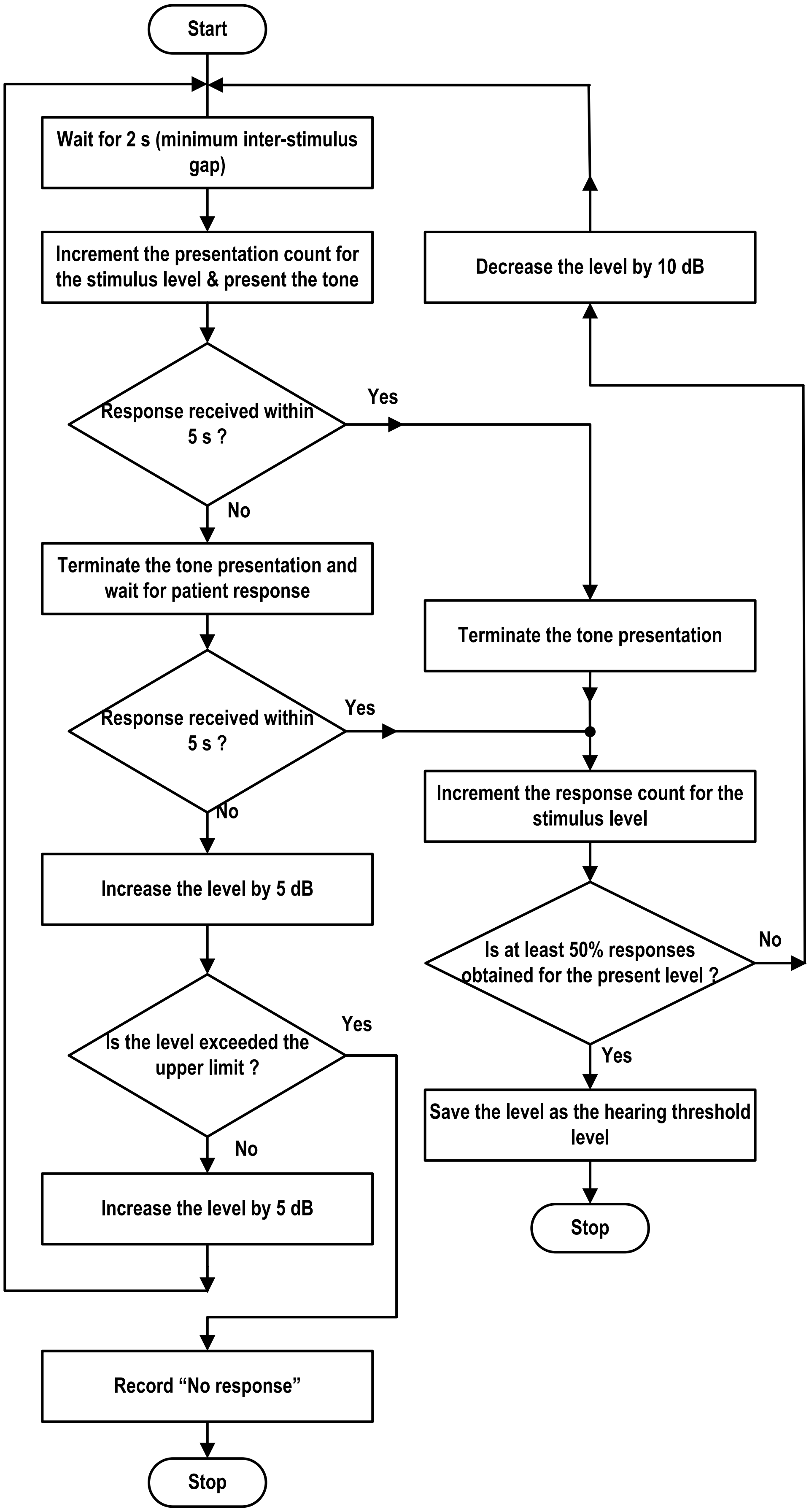


Figure 5.17 Flow chart for automated tone audiometry

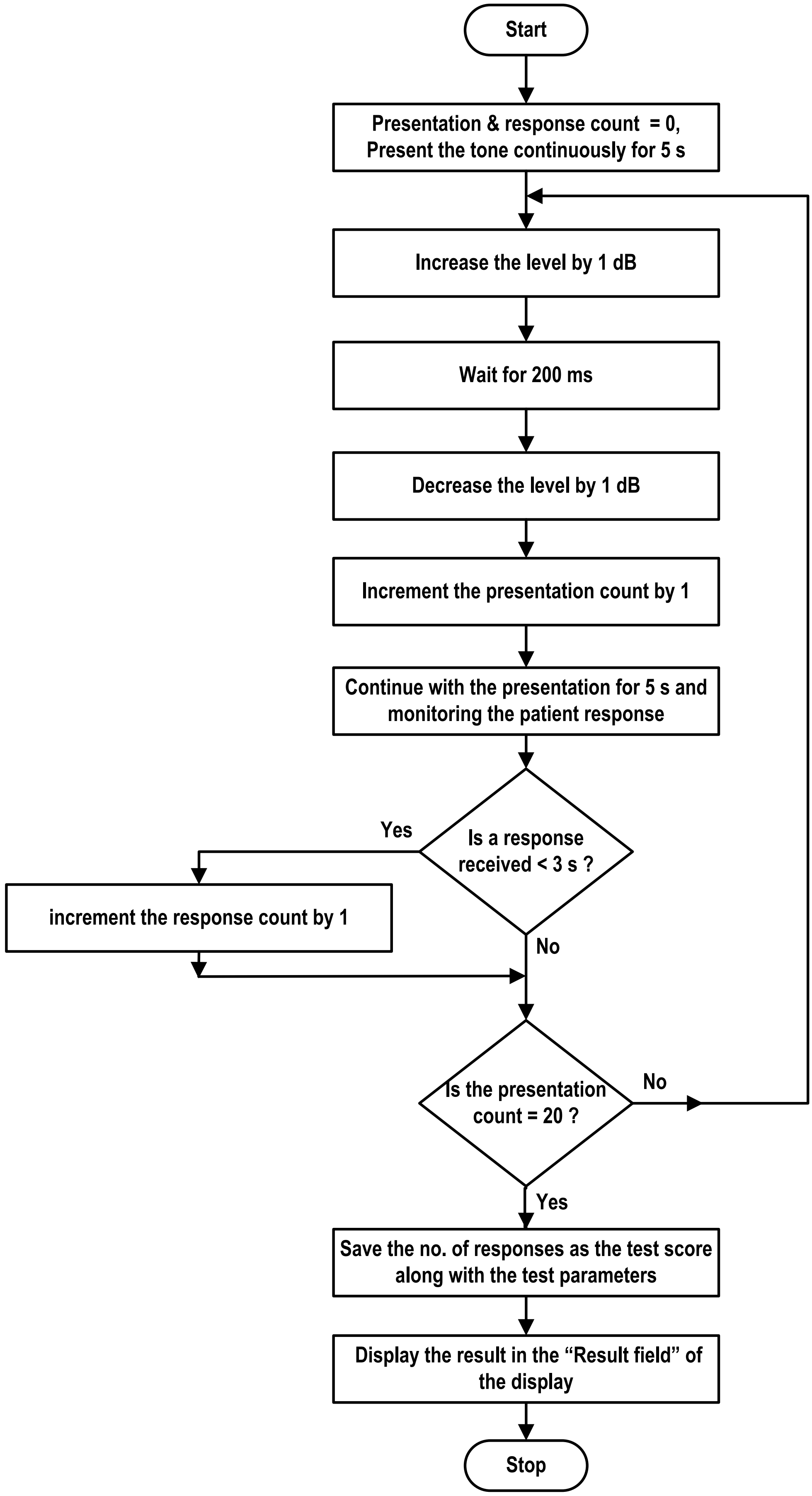


Figure 5.18 Flow chart for SISI test

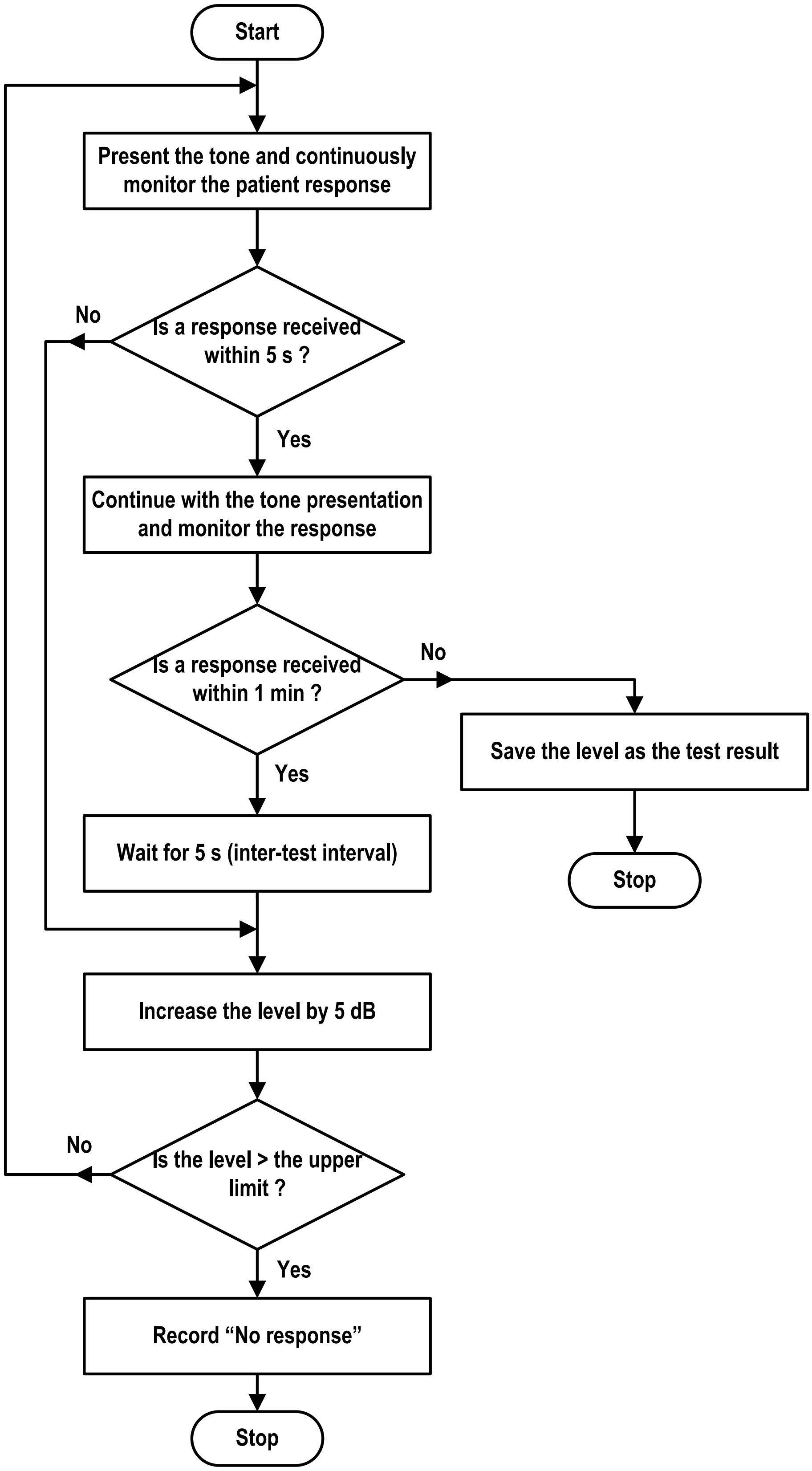
tone presentation starts after the ON key is pressed. At intervals of 2 s, the stimulus level is increased in steps of 5 dB until the patient response is received. The presentation is continued for 1 min. and if the second response is not received, the tone is terminated and the level is displayed as the test result. If the second response is received before the lapse of 1 min, it indicates that the stimulus became inaudible and hence the test is repeated by increasing the level by 5 dB. During the test, the status of presentation level and patient response is updated on the display. The final level along with the test parameters (noise type and its level) are saved automatically as the result for the selected frequency and for the selected test ear and conduction. The test can be terminated by pressing the OFF key. The flowchart representation of the tone decay test procedure is shown in Figure 5.19.

5.3.5 Speech Audiometry (manual)

Speech audiometry is commonly 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 tone audiometry. The patient is instructed to repeat the word as heard in the test ear. The test ear (left / right), the type of conduction (air / bone), the masking level, and the speech sound level are manually selected. The sound presentation starts after the ON key is pressed. The level is manually decreased in steps of 5 dB until a level is reached at which the patient identifies 50 % of the words correctly. In certain cases, all the words may be correctly identified at a particular level, but the score may be much lower than 50 % at the next lower level. In such cases the level at which all the words or more than 50 % of the words are identified correctly is taken as the SRT. In some cases, there may be a need to increase the presentation level for getting 50% response. The test can be terminated at any time by pressing the OFF key.The final level along with the test parameters (noise type and its level) generally conducted with the speech-spectrum shaped noise as the masking noise. The test is can be saved as the result for the selected frequency and for the selected test ear and conduction. The final level along with the test parameters (noise type and its level) are saved manually as the result for the selected frequency and for the selected test ear and conduction. The sequence of steps, written in a subroutine program, for conducting the test in automated mode, are represented as a flow chart as shown in Figure 5.20.

5.4 Operation sequence

The interface for administering the test is through a 4x4 keypad and a 128x64 graphical LCD. The keys are shown in Figure 4.1 and the selection of the test parameters through the keypad is given in Table 5.4.



Figue 5.19 Flow chart for Tone decay test

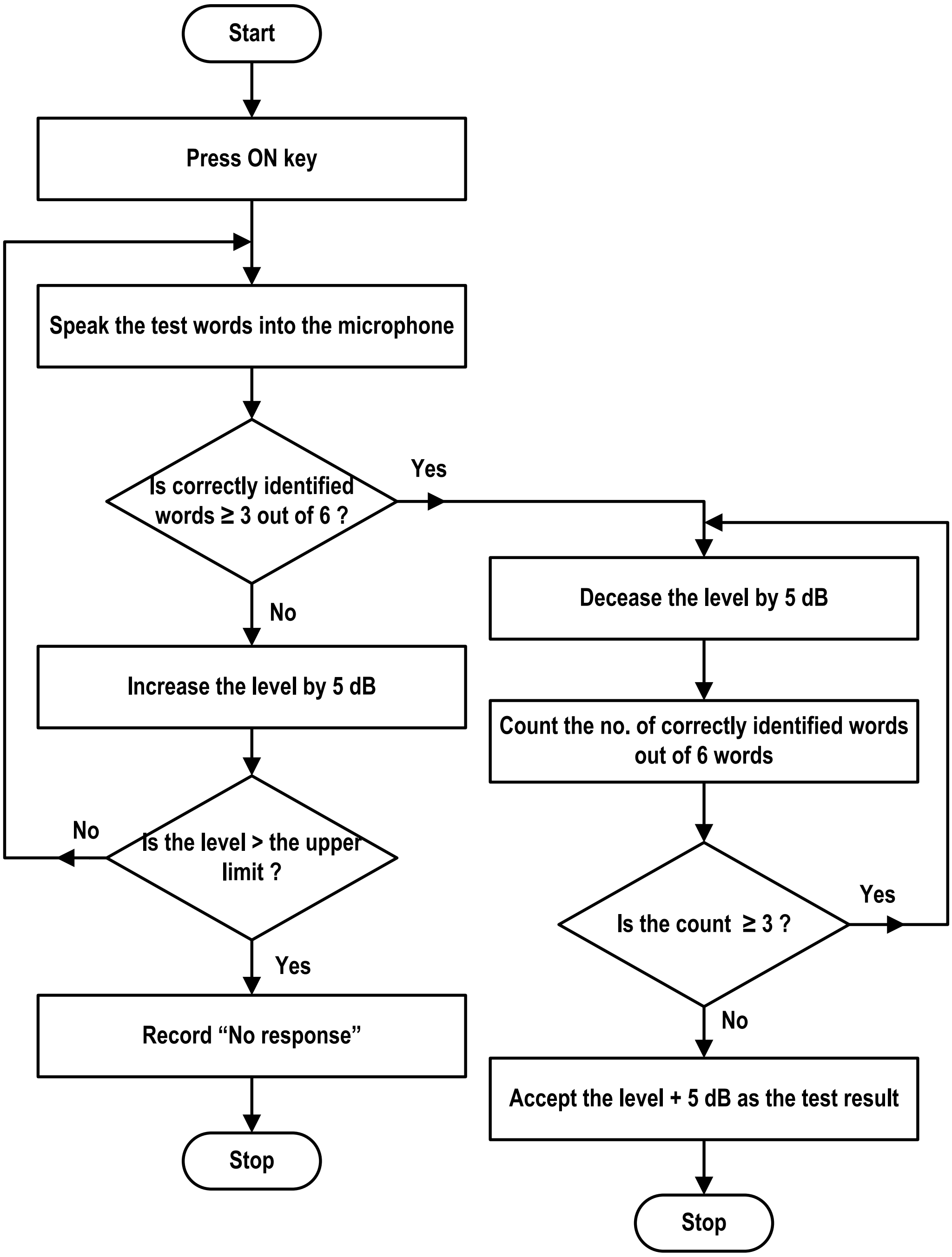


Figure 5.20 Flow chart for speech audiometry

Before turning on the instrument, the response switch, the headphones, the bone vibrator and the power supply has to be connected to it. For speech audiometry, the audio input (microphone or line) also needs to be connected. After power on, backlight of the display is turned on and the power-up display, as shown in Figure 5.5, appears. After pressing any key, the test menu appears on the display with default audiometric settings, as shown in Figure 5.6. The back-light can be turned off / on by pressing “□” key at any time. Keeping it off saves battery power.

The display has three fields: upper, lower left, and lower right. The upper field is the "options" field and it displays the test parameters as selected using the keypad. The lower left field is the "results" field and it displays the previously stored test results for the combination of the selected function (LAC / RAC / LBC / RBC) and type (TONE / SISI / TDECAY / SPEECH). The first line has 11 positions each corresponding to each of the audiometric frequencies, starting from the lowest frequency (125 Hz). A filled square indicates a previously stored test result at that frequency while a dash indicates that test result has not been stored. In case of speech audiometry, only the first position is displayed. The second line in this field displays the previously saved result for the selected frequency. Thus the two lines of this field can be used for browsing the previously stored results by using the function, type, and frequency keys. The lower right is the "status" field and displays information related to test progress and the battery voltage. Pressing of CLEAR key results in, after a confirmation as shown in Figure 5.7, clearing of the earlier test results and setting of default parameter values.

The combination of the test ear (left or right) and the type of conduction (air or bone) is selected by pressing the FUNC key. In case of air conduction, the stimulus will be presented to the headphone of the test ear and masking noise will be presented to the headphone of the non-test ear. In case of bone conduction, the stimulus will be presented to the bone vibrator while the masking noise will be presented to the headphone of the non-test ear. These stimulus and noise combinations are given in Table 5.5.

The desired test type can be selected by pressing the TYPE key: manual tone audiometry (TONE-M), automated tone audiometry (TONE-A), automated SISI test (SISI), automated tone decay test (TDECAY), or speech audiometry (SPEECH).

Next, the test type (TYPE), stimulus (STIM), noise (NOISE) are selected by repeatedly pressing the corresponding keys until the desired settings are displayed. The keys can be pressed in any order. For any key press, only the permitted options are displayed. In tone audiometry, the tone can be selected as pure, pulsed, or warble tone. The SISI and Tone Decay tests are conducted using pure tone. Speech audiometry is conducted using speech stimulus input from external audio input. Only speech-spectrum shaped noise is available for speech audiometry.

After selecting the function, type, the stimulus, and the noise, the stimulus frequency is selected by pressing frequency up or down keys. This selection is obviously not applicable for speech audiometry. For narrow-band noise, the center frequency of the noise is the same as the tone frequency.

The noise level is selected by using noise up or down keys. It may be noted that the selected noise is presented continuously to the non-test ear, irrespective of the status of stimulus presentation. The noise can be turned off by selecting “none” (XX) as the noise type or by decreasing the noise level to 0 dB.

Once the test for all the desired frequencies has been completed, the function or type may be changed. It may be noted that the order of the settings described here is a suggested sequence. However, there is no restriction on the order of settings for conducting a test. Specific operations to be followed for each of the test type are as the following.

5.4.1 Manual tone audiometry

For conducting this test, tone audiometry (TONE-M) needs to be selected as the type of audiometry. The tone can be selected as pure, pulsed, or warble. The masking noise can be selected as none, broad-band, or narrow-band. Select the tone frequency and appropriate noise level. The tone at the selected level is presented by pressing the ON key. It gets turned off if the patient-response button is pressed or if the OFF key is pressed. The following instruction has to be given to the patient.

“Pay attention to the tone in your test (left / right) ear. You may hear a noise in the other ear which should be ignored. Please press the response-button, whenever you hear the tone no matter how small it is. As you release the button, a light will indicate that your response has been recorded.”

Steps for finding the threshold at the selected frequency (using 5 dB up, 10 dB down method) are represented as a flow chart as shown in Figure 5.16. After finding the threshold for the selected frequency, the tone presentation is stopped and the result is saved by pressing the SAVE key. In addition to the stimulus level, the test conditions (stimulus type, noise type, noise level) also get saved. The stored result can be viewed in stored “result” field of the display by selecting the corresponding function, type, and frequency. The test may be repeated for each of the audiometric frequencies by following the above procedure.

5.4.2 Automated tone audiometry

For conducting this test, automated tone audiometry (TONE-A) needs to be selected as the type of audiometry. The tone may be selected as pure, pulsed, or warble. The masking noise can be selected as none, broad-band, or narrow-band. The following instruction has to be given to the patient before starting of the test.

“Pay attention to the tone in your test (left / right) ear. You may hear a noise in the other ear which should be ignored. Please press the response-button, whenever you hear the tone no matter how small it is. As you release the button, a light will indicate that your response has been recorded.”

Selecting the tone frequency, setting the initial stimulus level as 30 dB HL (or a level slightly higher than the expected threshold) and also setting an appropriate noise level has to

Table 5.4 Setting of test parameters through the keypad

Sr. No.	Setting	Key	Available options	
			Display	Selection/operation
1	Function	FUNC	RAC LAC RBC LBC	Right ear with air conduction Left ear with air conduction Right ear with bone conduction Left ear with bone conduction
2	Type of audiometry	TYPE	TONE-M TONE-A SISI TDECAY SPEECH	Tone audiometry - manual Tone audiometry - automated SISI test Tone decay test Speech audiometry
3	Stimulus	STIM	PURE WRBL PULS SPCH	Pure tone Warble tone (FM) Pulsed tone (AM) Speech (Ext. input)
4	Masking noise	NOISE	XX WB NB SN	No noise Wide-band noise Narrow-band noise Speech -spectrum shaped noise
5	Stimulus level increment	S.▲	“NN” dB	Increment the tone level by 5 dB
6	Stimulus level decrement	S.▼	“NN” dB	Decrement the tone level by 5 dB
7	Frequency increment	F.▲	“NNNN” Hz.	Change the frequency to the next higher audiometric frequency
8	Frequency decrement	F.▼	“NNNN” Hz.	Change the frequency to the next lower audiometric frequency
9	Noise level increment	N.▲	“NN” dB	Increment the noise level by 5 dB
10	Noise level decrement	N.▼	“NN” dB	Decrement the noise level by 5 dB
11	Presentation start	ON	►►	Stimulus on
12	Presentation stop	OFF		Stimulus off
13	Save result	SAVE	Saved	Save the test results (overwriting previously stored result)
14	Clearing the tests	CLEAR	Cleared	Clear all the test results after confirmation
15	Print	PRINT	Print	Transfer the test results to the PC
16	Back-light on/off	□		Toggles the back-light of the display

be done before starting of the test. The test is started in automated mode by pressing the ON key. Then the instrument finds out the hearing threshold level by an automated procedure (using 5 dB up, 10 dB down method) which is represented as a flowchart in the Figure 5.17. In addition to the stimulus level, the test conditions (stimulus type, noise type, noise level) also get saved after finding the HTL value. “No response” is recorded as a level 999 dB. The result is visible in the “stored result field” of the display whenever the corresponding function, type, and frequency get selected. It may be noted that during the test, all key presses other than OFF get ignored. The test may be terminated prematurely by pressing the OFF key and no result is saved in this case. The test is to be repeated for each of the audiometric frequencies by following the above procedure.

5.4.3 SISI test

For conducting this test, SISI needs to be selected as the type of audiometry. The test is conducted using pure tone in automated mode. The masking noise can be selected as none, broad-band, or narrow-band. The following instruction has to be given to the patient.

“Pay attention to the tone in your test (left / right) ear. You may hear a noise in the other ear which should be ignored. Occasionally you will hear a slight and brief increase in the loudness of the tone. Please press the response-button whenever you hear this increase in the loudness no matter how small the increment is. As you release the button, a light will indicate that your response has been recorded.”

The tone frequency and appropriate noise level should be selected before starting of the test. The stimulus level should be set as 20 dB supra-threshold for the selected tone frequency as determined earlier by tone audiometry. The test is started by pressing the ON key. The instrument finds the test result in an automated method by monitoring the responses during the test. The automated method is represented as a flowchart in the Figure 5.18. Test progress is updated in the “status” field of the display. 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 and conduction. It may be noted that during the test, all key presses other than OFF key get ignored. The test may be terminated prematurely by pressing the OFF key and no result is saved in this case. The test may be repeated for each of the audiometric frequencies by following the above procedure.

5.4.4 Tone decay test

For conducting this test, TDECAY has to select as the type of audiometry by using TYPE key. The test is conducted using pure tone in automated mode. The masking noise may be selected as none, broad-band, or narrow-band. The following instruction has to be given to the patient.

"Pay attention to the tone in your test (left / right) ear. You may hear a noise in the other ear which should be ignored. Please press the response-button whenever you hear the tone for the first time and again whenever you fail to hear it."

The tone frequency and appropriate noise level should be selected before starting of the test. The stimulus level should be set as 10 dB below the hearing threshold for the selected tone frequency as determined earlier by tone audiometry. The test is started by pressing the ON key. The instrument finds the test result in an automated method by monitoring the responses during the test. The automated method is represented as a flowchart in the Figure 5.19. Test progress is updated in the "status" field of the display. The level along with the test parameters (stimulus level, noise type, noise level) are automatically saved for the selected frequency and for the selected test ear and conduction. It may be noted that during the test, all key presses other than OFF get ignored. The test may be terminated prematurely by pressing the OFF key and no result is saved in this case. The test may be repeated for each of the audiometric frequencies by following the above procedure.

5.4.5 Speech Audiometry

For conducting this test, speech audiometry (SPEECH) needs to be selected as the type of audiometry. The test is conducted using a set of speech stimuli (consisting of standardized word lists in different languages) as stimulus. The masking noise may be selected as none, broad-band, or speech-spectrum shaped noise. The test can be terminated at any time by pressing the OFF key. The following instruction has to be given to the patient.

"Pay attention to the speech sound in your test (left / right) ear. You may hear a noise in the other ear which should be ignored. Please respond by speaking the word heard by you, whenever you can identify it."

The appropriate noise level is to be selected before starting of the test. The speech source (microphone or CD player) has to be connected to the audio input. The stimulus level should be set as 25 dB above the hearing threshold. Steps for finding the threshold at the selected frequency (using 5 dB up, 10 dB down method) are represented as a flow chart as shown in Figure 5.20. The test is terminated by the OFF key. The final level along with the test parameters (noise type and its level) can be saved as the result for the selected test ear and conduction.

5.5. Calibration

The instrument needs to be calibrated to compensate for the variation in the frequency response (acoustic output level for a given input rms voltage) of the transducer (headphone, bone vibrator, or speaker) used for generating the acoustic output. The calibration is carried out under software control using the keypad and display, without having to make any adjustments inside the instrument. It involves finding the setting of the internal digitally controlled attenuator to generate the acoustic output from the transducer at a specified level at each of the test frequencies. The settings are stored in the internal memory of the instrument and they are used for sound generation, until they are changed by a subsequent calibration. In some cases, the two transducers of the same make and model may differ somewhat in their frequency responses and hence whenever a transducer is changed, it should be calibrated. The instrument uses a separate calibration table for each of the transducers (left headphone, right headphone, bone vibrator), and hence the left and right headphones need not be a matched pair.

Calibration setup for the headphone consists of an artificial ear with a coupler for the headphone to be tested and a sound level meter with a microphone calibrated for use with the artificial ear. The sound level meter gives the output in dB SPL. The instrument is calibrated for 90 dB SPL at each frequency. The instrument uses an internal table for conversion from SPL to the audiometric HL scale. Calibration of the bone vibrator needs an artificial mastoid.

The instrument can be put into the calibration mode only by simultaneously pressing the response button and the OFF key on the keypad, followed by a confirmation. This avoids the possibility of unintentional modification of the calibration table which can make the test results unusable.

The transducer to be calibrated is selected by setting the corresponding function as given in Table 5.5. During calibration, the instrument outputs pure tone to the transducer to be calibrated irrespective of the stimulus type selected earlier and no masker is output. This mode uses frequency up/down, stimulus dB up/down, SAVE, CLEAR, ON and OFF keys. Other key presses are ignored. The calibration is carried out by using the following steps.

1. Connect the transducer to be calibrated to the audiometer. Couple the transducer to the test setup: artificial ear and sound level meter in case of a headphone, or artificial mastoid and sound level meter in case of a bone vibrator.
2. Select the function corresponding to the transducer to be calibrated, as given in Table 5.6.
3. Start calibration mode, by simultaneously pressing the Patient-Response button and the OFF key on the keypad. The display shows the status as "Calibration Mode" and prompts "* Proceed (ON)?" and "* Exit (OFF)?". If OFF key is pressed, the instrument exits back to the operation mode. If ON key is pressed, the instrument proceeds with calibration. The prompt display for calibration mode is shown in Figure 5.8 and calibration display is shown in Figure 5.9. It shows the previous attenuation setting (Att.) for the selected frequency (Freq.) and continuously outputs the corresponding tones through the transducer. The default frequency for calibration is 1 kHz and can be changed using frequency up/down keys. The attenuation setting

appears as a 3-digit number and it controls the voltage being output to the transducer. It can be changed using stimulus dB up / down keys.

4. Select the tone frequency by using frequency up/down key. Using stimulus dB up/down keys, adjust the acoustic output level such that the sound level meter reads 90.0 dB SPL. The calibration for the selected frequency may be stored by pressing SAVE key.
5. Go to step 4, for selecting another frequency for calibration. Alternatively, exit from the calibration mode back to the operation mode, by simultaneously pressing the Patient-Response button and the OFF key on the keypad.

It may be noted that in the calibration mode, the earlier attenuation setting is changed only if SAVE key is pressed. Calibration for a transducer may be carried out for any sequence of frequencies, and it need not be carried out for all the frequencies. The same calibration procedure is to be used for each of the transducers.

Table 5.5 Stimulus and noise presentation for different test functions

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

Table 5.6 Function codes for calibration of different transducers

Transducer to be calibrated	Left headphone	Right headphone	Bone vibrator
Function code	LAC	RAC	LBC or RBC

Chapter 6
SYSTEM ASSEMBLY AND TESTING

6.1 PCB design

Wherever possible the circuit blocks were individually bread-boarded and tested. Subsequently a PCB was designed. It consists of the TA microcontroller, the SG microcontroller, attenuator, latch, three multiplexers, two amplifiers, three power amplifiers, MAX232 IC, pre-amplifier, differential amplifier and regulators, keypad connector, graphical LCD connector, response button jack, two debugger connectors (one for each microcontroller), and SD card connector.

The PCB is a double sided PTH board with a size of 178 mm x 127 mm. Wherever needed, the top and bottom side tracks are connected by 0.8 mm dia. PTH. As most of 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 noise problems. Decoupling capacitors of 0.1 µF (ceramic) have been used for each IC close to its power supply pin. The supply entry points are decoupled by 100 µF/25 V electrolytic capacitor in parallel with 0.1 µF ceramic capacitor. The analog ground and digital ground are routed separately throughout the board and are shorted at the input power supply. Digital +5 V, digital +3.3 V and analog +5 V are distributed as a plane on the top side of the board. Similarly, copper planes of digital and analog ground are provided on the bottom side of the PCB. Shortest overall connection path is chosen while connecting pin-to-pin connections. The width of the signal tracks is 0.38 mm. The width of supply tracks is 1.27 mm. The minimum space between the two signal tracks is 0.33 mm. The minimum space between a signal and a polygon plane is 0.63 mm. The routing style at the corner is in 45^0 shape. The layouts of the PCB i.e. top layer, top over lay, bottom layer, are given in Appendix D. After receiving the PCB, a check on all the tracks was carried out to ensure continuity of tracks and also to verify that there were no shorts between any neighbouring track pairs.

6.2 System assembly

Required ICs, resistors and capacitors are placed in the respective positions of the board. Three 1/4” and one 1/8” mono phone jacks are mounted at the upper end of the board. One 1/8” mono phone jack is mounted on the left hand side of the board. 3-pin connecter is placed to receive the input power. 20-pin connector is placed to connect the graphical LCD. 8-pin connector is placed on the board to connect the 4x4 keypad. SD card connector is placed to connect the SD card. One DB-9 socket is placed at lower left hand side for RS232 communication. Two 6-pin connectors are placed to connect the PICkit™ 3 debugger for programming and / or debugging.

The keypad is mounted on the lower portion of the front panel of the PCB through a 6-pin connector. The graphical LCD is mounted on the upper portion of the front panel of the display through a 20-pin connector. To hold this display, screws are used at 4 corners of the display. Two mono headphones can be connected through the respective 1/4” mono phone jacks. Response button and microphone can be connected through the respective 1/8” mono phone jack.

6.3 Test results

Each hardware block was assembled and tested individually. All the software subroutines were tested. The stimulus tones i.e. pure tone, warble tone and pulsed tone were measured using sound analyzed using a spectrum analyzed using a spectrum analyzer. The wide-band noise and narrow band noise were generated and their frequency spectra were verified using a spectrum analyzer. The various waveforms are shown Figure 6.1 to 6.9. Live speech was input through microphone and heard through head-phone and the quality was found to be satisfactory. Tone levels were changed according to the test procedure. The changing of the level at zero crossing of the input was observed at the output of attenuator. The distribution of stimulus and / or masking noise to the transducers (left head-phone, right head-phone and bone vibrator) was tested and no inter transducer leakage was found. The quality of stimulus and masking noise was heard through the headphones and the bone vibrator. All the display features were verified. The saved test results were displayed and sent to PC. The instrument was calibrated and its effect on level change was confirmed.

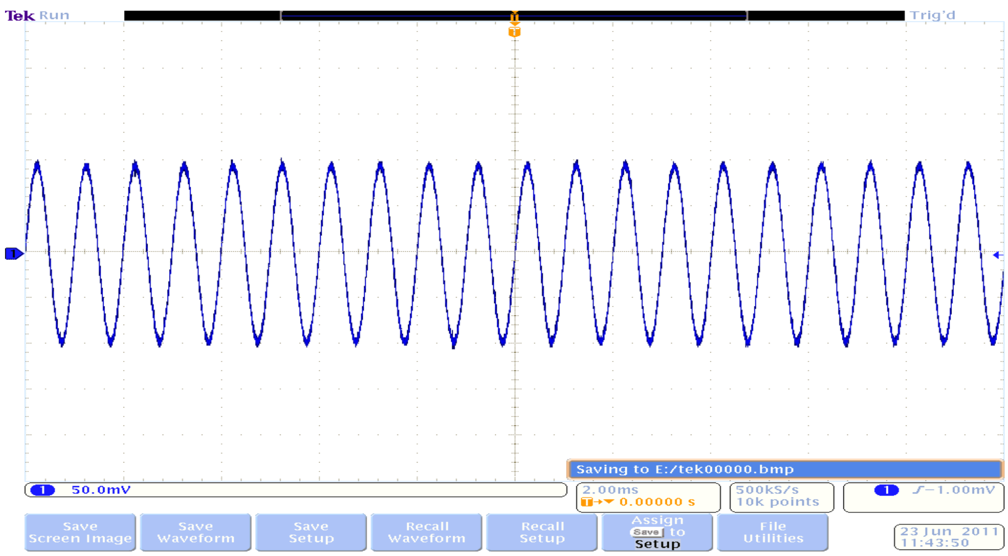


Figure 6.1 Pure tone waveform

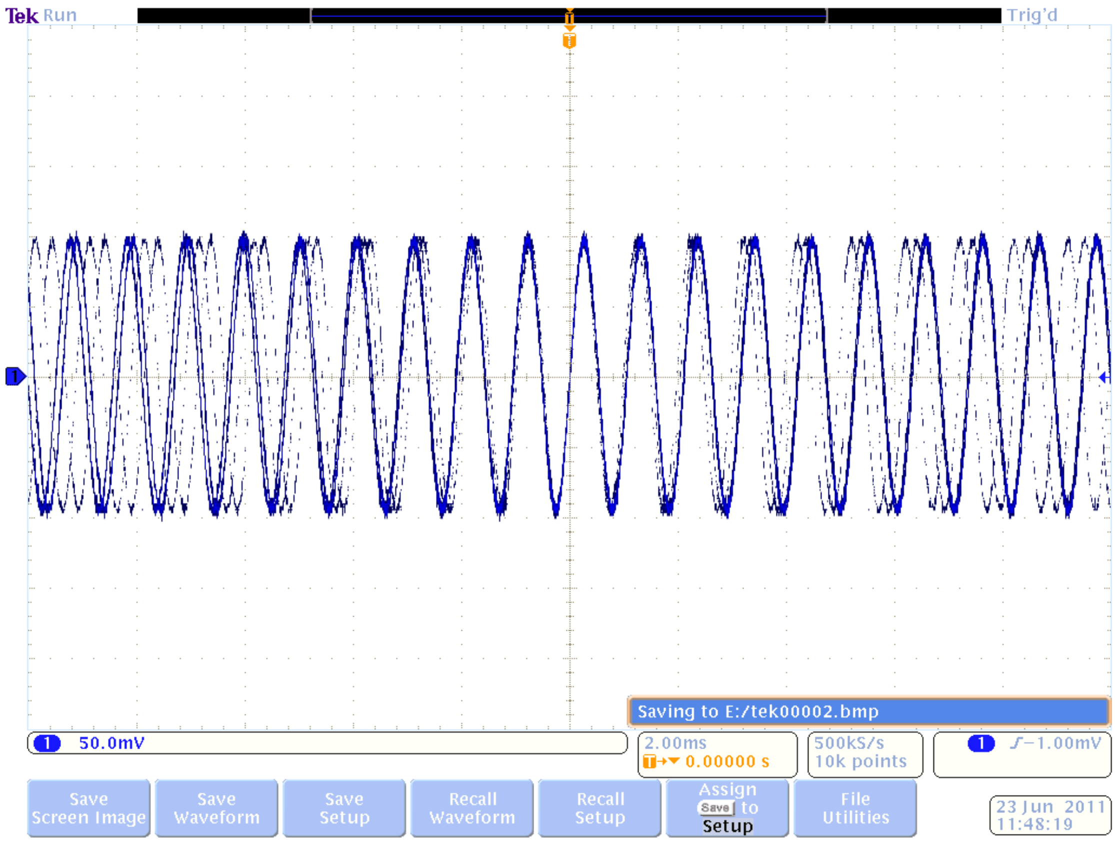


Figure 6.2 Warble tone waveform

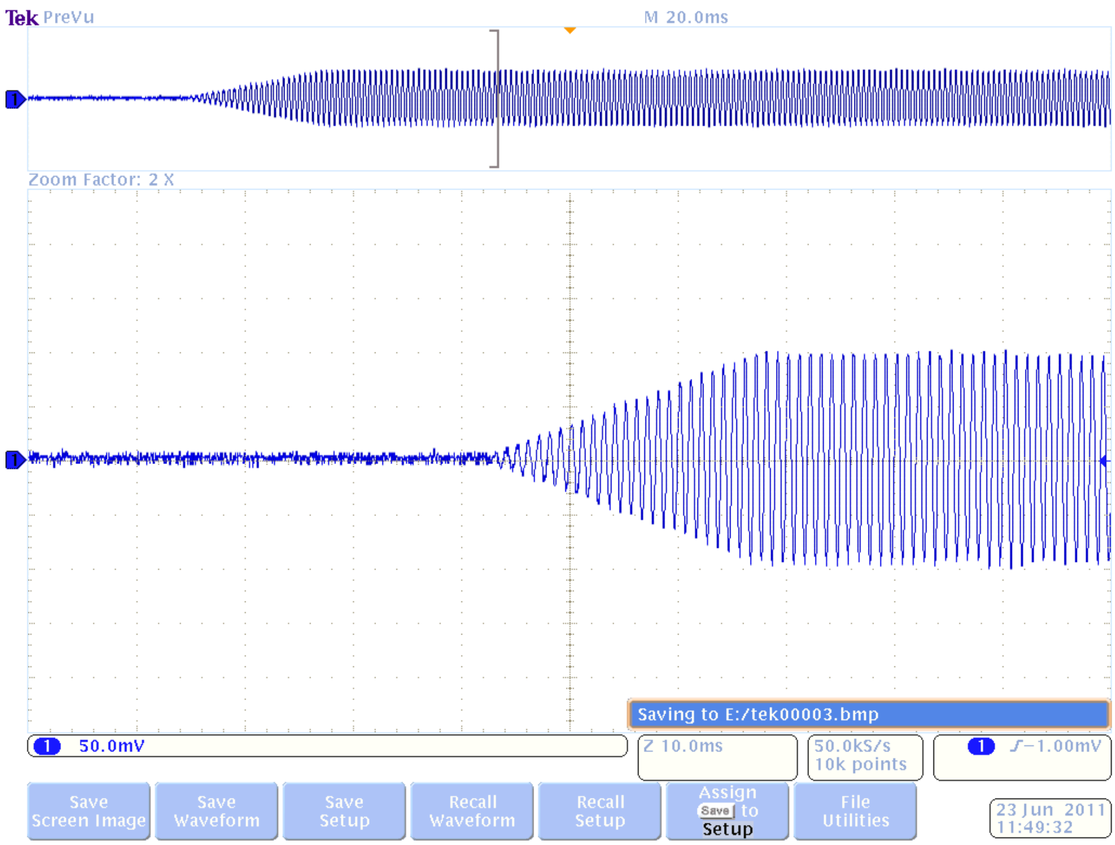


Figure 6.3 Rising edge of pulsed tone waveform

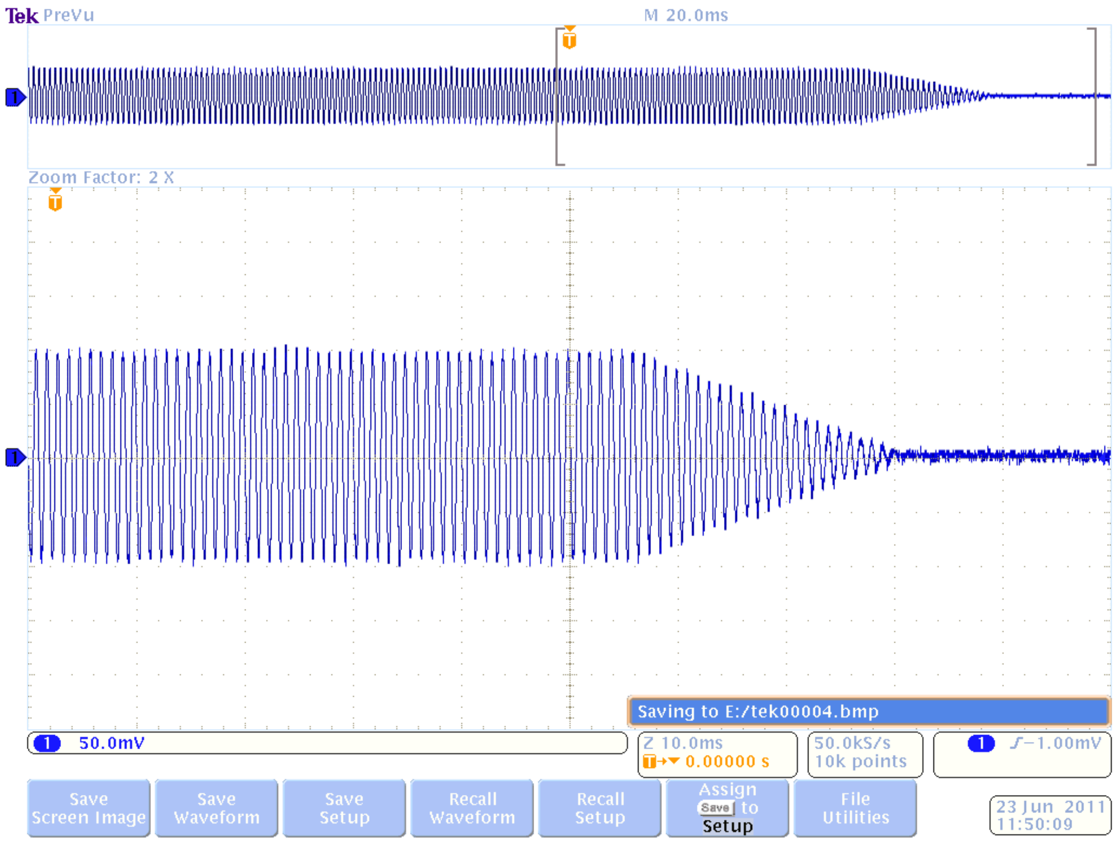


Figure 6.4 Falling edge of pulsed tone waveform

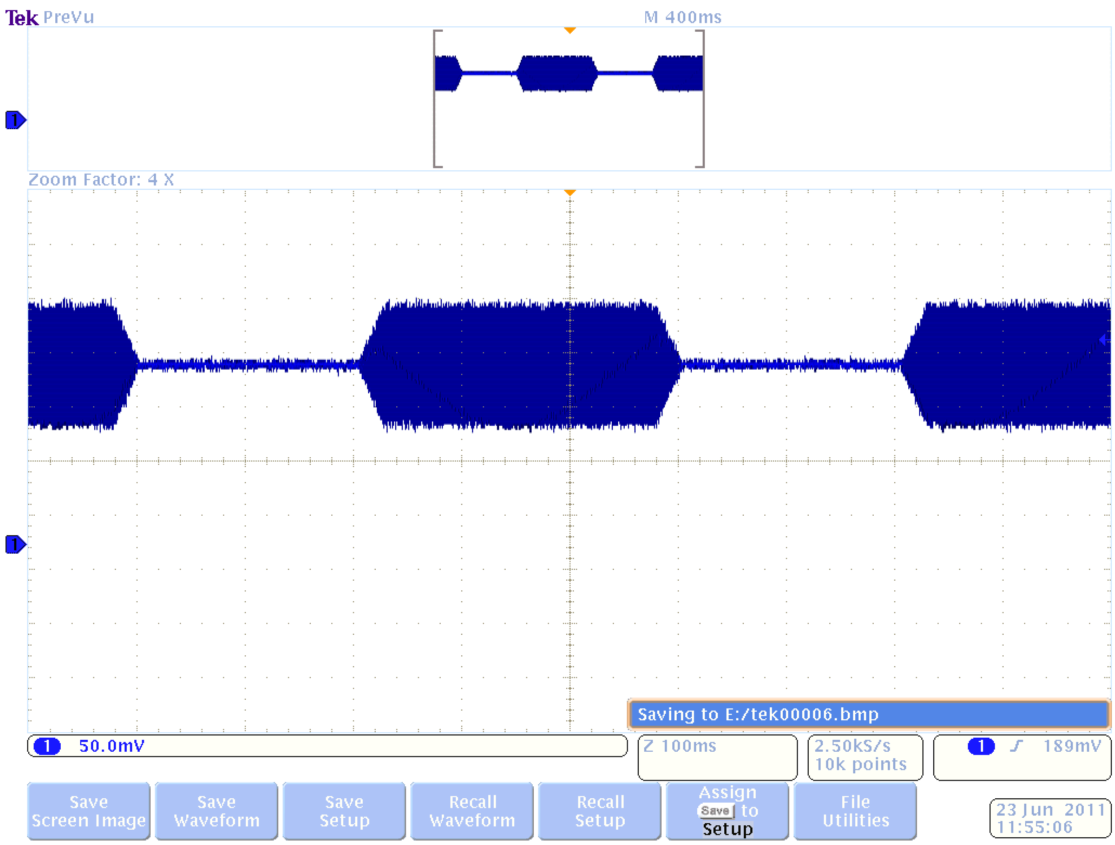


Figure 6.5 Pulsed tone waveform

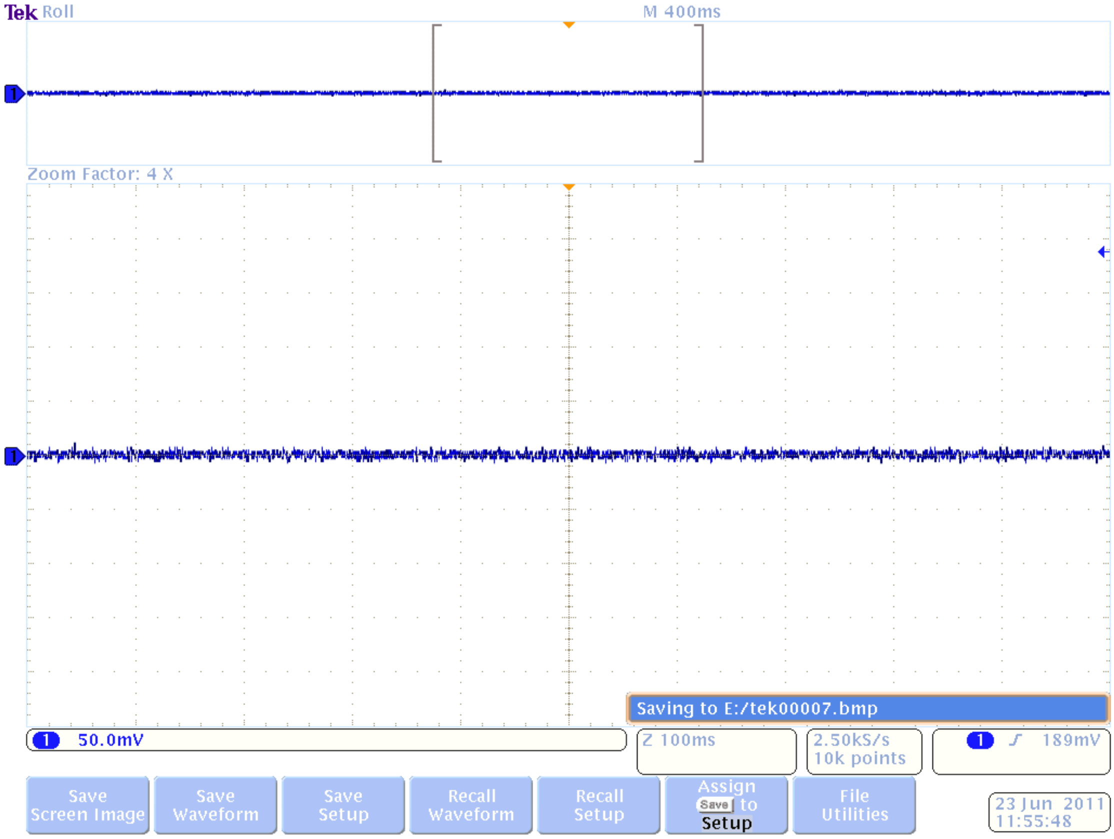


Figure 6.6 Noise in the absence of signal at the output of power amp.

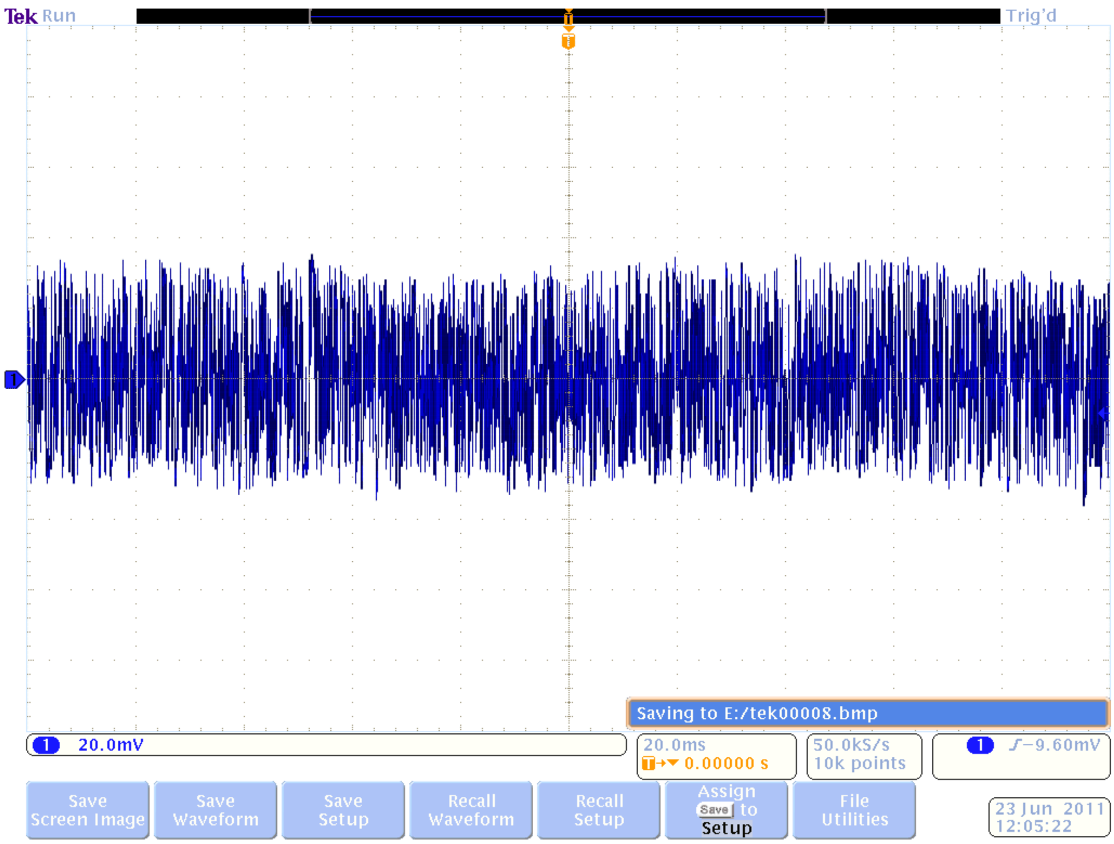


Figure 6.7 Broad-band noise waveform

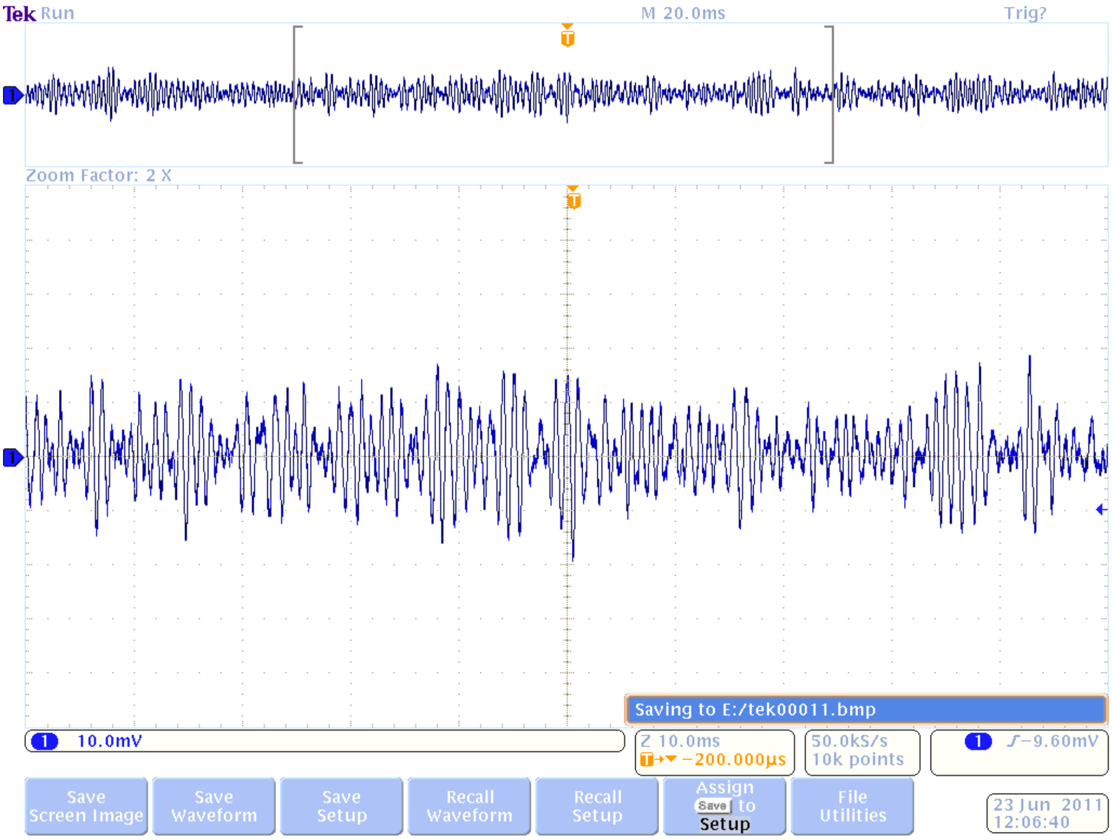


Figure 6.8 Narrow-band noise waveform

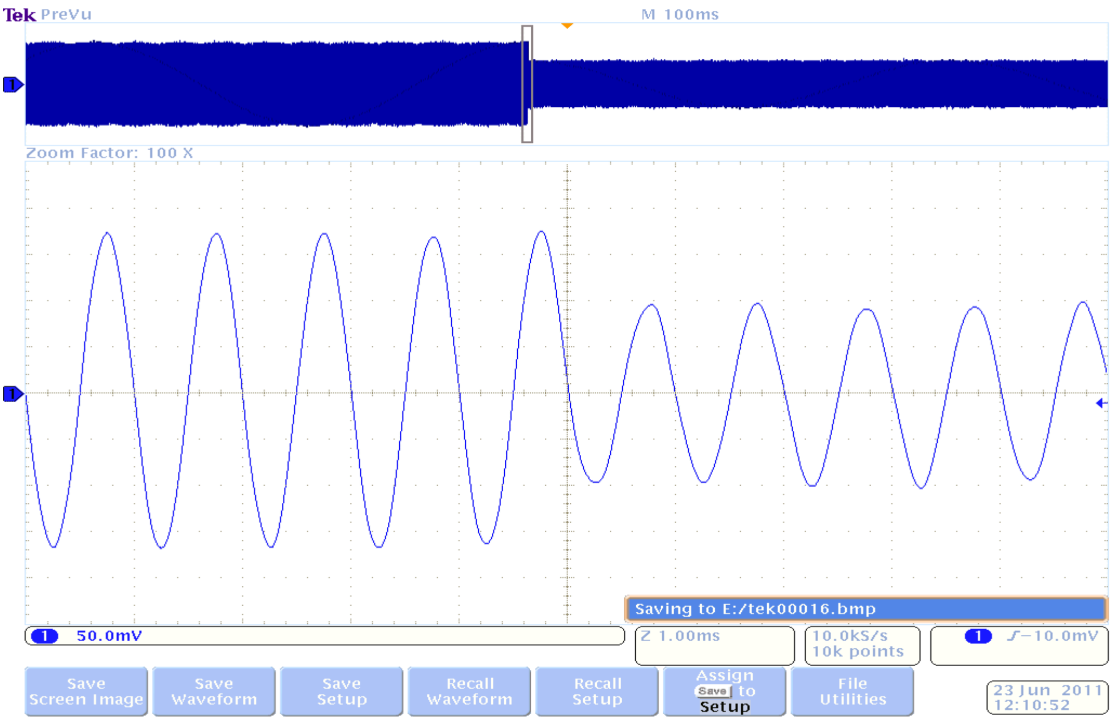


Figure 6.9 Tone level change waveform

Chapter 7
SUMMARY AND SUGGESTIONS

7.1 Summary

The objective of this project was to develop a portable audiometer for conducting subject-response based audiometry. After studying the various earlier designs of microcontroller based audiometers, a subject response based audiometer "IITB-2K11" has been designed. To facilitate modular program development without having the overheads of an operating system, two microcontrollers are used. One is used for handling user interface and controlling the test parameters, called as the TA microcontroller (PIC16F1939). The second microcontroller, called as the SG microcontroller (dSPIC33FJ128GP802), is used for generating stimulus and masker. The user interface is through a 4x4 keypad and 128x64 graphical display interfaced to the TA microcontroller.

Audiometric stimulus tones are generated digitally by using SG microcontroller which has sufficient internal memory and processing speed for synthesizing the waveform and an on-chip dual channel 12-bit DACs for outputting them. Masking noise is generated digitally by a pseudo-random binary sequence and digital filtering. The right channel DAC is used for stimulus and the left channel is used for masker. It was estimated that generation of warble tones could neither be carried out by real-time processing nor the corresponding samples could be stored in the internal program memory. Hence an SD card is interfaced and used as an external memory for storing one cycle of each of the stimulus waveforms. For conducting speech audiometry, speech input is given to ADC and then output through its right DAC. The two complementary outputs of each DAC are given to a differential amplifier for centering the audio signal at zero volt with a swing compatible with the input of the subsequent stages using direct coupling.

The test tone and masking noise parameters are sent from the TA microcontroller to the SG microcontroller using an asynchronous serial protocol. A programmable stereo audio volume control log attenuator chip (PGA2310) is used for outputting the selected stimulus and noise to be output at different levels. The right channel attenuator is used for stimulus and left channel attenuator is used for the masker. The attenuator chip is interfaced to the TA microcontroller and the gains for both the attenuators are set using a common SPI serial interfacing. The audiometer has three transducers for presenting the sounds: the left headphone, the right headphone, and the bone vibrator. Three multiplexers are used for coupling the sounds to the transducers. The digital controls, used for multiplexing are given from the TA microcontroller. In addition to selecting between the stimulus and masker, this stage is also used for selecting the stimulus gain as 0 or 20 dB.

In order to meet the power requirement, the output from each multiplexer is fed to a power amplifier realized using LM1877. As the signal outputs from the amplifier have no dc offset, the outputs are directly coupled to the transducers, thus avoiding the poor low frequency response due to ac coupling, and problem associated in using electrolytic capacitor. A response button is connected to the TA microcontroller for taking the responses from the patient. The interfacing circuitry is also used for giving feed back by turning on a light in the button enclosure. Calibration settings are stored in EEPROM of the TA microcontroller.

All the user interface operations on the TA microcontroller and the signal generation operations on the SG microcontroller and communication between the two microcontrollers are handled by a program running on each of them.

The instrument facilitates conducting of various types of subject response based audiometry in manual and automated modes and the test results can be transformed through a serial port to a PC for report preparation.

7.2 Suggestions for future work

Further improvement can be carried out to incorporate the features and specifications of an advanced diagnostic audiometer. The PCB has to assemble in an appropriate instrument cabinet. In the current instrument, speech audiometry requires speech input through the external mic Program can be modified to permit pre-stored speech material on the SD card for speech audiometry. SD card can be replaced by a memory chip. It can also be possible to control the audiometer by using a PC based graphical user interface.

Appendix A
USER MANUAL

A.1 Introduction

An audiometer is an electro-acoustic instrument for quantifying hearing impairment and for diagnosing its causes. Hearing test is conducted individually on the two ears by presenting a sequence of sound stimuli and receiving the patient's responses to these stimuli. The acoustic vibrations presented to the test ear may reach, through the skull bone, to the cochlea of the non-test ear, and it may result in cross-hearing. To prevent the non-test ear from detecting the stimulus presented to the test ear, a masking noise may need to be presented to the non-test ear.

The audiometer "IITB-2K11" can be used for conducting tone audiometry, SISI test, tone decay test, and speech audiometry, using air and bone conductions. It can be used for outputting the selected types of stimuli and masking noises at the selected levels (in dB HL) and monitoring patient's responses for conducting the hearing tests in the manual and automated modes. The user interface for conducting the tests is through a 4x4 keypad and a 128x64 graphical LCD. The patient's response is received through a response button which has an indicator light. The saved test results can be seen on the display and they can be transferred to a PC for preparing test reports. All the test stimuli are digitally synthesized with frequencies derived from a crystal controlled oscillator and the levels controlled by digital attenuators, avoiding the need for routine calibration. For a given set of transducers, the instrument can be calibrated through its user interface, and the calibration data is saved in the internal memory.

A.2 Hearing Tests

The instrument can be used for conducting tone audiometry, SISI test, tone decay test, and speech audiometry, using air and bone conduction. In manual mode, the level and presentation of the test stimulus is manually controlled. In automated mode, the instrument automatically controls the level and presentation of the test stimulus by monitoring the patient's response through the response button.

A.2.1 Tone audiometry (manual and automated)

Tone audiometry is used for finding the hearing threshold level (HTL) for tones at the audiometric test frequencies. The instrument permits selection of the test stimulus as pure tone (continuous), pulsed tone (amplitude modulated, AM), and warble tone (frequency modulated, FM). The test can be conducted using either the manual or the automated mode. The patient is instructed to press the response button whenever he/she hears a tone in the test ear. In both the modes, the test ear (left/right), the type of conduction (air/bone), the masking type and level, the stimulus frequency, and its type (pure/pulsed/warble), and the initial level are manually selected.

In manual mode, the tone is presented by pressing the ON key. It gets terminated if the patient response button is pressed or if the OFF key is pressed. The test level is manually changed by pressing the stimulus level up/down keys. The lowest level resulting in 50% response for 5 or more presentations is taken as the HTL. In automatic mode, once the ON key is pressed, the sequence of tone presentation and level is controlled by the instrument by monitoring the patient response button until the threshold level gets determined. The test progress (presentation level and patient response) is updated on the display, and the test can be terminated by pressing the OFF key.

In both the modes, the final test level along with the test parameters (stimulus type, noise type, noise level) can be saved as the test result at the selected frequency for the selected test ear and conduction.

A.2.2 SISI test (automated)

Short increment sensitivity index (SISI) test is used for determining the capacity to detect a brief 1-dB increment in a 20 dB supra-threshold tone. It is conducted with the pure tone as the stimulus. The stimulus level should be 20 dB supra-threshold for the selected tone frequency as determined earlier by tone audiometry. The patient is instructed to press the response button whenever he/she hears a short duration increase in the level of the tone in the test ear.

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 after the ON key is pressed. At intervals of 5 s, the stimulus level is increased by 1 dB for 200 ms and then the level is restored back. Pressing of the response key is recorded. Multiple presses between successive increments are ignored. The test is continued for 20 such increments and the percentage of the responses is displayed as the SISI score. 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 and conduction. The test may be terminated prematurely by pressing the OFF key.

A.2.3 Tone decay test (automated)

Tone decay test is used for determining the continuous listening capability for about 1 min. It is conducted with the pure tone as the stimulus. The stimulus level should be 10 dB below the hearing threshold for the selected tone frequency as determined earlier by tone audiometry. The patient is instructed to press the response button whenever he/she hears a tone for the first time in the test ear and then again whenever the tone disappears.

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 after the ON key is pressed. At intervals of 2 s, the stimulus level is increased in steps of 5 dB until the patient response is received. The presentation is continued for 1 min. and if the second response is not received, the tone is terminated and the level is displayed as the test result. If the second response is received before the lapse of 1 min, it indicates that the stimulus became inaudible and hence the test is repeated by increasing the level by 5 dB. During the test, the status of presentation level and patient response is updated on the display. The final level along with the test parameters (noise type and its level) are saved automatically as the result for the selected frequency and for the selected test ear and conduction. The test can be terminated by pressing the OFF key.

A.2.4 Speech Audiometry (manual)

Speech audiometry is commonly used for finding the speech reception threshold (SRT) for a set of speech sounds (consisting of standardized word lists in different languages). It is generally conducted with the speech-spectrum shaped noise as the masking noise. The test is started with the level set as 25 dB above the hearing threshold level (HTL), as determined earlier by tone audiometry. The patient is instructed to repeat the word as heard in the test ear.

The test ear (left/right), the type of conduction (air/bone), the masking level, and the speech sound level are manually selected. The sound presentation starts after the ON key is pressed. The level is manually decreased in steps of 5 dB until a level is reached at which the patient identifies 50 % of the words correctly. In certain cases, all the words may be correctly identified at a particular level, but the score may be much lower than 50 % at the next lower level. In such cases the level at which all the words or more than 50 % of the words are identified correctly is taken as the SRT. In some cases, there may be a need to increase the presentation level for getting 50% response. The test can be terminated at any time by pressing the OFF key.The final level along with the test parameters (noise type and its level) can be saved as the result for the selected frequency and for the selected test ear and conduction.

A.3 Operation procedure

The operation of the instrument is controlled using its 4x4 keypad and 128x64 graphical LCD.

Connect the response switch, the headphones, the bone vibrator and the power supply to the instrument. For speech audiometry, the audio input (microphone or line) also needs to be connected. After power on, backlight of the display is turned on and the power-up display, appears. After pressing any key, the test menu appears on the display with default audiometric settings. The back-light can be turned off/on by pressing "□" key at any time. Keeping it off saves battery power.

The display has three fields: upper, lower left, and lower right. The upper field is the "options" field and it displays the test parameters as selected using the keypad. The lower left field is the "results" field and it displays the previously stored test results for the combination of the selected function (LAC / RAC / LBC / RBC) and type (TONE / SISI / TDECAY / SPEECH). The first line has 11 positions each corresponding to each of the audiometric frequencies, starting from the lowest frequency (125 Hz). A filled square indicates a previously stored test result at that frequency while a dash indicates that test result has not been stored. In case of speech audiometry, only the first position is displayed. The second line in this field displays the previously saved result for the selected frequency. Thus the two lines of this field can be used for browsing the previously stored results by using the function, type, and frequency keys. The lower right is the "status" field and displays information related to test progress and the battery voltage. Pressing of CLEAR results in, after a confirmation, clearing of the earlier test results and setting of default parameter values.

Select the combination of the test ear (left or right) and the type of conduction (air or bone) by pressing the FUNC key. In case of air conduction, the stimulus will be presented to the headphone of the test ear and masking noise will be presented to the headphone of the non-test ear. In case of bone conduction, the stimulus will be presented to the bone vibrator while the masking noise will be presented to the headphone of the non-test ear.

The desired test type should be selected by pressing the TYPE key: manual tone audiometry (TONE-M), automated tone audiometry (TONE-A), automated SISI test (SISI), automated tone decay test (TDECAY), or speech audiometry (SPEECH).

Next select the test type (TYPE), stimulus (STIM), noise (NOISE) by repeatedly pressing the corresponding keys until the desired settings are displayed. The keys can be pressed in any order. For any key press, only the permitted options are displayed. In tone audiometry, the tone can be selected as pure, pulsed, or warble tone. The SISI and Tone Decay tests are conducted using pure tone. Speech audiometry is conducted using speech stimulus input from external audio input. Broad-band noise and speech-spectrum shaped noise are available for speech audiometry whereas for the rest of the tests, broad-band noise and narrow-band noise are available.

After selecting the function, type, the stimulus, and the noise, the stimulus frequency is selected by pressing frequency up or down keys. This selection is obviously not applicable for speech audiometry. For narrow-band noise, the center frequency of the noise is the same as the tone frequency.

The noise level is selected by using noise up or down keys. It may be noted that the selected noise is presented continuously to the non-test ear, irrespective of the status of stimulus presentation. The noise can be turned off by selecting "none" (XX) as the noise type or by decreasing the noise level to 0 dB.

Once the test for all the desired frequencies has been completed, the function or type may be changed. It may be noted that the order of the settings described here is a suggested sequence. However, there is no restriction on the order of settings for conducting a test. Specific operations to be followed for each of the test type are as the following.

A.3.1 Manual tone audiometry

For conducting this test, select the type of audiometry as manual tone audiometry (TONE-M). The tone may be selected as pure, pulsed, or warble. The masking noise may be selected as none, broad-band, or narrow-band. Select the tone frequency and appropriate noise level. The tone at the selected level is presented by pressing the ON key. It gets turned off if the patient-response button is pressed or if the OFF key is pressed. The following instruction may be given to the patient.

"Pay attention to the tone in your test (left / right) ear. You may hear a noise in the other ear which should be ignored. Please press the response-button, whenever you hear the tone no matter how small it is. As you release the button, a light will indicate that your response has been recorded."

Steps for finding the threshold at the selected frequency (using 5 dB up, 10 dB down method) are as the following.

1. Select the initial stimulus as 30 dB HL (or a level slightly higher than the expected threshold). If the response is received, go to step 5; else proceed to the next step.
2. Increase the level by 20 dB. If the response is received, go to step 5; else proceed to the next step.

3. Increase the level by 10 dB. If the response is received, go to step 5; else proceed to the next step.
4. If the level exceeds the upper limit, record “No response” and end the test, else go back to step 3.
5. Decrease the level by 10 dB (subject to the lower limit). If the response is received, repeat the step; else proceed to the next step.
6. Increase the level by 5 dB. If no response is received, repeat the step; else proceed to the next step.
7. If the number of responses at this presentation level is at least 50 % (3/3, 3/4, 3/5, 3/6, 4/7, 4/8, ..), record the level as the hearing threshold and end the test; else go to step 5.

After finding the threshold for the selected frequency, the tone presentation is stopped and the result is saved by pressing the SAVE key. In addition to the stimulus level, the test conditions (stimulus type, noise type, noise level) also get saved. The stored result can be viewed in stored “result” field of the display by selecting the corresponding function, type, and frequency. The test may be repeated for each of the audiometric frequencies by following the above procedure.

A.3.2 Automated tone audiometry

For conducting this test, select the type of audiometry as automated tone audiometry (TONE-A). The tone may be selected as pure, pulsed, or warble. The masking noise may be selected as none, broad-band, or narrow-band. The following instruction may be given to the patient.

“Pay attention to the tone in your test (left / right) ear. You may hear a noise in the other ear which should be ignored. Please press the response-button, whenever you hear the tone no matter how small it is. As you release the button, a light will indicate that your response has been recorded.”

Select the tone frequency. Set the initial stimulus level as 30 dB HL (or a level slightly higher than the expected threshold) and also set an appropriate noise level. The test is started in automated mode by pressing the ON key. The threshold is found by an automated procedure (using 5 dB up, 10 dB down method) involving the following steps of changing the tone level, tone presentation, and monitoring the patient response.

1. Wait for 2 s (minimum inter-stimulus gap). Increment the presentation count for the stimulus level. Present the tone and continuously monitor the patient response. If the response is received within 5 s, terminate the tone presentation and go to step 4; else proceed to the next step.

2. Terminate the tone presentation and wait for patient response. If the response is received within 5 s, go to step 4; else proceed to the next step.
3. Increase the level by 5 dB. If the level exceeds the upper limit, record "no response" and end the test; else go back to step 1.
4. Increment the response count for the stimulus level. If the number of responses for this level is at least 50% (i.e. 3/3, 3/4, 3/5, 3/6, 4/7, 4/8, ..), save the level as the threshold and end the test; else proceed to the next step.
5. Decrease the level by 10 dB (subject to the lower limit) and go back to step 1.

In addition to the stimulus level, the test conditions (stimulus type, noise type, noise level) also get saved. "No response" is recorded as a level 999 dB. The result is visible in the "stored result field" of the display whenever the corresponding function, type, and frequency get selected. It may be noted that during the test, all key presses other than OFF get ignored. The test may be terminated prematurely by pressing the OFF key and no result is saved in this case. The test may be repeated for each of the audiometric frequencies by following the above procedure.

A.3.3 SISI test

For conducting this test, select the type of audiometry as SISI. The test is conducted using pure tone in automated mode. The masking noise may be selected as none, broad-band, or narrow-band. The following instruction may be given to the patient.

"Pay attention to the tone in your test (left/right) ear. You may hear a noise in the other ear which should be ignored. Occasionally you will hear a slight and brief increase in the loudness of the tone. Please press the response-button whenever you hear this increase in the loudness no matter how small the increment is. As you release the button, a light will indicate that your response has been recorded."

Select the tone frequency and appropriate noise level. The stimulus level should be set as 20 dB supra-threshold for the selected tone frequency as determined earlier by tone audiometry. The test is started by pressing the ON key. The instrument finds the test result by using the following steps.

1. Set the presentation count and response count as zero. Present the tone continuously for 5 s. Proceed to the next step.
2. Increase the level by 1 dB, wait for 200 ms. and then decrease the level by 1 dB. Increment the presentation count by 1. Proceed to the next step.
3. Continue with the presentation for 5 s, while monitoring the patient response. If the response is received within 3 s, increment the response count by 1; else ignore the patient response. Proceed to the next step.

4. If the presentation count is less than 20, go back to step 2; else proceed to the next step.
5. Save the number of responses as the test score along with the test parameters (stimulus level, noise type, noise level). Display the result in the "result" of the display. End the test.

Test progress is updated in the "status" field of the display. 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 and conduction. It may be noted that during the test, all key presses other than OFF get ignored. The test may be terminated prematurely by pressing the OFF key and no result is saved in this case. The test may be repeated for each of the audiometric frequencies by following the above procedure.

A.3.4 Tone decay test

For conducting this test, select the type of audiometry as TDECAY. The test is conducted using pure tone in the automated mode. The masking noise may be selected as none, broad-band, or narrow-band. The following instruction may be given to the patient.

"Pay attention to the tone in your test (left/right) ear. You may hear a noise in the other ear which should be ignored. Please press the response-button whenever you hear the tone for the first time and again whenever you fail to hear it."

Select the tone frequency and appropriate noise level. The stimulus level should be set as 10 dB below the hearing threshold for the selected tone frequency as determined earlier by tone audiometry. The test is started by pressing the ON key. The instrument finds the test result by using the following steps.

1. Present the tone and continuously monitor the patient response. If the response is received within 5 s, go to step 3; else proceed to the next step.
2. Increase the level by 5 dB. If the level exceeds the upper limit, record "no response" and terminate the test; else go back to step 1.
3. Continue with the tone presentation and continuously monitor the patient response. If no response is received within 1 min, save the level as the test result and end the test; else proceed to the next step.
4. Wait for 5 s (inter-test interval) and go to step 2.

Test progress is updated in the "status" field of the display. The level along with the test parameters (stimulus level, noise type, noise level) are automatically saved for the selected frequency and for the selected test ear and conduction. It may be noted that during the test, all key presses other than OFF get ignored. The test may be terminated prematurely by pressing the OFF key and no result is saved in this case. The test may be repeated for each of the audiometric frequencies by following the above procedure.

A.3.5 Speech Audiometry

For conducting this test, select the type of audiometry as SPEECH. The test is conducted using set of speech sounds (consisting of standardized word lists in different languages) as stimulus. The masking noise may be selected as none, broad-band, or speech-spectrum shaped noise. The test can be terminated at any time by pressing the OFF key. The following instruction may be given to the patient.

"Pay attention to the speech sound in your test (left / right) ear. You may hear a noise in the other ear which should be ignored. Please respond by speaking the word heard by you, whenever you can identify it."

Select the appropriate noise level. Connect the speech source (microphone or CD player) to the audio input. The stimulus level should be set as 25 dB above the pure tone average (PTA) threshold (average of hearing threshold levels at 0.5, 1, 2 kHz) as determined earlier by tone audiometry. The following steps may be followed for finding the speech reception threshold (SRT).

1. Start the presentation by pressing ON key and speaking the test words into the microphone or outputting them from CD player or another audio device.
2. If the number of correctly identified words out of 6 words is 3 or greater, go to step 4; else proceed to the next step.
3. Increase the level by 5 dB. If the level exceeds the upper limit, record "No response" and end the test; else go back to step 2.
4. Decrease the level by 5 dB.
5. Count the number of correctly identified words out of 6 words. If the count is 3 or greater, go back to step 4; else accept the level + 5 dB as the test result.

The test is terminated by the OFF key. The final level along with the test parameters (noise type and its level) can be saved as the result for the selected test ear and conduction.

A.4. Calibration

The waveforms in the audiometer are digitally synthesized with crystal controlled oscillator and highly stable amplitude levels resulting in highly stable frequencies and levels. The output levels are controlled through digital attenuators. Thus the characteristics of the stimulus and the masker are highly stable. The instrument needs to be calibrated to compensate for the variation in the frequency response (acoustic output level for a given input rms voltage) of the transducer (headphone, bone vibrator, or speaker) used for generating the acoustic output. The calibration is carried out under software control using the keypad and display, without having to make any adjustments inside the instrument. It basically involves finding the setting of the internal digitally controlled attenuator to generate the acoustic output from the transducer at a specified level at each of the test frequencies. The settings are stored in the internal memory of the instrument and they are used for sound generation, until they are changed by a subsequent calibration. It may be noted that two transducers of the same make and model may differ somewhat in their frequency responses and hence calibration needs to be carried out whenever a transducer is changed. The instrument uses a separate calibration table for each of the transducers (left headphone, right headphone, bone vibrator), and hence the left and right headphones need not be a matched pair.

Calibration setup for the headphone consists of an artificial ear with a coupler for the headphone to be tested and a sound level meter with a microphone calibrated for use with the artificial ear. The sound level meter gives the output in dB SPL. The instrument is calibrated for 90 dB SPL at each frequency. The instrument uses an internal table for conversion from SPL to the audiometric HL scale. Calibration of the bone vibrator needs an artificial mastoid.

Unintentional modification of the calibration table can make the instrument unusable. To avoid such a possibility, the instrument can be put into the calibration mode only by simultaneously pressing the Patient-Response button and the OFF key on the keypad, followed by a confirmation.

The transducer to be calibrated is selected by setting the corresponding function as given in Table A.1. During calibration, the instrument outputs pure tone to the transducer to be calibrated irrespective of the stimulus type selected earlier and no masker is output. This mode uses frequency up/down, stimulus dB up/down, SAVE, CLEAR, ON and OFF keys. Other key presses are ignored. The calibration is carried out by using the following steps.

1. Connect the transducer to be calibrated to the audiometer. Couple the transducer to the test setup: artificial ear and sound level meter in case of a headphone, or artificial mastoid and sound level meter in case of a bone vibrator.
2. Select the function corresponding to the transducer to be calibrated, as given in Table A.1.
3. Start calibration mode, by simultaneously pressing the Patient-Response button and the OFF key on the keypad. The display shows the status as "Calibration Mode" and prompts "* Proceed (ON)?" and "* Exit (OFF)?". If OFF key is pressed, the instrument exits back to the operation mode. If ON key is pressed, the instrument proceeds with calibration. It shows the previous attenuation setting (Att.) for the selected frequency (Freq.) and continuously outputs the corresponding tones through the transducer. The default frequency for calibration is 1 kHz and can be changed

using frequency up/down keys. The attenuation setting appears as a 3-digit number and it controls the voltage being output to the transducer. It can be changed using stimulus dB up/down keys.

4 Select the tone frequency by using frequency up/down key. Using stimulus dB up/down keys, adjust the acoustic output level such that the sound level meter reads 90.0 dB SPL. The calibration for the selected frequency may be stored by pressing SAVE key.

5 Go to step 4, for selecting another frequency for calibration. Alternatively, exit from the calibration mode back to the operation mode, by simultaneously pressing the Patient-Response button and the OFF key on the keypad.

It may be noted that in the calibration mode, the earlier attenuation setting is changed only if SAVE key is pressed. Calibration for a transducer may be carried out for any sequence of frequencies, and it need not be carried out for all the frequencies. The same calibration procedure is to be used for each of the transducers. It may also be noted that the calibration of noise levels is automatically carried out internally according the stimulus tone calibration levels.

Table A.1 Function codes for calibration of different transducers

Transducer to be calibrated		Left headphone	Right headphone	Bone vibrator
Function code		LAC	RAC	LBC or RBC

A.5 Specifications

Tone Audiometry

- Stimuli
  Pure tones: total harmonic distortion: <1%
  Pulsed tones: 200 ms on, 200 ms off, with 50 ms rise/fall time
  Warble (FM) tones: frequency modulation with 5 Hz sine wave and ±5% deviation.

- Frequencies: 125, 250, 500, 750, 1 k, 1.5 k, 2 k, 3 k, 4 k, 6 k, 8 k Hz

- Test Administration: Manual & Automated

SISI Test

- Stimuli: Pure tone with level incremented by 1 dB for 200 ms, after every 5 s.

- Frequencies: 125, 250, 500, 750, 1 k, 1.5 k, 2 k, 3 k, 4 k, 6 k, 8 k Hz

- Test Administration: Automated

Tone Decay Test

- Stimuli: Pure tone presented for 1 min.

- Frequencies: 125, 250, 500, 750, 1 k, 1.5 k, 2 k, 3 k, 4 k, 6 k, 8 k Hz

- Test Administration: Automated

Speech Audiometry

- Stimuli: Speech signal from the microphone or the line input, or from the sound files stored on the internal SD card.

- Test Administration: Manual

Masking Noise

- Broad-band (BB) noise: Pseudo-random white noise, band-pass filtered with 3-dB cutoff of 100 Hz and 12 kHz

- Narrow-band (NB) noise: White noise band pass filtered with the center frequency equal to the stimulus tone frequency and 3-dB bandwidth of one-half octave.

Level control: Increment/decrement in steps, with limits as given Table.A.1.

User interface

- 128x64 LCD with switchable backlight
- 4x4 keypad

Data communication

Serial (RS 232) with TXD, RXD, ground pins for transferring of test results for preparation of test reports

Table.A.2 dB HL limits vs. frequency for two headphones

Freq. (Hz)	TDH39P BS1045 (Left)	TDH39P BS1055 (Right)
125	60	60
250	90	95
500	130	115
750	105	115
1000	100	115
1500	110	115
2000	100	120
3000	105	125
4000	110	115
6000	105	120
8000	90	105

Connections

- Power input with +9 V, -9 V, and ground: 3-pin connector
- Left headphone, right headphone, and bone vibrator: 3 mono jacks (1/4")
- Patient response button: mono jack (1/8")
- Ext. audio input: mono jack (1/8")
- Serial communication with PC: DB-9 (pins 2, 3 and 5)

Front panel

- Display and keypad

Side panel

- Sockets
    - Audio input
    - Bone conductor
    - Right headphone

- Left headphone
- Patient response button

- Connector
  - 3-pin power supply (DC ±9 V, Ground)
  - Serial connector

Accessories

- Headphones (2)
- Bone vibrator
- Response button with feedback light
- Microphone

Power

- Supply voltage: DC ±9 V, Ground,
- Current

Supply	Max presentation		No presentation	
	Backlight on	Backlight off	Backlight on	Backlight off
+9 V	490 mA	460 mA	360 mA	320 mA
-9V	360 mA	360 mA	200 mA	200 mA

Appendix B
HEAD-PHONE CHARACTERISTICS

Impedance as a function of different head-phone models was measured. Different types of headphones were calibrated by using B&K artificial ear type 4153 to determine the voltage levels for driving the headphone. The voltage levels in dBm were calculated using 0 dBm as a reference (i.e. a voltage required to produce 1 mW of power in 600 Ω load, i.e. 0.774 V_{rms}). The impedance and voltage levels for various transducers are given in Table B.1 to B.5.

Table B.1 Impedance and driving characteristics, model B51045

Freq. (Hz)	Impedance		Input for 100 dB SPL	
	Resistance (R)	Inductance (X)	mV p-p	dBm (ref = 774 mV rms)
125	9.80 Ω	291.5 mΩ	988	-6.9
250	9.84 Ω	591.4 mΩ	309	-17
500	9.99 Ω	1.62 Ω	127	-24.7
750	13.20 Ω	-9.93 Ω	50.9	-32.7
1000	9.96 Ω	-792 mΩ	336	-16.3
1500	10.03 Ω	49.52 mΩ	256	-18.6
2000	10.11 Ω	612.6 mΩ	216	-20.1
3000	10.25 Ω	1.01 Ω	245	-19.0
4000	10.32 Ω	1.48 Ω	191	-21.2
6000	10.55 Ω	2.31 Ω	409	-14.6
8000	10.866 Ω	3.056 Ω	709	-9.8

Table B.2 Impedance and driving characteristics, model B51055

Freq. (Hz)	Impedance		Input for 100 dB SPL	
	Resistance (R)	Inductance (X)	mV p-p	dBm (ref = 774 mV rms)
125	9.4 Ω	123 mΩ	1020	-6.6
250	9.5 Ω	228 mΩ	328	-16.5
500	9.5 Ω	419 mΩ	163	-22.6
750	9.5 Ω	597 mΩ	126	-24.8
1000	9.6 Ω	738 mΩ	114	-25.6
1500	9.8 Ω	925 mΩ	117	-25.4
2000	9.9 Ω	1.1 Ω	120	-25.2
3000	10.1 Ω	1.3 Ω	77	-29.1
4000	10.0 Ω	1.8 Ω	106	-26.3
6000	10.2 Ω	s2.5 Ω	195	-21
8000	10.6 Ω	3.3 Ω	363	-15.6

Table B.3 Impedance and driving characteristics, model DR52 (Black)

Freq. (Hz)	Impedance		Input for 100 dB SPL	
	Resistance (R)	Inductance (X)	mV p-p	dBm (ref = 774 mV rms)
125	7.5 Ω	155 mΩ	600	-11.2
250	7.6 Ω	130 mΩ	241	-19.2
500	7.5 Ω	174 mΩ	150	-23.3
750	7.5 Ω	366 mΩ	222	-19.9
1000	7.5 Ω	451 mΩ	120	-25.2
1500	7.8 Ω	600 mΩ	94	-27.3
2000	7.7 Ω	745 mΩ	89	-27.7
3000	7.8 Ω	1.1 Ω	83	-28.3
4000	7.9 Ω	1.4 Ω	103	-26.5
6000	8.1 Ω	2.0 Ω	259	-18.5
8000	8.2 Ω	2.4 Ω	541	-12.1

Table B.4 Impedance and driving characteristics, model DR52 (white)

Freq. (Hz)	Impedance		Input for 100 dB SPL	
	Resistance (R)	Inductance (X)	mV p-p	dBm (ref = 774 mV rms)
125	7.5 Ω	146 mΩ	594	-11.3
250	7.6 Ω	102 mΩ	231	-19.5
500	7.5 Ω	214 mΩ	133	-24.3
750	7.5 Ω	390 mΩ	142	-23.8
1000	7.6 Ω	556 mΩ	91	-27.6
1500	7.7 Ω	595 mΩ	76	-29.2
2000	7.7 Ω	789 mΩ	85	-28.2
3000	7.8 Ω	1.1 Ω	73	-29.4
4000	8.0 Ω	1.5 Ω	113	-25.7
6000	8.2 Ω	2.0 Ω	213	-20.2
8000	8.3 Ω	2.5 Ω	725	-9.6

Table B.5 Impedance and driving characteristics, model TDH39P-18778

Freq. (Hz)	Resistance (R)	Inductance (X)	Voltage (mV p-p) for 100 dB SPL	dBm (ref = 774 mV rms) for 100 dB SPL
125	9.6 Ω	146 mΩ	1240	-4.9
250	9.7 Ω	284 mΩ	388	-15.0
500	9.7 Ω	498 mΩ	198	-20.9
750	9.8 Ω	639 mΩ	159	-22.8
1000	9.9 Ω	803 mΩ	153	-23.1
1500	10.1 Ω	1.1 mΩ	169	-22.2
2000	10.2 Ω	1.3 mΩ	172	-22.1
3000	10.6 Ω	1.6 Ω	106	-26.3
4000	10.4 Ω	1.9 Ω	96.3	-27.1
6000	10.7 Ω	2.6 Ω	236	-19.3
8000	11.0 Ω	3.4 Ω	678	-10.2

Appendix C
COMPONENT LIST

Designator	Part Number/value	Component description	Quantity
C1, C3, C7, C8, C9, C10, C11, C12, C13, C14, C15, C17, C18, C19, C20, C21, C23, C25, C28, C29, C30, C31, C32, C33, C34, C35, C38, C39, C40, C41, C42, C43, C44, C46, C49, C50, C52, C53, C54, C57, C58, C59, C64, C65, C66, C67, C70, C71, C72, C73, C74, C75, C76, C82, C83, C84, C85, C86, C87, C88	0.1 µF	Capacitor (Ceramic, through hole-TH)	60
C2, C4, C6, C22, C24, C26	1000 µF/16 V	Capacitor (Electrolytic, TH)	6
C5, C16, C27, C45, C61, C63, C69, C77, C79, C81	100 µF/16 V	Capacitor (Electrolytic, TH)	10
C36,C37,C51	0.1 nF	Capacitor (Ceramic, TH)	3
C47, C48, C55, C56, C60, C62, C68	10 µF	Capacitor (Electrolytic, TH)	7
C78, C80	22 pF	Capacitor (Ceramic, TH)	2
R1, R2, R3, R4, R5, R6, R8, R9, R10, R11, R12, R13	33 Ω ± 5 %, ¼ W	Resistor (TH)	12
R7, R18, R20, R23, R29, R30	12 Ω ± 5 %, ¼ W	Resistor (TH)	6
R14, R15, R16, R17, R19, R21, R22, R24, R25, R26, R27, R28, R31, R38, R40, R42, R43, R49, R63, R64	100 kΩ ± 5 %, ¼ W	Resistor (TH)	20
R32, R34, R36, R37, R41, R44, R45, R48, R50, R51, R52, R53, R55, R56, R57, R58,R59, R60	10 kΩ ± 5 %, ¼ W	Resistor (TH)	18
R35, R39, R33	33 kΩ ± 5 %, ¼ W	kΩ, ¼ W (TH)	3
R46, R47	470 Ω ± 5 %, ¼ W	Resistor (TH)	2

R54	8 Ω ± 5 %, ¼ W	Resistor (TH)	1
R61	4.7 kΩ ± 5 %, ¼ W	Resistor (TH)	1
R62	3.9 kΩ ± 5 %, ¼ W	Resistor (TH)	1
R65	1 kΩ ± 5 %, ¼ W	Resistor (TH)	1
R66	5.6 kΩ ± 5 %, ¼ W	Resistor (TH)	1
RP1	10 kΩ, ¼ W	Resistor pack-8 (SIP-8)	1
RP2	10 kΩ, ¼ W	Resistor pack-7 (SIP-7)	1
D1, D2	1N4148	Diode (DO-35)	2
T1	BC557	p-n-p transistor (TO-92)	1
L1	LED	LED(5 mm cylindrical package)	1
U1, U2, U3	LM1877 (DIP-14)	Power amplifier	3
U4	LM7905C (TO-220)	-5 V regulator	1
U5, U19	LM7805C (TO-220)	+5 V regulator	2
U6, U9,U10	CD4053 (DIP-16)	Multiplexer	3
U7, U8, U14, U15	MCP602 (DIP-8)	Op Amp	4
U11	PGA21310 (DIP-16)	Attenuator	1
U12	SN74HC373 (DIP-20)	Latch	1
U13	MCP601 (DIP-8)	Op Amp	1
U16	PIC16F1939 (DIP-40)	8-bit PIC microcontroller	1
U17, U20	LM1117(SOT-223)	3.3 V regulator	2
U18	dSPIC33FJ128GP802 (DIP-28)	16-bit DSC controller	1
U21	MAX232 (DIP-16)	RS232 IC	1
Rout, Lout, Bout		1/4” mono phone jack	3
Switch, Speech in		1/8” mono phone jack	2
JP1, JP2		2-pin jumper	2
CN4		3-pin connector	1
CN5, CN6		6-pin connector	2
CN1		8-pin connector	1
CN2		20-pin connector	1
CN7		DB9 female connector	1
CN3	SD card connector (SMD)	SD card connector	1
G1	LGM12641BS1R	Graphical LCD	1
K1	4x4 Keypad	keypad	1
SD	SD card	SD card	1

Appendix D

PCB LAYOUT OF THE AUDIOMETER

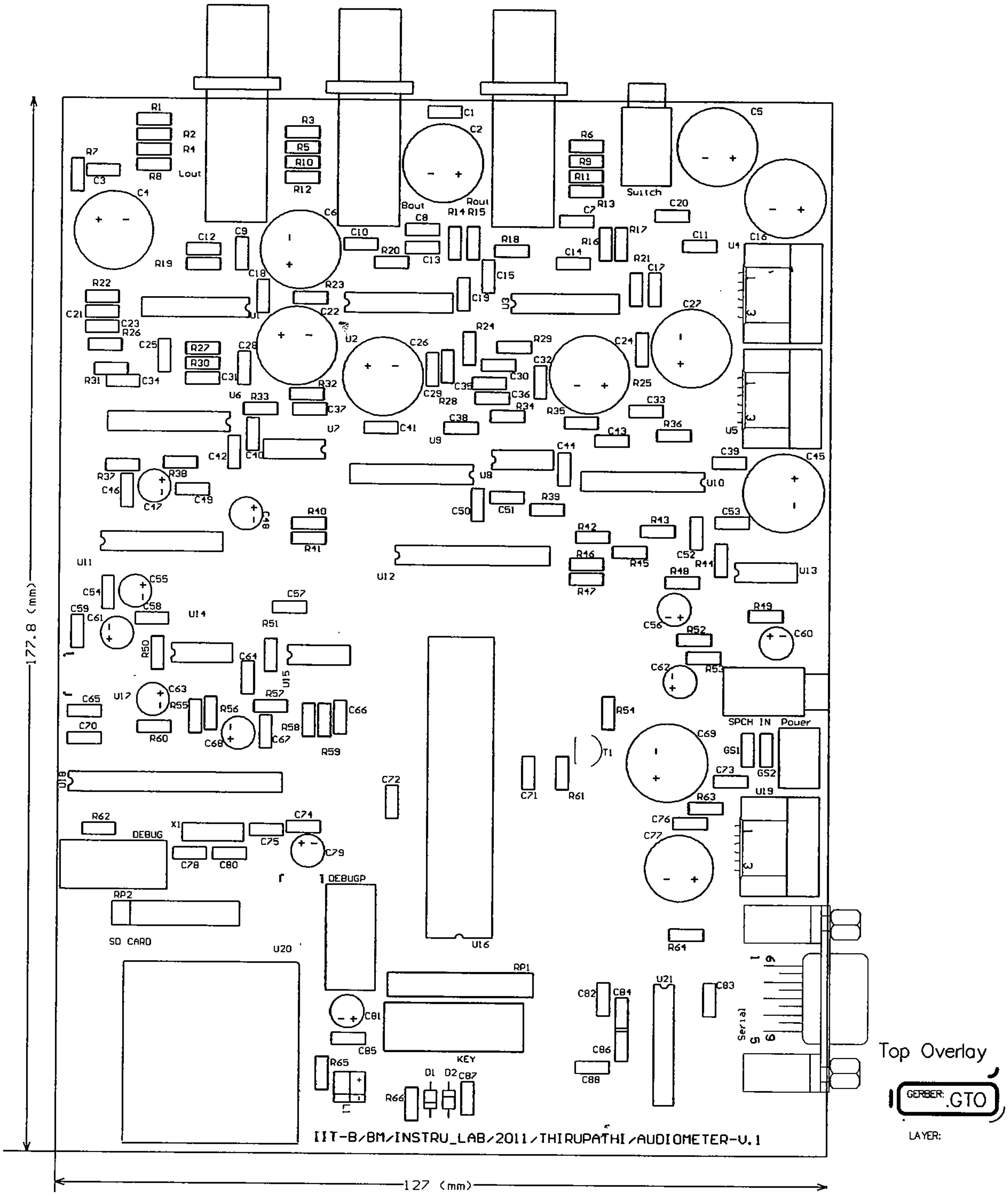


Figure D.1 Top over lay of the audiometer PCB.

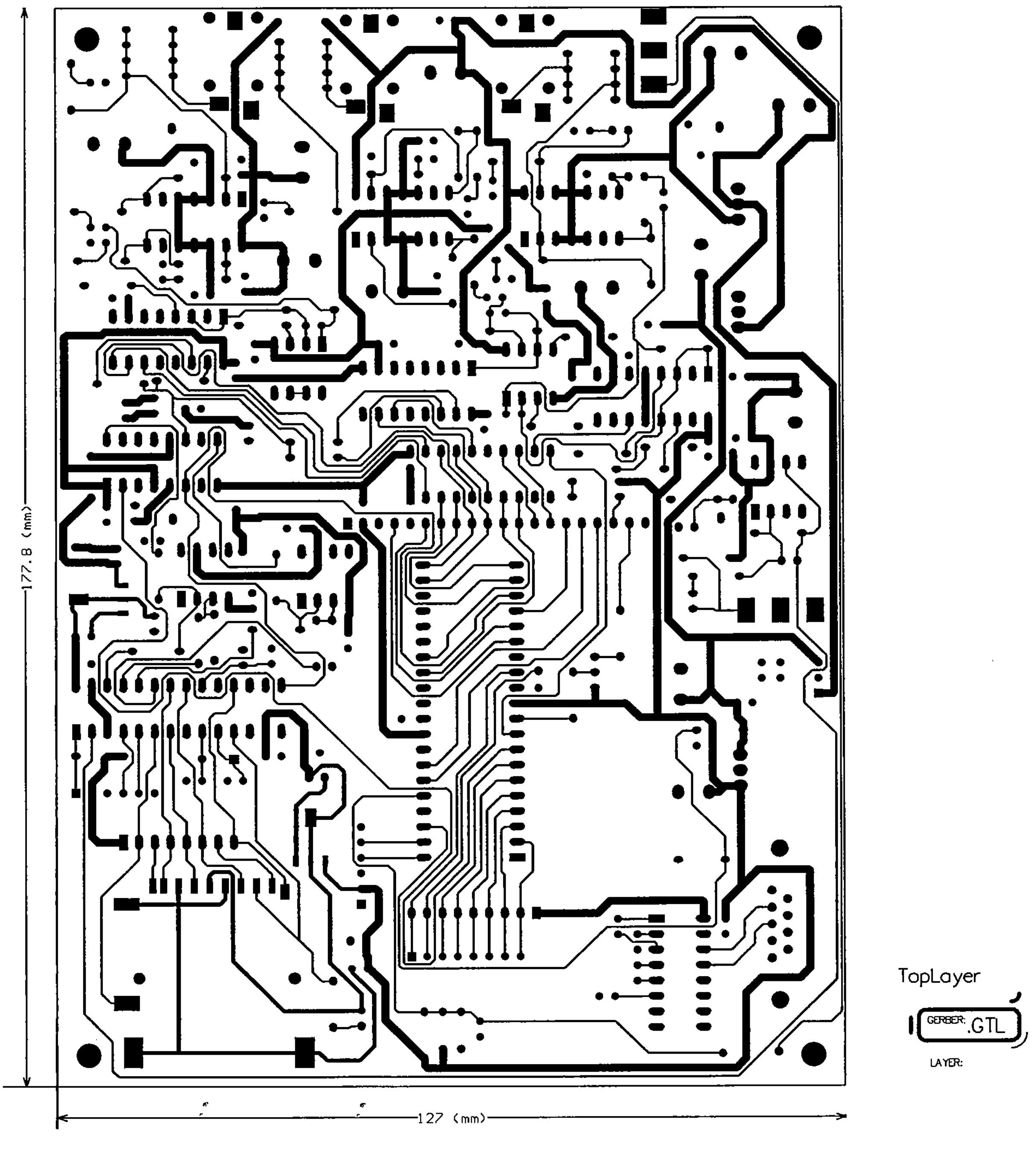


Figure D.2 Top layer of the audiometer PCB.

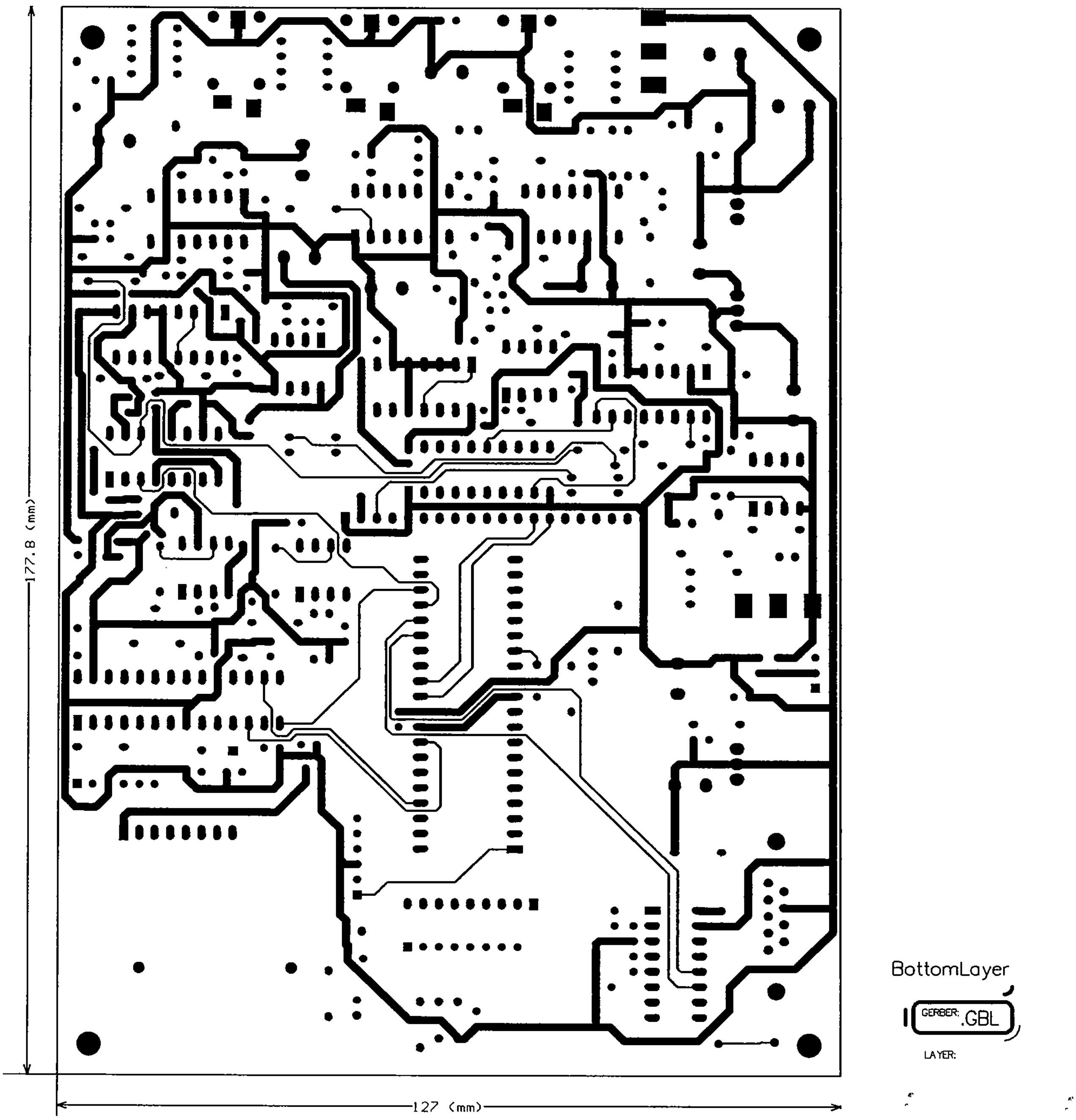


Figure D.3 Bottom layer of the audiometer PCB.

Appendix E

SCHEMATIC DIAGRAM OF AUDIOMETER

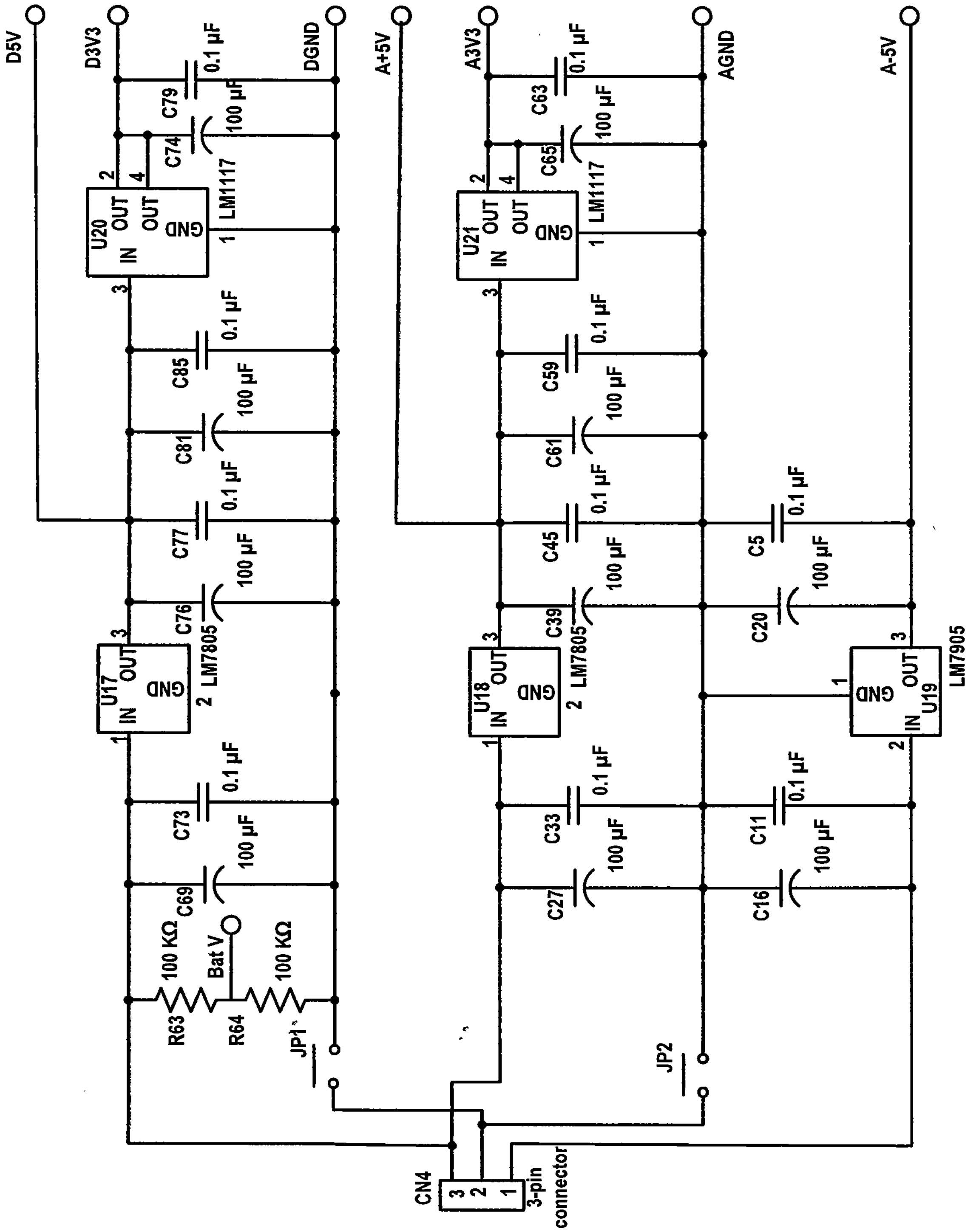


Figure E.1 Audiometer schematic sheet-1

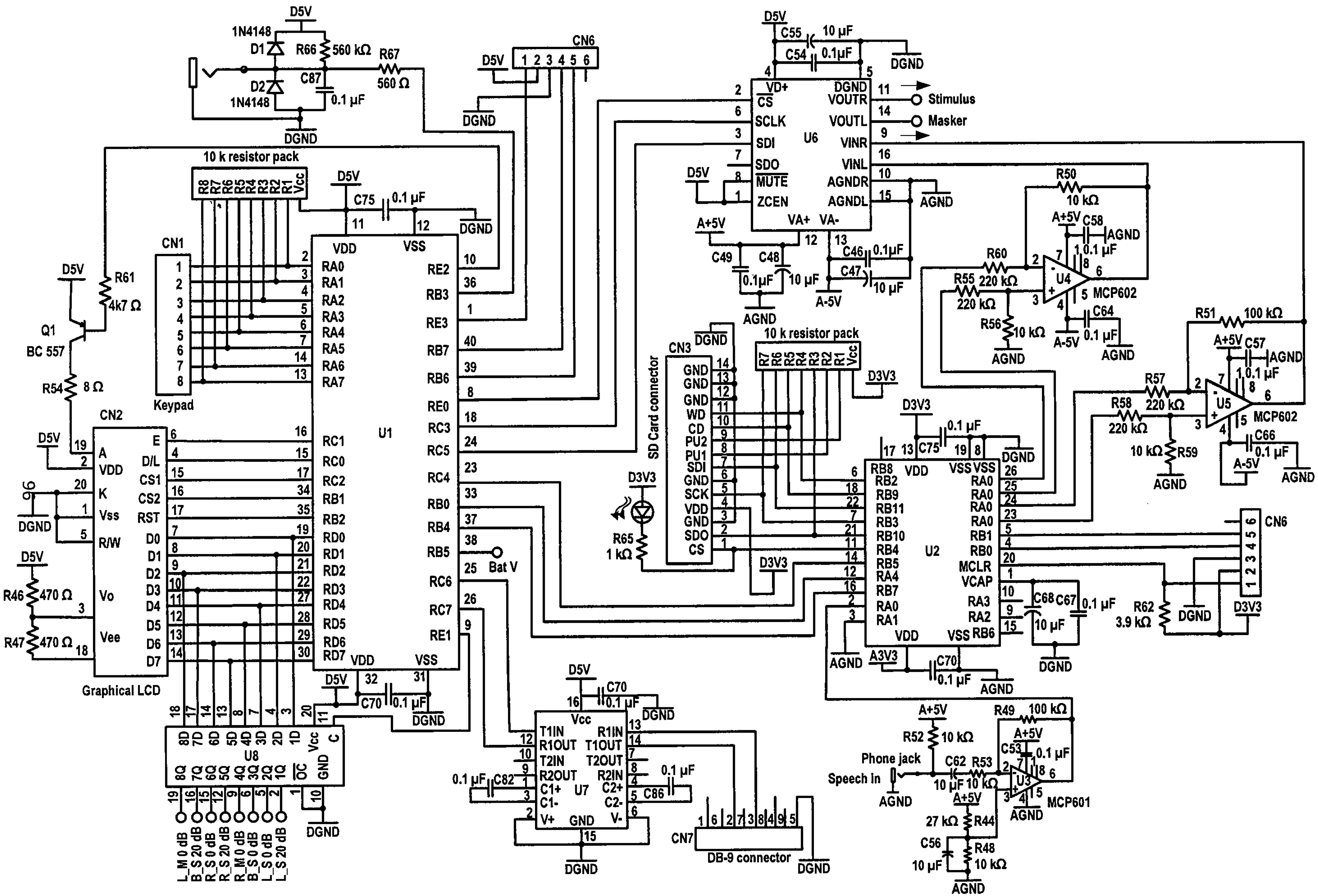


Figure E.2 Audiometer schematic sheet-2

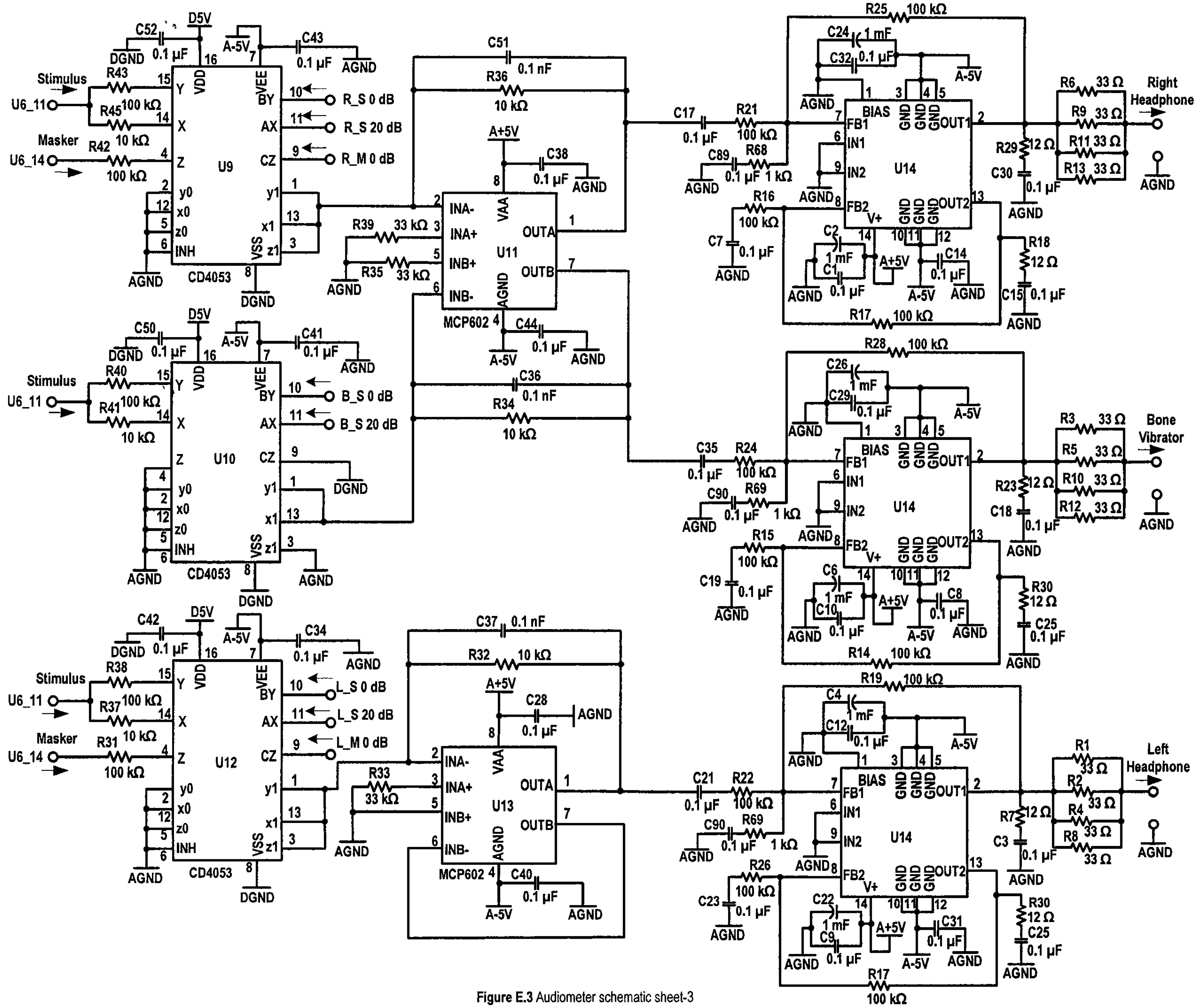


Figure E.3 Audiometer schematic sheet-3

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I dedicate my work to my parents and to my sister Sudha.

K. Thirupathi

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