A MICROCONTROLLER BASED AUDIOMETER

A dissertation submitted in

partial fulfillment of the requirements for the degree of

Master of Technology

by

Siraj Fulum Mossa

Roll Number 08307201

Under the supervision of

Prof. P. C. Pandey

Department of Electrical Engineering

Indian Institute of Technology Bombay

June 2010

Indian Institute of Technology Bombay

Dissertation Approval

This dissertation entitled **"A microcontroller based audiometer"** by **Siraj Fulum Mossa (Roll No. 08307201)** is approved, after the successful completion of viva voce examination, for the award of the degree of **Master of Technology in Electrical Engineering.**

Examiners:	M. C. Chand	(Prof. M. C. Chandorkar)
	L R Subramanya	(Prof. L. R. Subramanyan)
Supervisor:	PCPandey	(Prof. P. C. Pandey)
Chairman:		(Prof. K. P. Karunakaran)

Date: 30th June 2010

Place: Mumbai

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(Signature)

Siraj Fulum Mossa
(Name of the student)

08307201
(Roll No.)

Date: 11/7/2010

Siraj F. Mossa / Prof. P. C. Pandey (Supervisor), "A Microcontroller Based Audiometer", *M. Tech. dissertation*, Department of Electrical Engineering, Indian Institute of Technology Bombay, June 2010.

ABSTRACT

An audiometer is an instrument for quantifying hearing impairment. Test tones of different frequency and level are presented and hearing thresholds are determined on the basis of subject's response. In this project, a microcontroller based instrument is developed to provide two-channel speech and pure-tone audiometry with air conduction and bone conduction tests with the facilities of manual and automated test administration. Other audiometric tests like tone decay test and short increment sensitivity index test (SISI) can also be conducted. The instrument is developed using a general purpose 8-bit microcontroller as the core for all the interfacing and data handling. A direct digital synthesizer is used for generating low distortion tones and a DSP microcontroller with internal D/A converter is used for synthesizing white noise and band-pass filtered noise. Programmable attenuations are used for level control. It provides pure tone and warble tone stimuli in the frequency range of 125 Hz to 8000 Hz and acoustic output level of 0 to 140 dB in 5 dB steps. The instrument is controlled through a 4×4 keypad and 128×64 graphic display. The subject response can be given using the response switch or through the keypad. The instrument uses an internal look-up table for relating the dB HL levels to attenuator steps as a function of frequency and can be easily calibrated for any headphone or vibrator.

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Chapter 1
INTRODUCTION

1.1 Background

Pure-tone threshold audiometry is a procedure used in assessing a patient's threshold of hearing for pure tones at octave intervals from 125 to 8000 Hz and at intermediate frequencies as needed. Pure-tone threshold audiometry includes manual and automated modes with air-conduction and bone-conduction tests. In air-conduction tests, the sound is presented through headphones or speakers and gets conducted to the inner ear through the ear canal, the ear drum and the three bones in middle ear. In bone conduction test, sound is presented through a bone vibrator and it gets conducted to the inner ear through the cranial bones. The pure-tone audiometer consists of an oscillator, attenuator, multiplexer, power amplifiers and a response switch. Oscillator is used for generating pure-tone sounds of various frequencies usually 125, 250, 500, 750, 1000, 1500, 2000, 3000, 4000, 6000, and 8000 Hz. The pure tone is attenuated by the attenuator for level selection and applied to the power amplifier. The sound is presented through calibrated headphones or speakers for ear conduction and through a bone vibrator for bone conduction.

There are four audiometry testing methods, pure-tone threshold audiometry, tone decay test, speech audiometry, and short increment sensitivity index test. Pure-tone stimuli signals can be generated by either analog or digital circuits. Use of digital circuits provides precision in the control of frequency and level, and hence they are being increasingly employed in audiometery [1], [3].

1.2 Project objective

Objective of this project is to develop a microcontroller based diagnostic aduiometer by investigating the shortcoming and limitations of earlier designs. Earlier audiometers developed at IIT Bombay was studied. Switched-capacitor based oscillator circuit was used to generate pure tone of different frequencies. And maximal length feedback shift register was used to generate pseudo random binary sequence (PRBS). Output of PRBS was low pass filtered with cut of frequency 10 kHz, to generate wide band noise. The narrow band noise was generated using switched-capacitor based filter to make the frequency of pure tone as the center frequency of the band. This design put constraints on amplitude stability levels and spectral purity of the tones. To provide flexibility of using a variable frequency with high stable amplitude, a programmable waveform generator based on direct digital synthesizer (DDS) is tested and used. Further warble tone could be generated with small number of steps in frquency modulation of the tone. Log attenuator with 120 dB dynamic range in steps of 5 dB is used for tone and noise level control. Masking noise is generated using DSP chips with on-chip DAC.

1.3 Report outline

Chapter 2 covers the basics of the audiometer and audiometric techniques. Chapter 3 describes design approach. Work done on hardware is presented in the fourth chapter. Software, system assembly, test and result are presented in the fifth chapter. The last chapter summarizes the work and lists suggestion for future work.

Chapter 2
AUDIOMETER BASICS

2.1 Introduction

Hearing can be affected by a wide range of hearing defects, caused by exposure to intense sounds or infection. Hearing loss due to the outer and/or middle ear is called conductive loss; where as hearing loss due to the inner ear or auditory nerves is called sensorineural loss [2]. The nature and degree of the hearing impairment should be assessed in order to diagnose its causes so that suitable treatment or appropriate hearing aids can be prescribed. An audiometer is used for presenting the test tones of different frequency and levels to the subject and hearing thresholds are determined based on the subject's response.

Pure-tone audiometry is a technique of ascertaining the hearing threshold level of a subject for pure tone sounds of various frequencies. The result, when plotted graphically, is called pure-tone audiogram. Although this method can assess the degree of hearing loss, it cannot characterize the extent of impaired speech discrimination (understanding of speech) [1], [4]. Speech audiometry measures hearing sensitivity for spoken material. In addition, spoken material can be used to test for speech discrimination. Audiometric tests are generally conducted in an audiometry room, with a consistent and controlled acoustic environment and with background noise kept at an acceptably low level to provide a calibrated acoustic test.

2.2 Hearing loss

In hearing, the sound is conducted through three different parts of the ear: the outer ear, the middle ear and the inner ear, as shown in Fig. 2.1. The outer ear consists of the external ear and the ear canal which is terminated in the ear drum. Behind the ear drum is the middle ear, a small cavity with three small bones. The inner ear consists of two sections, the cochlea and the vestibular mechanism which is responsible for the sense of balance [5], [6]. When sound wave strikes the ear, it enters the ear canal and causes the eardrum to vibrate. The vibration is transmitted to the cochlea by the three bones of the middle ear. The hair cells in the cochlea sense the vibrations and generate electrical depolarizations which are converted by the fibres of the auditory nerve as electrical pulses and conducted to the brain where it is perceived as sounds.

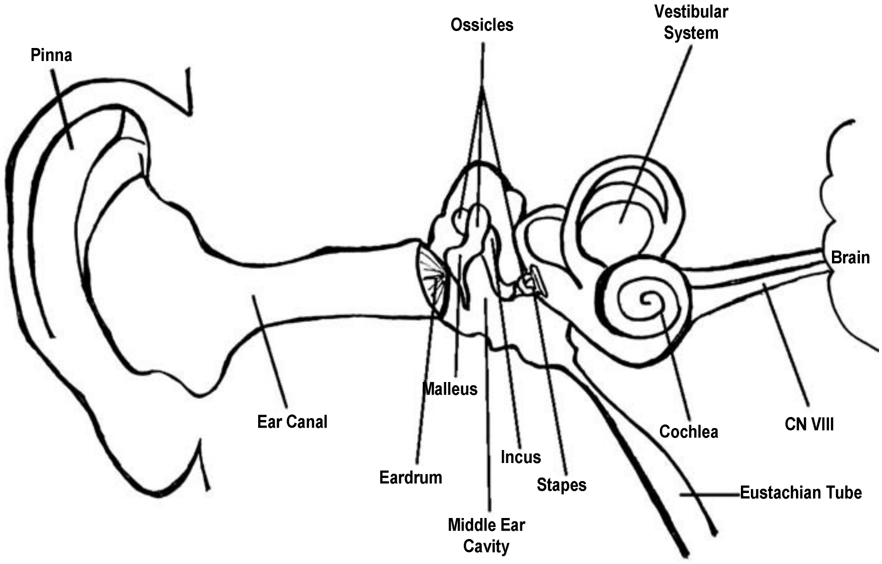

Fig. 2.1 The peripheral auditory system: outer ear, middle ear and inner ear [6].

The cochlea performs a frequency analysis on incoming sounds. Hair cells that respond to high-frequency sounds are located at the end of the cochlea closest to the middle ear, whereas those that respond to low frequency sounds are located in the end farthest from the middle ear, as shown in Fig. 2.2. The travelling wave on the basilar membrane peaks when it reaches the hair cells tuned to the frequency of the incoming sound, and then fades away. Damage to a specific place in the cochlea results in a hearing loss affecting specific frequencies. For example, damage to the hair cell in the base of the cochlea (closest to the middle ear) results in a high-frequency hearing loss.

The cochlea is the sensory portion of the auditory system, and CN VIII (cranial nerve eight) is the neural portion. Together, the cochlea and CN VIII make up the sensorineural mechanism. Problems affecting only the cochlea cause a sensory hearing loss, and those affecting only CN VIII cause a neural hearing loss. Problems affecting either (or both) cause a sensorineural loss. Sensory loss is far more common than neural loss [1], [2]. Problems in the sensorineural portion of the auditory system results in increased bone conduction thresholds. Air conduction thresholds are also increased because the air conduction pathway also includes the damaged cochlea or CN VIII. In a purely sensorineural loss, the conductive mechanism is perfectly normal. An example of a purely sensorineural loss is one caused by prolonged exposure to loud noise. When both air conduction and bone conduction thresholds are outside the normal range (>20 dB), and the bone conduction thresholds are better than air conduction thresholds, a conductive hearing loss and a sensorineural hearing loss exist in the same ear, and the condition is known as a mixed hearing loss. An example of a mixed hearing loss is one caused by prolonged exposure to loud noise and an infection in the middle ear.

Hearing problems caused by sensorineural loss are worse than those caused by conductive loss. In the case of conductive loss, sound information reaching the cochlea is essentially unchanged except for a reduction in loudness. This means that all sounds are softer, but they are not distorted. If sounds can be made louder (using hearing aids or an assistive listening device), they're usually quite clear. With sensorineural loss, the signal is likely to be distorted to some degree. Even when sound can be made loud enough, it may not be perfectly clear [1]. Conductive hearing loss tends to affect all frequencies more or less equally. In contrast, sensorineural loss is likely to be worse at the higher frequencies, reducing (or eliminating) important high-frequency information. The high-frequency part in the speech contributes roughly 80 percent of the information necessary for speech understanding, whereas the low-frequency part contributes roughly 20 percent. It has been estimated that low-frequency part of speech contributes about 80 percent of the volume of speech, whereas high-frequency part contributes only 20 percent [6].

2.3 Measurement of hearing loss

The procedure for determining hearing threshold begins with familiarizing the patient with the stimuli. The test proceeds with a protocol requiring positive responses to stimuli, pure or speech, presented in a sequence [7]. The weakest sound heard at a selected frequency is the hearing threshold for that particular frequency. The sound level is measured in dB hearing level (HL). The reference sound level of 0 dB HL corresponds to the average threshold for normal healthy young adults in the age group of 18 to 25 years. The sensitivity of the normal ear varies with frequency; therefore, 0 dB HL represents different levels of sound pressure at different frequencies [2].

In air conduction test, the test tones or speech is presented using headphone or a speaker. The patient is asked to respond by raising hand or pressing a response button. The patient should respond every time he/she hears a tone, even if it's very faint. Although the healthy human ear is capable of hearing across a wide range of frequencies, from 20 to 20000 Hz, only the range of 125 Hz to 8000 Hz is important for speech understanding. The audiologist measures thresholds at eleven different frequencies in each ear and plots the on a graph called audiogram. It gives a plot of the softest sounds the patient can hear at different frequencies in each ear. This shows the degree of hearing loss, hearing in the right ear as compared to the hearing in the left ear, and the shape of the hearing loss.

The bone conduction testing is carried out by presenting the sound through a bone vibrator that is placed on the bony bump behind the patient ear or on the forehead. The sound is coupled to the cranial bones and gets conducted to the inner ear bypassing the outer and middle ear.

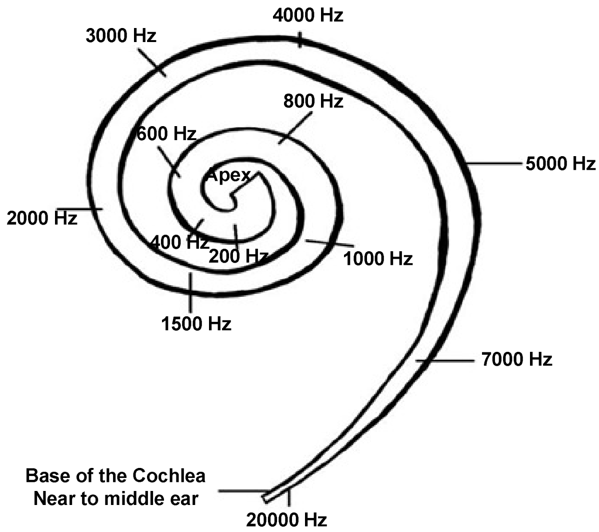

Fig. 2.2 Frequency organization of the cochlea [6].

The bone conduction testing is carried out by presenting the sound through a bone vibrator that is placed on the bony bump behind the patient ear or on the forehead. The sound is coupled to the cranial bones and gets conducted to the inner ear bypassing the outer and middle ear. Comparing the air conduction thresholds to the bone conduction thresholds allows the audiologist to decompose the hearing loss into conduction and sensorineural losses [2], [8].

We hear mostly by air conduction in everyday life. Air conduction testing tests the entire auditory pathway. The sound wave enters the ear canal, and it causes vibrations of the eardrum and the bones to which it is attached (the ossicles). Movement of the innermost ossicle, the stapes, causes waves in the cochlear fluids. Disturbance of the cochlear fluids cause tiny hairs on top of the hair cells to bend, and electrical depolarization in the hair cells, which trigger neural impulses in the fibres of the auditory nerve. These impulses are transmitted through the central auditory pathways to auditory areas in the brain. Air conduction thresholds reflect the healthiness of the entire auditory system. When we hear by the bone conduction, the outer ear (pinna and ear canal) and middle ear (eardrum and ossicles) are bypassed, and the cochlea is stimulated directly. Because the cochlear fluids are contained within the bones of the skull, the skull vibration causes a disturbance of the liquid inside the cochlea. Hearing through the bone conduction is not ear specific as the fluids inside both cochleas are disturbed irrespective of the position where the bone vibrator is placed. As with air conduction hearing, disturbance of the cochlear fluids by bone conduction causes the tiny hairs on top of the hair cells to bend and trigger neural impulses.

Subjective pure tone audiometry is the most common way to measure hearing loss and is used effectively at any age. It measures an individual’s threshold for hearing at different frequencies. Speech audiometer measures hearing acuity by presenting speech stimuli to an individual through the use of a headphone or insertion of ear phones. Speech audiometry is used to establish levels of detection, reception, and perception of speech [4].

The objective audiometric tests [6] do not require response from the patient. These tests are especially useful for infants, children, or individuals with severe multiple needs because they do not require the client to raise a hand or respond to words or pressing response switch.

For adults, thresholds below 20 dB are considered normal. Thresholds greater than 20 dB indicate some degree of hearing loss. Hearing loss can be characterized by its type, symmetry, configuration, and severity. A comparison of the air conduction thresholds to the bone conduction thresholds from the same ear tells us whether the hearing loss is conductive. Problems in the conductive mechanism (the outer ear, the eardrum and the middle ear), do not affect the bone conduction thresholds, because bone conduction testing bypasses the part of the auditory system where the problem lies. Air conduction thresholds will not be normal because air-conducted sound must travel through the entire auditory system, including the conductive mechanism. When an audiogram shows normal hearing by bone conduction (thresholds less than 20 dB) and a hearing loss by air conduction, the hearing loss is conductive. In a purely conductive hearing loss, the remaining parts of the ear (the cochlea, CN VIII, and beyond) are perfectly normal. An example of a purely conductive hearing loss would be one caused by an infection in the middle ear [6].

2.4 Pure tone audiometry

In pure tone audiometry, the subject's response for acoustic stimuli of different frequencies is measured. Although human hearing ranges from 20 Hz to 20 kHz, there is little speech information above 8 kHz, and the perception of the frequencies below 100 Hz is increasingly tactile in nature, making them difficult to assess [2]. Hearing loss starts at high frequency (8 kHz) and later on as the loss progresses, its effect is observed in the mid frequency region 1 to 2 kHz as well. Audiometric tests are generally conducted in the range of 125 Hz to 8 kHz, often at frequencies of 125, 250, 500, 750, 1000, 1500, 2000, 3000, 4000, 6000, and 8000 Hz [8].

Pure tone audiometry helps in determining whether the loss is conductive (disorder in external auditory meatus and / or middle ear) or sensorineural (disorder in the inner ear or in the nerve of hearing in the brain) or mixed. Initially a pure tone of 40 dB HL or another level as selected by the audiologist is presented to the subject. If the patient does not respond to the initial set point, the tone level is raised in steps of 10 dB until the patient gives a response. If the patient responds, the tone level is decreased by 10 dB. If the patient does not hear it, the tone is again raised by 5 dB. In this way, by several presentations, the hearing threshold is obtained as the minimum presentation level at which the patient responds at least 50 % [2], [7]. The hearing threshold is obtained for various test frequencies. Different shapes of audiogram are associated with different types of hearing loss. The difference between air conduction and bone conduction thresholds at a given frequency reflects conductive hearing loss. The difference between air conduction thresholds and bone conduction thresholds of 0 dB reflects sensorineural loss.

2.5 Tone decay test (TDT) and short increment sensitivity index test (SISI)

Tone decay test is used to diagnose the sensorineural deafness [1]. In this case a tone at a particular frequency, with intensity equal to the hearing threshold, is presented as a continuous tone. The time up to which the patient is able to hear the tone is recorded. If the patient is not able to hear the tone continuously for more than one minute, the tone intensity will be increased by 5 dB and test will be conducted for one minute. The lowest intensity for which the patient is able to hear the tone for at least one minute is considered as the threshold for tone decay test [9], [10]. Similarly, the testing is done for other frequencies and the relation between the threshold and the frequencies are obtained.

The short increment sensitivity index test determines the capacity of a patient to detect a brief 1 dB increment in 20 dB suprathreshold (above tone hearing threshold) tone at various frequencies [9], [10]. A brief increment in intensity is provided at 5 second intervals, and patient is asked to respond if he/she senses the increment in intensity. Twenty such increments are presented and the number of increments the patient is able to recognize correctly is noted. SISI audiogram is plotted as a percent score for each test frequency [1]. A score of 30 to 40 % is considered as normal score. This type of test is used to detect the pathology in cochlear or deafness due to nerve disorder [11].

2.6 Speech audiometry

Speech is a broadband signal, and important information for speech comprehension is concentrated between 1000 and 4000 Hz. An estimate of importance of various octave bands in speech is as given in Table 2.1. For most of the people, hearing is poorer at the high frequencies (1000 to 8000 Hz) than at the low frequencies (250 to 500 Hz), making speech difficult to understand. Speech audiometry is used to establish levels of detection, reception, and perception of speech [4].

Table 2.1 Speech information versus frequency [6].

Frequency (Hz)	250	500	1000	2000	4000	8000
Speech Information (%)	8%	10%	22%	33%	23%	0%

A speech reception threshold (SRT) test measures the sound levels at which speech is detected and individual words are distinguished from one another. Spoken material is introduced as an input signal to the audiometer from a microphone, tape, or CD player. Speech discrimination test (SDT), assesses the patient's ability to understand mono-syllable words at 30 dB above the SRT level [9], [10]. While the pure tone thresholds help in assessing the losses in the peripheral auditory systems, the speech audiometry assesses the integrity of the entire auditory system by assessing the ability to hear clearly and to understand speech [4]. Sensorneural loss affects both the reception as well as the discrimination of speech while conductive loss affects only the reception.

Method for finding speech reception threshold is the same as for finding hearing threshold in pure-tone audiometry except that different words with equal stress are used as the stimuli. 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 [1].

In speech discrimination test, lists of monosyllable speech discrimination words are presented at 30 dB above the speech reception threshold, using headphones for each ear. The patient is then asked to repeat the word. The percentage of the total number of words presented, which the patient is able to indentify correctly gives the speech discrimination score (SDS). The result of this test is from 0 to 100 %. Higher score is associated with normal hearing or conductive hearing loss, but in case of neural lesions SDS is very poor.

2.7 Physiological tests

Hearing evaluation tests can be conducted using two methods; subjective tests in which the patient provides response and physiological tests (objective tests) in which the response fully depends on the involuntary body response rather than the voluntary responses from the patient. Although physiological tests are extremely useful, they do not measure hearing in the perceptual sense. Physiological measures simply tell us whether the peripheral auditory system is working properly or not [6].

2.7.1 Auditory brainstem response (ABR)

Human brain continuously generates spontaneous, random, brain waves. A portion of this brain wave can be recorded on the surface of the head using electrodes on the skin. This is referred to as electroencephalographic activity, or EEG [11], [12]. Other than EEG, it is possible to record brain waves generated in response to sounds that are presented to the listener. Auditory brainstem response (ABR) is a response to clicking noises through the earphones on the patient's ear. ABR testing is used to measure the hearing threshold of infants and young children. A relaxed state is required to obtain accurate results.

2.7.2 Otoacoustic emission

The hair cells inside of the cochlea bend when the cochlear fluids are disturbed. This bending creates an effect that triggers neural impulses that are sent to the brain for interpretation. There are two types of hair cells; outer hair cells and inner hair cells. It is the inner hair cells that actually trigger most of the neural impulses. Outer hair cells work to make inner hair cells more sensitive to sounds by expanding and contracting or making themselves larger and smaller. This is called hair cell dance. It creates sounds that travel back through the middle ear and out into the ear canal, where they can be measured with a miniature microphone. These sounds are called otoacoustic emissions (OAEs) [6], [11]. The presence of measurable OAEs suggests normally functioning outer hair cells and hearing that is probably normal or nearly normal. Most hearing loss involves loss of outer hair cells; the lack of an emission is a good predictor of hearing loss. But OAE require healthy middle ear function and relatively normal hearing to be useful. The test is performed by putting a plug into the ear canal. The test is quick and simple. It is commonly used to screen the hearing of newborn babies.

2.8 Masking in audiometry

In audiometry, the two ears are tested separately. In both air and bone conduction test when the sound is presented to the test ear, the cochlea of non-test ear also gets stimulated by transmission through the bone of the skull. The stimulation of the non test ear is called cross-hearing. During the air conduction test, the part of the test tone that stimulates the non test ear is called interaural attenuation and varies between 45 to 80 dB [1]. However, during bone conduction test, the cochlea of both sides are equally stimulated i.e. the interaural attenuation is of 0 dB. Hence, cross-hearing is more of a concern in case of bone conduction test than it is for air conduction. Whenever cross hearing is suspected, it is necessary to mask the non-test ear, by presenting a noise to the non test ear through the corresponding headphone. The masking noise should be loud enough to prevent the test tone from stimulating the non test ear, but at the same time it should not mask the sensitivity of the test ear. The masking noise is often selected to be a wide band noise which has uniform power density spectrum over the entire audible frequency range. Narrow band noise is effectively used for masking the pure tone, by making its center of frequency same as the test tone frequency. Its band width is 1/3 or 1/2 octave about the center frequency with attenuation rate outside the pass band of 12 dB / octave [13].

2.9 Pure tone audiometer

An instrument used to carry out pure tone audiometric test is called pure tone audiometer. The three major parts of a pure tone audiometer are the tone generator, the level controller unit, and power amplifier. The instrument generally also has a provision for air and bone conduction test with masking noise to the non-test ear. Fig. 2.3 shows a block diagram of pure tone audiometer. It consists of the pure-tone generator/oscillator, masking noise generator, attenuator, selector switches, power amplifier and the control unit. The pure tone generator generates pure tones of different frequencies with stable amplitude. The frequency ranges from 125 Hz to 8 kHz and each of the frequency should be within 3 % of the indicated frequency. The output of pure-tone generator is fed to attenuator to have corresponding attenuation in dB HL, and the output level of any harmonic distortion should be at least 40 dB below the fundamental level. Wide band and narrow band noise are generated using the masking noise generator. A switch is used to selecting either of them.

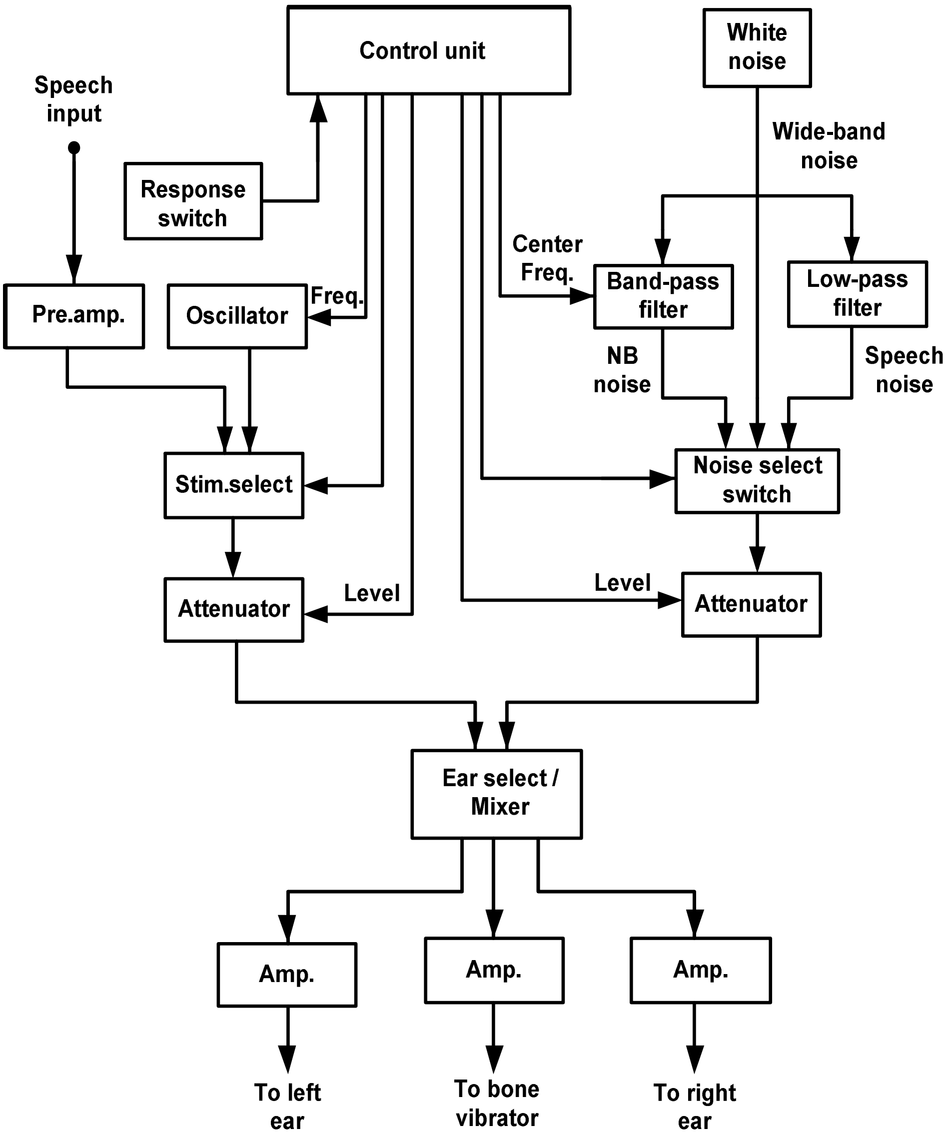

Fig. 2.3 Block diagram of a pure tone audiometer.

Pure tone audiometer is used to conduct pure-tone audiometric test, with the aim of finding the hearing threshold level for each frequency [7]. The attenuator should be capable of controlling the output sound pressure level over a desired range in steps of 5 dB [14].

Two pairs of attenuators are provided; one for pure-tone and one for masking noise. A mixer/selector unit diverts the stimulus and masker to the amplifier for presentation to the left and the right ear. The power amplifier should provide the output power required by the maximum sound pressure level from the headphone and the bone vibrator. Thus the power amplifier must have low distortion and a good signal-to-noise ratio to meet the standards. The type of masking noise and pure tone of different frequencies are selected using the control unit. It also controls the attenuation level in accordance with frequency.

2.10 Microcontroller based audiometer

In manual audiometry, the stimulus level is manually controlled by the audiologist after observing the responses of the patient. The patient hearing threshold for a discrete set of frequencies over the range of 125 Hz to 8 kHz is noted. For this purpose, the audiometer has dials or knobs with calibrated scale for frequency selection and for masking noise selection, and for varying the level of the stimuli. An interrupter switch is used for tone switching and the switch needs to be mechanically silent. Use of mechanical switches for selection of frequency, level, and tone interruption makes the instrument susceptible to wear and tear. Hence calibration is necessary at least once in six months.

Analog switches can be used for electronically controlled and noise free switching. Further the application of microcontroller in audiology offers many advantages in terms of flexibility and simplicity of use, over their conventional counterparts [15], [16]. In this case knobs and switches are replaced by a keypad. Calibrated scales and other indicators are replaced by display to show the various parameters and modes, and operation status. Using a microcontroller in this application increases accuracy and precision by removing the need for frequent calibration of audiometer.

Block diagram of a microcontroller based audiometer is shown in Fig. 2.4. All the digital controls are through the digital port outputs from the microcontroller. Here the microcontroller is employed to control the frequency output of the oscillator, which generates the test tones. An important feature of the microcontroller based audiometer is that it can easily carry out automated determination of the threshold. Masking noise for the non-test ear is generated using the noise generator block. At each specified stimulating period, the microcontroller changes the frequency and amplitude if necessary by sending control signals to oscillator and an attenuator respectively. While presenting the test tone of single frequency, the level of the test tone is varied according to the subject response, and then the threshold is determined.

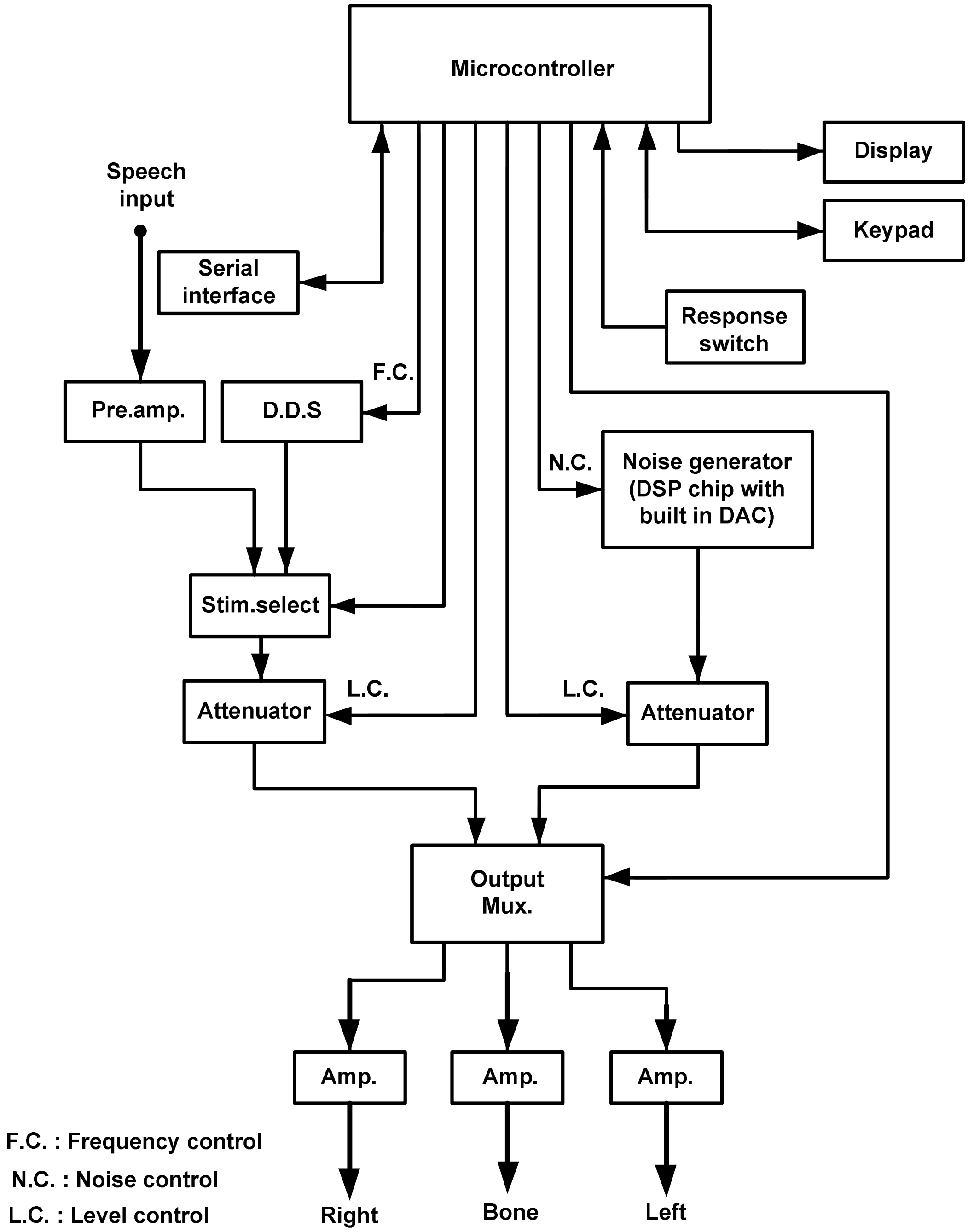

Fig. 2.4 Block diagram of a microcontroller based audiometer.

Chapter 3
DESIGN APPROACH

3.1 Audiometer specifications

The goal of this project is to develop an audiometer that will be well suited for a clinical and research purpose and which can be used in rural areas. It should be portable, without any mechanical parts in order to reduce repair and maintenance servicing. Further it should be easy and convenient to calibrate without opening the unit, under software control through its keypad. It should be capable of generating pure tone and warble tone, in continuous and interrupt modes of different frequencies. It should also have wide band and narrow band masking noises with the frequency of the narrow band automatically tracking the tone frequency. Frequency and level should be easily selectable and with provision for various audiometric tests. It should have keypad and graphical display for user interface and provisions for presenting speech from microphone or tape for speech reception and speech discrimination test. It should provide air and bone conduction test facilities with output levels of 0 dB HL to a maximum value as given in Table 3.1, in steps of 5 dB as per ANSI standards 1972.

The audiometer should have two types of masking noises, wide band/white noise and narrow band noise with sufficient level to mask the effect of cross hearing. Wide band noise should have a flat spectrum from 125 Hz to 10 kHz. Narrow band noise with centre frequency equal to test tone frequency, its band width is 1/3 or 1/2 octaves about centre frequency attenuation rate outside the pass band is 12 dB / octave [1], [8].

Some keys from 4x4 keypad will be dedicated for selecting the desired attenuation level, frequency and for choosing types of audiometric tests and types of masking noises. Graphical display, 128x64 LCD, will be used for displaying the stimulus (type /frequency and level), noise type level and for plotting the audiogram. Section 3.2, summarizes the technical specifications of the audiometer.

Table 3.1 Frequency and its maximum HL in dB as per ANSI standards 1972.

Frequency (Hz)	250	500	1000	1500	2000	3000	4000	6000	8000
Air cond. (dBHL)	90	100	100	100	100	100	100	90	80
Bone cond. (dBHL)	40	50	50	50	50	50	50		

3.2 Technical descriptions

The audiometer should have the following characteristics:

1. Channels: Two channel audiometer.
2. Operation modes: Manual and automated.
3. Test modes
    - Pure-tone threshold testing in manual mode (continuous and interrupted tones).
    - Speech: Live voice, Tape or CD input.
    - Pure-tone threshold automated testing (2/2, 2/3, 3/5, 4/6, 4/7…selectable response criteria).
    - Tone decay test.
    - SISI test.
4. Outputs: Right phone, Left phone, Bone.
5. Stimuli: Pure tone, speech and warble tones.
6. Masker: White noise, Narrowband noise, Speech-shaped noise.
7. Tone: Warble tone, ± 5 % frequency deviation around the centre frequency with one sweep per second.
8. Wideband noise: flat frequency spectrum in all aduiometric testing frquency.
    - Narrow band noise: 1/3 octave, center frequency equal to the tone frequency, roll-off 12 dB per octave.
9. Output level: Air-conduction -10 to 160 dB (SPL). Bone-conduction -10 to 90 dB (SPL).
    - Calibration: Internal calibration tables for mapping dB SPL to dB HL for specific headphone and bone vibrator.
10. Attenuator: 5 dB step.
11. Signal output: Tone on (manual test) or automatic (automated test).
12. Patient's response: one response switch and light indicator on panel.
13. Communication with patient: microphone.
14. Data communication: Built-in RS232 serial interface for in-system programming and bidirectional data transfer.
15. Accessories:
    - Test headset (TDH-39 earphones).
    - Bone vibrator B72.
    - Microphone.
    - Patient's switch.
16. Display: 128x64 monochrome LCD display.
17. Control: 4x4 keypad.

3.3 Design approach

In the block diagram shown in Fig. 2.4, the microcontroller is the central control unit of the audiometer. For a very compact design, the microcontroller should have sufficient programmable ROM, and data RAM, parallel I/O ports, UART, and a programmable timer/counter for handling all the operations without requiring additional chips. For this application it was decided to use P89V51RD2 microcontroller [4] from Philips 8-bit microcontroller family. It has the following features; up to 40MHz speed, 5 Volt supply voltage, 8051-based microcontroller with 32 I/O lines, 3 timers/counters, 9 interrupts/4 priority levels, 64K+8K flash, 1K on-chip RAM, SPI, dual data pointers and WDT [17].

Oscillator should be capable of generating test tones of different frequencies with a distortion level below 30 dB. For this purpose, it has been decided to use a direct digital synthesizer (DDS), a programmable waveform generator IC with very stable amplitude and frequency. The output frequency of the DDS can be varied by sending control signal from the microcontroller. The clock frequency of the DDS can be taken from the crystal clock of the microcontroller. For this application AD9833/AD9834 DDS IC is chosen to generate pure tone of different frequencies [18], [19]. A look-up table is used for sending control word and data to generate each frequency using the DDS.

The hearing threshold of the test tone is determined by systematically varying the level to find out the minimum level at which the subject fails to hear it. Therefore, the attenuator must be capable of adjusting output sound pressure level from -10 dB up to 120 dB in steps of 5 dB [1]. For this purpose, both linear and log attenuators were studied and tested. Quad 12-bit voltage output DAC, MAX536, from Maxim has been tested [20]. Since this is a linear attenuator, a look-up table was used to get 1 dB, 5 dB and 10 dB attenuation. Rounding errors occur in converting dB values to linear binary values and there error varies with the attenuation. For the attenuation range 0-80 dB with attenuation step of 0.5 dB, 19-bit DAC is required. Thus the two DACs of MAX536 was cascaded to have fine and coarse attenuation. The second method to have attenuation is using a log attenuator. Two log attenuators have been studied and tested; TDA8551and PGA2310. TDA8551 is an audio power amplifier capable of delivering 1 W output power to an 8 Ω load at THD = 10 % using a 5 V power supply [21]. The circuit contains a bridge-tied-load (BTL) power amplifier, a digital volume control and standby/mute logic. The gain of the amplifier can be set by the digital volume control. Using the MODE pin the device can be switched to the standby condition, the mute condition and the normal operating condition. TDA8551 has 80 dB dynamic range. Attenuation step of 1.25 dB is achieved by providing a pulse on the UP/DOWN pin. By cascading two TDA8551 ICs, the dB range requirement of the audiometer can be achieved.

The PGA2310 is a stereo audio volume control [22]. It is a log attenuator from Texas Instruments. The heart of the PGA2310 is a resistor network, an analog switch array, and a high performance bipolar op-amp stage. The switches are used to select taps in the resistor network which, in turn, determines the gain of the amplifier stage. Switch selections are programmed using a serial control port. The ability to operate from ± 15V analog power supplies enable it to process input signals with large voltage swings. PGA2310 uses high performance internal operational amplifier stages, for providing low-noise and distortion drive for 600 Ω loads. It has attenuation range from -95.5 dB to +31.5 dB; which covers 120 dB dynamic ranges. This log attenuator has two independent channels and gain for each channel is set using SPI serial interfacing mode.

In early audiometers, a low frequency saw tooth waveform was used as a masker, which was effective at low frequencies, but not at high frequencies. A diode can be used as a noise source, but it requires high gain amplifier, which may result in 50 Hz power line pick up [13]. These techniques have a possibility of generating unequal amplitude of masking noise. However, a digital noise generator generates noise of known spectrum and amplitude [13]. Hence, it was decided to use digital white noise generator based on the pseudo random binary sequence (PRBS) generator. After low pass filtering with cut off frequency equal to 10 kHz, it can be used as masking noise with flat power spectrum over the entire test frequency range. This noise can be band-pass filtered to get narrow band noise. Narrow band noise gives the same masking effect as wide band noise but at lower sound pressure level.

Masking noise is generated using DSP chip with built-in DAC inside. A software program is written for generating wide band noise and narrow band noise centred at tone frequency. A shift register with 16-bit width is used to generate pseudo random binary sequence. Shifting, xor-ing and loading is done inside the code. All this is done at a sampling frequency of 36 kHz. Coefficients of the narrow band noise are pre-calculated and stored in the code in a look-up table. This table has values for all frequencies. The code keeps track of the tone frequency and accordingly uses the corresponding coefficients from the look-up table to filter and generate narrow-band noise centred at the set tone frequency.

The output selection is done using the block called output select. It consists of analog switches controlled by the microcontroller. For switching between the stimuli and type of masking noises, a triple SPDT analog multiplexer, 4053, was chosen [9].

The audiometer has a response switch for getting the response to the stimuli. Depending on the response, the audiologist decides the next tone level. In automated mode, the instrument itself presents the stimuli and on the basis of the response it decides the level of next stimuli; by decreasing in steps of 10 dB if there is a response and increasing in steps of 5 dB if no response and checking if there is a 2 out of 2 or 2 out of 3 or 3 out of 5 responses to determine the hearing threshold at every tone level for each frequency. Hearing thresholds for all the standard test frequencies are stored and drawn to find out the corresponding hearing loss [8].

The user interface is provided using a 4x4 keypad and 128x64 graphical display. The 4x4 keypad is interfaced directly to Port 0 of the microcontroller. All the information on selection is provided on 128x64 graphical displays, LGM12641BS1R, which is a commercially available unit with on-board controller. It requires 8 data lines and 6 control lines, for interfacing to the microcontroller.

The microcontroller, P89V51RD2, controls all the operations. The test tone is generated using DDS, AD9833/AD9834, which needs SDATA, SCLK, FSYC and MCLK input signals for serial interfacing. The tone frequency of DDS can be changed, by enabling the DDS followed by sending data via SDATA pin. The data in the SDATA will be considered as data only if SCLK is held at logic low. The output of the attenuator is systematically attenuated to get a correct hearing threshold. Resistive gain of 20 dB and PGA2310 are cascaded to get dynamic attenuation range from 0 to140 dB with attenuation step of 5 dB. The output of the attenuator is fed to the power amplifier to meet the power requirement in the output and then directed to headphone or bone vibrator. Separate attenuator and power amplifier are provided to the masking noise. After selecting the type of noise it will be fed to the attenuator to have the required attenuation under the control of the microcontroller and then amplified using power amplifier; and finally it will be presented to the headphone corresponding to the non-test ear.

Chapter 4
AUDIOMETER HARDWARE

4.1 Introduction

An audiometer generally consists of a control unit, tone generator/oscillator, attenuator, masking noise generator, power amplifier, headphone, response switch, bone vibrator, internal and external Talk-Over Mic, Talk-Back Mic, internal and external monitor loudspeaker. Pure and warble tone is generated using tone generator/oscillator. The frequency of stimuli ranges from 125 Hz to 8 kHz. Logarithmic attenuation of the stimuli is achieved using the attenuator. The frequency and level of the stimuli are controlled by the control unit by sending control signals to the tone generator/oscillator and attenuator respectively. To cancel the effect of cross hearing, masking noise of wide band or narrow band (with center frequency made equal to presented tone) is applied to the headphone corresponding to the non-test ear. Finally power amplifier is used to meet the requirement of power in the output to derive the headphone or bone vibrator.

4.2 Oscillator

An oscillator is needed to generate test tone of different frequencies, with stable amplitude. There are many ways of generating tones of different frequency, one way could be using software means and putting DAC at the output, but the microcontroller will be busy all the time in generating the stimuli and high quality DAC is needed for generating tones with very low spectral distortion. Other methods of generating stimuli include switched-capacitor (SC) filter based oscillator and programmable waveform generator. The oscillator part of all earlier audiometers designed at IIT Bombay used SC filter ICs [9], [10]. A switched capacitor filter based oscillator requires only clock frequency as control input [13]. Thus the tone frequency can be easily changed by changing the clock frequency. These oscillators provide outputs with excellently high frequency stability and acceptable spectral purity and moderate amplitude stability. Sinusoidal output can also be generated using direct digital synthesizer ICs. Using direct digital synthesizer results in excellent frequency and amplitude stability and easy control of the frequency [3]. In this project a direct digital synthesizer, AD9833/AD9834, is used to generate pure tones with stable amplitude [17], [18].

The DDS is capable of generating sine, triangle and rectangle waves. The internal circuitry of the AD9833/34 consists of the following main sections: a numerically controlled oscillator (NCO), frequency and phase modulators (consists two 28-bit frequency registers and two 12-bit phase registers), SIN ROM, a digital to analog converter and a regulator. The theory behind generation of sine wave using the DDS can be described shortly as follows;

generation of sine waves can be carried out through piecewise construction, due to the fact that, magnitude of the sine wave is $Y(t) = \sin(wt)$. On the other hand, the angular information is linear in nature, that is, the phase angle rotates through a fixed angle for each unit of time. The angular rate depends on the frequency of the signal by the rate of $w = 2\pi f$. For the given clock period (f_{MCLK}), the phase rotation for that period can be determined.

$$\Delta\text{Phase}=2\pi f \Delta t \tag{4.1}$$

Thus the frequency can be controlled by varying ΔPhase.

$$f = \Delta\text{Phase} \times f_{MCLK} / 2\pi \tag{4.2}$$

The DDS chip builds the output based on the above simple equation. Continuous time signals have a phase range of 0 to 2π. Outside this range of numbers, the sinusoid functions repeat themselves in a periodic manner. The accumulator inside the DDS simply scales the range of phase numbers into a 28-bit digital word. The corresponding phase range is 0 to 2^{28} (i.e. $2\pi = 2^{28}$). Likewise, the ΔPhase term is scaled into this range of numbers:

$$0 < \Delta\text{Phase} < 2^{28} - 1 \tag{4.3}$$

The input to the phase accumulator is loaded from the frequency registers. As it generates continuous phase signals, it avoids any output discontinuity when switching between frequencies. Phase offset can be added to perform phase modulation using the 12-bit phase registers. The count in the phase accumulator is used as an address to a look-up table “SIN ROM” for converting the phase information into amplitude. The amplitude data is given to the DDS’s DAC [18], [24]. To smooth out the DAC output capacitor should be connected at its output. The frequency and the phase of the output can be controlled using synchronous serial interface. The output frequency and the phase are given as,

$$f_{out} = \frac{f_{MCLK}}{2^{28}} N_{FREQREG} \tag{4.4}$$

$$\theta_{out} = \frac{2\pi}{2^{12}} N_{PHASEREG} \tag{4.5}$$

where N_{FREQREG} and N_{PHASEREG} are the counts loaded in the frequency and the phase registers of the DDS. Fig. 4.1 shows the circuit connection of P89V51RD2 with AD9833.

Microcontroller sends data serially from a look-up table of the standard tone frequency through P3.5-SDATA pair. AD9833 was tested to generate a pure tone of single frequency both for sine and triangle waves. As shown in Fig. 4.1, the DDS IC2 is controlled and programmed by the microcontroller IC1 (P89V51RD2) through synchronous serial interface. The port pins P3.5, P3.6, and P3.7 are connected to the SDATA, the SCLK, and the FSYNC of the DDS respectively. The master clock, MCLK, of the DDS is connected to XTL1 pin 18 of the microcontroller.

AD9833 has single voltage output but AD9834 has two current outputs and it is capable of generating either sine or triangle wave in one output and rectangle wave in the other output at the same time. A resistance can be used to change the current output to voltage output. Circuit connection of AD9834 with P89V51RD2 is shown in Fig. 4.2. Microcontroller, IC1, receives the stimulating parameters from up and down frequency keys from keypad, and then it sends controlling signals to DDS, IC2 which changes the frequency. Table 4.1 shows the data and control word of DDS for both frequency registers and phase registers to generate the pure tone of different frequencies.

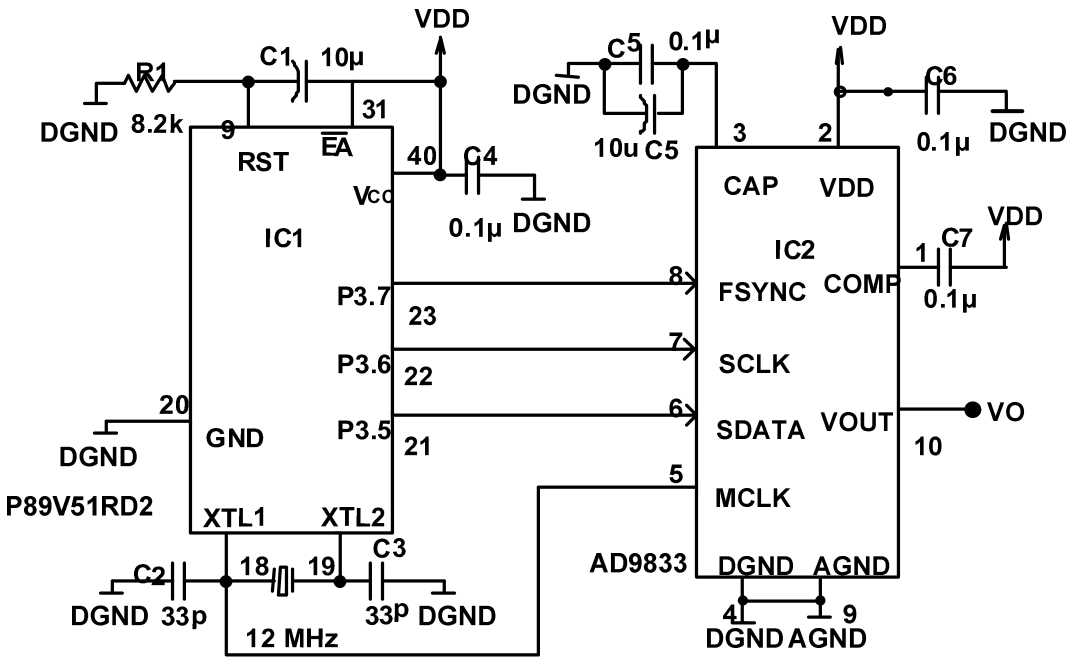

Fig. 4.1 Pure tone generator using wave form programmable IC, AD9833.

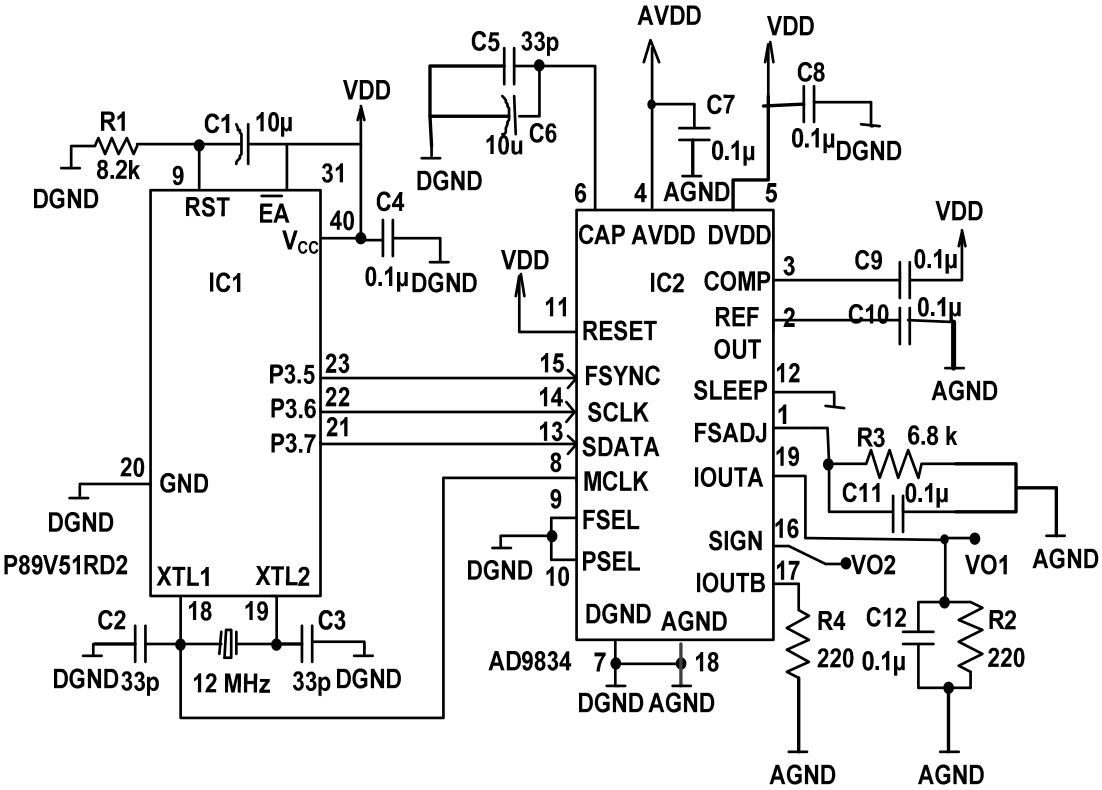

Fig. 4.2 Pure tone generator using wave form programmable IC, AD9834.

Table 4.1 Data loaded to frequency register of DDS for different frequencies.

Freq. Hz	Control word	Frequency Register 0 Data		Frequency Register 1 Data		Phase Register 0	Phase Register 1
		14 bits LSB	14 bits MSB	14 bits LSB	14 bits MSB		
125	0x2008	0x4AEC	0x4000	0x8000	0x8000	0xC000	0xE000
250	0x2008	0x55DB	0x4000	0x8000	0x8000	0xC000	0xE000
500	0x2008	0x6BB1	0x4000	0x8000	0x8000	0xC000	0xE000
750	0x2008	0x4AEC	0x4000	0x8000	0x8000	0xC000	0xE000
1000	0x2008	0x5762	0x4001	0x8000	0x8000	0xC000	0xE000
1500	0x2008	0x4312	0x4002	0x8000	0x8000	0xC000	0xE000
2000	0x2008	0x6EC3	0x4002	0x8000	0x8000	0xC000	0xE000
3000	0x2008	0x4625	0x4004	0x8000	0x8000	0xC000	0xE000
4000	0x2008	0x5D86	0x4005	0x8000	0x8000	0xC000	0xE000
6000	0x2008	0x4C4A	0x4008	0x8000	0x8000	0xC000	0xE000
80000	0x2008	0x7B0D	0x400A	0x8000	0x8000	0xC000	0xE000

4.3 Warble tone generation

Some patients may have a severe loss in a narrow frequency region. Since hearing loss at single frequency doesn't have significant implication for speech perception, in such cases the hearing needs to be tested at surrounding frequencies using a warble tone. Warble tone is also generated using the DDS. A ±5 % frequency modulation is chosen with one sweep per second. To avoid the possibility of discontinuity while switching from one frequency to other frequency, both 28-bit frequency registers of DDS are used alternatively. If frequency register0 is used to generate one discrete frequency, frequency register1 is used to generate the next higher discrete frequency. In this way warble tone is generated by loading two discrete frequencies to both frequency registers of the DDS simultaneously but alternatively choosing data from frequency register0 or data from frequency register1 to be output from the DDS. This is done for all discrete steps of frequencies. A Matlab simulation was used to check the number of steps needed per sweep.

4.4 Attenuator

Attenuator is a circuit used for level control of the tone generated by the oscillator. Sound pressure levels in audiometry are specified in decibel (dB) [2].

$$\mathrm{SPL(dB)} = 20\log(\frac{P_O}{P_R}) \tag{4.6}$$

where P_{O} is the measured RMS pressure and P_{R} is the reference pressure (20 μPa). The electronic amplifiers control their output voltage and hence, it is convenient to have an expression for sound pressure level that is related to the voltage gain of the amplifier. Because the output pressure is proportional to the output voltage, the relative sound pressure level produced by an amplifier operating into constant load impedance can be expressed as

$$\mathrm{SPL(dB)} = 20\log(\frac{V_O}{V_R}) \tag{4.7}$$

Where V_O is the output voltage of the amplifier and V_R is a reference input voltage which produces the reference acoustic pressure level P_R. The value of this voltage will depend upon such factors as the load impedance, losses in transmission, the efficiency of the acoustic transducer, and frequency [2], [8]. In practice, the input voltage to the amplifier is adjusted to produce a desired SPL while measuring the output of the acoustic transducer with a calibrated sound pressure level meter.

4.4.1 Log attenuator

Three attenuator chips were tried for designing the most appropriate attenuation circuits: MAX536 (dual 12-bit linear DAC), Philips TDA8551 (log attenuator with 80 dB dynamic range and bridge tied 1 W output) and Texas Instruments PGA2310 (120 dB log attenuation with 2-channel can drive 600Ω). After an extensive testing, it was decided to use PGA2310, as it provides a large dynamic range and has a serial interface. In this chip, a resistive network, an analog switch array and a high performance bipolar op-amp are used to get log attenuation. It has two independent channels. Each channel has 120 dB dynamic ranges.

On power up, the gain byte value for both the left and right channels are set to mute condition. The gain will remain at this setting until a new setting for each channel is programmed via the serial control port. The serial control port is utilized to program the gain settings for the PGA2310. The Serial control port includes three input pins and one output pin. The inputs include active low CS (pin 2), SDI (pin 3), and SCLK (pin 6). The output pin is SDO (pin 7). Data is written to the PGA2310 when CS is low. Control data is provided as a 16-bit word at the SDI pin, 8-bits each for the left and right channel gain settings. MSB first format is used. Control data is clocked into SDI on the rising edge of SCLK. The gain setting for each channel is set by its corresponding 8-bit code, either R [7:0] (i.e., 8-bit for the right channel) or L [7:0] (i.e., 8-bit for the left channel) as shown in Fig. 4.3.

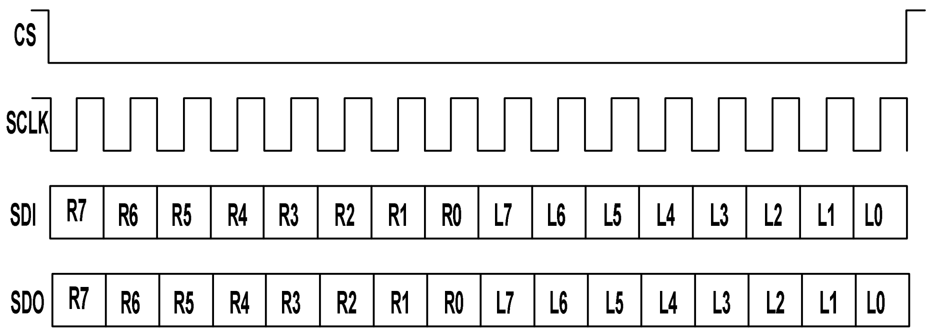

Fig. 4.3 Serial interface protocol of the attenuator, PGA2310.

The gain code data is straight binary format. Let N be the decimal equivalent of R [7:0] or L [7:0], then the following relationships exist for the gain settings: For N=1 to 255;

$$\mathrm{Gain(dB)} = 31.5 - \{0.5 \times (255 - N)\} \tag{4.8}$$

N = 0 results mute condition. The above formula, gives the dynamic range from +31.5 dB (with N = 255) to -91.5 dB (with N = 1). Another feature of PGA2310 is its ability to change the gain on the zero crossing if the zero crossing enable pin (ZCEN or pin 1) is enabled. This feature is provided for noise-free level transitions. The zero crossing detection takes effect with a change in gain setting for a corresponding channel. The new gain setting will not be latched until either two zero crossings are detected, or a timeout period of 16 ms has elapsed without detecting two zero crossings. In the case of a timeout, the new gain setting takes effect with no attempt to minimize level transition. There are two ways of putting the

IC in mute condition. Making the mute pin (pin 8) active low results in each channel muted following the next zero crossing event or timeout. The second method is by setting the gain byte value for one or both the channels to 0x00. This will immediately mute the corresponding channel. Fig. 4.4 shows the interfacing of the log attenuator, PGA2310 with the microcontroller.

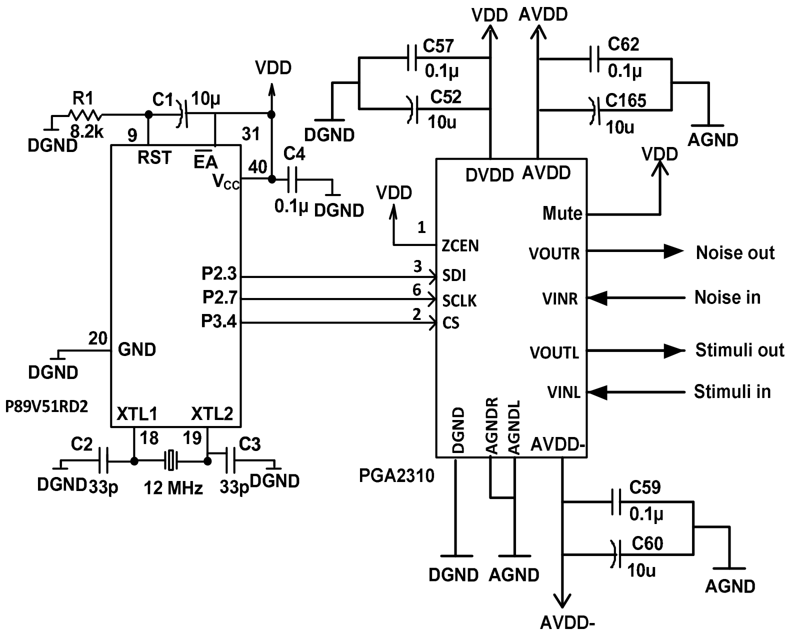

Fig. 4.4 Interfacing of the log attenuator, PGA2310, with the microcontroller.

4.5 Masking noise generation

A band-limited white Gaussian noise is used as a masking noise, which is generated by low pass filtering the digital output of a pseudo-random binary sequence (PRBS) [13]. Further the Gaussian noise is passed through a narrow band pass filter to get narrow band noise. In earlier designs [25]-[27], a 15-bit PRBS generator based on the maximal length feedback shift register, was used to generate wideband noise, with a clock input of 200 kHz and it gives a repetition rate of, $(200000/(2^{15}-1))$ which is approximately 6 Hz. This noise had a perceivable repeatability, which is not desirable. This problem is addressed by using 23-bit shift register [9], [10]. Using this long shift register the output of the PRBS repeats after every 42 s.

It was decided to use a DSP microcontroller to synthesize and filter the masking noise, and output using its internal DAC. For this purpose, Microchip dsPIC33FJ128GP804 [21] was selected. The code is written in C and it has two main parts. The first part is generating pseudo random sequence using 16-bit shift register and outputting to the DAC at the rate of 36 kHz. The second part is a bandpass filter code with each of the test frequencies (125, 250, 500, 750, 1000, 1500, 2000, 3000, 4000, 6000 and 8000 Hz) as the center frequency. The coefficient of the transfer function are pre-calculated and stored in the code. The second part takes an input from the first part and then filters for selected frequency and puts its output to the DAC. DAC output is connected externally to the noise attenuator to set the desired noise level and then fed to the power amplifier. Circuit for noise generation is shown in Fig. 4.5. Characteristic of the wideband noise is shown in Fig. 4.6.

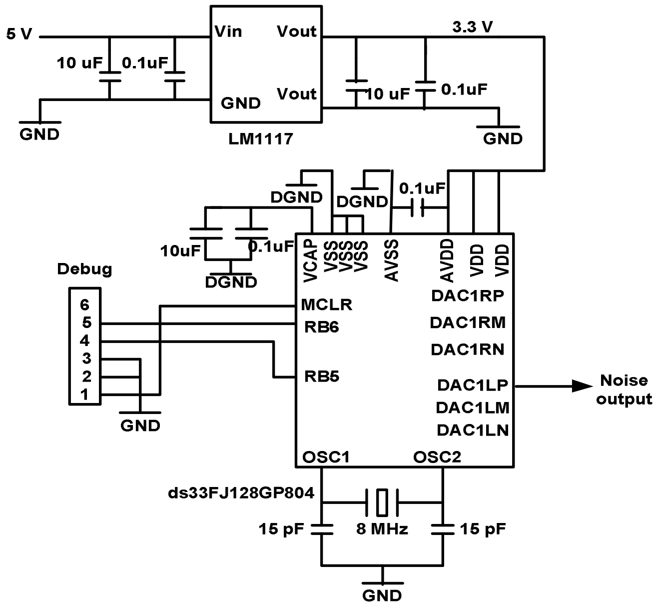

Fig. 4.5 Noise generation using ds33FJ128GP804.

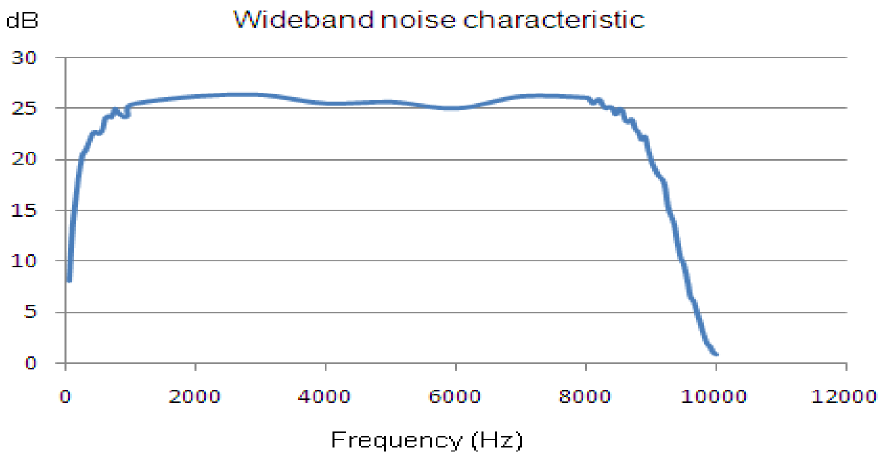

Fig. 4.6 Power spectrum of wideband noise.

4.6 Keypad

A 4x4 matrix keypad is used for user interfacing and it is directly connected to the microcontroller through its Port 1 as shown in Fig. 4.7. The microcontroller scans the keypad, using row-column matrix scanning method. Pressing a key physically involves two pins permanently connected to the port; thus it is not possible to multiplex this port for other operations.

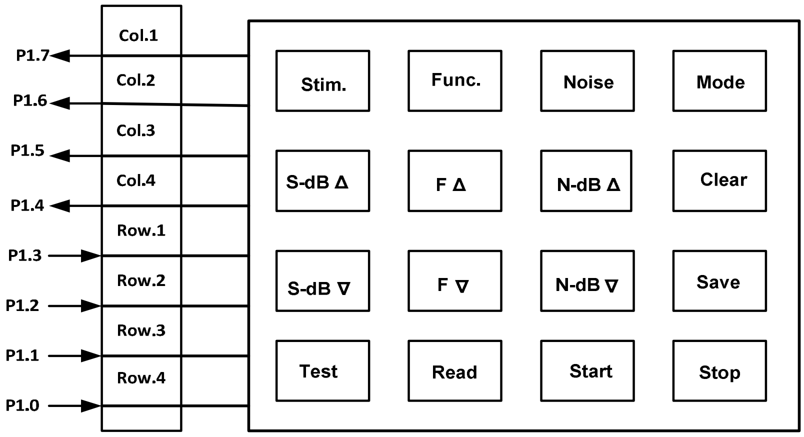

Fig. 4.7 Keypad layout.

4.7 Display

The display unit uses a 128x64 graphical display, LGM12641BS1R. It has an on-board controller, KS0108/7, that works on a single 5 V supply. A graphical display is selected to print alphanumeric display as well as plotting of the audiogram. The LCD interfacing software is written in C (using the Keil compiler) for displaying the type of test tone with its frequency in Hz and its level in dB, for displaying the masking noise type and its level in dB, for displaying the conduction type (air or bone conduction) and for displaying test tone duration (continuous or interrupted).

Considering the 128x64 pixels as a plane as shown in Fig. 4.8. The extreme left of the x-axis is taken as (0, 0) position. The built-in display controller, KS0108/7, performs all of the refreshing and data storage tasks of the LCD display. The display is split logically in half. It contains two controllers with chip select 1, controlling the left half of the display and chip select 2, controlling the right half of the display. Each controller must be addressed independently. The page addresses, 0-7, specify one of the 8 horizontal pages which are 8 bits high. The interfacing of the 128x64 with the microcontroller is shown in Fig. 4. 9. The interfacing involves 13 I/O pins of the microcontroller of which 8 I/O pins are used for data bus and 5 I/O pins are control pins. While connecting the display to the microcontroller's data bus and mapped into its memory area, the status should be tested to guarantee reliable operation.The I/0 pins P2.3, P2.4, P2.7, P2.6 and P2.5 are connected to the RS, E, CS1, CS2, RST pins of the graphical display respectively. RS is control register when it is held at logic high, D0-D7 is considered as data input but if RS is at logic low D0-D7 is treated as command input. RST is active low reset input and E is enabling input pin. The software flow chart for sending command and data to the display are shown in Fig. 4.10.

4.8 Multiplexer (output selector)

Digitally controlled analog switches are used to select the test tone or noise to either right ear or left or to both. For this 4053 analog multiplexer, triple single-pole double-on analog multiplexer, is selected. Three analog multiplex ICs are used, two for the two headphone outputs and one for the bone conduction output. The digital control for the muliplexer comes from the I/O port of the microcontroller which is Port 0. A latch IC, 74HC373, is used to multiplex Port 0 to the display and to the analog switches. Interfacing of the multiplexer to the microcontroller is shown in Fig. 4.11.

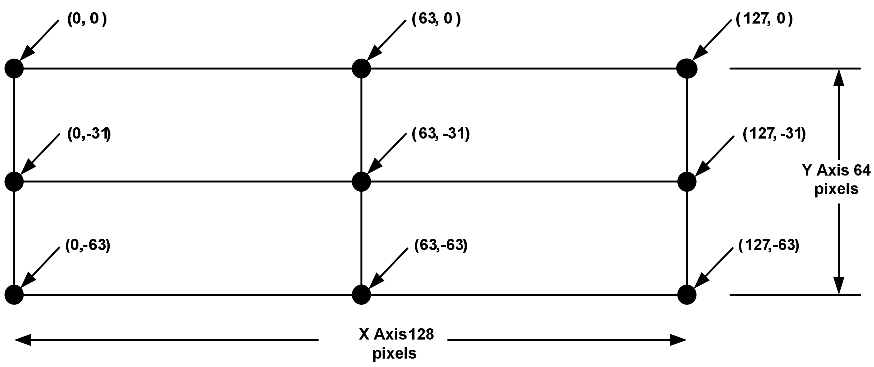

Fig. 4.8 Mapping of LCD, 128x64 screen to the cartesian plane.

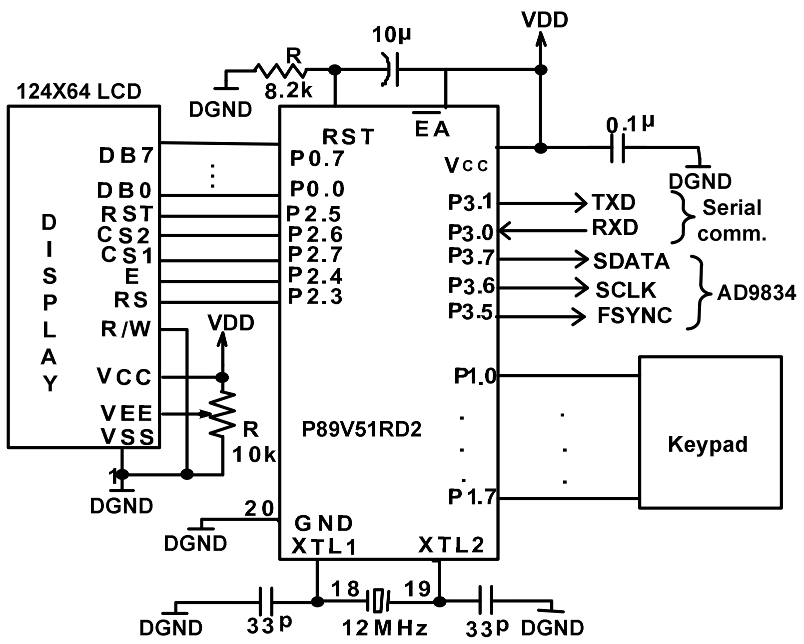

Fig. 4.9 Interfacing of 128x64 with P89V51RD2.

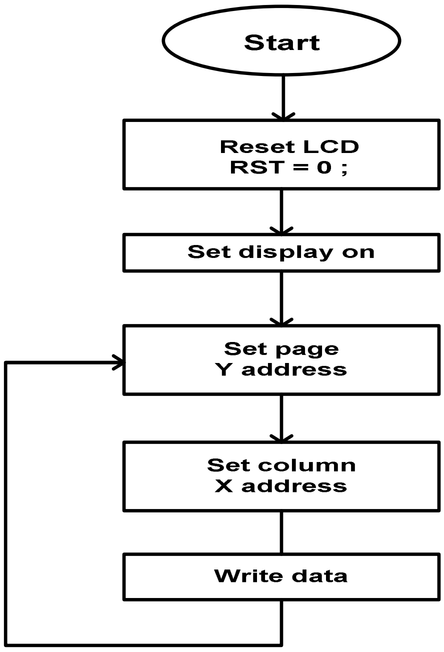

Fig. 4.10 Flow chart for initializing and writing data to LCD.

While data is being written to the display the latch is disabled. Microcontroller sends data to the multiplexer by disabling the display followed by writing the data to its port and then enabling the latch to latch it to its output. After some delay it disables the latch and enables the display. The function of the digital control pins of the three muliplexer ICs is shown in Table 4.2.

Table 4.2 Digital control pins of multiplexer and their functions.

Port pin	Latch		Function to be assigned
	I/P	O/P	
P0.0	D0	Q0	Bone stimuli
P0.1	D1	Q1	Noise to left
P0.2	D2	Q2	Stimuli to Left with 20dB more
P0.3	D3	Q3	Stimuli to Right
P0.4	D4	Q4	Stimuli to Right with 20dB more
P0.5	D5	Q5	Noise to Right
P0.6	D6	Q6	Stimuli to Left
P0.7	D7	Q7	Noise to Bone

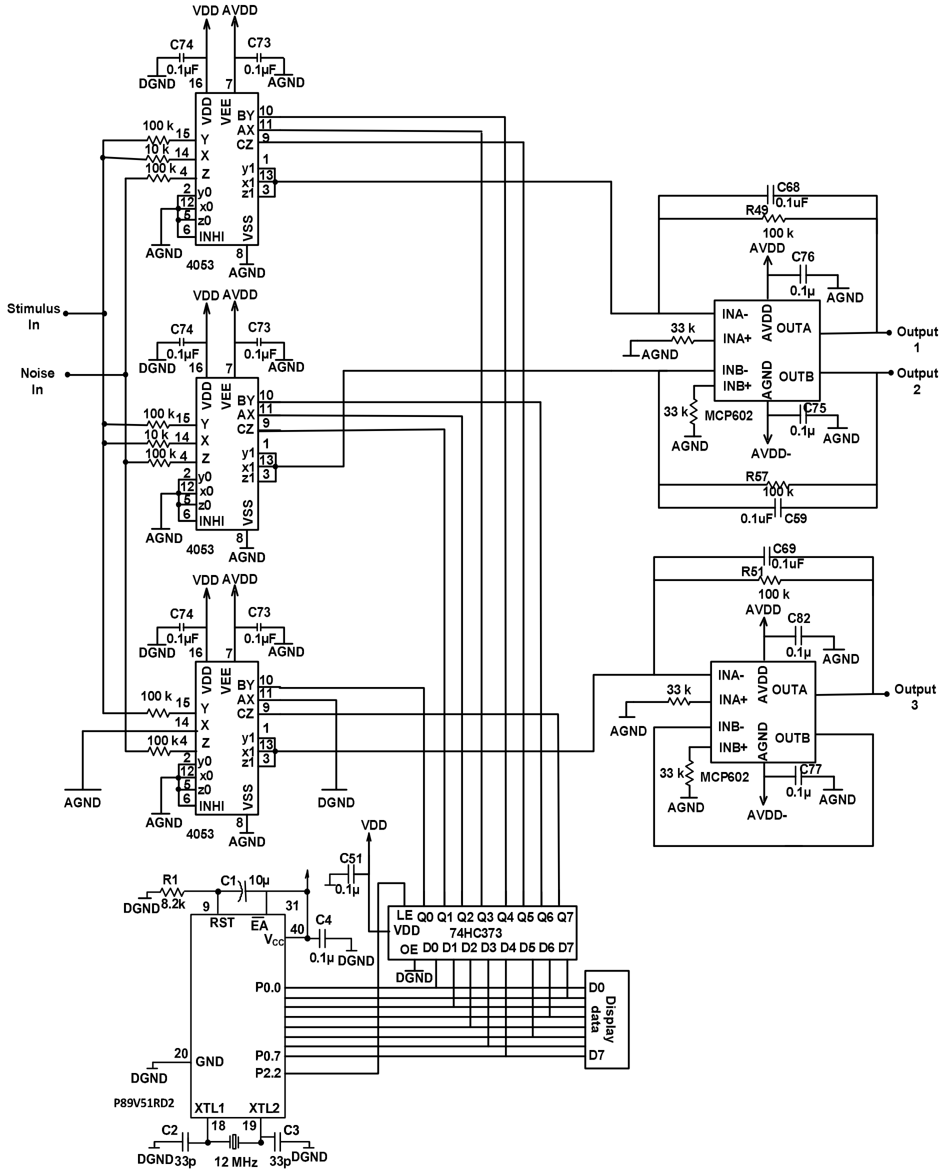

Fig. 4.11 Circuit for muliplexing.

4.9 A serial interface

A serial interface facility is provided for transferring data from the instrument to computer or printer. Thus the audiometer can be interfaced with a serial device through the serial interface. The microcontroller has two pins assigned specifically for transferring and receiving data serially. These two pins are called TxD and RxD and are part of Port 3 group (P3.0 and P3.1). Since the RS232 is not compatible with microcontroller, it needs a line driver to convert the RS232 signals to TTL voltage levels that will be acceptable to microcontroller's TxD and RxD pins. For this MAX232 is used to convert from RS232 voltage levels to TTL voltage levels, and vice versa. One advantage of using this driver IC is it needs only single power supply. Circuit connection of MAX232 with microcontroller is shown in Fig. 4.12.

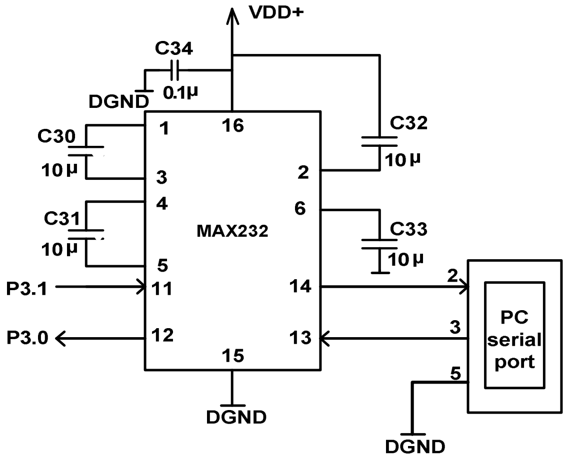

Fig. 4.12 Circuit for serial interfacing.

4.10 Response switch

Searching for hearing threshold in pure tone audiometer requires patient's response when the tone is heard. Before the audiometry, subject should be instructed to press the response switch every time he /she hears the sound emitted, even if it is very faint and then the subject should be correctly fitted with headphones to present the test tone. Here the response switch is connected to the external interrupt pin of the microcontroller (P3.2). Every time the start key is pressed (i.e. pure tone or warble tone or speech is presented) this external interrupt pin is activated and the microcontroller responds immediately when the response switch is pressed. The external interrupt pin of the microcontroller is protected against over voltage and under voltage using diodes as shown in Fig. 4.13.

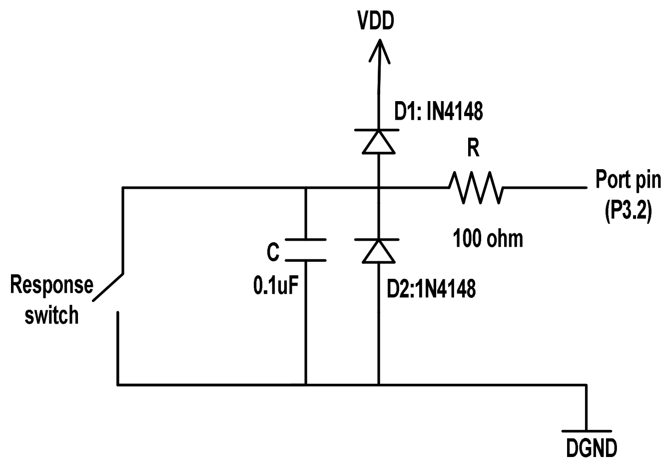

Fig. 4.13 Response switch interfacing to external interruppt I/O pin.

4.11 Power amplifier

Power amplifiers are used to drive the headphone and the bone vibrator. Separate power amplifiers are used for the right headphone, left headphone and bone vibrator. In this project a monolithic dual power amplifier, LM1877 is used. It can deliver 2 W per channel continuously into 8 ohm load. It has low cross-over distortion and AC short circuit protection. Channel separation refered to output is -65 dB [28]. Only one channel of the LM1877 is used to avoid the possibility of degradation of channel separation. The second channel is unused with the input grounded.

Headphone model TDH-39 was calibrated by using artificial ear type 4153 from B&K [9], [10]. The sensitivities of headphone and bone vibrator are different. Usually the hearing thresholds measured as electrical driving voltage obtained using bone vibrator are 45 to 50 dB more than those obtained using headphones. The range selected for bone conduction is 40 to 45 dB less than that of air conduction. Therefore, maximum output voltage required for bone vibrator is approximately same as that of the headphone. Power amplifier circuit for left headphone, right headphone and bone vibrator are shown in Fig. 4.14, Fig. 4.15 and Fig. 4.16 respectively. The circuit is operated on dual ± 5 V supply. LM 1877 is operated in inverting unity gain amplifier mode.

The parallel resistive network formed by R1, R3, R6 and R2 is used to match the output impedance to that of headphone or bone vibrator, which is 8 ohm. Four resistors are connected in parallel to meet the wattage requirement. The gain of the output amplifier is set such that the calibrated driving voltage from the output amplifier is the voltage obtained with the load connected at the output. All the three amplifiers have unity gain.

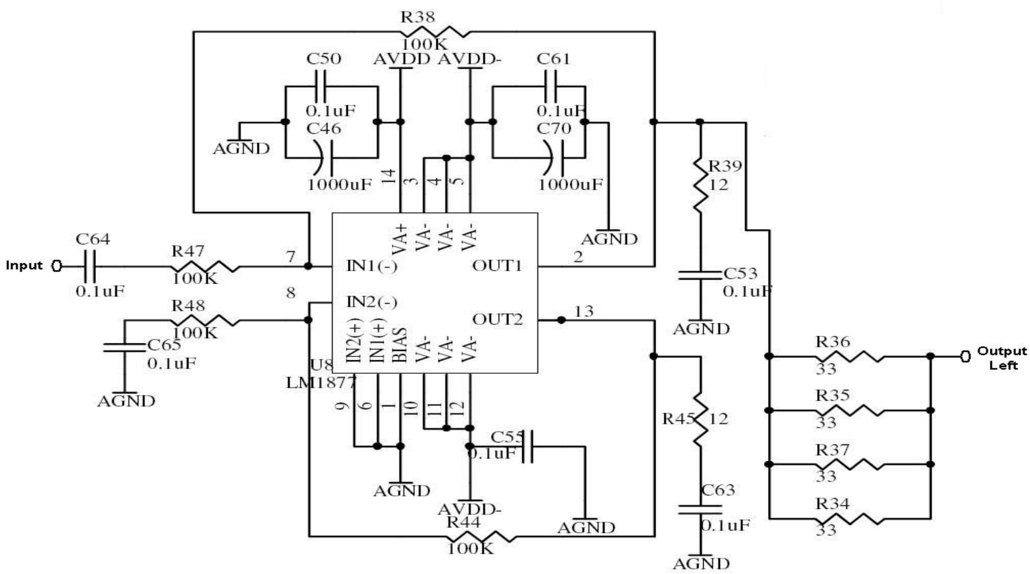

Fig. 4.14 Power amplifier for the left headphone.

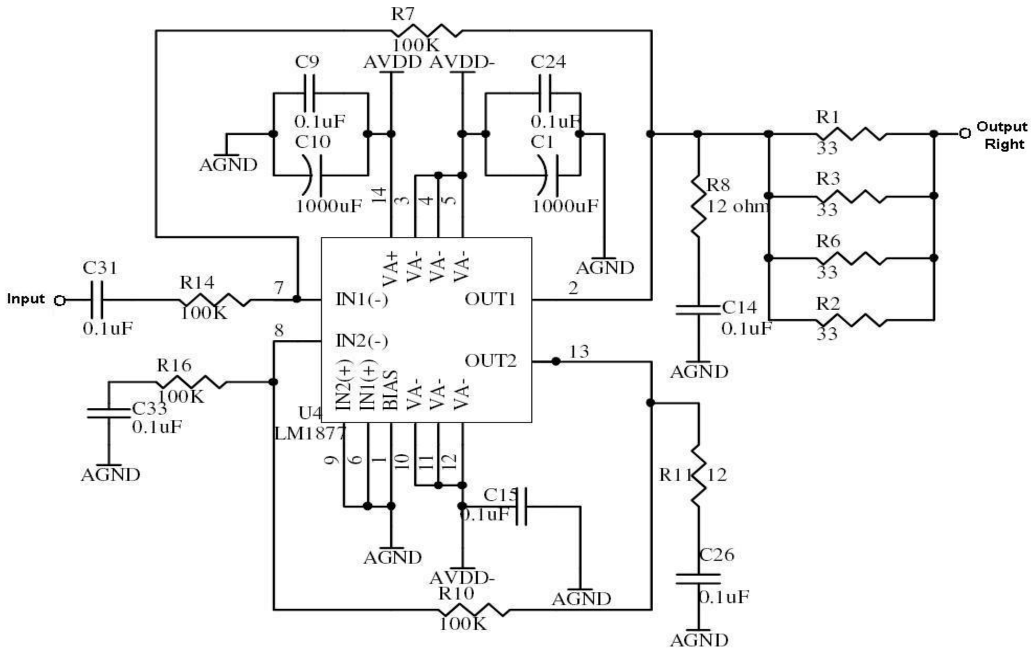

Fig. 4.15 Power amplifier for the right headphone.

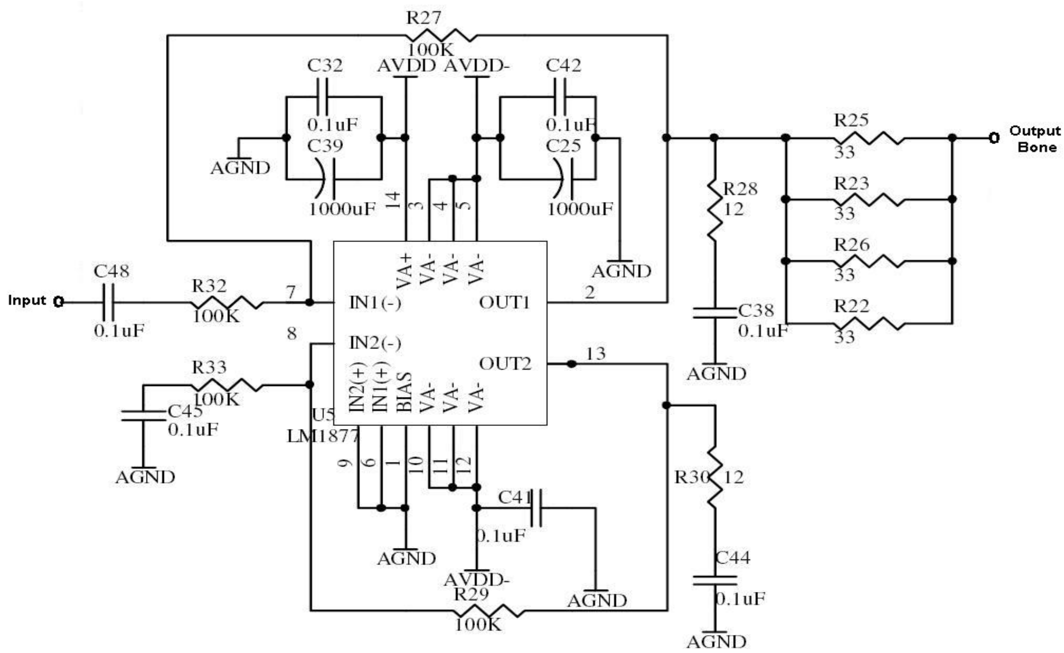

Fig. 4.16 Power amplifier for the bone vibrator.

4.12 Power supply

The circuit is powered by ± 8 V dc. Two voltage regulators, LM7805 and LM7905 are used to supply the analog sections of the hardware with regulated +5 V and -5 V as shown in Fig. 4.17. The digital section is powered by regulated +5V from a separate voltage regulator LM7805.

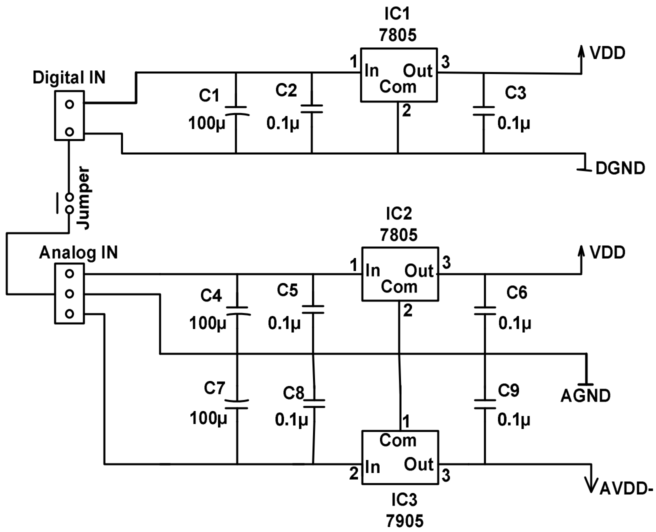

Fig. 4.17 Power supply.

4.13 Microcontroller interfacing

Connection of various port pins of the microcontroller is given in Table 4.3. Some port pins of the microcontroller are multiplexed in order to save I/O pins. All I/O port pins of Port 0 are multiplexed to display data pins and to the digital control of the three analog switches. This is achieved with the help of 8-bit latch IC, 74HC373.

Test tone of different frequency is generated using DDS IC; AD9834. Three I/O pins are required for serial interfacing of DDS to the microcontroller. The serial interfacing of the DDS with the microcontroller is done using P3.5-FSYNC, P3.6-SCLK and P3.7-SDATA pairs. The master clock, MCLK, of the DDS is directly connected from the XTL1, pin 18, of the microcontroller. Interfacing of the graphical display, 128x64, to the microcontroller requires 8 data I/O pins plus 5 control pins. Port 0 is used for the LCD data I/O and P2.3 to P2.7 are used as control pins. It requires 13 I/O pins for proper operation but Port 0, the data I/O pins, is multiplexed for other purposes.

A 4x4 keypad is directly connected to Port1. Since microcontroller continuously scans the keypad, these I/O pins are not multiplexed. When tone is ON an external interrupt pin P3.3 is enable and is used to stop the tone immediately when a “STOP” key is pressed. When tone is OFF this external interrupt is disabled. For this P1.0 is permanently connected to this external interrupt pin as well as to the fourth row of the keypad.

Interfacing of the test tone attenuator requires 3 I/O pins. These are SDI (serial data in), SCLK (serial clock), and E (chip select). P2.3 is multiplexed to SDI (serial data of the attenuator, i.e. pin 6) and to RS (control pin of the LCD, i.e. pin 16). P2.7 is also multiplexed to SCLK (serial clock of the attenuator, i.e. pin 3) and to CS1 (control pin of the LCD, i.e. pin 4). P3.2 is used for response switch. This pin is an external interrupt so it is automatically enabled the moment test tone is ON. While the test tone is presented, if the subject presses the response switch the microcontroller will go to the corresponding interrupt service routine to execute the user interrupt program. When the test tone is OFF this external interrupt pin is disabled all the time.

Two I/O pins (TxD and RxD of the microcontroller) are used for serial communication and in-system programming. Manually resetting the microcontroller is required for in-system programming.

Table 4.3 Function assigned to I/O port pins of the microcontroller, P89V51RD2.

I/O port pins	Function assigned
P0.0 to P0.7	Data bus for display
P1.0 to P1.3	Write to keypad row lines
P1.4 to P1.7	Read from keypad column lines
P2.3	RS control register pin 4 of the display and SCLK pin 6 of PGA2310
P2.6	CS2 chip select 2 pin 16 of the display and SDI pin 3 of PGA2310
P2.4	E, enable pin 6 of the display
P2.5	RST, reset pin 17 of the display
P2.7	CS1, chip select 1 pin 15 of the display
P3.4	Chip select pin 2 of PGA2310
P3.0	$R1_{OUT,}$ receive pin 10 of MAX232
P3.1	$T1_{IN,}$ transmit pin 9 of MAX232
P3.5	FSYNC, chip select pin15 of the AD9834
P3.6	SCLK, serial clock pin 14 of the AD9834
P3.7	SDATA, serial data input, pin 13 of the AD9834
P3.2	Interrupt 0, for response switch
P3.3	Interrupt 1, for stopping the test
XTL1	MCLK pin 8 of AD9834
P2.0	I^2C DATA
P2.1	I^2C CLK
P3.4	Latch enable pin 11 of 74HC373

Chapter 5
AUDIOMETER SOFTWARE

5.1 Introduction

This chapter gives a description of the microcontroller program and its control of the hardware blocks, implementation of the audiometric test and it's over all operation. The microcontroller is programmed to perform the following tasks: scan the keys, update the display, send data serially to the DDS to generate the pure tone, control the level of the test tone, scan the response switch, store the hearing threshold if needed and communicate serially with the computer.

The microcontroller program is written in C. The code is divided into two parts namely the main program and the interrupt service routine (ISR). The main program starts by initializing the I/O port pins, and then continues to initialize the graphical display, the DDS and the attenuator. Subsequently it periodically scans the input keys, and updates the variables and the display accordingly. Before the generation of the pure tone starts, the user can select the desired frequency and level for the pure tone. For this purpose, Timer 0 of the microcontroller is programmed in Mode 1 to generate an interrupt after a delay of 15 ms. After the elapse of this time, the main program goes to the waveform generation mode, with continuous scanning of the input keys and updating display.

The ISR is called every time an external interrupt occur. The external interrupt happens when the response switch is pressed. The user code in the ISR has two parts: one for manual operation and another for automated. Operation in manual mode, when response switch is pressed then external interrupt occurs and the ISR sets a flag to indicate that the sound is heard. The main program accordingly updates the display. But in automated mode the code in ISR is written in such a way that every time an interrupt occurrs it counts how many times a specific pure tone level was heard out of its total presentation. If the specific level is heard one of the following cases: 2 out of 2, 2 out of 3, 3 out of 5 or 4 out of 7… flag is set to indicate the hearing threshold level of the set frequency is found. The main program accordingly updates the display. Each software module is described in the following subsection.

5.2 Tone generation

Generation of pure tones of different frequencies using DDS needs data corresponding to the frequency to be generated and the background clock for clocking the data to the DDS. Both of them are provided by the microcontroller to the DDS. When a particular frequency is selected, that frequency will be displayed in the LCD and the pure tone of same frequency is generated. Though the pure tone is generated from the DDS and is given to the attenuator as an input, the attenuator is in mute condition till the user presses the “Start” key at the keypad. After the “Start” key is pressed, the pure tone generated by the DDS is attenuated by the attenuator. Then the pure tone is fed to the multiplexer. The muliplexer output once more is fed to the power amplifier to get power required to drive the headphone or the bone vibrator.

One software subroutine is dedicated for generation of warble tone using the DDS. Warble tone is generated as a frequency modulated tone with a sweep of $\pm 5\ \%$ of the tone frequency. It was decided to complete one sweep in a second. The number of sweep cycles in a second, and number of steps in each cycle was selected such that the frequency modulated tone is perceived as a warble, i.e. the frequency steps are not very distinct.

5.3 Test modes and algorithm

The audiometer has the following test modes:

1) Manual audiometry for pure tone, warble tone and speech thresholds.
2) Automated audiometry for pure tone thresholds.
3) SISI test.
4) Tone decay test.

In the manual or automated modes, we can select the following

a) STIM: for pure tone threshold audiometry (pure tone, warble tone, and speech).
b) Air / Bone: type of conduction (air, bone).
d) Masking noise: [wide band noise (WB), narrow band noise (NB), no noise (XX)].

All the above modes and various parameters are selected by using the corresponding keys on the keypad. Repeated pressing of a key changes the mode/ parameter setting cyclically. In speech audiometry, only the manual and continuous mode is available. In tone decay and SISI tests, only pure tone or warble tone can be used as stimuli. The test algorithm implemented in the software for the various modes are described in the following subsections.

5.3.1 Manual mode

The default setting for this audiometer is set to manual mode with pure tone as stimuli set at 40 dB level and 1000 Hz. In this manual mode, the operator can turn the tone on or off by using “Start” and “Stop” key respectively. The tone is turned off when tone duration is complete or “Stop” key is pressed. The operator can select frequency or tone level according to the subject’s response. After “Start” key is pressed all keys from the keypad are disabled except the Stop key, but when tone is turned off, the operator has access to all the keys, and can change the settings. After finding the hearing threshold for a particular frequency the operator can save it by pressing the "Save" key. The operator can display the test results through the key pad.

5.3.2 Automated mode

The flowchart representation of the normal audiometric procedure for threshold determination is shown in Fig. 5.1 [14], [29]. Initially a pure tone of 40 dB is presented to the subject. If the response is positive, the tone level is decreased in steps of 10 dB till the patient does not give response. On the other hand, after applying 40 dB tone at the first time, if the patient does not hear it, the level is raised in steps of 10 dB until it is heard for the first time. Once, the response is positive, the tone is decreased by 10 dB. If the patient hears this tone, the tone is again decreased by 5 dB. If the patient does not hear it, the tone is again raised by 5 dB. In this way by several presentations, the hearing threshold is obtained [13], [4]. The minimum presentation level at which the subject responds at least 50 % (2/2, 2/3, 3/5, 4/6, 4/7, 5/8,), is taken as the hearing threshold [29].

5.3.3 SISI test

This test is normally carried out after finding the pure tone hearing threshold using normal pure tone audiometry. In SISI test [1], the operator will select the test frequency and set the level to 20 dB suprathreshold level. The tone is presented with brief bursts of 1 dB modulation above the carrier tone at every 5 s. The 1 dB increments is presented for an interval of 300 ms, out of which the rise time and fall time are 50 ms each, the patient is asked to press the response button whenever he detects a change in the level. Twenty such bursts are presented and out of them, the number of bursts the patient is able to detect is recorded. The number of responses is converted to percentage and stored as the test result. The same procedure is repeated for each frequency, and the result is stored. For this test, masking noise selection is same as in other modes.

5.3.4 Tone decay test

This test can be carried out with or without the tone hearing threshold of the subject. This test should be carried out in automated mode to get accurate result. The operator selects the frequency and the subject is instructed to press the switch as soon as he/she hears the tone and he will once again press the switch if he doesn't hear the tone. The duration between these two responses is measured. The tone is presented and the level is incremented, starting from 30 dB HL, until the subject responds. If the subject is able to hear the tone for more than one-minute [9], the tone level is decremented in steps of 5 dB, and the same procedure is repeated until the tone is audible for less than a minute. But if the tone is not heard then, tone is incremented by 5 dB. The tone is either incremented or decremented without switching off the tone. The lowest level for which tone is audible for at least a minute is detected and stored as tone decay threshold for that particular frequency. The same procedure is repeated for each frequency and the result is stored. The result can be displayed through the keypad. For SISI and tone decay tests, only pure tone stimuli and air conduction test is adopted, but using of masking noise is same as that of other modes.

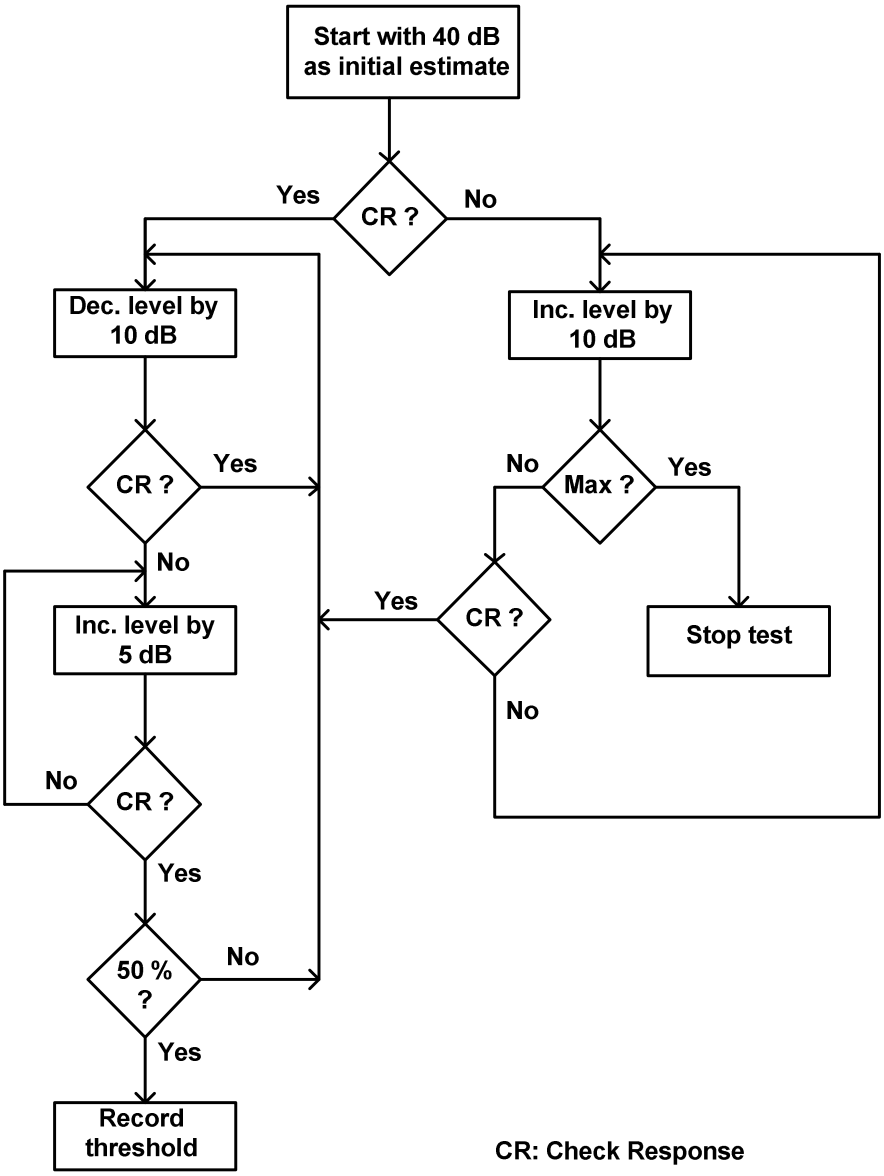

Fig. 5.1 Flow chart for automated test.

5.3.5 Speech audiometry

Since attenuator used for speech test stimuli is same as that of pure tones, switch control is provided for selection of each one of the pure and speech stimuli. Here the operation is same as that of the manual pure tone audiometry. Once speech audiometry mode is selected by the operator, the attenuation of the test stimuli is adjusted through the keypad. Manual volume control facility is provided for keeping proper level of the test stimuli [4].

5.4 System operation

Operation sequence of the audiometer is shown as in Fig. 5.2. After power ON the instrument displays "Audiometer and its version" and "Press any key to continue" as shown in Fig. 5.3. On detecting any key-press, the instrument displays the default settings as shown in Fig. 5.4. On this page, stimuli type, continuous or interrupt mode of operation, type of conduction and noise type can be selected using the keypad. Stimulus can be either pure, warble, or speech. The tone presentation mode can be continuous or interrupted. The mode of operation can be manual, auto, tone decay test, or SISI test. The output device can be either headphone (air conduction) or bone vibrator (bone conduction). There are three options for masking: wideband, narrowband and no noise. Once the initialization is over, the parameters for stimulus (e.g. frequency, tone level, and noise level) are set. The frequency range for air conduction test is from 125 Hz to 8 kHz. For bone conduction test, it is from 250 Hz to 4 kHz. Tone level for air conduction is from 0 to 100 dB HL and for bone conduction, it is 0 to 50 dB HL. The noise level ranges from 0 to 60 dB HL. Once the parameters have been fixed, the stimulus is presented. While presenting the tone, the instrument will show "##" in the display, at the bottom right of the display. After the tone on duration is over or tone is turned off, the instrument displays "XX" instead of "##". The operator can turn off the tone by pressing the "Stop" key. After presentation of the tone is over, the instrument will wait for approximately 1 s to receive the response from the patient. Pressing the response switch before the tone on duration is over or while the instrument is waiting for the response, will be considered as correct response and the instrument shows "$$" instead of "##" to show response is accepted and the presentation is terminated. If no response is obtained in that period, it is considered as the tone was not heard. Table 5.1 shows presentation of stimulus and mask to the left headphone, right headphone and bone. Since subject' response is communicated to the microcontroller using its external interrupt pin (INT0), it has higher priority than any other tasks. When an interrupt is invoked from this pin, the microcontroller runs its interrupt service routine (ISR).

In the manual mode, the audiologist will decide the next level of tone depending upon the previous responses. The test algorithm as shown in Fig. 5.5 is used for searching the hearing threshold for a selected frequency [30]. This threshold is saved in an array in the memory of the microcontroller. After completing the test, the result can be displayed sequentially for all the frequencies. It can also be downloaded to a computer or printed using the serial port of the instrument.

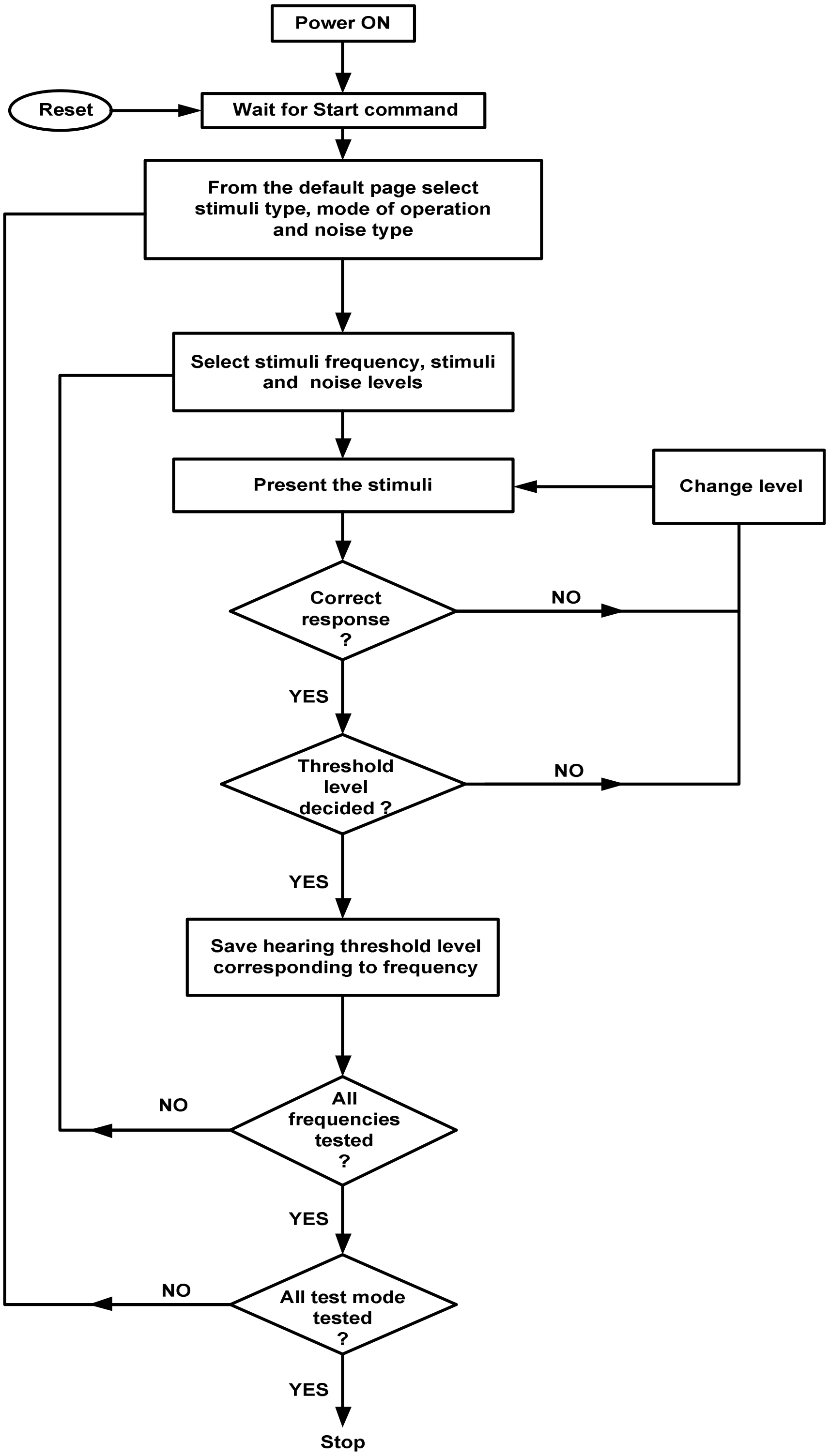

Fig. 5.2 Operation sequence of the audiometer.

Table 5.1 Presentation of stimulus and masking noise.

Mode	Left Phone	Right Phone	Bone Vibrator
LtAC	Stimulus	Masker	---
RtAC	Masker	Stimulus	---
LtBC	---	Masker	Stimulus
RtBC	Masker	---	Stimulus

In the automated mode, audiologist has to select the initial level and the frequency. The test frequency is to be set manually. The operator should press the “Start” key, to start the automated test. The instrument conducts the test according to the automated test algorithm as shown in Fig. 5.1. The hearing threshold obtained is stored automatically. There is one memory location for storing the test results for a particular frequency. The most recent result overwrites the previously stored result.

Before SISI test is carried out, test for hearing threshold for pure tone should be carried out otherwise 0 dB will be taken as tone threshold then the tone is set to 20 dB above the tone threshold. To start the test “Start” key should be pressed. Then the instrument will take care of the rest and will find out the score. In between operator can stop the test by pressing “Stop” key. SISI test is performed in automated mode. But the test frequency is selected manually.

In tone decay test, operator will set the frequency and a tone level of 30 dB HL. To start the test “Start” key should be pressed. Then the instrument will take care of the rest and will find out the threshold. In between operator can stop the test by pressing “Stop” key.

In speech audimetry test, the operator sets the level through keypad. The audiologist, depending upon the subject response determines the next level.

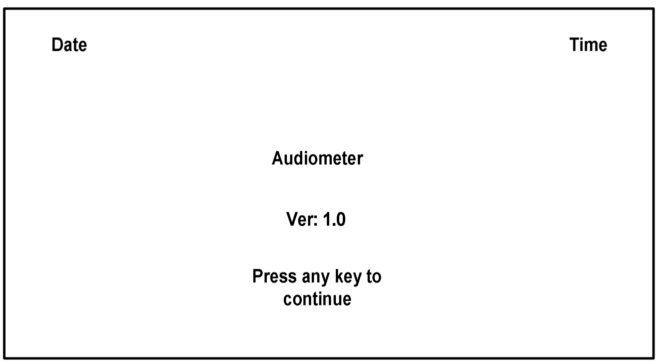

Fig. 5.3 Display, waiting for start command.

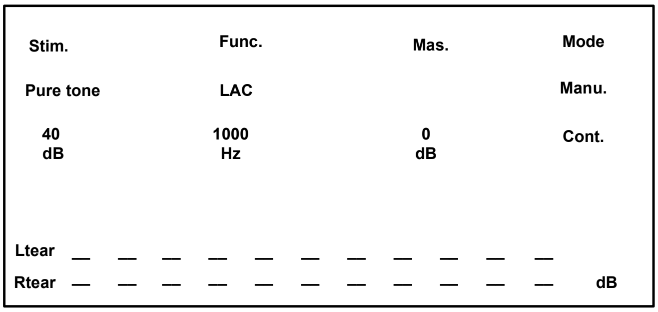

Fig. 5.4 Default graphical display of the audiometer.

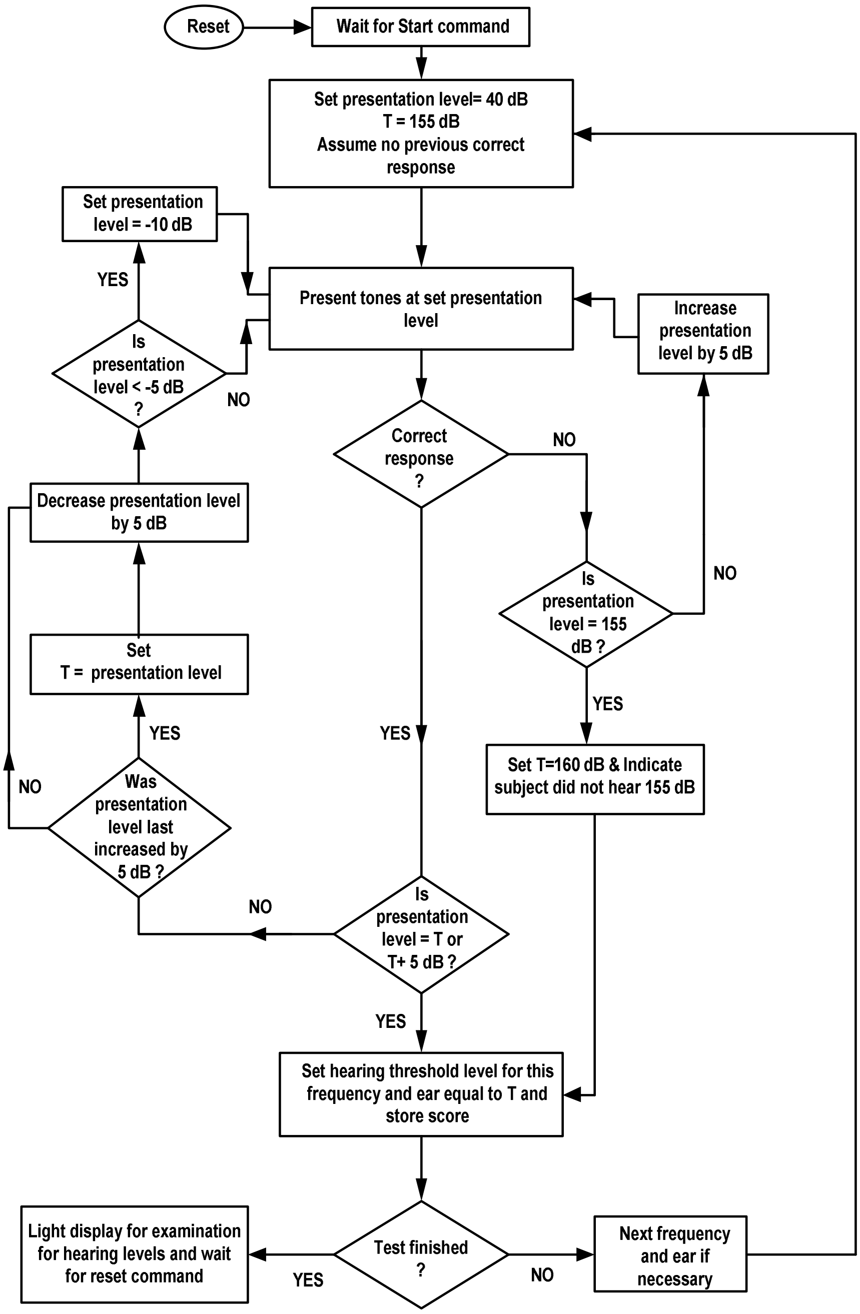

Fig. 5.5 Flow chart for pure tone audiometery.

5.5 Key functions

For user interface such as selection of stimuli type, noise type, frequency, test mode and level control, a 4×4 matrix keypad is interfaced to the microcontroller. The keypad is directly connected to Port 1 of the microcontroller. The keypad is scanned at an interval of 15 ms as a background process, by using interrupts generated by Timer 0. The scan process provides a debouncing interval of one scan cycle of 15 ms. Invalid presses are ignored using this technique. The functions assigned to the keys are given below.

- **Stim key:** The default stimulus setting at power up is "Pure tone". It can be change to "Warble tone" or "Speech" for live speech test or using recorded sources by pressing this key.
- **Func. key:** The default output is "Left Air Conduction" and you can select "Left Bone Condution", "Right Air conduction" and "Right Bone Conduction" or both "Left Air Conduction" and "Right Air Conduciton" using this key. There is just one output for all bone conduction tests.
- **Noise key:** The default noise setting at power up is "No noise" you can change it to "Wide band noise" or "Narrow band noise" by pressing this key.
- **Mode key:** There are four possible modes**;** manual mode**,** automated mode**,** SISI test mode and tone decay test mode. The default setting is "Manual" and you can change it to one of the above modes using this key.
- **S_dB ▲ key:** This key is used to increase stimuli level. Pressing this key will increment the current stimuli level by 5 dB. If the tone level reaches its maximum dB level, pressing this key has no effect.
- **S_dB ▼ key:** This key is used to decrease stimuli level. Pressing this key will decrement the current stimuli level by 5 dB. If the tone level reaches its minimum dB level, pressing this key has no effect.
- **N_dB ▲ key:** This key is used for noise level up. Pressing this key will increment the current stimuli level by 5 dB. If the tone level reaches its maximum dB level, pressing this key has no effect.
- **N_dB ▼ ke:** This key is used for noise level down. Pressing this key will decrement the current stimuli level by 5 dB. If the tone level reaches its minimum dB level, pressing this key has no effect.
- **F ▲ key:** This key is used to increase frequency. Pressing this key will change the frequency to the next higher audiometry frequency. The frequencies you can have are; 125, 250, 500, 750, 1000, 2000, 3000, 4000, 6000 and 8000 Hz. If the frequency reaches its maximum, which in this case is 8000 Hz, pressing this key has no effect. Every time new frequency is selected, the tone level is set to 40 dB.

Table 5.2 Data read for key presses.

Key pressed	Scan pattern	Key code
Stop	0xEE	0
Start	0xED	1
Read	0xEB	2
Test	0xE7	3
Save	0xDE	4
N_dB ▲	0xBD	9
N_dB ▼	0xDD	5
S_dB ▲	0xB7	11
S_dB ▼	0xD7	7
F ▲	0xBB	10
F ▼	0xDB	6
Clear	0xBE	8
Mode	0x7E	12
Noise	0x7D	13
Func.	0x7B	14
Stim	0x77	15

- **F ▼ key:** This key is used to decrease frequency. Pressing this key will change to the next lower audiometry frequency. If the frequency reaches its minimum, which in this case is 125 Hz, pressing this key has no effect. Every time new frequency is selected, the tone level is set to 40 dB.
- **Test key:** The default test is "Continuous test" and you can select "Interrupted test" using this key. Pressing this key again will keep recycling through the two options.
- **Save key:** This key will be used to save the hearing threshold level corresponding to the tested frequency.
- **Clear key:** This key will be used to clear the stored hearing threshold level corresponding to the tested frequency.
- **Start key:** This key is pressed to present the stimulus to the subject. The preselected tone type, frequency, level and duration will be presented accordingly. When the tone is on, a message "##" is displayed at the bottom right of the display.
- **Stop key:** This key is used to stop the presentation of tone and a message "XX" will be displayed at the bottom right corner of the display to show the tone has been stoped.
- **Read key:** This key will be used to read back the stored hearing threshold level corresponding to the tested frequency.

5.6 PC interface program

To download the test results from the audiometer using its serial port and to plot audiograms, a program has been written in C to run on a PC. It receives data from the audiometer over serial port of 9600 baud rate, 7 bit data with no parity, and 1 stop bit.

Patient information (name, age, sex, and test ear and test mode) is entered through PC keyboard. These data are stored in a file which can be printed. In the audiogram, patient information fields not filed are shown as blank and these can be written after taking a hardcopy of the audiogram. Through this program, a previously stored file can be used for displaying audiogram on screen. The plotted audiogram can be stored by saving the screen and it can be used as part of the document for obtaining hard copy.

5.7 System assembly and test results

The instrument was assembled and tested block by block on breadboard then designed and converted to PCB. There are two PCBs. PCB-1 consists of the microcontroller, DDS, attenuator, multiplexer, power amplifier, serial communication, circuit for graphical display, keypad, circuit for response switch and positive and negative power supplies. PCB-2 consists of ciruit for white band and narrow band noise and I^2C interfacing.

The sizes of PCB-1 and PCB-2 are 17.0 cm × 11.22 cm and 9 cm × 6.8 cm respectively. Both the PCBs are double sided with plated through holes (PTH). Most of the circuit blocks on PCB-1 have mixed signals (i.e. analog and digital); consequently special care was needed in the layout design. In PCB-1, the tone oscillator uses DDS, which needs digital clock. The data and control lines of the tone and noise attenuators (logarithmic resistor attenuator PGA2310) are interfaced to the microcontroller. Thus, almost everywhere analog is meeting digital. There is a great possibility of analog supply being corrupted by digital switching noise. To avoid this decoupling capacitor has been provided for each IC near its power supply pin. Also the supply has been routed carefully. The layouts of component solder side and component placement is given in Appendix F. The analog ground and digital ground are routed separately throughout the PCB. The two grounds are shorted by two-jumpers at the electrically most stable point on the PCB (just at the output of the regulators). Thick copper plane of ground is provided on one side of the PCB for having distributed capacitance for decoupling effect. Each IC is decoupled by 0.1 μF ceramic capacitor placed electrically as close as possible to the supply and ground pins of the particular IC. The layout is designed to provide ground shielding between the analog and digital pins of the IC. For observing the output of major blocks of the circuit, test points are provided. The supply entry points are decoupled by 100 μ F/25 V electrolytic capacitor in parallel to 0.1 μF ceramic capacitor. Care is taken to minimize the length of the supply path for power amplifier ICs.

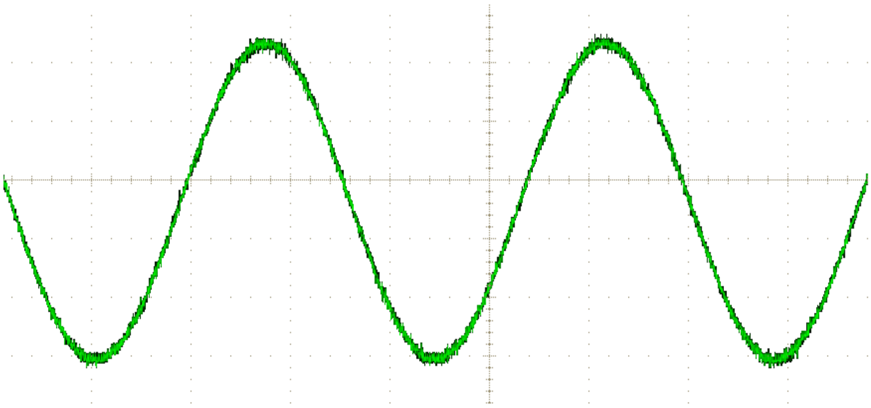

Fig. 5.6 Pure tone wave form, 1000 Hz.

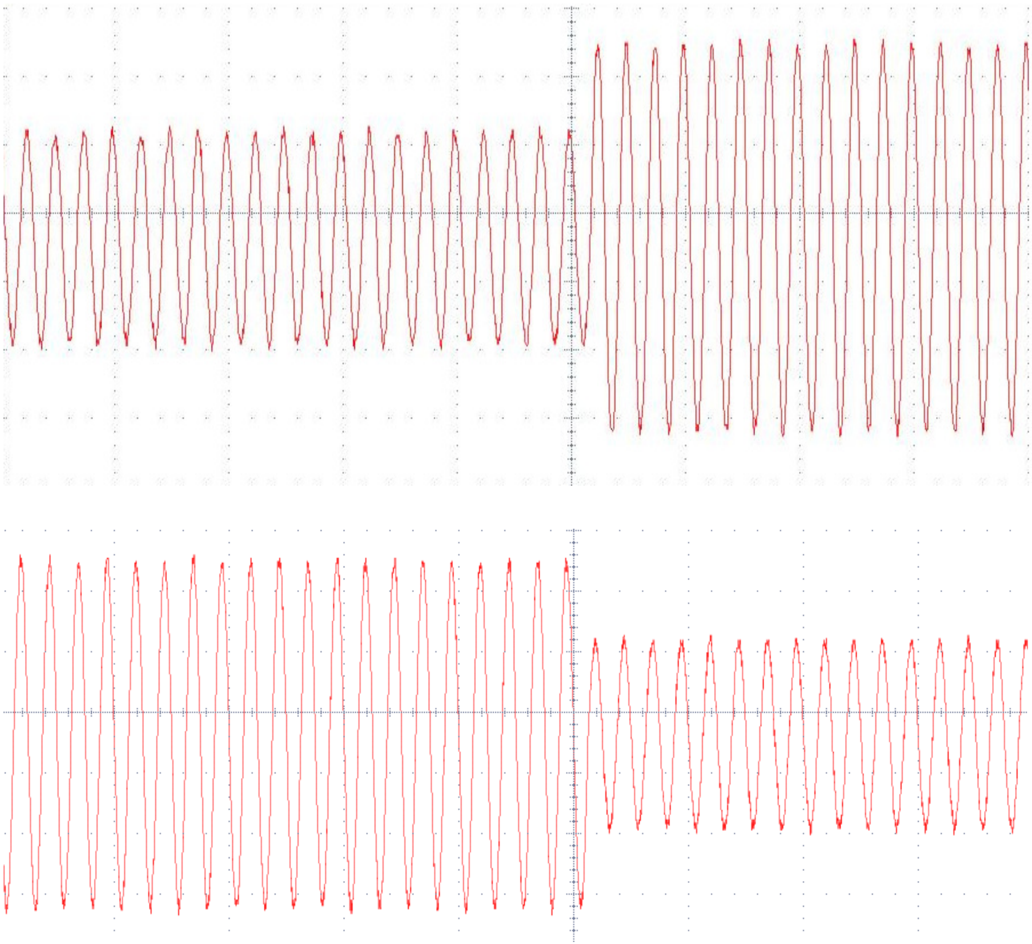

Fig. 5.7 Amplitude change of pure tone at zero crossing.

Chapter 6
SUMMARY AND CONCLUSION

6.1 Introduction

The objective of the project was to develop a microcontroller based audiometer, by investigating the limitations and shortcomings of the earlier designs. It was designed to provide different test modes like manual and automated tests, tone decay and SISI tests, and continuous tone and interrupted tone, air-conduction and bone-conduction tests facilities, facility for speech input from live speech, tape or CD for speech audiometry. The overall design of the audiometer has also taken care of portablity and low cost.

6.2 Conclusion

A scheme for a microcontroller based audiometer with speech audiometry was studied and all part of the audiometer were designed and tested. A microcontroller, P89V51RD2, was used as control unit of the audiometer. The microcontroller was interfaced with the audio oscillator, attenuator, noise generator, keypad and graphical display. Serial interface to transfer the stored data to a computer or printer is also provided. The audiometer can operate using menu driven software as well as automated mode. The microcontroller controlled circuit provides flexibility in selection of the frequency as well as tone level. The external hardware interrupt pin, one of the microcontroller port pin is connected to the subject's response button so that microcontroller will be able to respond immediately in the interrupt mode.

A programmable waveform generator based on a DDS AD9834 is used to generate pure tone (sinusoidal signal) of the frequency in the range of 125 Hz to 8 kHz. All the audiometry pure tone frequencies 125, 250, 500, 750, 1000, 1500, 2000, 3000, 4000, 5000, 6000 and 8000 Hz are generated as per the specifications. Warble tone is also generated using the programmable waveform generator, AD9834. A frequency modulation of $\pm$ 5 % is choosen with 64 discrete steps of frequency per sweep per second, to generate warble tone using DDS. To avoid the possibility of discontinuity while switching from one frequency to another frequency, both frequency registers of the DDS are used alternatively. Warble tone is generated for all eleven standard audiometry frequenies.

After testing attenuator circuits using linear and log attenuator ICs, finally log attenuator, PGA2310 was selected for implementing the attenuation for controlling the output levels. It has two independent channels with 120 dB dynamic range and uses three wire interfacing (SPI).

A DSP microcontroller is programmed to generate wideband noise and narrow band noise. A 16-bit software implemented shift register was used to generate pseudo-random binary sequence (PRBS). The output of the shift register is output at 36 kHz sampling rate through the on-chip DAC of the DSP. Some code is also written to filter the wideband noise to get narrow band noise centred at selected frequency. But this part need some work.

A 4x4 keypad and 128x64 graphical display is used for interface. Two keys (up/down) are dedicated for selecting frequency. Another two keys (up/down) are dedicated for selecting level of the tone in dB. Similarly another two keys (up/down) are dedicated for selecting level of the noise in dB. One key is dedicated for selecting type of test tone like pure tone, warble tone, and speech. Another key is dedicated for selecting type of noise either wideband noise or narrowband noise or no noise. Another key is dedicated for selecting the type of conduction mode: either air or bone conduction test. Two other keys; one for starting the test and one for stopping the test, are also used. All the parameters can be set using the 4 ×4 soft keys and the 128x64 LCD or by serial interface to the microcontroller.

For the microcontroller based audiometer hardware, a 2-layer PCB layout with PTH has been prepared and fully tested. The schematic for PCB-1 and PCB-2 and the PCB-1 layout (consists of DDS, attenuator, keypad, graphical display, and serial comunnication) and PCB-2 layouts (DSP controller for noise generation) are given in Appendix E and F respectively.

6.3 Suggestions

Suggestions for future work on making the audiometer fully functional can be listed as the following.

1. DSP has to be reprogrammed to generate PRBS using 24-bit or longer shift register. Wideband noise and narrow band noise should be generated and tested.
2. All parts of the audiometer should be assembled, tested and boxed. All its modes (manual, automated, SISI test and tone decay test) should be carefully checked.
3. A serial communication band graphical-user interface program, running on a PC has to be developed to enter subject’s information such as name, age, sex from PC and to get hearing threshold level, test ear and test mode from the audiometer.
4. The administration of the tone decay test has to be rechecked and tested.
5. Further improvement can be carried out to incorporate the features and specifications of an advanced diagnostic audiometer. In this design, attenuation table for the output device is to be hard coded in the program memory.
6. The software can be enhanced so that the instrument has a calibration mode, in which the user can load calibration via keypad. Two keys can be used for increasing or decreasing attenuation in the calibration mode. The audiometer output can be given to the headphone, which is placed on the artificial ear. The output tone level can be

monitored by the sound level meter, and the electrical signal from the meter is coupled to the tone level monitoring circuit. This can be achieved by introducing a calibration feedback input that gets in place of audio amplifier output, and by making appropriate software changes. A serial EEPROM interfaced to the microcontroller can be used for storing calibration table. The design at this stage can be further enhanced so that the EEPROM can store calibration tables for a number of output devices and user can select the table from amongst a number of output devices.

Appendix A

HEARING LEVEL IN dB

Human ear responds to an enormous range of pressure. Thus it is more convenient to express sound pressure in decibels. The sound pressure level (SPL) is expressed as;

$$\mathrm{L(dB)} = 20\log(P_o / P_R) \tag{A.1}$$

Where P_O is the sound rms pressure and P_R is the reference pressure. Generally, $P_R = 20$ µPa which is the least pressure required for an average listener to hear a tone of 1 kHz. Other means of expressing intensity is to specify sound pressure at a particular frequency relative to the pressure at absolute threshold for that frequency, and it is known as dBHL (decibel hearing level). It allows normal hearing to be operationally defined as a straight line at 0 dB HL. Thus, at different frequencies, the SPL levels corresponding to 0 dB HL are different. The sound stimuli are generated by presenting the signal in the form of electrical voltage to the output transducer, the headphone, insert earphone, the loudspeaker, or the bone vibrator. These devices have frequency dependent response. Thus the electrical voltages presented should be adjusted for the frequency dependent of the hearing level as well as for the transducer.

Appendix B

HEARING THRESHOLD DETERMINATION

Ambient noise should be measured using sound level meter immediately before the audiometry. Subjects should be correctly fitted with headphones. The subject should be instructed to press the response switch every time he/she hears the sound emitted, even if it is very faint. The searching protocol for hearing threshold is based on the one recommended by the American Speech and Hearing Association (ASHA) and adopted by the ANSI in 1963. The searching protocol for hearing threshold can be represented by flowchart as shown in Fig. 5.5. The presentation of sound started at 40 dB hearing level (dB HL) at 1 kHz. If there is no response at this threshold, it is raised in 5 dB increments until the subject responds to the sound. When the subject responded to a sound, the hearing threshold is then obtained, decreasing thresholds by 5 dB steps and increasing by 5 dB steps, until the threshold is established and confirmed on three consecutive occasions. These thresholds are established in the same manner at 2 kHz and 4 kHz, and then back to 1 kHz until they are within 5 dB of the original measurement at 1 kHz. If not, the entire procedure is repeated.

Start with the better-hearing ear (according to the patient's account). Start with 1000 Hz. Next proceed to test 2000, 4000, 8000, 500, 250 and 125 Hz. For the first ear only, retest at 1000 Hz. If the retest value is more than 5 dB more acute than the original value, retest the next frquency and so on, [7]. Take the more acute threshold as the final value. If needed, test also at intermediate frequencies 750, 1500, 3000 and 6000 Hz. Then test the opposite ear in the same order without the retest at 1000 Hz.

The basic procedure for determation of a patient's threshold starts with familiarizing the patient with the test. To do this present a tone of about 3 s duration at a level that is expected to be clearly audible to the patient, which will commonly be about 40 dB above the roughly estimated threshold, [8]. Check that the patient responds correctly. If there is no response, raise the level in 10 dB steps until a responses is obtained. After the patient is familiarize with the test the basic prcedure for determination of a patient's hearing threshold will be bone as follows.

1. Reduce the levels in 5 dB steps until the patient no longer responds.
2. The tone level of succeeding presentations is determined by the preceding response. After reaching no response from the patient in stage 1, and after each subsequent failure to respond to a signal, the level is increased in 5 dB steps until a response occures. After the response, the intesity is decreased 5 dB and another ascending 5 dB series is begun and so on until the threshold becomes evident. Threshold is defined as the lowest level at which responses occur in at least half

of a series of ascending trials with a minimum of two responses required at that level.

All possible auditory, visual and tactile clues should be eliminated from the tester and the apparatus, and it is essential for the tester to see the patient. The patient's response must not generate any audible sound and must involve minimal movement; acknowledgement of the tone by signalling with a push botton or by raising and lowering the finger is satisfactory. A vocal 'yes' or tap on the table are to be avoided. This aids the tester's precision in identifying valid responses [7].

Appendix C

HEADPHONE CALIBRATION

In order to determine the voltage levels to be generated to produce the audiometric range of sound pressure levels, headphone was calibrated using B&K artificial ear (type 4153) and spring pressure on the headphone was kept at 0.5 kg. The voltage level required to produce 100 dB SPL at different frequencies were noted. The voltage level (in dBm) required to produce 100 dB SPL are calculated using 0 dBm as a reference (i.e sinusoidal voltage required to produce 1mW to 600 Ω load which is 2.19 V peak to peak). The impedance characteristics of different headphones were tested and are shown in Table C.1 to C.7.

Table C.1 Headphone model B51055.

Freq. (Hz)	Impedance		Voltage for 100 dB SPL	
	R (Ω)	X (Ω)	Peak-peak (mV)	dBm
125	9.441	0.124	1328	2.4
250	9.456	0.228	507	-7.5
500	9.520	0.419	269	-13.5
750	9.589	0.597	198	-15.8
1.0 k	9.695	0.738	193	-16.6
1.5 k	9.838	0.925	173	-16.4
2.0 k	9.925	1.119	169	-16.2
3.0 k	10.117	1.299	90	-20.1
4.0 k	9.958	1.796	148	-17.3
6.0 k	10.207	2.495	350	-12.0
8.0 k	10.562	3.313	700	-6.6

Table C.2 Headphone model B51045.

Freq. (Hz)	Impedance		Voltage for 100 dB SPL	
	R (Ω)	X (Ω)	Peak-peak (mV)	dBm
125	9.820	0.290	988	2.1
250	9.840	0.591	309	-8.0
500	9.990	1.620	127	-15.7
750	13.300	-9.25	51	-23.7
1.0 k	9.958	-0.792	336	-7.3
1.5 k	10.031	0.049	256	-9.6
2.0 k	10.108	0.612	216	-11.1
3.0 k	10.250	1.012	245	-10.0
4.0 k	10.322	1.484	191	-12.2
6.0 k	10.553	2.312	409	-5.6
8.0 k	10.866	3.056	709	-0.8

Table C.3 Headphone model DR-52, White.

Freq. (Hz)	Impedance		Voltage for 100 dB SPL	
	R (Ω)	X (Ω)	Peak-peak (mV)	dBm
125	7.496	0.146	594	-2.3
250	7.550	0.102	231	-10.5
500	7.481	0.214	133	-15.3
750	7.478	0.389	142	-14.7
1.0 k	7.571	0.556	91	-18.6
1.5 k	7.694	0.595	76	-20.1
2.0 k	7.667	0.789	86	-19.1
3.0 k	7.755	1.138	74	-20.4
4.0 k	7.905	1.475	113	-16.7
6.0 k	8.228	2.073	213	-11.2
8.0 k	8.303	2.507	725	-0.6

Table C.4 Headphone model DR-52, Black.

Freq. (Hz)	Impedance		Voltage for 100 dB SPL	
	R (Ω)	X (Ω)	Peak-peak (mV)	dBm
125	7.499	0.155	594	-2.3
250	7.610	0.129	238	-10.3
500	7.504	0.174	150	-14.3
750	7.520	0.365	222	-10.9
1.0 k	7.534	0.451	122	-16.1
1.5 k	7.703	0.599	95	-18.2
2.0 k	7.600	0.744	90	-18.7
3.0 k	7.787	1.085	84	-19.3
4.0 k	7.860	1.415	103	-17.5
6.0 k	8.123	1.996	259	-9.5
8.0 k	8.277	2.376	541	-3.1

Table C.5 Headphone model T-30, Red.

Freq. (Hz)	Impedance		Voltage for 100 dB SPL	
	R (Ω)	X (Ω)	Peak-peak (mV)	dBm
125	3.85	0.434	158	-13.8
250	4.22	0.909	53	-23.3
500	4.77	0.381	22	-30.8
750	5.96	0.495	26	-29.6
1.0 k	4.20	-0.738	13	-35.7
1.5 k	3.97	0.065	31	-28.0
2.0 k	3.96	0.399	44	-24.8
3.0 k	4.04	0.884	96	-18.2
4.0 k	4.19	1.19	138	-15.0
6.0 k	4.30	1.79	268	-9.2
8.0 k	4.48	2.35	795	0.2

Table C.6 Headphone model T-30, Blue.

Freq. (Hz)	Impedance		Voltage for 100 dB SPL	
	R (Ω)	X (Ω)	Peak-peak (mV)	dBm
125	3.76	0.343	128	-15.6
250	4.03	0.723	33	-27.4
500	4.13	0.363	14	-34.6
750	6.56	0.780	17	-33.2
1.0 k	4.24	-0.809	8	-39.4
1.5 k	3.91	0.037	15	-34.2
2.0 k	3.88	0.359	23	-30.5
3.0 k	3.95	0.835	48	-24.2
4.0 k	4.09	1.145	69	-20.9
6.0 k	4.21	1.71	180	-12.7
8.0 k	4.40	2.23	491	-4.0

Table C.7 Headphone model TDH-39P.

Freq. (Hz)	Impedance		Voltage for 100 dB SPL	
	R (Ω)	X (Ω)	Peak-peak (mV)	dBm
125	9.590	0.146	1240	4.1
250	9.623	0.284	388	-6.0
500	9.747	0.498	198	-11.9
750	9.814	0.639	159	-13.8
1.0 k	9.880	0.802	153	-14.1
1.5 k	10.055	1.069	169	-13.2
2.0 k	10.190	1.300	172	-13.1
3.0 k	10.582	1.588	106	-17.3
4.0 k	10.424	1.899	96.3	-18.1
6.0 k	10.613	2.565	236	-10.3
8.0 k	10.964	3.369	678	-1.2

.

Appendix D

COMPONENT LIST

D.1 Component list of PCB-1 (8-bit microcontroller, display, keypad, serial communication, DDS, attenuator, multiplexer and power amplifier)

Component designator	Part Number/value	Component description	Quantity
C2,C36,C6	33pF	Capacitor(Ceramic)	3
C181,C30,C68,C69	0.1nF	Capacitor(Ceramic)	4
C14,C15,C17,C19,C20,C21, C23,C24,C26,C28,C29,C3, C33,C34,C37,C38,C40,C41, C42,C43,C44,C45,C47,C48, C49,C50,C51,C52,C53,C54, C55,C56,C57,C58,C61,C63, C64,C65,C73,C74,C75,C76, C77,C78,C79,C82,C83,C84,C9,	0.1uF	Capacitor(Ceramic)	57
C11,C12,C18	1uF/ 16V	Capacitor(Electrolytic)	3
C13,C16,C165,C22,C27,C35, C52,C60,C7	10uF/ 16V	Capacitor(Electrolytic)	9
C5,C66,C71,C72,C8,C81	100uF/ 16V	Capacitor(Electrolytic)	6
C1,C10,C25,C39,C46,C70	1000u F/ 16V	Capacitor(Electrolytic)	6
R5	8 Ω, ¼ W	Resistor	1
R12,R25,R27,R38,R42,R9	12 Ω, ¼ W	Resistor	6
R2,R28,R29,R3,R30,R31,R35, R36,R39,R4,R40,R7	33 Ω, ¼ W	Resistor	12
R1	100 Ω, ¼ W	Resistor	1
R17,R20	220 Ω, ¼ W	Resistor	2
R21	6.8 kΩ, ¼ W	Resistor	1
R23	8.2 kΩ, ¼ W	Resistor	1
R10,R14,R52,R54	10 kΩ, ¼ W	Resistor	4
R50,R53,R56	33 kΩ, ¼ W	Resistor	3
R11,R13,R15,R16,R18,R19,R22, R24,R26,R32,R33,R34,R37,R41, R43,R44,R45,R46,R47,R48,R49, R51,R55,R57,R6,R8	100 kΩ, ¼ W	Resistor	26
U1,U16	LM7805C	Regulator	2
U15	LM7905C	Regulator	1
U10,U12,U7	4053	Analog switch	3
U11	PGA2310	Attenuator	1
U13,U14,U6	MCP602	Op Amp	3
U17	AD9834	DDS	1
U2	P89V51RD2	Microcontroller	1
U3	MAX232	RS-232	1
U9	74HC373	8-bit latch	1
X1	12 MHz	Crystal	1
AV_IN, Serial		Three pin connector	1
S1		SPST switch	1
Audio1, Audio2, Noise in, SF		Phonejack	4
Bone, Lout, Rout		Phonomonjack	3

D.2 Component list of PCB-2(Noise generator)

Component designator	Part Number/value	Component description	Quantity
C7,C30,C29,C31	0.1uF	Capacitor(Ceramic)	4
C12	10uF/ 16V	Capacitor(Electrolytic)	1
U1	Ds33FJ129GP804	Digital signal control	1
U2	LM1117	+3.3 V regulator	1

Appendix E

SCHEMATIC DIAGRAM OF THE MICROCONTROLLER BASED AUDIOMETER

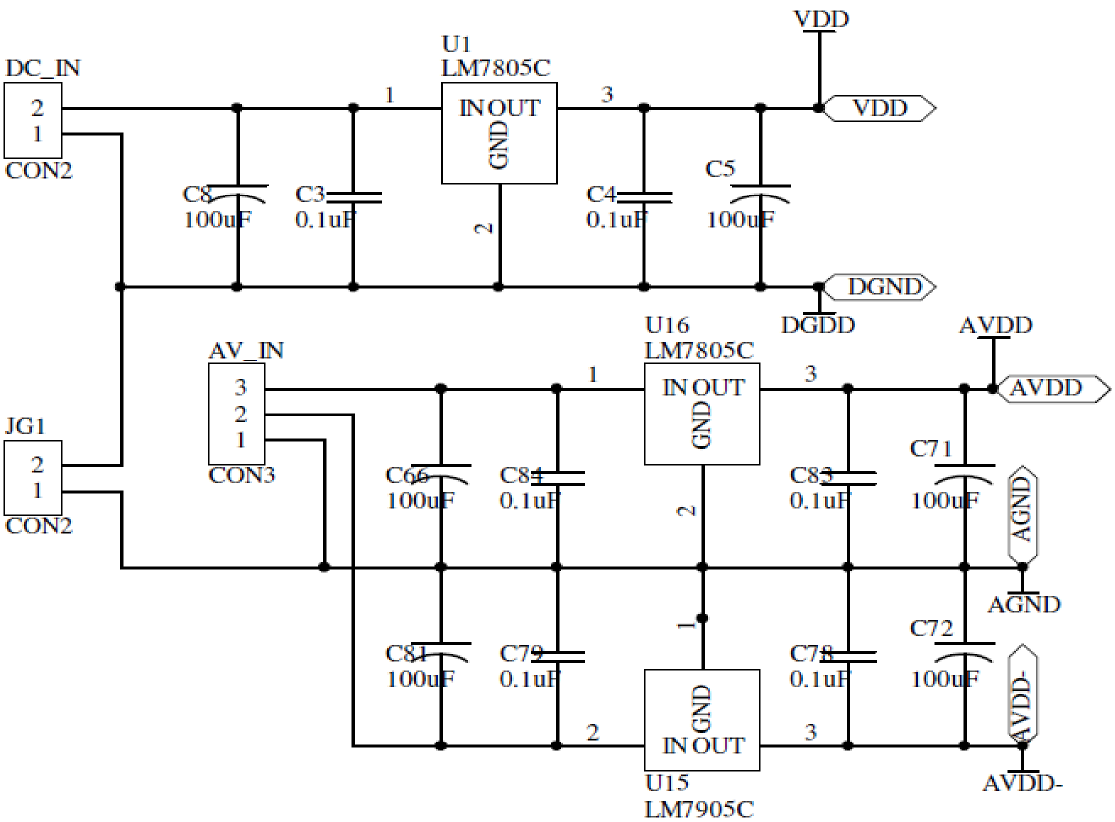

Fig. E.1 Schematic diagram of the power supply as the part of the microcontroller based audiometer.

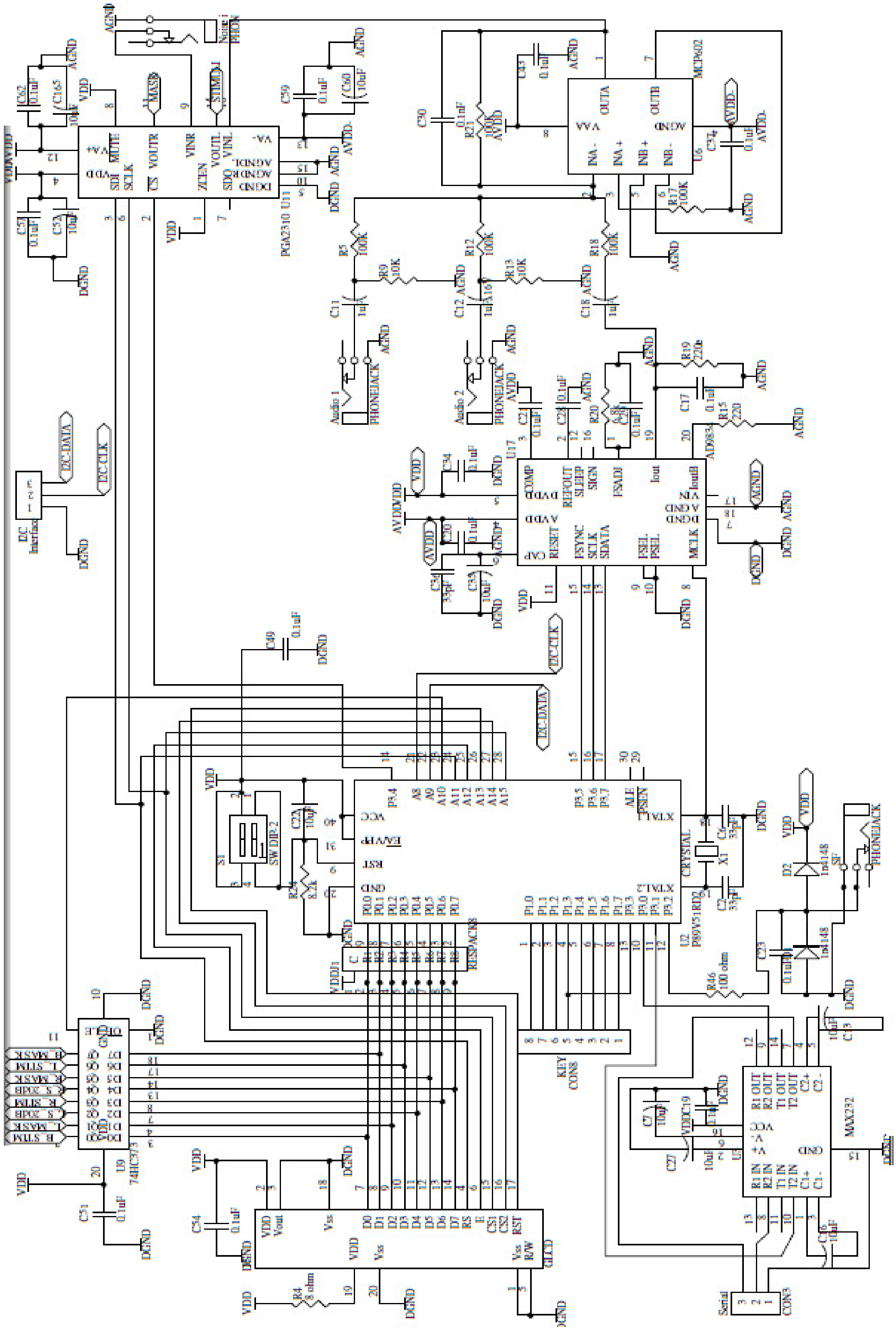

Fig. E.2 Digital part of the microcontroller based audiometer (for PCB-1).

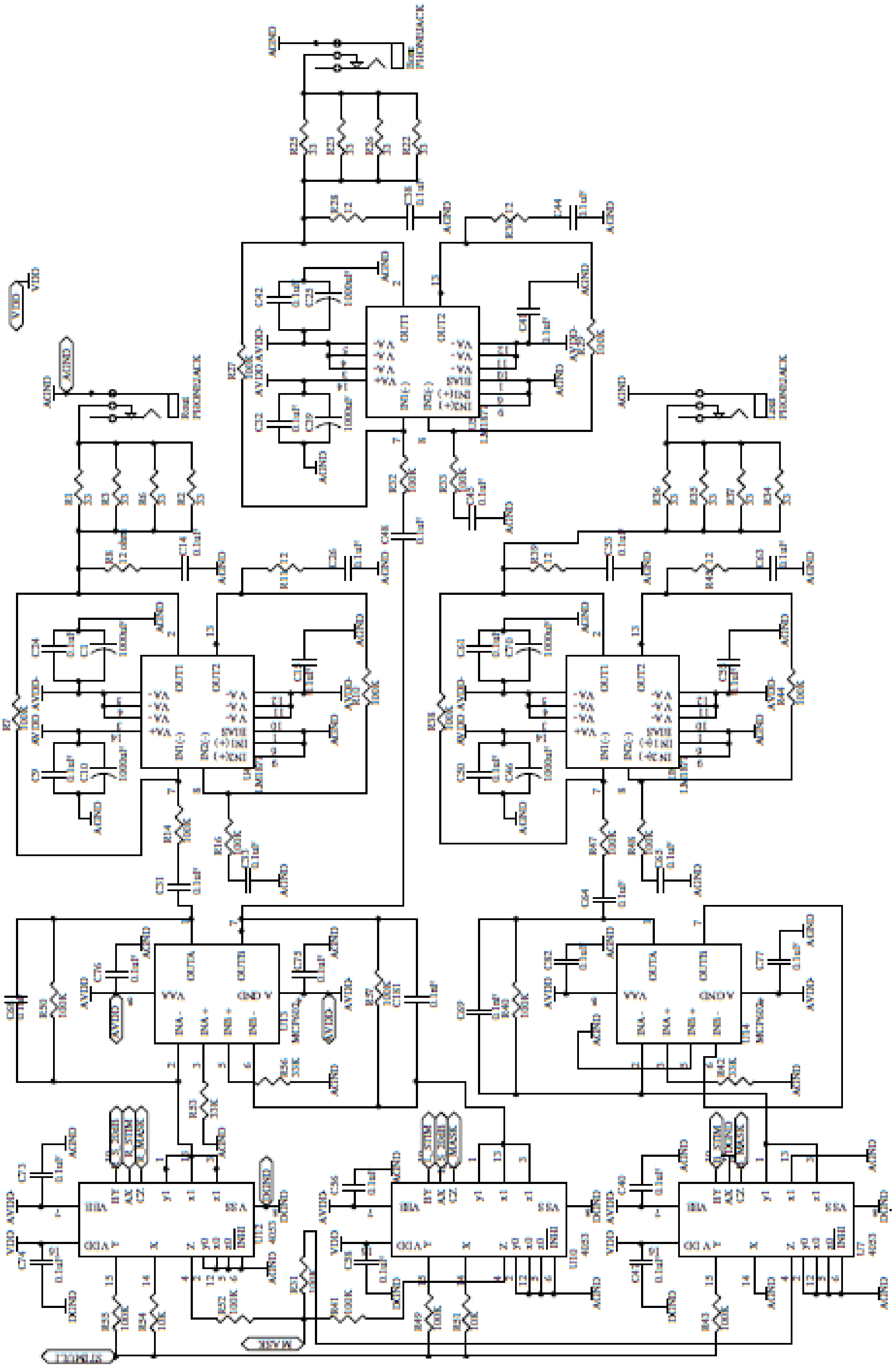

Fig. E.3 Circuits for multiplexer and power amplifiers as part of the microcontroller based audiometer (PCB-1).

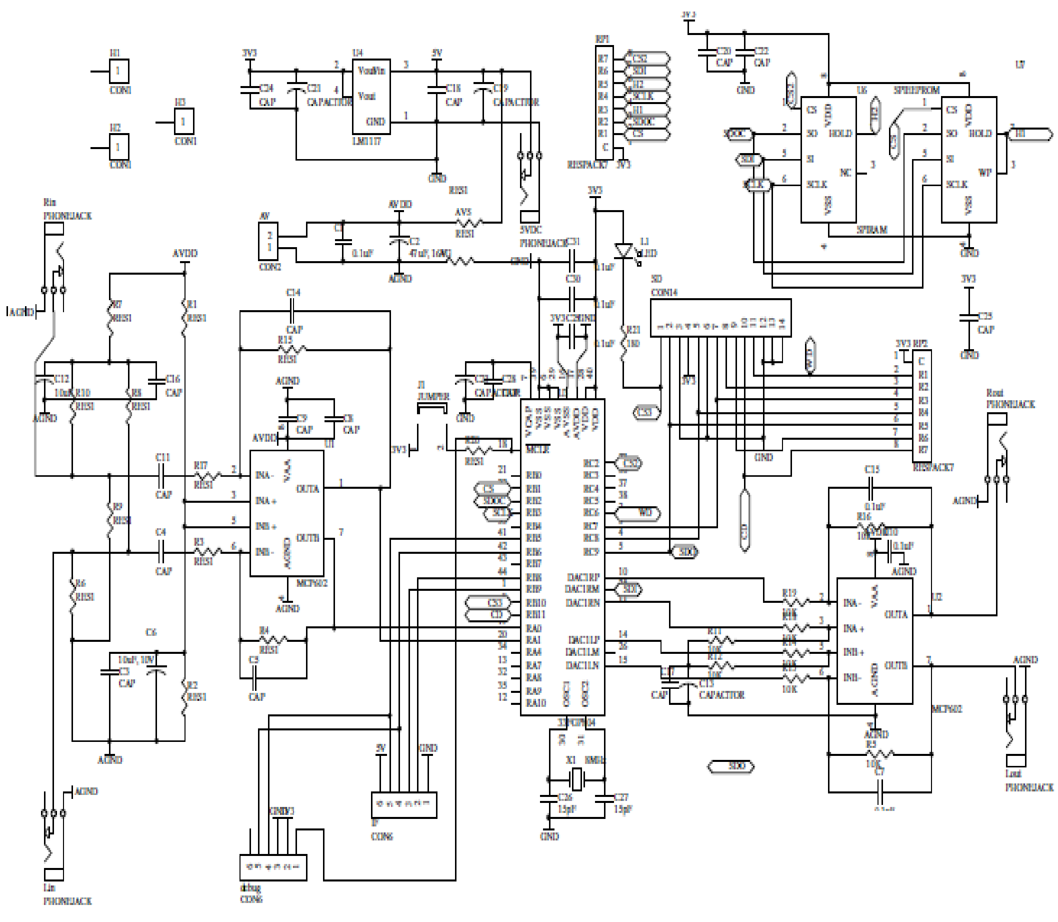

Fig. E.4 Circuits for noise generation circuit (PCB-2).

Appendix F

PCB LAYOUT OF THE MICROCONTROLLER BASED AUDIOMETER

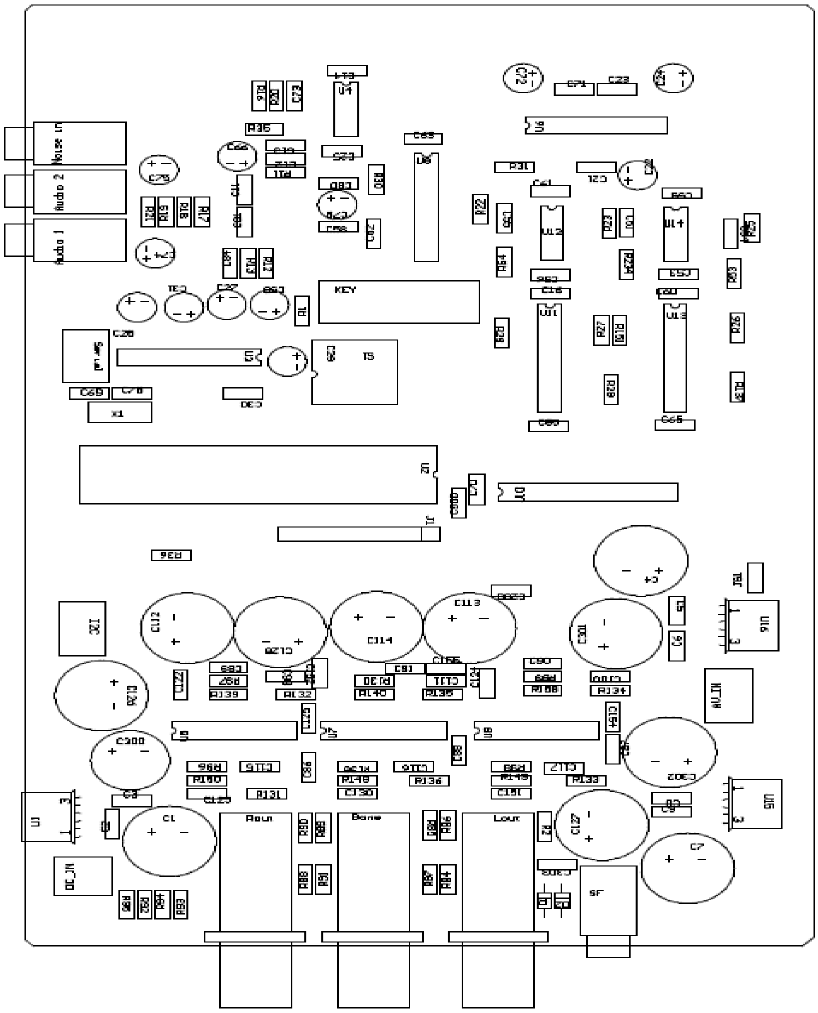

Fig. F.1 Component placement of the microcontroller based aduiometer PCB-1 (16 cm × 8 cm).

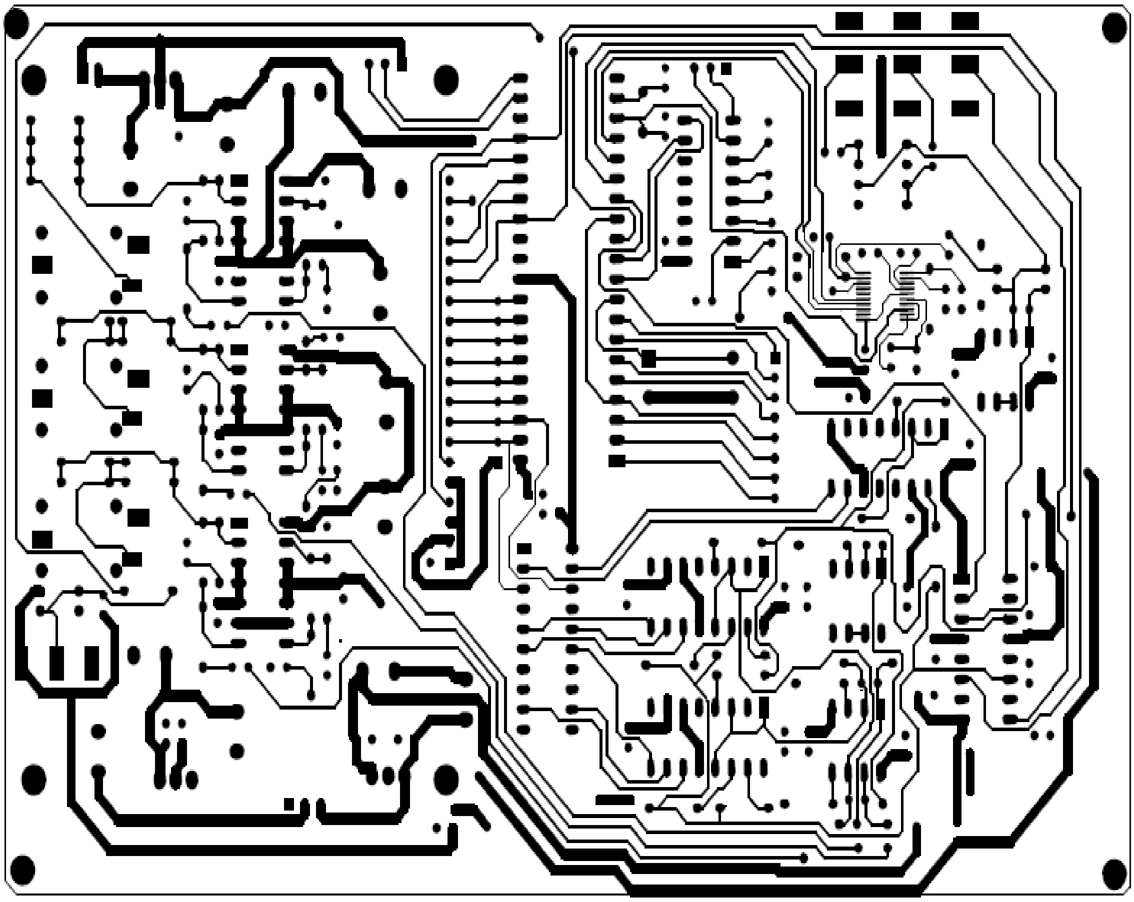

Fig. F.2 Component side of the microcontroller based aduiometer PCB-1 (17 cm × 11.22 cm).

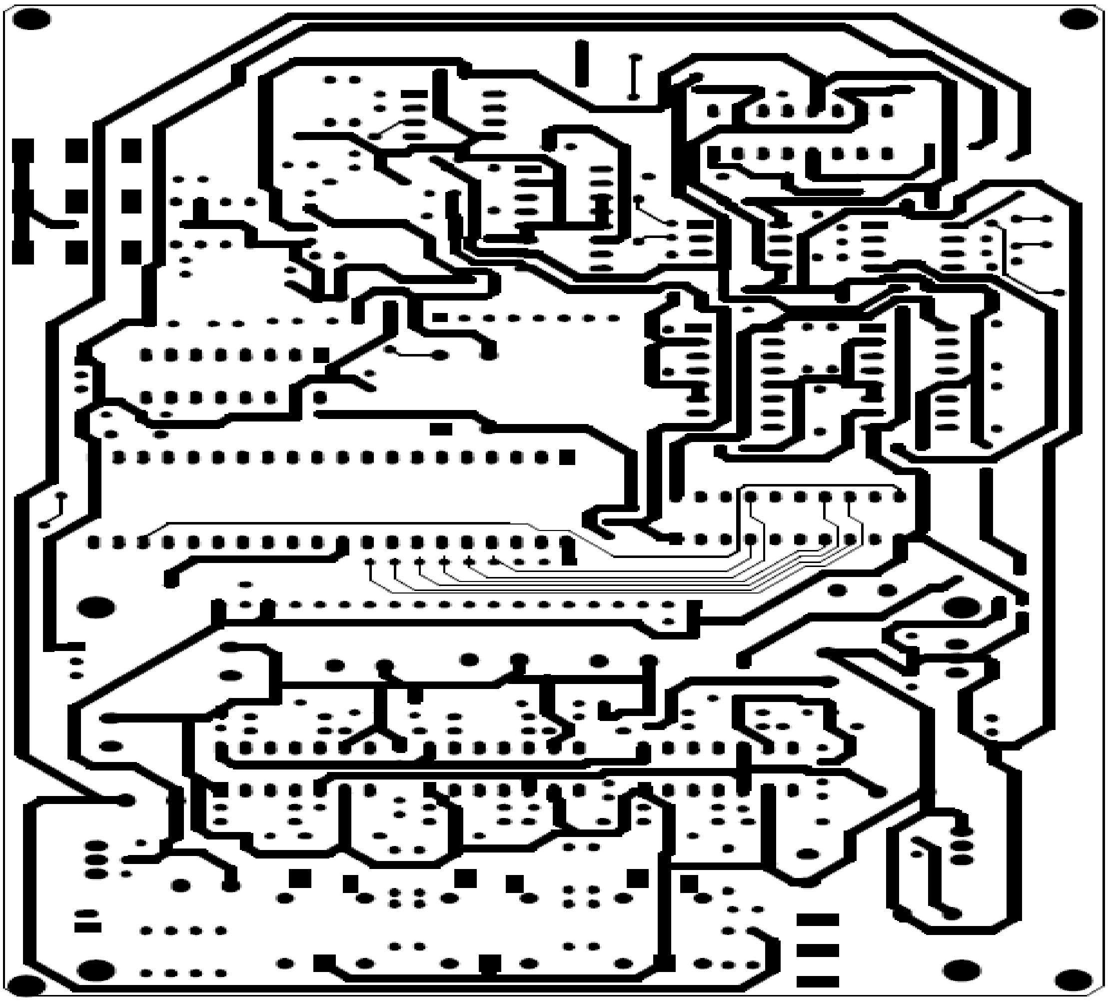

Fig. F.3 Bottom side of the microcontroller based aduiometer PCB -1(17 cm × 11.22 cm).

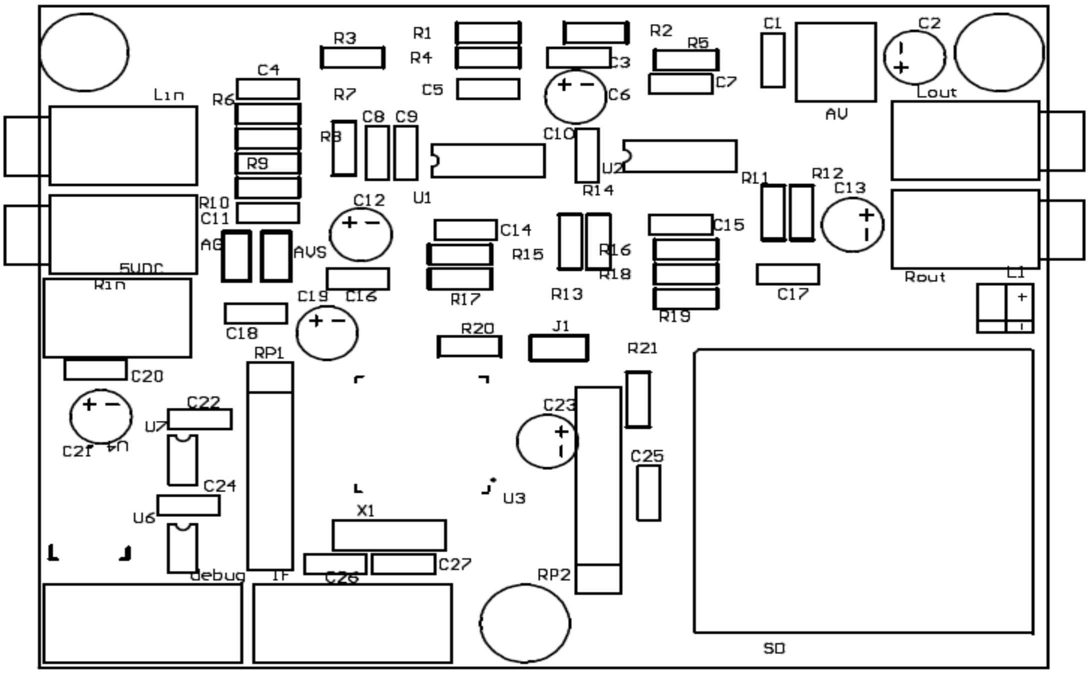

Fig. F.4 Component placement, top side, of the noise generation, PCB-2 (9.0 cm × 6.8 cm).

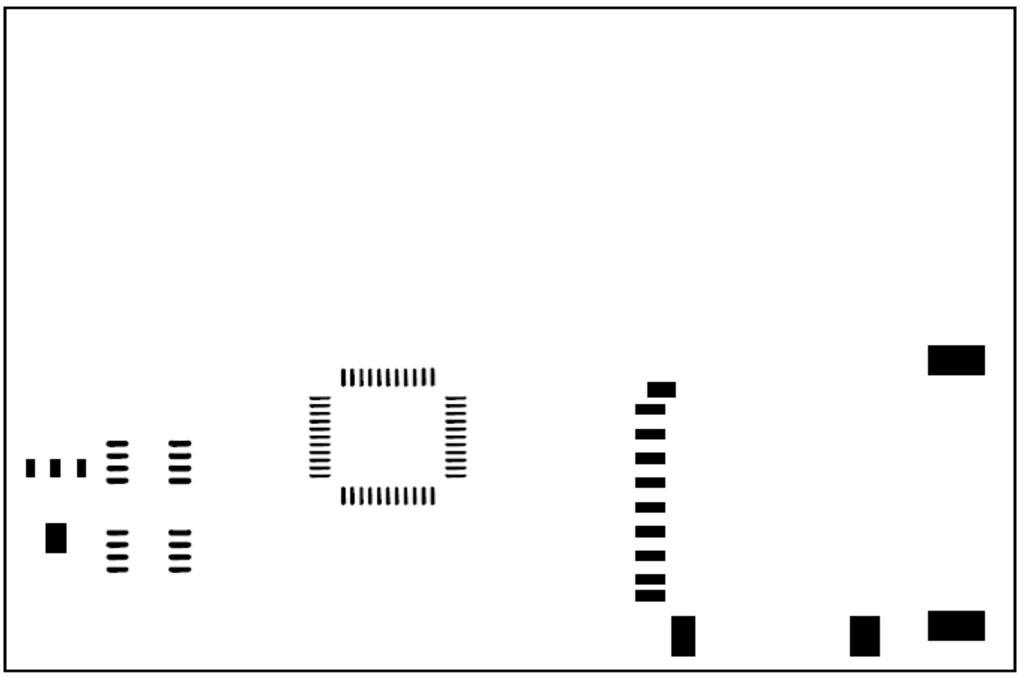

Fig. F.5 Components placement, bottom side, of the noise generation, PCB-2 (9.0 cm × 6.8 cm).

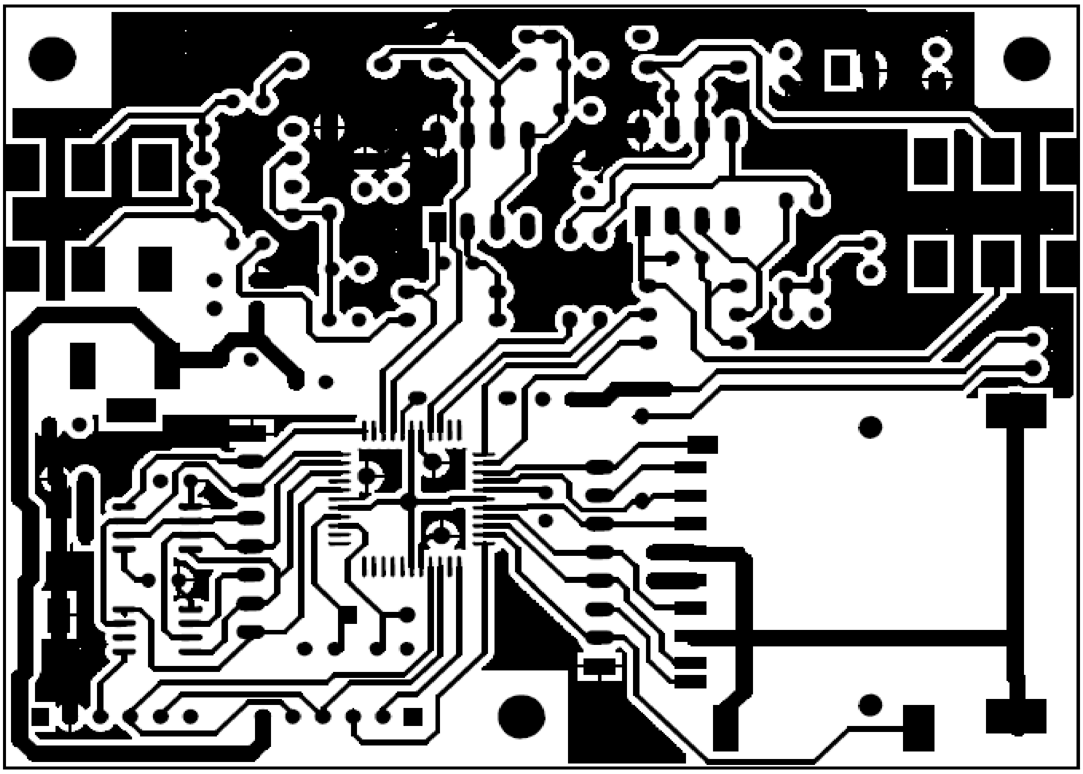

Fig. F.6 Components side of the noise generation, PCB-2 (9.0 cm × 6.8 cm).

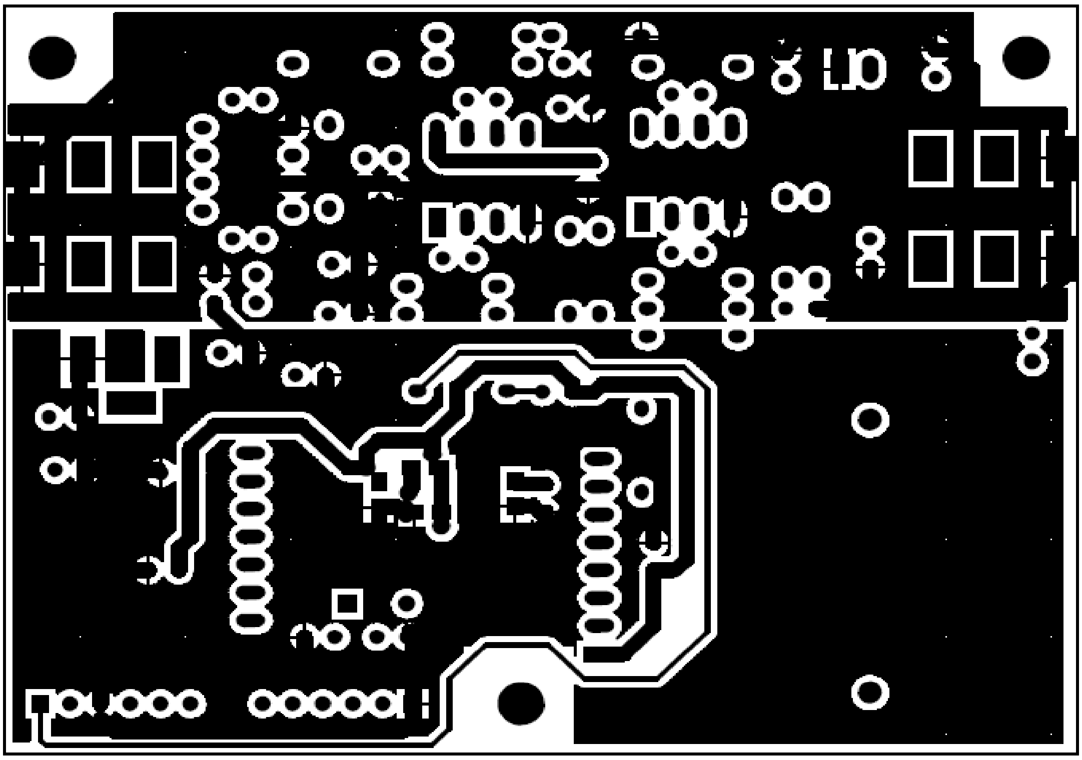

Fig. F.7 Bottom side of the noise generation, PCB-2 (9.0 cm × 6.8 cm).

REFERENCES

[1] A. Biswas, *Clinical Audio-Vestibulometry,* Mumbai: Bhalani, pp. 1-11, 1990.

[2] D. W. Kim, Y. C. Park, W. K. Kim, S. J. Park, W. Doh, S. W. Shin, and D. H. Youn, "Simulation of hearing impairment with sensorineural hearing loss", in *Proc. 19th Int. Conf. IEEE Eng. Med. Biol. Soc.*, Oct. 30 – Nov. 2, 1997.

[3] R. Giannini, S. Tonkovic and V. Dozina, "A DSP based method for audio stimuli generation", in *Proc. 19th Int. Conf. IEEE Eng. Med. Biol. Soc.,* pp. 1163-1166, Oct. 30 – Nov. 2, 1997.

[4] S. L. Tan, S. K. Loh, and W. C. Chee, "Speech-eanbled pure tone audiometer", in *Proc. International Symposium on Intelligent Signal Processing and Communication Systems* Nov. 28-Dec. 1, 2007 Xiamen, China.

[5] V. Aras / P. C. Pandey (Supervisor), "Microcontroller based audiometer user's manual", EE Dept., Indian Institute of Technology Bombay, 2003.

[6] S. Dalebout, *The Praeger Guide to Hearing and Hearing Loss,* Greenwood Publishing Group, 2009.

[7] British Society of Audiology, "Recommended procedures for pure-tone audiometry using a manually operated instrument", *British Jouranl of Audiology,* vol. 15, pp. 213 - 216, 1981.

[8] Y. Faycal, B. Wahiba, B. Lotfi, B. Ratiba, and A. Benia, "Computer audiometer for hearing testing", in *Proc. Int. Conf. on Advances in Electronics and Micro-electronics,* pp. 111-114, 2008.

[9] D. M. Patel / P. C. Pandey (Supervisor), "A microcontroller based audiometer", *M. Tech. dissertation*, EE Dept., Indian Institute of Technology Bombay, 2002.

[10] P. L. Reddy / P. C. Pandey (Supervisor), "A microcontroller based audiometer", *M. Tech. dissertation*, EE Dept., Indian Institute of Technology Bombay, 2001.

[11] K. Kochanek, L. Sliwa, J. Zajac, and H. Skarzynski, "A universal computer audiometer for objective hearing testing and screening", *Int. Workshop on Med. Meas. and Appl.*, May 4-5, 2007.

[12] G. G. Gentiletti-Faenze, O. Yanez-Suarez, J. M. Cornejo-Cruz, "Evaluation of automatic identification algorithms for auditory brainstem response used in universal hearing loss screening", in *Proc. 25th Annu. Int. Conf. IEEE Eng. Med. Biol. Soc.,* Sept. 17 – 21, 2003.

[13] P. Horowitz and W. Hill, *The Art of Electronics,* Cambridge: Cambridge Univ. Press, 1995.

[14] C. R. Meyer, and H. C. JR. Sutherl, “A technique for totally automated audiometery”, *IEEE Trans. Biomed. Eng.,* pp. 166-168, March 1976.

[15] L. Shao, B. E. Pfingst, “An intergrated real-time computerized hearing testing system”, *IEEE Eng. Med. Biol. Soc. 11th Annu. Int. Conf.*, 1967.

[16] W. H. JR. Avery, and C. Susskind, “Simple Bekesy type audiometer for clinical use”, *IEEE Trans. Acoustics, Speech Signal Proc.*, vol. ASSP - 25, no. 5, Oct. 1977.

[17] Philips Corporation, "8-bit microcontroller with 64K bytes in-system programmable flash P89V51RD2", Available online, http://www.keil.com/dd/docs/datashts/philips/p89v51rd2.pdf.

[18] Analog Devices, "Low power 20 mW 2.3 V to 5.5 V programmable waveform generator AD9833" Available online, www.analog.com/static/importedfiles/Data_Sheets/AD9833.pdf, downloaded on 4th July 2009.

[19] Analog Devices, "20mW power, 2.3 V to 5.5 V, 75 MHz complete DDS AD9834", Available online, www.analog.com/static/importedfiles/Data_Sheets/AD9834.pdf, downloaded on 28th July, 2009.

[20] Maxim Products, “Quad 12-bit Voltage-Output DACs with serial interface”, Available online, www.datasheetcatalog.org/datasheet/maxim/MAX536-MAX537.pdf, downloaded on 10th Aug. 2009.

[21] Michrochip Products, “Digital signal controller, dsPIC33FJ128GP804”, Available online, http://ww1.microchip.com/downloads/en/DeviceDoc/80371B.pdf.

[22] Texas Instrument Products, “Stereo audio volume control, PGA2310”, Available online, http://focus.ti.com/lit/ds/symlink/pga2310.pdf.

[23] H. C. Wu, L. P. Yang, C. H. Chen, S. T. Young, and T. S. Kuo, “A low-cost architecture of audiometer based on direct digital synthesizeer”, in *Proc.of the First Joint BMES/EMBS Conf. Serving Humanity, Advanced Technology* Oct. 13-16, 1999, Atlanta, GA, USA.

[24] IS: 10565, *Indian Standards Specification for Diagnostic Audiometers,* 1983.

[25] A. Kothari / P. C. Pandey (Supervisor), “A microcontroller based audiometer”, *M. Tech. dissertation*, EE Dept., Indian Institute of Technology Bombay, 1999.

[26] C. K. Singh / P. C. Pandey (Supervisor), “A microcontroller based audiometer”, *M. Tech. dissertation*, EE Dept., Indian Institute of Technology Bombay, 1997.

[27] V. Malini / P. C. Pandey (Supervisor), “A PC based audiometry system”, *M. Tech. dissertation*, EE Dept., Indian Institute of Technology Bombay, 1991.

[28] National Semiconductor Products, “Dual audio power amplifier, LM1877”, Available online, http://www.national.com/ds/LM/LM1877.pdf.

[29] M. E. Lutman, M. A. Cane, and P. A. Smith, “Comparison of manual and computer-controlled self-recorded audiometric methods for serial monitoring of hearing”, *British Jouranl of Audiology,* vol. 23, pp. 305 - 315, 1989.

[30] C. A. Martinez-Baez, L. F. Borjon, and A. Noyola, “Manual and automated programmable audiometer”, in *Proc. Annu. Int. Conf. IEEE Eng. Med. Biol. Soc.*, vol. 13, no. 4, pp. 1915-1916, 1991.

[31] Unisonic Technologies, “Analog multiplexer/ demultiplexer, 4053”, Available online, http://pdf1.alldatasheet.com/datasheet-pdf/view/117304/UTC/4053.html

[32] N. Nakamura, “Development of MobileAudiometer for screening using mobile phones”, in *Proc. 26th Annu. Int. Conf. IEEE Eng. Med. Biol. Soc.*, pp. 3369 – 3372, 2004.

[33] Texas Instrument Products, “Audio amplifier with digital control, TDA8551”, Available online, http://www.nxp.com/documents/data_sheet/TDA8551_T.pdf.

[34] National Semiconductor Product, ”Digital noise source”, Available online, http://pdf1.alldatasheet.com/datasheet-pdf/view/103155/NSC/MM5437.html

[35] S. D. Arlinger, “Comparison of ascending and bracketing methods in pure tone audiometry”, *Scand Audiol,* vol. 8, pp: 247 – 251, 1979.

[36] M. K. Nalamwar / P. C. Pandey (Supervisor), “Advanced audiometer: a novel signal generator technique”, *M. Tech Credit Seminar Report,* EE Dept., Indian Institute of Technology Bombay, 2004.

[37] M.L. Severns, “A computer controlled attenuator for audiological testing”, J. of Clinical Engg., vol. 10, no. 4, 1985.

[38] R. Prasad, T. Koike, and F. Matsuno, “Measurement of threshold of hearing using tone trains”, in *Proc. SICE Annu. Conf.* Aug. 20-22, 2008.

[39] G. Shangai and M. H. Loew, “An autoregressive model of the BAEP signal for hearing-threshold testing”, *IEEE Trans. Biomed. Eng.,* vol. 33, no. 6, June 1986.

[40] D. L. Hollway and F. P. Kelly, “A standard attenuator and the precise measurement of attenuation”, *IEEE Trans. Inst. Meas.*, pp. 33 - 44, March, 1964.

ACKNOWLEDGEMENT

I would like to thank my teacher Prof. P. C. Pandey for all the help provided by him during my stay at IIT. I am very fortunate to get his valuable guidance for the project work. The technical discussions I had with him improved my understanding of engineering. I also want to give many thanks to Mr. Nandakumar Pai for all the valuable time, help, suggestions and support he gave me for this project and particularly in desiging the PCB. I thank Mr. Vidyadhar Kamble for all help and support in the lab**.** I would like to thank Mr. Dinesh Choudhary for his valuable advices particularly on PCB layouts. I thank all my friends especially from the SPI Lab.

I dedicate my work to my mother Fatima Abkabr and to Ato Araya Zerihun.

Siraj F. Mossa

29th June 2010