A MICROCONTROLLER BASED AUDIOMETER

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by

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Guide

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ABSTRACT

The degree of hearing loss can be determined by audiometry, which measures a person’s hearing sensitivity. The instrument developed for carrying out such audiometric tests is known as audiometer. Using it, the test tones of different frequency and level are presented and hearing thresholds are determined on the basis of patient’s response. A microcontroller based audiometer has been under development at IIT Bombay since 1997. The objective of the project is to study the work done so far and build a fully functional instrument by overcoming shortcomings in earlier prototype. A microcontroller based pure tone diagnostic audiometer is developed which operates over full frequency range (250 Hz to 8 kHz) and acoustic output level of 0 to 100 dB HL. It can also generate warble tone having ±10% frequency deviation, and amplitude modulated tone with ± 5 dB modulation. It has a facility for speech audiometry. The instrument provides a broadband/narrow-band masking noise, with level selection. Facility of air and bone conduction is provided. All the controls are through a 4 × 4 membrane keypad and indications are using 16 characters × 2 lines LCD display. The instrument is menu driven and has option of manual and automated audiometry. At power on, it carries out a self test of the output levels. It has RS232 interface for downloading the test results to a computer.

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LIST OF ABBREVIATIONS

Abbreviation	**Term**
ac	alternating current
dc	direct current
dB	decibel
dBm	dB power level with ref. power = 1 mW (1 mW power in 600 ohm, Vref = 774 mV (rms))
dB SPL	Sound pressure level in dB with pressure reference = 20 μPa
dB HL	sound pressure level in dB with pressure reference as average hearing threshold for the young adult population for the test frequency
SC	switched capacitor
SCF	switched-capacitor filter
SISI	short increment sensitivity index
TDT	tone decay test
SRT	speech reception threshold
SDS	speech discrimination score
BPF	band pass filter
WB	wide band
NB	narrow band
DAC	digital-to-analog converter
LCD	liquid crystal display
PCB	printed circuit board
PTH	plated through holes
EFD	electro fluorescent display

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Chapter 1
INTRODUCTION

The method by which patient's hearing sensitivity can be determined is termed as audiometry [1][2]. It helps in assessing the nature, degree, and probable cause of the hearing impairment. In this technique, auditory stimuli with varying intensity levels are presented to the person who responds to these stimuli. The minimum intensity level of these stimuli to which consistent responses are obtained is taken as the “threshold of hearing”. Depending on this threshold, the patient’s hearing sensitivity can be estimated and best-suited medical treatment or hearing aid or other assistive devices can be prescribed. There are different audiometric procedures depending on the stimuli used. An audiometer is an instrument, which is used for carrying out these audiometric tests.

Audiometer can be of different types, depending upon the frequency range, range of acoustic output, mode of acoustic presentation, masking facility, and types of acoustic stimuli. Chandrakant Singh [3], Ashish Kothari [4] and Pratibha Reddy [5], as part of their M. Tech. dissertations at IIT Bombay, worked towards developing a microcontroller based pure tone audiometer with a provision for automated audiometry and computer/printer interface.

The objective of the project was to study the earlier developments thoroughly and develop a fully functional diagnostic audiometer based on same approach and by overcoming the shortcoming of the earlier prototype. Speech audiometry facility is also to be provided in the prototype.

In the second chapter, various audiometric techniques, details of pure tone audiometer, and need for an automated pure tone audiometer have been discussed. In the third chapter, design approach of the system and work done by Pratibha Reddy is described. Fourth chapter provides system hardware description. In the fifth chapter, software description is given. PCB design, system assembly, and test results are discussed in the sixth chapter. Summery and further work needed is discussed in the last chapter.

Chapter 2
AUDIOMETERY

Audiometry is a technique for identification and quantitative determination of hearing impairment. Before medical treatment can proceed, the nature and degree of hearing impairment should be assessed in order to quantify the hearing impairment and diagnose its causes so that suitable treatment or one of the appropriate hearing aids and assistive devices can be prescribed. This chapter provides a description of the various audiometric tests and procedures, and masking in audiometric tests, pure tone audiometers, and PC/microprocessor based pure tone audiometers.

2.1 Types of Audiometry

There are two types of audiometric techniques [2], subjective type and objective type. In subjective test, the patient has to respond when he hears the presented sound. Objective test only requires co-operation from the patient towards attachment of the measuring electrodes or probes [6]. Subjective type audiometric test involves presentation of systematically varying acoustic stimuli to the subject and recording the responses. There are different audiometeric procedures depending on the stimuli used. In pure tone audiometry, the subject's threshold for hearing is measured. In speech audiometery, the subject's threshold for the reception of speech is recorded. Different audiometric techniques are described into the following subsections.

2.1.1 Pure Tone Audiometry

Pure tone audiometry is a procedure for determination of the extent of hearing loss and the cause, i.e. conduction or sensorineural loss [2]. The subject's response for acoustic stimuli of different frequencies are measured. The initial level of the stimuli is selected by the audiologist. Although human hearing ranges from 20 Hz to 20 kHz, there is little speech information above 8000 Hz, and perception of frequencies below 100 Hz is increasingly tactile in nature, making them difficult to assess [7]. Also, the loss of hearing sensitivity is observed first at high frequency (8 kHz) and later on as the loss progresses, its effect is observed in the mid-frequency region (1-2 kHz) as well. By the time the loss is observed in the low frequency region, the subject will near to deafness. Hence, audiometric tests carried out in the low frequency region do not give any significant information about hearing loss. Therefore, audiologists routinely test only in the range of 250-8000 Hz, often in octave steps.

In acoustic measurements, sound level is often given in dB, taking sound pressure of 20 μPa as the reference level, and is known as sound pressure level (SPL). However, in audiometry the sound level of pure tones is given in dB by taking average hearing threshold of normal hearing young adults as the reference, and is known as hearing level (HL). The hearing threshold is frequency dependent, and hence SPL corresponding to a given HL varies with frequency. Since both HL and SPL are logarithmic units, a certain increment in HL corresponds to the same value increment in SPL also [1]. The following table gives the dB SPL values corresponding to 0 dB HL for standard frequencies

Frequency (Hz)	250	500	1k	1k5	2k	3k	4k	6k	8k
Sound level (dB SPL)	25.5	11.5	7	6.5	9	10	9	10.5	13

In this technique, single frequency stimulus at some presumed level is presented to the patient. The stimulus intensity is decreased in 10 dB steps until the subject no longer responds. After this point the level is increased or decreased in 5 dB steps until 50% response criterion has been met, i.e. subject responds to half of the tones presented to him/her. The minimum presentation level at which the subject responds at least 50% times, is taken as the hearing threshold [1][7]. The results of the audiometry are reported as an audiogram which is a plot of threshold intensity versus frequency. Different shapes of audiograms are associated with different types of hearing loss [1].

2.1.2 Tone Decay Test (TDT)

In this test, we try to quantify the deterioration in the auditory nerve. Here, a tone of particular frequency with threshold intensity is presented as a continuous tone and the time for which the subject is able to hear the tone is recorded. If the subject is not able to hear the tone continuously for more than one minute, the intensity is incremented by 5 dB and again tested for the same. The lowest intensity for which patient is able to hear the tone for about 1 min. is considered as threshold for tone decay test. Tone decay test is used to diagnose the sensorineural deafness [1].

2.1.3 SISI Test

The SISI (short increment sensitivity index) test is used to detect the pathology in cochlear or retrocochlear lesions [1]. This test determines the capacity of a patient to detect a brief 1 dB increment in intensity, provided at 5 seconds interval. The patient is asked to respond by using response switch if he senses the increments. The number of increments the patient is able to recognize correctly is noted. A SISI audiogram is plotted on the basis of percent score for each of the test frequencies.

2.1.4 Speech Audiometry

While pure tone threshold testing attempts to assess sensitivity, speech audiometry testing attempts to address the integrity of the entire auditory system by assessing the ability to here clearly and to understand speech communication [7]. The main use of speech audiometry is in the identification of neural types of hearing loss, in which both the reception as well as the discrimination of speech is impaired more markedly than in cochlear or conductive hearing loss. There are two types of speech audiometric tests, “speech discrimination test” and “speech reception threshold test”.

In speech discrimination test, lists of monosyllable speech discrimination words are presented over earphones for each ear which patient is asked to repeat. The percentage of the total number of words presented which the patient is able to identify correctly gives the speech discrimination score (SDS). The SDS is determined when the patient repeats 50% of the words correctly. The result of this test is from 0 to 100 %. Generally, a high score is associated with normal hearing or conductive hearing loss and low score is associated with sensorineural loss.

The speech reception threshold test is similar to the speech discrimination test except for the fact that this test uses two syllable words with equal stress (spondees) and the words are attenuated successively. The SRT (speech reception threshold) is the lowest hearing level in dBHL at which 50 % of a list spondee words are correctly identified by a subject. For estimating SRT, a group of 6 spondee words is presented at 25 dB above the average pure tone audiometry threshold for 500 Hz and 1000 Hz, and then at successively lower intensities. When the level is such that the subject is able to identify 3 words out of 6 correctly, the level is taken as SRT. The SRT of a normal subject is very closely related to his pure tone hearing threshold and the SRT is generally 2 dB lower than average of pure tone hearing level thresholds at 500 Hz and 1 kHz.

2.2 Masking in Audiometry

In audiometry, both ears are tested separately. In air and bone conduction audiometry where sound is applied to one ear, the contra lateral cochlea is also stimulated by transmission through the bone of the skull. In case the sound in one ear is sufficient to stimulate the second ear, it is called cross hearing. During the air conduction test, the tone while passing from test ear to cochlea of the non-test ear gets attenuated. This loss of sound energy is called interaural attenuation and varies between 45 to 80 dB [1]. However, during bone conduction test, the cochleae of both sides are equally stimulated i.e. the inter-aural attenuation is of 0 dB. Hence, cross hearing is a serious concern in case of bone conduction test than it is for air conduction. Whenever cross hearing is suspected, it is necessary to remove the non-test ear from procedure [2].

A simple procedure by which this can be done is to deliver a noise to the non-test ear in order to remove it from the test procedure by masking. Here masking noise which is loud enough to prevent the tone reaching and stimulating the non-test ear, but at the same time it should not mask the sensitivity of the test ear [1]. Thus, an audiologist should provide appropriate level of masking. The masking noise is often selected to be a wide-band noise, or narrow band noise with the band centered about the test frequency. The wide-band noise has uniform power density spectrum over all the audible frequency range. However the masking effect is actually contributed by frequency components centered on the test tone frequency, over a bandwidth of about 1/3 to 1/2 octave, known as critical band. Broadband noise bandpass filtered with a band approximately corresponding to the critical band is known as narrow band noise, and compared to wide band noise it gives the same masking effect at a lower sound pressure level.

2.3 Pure Tone Audiometer

The instrument used to carry out pure tone audiometric test is known as pure tone audiometer. A pure tone audiometer consists of three major parts: (a) the signal source, (b) the system for control of signals, and (c) the means of presenting the acoustic or vibratory test signal to the patient. The instrument may be provided with a calibrated noise source for masking the hearing in the non-test ear, and a bone conduction vibrator.

A block diagram of conventional audiometer is shown in Fig. 2.1. It consists of tone generator, noise generator, attenuator, equalization circuit, and power amplifier. The tone frequency should range from 250 Hz to 8 kHz. Each of the frequency should be within 3% of the indicated frequency. Also the output level of any harmonic should be at least 30 dB below the fundamental level [8].

Equalization circuit provides frequency dependent attenuation in order to calibrate the output sound level in dB HL for particular output devices used (headphone, loudspeaker, and vibrator). This is necessitated by frequency dependence of the reference used for dB HL scale, and also due to the frequency response of the electroacoustic device used for presenting sound (relationship between sound level output by the device and the voltage input applied to it). The attenuator, known as the as hearing level control, should be capable of controlling the output sound pressure level over a desired range in steps of 5 dB. Calibration should ensure the output sound level to be within $\pm$ 3 dB of the indicated value.

The noise generator should provide wide-band noise, which has energy spectrum equally distributed over the test frequency range i.e. up to 8 kHz. The amplitude of the noise level should be within $\pm$ 5 dB of the indicated value [5].

The output power available from the power amplifier determines the maximum sound pressure level available from the headphones and the bone vibrator. The amplifier must have low distortion and a good S/N ratio to meet the standard requirements. The audiometer should have some means of switching of the signal from one earphone to another and also to the bone vibrator.

2.4 Microprocessor/PC Based Audiometer

A conventional audiometer instrument has dials or knobs with calibrated scale for frequency selection and for tone masking noise level selection. The variation of the level of the stimulus is done manually by the audiologist after carefully observing the responses of the subject. An interrupter switch is used for tone switching and needs to be mechanically silent. The presence of mechanical parts makes the instrument more susceptible to wear and tear. Calibration is necessary, at least, once in six months.

The advancement in technology has made the various switching tasks simple, flexible, and noise free. Also the procedure can be automated. Application of microprocessor/PC in audiology offers many advantages in terms of flexibility and simplicity of use, over their conventional counterparts. In this type of audiometers, the knobs and switches are replaced by keypad. Calibrated scales and other indicators are replaced by display to show the various parameters and modes, and operation status. Microprocessor/PC based audiometry also offers automation of audiometric testing. It is possible to store test results and to print audiogram. Increased accuracy and precision removes the need for frequent calibration of audiometer, which was required for earlier audiometers.

The main task of microprocessor based audiometer is to generate tones of different frequencies. The tone generation in microprocessor/PC based audiometers can be achieved by any of the following means.

(1) The frequency of the tone generator is selected by switching R or C values through analog switches. Attenuator is a multiplying D/A converter (resistor network, opamps and analog switches). All the controls are through the digital outputs from a microprocessor/microcontroller with keypad/rotary switch interface and digital display. Alternatively, the digital controls are provided through a controller and interface to PC bus, enhanced printer port, or serial port of a PC.

(2) The tone is generated through software and output thorough the D/A converter of a PC based signal acquisition card, and a digitally controlled attenuator built with resistor network and analog switches, and controlled by the digital I/O lines of the signal acquisition card. Sound card of the PC can also be used to generate tones, and attenuator can be controlled through serial or parallel port.

(3) The generation of stimulus tone and masker is achieved by dedicated DSP chip(s) and D/A converters working under the control of microprocessor, with rest of the block as in (1) or (2).

Microprocessor/microcontroller based audiometer can be interfaced to the multimedia facility of the PC and include speech audiometry facility with some additional hardware changes in the pure tone audiometer.

Above discussed microprocessor/PC based audiometers often also incorporate the facility of automated audiometry, in which the output level of test stimuli is selected in accordance with the subject response and the threshold levels are determined.

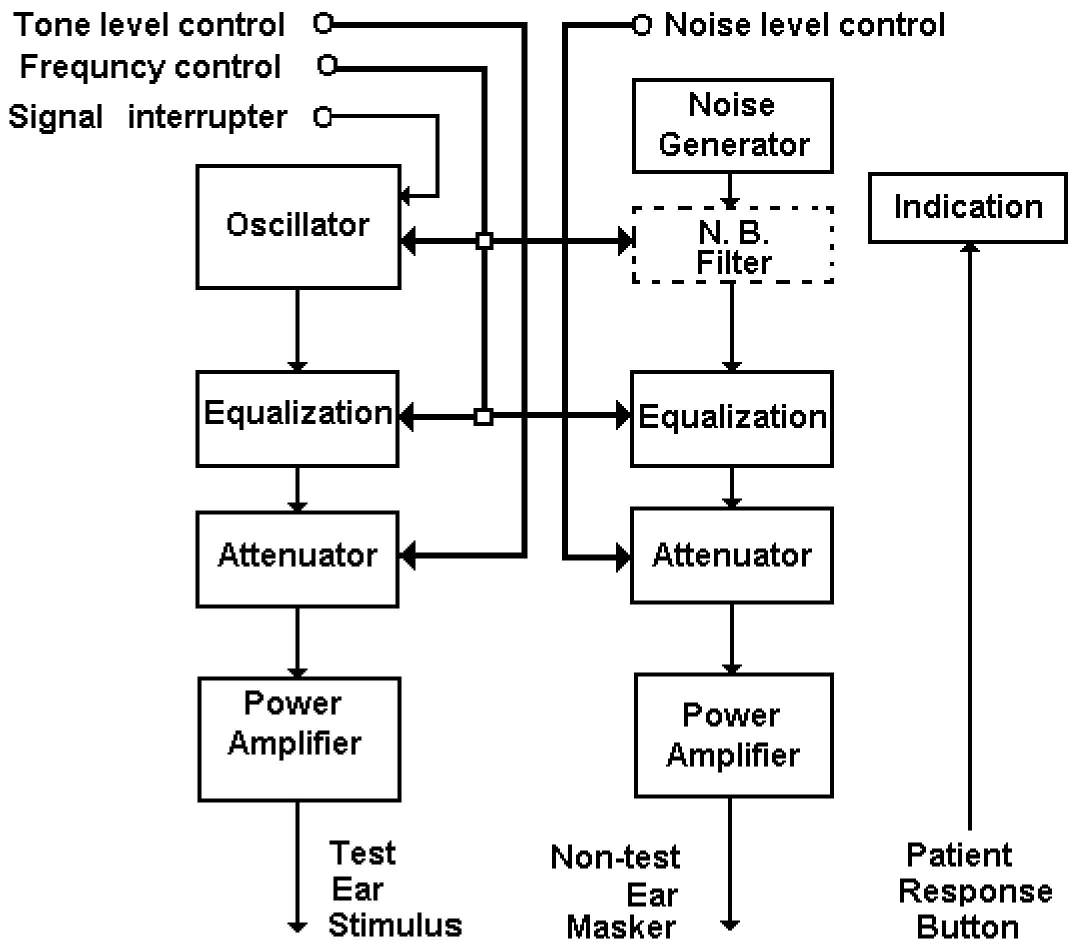


Fig. 2.1 Block diagram of a pure-tone audiometer

Chapter 3
DESIGN APPROACH

The objective of this project was to develop a portable and microprocessor based audiometer that is well suited for a variety of uses and can be used widely even in rural areas. The design has been carried out with the following considerations:

(1) The instrument should portable, so that it can be used in mobile clinics and in rural clinics.

(2) Convenient and easy calibration should be possible without opening the unit (possibility of software calibration through a keypad).

(3) It should not have mechanical parts that may require servicing and maintenance.

(4) Required frequencies and sound levels should be easily selectable, option of computer / printer interface.

(5) There should be a provision for storing the audiometric test results.

(6) The instrument should have features of a diagnostic audiometer, with a provision for automated audiometry.

(7) The design should use standard components, and the number of components should be kept low.

Keeping these considerations in mind the system specifications are finalized and are listed in appendix A. Pratibha Reddy worked (as part of her M. Tech. dissertation project at IIT Bombay) towards developing a microcontroller based portable diagnostic audiometer having these specifications [5]. However, all design objectives could not be met in the prototype developed. In this chapter worked done by Pratibha [5] is summarized and the design philosophy of the audiometer is described.

3.1 Earlier Development

Pratibha Reddy [5] has developed a prototype of a microcontroller based pure tone diagnostic audiometer, which operates over a frequency range of 250 Hz to 8 kHz and acoustic output of 0 dB HL to 100 dB HL. It can also generate warble tone having $\pm$10% frequency deviation. It has flexibility in selecting required frequencies and sound levels. It has two channels, one for pure tone and other for masking noise.

Full masking facility is provided, which includes both types of noise, wide-band and narrow-band. Facility of air and bone conduction is also provided.

The instrument is menu driven with option of manual as well as automatic mode. This was made possible by using microcontroller, programmable oscillator, and programmable attenuator. A programmable oscillator was designed using a switched capacitor filter. Frequency of sinusoidal oscillation is proportional to the digital clock input, which is generated by microcontroller. Digitally controlled attenuation in dB scale was obtained by using a logarithmic D/A converter. Wide-band noise is generated by low pass filtering of the output of a pseudo random binary sequence generator and is shaped to a narrow-band noise by a SCF based narrow band filter. The instrument has a serial port interface for transferring the test results to computer or printer. After power on, the instrument carries out a diagnostic test for the tone and noise level as controlled by attenuators.

The instrument has facilities of other audiometric tests like SISI (short increament sensitivity index) and TDT (tone decay test). Also there is a facility for generating amplitude modulated tone and warble tone which are also useful in audiological testing.

The scheme used by Reddy has been critically reviewed, and the circuit blocks have been thoroughly tested to establish the areas for further work, in order to develop a compact instrument. The following paragraphs summarize the design philosophy of each block in the system, as shown in Fig. 3.1.

3.2 Design Approach

A block diagram of the system is shown in Fig 3.1. For overall control of the instrument there is a microcontroller which control digitally all the operations of the instrument. For a very compact design, this microcontroller should have sufficient programmable ROM, and data RAM, parallel I/O ports, a serial port, and a programmable timer/counter in order to handle all the operations, without requiring additional chips. MCS-51 family of 8-bit microcontroller meets most of these requirements. It was decided to use Atmel AT89C52 microcontroller, having 8K bytes flash (electrically erasable) EPROM and 256×8-bit internal RAM. Its most important feature, of particular concern in this design, is that it can output 50 % duty cycle programmable clock as a background operation [9].

For generation of different frequency tones, a programmable oscillator is designed using a switched capacitor filter (SCF) which requires only one clock frequency as control input. The frequency of sinusoidal tone in this circuit is a fraction of the clock frequency. The clock frequency for the oscillator circuit is derived from the crystal clock by using the clock generation mode of the programmable timer/counter of the microcontroller. The generated tone frequency is highly stable and will not require calibration. Warble tone i.e. frequency modulated tone can be generated by introducing frequency modulation in the clock frequency to the oscillator. Alternatively, the sine wave can also be generated by software, and output using a DAC. But then processor will be busy all the time in generating stimulus, tone will be discontinued while executing other interrupt routines. So use of SCF based oscillator was considered to be suitable for this application.

Level control for generated tones is required for various tone levels. Hence, attenuator must be capable of adjusting output sound pressure level from below the threshold of hearing to some 100 dB above, and normally in steps of 5-dB [2]. For attenuation control 8-bit programmable monolithic logarithmic D/A converter AD7111 from Analog Devices was found suitable. It gives attenuation of 88.5 dB, with a resolution of 0.375 dB. Since, the attenuation range needed for audiometer was more than 88.5 dB; so one more attenuator chip is connected at the output of first attenuator chip. Amplitude modulated tone can easily be generated using this attenuator by software control.

In early audiometers, a low frequency saw tooth waveforms was used as a masker, which was effective at low frequency, but not at high frequency. Another drawback was the harmonic beating with the test tone frequency [2]. Diodes can also be used as a noise source. But, it needs high gain amplifier, which may result in 50 Hz power line peak up [7]. All these techniques had a possibility of generating uneven spectrum of masking noise. However, digital noise generators generate noise of known spectrum and amplitude, with adjustable bandwidth. Hence it was decided to use digital white noise generators based on the pseudo random binary sequence (PRBS) generator. The low pass filtering of PRBS output gives band limited white Gaussian noise, and 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 same masking effect as wide-band but at lower sound pressure levels.

A response switch is given to the patient, to indicate whether the tone is heard or not. While applying tone level to the patient, microcontroller continuously checks the response switch for the patient response. The closure or non-closure of response switch is indicated on the display, depending upon which the audiologist decides the next tone level. One LED is interfaced with the same circuit for better indication of subject response. In auto mode, the instrument itself presents the stimulus, and on the basis of the response it decides the level of next stimulus, and by following the procedure described earlier in subsection 2.1.1, the hearing threshold for each frequency is determined. These test results are stored in the data memory of microcontroller and these may be transferred to a printer or a computer through serial port.

The user interface is through a 4×4 keypad and 2 line×16 characters LCD display. The 4×4 keypad is interfaced directly to one of the I/O ports of the microcontroller. All the indications are through the LCD display, which is a commercially available unit, assembled on 8 cm × 3.5 cm size PCB with on-board controller. It requires 8 data lines and 3 control lines, which are interfaced to the microcontroller.

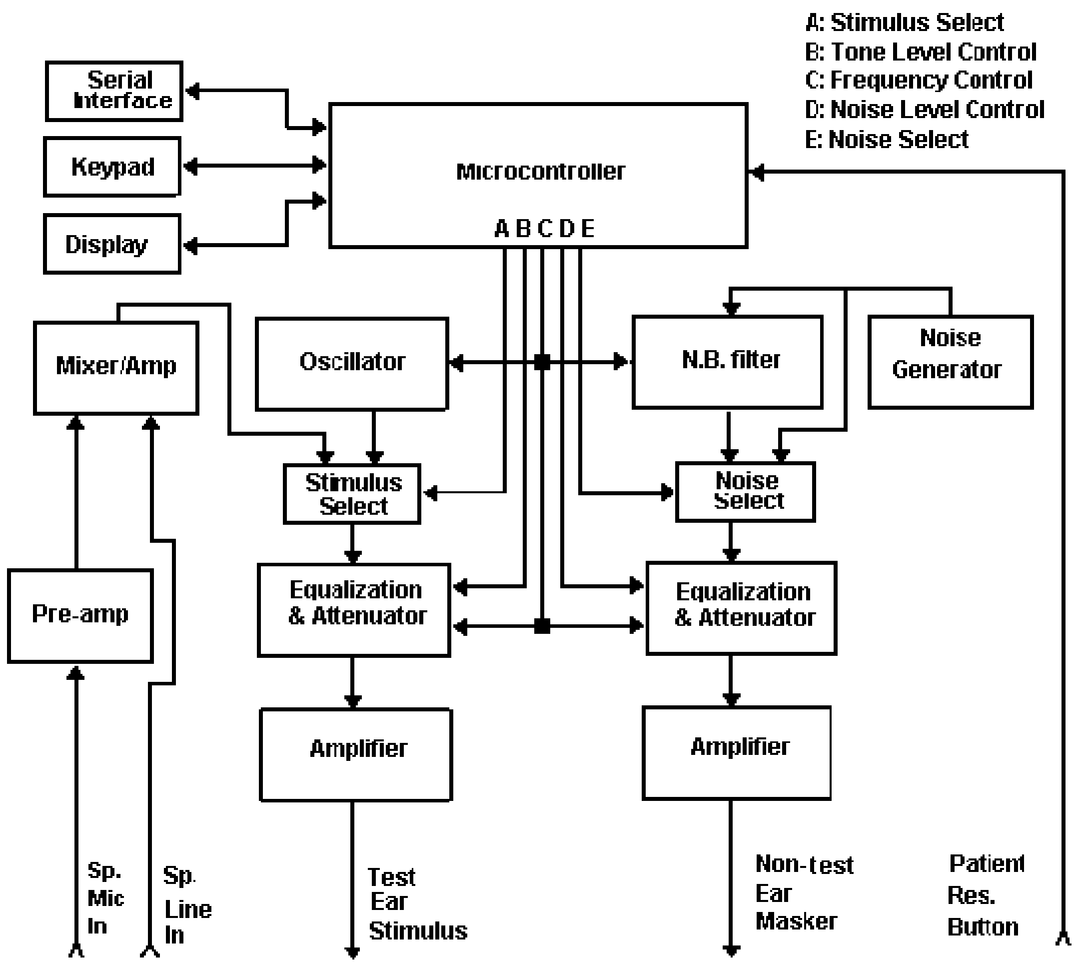


Fig. 3.1 Block diagram of microprocessor based audiometer

Chapter 4
HARDWARE DESCRIPTION

Chandrakant Singh [3], Ashish Kothari [4] and Pratibha Reddy [5], as part of their M.Tech. dissertations at IIT Bombay, worked towards developing a microcontroller based pure tone audiometer with design approach discussed in the last section of Chapter 3. The same design approach has been used in this project, with certain modifications to overcome existing shortcomings. This chapter provides a description of the complete hardware (including the blocks unmodified from previous designs).

The block diagram of the hardware of the audiometer is given in Fig. 4.1. All the controlling operations are handled by the microcontroller. The tone is generated by SCF based oscillator, which needs only one clock frequency as control input. Here the clock is provided by the microcontroller. The generated tone is attenuated by logarithmic attenuator IC AD7111. Since one 8-bit attenuator IC AD7111 gives maximum attenuation up to 88.5 dB, two chips are cascaded to get the required attenuation. Same attenuator circuit is used for attenuation of speech signal by additional switching arrangement. The output of attenuator is fed to the two power amplifiers, one for headphone and other for the bone vibrator. White noise is generated digitally, by low pass filtering the pseudo random binary sequence. The white noise is further band pass filtered to get narrow band noise. The band pass filter is realized by using SCF IC. Either of the two wide band or narrow band noise is selected by switch. The output of the power amplifiers for the tone/speech and for noise is verified by means of the self-test circuit. The response switch with LED flash interface is provided to the patient to communicate the response to the instrument. The operator interface is through a 4×4 keypad and 2 lines ×16 characters LCD display. Serial interface is used to download test results to the computer.

This chapter gives brief description of each block of the overall hardware of Fig. 4.1. The last section describes the interfacing of microcontroller to these blocks. Also, the hardware modification done so far is described in this chapter.

4.1 Oscillator

The main task of a microcontroller-based audiometer is to generate the tones of different frequency. The generation of tones of different frequency can be possible by software means and the output using a DAC. But then, the processor will be busy all the time in generating stimulus, tone will be discontinued while executing other interrupt service routines. The oscillator realization is implemented using switched capacitor (SC) filter IC. The switched capacitor filter (SCF) is based on the principle that a capacitor switched between two nodes at a sufficiently high rate is equivalent to a resistor connected between these two nodes. Thus, the time constant of an RC circuit block and hence the transfer function of a circuit realized using such RC circuit blocks depends upon the switching frequency and capacitor ratio not on the absolute value of the capacitor [10]. Switched capacitor filter (SCF) based oscillator requires only clock frequency as control input. Tone frequency can be easily changed by changing the clock frequency.

The basic principle of the tone oscillator is shown in Fig. 4.2. The circuit consists of a high-Q band pass filter connected in a positive feedback loop with a hard limiter. The output of the band pass filter will be a sine wave whose frequency is equal to the center frequency of the filter. The sine wave signal is fed to the limiter, which produces a square wave output with same frequency and amplitude determined by the limiting levels. This square wave is fed back to the band pass filter, which filters out all the harmonic components and provides sinusoidal output at the fundamental frequency. The amplitude of sine wave depends on the limiting levels of the square wave and the gain of the BPF. Higher the Q of BPF (selectivity) better the purity of the output sine wave [11]. Band pass filter uses two-integrator loop as shown in the Fig. 4.2. It consists of two integrators and a summer. The output of first integrator is BPF and that of the second integrator is LPF. These two outputs are 90-degree phase apart. Transfer function of the band pass filter is

$$H_{bp}(s) = \frac{V_{bp}}{V_i} = \frac{\alpha s/\omega_o}{(s/\omega_o)^2 + (1/Q)s/\omega_o + 1}$$

If V_i is a square wave of $2V_m$ peak-to-peak voltage with frequency f_o, it has fundamental component of $(4/\pi)V_m$ peak to peak. Output V_{bp} is a sinusoid with frequency f_o and its peak-to-peak voltage is given as

$$V_{bp} = (\alpha Q)\,(4/\pi)V_m$$

The SCF IC used here is LMF 100 which has high Q and its f_o*Q = 1.8 MHz. And also the output frequency is derived from input clock frequency as $f_o = k * f_{clk}$ here k can be set as 1/100 or 1/50. The variable clock frequency given to SCF is obtained using timer 2 of microcontroller. For a microcotroller clock frequency of 24 MHz, pure tones with frequencies 250 Hz, 500 Hz, 1 kHz, 1.5 kHz, 2 kHz, 3 kHz, 4 kHz, 6 kHz, and 8 kHz are generated. The IC LMF 100 has two identical structures consisting of two integrators (SC based) and one summer (SC based) and an op-amp.

The band pass filter implemented using LMF 100 is shown in Fig. 4.5. The filter transfer function is implemented using section B of the IC LMF100 (U15). Section A is used for realizing narrow band filter for noise, as described later in Section 4.4. The circuit provides two outputs of equal amplitude but 90º phase apart. Hence, it is also known as quadrature oscillator [12]. The BPF output rings at its resonance frequency in response to a step input change. The oscillation loop is sustained by an inverting Schmitt trigger, formed using an op-amp U11 (LF 356). A hysteresis of approximately 40 mV is provided to avoid jitter at the output. The output of the Schmitt trigger is stabilized by using 2.1 V zener diodes (D2, D3) connected back to back. This ensures stability of the BPF output. Peak-to-peak voltage of the square wave output V2 is

$$V_{2p\text{-}p} = V_{D2} + V_{D3} + V_{Z2} + V_{Z3}$$

where V_{D3}, V_{D4} are forward voltage drops of diode D3, D4 respectively and V_{Z3}, V_{Z4} are the zener voltage drop of diode D3, D4 respectively. Comparing the block diagram of Fig. 4.3 and circuit schematic of Fig. 4.5, we have

$$\alpha = - R_8 / R_9$$

$$Q = R_7 / R_8$$

$$H_{bp} = \alpha Q = - R_7 / R_9$$

where Q is the selectivity of the filter and hence determines the spectral purity. The peak-to-peak amplitude of sine output V_1 is given as

$$V_{1p\text{-}p} = (4/\pi)\, V_{2p\text{-}p}\, H_{bp}$$

It is desirable to increase Q so as to increase the selectivity of the BPF, but it should not result in clipping at the output. The resistor values selected are R_7 = 47 kΩ, R_8 = 1 kΩ, R_9 = 150 kΩ and thus result in

$$Q = 47,\ \alpha = - 1/150,\ H_{bp} = 0.313$$

with $V_{2P\text{-}P}$ = 5.8 V, we get $V_{1p\text{-}p}$ = 2.31 V.

Spectral purity of test tone for all frequencies was observed by using spectrum analyzer. It was found that the total harmonic content is around –50 dB below the fundamental level, which is well within the limits specified by standards.

BPF output is fundamental frequency component of the square wave input. The frequency of oscillation is

$$f_o = k\, f_{CLK}$$

where k = 1/50 or 1/100, depending on the control input at pin 12 of U15, which selects, the clock mode of LMF 100, either to 1/50 or to 1/100. It is to be noted that as the tone generation has been realized using SCF, the output waveform is made of steps at clock frequency, and thus the output tone has a certain component of the clock frequency. The clock frequency is obtained as the one of the output of microcontroller pins, as a division of the crystal clock frequency by a 16-bit integer (stored in an internal register). For 24 MHz crystal, 100:1 and 50:1 makes result in frequency resolution 250 and 500 Hz. As per 50:1 mode of LMF 100, for 250 Hz tone, clock frequency is 12.5 kHz, which falls within the audible range. To avoid this, the SCF is operated into 100:1 mode for lower frequencies, up to 1500 Hz. Thus, shifting the dominant component for 250 Hz to 25kHz, which is outside the audible range. If 100:1 mode is used at higher frequencies, the frequency resolution decreases. Hence for higher frequencies, 50:1 mode is used. The sensitivity of the output transducer (headphone, speaker, bone vibrator) reduces drastically for frequencies higher than 20 kHz. Thus, for high frequency tones the clock frequency is outside the range of reproduction by the headphone and also outside the range of audibility. The frequency resolution for frequency below 1500 Hz is about 250 Hz while for the high frequencies it becomes 500 Hz.

The square wave output V2' of the Schmitt trigger is used for generating interrupts for warble tone generation. The bipolar signal V2' is converted to unipolar output, labeled P3.3, using R5 and D1 (as shown in Fig. 4.5). The output P3.3 is connected to microcontroller.

4.2 Attenuator

Attenuator circuits are used for level control for tone generated by the oscillator circuit. In order to determine the voltage levels to be generated to produce the audiometric range of sound pressure levels, the TDH-39 headphone was calibrated using B&K artificial ear (type 4153). Spring pressure on the headphone was kept at 0.5 kg. The voltage level (in dBm) required for producing 100 dB SPL at different frequencies were noted. The voltage levels (in dBm) required for producing dB HL_{max} and 0 dB HL were calculated. Maximum dBm (–0.98 dBm) is required to produce 90 dB HL at 250 Hz and minimum dBm (–107 dBm) is required to produce 0 dB HL at 1000 Hz. Hence, the attenuator should have a dynamic range of at least 107 dB.

It was decided to design the circuit in such a way that the maximum sinusoidal output is 0.774 $V_{r.m.s.}$, or 2.19 V_{p-p}. This is the voltage corresponding to 0 dBm. Sensitivity of the headphone is a function of frequency and may vary (within a small range, for a given type of headphone) from piece to piece. Hence, calibration of headphones before testing is a must. The sound pressure level corresponding to the threshold of hearing varies with frequency and, if the zero on the audiometer is to be valid for each test frequency, the output sound pressure must be varied for each frequency, and this is taken care of in the software for the attenuator control.

The attenuator for tone has been realized using two logarithmic D/A converter AD 7111 from Analog Devices. Each attenuator chip gives attenuation of 88.5 dB, with a resolution of 0.375 dB [12]. Internally AD 7111 consists of 17-bit R-2R ladder network based multiplying D/A converter. An on-chip logic circuit converts 8-bit input into 17-bit data, which is used to drive the D/A converter. The 8-bit input is latched into the internal latch by $\overline{CS}$ and $\overline{WR}$ control signals. Analog input is attenuated according to the data latched. The attenuator circuit for tone is shown in the Fig. 4.6a. The current output of both chips AD 7111 is converted to voltage by using op-amp U4 and U6 (LM 356) respectively. Resistors R4 and R12 are the gain trim resistors, used to set output equal to input when there is 0-dB attenuation. In the present design, metal film resistors (MFR) are used instead of trim resistors. Capacitors C15 and C30 are used for phase compensation. It is to be noted that the IC AD 7111 needs only a single 5V supply, and it handles bipolar signals.

One more attenuator chip IC AD7111 is used for attenuation of masking noise, as explained in the next section.

4.3 Attenuator Selector for Pure Tone and Speech Stimuli

For speech audiometry facility test stimuli generated from audio amplifier circuit is fed through the same attenuator circuit, which is used for pure tone. The circuit diagram for this scheme is as shown in Fig. 4.6b. Here four analog switches in IC U16 (CD 4066) are used for selection of test signal. Control signals of U16a, U16d are complementary to control signals of U16b, U16c. U10d (CD 4030) is used as an inverter to get complementary signal. If pure tone signal stimuli are to be selected, the control labeled P3.1 is made high. Thus, making U16c, U16b on and U16a, U16d off. If speech signal stimuli is to be selected then P3.1 is made low. Here pin P3.1 of microcontroller TXD (transmit pin) is used as a control pin.

4.4 Masking Noise Generator

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]. The band of noise will be flat up to 12 % of the clock frequency driving the shift register. Further the Gaussian noise is passed through a narrow band pass filter to get narrow band noise. In earlier design, a 15-bit PRBS generator based on the maximal length feedback shift-register was used, with a clock of 200 kHz. It gives repetition rate of $(200000/ 2^{15} - 1) \approx 6$ Hz. The noise had a perceivable repeatability, which is not desirable. The effect of variation in bit length on perceived repeatability was studied by software implementation of the PRBS generator. The results indicated that with a length of 23 bits, the repeatability was not perceivable. So an additional chip of 8-bit shift register has been added to the existing 15-bit register sequence length. The output of the PRBS repeats after every $2^{23}-1$ clock pulses. The clock frequency used is 200 kHz. Thus, we get a noise with flat spectrum in the range of $f_1 - f_2$ [13], where

$$f_1 \approx 10\, f_{CLK} / (2^{23}-1) \approx 0.2 \text{ Hz}$$

$$f_2 \approx 0.12\, f_{CLK} = 24 \text{ kHz}.$$

The circuit is shown in Fig. 4.8. Three dual 4-bit shift registers IC CD 4015 (U12, U14 and U23) and one quad XOR gate IC CD 4030 (U13a and U13b) is used to make the PRBS generator. The tapping is made at 18th and 23rd bit and output is taken from 23rd bit ($Q3_B$ of U23) thus making it a 23 bit PRBS. At the time of power on, the output of shift registers is undetermined. Hence, a power-on reset (R24 and C24) is provided to ensure that all outputs are zero. If XOR feedback is provided, the output states will be again zero. Hence, it was decided to put XNOR in feedback loop. U13B is an XOR gate acting as an inverter to provide XNOR operation in the feedback path. The clock input to the PRBS generator circuit is provided by the astable pulse generator U9 (IC7555). Pulling the control input to U9, marked P3.7, low stops the clock.

To get wide band white noise, a second order low pass filter built using op-amp U14 with a 10 kHz cutoff frequency and unity gain is used [14]. The dc blocking capacitor C31 is used to eliminate the dc offset present at the output of PRBS generator. The transfer function of the second order low pass filter is

$$H_{lp}(s) = \frac{V_{lp}}{V_i} = \frac{A_o}{(s/\omega_o)^2 + \sqrt{2}(s/\omega_o) + 1}$$

The cutoff frequency is

$$f_o = 1 / (2\pi \sqrt{R_{25}\, R_{26}\, C_{30}\, C_{26}}).$$

By selecting R_{25} = 22 k, R_{26} = 12 k, R_{27} = 22 k, C_{26} = 470 pF, C_{30} = 2.2 nF, we get $f_o \approx$ 10 kHz and dc gain A_o is unity.

To get the narrow band noise with a center frequency same as that of the test tone, section A of the SCF IC LMF-100 (U15) is used for realizing band pass filter and the clock to this block is the same as that for the oscillator. The bandwidth requirement is in between one-third to one-half octave of the center frequency. The transfer function of band pass filter is

$$H_{bp}(s) = \frac{V_{bp}}{V_i} = \frac{s/\omega_o}{(s/\omega_o)^2 + (1/Q)s/\omega_o + 1}$$

For this circuit the center frequency f_o, selectivity Q, and bandwidth Δf are given as

$$f_o = f_{clk}\, k\,, \qquad \Delta f = f_o / Q, \qquad Q = R_{21}/R_{23}$$

where k = 1/50 or 1/100. With R_{21} = 3.3 k and R_{22} = 1.2 k. We get Δf / f_o = 0.363, which gives bandwidth of 0.53 octave.

Here, the stimulus oscillator and masking noise filter are realized using the two independent sections of a single SCF IC LMF100. Both the output levels are about similar and therefore cross-coupling of the stimulus and masker will not be a serious concern as long as it is lower than -50 dB or so. Satisfactory operation (absence of appreciable cross-coupling) was experimentally verified.

4.5 Masking Noise selector and Attenuator

Masking noise selector and attenuator circuit is shown in Fig. 4.9. If masking is not selected, clock to PRBS generator is blocked by pulling the input of U9, labeled P3.7, low. If masking is selected, P3.7 is made high which is shown in Fig. 4.9. Either the broad band or narrow band noise is selected. Four analog switches in IC (CD 4066) U17 are used for selection of noise. Control signals of U17a, U17d are complementary to control signals of U17b, U17c. U10c (CD 4030) is used as an inverter to get complementary signal. If wide band noise is selected, the control labeled P1.4 is made high. Thus, making U17c, U17b on and U17a, U17d off. This passes wide band noise through unity gain inverting op-amp U18 (LM 356). If narrow band noise is to be selected then P1.4 is made low. Attenuator circuit for noise provides a programmable attenuation in the range of 0 – 88.5 dB and has been realized using AD 7111, with a circuit identical to that for tone. The noise selector and attenuator circuit works with single 5 V supply.

4.6 Speech Preamplifier

Preamplifier circuit is used for speech audiometry. The circuit is given in Fig. 4.10. The LM 381 is a dual preamplifier for the amplification of low signals. One section of LM 381 is used for amplification of speech signal coming from the microphone. The output of preamplifier is passed through the audio mixer circuit. The other section of the LM 381 is used for the audio mixer circuit. Other input for the audio mixer circuit is coming from the other interfacing devices used for speech audiometry like tape recorder, CD player or multimedia PC sound card. Here for both the channels, volume control is provided by means of potentiometers P6 and P7. The output of preamplifier is further passed through the same attenuator circuit, which is used for the pure tones. The selection of pure tone or speech signal for attenuation is carried out using analog switches in U16 (IC 4066). Since all the port pins are already assigned for specific tasks, TXD pin is used for controlling the switches in U16, in addition to the use of this pin to transmit serial data. In order to avoid switching of stimulus by TXD pin during serial data transmission, all the attenuator chips are loaded with maximum attenuation count value before start of serial transmission. Use of RXD pin for the purpose would not have been appropriate, because an output on this pin will conflict with incoming data (receive output of RS232 driver U1). Further, as the data transfer is asynchronous, information for setting maximum attenuation cannot be obtained. In case the hardware is to be designed without serial data receive feature, RXD pin can be assigned for switching of U1, leaving TXD for transmission.

Level indicator circuit continuously monitors the output of the preamplifier circuit. For indication of signal level, display bar LEDs are interfaced with IC LM3915. The IC LM3915 provides a logarithmic 3 dB/step analog display.

4.7 Power Amplifiers

To drive the headphones and the bone vibrator, three power amplifiers are needed, one each for producing tone and noise and one for pure tone to the bone vibrator. It was decided to use a power amplifier IC LM 1877, a monolithic dual power amplifier. It can deliver 2W/channel continuous into 8 ohm loads. It has low cross-over distortion and ac short circuit protection. Channel separation referred to output is – 65 dB [17].

The operation of power amplifier was tested for headphone and bone vibrator. At a time either of the two will be connected. The amplifier circuit for both is identical, except gain. While testing the two-channel amplifier, it was observed that the effective channel separation is much less than 65 dB and depends on the load connected. Possibly, the degradation in channel separation is caused by modulation of the supply voltage due to load current. Hence, it was decided to physically separate the power amplifiers for the tone and the masking noise. One IC is used for realizing the two-tone amplifiers. One section of another IC is used as the noise amplifier, while the other section is unused with the input grounded.

TDH-39 headphone was calibrated by using Artificial Ear type 4153 from B&K [16]. The maximum peak-to-peak voltage needed is 2.2 V. 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. Therefor, maximum output voltage required for bone vibrator is approximately same as that of the headphone.

The circuit for headphone and bone vibrator tone amplifier is shown in Fig. 4.11(a) and (b) respectively. The circuit is operated on dual ± 5 V supply. IC LM

1877 is operated in inverting unity gain amplifier mode. R43 and R44 are used to achieve the output impedance same as that of headphone or bone vibrator, which is approximately 10 ohms (measured at 1 kHz). Two resistors are connected in parallel to meet the wattage requirement. The circuit for noise amplifier is shown in Fig. 4.12. Input of U5a is grounded, since it is not used.

It is to be noted that 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. The three amplifiers have individual gain adjustments in the range of 1.5 to 2.5.

4.8 Circuit for Self-test

The task of the self-test circuit is to verify the output levels at the power amplifier of the tone and masking noise, and thus to verify the operation of the two attenuators. For this purpose, the full wave rectified and averaged value of the output voltage is compared with a reference voltage. The attenuation is varied, and the attenuator level for which the tone level matches the reference is found. This attenuation value is checked for each frequency tone, as well as for the wide band noise. Two separate circuits are used, one each for the tone and the noise. The circuit for self-test for the tone is shown in Fig. 4.12 (a). The tone signal V10 is converted to dc of amplitude $2V_p/\pi$, where V_p is the peak value of V10. Potentiometer P1 is adjusted for making the rectifier gain identical in both the half cycles. The value of R5 and C1 is selected such that the ac ripple is very small, even at the lowest tone frequency (250 Hz).

$R_5C_1 \geq 10/(2\pi \times 250)$.

A reference voltage of 1.2 V is obtained by using temperature compensated reference diode ICL 8069 (D3) from Intersil [9]. Voltage divider comprising of R8 and R9 gives 0.5 V reference to the negative input of comparator U3 (LM 311). Capacitors C4 and C5 are used for bypassing high frequency noise. A hysteresis of 50 mV is provided by R10 and R11 around the comparator, to avoid the jittering at the output of comparator. Since, LM 311 is an open collector comparator; a pull up resistor R12 = 6.8 k has been used. Pin 1 of the comparator is grounded. Thus, the output swings between 0 and +5 V that is compatible to the microcontroller. The algorithm for self-test is explained in the next chapter.

The same circuit is duplicated for self test for the noise channel, as shown in Fig. 4.12(b). The noise is pseudo random with a repetition period of 0.16 sec. Hence,

$R_{21}C_8 >> 0.16$. While testing, a considerable amount of software delay needs to be given between two successive increments. The two comparator outputs labeled P3.4 and P3.5, from the tone and noise circuits respectively, are connected to the appropriate port pins of the microcontroller.

4.9 Keypad

For changing various functional parameters of the audiometer, a 4×4 matrix keypad is used. It is interfaced to the microcontroller through its port 2, as shown in Fig 4.14. Since, the key pressing physically shorts two port pins, it is not possible to multiplex the port for other operations. The scanning of keypad is carried by a row-column matrix scanning technique. The various functions and settings related to tone can be set through keypad.

4.10 Display

The display used is 16 characters × 2 lines display model LCD ODM-16216S from Oriole [11]. It has an on-board CMOS based controller that works on a single 5V supply. A pot is used to adjust the LCD driving voltage, which controls the intensity. The hardware interface of display consists of 8 data lines and 3 control lines (RS, $R/\overline{W}$, $\overline{EN}$), which are interfaced to microcontroller, as shown in Fig 4.15. Control pin $R/\overline{W}$ is used for writing data/control word or reading the status of the display controller. The RS control pin is used to distinguish between 8-bit data word and control word that is sent to display. The control pin $\overline{EN}$ is used to latch the data to the display.

4.11 Serial Interface

For transferring data from device to computer or printer serial interface facility is provided. The instrument can be interfaced with a serial device through the serial interface. The serial data coming from the microcontroller serial pins is converted in to RS232 levels by driver chip IC U1 (MAX 232), as shown in Fig 4.16. The instrument can transfer data to the computer with RS232 compatible serial port. The TXD pin is also used for switching the analog switch IC U16 as we have discussed earlier.

4.12 Response Switch

The push button switch is given to patient to communicate the response to the instrument. The switch is connected to the external interrupt pin $\overline{\mathrm{INT0}}$ through debounce circuit, as shown in Fig 4.17. For the indication of subject response, a LED flasher is provided, so that operator doesn't have to look for it on the display.

4.13 Microcontroller Interfacing

Connection of various port pins of the microcontroller AT 89C52 given in Table 4.1. Some of the port pins of the microcontroller is used for the dual purposes as we discussed in the earlier sections.

Clock generation required for the SCF IC is generated by the microcontroller as a background processes by using timer 2 of the microcontroller. Timer 2 is programmed in the clock generator mode. Thus, generating a 50% duty cycle clock at P1.0. This clock is applied to the programmable oscillator. P3.6 is used for selecting either 50:1 or 100:1 mode of operation for SCF. The tone frequency is dependent on crystal. The present system uses 24 MHz crystals. External interrupt pin $\overline{\mathrm{INT1}}$ is used for warble tone generation. Response switch is connected to external interrupt pin $\overline{\mathrm{INT0}}$.

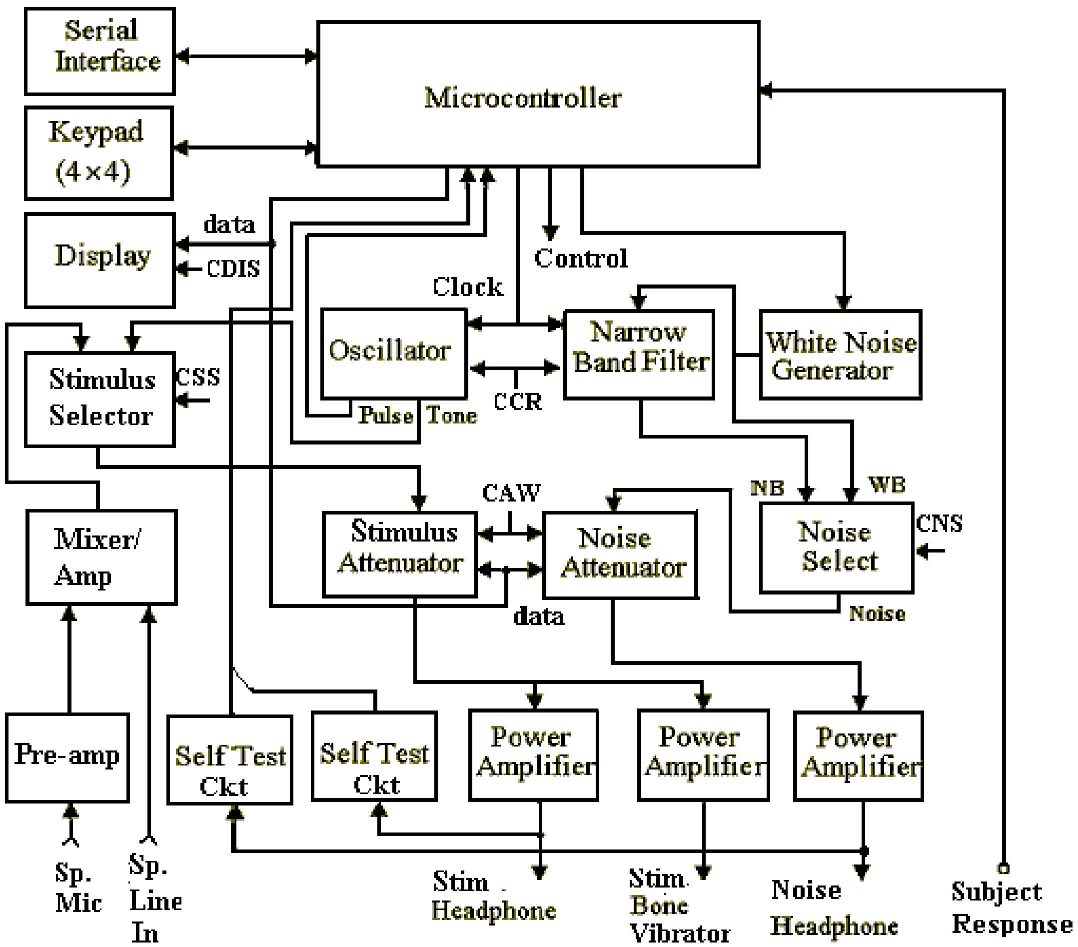
A 4×4-matrix keypad is directly connected to port 2. Keypad is scanned repeatedly after 60 ms. the interrupt for scanning is generated using Timer 0.

Port pin P1.5 is common for R/$\overline{\mathrm{W}}$ of display and $\overline{\mathrm{WR}}$ of three attenuator chips. Control signals, RS and $\overline{\mathrm{EN}}$, for display are provided using P1.6 and P1.7 respectively. Clock to the PRBS generator is disabled when noise is not needed. This is done using P3.7. The clock to the BPF (for generating narrow band noise) is same as that for the tone oscillator. Noise selection is made using P1.4. Port pins P3.4 and P3.5 are used for polling the output of comparators of the level monitoring circuit. This is done only during the level-monitoring mode after power on. The response switch can be closed for terminating the self-test in between. The RxD and TxD lines of port 3 are used for serial communication with printer or computer.

Table 4.1

Functions assigned to I/O port pins of microcontroller 89C52 [1].

I/O Port pins	Functions assigned
P0.0 to P0.7	Data bus for tone and noise attenuator, display
P1.0	Clock to oscillator and BP filter (U15- 10,11)
P1.1	$\overline{CS}$ Of tone attenuator (U3- 12)
P1.2	$\overline{CS}$ of noise attenuator (U5-12)
P1.3	$\overline{CS}$ Of tone attenuator (U4- 12)
P1.4	Wide / narrow band noise selection (U10-9 and U17-5,6)
P1.5	$R/\overline{W}$ of display and $\overline{WR}$ of both attenuators (CN5-5, U3,U4,U5-13)
P1.6	RS of Display (CN5-4)
P1.7	En of Display (CN5-6)
P2.0 to P2.3	Write to keypad row lines (CN3-1 to 4)
P2.4 to P2.7	Read from keypad column lines (CN3-5 to 8)
P3.0	Serial interface RxD (U1-9)
P3.1	Serial interface TxD (U1-11) + Stimuli select (U16-12, 13)
P3.2	Interrupt 0, subject response
P3.3	Interrupt 1, pulse i/p from tone generator (for warble)
P3.4	Input from self-test of tone circuit (CN4-1)
P3.5	Input from self-test of noise circuit (CN4-2)
P3.6	50:100 clock control (U15-12)
P3.7	PRBS clock control (U9-4)



Control: CCR (Clock ratio control for SCF), CSS (Control for stimulus select)
CNS (Control for noise select), CAW (Control for atten. write),
CDIS (Control for LCD display)

Fig. 4.1 Block diagram of system hardware, adapted from [1].

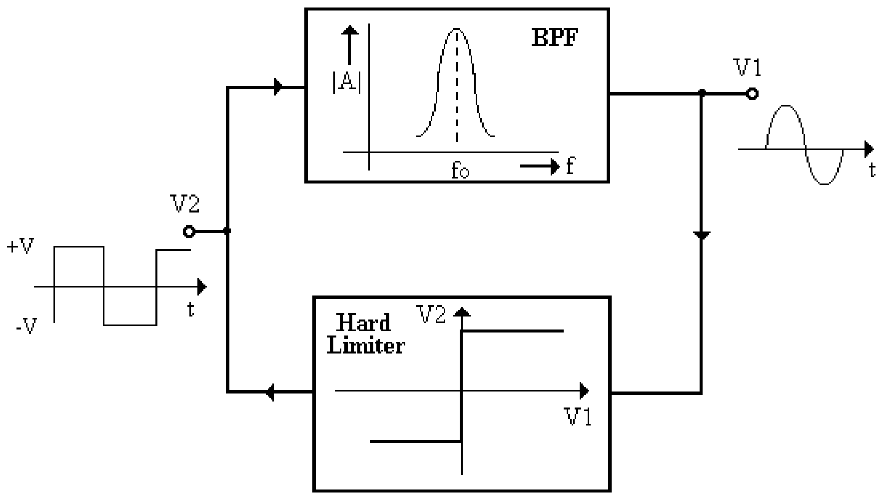


Fig. 4.2 Block diagram for quadrature oscillator.

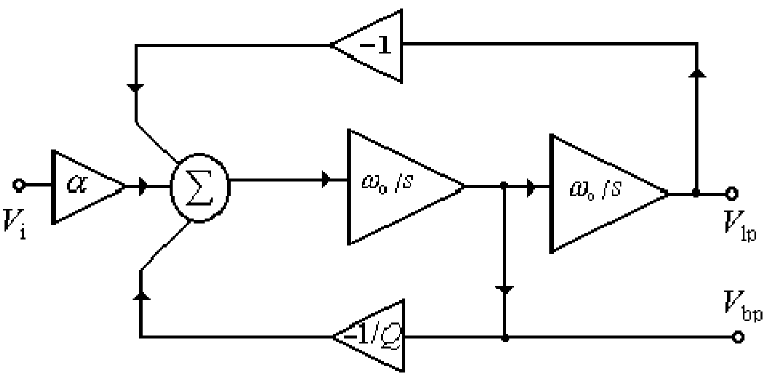


Fig. 4.3 Band pass filter realization using two non-inverting integrator loop.

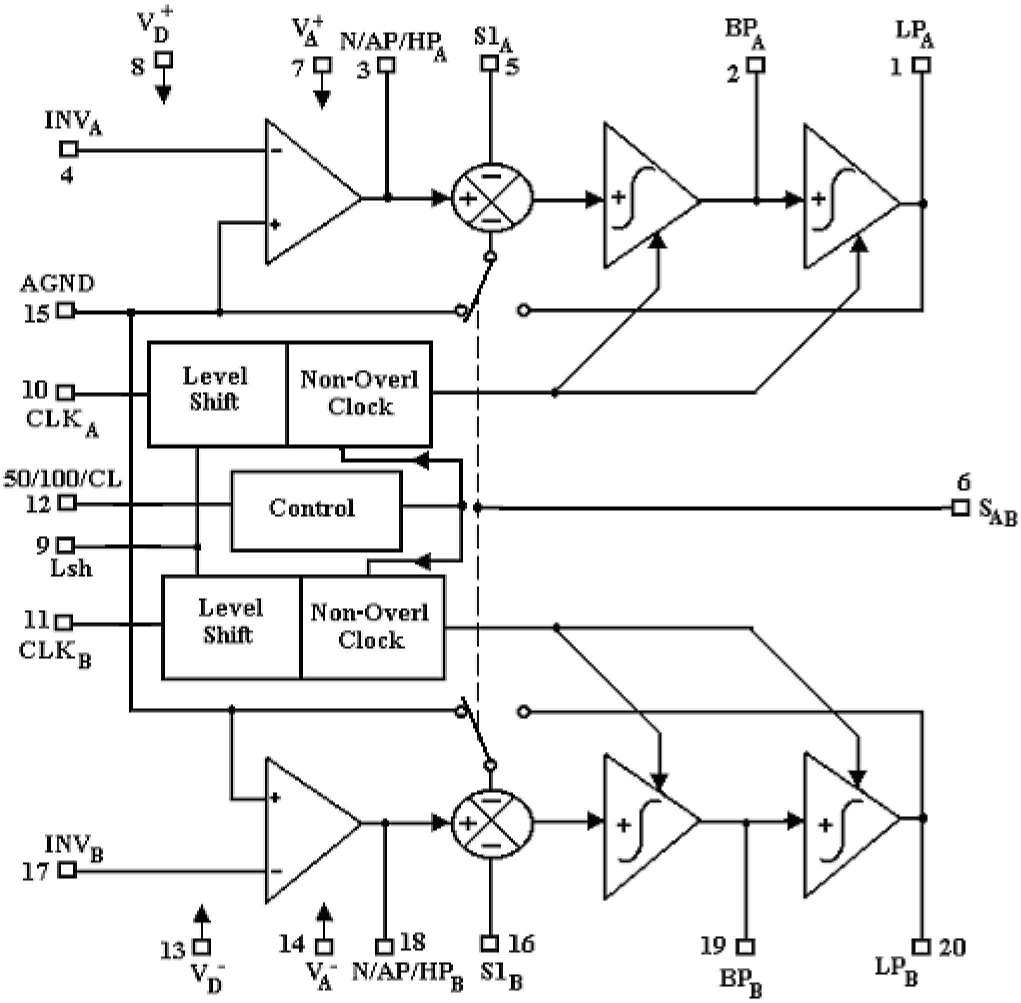


Fig. 4.4 Block diagram of SCF LMF 100. Source [8].

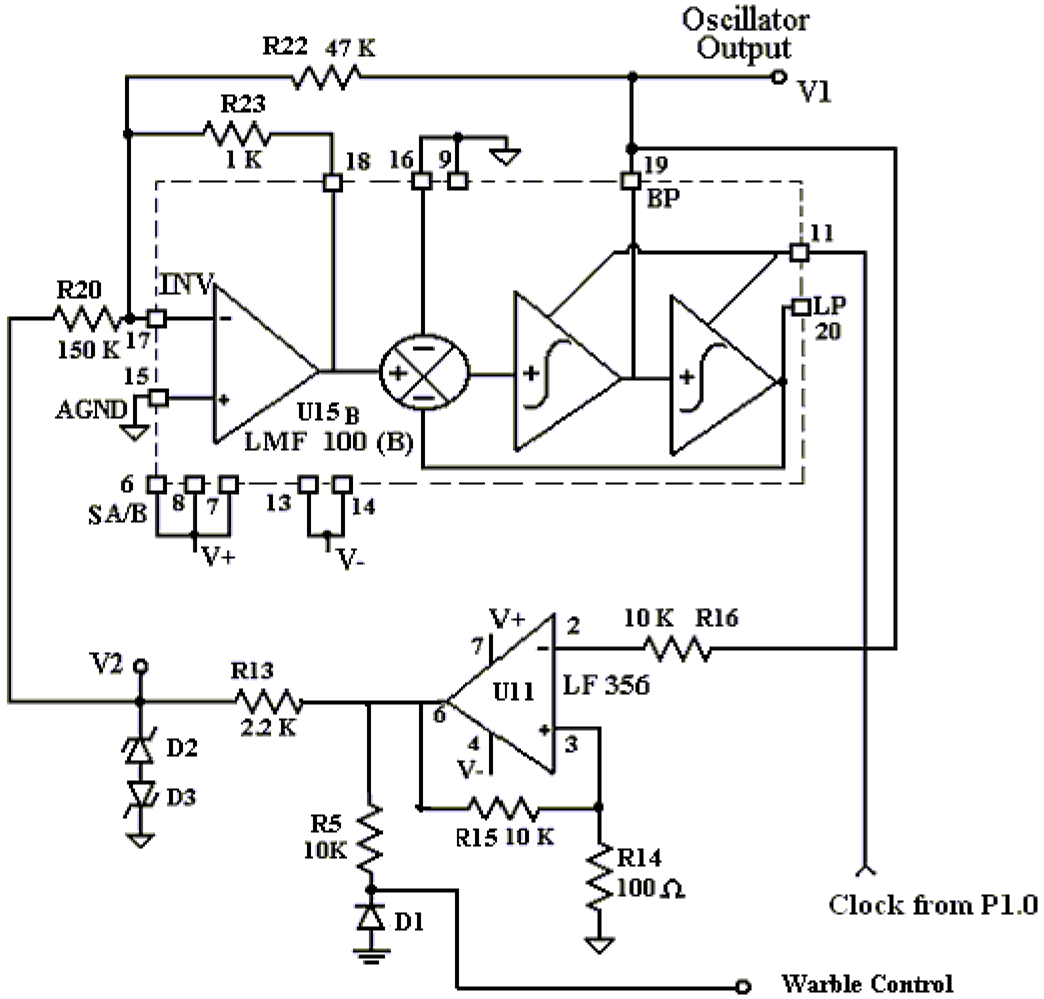


Fig. 4.5 SCF based quadrature oscillator using LMF 100.

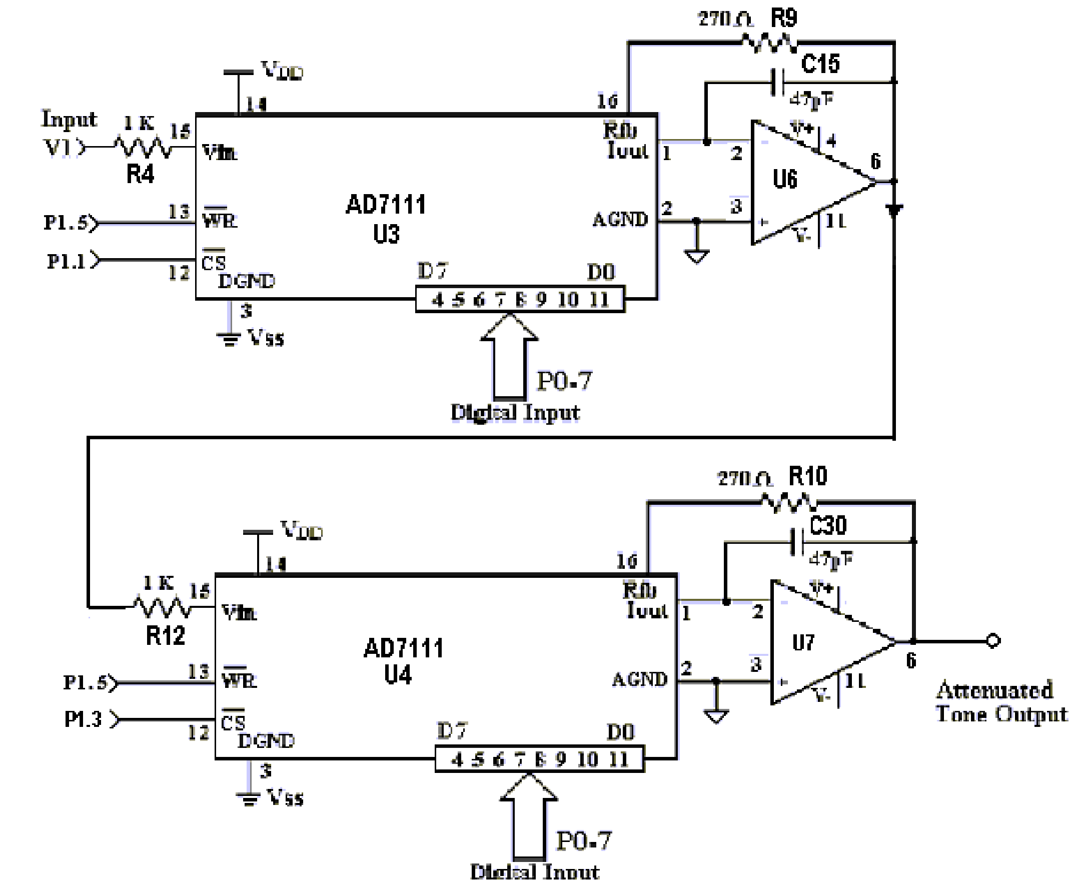


Fig. 4.6a Attenuator circuit for stimulus

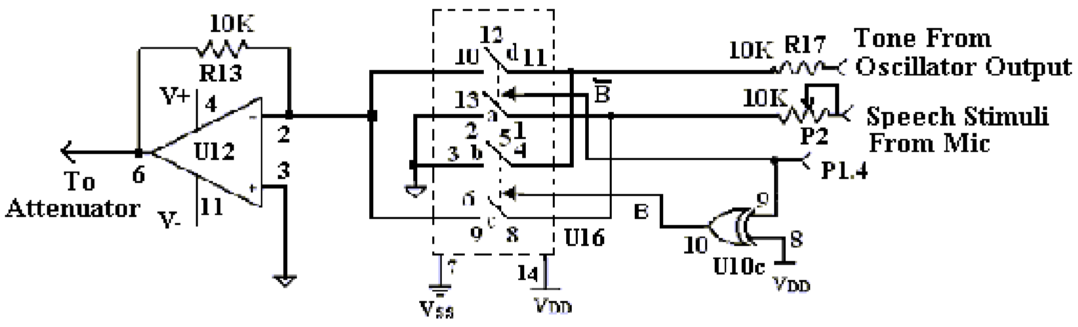


Fig. 4.6b Circuit for stimulus selector

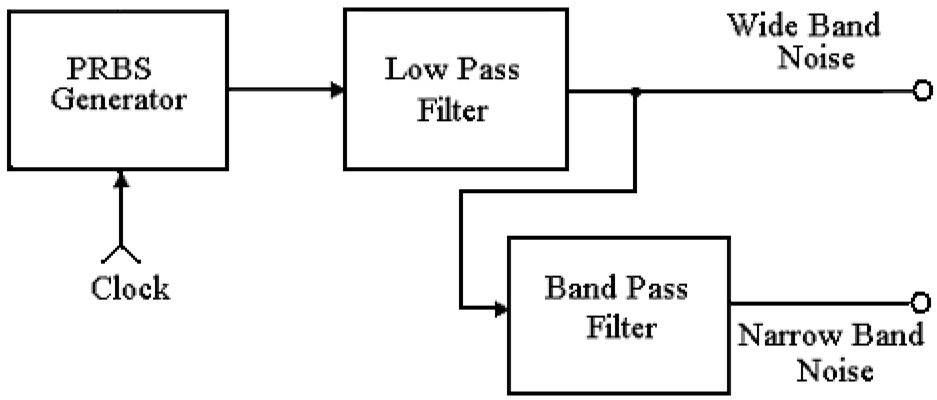


Fig. 4.7 Block diagram for noise generator

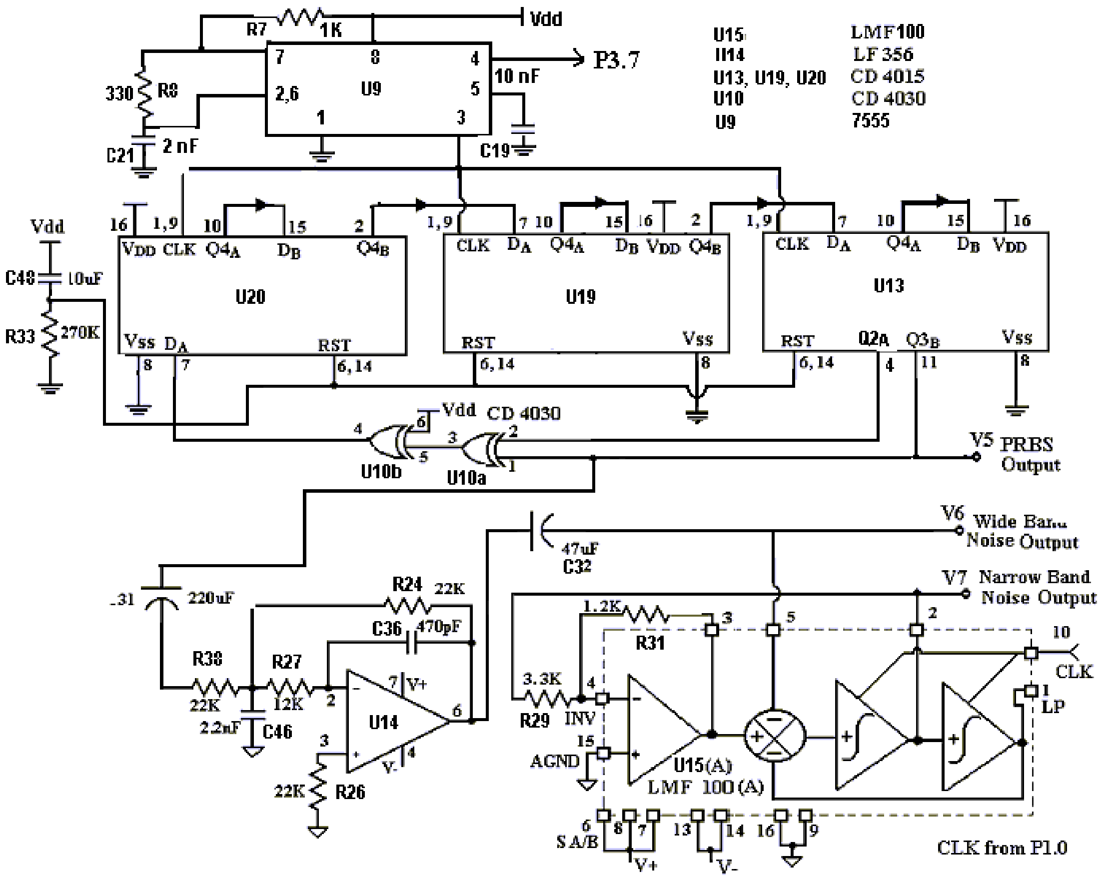


Fig. 4.8 Circuit for noise generator.

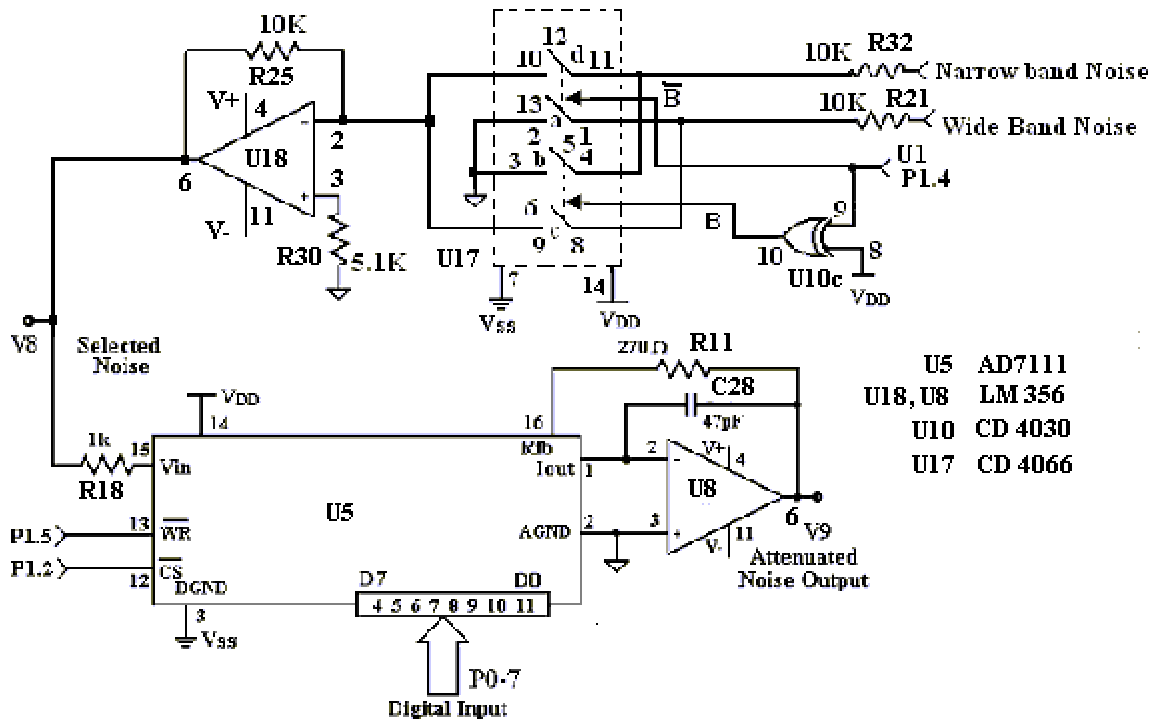


Fig. 4.9 Noise selector and attenuator circuit.

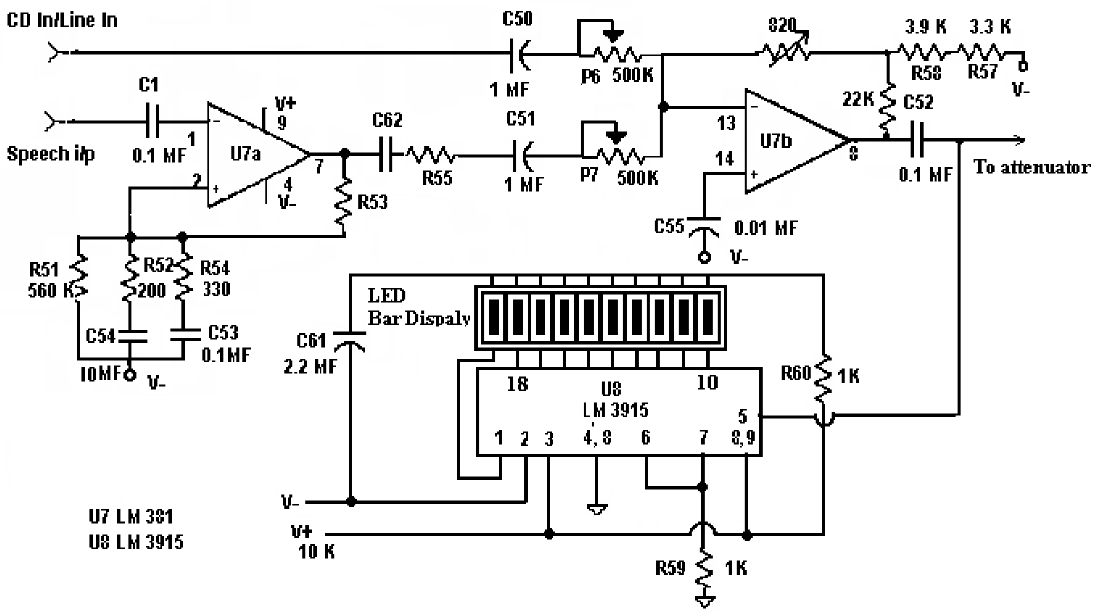


Fig. 4.10 Speech preamplifier circuit.

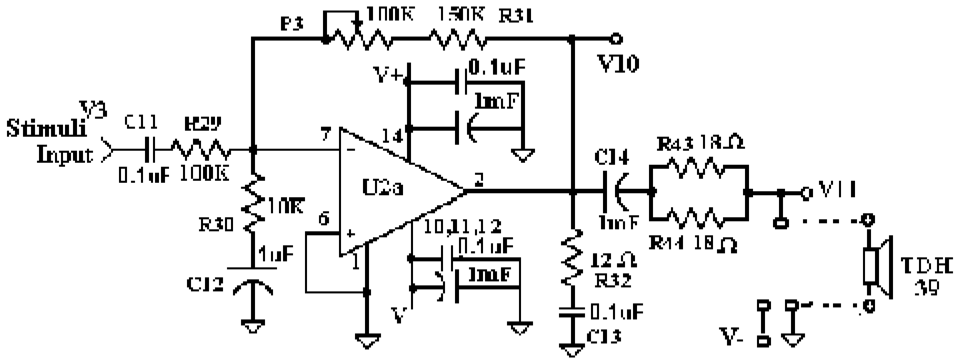


Fig. 4.11(a) Power amplifier for headphone.

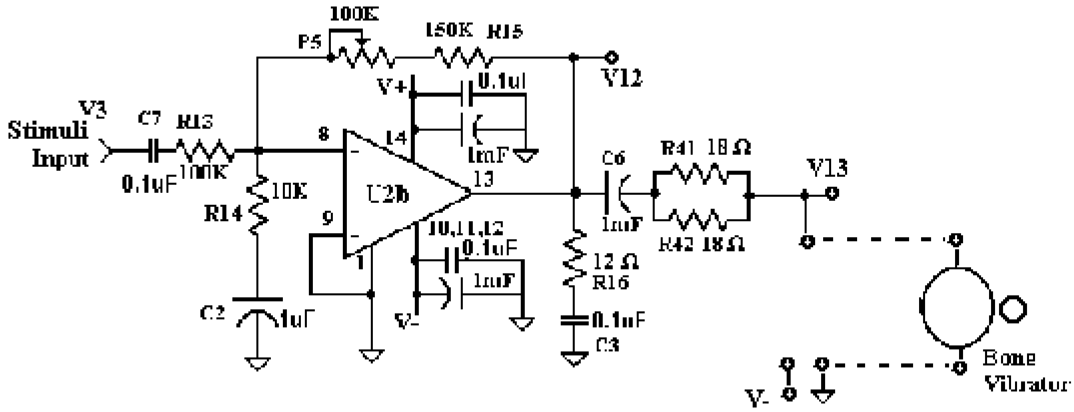


Fig. 4.11(b) Power amplifier for bone vibrator.

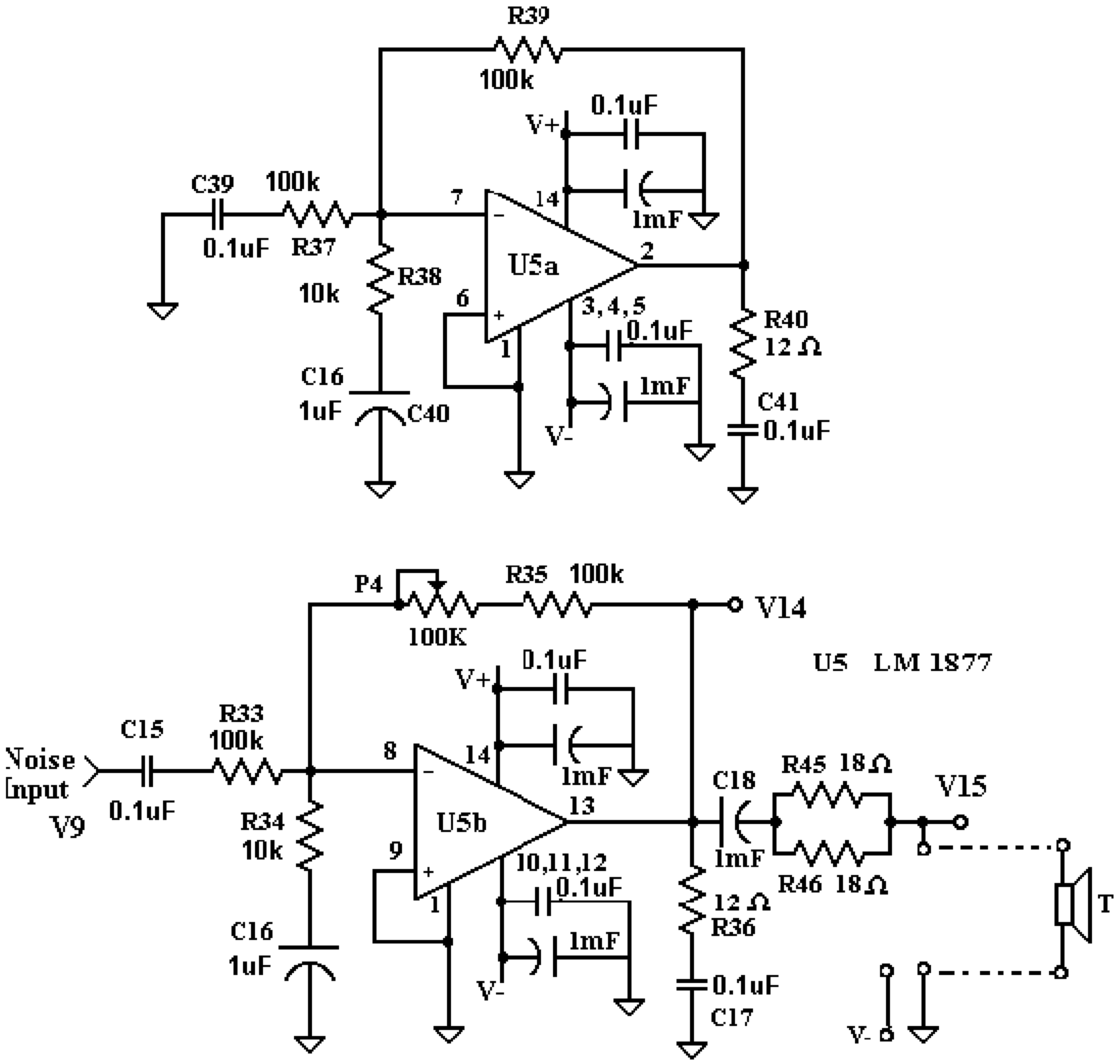


Fig. 4.12 Power amplifier for noise.

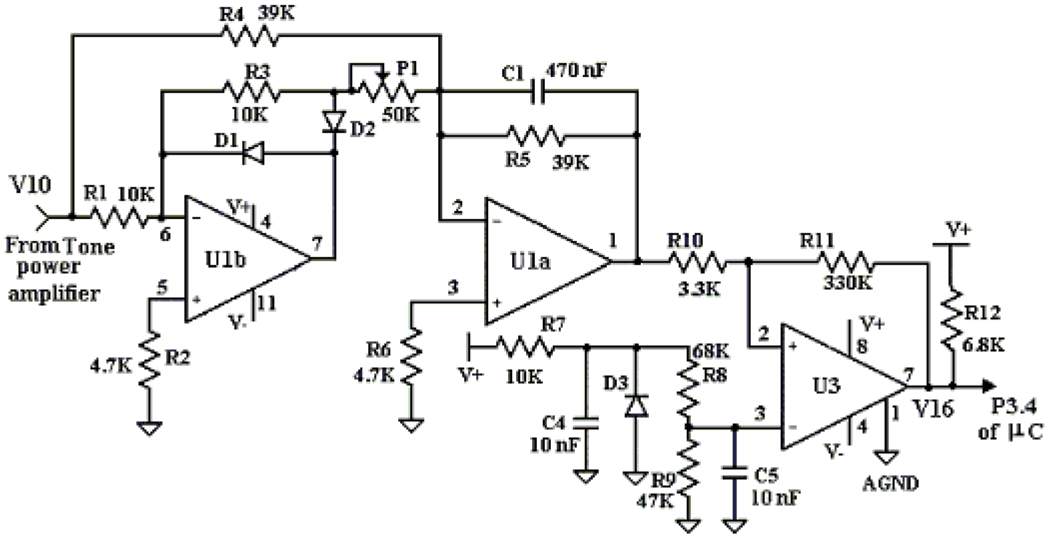


Fig. 4.13(a) Level monitoring circuit for tone channel.

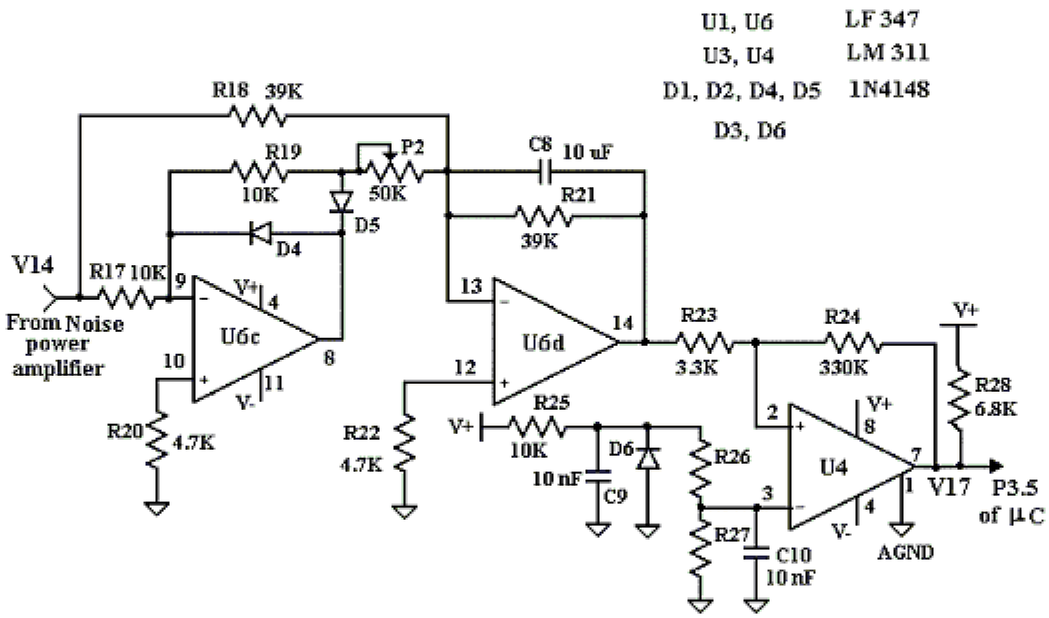


Fig. 4.13(b) Level Monitoring circuit for noise channel.

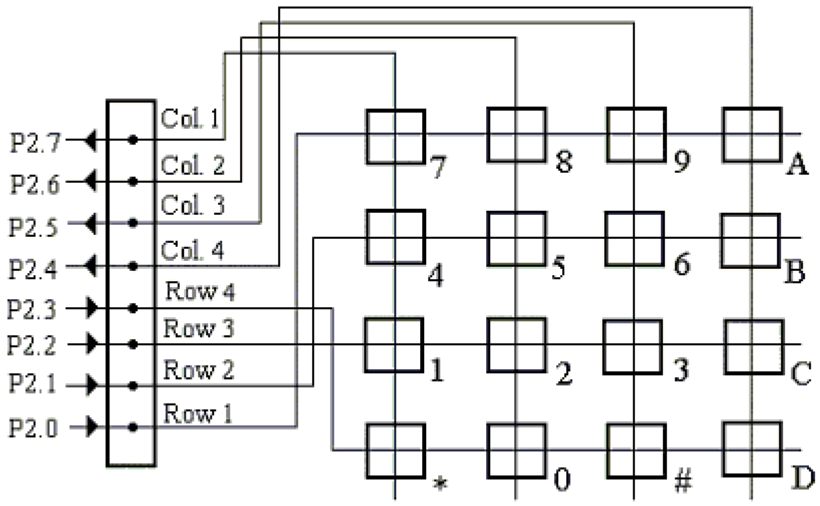


Fig. 4.14 4X4 keypad Layout.

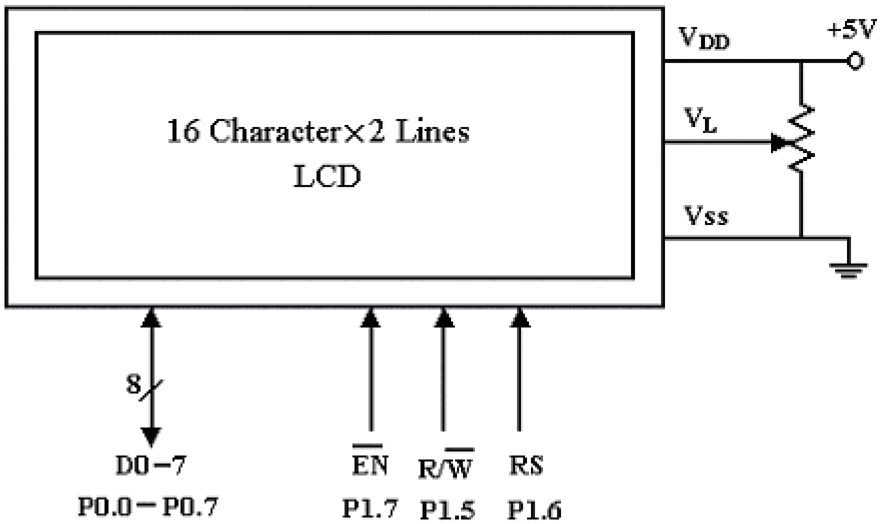
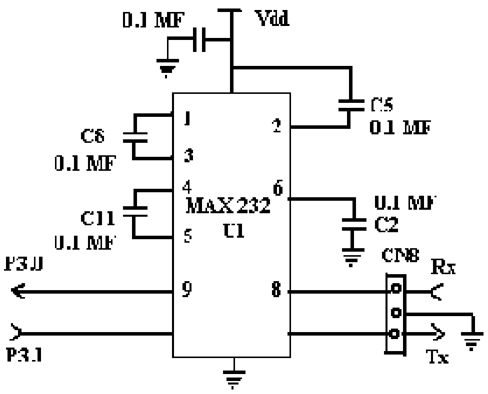


Fig. 4.15 16 characters × 2 lines ODM-16216S LCD.



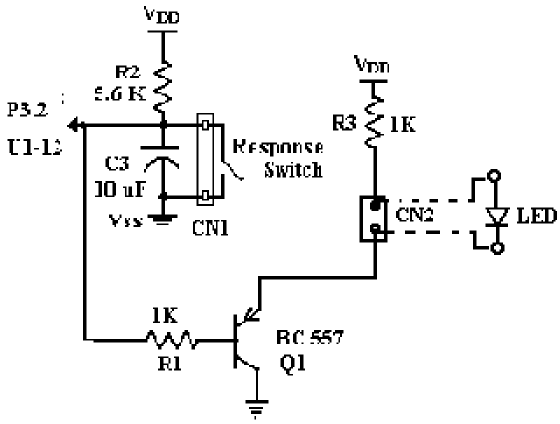


Fig. 4.16 Circuit for serial interface.   Fig. 4.17 Debounce for response switch.

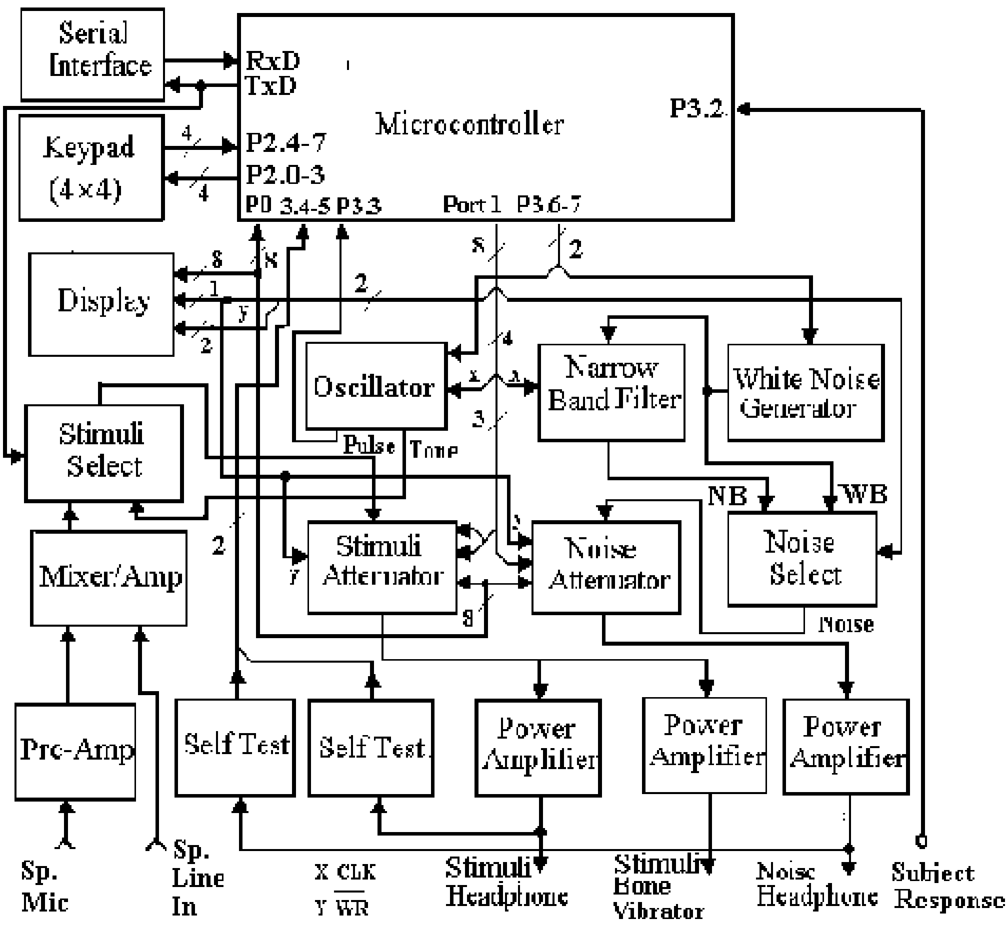


Fig. 4.18 Microcontroller interfacing with various blocks.

Chapter 5

SOFTWARE DESCRIPTION

The software has been developed by retaining the basic structures and properly functioning modules of the software developed earlier by Ashish Kothari [4] and Pratibha Reddy [5], some of the modules have been modified and some have been developed afresh. This chapter provides a description of the software in its present form.

Generation of tones of different frequencies switched capacitor filter (SCF) IC requires clock frequency as input. Microcontroller timer 2 as a background process provides this clock. The clock frequency can be easily changed by changing the count value loaded in to the timer 2 of the microcontroller. The pin assignments of the microcontroller given in Table 4.1. Fig. 4.18 shows the block diagram of the system along with the interfacing details. A 50% duty cycle clock (using Timer 2) generated at P1.0. is applied to the programmable oscillator. P3.6 is used for selecting either 50:1 or 100:1 mode of operation for SCF. External interrupt pin $\overline{\mathrm{INT1}}$ is used for warble tone generation. Response switch is connected to external interrupt pin $\overline{\mathrm{INT0}}$. The generated tone frequency signal is passed through the attenuator IC for changing the proper level of the tone. Port pins P1.1, P1.2 and P1.3 are used for selecting ($\overline{\mathrm{CS}}$) of tone and noise attenuator ICs.

Port pin P1.5 is common for R/$\overline{\mathrm{W}}$ of display and $\overline{\mathrm{WR}}$ of both attenuator chips. Control signals, RS and $\overline{\mathrm{EN}}$, for display are provided using P1.6 and P1.7 respectively. P3.7 controls the clock applied to the PRBS generator. Noise selection is made using P1.4. Port pins P3.4 and P3.5 are used for polling the output of comparators of the level monitoring circuit. The RxD and TxD lines of port 3 are used for serial communication with computer. TXD pin is also used for control of tone/speech signal switching.

This chapter provides a description of the control of the hardware blocks by the program on the microcontroller, implementation of the test algorithm, and overall operation.

5.1 Software modules

Each software module is described in the following subsection.

5.1.1 Tone generation

Since timer 2 of a microcontroller is used as a clock generator and as a background process no additional software routines are required. When a particular frequency is selected, the register RCAP2L of timer 2, which is running in clock generator mode will be loaded with a number corresponding to that frequency. Contents of RCAP2H is always 0FFH. Timer 2 will output a clock frequency given by [6]

f_{CLK} = (Crystal Freq./4) / (65536 – $(RCAP2)_D$)

Where $(RCAP2)_D$ is the decimal equivalent of the hex numbers in registers (RCAP2H, RCAP2L), taken as a single 16-bit register. Circuit uses a crystal of 24 MHz. Modes and the number to be loaded in RCAP2 register for pure tones test frequency counts are given in Table 5.1. This results in a frequency resolution of 500 Hz and 250 Hz for 100:1 and 50:1 modes of SCF oscillator respectively. At lower frequency, 50:1 mode is used, while at high frequency we use 100:1 mode.

In order to modify the tone signal as a frequency modulated tone (warble tone) or amplitude modulated tone software subroutines are written. Warble tone is generated as a frequency modulated tone with a sweep of ±10% of the tone frequency. It was decided to complete one sweep in two seconds at lower frequencies, and more sweeps at higher frequencies [4]. The number of sweeps 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. frequency steps are not very distinct. This has been done within the constraint of total modulation of ±10% and the constraint of smallest frequency step as determined by change of 1 count in the counter register. The time interval between the frequency steps is obtained by counting the tone cycles. The output labeled P3.3 from the tone generator is a unipolar square wave in synchronism with the sinusoidal tone, and it is applied as external interrupt $\overline{\text{INT1}}$, which is operated in the negative edge triggered mode. Table 5.2 shows, for each frequency, the number of sweeps/s, number of steps/s, and number of tone cycles (number of INT1 interrupt) per step, along with the range of actual count to be loaded in RCAP2L register for changing the frequency. The sweep starts from the lowest frequency. The number of INT1 interrupt are counted, and after the specific number of counts, the RCAP2L content is changed, resulting in a new frequency step, and the process of counting INT1 is repeated. The frequency sweep steps for 500 Hz, 2000 Hz, and 8000 Hz are shown in Fig. 5.1. The number of tone cycles in the time interval of each frequency step is the same, and consequently the time intervals vary over a range of ±10%. The external interrupt 1 will be enabled and the stimuli will be frequency modulated. For this, a flag 'purewarb' is used. When this flag is set, tone will be warble type, otherwise pure or amplitude modulated. The selection of tone can be done by using key pad.

Amplitude modulated tone generation is also implemented using software. Here amplitude modulated tone is generated by modulating the level within ±1 dB around the tone level. This modulation is provided in steps of 0.375 dB. Total 12 steps are there in one sweep. It is decided to complete one sweep in one second. Hence each step lasts approximately for 80 ms. This delay is generated using a software subroutine, which polls for a flag that is set after 80 overflows of timer1 set for 1 ms. Amplitude modulated tone for 20 dB level is shown in Fig. 5.2.

5.1.2 Attenuator Control

Tone generated from oscillator circuit, which requires attenuation as per the sound level requirement in logarithmic scale. The desired tone level attenuation is obtained by loading the attenuator count into the logarithmic D/A converter chip IC AD7111 by software. As mentioned earlier in section 4.1.2, "0" attenuation count corresponds to 0dBm voltage level (0.775 V rms). Attenuation counts *A* results in voltage level of -0.375*A* dBm. The attenuator count, *A*, is a function of hearing level *L* and the tone frequency *F*. Corresponding attenuation counts are calculated. Thus, the table of counts, *A*30(*F*), corresponding to 30 dB HL for different frequencies is prepared. The counts are also given in Table 5.3, for a specific piece of TDH39P headphone. From these values, the voltage level in dBm and corresponding count for *L*=30 are calculated and given in Table 5.3. The attenuation count for a specific L and F is obtained as

$$A(L, F) = A(30, F) + \left(\frac{30 - L}{0.375}\right)_{\text{rounded}}$$

Since *A* needs to be calculated for the value of *L* in steps of 5 dB over the range of 0-100 dB, using the above formula, an array of 20 values is used. Further since A cannot be accommodated as a signed number in 1 byte, it was decided to use the following method. The values to be stored in array are calculated as

$$A_l(L) = \text{abs}\left(\frac{30 - L}{0.375}\right)_{\text{rounded}}$$

for L varying from 0-100 in 5 dB steps. The attenuation count is calculated as

$A(L, F) = A(30, F) + A_l(L);$ for $L \leq 30$

$A(30, F) - A_l(L);$ for $L > 30$

Here A is a 9-bit integer. It is to be noted that attenuator chip AD7111 can provide a maximum attenuation of 88.5 dB, i.e. 236 decimal count. So, to increase the range of dB level second attenuator chip AD7111 is used. Hence if A exceeds 236, then second attenuator chip is turned on and corresponding count is loaded into the second IC. In the earlier version, a circuit using CD4066 analog switch was used for additional attenuation of 40 dB. But it was giving a click while turning the tone off. So it was decided to use one more additional attenuator chip IC for tone attenuation. This implementation can be used for increasing the attenuation range for a research model of the instrument.

While testing, sound level is increased or decreased in steps of 5 dB. Since, the resolution of AD 7111 is 0.375 dB; the previous count should be changed by 13 to get change of 5 dB. If the level is to be increased by 5 dB, then decrease the previous count by 13 (0dH) and if the level is to be decreased, then increase the previous count by 13 (0dH). Actual number is 13.33, which is rounded to 13. Hence, for a change of 15 dB, the rounding error will be 1, which is nullified by adding 1 to the corresponding number. Hence, the rounding error does not go beyond 0.66 i.e. 0.2475 dB.

The attenuation range of 0 - 88.5 dB is sufficient for masking noise. Hence, the scheme for attenuation of masking noise is same as for the tone, except that there is only one attenuator chip. For both types of noise, the frequency dependence of the count is the same as for the tone frequency selected.

Throughout the testing subroutine operation clock signal to the programmable oscillator is being applied. Because when tone is to be presented, the flag TCON.2 is set then the clock signal is presented to the oscillator chip, which generates tone signal at the output. To stop this generated tone, we can reset TCON.2 flag. But this technique gives rise to a “thump” at the start and at the end of the tone. Therefore when tone is not presented the attenuation count is made maximum and the clock frequency can be continuously on and when tone is not presented the attenuation count can be made maximum. Since the attenuator output suddenly changes from inaudible to audible tone, we hear a "click" sound instead of thump. Therefore instead of putting the tone off abruptly, the attenuation should be gradually increased from present value to the maximum attenuation within a time span of about 100 ms. Then the clock can be put off to save power. When the next tone starts again (by providing clock), attenuation should be decreased from maximum to the desired value within time span of about 100 ms. Since, the voltage change achieved is logarithmic, in order to have similar delay at all voltage levels, the delay is increased as the attenuation decreases i.e. at higher dBHL values delay is more. This approach will keep the dc level at the oscillator output to minimum, thus avoiding thump. It should be noted that at higher frequencies, oscillator resumes its operation at a faster rate than that at lower frequencies. Hence at lower frequencies a longer delay should be provided. Fig. 5.3 shows the various wave envelopes observed with direct tone presentation (thump), tone presented with slowly increasing intensity (without thump), and tone in which clock is kept continuously on and attenuation is kept maximum in case of ‘tone off’. Experiments were carried out for incrementing or decrementing the different dB count values for smooth 'tone on' and 'tone off' and the count values for best perceived tone have been worked out.

5.1.3 Keypad, Display, and Serial Interface

As a user interface facilities and for selection of tone, noise, and other parameters, a 4×4-matrix keypad is interfaced. This keypad is connected to port 2 of microcontroller. Keypad is scanned at intervals of 10 ms as a background process, by using interrupts generated by Timer0.

The scan result is communicated via flag `kbhit` and keycode in memory location `key` with value in the range of 0-f h. Thus at a time only the most recent valid key is available and the foreground process resets the flag `kbhit` after reading the key. The scan process provides a debouncing interval of two scan cycles (i.e. 20 ms). Invalid key presses are ignored, and a valid key press in registered only once. Key must be released for one scan cycle before its pressing can be registered. The background scan process itself is described in appendix B. For each key there are specific functions associated with it, also described in appendix B.

The LCD display consists of 8 data lines and 3 control lines (RS, R/$\overline{W}$, $\overline{EN}$), which are interfaced to microcontroller. Control pin R/$\overline{W}$ is used for writing data/control word or reading the status of the display controller. The RS control pin is used to distinguish between 8-bit data word and control word that is sent to display. The data are latched on the falling edge of the pulse $\overline{EN}$. The pulse width of $\overline{EN}$ should be greater than 450 ns. The display takes varying amounts of time to accomplish different functions. Data bit 7 is monitored for logic high (busy) to ensure that the display is not overwritten.

The response switch is connected to external interrupt pin INT0. When the output is low, the interrupt is generated. It will set the 'resflag', which is polled in main subroutine to detect the subject response. An LED is connected to the same port pin to indicate pressing of the response button.

The serial communication is used to download results to the PC by using a program, "*audiogram*" on the PC. This program is described in a later section. Download is done with the data transfer rate of 2400 bps, even parity as MSB followed by 7-bit data. Timer 1 is used to generate the data transfer rate. TXD line of microcontroller also serves the purpose of switching pure tone mode or speech audiometry mode selection.

5.1.4 Level Monitoring Routine

For checking of the operation of the power amplifier, a self test routine is executed at the power-on reset. This routine makes use of "level monitoring" circuit. In this mode, the level of test tone is increased in steps of 1 dB. The rectified average value of tone from the power amplifier (headphone amplifier) is compared with the reference. Microcontroller checks for pin P3.4, which is the output of the comparator, to find the tone level which just exceeds the reference. The attenuator control count is stored in RAM. The routine is executed for each of the nine test frequencies. The result is displayed on the LCD display. For the purpose of display, when the tone level increases from 0 dB starting with lowest frequency i.e. 250 Hz, the incrementing counter is displayed on the upper left corner. And when the value reaches to the referance voltege, that value is frozen on the display and for next frequency curser moves towords right (in the 2 line x 16 character display) and again the same procedure repeats. At the end, all the values corresponding to the referance voltage are displayed. Operator can proceed further by pressing “Ok” key on the keypad. Intermidiately the test can be terminated by pressing either the subject response switch or by pressing “Ok” key on keypad. The level monitoring circuit and routine here is checking for any frequency dependent variability. The calibration table may very for various types of headphones.

5.2 Test Modes and Algorithm

After power on, a self-test routine is executed. The self-test routine can be terminated by pressing the response switch. Once the self-test is over, various parameters for the test are selected. The various test mode options are,

1) Man: Manual audiometry for tones and speech
2) Aut: Automated for tones
3) SISI: SISI test
4) TDT: Tone decay test

In the manual audiometry mode, we can select the following

a) STIM: Stimulus type [pure tone, warble tone (FM), amplitude modulated tone (AM), speech (SP)]
b) Air / Bone: type of conduction [air, bone]
c) Duration: Tone duration [2 sec, 3 sec, 4 sec, continuous]
d) Noise: Mask [wide band noise (WB), narrow band noise (NB), no noise (__)].

All 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 is available only in the manual mode. Also, speech audiometry sets the duration to continuous.

All the parameter option of manual mode is also available in automated mode. In tone decay and SISI tests, only pure/warble tones can be used as stimuli. The test algorithm implemented in the software for the various modes are described in the following subsections

5.2.1 Manual Pure Tone Test

In the manual pure tone test, the operator can turn the tone on or off by using "tone on" key. The tone is turned off when tone duration is complete or "tone off" key is pressed. Operator can select frequency or tone level according to the subject's response. When tone is turned off, the operator has access to all the keys. So he can change the settings. When operator presses "Save" key, the level value is stored as threshold for that particular frequency. If in the middle of the test, operator changes the test mode, no result is stored. Operator can display the test results through the key pad.

5.2.2 Automated Pure Tone Test

The flowchart representation of the normal audiometric procedure for threshold determination is shown in Fig.5.4. Initially a pure tone of 30 dB HL 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 30 dB tone at first time, if the patient does not hear it, the level is raised is steps of 10 dB step until it is heard for 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. The minimum presentation level at which the subject responds at least 50% times (3 responses out of 6 tone presentations), is taken as the hearing threshold [1][2]. This algorithm has been implemented for the automated mode.

5.2.3 SISI Test

This test is normally carried out after finding the pure tone hearing threshold using normal pure tone audiometry. In SISI test, 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 increment 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 [1]. Twenty such bursts are given and out of them, the number of bursts the patient is able to detect is recorded. The no. of responses is converted to percentage and stored as the test results. 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.

5.2.4 Tone Decay Test

This test can be carried out with or without detecting the hearing threshold of the subject. This test should be carried out in auto mode to get accurate result. The operator selects the frequency. The subject is instructed to press the switch as soon as he 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 form 30 dB HL, until the subject responds. If the subject is able to hear the tone for more than one-minute [1], 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. If reverse is the case 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.

For SISI and tone decay tests, tone type is pure tone and type of conduction is air conduction, and masking noise selection is same as that of other modes. When the instrument goes from SISI or tone decay test mode to pure tone mode, the earlier set of parameter in pure tone audiometry are restored. After that the frequency and tone level values are displayed, operator can change the frequency and tone level value. When the new frequency is selected the default value for level is set at 30 dB. Operator will press the "Tone On" key to start the test.

5.2.5 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 adjusted through the keypad. The calibration for attenuator is the same as that for 1 KHz tone. Manual volume control facility is also provided for keeping proper level of the test stimuli.

5.3 Operation Sequence

Operation sequence of the instrument is given in Fig. 5.5. After power on, a self-test routine is executed. The self-test routine can be terminated either by pressing the response switch or “Ok” key on the keypad. Once the self-test is over, various options (viz. stimulus type, tone duration, mode of operation, mode of conduction, and noise type) are selected. Stimulus can be either pure, warble, or amplitude-modulated tones, or speech. The tone duration can be 2s, 3s, 4s, or continuous. 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, or no noise.

Once the initialization is over, the parameters for stimulus (e.g. frequency, tone level, and noise level) are decided by the audiologist. The frequency range for air conduction is from 250 Hz to 8 kHz. Whereas, for bone conduction 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 tone, display will show ‘PR’ at the bottom right corner. Once, the tone on duration is over, tone is turned off. The operator can turn off the tone by pressing the ‘tone-off’ key. Once, the presentation of tone is over, the instrument will wait for approximately 1s to receive the response from the patient. While waiting for the response ‘WT’ will be displayed at the bottom right corner. The subject response is communicated to the microcontroller interrupt $\overline{\mathrm{INT0}}$. If no response is obtained in the wait period, ‘NR’ will appear at the bottom corner of the display. If the response is positive, ‘PR’ will be displayed. Subject can even press the response switch before the tone on duration is elapsed. Closure of the response switch will interrupt the tone and a positive response will be considered.

In the manual mode, the audiologist will decide the next level of tone depending upon the previous responses. The test algorithm should be followed for finding the hearing threshold for a particular frequency. This threshold is saved in the data RAM of the microcontroller. After completing the test, the result can be displayed sequentially for all frequencies. It can also be downloaded to a computer or printed using the serial port of the instrument.

In the automated mode, audiologist has to select the initial parameters and the frequency. The instrument does the rest all in accordance with the test algorithm. The threshold obtained is stored automatically. The test frequency is to be manually set. There is only one memory location for storing the test results for a particular frequency. The most recent result overwrites the previously stored result.

In SISI test, operator will select the frequency and 20 dB super threshold tone level. Then the instrument will take care of the rest and will find out the score. In between operator can stop the test by pressing “tone off” key.

In tone decay test, operator will set the frequency and tone level 30 dB above threshold. Then the instrument will take care of the rest and will find out the threshold. In between operator can stop the test by pressing “tone off” key.

In speech audimetry test, operator will set the level control through keypad. The audiologist depending upon the subject response determine the score for the speech audiometry facility.

5.4 PC Interface Program

To download the test results from the audiometer using its serial port and plot audiograms, a program has been written in C to run on a PC. It receives data from the audiometer over serial port with the settings of baud rate: 2400 bps, 7 bit data with even parity, 1 stop bit. Patient information (name, age, sex, test ear, test mode) is entered through PC keyboard. These data are stored in a file which can be printed, used in another document, or standard package. In the audiogram, unentered patient information fields 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.

Table 5.1

Operation Mode & Count loaded in RCAP2 for different frequencies for 24 MHz crystal frequency

Frequency (Hz)	Operation mode	Number to be loaded (Hex)	Frequency (Hz)	Operation mode	Number to be loaded (Hex)
250	100:1	FF10	3000	50:1	FFD8
500	100:1	FF88	4000	50:1	FFE2
1000	100:1	FFC4	6000	50:1	FFEC
1500	100:1	FFD8	8000	50:1	FFF1
2000	50:1	FFC4			

Table 5.2

Different parameters for warble tone.

Frequency (Hz)	Steps per sweep	No. of sweeps for tone of 2 Sec.	No. of INT1 interrupts per step (Dec.)	Actual count in RCAP2L (Hex) for ± 10 % deviation
250	13	1	27	03 to 1d
500	13	1	55	77 to 84
1000	9	1	143	BD to C6
1500	5	1.5	200	D6 to DB
2000	9	1	285	DF to E8
3000	5	1.5	400	D6 to DB
4000	5	1.5	533	DF to E4
6000	3	2	1000	EB to EF
8000	3	2	1250	F0 to F2

Table 5.3

Attenuator counts for 30 dB HL acoustic output for TDH 39.

Frequency *F* **(Hz)**	**Voltage in dBm for 30 dB HL**	**Attenuator Count** $A_{30}(F)$	
		Decimal	**Hex**
250	-60.98	163	A3
500	- 74.75	199	C7
1000	- 77	205	CD
1500	- 75.25	201	C9
2000	- 71.99	192	C0
3000	- 75.37	201	C9
4000	- 75.49	201	C9
6000	- 68.33	182	B6
8000	- 59.39	158	9E

Table 5.4

Data read for key presses

Data read for key presses																
Key pressed	*	0	#	D	1	2	3	C	4	5	6	B	7	8	9	A
Scan Pattern	EE	ED	EB	E7	DE	DD	DB	D7	BE	BD	BB	B7	7E	7D	7B	77
Keycode	00	01	02	03	04	05	06	07	08	09	0A	0B	0C	0D	0E	0F

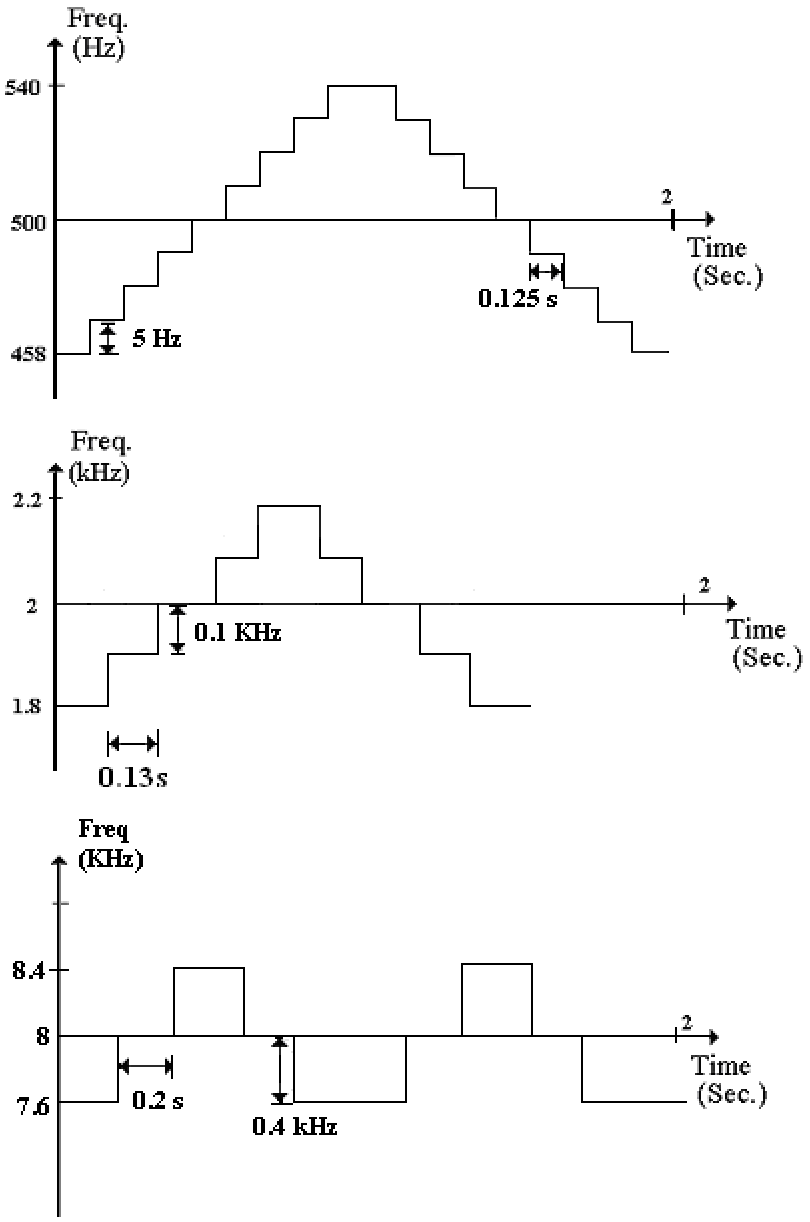


Fig. 5.1 Frequency sweeps for 1 kHz, 3 kHz, and 8 kHz

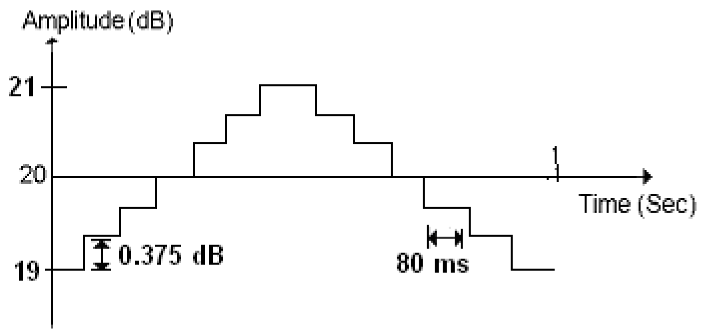


Fig.5.2.a Amplitude modulated tone for 20 dB level

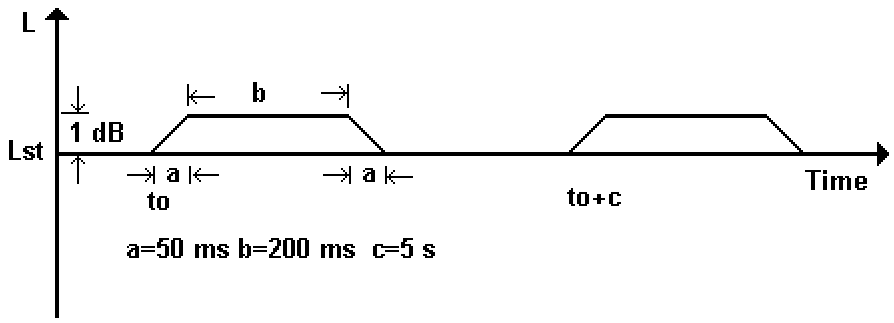


Fig 5.2.b Amplitude levels for SISI test

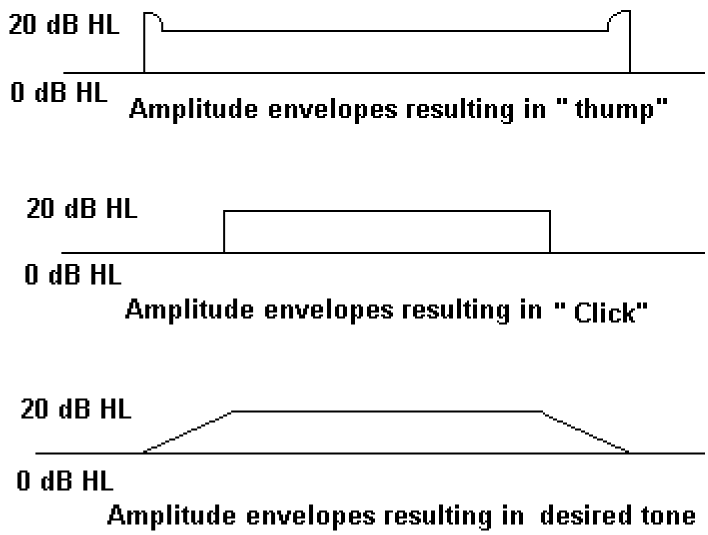


Fig. 5.3 Tone envelops

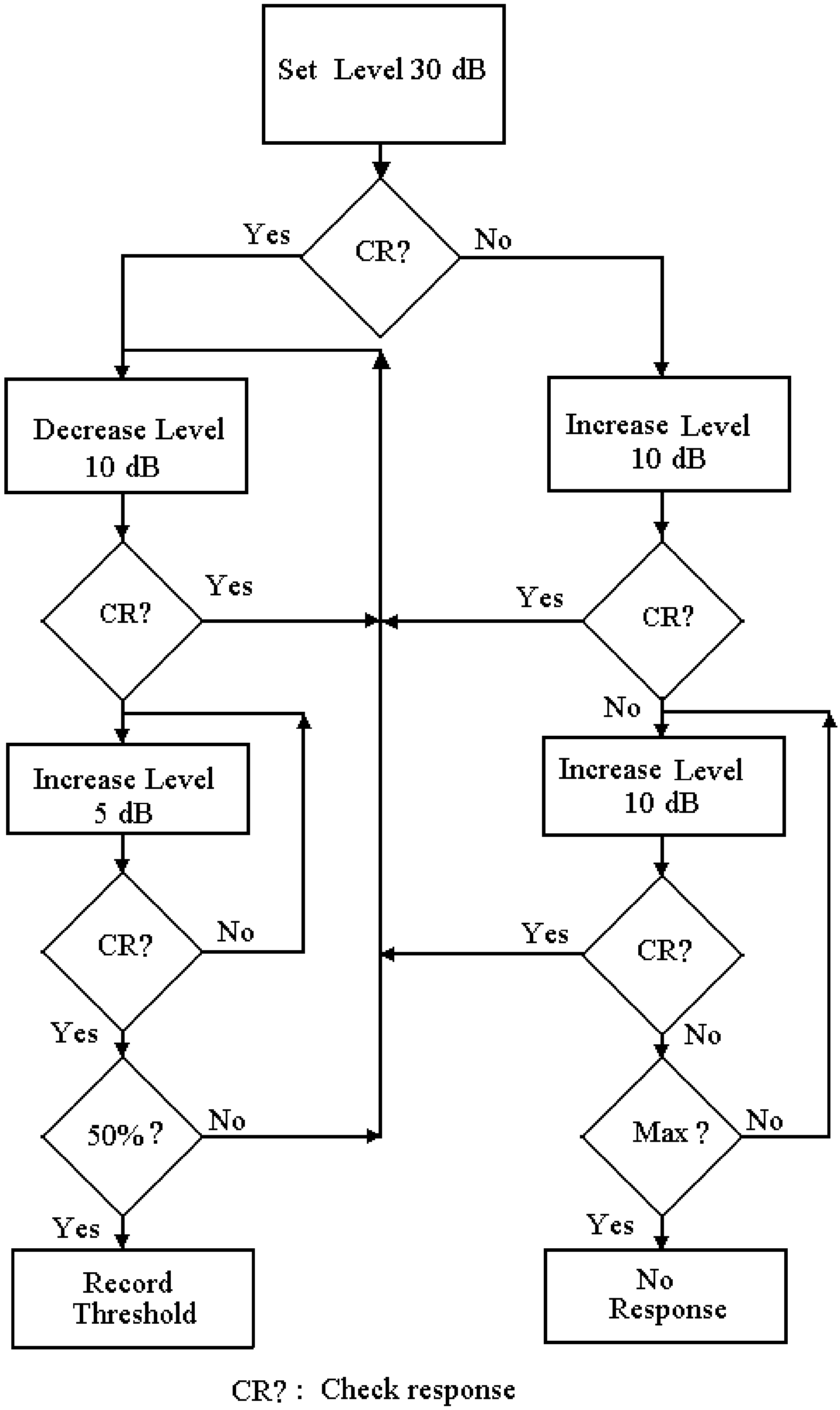


Fig. 5.4 Flowchart for automated test

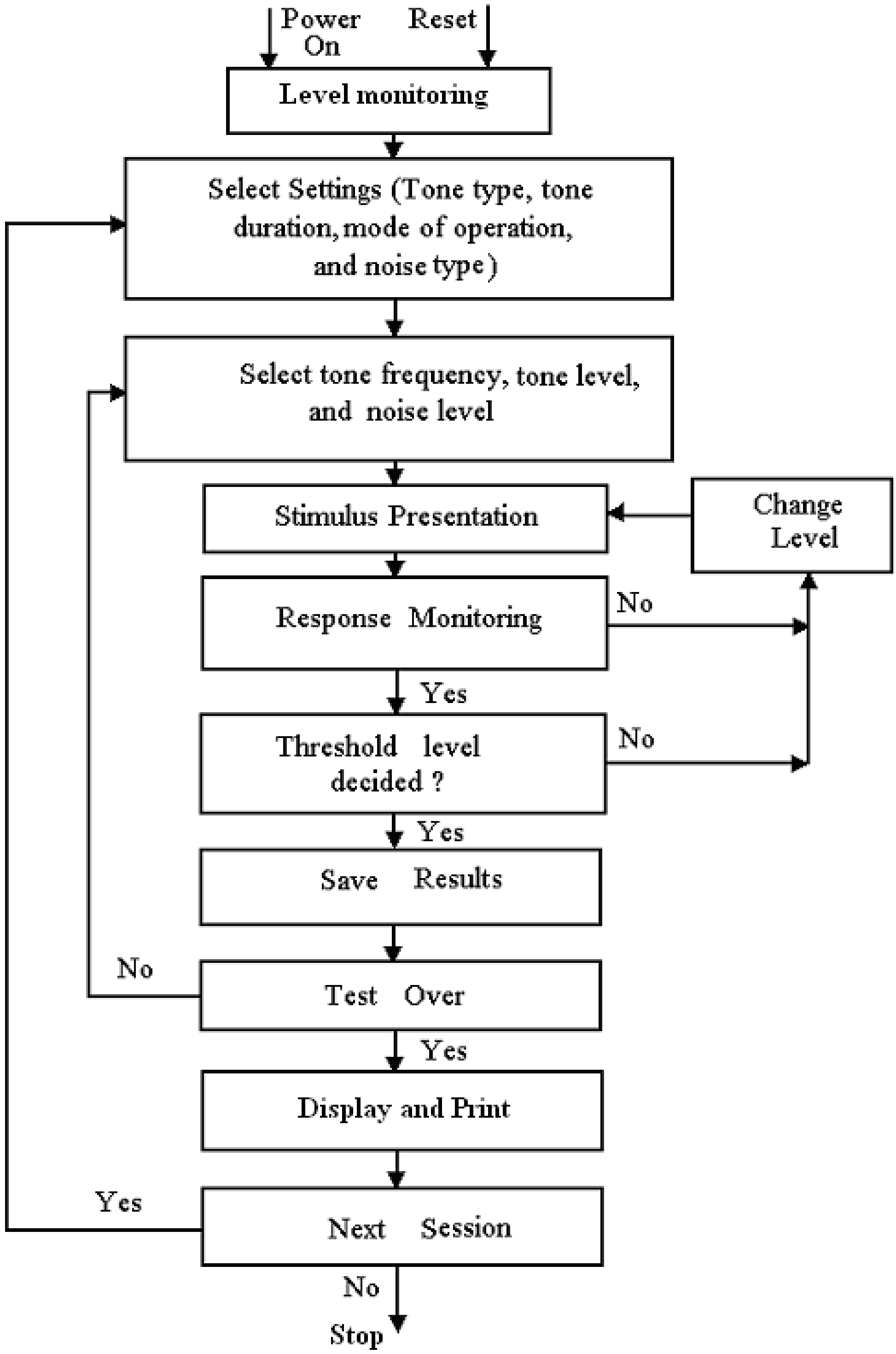


Fig. 5.5 Operation sequence for threshold search

Chapter 6
SYSTEM ASSEMBLY AND TESTING

As the hardware was substantially modified, new PCBs were designed. There are two PCBs. PCB-1 consists of the tone generator circuits, and the microcontroller. PCB-2 consists of the power amplifier and the circuit for level monitoring. LCD display unit and keypad are connected to PCB-1. Partitioning of the circuit was done keeping in view (a) interconnection between the boards, (b) modular upgradation of the design in the future. The PCB's have been assembled and mounted on a Bakelite sheet and appropriate connectors and switches have been provided. The instrument has been tested for the operations of its blocks and for its overall operation. This chapter provides a description of PCB design, assembly, and testing.

6.1 PCB Design

The sizes of PCB-1 and PCB-2 are 16.0 cm × 9.25 cm and that 16.0 cm × 7.0 cm respectively. Both the PCBs are double sided with plated through holes (PTH). Most of the circuit blocks on both the PCB’s have mixed signals (i.e. analog and digital), and consequently special care was needed in the layout design.

In PCB-1, the tone oscillator uses SCF, which needs digital clock. The wide band noise is generated by analog low pass filtering of digital PRBS. The data and control lines of the tone and noise attenuators (logarithmic DAC AD7111) are interfaced to microcontroller. Thus, almost everywhere analog is meeting digital. There is a great possibility of analog supply being corrupted by digital switching noise. Even in the digital parts, proper decoupling of power supply of each chip is essential. Also the supply routing should be done carefully. The layouts of component, solder side and component placement are given in Appendix C. The analog ground and digital ground are routed separately throughout the PCB. The two grounds are shorted by two-jumper settings at the most electrically stable point in the PCB. Any one of the two jumpers can be shorted. The entry points of the analog and digital supplies on PCB are decoupled by 220 μF/50 V electrolytic capacitor in parallel with 0.1 μF ceramic disc capacitor. Thick copper plane of ground is provided on one side of the PCB with overlap to the extent possible, for having distributed capacitance for decoupling effect. Each IC is decoupled by 0.1 μF ceramic disc capacitor placed electrically as close as possible to the supply and ground pins of the particular IC. Special care is taken while track routing for attenuator AD 7111. The layout is designed to provide ground shielding between the analog and digital pins of the IC. A ground track is run between input V_{in} and output of DAC chip in order to minimize feed through from input to output [14]. For observing the output of major blocks of the circuit, test points are provided.

PCB-2 has three blocks: stimulus (tone/speech) amplifier and self-testing circuit, masker and amplifier and self test circuit, speech pre-amplifier/mixer. In order to minimize cross talk between stimulus and masker channels, the supply routing of these blocks is different and a thick coupling (copper plane) is provided between analog grounds of both channels. The supply entry points are decoupled by 1000 μF/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. Shielded coaxial cable is used for transferring audio signals between the two boards.

6.2 Assembly

The 2 PCBs, display, and keypad have been mounted on a Bakelite sheet. 2 PCBs are mounted on the bottom plate on the left hand side. On the right hand side, the display is mounted and below it the keypad is mounted with slight elevation compared to the PCBs. And to hold this board at 30-degree elevation a special wooden stand with a hinge at the center is made. Whenever this stand is not in use, it can be folded and kept aside. The various connectors and switches are provided on the left hand side in proper slots. Later the PCB's and other parts should be assembled in an appropriate instrument cabinet.

6.3 Testing

The testing of individual blocks was carried out. The results and observations are discussed in this section.

The spectral purity of tone signal was observed on dynamic signal analyzer HP 3561A. All the harmonics are at -54 dB or lower with respect to the test tone frequency. To test the stability of the oscillator output, the analog supply voltages were varied from ± 4.5 V to ± 5.5 V and oscillator output amplitude was measured. Table 6.1 shows the variation in the output. The output is not frequency dependent.

However, it varied by 250 mV for a supply variation from ±4.5 to ±5.5V. These variations are due to change in the current in the voltage level stabilizing diodes (D2, D3). Thus to maintain proper levels, it is necessary to use stabilized supplies.

The spectrum of wide band noise was observed on the signal analyzer. Fig. 6.1 shows that the power spectrum for the wide band and narrow band noise for 250, 1 k, and 8 kHz. The analysis bandwidth is kept at 119.5 Hz. The wideband noise is flat up to 8 kHz. The roll-off outside the pass band is 12 dB/octave. For the narrow band noise, from the component values used in the circuit, bandwidth comes out as 0.53 octaves.

The operation of power amplifiers was tested, and the gain variation was verified. The gains are set by adjusting pots, so that there is unity gain under rated load condition.

The serial port interface was tested by connecting the audiometer to a PC running the program "*audiogram.c*" for downloading the test results from audiometer and displaying these as an audiogram. The sample audiogram is shown in Fig 6.2a and the file data format is shown in Fig 6.2b.

Table 6.1

Oscillator output (p-p) at different frequencies for different supply voltages.

Frequency (Hz)	V_{p-p} at $V_s = \pm 4.5$ V	V_{p-p} at $V_s = \pm 5.0$ V	V_{p-p} at $V_s = \pm 5.5$ V
500	1.88 V	1.94 V	1.95 V
2000	1.88 V	1.91 V	1.95 V
6000	1.89 V	1.91 V	1.95 V

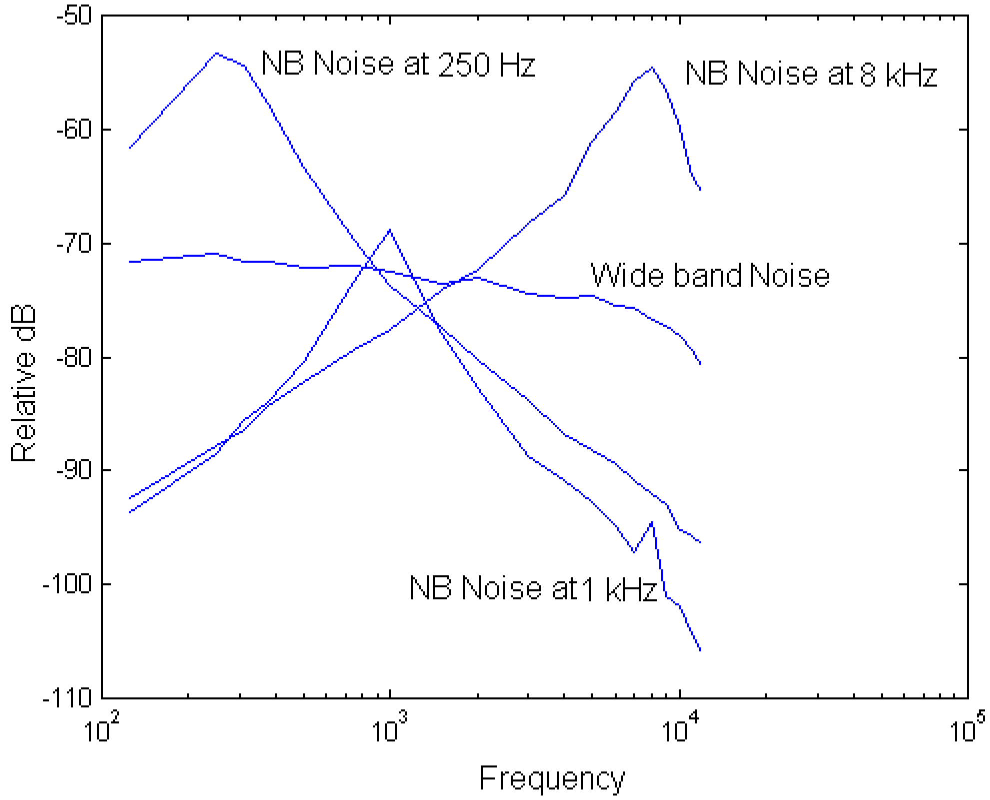


Fig. 6.1 Averaged power spectra of wide and narrow band noise taken at output of power amplifier, for 6o dB HL with analysis bandwidth as 119.5 Hz.

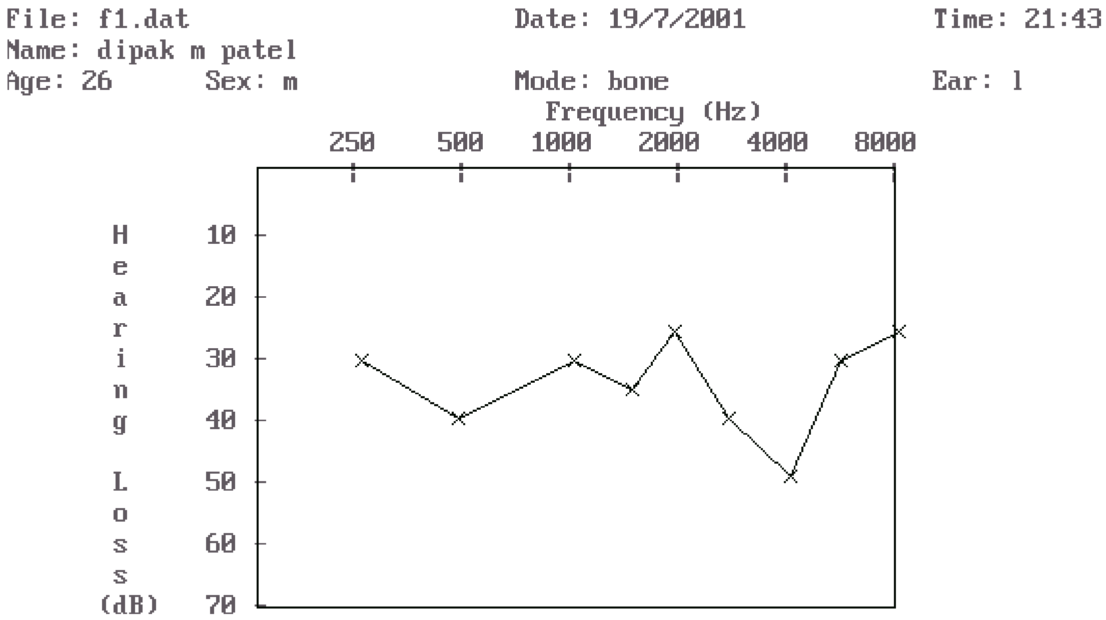


Fig. 6.2a Audiogram generated by “audiogram.C”

```
19 7 2001 21 43 39
dipak m patel
m 26 bone l
30
40
30
35
25
40
50
30
25
```

Fig. 6.2b Data file format of file “f1.dat”

Chapter 7
SUMMARY AND SUGGESTIONS

The objective of this project was to develop a fully functional portable diagnostic audiometer, with operation over full range of frequency and hearing level, It should have full masking facility, and air and bone conduction. It should also provide the facility for speech audiometry by interfacing of the instrument with a multimedia PC. It should be usable in mobile clinics and even in rural areas.

7.1 Work Done

A microcontroller based pure tone diagnostic audiometer is developed which operates over a frequency range of 250 Hz to 8 kHz and acoustic output of 0 dB HL to 100 dB HL. It can generate warble tone having $\pm$10% frequency deviation and amplitude modulated tones. It has two channels, one for pure tone and other for masking noise. Full masking facility is provided, which includes both, wide-band and narrow-band noise (centered at test tone frequency). Facility of air and bone conduction as well as speech audiometry is provided.

The instrument is menu driven with option of manual and automatic modes. This is made possible by using microcontroller, SCF-based programmable oscillator, and programmable attenuator. Digitally controlled attenuation in dB scale is obtained by using logarithmic D/A converter. Wide-band noise is generated by low pass filtering of the output of a pseudo random binary sequence generator and is shaped to a narrow-band noise by a SCF based narrow band filter, whose centre frequency is the same as the test frequency.

The range of tone level control is increased by the additional attenuator IC, so that the instrument can be used for research laboratory.

In the earlier prototype the masking noise has a perceivable repeatability. This noise quality is improved by making 23 bit shift register PRBS generation.

Speech audiometry facility is provided by additional hardware on the analog PCB. Attenuation for this test stimuli is provided through same hardware which is used for pure tones on the digital board. Level monitoring of test stimuli is indicated by the bar display LEDs. Manual volume control for test stimuli is also provided.

For the subject response indication, an LED is interfaced with the circuit such that when subject gives positive response it flashes, so operator doesn't have to look at the LCD display.

The newer hardware provides standard RS232 serial data transfer. The instrument has a serial port interface for transferring the test results to computer or printer. The program "*audiogram.c*" can be used for receiving the test results over serial port and storing these and patient information in a file. The test results can be plotted as an audiogram, and a hard copy can be obtained.

7.2 Suggestions for Future Work

Further improvement can be carried out to incorporate the features and specification of an advanced diagnostic audiometer. To improve the audiometer further, the following features can be added.

Facility of EFD type of display interface with this instrument, may improve its usability in lab models. However, for battery operated portable unit, LCD display should be retained.

In the present design, attenuation table for the output device is hard coded in the program memory. The design 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 / 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 NVRAM interfaced to the microcontroller can be used for storing this table. The design at this stage can be further enhanced so that the NVRAM 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

System Specifications of Audiometer "IITB-AUD2k1"

Audiometer type: dual channel microcontroller based audiometer, with pure/warble tone/AM tone stimulus and wide-band/narrow-band masking noise. Facility of air and bone conduction. Facility of SISI test and tone decay test.

Circuit size: two double-sided PCBs with PTH. PCB-1 of 14.5 cm × 13.5 cm and PCB-2 of 10 cm × 13.5 cm.

Stimulus: crystal controlled test tone frequencies, with intensity level controlled in 5 dB steps. The ranges of tone output for air conduction and bone conduction are 0 to L_{max}(dBHL) for different frequencies as given below

Frequency (Hz)	250	500	1000	1500	2000	3000	4000	6000	8000
Air L_{max} (dBHL)	90	100	100	100	100	100	100	90	80
Bone L_{max} (dBHL)	40	50	50	50	50	50	50		

Warble tone: frequency deviation of ± 10% with one sweep in two seconds.

Amplitude modulated tone: amplitude deviation of ± 5 dB with one sweep in one second.

Masking noise: broadband/narrow-band noise over 0-60 dBHL range in 5 dB step. Wide-band noise: flat spectrum up to 8 kHz, with approx. 12 dB/octave roll off on the higher side. Narrow-band noise: centered at test tone frequency, 3-dB BW ≈ 0.55 octave, 20-dB BW ≈ 4 octave.

Output Devices: Headphone type TDH-39 (software calibration for other headphones, by changing a table). Bone Vibrator type Oticon 70127 (software calibration for others)

Control and indication: control through 4×4-matrix keypad of size 9×9 cm.

16 characters × 2 lines LCD display with font 5×7 or 5×10 dots.

Operation: software controlled menu driven manual / automated modes.

Result Storage: for one set of the test results with rewrite facility.

Interfacing: serial port (TxD, RxD, and GND), TTL level, baudrate of 2400 bits per second, 7 bit data, and even parity.

Self test: internal monitoring of output levels.

Power supply: +5V, 20 mA for digital and ± 5V, 120 mA for analog.

Appendix B
Keypad Scan Algorithm and Key Functions

B.1 Keypad Scan Algorithm

Scanning of the 4×4 matrix keypad connected to P2 is done as a background process, initiated by interrupts periodically generated by Timer 0. P2 has internal FET pull-ups. When 1's are written to P2 pins, they are pulled high by the internal pull-ups and these can be used as inputs: an external circuit can overcome the high impedance pull-ups and drive the pin low to input 0 or leave the pin high to input 1. As shown in Fig 4.13, the pins 2.0 to 2.3 are connected to the 4 rows and pins 2.4 to 2.7 are connected to the 4 columns for row-column scanning for determining the key pressed by sensing row-column shorting. Scanning and debouncing is handled by software and no external hardware is needed.

Timer 0 is dedicated for generating interrupts for keypad scanning. It is used in mode-1, and 16-bit count " N " loaded in the timer register results in periodic interrupts at the interval of

$$Ts = \frac{2^{16} - N + 1}{f_{clk}/12}$$

where f_{clk} = crystal frequency. With f_{clk} = 24 MHz and $N = 45535$ we get T_{s} = 10 ms. It is to be noted that the interrupt service routine has to load count N each time for timing the next scan cycle.

The scan result is communicated by the interrupt service routine to the foreground program by setting flag `kbhit` and placing the scan code in memory location `key`. The key code is in the range 0-F h and no further processing is normally needed for code conversion. At a given time, only the most recent key is available and the foreground program resets `kbhit` after reading `key`. The scan process provides key debounce of two scan cycles (20 ms). Invalid key presses are ignored and a valid key is registered only once. A key must be released at least for one scan cycle (10 ms), before it can be registered again.

The scan routine uses `a`, `r0`, `key_state` (key press count). It returns the results using flag `kbhit` for valid key scan and location `key` for scanned key code. The scanning is done in 5 steps. In the first step, 0's are written to all the rows and 1's to the columns, and the port pins are read. 1's on the columns indicate no key press. In the next step, first row is made 0, and 1 is written to the other rows as well as column, the port pins are read, and column shorted to the row is determined. The same process is repeated for the other three rows in the three subsequent steps. For valid key press, in the 2^{nd} to 5^{th} steps, only one column should be found as zero. If no key is found, the routine returns with `key_state` = 0. If a valid key is found, the key code is determined. If `keystate` = 0, it is incremented. If `key_state` = 1, it is incremented, flag `kbhit` is set, and the key code is placed in `key`. If `key_state` = 2, no action is taken.

Thus a key is registered when the code remains valid in two consecutive scan cycles and then onwards it is ignored. It is to be noted that the two consecutive key codes need not be equal, the accepted code is the second code. An algorithmic flowchart of the routine is given in Fig B.1. The routine has been practically tested and works satisfactorily.

B.2 Key Functions

Keys named "Tone Type", "Noise Type", "Tone Dur.", "Mode A/M", and "Air/Bone" will be used only during the initialization. Once all parameters/modes are selected, key "Ok" is pressed. This ends initialization routine and starts with the actual test routine. The frequency of tone and level of tone and noise is set. "Tone On" key is pressed to present tone to the subject. While tone is being presented, only "Tone Off" key is accessed. All other keys do not affect the operation. If "Tone Off" key is pressed, tone is interrupted and then all other keys are also accessible.

Tone Type. This key will be used at the time of initialization of tone presentation to select the tone type.

Freq. Up. Whenever this key is pressed, frequency of tone is increased. Display shows the next frequency with a level of tone initialized at 30 dB HL.

Freq. Down. This key is similar to the 'Frequency up' key except that when ever this key is pressed frequency of the tone decreases.

Ok. This key is used to confirm the message displayed and to proceed further.

Noise Type. This key will be used during the initialization to select the noise type. It may be wide band (WB), narrow band (NB), or no noise (--).

Tone Up. This key is used to increase tone level in 5 dB steps

Tone Down. This key is used to decrease tone level in 5 dB steps

Save. This key is used to save the hearing threshold for the selected frequency into the RAM.

Tone Dur.. This key is used during initialization to select the duration up to which the tone will be presented. Four options are available viz. 2 sec, 3 sec, 4 sec, and continuous. When this key is pressed, the present tone duration will be displayed. To change the tone duration, the same key has to be repressed.

Noise up. This key is used to increase the level of masking noise in 5 dB steps.

Noise down. This key is used to decrease the level of masking noise in 5 dB steps.

Air / Bone, Cancel. This key decides mode of conduction during test. Once, the test is over, the same key is used for another purpose. In the print/display routine, cancel key is pressed to come out of that option.

Mode A/M. This key is pressed to select the mode of operation of the audiometer, either auto or manual.

Tone on. This key is pressed to present the tone to the subject. The tone type, duration, and level will be preselected and according to that the tone will be presented. When the tone is on, a message ‘PR’ is displayed at the bottom right corner.

Tone Off. This key is pressed to interrupt the tone before the tone on duration has lapsed. ***Recall/Print***. This key is used to display the results stored and to transfer the results to the serial device. When this key is pressed, a message ‘Display Results?’ will be displayed. If this key is repressed, message ‘Print Results?’ will be displayed. To confirm the message, “Ok” key has to be pressed. If display mode is selected, all the threshold values are displayed starting with value corresponding to lowest frequency at the upper left corner. To proceed press “Ok” key. It enters into the printing mode asking for printing. If print mode is selected, the entire set of data will be transferred to the serial device, and ‘Printing’ will be displayed.

Cancel. This key is pressed to come out of display or print mode.

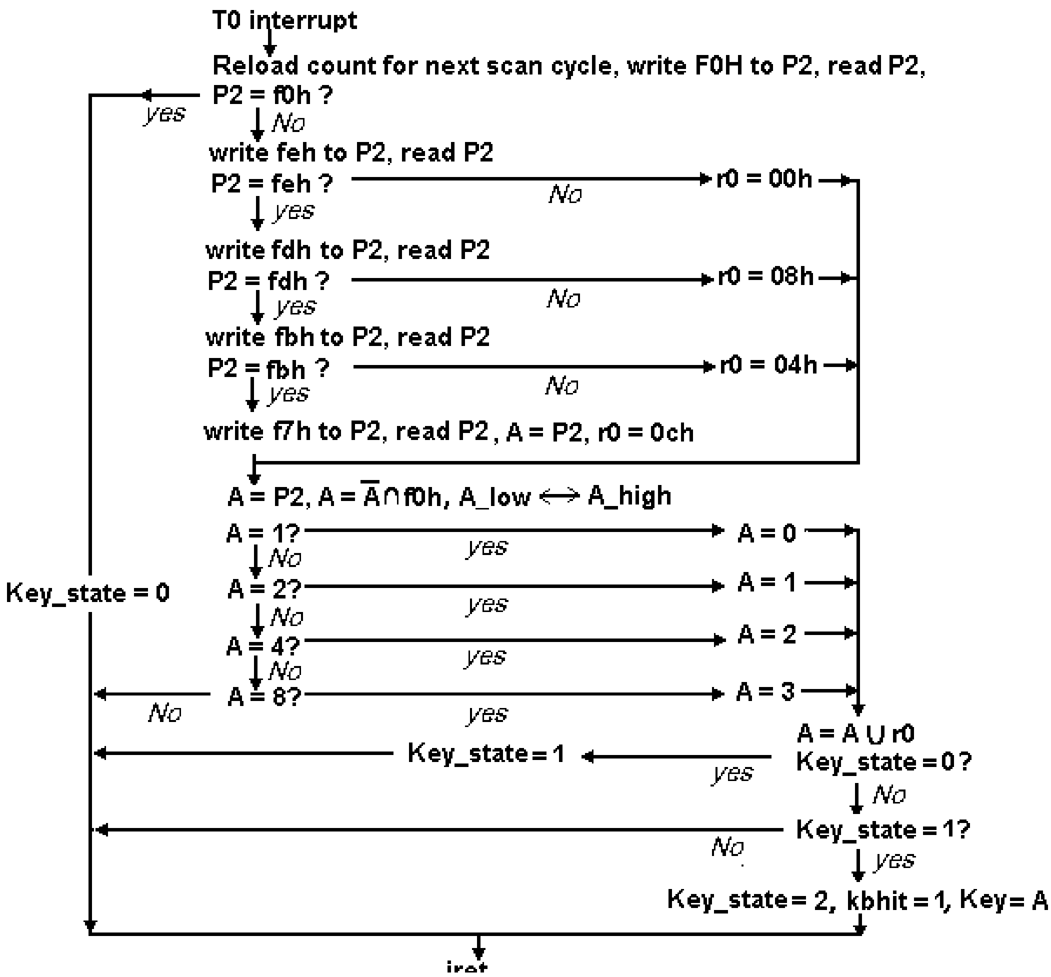


Fig B.1 Interrupt sequence routine for key scan

Appendix C

C.1 Component List for PCB1 (Oscillator, attenuators, masker, display, keypad interfaces, serila port)

Component designator	Part Number/value	Component description	Quantity
C2, C5, C8, C11, C46	0.1 μF/50V	Capacitor (electrolytic)	5
C15, C28, C30	47 pF	Capacitor (Ceramic)	3
C19	10 nF	Capacitor (Ceramic)	1
C36	470 pF	Capacitor (Ceramic)	1
C31, C32	47 μF/63V	Capacitor (electrolytic)	2
C48, C4, C3	10 μF/63V	Capacitor (electrolytic)	3
C21	2.2 nF	Capacitor (Ceramic)	1
C1, C3, C12, C34, C35, C36, C37, C42,C43, C43, C44,C45, C46, C50, C51, C54, C57, C58, C59, C60	0.1 μF	Capacitor (Ceramic)	20
CN1, CN2, CN6, CN9, CN10, CN11		2 pin connector	6
CN4, CN8		3 pin connector	2
CN3		8 pin connector	1
CN5		15 pin connector	1
D1	IN 4148	Diode	1
D2, D3	2.1 V Zener	Zener diode	2
R1, R3, R7, R4, R12, R18, R23	1 kΩ, ¼ W	Resistor	8
R8	300 Ω, ¼ W	Resistor	1
R2	5.6 kΩ, ¼ W	Resistor	1
R5, R13, R15, R16, R25, R17, R32, R21	10 kΩ, ¼ W	Resistor	8
R9, R10, R11	270 Ω, ¼ W	Resistor	3
R14	100 Ω, ¼ W	Resistor	1
R13	2.2 kΩ, ¼ W	Resistor	1
R20	150 kΩ, ¼ W	Resistor	1
R22, R26, R38, R24	22 kΩ, ¼ W	Resistor	4
R31	1.2 kΩ, ¼ W	Resistor	1
R29	3.3 kΩ, ¼ W	Resistor	1
R33	270 kΩ, ¼ W	Resistor	1
P2	10 kΩ	Potentiometer	1
P4	50 kΩ	Potentiometer	1
U1	MAX 232	RS232 Driver	1
U2	AT 89C52/89C55	Microcontroller	1

U3, U4, U5	AD7111	Logarithmic D/A	3
U6, U7, U8, U11, U12, U14, U18	LM356	Op-amp	7
U9	C7555	Timer	1
U13	CD 4030	XOR gate	1
U15	LMF100	SC Filter	1
U16, U17	CD 4066	Analog switch	2
U13, U19, U20	CD 4015	Shift-register	3
XTAL	24 MHz	Crystal	1
	ODM 16216	LCD 16 char. × 2 lines	1
		4 × 4 Keypad	1

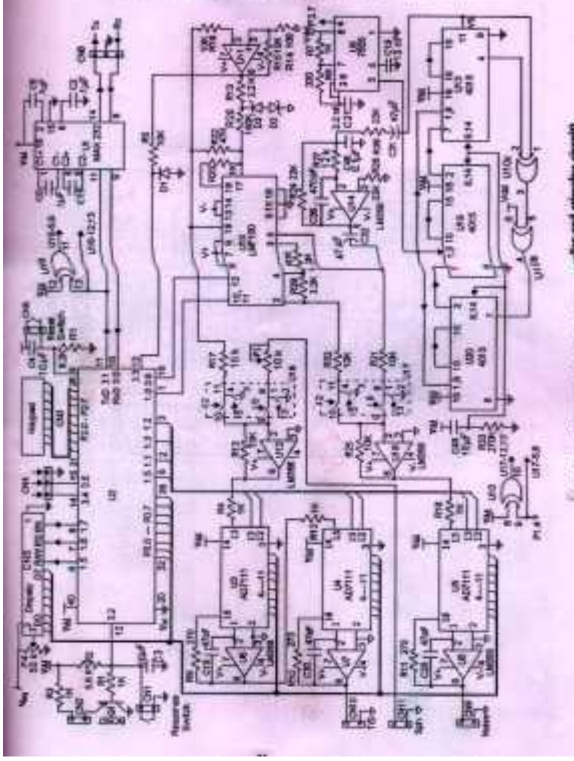
C.2 Component List for PCB2 (Speech pre-amp, power amplifiers, self test circuirs)

Component designator	Part Number/value	Component Description	Quantity
C1	470 nF	Capacitor (Polyster)	1
C2, C12, C16, C40, C50, C51	1 μF/63V	Capacitor (electrolytic)	4
C3, C4, C5, C7, C9, C10, C11, C13,C15, C17, C23,C24, C25, C26,C27, C28, C29, C30, C31, C32, C33 C34, C35, C36, C37 C38, C39, C41, C52 C62	0.1 μF	Capacitor (ceramic)	28
C6, C14, C18, C19, C20, C21, C22	1000 μF/25V	Capacitor (electrolytic)	7
C8, C54	10 μF/63V	Capacitor (electrolytic)	1
C52, C53	0.1 μF	Capacitor (ceramic)	1
C55	0.01 μF	Capacitor (electrolytic)	1
C61	2.2 μF	Capacitor (electrolytic)	1
CN1, CN2, CN5, CN9, CN10, CN11		2 pin connector	6
CN8, CN3, CN4, CN7		3 pin connector	4
D1, D2, D4, D5	IN 4148	Diode	4
D3, D6	ICL 8069	Ref. Diode	2
R1, R3, R7, R14, R17, R19, R25, R30, R34, R38, R55	10 kΩ, ¼ W	Resistor	10
R2, R6, R20, R22	4.7 kΩ, ¼ W	Resistor	4
R4, R5, R18, R21	39 kΩ, ¼ W	Resistor	4
R8, R26	68 kΩ, ¼ W	Resistor	2
R9, R27	47 kΩ, ¼ W	Resistor	2
R10, R23	3.3 kΩ, ¼ W	Resistor	2
R11, R24	330 kΩ, ¼ W	Resistor	2
R12, R28	6.8 kΩ, ¼ W	Resistor	2
R13, R29, R33, R37	100 kΩ, ¼ W	Resistor (MFR)	4
R15, R31, R35, R39	150 kΩ, ¼ W	Resistor	4
R16, R32, R36, R40	12 Ω, ¼ W	Resistor	4
R59, R60	1 kΩ, ¼ W	Resistor	2
R41, R42, R43,	18 Ω, ¼ W	Resistor	6

R44, R45, R46			
R51	560 kΩ	Resistor	1
R52	200 Ω	Resistor	1
R53	1 MΩ	Resistor	1
R54	330 Ω	Resistor	1
R57	3.9 kΩ	Resister	1
R58	3.3 kΩ	Resister	1
R56	22 kΩ	Resister	1
P1, P2	50 kΩ	Potentiometer	2
P3, P4, P5	100 kΩ	Potentiometer	3
P6, P7	500 kΩ	Potentiometer	2
P8	1 kΩ	Potentiometer	1
U9 (Socket)	10 LED strip	LED display bar	1
U1, U6	LF 347	Quad Op-amp	2
U2, U5	LM 1877	Power Amplifiers	2
U3, U4	LM 311	Comparator	2
U7	LM 381	Preamplifier	1
U8	LM 3915	Logarithmic dot/bar display driver	1

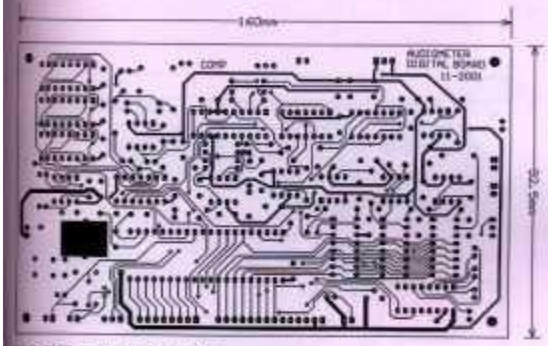
Appendix D

Circuit diagram and PCB layouts



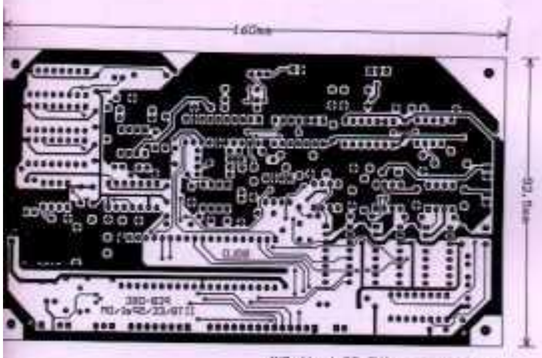
D.1 Circuit Diagram for PCB1 (microcontroller and stimulus circuit)

B.2 Component side layout of PCB1

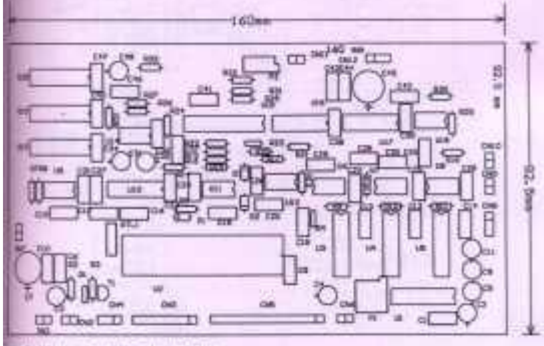


COMP SIDE - AUD-DC-1 11/2K1

D.3 Solder side layout of PCB1



B.4 Component placement of PCB1



COMP SILK - AUD-DC-1 11/2K1

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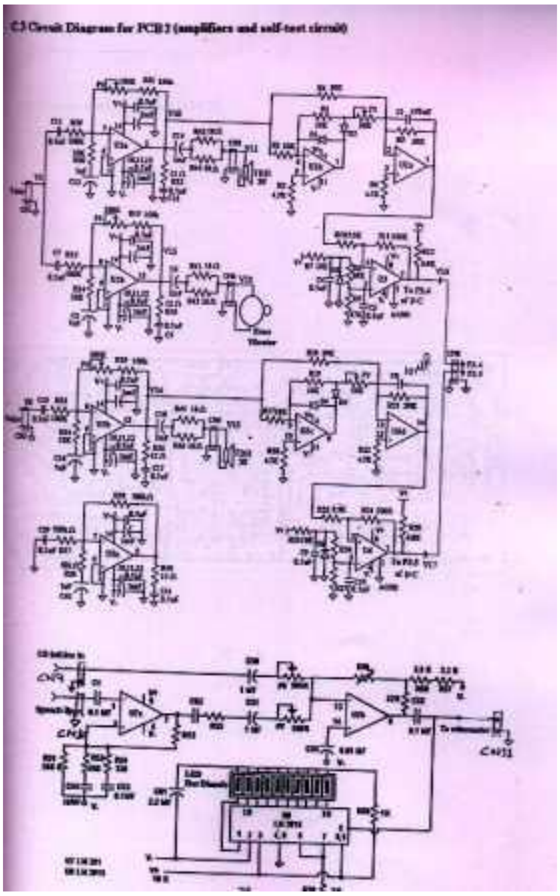
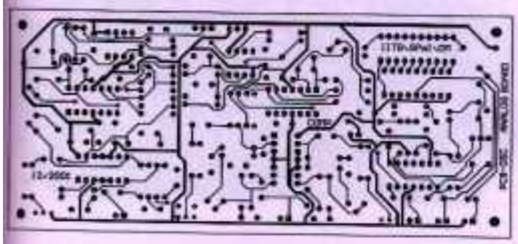
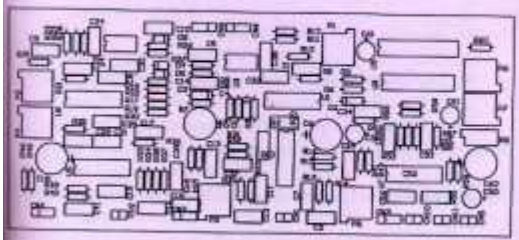

C3 Circuit Diagram for PCB 2 (amplifiers and self-test circuit)

D.6 Component side layout of PCB2



D.7 Solder side layout of PCB2

D.8 Component placement of PCB2



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