A MICROCONTROLLER BASED AUDIOMETER

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by

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Guide

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ABSTRACT

An audiometer is an electroacoustic instrument for quantifying hearing impairment. Using it, the test tones of different frequency and level are presented and hearing thresholds are determined on the basis of patient's response. The objective of this project is to develop a portable audiometer, which can be used in mobile clinics and even in rural areas. 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 provision for tone decay test and SISI test. 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 with option of manual and automated audiometry, and it carries out a self test of the output levels.

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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)
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

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Chapter 1

INTRODUCTION

The audiometry is a technique for measuring hearing sensitivity of a person and it is used to estimate the degree of the hearing loss. It helps in assessing the nature, degree, and probable cause of the hearing impairment. In this technique, some 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 severity of the hearing loss can be estimated and the best-suited medical treatment or sensory aid can be given. There are different audiometry procedures depending on the stimuli used. An audiometer is an electroacoustic instrument, for quantifying hearing impairment by carrying out audiometric tests.

Audiometers can be of different types, depending upon the frequency range, range of acoustic output, mode of acoustic presentation, masking facility, and types of acoustic test stimuli. Chandrakant Singh and Ashish Kothari, as part of their M. Tech. dissertations at IIT Bombay, worked towards developing a microcontroller based diagnostic audiometer with a provision for automated audiometry and computer/printer interface [1].

The object of the project is to study the earlier development thoroughly and develop a fully functional diagnostic audiometer by overcoming the shortcomings of the earlier prototype. The instrument will be basically for pure tone audiometry. It will have other tests like SISI and tone decay test, which help in further diagnosis of the hearing impairment.

In the second chapter, various audiometric techniques, details of pure tone audiometer, and need for an automated portable pure tone audiometer have been discussed. In the third chapter, design approach of the system and work done by Kothari [1] is described. Fourth chapter provides system hardware description. In the fifth chapter detailed software description is given. Specifications of the existing system and further work planned are discussed in the sixth chapter.

Chapter 2
AUDIOMETERIC TECHNIQUES

Audiometry is used to measure hearing ability and to evaluate hearing pathology in order to provide diagnostic information and rehabilitation. 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 Audiometry

The various measurement techniques can be broadly grouped into two categories, "subjective" and "objective". Subjective tests rely on the patient performing a task according to the instructions. The objective tests only require co-operation from the patient in the attachment of measuring electrodes or probes [2]. In subjective tests, routine clinical assessment includes measurement of hearing sensitivity as a function of frequency, measurement of dynamic range of intensity, assessment of speech reception as a function of speech intensity, etc.

The stimuli can be presented either by air conduction or by bone conduction. For air conduction test, audiometric headphones or loudspeakers (free field) are used, whereas in bone conduction test, a special vibratory transducer is placed on the forehead or over the mastoid bone. In this test the tone is directly coupled to the inner ear, bypassing the outer ear and middle ear. Therefore, bone conduction threshold is a function of the inner ear pathology only and is not affected by problems in the outer ear and middle ear. Tests using both air and bone conduction help in diagnosing the source of hearing losses. Some of the audiometric tests are described below [2].

2.1.1 Pure Tone Audiometry

Pure tone audiometry tests are carried out primarily to obtain air conduction and bone conduction thresholds of hearing [2] [3]. The frequency of test tone presented is in the range of 250-8000 Hz. 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 [2]. 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 be 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 this is known as sound pressure level (SPL). However, in audiometry the sound level of pure tones is given in dB by taking the average hearing threshold of normal hearing young adults as the reference, and this 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 [4]. 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 pure tone audiometry, single frequency stimulus at some presumed level is presented to the patient. The tone can be presented using audiometric headphones, or through loudspeakers (free field). Initially a pure tone of 30 dB HL is presented to the subject. If the response is positive, then the tone level is decreased in step of 10 dB till the patient does not give response. On the other hand, after presenting the tone the first time, if the patient does not hear it, the level is raised in steps of 10 dB until it is heard for the first time. Once, the response is positive, the tone is decreased in steps of 10 dB till the patient does not hear it. Now, tone is raised in steps of 5 dB till the response is again positive. The minimum presentation level at which the subject responds at least 50% times (minimum 3 responses out of 6 presentations) is taken as the hearing threshold [4].

The test results are represented in the form of a plot of hearing threshold as a function of frequency, and this plot is known as audiogram. Different shapes of audiograms are associated with different types of loss [4]. A typical audiogram is shown in Fig 2.2

2.1.2 Tone Decay Test (TDT)

Tone decay test is used to diagnose the sensorineural deafness [4]. In this test, 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 tone intensity is incremented by 5 dB and again tested for the same. The lowest intensity for which patient is able to hear the tone about 1 min. is considered as threshold for tone decay test. The tone decay of 30 dB and above is considered as severe tone decay and it is considered as suggestive of retrocochlear lesion [4].

2.1.3 SISI Test

The SISI (short increment sensitivity index) test determines the capacity of a patient to detect a brief 1 dB increment in 20 dB suprathreshold tone (also called as carrier tone) at various frequencies. While performing this test a brief increment in intensity is provided at 5 s intervals, and patient is asked to respond if he senses the increment in intensity. Twenty such increments are presented and the number of increments, the patient is able to recognize correctly is noted. The result of this test can range anywhere between 0 to 100%. Normally the score is 30 to 40 %. The scores between 70 to 100 % indicate cochlear pathology, whereas the score of 0 to 20 % suggest retrocochlear pathology [4].

2.1.2 Speech Audiometry

In speech audiometry, a person's ability to hear and understand speech, and thereby the integrity of the auditory system, is assessed [2] [4]. The main use of speech audiometry is in identification of neural type of hearing loss. In this both the reception and discrimination of speech is impaired more markedly than any other type of hearing loss. It is also useful in finding the assesment of central auditory function. It comprises of "speech reception threshold " and "speech discrimination score" test.

Speech reception threshold (SRT) is the lowest hearing level in dBHL, at which 50% of a list of spondee words (two-syllable word with equal stress on each syllable) are identified correctly by the 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 the at succesively 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. It is generally 2 dB lower than average of pure tone hearing level thresholds at 500 Hz and 1 kHz.

In "speech discrimination score test", list of monosyllabic speech discrimination words are presented at 35 dB above the speech reception threshold, over headphones for each ear and the subject is requested to repeat back what the patient heard. The percentage of the total number of words presented which the patient is able to identify correctly gives the speech discrimination score (SDS). In case of neural lesions, SDS is very poor [4].

2.2 Masking in Audiometry

In air and bone conduction audiometry where sound is applied to one ear, the cochlea of the other ear is also stimulated by transmission through the bone of the skull. This 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 [4]. 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.

To offset the risk of cross hearing, masking is done in the non-test ear by presenting a 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 [4]. 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 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 [11]. 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. Broad band noise was preferred earlier because narrow band noise requires a band pass filter whose center frequency is the same as the test tone frequency.

2.3 Pure Tone Audiometer

A pure tone audiometer consists of (a) tone generator (test stimulus), (b) noise generator (masker), (c) two attenuator/equalization circuits for controlling the stimulus and masker levels in dBHL scale, d) two power amplifiers, e) appropriate controls, displays, and interfaces, f) means of presenting the acoustic or vibratory test signal to the patient's test and non-test ears, g) means of obtaining patient response. A block diagram of basic audiometer is shown in Fig. 2.1

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 [5].

Equalization circuit provides frequency dependent attenuation in order to calibrate the output sound level in dB HL for particular output device used (headphone, loudspeaker, 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 the sound (relationship between sound level output by the device and voltage input applied to it.) The attenuator also called, 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 that the output sound level should be within ± 3 dB of the indicated value. Equalization circuit is switched by frequency selector switch.

The noise generator for masking 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 noise level should be within ± 5 dB of the indicated value [5]. Equalization and attenuator circuits are similar to that of stimulus however attenuation range is generally smaller.

The output power amplifier for tone and masking noise should have low distortion and high efficiency. The audiometer should have some means of switching the signal between output devices for the two ears.

2.4 Microprocessor Based Audiometer

Basic blocks of pure tone audiometer are shown in Fig. 2.1. A conventional audiometer has dials or knobs with calibrated scale for frequency selection and for tone and masking noise level selection. 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 size of instrument is also large. It is to be noted that the output sound level has to be calibrated in HL, and the electrical output for a given HL varies with frequency.

The tone generator can be an R-C oscillator. Different frequencies are obtained by switching different values of resistors or capacitors. More the frequency steps required, more would be the passive components, switches, and control lines. To get frequency dependent attenuation the instrument should have number of tone generators, with levels adjusted in accordance with the desired frequency or the tone generator can be a single circuit, with equalizing attenuators switched by the frequency selector knob. Further output level adjustments are needed for different type of acoustic devices, because each device has different sensitivity and frequency response.

For generating the masking noise, a low frequency saw tooth waveform was used earlier. However it is effective at low frequency but not at high frequency. Another drawback was the harmonic beating with the test tone frequency [3]. Discharge tubes as a noise source had a better performance than the earlier ones. Diodes have also been used as a noise source, but, the circuit needs high gain amplifier, which may result in 50 Hz power line pick up [3]. All these techniques had a possibility of generating nonuniform spectrum of masking noise.

The advancement in technology has made the various switching tasks simple, flexible, and noise free. The newer generation audiometers use analog switches controlled by a digital processor (a microprocessor/PC) for frequency and tone selecting. 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 audiometers also offer automation of audiometric testing, storage of test results, and audiogram printing. Increased accuracy and precision removes the need for frequent calibration. In PC/microprocessor based audiometers, tones may be generated by various methods, e.g.

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 through the D/A converter of a PC bus 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. Alternatively, stimulus and masker can be output using sound card of a PC and attenuator can be controlled through serial or parallel port. This requires an external interface with a microprocessor / microcontroller to communicate with the PC.

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With a large number of bits in the output D/A converter, level control may be obtained by software scaling of the waveform.

3) The generation of stimulus tone and masker is done 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).

These audiometers may 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. Advanced audiometers also provide the option of frequency and amplitude modulated tones and facility for carrying out some of diagnostic tests: tone decay test, SISI (short increment sensitivity index) test, etc.

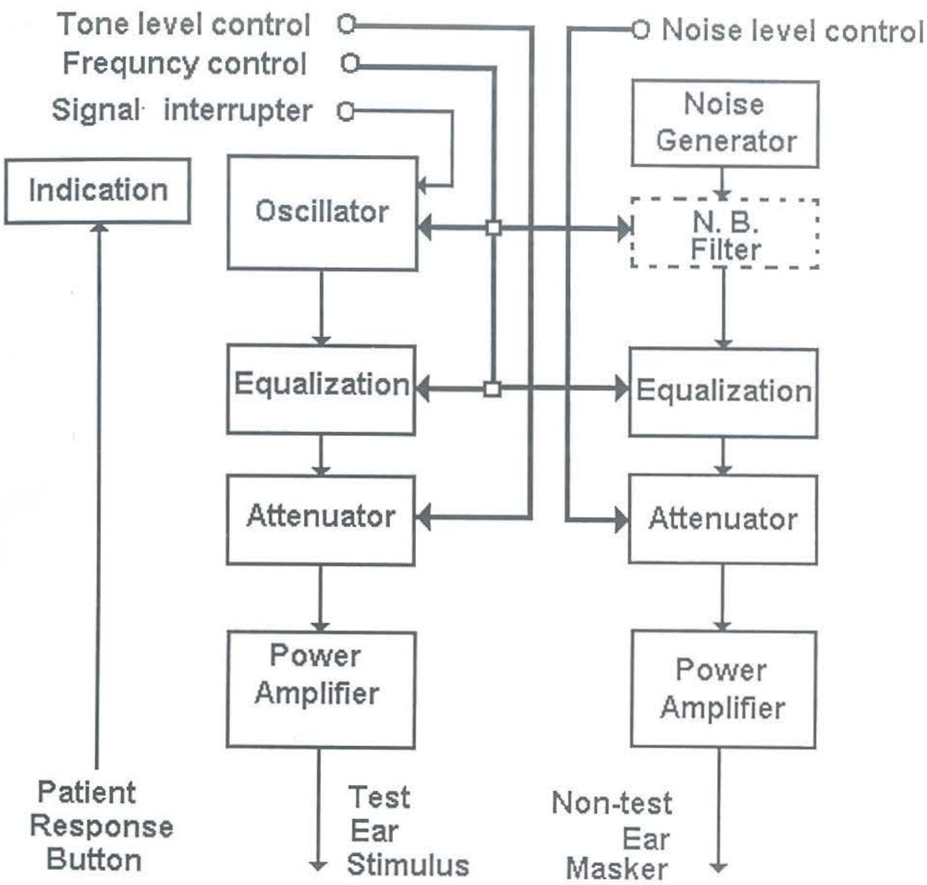

Fig. 2.1 Block Diagram of an audiometer

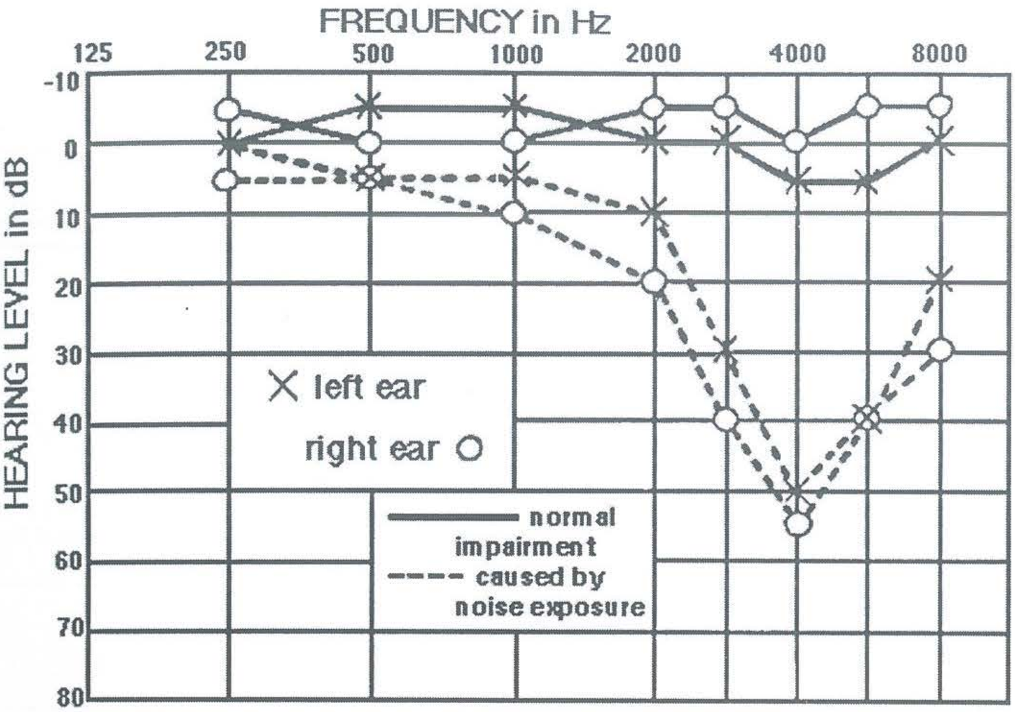

Fig 2.2. A typical audiogram [4]

Chapter 3
DESIGN APPROACH

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

(1) 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) Provision for storing the audiometric test results.

(6) Provision for automated audiometry.

(7) Features of a diagnostic audiometer.

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

3.1 System Specifications

With the above general considerations, the technical specifications for the final system have been selected as follows.

Audiometer type: dual channel microcontroller based audiometer with pure, warble, and amplitude modulated tone stimulus, wide-band/narrow-band masking noise, air and bone conduction facility, and provision for tone decay test and SISI test.

Circuit size: suitable for a compact instrument.

Stimulus: crystal controlled test tone frequencies, with intensity level from 0 dB HL to a maximum value as given below, in steps of 5 dB (as per ANSI standards 1970).

Frequency (Hz)	250	500	1000	1500	2000	3000	4000	6000	8000
$(HL)_{max}$ (dB)	90	100	100	100	100	100	100	90	80

Headphone: type TDH-39

Warble tone: frequency deviation of ± 10% and one sweep per second.

Amplitude modulated tone: amplitude deviation of ± 1 dB and one sweep per second.

Masking noise: broad-band / narrow-band noise, with level sufficient to mask test tones.

Intensity level variation in steps of 5 dB.

Wide-band noise: flat spectrum from 250 Hz to 10 kHz.

Narrow-band noise: center frequency = test tone frequency

Band width = 1/3 to 1/2 octave about center frequency

attenuation rate outside the pass band: 12 dB / octave.

Control and indication: control through membrane keypads and appropriate digital display for various indications.

Operation: software controlled menu driven manual / automated modes and provision for tone decay test and SISI test.

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

Interfacing: serial port (TxD, RxD, and GND at TTL level) at a data transfer rate of 1200 bits per second for downloading the test results.

Self test: provision for internal monitoring of output levels.

Power supply: ± 5 volt (with provision for battery based operation with single 3/6/9 V battery with dc/dc converter, or 230V mains with a power adapter.)

3.2 Design Approach

The following paragraphs summarize the design philosophy of each block in system, as shown in Fig. 3.1.

At the center of the instrument is a microcontroller. 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 [6].

Stimuli in a pure tone audiometer can be generated by an R-C oscillator. Different frequencies can be obtained by switching different values of resistors and capacitors. More the frequency steps required, more would be the passive components, switches, and control lines. Alternatively, the sine wave can also be generated by software, and 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. It was decided to use a switched capacitor filter (SCF) based oscillator, 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. This means that the tone frequency will be 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.

The hearing threshold for the test tone is determined by systematically varying the level to find out the minimum level at which the subject fails to perceive it. Hence, attenuator must be capable of adjusting output sound pressure level from below the normal threshold of hearing to some 100 dB above, and normally in steps of 5 dB [3]. An 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; an additional attenuation of 40 dB is achieved by using a resistive network and analog switches. Amplitude modulated tone can easily be generated using this attenuator.

In early audiometers, a low frequency saw tooth waveform 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 [3]. Discharge tubes as a noise source had a better performance than the earlier ones. Diodes can also be used as a noise source. But, it needs high gain amplifier, which may result in 50 Hz power line pick up [3]. All these techniques had a possibility of generating unequal amplitude 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 the same masking effect as wide-band but at lower sound pressure levels. The noise selection is done by operating analog switches that are controlled by microcontroller.

A response switch is given to the patient, to indicate whether the tone is heard or not. The closure or non-closure of response switch is indicated on the display, depending upon which the audiologist decides the next tone level. 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 lines×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.

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. The tone frequency can be easily changed, by changing the clock frequency. The clock is provided by microcontroller. The tone output is systematically attenuated to get a correct hearing threshold. The tone attenuator consists of an 8-bit logarithmic attenuator chip and a switchable 40-dB attenuator. The attenuator chip gives attenuation of 88.5 dB in steps of 0.375 dB. The output of attenuator is fed to two power amplifiers, one for the headphone and the other for 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. The clock to the filter is same as that of the tone oscillator. Either of the two noises is selected and fed to the attenuator. It is to be noted that the SCF IC selected has 2-integrator filter sections. One section is used for realizing the oscillator, and the other one is used as a band pass filter. Thus both the operations are handled by same chip. The noise is fed to the power amplifier that drives the headphone. The output of power amplifiers for tone (for headphone) and noise is verified by means of the self-test circuit. A push button switch 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 either a computer or a printer.

3.3 Earlier Development

Ashish Kothari [1] and Chandrakant Singh [15] worked as part of their M. Tech. dissertation project at IIT Bombay towards developing a microcontroller based portable diagnostic audiometer having these specifications [1]. However, all the design objectives could not be met in the prototypes developed. In this chapter the work done by Kothari [1] is summarized.

Kothari [1] has developed a prototype of a microcontroller based pure tone diagnostic audiometer, which operates over a frequency range of 250 Hz to 7.5 kHz and acoustic output of 0 to 100 dB HL. It can also generate warble tone having ±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 is made possible by using microcontroller, programmable oscillator, and programmable attenuator. A programmable oscillator is 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 is 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.

3.4 Further Development Needed

The scheme used by Kothari 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 oscillator circuit needs some improvement, it should be able to generate the tone frequency of 8 kHz which is a standard frequency. For that possibility of using higher clock rate should be checked. If feasible, then accordingly software modifications have to be done. It may help in improving warble tone also.

In the tone generation there is generation of "thump" at the time of turning on and off of the tone. This is highly undesirable. So to remove this problem software modification has to be done.

Along with pure tone and warble tone mode, provision for selecting amplitude modulated tone mode should be provided. That needs software implementation. Also it is decided to implement two new tests, which are helpful in further diagnosis of the hearing impairment. These are "tone decay test" and "short increment sensitivity index" (SISI) test [4] as described earlier in section 2.9. This is done by introducing additional software. The whole software has to be modified to make it more modular. The serial interface with PC has to be reviewed and needs some changes. A program has to be developed on PC, which will download the test results from audiometer and display this in the form of audiogram.

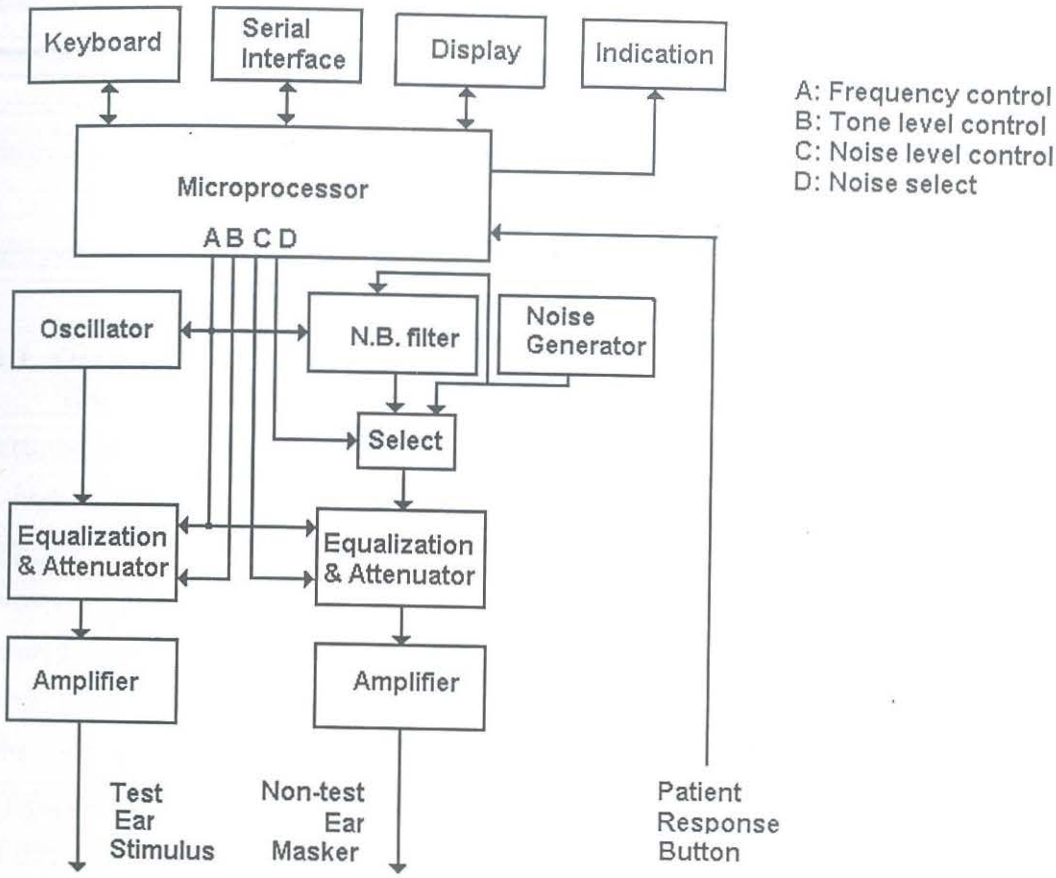

Fig. 3.1 Block Diagram of Microprocessor Based Audiometer

Chapter 4
HARDWARE DESCRIPTION

The block diagram of the hardware of the audiometer is given in Fig. 4.1. All the controlling operations are handled by the microcontroller. These various blocks are tone oscillator, tone attenuator, PRBS based noise generator, noise selector and attenuator, power amplifiers for headphone and bone vibrator, self test circuit for tone and noise, 4x4 matrix keyboard, LCD display, serial interface, response switch, and microcontroller.

This chapter provides hardware details of each block and the interfacing of microcontroller to these blocks, and power supply circuit.

4.1 Oscillator

The oscillator circuit used is a switched-capacitor based quadrature phase oscillator, with the basic principle of tone oscillator as illustrated in Fig. 4.2. The circuit consists of a high-Q band pass filter connected in a positive feedback loop with a hard limiter. Assuming the band-pass filter output V1 to be sinusoidal, the limiter produces a square wave output V2 with a frequency same as that of V1 and amplitude determined by the limiting levels. This square wave is fed back to the BPF, which filters out all the harmonic components and provides sinusoidal output. Due to high Q nature of band-pass filter, the oscillator frequency gets set to the BPF center frequency f_0. The amplitude of V1 depends on the amplitude of the input square wave V2 and gain of BPF. Higher the Q of BPF (selectivity) better the purity of the output sine wave [10].

Band pass filter uses two-integrator loop. 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

$$H_{bp}(s) = \frac{V_{bp}}{V_i} = \frac{\alpha s/\omega_0}{(s/\omega_0)^2 + (1/Q)s/\omega_0 + 1}$$

If V_i is a square wave of $2V_m$ peak-to-peak voltage with frequency f_0, it has fundamental component of $(4/\pi)V_m$ peak to peak. Output V_{bp} is a sinusoid with frequency f_0 and its peak-to-peak voltage is given as

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

This oscillator realization is implemented using switched capacitor (SC) filter IC. The switched capacitor filter (SCF) is based on the principle that a capacitor switched

Chapter 4

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This 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 capacitors [7]. Singh had used SCF chip MF 10, with $f_o \times Q = 200$ kHz. This has been replaced by LMF 100, with $f_o \times Q = 1.8$ MHz [8]. This helps in overall performance improvement, because this chip can be used at higher Q for a given clock rate. The block diagram of LMF 100 is shown in Fig. 4.4. It has two identical sections. Each of them consists of two integrators (SC based), a summer (SC based), and an op-amp. Thus, it is possible to implement circuit with various second order functions.

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 U6. Section A is used for realizing narrow band filter for noise, as described later in Section 4.3. The circuit provides two outputs of equal amplitude but 90° phase apart. Hence, it is also known as quadrature oscillator [7]. 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 U8 (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 (D3, D4) connected back to back. This ensures stability of the BPF output. Peak-to-peak voltage of the square wave output V2 is

$$V_{2\text{p-p}} = V_{D3} + V_{D4} + V_{Z3} + V_{Z4}$$

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

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

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 U6, 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. 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 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 R31 and D5 (as shown in Fig. 4.5). The output P3.3 is connected to microcontroller.

4.2 Attenuator Circuit for Tones

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. From the Table A.1, 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 tones has been realized using a logarithmic programmable attenuation for 88.5 dB and a switchable 40 dB attenuator. An 8-bit logarithmic D/A converter AD 7111 from Analog Devices is used as a programmable attenuator. It gives attenuation of 88.5 dB, with a resolution of 0.375 dB [9?]. 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. 4.5. The current output of AD 7111 is converted to voltage by using op-amp U16d (LF 347). Resistors R3 and R4 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. Capacitor C7 is 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.

The test tone level should range from 0 dB HL to 100 dB HL. Since, AD 7111 can attenuate only up to 88.5 dB, additional switchable attenuation of 40 dB is required. It has been realized using op-amp U16a, resistors and analog switches from IC CD 4066. This IC requires bipolar supply and control voltages for handling bipolar signals. However, attenuator circuit has been devised in such a way that bipolar control and supply voltages are not needed. In this circuit switch U17a and U17d are controlled by complementary signals i.e. either of the two is on at a time. When switch U17a is on and U17d is off, op-amp U16c (LF 347) acts as a unity-gain inverting amplifier. In the second case, when switch U17d is on, U17a is off, gain of the amplifier is 1/100, giving an attenuation of 40 dB. In both the cases, voltage across switch U17a and U17d is always zero, irrespective of being on or off. This makes unipolar supply and control signal capable of making switches on and off.

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

4 3 Noise Generator

The block diagram of noise generator scheme is shown in Fig. 4.7. A low pass filtering of the digital output of a pseudo random binary sequence (PRBS) generator gives a band limited white Gaussian noise [7]. 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.

A 15-bit PRBS generator based on the maximal length feedback shift register is used. It has an XOR feedback with tapping at bit no. 14 and 15, and output of the PRBS is taken from the 15th bit. The output of the PRBS repeats after every $2^{15}-1$ clock pulses. The clock frequency used is 200 kHz. Thus, we get a noise with flat spectrum in the range of $f_1 - f_2$, where

$$f_1 \approx 10\, f_{CLK} / (2^{15}\text{-}1) \approx 60 \text{ Hz}$$

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

The circuit is shown in Fig. 4.8. Two dual 4-bit shift registers IC CD 4015 (U12 and U14) and one quad XOR gate IC CD 4030 (U13a and U13b) is used to make the PRBS generator. The tapping is made at 14^{th} and 15^{th} bit and output is taken from 15^{th} bit ($Q3_B$ of U14) thus making it a 15 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. An astable multivibrator is used to generate the clock that is applied to the PRBS. The circuit consists of R15, C15, and NAND gate IC CD 4011 (U4b). With R15 = 15 k and C15 = 22 nF, we get clock frequency of about 200 kHz. In order to save power in shift registers, a clock control is provided (by using U4a) so that whenever masking is not required the clock is not applied to PRBS generator. This is done by pulling low the input to U4a, marked P3.7.

To get wide band white noise, a second order low pass filter built using op-amp U11 with a 10 kHz cutoff frequency and unity gain is used [12]. 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, the other half of the LMF-100 (U6a) 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 $\Delta f / f_o$ = 0.363, which gives bandwidth of 0.53 octave.

4.4 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 U4a, labeled P3.7, low. If masking is selected, P3.7 is made high. Either the broad band or narrow band noise is selected. Four analog switches in IC (CD 4066) U15 are used for selection of noise. Control signals of U15a, U15d are complementary to control signals of U15b, U15c. U13c (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 U15c, U15b on and U15a, U15d off. This passes wide band noise through unity gain inverting op-amp U16b (LF 374). 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. Additional switchable attenuation of 40 dB is not required for noise. The noise selector and attenuator circuit works with single 5 V supply.

4.5 Power Amplifiers

Earlier Singh had used discrete transistorized class B push-pull amplifier [1]. 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 [8].

Three amplifiers are needed; one each for producing tone and noise to the headphones, one for pure tone to the bone vibrator. 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 [13], and the electrical voltages needed for driving the headphone are given in Table A.1. The maximum peak-to-peak voltage needed is 2.2 V. The sensitivities of headphone and bone vibrator are different. An artificial mastoid for calibration of bone vibrator was not available, and hence biological calibration was carried out for finding the driving voltage for the bone vibrator (Oticon 70127). This was done with the help of normal hearing subjects. The hearing threshold for any one of the ears is determined for air conduction, over a defined range. The same procedure is repeated for bone conduction by placing the vibrator behind the same ear. It was found that, the hearing thresholds measured as electrical driving voltage obtained using bone vibrator were 45 to 50 dB more than those obtained using the headphone. Since, the maximum output voltage from power amplifier is restricted to 12 Vp-p for 8 ohm load, the output range for bone conduction mode is 45 to 50 dB less than that for the air conduction.

The circuit for headphone and bone vibrator tone amplifier is shown in Fig. 4.10(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. 4.11. 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.

4.6 Level Monitoring Circuit

The task of the level monitoring 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.

4.7 Keypad

A 4×4 matrix keypad is used. It is interfaced to the microcontroller through its port 2. 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 as shown in Fig. 4.13. The scanning procedure is explained in detail in appendix B.

4.8 Display

The display used is 16 characters × 2 lines display model LCD ODM-16216S from Oriole [10], as shown in Fig. 4.14. 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. 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.

4.9 Serial Interface

The instrument can be interfaced with a serial device through the serial interface. It consists of an inverter which buffers the transmit and receive signal lines of the microcontroller, as shown in Fig. 4.15. Normal TTL levels are used for the data transfer.

4.10 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 $\overline{INT0}$, as shown in Fig. 4.16. The pull up resistor R2 of 5.6k is provided for pulling the pin P3.2 high and charging C4. Capacitor C4 of 2.2 μF provides debounce of about 50 ms to the response switch.

4.11 Microcontroller Interfacing

Digital data and control of all the circuit blocks described in Section 4.1 to 4.10 are interfaced to the microcontroller. A microcontroller should have sufficient program 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 AT 89C55 microcontroller from Atmel, having 20 K 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 [8]. Another microcontroller AT 89C52, from the same series has flash EPROM of 8 K bytes, which is enough for the program, which is around 4 K Bytes. Hence, 89C52 is preferred as it provides same performance at lower cost.

The pin assignments of the microcontroller are given in Table 4.1. Fig. 4.17 shows the block diagram of the system along with the interfacing details. Port 0 is multiplexed between display, tone and noise attenuators' data bus. Port pins P1.1 and P1.2 are used for selecting ($\overline{CS}$) of tone and noise attenuator chip respectively. The extra 40-dB attenuator for tone is controlled by P1.3.

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 $\overline{INT1}$ is used for warble tone generation. Response switch is connected to external interrupt pin $\overline{INT0}$.

Port pin P1.5 is common for $R/\overline{W}$ of display and $\overline{WR}$ of both attenuator chips. Control signals, RS and $\overline{EN}$, 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.

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.

4.12 Power Supply

The audiometer needs ± 5 V DC power supply. This is obtained by full wave rectifying the mains AC. The circuit is shown in Fig 4.18. A 9-0-9 transformer is used to step down the AC mains voltage. This stepped down voltage is full wave rectified using bridge rectifier. Hence output of rectifier will give positive as well as negative voltage with respect to center tap. The voltages are filtered by two electrolytic capacitors of 1000 μF and two 0.1 μF ceramic disc capacitors are placed in parallel with the electrolytic capacitors. These voltages are regulated using 3 separate regulators. Two of them are LM 7805 (5V regulator) and one LM 7905 (-5V regulator). Out of these three regulators, one 5V regulator will give 5V supply for digital circuitry, while others two will act as dual supply for analog circuit. The supply for analog and digital circuit are taken separate to eliminate the modulation of analog supply due to switching in digital circuit.

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 (U6- 10,11)
P1.1	$\overline{CS}$ Of tone attenuator (U3- 12)
P1.2	$\overline{CS}$ of noise attenuator (U10-12)
P1.3	40 dB attenuator control (U17- 12 and U13- 5)
P1.4	Wide / narrow band noise selection (U13-9 and U15-5,6)
P1.5	R / $\overline{W}$ of display and $\overline{WR}$ of both attenuators (CN2-5, U3-13, U10-13)
P1.6	RS of Display (CN2-4)
P1.7	En of Display (CN2-6)
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 (U4-10)
P3.1	Serial interface TxD (U4-13)
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 (CN11-1)
P3.5	Input from self-test of noise circuit (CN11-2)
P3.6	50:100 clock control (U16-12)
P3.7	PRBS clock control (U4-1)

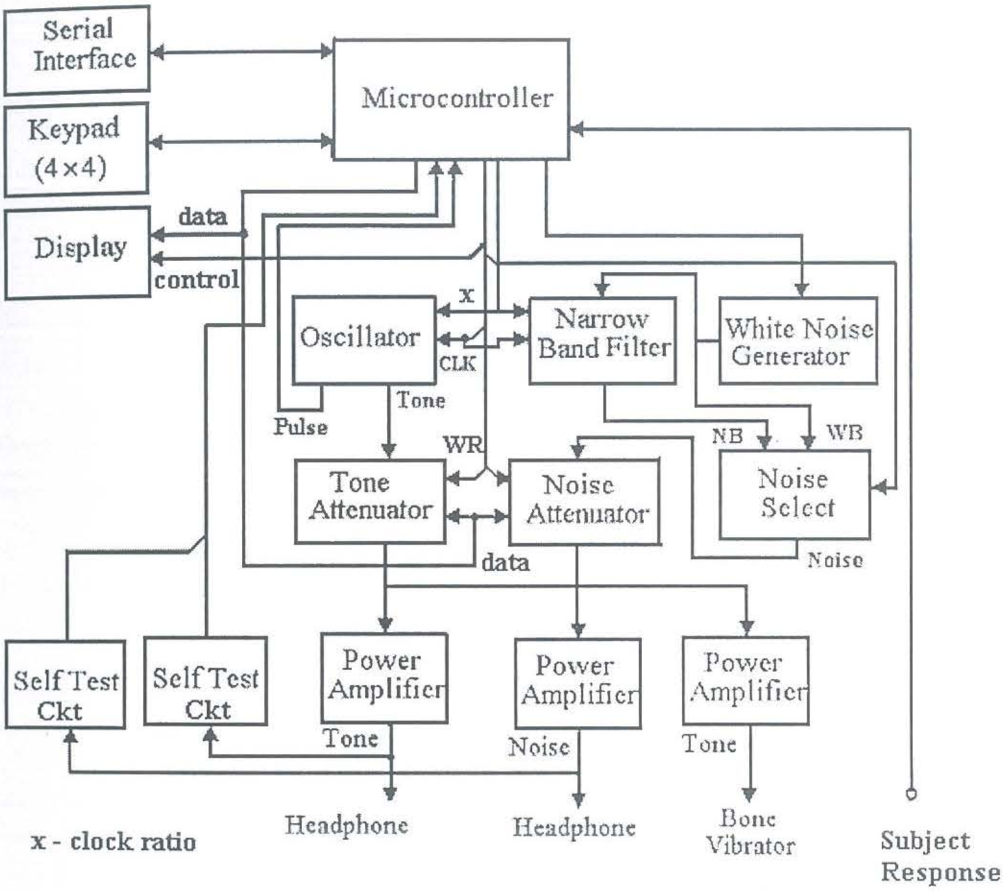
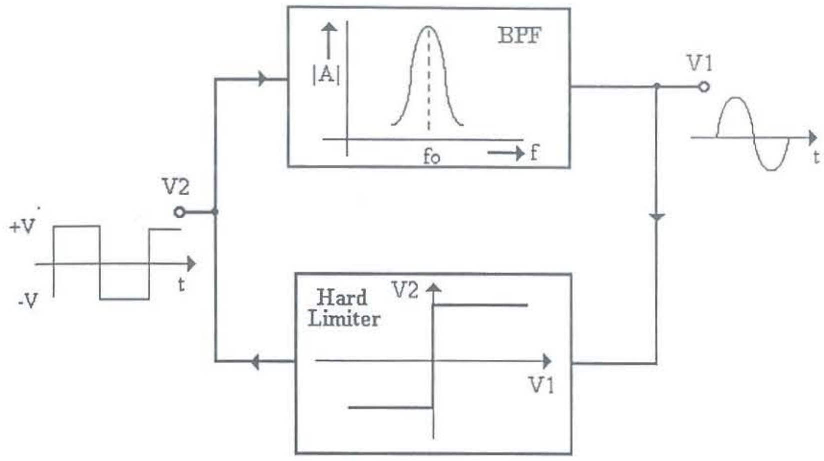


Fig. 4.1 Block diagram of system hardware [1]



4.2 Block diagram for quadrature oscillator

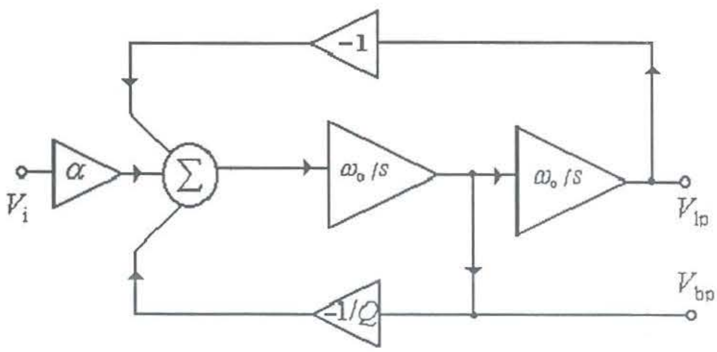


Fig. 4.3 Band pass filter realization using two non-inverting integrator loop.

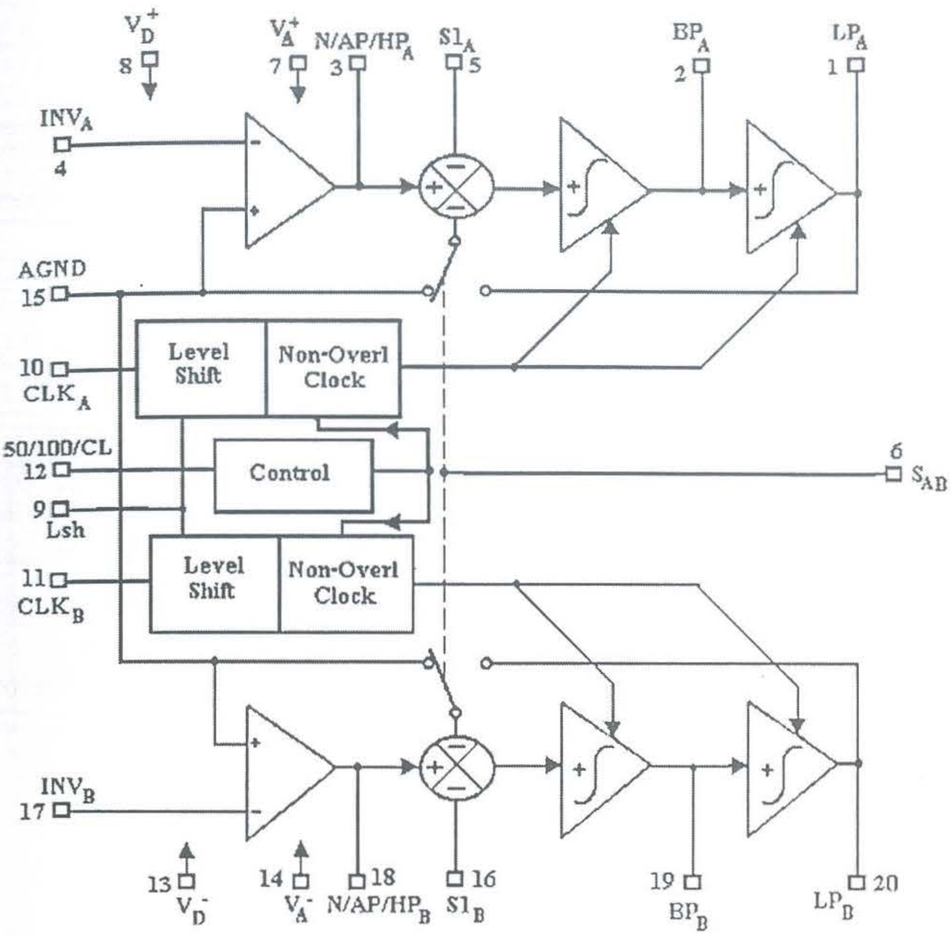


Fig. 4.4 Block diagram of SCF LMF 100. Source [8].

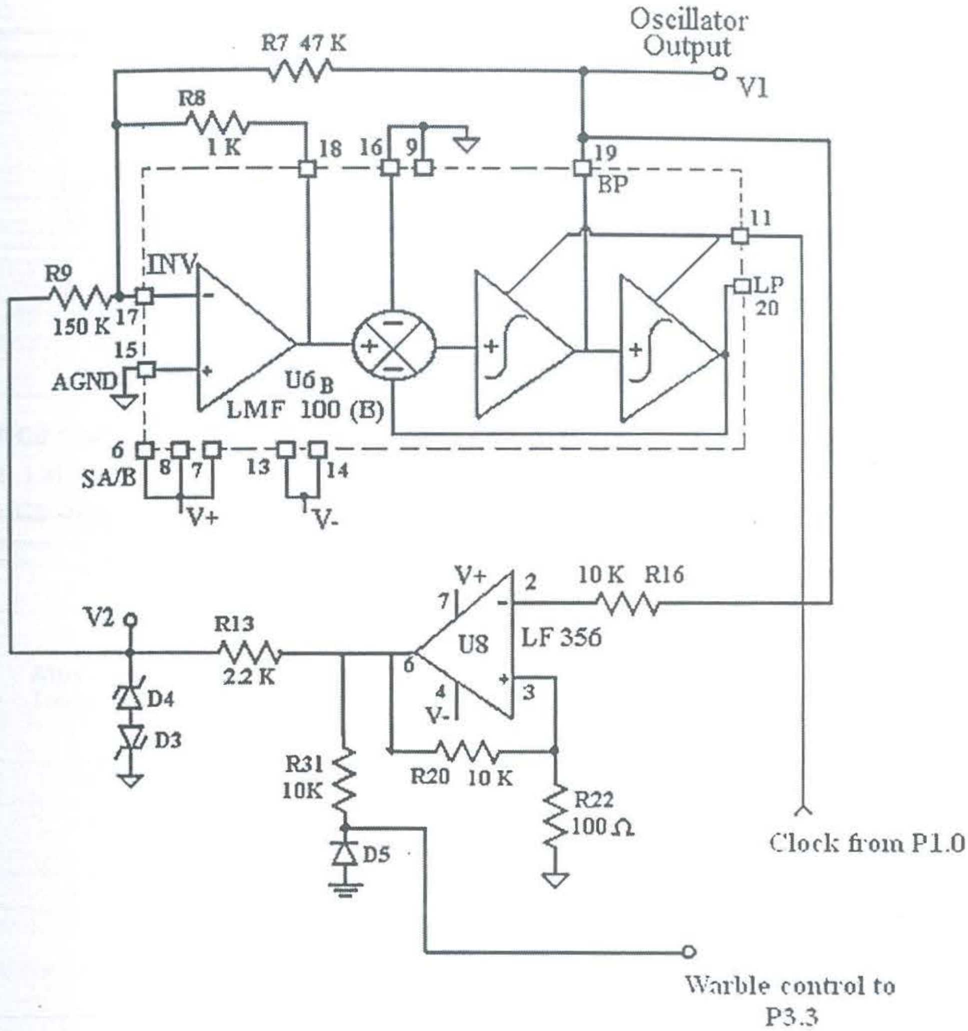


Fig. 4.5 SCF based quadrature oscillator using LMF 100

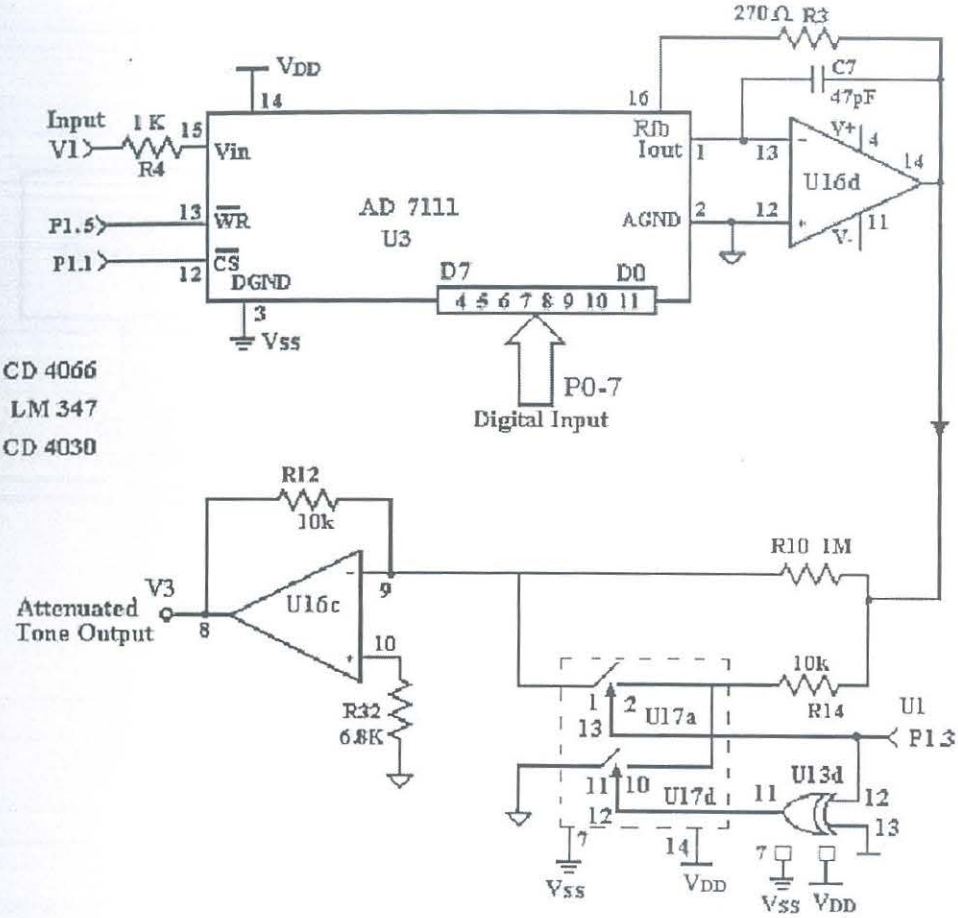


Fig. 4.6 Attenuator circuit for tone

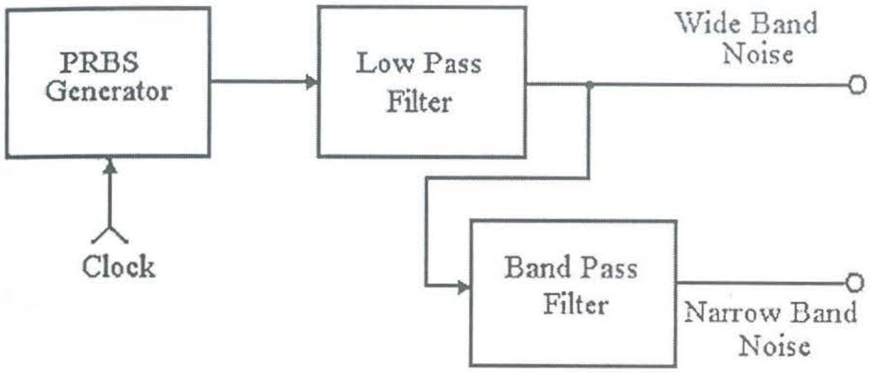


Fig. 4.7 Block diagram for noise generator

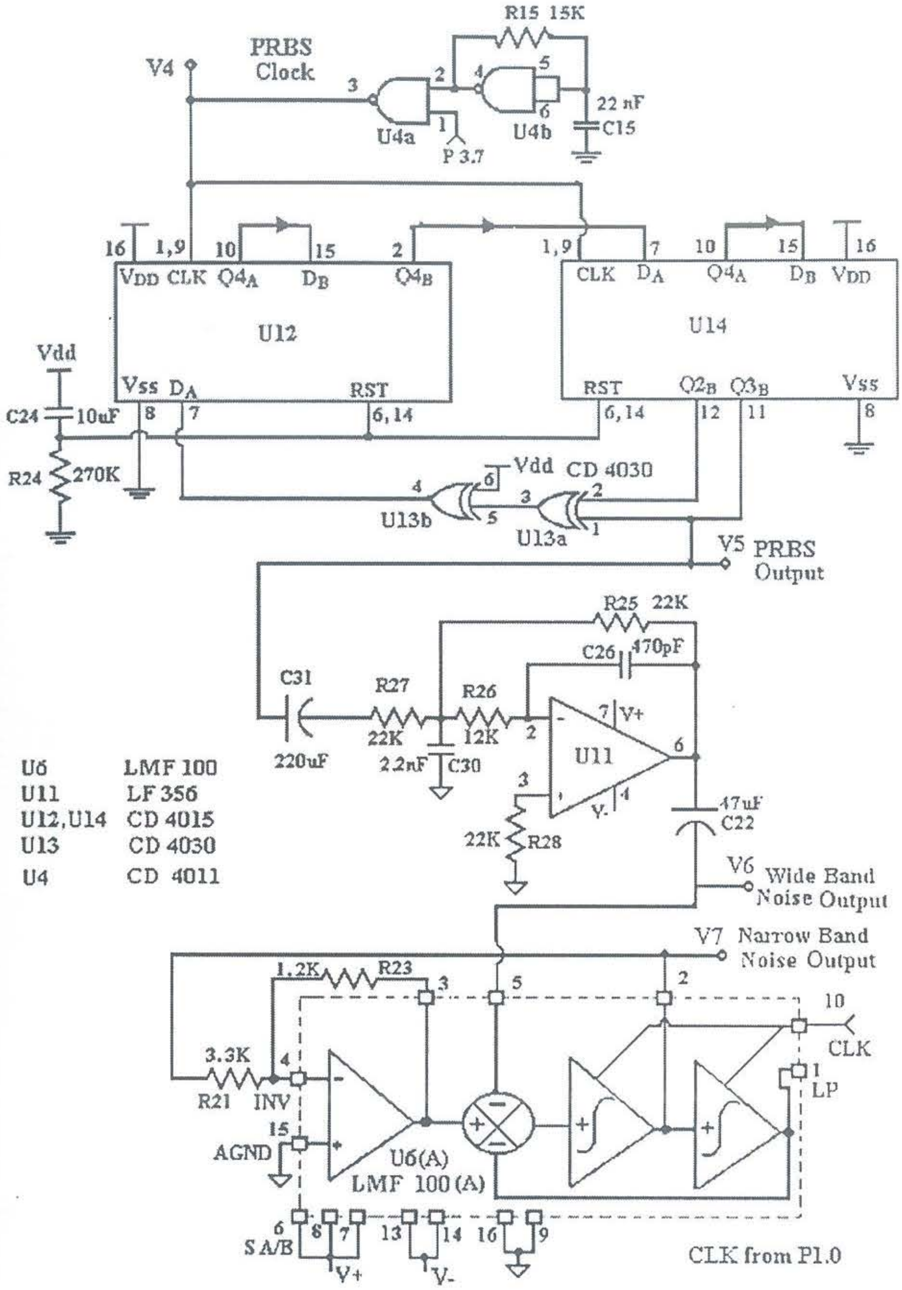


Fig. 4.8 Circuit for noise generator

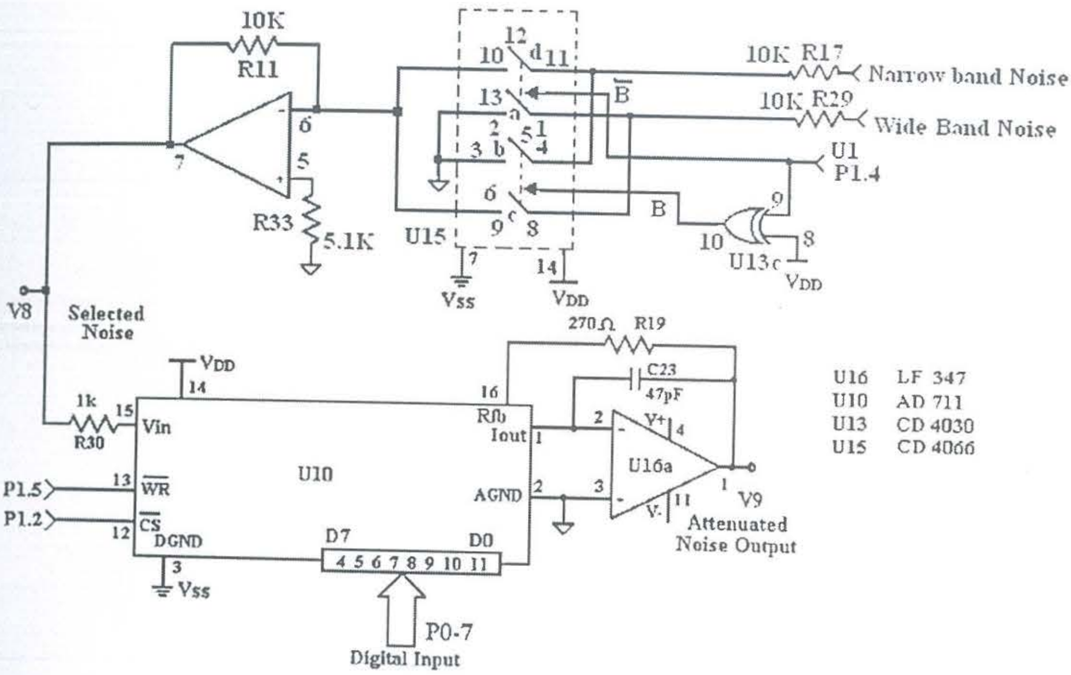


Fig. 4.9 Noise selector and attenuator circuit

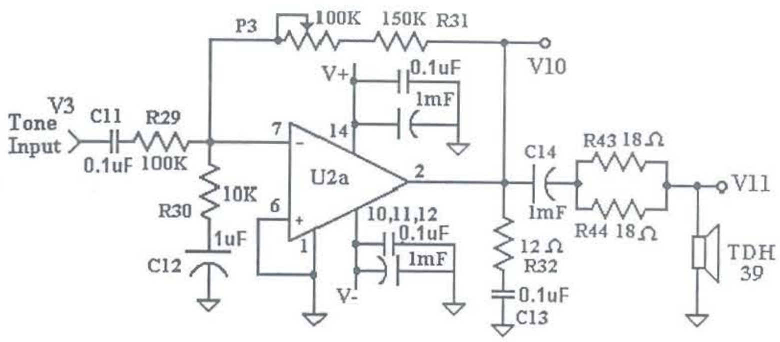


Fig. 4.10(a) Power amplifier for headphone

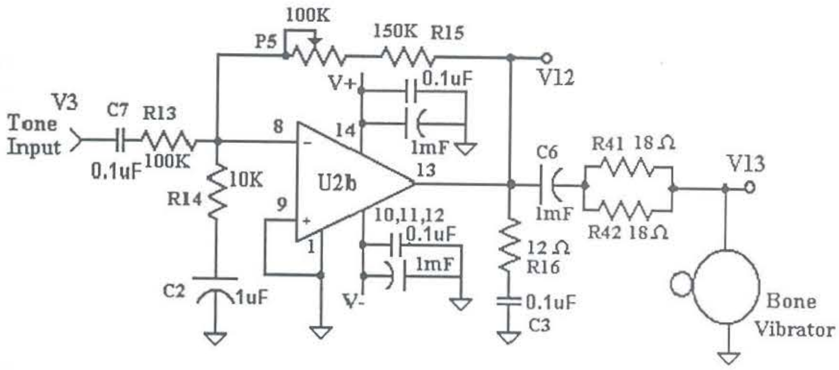


Fig. 4.10(b) Power amplifier for bone vibrator

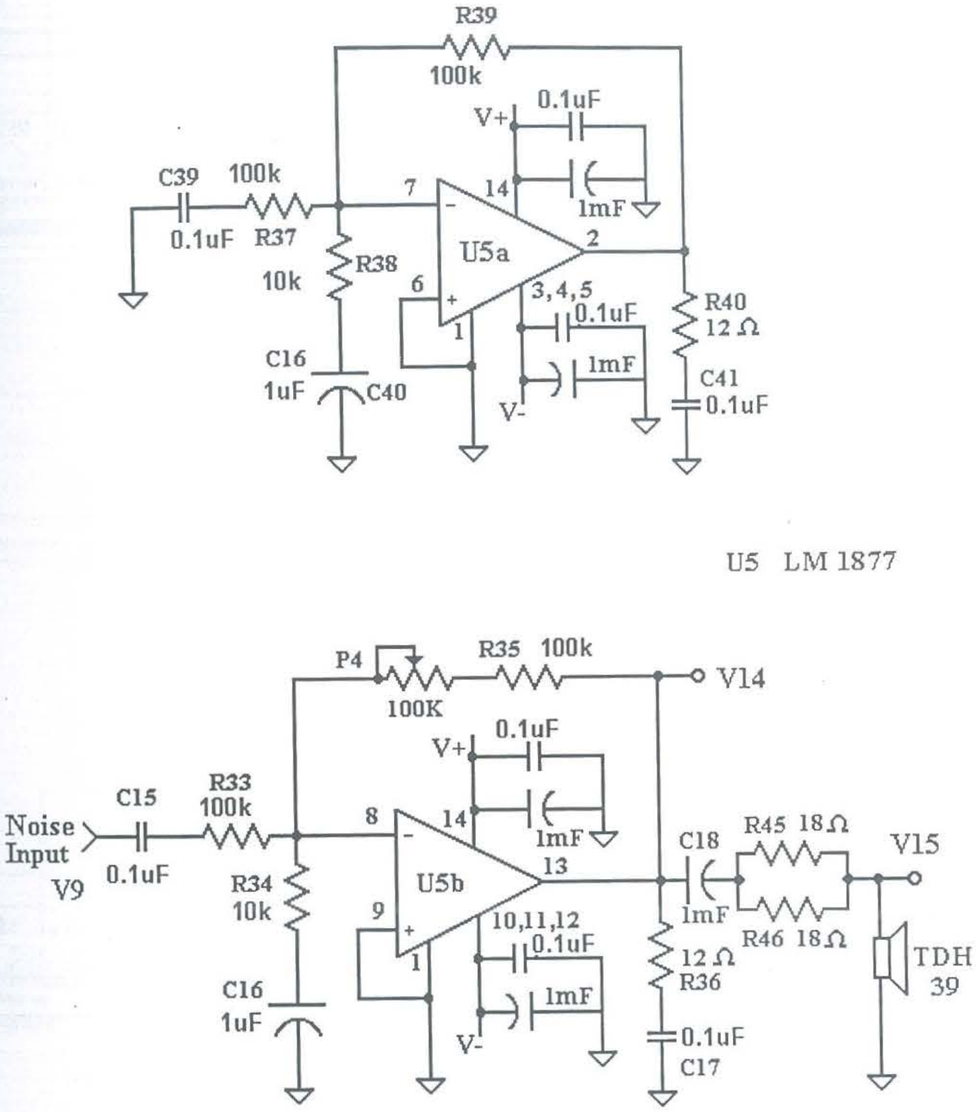


Fig. 4.11 Power amplifier for noise

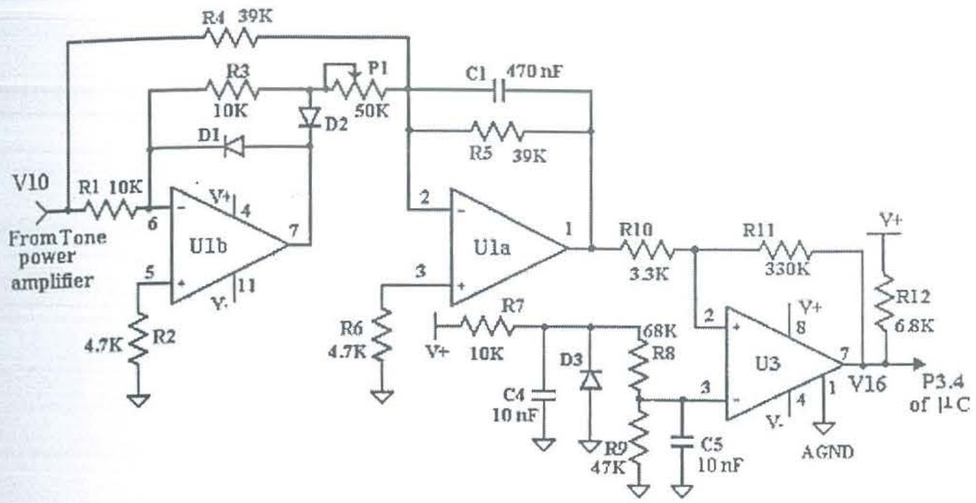


Fig. 4.12(a) Level monitoring circuit for tone channel

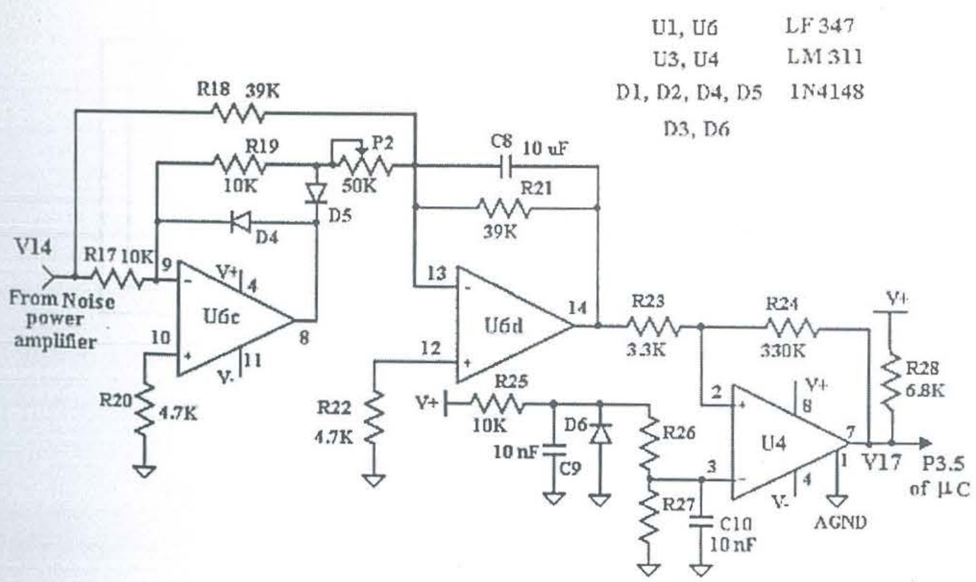


Fig. 4.12(b) Level Monitoring circuit for noise channel

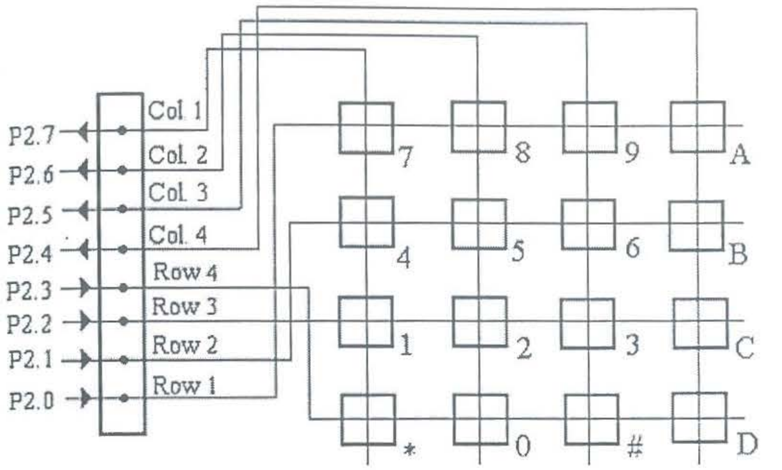


Fig. 4.13 4X4 keypad Layout

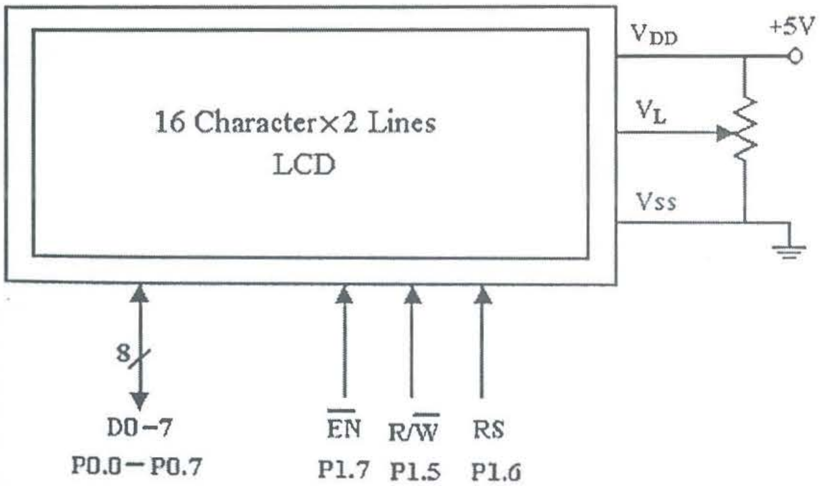


Fig. 4.14 16 characters × 2 lines ODM-16216S LCD

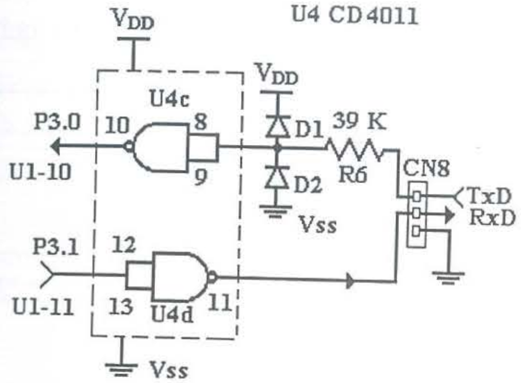


Fig. 4.15 Circuit for serial interface

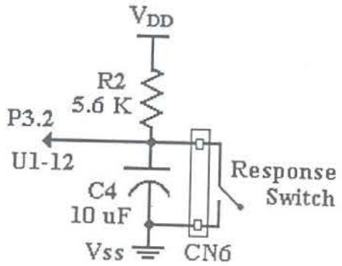


Fig. 4.16 Debounce for response switch

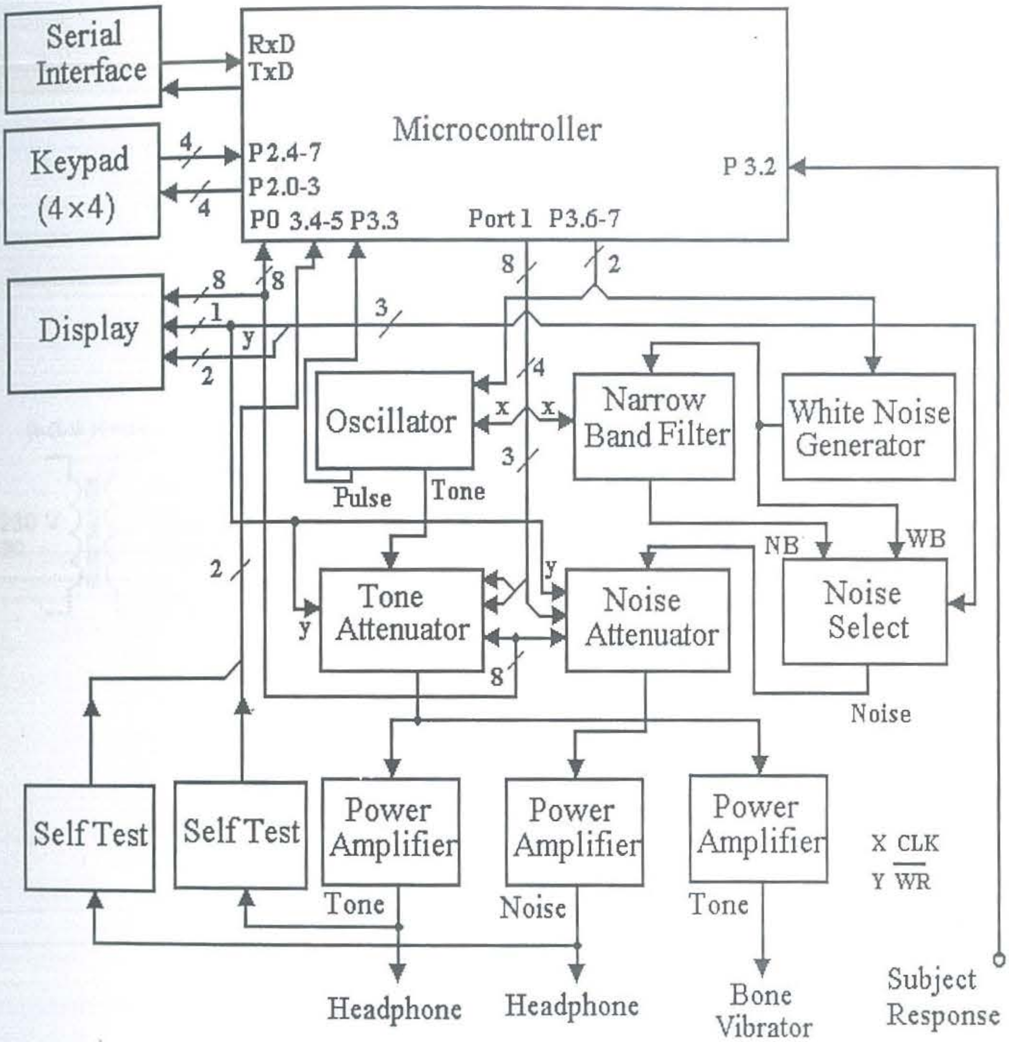
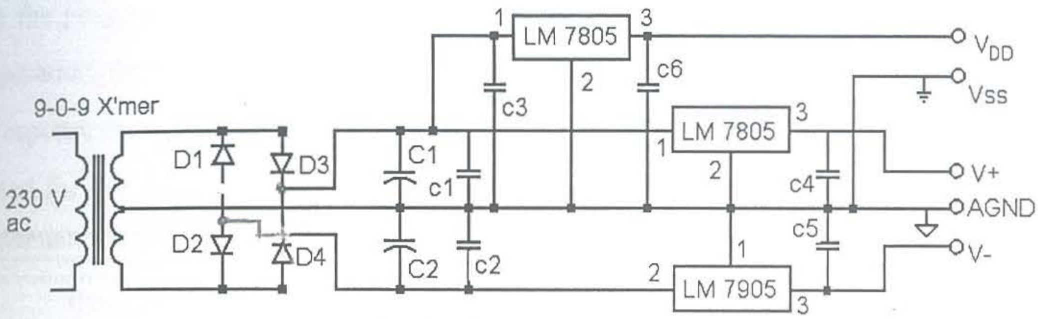


Fig. 4.17 Microcontroller interfacing with various blocks



C1=C2= 1000 μF

c1-6 = 0.1 μF

Fig. 4.18 Power Supply

Chapter 5

SOFTWARE DESCRIPTION

The hardware consisting of tone oscillator, noise generator, attenuator, amplifier, circuit for level monitoring, and microcontroller alongwith the interfacing has been described in the previous chapter.

The pin assignments of the microcontroller are given in Table 4.1. Fig. 4.1 shows the block diagram of the system along with the interfacing details. Port 0 is multiplexed between display, tone and noise attenuator's data bus. A 4×4-matrix keypad is directly connected to port 2. 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 $\overline{\text{INT1}}$ is used for warble tone generation. Response switch is connected to external interrupt pin $\overline{\text{INT0}}$. Port pins P1.1 and P1.2 are used for selecting ($\overline{\text{CS}}$) of tone and noise attenuator chip respectively. An extra 40-dB attenuator for tone is controlled by P1.3.

Port pin P1.5 is common for R/$\overline{\text{W}}$ of display and $\overline{\text{WR}}$ of both attenuator chips. Control signals, RS and $\overline{\text{EN}}$, 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 printer or computer.

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. Some parts of the program have been retained as developed earlier by Kothari [1], some parts are modified and some parts are written afresh.

5.1 Tone Generation

Pure tones with frequency 250 Hz, 500 Hz, 1 kHz, 1.5 kHz, 2 kHz, 3 kHz, 4 kHz, 6kHz and, 8 kHz are generated. 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. Content of RCAP2H is always 0FFH. Timer 2 will output a clock of 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. The crystal frequency used was 12 MHz in previous instrument [1]. One bit change in the RCAP2L register changes the frequency by approximately 1 kHz. Three successive counts produce frequencies of 6.6 kHz, 7.5 kHz, and 8.4 kHz. By using a microcontroller running at a higher clock rate, finer steps of frequencies are obtained, and this will result in improved quality of warble tone. Hence the crystal of 24MHz frequency has been used. For this frequency, number, which will be loaded for each frequency, is given in Table 5.1.

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. 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, alongwith 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.

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 80ms. 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.

5.2 Attenuator Control

As mentioned earlier in section 4.2, "0" attenuation count corresponds to 0dBm voltage level (0.775 V rms). Attenuation counts A results in voltage level of -0.375A dBm.

The desired tone level is obtained by loading the attenuator count A, which is a function of the hearing level L and tone frequency F. The voltage level (in dBm) required for producing L=0, as a function of frequency are given in Table 4.1, for a specific piece of TDH39 P headphone. From these values, the voltage level in dBm and corresponding attenuation count for L=30 are calculated and given in Table 5.3. The attenuation count for a specific L and F is obtaining as

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

Since A need 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

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

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

For $L \leq 30$,

$$A(L, F) = A(30, F) + A_l$$

For $L > 30$,

$$A(L, F) = A(30, F) - A_l$$

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. Hence if A exceeds 236, then 40-dB attenuator is turned on and corresponding count 40/0.375 =107 is subtracted from the value of A. This value is loaded to the attenuator latch of the chip AD 711. The data is latched on the rising edge of the $\overline{WR}$. Before latching the data, $\overline{CS}$ of the chip should be made active. The data set up and hold requirements are satisfied [9].

The attenuation of 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 no switchable 40-dB attenuator. For both types of noise, the frequency dependence of the count is the same as for the tone frequency selected.

When the tone is presented, the flag TCON.2 is set and then corresponding attenuator count is latched into the chip. This tone is kept on for the specified duration, which is selected at the time of initialization. After that duration the tone is turned off by clearing TCON.2 flag. But this technique gives rise to a "thump" sound at the start and at the end of tone. When tone is not presented the clock given to the oscillator is disconnected. As the oscillator stops operation, a dc level exists at the output of oscillator. Next time when tone is presented, again clock is made on, and the attenuator count related to desired value is fed. The oscillator takes finite time to resume operation and hence dc level superimposed upon the tone frequency sounds like a thump. Similarly when tone is turned off, the oscillator clock is disconnected immediately, at the same time the attenuater has not attained its maximum attenuation instantaneously hence the oscillator will have some dc level at its output. This dc level along with some intermediate attenuation level will produce thump.

So to solve this problem, the clock can be kept continuously on and when tone is not presented the attenuation count can be made maximum. But in this case when we resume the tone presentation, the output of attenuator will change from no value to a sinusoidal output of desired level. Here since the attenuator output suddenly changes from inaudible to audible tone, so we can hear a "click" sound instead of thump, which is not desirable. So another approach could be that 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. Since the attenuator takes some time to stabilize its output. 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 of 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'.

5.3 Keypad, Display, Subject Response and Serial Interface

A 4×4-matrix 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 it's 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.

The serial communication is used to download results to the PC. For that the data transfer rate of 2400 bps is used. For generating this rate timer1 is used in 8-bit "auto-reload" mode. The count to generate this rate is calculated using following formula.

$$\text{BaudRate} = \frac{K \times \text{Crystal Frequency}}{12 \times 32 \times [256 - \text{Th1}]}$$

Here the crystal frequency used is 24 MHz. *K* is 1, which is set by clearing SMOD bit in PCON register. The serial subroutine polls for the Ti bit. It indicates that the given byte in the serial buffer is transmitted completely and it is ready to accept next byte. The data is downloaded to PC through a program running on PC. Its description is given afterwards.

5.4 PC Interface Program

To download the test result from audiometer, a program is written in C to run on PC, which will receive data over serial port with the settings = 2400 bps, 7 bit data with even parity, and one stop bit. Then these data are stored in one file along with other subject details like name, age, test ear etc (collected through PC console). The data are used for plotting audiogram. In the audiogram whichever details are not entered are kept blank, so that the operator can enter these manually. Through this program a previously stored file can be opened for displaying audiogram on screen (like a typical audiogram shown earlier in Fig. 2.2). The code for displaying audiogram is written in VC++. The audiograms are displayed in a window, and "Save" and "Print" command can be used for using audiogram as a part of document or for obtaining hard copy.

5.5 Level Monitoring Routine

When the instrument is switched on, a self test routine is executed, making 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. And at the end all the values corresponding to the referance voltage are displayed. And then 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 here is checking for any frequency dependent variability. The calibration table may very for various types of headphones.

5.6 Test Modes

Once the level monitoring is over, the various parameters for the test are selected. The various test mode options available are

1) Manual pure tone audionmetry
2) Automated pure tone audiometry
3) SISI test
4) Tone decay test

For first two options other parameters that can be set are

a) Tone type: pure tone, warble tone (FM), amplitude modulated tone (AM)
b) Type of conduction: air, bone
c) Tone duration: 2 sec, 3 sec, 4 sec, continuous
d) Masking: Wide band noise, narrow band noise, no noise.

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. Hence when these parameters are set and in between the tone mode is changed to either of these two, the parameter settings are preserved and when test mode is to be selected again instrument will show these stored settings. These parameters are set by selecting the appropriate value for each parameter and after that "Ok" key is pressed. 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. The operation in each of the four modes is described in the following subsections.

5.6.1 Manual pure tone test

Once the tone is started, when tone duration is complete or "tone off" key is pressed, the tone is turned off. 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.

5.6.2 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 [4] [2]. This algorithm has been implemented for the automated mode.

5.6.3 SISI test

This test is normally carried out after finding the hearing threshold using normal pure tone audiometry. In SISI test the operator will select the frequency 20 dB suprathreshold tone levels. The tone is presented with brief bursts of 1dB modulation above the carrier tone at every 5 s. The 1dB increment is presented for an interval of 300 ms. Out of which the rise time and fall time are 50 ms each, and the incremented tone will remain on for 200 ms [4]. Twenty such bursts are given and out of them, the number of bursts the patient is able to detect is recorded, stored as the result. The same procedure is repeated for each frequency. And the result will be stored. For this test masking noise selection is same as in other modes.

5.6.4 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. Here the operator will select the frequency, starting from the 0 dB HL tones level, a continuos pure tone is presented. The subject is told 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. This duration between these 2 responses is measured, if the subject is able to hear the tone for more than one minute [10], the tone level is decremented in steps of 5 dB. And 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. It has to be seen that the level shift should be continuous i.e. the tone will be either incremented or decremented without switching off the tone. So in this test the lowest level for which tone is audible for atleast 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 in RAM. Here also the masking noise selection is same as in other test modes.

5.7 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 intermediately either by pressing the response switch or "Ok" key on the keypad. Once the self-test is over, various options (viz. tone type, tone duration, mode of operation, mode of conduction, and noise type) are selected. Tone can be either pure, warble. Or amplitude modulated type. 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 or 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 $\overline{INT0}$. 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 RAM memory of microcontroller. After completing the test, the result can be displayed sequentially for all frequencies. It can also be printed on to the serial printer.

In the automated mode, audiologist has to select the initial parameters and the frequency. The instrument does rest all in accordance with the test algorithm. The threshold obtained is stored automatically. Then again it will ask for frequency. For any frequency if the operator selects more than one test mode, then the most recent result will be stored in the memory, as it will overwrite the previously stored result.

In the automated mode, the subject response has to be obtained through the subject response button only. In the manual mode, the audiologist need not rely upon the response through the push button and can use an alternative depending on subject's convenience.

In SISI test, operator will select the frequency and 20 dB suprathrshold tone level. Then the instrument will take care of the rest and will find out the score. In between operator can always 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 always stop the test by pressing "Tone off" key.

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)
250	100:1	FF10
500	100:1	FF88
1000	100:1	FFC4
1500	100:1	FFD8
2000	50:1	FFC4

Frequency (Hz)	Operation mode	Number to be loaded (Hex)
3000	50:1	FFD8
4000	50:1	FFE2
6000	50:1	FFEC
8000	50:1	FFF1

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

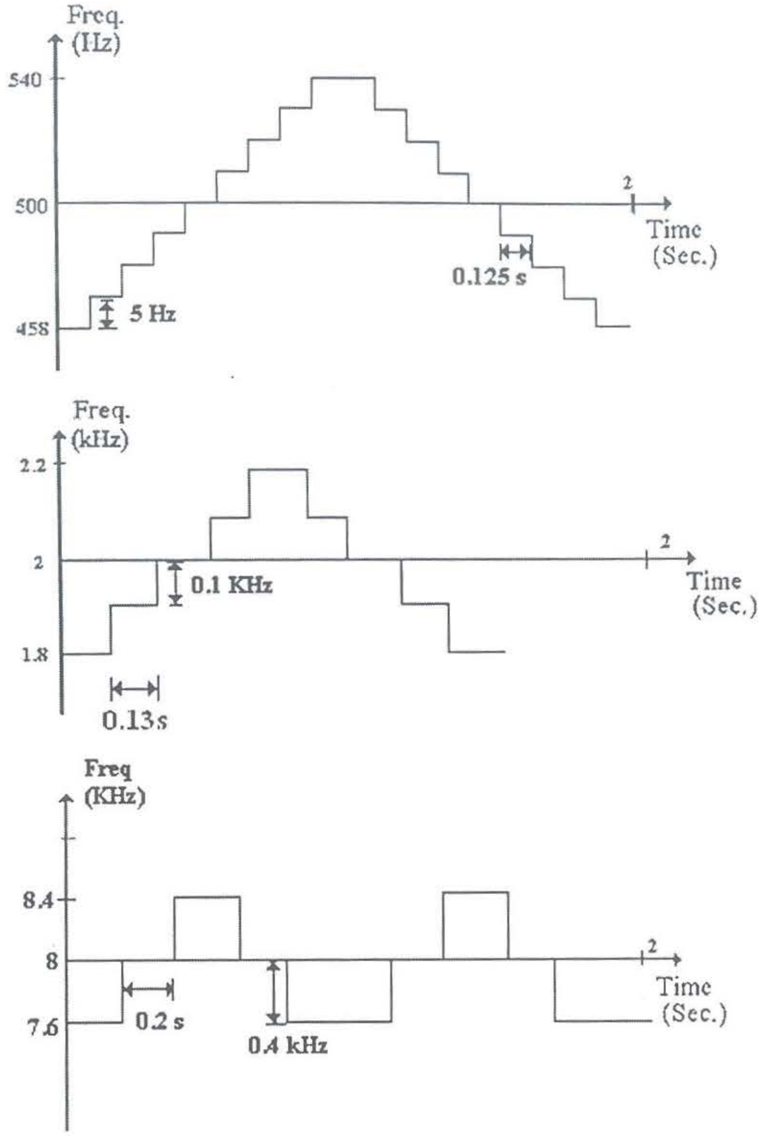


Fig. 5.1 Frequency sweeps for 1 kHz, 3 kHz, and 8 kHz

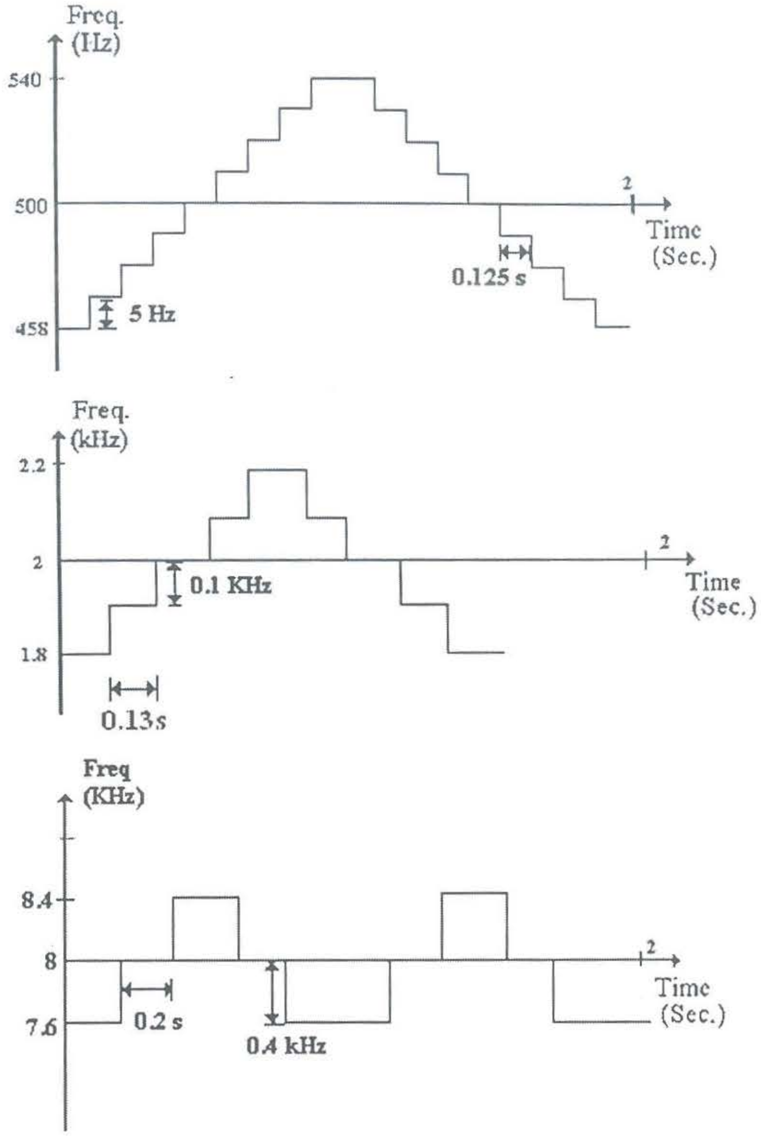


Fig. 5.1 Frequency sweeps for 1 kHz, 3 kHz, and 8 kHz

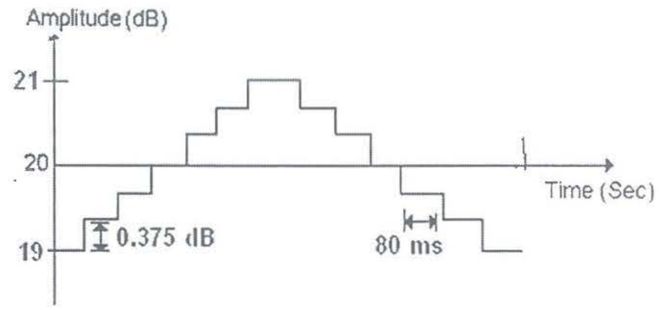


Fig.5.2.a Amplitude Modulated Tone for 20 dB level

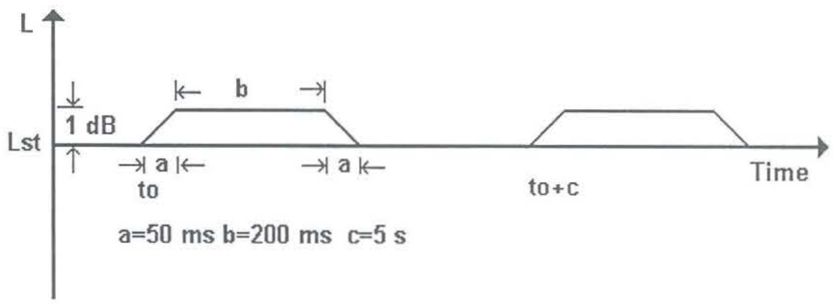


Fig 5.2.b Amplitude levels for SISI test

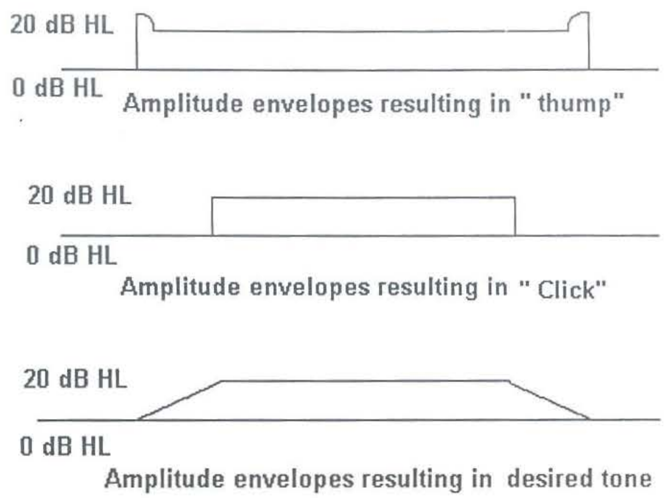


Fig. 5.3 Tone envelops

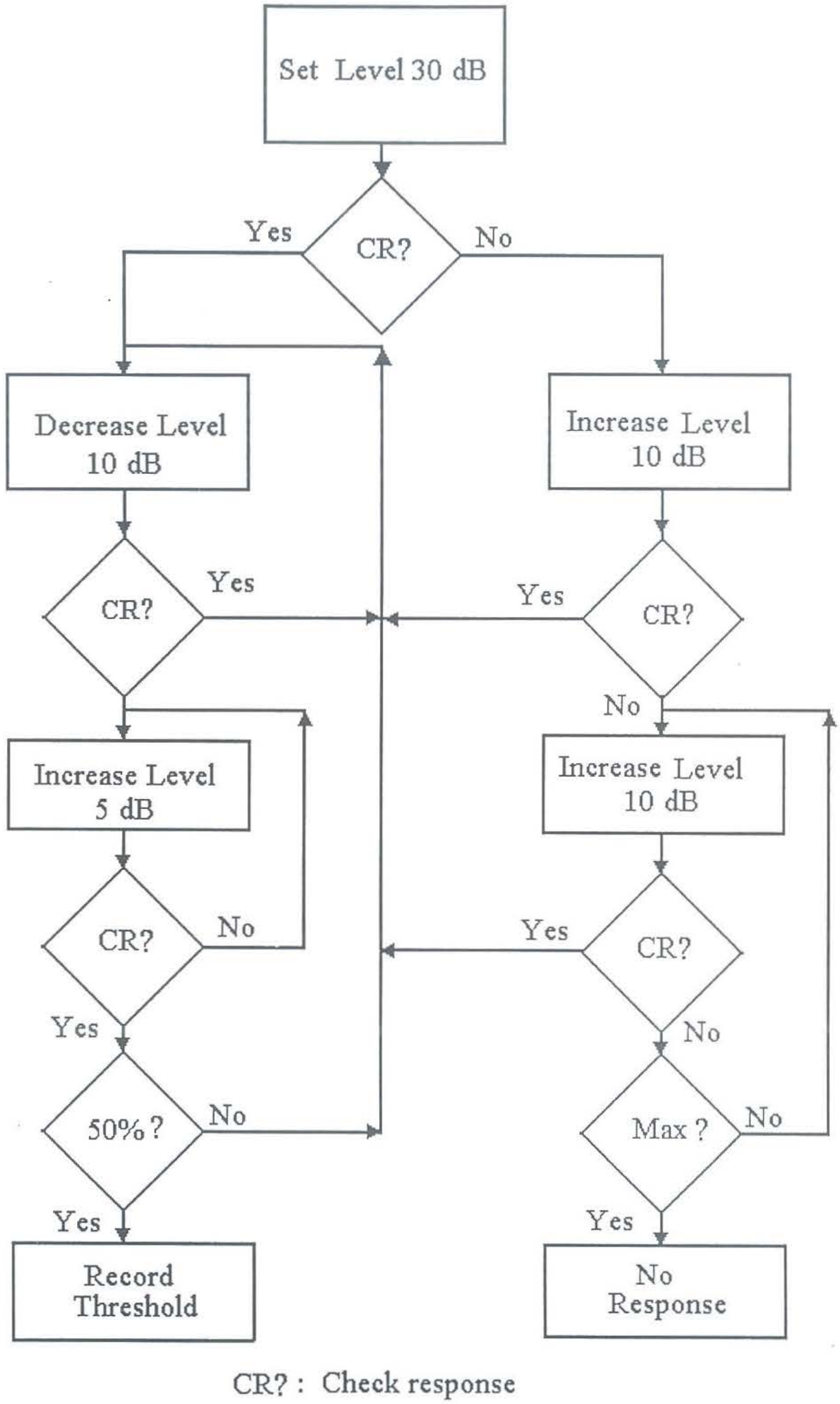


Fig.5.4 Flowchart for automated test

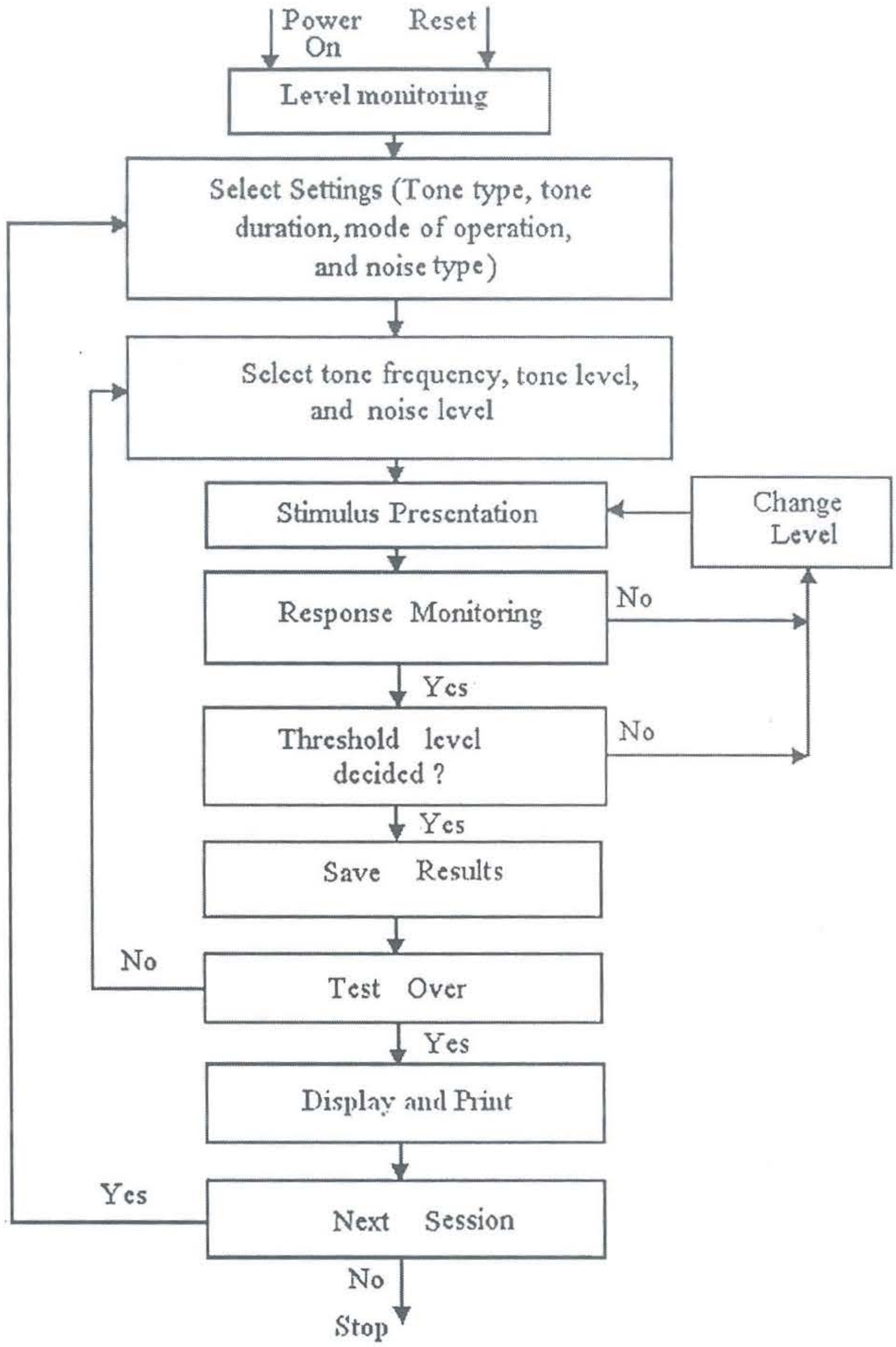


Fig. 5.5 Operation sequence for threshold search

Chapter 6
SYSTEM ASSEMBLY AND TESTING

For the purpose of PCB assembly, the circuit was divided in two parts. PCB-1 consists of the tone generator, noise generator circuits, and the microcontroller. PCB-2 consists of the power amplifier and 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. For the two PCBs, the double-sided layouts designed earlier by Kothari [1] were used. 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, and assembly. Finally the test results are given.

6.1 PCB Design

The size of PCB-1 is 14.5 cm × 13.5 cm and PCB-2 is of 10 cm × 13.5 cm. Both the PCBs are double sided with plated through holes (PTH). It should be noted that each block in the circuit has mixed signals i.e. analog and digital, and consequently special care is 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 tone and noise attenuators are realized using DAC chips, the data and control lines of which are interfaced to microcontroller. Thus, almost everywhere analog is meeting digital. There is a great possibility of analog supply being modulated 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 thick copper plane at the most electrically stable point in the PCB. 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 planes of supply and ground are provided on opposite sides 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 AD 7111. The layout is designed to provide ground shielding between the analog and digital parts 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].

The PCB-2 is divided in two channels, one for tone and the other for noise. In order to minimize cross talk between two channels, the supply routing of both channels 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 display is mounted and bellow 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.

6.3 Testing

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

The spectral purity of tone signal was observed on dynamic signal analyzer HP 3561A. Total harmonic distortion (THD) was approximately 51 dB below the fundamental, for all the test frequencies. 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 (D3, D4). Thus to maintain proper levels, it is necessary to use stabilized supplied for the measurement.

The tone level in the instrument operation was changed in steps of 5 dB. The level change was confirmed by actually measuring the output voltage. The linearity of attenuator is confirmed over 20 dB HL to 100 dB HL. For measuring the output for levels below 20 dB HL, it was found necessary to use tuned band pass filter before the voltmeter, which was not available. The working of 40-dB attenuator was confirmed. The attenuation provided by the switchable 40-dB attenuator is within ± 0.2 dB of 40 dB. Thus, the monotonicity of the attenuator was confirmed over the whole range.

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 viz. The spectrum is plotted by taking the individual peak voltage in dBm as reference . The analysis bandwidth is kept at 62.5 Hz. The wideband noise is flat up to 8 kHz The roll-off out side 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. Practically the bandwidth for 1 kHz, 8 kHz noise is 0.55, 0.57 octave respectively. The 20 dB bandwidth is measured as 4 ocatve about the center frequency.

In the level monitoring circuits, the ripple in the rectifier-averager output affects the operation of the circuit. The ac ripple for lowest operating frequency (250 Hz) was measured to be 20 mVp-p. Higher the frequency, lower is the ripple. The comparator is designed with a hysteresis of 50 mV, and it was found to be around 40 mV. Therefore, ripple in dc output does not result in chatter in the operation of comparator circuit.

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.cpp" for downloading the test results from audiometer and displaying these as an audiogram.

The instrument is supplied by two power supplies, +5V for digital and ±5V for analog. The currents under no load and full load conditions were measured, and these are as given in Table 6.2. If the current is supplied from a ± 5V source, the standby currents are 70 mA and 56 mA on the +ve and -ve side respectively. Under full load condition (max sound level is generated and output connected to headphones) the currents are 96 mA and 110 mA for -ve and +ve side respectively.

Table 6.1

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

Frequency (Hz)	**$V_{p\text{-}p}$ at $V_s = \pm 4.5$ V**	**$V_{p\text{-}p}$ at $V_s = \pm 5.0$ V**	**$V_{p\text{-}p}$ at $V_s = \pm 5.5$ V**
500	2.25 V	2.35 V	2.5 V
2000	2.25 V	2.35 V	2.5 V
6000	2.25 V	2.35 V	2.5 V

Table 6.2

Current drawn from power supplies.

Full load: highest sound level delivered to both headphones.

No load: no sound generation.

Supply Source	**Current drawn (mA)**	
	No load	**Full load**
Digital + 5V	14	14
Analog + 5V	56	96
Analog – 5V	56	96

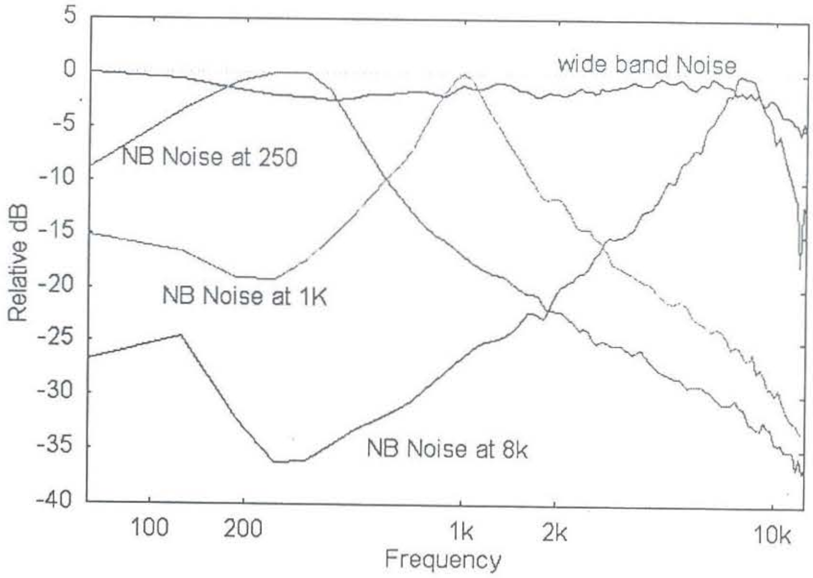


Fig. 6.1 Power spectrum of wide and narrow band noise taken at output of power amplifier with analysis bandwidth as 62.5 Hz

Chapter 7
SUMMARY AND SUGGESTIONS

7.1 Work Done

The objective of this project was to develop a portable diagnostic audiometer, which can produce tone of full frequency range and hearing level, have full masking facility, facility of warble, of amplitude modulated tone, and facility of air and bone conduction. It should have a provision for conducting tone decay test and SISI test. It should be usable in mobile clinics and in rural areas.

A microcontroller based pure tone diagnostic audiometer is developed which operates over the 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 ±10% frequency deviation, and amplitude modulated tone with amplitude deviation of ±1 dB. It has flexibility in selecting required frequencies and sound levels. It has two channels, one for pure tone and other for masking noise. Masking facility is provided, which includes both types of noise, wide-band and narrow-band. Facility of air and bone conduction is provided. A prototype developed earlier by Kothari [1] was tested for all these features, appropriate software corrections were made. The hardware was duplicated and the implementation of tone decay test and SISI test was tested on hardware.

The instrument is menu driven with option of manual as well as automatic mode. This is made possible by using microcontroller, programmable oscillator, and programmable attenuator. A programmable oscillator is 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 is 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. There is one PC based program, which will download the test results on audiometer and display it in the form of audiogram. After power on, the instrument carries out a diagnostic test for the tone and noise level as controlled by attenuators.

Specifications of the desired system have been given earlier in Section 3.1. The specifications of the actual system developed, on the basis of performance test on it are as given in Appendix B.

7.2 Further Work Needed

The instrument can be made more versatile and useful by providing following features.

a) Improvement in the indication of subject response
b) Improvement in the masking noise quality
c) Increasing the range of output level for the tone.
d) Implementation of speech audiometry
e) Selectable calibration tables for different output devices
f) Facility for generating the calibration data for an output device
g) Use of dc/dc converter for powering the instrument from a single battery to increase its portability.
h) PCB revision

Suggestions for implementing these features are explained in the following paragraphs

For the indication of subject response, a LED flasher along with a low intensity buzzer can be provided, so that operator doesn't have to look for it on the display.

The noise generator circuit has to be tested thoroughly. Since we are using 15-bit PRBS generator operated at 200kHz, it gives repetition rate of $(200000/ 2^{15} - 1)$ which is approximately 6 Hz, and the noise has a perceivable repeatability, which is not desirable. Hence the noise generator circuit needs modification, by increasing the number of bits in the PRBS generator.

For the screening audiometer, the output level range of 0 to 100 dB HL is adequate. But for the research facility, this range should be –10 to 120 dB HL. For that the fixed 40 dB attenuator can be changed to 60 dB attenuator.

While implementing speech audiometry, we may want to retain the feature of portability of the instrument and we may as well need a good interface with multimedia PC (which has sound card and media utilities). For this the basic hardware requirement is the same as that of the pure tone audiometer. Instead of applying pure tone, speech input is s applied, either from a microphone and amplifier or a line input (tape recorder/ audio CD player/ PC multimedia card).

It will be useful to incorporate calibration facility so that the operator can select the calibration table for the output device from among a number of device types. This will need interfacing a serial NVRAM or flash programmable EEPROM.

Instrument can have a separate mode for calibration in which, the user can load calibration data 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.

7.3. Present Status (at the time of final Submission)

1) Power supply circuit to be debugged
2) In duplicated unit a) masking noise gets superimposed on tone, b) tone has a thump (more pronounced compared to first unit).
3) Self test routine sometimes goes in to infinite loop, needs to be debugged.
4) In SISI and "Tone Decay" tests, test termination in response to "Tone Off " has to be implemented.
5) In SISI test, "Save" should save "score", at present it saves dBHL, and this needs to be corrected.
6) Audiogram plotting program has to be made "windows" compatible.
7) Tone quality in AM tone needs to be checked.

Appendix A
System Specifications

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 4 rows and pins 2.4 to 2.7 are connected to 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

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

the interval of

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 lead the 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 needs 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. At that time 'Printing' will be displayed.

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

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.

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

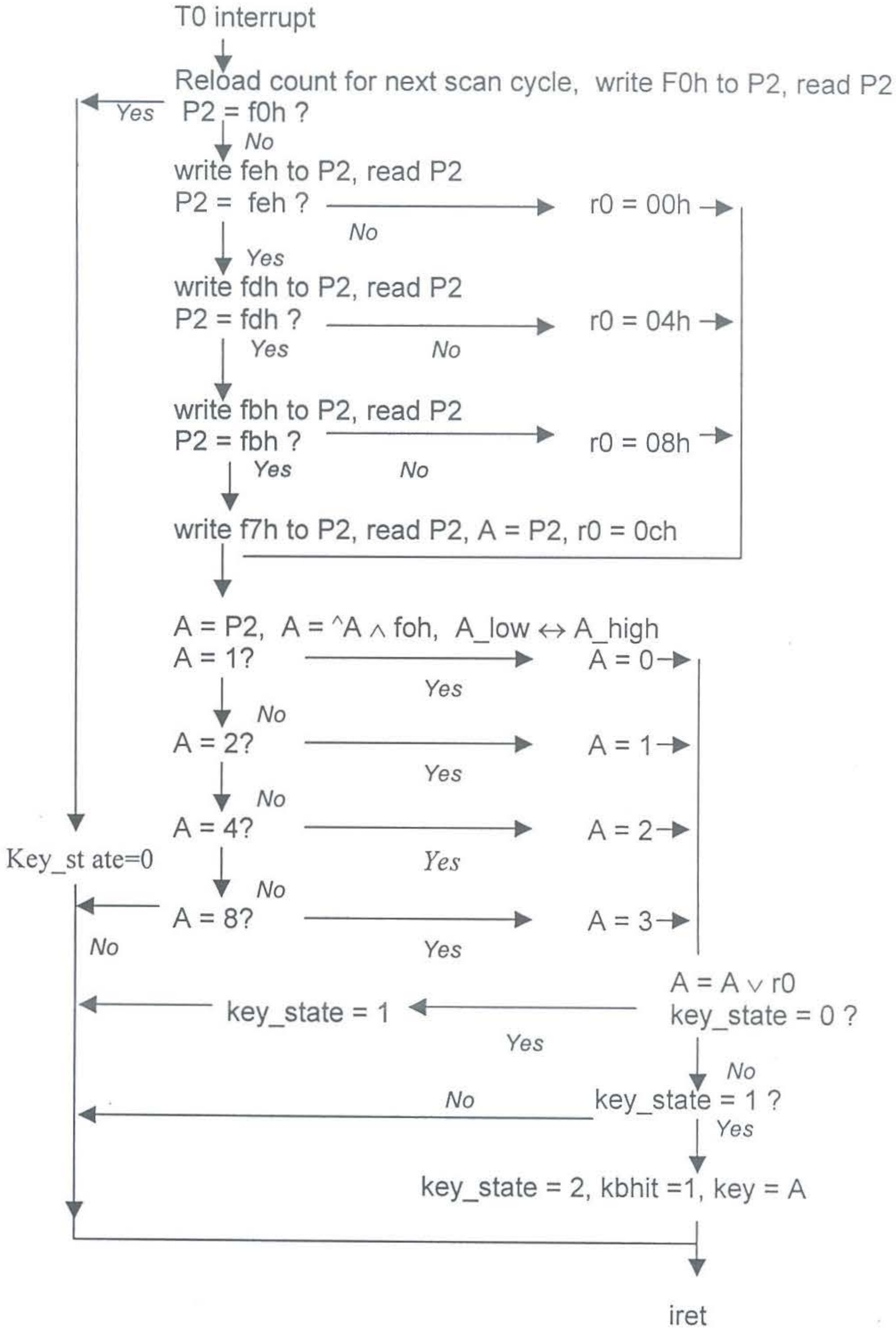
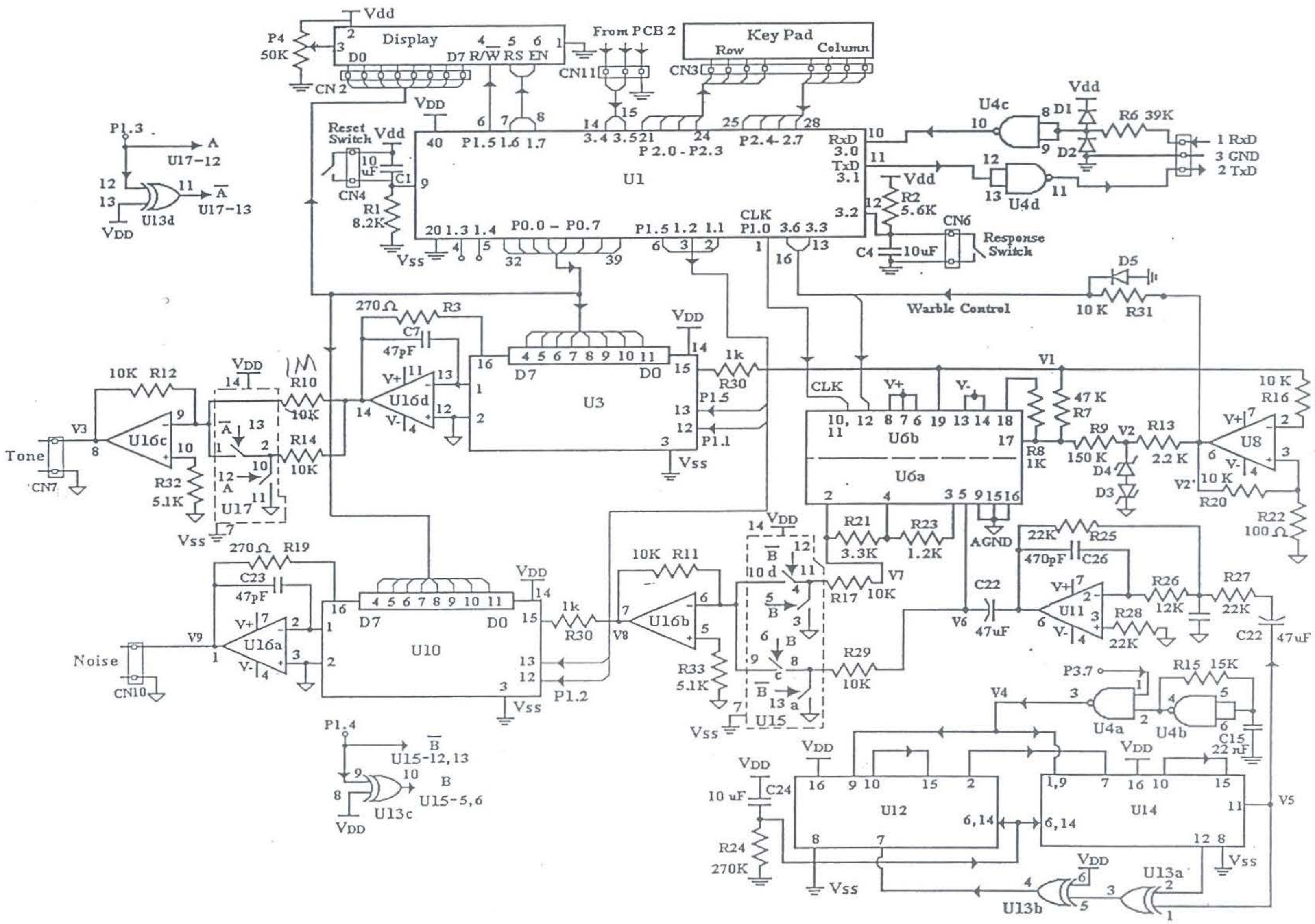


Fig B.1 Interrupt sequence routine for key scan

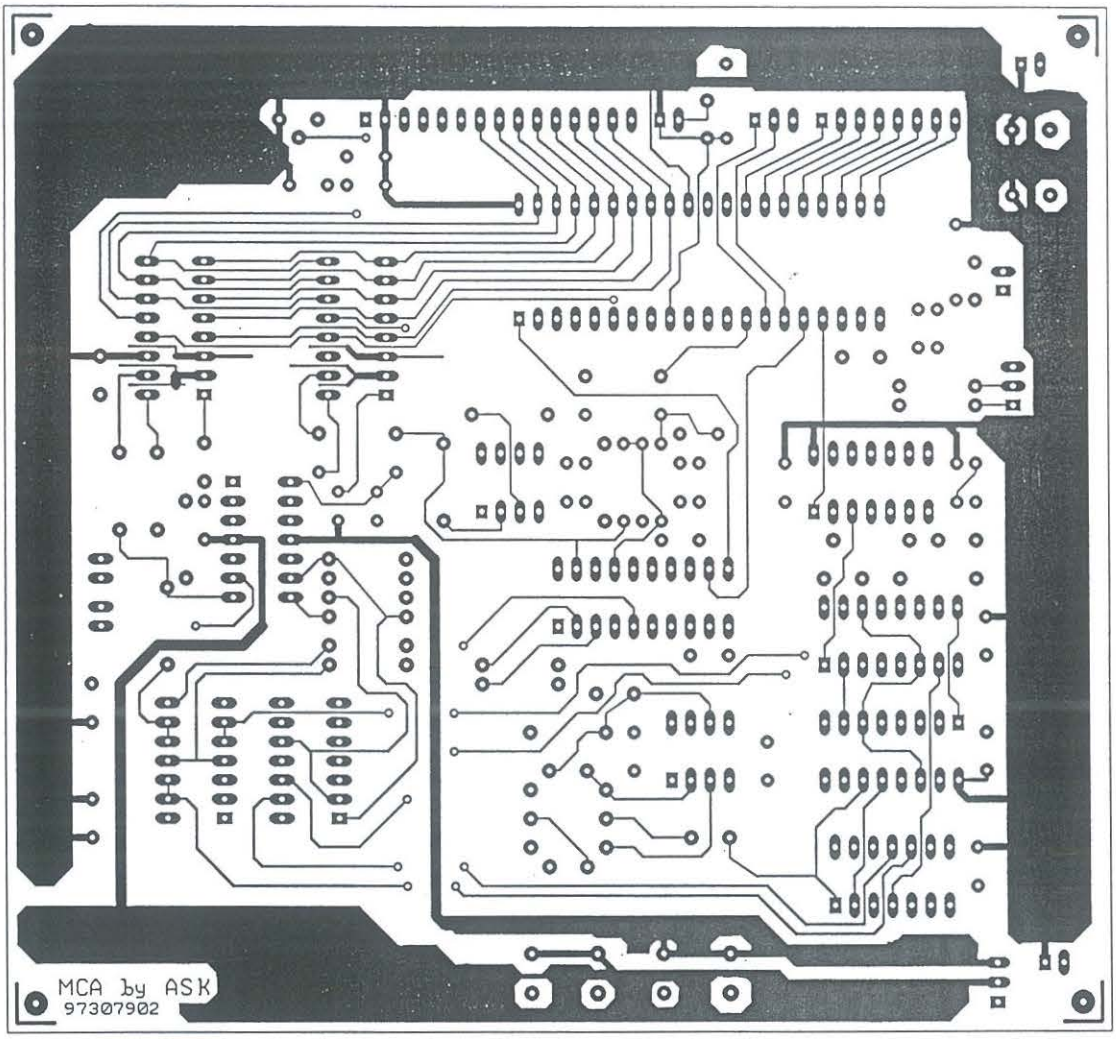
Appendix C

Circuit Diagrams and PCB Layouts

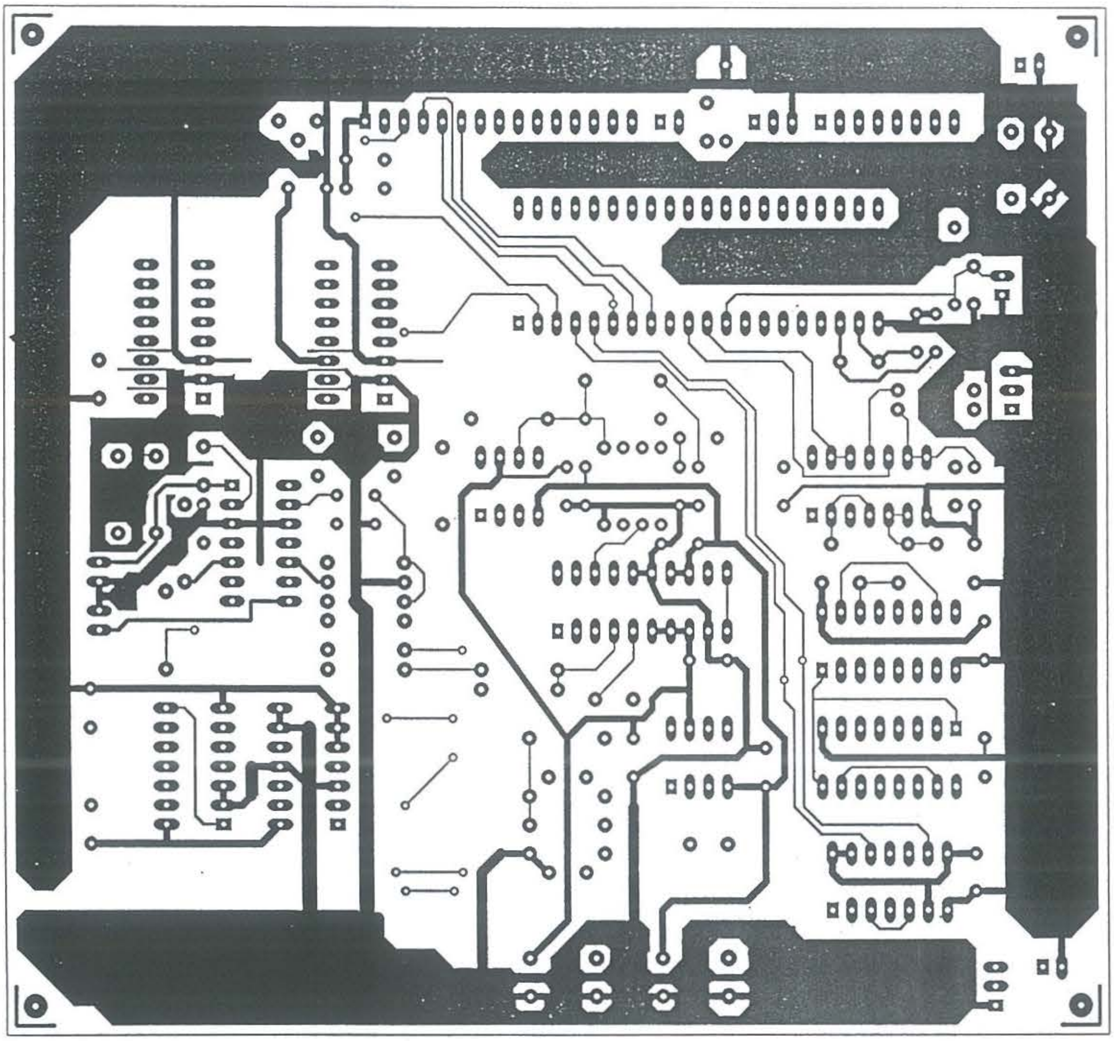


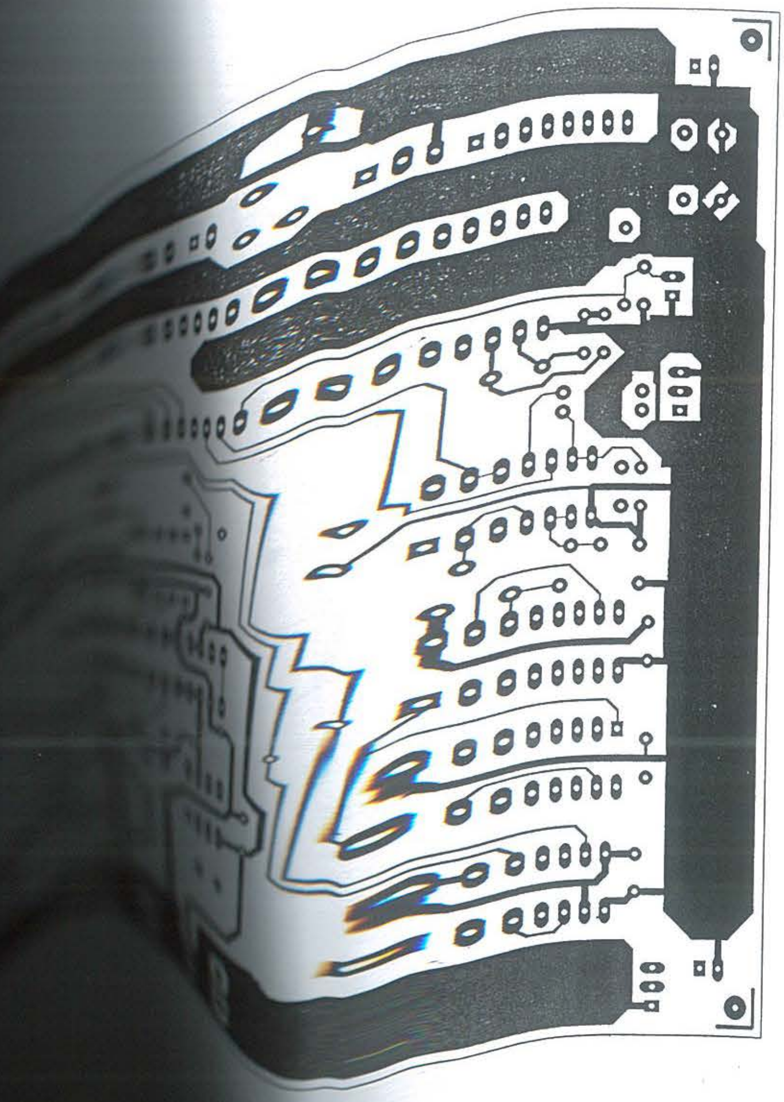
C.1 circuit Diagram for PCB1 (microcontroller and stimulus circuits)

C.2 Componente side layout of PCB1

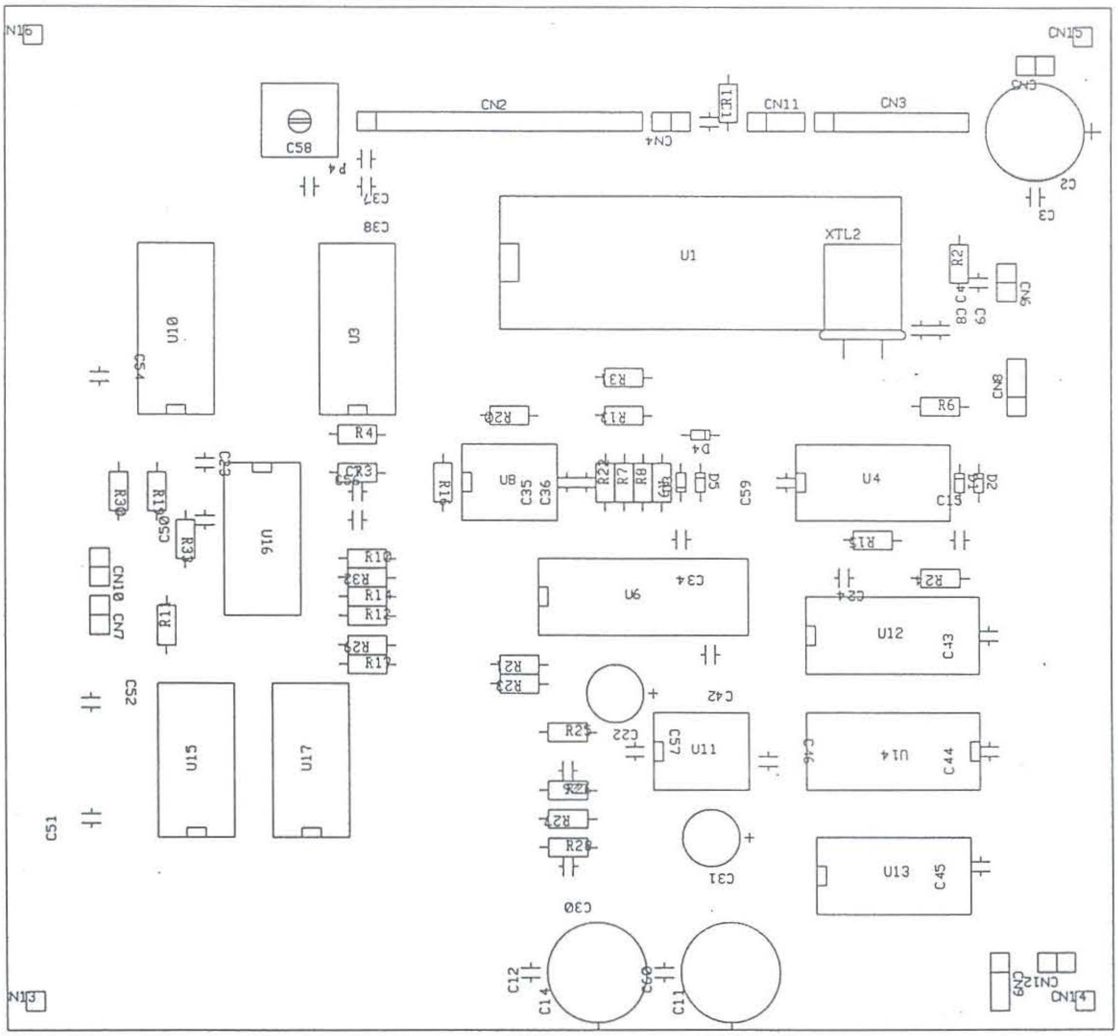


C.3 Solder side layout of PCB1

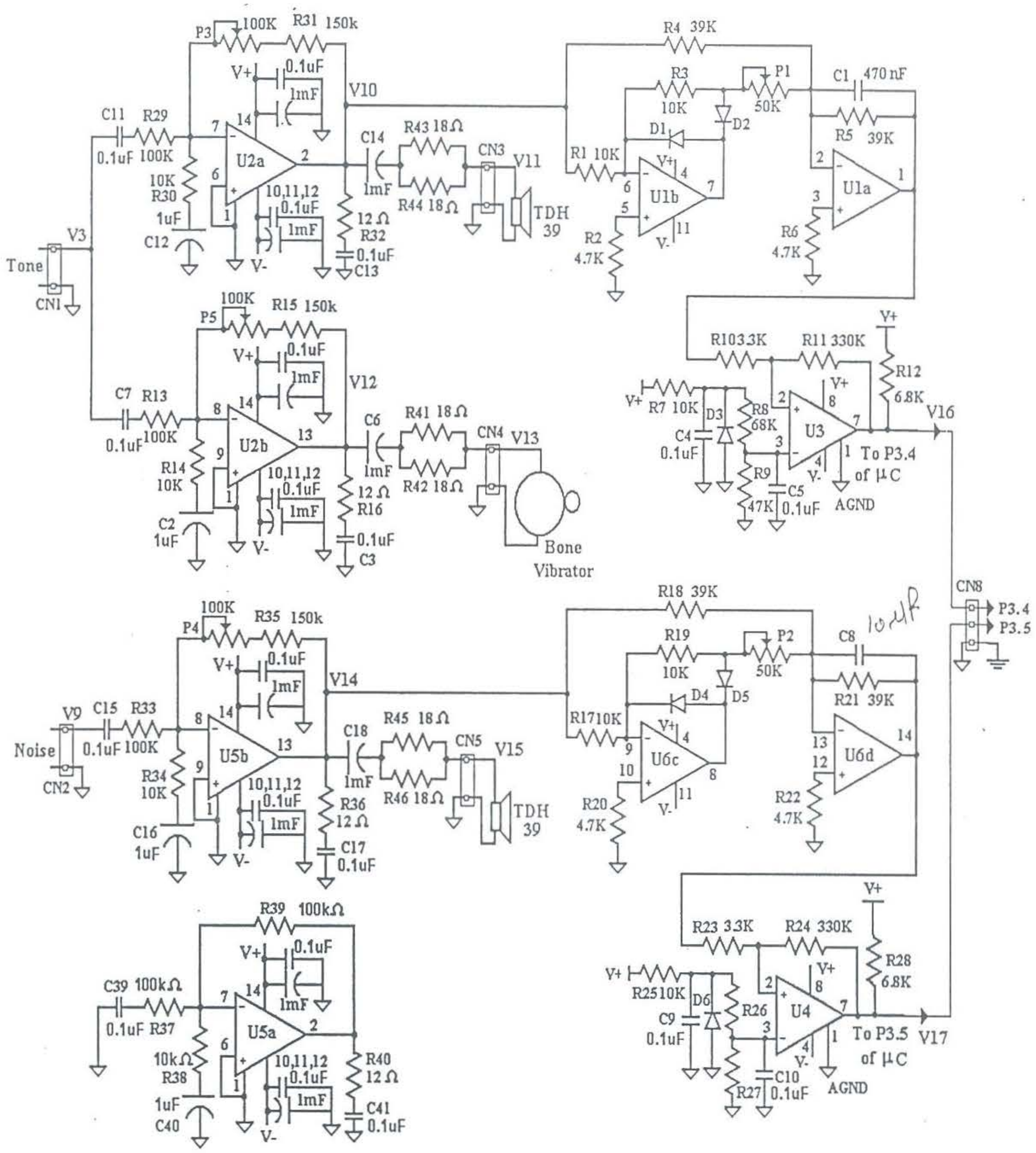




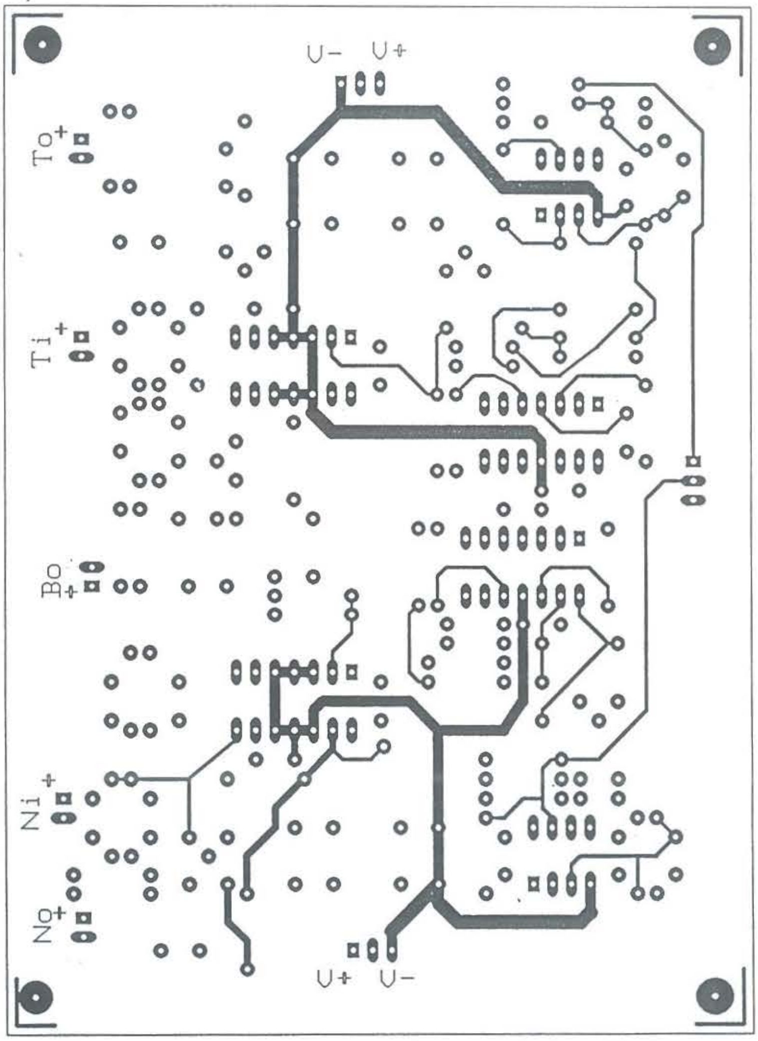
C.4 Component placement of PCB1



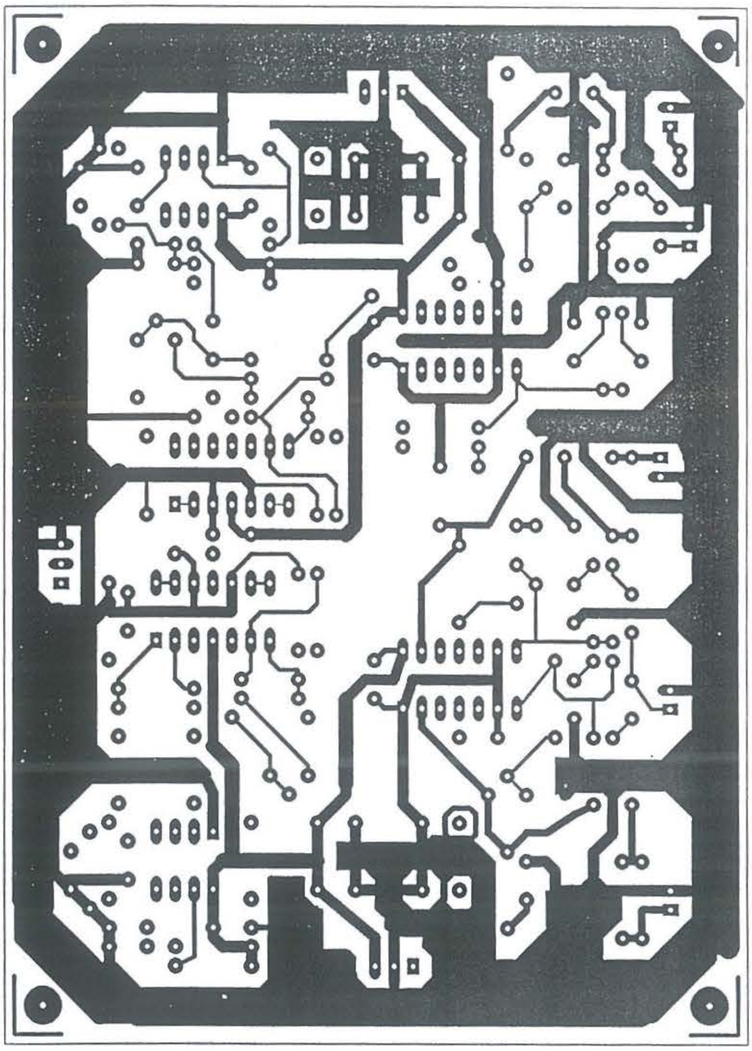
C.5 Circuit Diagram for PCB2 (amplifiers and self-test circuits)



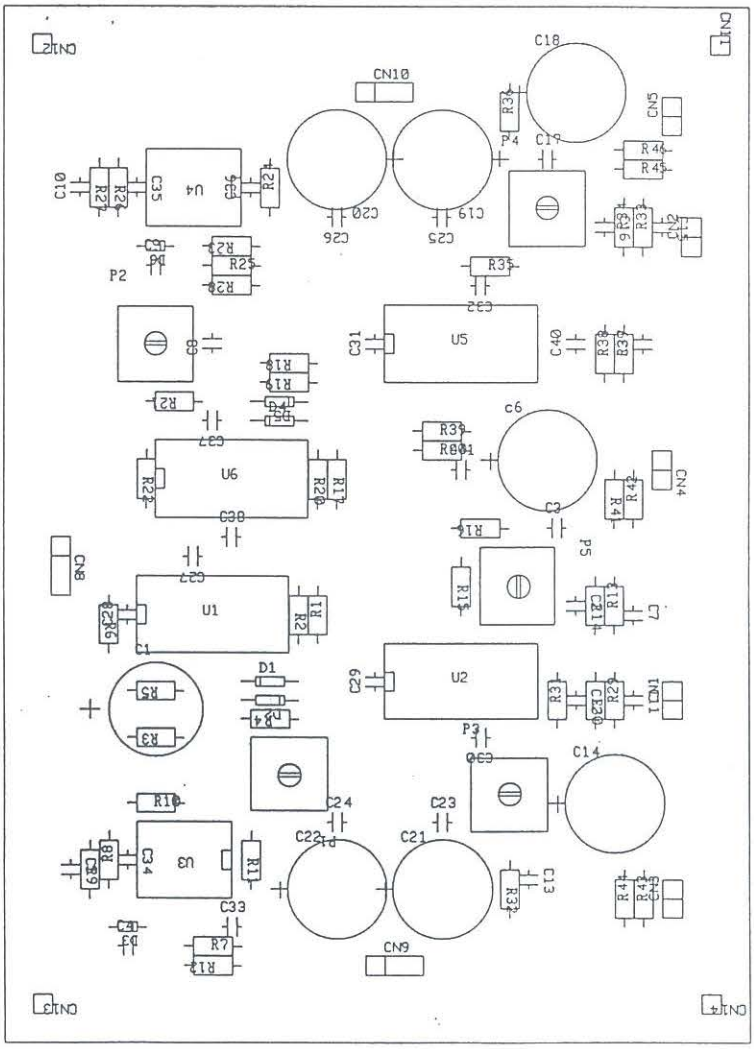
C.6 Component side layout of PCB2



c.7 Solder layout of PCB2



C.8 Component placement of PCB2



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