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A MICROCONTROLLER BASED AUDIOMETER

A dissertation
submitted in partial fulfillment of the requirements for
the degree of
Master of Technology

by

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Dissertation Approval

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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 a full frequency range (250 Hz to 7.5 kHz) and acoustic output (0 dB HL to 100 dB HL). It can also generate warble tone having ±10% frequency deviation. The instrument provides a broad-band / 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
dB	deci Bel
dB HL	deci Bel hearing level
dBm	dB above or below 1 mW (1 mW power in 600 ohm)
SC	switched capacitor
SCF	switched capacitor filter
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 degree of hearing loss can be determined by audiometry, which measures a person's hearing sensitivity. Before medical treatment can proceed, the nature, degree, and probable cause of hearing impairment must be assessed. Measurement of the responses of a person to auditory stimuli helps in such assessment. The minimum intensity level to which consistent responses are obtained is taken as the "threshold of hearing". There are different audiometry procedures depending on the stimuli used. An audiometer is an instrument, which is used for carrying out these 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, as part of his M. Tech. Dissertation at IIT Bombay, worked towards developing a microcontroller based pure tone audiometer with a provision for automated audiometry and computer/printer interface [1]. The objective of project was to test this instrument thoroughly and to develop a pure tone diagnostic audiometer based on the same approach.

The instrument has to have tone over full frequency range i.e. from 250 Hz to 8 kHz, pure and warble tones, wide-band and narrow-band masking facility, and air and bone conduction facility.

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 Singh is described. Fourth chapter describes hardware design of various blocks of the modified system. The fifth chapter provides software description. PCB design, system assembly, and test results are discussed in chapter six. Specifications of the existing system and further work needed is discussed in the seventh chapter.

Chapter 2
PURE TONE AUDIOMETER

Audiometry is required 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 identification as a function of speech intensity.

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 for 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-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].

In pure tone audiometry, a series of bursts of single frequency stimuli 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 steps 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. In this way by several presentations, the hearing threshold is obtained. The minimum presentation level at which the subject responds at least 50% times, is taken as the hearing threshold [4] [2].

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 losses [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]. Stimulus is presented through microphone, or tape recorder. In "speech discrimination test", list of monosyllable speech discrimination words are presented over headphones for each ear and the subject is requested to repeat back what he or she heard. The "speech reception threshold test" is similar to the "speech discrimination test" except for the fact that this test uses two syllable words with equal stress (spondees) and the words are attenuated successively. A threshold is determined when the patient repeats 50 % of the words correctly.

2.1.3 Evoked Response Audiometry

In response to well-defined stimuli, several clinically important potentials can be evoked from the brain and used to assess the integrity of the central auditory system. No conscious response from the patient is required in this test. "Brain stem evoked response" testing assess the pathology in the brain stem region [5]. Basically, the skin surface potential is detected, which is caused by electric impulses occurring in the area of brain stem in response to very short rise/fall time clicks of stimuli.

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 dB HL to 80 dB HL [4]. However, during bone conduction test the cochleas of both sides are equally stimulated i.e. the interaural attenuation is of 0 dB. Hence, crosshearing is a serious concern in case of bone conduction test than it is for air conduction.

To offset the risk of crosshearing, 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 broad-band noise, or band pass filtered noise, with the band centered about the test frequency.

2.3 Pure Tone Audiometer

A pure tone audiometer consists of three major parts: (a) the source signal to be presented, (b) the system for control of this signal, and (c) the means of presenting the acoustic or vibratory test signal to the patient. The tone levels are controllable in dB HL. The instrument may be provided with a calibrated noise source for masking the hearing in the non-test ear, and a bone conduction vibrator. The pure tone audiometers can be classified in the following four categories on the basis of facilities provided [3].

(1) Screening audiometer: limited frequency range up to 6 kHz and acoustic output up to 60 dB, only air conduction.

(2) Simple diagnostic audiometer: full frequency range up to 8 kHz but limited acoustic output up to 100 dB, air and bone conduction with wide-band noise for masking facility.

(3) Diagnostic audiometer: full range and acoustic output, air and bone conduction, wide and narrow-band masking noise facility for speech and pure tones, and speech audiometry inputs.

(4) Advanced diagnostic audiometer: facilities as for the diagnostic audiometer having two independent channels, having both amplitude and frequency modulation of the test tones. It provides ability to switch signals in a more flexible way.

A simple pure tone audiometer consists of tone generator, noise generator, attenuator, equalization circuit, and power amplifier, as 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 [6]. The interrupter switch is used to present or interrupt the tone.

The noise generator should provide wide-band noise, which has energy spectrum equally distributed over the test frequency range i.e. up to 8 kHz. The amplitude of noise level should be within $\pm$ 5 dB of the indicated value [6].

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. There is also a provision for calibrating the output level for different devices. The need of equalization for different frequencies and output devices is explained a little later.

The output power amplifier for tone and masking noise should have low distortion and better efficiency. The audiometer should have some means of switching the signal from one earphone to another and also to the bone vibrator.

2.4 Microprocessor Based Audiometer

A conventional audiometer, shown (block diagram shown in Fig. 2.1) needs 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 big. It is to be noted that the output sound level has to be calibrated in HL, and the electrical output for the same HL varies with frequency. This means that either, 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.

The advancement in technology has made the various switching tasks simple, flexible, and noisefree. The knobs and switches are replaced by keypad. Calibrated scales and other indicators are replaced by display to show the various parameters and modes, and operation status. Microprocessor / PC based audiometry also offers automation of audiometric testing. It is possible to store test results and to print audiogram. Increased accuracy and precision removes the need for frequent calibration of audiometer, which was required for earlier audiometers.

In PC based audiometers, tones may be generated by 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 [7]. One can generate tones by using sound card of a PC and attenuator can be communicated through serial port. Alternatively, the tone generator circuit can be similar to those in the conventional circuits, with all the controls carried out by analog switches, digitally controlled by a controller interfaced to the PC. PC based audiometers are not portable. Compact portable audiometers have become available, in which all the interfacing with the audiometer as well as controlling of the stimulus is handled by microprocessor/microcontroller. These audiometers often also incorporate the facility of automated audiometry, in which the output level of test stimuli is selected in accordance with the subject response and the threshold levels are determined. Use of programmable oscillator, and attenuator reduces the chip count resulting in simpler assembly, higher reliability, and also cost reduction.

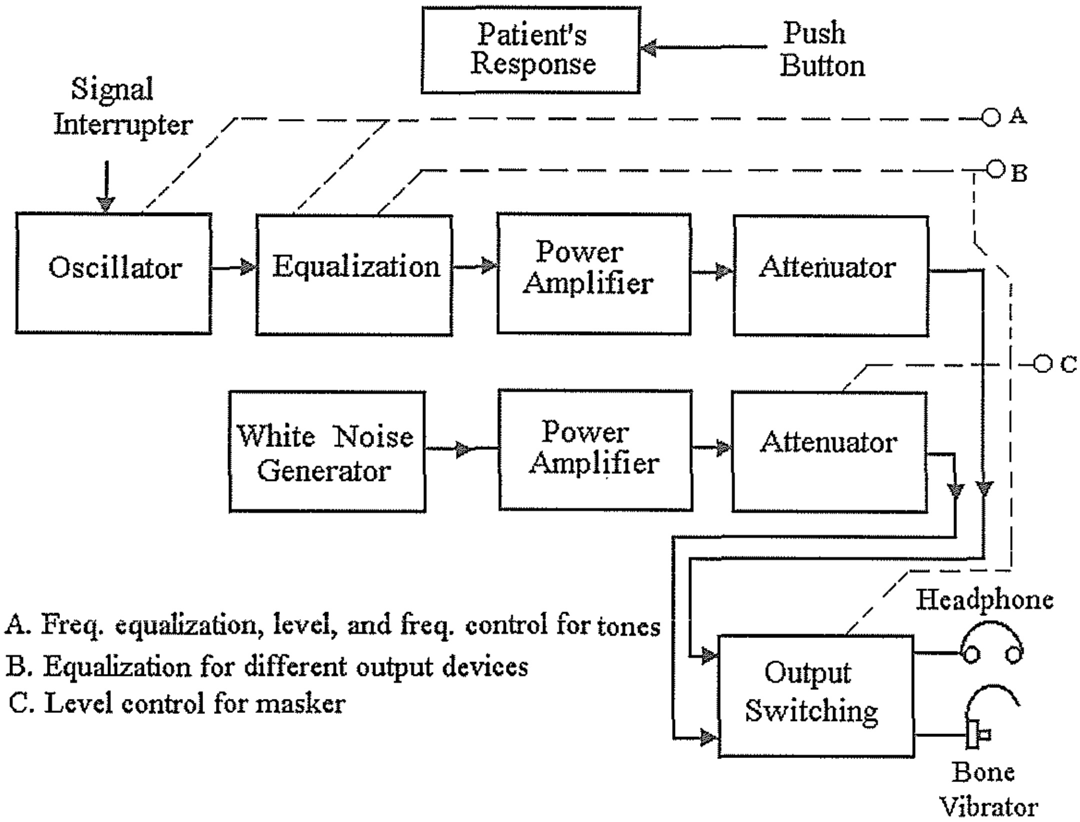


Fig. 2.1 Block diagram of conventional audiometer. Adapted from [3].

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 widely even 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 tone stimulus and wide-band / narrow-band masking noise. It should provide air and bone conduction facility.

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: warble tones with frequency deviation of ± 10%, steps are frequency dependent 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.

Storage memory: for one set of the 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 down loading the test results.

Self test: provision for internal monitoring of output levels.

Power supply: ± 5 volt (with provision for battery based operation).

Likely to be operated with single 3/6/9 V battery with dc/dc converter, or 230V mains with a power adapter.

3 2 Earlier Development

A design containing programmable oscillator, and attenuator around microcontroller can help in reducing the total number of components. Chandrakant Singh worked (as part of his M. Tech. dissertation project at IIT Bombay) towards developing a microcontroller based portable diagnostic audiometer based on this approach [1]. The following paragraphs summarize work done by Singh and also the design philosophy of each block in system, as shown in the 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, 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 AT 89C55 microcontroller, 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].

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 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 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 sensitivity of the test tone is systematically varied to determine the level at which the subject fails to perceive it. Hence, attenuator must be capable of adjusting output sound pressure level from below the threshold of hearing to some 100dB above, and normally in steps of 5 dB [3]. An 8-bit programmable monolithic logarithmic D/A converter AD 7111 from Analog Devices was found suitable. It gives attenuation of 88.5 dB, with a resolution of 0.375 dB [9]. 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, controlled by microcontroller.

A response switch is given to the patient, to indicate whether the tone is heard or not. The closure or nonclosure 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 Section 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 a 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.

3.3 Further Development Needed

The scheme used by Singh 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, particularly for its amplitude stabilization and improvement in the purity of the tone. The warble tone generation was not proper and its proper implementation requires hardware and software modification.

The attenuator circuit for the tone consists of 88.5 dB programmable attenuator using AD 7111 and an additional 40-dB attenuator using analog switches and R-network. This 40-dB attenuator needs modification. The frequency sensitive tone attenuation was not incorporated in the software, and has to be implemented. The noise attenuation control has to be worked out.

The performance of the tone and noise amplifier has to be improved, if necessary by using a different circuit. The amplifier for bone vibrator has to be built. Self-test circuit for monitoring the output level should be developed. Finally the PCB(s) should be redesigned, assembled, and appropriately packaged for building an audiometer.

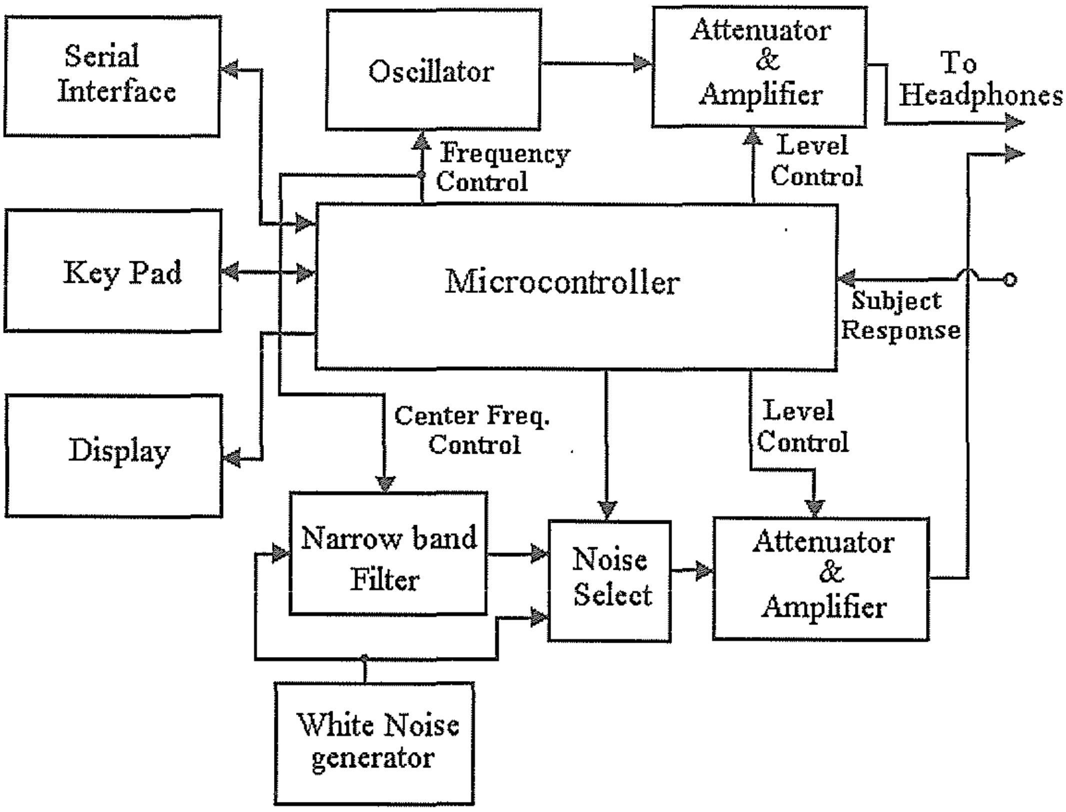


Fig. 3.1 Block diagram of system developed by Singh [1].

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. 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 noise is selected and fed to the attenuator. The noise attenuator uses the same attenuator chip, which is used for tone. 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.

This chapter provides hardware details of each block. The last section describes the interfacing of microcontroller to these blocks.

4.1 Oscillator

The basic principle of tone oscillator is illustrated in Fig. 4.2(a). The circuit consists of a high-Q band pass filter connected in a positive feedback loop with a hard limiter. The output of the band pass filter will be a sine wave whose frequency is equal to the center frequency of the filter, f_0. The sine wave signal V1 is fed to the limiter, which produces a square wave output V2 with a frequency, same as that of the 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 at the fundamental 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 can be realized using two-integrator-loop, as shown in the Fig. 4.2(b). 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, the output V_{bp} is a sinusoid with frequency f_0 and its peak-to-peak voltage is given as

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

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 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 [10]. Singh had used SCF chip MF 10, with $f_0 \times Q = 200$ kHz. This has been replaced by LMF 100, with $f_0 \times Q = 1.8$ MHz [11]. 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. 4.3. 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. 4.4. 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 [12]. The BPF output rings at its resonance frequency in response to a step input change. The oscillation loop is sustained by an inverting Schmitt trigger, formed using an op-amp 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. 4.2 and circuit schematic of Fig. 4.4, we have

$$\alpha = -R_8 / R_9$$

$$1/Q = R_8 / R_7$$

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

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

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

We want 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 $V_{2\text{P-P}} = 5.8$ V, we get $V_{1\text{p-p}} = 2.31$ 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 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. For higher frequencies 50:1 mode is used. Thus, shifting the dominant component for 250 Hz to 25kHz, which is outside the audible range. 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.

If a very high spectral purity of tone is required, then one can go for active filter having a very sharp cut off characteristics. A filter design is given briefly in Appendix C. This filter was assembled and tested, but has not been used in the present system.

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. 4.4). 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 $V_{p\text{-}p}$. 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 [3], 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. 4.6. A low pass filtering of the digital output of a pseudo random binary sequence (PRBS) generator gives a band limited white Gaussian noise [12]. 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

$$f_1 \approx 10 f_{CLK} / 215 \approx 60 \text{ Hz}$$

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

The circuit is shown in Fig. 4.7. 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 [13]. 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

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

The cutoff frequency is

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

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 transfer function

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. 4.8. 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 make P1.4 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 [14].

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 [15], 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 by air conduction, over a defined range. The same procedure is repeated by 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 maximum output voltage for bone vibrator is approximately the same as that for the headphone.

The circuit for headphone and bone vibrator tone amplifier is shown in Fig. 4.9 (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 head phone 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. 4.10. 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 Circuit for Self Test

The task of the self-test circuit is to verify the output levels at the power amplifier of the tone and masking noise, and thus to verify the operation of the two attenuators. For this purpose, the full wave rectified and averaged value of the output voltage is compared with a reference voltage. The attenuation is varied. The attenuator level where 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. 4.11 (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 [16]. 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. 4.11(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 has been 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. 4.12(a). Port pins P2.0 through P2.3 are used for outputting row scan pattern and P2.4 through P2.7 are used for reading column pattern. The scanning procedure is given in Section 5.3 of the next chapter.

4.8 Display

The display used is 16 characters × 2 lines display model LCD ODM-16216S from Oriole [17], as shown in Fig. 4.12(b). It has an on-board CMOS based controller that works on a single 5V supply. The 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/W, EN), which are interfaced to microcontroller. Control pin R/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 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. 4.13. 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. 4.14. 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 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 with reduced cost.

The pin assignments of the microcontroller are given in Table 4.1. Fig. 4.15 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. An 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 12 MHz crystal. External interrupt pin INT1 is used for warble tone generation. Response switch is connected to external interrupt pin INT0. Port pins P1.1 and P1.2 are used for selecting (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/W of display and WR of both attenuator chips. Control signals, RS and 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 self test circuit. This is done only during the self-test 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 5 ms. The interrupt for scanning is generated using Timer 0.

Table 4.1

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

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)

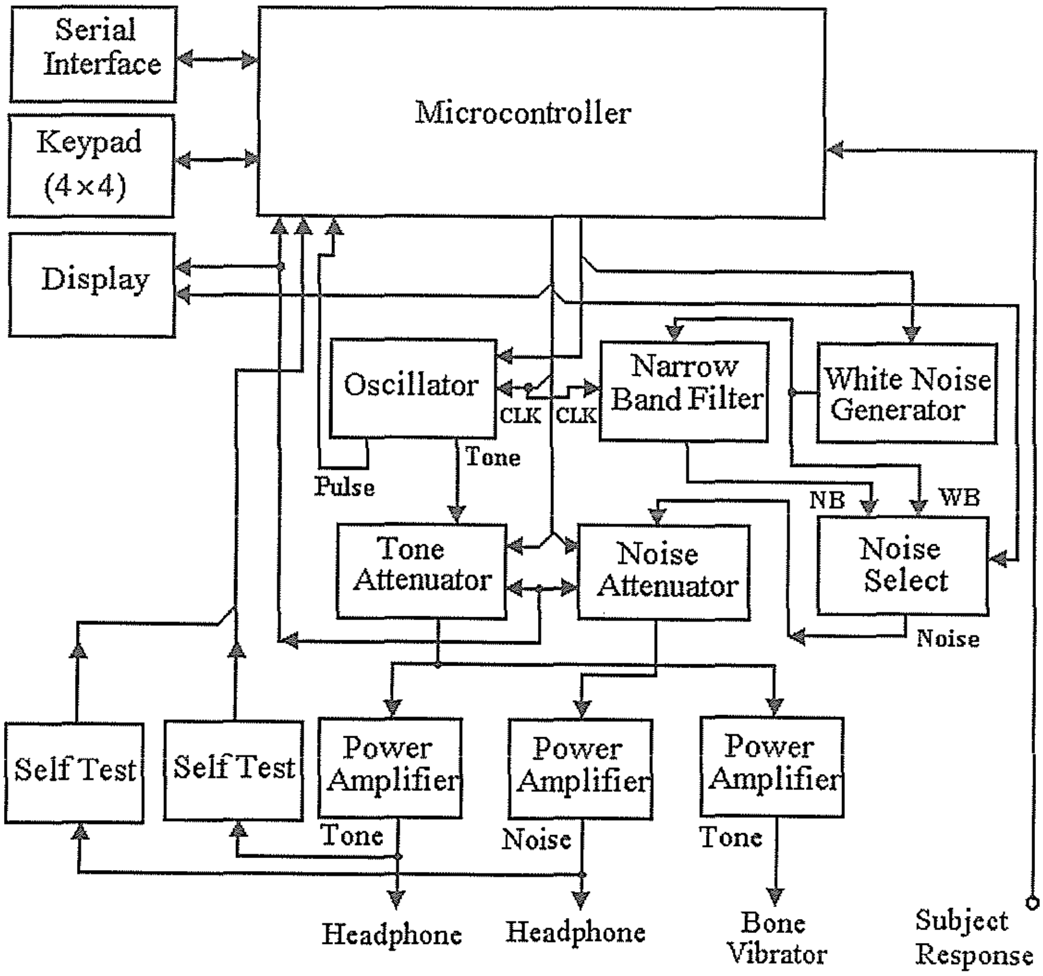


Fig. 4.1 Block diagram of system hardware

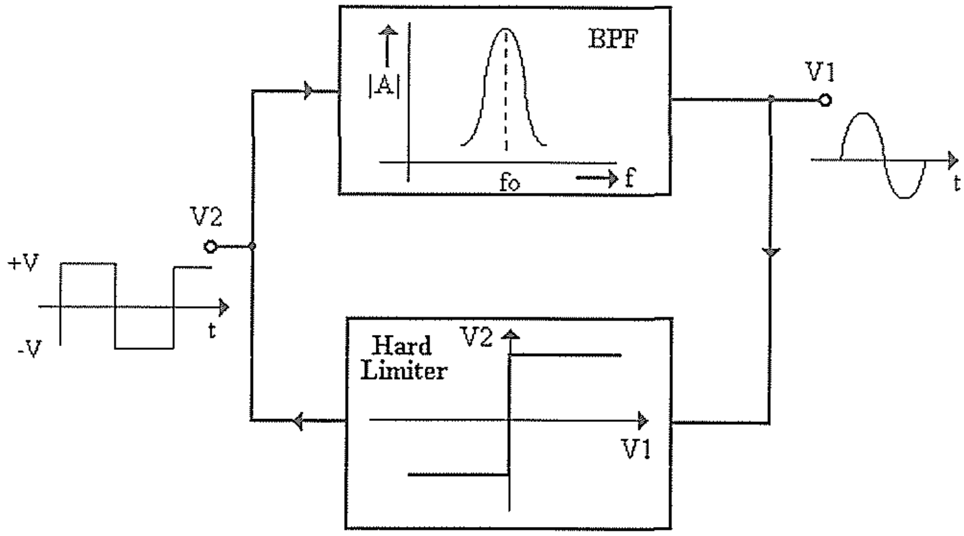


Fig. 4.2(a) Oscillator based on tuned filter and hard limiter. Source [10].

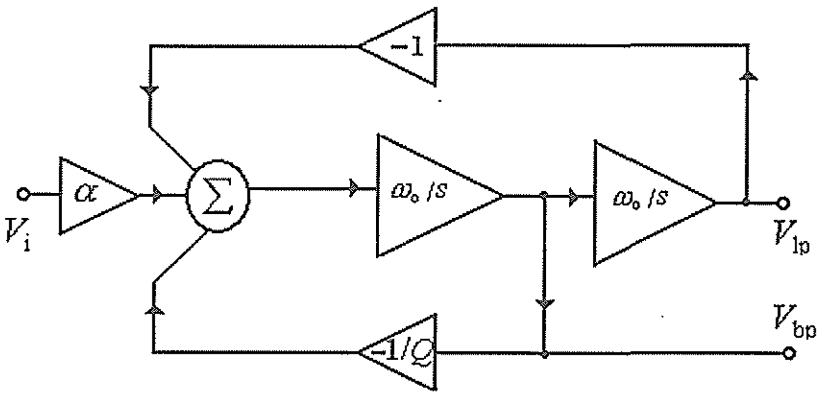


Fig. 4.2(b) Band pass filter realization using two non-inverting integrator loop.

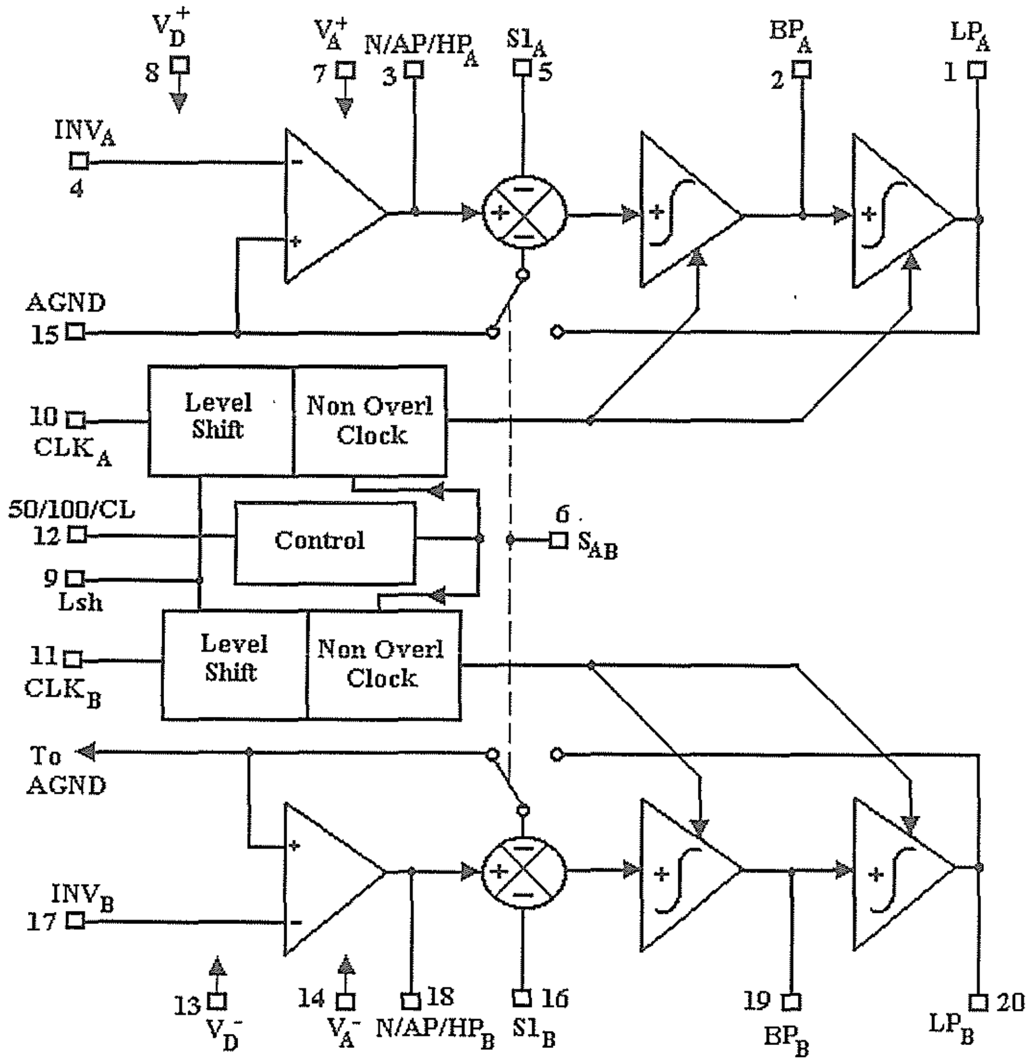


Fig. 4.3 Block diagram of SCF LMF 100. Source [11].

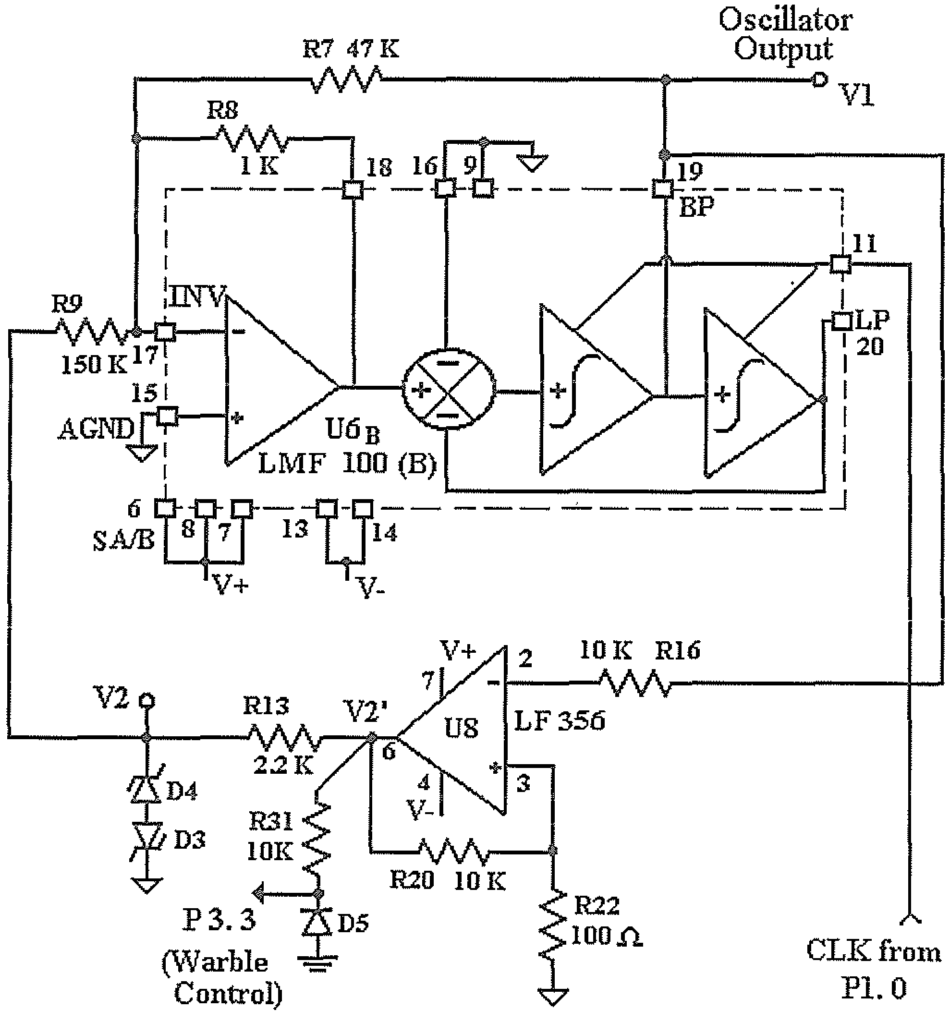


Fig. 4.4 SCF based quadrature oscillator using LMF 100

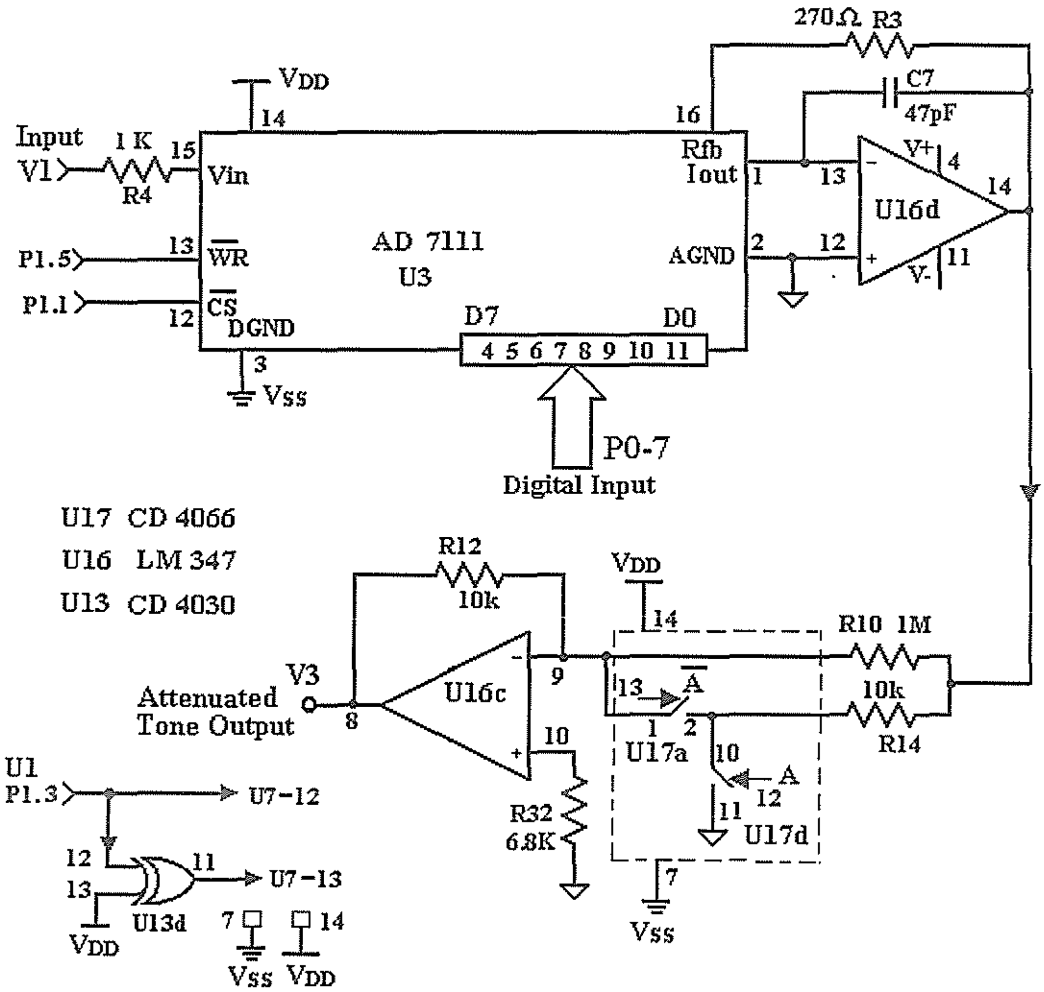


Fig. 4.5 Attenuator circuit for tone

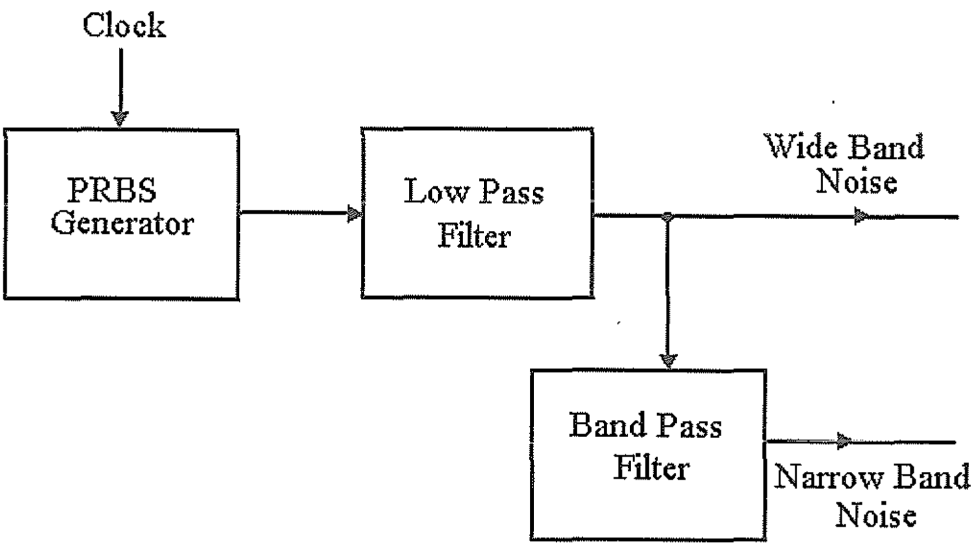


Fig. 4.6 Block diagram for noise generator

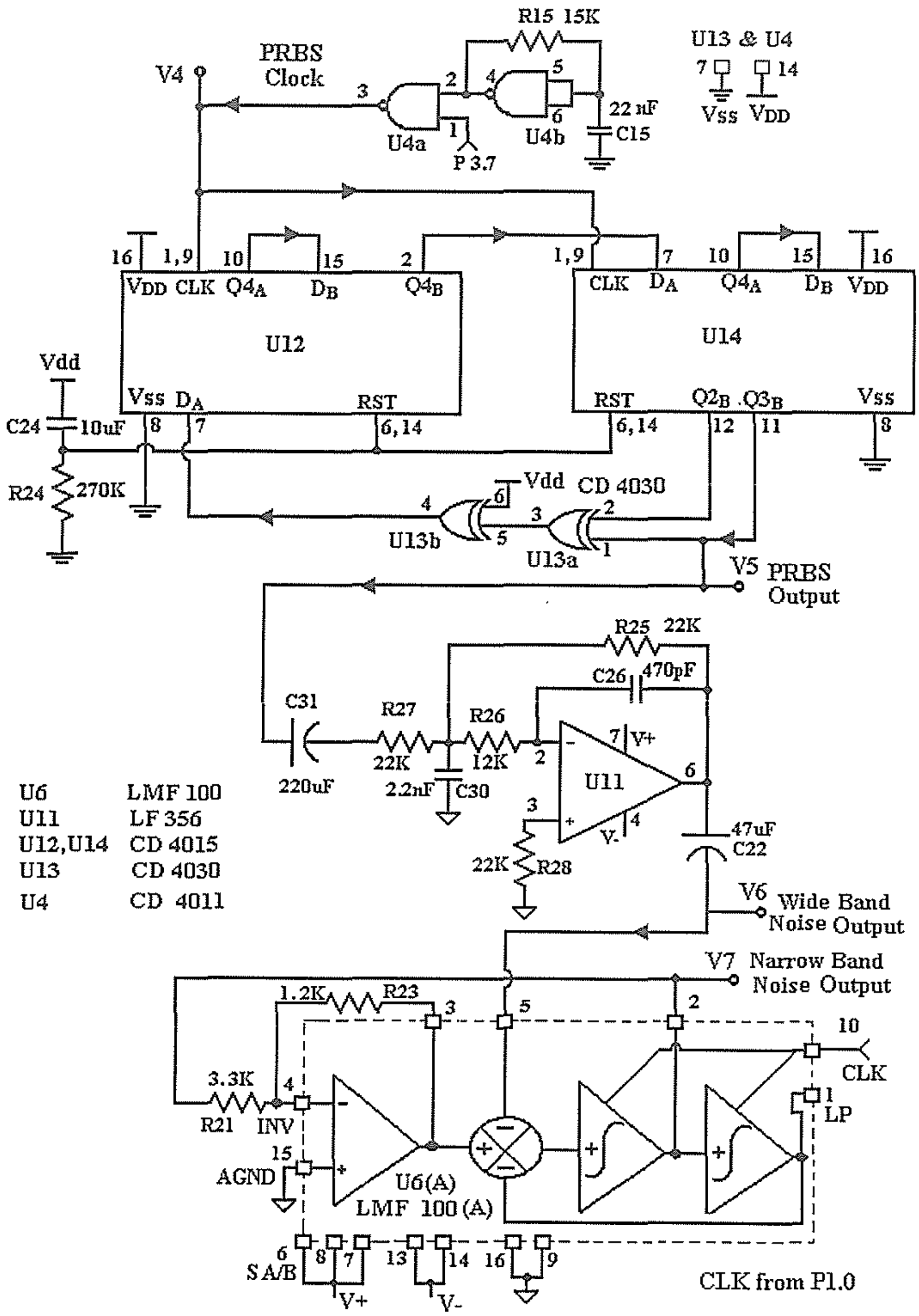


Fig. 4.7 Circuit for noise generator scheme

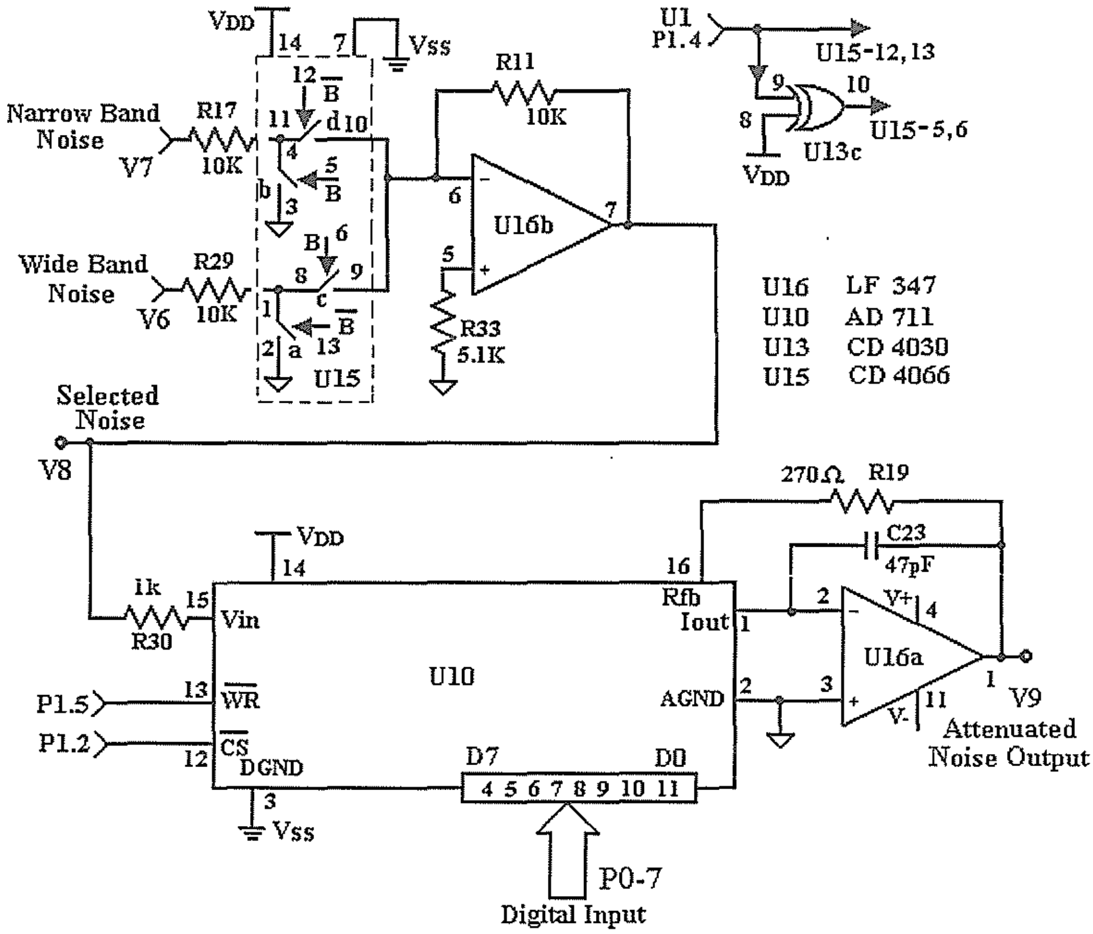


Fig. 4.8 Noise selector and attenuator circuit

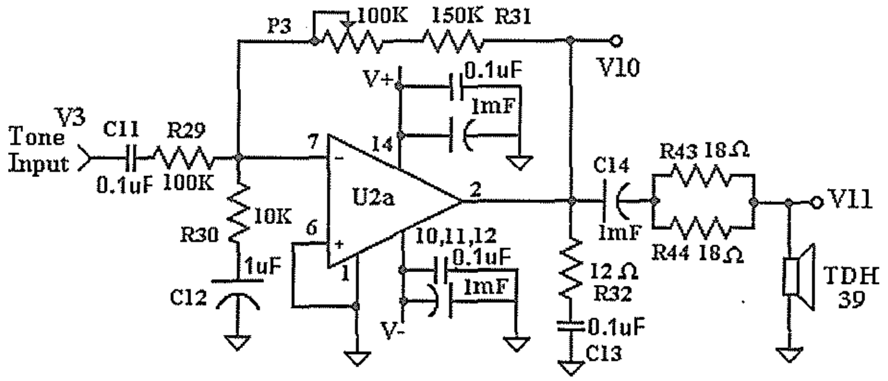


Fig. 4.9(a) Power amplifier for headphone

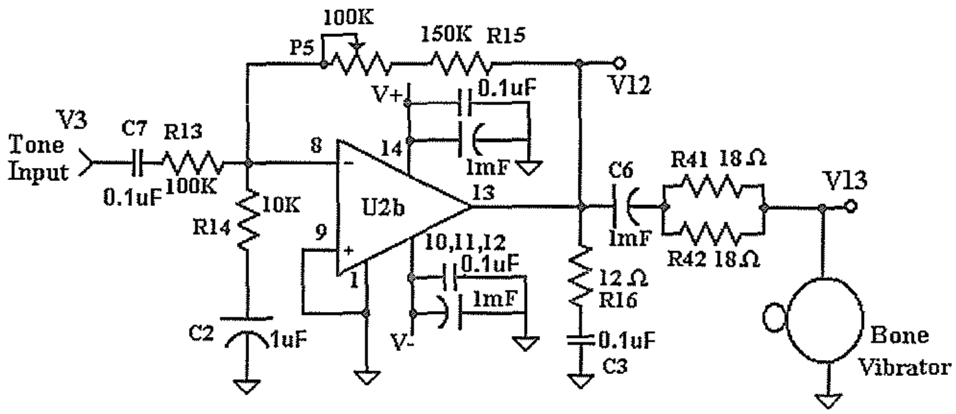


Fig. 4.9(b) Power amplifier for bone vibrator

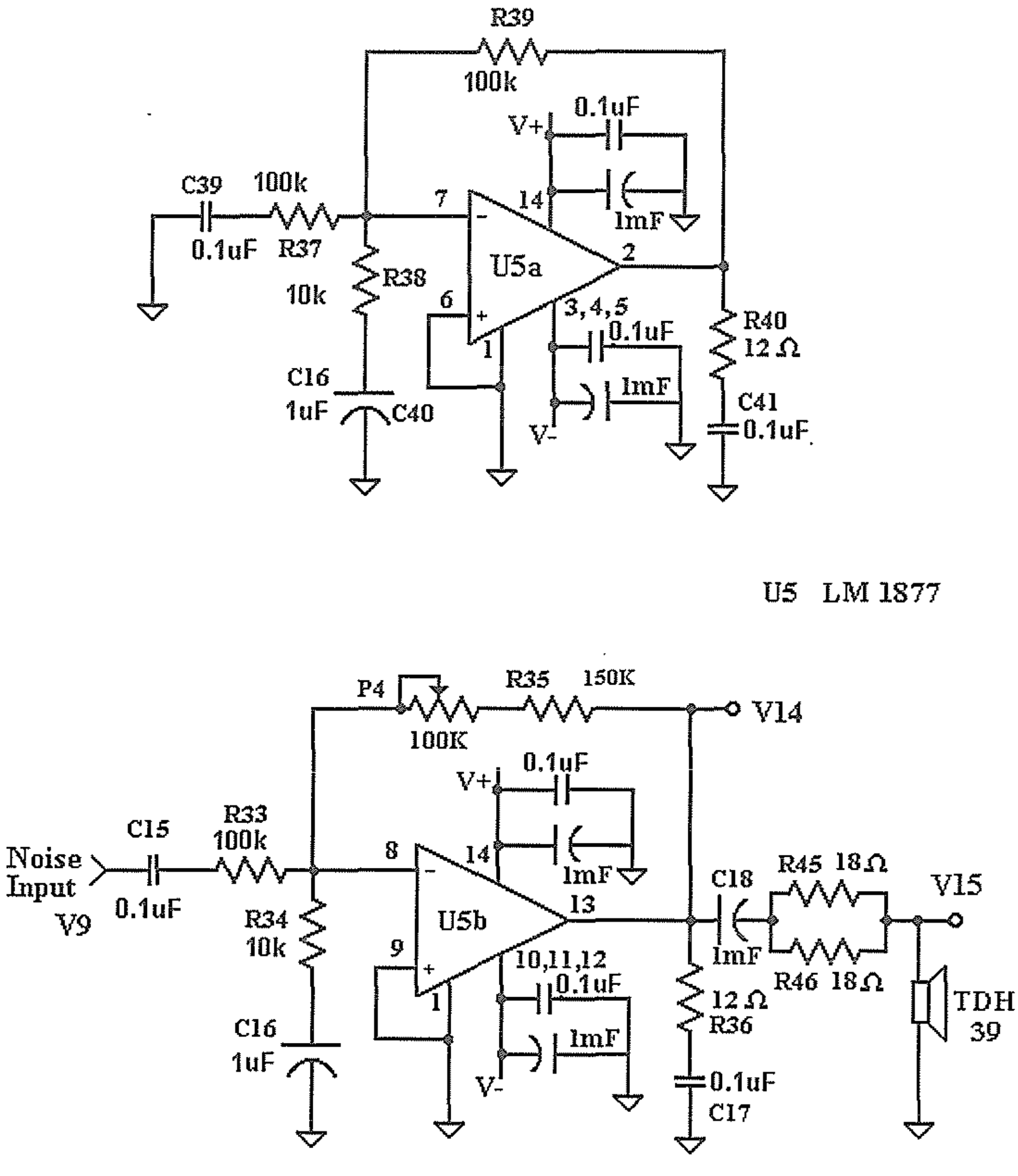


Fig. 4.10 Power amplifier for noise

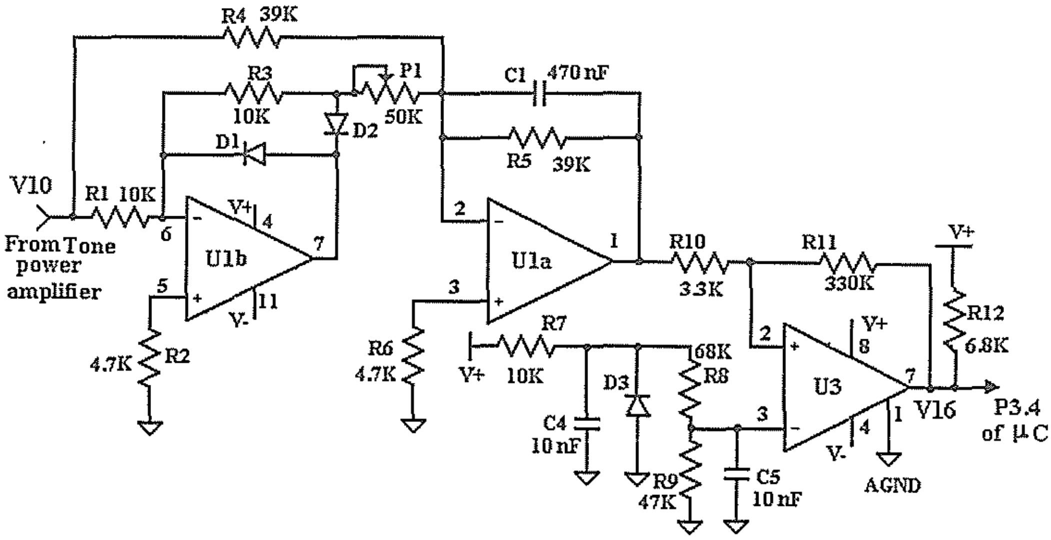


Fig. 4.11(a) Self-test circuit for tone channel

U1, U6	LF347
U3, U4	LM311
D1, D2, D4, D5	1N4148
D3, D6	

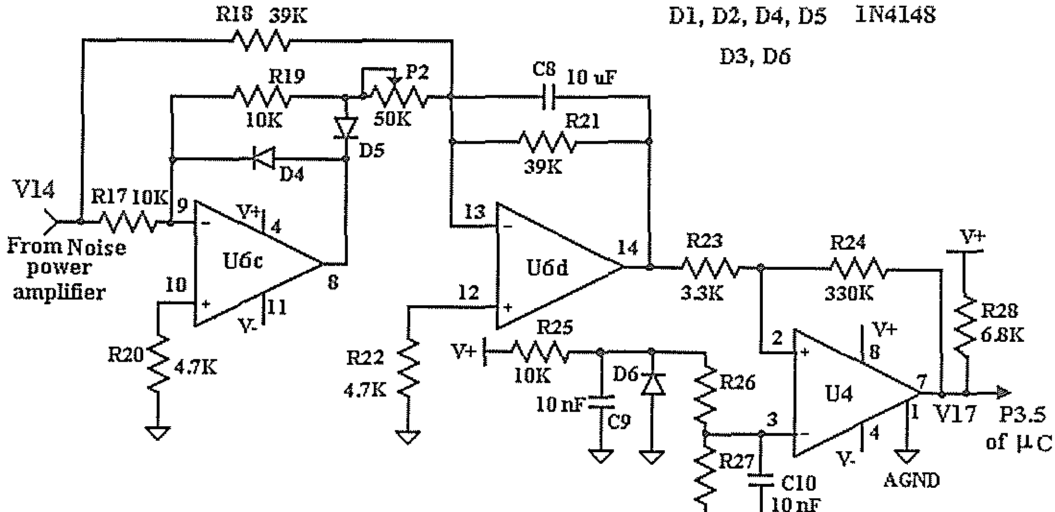


Fig. 4.11(b) Self-test circuit for noise channel

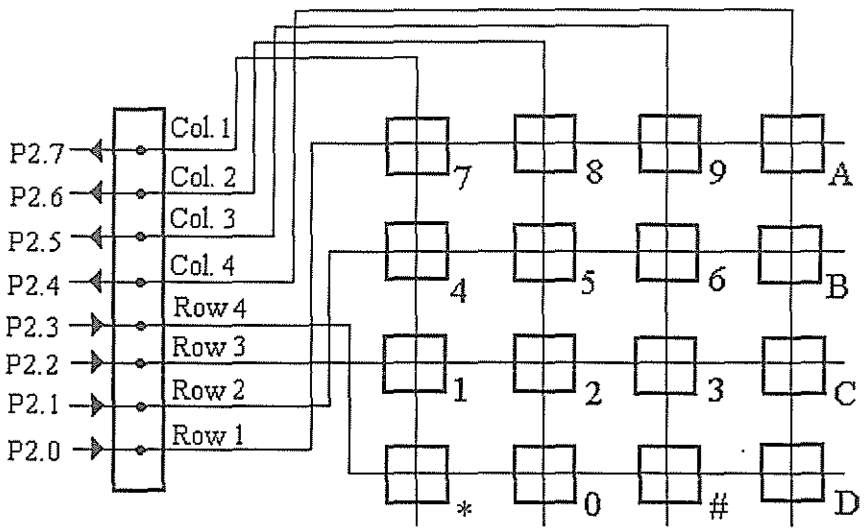


Fig. 4.12(a) Keypad interfacing with microcontroller

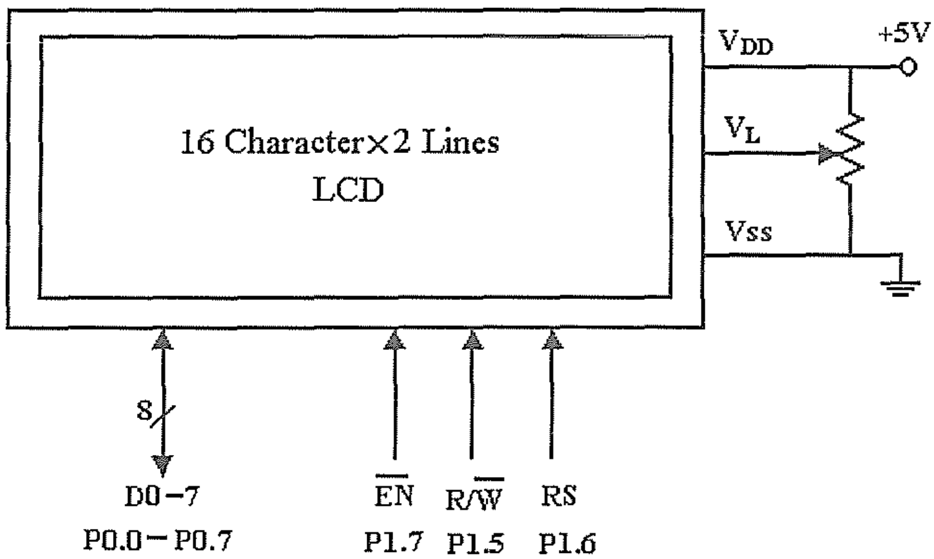
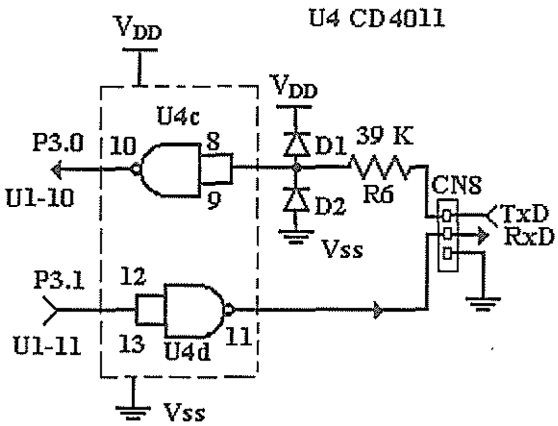


Fig. 4.12(b) 16 characters × 2 lines ODM-16216S LCD



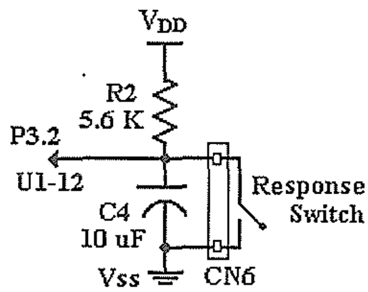


Fig. 4.13 Circuit for serial interface

Fig. 4.14 Debounce for response switch

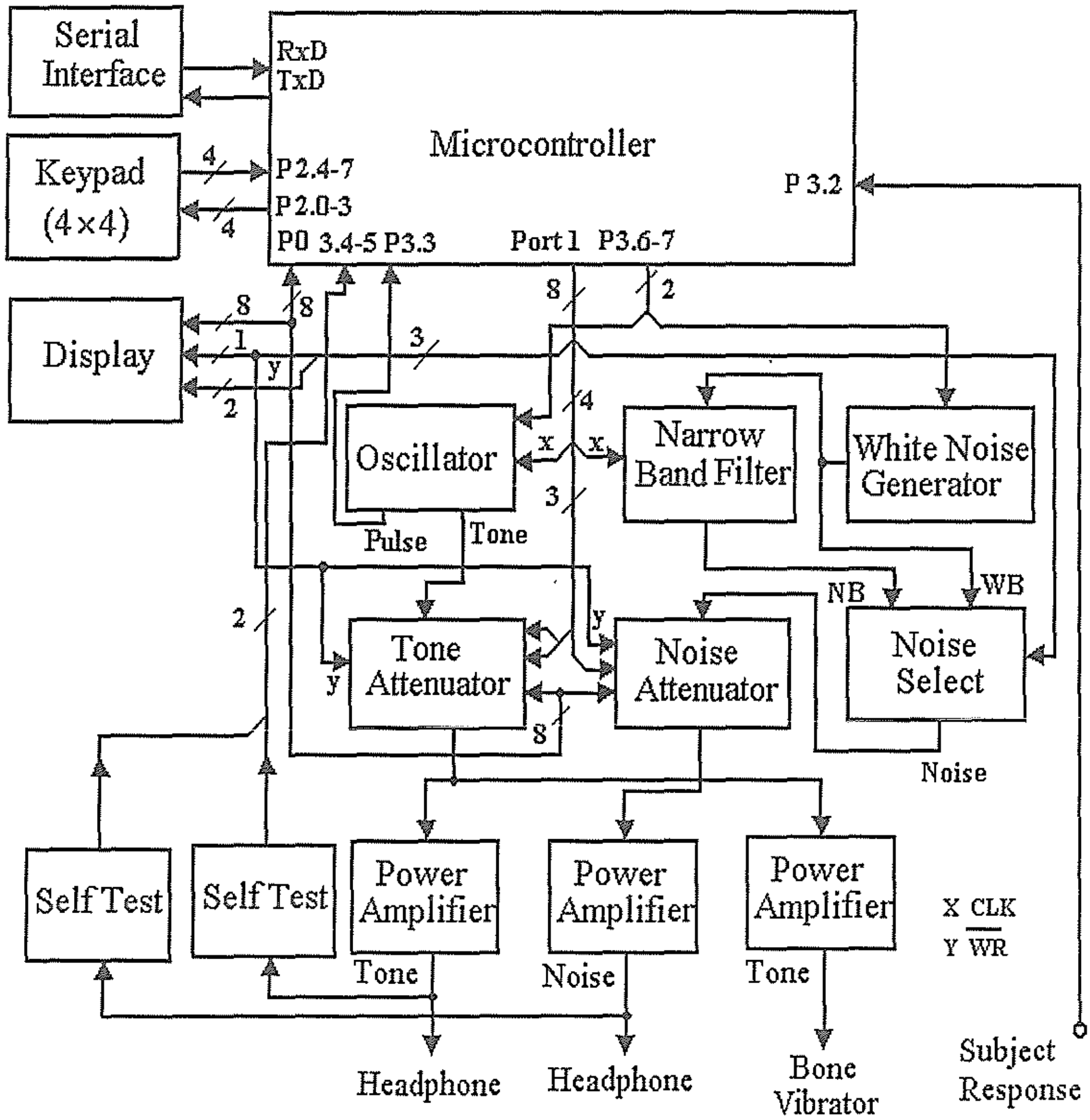


Fig. 4.15 Microcontroller interfacing with various blocks

Chapter 5

SYSTEM DESCRIPTION

The hardware consisting of tone oscillator, noise generator, attenuator, amplifier, circuit for self test, 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. 4.15 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 $\overline{INT1}$ is used for warble tone generation. Response switch is connected to external interrupt pin $\overline{INT0}$. Port pins P1.1 and P1.2 are used for selecting ($\overline{CS}$) 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 $R/\overline{W}$ 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. 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 self test 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.

5.1 Tone Generation

Pure tones with frequency 250Hz, 500Hz, 1kHz, 1.5kHz, 2kHz, 3kHz, 4 kHz, 6kHz, and 7.5 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

$$f_{CLK} = (\text{Oscillator Freq.}/4) / \{65536 - (\text{RCAP2})_D\}$$

where $(\text{RCAP2})_D$ is the decimal equivalent of the hex numbers in registers (RCAP2H, RCAP2L), taken as a single 16-bit register. The 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 sweep cycles in a second, and number of steps in each cycle was selected such that the frequency modulated tone is perceived as a warble, i.e. 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 7500 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%.

In pure tone mode, the external interrupt 1 will be disabled and the stimuli will be of pure tone type. In warble tone mode, 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.

5.2 Attenuator Control

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 dBm required for producing 30 dB HL at the headphone output for each frequency is calculated by referring to the Table A.1. Corresponding attenuation counts are calculated. Thus, a table of counts, $A_{30}(F)$, corresponding to 30 dB HL for different frequencies is prepared. The counts are given in Table 5.3.

The attenuation count for a specific dB HL can be calculated with respect to the count for 30 dB HL, in a frequency independent manner.

$$A_L = (L - 30)/ 0.375$$

The dB HL values actually used are in steps of 5 dB, and therefore in order to avoid the program code for calculation of A_L, a table has been used.

For $L \leq 30$,

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

$$A(L,F) = A_{30}(F) + A_{Lp}$$

For $L > 30$

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

$$A(L,F) = A_{30}(F) - A_{Ln}$$

For $L \leq 30$, the value of count dB_Val = $A(L,F)$ should be checked every time, because AD 7111 can provide maximum attenuation of 88.5 dB i.e. 236 decimal count. Hence, if dB_Val $\leq$ 0ECH (236 decimal), then don't make 40 dB passive attenuator on. Otherwise make 40 dB attenuator on. The attenuation of 40 dB corresponds to attenuation count of 40/0.375 = 107.Hence, this value should be subtracted from dB_Val, if the 40-dB attenuator is on. The dB_Val is loaded to the attenuator latch of the chip AD 711. The data is latched on the rising edge of the WR. Before latching the data, 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.

5.3 Keypad, Display, and Serial Interface

A 4×4-matrix keypad is directly connected to port 2 as shown earlier in Fig. 4.12(a). Keypad is scanned at intervals of 5 ms. The interrupt for scanning is generated using Timer 0. Keys are scanned by outputting a row scan pattern and reading back column pattern. The row scan pattern makes one row low at a time.

The scanning is done in the background, and scan result is communicated via flag "newflg" and memory locations "newrow" and "newcol". After each scan, the column data are checked for keypress (there will be one zero for a valid key press). This is compared with previous values of "newrow" and "newcol". The keypress value is accepted if the valid data remain the same in three successive scan cycels. This results in a software implemented debounce interval of 60 ms.

The main program, when looking for a key pressed, checks the status of "newflag" and if it is set, it takes the key value from "newrow" and "newcol". Multiple key presses are rejected as invalid data pattern or a failure to match for three cycles. Table 5.4 provides the data write and read data on port 2 pins during the scan cycle, for each key pressed. One complete scan cycle, for all four rows takes 20 ms.

The LCD display consists of 8 data lines and 3 control lines (RS, R/W, EN), which are interfaced to microcontroller, as shown earlier in Fig. 4.12(b). Control pin R/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 EN. The pulse width of 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 data transfer rate of 1200 bits/s is selected for serial communication with the printer EPSON LX 800. The printer is set in the mode of receiving 7-bit data with even parity check . The program on the microcontroller sends even parity as MSB followed by 7-bit data. Timer 1 is used to generate the data transfer rate.

5.4 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.

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 at all.

Tone Up. When this key is pressed, the attenuation of the tone is reduced by 5 dB thus, increasing level of the tone by 5 dB.

Tone Down. When this key is pressed, the attenuation of the tone is increased by 5 dB, and thus the level of the tone reduces by 5 dB.

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.

Noise down. This key is used to decrease the level of masking noise.

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 been lapsed. For switching off the tones, microcontroller stops the timer2.

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, the threshold value for the frequency 250 Hz will be displayed and device will display the message 'Next?'. If user wants to see the threshold obtained for the next frequency, press "OK" key again. If 'CANCEL' key is pressed, the device will come out of the recall mode and 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.

5.5 Self-test Routine

When the instrument is switched on, a self test routine is executed, making use of "self test" circuit as shown earlier in Fig. 4.11(a) and (b). In this mode, the level of test tone is increased in steps of 0.75 dB. The rectified average value of tone V10 from the power amplifier (headphone amplifier) is compared with the reference (Fig. 4.9 (a)). 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 attenuator counts for frequencies of 250 Hz, 1 kHz, 3 kHz, and 6 kHz are averaged. Then, the count for each frequency is compared with the averaged one. A tolerance of two counts i.e. 0.75 dB is permittted. If the count does not differ more than 2 counts, 'pass ok' and the corresponding frequency is displayed. If the difference is more than 2 counts, 'No pass' is displayed alongwith that particular frequency. The display for various frequencies is shown with an interval of 1s. The self test here is checking for any frequency dependent variability. By pressing the response switch, one can come out of the self-test mode.

For noise, the test is done by using port pin P3.5. Here the attenuator count is compared with a reference count and a tolerance of ± 2 is accepted for "Pass-WB Noise"

The self test circuit can be used, by using appropriate modification in the program to check for the calibration for various type of headphones. By pressing the response switch, one can come out of the self-test mode.

5.6 Test Algorithm in Automated Mode

The flowchart representation of the normal audiometric procedure for threshold determination is shown in Fig.5.2. 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 10 dB. If the patient does not hear it, the tone is again raised by 5 dB. In this way by several presentations, the hearing threshold is obtained. The minimum presentation level at which the subject responds at least 50% times, is taken as the hearing threshold [4] [2]. This algorithm has been implemented for the automated mode.

5.7 Operation Sequence

Operation sequence of the instrument is given in the Fig. 5.3. After power is made on, a self-test routine is executed. The self-test routine can be terminated by pressing the response switch. Once the self-test is over, various options (viz. tone type, tone duration, mode of operation, mode of conduction, and noise type) are selected. Tone can be either pure or warble type. The tone duration can be 2s, 3s, 4s, or continuous. The mode of operation can be either manual or auto. The output device can be either headphone (air conduction) or bone vibrator (bone conduction). The noise can be of wide band or narrow band type.

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 7.5 kHz. Whereas, for bone conduction it is from 250 Hz to 4 kHz. Tone level for air conduction is from 0 dB HL to 100 dB HL and for bone conduction it is 0 dB HL to 50 dB HL. The noise level ranges from 0 dB HL 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 interrupted by software. 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 only the initial parameters. The instrument does rest all in accordance with the test algorithm. The thresholds obtained are stored automatically.

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.

Table 5.1

Number to be loaded in RCAP2 for different frequencies (for 12 MHz crystal).

Frequency (Hz)	Number to be loaded (Hex)
250	FF88
500	FFC4
1000	FFE2
1500	FFEC

Frequency (Hz)	Number to be loaded (Hex)
2000	FFE2
3000	FFEC
4000	FFF1
6000	FFF6
7500	FFF8

Oscillator is operated in 100:1 mode from 250 to 1500 Hz.

Oscillator is operated in 50:1 mode from 2000 to 7500 Hz.

Table 5.2

Different parameters for warble tone. (for 12 MHz crystal)

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	9	1	27	7B to 93
500	9	1	55	C0 to C8
1000	7	1	142	DF to E5
1500	5	1.5	200	EA to EE
2000	7	1	285	DF to E5
3000	5	1.5	400	EA to EE
4000	5	1.5	533	EF to F3
6000	3	2	1000	F5 to F7
7500	3	2	1250	F7 to F9

Oscillator is operated in 100:1 mode from 250 to 1500 Hz.
Oscillator is operated in 50:1 mode from 2000 to 7500 Hz.

To obtain ± 10 % deviation for 250 Hz, the count is increased from 7B H to 93 H in steps of 3. Whereas, for all other frequencies the count increment is in steps of 1.

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

Row No.	Scan Data	Data read for key presses															
		0	1	2	3	4	5	6	7	8	9	A	B	C	D	*	#
1	FE	FF	FF	FF	FF	FF	FF	FF	7E	BE	DE	EE	FF	FF	FF	FF	FF
2	FD	FF	FF	FF	FF	7D	BD	DD	FF	FF	FF	FF	ED	FF	FF	FF	FF
3	FB	FF	7B	BB	DB	FF	FF	FF	FF	FF	FF	FF	FF	EB	FF	FF	FF
4	F7	B7	FF	FF	FF	FF	FF	FF	FF	FF	FF	FF	FF	FF	E7	77	D7

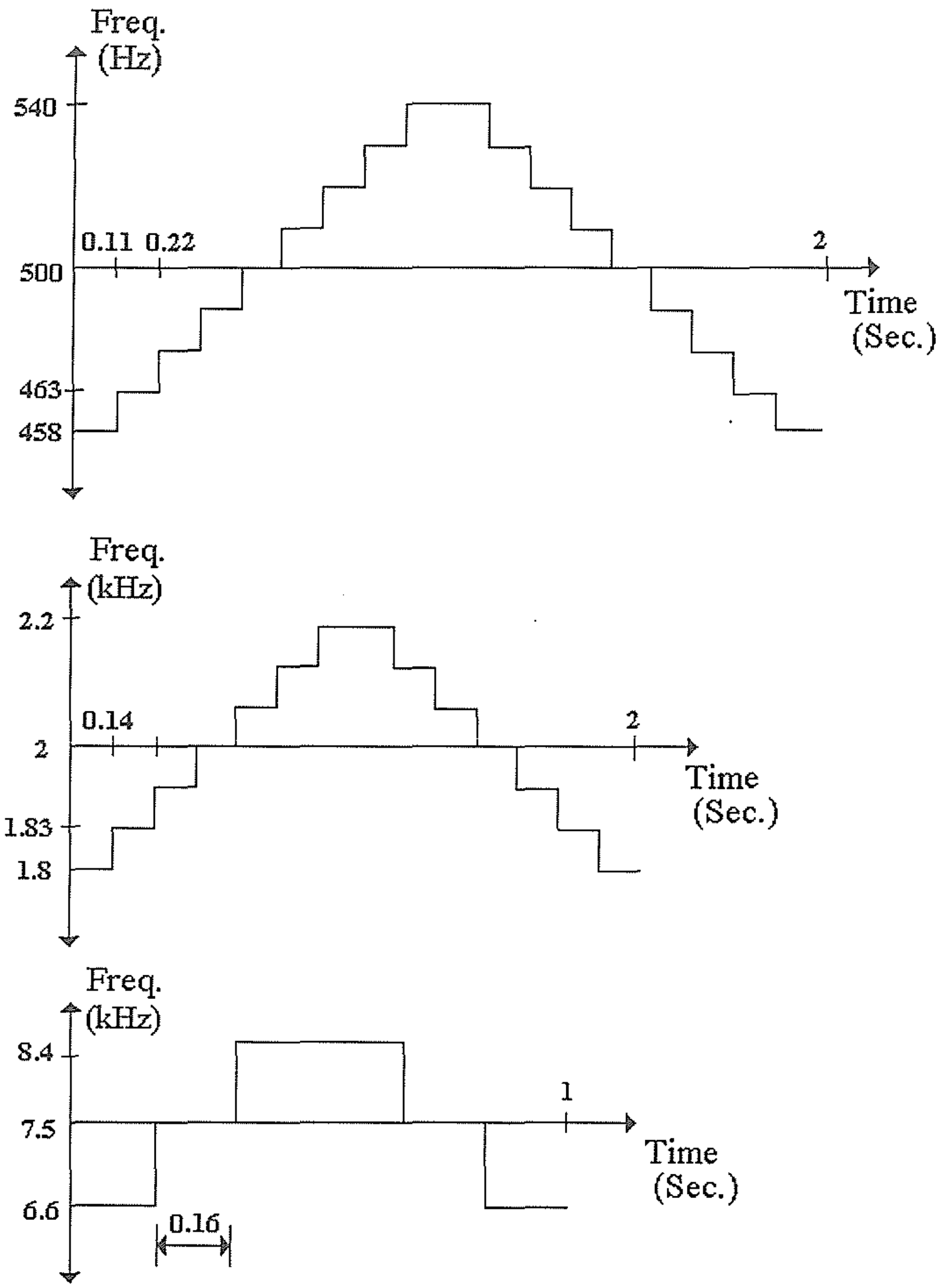
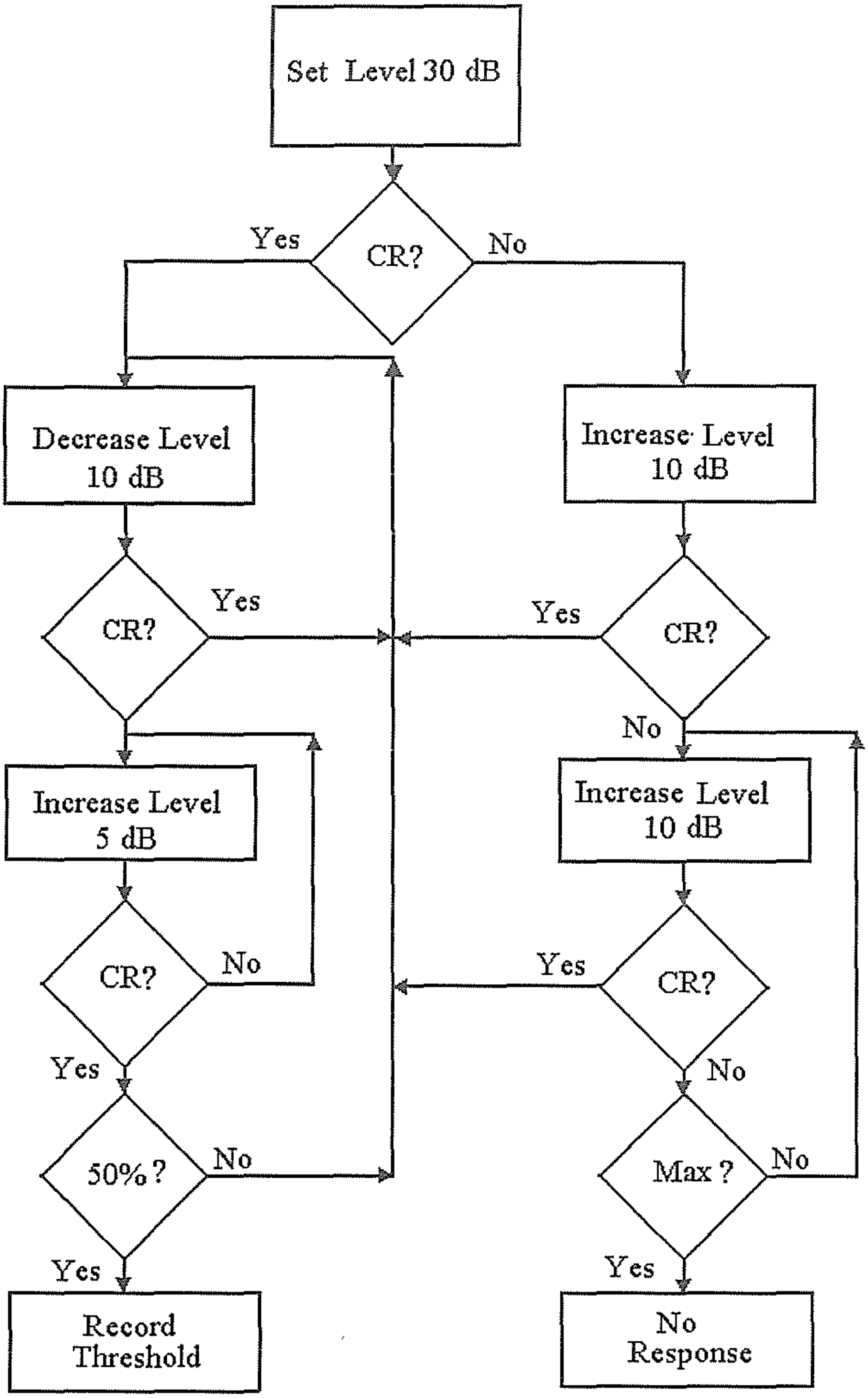


Fig. 5.1 Frequency sweep for 500 Hz, 2kHz, and 7.5 kHz



CR? : Check response

Fig. 5.2 Flow chart for pure tone audiometry

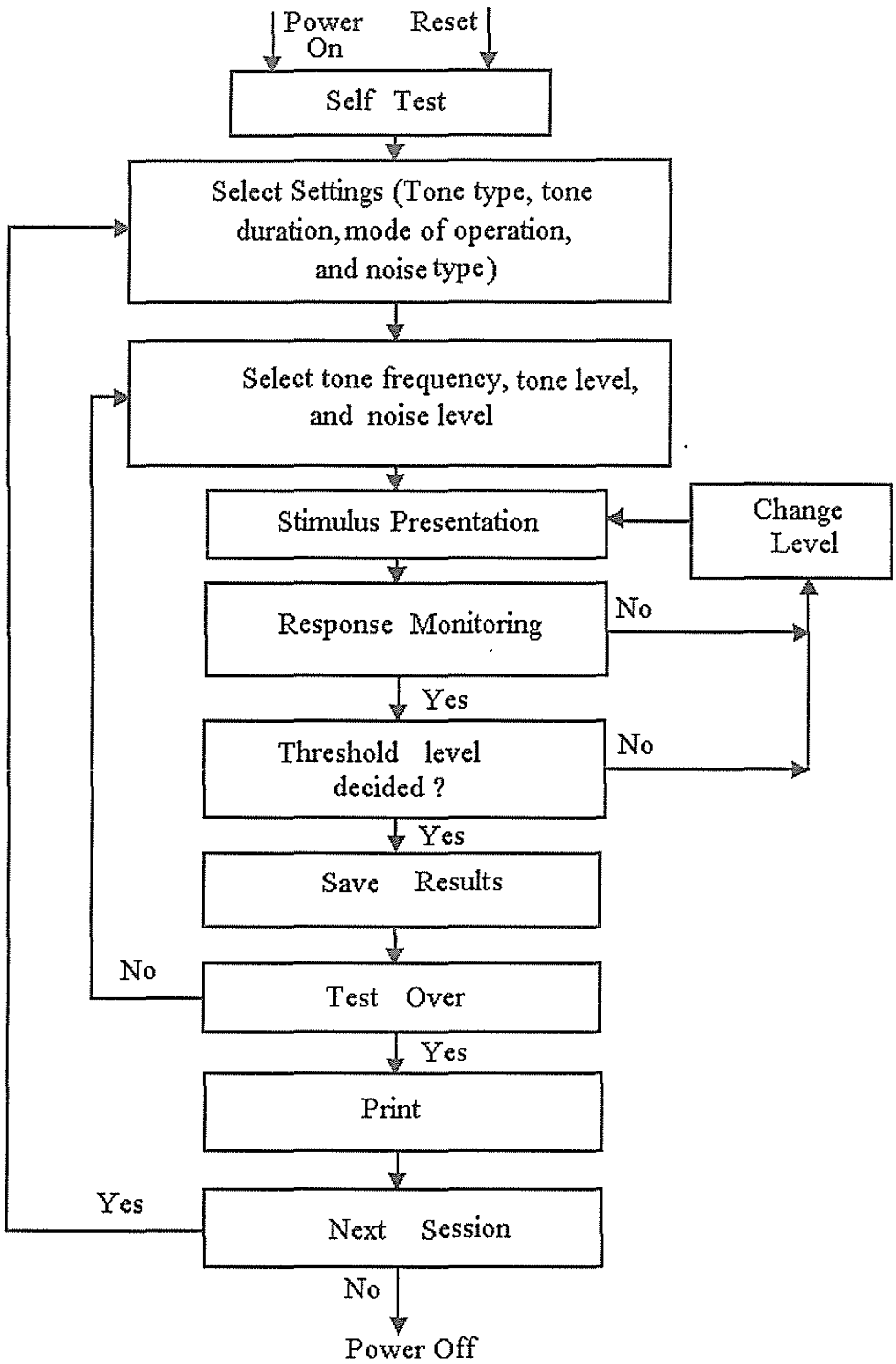


Fig. 5.3 Operation sequence of the instrument

Chapter 6

SYSTEM ASSEMBLY AND TESTING

Almost all the circuit blocks were tested separately for satisfactory operation, under appropriate interfacing/control/load conditions. Subsequently, for the purpose of PCB assembly, the circuit was divided in two parts. PCB-1 consists of the tone, noise circuits, and the microcontroller. PCB-2 consists of the power amplifier and circuit for self test. LCD display unit and keypad are connected to PCB-1. Partitioning of the circuit was done keeping in view (a) interconnection between the boards, (b) modular upgradation of the design in the future. The PCB's have been assembled in a specially fabricated cabinet 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. 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 every where 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 D. 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 capacitor. Thick copper planes of supply and ground are provided on opposite sides of the PCB. This gives the effect of distributed capacitance across the PCB, thus improving the decoupling effect. Each IC is decoupled by 0.1 μF ceramic capacitor placed as close as possible to the IC. A Care is taken that it is connected right at the supply and ground pins of the particular IC. A Special care is taken while 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 feedthrough from input to output [9].

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.

6.2 Assembly

A special cabinet is fabricated for assembling the circuit. Two sheets of acrylic of 3-mm thickness are used to make cabinet. The dimensions and other details of top and bottom plate of the cabinet are shown in Appendix E. Two PCBs are mounted horizontally on the top plate. The keypad and display are also placed on the top plate. Opening the cabinet and testing the circuit becomes easier due to connection of all PCBs and accessories on top plate. Since the display needs some elevation for proper vision, a stand giving an angle of 30 degree is made. The stand covers the front portion of top plate. Hence, all slots for various connectors and switches are provided on the top plate backside. The slots for power supply, power on switch, and reset switch is provided on the left side. The slots for tone outputs (air and bone), noise, and response are provided in the middle. The extreme right side slot is for serial transmission. A slot exactly equal to the size of LCD display is cut on the top plate (top side). The keypad is placed below it. The keypad and display are placed towards left leaving enough space for printing instructions to the user on the right side.

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 is observed on dynamic signal analyzer 3561A from HP. Total harmonic distortion (THD) is approximately 51 dB below the fundamental, for all the test frequencies. The stability of the oscillator output is also of a concern. The analog supply voltages are varied from ± 4.5 V to ± 5.5 V and oscillator output amplitude is measured. The Table 6.1 shows the variation in the output. The output is not frequency dependent. However, it varies by 250 mV for a supply variation from ±4.5 to ±5.5V. These variations occur due to change in the current in the voltage level stabilizing diodes (D3, D4).

The tone level in the instrument operation is 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 is observed on the signal analyzer. Fig. 6.1 shows that the power spectrum is flat up to 8 kHz (power spectrum obtained using analysis bandwidth $\Delta f = 62.5$ Hz). The roll-off out side the pass band is 12 dB/octave. The spectrum of narrow band noise for different tone frequencies was observed. Fig. 6.2, 6.3 show the power spectrum of noise centered around 1 kHz, 4 kHz respectively (analysis bandwidth $\Delta f = 31.25$ Hz). From the component values used in the circuit, bandwidth comes out as 0.53 octave. Practically the bandwidth for 1 kHz, 4 kHz noise is 0.55, 0.57 octave respectively. The 20 dB bandwidth is measured as 4 ocatve about the center frequency.

In the self-test circuits, the ripple in the rectifier-averager output affects the operation of the circuit. The a.c. ripple for lowest operating frequency (250 Hz) is 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 has been found to be around 40 mV. Therefore, ripple in dc output does not affect the operation of comparator circuit.

The operation of power amplifiers is tested, and the gain variation has been 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 PC running a terminal emulation program on the PC for capturing the audiometer output of the test results. The instrument was also interfaced to a printer, an EPSON LX 800 printer, fitted with serial interface card (type 8145) for providing RS 232 port. The data transfer rate, number of data bits are set by DIP switches on the serial interface card of the printer [18]. Printing of audiometer test results was obtained with a data transfer rate of 1200 bit/s and even parity check.

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.

Then if the current is supplied from a ± 5V source, the stand by current is 70 mA and maximum current supplied during maximum intensity sound generated is 110 mA.

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

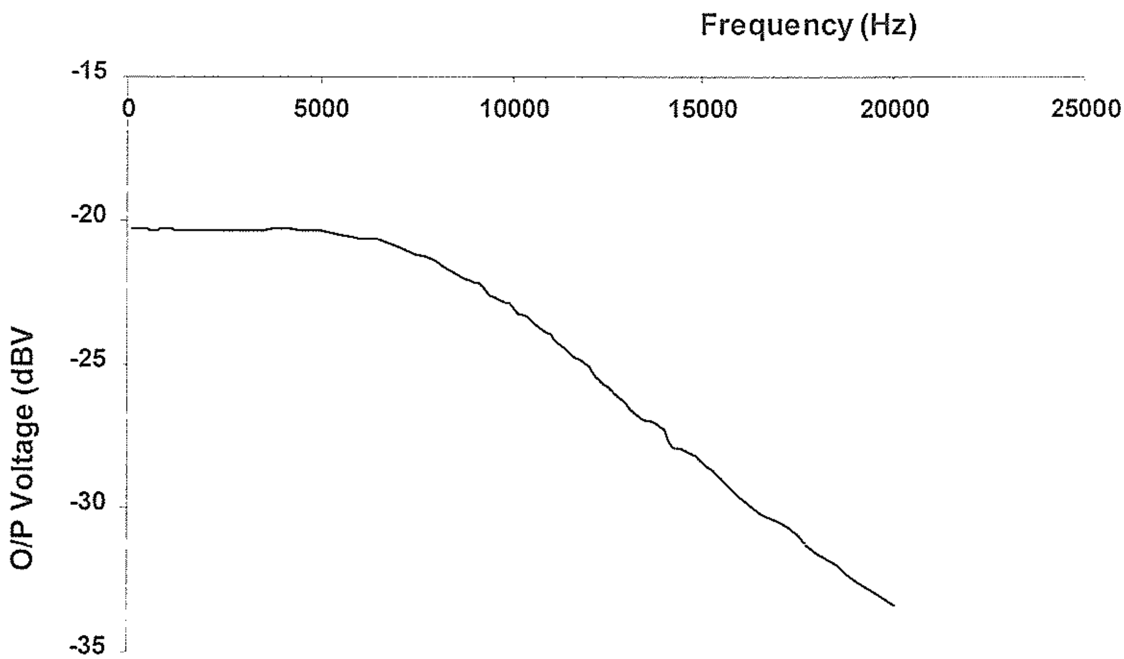


Fig. 6.1 Power spectrum of wide band noise (m/s using Δf = 62.5 Hz)

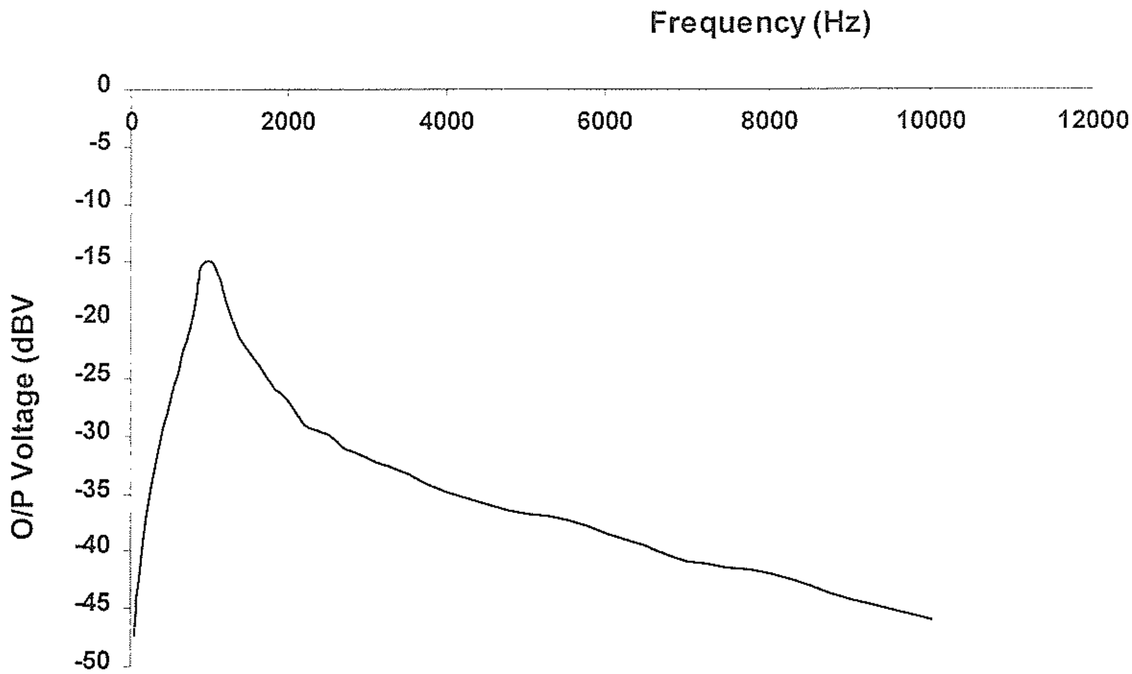


Fig. 6.2 Power spectrum of narrow band noise centered around 1 kHz (m/s using $\Delta f = 31.25$ Hz)

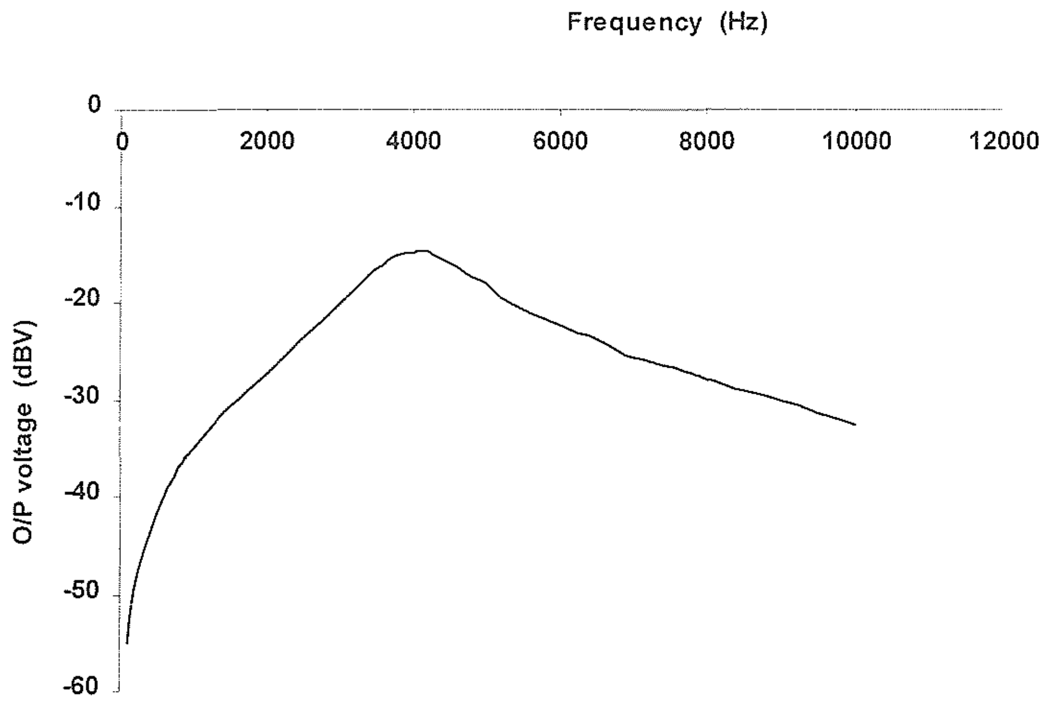


Fig. 6.3 Power spectrum of a narrow band noise centered around 4 kHz (m/s using $\Delta f = 31.25$ Hz)

Chapter 7
SUMMARY AND CONCLUSION

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 tone, and facility of air and bone conduction. It should be usable in mobile clinics and even in rural areas.

A microcontroller based pure tone diagnostic audiometer is developed which operates over a full frequency range of 250 Hz to 7.5 kHz and acoustic output of 0 dB HL 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 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.

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 oscillator output level stabilizing circuit makes use of two back-to-back zener diodes. A change in the power supply changes the comparator output levels, affects the current level in the diodes, and hence the voltage levels. Temperature variations can also change the voltage levels. These values are well within the acceptable limits. The self-test circuit consists of temperature compensated reference diode. It can be used for auto calibration of the instrument. The spectral purity of tone can also be improved by using a high order elliptic filter, which provides sharp cut-off. The design of such type of filter is given in Appendix C.

The highest tone frequency was desired as 8 kHz. But, it could not be achieved since, the one bit change in the RCAP2L register changes the frequency by approximately 1 kHz. Three successive counts produce frequencies 6.6 kHz, 7.5 kHz, and 8.4 kHz. Hence, it was decided to use 7.5 kHz as a test frequency, and the other frequencies are used as steps on either side for warble tone. By using a microcontroller running at a higher clock rate, it may be possible to generate 8 kHz tone. The quality of warble tone will also improve due to higher frequency operation.

The system can be redesigned so that the operator can select the calibration table for the output device from among a number of device types. A calibration chart for each device can be loaded into the serial NVRAM or flash programmable EEPROM. The 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. The counts can be determined for each frequency for a particular device and can be stored in the NVRAM. Further, a flasher along with a low intensity buzzer can be provided to indicate the subject response, so that the operator does not have to look for it in the display.

APPENDIX A

Table A.1

Relation Between HL, SPL, and Input Voltage to the Headphone.

(A) SPL corresponding to 0 dB HL as a function of frequency.

(B) Prescribed $(HL)_{max}$ for an audiometer.

(C) RMS voltage (in dBm) required for producing 100 dB SPL in a TDH-39 headphone.

(D) RMS voltage (in dBm) required for producing 0 dB SPL in a TDH-39 headphone.

(E) RMS voltage (in dBm) required for producing HL_{max} (as given in A) in a TDH-39 headphone.

Frequency (Hz)	**A RETSPL(dB) relative to 20 μPa**	**B Min. upper limit of hearing threshold HL_{max} (dB)**	**C dBm required to produce 100 dB SPL**	**D dBm required to produce 0 dB HL**	**E dBm Required to produce $(HL)_{max}$**
250	25.5	90	– 16.48	– 90.98	– 0.98
500	11.5	100	– 16.25	– 104.75	– 4.75
1000	7	100	– 14	– 107.0	– 7.0
1500	6.5	100	– 11.75	– 105.25	– 5.25
2000	9	100	– 10.99	– 101.99	– 1.99
3000	10	100	– 15.37	– 105.37	– 5.37
4000	9	100	– 14.49	– 105.49	– 5.49
6000	10.5	90	– 8.83	– 98.33	– 8.33
8000	13	80	– 2.39	– 89.39	– 9.39

APPENDIX B

System Specifications

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

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 range of tone output for air conduction is 0 to $(HL)_{max}$, with $(HL)_{max}$ for different frequencies as given below

Frequency (Hz)	250	500	1000	1500	2000	3000	4000	6000	7500
$(HL)_{max}$ (dB)	90	100	100	100	100	100	100	90	80

The range of tone output for bone conduction is 0 to $(HL)_{max}$, with $(HL)_{max}$ for different frequencies as given below

Frequency (Hz)	250	500	1000	1500	2000	3000	4000
$(HL)_{max}$ (dB)	40	50	50	50	50	50	50

Headphone: Instrument has been calibrated for headphone type TDH-39. It can be calibrated for other headphones, by changing a table in the software.

Bone Vibrator: Calibrated for bone vibrator Oticon 70127.

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

Masking noise: broad-band / narrow-band noise, with 0 to 60 dB HL, with intensity level controlled in 5 dB steps.

Wide-band noise: flat spectrum up to 8 kHz, with approx. 12 dB/octave roll off on the higher side.

Narrow-band noise: center frequency = test tone frequency. The 3 dB bandwidth is approximately 0.55 octave about center frequency. The 20 dB bandwidth is 4 octave about center frequency.

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.

Storage memory: for one set of the 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, 7 bit data, and even parity.

Self test: provision for internal monitoring of output levels. Facility of auto calibration.

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

Power consumption: 630 mW for no sound generated

1 W for maximum sound generated

APPENDIX C

Low Pass Filter for Tone Generator

The output of oscillator is a staircase waveform, which is to be converted into a pure sinusoidal signal. The signal frequency ranges from 250 Hz to 7.5 kHz. Since, the SCF IC LMF-100 is operated in 50:1 mode, for 250 Hz the switching frequency is 12.5 kHz. If 12.5 kHz is to attenuated, that too by allowing 7.5 kHz (maximum tone frequency) to pass without attenuation, gives very stringent conditions for filter. Hence, it is decided to use LMF-100 chip in 100:1 mode for lower frequencies i.e. up to 1500 Hz and then use 50:1 mode for higher frequencies. Now the dominant component for 250 Hz is 25 kHz.

Thus, the filter should pass frequencies up to 10 kHz and should attenuate signal by at least 40 dB above 20 kHz. Hence, the filter having sharp roll-off characteristics is needed. A fifth order elliptic filter was built for this purpose [19].

The filter has been realized as a cascade of two biquad sections, each tuned separately. The filter circuit diagram with component values along with section resonant frequency $\mathbf{f_o}$, Quality factor **Q**, and notch frequency $\mathbf{f_n}$ are shown in Fig.C.1. To tune each section to its $\mathbf{f_o}$, **Q**, and $\mathbf{f_n}$; following steps are followed:

(1) $\mathbf{f_o}$: R3 is adjusted to get the section resonant frequency tuned to $\mathbf{f_o}$, by monitoring the band pass output at node 3. There should be 180° phase shift between input output at $\mathbf{f_o}$.

(2) **Q:** R1 is tuned for unity gain (at node 3) at the resonant frequency.

(3) $\mathbf{f_n}$: R5 is tuned to null node 4 output at the notch frequency $\mathbf{f_n}$.

The magnitude transfer function of the filter is shown in Fig. C.2. The signal frequency components below 10 kHz are within 0.01 dB, while components above 20 kHz are attenuated by at least 40 dB.

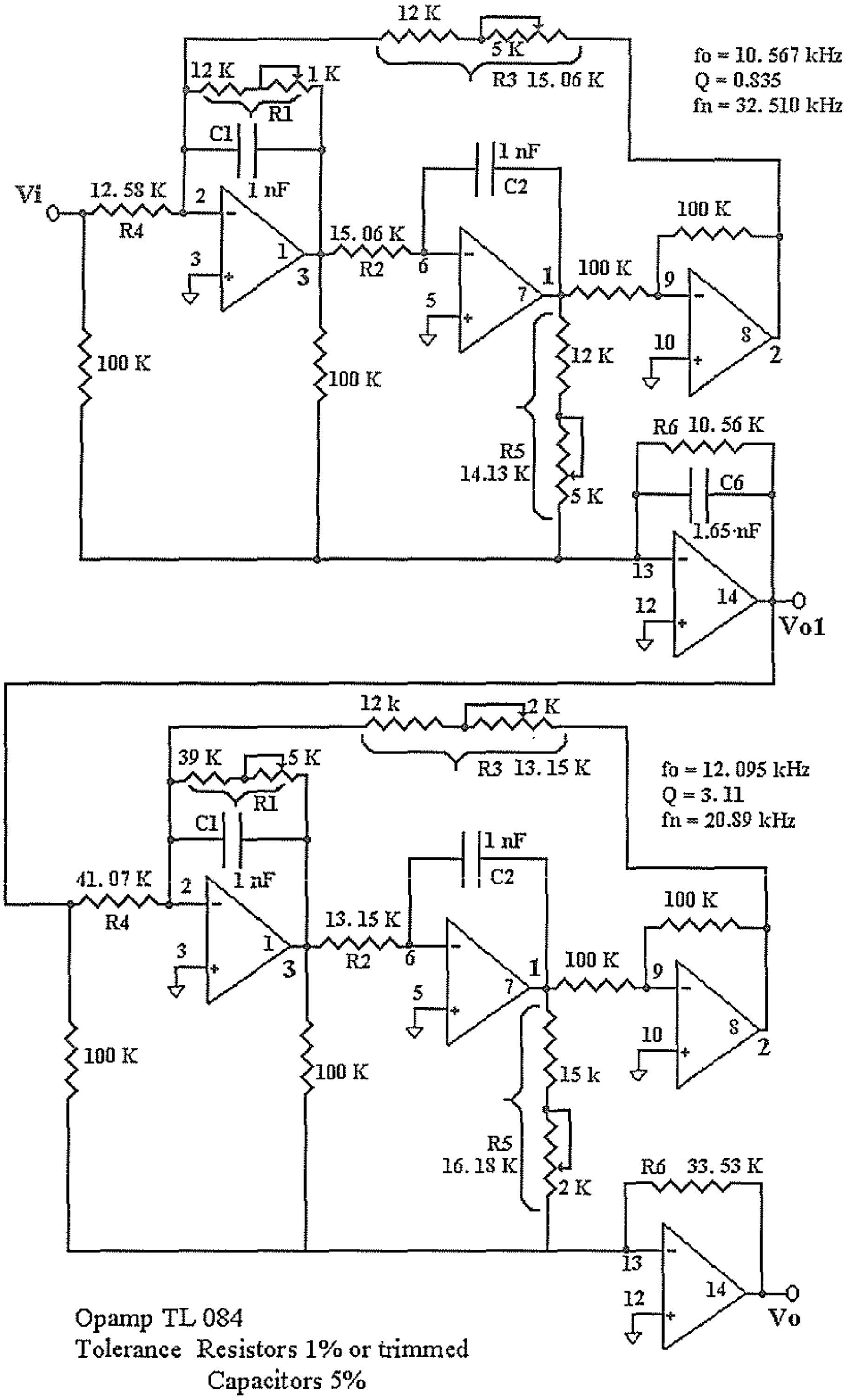


Fig. C.1 5^{th} order elliptic low pass filter

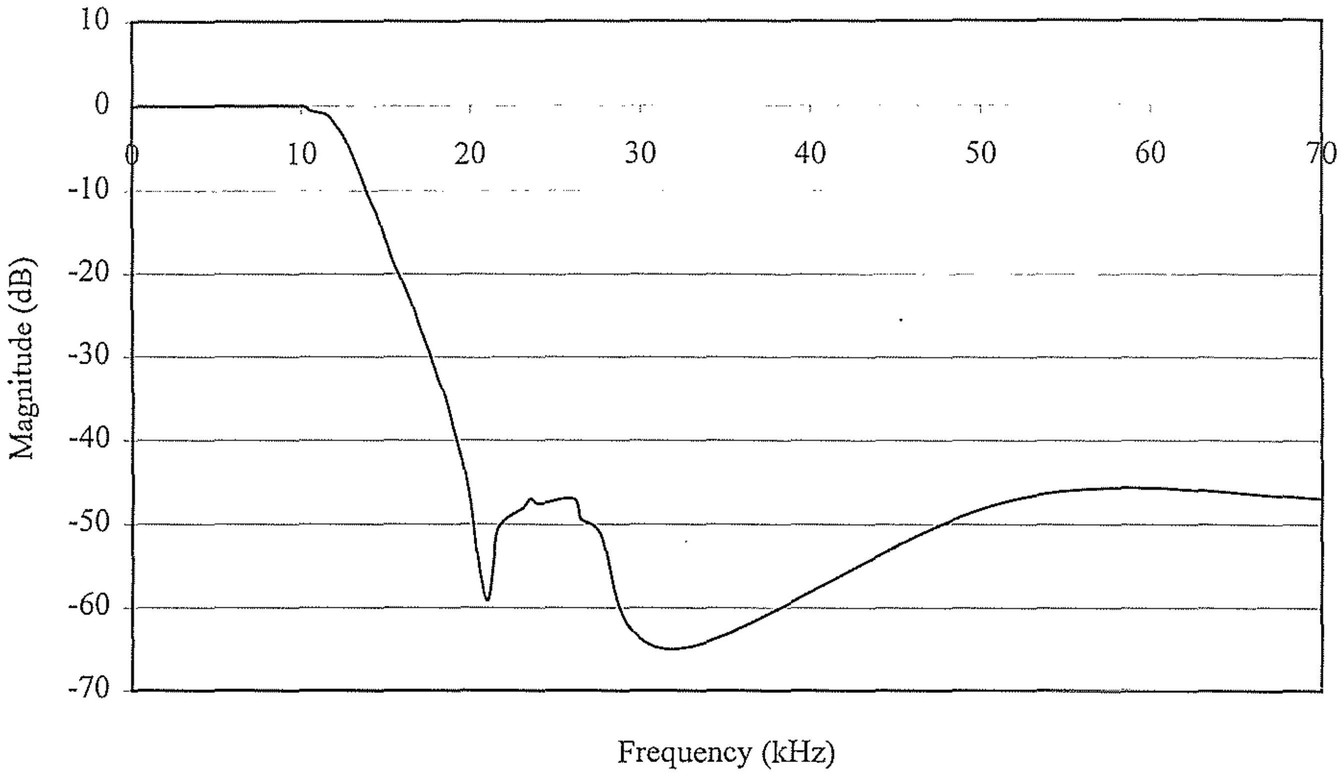
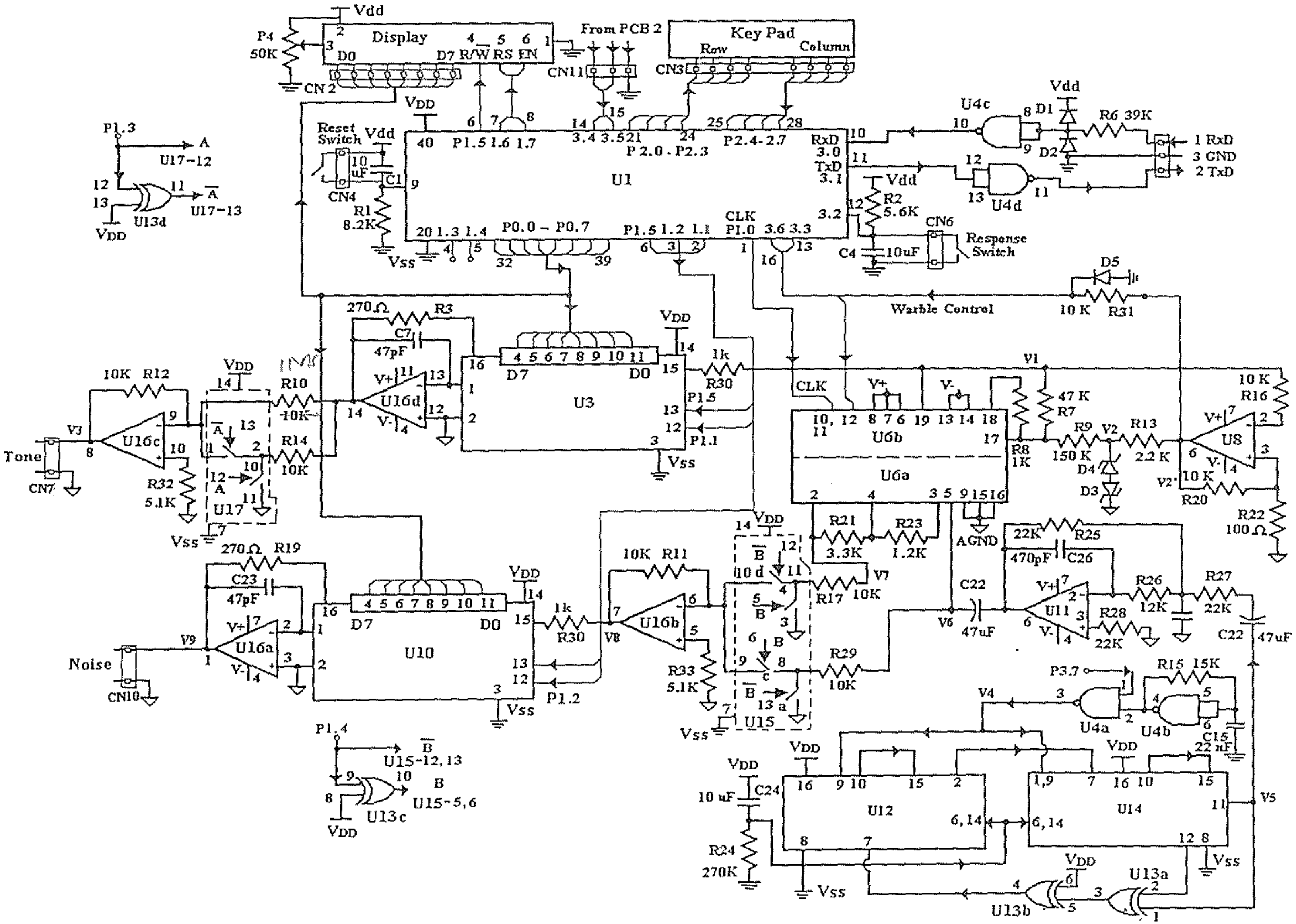


Fig. C.2 Frequency response of the 5^{th} order elliptic low pass filter

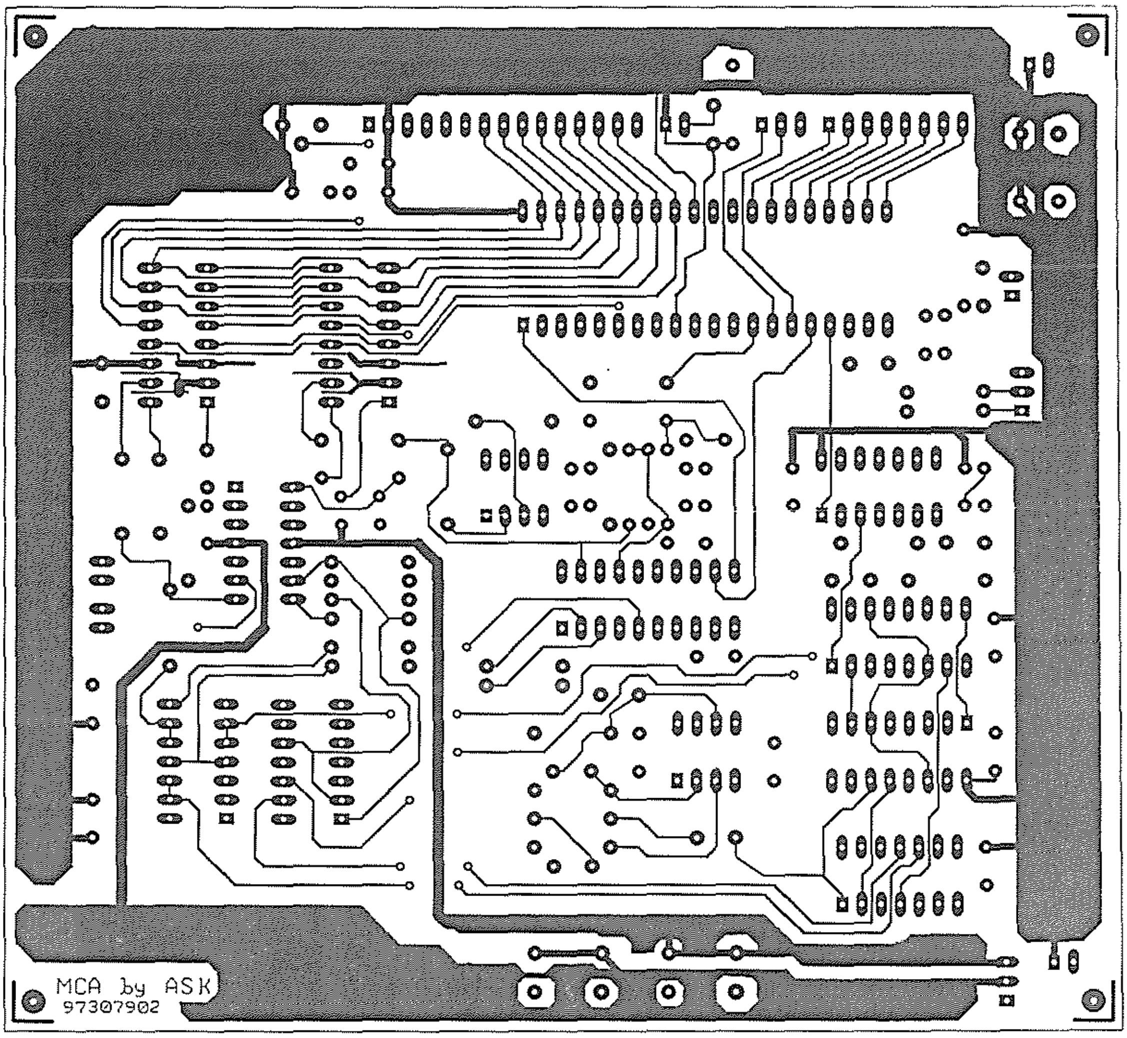
APPENDIX D

Circuit Diagrams and PCB Layouts

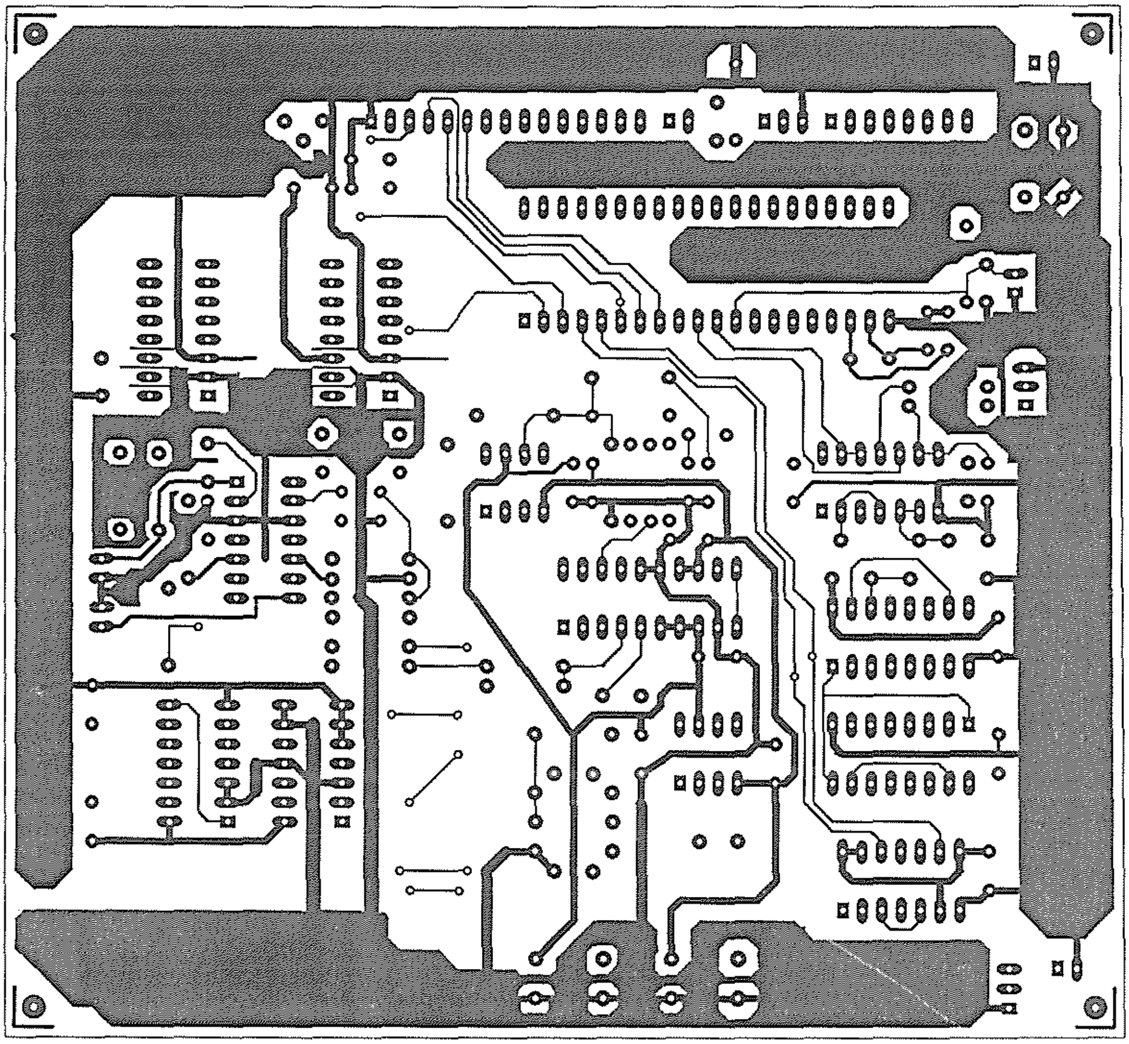


D.1 Circuit diagram for PCB1 (microcontroller and stimulus circuits)

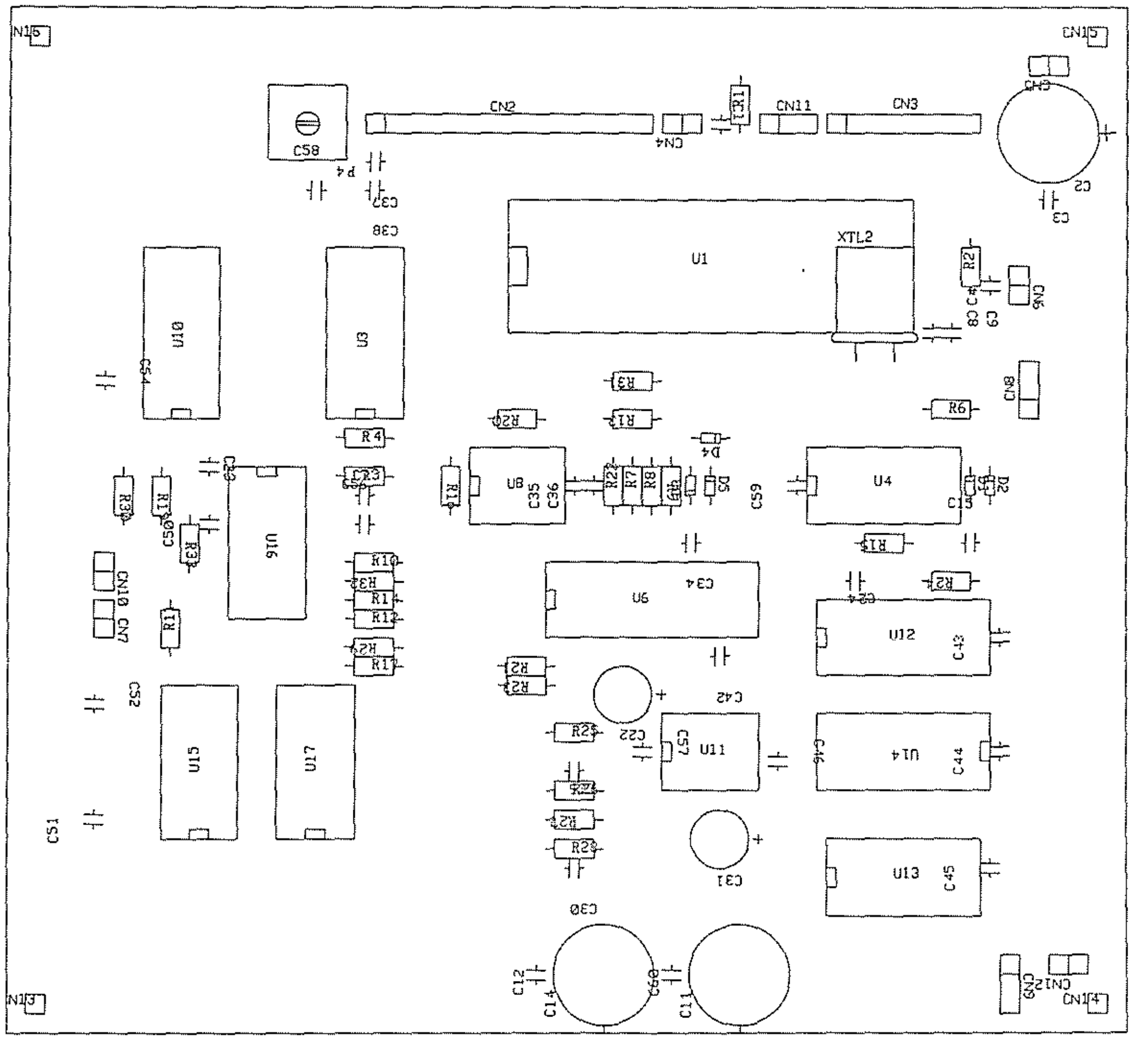
D.2 Component side layout of PCB1



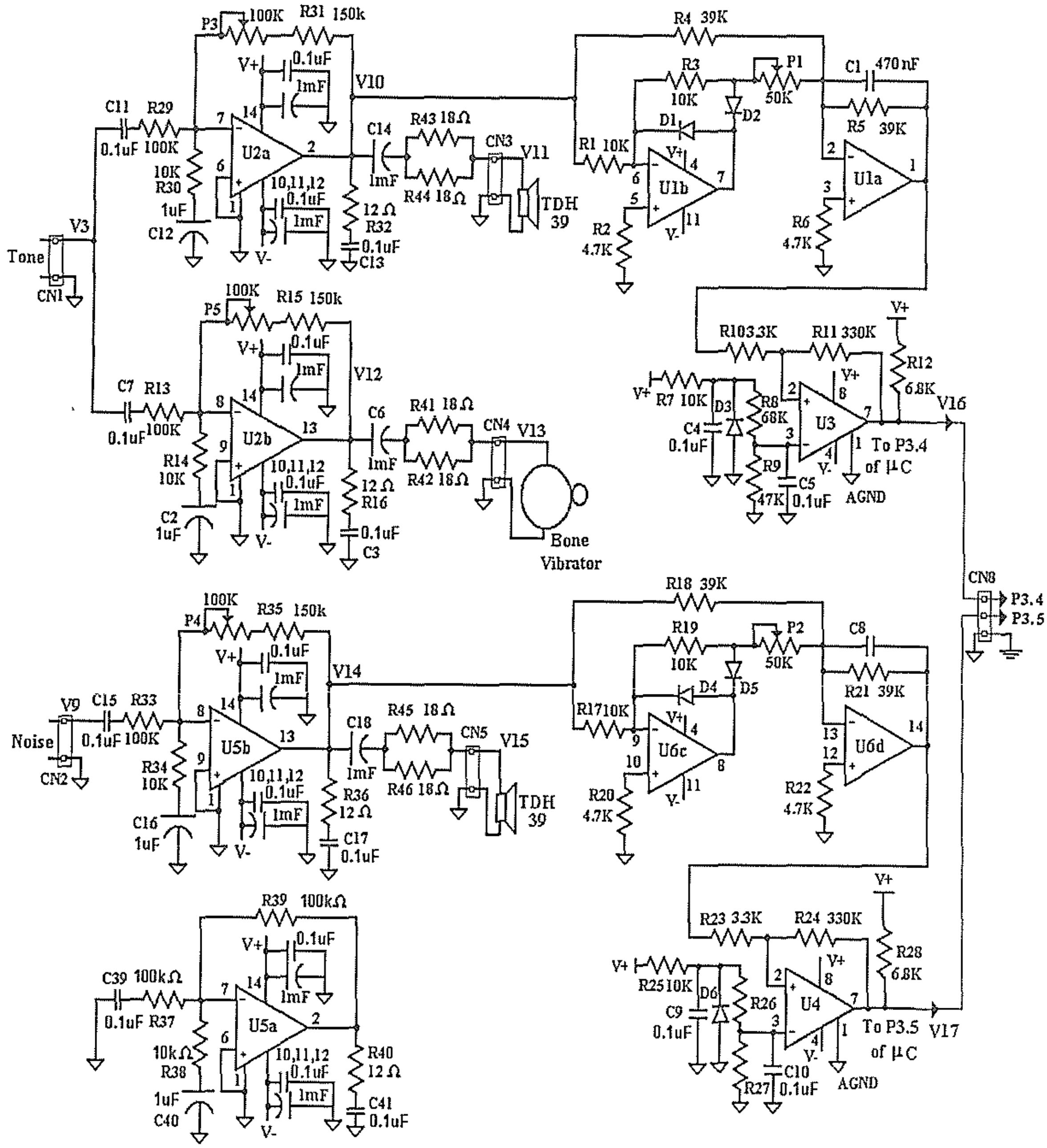
D.3 Solder side layout of PCB1



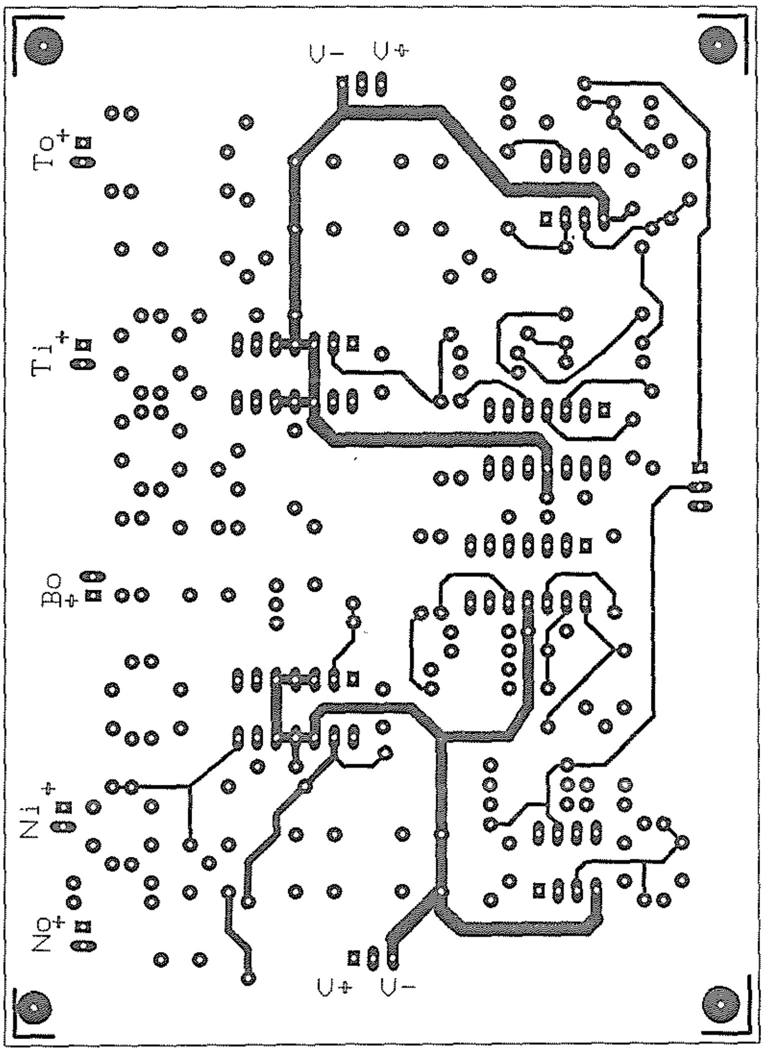
D.4 Component placement of PCB1



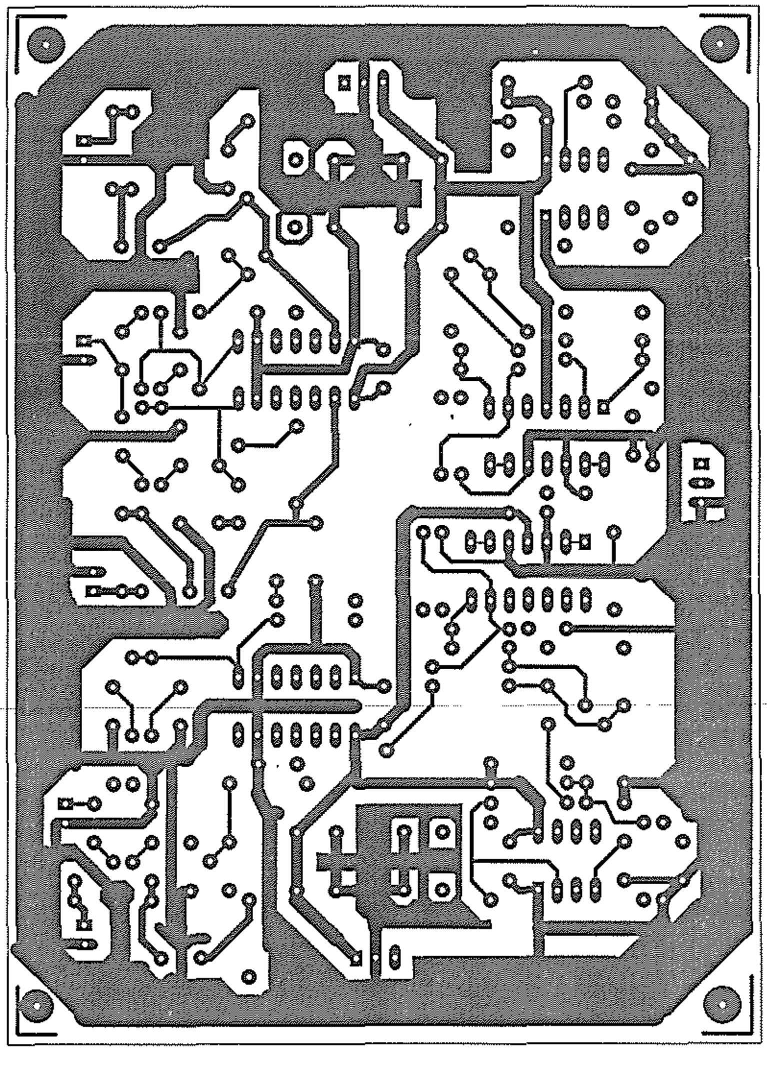
D.5 Circuit diagram for PCB2 (amplifiers and self-test circuits)



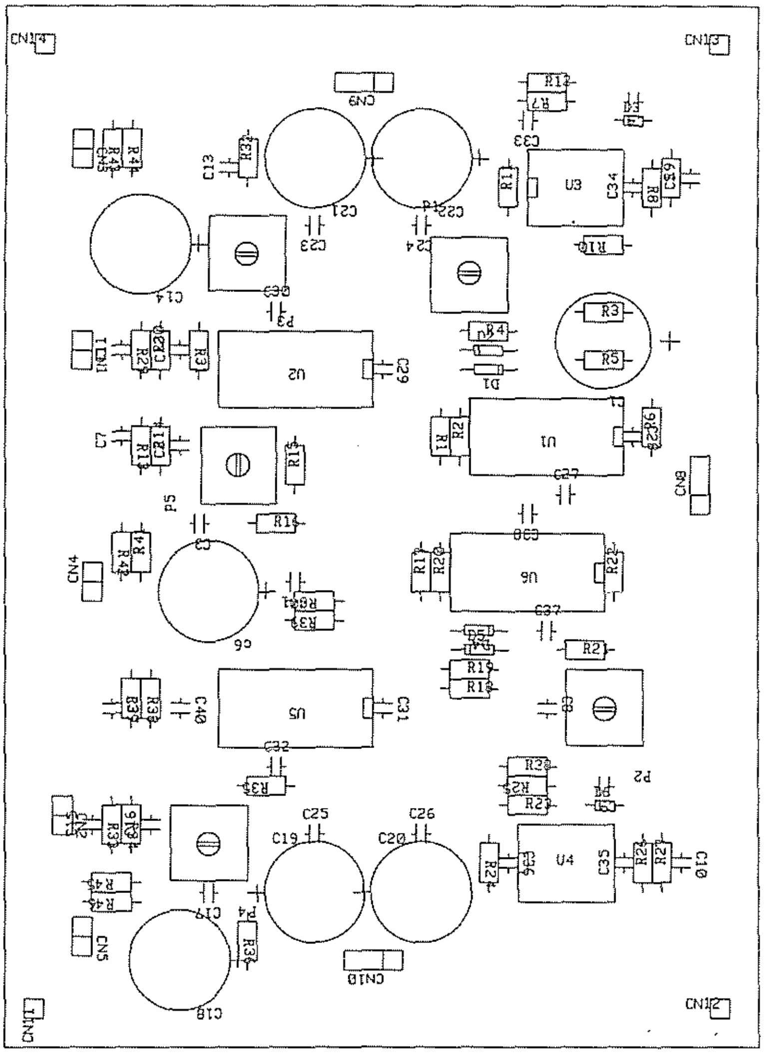
D.6 Component side layout of PCB2



D.7 Solder side layout of PCB2



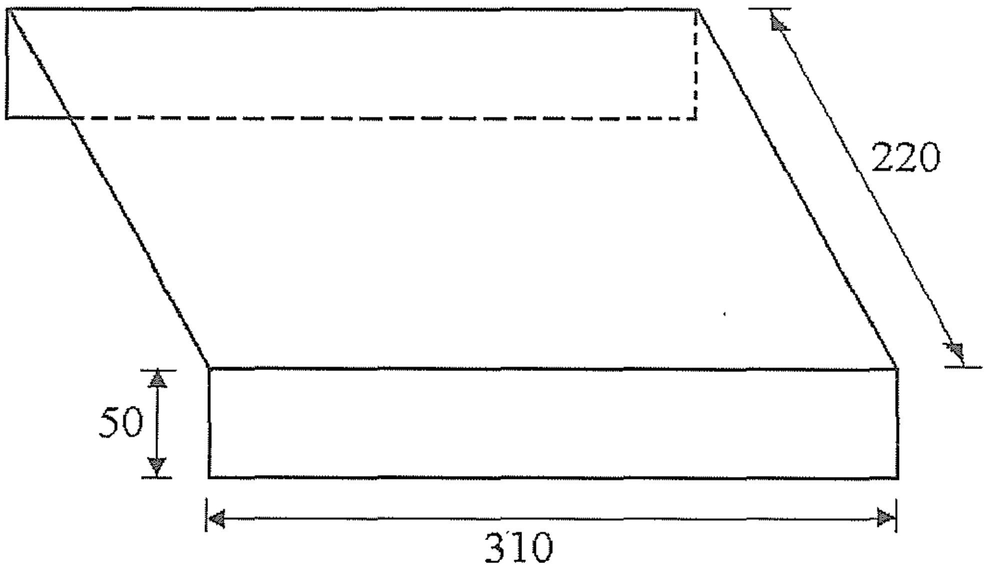
D.8 Component placement of PCB2



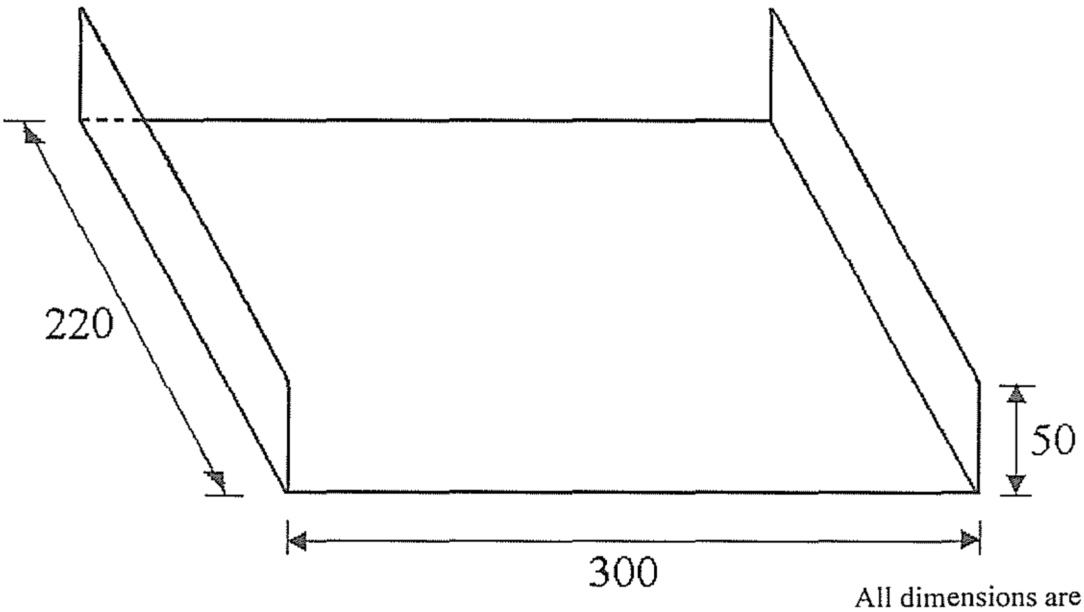
APPENDIX E

Panel design and cabinet dimensions

E.1 Cabinet dimensions



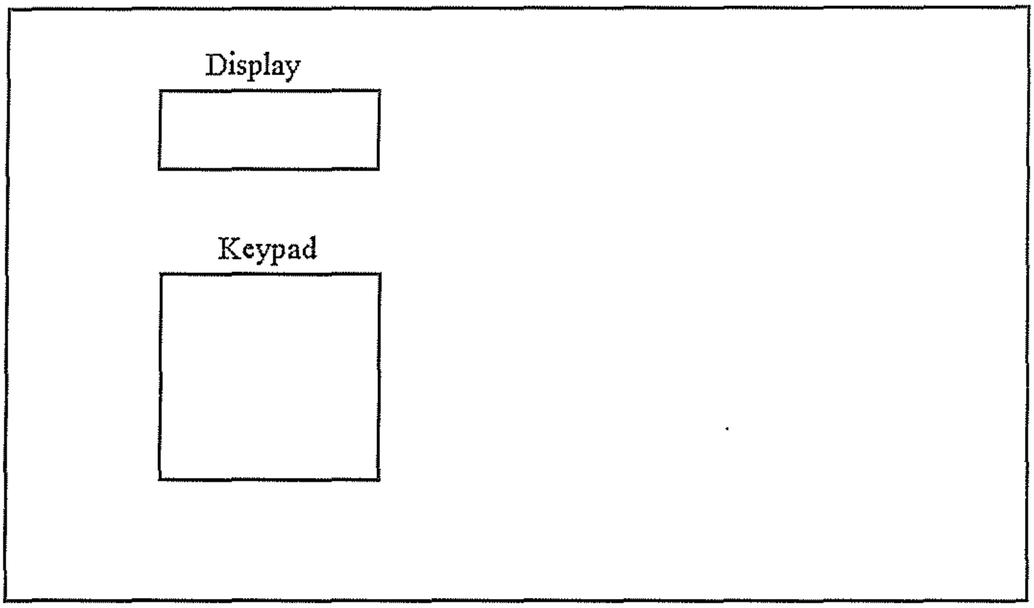
Dimensions of top plate



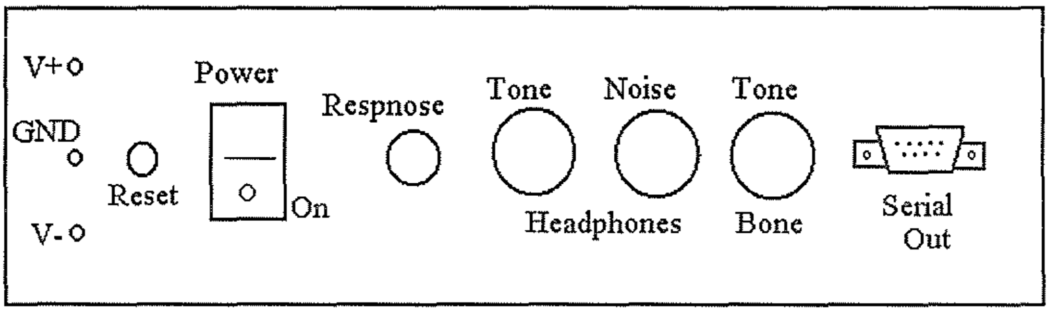
All dimensions are in mm

Dimensions of bottom plate

E.2 Panel design



Top view



rear view

E.3 Keypad mapping

*	1	4	7
0	2	5	8
#	3	6	9
D	C	B	A

Tone Type	Noise Type	Tone Dur.	Mode A/M
Freq. Up	Tone Up	Noise Up	Tone On
Freq. Down	Tone Down	Noise Down	Tone Off
OK	Save	Air/Bone Cancel	Recall/ Print

APPENDIX F

Cost Estimate

F.1 Component List for PCB1

Component designator	Part number/ value	Part description	Approx. price per part (Rs.)
C1	10 μF/50V	Capacitor (Electrolytic)	4.00
C2, C11, C14, C31	220μF/50V	Capacitor (Electrolytic)	4.00
C3, C12, C34, C35, C36, C37, C42, C43, C44, C45, C46, C50, C51, C54, C57, C58, C59, C60	0.1 μF	Capacitor (Ceramic)	1.00
C4	2.2 μF/63V	Capacitor (Electrolytic)	4.00
C7, C23	47 pF	Capacitor (Ceramic)	1.00
C8, C9	33 pF	Capacitor (Ceramic)	1.00
C15	22 nF	Capacitor (Polyester)	4.00
C22	47 μF/63V	Capacitor (Electrolytic)	4.00
C24	1 μF/63V	Capacitor (Electrolytic)	4.00
C26	470 pF	Capacitor (Ceramic)	1.00
C30	2.2 nF	Capacitor (Ceramic)	1.00
CN2		16 pin connector	26.00
CN3		8 pin connector	12.00
CN4,CN5,CN6,CN7,CN10,CN12		2 pin connector	2.50
CN8, CN9, CN11		3 pin connector	3.75
D1, D2, D5	1N4148	Diode	0.50
D3, D4	2.1 V Zener	Zener Diode	1.50
R1	8.2 kΩ, ¼ w	Resistor	0.50
R2	5.6 kΩ, ¼ w	Resistor	0.50
R3, R19	270 Ω, ¼ w	Resistor	0.50
R4, R8, R22, R30	1 kΩ, ¼ w	Resistor (MFR)	1.00
R6	39 kΩ, ¼ w	Resistor	0.50
R7	47 kΩ, ¼ w	Resistor	0.50
R9	150 kΩ, ¼ w	Resistor	0.50

R10,	330 kΩ, ¼ w	Resistor	0.50
R11, R17, R29	10 kΩ, ¼ w	Resistor	0.50
R12, R14	3.3 kΩ, ¼ w	Resistor	0.50
R13	2.2 kΩ, ¼ w	Resistor	0.50
R15	15 kΩ, ¼ w	Resistor	0.50
R16, R20	100 kΩ, ¼ w	Resistor	0.50
R21	3.3 kΩ, ¼ w	Resistor	0.50
R23	1.2 kΩ, ¼ w	Resistor	0.50
R24	270 kΩ, ¼ w	Resistor	0.50
R25, R27, R28	22 kΩ, ¼ w	Resistor	0.50
R26	12 kΩ, ¼ w	Resistor	0.50
R33	5.1 kΩ, ¼ w	Resistor	0.50
P4	50 kΩ	Pot	15.00
U1	AT 89C52	Microcontroller	250.00
U3, U10	AD 7111	Logarithmic DAC	800.00
U4	CD 4011	NAND gate	15.00
U6	LMF 100	SC Filter	355.00
U8, U11	LF 351	Op-amp	15.00
U12, U14	CD 4015	Shift register	15.00
U13	CD 4030	XOR gate	15.00
U15, U17	CD 4066	Analog switch	15.00
U16	LF 347 (or TL 084)	Quad Op-amp	18.00
XTL2	12 MHz	Crystal	12.00
	ODM 16216	LCD 16 char. × 2 Lines	800.00
		4 × 4 Keypad	300.00
		PCB- double sided PTH	716.00

F.2 Component List for PCB2

Component designator	Part number/Value	Component description	Approx. price per part (Rs.)
C1	470nF	Capacitor (Polyester)	4.00
C2, C12, C16, C40	1 μF/63V	Capacitor (Electrolytic)	4.00
C3, C4, C5, C7, C9, C10 C11, C13, C15, C17, C23, C24, C25, C26, C27, C28, C29, C30, C31, C32, C33, C34, C35, C36, C37, C38, C39, C41	0.1 μF	Capacitor (Ceramic)	1.00
C6, C14, C18, C19, C20, C21, C22	1000 μF/25V	Capacitor (Electrolytic)	4.00
C8	10 μF/63V	Capacitor (Electrolytic)	4.00
R1, R3 ,R7, R14, R17,R19, R25, R30, R34, R38	10 kΩ, ¼ w	Resistor (MFR)	1.00
R2, R6, R20, R22	4.7 kΩ, ¼ w	Resistor	0.50
R4, R5, R18, R21	39 kΩ, ¼ w	Resistor	0.50
R8	68v kΩ, ¼ w	Resistor	0.50
R9	47 kΩ, ¼ w	Resistor	0.50
R10, R23	3.3 kΩ, ¼ w	Resistor	0.50
R11, R24	330 kΩ, ¼ w	Resistor	0.50
R12, R28	6.8 kΩ, ¼ w	Resistor	0.50
R13, R29, R33, R37	100 kΩ, ¼ w	Resistor (MFR)	1.00
R15, R31, R35, R39	150 kΩ, ¼ w	Resistor	0.50
R16, R32, R36, R40	12 Ω, ¼ w	Resistor	0.50
R26		Resistor	0.50
R27		Resistor	0.50
R41, R42, R43, R44, R45, R46	18Ω, ¼ w	Resistor	0.50
P1, P2	50 kΩ	Pot.	15.00
P3, P4, P5	100 kΩ	Pot.	15.00

CN1, CN2, CN3, CN4, CN5,		2 pin connector	2.50
CN8, CN9, CN10		3 pin connector	3.75
D1, D2, D4, D5	1N 4148	Diode	0.50
D3, D6		Ref. Diode	
U1, U6	TL 084	Quad Op-amp	18.00
U2, U5	LM 1877	Power Amplifier	175.00
U3, U4	LM 311	Comparator	15.00
		PCB- double sided PTH	390.00
		Mono jack-socket	12.00
		Enclosure	225.00

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