A MICROCONTROLLER BASED AUDIOMETER

A dissertation
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the degree of
Master of Technology

by

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Guide

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M. Tech. Dissertation Approval

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ABSTRACT

An audiometer is an electroacoustic instrument for quantifying hearing impairment and for diagnosing its causes. Using it, the test tones of different frequency and levels are presented to the subject and hearing thresholds are determined after judging the responses of the subject. In conventional audiometers, the frequency and level of the tones are changed by manually switching the circuit components in the osciilator and attenuator. In more recent designs, digitally controlled oscillators and attenuators are used. In this project, a microcontroller based simple diagnostic audiometer is developed which does not have any manually operated switches and moving parts, is compact in size, and will give maintenance free operation. It has two channels, one for pure tone, and other for masking noise. Facility for both types of noise, wide band and narrow band noise, is provided.

Here, a programmable oscillator is designed using a switched capacitor filter, and frequency of sinusoidal oscillation is proportional to that of the digital clock input, thus needing only one control line. For programmable attenuators, monolithic logarithmic D/A converters are used. 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 switched capacitor band pass filter. The instrument also has a serial port interface for transferring the test result to a printer or computer. The instrument can be used either as conventional operator controlled audiometer or as an automated audiometer. In operator controlled audiometer, the operator has full control over stimulus parameter selection and threshold determination. In automated mode, the threshold parameters are determined by monitoring the subject response to the stimulus in accordance with audiometric procedure.

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Chapter 1

INTRODUCTION

1.1 Overview

Hearing impairment is characterized by either loss in perceived loudness (loss in sensitivity) or loss in the ability to discriminate different speech sounds or both. Many persons who suffer from significant hearing loss can be helped with medical and surgical procedures and use of hearing aids. Before medical treatment can proceed or appropriate type of hearing aid is prescribed, the nature and degree of hearing impairment should be assessed in order to quantify the hearing impairment and diagnose its causes so that suitable treatment can be prescribed wherever necessary and possible. Some of the earliest hearing tests include observing an individual's response to vocal sounds or sounds produced by clapping the hands. But these tests were limited in scope [2]. Later the tuning forks were used to test hearing. A tuning fork oscillates at its natural frequency and radiates a pure tone. But, it radiates uncontrolled intensity of sound of fixed frequency and intensity decreases exponentially with time [3].

Audiometry is a technique for identification and quantitative determination of hearing impairment. It involves presentation of systematically varying acoustic stimuli to the subject and recording the responses. 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. In pure tone audiometry, the subject's threshold for hearing is measured. In speech audiometry, the subject's threshold for the reception of speech is recorded. In addition to the hearing threshold level, the most comfortable listening level and the uncomfortable loudness level may be estimated. An audiometer is an instrument for carrying out audiometric tests.

1.2 Project Objective

The objective of this project is to develop a microcontroller based automated portable pure tone audiometer that will permit nearly maintenance free simple operation. The instrument will have manual as well as automated operation, with a serial port interface for transferring test results to a printer or computer.

1.3 Report Outline

In Chapter 2, various audiometric techniques, details of pure tone audiometer and need for an automated portable pure tone audiometer have been discussed. In third chapter, proposed instrument and its implementation has been presented. Chapter 4 gives details of oscillator, attenuator, and noise generator circuits. In Chapter 5, the system hardware and interfacing of various blocks of the instrument has been discussed. The last chapter summarizes the work done and provides some suggestions for further development.

Chapter 2

PURE TONE AUDIOMETER

2.1 Audiometric Tests and Procedures

Before medical treatment and aural rehabilitation for any hearing impairment can proceed, the nature, degree and probable causes of the hearing impairment should be assessed. The various measurement techniques can be grouped in two categories, "subjective or objective". Subjective tests rely on the patient performing a task according to the instructions, viz. pressing a button after hearing a particular sound. The objective methods only require cooperation from the patient in the attachment of measuring electrodes or probes [4].

The majority of tests of auditory function are of the subjective type. Routine clinical assessment usually includes measurement of hearing sensitivity as a function of frequency, measurement of dynamic range for intensity, measurement of adaptation to low intensity continuous tones and assessment of speech identification performance as a function of the speech intensity.

2.1.1 Pure tone audiometry

In pure tone audiometry, the subject's response for acoustic stimuli of different frequencies are measured. The subject responds after hearing the tone. The initial level of the stimuli is selected by the audiologist. The stimulus intensity is decreased in 10 dB steps until the subject no longer responds. After this point the level is increased and decreased in 5 dB steps until 50% response criterion has been met, i.e. subject responds to half of the tones presented to him/her. The results of the audiometry are reported as an audiogram which is a plot of threshold intensity versus frequency. The frequency of the test tone is plotted in the range of 125 Hz to 8 kHz and intensity is shown in decibel hearing level (dB HL). Pure tone audiometry tests are carried out primarily to obtain air conduction and bone conduction thresholds of hearing. Pure-tone screening tests are employed extensively in military, industrial, and school hearing conservation programmes.

In **pure tone air conduction test**, a series of 0.5 sec. bursts of single frequency stimuli are presented to the subject through calibrated headphones placed on the head. Subject responds after hearing the sound. In **pure tone bone conduction test**, a special vibrating transducer is placed on the mastoid process or on the forehead and a threshold is determined. In this test, the sound gets applied directly to inner ear, bypassing the outer and middle ears. Therefore, bone conduction threshold is a function of the inner ear pathology [5].

Bekesy audiometry is different from conventional pure tone audiometry wherein the stimulus intensity and subject's response are interdependent. It is an audiometer with a motor driven attenuator. Initially the direction of rotation of motor is such as to increase the level. Whenever subject presses the key, the direction of rotation reverses so as to decrease the level until the subject releases the key because he no longer hears the tone. A pen connected to the attenuator traces a continuous record of patient's intensity adjustments on an audiogram chart. The threshold is determined from the audiogram chart [1]. Now a days, by using programmable oscillator and attenuator, audiometers can be developed with options for conventional audiometry, or Bekesy audiometry, or automated audiometry.

For various clinical considerations, sometimes it is advisable to use a frequency modulated tone (warble tone) as stimulus and measure the impairment. Warble tones cause less mental fatigue and also help in avoiding standing waves in the sound field during free fielding testing [5].

2.1.2 Speech audiometry

In speech audiometry, a person's ability to hear and understand speech, and thereby the integrity of auditory system, is assessed. Stimulus is presented by microphone, tape recorder, or a phonograph. Speech audiometry is normally carried out to determine speech reception thresholds for diagnostic purposes and to assess and evaluate the performance of hearing aids [5,8].

In **speech discrimination test**, lists 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 result of this test is a score from 0 to 100%. Generally, a high score is associated with normal hearing or conductive hearing loss and a low score is associated with sensorineural loss. **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 [5].

2.1.3 Evoked response audiometry

It is a technique to evaluate the integrity of the central auditory system by the measurement of electrophysiological activity evoked by a transient acoustic stimulus. It is objective in the sense that no conscious response from the patient is required. Information about the different levels of the system can be obtained from sub techniques such as electrocochleography, brainstem response audiometry and electroencephalic audiometry [8].

(1) Electrocochleography measures the electrical activity generated in the cochlea in response to a click stimulus. The response, auditory nerve action potential, is taken by an electrode placed at promontory.

(2) Brain stem electric response audiometry is used to detect the skin surface potentials caused by electric impulses which occur in the area of the brain stem in response to the auditory stimulus.

(3) Electroencephalic audiometry is primarily concerned with the change of electrical activity at the cortex, evoked by an audiotory stimulus.

2.2 Masking in Audiometry

In air and bone conduction audiometry where sound is applied to one ear, the contralateral cochlea is also stimulated by transmission through the bone of the skull. This is called cross hearing. In the simplest case where there is a complete unilateral hearing loss and other ear is normal, a shadow audiogram will be obtained for the poorer ear which shows threshold levels lower than the true value. The cross hearing can be reduced by using miniature aid earphone, but the sensitivity changes with placement. Hence, to avoid cross hearing while testing the poorer ear, a masking noise is applied on the better ear. The threshold level of the poor ear when the better ear is unmasked, is always lower than the one when the better ear is fully masked. During testing, masking level of the better ear is increased till the threshold level of the poor ear becomes constant. In speech audiometry, some noise is deliberately added to the signal as we seldom hear speech sound in the quiet [3].

In pure tone audiometry, the masking noises are either (1) broad band noise covering the full range of the headphones and having a flat frequency response, or (2) narrow band noise. The narrow band noise may be of constant bandwidth, approximate critical bands, or a constant percentage bandwidth. The center frequency of the narrow band should be same as the test tone. Narrow band noise has the advantage of producing the same masking effect as a wide band noise but with a lower sound pressure level of the noise [6].

In speech audiometry, broad band white noise or pink noise is necessary as a masking signal. The use of competing speech signals as masking for speech has been shown to be more effective than the white or pink noise or white noise shaped to a speech spectrum [6]. A wide band noise filtered by a first order low pass filter with a corner frequency of the 500 Hz produces noise which spectrally approximates the noise generated by many people talking at the same time [5].

2.3 Pure Tone Audiometer

A pure tone audiometer mainly consists of an oscillator and attenuator and output of the audiometer is calibrated in terms of frequency and acoustic level. The frequency is selectable over the range of 125 Hz to 8 kHz. The instrument is also 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 [6].

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

(2) Simple diagnostic audiometer: Full frequency range but limited acoustic output (up to 100dB), air and bone conduction with simple masking facility.

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

(4) Advanced diagnostic audiometer: facilities in the diagnostic audiometer but amplitude and frequency modulation of the test tones, intermediate frequency tones can also be generated .

The block diagram of a simple pure tone audiometer is given in Fig. 2.1. It consists of a pure tone generator, noise generator, interrupter switch, hearing level control (attenuator), power amplifier, switching system, and power supply [6]. These blocks will be discussed separately.

2.3.1 Pure tone generator

The pure tones are generated by an oscillator with an accurate, stable frequency output with low harmonic distortion. A further requirement is that oscillator output should not have switching transients when it is switched on and off by the interrupter switch. Tones of at least 250 Hz, 500 Hz, 1 kHz, 1.5 kHz, 2 kHz, 3 kHz, 4 kHz, 6 kHz, and 8 kHz have to be produced. Each of the above frequency should be within 3% of indicated frequency. Also, the output level of any harmonic should be at least 30 dB below the fundamental level [13].

2.3.2 Noise generator

The noise generator should produce wide band noise which have a flat frequency response up to 10 kHz and amplitude should be within ±5 dB of the indicated value. The center frequency of the narrow band noise should be equal to the frequency of the test tone. The bandwidth of the narrow band noise must be such that when it is sufficiently intense, it will mask the sensation of the pure tone. This bandwidth is called critical bandwidth. The bandwidth should be between one third octave to one half octave of the center frequency [13].

2.3.3 Interrupter switch

The interrupter switch on the manual audiometer is operated by the audiologist and allows the tone to be presented to the subject. The switch itself must be mechanically silent.

2.3.4 Hearing level control (attenuator)

The hearing level control must be capable of adjusting the output sound pressure level in 5 dB steps. The output is specified in dB HL. The sound pressure level for 0 dB HL is different for different frequencies. Also, the gain of the headphones varies with frequency. Therefore, the attenuator output should be compensated against the change in SPL for zero dB HL for that frequency as well as the change in the gain of the headphones. Hence, there should be some provision of changing the attenuator setting when the frequency is changed.

2.3.5 Power amplifier

The output power available from the power amplifier determines the maximum sound pressure level available from the headphones and the bone vibrator. The amplifier must have low distortion and a good S/N ratio to meet the standard requirements.

2.3.6 Power supply

The power supply for the audiometers may either be mains electricity or batteries. The design requirements for the power supply are that it should be able to supply sufficient current to the audiometer at high outputs without causing any undesired effects and should not induce any hum or other spurious signals at any setting.

2.3.7 Switching system

All double channel audiometers must have a means of switching the signal from one headphone to other headphone and from headphone to bone vibrator.

2.4 Microprocessor Based Audiometer

In pure tone audiometry, as described earlier in this chapter, the subject responds to varying intensities of the acoustic stimuli of a discrete frequency over the range of the 125 Hz to 8 kHz. The variation of the level of the stimulus is done manually by the audiologist after carefully observing the responses of the subject. The audiologist then decides the threshold for that particular frequency.

Application of microprocessor / PC in audiology offers many advantages in terms of flexibility and simplicity of use, over their conventional counterparts. Some PC based and portable audiometers are available in the market. In these audiometers, for each frequency, one RC oscillator is used. In other types, the test tones are generated by software and output by a D/A converter. Output of these oscillators are calibrated for 0 dB HL and attenuator is passive attenuator. For switching in each of the oscillators and to give desired attenuation, analog multiplexers and electronic switches are used and for switching on and off the tones, a mechanical relay is used which connects and disconnects the signal to the headphones. For noise generation, the noise of diode and transistor are amplified and given to attenuator. Large number of R's and C's increases the size and while switching in the signal, noise is produced. Earlier, Malini [8] has developed a PC based audiometer. The stimuli are 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 active R network and analog switches. A portable system can be made using microproccessors or microcontrollers, programmable oscillator, and programmable attenuator.

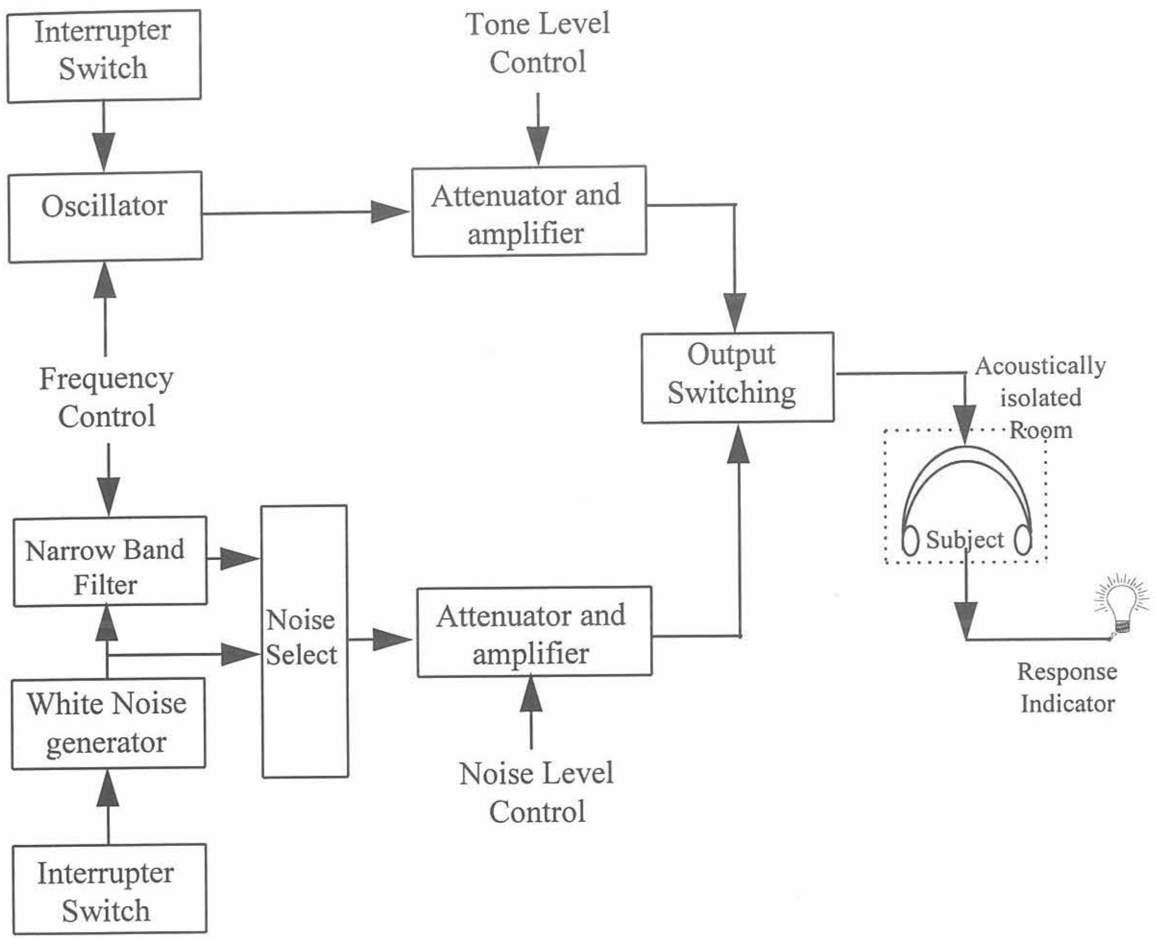


Fig.2.1: Block diagram of pure tone audiometer [2].

Chapter 3

SYSTEM IMPLEMENTATION

3.1 Introduction

A portable microprocessor based audiometer can be realized by using a microprocessor, programmable oscillator, noise generator, programmable attenuator, a keypad, display, and a computer interface as shown in the Fig. 3.1. The stimulus generation is carried out by the oscillator, and attenuator and amplifier unit. The stimulus presentation, response monitoring and display of stimulus parameters (frequency and levels) are carried out by the microcontroller. The change in stimulus parameters viz. frequency and level, storage of stimulus parameters, and transfer of the result through serial port can be made through key board by pressing the respective keys. The main functions of the instrument are:

(i) Stimulus generation and presentation,

(ii) Subject response monitoring,

(iii) Display of stimulus parameters,

(iii) Storage of subject's threshold values.

In automated mode, in addition to the above functions, the instrument also decides the threshold levels.

3.2 System Specification

Audiometer type:	Single channel microcontroller based audiometer, with pure / warble tone stimulus and wideband / narrowband masking noise.
Circuit size:	Suitable for a compact instrument.
Controls:	4 X 4 keypad for controlling all operations of the instrument.
Display:	16 character X 2 line LCD display, for displaying the stimulus and masking noise level, subject response, and menu options.

Power supply: $\pm$ 5 volt (with provision for battery based operation).

Stimulus: Crystal controlled test tone frequencies, with intensity level from 0 dB HL to a maximum value given below for each frequency (as per IS 10565: 1983).

Frequency (Hz)	125	250	500	1000	1500	2000	3000	4000	6000	8000
$(HL)_{max}$ (dB)	70	90	100	100	100	100	100	100	90	80

Warble tones with a modulation rate of 8 per second and frequency deviation of $\pm$ 10% (as per IS 10565: 1983).

Intensity level variation in steps of 5 dB, and intensity level calibration in steps of 0.5 dB.

Masking noise: Broadband / narrowband noise, with level sufficient to mask tones at 60 dB HL at 250 Hz, 75 dB HL at 500 Hz, and 80 dB HL from 1kHz to 4 kHz (as per IS 10565: 1983).

Intensity level variation in steps of 5 dB.

Narrow-band noise :

Center frequency = Test tone frequency.

Band width = One-third to one-half octave about center frequency.

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

Operation: Software controlled menu driven manual / automated mode.

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

Interfacing: Serial port (TxD, RxD, GND, TTL level) at a baud rate of 1200 for down loading the test results.

Headphone: Type TDH-39

3.3 System Implementation

The system implementation of the Fig. 3.1 is given in Fig. 3.2. The instrument will be operated through a menu driven software, keypad and display. The serial interface is used to transfer the test result to the printer or a computer.

A microprocessor based system typically requires program ROM, data RAM, timers, and peripheral interface chips. Various microprocessors and microcontrollers were considered with the objective of keeping the chip count low. We have decided to use Atmel's 89C55-24PC microcontroller which is pin to pin compatible to Intel's 8751 microcontroller. It has on-chip 20K bytes of flash programmable and erasable ROM and 256 bytes of RAM, [17]. Also, it has one more timer than 8751which can be operated to generate a clock with 50% duty cycle.

In microcontroller / PC based audiometers, the oscillator and attenuator are programmable. The oscillator can be of RC type and for each frequency, a set of R and C are selected using multiplexers or by employing programmable R's and C's. In both cases number of control lines required increases with the number of frequency steps. The sine wave can also be generated by a D/A converter but to reduce the harmonic level, D/A converter with large number of bits is required which in turn will require more number of microcontroller pins. In this project, switched capacitor filter based oscillator is used which requires only one control line, namely a 50% duty cycle clock. Frequency of the output signal is in direct proportion to the clock. Thus, any frequency can be generated by changing the clock frequency.

To make the programmable attenuator, a simple passive attenuator of fixed attenuation for each level can be made and signal can be attenuated by selecting the corresponding attenuator by multiplexer or electronic switches. These type of attenuators suffer from the fact that accurate attenuation can not be obtained. Also, the resistance values change with temperature and time. Multiplying type D/A converter (MDAC) can also be used as attenuator but this attenuator gives attenuation on a linear scale while the requirement is in dB. Thus a large D/A converter will be required e.g. a 19 bit MDAC will give attenuation of 80 dB within 0.5 dB [9]. In this project, an 8-bit logarithmic D/A converter is used as an attenuator which gives attenuation of 88.5 dB in steps of 0.375 dB.

For masking purpose, the white noise is generated by low pass filtering of the 15-bit pseudo random bit sequence (PRBS) generator. To get narrow band noise, a programmable band pass filter will be required whose center frequency should be same as that of the tone. Here, a switched capacitor filter configured as a BP filter whose center frequency is proportional to the frequency of the clock driving it, is used. The clock which is driving the oscillator also drives the filter so the center frequency of the band pass filter and that of tone remains same.

The warble tones can be generated by varying the frequency of the digital clock to the switched capacitor based oscillator.

Details of the various blocks of the instrument are discussed in the following two chapters.

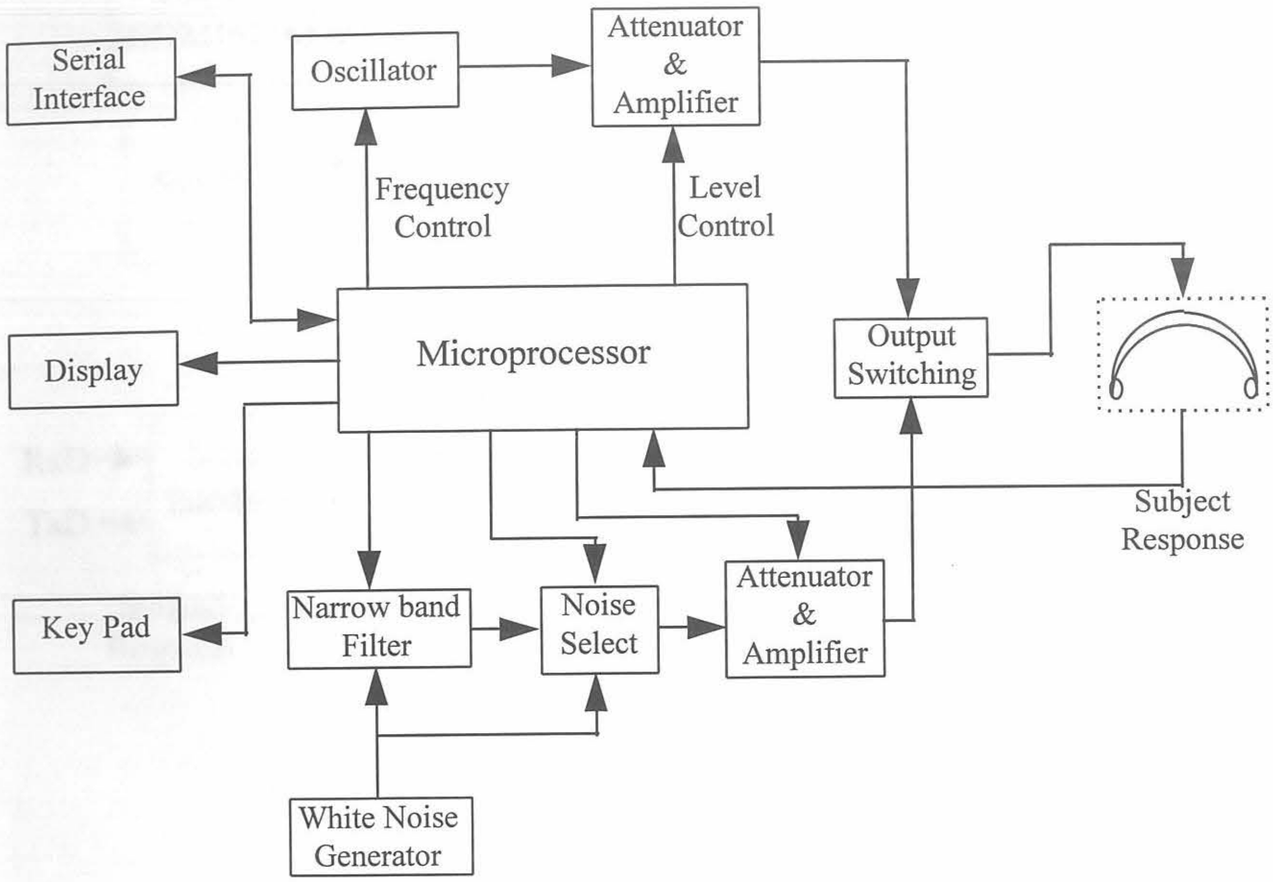


Fig. 3.1: Block diagram of a microprocessor based audiometer.

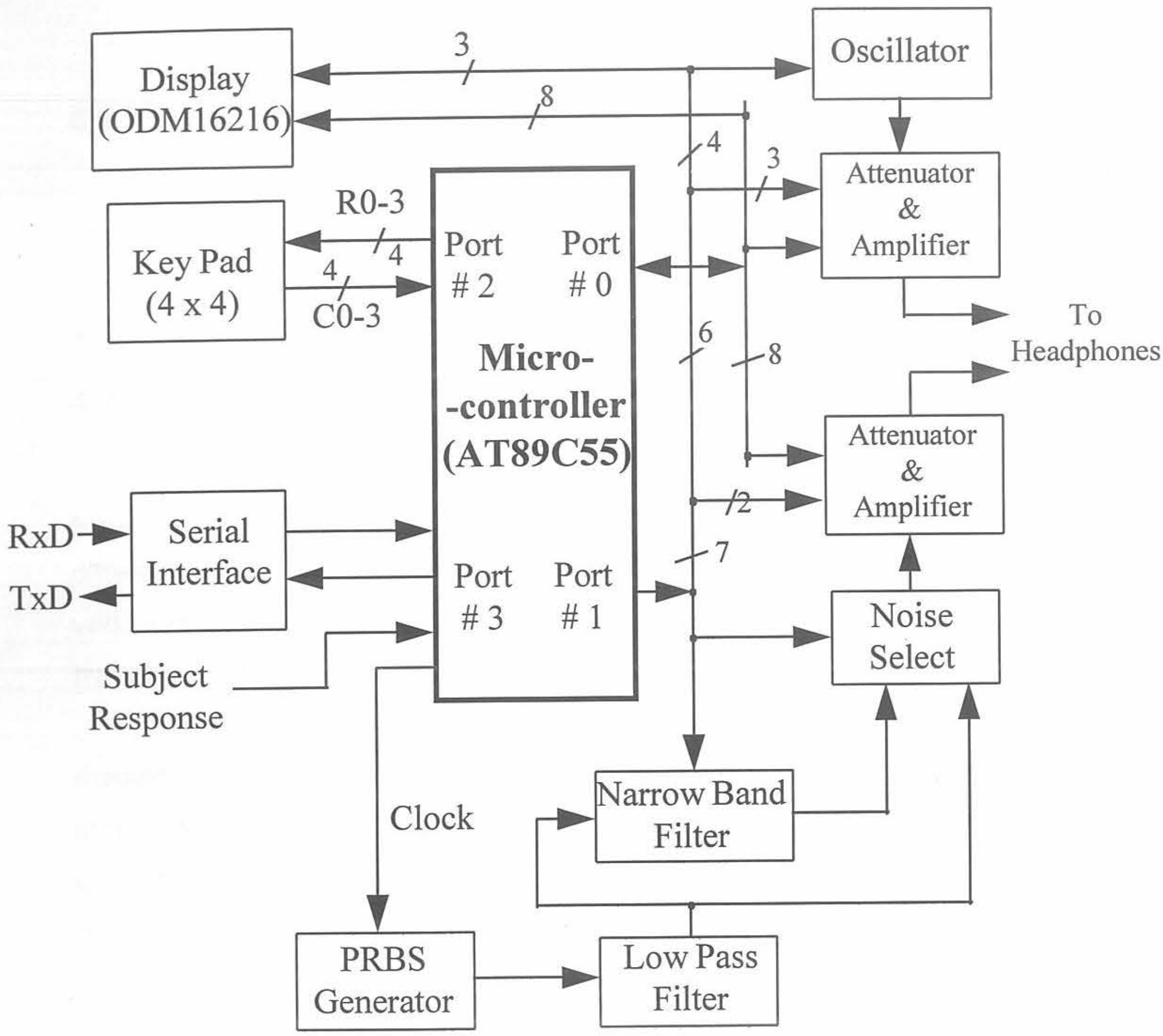


Fig. 3.2 : Block Diagram of the System Hardware

Chapter 4

STIMULUS GENERATOR & ATTENUATOR

4.1 Introduction

In pure tone automated audiometry, the stimulus intensity and frequency are varied between -10 dB to 100 dB and 125 Hz to 8 kHz respectively. Also, the center frequency of the masking noise should be the same as the test tone. This can be achieved by employing programmable attenuator, oscillator, and filter.

4.2 Oscillator

The oscillator can be of RC type. The RC circuits can be tuned to get desired frequency and frequency can be changed by switching in the resisters or capacitors of different values. The number of lines required for switching in the R or C increases with increase in the frequency steps required. These circuits have generally poor frequency stability. Further, these can not be easily used for generating warble tones.

The sine wave can also be generated by software. The advantage of this method is that no additional hardware is required except a D/A converter [8]. But processor will be always busy in generating the stimulus. It will stop the stimulus generation while placing the data to the attenuator and acknowledging the subject response.

It has been decided to use a switched capacitor based quadrature oscillator, which requires only one control input, as programmable oscillator to generate stimulus. The output of this oscillator circuit can be easily stabilized [10].

Quadrature oscillator is a linear oscillator with some form of nonlinearity introduced for amplitude control. Sine waves are generated by the resonance phenomenon in a frequency selective network, employed in positive feedback. The basic building blocks are two integrator loops. The circuit provides two sinusoids with 90 degree phase shift. These can be designed using switched capacitor filters [10].

4.2.1 Switched capacitor filter

A conventional filter circuit uses RC time constant to establish the frequency characteristic. Precise values of R and C are difficult to fabricate using MOS technology. These limitations are overcome by switched capacitor filters (SCF) by exploiting the fact that a resistor can be stimulated with a periodically operated MOSFET switch and MOS capacitor. And the time constant of the circuit involving switched capacitor are not RC product but capacitor ratios. SC filter IC's do not require external R or C [11].

A switched capacitor integrator is shown in Fig. 4.1. The switch, actually a MOS transistor, periodically connects the sampling capacitor to input and to op amp. The capacitor samples the input and applies it to op amp. The two MOS transistor are driven by non overlapping two phase clock to prevent the simultaneous operation of two transistors.

Let the voltage of the source be Vs and during each sample capacitor C_1 charges up to Vi. Since clock frequency is f_{CLK}, thus sampling period is $Tc = 1/f_{CLK}$. The charge transferred during Φ_1 from the source to capacitor C_1 is

$Qi = C_1(Vs - Vi)$

Therefore, equivalent current $Ieq = (Vs - Vi) \times (C_1 / Tc) = C_1 \times (Vs - Vi) \times f_{CLK}$

or $Ieq = (Vs - Vi)/Req$

where $Req = 1/C_1 f_{CLK}$

The output voltage is $V_0 = -(1/C_2) \int Ieq\, dt$

therefore, equivalent time constant of the integrator is

$Req\ C = (C_1/C_2)\ Tc$

Thus, the time constant and hence the transfer function depends upon the capacitor ratio C_1/C_2, not on the absolute value of the capacitors [10,11].

In this project, National Semiconductor's MF 10 has been used as a SCF. It is a universal monolithic SCF. It consists of two dual integrator loop sections that can be independently configured for LP, HP, BP, AP, and BS responses. The resonant frequency, f_0, of the filter is controlled by a separate clock of 50% duty cycle and it can be either $f_0 = f_{CLK}/50$ or $f_{CLK}/100$ [18].

4.2.2 Switched capacitor based oscillator

The MF 10 based quadrature oscillator circuit was earlier developed and tested by Sarvaiya [6]. The schematic diagram of this circuit is given in Fig 4.2. Two integrator loops of MF 10 are intended to have its poles on jw axis. The MF 10 BPF will ring at its resonance frequency in response to a step input change [6]. The step input is provided by the comparator circuit in feedback path. If the output of comparator is stable, the output of BPF will be stable. To make comparator output stable, two zener diodes of 3.9 V each are employed in back-to-back configuration. BPF output is fundamental frequency component of the square wave input. The 90 degree phase shifted output of the BP is obtained at LPF output [18].

From the circuit analysis, the frequency of oscillation

$f_0 = k\, f_{CLK}$.

where k = 1/50 or 1/100.

$Q = R_3 / R_2$

The BP gain $H_{BP} = -(R_3 / R_1)$.

We want to increase Q so as to increase the selectivity of the BPF. But large Q produces clipping at the output as the o/p is amplified by the same amount. By judiciously varying R_1, R_2, and R_3, the maximum gain without clipping was obtained for the following resistance values.

$R_1 = 150\ k\Omega$, $R_2 = 1\ k\Omega$, $R_3 = 47\ k\Omega$.

Since we need oscillator frequency from 125 Hz to 8 kHz, the MF 10 is used in 1:50 mode as maximum clock frequency will be 400 kHz, well below the range of 1 MHz. It was found that oscillator output decreases with increase in frequency if used in 1:100 mode. Also, in 1:50 mode, DC offset voltage will be low.

4.2.3 Test results

The oscillator circuit was tested for its amplitude stability, and for its output distortion and levels of second and third harmonics were noted. The results are given in Tables 4.1 and 4.2. We see that the fundamental level (p-p) variation is about 0.1 V when the supply voltage changes from ± 4.5 V to ± 5.5 V and third harmonic has the highest level and it is around 45 dB below the fundamental level. Second harmonic is around 57 dB below the fundamental. Hence, harmonic distortion is very low.

Table 4.1

Strength of various harmonics in the output for supply voltage Vs = ± 5.0 volts. The fundamental at 125 Hz is taken as the reference.

Frequency (Hz)	Fundamental level (dB)	2^{nd} Harmonic level (dB)	3^{rd} Harmonic level (dB)
125	0.0	– 58.7	– 45.1
250	– 0.02	– 58.6	– 44.1
500	0.00	– 58.1	– 44.1
1000	– 0.02	– 55.5	– 44.6
1500	– 0.07	– 56.4	– 44.5
2000	– 0.02	– 57.2	– 44.7
3000	– 0.05	– 57.8	– 44.8
4000	– 0.13	– 57.5	– 44.7
6000	– 0.13	– 57.8	– 44.7
8000	– 0.15	– 57.8	– 44.6

Table 4.2

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

Frequency (Hz)	V(p-p) at Vs = ±4.5 V	V(p-p) at Vs = ±5.0 V	V(p-p) at Vs = ±5.5 V
500	4.45	4.49	4.56
2000	4.44	4.48	4.53
6000	4.39	4.42	4.46

4.3 Attenuator Circuit

In pure tone audiometry, the intensity of the acoustic stimuli is systematically varied to determine the threshold level of the subject. In microcontroller based audiometer, the attenuator should be programmable. In programmable attenuator, the output is a fraction of input and the attenuator setting can be controlled by a computer / microcontroller. Thus multiplying type D/A converters can be used as a programmable attenuator whose output is proportional to digital input byte [11].

From an 8 bit multiplying type D/A converter the attenuation range is 0 to 48 dB while 16 bit D/A converter can give attenuation up to 96 dB. But 16 bit D/A converter will require two, 8 bit ports of microcontroller. Also, linear multiplying type D/A converters give output values with unequal step size and thus can not be used for logarithmic step size in the attenuator. On the other hand, an 8 bit logarithmic D/A converter will give attenuation up to 88 dB in steps of 0.375 dB. Various designs have been developed for log. D/A converter. One such design was developed by Malini [8] in a PC based audiometry scheme. But, disadvantage of this circuit was that precise attenuation can not be obtained and hardware is complicated. It has been decided to use a monolithic logarithmic D/A converter, `AD 7111' from the Analog Devices. It has a dynamic range of 87.5 dB and resolution of 0.375 dB. Internally `AD 7111' consists of 17-bit R-2R ladder network. An on the chip logic circuit converts the 8 bit data into 17 bit data, which is used to drive the D/A converter. The 8 bit data are latched into the internal latches by CS and WR.

The output

$$Vo = V_i \; 10^{-(0.375\,N/20)}$$

or $$|Vo/Vi|_{dB} = -\,0.375\,N$$

where N is the input code in decimal. The output is 0 for the $240 \leq N \leq 255$ [19].

The range of test tone levels should be from 0 dB HL to 100 dB HL. From the Appendix A, we see that for this range, the maximum SPL requirement is 115 dB. But, 'AD 7111` can attenuate only up to 88.5 dB. Thus, there should be another attenuator (A2) which can give additional attenuation of 30 dB. It may be a simple resistive attenuator which can be controlled by microcontroller. To make this attenuator, an analog switch CD 4066 is used . It will give a fixed attenuation of 40 dB. The attenuator circuit along with the amplifier is shown in Fig. 4.3.

4.4 Amplifier Circuit

A class B push-pull type amplifier has been used for high efficiency. The class B push-pull amplifier has been made part of the negative feedback loop of the opamp in order to keep the distortion low[12].

4.5 Noise Generator

Early audiometers provided masking in the form of low frequency sawtooth waveforms which were effective at low frequencies but much less so at higher frequencies. Another drawback was that harmonics strength was comparable with test tones. After that white noise generators using the discharge tube as the noise source was used which had a better performance than the earlier ones. Now a days, audiometers use diodes and /or transistors as the noise source followed by a high gain amplifier [6].

For this project, noise generators based on diode and transistors as a noise source were tested. It was found that the spectrum of the noise was not flat. Also, due to the high amplification, of the order of 10^3 , the power line interference was high in the low frequency region. Hence it was decided to use digital white noise generators based on the pseudo random binary sequence (PRBS) generator. The simple low pass filtering of the digital output of the PRBS will give a band limited white Gaussian noise. It will have a flat frequency response up to 12% of the clock frequency driving the shift registers. Digital noise generators generate noise of known spectrum and amplitude, with adjustable bandwidth (by clock frequency adjustment). The diode and transistor noise changes with time which in turn change the output noise level. This will not happen to digital noise generators. Also, there will be no pick up problem due to the high amplitude level in the digital noise generators [14]. Block diagram of the digital noise generator is given in the Fig. 4.4.

There are PRBS based noise generator IC's, such as MM 5437 from National Semiconductor, but these could not be locally procured. Here, a 15 bit PRBS generator based on the maximal length feedback shift register is used. It has an XOR feedback with tappings at bit no. 14 and 15, and output of the PRBS is taken from the 15th bit.

The output of the PRBS is not true random but repeats after every 2^{15} clock pulses. If the clock frequency is 200 kHz, then the repeatation rate will be around 0.2 s. At the power on, the state of the registers are undetermined. Hence, a power on reset circuit is used so that registers can be simply initialized at the stage of all zeroes. But, with XOR feedback, the state of all 0's state are excluded, so an XNOR feedback is used to make the state of all 1's as excluded state [15].

Here, two dual 4 bit shift register IC CD4015 and one quad XOR gate IC CD4030 is used to make the PRBS. The XNOR gate is implemented by an XOR gate followed by a NOT gate. Since the CMOS devices consume power only when they are being clocked, so clock is made using NAND gates as inverters. Thus clock can be enabled or disabled depending on the requirement.

To get wide band white noise, a second order low pass filter with a 10 kHz cutoff frequency is used [16]. The D.C. offset of the digital output is removed by RC coupling. The PRBS generator and low pass filter circuit is given in the Fig. 4.5.

The DC gain of the circuit is

$A_{dc} = -(R_2/R_1)$

and the cutoff frequency is $f_c = 1/2\Pi\sqrt{(R_2 R_3 C_1 C_2)}$

By selecting the values of the $R_1 = 24\ k\Omega$, $R_2 = 24\ k\Omega$, $R_3 = 12\ k\Omega$, $C_1 = 1800\ pF$, and $C_2 = 470\ pF$, we get a cutoff frequency of approx. 10 kHz and D.C. gain of unity.

To get the narrow band noise with a center frequency of that of the test tone, the other half of the MF-10 will be used as a band pass filter and the clock to this block will be same as that to the oscillator. The circuit diagram is given in Fig. 4.6.

The band width of the circuit is given by

$BW = f_0 / Q$

and $Q = R_3 / R_2$

Since we want the BW to be between one third octave (26%) to one half octave (41%) of the center frequency. So, the values of $R_3 = 3.3\ k\Omega$ and $R_2 = 1.2\ k\Omega$ are chosen so as to give a BW of 36%.

An electronic switch is used to connect either wide band or narrow band noise to noise attenuator. The attenuator and power amplifier for the noise is same as that for the test signal except that 40 dB fixed attenuator is not present.

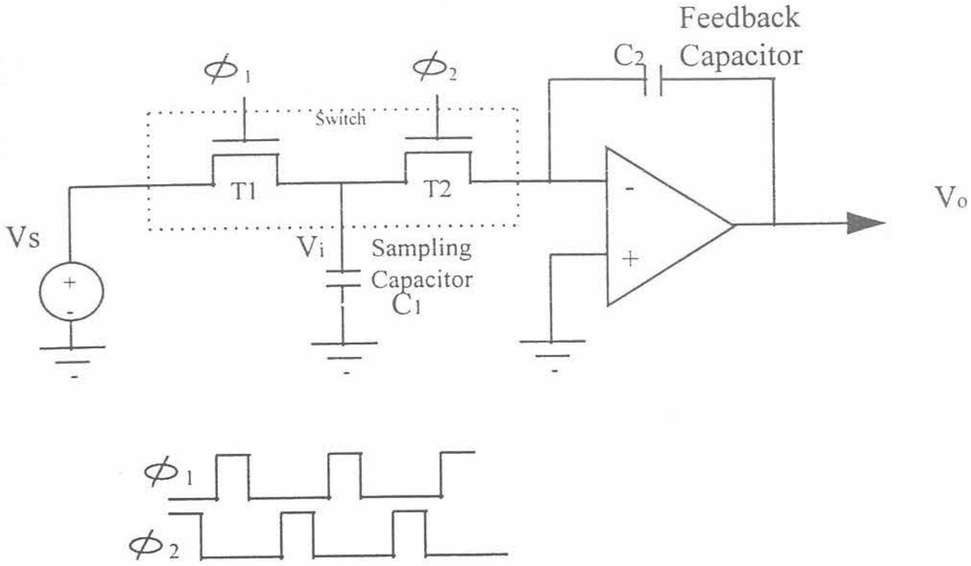


Fig. 4.1 : Switched capacitor inverting integrator with its two phase (non overlapping) clock [11].

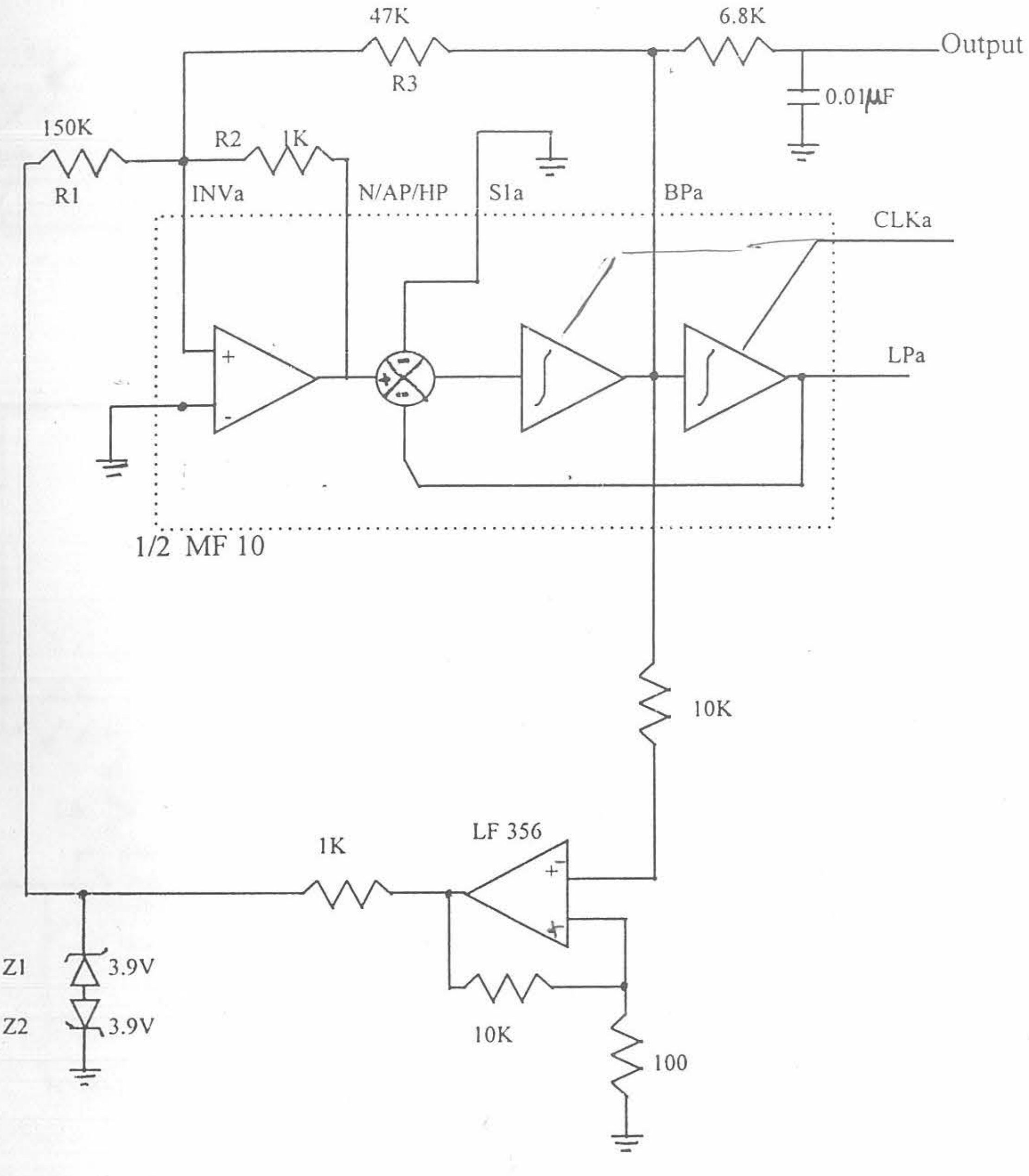


Fig. 4.2: SCF Based Quadrature Oscillator.

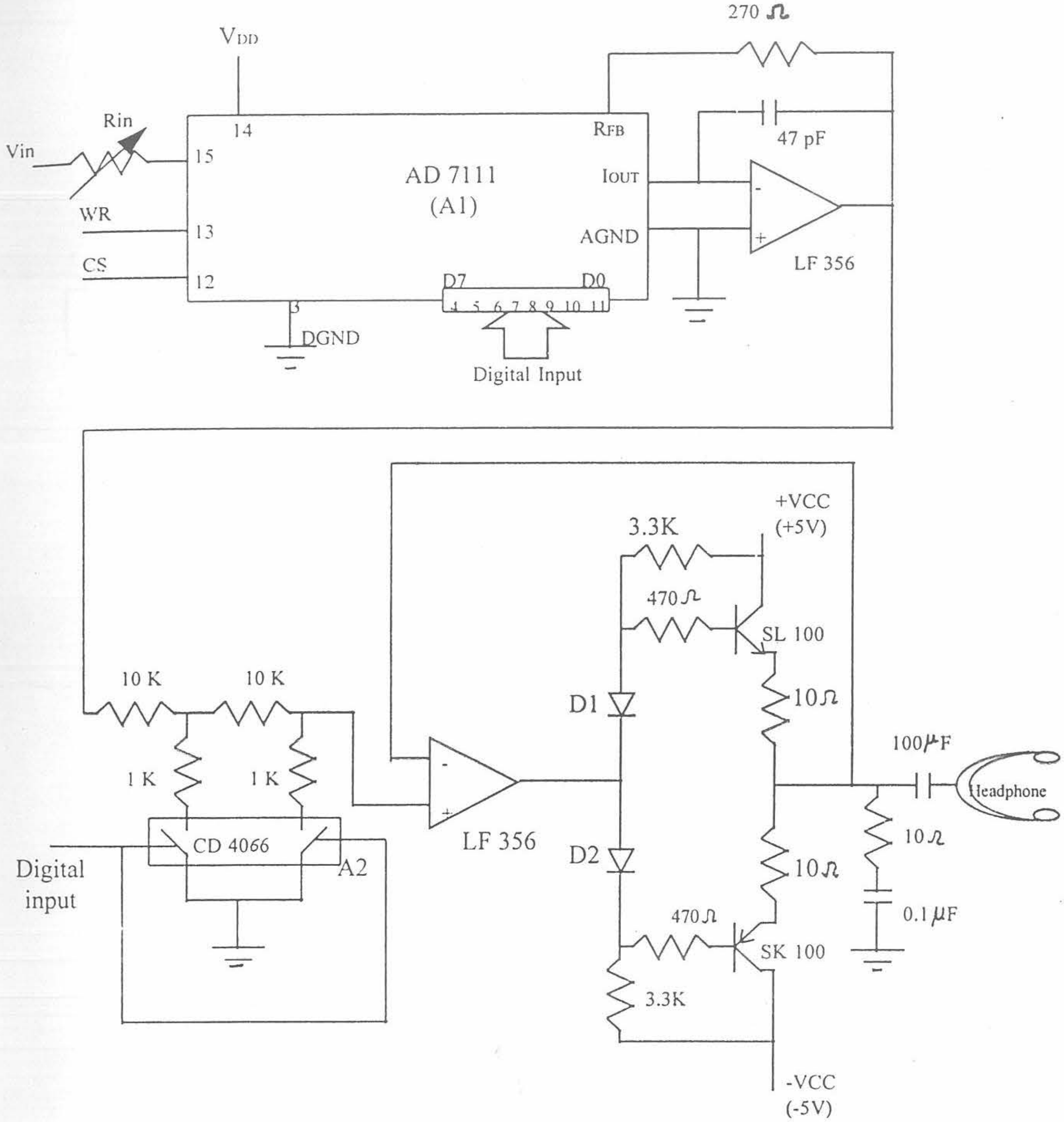


Fig. 4.3 : Attenuator and Amplifier Circuit

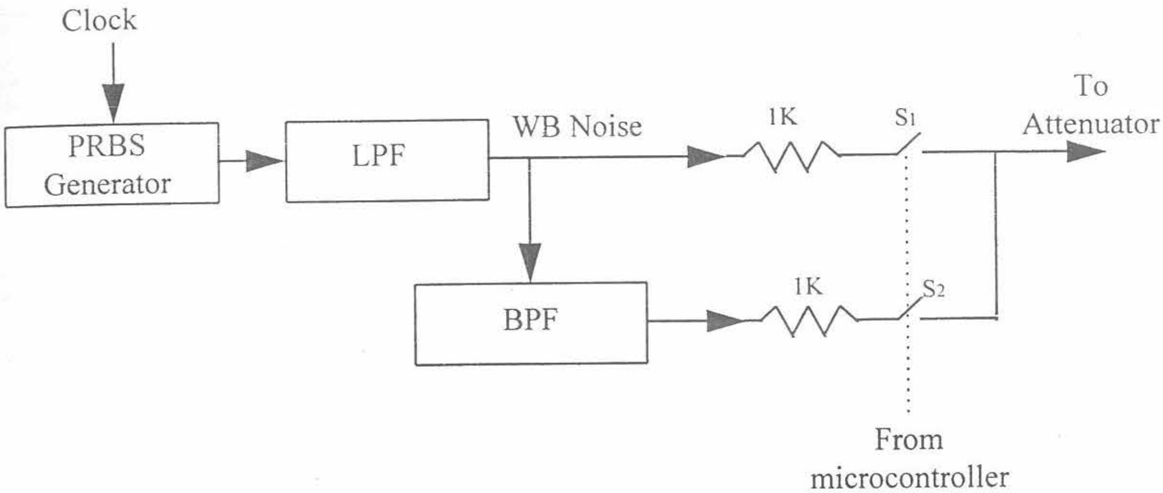


Fig. 4.4 : Block diagramof the noise generator

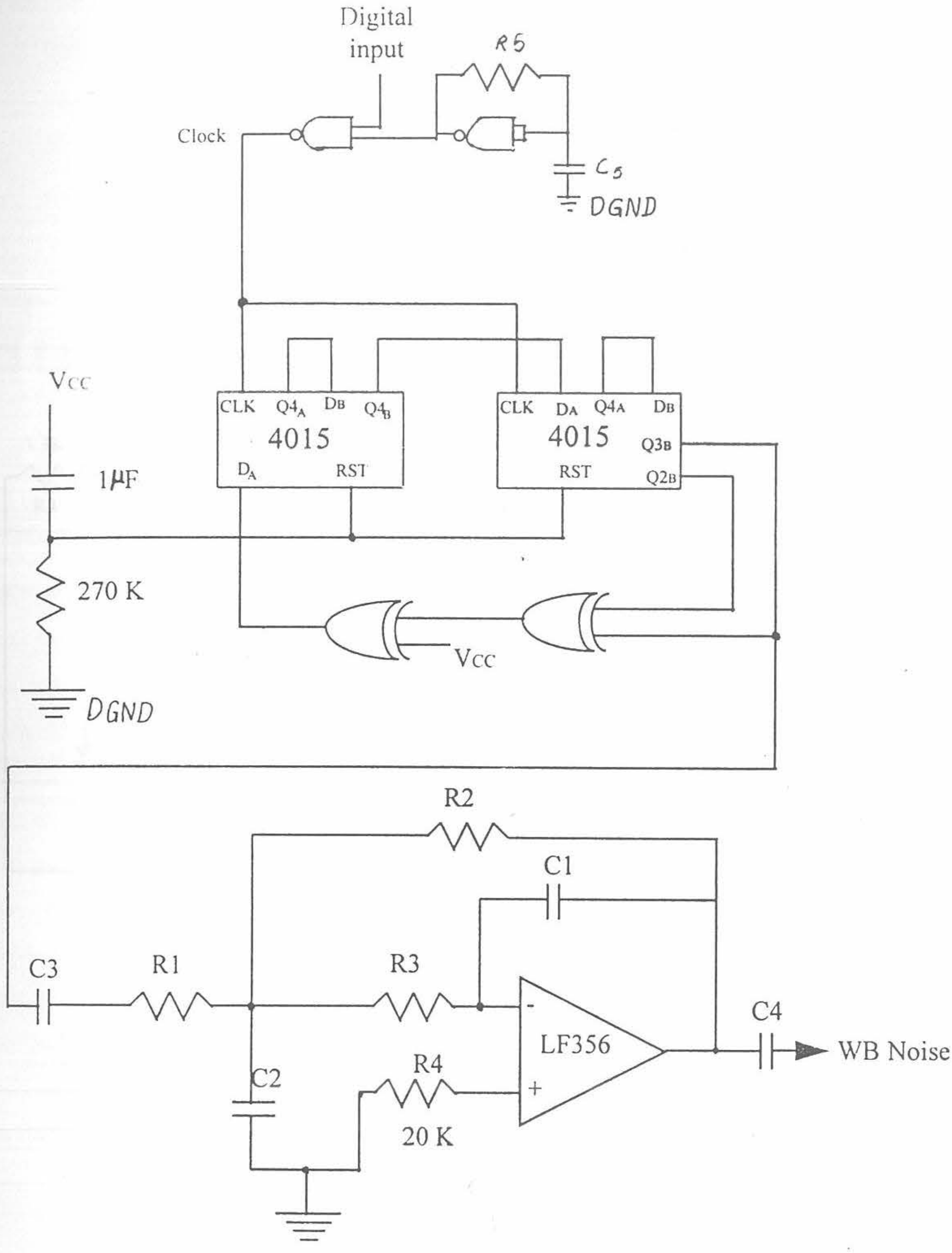


Fig. 4.5 : Wide Band Noise Generator

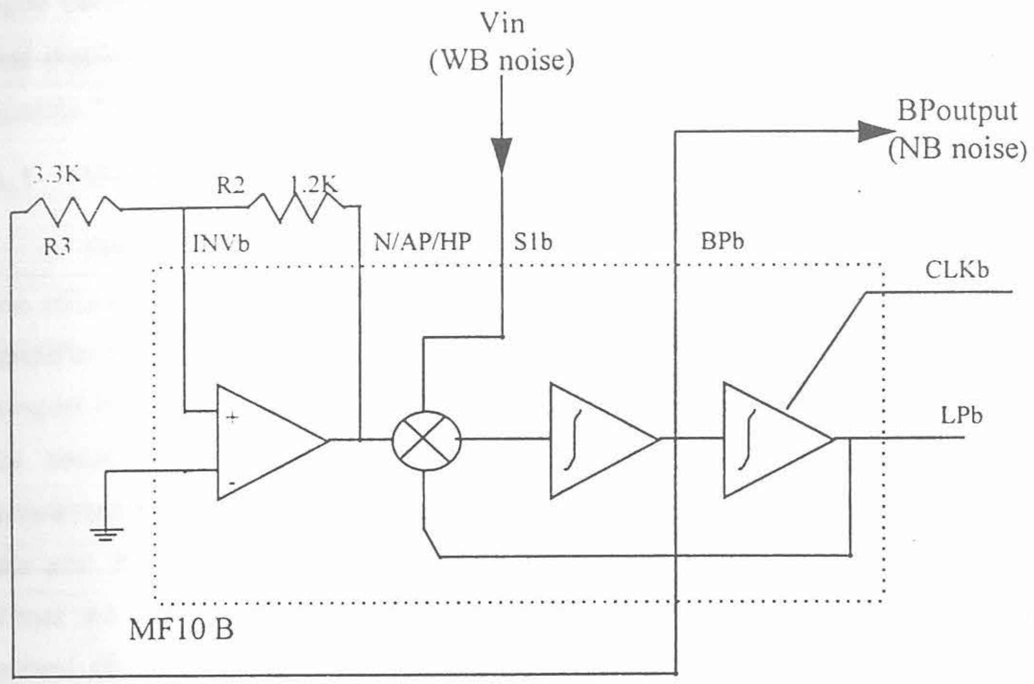


Fig. 4.6: Band pass filter for narrow band noise

Chapter 5

SYSTEM DESCRIPTION

The oscillator, noise generator circuits, and attenuator and amplifier circuits have been described in the previous chapter. Interfacing of these circuits, key pad, and display to the microcontroller is described in this chapter. The entire circuit will operate from a bipolar ± 5 V supply.

5.1 Microcontroller Interfacing

The diagram of system hardware is shown in Fig. 5.1. The pin connections of the microcontroller are given in Table 5.1. The port # 0 drives the attenuator and amplifier units of noise generator and tone generator, and the display. A 4 X 4 matrix keypad is directly connected to port #2. The RxD and TxD lines of port #3 are used for serial communication with a printer or computer. Tone response switch is connected to the external interrupt IE_1 for monitoring the subject response. The rest of the port #1 and port #3 pins are used to generate control signals to various chips. Timer #2, which is programmed in clock generator mode, will generate the clock pulses at pin P1.0 to be used as clock to the oscillator circuit. Since, the CMOS devices consume power only when they are being clocked, so when there is no requirement of masking, the clock to the PRBS is disabled. For this, control line to clock is made low so that the output of NAND gate remains high. To control the two switches selecting the noise type, two control lines are derived from a single line by inversion. To invert the line, it is given to one input of the XOR gate while other input is tied high. Two lines, one direct and other inverted, are given to switch control lines.

Warble tone generation is done by software. For warble tone generation, an interrupt is generated from the output of the op-amp present in the feedback path of the oscillator and has been given to the external interrupt zero pin. When external interrupt zero is enabled, a high to low transition at this pin interrupts the processor and the content of the RCAP2 registers are changed before the arrival of the next interrupt. Firstly, on each interrupt, a number is added to the RCAP2 register till the contents of RCAP2 register corresponds to the upper limit for that frequency.

Table 5.1

Functions assigned to I/O port pins of microcontroller.

I/O Port Pins	Functions Assigned
P0.0 to P0.7	Data bus of tone and noise attenuators, Display
P1.0	Clock to oscillator and BP filter
P1.1	WR lines to Display, tone and noise attenuators
P1.2	Switch control line of 40 dB tone attenuator
P1.3	CS of noise attenuator
P1.4	Display control line, RS
P1.5	Display control line, EN
P1.6	Noise select line
P1.7	CS of tone attenuator
P2.0 to P2.3	Write to keypad row lines
P2.4 to P2.7	Read to keypad column lines
P3.0	Serial Interface (RxD)
P3.1	Serial Interface (TxD)
P3.2	Subject Response
P3.3	Interrupt line for warble tone generation
P3.4	PRBS clock control
P3.5 to P3.7	Not Used

After that, the contents of the RCAP2 register are decreased by the same number till the content of the RCAP2 register corresponds to lower limit of the frequency. The number by which the RCAP2 register contents are changed is different for each frequency and are given in the table 5.2. The sweep period for one cycle is around 120 ms. Thus, there will be more than 8 sweep cycles per second. When the external interrupt bit of interrupt zero is disabled, it has no effect on the clock frequency.

Table 5.2

The number by which the contents of the RCAP2 register will be changed for each frequency.

Frequency (Hz)	**Number**	**Frequency (Hz)**	**Number**
125	5	2000	1
250	3	3000	1
500	2	4000	1
1000	1	6000	1
1500	1	8000	1

5.2 Keypad

Here, a 4 X 4 matrix keypad is used. It is interfaced to the microcontroller through port #2. The keypad scanning is carried by a row-column matrix scanning technique as shown in Fig. 5.2. Keys are scanned by outputting data at the microcontroller port pins P2.0 through P 2.3 to poll each key column for a key press. Column read back data are read by the microcontroller at port pins P2.4 through P2.7 for decoding and taking the actions accordingly.

5.3 Display

The display is a 16 character X 2 lines LCD module from Oriole electronics, Mumbai. The display is interfaced to the microcontroller through port #0. It has three control lines (RS, R/W, EN) which are connected to other pins of microcontroller (Table 5.1). The display works on a single 5 V supply. Fig. 5.3 shows the block diagram of the display.

5.4 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 the Fig. 5.4. Normal TTL levels are used for the data transfer and reception.

5.5 Operation Sequence

Initially, the settings (viz. tone type, tone duration, mode of operation, and noise type) will be selected. After this, the stimulus parameters (e.g. frequency, tone level, and noise level) are decided by the audiologist. Once the parameters have been fixed, the stimulus is presented. After presenting the tone to subject, the instrument will wait for 4 seconds to receive the response from the subject. If no response is obtained within that period, 'N' will appear at the bottom corner of the display. If the response is positive , 'P' will be displayed. In manual mode, the threshold is decided by the audiologist while in automated mode, threshold is determined by the instrument. Also, in the automated mode, the number of total tones presented (Y) and responded (X) will be displayed in the form of X/Y. If the threshold has been decided, the levels for that frequency will be saved in the RAM area 80H to FFH. The operation sequence is given Fig. 5.5 which will be followed in both, manual as well as automated modes.

5.6 Key Functions

The functions assigned to the keys are given in Table 5.3. Keys numbered 1, A, and B will be used only during the initialization. If these keys are pressed in between the testing, they will be ignored. Key no. 8 will be used during both, initialization and testing. During initialization, it will be used to change the tone on duration while during testing, it will be used to increase the masking level.

Table 5.3

Functions assigned to various keys.

Key	Function assigned	Key No.	Function assigned
0	Reset	8	Noise level up / Tone duration
1	Auto / Manual	9	Noise level down
2	Frequency up	A	Noise type (WB, NB, None)
3	Frequency down	B	Pure tone / Warble tone
4	Level up	C	Save
5	level down	D	Recall / Print
6	Tone on	E	OK
7	Tone off	F	Cancel

(0) Reset

When this key is pressed, control will be transferred to 0000H location in the program memory which is the starting point of the program, thus reinitializing the program execution.

(1) Auto / Manual

This key is pressed to select the mode of operation of the audiometer. In manual mode, all the parameter changes and threshold determination are done manually by operator. In auto mode, the program will take all the decisions. A flag 'Automan' has been used to tell the processor in which mode the instrument is going to be operated. When the flag is set, the instrument will be in auto mode and when the flag is cleared, it will be in manual mode.

(2) **Frequency up**

Whenever this key is pressed, present frequency of the tone will be displayed on the display. If, the audiologists want to increase the frequency, he/she will have to repress this key and the new frequency will be displayed. Frequency which can be generated are 125 Hz, 250 Hz, 500 Hz, 1 kHz, 1.5 kHz, 2 kHz, 3 kHz, 4 kHz, 6 kHz and 8 kHz. Each time a new frequency is selected, the level of tones is set at 50 dB.

When a particular frequency is selected, the registers (RCAP2H, RACP2L) of timer #2, which is running in clock generator mode will be loaded with a number corresponding to that frequency. The numbers, which will be loaded for each frequency, are given in Table 5.4.

The timer #2, when running in clock generator mode, will output a clock of frequency given by

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

where $(RCAP2)_D$ is the decimal equivalent of the hex numbers in registers (RCAP2H,RCAP2L), taken as a single 16-bit register.

(3) **Frequency down**

This key is similar to the 'Frequency Up` key except that when ever this key is pressed frequency of the tone decreases.

Table 5.4

Numbers to be loaded for different frequencies (for 12 MHz crystal).

Frequency (Hz)	Number to be loaded (Hex)	Frequency (Hz)	Number to be loaded (Hex)
125	FE1F	2000	FFE2
250	FF10	3000	FFEC
500	FF88	4000	FFF1
1000	FFC4	6000	FFF3
1500	FFD8	8000	FFF8 / FFF9

(4) **Tone level up**

When this key is pressed, the attenuation of the tone is reduced by 5 dB and thus, level of the tone increases by 5 dB. The 'AD7111' attenuates the signal corresponding to the data latched into the chip. The resolution of the attenuator is 0.375 dB and for 5 dB increase in attenuation, the latched data has to be increased by 13D which in fact increases the attenuation by 4.875 dB, an error of around 2.5%. Each time this key is pressed, a number 0DH is added into the previously latched data and new data is latched into the attenuator.

(5) **Tone level down**

When this key is pressed, the attenuation of the tone is increased by 10 dB, and thus the level of the tone reduces by 10 dB. For this, no. 1AH is subtracted from the previous latched data and the new data is latched into the attenuator.

(6) **Tone on**

This key is pressed to present the stimuli to the subject. When this key is pressed, tone will be presented to the subject by turning on the timer. Tones will be presented for a preselected duration and after this period, the tone will be switched off. When the tone is on, a message 'T' will be displayed at the bottom right corner.

(7) **Tone off**

This key is pressed to remove the stimuli from the subject before the tone on duration has been lapsed. For switching off the tones, microcontroller simply stops the timer.

(8) **Noise Level up/ tone duration**

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

After the initialization, this key will be used to increase the noise level. When the noise is wide band, the attenuation will be increased in similar manner as the tone level. For narrow band noise, the BW increases with increase in the frequency. Thus, increase the level by 5 dB, the decrease in attenuation will not be 5 dB but it will be less.

(9) Noise level down

When the noise is wide band, decrease in tone level will be similar to the decrease in tone level. However, if the noise is narrow band, more attenuation will be required when frequency increases, to decrease the acoustic level by same amount.

(10) Noise Type

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

(11) **Pure Tone / Warble Tone**

This key will be used at the time of initialization of tone presentation to select the tone type. In "pure tone" mode, the external interrupt zero will be disabled and the stimuli will be of pure tone type. In "warble tone" mode, the external interrupt zero 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 and when this flag is cleared, tone will be pure tone.

(12) **Save**

This key will be used to save the threshold level and masking level in the RAM area starting at 80H.

(13) **Recall / Print**

This key will be used to display the results stored and to transfer the results to the serial devices. 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 values for the frequency 250 Hz will be displayed and device will display the message 'Next ?' for displaying the next set of data. If OK key is pressed, next set of data will be displayed. If 'CANCEL' key is pressed, the device will come out of the recall/print mode. If print mode is selected, The entire set of data will be transferred to the serial device.

(14) **OK and (15) cancel**

These keys will be used to confirm and cancel the message displayed.

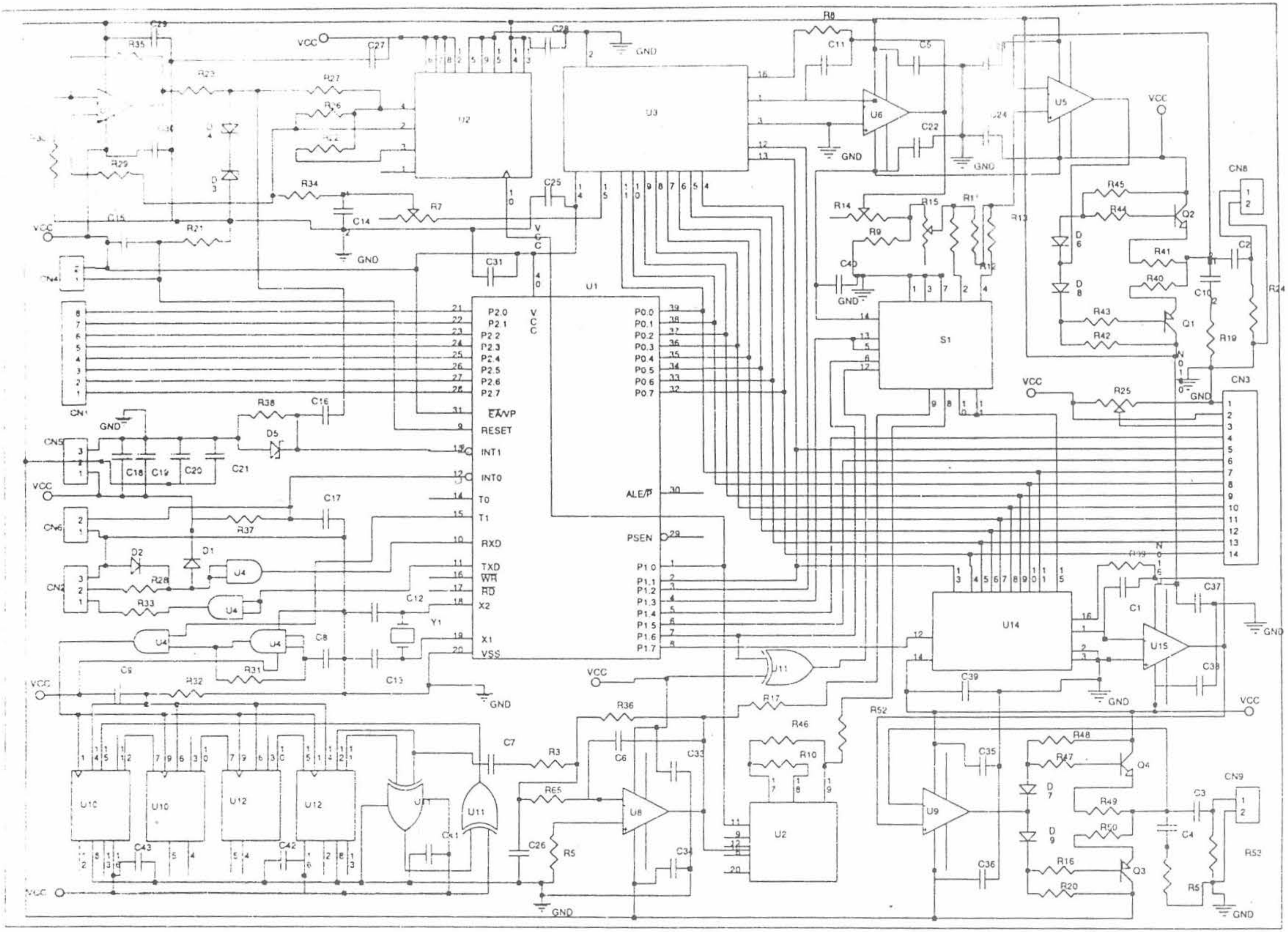


Fig. 5.1 : System Hardware

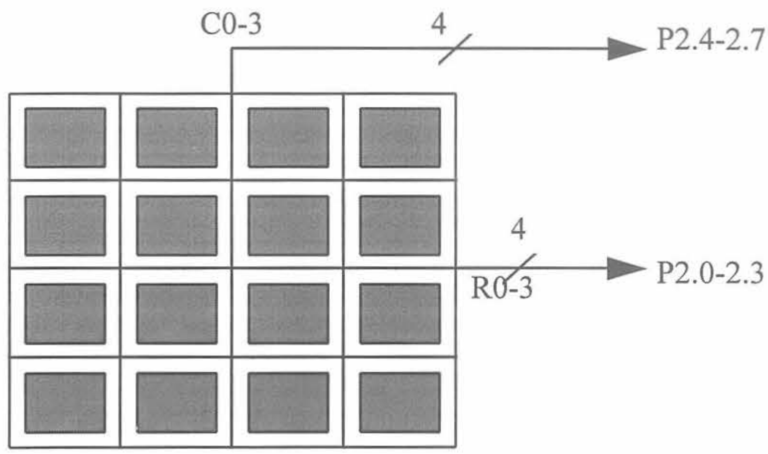


Figure 5.2: Keypad interfacing with microcontroller

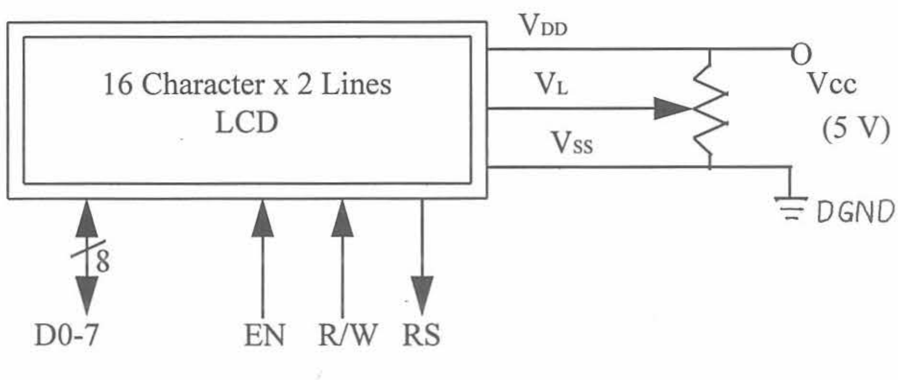


Figure 5.3 : 16 Character x 2 Lines LCD (ODM-16216S)

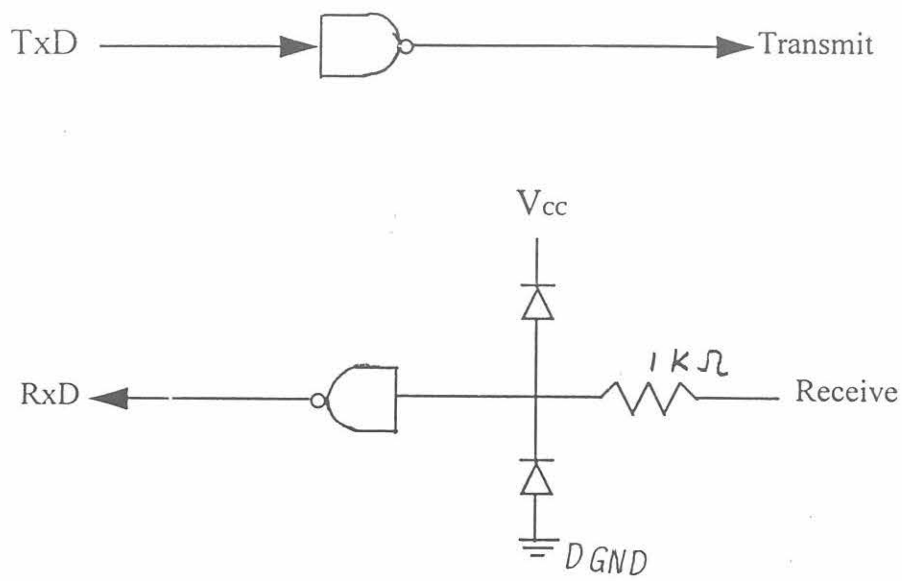


Fig. 5.4 : Serial Interface

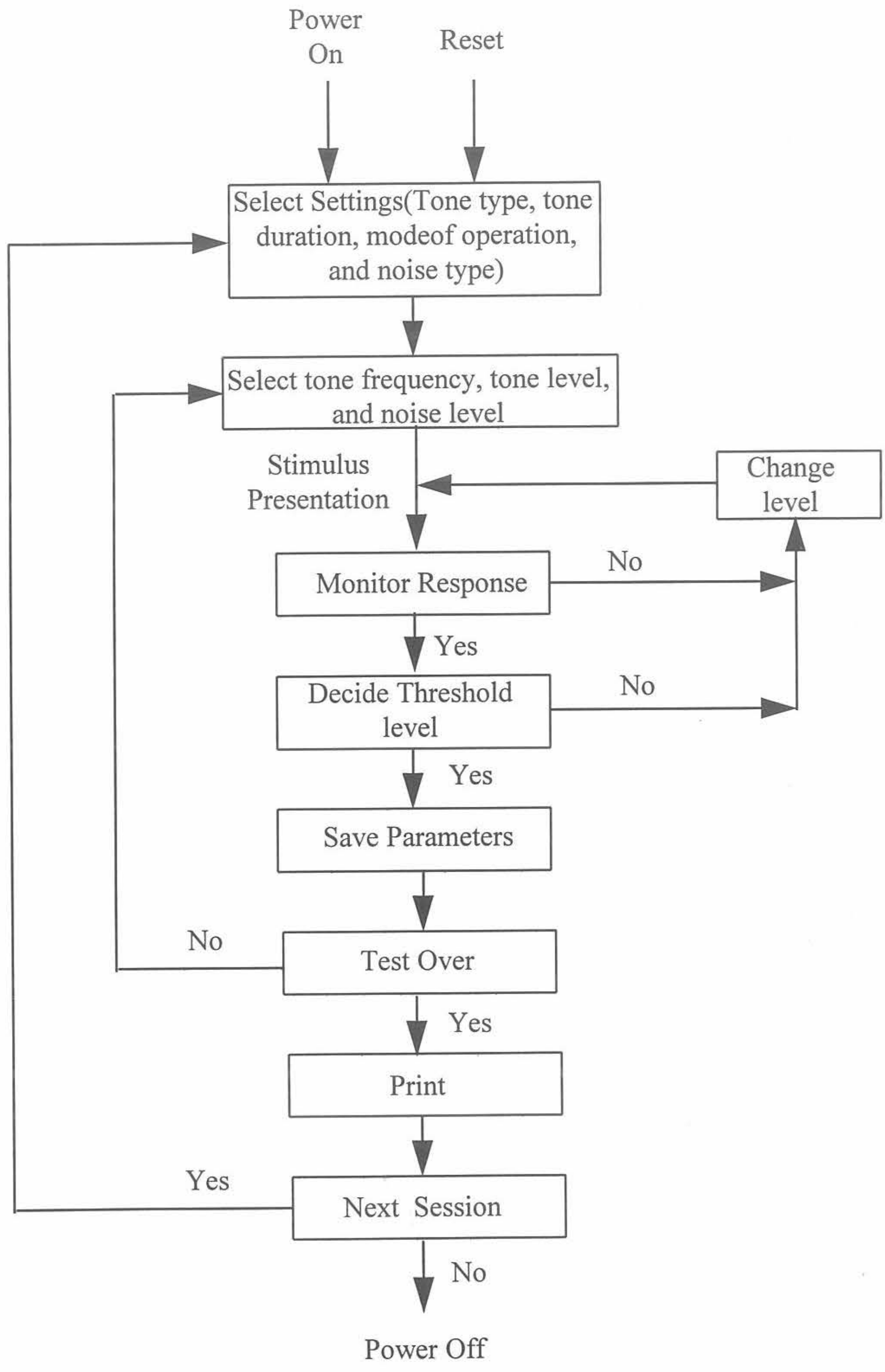


Fig. 5.5 : Operation Sequence of the Instrument.

Chapter 6

SUMMARY AND CONCLUSIONS

6.1 Work Done

A scheme for a portable audiometer was worked out and implemented using a microcontroller ATMEL 89C55. The microcontroller was interfaced with the audio attenuators, oscillator, noise generator, keypad, and display. Provision is also provided for the serial interfacing to transfer the stored data to a computer or serial printer. The audiometer is operated through a menu driven software, permitting both manual and automatic responses. All the settings are entered through keypad. For automated operation, the subject response button is monitored by the microcontroller.

Stimulus generation is done by a switched capacitor filter based oscillator. All tone frequencies are as per the specification, except that in place of 8 kHz, 7.5 kHz tones will be generated. The harmonic levels of the tones are more than 45 dB below the fundamental level. A logarithmic D/A converter has been used as attenuator, and power amplifier is of class B push pull type. The intensity level of the tones are from 0 dB to a maximum level as given in the Appendix A. Facility of warble tone is also provided. Here, the frequency sweep is not continuous but is in steps. As the center frequency increases, the number of steps in the warble decreases.

Masking facility is also provided. The masking can be either wideband, or narrow band. For noise generation, a PRBS based noise generator is used. Narrow band noise is generated by passing the wide band noise through a BP filter. The BW is adjusted to 36% of the center frequency (more than the one third octave band). The center frequency of the narrow band noise is equal to the test tone frequency. The attenuation rate outside the passband is 6 dB / octave.

6.2 Suggestion for Further Development

A number of design modifications can be carried out to incorporate the features and specifications of an advanced diagnostic audiometer.

The bone conduction facility can be provided by an additional amplifier for driving the bone vibrator. Speech audiometry feature can be provided by an option for connecting input from a speech source. Both these facilities will require modification in the program for using appropriate attenuation tables.

In the present design, narrow band masking noise has been obtained by bandpass filtering the broadband noise. The filter has been realized by using half of the SC filter IC MF-10 used for stimulus tone generation. This results in a filter with 6 dB/octave roll-off outside the passband. To get an attenuation rate of 12 dB/octave outside the pass band (as per IS 10565: 1983), a fourth order programmable BP filter will be required. This can be achieved by using another MF-10 IC.

As mentioned earlier, 7.5 kHz tone is produced in place of 8 kHz, and for warble tones, modulation is not continuous but is in steps. As frequency increases, the step size increases. This is due to the limitation on clock frequency generation and variation. Here, the clock is generated by a 16 bit timer. In order to get a smoother frequency variation and to generate precise frequencies, a PLL based frequency multiplier can be used to multiply the clock frequency before it is given to oscillator. Alternatively, we can use a hardware set-up with a timer having 18 or higher number of bits.

The stimulus can be better stabilized by using precision zeners. Further, an output level monitoring and display unit, independent of the microcontroller controlling the audiometer operation should be provided.

Appendix A

A.1 Hearing Thresholds

The human ear is sensitive to enormous range of pressures (about 10^5 fold). Thus it is more convenient to express sound intensity in decibels than in pressure units. The intensity level is expressed as

$$L\ (dB) = 20 \log(P/Po)$$

where P is the sound pressure and Po is the reference pressure. Generally, Po = 20 μPa which is the least pressure required for an average listener to hear a tone of 1 kHz. Intensities expressed in dB using this reference are called sound pressure levels (SPL). Other means of expressing intensity is to specify sound pressure at a particular frequency relative to the pressure at absolute threshold for that frequency. Reference level is different for each frequency as sensitivity of ear varies with frequency. This level is known as dB HL (decibel hearing level). It allows normal hearing to be operationally defined as a straight line at '0' dB HL. Thus, at different frequencies, the SPL levels corresponding to `0' dB HL (also called RETSPL) are different and are given in Table A.1. Also, the table A.1 shows the maximum level (dB HL) of the test tones at the different frequencies.

A.2 Calibration of the Teledyne TDH - 39 Headphone

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) and spring pressure on the headphone was kept at 0.5 kg. The voltage level(in dBm) required to produce 100 dB SPL at different frequencies were noted . The voltage level(in dBm) required to produce (HL)max and 0 dB HL was also calculated. The results are given in Table A.1

-42 dBm
66 mV ac

Table A.1

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

(B) Prescribed $(HL)_{max}$ for a simple diagnostic 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.

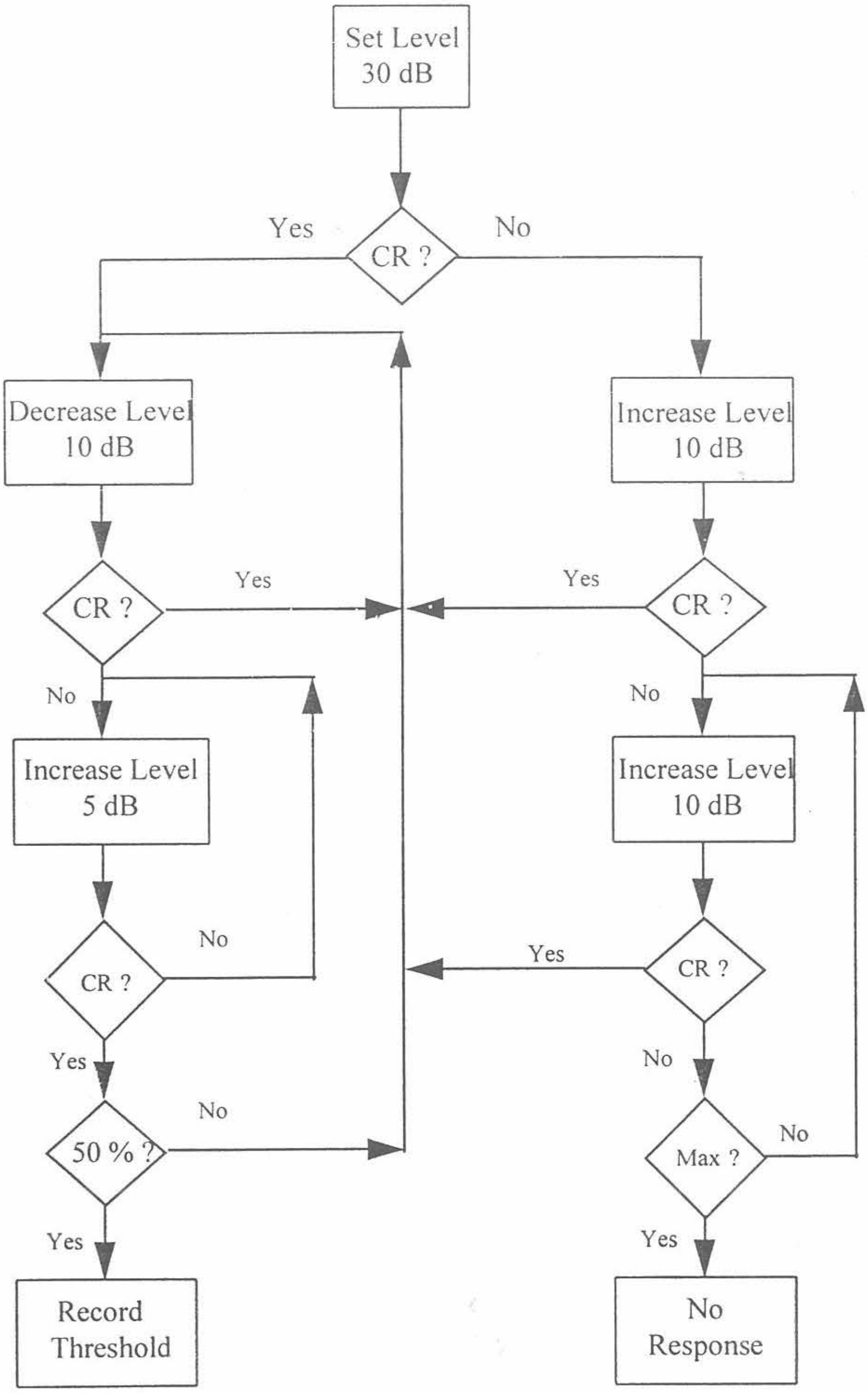
(E) RMS voltage (in dBm) required for producing $(HL)_{max}$ (as given in A) in a TDH-39 headphone.

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 $(HL)_{max}$
125	45	70	-14.63	-69.63	0.37
250	25.5	90	-16.48	-90.48	-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	-104.59	-5.49
6000	10.5	90	-8.83	-93.33	-3.33
8000	13	80	-2.39	-90.69	-10.69

Appendix B

B.1 Audiometric Procedure

The adaptive algorithm according to which the threshold determination is made is based on the one recommended by the American Speech and Hearing Association (ASHA) and adapted by the ANSI in 1978. The flowchart representation of the algorithm is shown in Fig. B.1. Initially a pure tone is presented to the subject at 30 dB. If the response is positive, then the limit is decreased in 10 dB step, else the limit is raised to 50 dB. After this, increase in the level is made in 10 dB step. If the response is positive then the level is decreased in 10 dB steps and increased in 5 dB steps. After the tone is presented, it is maintained for 2 or 3 seconds. until 50% criterion has been met [2].



CR ?: Check response

Fig. B.1: Pure Tone Audiometry Procedure [2].

Appendix C

C.1 System Specification

Audiometer type: Single channel microcontroller based audiometer, with pure / warble tone stimulus and wideband / narrowband masking noise.

Circuit size: Suitable for a compact instrument.

Controls: 4 X 4 keypad for controlling all operations of the instrument.

Display: 16 character X 2 line LCD display, for displaying the stimulus and masking noise level, subject response, and menu options.

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

Current Drain:

(a) Quiscent: mA for +ve supply and mA for -ve supply.

(b) With stimulus: mA for +ve supply and mA for -ve supply

Stimulus: Crystal controlled test tone frequencies, with intensity level from 0 dB HL to a maximum value given below for each frequency (as per IS 10565: 1983).

Frequency (Hz)	125	250	500	1000	1500	2000	3000	4000	6000	8000*
$(HL)_{max}$ (dB)	70	90	100	100	100	100	100	100	90	80

* actually 7.5 kHz

Warble tones (stair case modulation) with a modulation rate of 8 per second and frequency deviation within ±10% (as per IS 10565: 1983).

Intensity level variation in steps of 5 dB and intensity level calibration with 0.375 dB.

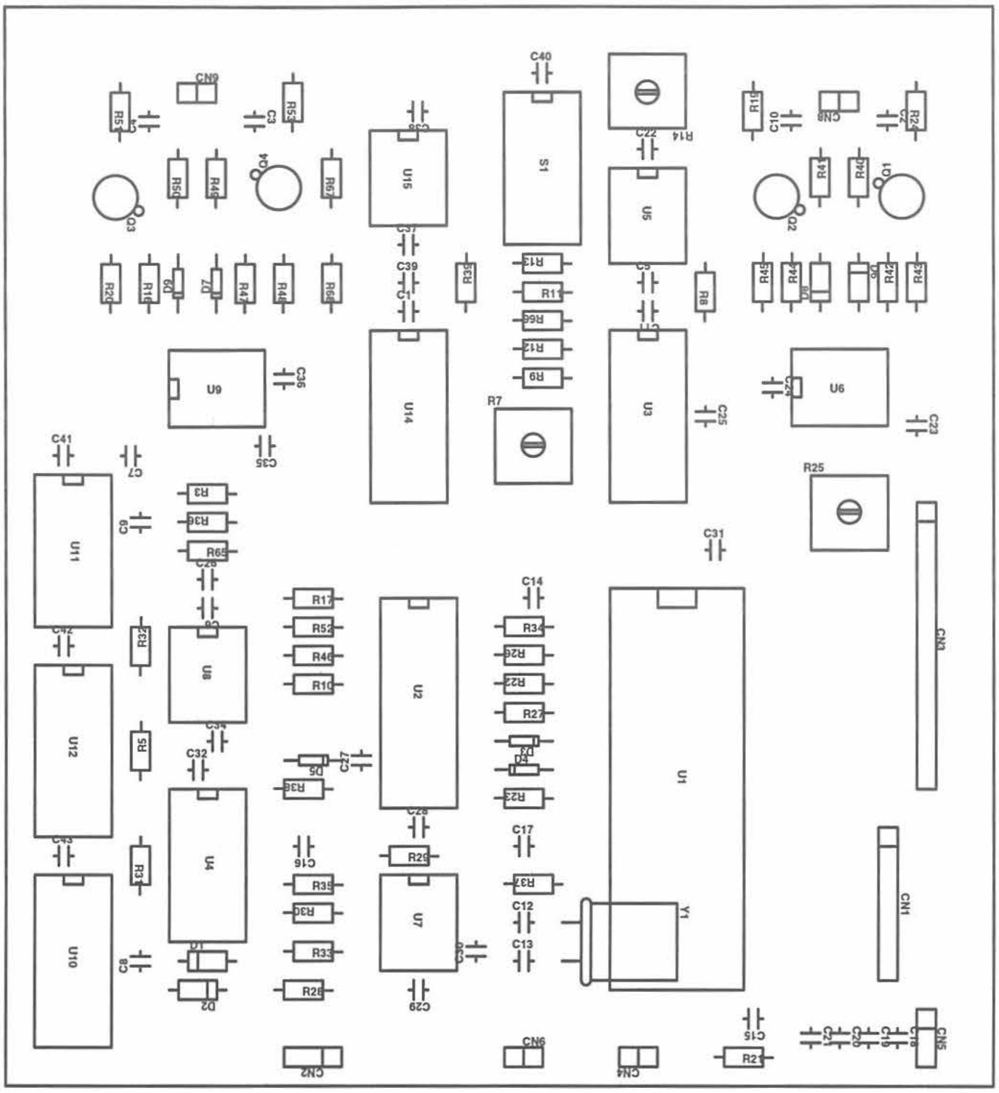
Masking noise:	Broadband / narrowband noise, with level sufficient to mask tones at 60 dB HL at 250 Hz, 75 dB HL at 500 Hz, and 80 dB HL from 1kHz to 4 kHz (as per IS 10565: 1983). Intensity level variation in steps of 5 dB. Narrow-band noise : Center frequency = Test tone frequency. Band width = 36% of center frequency. Attenuation rate outside the pass band : 6 dB / octave.
Operation:	Software controlled menu driven manual / automated mode.
Storage memory:	For one set of the test results with rewrite facility.
Interfacing:	Serial port (TxD, RxD, GND, TTL level) at a baud rate of 1200 for down loading the test results.
Headphone:	Type TDH-39 (calibration for multiple headphones can be stored in the software.

C.2 Cost Estimation

Component List (referance Fig.5.1)

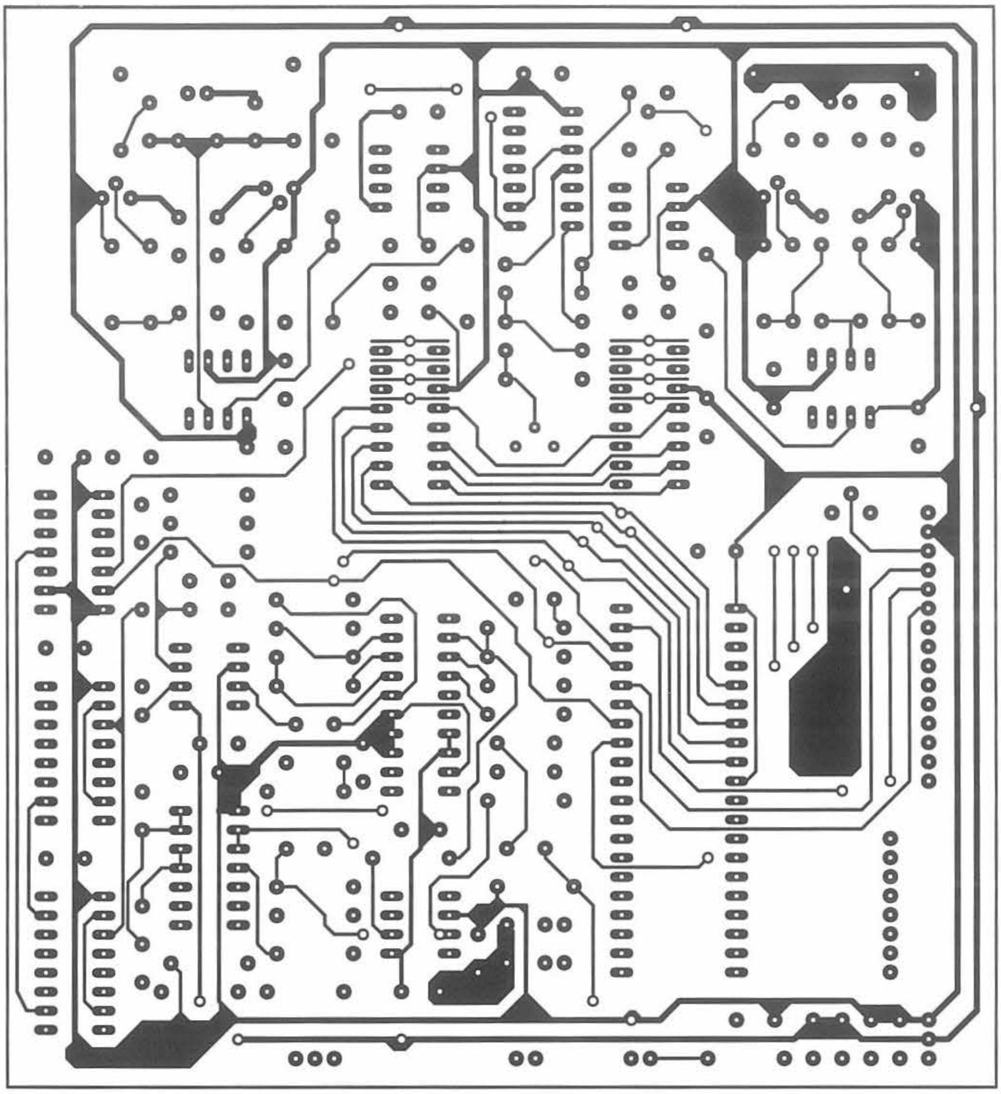
Reference Designator	Part Number/ Value	Part Description	Approximate price (Rs.)
C1, C11	47 pF	Capacitor	1.00 each
C2, C3	220 μF, 50 V	Capacitor (polarity)	4.00 each
C6,	1800 pF	Capacitor	1.00
C7	220 μF, 50 V	Capacitor (polarity)	4.00
C9	1 μF, 63 V	Capacitor (polarity)	4.00
C12, C13	39 pF	Capacitor	1.00 each
C14	22 nF	Capacitor	1.00
C15, C17	10 μF, 50 V	Capacitor (polarity)	4.00 each
C26	470 pF	Capacitor	1.00 each
C4, C5, C10, C18, C20 to C25, C27 to C43	0.1 μF	Capacitor	1.00 each
CN1		8 pin connector	
CN2, CN5		3 pin connector	
CN4, CN6, CN8, CN9		2 pin connector	
CN3		15 pin connector	
D1, D2, D6, D7, D8, D9	IN 4148	Diode	0.50 each
D3, D4	3.9 V Zener	Zener Diode	1.5 each
D5	5.1 V Zener	Zener Diode	1.5 each
R3, R5, R36, R38,	22 kΩ	Resistor	0.5 each
R7, R14	1kΩ	POT	15.0 each
R8, R39	270 Ω	Resistor	0.5 each
R9, R30	100 Ω	Resistor	0.5 each
R10, R17, R28, R33, R52	1.2 kΩ	Resistor	0.5 each
R11, R13	1 kΩ	Resistor (MFR)	1.00 each

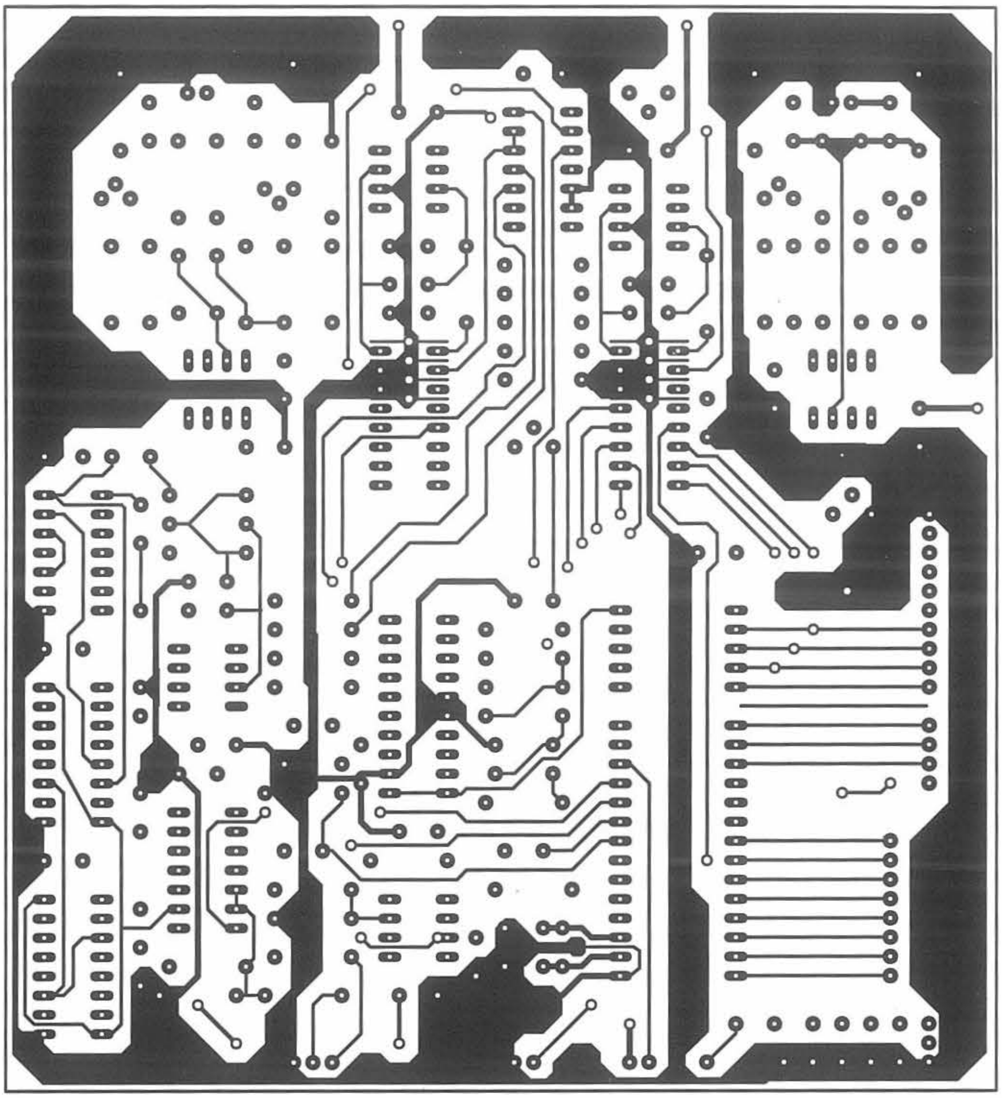
R12, R15	10 kΩ	Resistor (MFR)	1.00 each
R16, R43, R44, R47	470 Ω	Resistor	0.50 each
R19, R51, R40, R41, R49, R50	10 Ω	Resistor	0.50 each
R20, R42, R45, R46, R48	3.3 kΩ	Resistor	0.50 each
R21, R37	6.8 kΩ	Resistor	0.50 each
R22, R23, R24, R53	1 kΩ	Resistor	0.50 each
R25	50 kΩ	POT	15.0
R26	47 kΩ	Resistor	0.50
R27	150 kΩ	Resistor	0.50
R29	10 kΩ	Resistor	0.50
R32	220 kΩ	Resistor	0.50
R34	680 Ω	Resistor	0.50
S1	CD 4066	Electronic switch	15.0
U1	AT89C55-24pc	Microcontroller	800.0
U2	MF-10	SC Filter	355.00
U3, U14	AD 7111	Logarithmic DAC	855.00 each
U4	7400	NAND gate	15.00
U5, U6, U7, U8, U9, U15	LF 356	Op-amp	15.00 each
U10, U12	CD 4015	Shift register	15.00 each
U11	CD 4030	XOR gate	15.00
Y1	12 MHz	Crystal	12.00
	ODM 16216	LCD 16 char. x 2 L	800.00
		4 x 4 Keypad	400.00
		PCB - double sided PTH	375.00



U1
U2
U3
U4
U5
U6
U7
U8
U9
U10
U11
U12
U14
U15
S1
Y1
R7
R14
R25
CN1
CN2
CN3
CN4
CN5
CN6
CN8
CN9

Component Side





Solder Side

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