PURE TONE AUDIOMETER DESIGN

by

Yusuf ARPAT

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ABSTRACT

PURE TONE AUDIOMETER DESIGN

Audiometry is a technique which is used to measure where the hearing deficit stems from in aural way and what the degree of severity of deficit is.

This thesis work basically focused on a prototype of pure tone audiometer generating pure tone, wide and narrow band noise and warble tone. Direct Digital Synthesis (DDS) and logarithmic shaping methods are utilized for sine wave generation. For noise, firstly wide band noise is generated and then it is filtered by switched capacitor filter for narrow band noise. Warble tone is frequency modulated signal. Frequency modulation is implemented by Voltage Controlled Oscillator. All the obtained signals are made suitable to output apparatus and opening (80 dB) level via Digitally Controllable Potentiometer (DCP). Desired signal is selected via the multiplexer for right or left channel. Sequentially in order to match the signal between the -10 dB and 100 dB, its volume is readjusted in a special functional unit. Ultimately to supply the desired current drive for the TDH39 and bone vibrator, a current feedback operational amplifier is used. All the functional units are managed with the microcontroller coded as MSP430FG439 and the status of the device is showed on Liquid Crystal Display (LCD).

In Turkey, audiometer is not manufactured. Despite the fact that this prototype of audiometer is not appropriate to come on the market, it may give idea for production of more complex ones. Moreover, because of the fact that many hearing testing devices have analogous units with this prototype, inspecting this work carefully may give useful ideas about implementation of those devices. This prototype is tried to work out based on the standard of TS 9595-1-4 (EN60645-1-4) of Turkish Standards Institution.

Keywords: Pure tone audiometer, MSP430, DDS, Warble tone, Narrow band noise.

ÖZET

SAF TON ODYOMETRE TASARIMI

Odyometri kişide, işitme bozukluğunun, işitme sisteminin neresinden kaynaklandığının ve bu bozukluğun derecesinin ölçülmesinde bir teknik olarak kullanılmaktadır.

Bu çalışmada saf ton; dar band, geniş band gürültü ile warble ton üretebilen saf ton odyometresi prototipi üzerinde çalışılmıştır. Saf ton üretimi için Doğrudan Sayısal Sentez (DSS) ve logaritmik sinus benzetimi yöntemleri kullanılmıştır. Gürültü için; öncelikle geniş band gürültü üretilmiştir. Daha sonra anahtarlamalı kapasitör filtresi ile ilgili bandlarda filtrelenerek dar band gürültü elde edilmiştir. Warble ton ise frekans modüleli sinyaldir. Frekans modülayonu ise Voltaj Kontrollü Salınıcı kullanılarak gerçekleştirilmiştir. Elde edilen tüm sinyaller sayısal kontrol edilebilir potansiyometre'den geçirilerek başlangıç sinyal seviyesine (80 dB) ve çıkış aparatına uygun hale getirilmiştir. İstenen sinyal sağ veya sol kanal için seçicilerden seçilir. Daha sonra seçilen sinyallerin -10 dB ve 100 dB seviyeleri arasında herhangi bir değeri alabilmesi için, ses şiddeti, özel bir fonksiyonel birimde ayarlanmıştır. Son olarak ise çıkış aparatı olan TDH39 ve kemik titreştiricisini sürebilmek için akım geri beslemeli güç yükselticisi kullanılmış gerekli akım sağlanmıştır. Tüm birimler MSP430FG439 kodlu mikrokontrolcü ile kontrol edilmiştir ve işlemler Likit Kristal Ekran' da gösterilmiştir.

Türkiye'de odyometre üretimi yapılmamaktadır. Bu prototip ticari olarak piyasaya uygun olmasa da yerli malı kompleks odyometre üretimi için fikir verebilir. Ayrıca duyma üzerine yapılan bir çok test cihazı bu prototipteki işlevsel birimlere sahip olduğundan, prototip incelenerek bu cihazların gerçekleştirimi hakkında da fakir sahibi olunabilir. Bu çalışmadaki prototip Türk Standartları Enstitüsü 'nün TS 9595-1-4 (EN 60645-1-4) standardına uygun olarak gerçekleştirilmeye çalışılmıştır.

Anahtar Sözcükler: Saf ton odyometre, MSP430, DSS, Warble ton, Dar band gürültü.

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LIST OF SYMBOLS

ω	Angular Velocity
Δ	Change
R	Resistor
V	Voltage
C	Capacitance
T	Period
U	Integrated Circuit
t	Time
f	Frequency
Q	Quality Factor
I	Current
f_{MCLK}	Frequency of Master Clock
f_{ACLK}	Frequency of Auxiliary Clock

LIST OF ABBREVIATIONS

OSHA	Occupational Safety and Health Administration
TDT	Tone Decay Test
SISI	Short Increment Sensitivity Index
SRT	Speech Reception Threshold
SDS	Speech Discrimination Score
PTA	Pure Tone Audiometer
LCD	Liquid Crystal Display
TSE	Turkish Standards Institution
PCB	Printed Circuit Board
SPL	Sound Pressure Level
HL	Hearing Level
ISO	International Organization for Standardization
IEC	International Electrotechnical Commission
ANSI	American National Standards Institute
THD	Total Harmonic Distortion
LFSR	Linear Feedback Shift Register
DAC	Digital to Analog Converters
IC	Integrated Circuits
CFA	Current Feedback Amplifier
VFA	Voltage Feedback Amplifier
DIP	Dual in-Line package

1. INTRODUCTION

1.1 Audiometry and Audiometers

Humankind has been experiencing hearing loss for centuries due to the different kind of reasons. It is reported that there are many people who are suffering from noise induced hearing loss because of to be exposed to elevated sound levels as a result of living very closely to the Nile Waterfalls in Ancient Egypt [1]. However this is unintentional and involuntary; today, people inevitably exposed to very high sound levels in discos, in buildings under construction, and even in the roads while they are walking. According to the Occupational Safety and Health Administration (OSHA) about 1 million industry workers in the USA have noise-induced hearing loss particulary in frequencies 1000, 2000 and 3000 Hz. Furthermore there are some sensorial hearing loss so-called prespyacusis which arises with the increasing age. It is also known that some hearing aid devices also give some damage to the auditory organs as a side effect. Audiometry provides valuable data when assessing the degree of the hearing loss or diagnosing the reason of the communication deficiency [1].

Audiometry is the technique to identify and quantitatively determine the degree of hearing loss of a person by measuring his hearing sensitivity, so that suitable medical treatment or one of the appropriate hearing aids and assistive device can be prescribed. In this technique, auditory stimuli with varying intensity levels are presented to the person who responds to these stimuli. The minimum intensity level of these stimuli to which consistent responses are obtained is taken as the threshold of hearing. Depending on this threshold, the subject's hearing sensitivity can be estimated by obtaining an audiogram (An audiogram is a plot of threshold intensity versus frequency.) [2].

Depending on the procedure, logic and objective; audiometry can be performed with distinctively equipped audiometers. That is to say; thanks to the developing technology from the primitive audiometry to today's state of the art audiometers, developers have been presenting different types of audiometric procedures. Essentially they are Tone Decay Test (TDT), Short Increment Sensitivity Index (SISI), Bekesy Audiometry, Speech Audiometry and Pure Tone Audiometry.

All auditory tests are proposed to correctly locate the defected sensorineural pathway related to hearing organs. Of all these, most commonly used one is Tone Decay Test (TDT). The device used for this test is as the same as the one used in the Pure Tone Audiometry. In TDT for a certain frequency signal level which is 10 dB below the hearing threshold in related frequency is applied to the subject. And a timer is started. If subject responds within the 1 minute after the tone application, object is said to be all right in this frequency. But if s/he does not, instant, when the subject ceased to hear, is recorded and the intensity level is incremented 5 dB. These increments keep on above the 30 dB of the threshold for the related frequency. If subject still can not maintain hearing for 1 minute, test is halted and subject is said positive [3].

SISI Test is another audiometric test which is used to ascertain the deficiency in cochlea. After finding normal threshold object is subjected to the 1 dB signal increase in particular frequency. In SISI test, modulated signal is applied with a burst of 5 seconds interval and 1 dB modulating signal increase. 20 bursts are given to the subject and the number of the hearing events are noted. Then this is plotted as percent, as hearing representation in corresponding frequency [3]. It is freely asserted that this procedure requires extra hardware for audiometer.

Moreover Bekesy Audiometry comprised of nearly the same type of test steps but it necessitates a little bit complicated hardware and it is convenient for self-test. In this test audiometer is programmed to either increase or decrease the frequency. When subject responds, tone is ceased and device sets itself to the consecutive frequency. This tone can be pulsed or continuous and useful to detect the middle ear or 8^{th} nerve lesions [3, 4].

One of the other important hearing testing is Speech Audiometry. It is mainly used to identify the neural dysfunction instead of cochlear or conductive path dysfunctions. It is well suited to determine both receptive and discriminative speech inability. One subsection of the Speech Audiometry is Speech Reception Threshold (SRT) Test. In this test 6 spondee words between predefined 36 is given to the subject nearly 25 dB above the normal hearing threshold (for 500 Hz and 1 kHz) and consecutively signal intensity is decreased. If the subject pronounces the 3 out of 6 correctly than this level is marked as SRT. In Speech Discrimination Test at this time monosyllable words are presented via the headphones to the subject and asked to repeat after the device. By doing so, a percentage is obtained which shows how correctly the object discriminates the words and that percent is noted as Speech Discrimination Score (SDS). After that, plot of the intensity versus percentage of the words discriminated correctly (SDS), is used to separate the neural or other types of hearing loss [4].

Pure Tone Audiometry can be regarded as basic and indispensable procedure in clinical environment. Pure tone audiometric air conduction testing is performed by presenting a pure tone to the ear through an earphone and measuring the lowest intensity in decibels (dB) at which this tone is perceived 50% of the time. This measurement is called threshold. The testing procedure is repeated at specific frequencies from 250 to 8000 Hertz (Hz, or cycles per second) for each ear, and the thresholds are recorded on a graph called an audiogram. Bone conduction testing is carried out by placing a vibrator on the mastoid and marking the exact spots where the subject is able to sense. Masking noise is sometimes used in the non-tested ear to prevent its interference to the tested one. Furthermore sometimes warble tones are highly desirable because of the induced standing waves between source and destination. Standing wave formation is one phenomenon when the wavelength of the test tone is the integer multiple of the distance between surfaces of the test booth. In spite of high sound absorption constant of the surfaces of the booths, induced reverberation brings about some areas where the sound amplitude changes unintentionally. This leads wrong audiograms and wrong diagnosis.

By the way, needless to say, all these techniques mentioned above can be put into the class of subjective audiometric techniques.

It has been already said that basic audiometer is designed to give some predetermined stimulus to the ear or mastoid which are very important in terms of evaluating the efficiency of the human hearing system. Fundamentally, an audiometer composed of signal generators (sine wave, noise, warble, short tone), equalization circuitry, attenuator and power amplifier (see Figure 1.1). In generator sections pure tone sine wave is generated with suitable structure of oscillators starting from 125 Hz to the 20 kHz as an upper edge. In addition, generated sine wave must obey the frequency accuracy and distortion constraints of the standards if it is thought as a commercial device.

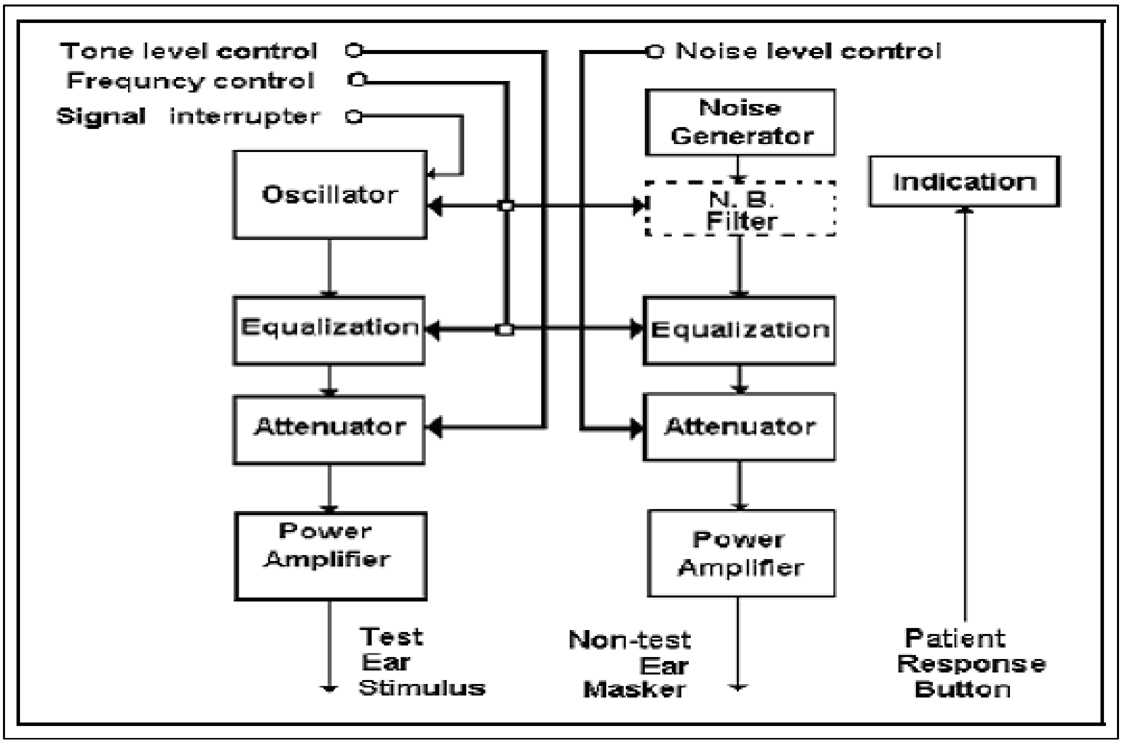

Figure 1.1 Basic Audiometer Architecture

Noise Generator is the source of the masking noise. In audiometry, however the signal is supplied only one ear there arises interaural interference owing to the bone skull. In air conduction audiometry the level of attenuation is in the range of 45, 80 dB (Even so sometimes masking is used.). On the other hand in bone conduction audiometry, this level is on the order of 0 dB, means interference is unacceptably high [4]. Therefore non tested ear channel is masked with the narrow band noise whose frequency is centered to tested ear pure tone. Warble tone is another option when proper energy transfer to aural way is the primary interest in the existence of the standing waves. In fact, warble tone is nothing but frequency modulation of two signals; pure tone sine wave as carrier and another sine wave generated by dedicated source as modulating wave.

Equalization circuitry is mainly used to make a calibration on the generated signals' amplitude on account of the fact that power amplifier not only drives a unique apparatus but also drives mastoid vibrators. Hence it is apt to adjust the signals before sending the attenuator in order to make their apparent amplitude equal but absolute amplitude customized according to device connected to the power amplifiers. Then signal is applied to the attenuator. Attenuator simply takes the signal and intensifies or lessens the voltage of the signal according to the preset value by the audiologist. Then signal is send to the power amplifier. Amplifier ensures the necessary current drive instead of voltage amplification. Then signal is conducted to the jacks. Coordinating all this blocks may be achieved by some sort of rotary encoders, buttons and dials as in the case in older designs. But contemporary audiometers accommodate microcontrollers for handling all these functional blocks as well as user interface. That is, user is not direct collocutor for the blocks but s/he only addresses the orders to the microcontrollers via user interface. With regard to integrity; hardware is not formed merely these blocks. There should be an interrupter which stops the signal application to the object. Apart from this there should be some kind of memory unit to store the obtained hearing levels from each individual object. Again there should be a screen to denote the status of the audiometer.

1.2 Aim and Motivation

Audiometry consists of quite many branches and hardware. In this thesis, it is planned to work out basic pure tone audiometer. It is dual channel. Its frequency range extends from 125 Hz to 16 kHz. It also has noise masking capability. Besides it is planned to show the status of the audiometer with 2x40 Liquid Cyrstal Display (LCD). All the cooperation, regulation and communication will be managed by a microcontroller. After hardware implementation some measurements proving the system accuracy, adherence to the some distortion obligations and overall system performance will be examined.

After all; it is important to note that there is no commercially available audiometer device manufactured in Turkey. But substantial volume of audiometer market in Turkey can not be neglected. A device complying with the international standards has the potential to be marketed world wide. Although it is not easy to compete against well established, powerful audiometer brands, our design will have significant differences utilizing the state of the art technology. As a result of this thesis prototype will be developed to be introduced to the local manufacturers.

Particularly some devices are, highly analogous with the internal architecture of the basic pure tone audiometer. One of them is Hearing Aid Analyzer manufactured by numerous firms. This device checks the proper working of the hearing aids. Similarly Hearing Aid Analyzer use pure tones warble tones and different types of noise. In the same way with the audiometer this device uses attenuator and equalization circuitry and some drivers. All in all when audiometer is well designed than it is no hard to redesign the Hearing Aid Analyzer inspiring from the audiometer. Therefore such a company with its well developed audiometer architecture it may easily broaden its product portfolio.

In addition designing a cheap and sensitive audiometer, it can be important by the aspect of early detection of ear deficiency or illness and precise diagnosis can lead the future therapies in the right way.

1.3 Thesis Outline

The thesis contains different chapters summarizing topics related to the hardware and some other theoretical explanations of different functional blocks of the Pure Tone Audiometer (PTA).

In the first chapter, definition of the audiometry and PTA is given. Some contemporary types of audiometry and internal structures of audiometer are mentioned roughly. The objectives and motivation of this thesis work is clearly indicated.

In the second chapter, physiology of hearing and some certain characteristics (anatomy) of human ear is briefly presented. Theory of audiometry according to the physiology of ear, some aural diseases that can be detected by the audiometric tests and their characteristics in the audiogram are explained.

In the third chapter, the specifications of the PTA based on the TSE (Turkish Standards Institution) are expressed. Apart from this our design matching and mismatching to the standards is highlighted.

The forth chapter may be the most important chapter of this thesis work. In this part the functional parts of the designed PTA, are investigated. Furthermore the instruments in the design explained in detail. Connections between the functional parts and microcontroller are explained. Moreover, Printed Circuit Board (PCB) implementations of the circuits are presented. Program utilized while producing the PCBs is shortly mentioned. At the same time some limitations of the microcontroller are included in this section. Additionally one can easily find the information about the firmware used to form the C codes of the microcontroller.

In chapter five, software tools are introduced and their pros and cons in this design are discussed.

In chapter six, results of the measurements related to the distortion, accuracy and overall efficiency of the device are reported.

The last chapter underscores the deviations (if there is any) of obtained results with respect to desired results. Chapter seven concludes with the future works about the device. This chapter also includes some renovations, some cost and design optimizations pertaining to the device.

2. HEARING MECHANISM

2.1 A Brief Introduction to Ear Anatomy

"The ear is marvelously sensitive structure. Its sensory receptors can convert sound vibrations into electrical signals 1000 times faster than photoreceptors can respond the light [5]." As well as these fast responding receptors it is equipped with some balancing formations in the inner and middle ear but these structures are beyond the scope of this thesis.

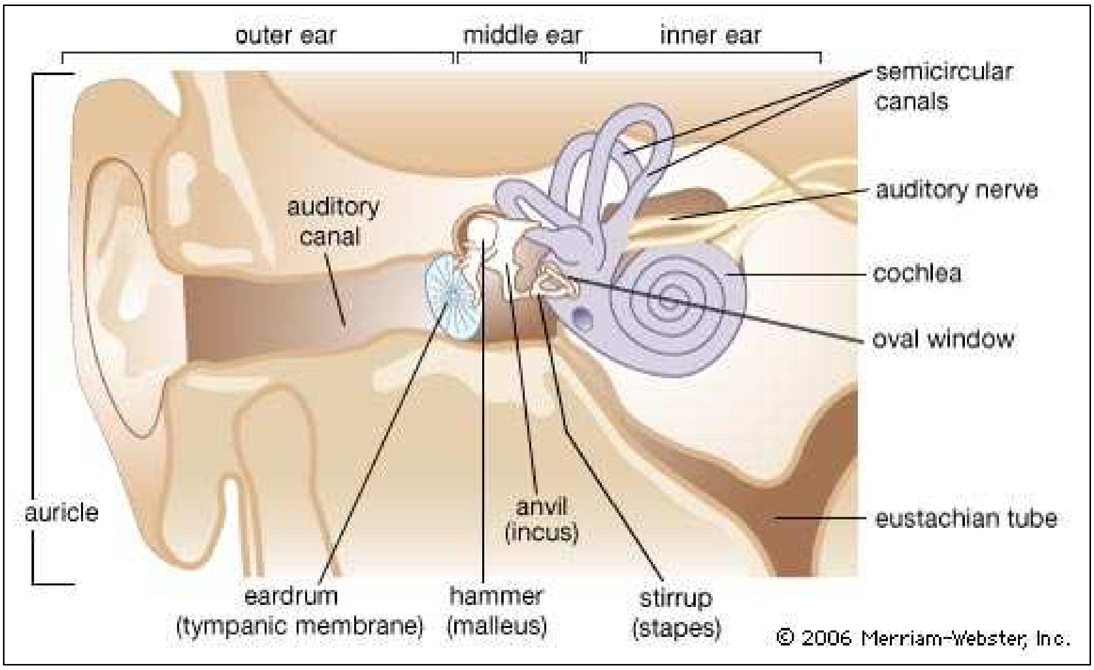

Figure 2.1 Anatomy of Ear [7].

Anatomic structure of the ear is divided into three parts; outer ear, middle ear and inner ear. Outer ear is the unique part of the ear that directly contacts with the physical world. It accumulates the sound waves and directs them into the middle ear. In the middle ear sound is actually transmitted as vibrations through three specialized bones. Finally, these vibrations conducted into the inner ear which is a formation of bone canals filed with some kind of bodily fluid and converted into the neural impulses by the Organ of Corti [6].

2.1.1 Outer Ear and Middle Ear

Outer ear composed of Auricle, external auditory canal and eardrum (see Figure 2.1). Thanks to the specialized structure auricle collects the sound. Actually it plays notable role collecting the sound waves. Apart from the Auricle, External Auditory Canal, a tube like structure, collects and directs the sound waves to the Eardrum. Eardrum converts the sound waves into the bone vibrations and passes them inward to the middle ear. Middle ear has the three rigid bone fragments so-called auditory Ossicles; malleus, incus, stapes. They conduct the vibrations loss free in the air filled environment. There are also two important striated muscles, tensor tympani and stapedius. Contraction of the former pulls the manabrium of the malleus medially and decreases the vibrations of the tympanic membrane, however contraction of the latter pulls the footplate of the stapes out of the oval window [6].

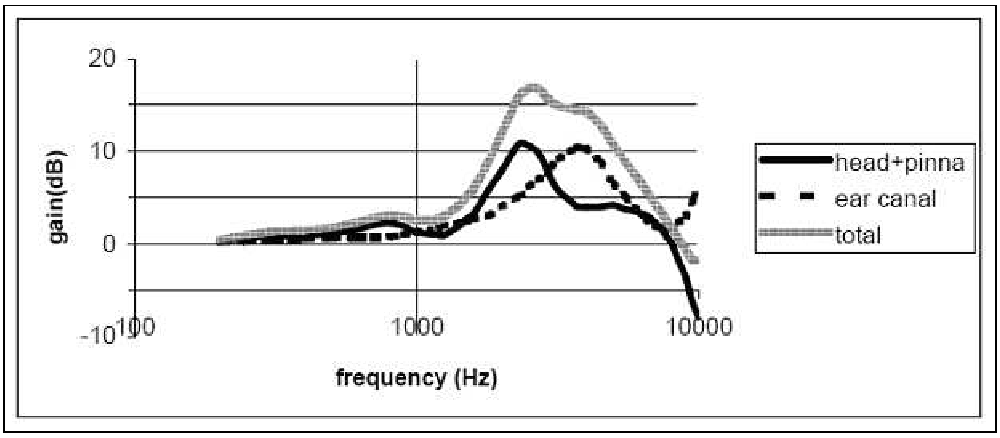

Figure 2.2 Frequency response of structures pertaining to the outer ear [8]. Ear canal and Pinna strengthens the signals between 1000 Hz and 6000 Hz.

2.1.2 Inner Ear

Inner ear is part of the ear where pressure changes are translated into the neural impulses via some specialized cells named as hair cells. Inner ear contains cochlea and semicircular canals. The former one is responsible from achieving the hearing and contains Organ of Corti; latter one is the sense organ for equilibrium and balance (beyond the scope of this work). In addition to these there is another bony structure named as vestibule. Similarly it is another organ of equilibrium.

A transverse cut through the cochlea, shows that it is divided into three canals: the scala vestibule, the scala tympani and scala media (see Figure 2.3). Scala vestibule starts with the oval window and its leftmost canal between these there canal systems. However scala tympani ends in the round window which is located right under the oval window. Both canals contain perilymph and totally separated from each other except the point the apex named as helicothrema. On the other hand, the third canal named as scala media and differently it is full with the endolymph which is structurally different but functionally same fluid. Most importantly, scala media surrounds the Organ of Corti [6].

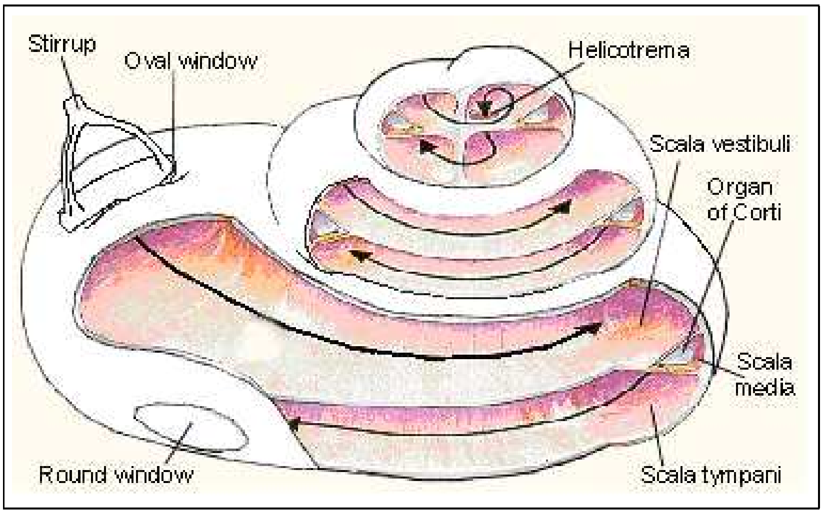

Figure 2.3 Cochlea.

Organ of Corti consists of some supporting cells and hair cells which are aligned as outer and inner cells. Organ of Corti is in contact with basilar membrane from the tympanic side and vestibular membrane from the vestibular side. Moreover the outer hair cells are shielded with the tectorial membrane [6].

2.2 Physiology of Hearing

Firstly sounds are collected and directed to the external auditory pathway by the auricle. Then external pathway leads the sound waves to the eardrum, this causes some vibrations in the eardrum. The louder the sound the stronger the vibrations induced by it. Nevertheless eardrum vibrates more slowly in response to low-pitched sounds and more fastly in response to high-pitched sounds. Owing to the fact that the eardrum binds the malleus, sound vibrations transmitted malleus, incus and steps sequentially. Further vibrations cause back and forth movements in the oval window and forms pressure in the perilymph (in the vestibular canal). Because of the fact that the fluids can't be compressed, pressure leaded to the tympanic duct also and bulge the round window towards the middle ear. At the same time as the scale tympani and scale vestibule are deformed by the pressure they push the vestibular membrane back and forth causing pressure waves in the endolymph. As a result basilar membrane moves the hair cells towards the tectorial membrane, hairs of these cells undulate which is a self-triggering effect to release the neurotransmitters to the synapses between the neurons belonging to the vestibulocochlear nerve and hair cells (see Figure 2.4) [5].

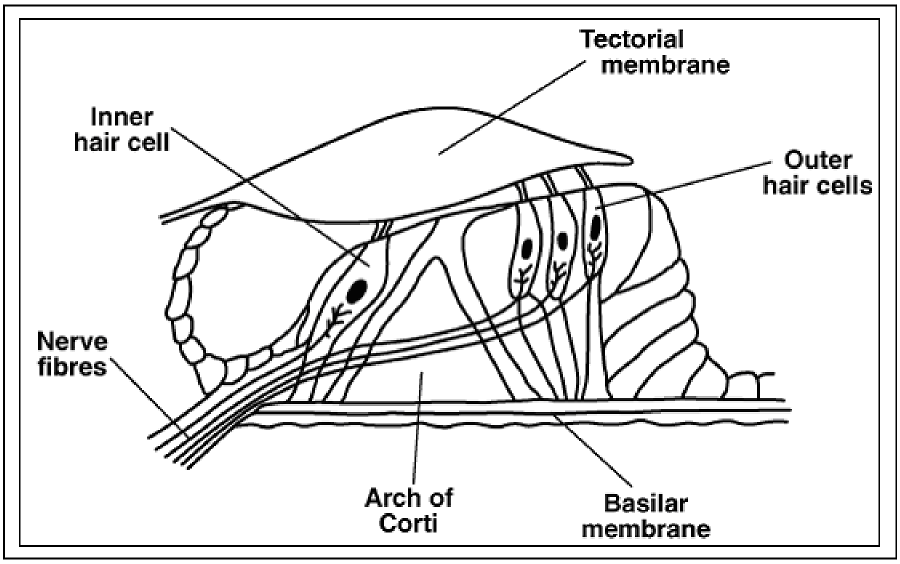

Figure 2.4 Organ of Corti.

The inner hair cells consist of a single row running the length of the cochlea and lying closer to the core of the cochlea. Sensor neurons are in close contact with the top surface of each receptor hair cell and they transmit electrical changes to the brain through the basilar membrane when sound waves disturb the fluids in the vestibular and tympanic canals. There are three or four rows of outer hair cells, which are further away sideways to the length of the cochlea, through which the brain sends messages back to the cochlear to mediate activity that the inner hairs have transmitted. The outer hair cells complete the feedback loop from the brain back to the inner ear. The returning messages refine the sensitivity and frequency selectivity of the mechanical vibrations of the fluids in the cochlea by causing the outer hairs to inject inhibitor chemicals to counteract the action of the inner hairsti fluids [6].

Basilar membrane has specified regions for different signals whose pitch rate is different from each other. The base field of the cochlea is stiffer and tiny with respect to the other side so maximum energy transfer occurs between high pitch sound waves and basilar membrane nonetheless base side is more flexible and bold respectively. Hence low pitch signals transfer maximum energy to the basilar membrane there. This is nothing than resonance actually.

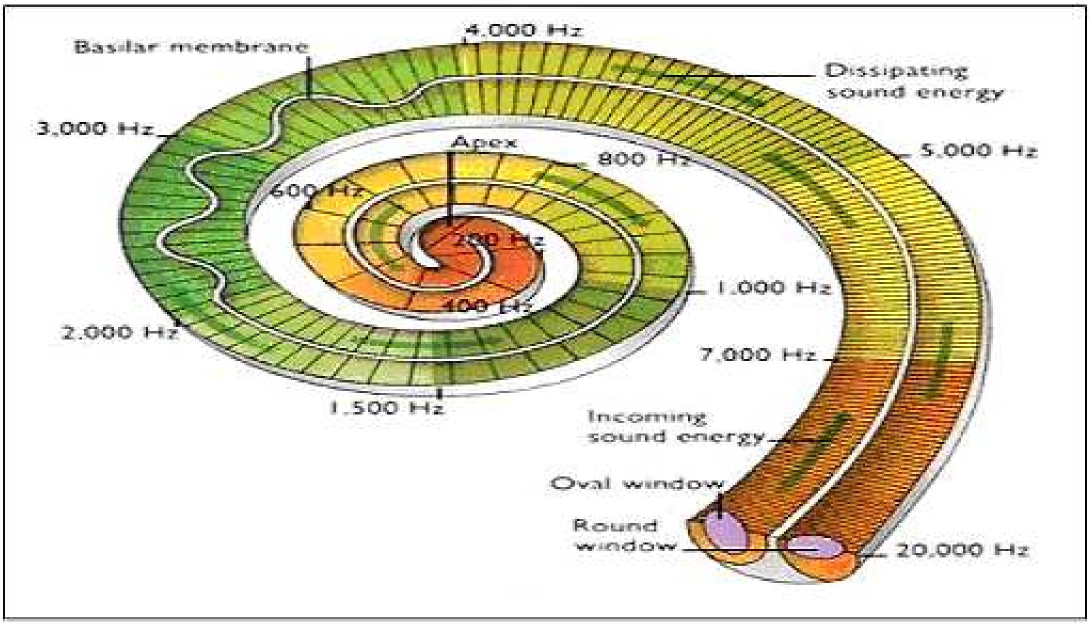

Figure 2.5 Sensitivity map of the cochlea.

Incidentally it is beneficial to say that the loudness counter-equivalent of the intensity. If the intensity of the sound waves increases so the basilar membrane vibrations elevated and this increase the firing rate of the nerves. This also means higher frequency of impulses reaching the cerebrum [6].

2.3 Frequency and Amplitude Dependent Characteristics of Ear

Human ear is capable of getting and processing sounds in the frequency range of 20 Hz - 20 kHz. As to intensity, 140 dB is the sound level generally accepted as high as to cause permanent hearing loss. Also it is stated that the intensity of the ear is best between the frequencies 1 kHz and 5 kHz. To absolutely specify the degree of the magnitude of the signal a reference level must be stated. This reference level is determined as testing healthy young people in several times. According to ANSI S1.1-1994, 20 μPa at 1000 Hz is accepted as the best available pressure level [9]. In general any sound level is measured against 20 μPa in a logarithmic scale and has a unit of dB SPL. Mathematically it is defined as [10]:

$$L_p = 20 \times \log \frac{p_{rms}}{p_{ref}} \tag{2.1}$$

where p_{rms} is measured sound pressure level and p_{ref} is 20 μPa. For instance according to the formula we can determine the 0.002 Pa Sound Pressure Level as 40 dB (a quiet library), 6.3 Pa as 110 dB (chainsaw 1m distance), 200 Pa as 140 dB (jet aircraft 50 meters away) [11].

Human ear is not sensitive to all frequencies in its sensing band. The sensitivity is best at 1 kHz and 5 kHz but gradually worsens at lower and higher frequency. Figure 2.5 illustrates the frequency response of ear to the sounds having different frequencies.

This graph is a reference graph as newly updated equal loudness contours which are determined by the ISO 226:2003. An equal-loudness-level contour represents a frequency characteristic of the sensitivity of human auditory system, to be drawn by connecting sound pressure points sounding identically loud for different frequencies, representing an equal sensation contour in the sound-pressure-level and frequency plane

[12].

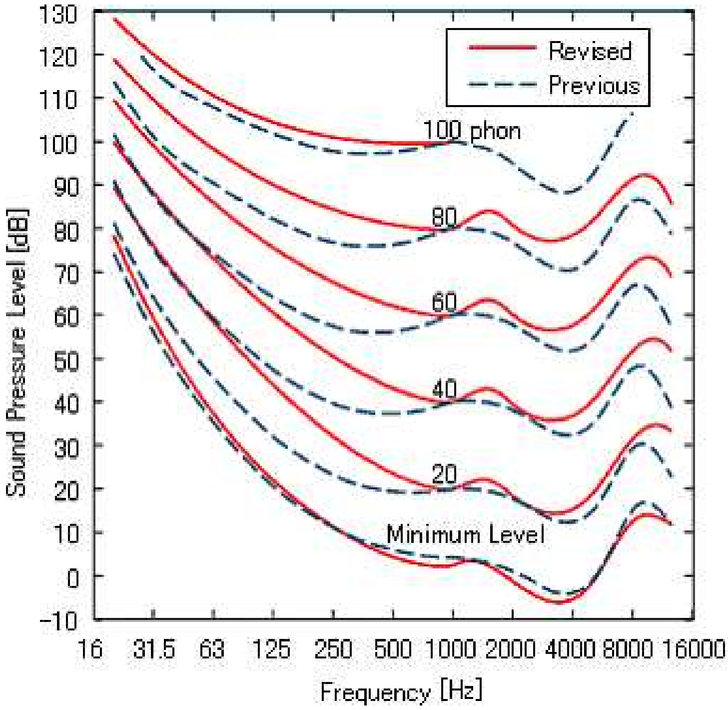

Figure 2.6 Redetermined Equal Loudness Levels and Hearing Thereshold [26].

"By definition any two sine tones that have equal phons are equally loud. The curves of equal phons are called equal-loudness contours." In other words, frequency effect on loudness is compensated classifying the two signals in same phon level [12]. For example on the basis of the graph above 60 phons sound level at 2000 Hz equals to 60 dB SPL however 60 phons sound level at 90 Hz equals to 80 dB SPL. Namely the latter one must be 20 dB SPL greater than the former to be heard at the same phon level.

However there is another unit to express the intensity of hearing, named as Hearing Level (HL), usually written as dB HL. This term is the threshold of hearing with respect to the normal hearing level (SPL). For example if somebody hears certain frequency of sound at 70 dB HL; this means that this subject hears 70 dB as bad as the normal hearing subject does [10]. Figure 2.7, shows the corresponding conversion values between dB SPL and dB HL.

Based on the International Organization for Standardization (2003), Figure 2.7

exhibits 0 dB HL and its corresponding dB SPL for some frequencies [13].

Frequency [Hz]	dB SPL	dB HL
250	12	0
500	5	0
1000	2	0
2000	-2	0
4000	-5	0
8000	13	0

Figure 2.7 Chart showing the relation between dB SPL and dB HL for some frequencies [13].

2.4 Audiogram

Audiogram is a chart showing the hearing characteristics of the subject. It is an indispensable tool clueing the physician about the possible hearing deficiency of the subject. It is used in the PTA. In this type of audiometric test, tones having different frequencies applied to the subject and when the subject hit the tone interrupter, applied test tone amplitude is marked to the chart as "O" or "X" as representative shapes right and left ears respectively. Apart from this "["and "]" shows test tone amplitude in bone conduction of right and left mastoid respectively. In a standard chart frequencies from 250 Hz to 8000 Hz are shown in the x axis. Amplitude levels are shown in the y axis from -10 dB HL to 120 dB HL.

Some patterns in the audiogram belong to a specific disease so when these kinds of patterns are encountered physician may easily diagnose the patient because this pattern is peculiar to that unique disease. For example Figure 2.8 shows the characteristic audiogram of Presbycusis (left chart). This disease is age related and sensorineural and it causes remarkable loss in the high frequencies. In Figure 2.8 in the left chart both ears exhibit same characteristic so there is only one curve. In the same figure, the chart in the left exhibits a normal hearing characteristic. In this chart "X" indicates left "O" indicates right ear.

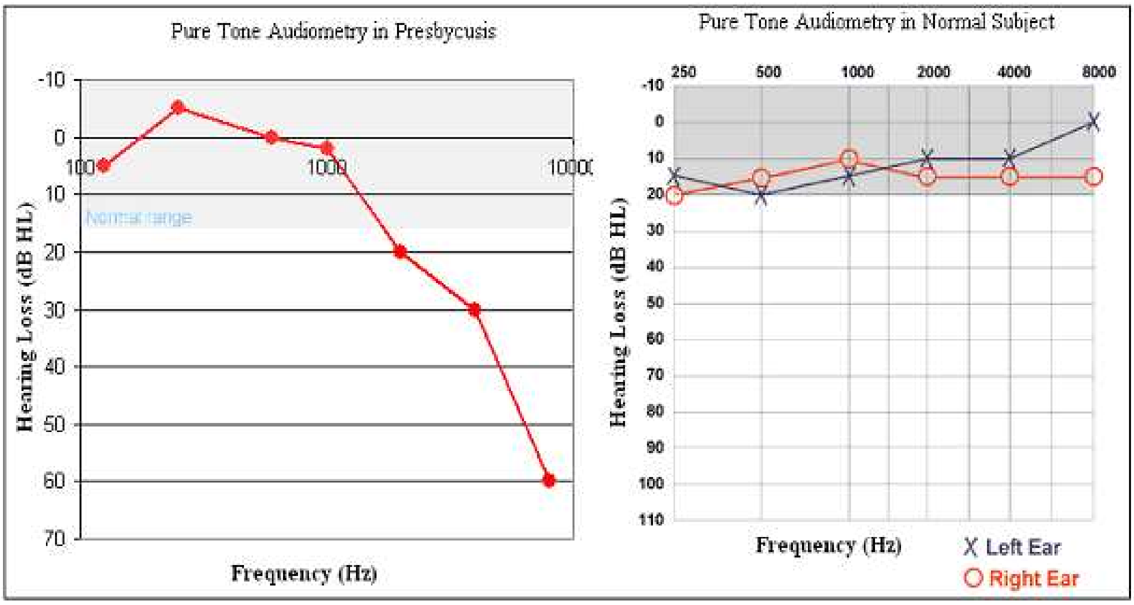

Figure 2.8 Audiograms representing the characteristics of a subject who is suffering from Presbycusis (left) and representing Normal Hearing Subject (right). Adapted from [14].

3. STANDARDS AND SPECIFICATIONS OF MATERIALIZED DEVICE

3.1 Requirements for PTA Based on Turkish Standards Institute

The first induction coil audiometer was developed by Hartman (1878) in Du Bois Raymonds Laboratory in Berlin. Others were developed in a variety of countries in an independent manner over the next couple of years and the ones produced by Högyes (1879) in Hungary and Hughes (1879) in London may be asserted as the best [15]. In 1937 the Maico D-5 audiometer was introduced. This was the first audiometer with a zero reference level that was adjusted automatically for each frequency. The first articles that dealt with the design of soundproofed rooms appeared in the professional literature in 1938. One of the greatest breakthroughs in terms of diagnostic audiology occurred when the two-channel audiometer was developed [16]. And up till today many manufacturers are included to the contest of producing new audiometers in terms of technology, functionality and modularity. Because of huge number of manufacturers, institutions like IEC, ANSI published some standards about different parts of the audiometers. These are the obligatory and international rules for the commercial audiometer brands.

Audiometer devised in this study is based on the EN 60645-1 (2003) and TS 9595-1 (2006) and EN 60645-4 (partially). In reality TSE is customized form the EN Standards. Some parts of the mentioned standards will be refered and clearly indicated in this part and than specifications of audiometer implemented in this work, will be clarified.

The objective of thesis project is to design a PTA instrument in compliance with Type 3 (see Table 3.1) of TS 9595-1 standard which is classified as Basic Diagnostic.

Table 3.1
Minimum requirements for fixed tone Audiometers [17].

Function	type 1 advanced clinical research	type 2 clinical	type 3 basic diagnosis	type 4 monitoring
air conduction				
-two earphones	x	x	x	x
-plug earphone	x			
bone conduction	x	x	x	
narrow band masking	x	x	x	
external input	x			
tone switching				
-tone presentation	x	x	x	x
-tone interruption	x			x
-pulsed tone	x			
reference tone				
-variable presentation	x	x		
-syncronized presentation	x			
subject response system	x	x	x	x
electrical signal output	x	x		
signal display	x	x		
speechless communication				
-from operator to subject	x	x		
-from subject to operator	x			
audible monitoring component for signal				
-pure tone and noise	x			
-external input	x			

An audiometer of Type 3 should yield minimum and maximum sound levels according to Table 3.2.

Table 3.3 summarizes the maximum allowable THD at the output of the vibrator and the headphones [17].

Table 3.4 is the list of cut-off frequencies for the applied narrow band noise sounds according to TS 9595-1. Noise bands that are thought to be produced must be selected between the frequencies of signals which are generated by the pure tone section. Therefore bands between the 250 Hz and 8000 Hz are accepted as center frequencies of narrow band noise sounds. It is decided that 16000 Hz narrow band noise not

Table 3.2
Minimum and maximum hearing levels [17].

Frequency(Hz)	Hearing Levels (dB)						
	Type 1		Type 2		Type 3		Type 4
	Air	Bone	Air	Bone	Air	Bone	Air
125	70	-	60	-	-	-	-
250	90	45	80	45	70	35	70
500	120	60	110	60	100	50	70
750	120	60	-	-	-	-	-
1000	120	70	110	70	100	60	70
1500	120	70	110	70	-	-	-
2000	120	70	110	70	100	60	70
3000	120	70	110	70	100	60	70
4000	120	60	110	60	100	50	70
6000	110	50	100	-	90	-	70
8000	100	-	90	-	80	-	-

Table 3.3
Maximum allowable THD as percentage of vibrating force or sound pressure [17].

Frequency Region(Hz)	Air Conduction			Bone Conduction		
	125-250	315-400	500-5000	250-400	500-800	1000-4000
Hearing Level(dB)	75	90	110	20	50	60
TDH (%)	2,5	2,5	2,5	5,5	5,5	5,5

necessary at present. Nonetheless implemented device is capable of producing that band of narrow band noise.

"Frequency deviation should be no higher than %2 for Type 3 and Type 4 audiometer." This is another important statement written in EN 60645-1 (TS9595-1). And the implemented device will be designed based on this limitation both in pure tone generation and central frequency assigning in narrow band noise generation.

Another important requirement of the standard is the adjustability of the output

Table 3.4
Upper and lower frequency roll-off edges for band pass filters [17].

Central Frequency (Hz)	Lower cut-off Frequency (Hz)		Upper cut-off Frequency (Hz)	
	Lowest	Highest	Lowest	Highest
125	105	111	140	149
250	210	223	281	297
500	420	445	561	595
750	631	668	842	892
1000	841	891	1120	1190
1500	1260	1340	1680	1780
2000	1680	1780	2240	2380
3000	2520	2670	3370	3570
4000	3360	3560	4490	4760
6000	5050	5350	6730	7140
8000	6730	7130	8980	9510
Based on the IEC 61260.				

of the audiometer with a 5 dB SPL step size. This is also accomplished in our design. Furthermore as an option to the device, it can achieve 1 dB amplitude resolution at the output. Other specifications are listed below:

- Masking level attenuation : Variable intensity with a 5 dB step,
- Voltage requirements : 220 volts AC, 50 Hz
- Frequency modulated output distortion : 2%
- Frequency modulating signal accuracy : 2%

4. DESIGN AND INSTRUMENTATION

Audiometers are among the state of the art health devices. Doubtlessly, they accommodate very strict THD and frequency accuracy standards. Thus, in some parts of the audiometers precision amplifiers are widely used instead of general purpose operational amplifiers which rise up the production cost. Human ear has very large dynamic range ("The dynamic range of ear exceeds 130 dB, but at the extremes of this range, the ear is either straining to hear or is in pain [18]."). The circuitry in the audiometers is such a circuitry that it can yield signals fitting the aforementioned dynamic range. Hence many op amps have limitations when such high dynamic range is considered, it could be alleged that designing the circuitry supplying this specification to the audiometer is the most challenging part of it. The output of a PTA is obtained through jack sockets. Different sorts of tones are available at the outputs, like pure tone, noise etc. Some converters like headphones, mastoid vibrators are inserted to the jack sockets in order to convert the electrical energy to sound energy or mechanical vibrations. At this point, there suddenly appears another important issue; driving these converters. Assume that audiometer settings are tuned to output sine wave having 100 dB SPL at 16000 Hz. Output amplifier must render this signal ready in jack sockets. However if power amps are investigated, designer will easily comprehend that power amplifiers can output high current but their dynamic range flatness is sustained for only few hundreds of hertz even less than this. However employing different combination of elements makes it easy to attain such dynamic ranges in higher frequencies. On the other hand if commercially available op-amps are considered current feedback amplifiers can be the most suitable choice for implementing the power stages.

There are more than one apparatus connected to the output of the audiometers. They are different intrinsic characteristics so they have different voltage and current needs. However, they can be calibrated for the same hearing level at the output. This job is processed in "Equalization" module (see Figure 1.1). In this section signal amplitude is set to a suitable absolute amplitude level thanks to the available circuitry like resistor divider networks. In this work this part is implemented using Digitally Controllable Potentiometers. Signal levels regulated discretely utilizing such component. Despite the fact that analog potentiometers offer more precise regulation, they are not preferred due to the scarce of the ports of the microcontroller (One potentiometer is required for every single type of signal.). Figure 4.1 shows the block diagram

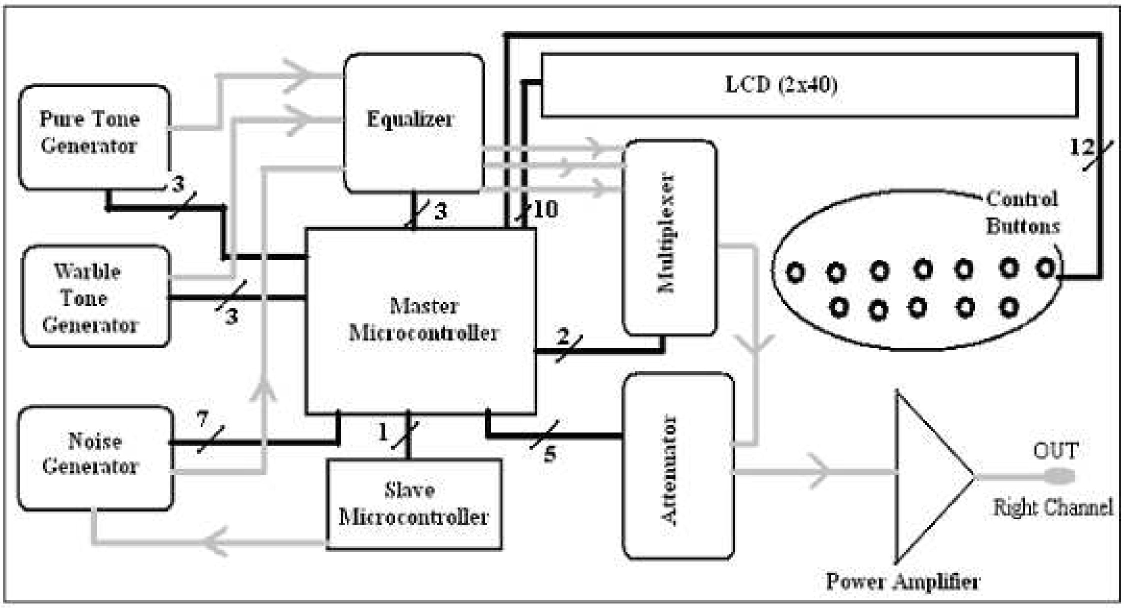

Figure 4.1 Block diagram of Right channel (Pale lines demonstrate signal propagation direction, however bold ones demonstrate control bus or line).

representation of one of the channels. The bold lines indicate the connections between the microcontroller and pertaining functional unit. They include more than one line to control the unit. However grey lines (pale) shows the output of the unit and it is a single line. As realized there are two microcontroller established as master slave structure. Slave one is dedicated to the noise block to feed the pertaining IC with its clock output. Microcontroller's finite loop necessity and main program's firmware structure cause indispensably doubling the number of the microcontrollers. This point will be clarified later. Other functional units are LCD (2 x 40) and buttons thought as separate units to achieve the user interface for this prototype. LCD lets the user know the state of the device. Microcontroller locates the cursor to the dedicated place and writes data at the cursor position. State of the device is shown in the lower line in LCD screen, however warnings about the functioning of the device is shown in the upper line for about five seconds to supply additional time to user to read it correctly and cleared later (see Figure 4.2). Additionally buttons changing the gain and frequency of the pertaining channel and toggle active channels as well. For the sake of simplicity

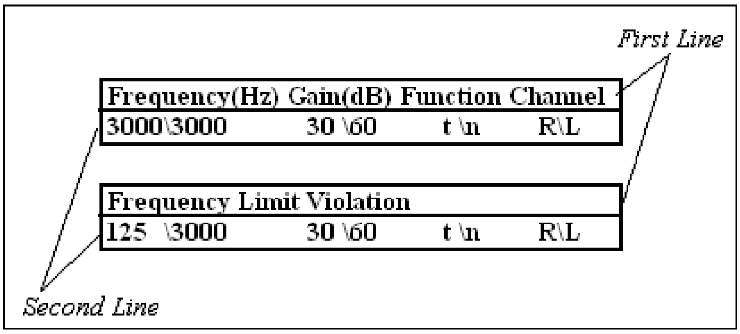

Figure 4.2 Representative showing of two different states of LCD screen.

and cost alleviation, in this prototype, stop button (tone interrupter) which must be in the hand of the patient, placed near the other buttons. Power amplifier is thought to be as a voltage follower. Its output is as the same as the attenuator's output however it provides the required current to drive the headphones and mastoid vibrator.

4.1 Pure Tone Generator (PCB-1)

Pure tone generation is one of the most important tasks when the system accuracy is considered. It is strongly desired that the frequency deviation and Total Harmonic Distortion (THD) of the generated waveform should be as low as possible. The first circuit tried in this thesis work is explained in Section 4.1.1. Yet it is not preferred as the pure tone generator circuit. Because; in Figure 4.3 the variable resistor R1 is replaced 16 x 1 multiplexer and related resistors in each channel to obtain desired frequencies from 125 Hz to 16000 Hz. And it is thought to activate the suitable channel by the four address line of multiplexer. Unfortunately, when the circuit constructed over this architecture, channels of the multiplexer are not separated from each other because of the low "off" resistance (22 Mohms) of the related channel. This brings about unbearable crosstalk between the channels. Thus output is observed as summation of band of frequencies between 125 Hz and 16000 Hz. As an alternate approach Direct Digital Synthesis (DDS) method is tried. As the name implies it is not an analog method of generating sine wave. It needs more complicated firmware procedure on the microcontroller. Yet, as will be seen later it is more accurate and stable when compared with the former one.

4.1.1 Sine Approximation: Logarithmic Shaping

One of the pure tone generators designed for the audiometer in this thesis is shown in Figure 4.3. Generator is carried out using 2 different topologies. One of them is responsible from triangle wave generation and the other one is responsible from logarithmic shaping of the triangle wave to obtain sine wave. There are two reasons to prefer this circuit as a pure tone generator. Firstly this circuit is capable of generating frequencies in the range of 125 Hz to 16000 Hz only by changing one resistor, R1. Secondly circuit generates sine waves with very low distortion of about 2%. With special concern this ratio can be reduced to 1% [19].

Triangle wave is generated thanks to the loop formed by LF356 and LM311 (see Figure 4.3). LF356 works as constant current driver and LM311 as comparator. Furthermore diode bridge with Zener diode forms a voltage clamp (V_{clp}) at voltage;

$$+V_{clp} = +(V_z + 2V_{don}), \tag{4.1}$$

and

$$-V_{clp} = +(V_z + 2V_{don}), \tag{4.2}$$

where V_{don} and V_z diode on voltage and Zener voltage respectively. LM311 functions as a Schmidt trigger and R_b and R_a constitute its switching thresholds as [19]:

$$+V_{sw} = +(R_b/R_a)V_{clp}, \tag{4.3}$$

morover,

$$-V_{sw} = (-R_b/R_a)V_{clp}, \tag{4.4}$$

When the circuit is powered on, suppose that the LM311's output swings to $-V_{clp}$. Then the capacitor will be charged by the current;

$$I = -(V_{clp} + 2V_{don})/R_1. \tag{4.5}$$

Hence LF356 inverts the polarity, the voltage at V_x , equals:

$$V_x = \frac{-(-V_{clp} + 2V_{don})}{RC} t, V_x = 0. \tag{4.6}$$

It is seen that the output goes positive when the output of the LM311 negative (note that $2V_{don} < V_{clp}$).Similarly when the output of the LM311 is at $+V_{clp}$, a negative ramp is seen at the output of the LF356,$V_x = 0$. To find the circuit frequency, steps

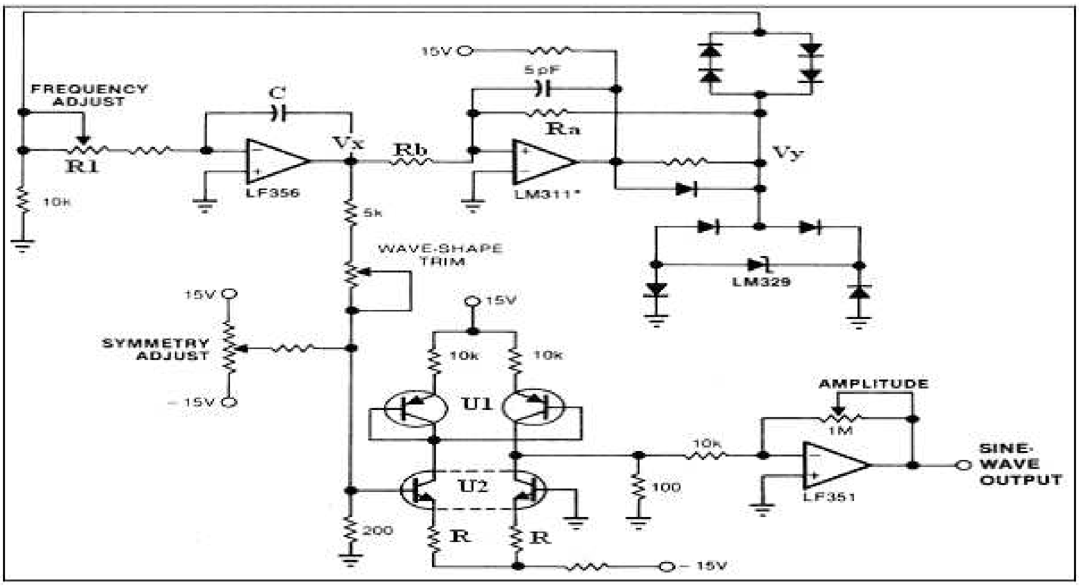

Figure 4.3 Pure tone generator.

can be written as follows:V_x ramps up from $-V_{sw}$ to $+V_{sw}$ at about $T/2$ seconds and;

$$C\Delta V = I\Delta t, \tag{4.7}$$

so we have;

$$C(2V_{sw}) = \frac{V_{clp}}{R_1} T/2. \tag{4.8}$$

Using the fact that

$$\frac{V_{clp}}{V_{sw}} = \frac{R_a}{R_b}, \tag{4.9}$$

and expressing $f = 1/T$ we get:

$$f = \frac{(R_a/R_b)}{4R_1C}. \tag{4.10}$$

LF356 itself has low input bias current and reasonable slew rate specifications therefore suitable choice for that stage. Apart from this comparator must be high speed which is enough for this application.

Triangular output is divided using 5k wave shape trimpot and 200 ohms. Because experimentally it is inspected that larger triangular waves cause more distortion at the output of the log shaper. Log shaper is formed by the two IC namely; U1 and U2 and it is the most expensive part of the circuit. Both U1 an U2 are matched pairs, that's why they are the most expensive. U2 is connected to the circuit in the form of difference amplifier. And it has logarithmic voltage transfer function .Thus it rounds the tips and verges of the triangular wave that is it gradually converts the triangular wave to the sine wave [19]. This is the non-linear effect of the U1. U2 is connected as a current mirror to convert the U1's collector current difference to the single ended current. Then this current is divided to match it to the input of the LF351 and then converted to the output voltage by the feedback resistor. U1 is selected as MAT01 and U2 as SSM2220. In addition amplitude-adjust resistor and frequency-adjust resistor R1 is multiplexed to obtain 12 different frequencies at different amplitude. In fact amplitude adjustment has another important role. When a sine is generated in a certain frequency than it is equalized to 80 dB equivalent of it at the output of the TDH39 (headphone) Then attenuator only elevate it 100 dB or decrease it -10 dB to attain full dynamic range clarified in the standards. Trimpots are sensitive to allow precise adjustment. All the resistor values, obtained frequencies and THDs will be given in the Results section.

4.1.2 Direct Digital Synthesis (DDS)

Sinusoid functions are not linear regarding their amplitudes but they are linear in the sense of their phase values. The phase of a sinusoid function changes between 0 and 2π; and it repeats itself in every step of 2π. AD9833 which is selected from the DDS product portfolio of Analog Devices benefits this principle. This chip gets a reference sinusoid signal, therefore basic phase equation becomes the starting point as:

$$\omega = \Delta Phase/\Delta t = 2\pi f, \tag{4.11}$$

where, ω is the angular velocity and f is the frequency.Solving for f and substituting the reference clock frequency (f_{MCLK}) for the reference period:

$$f = \frac{\Delta Phase \times f_{MCLK}}{2\pi} \tag{4.12}$$

Because of the fact that 2π radians are coded with 28 bits registers the Equation 4.12 can be written as:

$$f = \frac{\Delta Phase \times f_{MCLK}}{2^{28}}. \tag{4.13}$$

A functional unit, Numerically Controlled Oscillator (see Figure 4.4), inside the chip produces signals according to the Equation 4.13. $\Delta Phase$ is also coded by the help of FREQ0 or FREQ1 which are 28-bit registers to select the desired frequency. Signal is applied to SIN ROM and DAC after Numerically controlled Oscillator. In SIN

ROM phase information (signal) is replaced with the sine wave amplitude information based on its phase signal. This is nothing but a lookup table that converts the digital information to amplitude information. Finally DAC gets the words from the SIN ROM and converts them to corresponding analog voltages [20]. But before writing the related

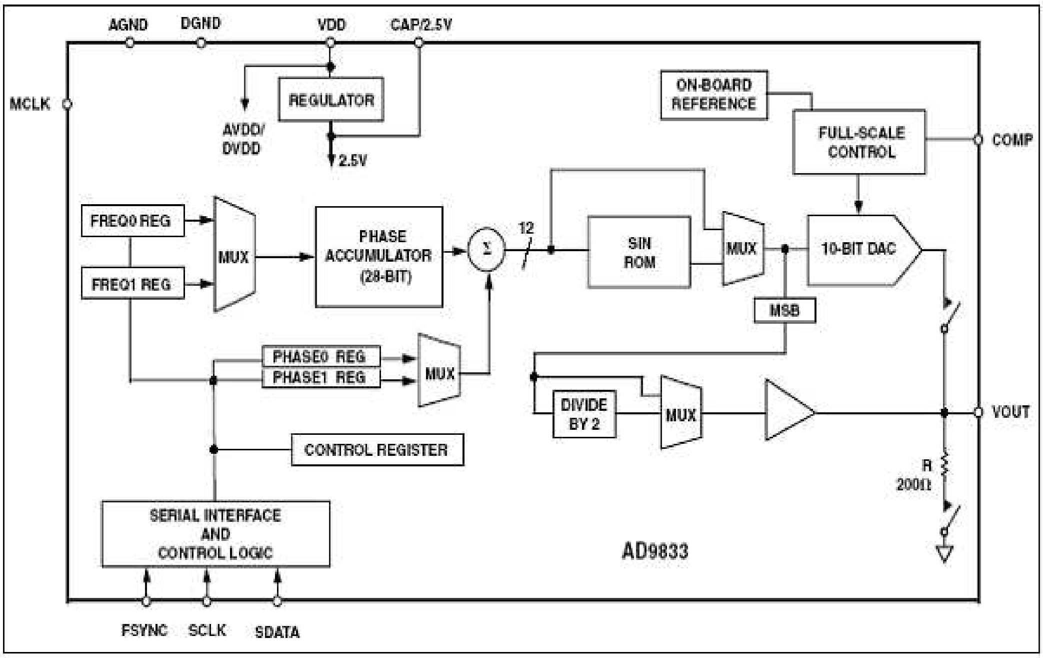

Figure 4.4 Internal structure of AD9833.

registers, control register of the chip must be programmed. It contains a set of bits which determines which frequency register will be used between two, or whether the phase modulation will be performed, or what kind of signal (triangle, sine, etc.) will be ready at the output. The circuit set up for this work is depicted in Figure 4.5 [20]. The circuit composed of some decoupling capacitor combination which eliminates supply noise, a reference capacitor which sets the internal reference voltage (pin 1), 25 MHz oscillator and its ground, supply connections, and octal buffer so as to remove the noise due to transitions at logic states at interface pins. All the frequencies produced by the chip have 608 mV peak to peak amplitude and a considerable offset voltage at about 300-400 mV. The **measured** total harmonic distortions for the targeted frequencies are tabulated in Table 4.1.

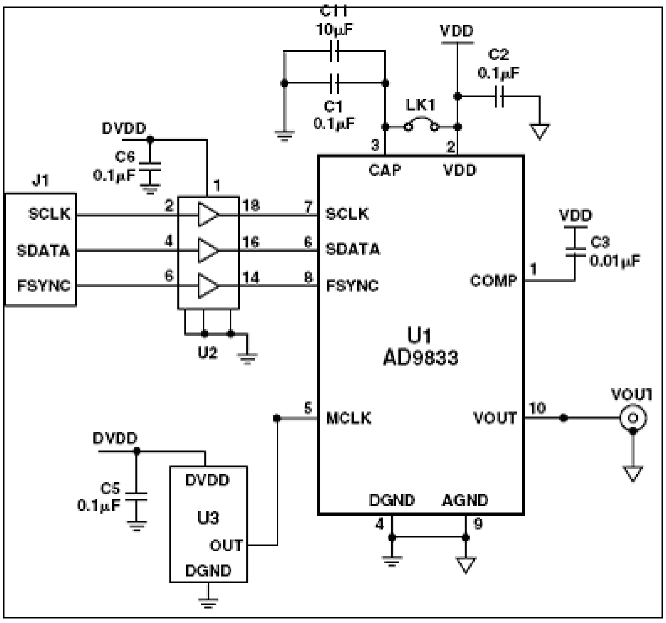

Figure 4.5 Schematic of Sine Wave Generator.

4.2 Noise Generator and Noise Filter (Main Board)

Noise generator is the other functional block in the designed audiometer. First of all it is beneficial to say noise is used to mask the ear that is not being tested. In hearing testing (air conduction), contribution to the tested ear by the non-tested ear is low but it is extremely high in bone conduction testing. Namely, non-tested ear, undesirably helps the tested ear in hearing because of the sound vibrations conducted by the skull. Therefore subject under examination may have good hearing results though, s-he has insufficient hearing capability. Audiologists always advice commencing the process from the better ear. When test is completed for the better one; routine is repeated again for badly hearing ear. If the HL difference between the two is higher than 20 dB process is repeated for badly hearing ear again with the masking on the better ear. Also it is recommended that the applied noise amplitude must surpass the pure tone or warble or pulsed tone applied to the tested ear. The word "noise" in this study refers to narrow band white noise. However in some audiometers wide band noise is used. The proposed design, can easily be modified to implement wide band noise. White noise is obtained in the frequency range of 0 to 100 kHz reasonably flat. White noise generating module is analog. A reverse biased Zener diode (LM336) is the source

Table 4.1
Total Harmonic Distortion of all the frequencies.

Frequency(Hz)	**Total Harmonic Distortion (%)**	**Frequency(Hz)**	**Total Harmonic Distortion (%)**
125	0.092	**2000**	0.081
250	0.090	**3000**	0.079
500	0.088	**4000**	0.078
750	0.083	**6000**	0.078
1000	0.084	**8000**	0.065
1500	0.082	**16000**	0.023

of the noise. The Zener noise is amplified using LM386 known as low current power

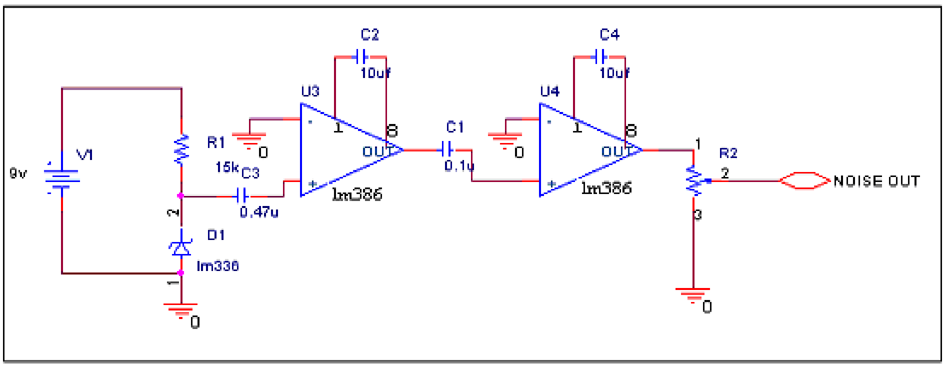

Figure 4.6 White noise generator.

operational amplifier. It has a voltage gain of 200 as its maximum. The noise at the output of the generator was 8 volts peak to peak. Therefore the original Zener voltage is: $8/(200{\times}200) = 50\mu V$.If more current is injected to the cathode of the Zener then the noise current will increase and so the noise voltage at the output. C2 and C4 in Figure 4.6 are advised as 10 μF to reach to the maximum gain of the LM386. If the obtained noise voltage is higher than the desired voltage level, then it can be simply decreased by a trimpot connected to the output of the second LM386. Another widely used method in the commercially available method implemented in the audiometer's noise generator, pseudorandom pattern generation by using linear feedback shift registers, LFSR, Figure 4.7. Using Digital to Analog Converter (DAC) one can obtain white noise. So clock rate is the dominant parameter of the circuit in determining the noise

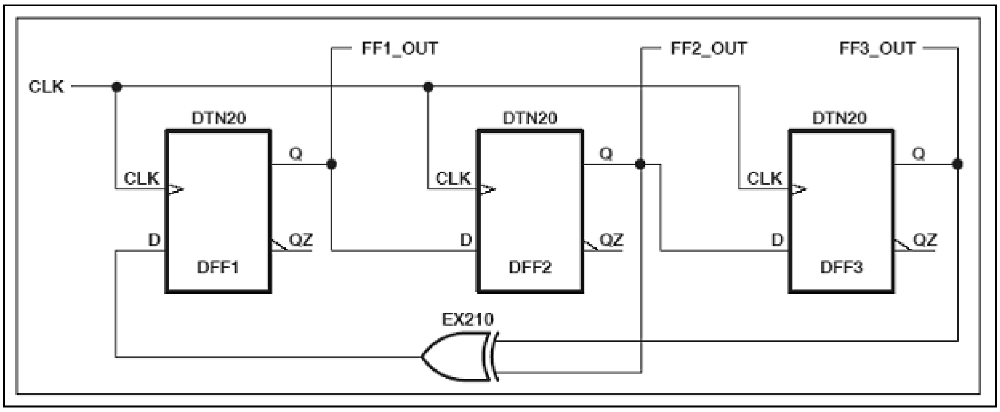

Figure 4.7 Linear feedback shift register.

behaviour. The user would create an LFSR that is much bigger than three bits to get a large number of pseudorandom patterns, before the patterns are repeated. However, there are some practical restrictions to the length of the LFSR. A 32-bit maximal-length LFSR would create over 4 billion patterns, at a 16-MHz clock rate, would take almost 5 minutes to generate the whole pattern set [21]. Thus the white noise generator used in this study is preferred over the LFSR based designs, in terms of efficiency and cost. Noise generator provides wide band noise to a 4^{th} order bandpass filter module as seen in Figure 4.8. It is a switched capacitor filter manufactured by Maxim (MAX261). Due to the fact that it is switched device sampling is performed. On account of the fact that this 4^{th} order filter makes sampling, frequency components over half of the sampling rate cause aliasing. This is an obstructive condition which is eliminated by the higher order anti aliasing filter. Minimum clock frequency utilized by the 4^{th} order noise filter is 32767 Hz. 32767 Hz is divided by 2, for some reason internally. Thus the lowest sampling rate is 16384 Hz and frequencies above the 8192 Hz will cause aliasing [22]. For example 9000 Hz component of the noise cause reflection back to the 808 Hz $(9000 - 8192)$. But with the roll-off frequency of the anti-aliasing filter set at 8300 Hz aliasing will be totally eliminated. Furthermore if this interference is disregarded and roll-off frequency set to; let's say 16250; then the circuit will produce a narrow band noise in the verge of 16000 Hz. Additionally anti-aliasing filter used in this design is an 8^{th} order filter, hence it eliminates important part of the interfering element. Fourth order noise filter is controlled by the microcontroller. Because of the fact that there are many different bands of noise centered at the different frequencies, it will be

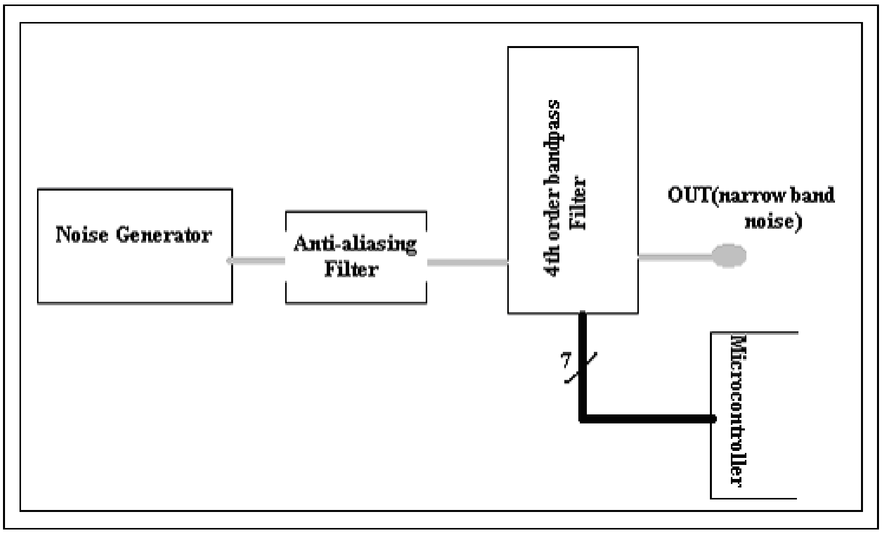

Figure 4.8 Noise block.

wise to show only the calculations belonging to the one of them, let's say band centered around the 2000 Hz. First of all from EN60645-1, upper and lower cutoff frequencies, as well as attenuation characteristics of the filter are determined. They are: Lower cutoff frequency : between 1680 Hz and 1780 Hz (1730 Hz is assigned) Upper cutoff frequency : between 2240 Hz and 2380 Hz (2300 Hz is assigned) Attenuation band requirement : at least 12dB/Octave Frequency accuracy : 2% Then second microprocessor provides clock frequency as 257 kHz. If f_{clk}/f_0 is selected as 128.81 f_0 is obtained as 1995.2 Hz [18]. Q(quality factor) is defined as:

$$Q = \frac{f_0}{(f_u - f_l)}. \tag{4.14}$$

so Q is obtained as: $Q = 2000/(2300 - 1730) \cong 3.51$, where; f_0 is center frequency, f_{clk} is clock frequency, f_l is lower cut-off frequency, f_u is upper cut-off frequency. Both of the 2^{nd} order sections of the 4^{th} order filter (MAX261) are used as cascade to ensure the desired attenuation. Also they operate at Mode 1. Thus Q of each section is assessed as its gain at the same time [22]. On the other hand because of its cascaded structure, Q is multiplied by 1.55, therefore Q should be selected as $3.51/1.55 = 2.26$. Thus 2.29 is accepted as a Q value presented in the table (see [22] for the mentioned Q value table). Then binary codes of f_{clk}/f_0 ratio and Q are sent to the filter. Also note that during the experiments it is noticed that the microcontroller MSP430FG439 isn't able to provide correct clock frequency. This made the design a little bit sophisticated.

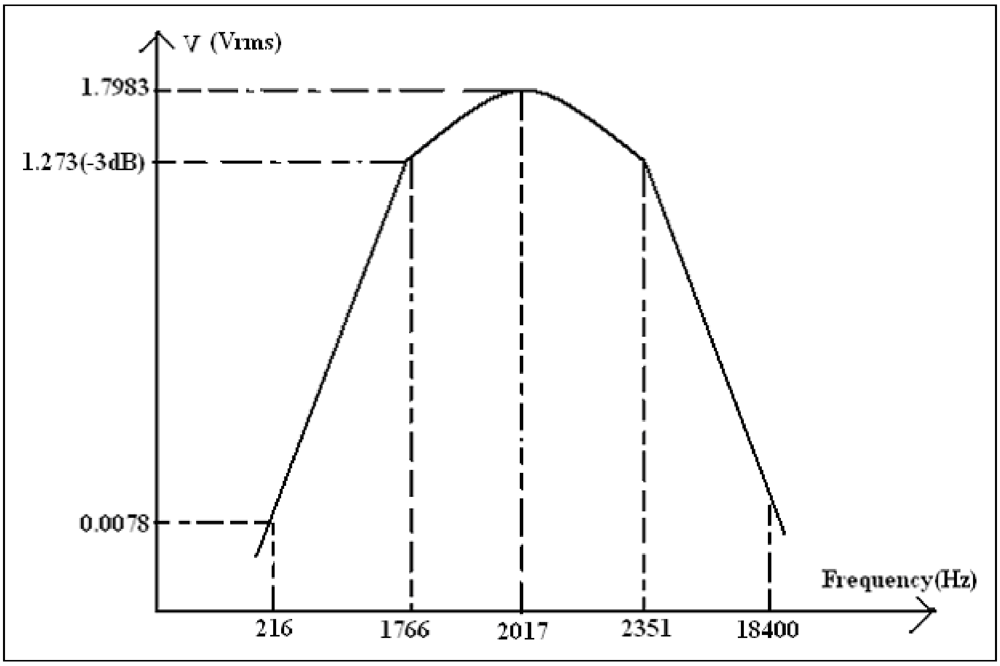

Figure 4.9 Obtained results for 2000 Hz using 4^{th} order switched capacitor filter from Maxim.

Besides designer must consider the amplitudes at the end of the noise block, because the 4th order noise filter multiply the amplitude of the signal with 3.51. For 2000 Hz results are shown in Figure 4.9. Besides 216 Hz and 18400 Hz in Figure 4.9 are 3 octaves smaller forms of 1766 Hz and 3 octaves bigger form of 2351 Hz, respectively. Notice that frequency deviation is not larger than 2% of center frequency and attenuation is greater than 36 dB after 3 octaves far from the lower and upper cutoff frequencies.

4.3 Warble Tone Generator

Warble tones are frequency modulated waves. They are used to prevent formation of standing wave patterns which are confronted when using single-frequency test tones. Audiometric tests are carried out in nearly full absorbent booths but they are not perfect. These imperfections cause echoing from the walls of the booth. Owing to the fact that wall of the booth may be assigned as fixed or closed end, incident wave is reflected back as 180 degrees out of phase. Therefore standing nodes and sweeping antinodes are induced in every step of $\lambda/2$ and $\lambda/4$ respectively. This has the effect of producing ambiguous results when attempting to measure the energy gradient of the directly radiated signal. On the other hand warble tone contains band of frequencies instead of one kind of frequency. So every single frequency inside this band changes the locations of nodes and antinodes and produces a substantially or completely homogeneous energy gradient that is measurable and predictable [23, 25]. The circuit implementing Frequency Modulation (FM) is given in Figure 4.10. In this circuit all the frequencies generated in pure tone generator, generated again by LM565 as square or triangle wave. As a matter of fact LM565 is Phase Locked Loop (PLL) IC however, it contains a Voltage Controlled Oscillator (VCO) inside and this structure is taken advantage of to reproduce all the frequencies from 125 Hz up to 16000 Hz as carrier waves. This is achieved scaling the DC voltage applied to the 7th pin of the IC. Every

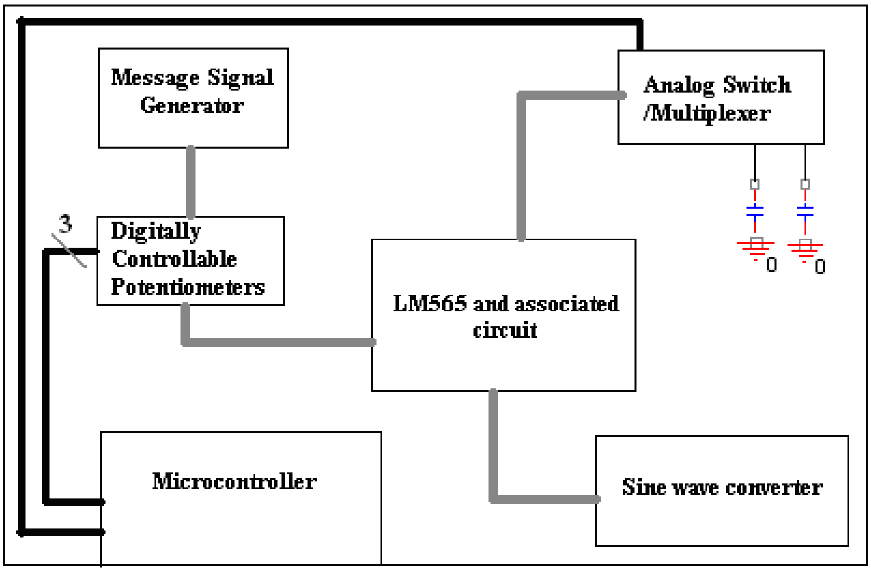

Figure 4.10 Block diagram of Warble Tone Generator.

single level in this pin stands for a periodic signal at unique frequency. Besides to expand the range of generated frequencies two capacitor switching is required. Therefore as switching circuit, a multiplexer is used. Seventh pin of the microcontroller implements another important job. Thanks to the message signal superposed over its DC component it changes its own output frequency which is available at pin 4 in reverse manner i.e. when the message signal amplitude increasing the frequency of the output wave is decreasing and vice versa.

Figure 4.11 shows more detailed circuit of the important parts of the warble tone generator. According to this representation output frequency (before message signal application, only carrier is available at the output) can be formulated as:

$$f = \frac{2.4(V_{cc} - V_s)}{R_1 C_c V_{cc}}, \tag{4.15}$$

where; V_{cc} is the supply voltage, V_s is the voltage on pin 7 and C_c is the effective capacitance on pin 9.

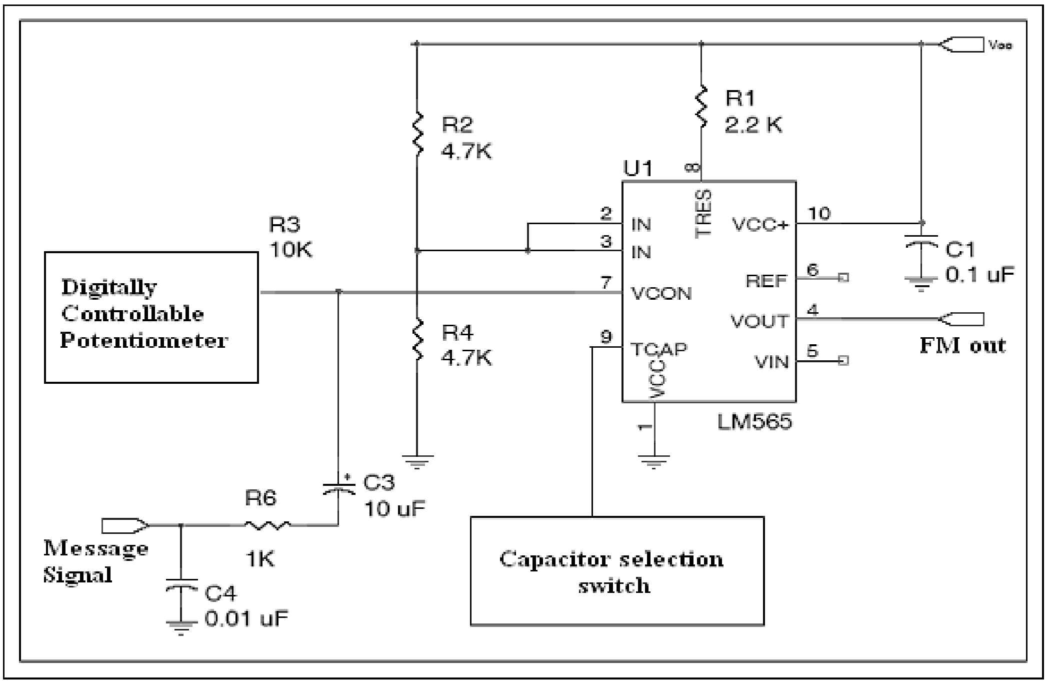

Figure 4.11 Detailed depiction of LM565 and its connections.

Furthermore, effective capacitance on pin 9 and sets the modulator sensitivity. It is assigned as 500 $Hz/Volts$ in this circuit. Besides message signal frequency is selected as 10 Hz, and maximum frequency deviation is selected as 10%. Incidentally, message signal is generated by ICL8038 which is obsolete today. But few of them are available and it simplifies the overall circuit because it needs only few capacitors and resistors to output the desired frequency. On the other using AD9833 in place of ICL8038 gives more and more excellent results but microcontroller doesn't have enough number of pins for AD9833 to program it via its digital interface.

4.4 Equalizer

Equalizer is one of the requisite module, when the design topology is regarded. Equalizer implements equalization of the output levels of each generator and for each specific signal. This level is assigned as 80 dB for pure tone signal and 60 dB for noise at the start up. For example pure tone generator outputs 608 mV peak to peak sine wave in every frequency. However in 125 Hz 799.2 mV (absolute level) corresponds to 80 dB SPL but in 250 Hz only 198 mV rises up the sound pressure level to 80 dB SPL. Therefore produced level (608mV) must be equalized to its corresponding dB SPL absolute level. This is also required for noise signal and warble tones so this section includes dedicated channels for each kind of signal. Initially incoming signal is applied to the high pass filter to eliminate the offset of the signal. Because this offset may elevate the signal and cause clipping at he upper or lower side of the signals. Furthermore noise signal is low pass filtered to weaken the effect of clock feedthrough and sampling frequency as a component inside the noise. Secondly suitable offset is added to the signals to match them to the input range of the Digitally Controllable Potentiometer (DCP). (Note that DCP is powered by 5 Volts only, not symmetric voltages i.e. +5, -5 Volts.) AD8403 (Analog Devices) is used as DCP. It has got 4 channels namely 4 potentiometers each of which is 10 Kohms. Each potentiometer has 256 tabs. IC has 10 bit control register whose most significant two bits are utilized for selecting the channel and rest is used for setting the tabs in predefined potentiometers. Every potentiometer inside the IC has 3 endings, named as A, B and W and the least significant 8 bits set the resistance between B and W terminal as:

$$R_{WB}(D) = \frac{D}{256} \times R_{AB} + R_W, \tag{4.16}$$

where; D is data that stores the tap information; R_{WB}, R_{AB}, R_W are resistances between W and B terminals, A and B terminals and wiper resistance, respectively [24]. So, thirdly one of the op-amp is configured as non-inverting operational amplifier but its feedback path is totally constructed by one potentiometer of the DCP. Doing so, it is intended to amplify signal if desired output is higher than the incoming signal; on the other hand unless the desired output signal is higher than the incoming signal, then

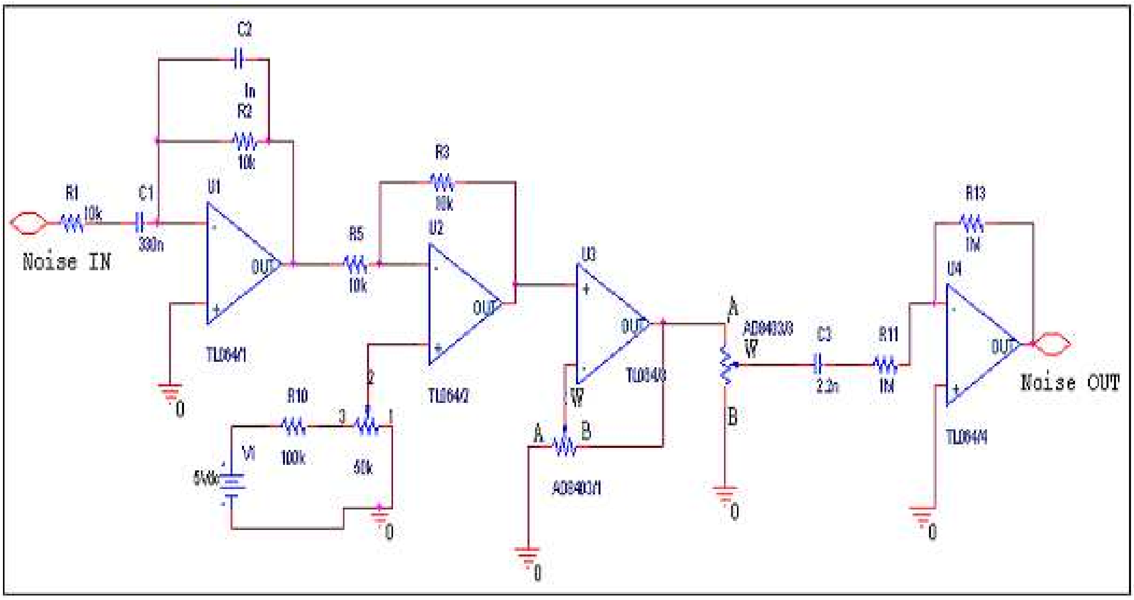

Figure 4.12 Schematic of Equalizer section (Single channel).

non-inverting amplifier exhibits buffer behavior. Next the signal is applied to another channel of the DCP for a more precise setting. Hence two channel of the DCP is used for one type of a signal. As final step signal is directed to a high pass filter to remove its dc part. For one channel of signal (e.g. noise) four op amps are used. In order to alleviate the tangle of op-amps one IC that contains four separate op-amps (Quad) is used that is named as TL064. AD8403 has serial port interface, therefore only three lines are enough to program it. These pins have standard names as Serial Data Input (SDI), Clock (CLK), and Chip Select (CS). Furthermore AD8403 has Serial Data Output (SDO) pin which renders programming with only there lines for more than one chip. Between AD8403 and microcontroller one pin of port 3 of microcontroller is assigned as clock line. Writing the desired word to the chip is implemented as activating chip first by CS line, loading the SDI pin and shifting the CLK line to high level sequentially.

4.5 Attenuator (Main Board)

Attenuator is almost the most important part of the circuit. Hearing system has a large dynamic range so that the audiometers must also comply with. This makes the audiometer design a challenge. When signal level control required it can be performed using digital potentiometers. Digital audiometers can fix the signals to the user defined values, free of mechanical problems which are great concern in analog potentiometers. Nevertheless, digital audiometers suffer from lack of large dynamic range for audiometric applications. Their easy-to-use feature rapidly becomes senseless because of this. There is a second configuration exploited to build the attenuator stage in audiometers. It is nothing but a little bit complicated voltage divider network with resistors. Assume that for a headphone connected to the output of the audiometer, 80 dB SPL corresponds to 0.5 Vrms signal amplitude level in 1 kHz. This signal can be diminished or enhanced by a series of resistor networks. Resistors have such a placement that they provide for example 5 dB steps in each switch action. Such a switch can easily be assigned as rotary encoder. In each step, it contacts one output tab of the resistor network to provide the desired signal amplitude, relatively. In this design, Burr Brown's PGA2310 is used. It has digital gain control. It has two channels handling capacity, hence named as stereo but not facilitated in this manner. There are three crucial reasons preferring this IC. It has 120 dB dynamic range which matches our expectation exceedingly (Our range stretches from -10 dB to 100 dB.). It has 0.5 dB gain control (step size) which makes these chips alluring for audiometric application. Incidentally, as a part of standards, 5 dB step size is definitely required. It has very low THD (0.0004 %). On the other hand internal noise is somewhat higher than the expected (13 $\mu Vrms$). Since input signal shrinks to the order of few microvolts, internal noise of the chip may possibly deteriorate the quality of the signal extremely. But experiments showed no such indication. However noise addition will be more appreciable when the signals applied to the headphones by the power amplifier. Moreover, this contributes minimally to noise at the instant of digital gain setting by the controller, thanks to the zero-crossing-enable function. Due to this function new gain is set at only the zero crossing of the signal. Partially some tests are performed with a sine wave having 2 Volts peak to peak amplitude. Signal state is observed in very low gains at -60 dB and -80 dB. Figure 4.13 summarizes the connections of the attenuator for one channel. The attenuator has got two channels which are dedicated

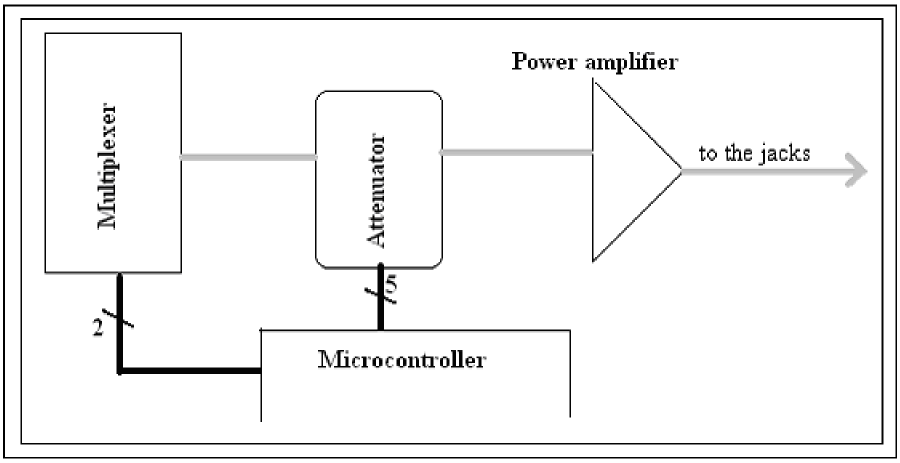

Figure 4.13 Attenuator and its connections.

for stereo volume control. These two channels are not used to utilize the stereo feature of the IC, instead these channels simply allocated for right and left channel. Multiplexer in the Figure 4.13 is precision multiplexer used to select, pure tone, warble tone or noise signals. For three different line selection 2 control lines are enough therefore a 4 to 1 multiplexer is used. Attenuator is also controlled by (master) microcontroller. Five lines from microcontroller are connected to the attenuator as SDI, $\overline{ZCEN}$, $\overline{MUTE}$, CLK and $\overline{CS}$. SDI is data line. Gain information of each channel is sent to the attenuator chip as 8 bits stream and right channel is conceded first. $\overline{CS}$ is another line between the microcontroller and PGA2310 and activates the chip when it is active low. ZCEN is a line to achieve noise free gain transition inside the chip. Additionally $\overline{MUTE}$ is a pin of the chip which is binds the output of the PGA2310 to the analog ground of the system. CLK line is utilized by the chip to acquire the data in the SDI pin in its rising edge. So in each programming of the gain, for both channels, sixteen clock cycles sequentially are sent to the PGA2310. The chip is deactivated using $\overline{CS}$ pin. Output of the attenuator is directed to the power amplifier to drive the headphones and the mastoid vibrator.

4.6 Power Amplifier (Main Board)

After the attenuator, the rest of the blocks that the signal directed must possess at least the same dynamic range with the attenuator as well as their owns feature pertaining to the functions. But we have only one functional block, after the attenuator. It is power amplifier. Some measurements carried out about the current requirement of the headphone for pure tone and noise conditions. 125 Hz, 16000Hz (for pure tone); 8000 Hz (for noise) sine waves and noise bands necessitate the highest current drives. Besides, in case of sine wave application to the bone vibrator (in 60 dB) highest current requirement is observed at 16000 Hz. While headphone driving is not a problem; current requirement of the mastoid is a problem. Its current requirements can be as high as 400 mA based on the measured data. Therefore Current Feedback Amplifier (CFA-AD815) from Analog Devices is suitable for this stage which can supply 500 mA for one channel over a load of 10 ohms. It is a dual channel IC. In higher current requirements input may be connected differentially. As expected its THD is a little bit high (66 dB) but not unacceptable. The favored feature of this amplifier is its wide dynamic range even at higher frequencies. CFAs, in contrast to traditional voltage feedback amplifiers (VFA), have no dominant pole capacitor and therefore can operate much more closely to their maximum frequency at higher gain. Stated another way, the gain/bandwidth dependence of VFA has been broken. It enables the system to drive the apparatus even in higher gain settings, in higher frequencies with a flat open loop gain (It is nearly flat up to 20 kHz, see Figure 4.14). The other worthy to say advantage of the CFAs is they have no slew rate limitations. But this design does not deal with high speed signals, this feature as an advantage does not appeal us. Conversely the disadvantages of CFAs are high offset voltages which urge the designer to insert the offset adjustment circuitry before the CFA.

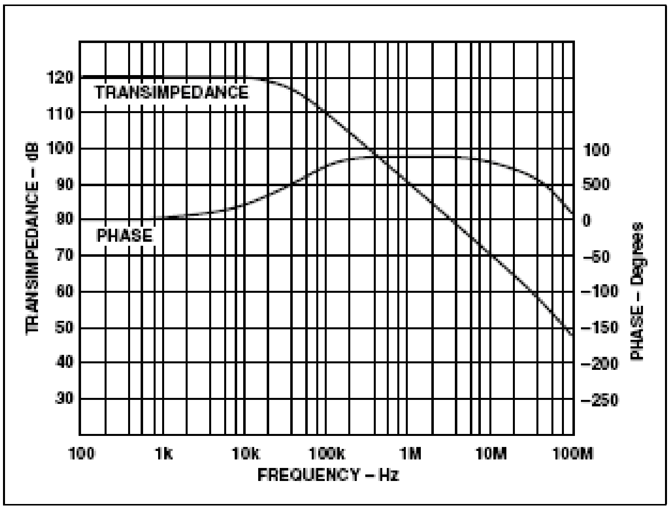

Figure 4.14 Open loop characteristic of the power amplifier Note that the amplifier is a current feedback amplifier, therefore open loop gain is represented as transimpedence.

4.7 Control Panel (PCB-3)

Interface between the device and user is implemented by this block. It is separated from the other modules and implemented on a board peculiar to it. So that this block completely free from the others and if an encasing is thought than it will easily put to a visually convenient place. This will ease the control also, because the board includes control buttons on it. This board contains master microcontroller, control buttons, LCD, switch debouncing circuitry and connectors providing communication between microcontroller, main board and control panel themselves. Microcontroller in this board is attached to the solder side to prevent any blockage of view and control. LCD is just over the buttons and buttons are grouped based on their tasks.

LCD is 2 x 40 characters wide, manufactured by Yes Optoelectronics (YMS 402) and controlled with 16 lines (Sixteen lines are standard for all character LCDs). It is powered from 5 Volts. The R/W line is connected to the ground to maintain the LCD always in write mode. However, only 2 control lines are used to manage the LCD. Enable pin (E) and register select (RS) pins are used to activate the LCD and to inform it whether the information sent is data or instruction. Additionally eight bit lines are connected between LCD and microcontroller to accomplish parallel communication. According to the state of the RS pin transferred bits are assessed as data or instruction. Also note that if instruction command is selected by the RS pin, programmer must be careful about the execution duration of each instruction which constitutes no problem when data mode is selected by the RS pin.

Buttons are placed to select the tone type as noise, pure tone, and warble; to active or deactivate right or left channels and decrease/increase the gain or frequency of the test signal. Hence the firmware is interrupt-driven. Each of these buttons trigger an interrupt event to execute a certain job. Furthermore there are "apply" and "stop" buttons to apply and cease the adjusted signal to the subject. "apply" button should be located to the controller side. This is as expected. But tone interrupting button (stop button) is also located on the control panel. This is not a suitable format but this version of the device is only a prototype at the moment. Buttons are simple push buttons and they produce bounce at the time of pressing which causes the wrong interpretation by the microcontroller. Sometimes this can cause skipping more then one tap for frequency or amplitude adjustment. One of the debouncer circuit named as "RC debouncer" is a useful circuitry to diminish the unwanted signal transitions during the pressing. RC debouncer is comprised of a RC network and a Schmiddt trigger. The key to get bounce free transitions when switch is closed, is to scale the time constant of the circuit below the upper threshold of the Schmiddt trigger. The

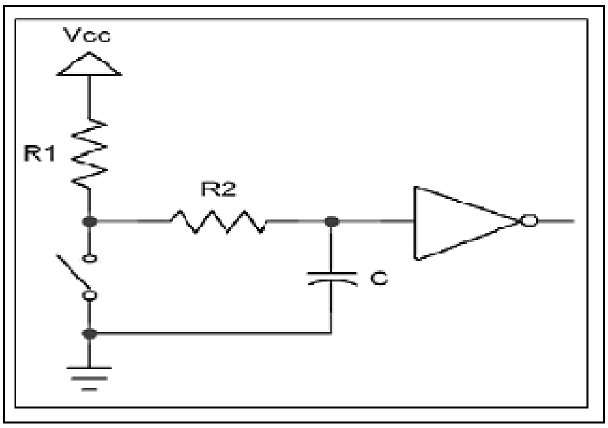

Figure 4.15 Debouncer [27].

voltage of the capacitor can be formulated as follows (see Figure 4.15):

$$V_{cap} = V_{cc} \exp(-t/R_2C), \qquad (4.17)$$

where; V_{cap} is the capacitor voltage.Then R_2 can be obtained as:

$$R_2 = \frac{-t}{C \ln(\frac{V_{cap}}{V_{cc}})} \tag{4.18}$$

Assuming bounce time as 20 ms, in the worst case, V_{cc} volts, V_{cap} as upper threshold of 1.7 volts R_2 is obtained as 18.5 Kohms. 18 Kohms is used instead. Likewise when charging equations are written, R_1can be extracted from these equations. The same values are utilized to calculate the R_1, except V_{cap} which must be set as 0.9 volt as lower threshold. Then R_1 is obtained as 100 Kohms; however 82 Kohms is used instead. So even higher switching structure yields bounce free transitions. In Figure

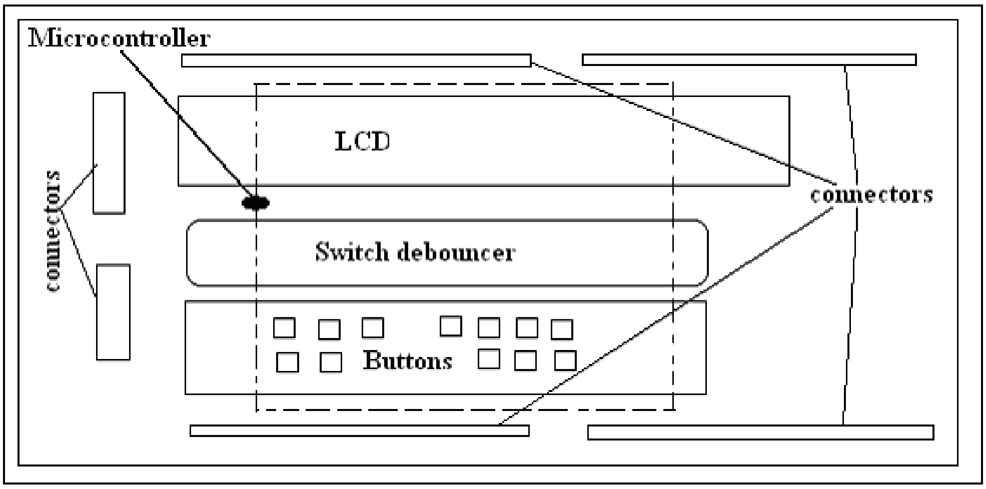

Figure 4.16 Control panel.

4.16 four of the connectors located at the bottom of the board are allocated to the microcontroller. The microcontroller (master) that is mounted to the solder side of the control panel (board) is depicted with dashed lines in Figure 4.16. Actually not only the microcontroller but its evaluation board (MSP-TS430PN80 Rev1.0) whose size is 72 x 72 mm is also completely mounted on to the control panel (board). However it is not smart; high coast of the IC socket of the microprocessor impels the over all design in this way. Moreover there are two connectors at the left side of the Control Panel. The one soldered to the upmost provides the communication between LCD and microcontroller and the other provides the communication between main board and control panel (board). The former one is 2 x 8 and later one is 2 x 13.

4.8 Microcontrollers and Supply Voltages

Microcontrollers are also crucial parts of the device. This audiometer design is totally based on the Texas Instrument's MSP430FG439 microcontroller. It has 80 pins surface mount (SO) package. But it is mounted on MSP-TS430PN80 Rev1.0 evaluation board, supplied by TI. Two MSP-TS430PN80 Rev1.0 boards are totally mounted to the designed PC boards. MSP430FG439 has many functional modules like DAC, ADC, timers, operational amplifiers, LCD control units, serial communication ports etc. It has six input/output ports. Each pin of the port can be programmed as an input, output or a special function pin. Apart form these, port 1 and port 2 accept interrupts from the external components facilitating an interrupt driven programs. Another block of the microcontroller is its phase locked loop (FLL) clock module control block. This part controls the system clock. It has 3 different clock signals so-called MCLK (Master clock) SMCLK (Sub-main clock) and ACLK (Auxiliary clock). ACLK needs a special crystal. When the crystal is connected to the evaluation board it provides 32767 Hz clock pulse. Furthermore the clock frequency can be divided by 2, 4, 8 by the internal registers of the PLL module. SMCLK is another clock line but again it needs a special crystal connection to function properly. MCLK is the system clock. It operates as certain integer multiplies of the 32767 when watch crystal for ACLK is functional. In this work, clock utilization is particularly important. Sixty fifth pin of MSP430FG439 is allocated to the MCLK that is appointed for noise masking module of the audiometer. Owing to the fact that a switched capacitor filter is used and it requires clock pulses to perform sampling, MCLK output is directly connected to clock input of the switched capacitor filter. One of the microcontrollers is the master microcontroller which manages all the functional blocks as well as the slave microcontroller. The slave microcontroller that responsible from the generation of clock pulses for the switched capacitor filter. Communication between these two microcontrollers is provided using the USART interface of each microcontroller in UART mode. It will also be explanatory that there is no reason to call the controllers as master and slave when the mediating communication type concerned by the controllers but this renders easy when somebody wants to discriminate them. Merely the transmit pin (UTX0) is enabled in master; however in slave, receive pin (URX0) is enabled. Hence communication is unidirectional. Baud rate is set to 2400 baud and ACLK is used as UART module clock. Receiver continuously waits for an interrupt. As an interrupt is captured, then it sets its MCLK to the desired frequency and reverts back to its infinite loop. Note also that master disables its UART module after the transmission of the pertaining control character. This impedes unintentional transmissions. A problem encountered during the design is the accuracy of the microcontroller while generating clock pulses. SCFI0 is one of the register of the FLL module. When it is programmed using FLLD8 (symbolic representation of dividing 8) and FLLD4 (symbolic representation of dividing 4) system clock deviates from its actual value for about 5 kHz. The problem is overcome thanks to the wide range of clock programming coefficients of the switched capacitor filter.

Symmetric voltages, used in the system are; $\pm 15V$, $\pm 5V$, $\pm 9V$. They are obtained using the LM317 (for positive voltages) and LM337 (for negative voltages). Basic circuitry recommended in the datasheet is set up. Besides $\pm 15V$ and $\pm 5V$ volts are supplied from the power supplies (GE Dual tracking power supplies) in the laboratory. Heatsinks are used for the LM317 and LM337 pair which generates $\pm 9V$, according to the calculations declared in the "heatsink requirement" section of the related components.

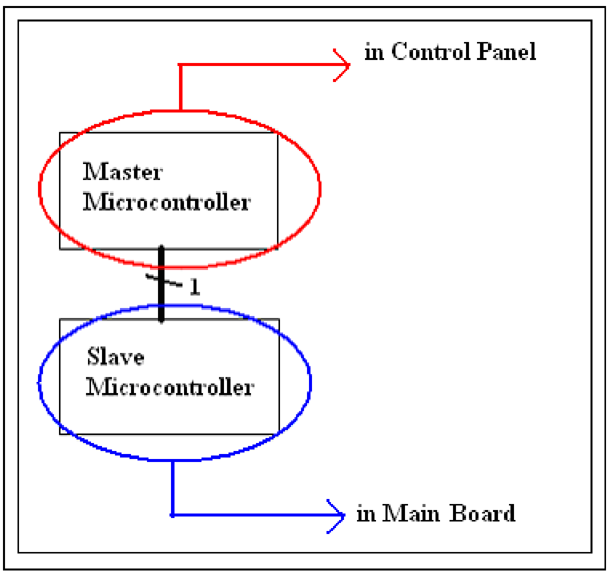

Figure 4.17 Location and connections of Microcontrollers.

5. SOFTWARE/FIRMWARE

5.1 Program for Printed Circuit Board (PCB) Production

The system comprises three different circuit boards. First board is a pure tone sine wave generator. Sometimes it is called PCB-1 as well, in this document. It is a 80 x 160 mm, dual layer PCB. Unfortunately PCB-1 is discarded because of a design error and an alternative one is designed and produced via a more classical method (etching and exposing ultraviolet light). This new version of PC board contains a sine wave generator with DDS method (see Section 4.1.2). There is also a second board which is produced via classical method, called Equalizer. Yet it is discarded and its breadboard version is adapted to the overall circuit. Second board is Main Board. Sometimes it is called PCB-2 as well, in this document. Noise source and filters, signal selector multiplexers, attenuator, power amplifier and jack sockets are placed on PCB-2. It is also dual layer and both layers are partially copper covered. The region where power amplifier located particularly is not covered with copper. It is recommended in power amplifier's datasheet that, coverage and pins of amplifier must be separated from each other for about 5 mm. Hence settings are valid for entire board, this causes formation of many ground islands connected to the nowhere. Therefore locus of power amplifier is kept uncovered. Main board is 160 x 160 mm. Third board is the control panel. Sometimes it is called PCB-3 in the document. It is a dual layer design. It is fully copper covered and this coverage is assigned as digital ground. It accommodates LCD, interrupt buttons, master microcontroller with its specific PCB. It is 100 x 160 mm in dimension.

All these there boards are worked out by Eagle 5.0 professional edition. Other versions of Eagle are impotent of implementing the boards in terms of dimension. Eagle is preferred against the others like Proteus, Orcad. Because it is very easy to learn. Apart from this, almost none of the IC packages used in this design are available in any of the PCB design tools. Designing these packages as a library file is very cumbersome in other programs. Nearly 10 packages have been designed as library files during this project. Moreover, many component libraries are downloaded from the official site of Eagle. Another useful feature of the Eagle is that, it has an autorouter whose settings are declared by the user and it connects the airwires in a whole PC board in a very short time, even for remarkably large board. After that user may do some arrangements on the PC board to obey some widely accepted rules to optimize the efficiency and its cost. Namely, autorouter saves your time woks as a second person on a PC board implementation in a project. But one handicap of Eagle is, it does not offer an opportunity to run a functional simulation to see the results before the PCB fabrication.

5.2 Program on The Master Microcontroller

Programs running on the microcontrollers (on both microcontrollers) are initially compiled using IAR Workbench v4.11. This is free edition of IAR with 8 kBytes of code memory limitation. The final program is about 24 kBytes long which was compiled with. That's why IAR is omitted for a while and Code Composer Essentials v3.0 (Professional Edition) IAR is used to write programs, to compile and run them on the microcontroller for peripheral units which have digital control interface with microcontroller at the beginning of the project. Formed code segments are saved as ".cpp", however they exhibit the characteristic features of C codes. Namely they are not object oriented. The main program actually formed by the patches. These patches are nothing but the previously written code segments. They are only merged to constitute an interrupt-driven C program. It is worthy to say that in some sites of the code there are function calls inside the other function. Branching increases up to 4th order leading inefficient stack use but this does not bring forth any problem.

Figure 5.2 summarizes the logic of the main program running on the master microcontroller. The program is founded on a variable whose type is "struct" in C programming language. It is defined as in Figure 5.1: Next, two variables (r, l) defined as type is "struct alan". Program begins with setting all peripheral units like LCD,

```
struct alan
        {
        short int freqind;
        short int gainind;
        short int t;
        short int n;
        short int w;
        short int active;
        }
```

Figure 5.1 Snapshot of 'struct alan'.

attenuator as well as microcontroller's ports. Initial values are reflected on to the screen (LCD). Interrupts are activated during the initialization. Then program enters in an infinite loop and continuously monitors the pins which buttons are connected in order to catch an interrupt. Assume that there is an interrupt from the button of right channel "r.w" variable. That means activate short tone by settings its value to 1. As another example providing that there is an interrupt from the gain buttons, firstly it is determined whether it is a gain increase or decrease button. If it is a gain increase button, then "gainind" which is the member of the variable r or l is incremented by 1. So it is value, presumably may increase from 7 to 8 means gain is increased from 55 dB to 60 dB. The induced changes are shown on the LCD. If the interrupt source is "apply" or "stop" button, no changes are done on the members of the variables "r" or "l". If "apply" button is pressed, (according to the settings done before); peripheral units are configured and lastly the attenuator is activated. For the activation "mute" pin of the PGA2310 is deactivated. If interrupt is triggered by the "stop" button, then the attenuator output is simply connected to the ground. When these operations are completed, microcontroller program flow is directed to the site where the infinite loop starts. This flow maintains itself during the "on" state of the device.

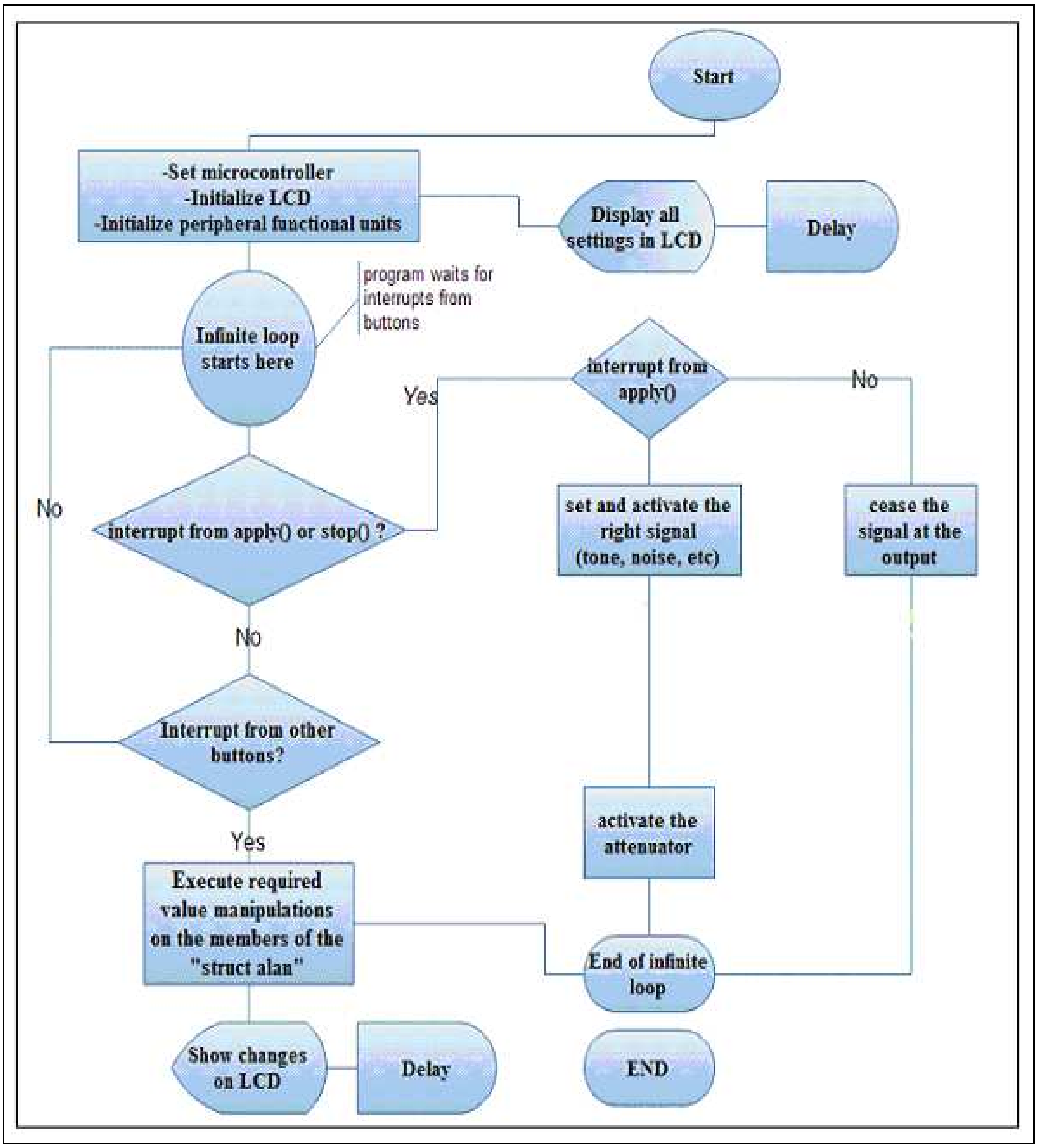

Figure 5.2 Flow chart of the main program.

6. CALIBRATION

Similar to the other audiometers, this audiometer requires two types of calibration. One of them is frequency calibration and the other one is amplitude calibration. Frequency calibration can be applicable for only pure tone generator and warble tone generator. On account of the fact that, warble tone generator is not ready for this work, frequency calibration is the prime interest just for the pure tone generator. Digital codes, sent to the AD9833, are kept in a two dimensional array which is called as "hold" in the program of the master microcontroller. The AD9833 is programmed at either .1 or .3 Hz above the desired frequency. For instance if 125 Hz is desired to be generated, the device is set to generate 125.3 Hz. This means a frequency error of 0.24% which is better than the requirements of the standard (2%). Deviations in other frequencies are better than this as percentage error. .1 or .3 Hz deviations are selected by the programmer. A few amount of frequency deviation is suggested if the desired frequency is an integer multiply of the clock source in the design tool, ADI simDDS [20] of the AD9833 with a caution as; "The desired output frequency should not have an integer or near integer relationship to the internal clock frequency." (Clock frequency is 25 MHz in this design.). In this tool it is asserted that violation of this suggestion may lead to increased spur levels and excessive jitter. Therefore error rate in frequency accuracy is firmware programmable. Amplitude calibration with achieved as firmware driven adjustments again: Every single frequency is set to 80 dB SPL as the opening level. The initial parameters are kept in a two dimensional array that is described as in Figure 6.1 Each column in this array is formed by a method of trial and error measurements and each of them represents the 80 dB SPL of pure sine wave and noise respectively except the last column which represents the 40 dB SPL (opening level for bone conduction) of pure sine wave for mastoid drive. Some frequencies exhibits different characteristics and need high voltage levels. Difference between the voltage levels makes this design challenging because every level in this array corresponding single frequency, is attenuated or amplified by the same amount without saturation or loss. Amplitude calibration can be achieved by manipulating these levels for 80 dB.

```
volatile float reference[][3]= {
                                61.0,   0,      295.0,     //125
                                57.2,   26.2,   49.0,      //250
                                54.3,   24.0,   8.0,       //500
                                52.2,   23.0,   14.5,      //750
                                54.0,   22.0,   12.65,     //1000
                                51.0,   21.0,   8.4,       //1500
                                59.0,   20.08,  20.0,      //2000
                                34.0,   14.3,   40.8,      //3000
                                55.5,   17.9,   32.43,     //4000
                                21.0,   13.91,  327.0,     //6000
                                180.0,  31.3,   377.0,     //8000
                                271.0,  0,      1060.3,    //16000
                        };
```

Figure 6.1 Snapshot of 'Amplitude Calibration Matrix'

At other levels calibration is achieved by a dedicated array for tone and noise outputs. For example in pure sine wave 85 dB SPL level is reached by 5.5 dB augmentation. However, theoretically 85 dB level must be reached only 5 dB augmentation over 80 dB.

Calibration for center frequency in every band of noise (there are 10 different bands of noise) could be achieved by reprogramming the MAX261 via the function "setnoisesettings()". It has special sections for every band of frequencies, but note that center frequency determination is done by applying single frequency to the MAX261 and inspecting center and 3 dB frequencies for every band. Shortly center frequency calibration for noise is not needed because best settings are obtained by trial and error method after many attempts.

7. RESULTS

An audiometer applied on a patient through:

a) Headphones (air conduction),

b) Mastoid Vibrator (bone conduction),

c) Free field.

The proposed device aims at two widely used media "a" and "b". For the designed audiometer to be considered as a successful design we must verify the set frequencies, amplitudes within the specifications of standard TS 9595-1. The verification system is also a well established calibration and test system for audiometric applications. The calibration/test system comprises:

(i) Headphone transducer (TDH39): Electrical signals which are available at the jack sockets are transmitted to the TDH39, headphone, via its jack and cable connection. It is used in air conduction.

(ii) Mastoid vibrator transducer: It converts electrical energy to mechanical energy. It is placed just over the mastoid bone right behind the ear. It is connected to the jack socket in the Main Board via its jack. It is used in bone conduction.

(iii) Artificial Ear (Coupler Set) is composed of a very sensitive membrane microphone, a steel body, and a squeezing arm for TDH39. It is used as artificial ear and the headphone is mounted on it. Artificial Ear is manufactured by GRAS.

(iv) Artificial Mastoid is a cylindrical steel body which imitates the mastoid bone. It has a squeezing unit for the mastoid vibrator and is connected to the Noise meter by a very sensitive and fragile cable. It is manufactured by Brüel & Kjaer.

(v) Pre amplifier: This unit undertakes the amplification of signals in air conduction. It feeds the Noise meter for signal analysis. It is manufactured by GRAS.

(vi) Noise meter (Measuring Amplifier): it has an analog display that shows the relative amplitude of an electrical signal. It has a very high dynamic range (10 to 170 dB SPL). The Noise meter also has built-in weighting filters according to the type of input. Moreover it has a sensitivity knob which is set according to the type of measurement. Its frequency range is between 5 Hz and 500 kHz. It is manufactured by the NF (M 2174). Manuel calibration is possible for Noise meter.

(vii) Distortion analyzer: it is used for measuring the distortion of a sine wave. It gives the distortion either as percentage or dB. It could also be used as an oscilloscope. It is manufactured by NTI and named as Minilyzer ML1.

(viii) Calibrator: it produces 1 kHz audio sine waves at two different amplitudes (94 and 114 dB SPL). It is used for calibration of the Noise meter at the beginning of the tests/calibration.

7.1 Air Conduction

Figure 7.1 depicts the test/calibration setup for the air conduction. TDH39 is inserted to the jack sockets.

TDH39 headphone converts the electrical energy into sound energy. On the contrary GRAS Coupler Set converts sound energy into electrical signal. Resultant signals are amplified and displayed in the Measuring amplifier. THD is measured for selected frequencies and tabulated in Table 7.1. Moreover set amplitude levels for in 5 dB SPL steps are tabulated in Table 7.2 and in Table 7.3 . But in Table 7.2 and in Table 7.3 measurements which is below 75 dB SPL have "*". This means these measurements are carried out with a voltmeter (Fluke True rms 189). Actually environmental noise dominates the signal in the laboratory and this hinders Measuring amplifier (Noise

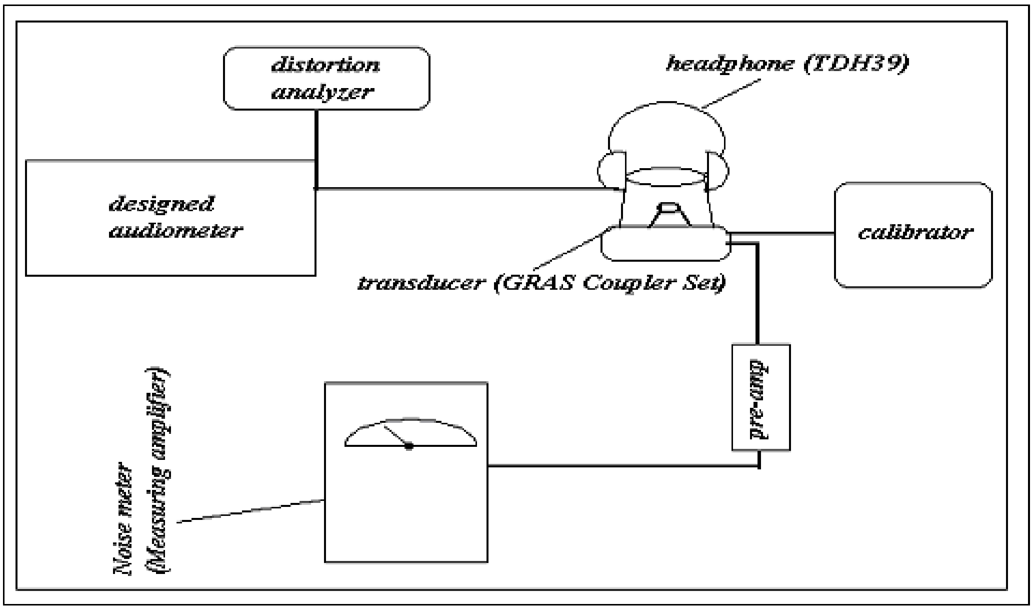

Figure 7.1 Representative showing of the air conduction calibration/test system.

Table 7.1
THD in pure sine wave application to TDH39(According to the TS 9595-1 THD must be smaller than 2.5%).

Measurement Level (dB SPL)	Frequency (Hz)	THD (%)
95	125	0.310
	250	0.338
100	500	0.312
	750	0.362
	1000	0.295
	1500	0.295
	2000	0.270
	3000	0.430
	4000	0.262
	6000	0.530
	8000	0.100
	16000	0.048

meter) to detect and show the signal in its screen correctly. Same problem is valid for noise application to the headphones. Considering this fact multimeter readings are converted to Noise meter readings as forming a linear equation.

Table 7.2
Measured signal levels for pure tone (Preceding 6 frequencies) when TDH39 is connected to device (Look at the Table 7.3 for the rest of the 6 frequencies. Signal level deviation must be smaller than ±3 dB SPL.).

Signal Level(dB SPL)	Measured Signal Level dB SPL					
	125 Hz	**250 Hz**	**500 Hz**	**750 Hz**	**1000 Hz**	**1500 Hz**
100	99.8	99.9	99.6	99.8	99.7	99.8
95	94.8	94.9	94.6	94.8	94.8	94.8
90	89.8	89.9	89.6	89.7	89.7	89.8
85	84.9	85	84.6	84.8	84.8	84.8
80	80.2	80.1	79.8	79.9	80	80
75	75.4	75.4	75.1	75.2	75.2	75.2
70	(69.35)*	(70.3)*	(70.4)*	(69.7)*	(70.48)*	(70.44)*
65	(64.33)*	(65.4)*	(65.6)*	(64.8)*	(65.8)*	(65.75)*
60	(60.1)*	(60.7)*	(61.4)*	(60.1)*	(61.35)*	(61.26)*
55	(55.6)*	(56.4)*	(57.3)*	(56.1)*	(56.5)*	(56.47)*
50	(51.1)*	(51.9)*	(52.7)*	(52.5)*	(52.8)*	(52.8)*

Table 7.3
Measured signal levels for pure tone (Following 6 frequencies) when TDH39 is connected to device (Look at the Table 7.2 for the preceding 6 frequencies. Signal level deviation must be smaller than ±3 dB SPL.).

Signal Level(dB SPL)	Measured Signal Level dB SPL					
	2000 Hz	**3000 Hz**	**4000 Hz**	**6000 Hz**	**80000 Hz**	**16000 Hz**
100	100.2	99.8	99.7	100	99.9	99.4
95	95.1	94.8	94.8	95	95	94.4
90	90.4	89.8	89.7	90	89.8	89.4
85	85.2	84.8	84.6	85	84.8	84.7
80	80.3	80	79.9	80.2	80	80
75	75.6	75.4	75.2	75.4	75.2	75.0
70	(70.96)*	(70.85)*	(70.6)*	(70.88)*	(70.1)*	(70.3)*
65	(66.37)*	(66.4)*	(65.94)*	(66.36)*	(65.3)*	(64.98)*
60	(61.93)*	(62.2)*	(61.65)*	(62.4)*	(60.4)*	(60.03)*
55	(57.28)*	(57.46)*	(56.8)*	(57.7)*	(54.52)*	(54.28)*
50	(52.69)*	(—)	(52.16)*	(—)	(48.2)*	(47.45)*
45	(—)	(—)	(—)	(—)	(44.72)*	(43.69)*
40	(—)	(—)	(—)	(—)	(41.2)*	(39.93)*

Table 7.4
Measured signal levels for noise (preceding 5 bands) when TDH39 is connected to device. Single channel is active (Look at the Table 7.5 for the rest of bands. Signal level deviation must be smaller than ±3 dB SPL.).

Signal Level (dB SPL)	Measured Signal Level dB SPL									
	250 Hz		500 Hz		750 Hz		1000 Hz		1500 Hz	
	lower	upper	lower	upper	lower	upper	lower	upper	lower	upper
	dB SPL		dB SPL		dB SPL		dB SPL		dB SPL	
100	99	100.5	99.5	100.5	99.8	100	100	100.2	99.6	100
95	95	95.8	94	95	94.6	94.8	94.4	94.8	94.6	94.8
90	89	90	89	90	89.6	90.2	89.6	90	—	89.6
85	84.5	85.5	84.4	85	84.6	85	84.8	85	84.6	84.8
80	80	81	79.8	80.5	80	80.4	—	80	80	80.4
75	—	76	75	75.6	—	75	—	75	—	75
70	(—)	(70.55)*	(—)	(71.74*)	(—)	(71.53)*	(—)	(69.7)*	(—)	(70.61)*
65	(—)	(66.34)*	(—)	(66.14)*	(—)	(67.01)*	(—)	(64.8)*	(—)	(65.77)*
60	(—)	(61.45)*	(—)	(61.4)*	(—)	(62.86)*	(—)	(59.7))*	(—)	(61.4)*
55	(—)	(56.52)*	(—)	(56.6)*	(—)	(57.3)*	(—)	(54.4)*	(—)	(56.6)*
50	(—)	(51.98)*	(—)	(52.3)*	(—)	(—)	(—)	(50.22*)	(—)	(52.34)*

This linear equation is written using two different points which are *measurable* by Noise meter (Measuring amplifer) as well as multimeter (Fluke). Multimeter's readings are appointed as abscissa and Noise meter readings are appointed as corresponding ordinate of the mentioned points. Afterwards every multimeter readings are converted to the corresponding Noise meter readings. Therefore the values below the 75 dB SPL in Table 7.2, 7.3, 7.4 and 7.5 are marked with "*".

Table 7.5

Measured signal levels for noise (following 5 bands) when TDH39 is connected.Single channel is active (Look at the Table 7.4 for the preceding bands. Signal level deviation must be smaller than ±3 dB SPL.).

Signal Level (dB SPL)	Measured Signal Level dB SPL									
	2000 Hz		3000 Hz		4000 Hz		6000 Hz		8000 Hz	
	lower	upper	lower	upper	lower	upper	lower	upper	lower	upper
	dB SPL		dB SPL		dB SPL		dB SPL		dB SPL	
100	—	99.8	99.4	99.9	—	99.8	99.5	100	—	99.8
95	—	94.5	—	94.4	—	94.7	—	94.4	—	94.7
90	89.8	90.1	89.4	90	—	89.7	—	89.6	—	89.8
85	—	84.7	—	85	—	84.8	—	84.6	—	85
80	79.6	80.2	—	80	—	80	80	80.2	80	80.2
75	75.2	76	75.5	76		75	74.6	75		75.2
70	(—)	(70.0)*	(—)	(70.0)*	(—)	(70.1)*	(—)	(69.48)*	(—)	(70.2)*
65	(—)	(65.2)*	(—)	(65.3)*	(—)	(65.3)*	(—)	(64.25)*	(—)	(65.4)*
60	(—)	(60.45)*	(—)	(60.5)*	(—)	(60.8)*	(—)	(59.54)*	(—)	(60.4)*
55	(—)	(55.6)*	(—)	(56.2)*	(—)	(56.0)*	(—)	(54.82)*	(—)	(55.01)*
50	(—)	(—)	(—)	(52.7)*	(—)	(52.3)*	(—)	(52.04)*	(—)	(49.97)*

7.2 Bone Conduction

In bone conduction testing, mastoid vibrator is connected to the jack sockets in the designed audiometer. Vibrator transduces the electrical energy into mechanical energy. On the contrary Artificial Mastoid converts mechanical energy into electrical signal and this signal is directed to the Noise meter via a very sensitive cable (see Figure 7.2). Resultant signals are amplified and displayed on the Measuring amplifier. THD is measured for every frequency as tabulated in Table 7.6. Moreover set amplitude levels for each 5 dB SPL step change of designed audiometer is tabulated in Table 7.7 and Table 7.8.

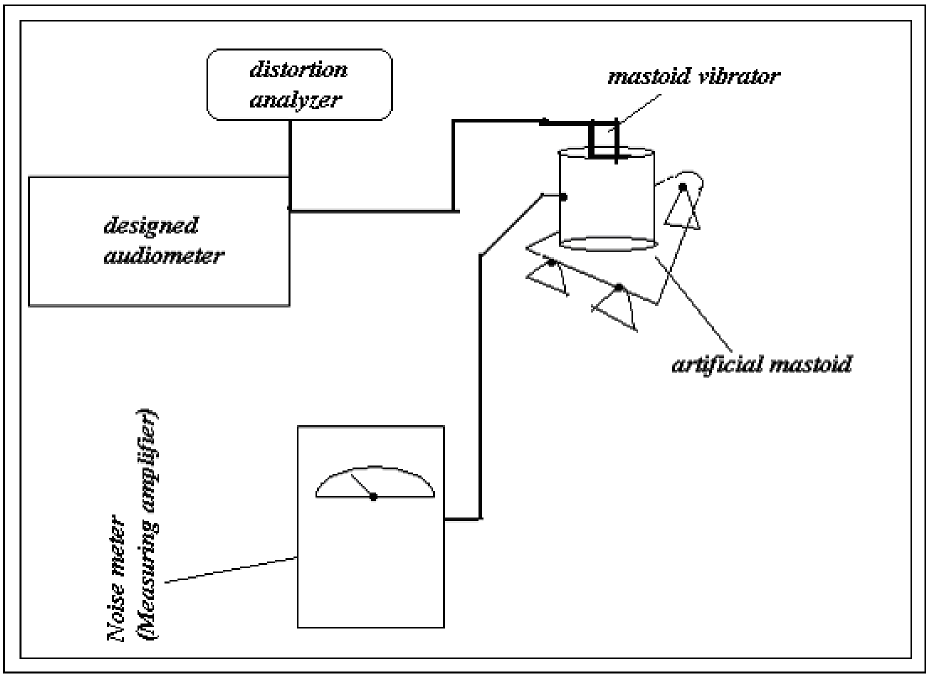

Figure 7.2 Representative showing of the bone conduction calibration/test system.

Table 7.6
THD in pure sine wave application to Mastoid vibrator (According to the TS 9595-1 THD must be smaller than 2.5%).

Measurement Level (dB SPL)	Frequency (Hz)	THD (%)
40	125	0.716
	250	2.9
60	500	4.9
	750	2.1
	1000	2.01
	1500	2.1
	2000	0.9
	3000	0.46
	4000	0.45
	6000	1.06
	8000	0.95
	16000	0.017

Table 7.7

Measured signal levels for pure tone (Preceding 6 frequencies) when Mastoid vibrator is connected to device (Look at the Table 7.8 for the rest of frequencies. Signal level deviation must be smaller than ±3 dB SPL.).

Signal Level(dB SPL)	Measured Signal Level dB SPL					
	125 Hz	**250 Hz**	**500 Hz**	**750 Hz**	**1000 Hz**	**1500 Hz**
60	59.9	60.4	59.3	60.5	59.8	61
55	55.9	56.4	55.3	56.5	55.6	56.8
50	50	50.4	49.3	50.5	49.7	50.9
45	44.9	45.3	44.3	45.5	44.8	45.8
40	39.9	40.3	39.3	40.4	39.6	40.9
35	34.8	35.2	34.2	35.5	34.6	35.7
30	29.9	30.1	29.2	30.4	29.5	30.6
25	24.7	25	24.2	25.3	24.5	25.7
20	19.8	20.5	19.3	20.4	19.6	20.8
15	13.5	14.5	13.5	14.2	14.2	13.6
10	10.2	10	10	10.4	9.5	10.5
5	—-	—-	—-	—-	—-	—-
0	—-	—-	—-	—-	—-	—-

Table 7.8

Measured signal levels for pure tone (Following 6 frequencies) when Mastoid vibrator is connected to device (Look at the Table 7.7 for the preceding frequencies. Signal level deviation must be smaller than ±3 dB SPL).

Signal Level(dB SPL)	**Measured Signal Level dB SPL**					
	2000 Hz	**3000 Hz**	**4000 Hz**	**6000 Hz**	**8000 Hz**	**16000 Hz**
60	60.5	60.1	60.2	60.2	60.4	60.2
55	56.2	56	56	56	56.4	57
50	50.4	50.1	50.2	50	50.4	50.6
45	45.3	45	45	45	45.4	45.3
40	40.2	40	40	40	40.4	40
35	35.3	35	35	35	35.3	34.9
30	30.3	29.9	30	29.9	30.3	29.8
25	25.2	25	25	24.9	25.3	24.8
20	20.2	20	20	19.9	20.3	19.8
15	13	13	13	13	13	13
10	10.7	10.2	10.2	10	10.3	10.2
5	—-	—-	—-	—-	—-	—-
0	—-	—-	—-	—-	—-	—-

8. CONCLUSIONS AND DISCUSSION ABOUT THE FOLLOWING IMPROVEMENTS

It is intended to devise basic diagnostic Type 3 pure tone audiometer in this work. Its design process divided into many threads like pure tone generator, (narrow band) noise generator, warble tone generator, multiplexers (signal selectors), attenuator and power operational amplifier. At the end 12 kinds of frequencies for air and bone conduction at 11 different levels are measurable. However there are 11 unmeasurable levels for air conduction and 3 for bone conduction. Narrow band noise is produced by device as non-test ear masker. There are 10 frequency bands which can be presented at 11 different measurable levels. However there are also 11 unmeasurable levels between 50 dB and -10 dB with the devices currently available for masker. This audiometer changes the signal level with five (dB SPL) steps with a deviation of less than $\pm$1dB for every step changes. Considering this feature it can be anticipated that device actually provides unmeasured levels with the desired step changes. Moreover, pure tone distortions for every single frequency don't exceed the 2.01% for air conduction, 4.9% for bone conduction. These limits are 2.5% for air conduction and 5.5% for bone conduction according to TS 9595-1 standard (see Table 3.3). This is another fulfilled feature which is strongly highlighted in standards.

Warble tone generator section is accidentally designed as short tone generator in main board. Short tone capability is also important but, on account of the fact that warble tone presentation becomes a "must" of commercial audiometers short tone section is omitted and warble tone generator is constructed to a board and will be adapted to the circuit later. All hardware and codes running on the master microcontroller are planned according to existence of a warble tone generator. This will make the device more competitive when compared with the contemporary ones.

All modular units are tried to determine meticulously. So as to meet the THD requirements for pure tone; seemingly well sine wave generation technique that is named as logarithmic shaping is attempted. Nevertheless, it was frustrating for pure tone generation module due to on-board calibration trimpots which are selected by the 16 x 1 multiplexers. This leads to waste of money and lots of time. But, consequently DDS method has supplied the desired frequencies with the desired accuracy and stability. DDS section is also capable to generate the frequencies necessary for High Frequency Audiometry. Moreover logarithmic shaping method that was abandoned might violate the THD limits with its 0.8% THD just at its output.

Noise (narrow band) generator module is the most original part of the design. It is the most expansive unit with its dedicated microcontroller and 4th order filter. Microcontroller is only responsible to generate the clock for the 4th order filter, and filter is meticulously programmed to abide by the determined cut-off frequencies by the standards. Nonetheless the cost of this module can be reduced as simply by replacing the development board with the microcontroller itself.

The accuracy of the system may be increased by replacing the 256 steps digital potentiometer located in the equalizer section with 1024 steps potentiometer. This will especially improve the opening level (80 dB) accuracy of the system. 256 steps digital potentiometer had been obtained before the 1024 steps potentiometer namely, timing prevented some optimizations. Also note that, attenuator has 1 dB resolution that is, it can amplify or diminish the signal with 1 dB step size. This can also be a nice feature of the system. Besides power amplifier supplies the needed current drive and needs no replacement.

Overall system costs nearly 700 TL excluding the microcontrollers which are located on and under the PCBs. Each of them is $99. Furthermore a software which costs $499 is ordered but no charge is paid because of collaboration with Texas Instruments. Despite the expenditures PCBs are not perfect. Some modules' necessity is understood in the course of time and they are integrated to the system. And some modules are transported and reconstructed on board like noise generator and noise masker. This part was problematic because of the non-conductive wire on PCB and caused the loss of one and a half month. Such errors repeated in different parts of the board.

APPENDIX A. SCHEMATICS

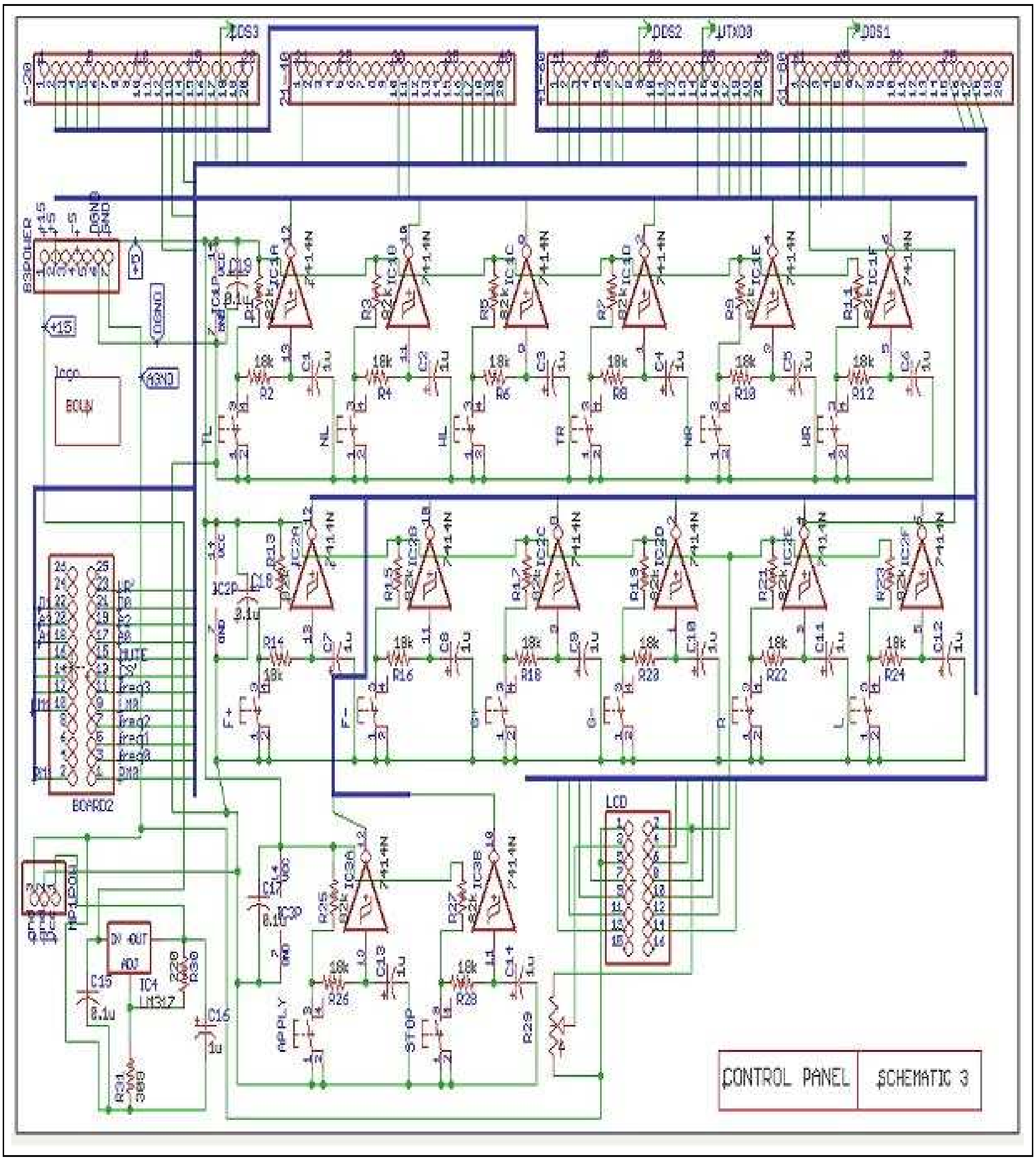

Figure A.1 Schematic of Control Panel (Drawn in Eagle 5.0.0 Schematic Editor)

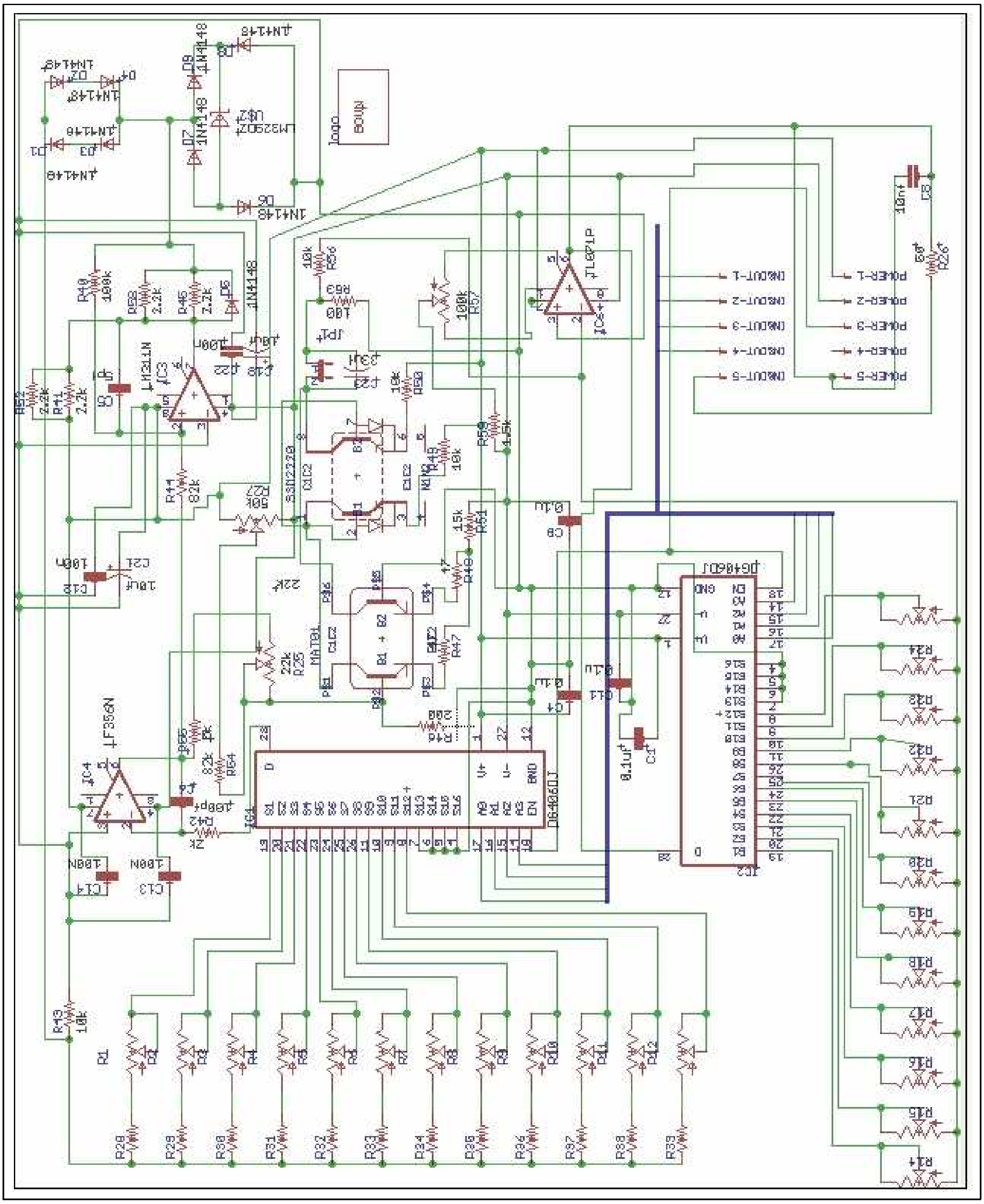

Figure A.2 Schematic of discarded Signal Generator (Drawn in Eagle 5.0.0 Schematic Editor)

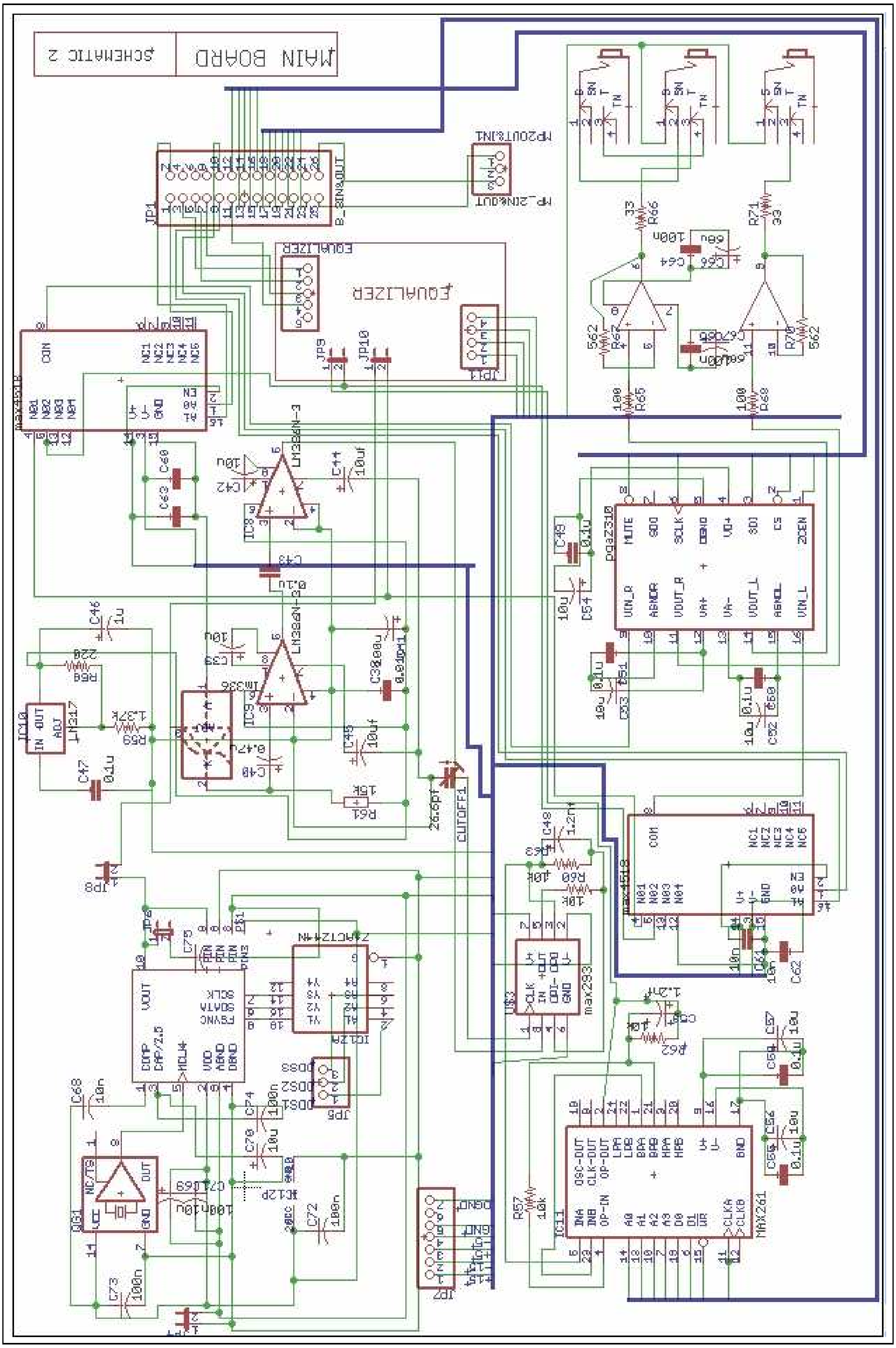

Figure A.3 Schematic of revised Main Board (Drawn in Eagle 5.0.0 Schematic Editor)

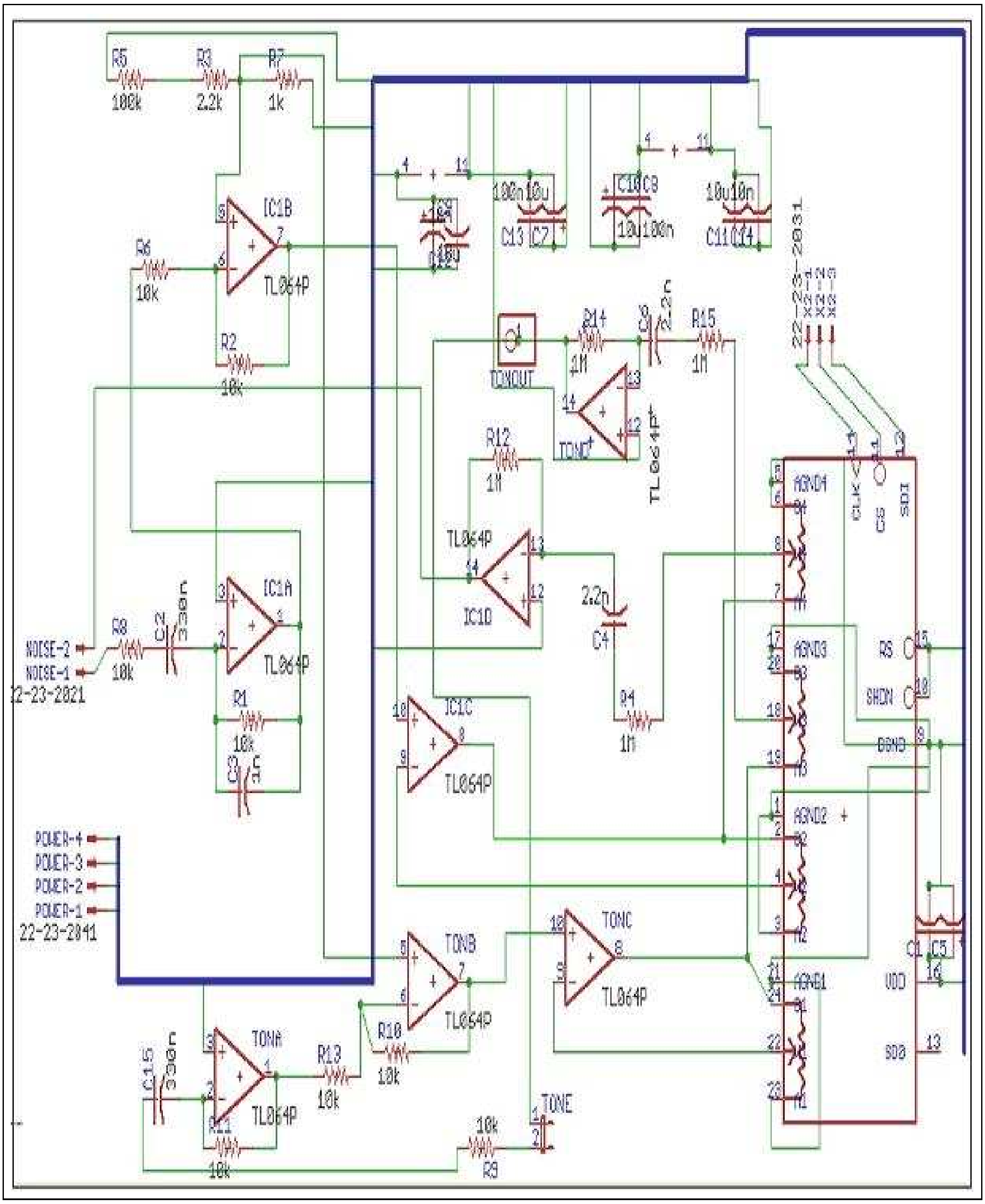

Figure A.4 Schematic of Equalizer (Drawn in Eagle 5.0.0 Schematic Editor)

APPENDIX B. PRESENTATION OF THE SYSTEM

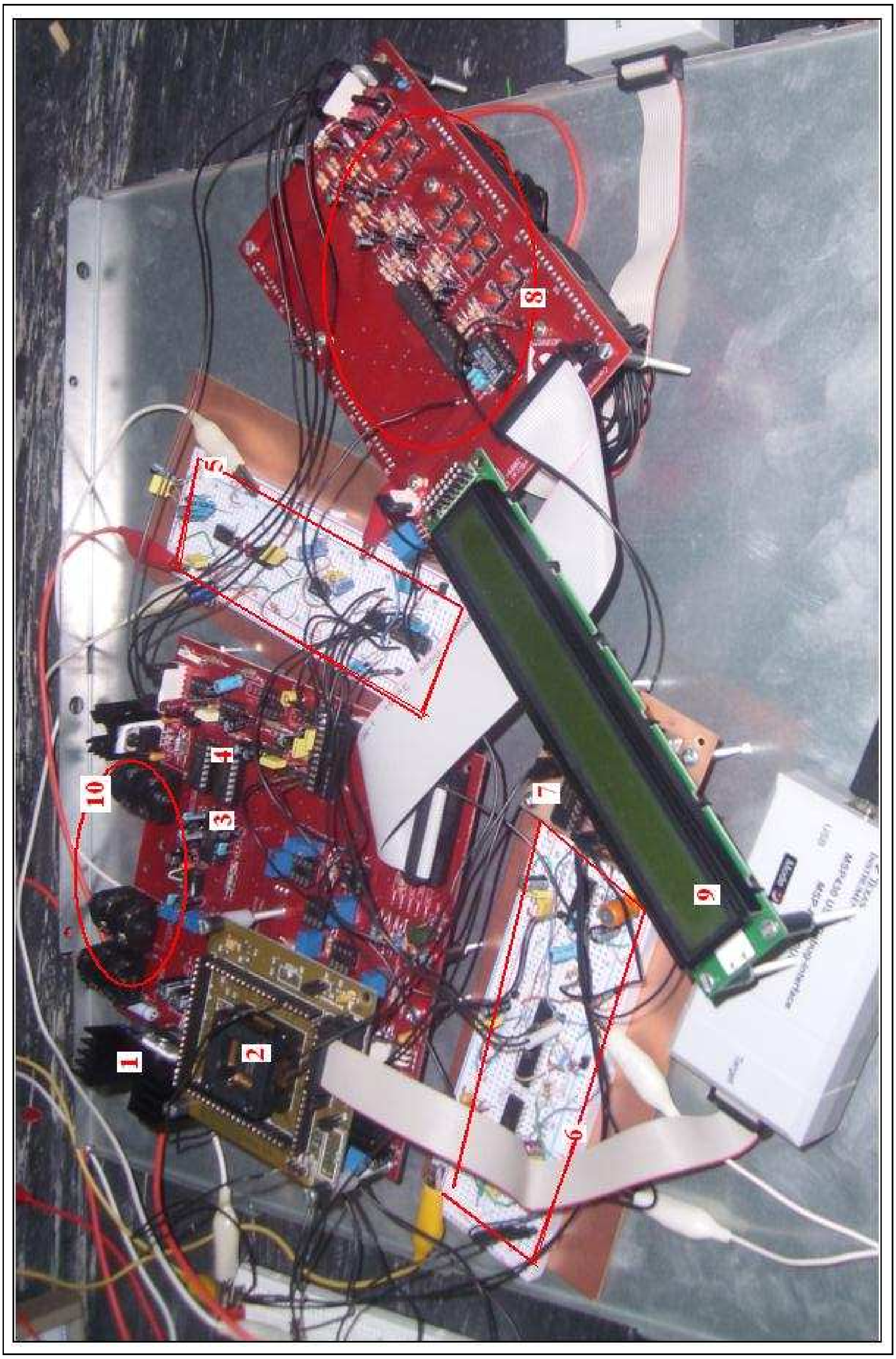

Figure B.1 Picture of the System

1 . Power amplifier

2 . Slave microcontroller

3 . Attenuator

4 . Multiplexers (Right and Left)

5 . Noise generator and noise masker

6 . Equalizer (Noise and tone channel)

7 . Pure tone generator

8 . Control panel

9 . LCD

10 . Jacks

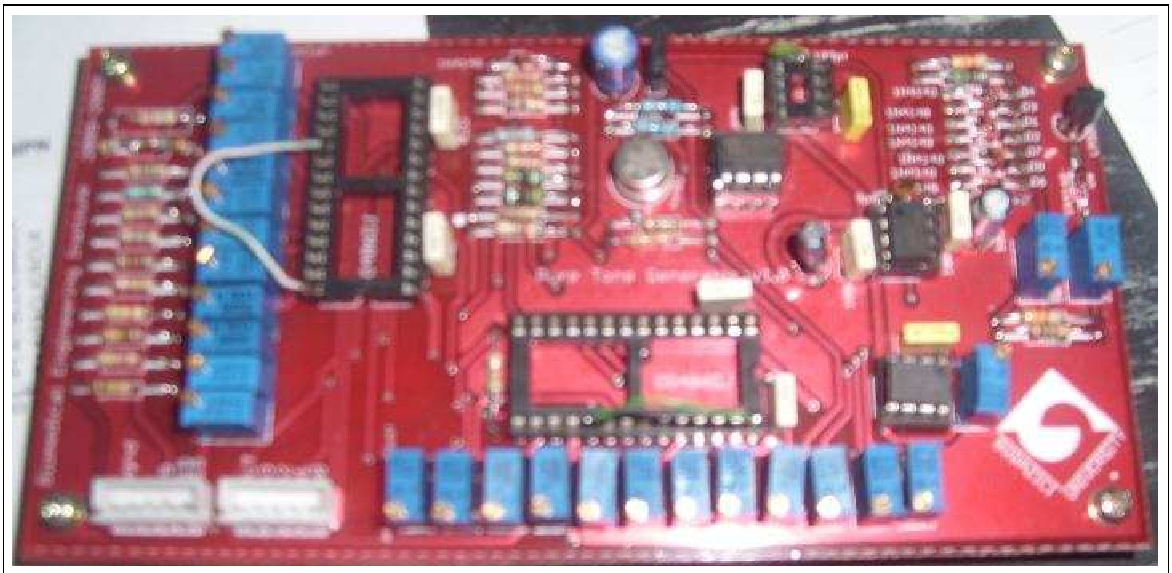

Figure B.2 Picture of the discarded Signal Generator

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