A PC-BASED AUDIOMETRY SYSTEM

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Master of Technology

By

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ABSTRACT

Audiometry involves quantitative determination of the degree and nature of hearing impairment. In this project, a PC-based screening audiometry system has been developed. The system consists of a PC with a data acquisition peripheral, and an audio attenuator-amplifier unit which can be controlled by the digital output of the data acquisition peripheral. The system generates and presents the stimulus to the subject. The intensity variation is brought about by the attenuator. The thresholds aredetermined according to a standard adaptive algorithm.

The system can implement three types of audiometry: operator controlled, automated, and subject controlled. In the operator controlled mode, the operator has complete control over the stimulus parameter selection and threshold determination. In the automated and the subject controlled modes, the threshold is determined automatically after the operator starts the program and specifies some experimental parameters.

In addition to providing advantages in terms of better reliability, flexibility, speed, and simplicity of use, this audiometry system is intended to facilitate easy data storage and retrieval for mass audiometry testing in schools and factory workforce.

ACKNOWLEDGEMENT

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V. Malini

[V. MALINI]

LIST OF SYMBOLS AND ABBREVATIONS

AC	Air Conduction
BC	Bone Conduction
C	Capacitors
CN	Connector
CR	Check Response
D	Diodes
DAP	Data Acquisition Peripheral
D/A	Digital-to-Analog
dB	Decibel
D/I	Digital Input
DL	Decrease Level
D/O	Digital Output
ERA	Evoked Response Audiometry
Gnd	Ground
HL	Hearing Level
Hz	Hertz
IL	Increase Level
I/P	Input
JP	Jumper in the DAP
KHz	Kilo Hertz
LSB	Least Significant Bit
MCL	Most Comfortable Loudness Level
MSB	Most Significant Bit

NC	No Connection
O/P	Output
PC	Personal Computer
Q	Transistors
R	Resistors
S	Switches
SDT	Speech Detection Threshold
SL	Set Level
SPL	Sound Pressure Level
SRT	Speech Reception Threshold
UCL	Uncomfortable Loudness Level
V_c	Control Voltage
V_{ref}	Reference Voltage
V^+	Positive Supply Voltage
V^-	Negative Supply Voltage

C O N T E N T S

CHAPTER 1

INTRODUCTION

1.1 OVERVIEW OF THE PROBLEM

Many persons who suffer from significant hearing impairment can be helped with modern medical and surgical procedures, and instrumentation. The testing of a person´s hearing is done to quantify the hearing impairment and diagnose its cause so that suitable treatment may be prescribed wherever necessary and possible.

Some of the earliest hearing tests included observing an individual´s response to vocal sounds or sounds produced by clapping hands etc. But these tests provided limited qualitative or quantitative information [Martin, 1986]. Later the tuning fork was used to test hearing because it provided quantitative information regarding the frequency emitted. But any diagnosis based on such a test is limited to the frequency of the fork used because the ear is not equally sensitive for different frequencies. The advent of electronics led to the development of audiometry, a technique for identification and quantitative determination of hearing impairment. It involves presentation of systematically varying acoustic stimuli to the subject and recording the responses. The minimum intensity level to which consistent responses are obtained is taken as the threshold of hearing.

There are different audiometry procedures depending on the stimuli used. In pure-tone audiometry, the subject's threshold for hearing of pure-tones is measured. In speech audiometry, the subject's threshold for reception of speech is recorded. In addition to the hearing threshold level (HTL), the most comfortable listening level (MCL) and the uncomfortable loudness level (UCL) also may be estimated.

In the conventional instruments for both the above audiometry procedures, the variation of the level of the stimulus is done manually after carefully observing the responses of the subject. The performance and interpretation of results of such tests require experience and clinical expertise. The correct determination of thresholds, etc. becomes subjective and this suggests a need for audiometry techniques which would be less audiologist-dependent [Meyer & Sutherland, 1976].

The audiometry procedures can be carried out in a methodical and scientific manner using the process of logical decision making. They also require control of the stimulus presentation in terms of its intensity. The stimulus parameters (intensity and frequency) are to be changed according to a standard algorithm which depends on the responses of the subject. Such parameters are well-suited to be controlled by computers [Lutman, 1983].

1.2 SCOPE OF THE PROJECT

The objective of this project is to develop a PC-based audiometry system that would offer advantages over conventional instruments in terms of increased speed of testing, simplicity of operation, higher reliability, greater flexibility and will hopefully be less expensive.

The system as shown in Fig. 1.1 mainly consists of a PC with a data acquisition peripheral (DAP) and an audio attenuator and amplifier unit. The PC serves as the overall controller of the audiometry procedure. The DAP is programmed for generation and intensity control of the stimulus, and subject response monitoring. The audio attenuator and amplifier unit is needed for varying the stimulus intensity. After the generation and presentation of the stimulus to the subject, his or her response is monitored as a threshold determining criterion. Accordingly, the intensity value is varied and the threshold value determined. At the conclusion of the test, the threshold values are displayed graphically as well as in a tabular form. The subject data and the test results are stored for any further analysis.

The system as implemented now performs three (operator controlled, automated and subject controlled) types of pure-tone audiometry. The system can be field-tested and evaluated by audiologists with hearing impaired subjects. Based on the experience thus given, further improvements in the system can be made. With the same hardware setup, the software can be modified to also incorporate speech audiometry by using digitized speech waveform as the stimulus.

1.3 OUTLINE OF THE REPORT

Chapter 2 provides an overview of various audiometry techniques, and the advantages of employing a PC in audiological investigations. The scheme for the PC-based audiometry system and its software details have been dealt with in the third chapter. In Chapter 4, the hardware aspects of the system have been discussed. The fifth chapter provides an explanation of the implementation of the system as a whole. Chapter 6 summarizes the work done along with the features of the system, and suggestions for further improvements.

Appendix A gives a brief review of the hearing mechanism, its impairments, and thresholds. The data acquisition peripheral is described in Appendix B. Appendix C contains the listing of the program "PC-AUD", for the audiometry testings. Appendix D provides a brief explanation of the low-noise operational amplifier LF 356 and the analog switch DG 201 used for the development of the digital attenuator.

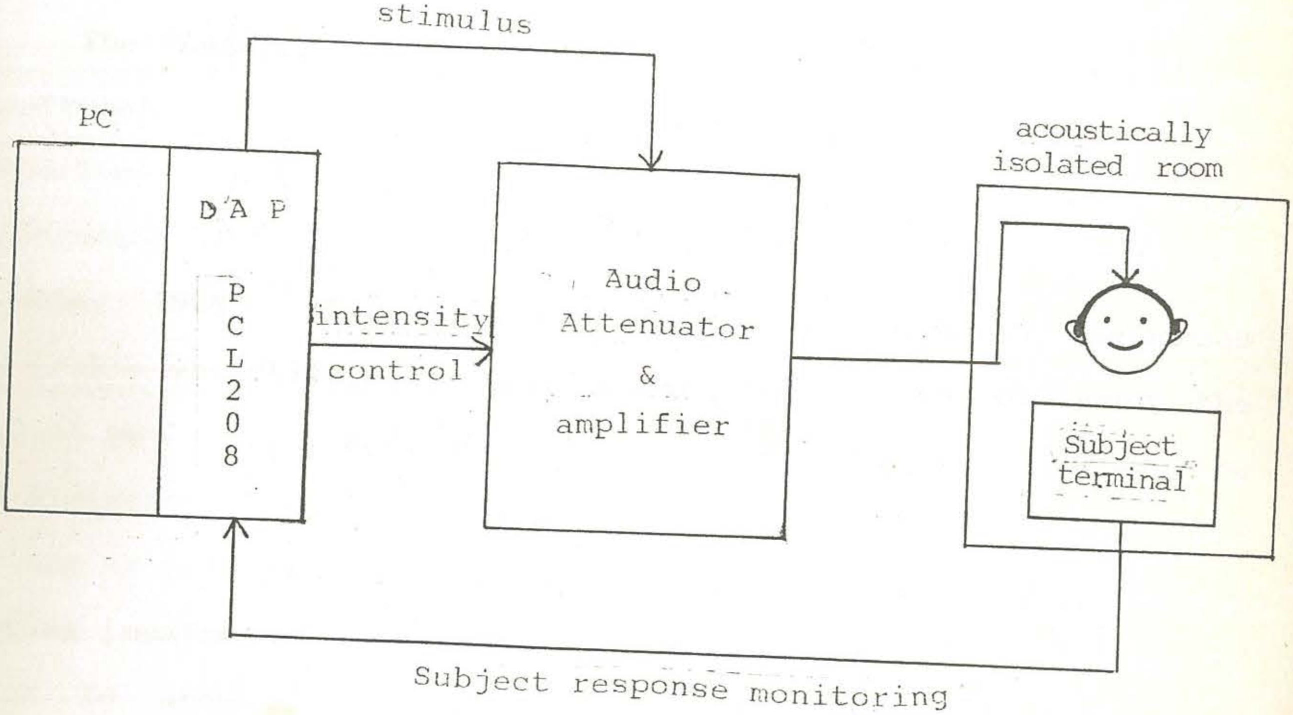


Fig.1.1 Simplified block diagram of the PC-based audiometry system

CHAPTER 2

AUDIOMETRY TECHNIQUES

2.1 INTRODUCTION

The nature, degree and probable cause of the hearing impairment should be assessed before a treatment may be prescribed. An important component of such an assessment is the measurement of the responses of a person to acoustic stimuli [Webster, 1988]. These responses may be physiological responses measurable objectively or subjective indications of what the subject perceives. Audiometry is the science and art of measuring and evaluating both the objective and subjective responses. The testing is performed by a qualified audiologist who quantifies any hearing impairment and evaluates any existing hearing pathology in order to provide diagnostic information for treatment and rehabilitation. The first few sections of this chapter provide an overview of various audiometry techniques - pure-tone audiometry, speech audiometry, evoked response audiometry and computerized audiometry. A reader may first refer Appendix A which briefly describes the mechanism of hearing, its impairments, hearing thresholds and frequency selectivity of the ear. The concluding sections deal with a few applications of microcomputers in audiology and the need for a PC-based audiometry system.

2.2 PURE-TONE AUDIOMETRY

In pure-tone audiometry, the subject´s responses for acoustic stimuli of different frequencies are measured. The test tones presented are in the range of 125 Hz to 8000 Hz. The pure-tone audiometer, shown in Fig. 2.1, consists of an oscillator, as a source of sinusoidal test signals in the audio frequency range. The specific test frequency is determined by the frequency selector control. The hearing level dial is used to alter the intensity of the test signal. The initial frequency and intensity are selected by the audiologist. The attenuator can be adjusted in 5 dB steps in the range of -10 dB to 110 dB. The output signal is presented to the subject seated in an acoustically isolated room, through earphones.

The subject acknowledges the hearing of a tone by raising a hand or a finger or by pressing a push-button switch which lights a lamp on the audiometer. The intensity of the stimulus is decreased in 10 dB steps until the subject no longer responds and then gradually increased in 5 dB steps until the subject responds to its hearing. The audiometer hearing level dial is adjusted until the person responds correctly to a test tone in 50% of the trials. The minimum sound level of a particular frequency which is heard by an individual in 50% of the trials is defined as the threshold at that frequency. The intensity is decreased in 10 dB steps and increased in 5 dB steps to obtain two agreeing ascending readings of the threshold. The whole procedure is repeated over the frequency range of clinical interest and thresholds are determined at 125 Hz, 250 Hz, 500 Hz, 1 kHz, 2 kHz, 3 kHz, 4 kHz, 6 kHz, and 8 kHz. The stimulus is then presented to the other ear and the thresholds determined similarly. The results are recorded as a graph called an audiogram. An audiogram is a plot of the threshold versus the frequency of the stimulus. A typical audiogram representing normal hearing is shown in Fig. 2.2.

Bekesy audiometry is a variation of pure-tone audiometry wherein the subject tracks his own pure-tone thresholds. This technique uses an audiometer with a motor driven attenuator and pen recorder. The intensity of the tone increases until the subject hears the tone through his earphones and presses his response button. When the button is depressed, the attenuator is driven in the opposite direction and the intensity of the tone decreased until the subject releases the button because he can no longer hear the tone. This technique provides several threshold crossings for each frequency being tested. The procedure is repeated separately for each frequency and ear. At the conclusion of the test, the audiologist examines the pen recording and determines the thresholds for both ears and for each frequency [Meyer & Sutherland, 1976].

2.3 SPEECH AUDIOMETRY

In speech audiometry, an individual's ability to hear and understand speech and thereby the integrity of the auditory system is assessed. A speech audiometer is an instrument designed specifically to deliver test material to the subject at a known intensity.

The block diagram of the speech audiometer is shown in Fig. 2.3. The stimulus is provided by a microphone, tape recorder or a phonograph. The intensity of the stimulus is adjusted using a hearing level dial [Martin, 1986]. The speech audiometry is performed in a two-room acoustically isolated suite. The subject is fitted with earphones and is instructed by the audiologist regarding his task during the test session. The mode of instruction may be written and/or verbal. The stimulus may either be cold-running speech (rapidly delivered speech, either pre-recorded or by monitored live voice such that the output is monotonous) or spondees (two-syllable words having common usage in the language, pronounced with equal stress on both syllables). Usually spondees are used. After familiarizing the subject with spondees, one spondee is presented at an intensity decided by the audiologist. A different spondee is presented at each level in 10 dB steps until the subject repeats the word correctly. The level is then reduced by 15 dB below the level at which the correct response was obtained and four words are presented at each level as the intensity is increased in 5 dB steps. When the subject identifies three out of four words correctly, the level is reduced by 10 dB. The procedure is repeated with four spondees at each level. The speech reception threshold (SRT) is defined as the minimum intensity at which speech can be understood. SRT is determined as the level at which at least half of the spondees are correctly repeated.

The stimulus intensities can be varied between -10 dB to 110 dB using the attenuator which is calibrated in dB with respect to audiometric zero for speech.

2.4 EVOKED RESPONSE AUDIOMETRY

Evoked response audiometry, also known as electric response audiometry (ERA) is a technique to evaluate the integrity of the central auditory system by measurement of electrophysiological activity evoked by a transient acoustic stimulus. ERA is objective in that no conscious response from the patient is required [Birrel, 1983]. The block diagram of evoked response audiometer used in clinical practice is shown in Fig. 2.4. The signal generation and presentation are similar to those in pure-tone audiometry. Additional timing circuitry is provided in ERA to generate precisely timed tone pips or clicks. The scalp electrode array consists of the active and the reference electrodes. An individual response is rarely of sufficient magnitude to be detectable in the concurrent spontaneous electrical activity and therefore time-domain averaging is done to extract the response. Information about the different levels of the system can be obtained from sub techniques such as electro-cochleography, brain-stem electric response audiometry and electroencephalic audiometry [McAinsh, 1988] :

(i) Electrocochleography measures the electrical activity generated in the cochlea in response to a click stimulus. A needle-shaped active electrode is surgically passed through the tympanic membrane and held in place against the promontory near the round window [Birrel, 1983]. The responses include the cochlear microphonics and the auditory nerve action potential. Being invasive, this technique is being replaced by brain-stem electric response audiometry. Its main clinical use is in the threshold determination in children who cannot be tested by other means.

(ii) Brain-stem electric response audiometry is used to detect electric impulses which occur in the area of the brain stem in response to sound. It uses surface electrodes placed on the mastoid and the vertex and hence is non-invasive. This technique can be used to determine the site and nature of a lesion in the auditory pathway and also to get a rough estimate of hearing threshold.

(iii) Electroencephalic audiometry is primarily concerned with the change of electrical activity at the vertex, evoked by a sound stimulus. It is recorded using surface electrodes placed over the central and frontal areas of the scalp. It is a very good objective measurement of hearing threshold. The application of this technique is mainly in medico-legal work and in the evaluation of non-organic or psychogenic hearing loss wherein the subjects are unable or unwilling to co-operate.

2.5 COMPUTERISED AUDIOMETRY

Most audiometry procedures involve systematic and repeated presentations of stimulus and are therefore amenable to computer control. Computerization is particularly suited to interactive procedures, those in which the presentation of a particular stimulus is dependent on the outcome of the preceding test. A self recording system based on the presentation of fixed frequencies has been described by Berry et al., (1979),[McAinsh, 1988].

Systems which allow programmed presentation of stimulus and provide automatic recording of the subject's responses have shown improved sensitivity over the conventional procedures with regard to the consistency in application of auditory decision criteria.

2.6 COMPUTER CONTROLLED AUDIOLOGICAL INVESTIGATIONS

Many applications of computers in audiology have been reported in the literature [Lutman (1983), Mason (1988), Coleridge Smith & Scurr (1988)]. A few of them are discussed in this section, with a special reference to the role of computers and the accompanying advantages.

Lutman (1983) has reported a microcomputer based system for assessing temporal and frequency resolution of the ear. Both these abilities are impaired in sensorineural hearing impairment. Therefore their accurate determination is very important for a reliable diagnosis.

The brain-stem evoked response audiometry, as explained earlier in this chapter is used for the identification of hearing impairment in children and particularly in neonates who are in the special baby care unit. Mason (1988) reported the development of a system for brain-stem evoked response audiometry. In this system, the parameters of the stimulus and data acquisition are controlled by a BBC Microcomputer, and a recording unit interfaced to it. The author has reported that this is quick, simple to operate and provides unambiguous results, without requiring a highly experienced operator. The data can be stored on a floppy disk for further off-line analysis.

The microcomputer analysis of pure-tone audiometry data has been done by Coleridge Smith & Scurr (1988). The system, based on a microcomputer, receives the data from a pure-tone audiometry test and generates a display which gives a standard representation, thus helping in immediate interpretation. The analysis described here takes into account the effect of presbyacusis (loss of hearing due to aging). This factor is of importance in deciding whether any hearing loss experienced is due solely to presbyacusis or whether some other disorder is also present. The graphical representation of the audiogram makes use of graphics capability of the microcomputer and also provides more information than the conventional audiogram as shown in Fig. 2.5. This is in the form of subject data and numerical presentation of hearing threshold levels. The effect of presbyacusis is represented graphically in the form of a normal band and a figure of merit labelled as deviation is also provided. This figure is an assessment of the deviation of the subject's hearing threshold levels from the expected levels given by the center of the band.

2.7 NEED FOR A PC-BASED AUDIOMETRY SYSTEM

In pure-tone audiometry, as described earlier in this chapter, the subject responds to varying intensities of the acoustic stimuli of a discrete frequency over the spectrum of

125 Hz to 8000 Hz. The variation of the level of the stimulus is being done manually by the audiologist after carefully observing the responses of the subject. The audiologist then decides the threshold for that particular frequency. The whole procedure is repeated for each of the frequencies and the second ear. The audiogram is plotted on a audiogram sheet. The performance and interpretation of results in this case require clinical expertise which makes the threshold determination subjective. This suggests a need for other audiometry techniques which are less subjective [Meyer & Sutherland, 1976].

The pure-tone audiometry can be carried out in a systematic and scientific manner using the process of logical decision-making. In this procedure, it is very essential to control the presentation of the stimulus accurately and repeatably in terms of its intensity. The stimulus parameters are also to be changed according to a standard algorithm which depends on the responses of the subject.

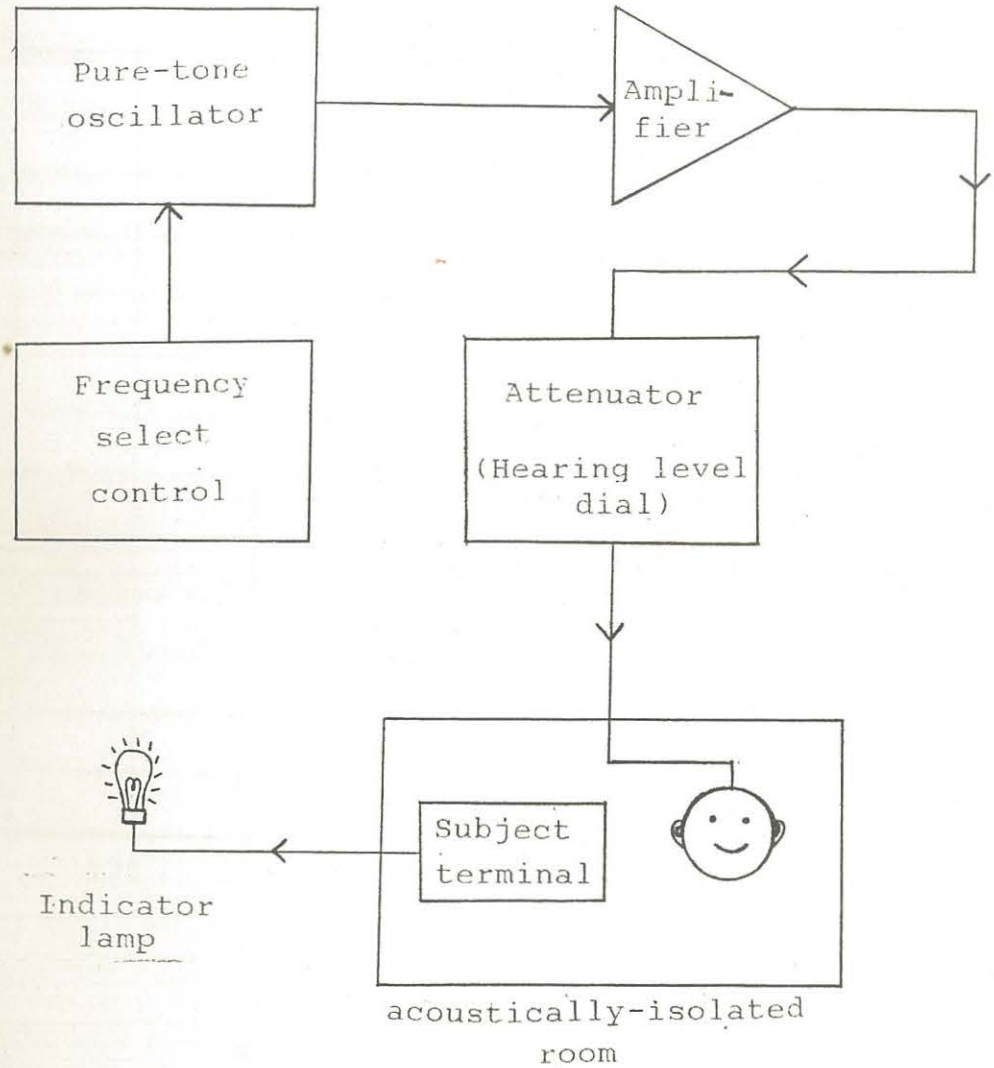


Fig.2.1 Block diagram of a pure-tone audiometer

Adapted from, Martin (1986)

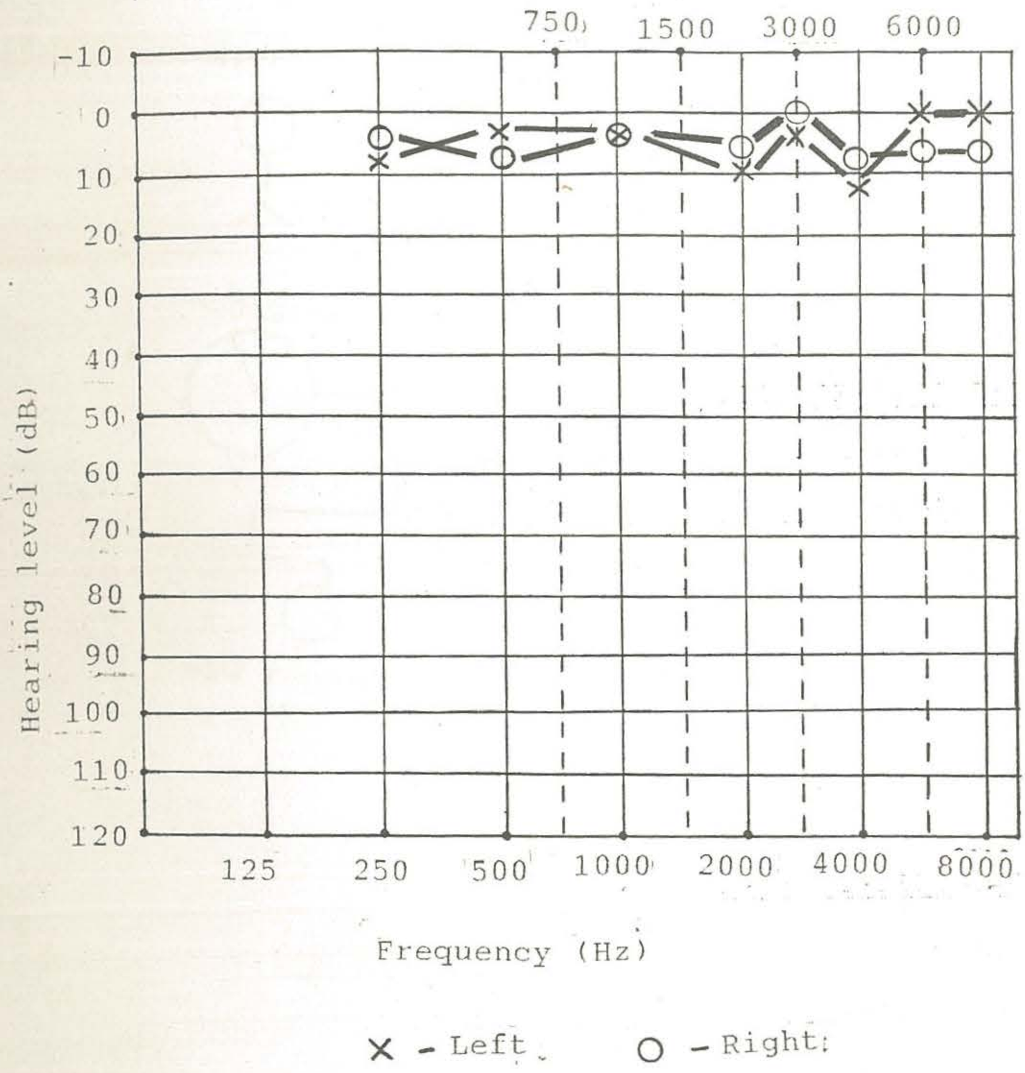


Fig.2.2 Audiogram illustrating normal hearing for both ears

Adapted from, Martin (1986)

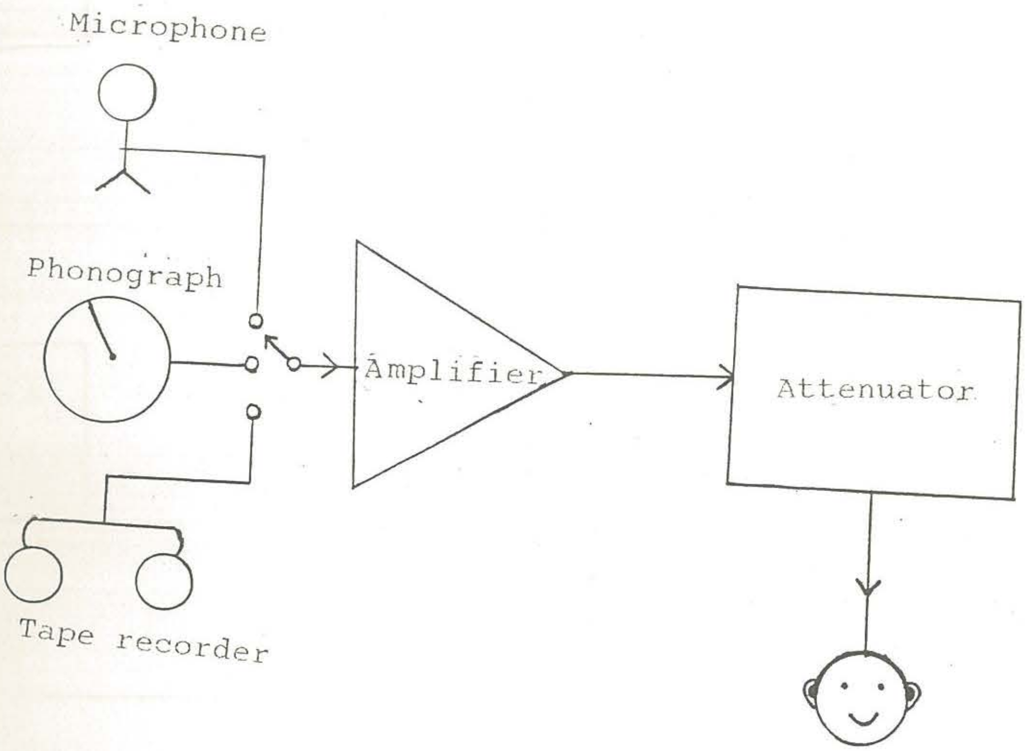


Fig.2.3 Block diagram of a speech audiometer
Adapted from, Martin (1986)

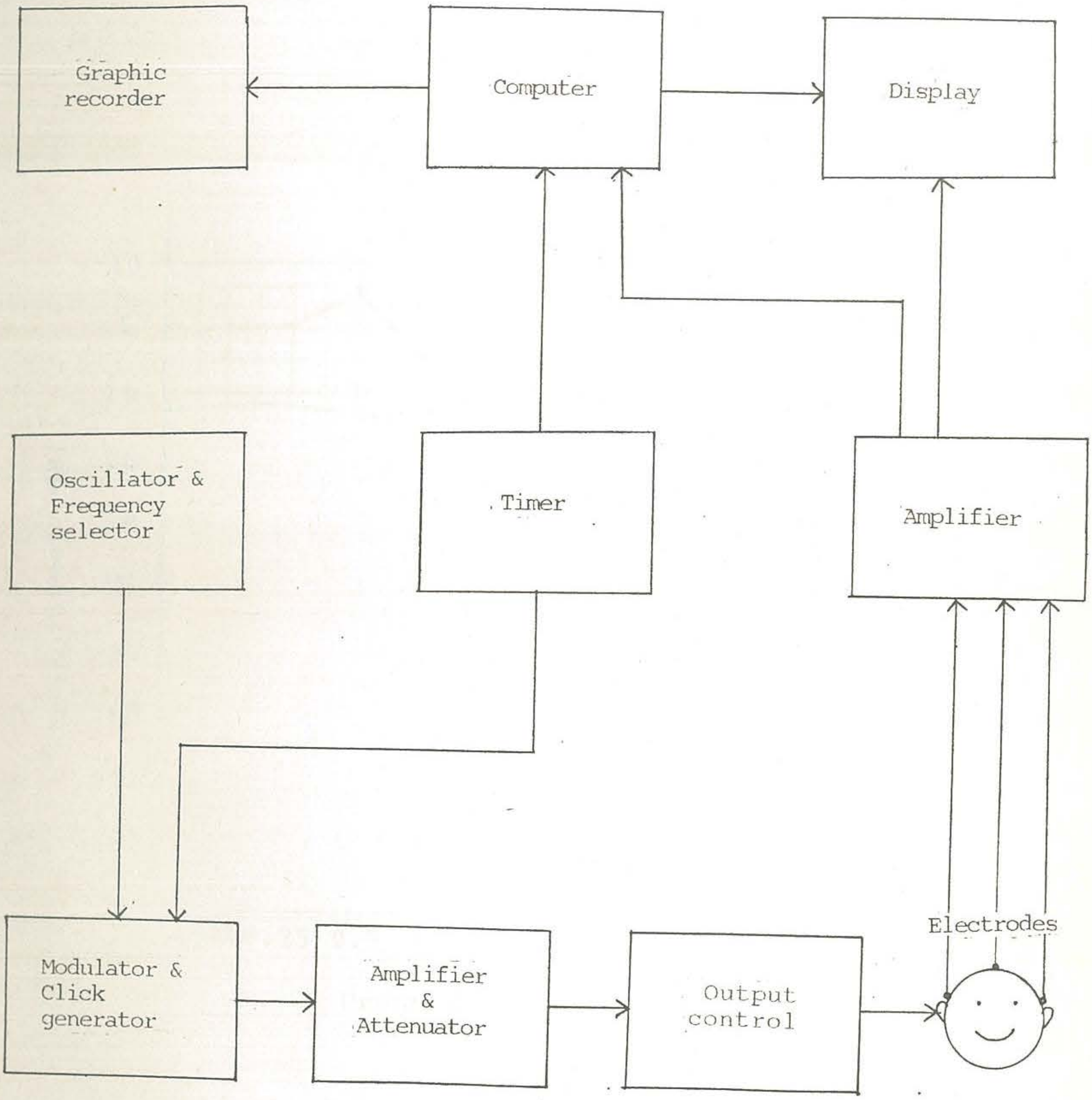
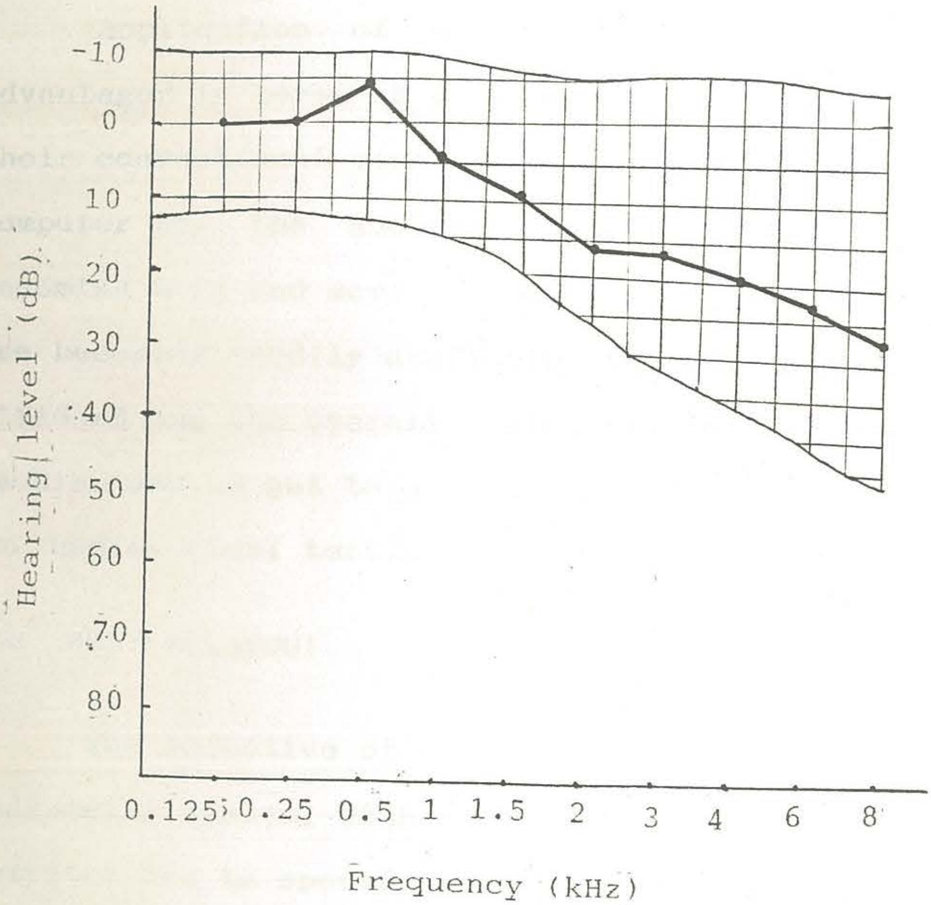


Fig.2.4 Block diagram of an evoked response audiometer

Source : Webster (1988)



Patient: G.D.L. (Male) Age: 55

Pure-tone responses: (Deviation=49 dB)

.125	.25	.5	1	1.5	2	3	4	6	8
0	0	-5	5	10	20	25	25	35	40

Fig.2.5 Audiogram recorded using a microcomputer

Adapted from, Coleridge Smith & Scurr (1988)

CHAPTER 3

A PC-BASED AUDIOMETRY SYSTEM

3.1 INTRODUCTION

Application of microcomputers in audiology offers many advantages in terms of flexibility, and simplicity of use, over their conventional counterparts. One may use a PC as a microcomputer for the audiometry system. This is because of its becoming more and more commonplace, different peripherals for it are becoming readily available, the bus structure has become a standard and the overall system can be a low cost one. The machine can be put to a variety of other uses when not being used for audiological testing.

3.2 SYSTEM LAYOUT

The objective of this project is to develop a pure-tone audiometry system, with the PC as an overall controller. The operator has to specify only some experimental parameters. The minimal involvement of the operator during the investigation reduces the element of subjectivity on the part of the operator. The layout of the system is shown in Fig. 3.1. The complete system consists of an audio attenuator and amplfier unit, interfaced to an IBM PC/AT with a data acquisition peripheral (DAP), PCL 208 installed in it. PCL 208 can handle digital-to-analog (D/A) conversion, digital output (D/O), digital input (D/I) and related operations. Appendix B gives the details of this DAP in brief. The subject is seated in an acoustically isolated room and is fitted with earphones. The DAP is programmed for stimulus generation. The 8-bit D/O port of PCL 208 is used to specify the intensity of the stimulus. The intensity variation is brought about by the audio attenuator and amplifier unit which is controlled by the D/O port. The system presents the stimulus through the D/A port of the DAP. The stimulus is presented monaurally to the subject through a pair of earphones. The subject responses are recorded using a response switch connected to the D/I of PCL 208. Based on the responses, the D/O specifies a different intensity for the stimulus. Thus varying the intensity systematically and logically according to a standard algorithm, the thresholds are determined. The subject data and the test results are stored for further off-line analysis.

The hardware details of the system are explained in Chapter 4. The software required to implement the audiometry testing is described in the next section.

3.3 SOFTWARE

The program "PC-AUD" in this project is written in GW-BASIC so that the I/O driver routines of the DAP can be easily accessed. The listing of "PC-AUD" is given in Appendix C. The main functions implemented using the program are :

(i) Stimulus generation and presentation

(ii) Subject response monitoring

(iii) Estimation of the threshold by implementing the adaptive algorithm

(iv) Storage of subject data and threshold values

(v) Display of the results in a graphical form

3.3.1 Adaptive Algorithm

The adaptive algorithm according to which the threshold determination is made is based on the one recommended by the American Speech and Hearing Association (ASHA) and adopted by the American National Standards Institute in 1978 [Martin, 1986]. The flow chart representation of the algorithm is shown in Fig. 3.2. After the stimulus is presented at a preset intensity initially, the algorithm calculates the subsequent intensity values depending upon the subject's previous responses.

A pure tone is presented initially at 30 dB HL. Response obtained at this stage indicates that the 30 dB tone is at or above the subject's threshold. If no response is seen, the level is raised to 50 dB, then in 10 dB steps until a response is obtained or the maximum limit of the audiometer is reached for the test frequency.

After a response is obtained, the level is reduced in 10 dB steps. Each time a tone is presented, it is maintained for one to two seconds. All increments in the sound level from this point are made in 5 dB steps. When the tone is lowered below the subject's threshold, it is then raised in 5 dB steps until it is audible again, then lowered in 10 dB steps, and raised in 5 dB steps until the 50% threshold response criterion has been met. The threshold is taken as the lowest level at which the subject can correctly identify three out of a theoritical six tones. The same procedure is repeated with tones of other frequencies.

3.3.2 Types of Audiometry

The software implements three types of audiometry : operator controlled (Conventional), automated and subject controlled (Bekesy). The three types vary in the mode of implementation of the various functions as summarised in Table 3.1.

The operator controlled type is very similar to the conventional audiometry in that the audiologist has complete control over the stimulus parameter variation and threshold determination. This type requires the active participation of the audiologist throughout the duration of the testing. His task is to select the stimulus parameters, observe the subject for the response and vary the intensity accordingly. The threshold is decided by the audiologist. After the testing is complete, the audiogram is plotted. The flow diagram for this type of audiometry is given in Fig. 3.3.

In the automated and the subject controlled (Bekesy) types, the estimation of threshold is program controlled according to the adaptive algorithm, discussed in Section 3.3.1. The initial value of the stimulus intensity is preset. The subsequent values of the intensity are calculated by the algorithm which takes the subject's previous responses into account. The audiograms are displayed on the screen. Fig. 3.4 shows the flow diagram for the automated and subject controlled types of audiometry.

In all the three types, the subject data along with the test results are stored.

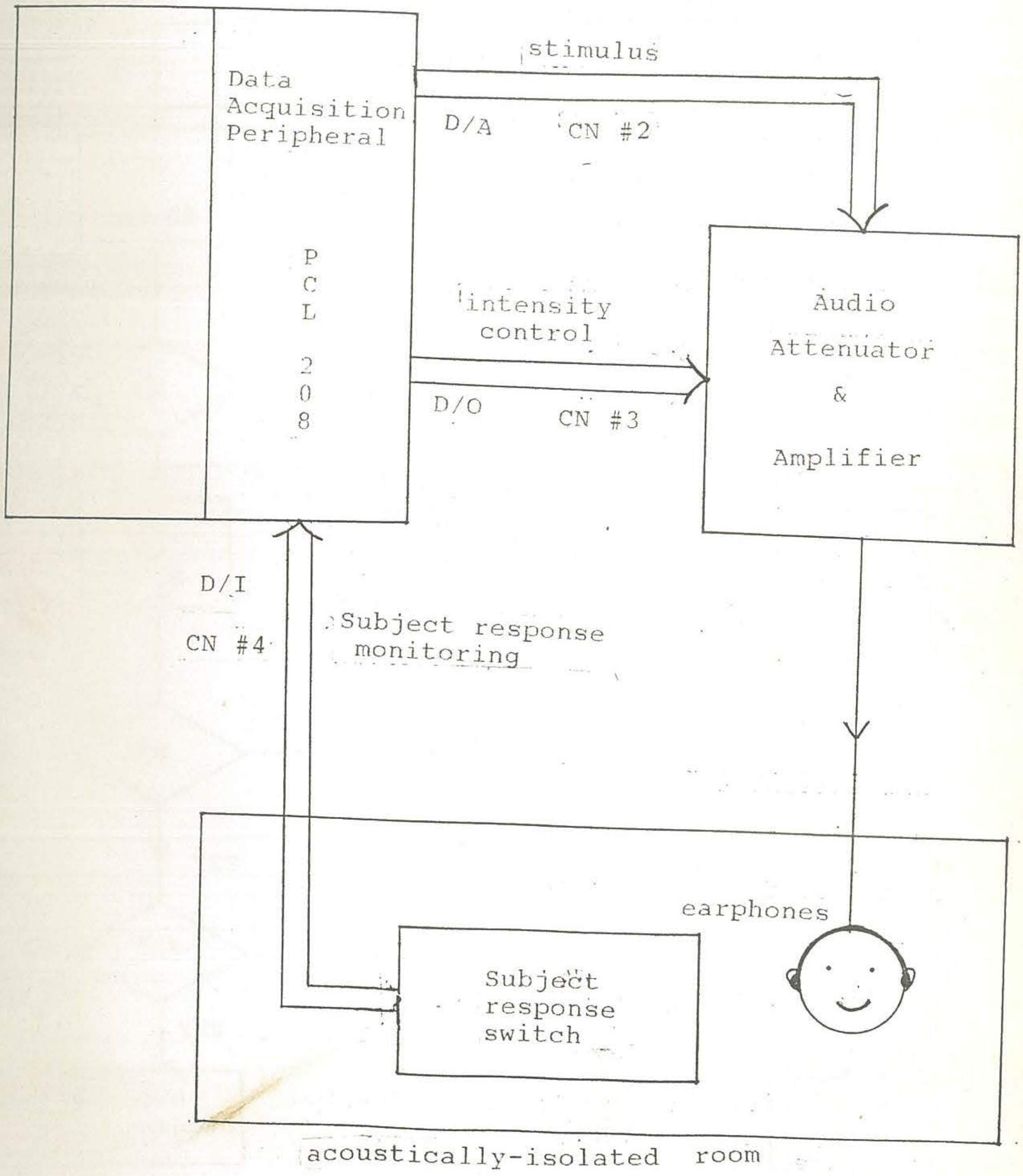


Fig.3.1 Layout of the PC-based audiometry system

Adapted from, Pandey (1987)

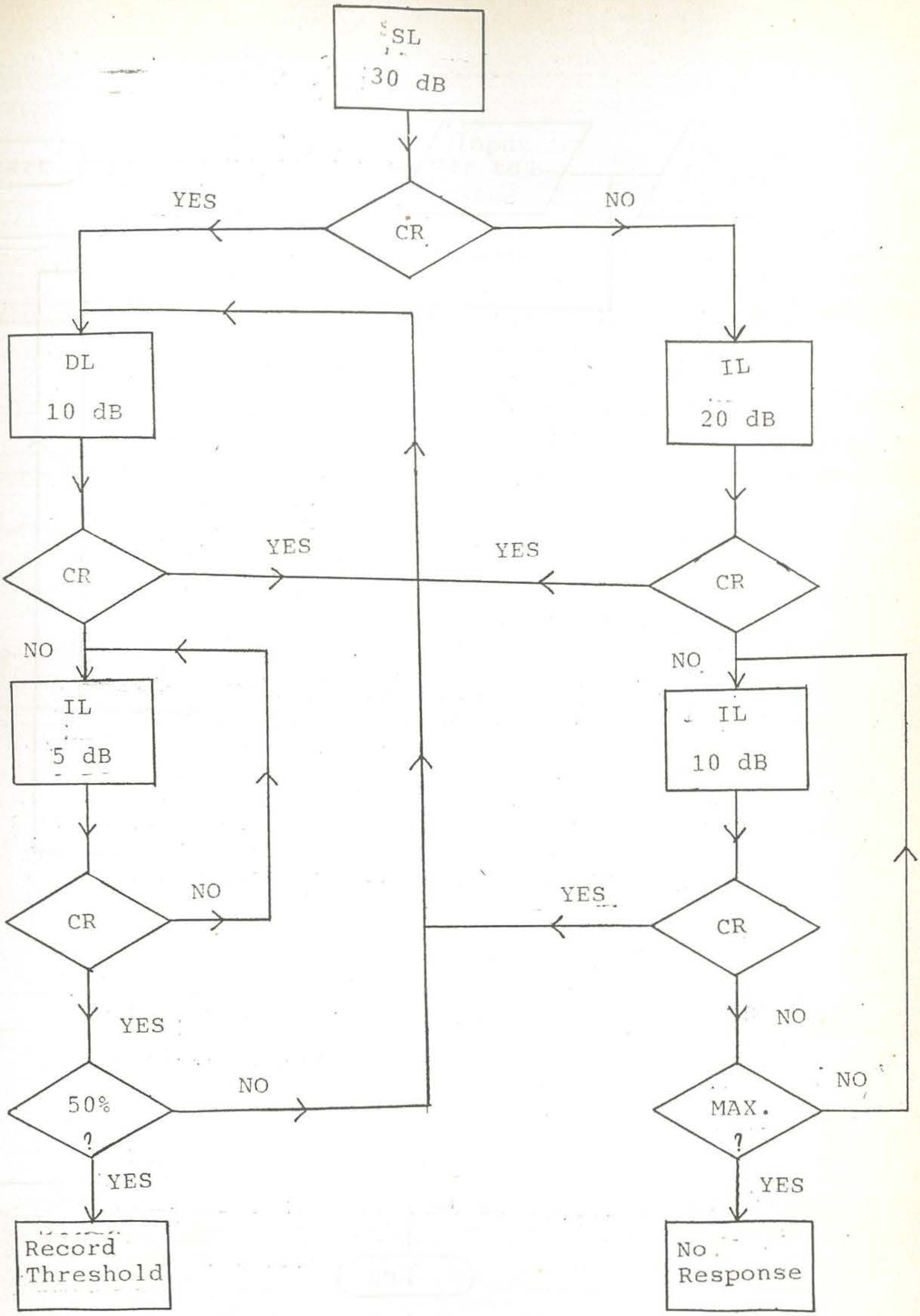


Fig.3.2 Adaptive algorithm for pure-tone audiometry

Source : Martin (1986)

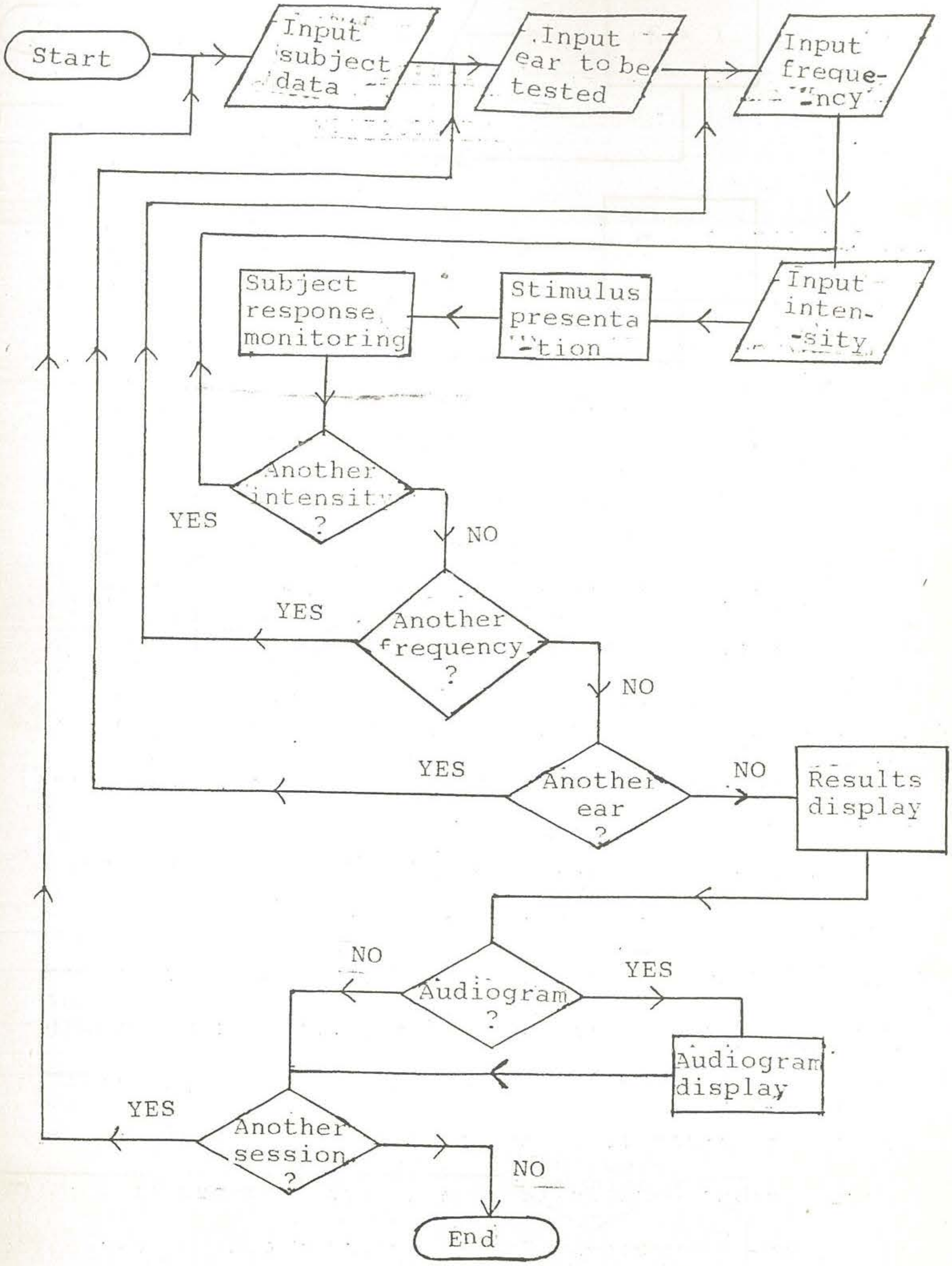


Fig.3.3 Flow diagram of operator controlled audiometry

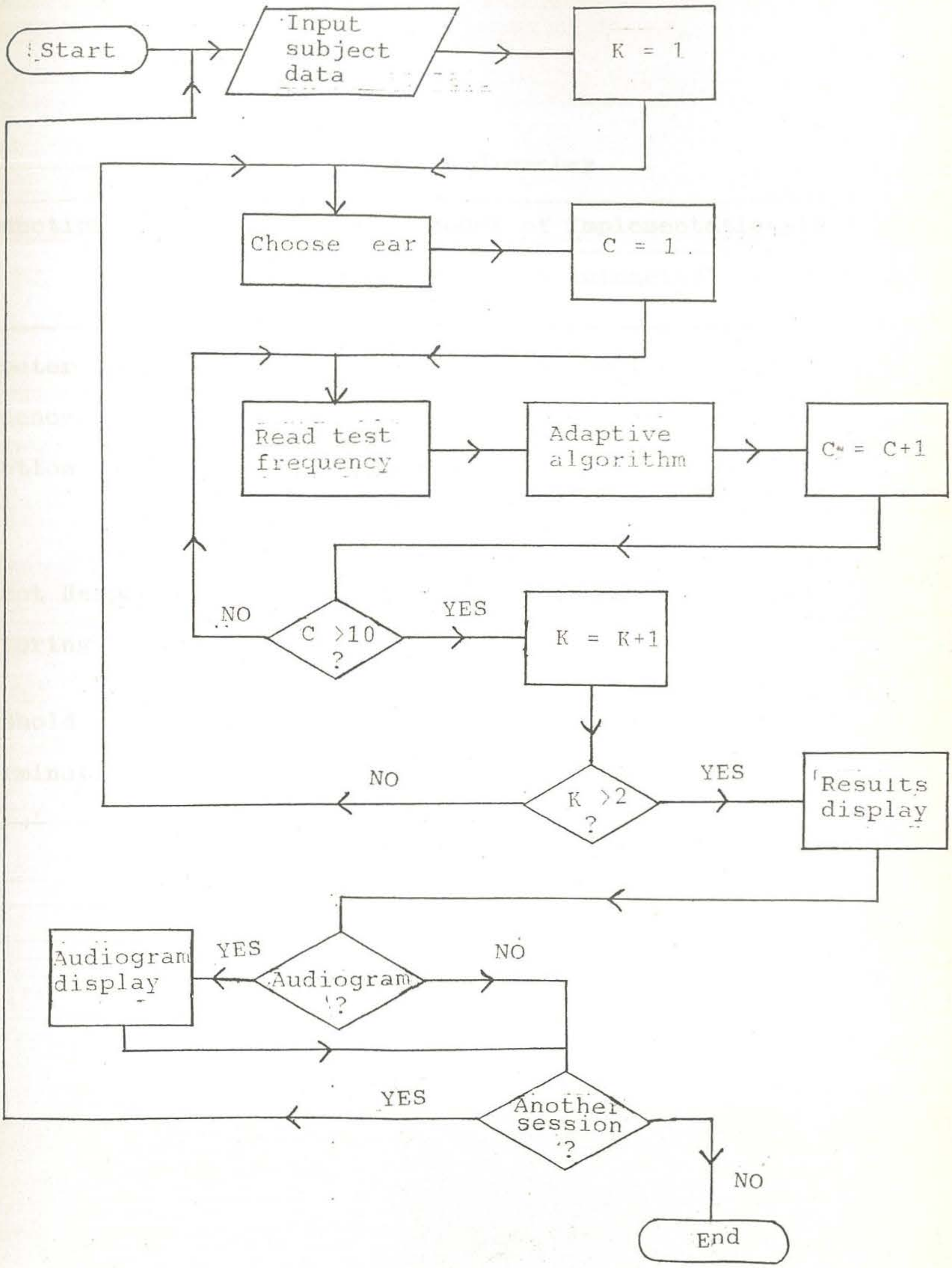


Fig.3.4 Flow diagram of automated and subject controlled audiometry

Table 3.1 Types of Audiometry

Function	Model of Implementation in		
	Operator controlled	Automated	Subject controlled
Parameter (ear, frequency, intensity) selection	Manual	Automatic	Automatic with simultaneous plotting of the intensity
Subject Response Monitoring	Manual	Automatic	Automatic
Threshold determination	Manual	Automatic	Automatic

CHAPTER 4

SYSTEM HARDWARE

4.1 INTRODUCTION

The layout of the PC-based audiometry system is described in Section 3.2. The major hardware components of the system are the PC, the DAP and its I/O ports, the audio attenuator and the amplifier unit, and the subject response switch. This chapter describes these functional blocks.

4.2 COMPUTER

The Computer used to control the overall procedure in our system is an IBM PC/AT with monitor and a disc drive. One of the expansion slots on the AT bus holds the data acquisition peripheral (DAP). The PC implements audiometry by executing the program. At the conclusion of the test, the subject data and the results of audiometry are stored and displayed on the monitor. The monitor also displays the audiogram.

4.3 DATA ACQUISITION PERIPHERAL

The DAP, PCL 208 has capabilities for operations like digital-to-analog (D/A) conversion, analog-to-digital(A/D) conversion, digital output and digital input. The specific details of the peripheral are provided in Appendix B of this report. All the operations of this card are managed via the IBM PC/AT data bus. The program developed in this project calls the assembly language driver routines of the DAP, which set up the operations. In this project, the D/A conversion, digital output, and digital input are utilized.

One of the two-channel D/A converters with 12-bit resolution is used to generate the stimulus. The stimulus waveforms are sinusoids in the frequency range of 125 Hz to 8000 Hz. It generates sinusoids by sending previously prepared data in an array to the D/A converter at A/D interrupts.

The 16-bit digital output port of the DAP specifies the intensity of the stimulus that is to be presented to the subject. The required intensity range extends from 0 dB to 110 dB. Only the least significant 8 bits are utilized as they are sufficient to cover this range. The subject response switch is connected to the digital input of the DAP. The least significant bit of the 16-bit digital input monitors the switch position and records the subject response.

4.4 AUDIO-ATTENUATOR AND AMPLIFIER UNIT

In pure-tone audiometry, the intensity of the acoustic stimulus is systematically varied to determine the threshold of the subject. In the PC-based audiometry system, the requirement is that of an attenuator, which can provide attenuation over the range of 0 to 110 dB, under computer control. The circuit diagram of computer controlled attenuator is shown in Fig. 4.1. It is a modification of the circuit developed by Severns, 1985.
The circuit is reported to have been used in audiological testing of human subjects with encouraging results.

The circuit allows the presentation acoustic stimuli at varying intensities under the control of an 8-bit digital output. The attenuation of the input signal can be specified between 0 dB and 127.5 dB in 0.5 dB steps. The circuit has four active stages, each providing different levels of attenuation. Each stage of the circuit has a low-noise operational amplifier LF 356 and the switching element to vary the gain of LF 356. DG 201 with 4 built-in analog switches is used in this project. Appendix gives the details of LF 356 and DG 201. The first stage provides attenuation in the range of 0 to 7 dB in 0.5 dB steps. 0 to 24 dB attenuation in steps of 8 dB is brought about the second stage. The third stage provides an attenuation of 0 to 32 dB in a single step. The final stage, in addition to providing a 64 dB attenuation, also provides current amplification. The output stage is that of a standard push-pull configuration. It has very low noise level at the output and drives the headphones directly. The individual stages of attenuation increases by powers of two. This allows the attenuation in dB to be specified as a binary number, which is sent as an 8-bit digital output. The condition of the individual switches and the associated attenuation is given in Table 4.1. Table 4.2 shows the digital output required to specify the attenuation between 0 to 110 dB in 5 dB steps.

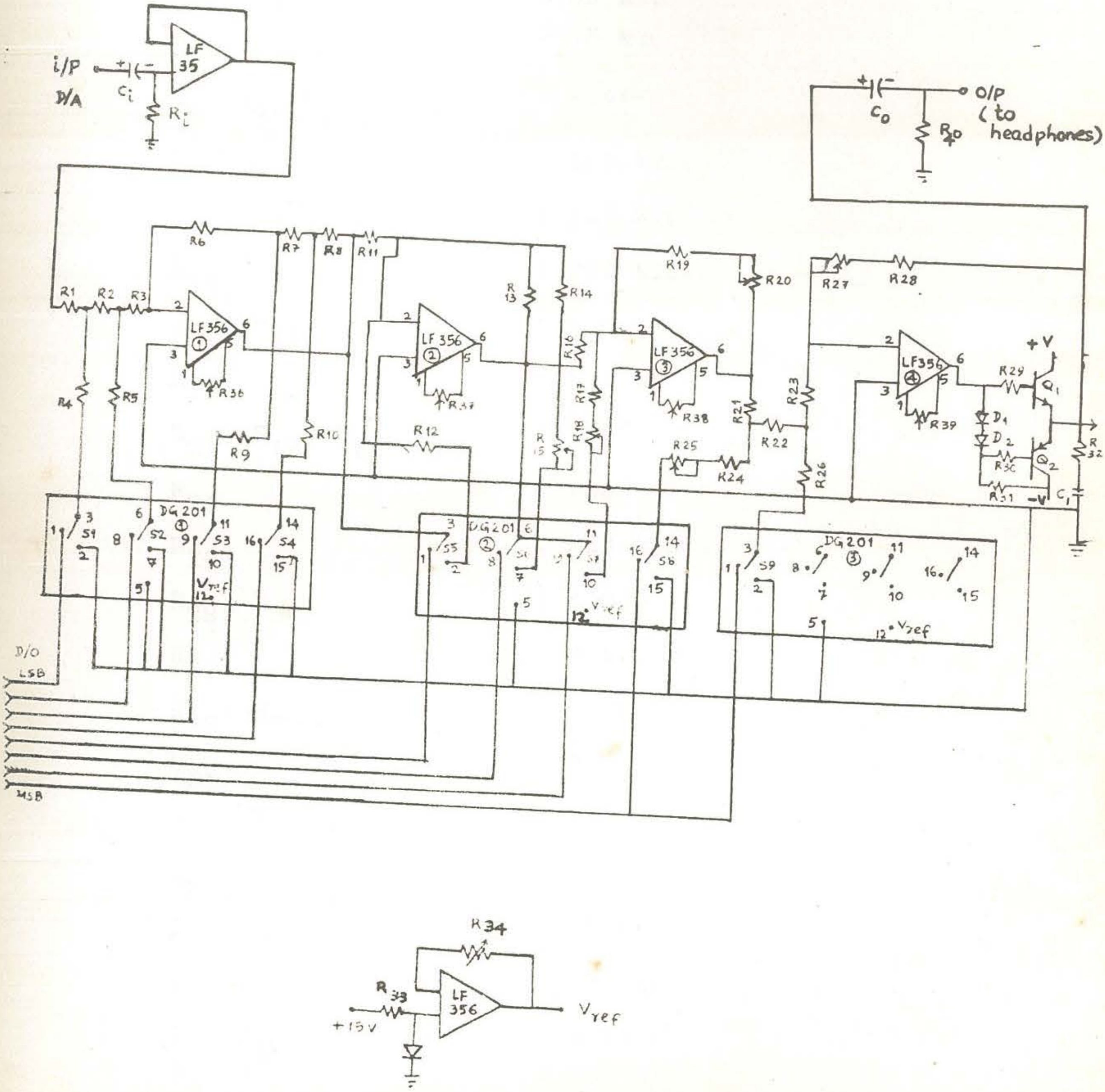


Fig. 4.1. Circuit diagram of the computer controlled attenuator (details on the next page). Adapted from, Severns (1985)

R_1, R_2, R_3, R_7, R_{10}, R_{31}	: 3.3kΩ
R_4, R_{13}	: 39 kΩ
R_5	: 18 kΩ
R_6, R_8, R_{26}	: 1 kΩ
R_9	: 1.5 kΩ
R_{11}, R_{34}, R_i	: 100 kΩ
R_{16}	: 247 kΩ
R_{14}, R_{17}	: 6.8 kΩ
R_{19}	: 5.6 kΩ
R_{21}, R_{12}, R_{22}, R_{23}	: 68 kΩ
R_{24}	: 780Ω
R_{28}	: 200 kΩ
R_{29}, R_{30}	: 470Ω
R_{32}	: 2.5Ω
R_{36}- R_{39}	: 25 kΩ
R_{25}	: 250Ω
R_{15}, R_{18}	: 500Ω
R_{20}, R_{40}	: 1 kΩ
R_{33}	: 10 kΩ
R_{27}	: 20 kΩ
C_1, C_i	: 0.1μF
C_0	: 100 μF
D_1	: 1N 914
Q_1	: 40409
Q_2	: 40410

Pins 7 and 4 of LF 356 are connected to + 15 V and -15 V respectively. Pins 4 and 13 of DG 201 are connected to -15 V and +15 V respectively.

Table 4.1 Switches and their individual attenuations

Switch	Attenuation (dB)	
	1(OFF)	0(ON)
S_8	0	64
S_7	32	0
S_6	0	16
S_5	8	0
S_4	2	0
S_3	4	0
S_2	0	1
S_1	0	0.5

Table 4.2 Digital words for various levels of attenuation.

Digital word	Attenuation (dB)
1 0 1 0 0 0 1 1	0
1 0 1 0 0 1 1 1	5
1 0 1 1 1 0 1 1	10
1 0 1 1 1 1 0 1	15
1 0 0 0 0 1 1 1	20
1 0 0 1 0 0 0 1	25
1 0 0 1 1 1 1 1	30
1 1 1 0 1 0 0 1	35
1 1 1 1 0 0 1 1	40
1 1 1 1 0 1 0 1	45
1 1 0 0 1 0 1 1	50
1 1 0 0 1 1 0 1	55
1 1 0 1 0 1 1 1	60
0 0 1 0 0 0 0 1	65
0 0 1 0 1 1 1 1	70
0 0 1 1 1 0 0 1	75
0 0 0 0 0 0 1 1	80
0 0 0 0 0 1 0 1	85
0 0 0 1 1 0 1 1	90
0 0 0 1 1 1 0 1	95
0 1 1 0 0 1 1 1	100
0 1 1 1 0 0 0 1	105
0 1 1 1 1 1 1 1	110

CHAPTER 5

SYSTEM IMPLEMENTATION

5.1 INTRODUCTION

The setup of the PC-based audiometry system, developed in this project, has been described in the previous two chapters. The program "PC-AUD" is written to implement the three types of pure-tone audiometry. Appendix C provides the listing of this program. The program runs on an IBM PC/XT/AT or compatible machine with PCL 208 Peripheral. All the values of thresholds used for results presentation and audiogram display are arbitrarily chosen. At startup, the program asks for the subject data, and these data are stored in a file. The next screen provides the choices for the type of audiometry :

1. Operator controlled (Conventional)
2. Automated
3. Subject controlled (Bekesy)

Depending upon the choice, different screens appear.

5.2 OPERATOR CONTROLLED AUDIOMETRY

When the operator controlled type is chosen, the program asks for the ear to be tested. At this stage, the screen appears as shown in Fig. 5.1. Successive screens seek the selection of frequency and intensity to be made. Any invalid entry is rejected with a short beep. When valid entries are made, the stimulus is presented to the subject through the earphones. The operator's task now is to observe the subject for his response and enter it. He is given the options of choosing other intensity, frequency or ear in the order. An attempt to quit the ear selection option terminates the testing procedure. The values of the parameters, after being stored, are displayed on the screen along with the subject data as shown in Fig. 5.2. Along with the presentation of these results, the screen asks if the operator wishes to obtain an audiogram. An entry of `Y', clears the results and displays the audiogram. The audiograms for both the ears are drawn on the same screen with different symbols, as shown in Fig. 5.3. Any key press quits the audiogram display and enquires if another session has to to be carried out. Entry of `Y ' starts another round of testing. The whole procedure repeats, starting from the entry of the patient data. Entry of `N' in response to the question "Another session (Y/N)", quits the testing.

5.3 AUTOMATED AUDIOMETRY

To implement the automated type of audiometry, the operator has to enter `2' as the choice for the type of audiometry. Once chosen, this type does not require the active participation of the operator, till the threshold determination for both the ears is complete. The left ear is chosen first and is tested for the frequencies in the ascending order from 125 Hz to 8000 Hz. At each frequency, the initial value of the intensity of the stimulus is set at 30 dB. The stimulus is presented to the subject through the earphones. The screen indicates that the stimulus is being presented and seeks the response of the subject, as shown in Fig.

5.4. The program monitors the subject response and adjusts the intensity accordingly. By following the adaptive algorithm, the threshold is determined or the testing at that frequency is terminated because the maximum intensity has been reached. The program checks the left ear for the remaining frequencies and then the right ear for all the frequencies. When the testing of both the ears is complete, the results of the test are displayed on the screen, as shown in Fig. 5.5. The operator is now required to respond to the question "Wish to obtain the audiogram (Y/N)". If `Y´ is entered, the audiograms for both the ears appear on the same screen but are differentiated by using different symbols to plot them, as in Fig. 5.6. Any key press leads to the option of another session, as in the previous case. The audiogram display can be bypassed by typing `N´ in response to that question.

5.4 SUBJECT CONTROLLED AUDIOMETRY

If the subject controlled (Bekesy) type is selected, the screen, prepared for the tracking of the subject´s threshold, appears as in Fig. 5.7. The parameter selection and threshold determination are identical as in the automated type but in this case, each intensity selected is plotted on the screen. This plotting along with the question for the subject "Did you hear (Y/N)" appears on the screen. After the completion of the test, the results are displayed on the screen. This type of testing is important if it is required to know the sequence of how the intensity has been varied in order to reach the threshold.

5.5 TEST RESULTS

The results of the audiometry tests performed on subjects along with their data are stored in files for further analysis. The storage of information is done in separate files for different subjects. Fig. 5.2 shows the presentation of results in the operator controlled type. In the automated and the subject controlled types, the results are presented as shown in Fig. 5.5.

Choose the ear to be tested:

1. Left ear

2.Right ear

Your choice(1,2): █

Fig.5.1 Screen seeking the choice of the ear in operator controlled audiometry

RESULTS

Name : AA
Sex : M
Date : 06-26-1991
Type : Operator-Controlled(Conventional)
Comments :

Id.No : A-531
Operator :
Time : 21:33:06

Age : 62

Left Ear: Frequency (Hz)	Left Ear: Threshold (dB)	Right Ear: Frequency (Hz)	Right Ear: Threshold (dB)
125	30	125	35
250	35	250	30
500	25	500	20
1000	20	1000	25
1500	25	1500	25
2000	30	2000	30
3000	30	3000	35
4000	35	4000	25
6000	35	6000	30
8000	30	8000	25

Wish to obtain the audiogram?

Fig.5.2 Results presentation in operator controlled audiometry

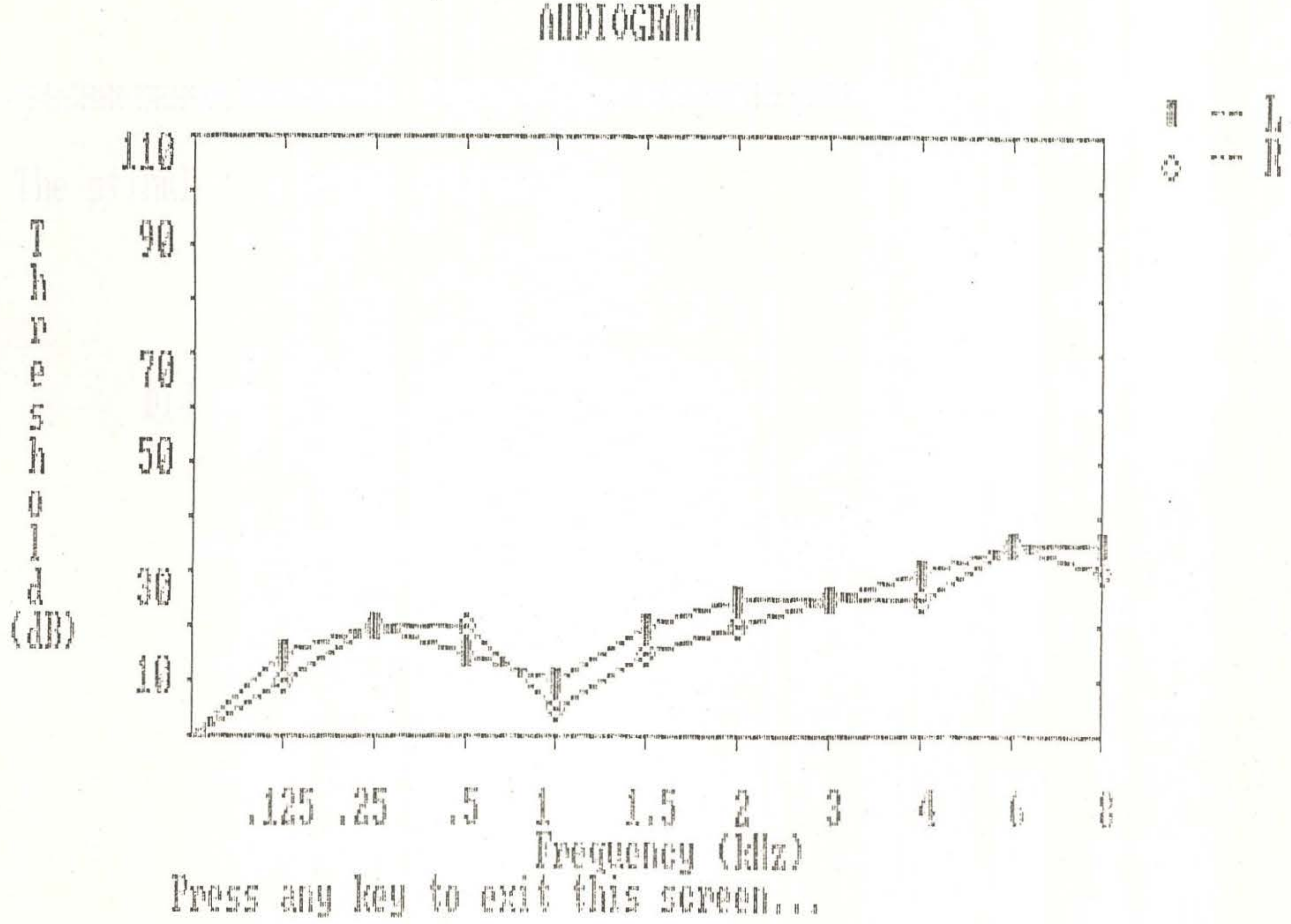


Fig.5.3 Audiogram display in operator controlled audiometry

SUBJECT TERMINAL

The stimulus is being presented....

Did you hear?

Fig.5.4 Screen seeking the subject response in automated audiometry

RESULTS

Name : AA
Sex : M
Date : 06-26-1991
Type : Automated
Comments :

Id.No : A-713
Operator :
Time : 21:06:30

Age : 62

Frequency(Hz)	Threshold(dB) L	Threshold(dB) R
125	10	15
250	20	15
500	15	10
1000	10	5
1500	20	15
2000	15	20
3000	20	25
4000	25	25
6000	30	35
8000	25	30

Wish to obtain the audiogram?

Fig.5.5 Results presentation in automated and subject controlled audiometry

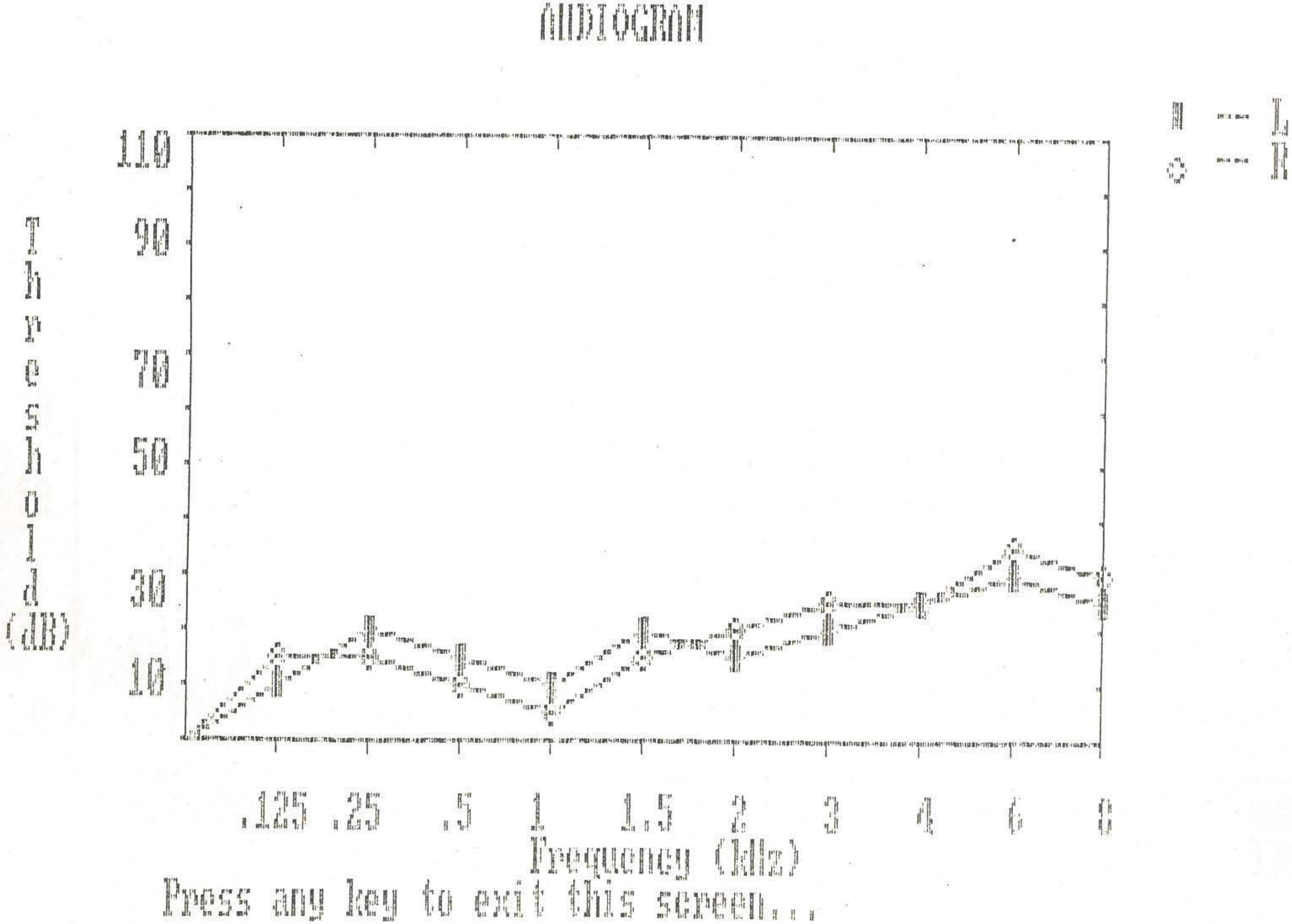


Fig.5.6 Audiogram display in automated and subject controlled audiometry

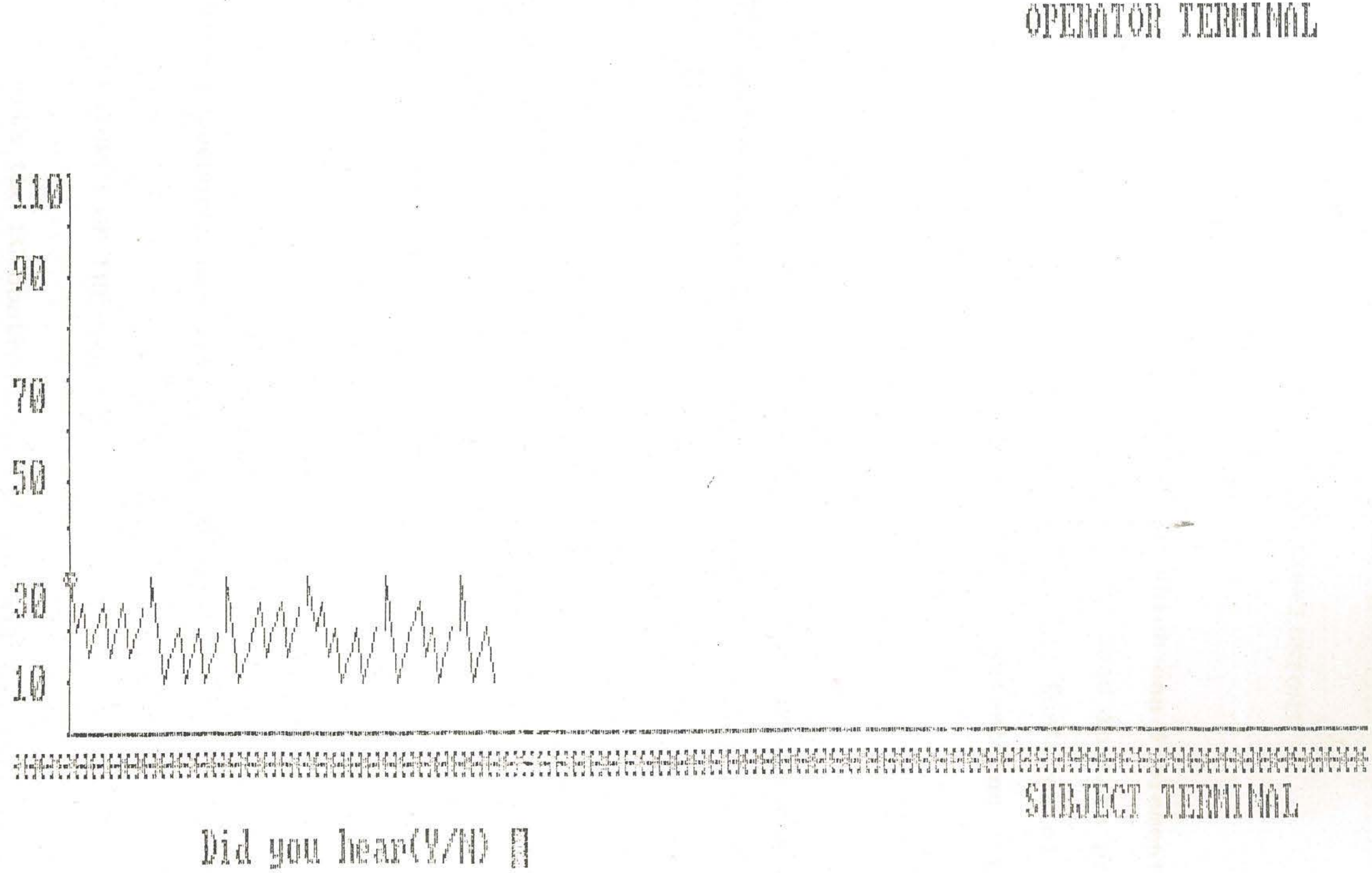


Fig.5.7 Screen tracking the threshold in subject controlled audiometry

CHAPTER 6

SUMMARY AND CONCLUSIONS

6.1 INTRODUCTION

A PC-based audiometry system which can implement three types of pure-tone audiometry has been developed in this project. Here, the work done will be summarised and the features of the system will be discussed. Finally, some suggestions for further improvements in the system will be made.

6.2 WORK DONE

A scheme for the PC-based audiometry system was worked out. The system consists of a PC with a data acquisition peripheral, and an audio attenuator-amplifier unit. The PC serves as the overall controller. The DAP is programmed for generation and intensity control of the stimulus, and subject response monitoring. The audio attenuator-amplifier unit is responsible for the stimulus intensity variation under the DAP control. The computer controlled audio attenuator-amplifier unit was built based on the circuit developed by Severns, 1985. The system performs three types of pure-tone audiometry operator controlled, automated, and subject controlled. The software necessary to implement these procedures was written in GW-BASIC.

6.3 FEATURES OF THE SYSTEM

With the PC controlling the overall testing procedure in the automated and the subject controlled types, the subjectivity of the operator is minimized. These two types are still subjective in the sense that they require the active co-operation of the subject. The system also provides the conventional type of audiometry, if an application demands it. If in any case, it is required to know the manner in which the threshold has been reached, the subject controlled (Bekesy) type can be selected.

For the threshold determination, the system follows the algorithm based on the one recommended by the American Speech and Hearing Association (ASHA) and adopted by the American National Standards Institute (ANSI, 1978). But specific testing protocols vary among institutions. And also the protocols may require special adaptation to accommodate certain physical or psychological limitations in certain cases. Since the system developed here depends on the software for the implementation of the algorithm, it can be easily adapted to suit a specific need. This increases the flexibility of the system, which is one of the major advantages of the software-based systems.

The system checks the input of the data, for any obvious errors. The presentation of results is also clear so as to minimize any possibility for misinterpretation. Because the system uses the PC, the data storage and retrieval are easy. The data are stored in separate files for each subject. The means of interaction with the system is made simple by providing the hints for the entries.

6.4 SUGGESTIONS FOR FURTHER WORK

The system as implemented now detects the hearing loss associated with air conduction. The output circuit can be modified to provide a higher power output to stimulate a bone conduction vibrator. The system can then perform pure-tone audiometry by bone conduction as well. Pure-tone audiometry by air conduction and bone conduction provides more diagnostic information in terms of the site of pathology. Now that the system has been tested for one channel, it can be easily extended to a two channel system. With a two channel system, masking tone can be introduced in the non-test ear.

After proper calibration of the system over the whole frequency range to provide accurate intensity,it can be evaluated by audiologists on hearing impaired subjects. The feedback thus obtained can be used for further improvements in the system. By using digitized speech waveforms as the stimulus and modifying the software accordingly, the system can be used to implement speech audiometry, with the same hardware setup.

The effect of presbyacusis can be incorporated in the audiogram display as reported by Coleridge Smith & Scurr (1988), to give a ready indication of any pathology responsible for the hearing loss. The system can be made more user-friendly by making the procedure entirely menu-driven.

APPENDIX A

OVERVIEW OF HEARING

A.1 INTRODUCTION

The sense organs, in general, serve as receptors of information from the environment. The sense of hearing, in particular, responds to the acoustic stimuli over a frequency range of 20 Hz to 20000 Hz. This appendix deals with the mechanism, impairments, and thresholds of hearing and the frequency sensitivity of the ear.

A.2 MECHANISM OF HEARING

Anatomically, the ear is divided into the external, middle and the inner ears [Ross & Wilson, 1986] as shown in Fig. A.1. The external ear comprises the auricle and the external acoustic meatus. The auricle is the portion of the ear which projects from the side of the head. The external acoustic meatus is a tube from the auricle to the ear drum, which is a membrane separating the external acoustic meatus from the middle ear. The middle ear is an irregular shaped cavity situated onthe medial side of the ear drum and is filled with air. The presence of air at atmospheric pressure on both sides of the ear drum enables it to vibrate when sound waves strike it. The medial wall of the middle ear cavity has two openings - an oval window and a round window. Three small bones (the malleus, incus and stapes) known as the auditory ossicles form movable joints with the ear drum, with each other and with the oval window. The handle of the malleus is in contact with the medial wall of the ear drum, and the base of the incus fits into the oval window. The handle of the malleus is constantly pulled inward by the tensor tympani muscle. This keeps the ear drum under tension so that sound vibrations striking any portion of the ear drum are transmitted to the malleus.

The inner ear has two parts- the bony labyrinth and the membranous labyrinth. The bony labyrinth is a cavity in the temporal bone which houses the smaller membranous labyrinth. The bony walls and the membranous tube are separated by a fluid known as perilymph. The membranous labyrinth is filled with a fluid called endolymph. The cochlea is a system of three tubes (scala vestibuli, scala media and scala tympani) coiled together. Scala vestibuli and scala tympani are filled with perilymph, and scala media with endolymph. The scala media and scala tympani are separated by the basilar membrane. The surface of the basilar membrane contains a series of mechnically sensitive hair cells. The hair cells along with their nerve fibres form the true organ of hearing known as the Organ of Corti. These nerve fibres combine to form the auditory nerve which reaches the hearing area of the cerebral cortex.

The auricle concentrates and directs the sound waves through external acoustic meatus towards the ear drum [Guyton, 1971]. The ear drum vibrates in harmony with the sound waves. The vibrations set up in the ear drum are conveyed through the middle ear by the three auditory ossicles. As the ear drum vibrates, a corresponding movement takes place in the malleus. The movements of the malleus are transmitted via the incus to the stapes. The base of the stapes then rocks to and fro in the oval window. This sets up wave motion in the perilymph. The ear drum and the ossicular system cause approximately 22 times as much pressure on the perilymph as that exerted by the sound waves against the ear drum. Since fluid has greater inertia than air, greater amount of pressure is needed to cause vibration in the fluid. Thus the ear drum and the ossicles provide impedance matching between the sound waves in air and the sound vibrations in the fluid of the cochlea. The vibrations of the membranous labyrinth stimulate movement of the endolymph in the scala media. The hair cells of the Organ of Corti are then stimulated by the sound vibrations and hence generate nerve impulses. These nerve impulses are transmitted through the auditory nerve to the hearing area of the cerebral cortex, where they are perceived as sound.

A.3 HEARING IMPAIRMENTS

Hearing impairment can be defined as any significant loss of sensitivity of the ear, which may be of moderate to long duration [Gulick, 1971]. Hearing impairments can be classified by their severity, duration, or more commonly by the site of pathology in the auditory system. Based on the site of pathology, the hearing impairments fall in three major categories. They are conductive hearing loss, sensorineural hearing loss and mixed hearing loss.

For the purpose of assessment of hearing, the ear can be divided into the conductive portion, consisting of the outer and the middle ears and the sensorineural portion, consisting of the inner ear and the auditory nerve as shown in Fig. A.1. Normally the sound is heard as a result of air conduction, i.e., transmission of sound vibrations to the inner ear through the external ear, ear drum and the middle ear bones. An alternative path, bone conduction, also exists, in which case the sound vibrations are directly coupled from the skull to the inner ear.

Some amount of hearing loss results due to the perforation of the ear drum or pathological changes in the middle ear that attenuate the vibrations of the ossicular system [Webster, 1988]. If the sensorineural portion of the ear is normal, the sound introduced by bone conduction is heard directly by the sensorineural mechanism. This type of hearing impairment which arises due to impaired air conduction is called conductive hearing loss. It is medically treatable, and may in some cases resolve on its own.

Due to the pathology of the cochlea and the associated nerves, both air conduction and bone conduction result in same amount of hearing loss. This type of hearing impairment is called sensorineural hearing loss. It is not medically treatable, and depending on the etiology, may become progressively worse, over time. Such hearing loss may be caused by basilar membrane discontinuity, vascular deterioration or hair cell and supporting structure pathology.

Abnormality in the both the conductive and the sensorineural portions of the ear results in hearing loss by bone conduction and an even greater loss by air conduction. Such a hearing impairment is termed as mixed hearing loss. Fig. A.2 illustrates these three clinical conditions.

In contrast to the above mentioned losses, functional loss is the apparent loss of hearing sensitivity without any organic pathology to explain the loss [Martin 1986].

A.4 HEARING THRESHOLDS

The human ear is sensitive to an enormous range of pressures (about 100000 fold) [Gulick, 1971]. This fact makes it more convenient to express sound intensity in terms of decibels than it is to express it directly in pressure units. The advantage of the decible scale is that it offers a means to express a large pressure range on a conveniently abbreviated scale. The sound intensity in decibels is defined as

$$N = 20 \log \frac{P}{P_o}$$

where N is the number of decibels of the sound intensity, P is the corresponding sound pressure and P_o is the reference sound pressre.

A standard, now widely accepted, reference sound pressure is 0.0002 dyne/cm^2. This value was chosen because it approximates the least pressure required for the average human listener to hear a tone of 1000 Hz. Intensities expressed in decibels using this reference are called sound pressure levels (SPL).

The other means of expressing intensity is to specify sound pressure at a particular frequency relative to the pressure at absolute threshold for that frequency. Since the ear is not equally sensitive to all the frequencies, the hearing threshold varies as a function of frequency. Therefore a different reference pressure is used for each frequency. This scale is known as dB HL (hearing level) scale. It allows normal hearing to be operationally defined as a straight line at `0´ dB HL.

In pure-tone audiometry, the threshold is defined as the minimum sound intensity which is heard by an individual in 50% of the trials. The threshold is measured and an audiogram is plotted, so that the threshold for any frequency is indicated as a certain number of decibels of hearing loss or hearing gain below or above the normal average.

In speech audiometry, speech detection threshold (SDT) is defined as the minimum intensity at which speech is just detected but not understood. The usually measured parameter is the more important speech reception threshold (SRT). SRT is defined as the minimum intensity at which speech can be understood. It is determined as the level at which atleast half of the spondees are correctly repeated. The SRT should be within 5-10 dB of the average of the air-conducted pure-tone thresholds for the middle frequencies (500 Hz, 1000 Hz and 2000 Hz) [Webster, 1988]. The other two parameters of interest are the most comfortable loudness level (MCL) and the uncomfortable loudness level (UCL). MCL is defined as the intensity of sound designated by the listener as the most comfortable listening level for speech. It lies in the range of 40-55 dB above SRT in persons with normal hearing. UCL is defined as that level of sound at which speech becomes uncomfortably loud. It is also called the threshold of discomfort or the tolerance level. For persons with normal hearing, UCL extends to about 100 to 110 dB.

A.5 FREQUENCY SENSITIVITY

An unimpaired human ear is sensitive to sounds in the frequency range of 20 Hz to 20000 Hz. The minimum perceptible frequency change by a listener is termed as the differential frequency discrimination. It can either be expressed as the absolute difference (Δf) between the two frequencies, or as the relative difference ($\Delta f/f$). Though the duration of the tones and the time interval between them have an effect upon Δf, the two overriding factors that influence differential sensitivity are the frequency and intensity of the tones. At an intensity of 40 dB HL, Δf is above 3 Hz for frequencies of 125 Hz to 2000 Hz. At the same intensity it increases to 12 Hz by 5000 Hz, by 10000 Hz to 30 Hz and by 15000 Hz to 187 Hz [Gulick, 1971]. This clearly shows that man's discrimination of higher frequencies is poor. The ability of discrimination between sounds of different frequencies improves with the gradual increase in the sound intensity. The range of frequencies to which the human ear is most sensitive extends from 1000 Hz to 3000 Hz with a minimum perceptible fractional difference of 0.3% [Keele et al., 1988].The ability of frequency discrimination is degraded in sensorineural hearing impairments and hence its determination is of diagnostic importance.

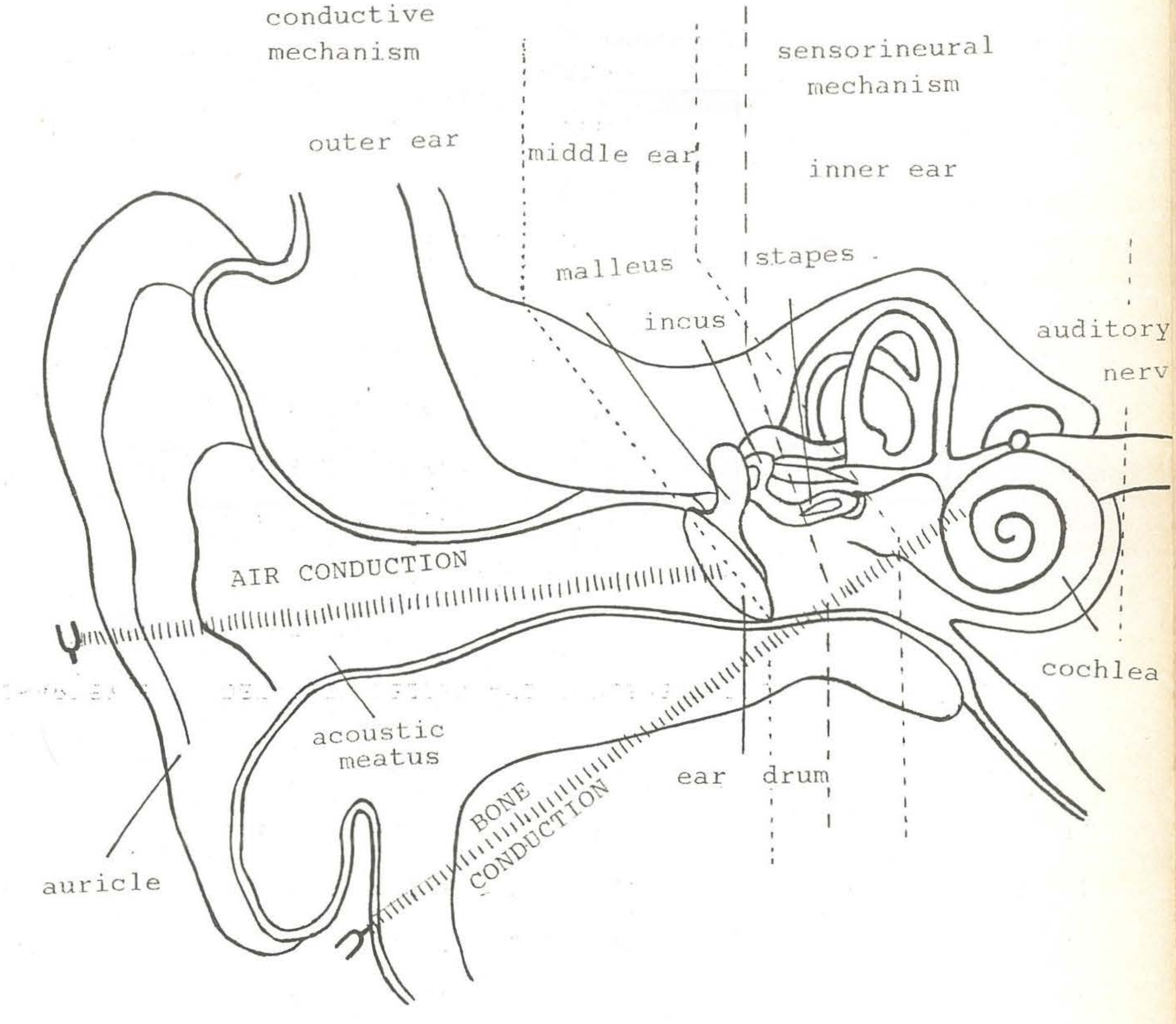


Fig.A.1 Functional anatomy of the ear

Adapted from, Brooks (1986)

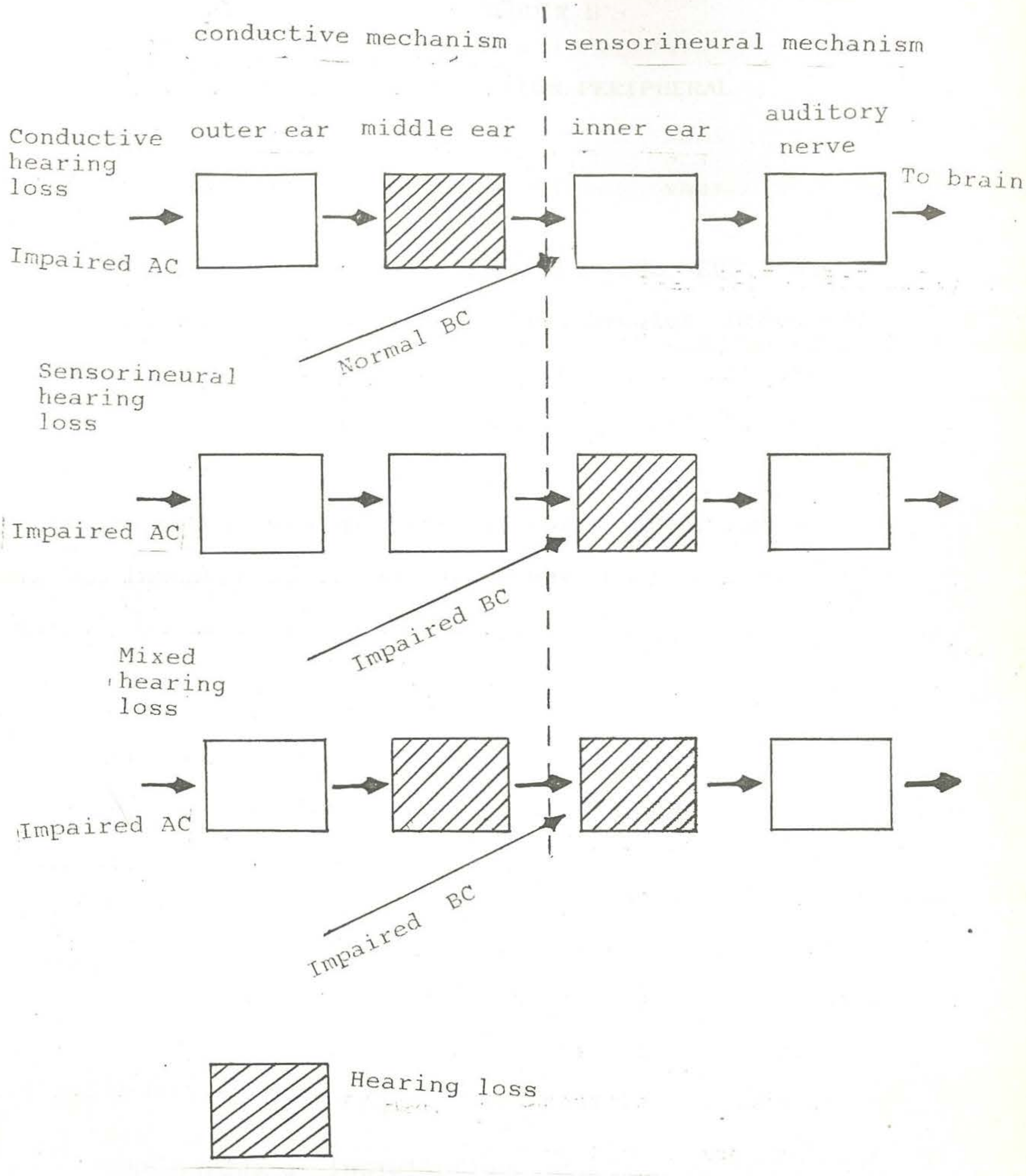


Fig.A.2 Block diagram illustrating types of hearing impairments

Source : Martin (1986)

APPENDIX B

DATA ACQUISITION PERIPHERAL

B.1 INTRODUCTION

The Data Acquisition Peripheral PCL 208, manufactured by Advantech Co. Ltd., and marketed by Dynalog Micro Systems, has been utilized for the implementation of the PC-based audiometry system. PCL 208 is a high performance, high speed multifunction data card for the IBM PC/XT/AT or compatibles. The associated software makes the card suitable for a wide range of applications in the industrial and laboratory environment, like data acquisition, process control, automatic testing and factory automation.

PCL 208 performs digital-to-analog conversion, analog-to-digital conversion, digital input, and digital output and includes a pacer clock for accurate timing of the operations. This appendix describes the features and basic concepts of PCL 208 which are necessary to understand its utility in the present project. Further details about the card can be obtained from the User's Manual of PCL 208.

B.2 INSTALLATION

The hardware installation of PCL 208 requires an unused expansion slot in the rear unit of the computer where the board can be inserted. The software installation involves the loading of PCL 208 driver routines into memory. The PCL 208 I/O driver is written in assembly language and compiled into a machine language program called "PCL208.BIN". Depending upon the memory space available in the PC, the driver can be loaded in two ways :

(i) The first method is loading the machine language driver immediately after the normal BASIC work space area. This method requires the calculation of free bytes available in the BASIC work space area and the starting address of the driver routine.

(ii) The second method is loading PCL208.BIN to a memory segment which is independent of the BASIC work area. Because the driver is located in an area outside the BASIC work area, a DEF command should be used to link the driver immediately before each function.

Once loaded, the driver routines can be easily accessed by the BASIC CALL statement, which has the following format :

CALL PCL208 (FUN%, DAT%(0), ER%)

PCL208 : It is a variable specifying the memory offset of the starting address of the PCL 208 driver routine from the most recently defined current segment. In all our applications, it is loaded from the begining of a memory segment so that PCL 208 is always set to 0.

FUN% : It is a variable indicating the driver function to be executed. The PCL 208 can carry out 22 different functions. The software developed in this project makes use of FUNCTION 0,7,14,17,18 and 21.

DAT%(0) : This variable specifies the starting location of the entire data array.

ER% : The return message code is contained in this variable. The ER% value is always set to 0 which indicates good return. Any other value of ER% halts the program.

B.3 BASE ADDRESS

The operation of PCL 208 is controlled through the input and the output ports. The I/O port address space is used to address these ports. The PCL 208 requires 16 consecutive address locations in I/O space. The factory setting for base address (BASE) is hex 300. It is selectable over the valid range from hex 000 to hex 3F0.

B.4 DIGITAL-TO-ANALOG CONVERSION

The PCL 208 provides two analog output channels each using a 12-bit multiplying D/A converter. The digital data which are to be converted into the analog form has to be written into the D/A registers which use the address locations BASE+4, BASE+5, BASE+6 and BASE+7. The D/A channel-0 uses BASE+4 and BASE+5. BASE+6 and BASE+7 are allocated for D/A channel-1.

The least significant byte of the data is written first and is temporarily held by a register in the D/A converter. After the most significant byte is written, the low and high bytes are simultaneously passed on to the D/A converters. The specifications of the D/A converter are given in Table B.1.

Connector #2 is used for analog output and its pin assignment is as shown in Fig. B.1. The D/A signal connections are shown in Fig. B.2. The PCL 208 driver FUNCTION 18 is used in our program to generate the stimulus waveforms which are sinusoids in the frequency range of 125 Hz to 8000 Hz.

The analog output is unipolar and extends from 0 to $-V_{ref}$, where V_{ref} is the reference voltage. If the on-board fixed reference voltage (-5V) is used, the analog output ranges from 0 to +5V. Any external reference voltage between -10V and +10V can be used.

B.5 DIGITAL OUTPUT AND INPUT OPERATIONS

PCL 208 provides 16 digital output (D/O) channels and 16 digital input (D/I) channels. These channels use the output and input registers at BASE+3 and BASE+11. The low byte of the D/O data is written into BASE+3 and the high byte into BASE+11. The specifications are given in Table B.2.

Connector #3 is used for the digital output operation. Its pin assignment is given in Fig. B.3. In our program, FUNCTION 21 of the PCL 208 driver is used to generate the digital word indicating the level of attenuation. Only the least significant 8 bits are used as they are sufficient to specify the required attenuation range.

The digital input operation is utilized for the subject response monitoring in the program. The least significant bit is checked each time. If it is `1´, the response is taken as `NO´. Otherwise it is taken as `YES´. Connector #4 is used for digital input operation. Its pin assignment is given in Fig. B.4. The specifications are given in Table B.3. FUNCTION 14 of PCL 208 is used to read the digital input and monitor the subject response.

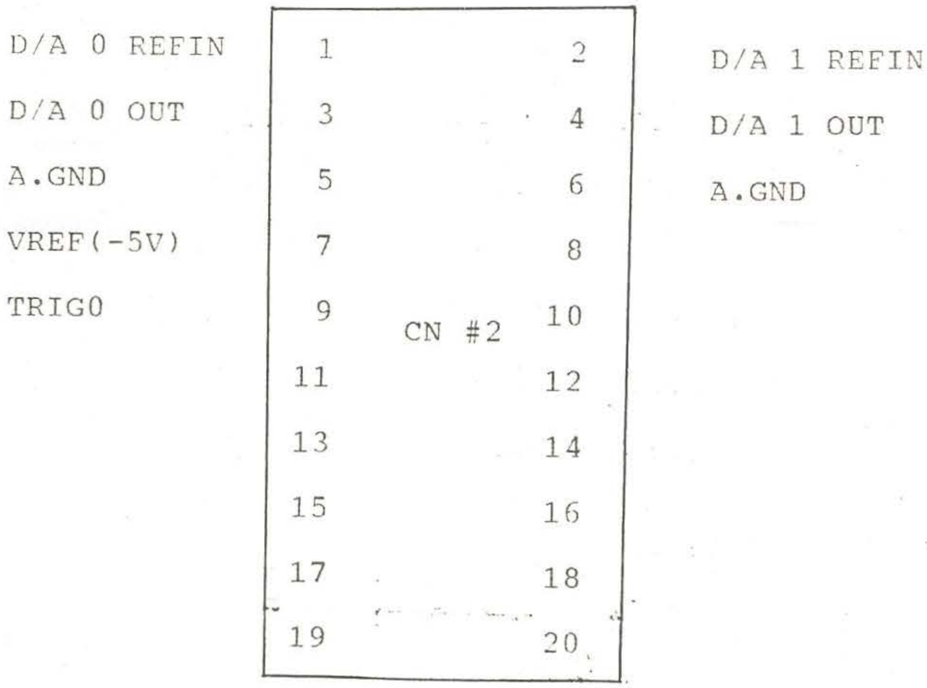


Fig.B.1 Pin assignment of Connector # 2

Source : PCL 208 User's Manual

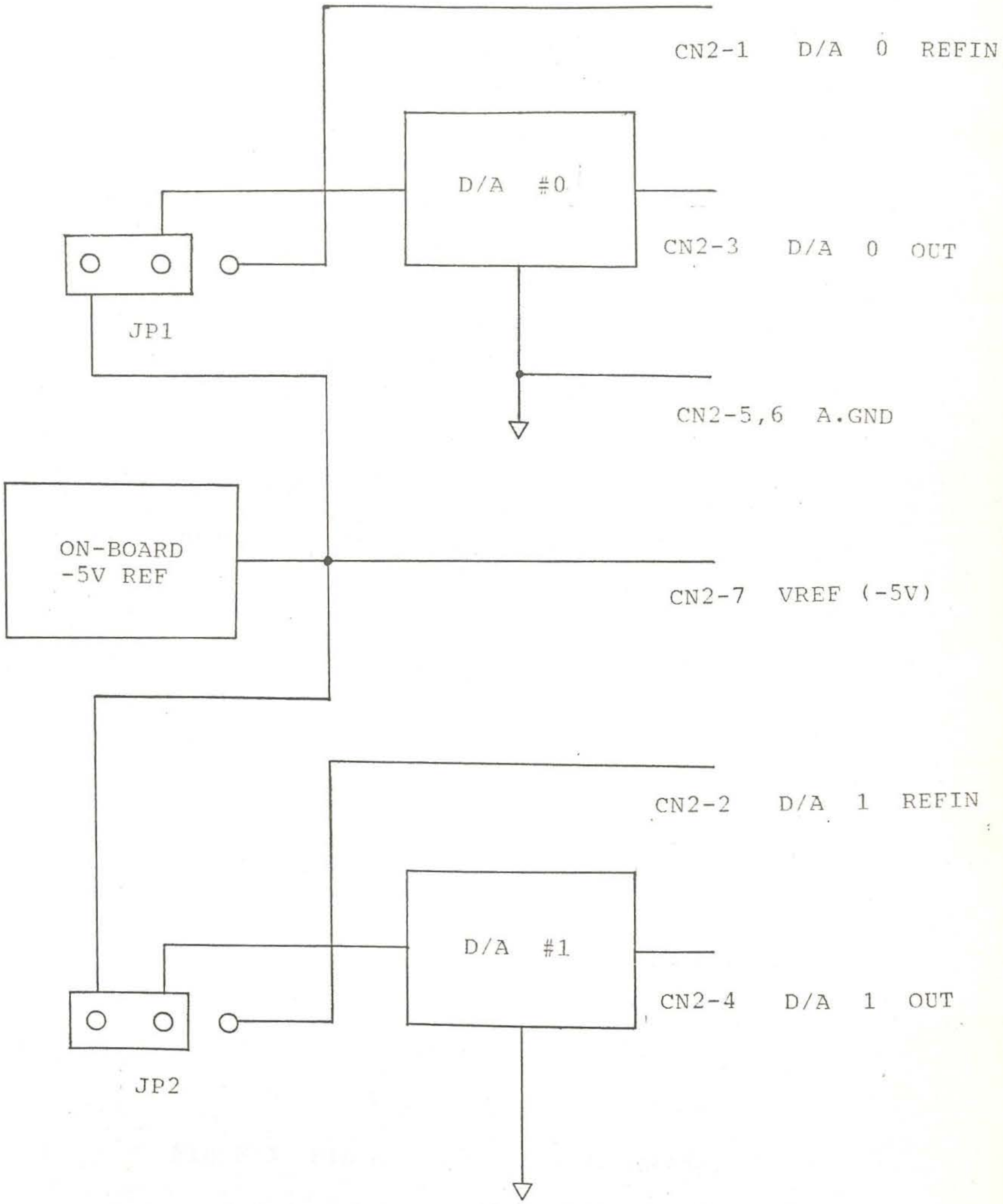


Fig.B.2 D/A converter signal connections

Source : PCL 208 User's Manual

CN #3

Signal	Pin	Pin	Signal
D/O 0	1	2	D/O 1
D/O 2	3	4	D/O 3
D/O 4	5	6	D/O 5
D/O 6	7	8	D/O 7
D/O 8	9	10	D/O 9
D/O 10	11	12	D/O 11
D/O 12	13	14	D/O 13
D/O 14	15	16	D/O 15
D.GND	17	18	D.GND
+ 5V	19	20	+ 12V

Fig.B.3 Pin assignment of Connector # 3

Source : PCL 208 User´s Manual

CN#4

D/I 0	1	2	D/I 1
D/I 2	3	4	D/I 3
D/I 4	5	6	D/I 5
D/I 6	7	8	D/I 7
D/I 8	9	10	D/I 9
D/I 10	11	12	D/I 11
D/I 12	13	14	D/I 13
D/I 14	15	16	D/I 15
D.GND	17	18	D.GND
+ 5V	19	20	+ 12V

Fig.B.4 Pin assignment of Connector # 4

Source : PCL 208 User's Manual

Table B.1 : Specifications of D/A Converter

Source : PCL 208 User's Manual

Channels	:	2 channels
Resolution	:	12 bits
Output Range	:	0 to + 5V with on-board - 5V reference. +/- 10V maximum with external DC or AC reference.
Reference	:	Internal - 5V (+/- 0.05V) or external DC or AC, +/- 10V maximum.
Conversion type	:	12 bit monolithic multiplying
Linearity	:	+/- 1/2 bit
Output drive	:	+/- 5 mA maximum
Settling time	:	5 microseconds

Table B.2 : Specifications of D/O

Source : PCL 208 User's Manual

Channel	:	16 bits
Level	:	TTL compatible
Output voltage	:	Low - Sink 8 mA at 0.5V maximum
		High - Source - 0.4 mA at 2.4V minimum

Table B.3 : Specifications of D/I

Source : PCL 208 User´s Manual

Channel	:	16 bits
Level	:	TTL compatible
Output voltage	:	Low - 0.8V maximum
		High - 2.0V minimum
Input Load	:	Low - 0.4mA maximum at 0.5V
		High - 0.05mA minimum at 2.7V

APPENDIX C

PROGRAM LISTING OF "PC-AUD"

The program "PC-AUD" implements the three modes of pure-tone audiometry: operator controlled, automated and subject controlled. The program requires GW-BASIC interpreter and the driver routines of the data acquisition peripheral PCL 208. The specific driver routines include FUNCTIONs 0,7,14,17,18 and 21. The execution of "PC-AUD" requires any IBM PC/XT/AT with the DAP, PCL 208.

```
10 '******************** PROGRAM : PC-AUD ***********************************
20 '*********** AUTHOR : V.MALINI, MTP'91 ,SBME ,I.I.T. BOMBAY **************
30 '*********************** DATE : 31-07-1991 *******************************
40 '*** PROGRAM DESCRIPTION : PC-AUD IMPLEMENTS THREE MODES OF PURE-TONE ***
50 '*** AUDIOMETRY: OPERATOR CONTROLLED, AUTOMATED, & SUBJECT CONTROLLED ***
60 '*** THIS PROGRAM RUNS ON ANY PC WITH THE DATA ACQUISITION PERIPHERAL ***
70 '*** PCL 208. IT REQUIRES  GW-BASIC INTERPRETER & THE DRIVER ROUTINES ***
80 '*************************** OF PCL 208 **********************************
90 KEY OFF :CLS
100 DEF SEG=&H5000
110 BLOAD "pcl718.bin",0
120 DIM DAT%(4)
130 DAT%(0)=&H300
140 DAT%(1)=2
150 DAT%(2)=1
160 PCL718=0
170 ER%=0
180 FUN%=0
190 DEF SEG=&H5000
200 CALL PCL718(FUN%,DAT%(0),ER%)
210 IF ER%<>0 THEN PRINT "failed":STOP
220 '******************* SUBJECT DATA ***************************************
230 CLS:LOCATE 3,5
240 PRINT "PATIENT DATA:"
250 LOCATE 5,5
260 INPUT "Name       : ",N$
270 IF N$="" THEN 3870
280 OPEN "O",#2,N$
290 LOCATE 6,5
300 INPUT"Id. No     : ",I$
310 X$=STRING$(10,0)
320 LOCATE 7,15
330 PRINT X$
340 LOCATE 7,5
350 INPUT"Age        : ",A
360 IF (A<1 OR A>100)THEN SOUND 1500,4:GOTO 310
370 X$=STRING$(10,0)
380 LOCATE 8,15
390 PRINT X$
400 LOCATE 8,5
410 INPUT"Sex  (M/F): ",S$
420 IF S$ ="m" OR S$ = "M" OR S$="F" OR S$="f" THEN 430 ELSE SOUND 1500,4:
    GOTO 370
430 LOCATE 9,5
440 INPUT"Operator  : ",O$
450 D$=DATE$
460 T$=TIME$
470 LOCATE 10,5
480 INPUT"Comments  : ",C$
490 PRINT #2,N$
500 PRINT #2,I$
```

```
510 PRINT #2,A
520 PRINT #2,S$
530 PRINT #2,O$
540 PRINT #2,D$
550 PRINT #2,T$
560 PRINT #2,C$
570 '***************** TYPE SELECTION **********************************
580 CLS:LOCATE 7,20:PRINT "Types of Audiometry:"
590 LOCATE 10,20:PRINT "1. Operator-Controlled (Conventional)"
600 LOCATE 12,20:PRINT "2. Automated"
610 LOCATE 14,20:PRINT "3. Subject-Controlled (Bekesy)
620 X$=STRING$(20,0)
630 LOCATE 20,20
640 PRINT X$
650 LOCATE 22,20
660 PRINT X$
670 LOCATE 18,41
680 PRINT X$
690 LOCATE 18,20
700 INPUT "Your choice (1,2,3): ",M
710 FOR I=1 TO 3
720 IF M=I THEN CLS:GOTO 800
730 NEXT I
740 SOUND 1500,4
750 LOCATE 20,20
760 PRINT "INVALID CHOICE!"
770 LOCATE 22,20:INPUT "Re-choose(Y/N): ",R$
780 INPUT "Re-choose(Y/N): ",R$
790 IF R$="Y" OR R$="y" THEN 620 ELSE END
800 IF M=1 THEN M$="Operator-Controlled(Conventional)
810 IF M=2 THEN M$="Automated"
820 IF M=3 THEN M$="Subject-Controlled(Bekesy)"
830 PRINT #2,M$
840 IF M=3  THEN GOSUB 5100 ELSE  850
850 IF M=1 THEN  5690
860 '******************** EAR SELECTION ********************************
870 E$="L"
880 GOSUB 990
890 LOCATE 13,5
900 IF M=3 THEN LOCATE 22,1:X$=STRING$(80,0):PRINT X$:LOCATE 22,1
910 LOCATE 13,5
920 X$=STRING$(22,0)
930 PRINT X$
940 IF M=3 THEN GOSUB 5100
950 E$="R"
960 IF M=3 THEN GOSUB 5100
970 GOSUB 1070
980 GOTO 3200
990 '******************** FREQUENCY SELECTION ****************************
1000 DATA .125,.25,.5,1,1.5,2,3,4,6,8
```

```
1010 FOR C=1 TO 10
1020 READ F(C)
1030 FTH(C)=F(C)
1040 PRINT #2,FTH(C)
1050 NEXT C
1060 RESTORE
1070 IF E$="l" OR E$="L" THEN P=6 ELSE P=7
1080 Q=15
1090 FOR C=1 TO 10
1100 READ F(C)
1110 GOSUB 1230
1120 IF M=2 THEN 1140
1130 CIRCLE (X,Y),4
1140 FTH(C)=F(C)
1150 IF E$="L" OR E$="l" THEN ITHL(C)=I% :PRINT #2,ITHL(C)
1160 IF E$="R" OR E$="r" THEN ITHR(C)=I% :PRINT #2,ITHR(C)
1170 Q=Q+6
1180 LOCATE P,Q:IF M=3 THEN 1200
1190 IF E$="l" OR E$="L" THEN PRINT ITHL(C) ELSE PRINT ITHR(C)
1200 NEXT C
1210 RESTORE
1220 RETURN
1230 '********************* INTENSITY SELECTION **************************
1240 N=0
1250 K=0
1260 I%=30
1270 IF M=3 THEN GOSUB 5640
1280 GOSUB 1800
1290 GOSUB 3010
1300 IF R$="Y" OR R$="y" THEN GOSUB 1340:GOTO 1330
1310 IF R$="N" OR R$="n" THEN GOSUB 1430:GOTO 1330
1320 SOUND 1500,4:GOTO 1290
1330 RETURN
1340 I%=I%-10
1350 IF M=3 THEN GOSUB 5640
1360 IF I%<0 THEN GOSUB 1760
1370 GOSUB 1800
1380 GOSUB 3010
1390 IF R$="Y" OR R$="y" THEN 1340:GOTO 1420
1400 IF R$="N" OR R$="n" THEN GOSUB 1510:GOTO 1420
1410 SOUND 1500,4:GOTO 1380
1420 RETURN
1430 I%=I%+20
1440 IF M=3 THEN GOSUB 5640
1450 GOSUB 1800
1460 GOSUB 3010
1470 IF R$="Y" OR R$="y" THEN GOSUB 1340:GOTO 1500
1480 IF R$="N" OR R$="n" THEN GOSUB 1640:GOTO 1500
1490 SOUND 1500,4:GOTO 1460
1500 RETURN
```

```
1510 I%=I%+5
1520 IF M=3 THEN GOSUB 5640
1530 IF I%>105 THEN 1740
1540 N=N+1
1550 GOSUB 1800
1560 GOSUB 3010
1570 IF R$="N" OR R$="n" THEN 1510:GOTO 1600
1580 IF R$="Y" OR R$="y" THEN K=K+1:GOTO 1600
1590 SOUND 1500,4:GOTO 1560
1600 IF N<6 THEN 1340
1610 IF K<3 THEN GOSUB 1340
1620 GOSUB 1720
1630 RETURN
1640 I%=I%+10
1650 IF M=3 THEN GOSUB 5640
1660 GOSUB 1800
1670 GOSUB 3010
1680 IF R$="Y" OR R$="y" THEN GOSUB 1340:GOTO 1710
1690 IF R$="N" OR R$="n" THEN GOSUB 1780:GOTO 1710
1700 SOUND 1500,4:GOTO 1670
1710 RETURN
1720 IF E$="L" OR E$="l" THEN T=ITHL(C) ELSE T=ITHR(C)
1730 RETURN
1740 LOCATE 13,5:PRINT "Max. Limit reached. No Response!"
1750 LOCATE 13,5:PRINT STRING$(40,0):RETURN
1760 LOCATE 13,5:PRINT "Min. Limit reached!"
1770 LOCATE 13,5:PRINT STRING$(40,0):RETURN
1780 IF I%>105 THEN GOSUB 1740 ELSE 1640
1790 RETURN
1800 '***************** STIMULUS PRESENTATION SUBROUTINE *****************
1810 F(C)=FTH(C)
1820 L%=50!/F(C)
1830 IF I%=110 THEN DAT%(0)=139:GOTO 1990
1840 IF I%=105 THEN DAT%(0)=172:GOTO 1990
1850 IF I%=100 THEN DAT%(0)=187:GOTO 1990
1860 IF I%=95 THEN DAT%(0)=189:GOTO 1990
1870 IF I%=90 THEN DAT%(0)=156:GOTO 1990
1880 IF I%=85 THEN DAT%(0)=130:GOTO 1990
1890 IF I%=80 THEN DAT%(0)=128:GOTO 1990
1900 IF I%=75 THEN DAT%(0)=160:GOTO 1990
1910 IF I%=70 THEN DAT%(0)=135:GOTO 1990
1920 IF I%=65 THEN DAT%(0)=164:GOTO 1990
1930 IF I%=60 THEN DAT%(0)=144:GOTO 1990
1940 IF I%=55 THEN DAT%(0)=176:GOTO 1990
1950 IF I%=50 THEN DAT%(0)=180:GOTO 1990
1960 IF I%=45 THEN DAT%(0)=148:GOTO 1990
1970 IF I%=40 THEN DAT%(0)=200:GOTO 1990
1980 IF I%=35 THEN DAT%(0)=200 ELSE DAT%(0)=215
1990 DAT%(1)=0
2000 FUN%=21
```

```
2010 ER%=0
2020 DEF SEG=&H5000
2030 CALL PCL718(FUN%,DAT%(0),ER%)
2040 IF ER%<>0 THEN PRINT "failed":STOP
2050 C% = 1
2060 DIV=20
2070 Z=2
2080 WHILE 1=1
2090 CTR1=Z
2100 CTR2=INT(DIV/Z)
2110 IF CTR2 > 65535! THEN Z=Z+50
2120 IF CTR2 < 65535! THEN 2140
2130 WEND
2140 NDIV=CTR1* CTR2
2150 IF CTR1 > 32767 THEN CTR1 = CTR1 - 65536!
2160 IF CTR2 > 32767 THEN CTR2 = CTR2 - 65535!
2170 FUN%=17
2180 DAT%(0)=CTR1
2190 DAT%(1)=CTR2
2200 DEF SEG = &H5000
2210 CALL PCL718(FUN%,DAT%(0),ER%)
2220 IF ER%<>0 THEN PRINT "fail":STOP
2230 DIM  DA%(400)
2240 FOR I =0 TO (L%-1)
2250 DA%(I)=CINT(2048+2047*SIN(2*3.14159*I/(L%/2)))
2260 NEXT I
2270 IF M=3 THEN 2790
2280 LOCATE 1,64
2290 PRINT "OPERATOR TERMINAL"
2300 LOCATE 2,1:X$=STRING$(80,45):PRINT X$
2310 X$=STRING$(80,45)
2320 PRINT X$
2330 LOCATE 3,3
2340 PRINT "Frequency (kHz):"
2350 LOCATE 3,20
2360 PRINT ".125"
2370 LOCATE 3,26
2380 PRINT ".25"
2390 LOCATE 3,32
2400 PRINT ".5"
2410 LOCATE 3,40
2420 PRINT "1"
2430 LOCATE 3,45
2440 PRINT "1.5"
2450 LOCATE 3,52
2460 PRINT "2"
2470 LOCATE 3,58
2480 PRINT "3"
2490 LOCATE 3,64
2500 PRINT "4"
```

```
2510 LOCATE 3,70
2520 PRINT "6"
2530 LOCATE 3,76
2540 PRINT "8"
2550 LOCATE 4,1
2560 PRINT X$
2570 LOCATE 5,3
2580 PRINT "Threshold (dB)
2590 LOCATE 6,17
2600 PRINT "L:"
2610 LOCATE 7,17
2620 PRINT "R:"
2630 LOCATE 8,1
2640 PRINT X$
2650 LOCATE 9,20:PRINT STRING$(5,0):LOCATE 9,2
2660 PRINT "Frequency (kHz)  :";F(C)
2670 LOCATE 10,2
2680 PRINT "Sound level(dB)  :";I%
2690 LOCATE 11,2
2700 PRINT "Ear              : ";E$
2710 LOCATE 15,1
2720 X$=STRING$(80,42)
2730 PRINT X$
2740 LOCATE 16,65
2750 PRINT "SUBJECT TERMINAL"
2760 LOCATE 22,10:PRINT STRING$(50,0)
2770 LOCATE 17,5
2780 PRINT "**** SOUND PRESENTATION ****"
2790 I.MSR%=INP(&H21)
2800 OUT &H21,(I.MSR% OR 1)
2810 FUN%=18
2820 DAT%(0)=C%
2830 DAT%(1)=L%
2840 DAT%(2)=0
2850 DAT%(3)=VARPTR(DA%(0))
2860 DAT%(4)=0
2870 DEF SEG =&H5000
2880 CALL PCL718 (FUN%,DAT%(0),ER%)
2890 IF ER%<>0 THEN PRINT "failed":STOP
2900 FOR I=1 TO 2
2910 GOSUB 7150
2920 NEXT I
2930 'IF INKEY$="" THEN 2545
2940 FUN%=7
2950 DEF SEG=&H5000
2960 CALL PCL718 (FUN%,DAT%(0),ER%)
2970 IF ER%<>0 THEN PRINT "failed":STOP
2980 OUT &H21,IMR%
2990 ERASE DA%
3000 RETURN
```

```
3010 '*************** RESPONSE MONITORING ******************************
3020 X$=STRING$(20,0)
3030 IF M=3 THEN 3050:LOCATE 13,36:PRINT X$:LOCATE 13,8
3040 LOCATE 14,8:IF M=1 THEN INPUT "Is the patient able to hear(Y/N)",R$
    :GOTO 3090
3050 IF M=3 THEN LOCATE 20,1:Y$=STRING$(80,42):PRINT Y$:LOCATE 21,60:
     PRINT "SUBJECT TERMINAL":LOCATE 2,60:PRINT "OPERATOR TERMINAL"
3060 IF M=3 THEN 3080 ELSE LOCATE 17,5:PRINT STRING$(50,0)
3070 DAT%(0)=0
3080 LOCATE 22,30:PRINT X$:LOCATE 22,12:PRINT "Press the button if you
     could hear."
3090 FOR I=1 TO 2
3100 FUN%=14
3110 ER%=0
3120 DEF SEG=&H5000
3130 CALL PCL718(FUN%,DAT%(0),ER%)
3140 IF ER% <>0 THEN PRINT "failed":STOP
3150 IF DAT%(0)=14 THEN R$="y" :GOTO 3190
3160 IF DAT%(0)=15 THEN R$="n":GOTO 3190
3170 GOSUB 7150
3180 NEXT I
3190 RETURN
3200 '*************** RESULTS PRESENTATION *******************************
3210 CLOSE #2
3220 LOCATE 13,5:IF M=3 THEN LOCATE 22,1:X$=STRING$(80,0):PRINT X$:
     LOCATE 22,1
3230 X$=STRING$(20,0)
3240 PRINT X$
3250 INPUT "File Name:",N$
3260 OPEN "I",#2,N$
3270 CLS:LOCATE 2,30
3280 PRINT "RESULTS"
3290 LOCATE 4,2
3300 INPUT #2,N$
3310 PRINT "Name     : ";N$
3320 LOCATE 4,34
3330 INPUT #2,I$
3340 PRINT "Id.No    : ";I$
3350 LOCATE 4,66
3360 INPUT #2,A
3370 PRINT "Age      : ";A
3380 LOCATE 5,2
3390 INPUT #2,S$
3400 PRINT "Sex      : ";S$
3410 LOCATE 5,34
3420 INPUT #2,O$
3430 PRINT "Operator : ";O$
3440 LOCATE 6,2
3450 INPUT #2,D$
3460 PRINT "Date     : ";D$
3470 LOCATE 6,34
3480 INPUT #2,T$
3490 PRINT "Time     : ";T$
3500 LOCATE 8,2
```

```
3510 INPUT #2,C$
3520 PRINT "Comments : ";C$
3530 LOCATE 7,2
3540 INPUT #2,M$
3550 PRINT "Type     : ";M$
3560 LOCATE 9,7
3570 Y$=STRING$(50,45)
3580 PRINT Y$
3590 LOCATE 10,10
3600 PRINT "Frequency(Hz)"
3610 LOCATE 10,30
3620 PRINT "Threshold(dB)"
3630 LOCATE 11,31
3640 PRINT "L"
3650 LOCATE 11,41
3660 PRINT "R"
3670 LOCATE 12,7
3680 PRINT Y$
3690 FOR C=1 TO 10:LOCATE C+12,14:INPUT #2,FTH(C): PRINT FTH(C)
3700 NEXT C
3710 FOR C=1 TO 10:LOCATE C+12,30:INPUT  #2,ITHL(C):PRINT ITHL(C):NEXT C
3720 FOR C=1 TO 10:LOCATE C+12,42:INPUT #2,ITHR(C):PRINT ITHR(C):NEXT C
3730 LOCATE 23,7
3740 PRINT Y$
3750 '********************* MISCELLANEOUS *******************************
3760 LOCATE 24,5
3770 INPUT "Wish to obtain the audiogram?",R$
3780 IF R$="Y" OR R$="y" THEN GOSUB 3930
3790 X$=STRING$(5,0)
3800 PRINT X$
3810 CLS:LOCATE 10,12
3820 INPUT "Another session(Y/N)?",R$
3830 IF R$="y" OR R$="Y" THEN 3840 ELSE 3870
3840 CLOSE #2
3850 GOTO 230
3860 GOTO 2820
3870 CLS:LOCATE 10,12
3880 INPUT "Wish to see the records(Y/N)?",R$
3890 IF R$="y" OR R$="Y" THEN 3200
3900 RESET
3910 END
3920 '******************** AUDIOGRAM PLOTTING ***************************
3930 CLS:KEY OFF
3940 SCREEN 1:SCREEN 2
3950 LOCATE 3,30
3960 PRINT "AUDIOGRAM"
3970 LINE (508,33)-(512,37),,BF
3980 LOCATE 5,67
3990 PRINT "-- L"
4000 CIRCLE (510,45),4
```

```
4010 PRINT "-- L"
4020 CIRCLE (510,45),4
4030 LOCATE  6,67
4040 PRINT "-- R"
4050 LINE(80,40)-(80,150)
4060 LINE(78,40)-(80,40)
4070 LINE(78,50)-(80,50)
4080 LINE(78,60)-(80,60)
4090 LINE(78,70)-(80,70)
4100 LINE(78,80)-(80,80)
4110 LINE(78,90)-(80,90)
4120 LINE(78,100)-(80,100)
4130 LINE(78,110)-(80,110)
4140 LINE(78,120)-(80,120)
4150 LINE(78,130)-(80,130)
4160 LINE(78,140)-(80,140)
4170 LINE(78,150)-(80,150)
4180 LINE(80,150)-(480,150)
4190 LINE(480,150)-(480,40)
4200 LINE(480,40)-(80,40)
4210 LINE(482,40)-(480,40)
4220 LINE(482,50)-(480,50)
4230 LINE(482,60)-(480,60)
4240 LINE(482,70)-(480,70)
4250 LINE(482,80)-(480,80)
4260 LINE(482,90)-(480,90)
4270 LINE(482,100)-(480,100)
4280 LINE(482,110)-(480,110)
4290 LINE(482,120)-(480,120)
4300 LINE(482,130)-(480,130)
4310 LINE(482,140)-(480,140)
4320 LINE(482,150)-(480,150)
4330 X=80
4340 IF M=1 THEN 6960
4350 FOR C=1 TO 10
4360 YL(C)=(200-(ITHL(C)+50))
4370 YR(C)=(200-(ITHR(C)+50))
4380 X(C)=X+C*40
4390 LINE (X(C)-2,YL(C)-2)-(X(C)+2,YL(C)+2),,BF
4400 CIRCLE (X(C),YR(C)),4
4410 LINE (X(C),150)-(X(C),152)
4420 LINE (X(C),38)-(X(C),40)
4430 LINE(X(1),YR(1))-(80,150),0
4440 NEXT C
4450 FOR C=1 TO 10
4460 LINE -(X(C),YL(C))
4470 NEXT C
4480 LINE -(80,150),0
4490 FOR C=1 TO 10
4500 LINE -(X(C),YR(C))
```

```
4510 NEXT C
4520 LOCATE 6,7
4530 PRINT "110"
4540 LOCATE 8,8
4550 PRINT "90"
4560 LOCATE 11,8
4570 PRINT "70"
4580 LOCATE 13,8
4590 PRINT "50"
4600 LOCATE 16,8
4610 PRINT "30"
4620 LOCATE 18,8
4630 PRINT "10"
4640 LOCATE 8,2
4650 PRINT "T"
4660 LOCATE 9,2
4670 PRINT "h"
4680 LOCATE 10,2
4690 PRINT "r"
4700 LOCATE 11,2
4710 PRINT "e"
4720 LOCATE 12,2
4730 PRINT "s"
4740 LOCATE 13,2
4750 PRINT "h"
4760 LOCATE 14,2
4770 PRINT "o"
4780 LOCATE 15,2
4790 PRINT "l"
4800 LOCATE 16,2
4810 PRINT "d"
4820 LOCATE 17,1
4830 PRINT "(dB)"
4840 LOCATE 22,30
4850 PRINT "Frequency (kHz)"
4860 LOCATE 21,14
4870 PRINT ".125"
4880 LOCATE 21,19
4890 PRINT ".25"
4900 LOCATE 21,25
4910 PRINT ".5"
4920 LOCATE 21,30
4930 PRINT "1"
4940 LOCATE 21,35
4950 PRINT "1.5"
4960 LOCATE 21,41
4970 PRINT "2"
4980 LOCATE 21,46
4990 PRINT "3"
5000 LOCATE 21,51
```

```
5010 PRINT "4"
5020 LOCATE 21,56
5030 PRINT "6"
5040 LOCATE 21,61
5050 PRINT "8"
5060 LOCATE 23,10
5070 PRINT "Press any key to exit this screen..."
5080 IF INKEY$="" THEN 5080
5090 SCREEN 0:WIDTH 80
5100 IF M=1 THEN 3810
5110 RETURN
5120 '***************** SCREEN PREPARATION ******************************
5130 KEY OFF:CLS
5140 SCREEN 2
5150 'WINDOW SCREEN (0,0)-(2000,200)
5160 VIEW (0,0)-(639,199)
5170 X=80
5180 CIRCLE (80,120),4
5190 LINE(80,40)-(80,150)
5200 LINE(78,40)-(80,40)
5210 LINE(78,50)-(80,50)
5220 LINE(78,60)-(80,60)
5230 LINE(78,70)-(80,70)
5240 LINE(78,80)-(80,80)
5250 LINE(78,90)-(80,90)
5260 LINE(78,100)-(80,100)
5270 LINE(78,110)-(80,110)
5280 LINE(78,120)-(80,120)
5290 LINE(78,130)-(80,130)
5300 LINE(78,140)-(80,140)
5310 LINE(78,150)-(80,150)
5320 LINE(2000,150)-(80,150)
5330 LOCATE 6,6
5340 PRINT "110"
5350 LOCATE 8,6
5360 PRINT "90"
5370 LOCATE 11,6
5380 PRINT "70"
5390 LOCATE 13,6
5400 PRINT "50"
5410 LOCATE 16,6
5420 PRINT "30"
5430 LOCATE 18,6
5440 PRINT "10"
5450 LOCATE 8,2
5460 PRINT "T"
5470 LOCATE 9,2
5480 PRINT "h"
5490 LOCATE 10,2
5500 PRINT "r"
```

```
5510 LOCATE 11,2
5520 PRINT "e"
5530 LOCATE 12,2
5540 PRINT "s"
5550 LOCATE 13,2
5560 PRINT "h"
5570 LOCATE 14,2
5580 PRINT "o"
5590 LOCATE 15,2
5600 PRINT "l"
5610 LOCATE 16,2
5620 PRINT "d"
5630 LOCATE 17,1
5640 PRINT "(dB)"
5650 RETURN
5660 '*************** THRESHOLD TRACING ******************************
5670 Y=200-(I%+50)
5680 LINE -(X,Y)
5690 X=X+3
5700 RETURN
5710 '************** T Y P E-1  O P E R A T I O N ********************
5720 '****************** EAR SELECTION *******************************
5730 L=0:R=0
5740 CLS:LOCATE 8,20:PRINT "Choose the ear to be tested:"
5750 LOCATE 10,20:PRINT "1. Left ear"
5760 LOCATE 12,20:PRINT "2.Right ear"
5770 X$=STRING$(20,0):LOCATE 20,20:PRINT X$:LOCATE 22,20:PRINT X$:
     LOCATE 16,38:PRINT X$:LOCATE 16,20:INPUT "Your choice(1,2): ",E
5780 IF E=1 THEN E$="L"
5790 IF E=2 THEN E$="R"
5800 FOR I=1 TO 2
5810 IF E=I THEN 5860
5820 NEXT I
5830 SOUND 1500,4:LOCATE 20,20:PRINT "INVALID CHOICE!"
5840 LOCATE 22,20:INPUT  "Re-choose(Y/N): ",R$
5850 IF R$="Y" OR R$="y" THEN 5770 ELSE END
5860 '****************** FREQUENCY SELECTION ***************************
5870 CLS:LOCATE 3,20:PRINT "Choose the frequency (KHz):"
5880 LOCATE 5,20:PRINT "1.  .125           2.  .25"
5890 LOCATE 7,20:PRINT "3.  .5              4. 1"
5900 LOCATE 9,20:PRINT "5. 1.5              6. 2"
5910 LOCATE 11,20:PRINT "7. 3               8. 4"
5920 LOCATE 13,20:PRINT "9. 6              10. 8"
5930 LOCATE 17,20:INPUT "Your choice(1,2,3...)";F
5940 IF F=1 THEN FTH(C)=.125
5950 IF F=2 THEN FTH(C)=.25
5960 IF F=3 THEN FTH(C)=.5
5970 IF F=4 THEN FTH(C)=1
5980 IF F=5 THEN FTH(C)=1.5
5990 IF F=6 THEN FTH(C)=2
6000 IF F=7 THEN FTH(C)=3
```

```
6010 IF F=8 THEN FTH(C)=4
6020 IF F=9 THEN FTH(C)=6
6030 IF F=10 THEN FTH(C)=8
6040 PRINT "RESULTS"
6050 FOR K=1 TO 10
6060 IF F=K THEN 6100
6070 NEXT K
6080 SOUND 1500,4:LOCATE 20,20 :PRINT "INVALID CHOICE!"
6090 IF R$="Y" OR R$="y" THEN 5930 ELSE CLS:GOTO 5750
6100 F(C)=FTH(C):PRINT #2,F(C)
6110 IF E=1 THEN L=L+1:F(L)=F(C)
6120 IF E=2 THEN R=R+1:F(R)=F(C)
6130 '***************** INTENSITY SELECTION ******************************
6140 CLS:INPUT "Intensity : ",I%
6150 GOSUB 1800
6160 GOSUB 3010
6170 IF R$="Y" OR R$="y" OR R$="N" OR R$="n" THEN 6180 ELSE SOUND 1500,4
     :GOTO 6160
6180 LOCATE 14,8:X$=STRING$(40,0):PRINT X$:LOCATE 14,8:INPUT "Another
     intensity(Y/N): ",R$
6190 IF R$="Y" OR R$="y" THEN 6140
6200 IF E$="L" OR E$="l" THEN ITHL(L)=I% :PRINT #2,ITHL(L)
6210 IF E$="R" OR E$="r" THEN ITHR(R)=I% :PRINT #2,ITHR(R)
6220 LOCATE 14,8:X$=STRING$(32,0):PRINT X$:LOCATE 14,8:INPUT "Another
     frequency(Y/N): ",R$
6230 IF R$="y" OR R$="Y" THEN 5870 ELSE 6240
6240 LOCATE 14,8:X$=STRING$(32,0):PRINT X$:LOCATE 14,8:INPUT "Another
     ear(Y/N): ",R$
6250 IF R$="Y" OR R$="y" THEN 5720
6260 '****************** RESULTS PRESENTATION ***************************
6270 CLOSE #2
6280 LOCATE 14,8:PRINT X$:LOCATE 14,8:INPUT "File Name:",N$
6290 OPEN "I",#2,N$
6300 CLS:LOCATE 2,30
6310 PRINT "RESULTS"
6320 LOCATE 4,2
6330 INPUT #2,N$
6340 PRINT "Name     : ";N$
6350 LOCATE 4,34
6360 INPUT #2,I$
6370 PRINT "Id.No    : ";I$
6380 LOCATE 4,66
6390 INPUT #2,A
6400 PRINT "Age      : ";A
6410 LOCATE 5,2
6420 INPUT #2,S$
6430 PRINT "Sex      : ";S$
6440 LOCATE 5,34
6450 INPUT #2,O$
6460 PRINT "Operator : ";O$
6470 LOCATE 6,2
6480 INPUT #2,D$
6490 PRINT "Date     : ";D$
6500 LOCATE 6,34
```

```
6510 INPUT #2,T$
6520 PRINT "Time     : ";T$
6530 LOCATE 8,2
6540 INPUT #2,C$
6550 PRINT "Comments : ";C$
6560 LOCATE 7,2
6570 INPUT #2,M$
6580 PRINT "Type     : ";M$
6590 LOCATE 12,7
6600 'Y$=STRING$(50,45)
6610 PRINT Y$
6620 LOCATE 9,1:Y$=STRING$(80,45):PRINT Y$
6630 LOCATE 10,15:PRINT "Left Ear: "
6640 LOCATE 10,55:PRINT "Right Ear: "
6650 LOCATE 11,5:PRINT "Frequency (kHz)"
6660 LOCATE 11,25:PRINT "Intensity (dB)"
6670 LOCATE 11,45:PRINT "Frequency (kHz)"
6680 LOCATE 11,65:PRINT "Intensity (dB)"
6690 LOCATE 12,1:PRINT Y$
6700 X=80
6710 FOR C= 1 TO L
6720 LOCATE C+12,8
6730 INPUT #2,F(L):PRINT F(L)
6740 IF F(L)=.125 THEN F=1
6750 IF F(L)=.25 THEN F=2
6760 IF F(L)=.5 THEN F=3
6770 IF F(L)=1 THEN F=4
6780 IF F(L)=1.5 THEN F=5
6790 IF F(L)=2 THEN F=6
6800 IF F(L)=3 THEN F=7
6810 IF F(L)=4 THEN F=8
6820 IF F(L)=6 THEN F=9
6830 IF F(L)=8 THEN F=10
6840 X(C)=X+F*40
6850 INPUT #2,ITHL(L):LOCATE C+12,18:PRINT ITHL(L)
6860 YL(C)=(200-(ITHL(L)+50))
6870 NEXT C
6880 GOTO 3780
6890 LINE (120,150)-(120,152)
6900 LINE (160,150)-(160,152)
6910 LINE (200,150)-(200,152)
6920 LINE (240,150)-(240,152)
6930 LINE (280,150)-(280,152)
6940 LINE (320,150)-(320,152)
6950 LINE (360,150)-(360,152)
6960 LINE (400,150)-(400,152)
6970 LINE (440,150)-(440,152)
6980 LINE (480,150)-(480,152)
6990 LINE (120,38)-(120,40)
7000 LINE (160,38)-(160,40)
```

```
7000 LINE (160,38)-(160,40)
7010 LINE (200,38)-(200,40)
7020 LINE (240,38)-(240,40)
7030 LINE (280,38)-(280,40)
7040 LINE (320,38)-(320,40)
7050 LINE (360,38)-(360,40)
7060 LINE (400,38)-(400,40)
7070 LINE (440,38)-(440,40)
7080 LINE (480,38)-(480,40)
7090 FOR C=1 TO L
7100 LINE (X(C)-2,YL(C)-2)-(X(C)+2,YL(C)+2),,BF
7110 LINE(X(1),YR(1))-(80,150),0
7120 NEXT C
7130 FOR C=1 TO L
7140 LINE -(X(C),YL(C))
7150 NEXT C
7160 GOTO 4520
7170 '************************* DELAY **********************************
7180 FOR K=1 TO 500
7190 NEXT K
7200 RETURN
```

APPENDIX D

OPAMP-LF 356 AND ANALOG SWITCH -DG 201

D.1 INTRODUCTION

The audio attenuator and amplifier unit of the PC-based audiometry system is described in Section 4.4. The two major components of the unit are the low-noise operational amplifier LF 356 and the Analog switch DG 201. This appendix deals with these two components.

D.2 OP AMP - LF 356

LF 356 is a low noise operational amplifier with a pin configuration as shown in Fig. D.1. Pin 7 is connected to the positive supply and Pin 4 to the negative supply. Pin 2 and 3 form the inverting and non-inverting input terminals. Pins 1 and 5 are used to balance the offset voltage. There is no internal connection to Pin 8. Four op amps are used in the circuit and each provides different levels of attenuation.

D.3 ANALOG SWITCH - DG 201

DG 201 has four built-in CMOS analog switches. Its pin configuration is shown in Fig. D.2. The state of any switch depends on its control voltage (V_c). If V_c is 0V, the switch is `ON'. A control voltage of 5V changes the switch state to `OFF'. In the present setup, each bit of the D/O from the DAP is used as the control voltage for changing the state of a particular switch. The specifications of DG 201 are given in Table D.1. The 'off' resistance of DG 201 was measured and found to be more than 400 $k\Omega$..

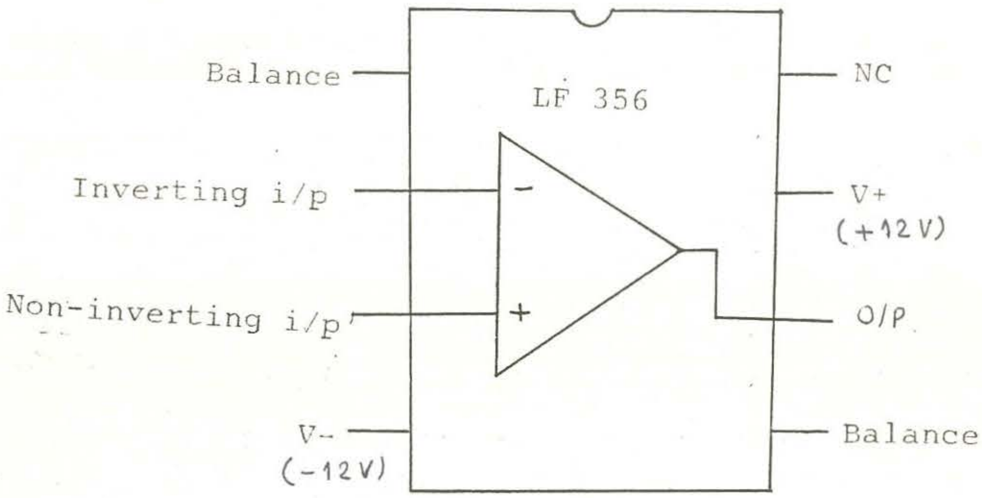


Fig.D.1 Pin configuration of LF 356

Source : UTP (1978) IC Master

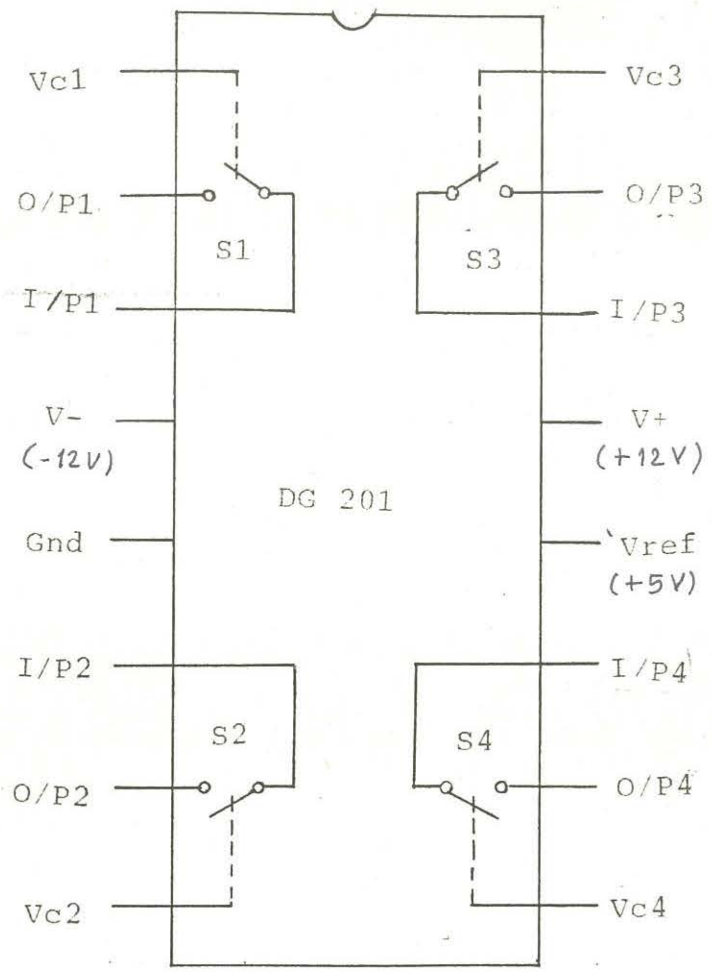


Fig.D.2 Pin Configuration of DG 201

Source : Intersil (1983). Hot Ideas in CMOS

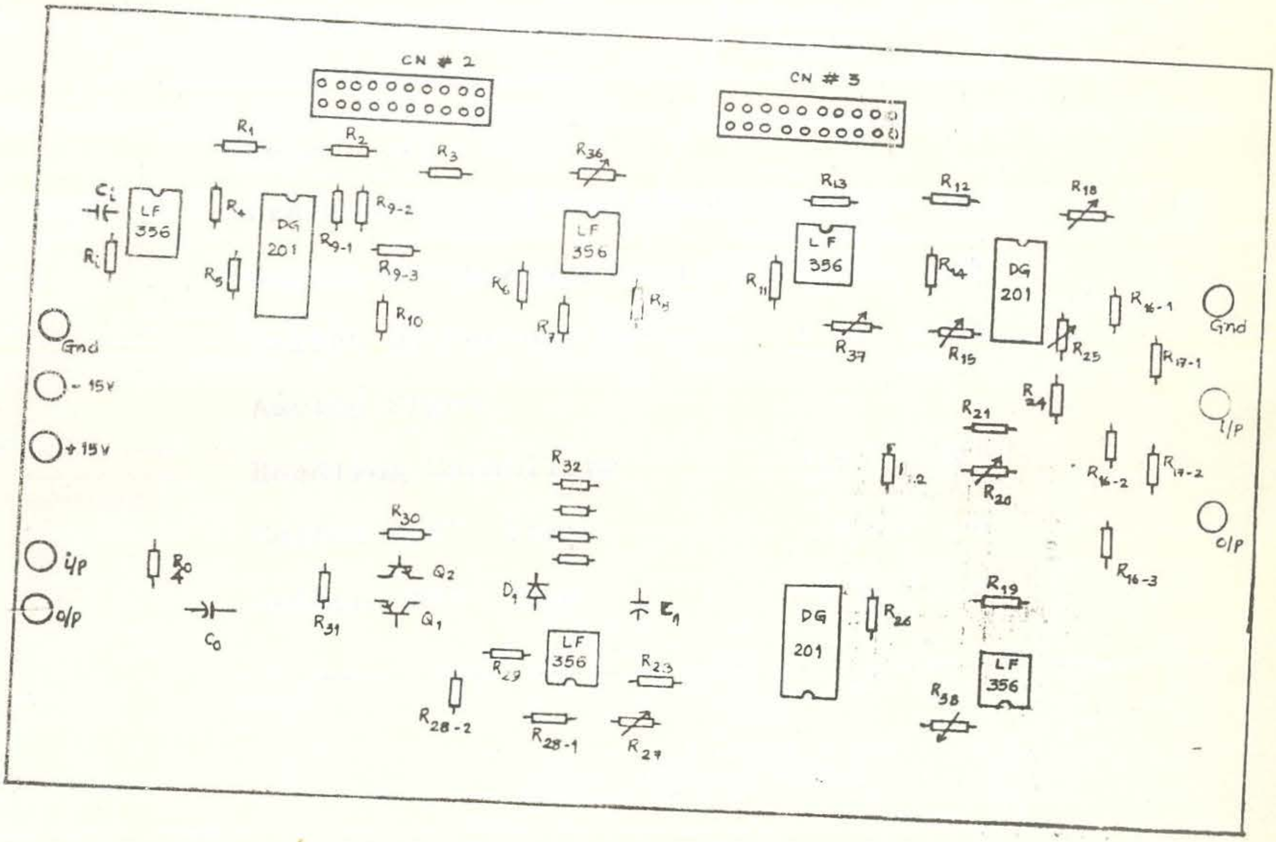


Fig. D.3. Component layout of the computer controlled attenuator.

Table D.1. Specifications of DG 201 at + 25^{o} C

Characteristic	Min/Max Limits	
Input Logic Current	$\pm$ 1	μA
Drain-source ON Resistance	80	Ω
Switch OFF Leakage current	$\pm$ 1	nA
Switch ON Leakage Current	$\pm$ 2	nA
Analog Signal		
Handling Capability	$\pm$15	V
Switch "ON" Time	1	μS
Switch "OFF" Time	0.5	μS

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