COMPUTER ASSISTED AUDIOMETRIC EVALUATION SYSTEM

by

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A dissertation submitted to the Faculty of Medicine at the University of Cape Town in partial fulfilment of the requirements for the degree of Master of Science in Medicine in the field of Biomedical Engineering

August 1991

ABSTRACT

A computer-based audiometric evaluation system has been developed. The system makes use of an IBM PC/XT/AT compatible personal computer to perform pure tone and speech tests and comprises a plug-in card and custom software. The card contains pure tone and masking noise generators, together with amplifiers for a set of headphones and bone conduction transducer, patient and audiologist microphone amplifiers and a hand-held infra-red remote-control unit. A voice-operated gain-adjusting device on the audiologist's microphone eliminates the need for a sound pressure level meter during speech tests.

The software-based user-interface makes use of overlaid pop-up menus, context sensitive assistance and a text editor on a graphics screen. Pure tone and speech data are acquired and displayed on a dynamic audiogram and speech discrimination gram respectively. This data may be stored and later retrieved from a patient data base. Further audiometric tests may be incorporated at a later stage.

ACKNOWLEDGEMENTS

The Speech and Hearing Departments at Groote Schuur Hospital (Cape Town), H F Verwoerd Hospital (Pretoria) and Tygerberg Hospital (Bellville) allowed me to sit in on audiological tests and pester the audiologists with questions until I understood, procedurally, what was required. Special thanks to Susan Smuts-Muller (until recently head of the Audiology Unit at Groote Schuur Hospital) for providing me with descriptions of the test procedures and for vital ideas and advice in the development of user-friendly software.

My mentor and thesis supervisor, Mladen Poluta at Biomedical Engineering, UCT, whose consistent red-ink comments kept me up nights un-splitting my definitives and re-arranging paragraphs and who has been a great help and support through this whole process.

Financial assistance for this dissertation was obtained from the Research Institute for Environmental Diseases (S A Medical Research Council, Pretoria) and the University of Cape Town and is gratefully acknowledged.

Finally my mother, head of the Speech and Hearing Department at H F Verwoerd Hospital, whose original idea it was to interface audiological tests and computers. Her inspiration and positive attitude towards the project saved me from quitting on a number of occasions. This project is dedicated to her memory.

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LIST OF ABBREVIATIONS AND SYMBOLS

ANSI	American National Standards Institute
ASHA	American Speech and Hearing Association
BIOS	Basic Input Output System
DAC	Digital to Analogue Converter
dB	decibels (A logarithmic representation of the ratio between two quantities or values, eg two SPL's vary by one decibel when $20\log_{10}(SPL_2/SPL_1) = 1$).
DOS	Disk Operating System
Hz	hertz (cycles per second)
IBM-PC	International Business Machines - Personal Computer
IEC	International Electrotechnical Commission
MCL	Most Comfortable Level
MUX	Multiplexer
PTA	Pure Tone Average (The average of the hearing threshold of one ear at 500Hz, 1000Hz and 2000Hz)
SABS	South African Bureau of Standards
SRT	Speech Reception Threshold
TD	Threshold of Discomfort
V	volts
VCA	Voltage Controlled Amplifier
VCO	Voltage Controlled Oscillator

SPL — Sound Pressure Level. The ratio, expressed in dB, of the sound pressure in question to a reference pressure, SPL (dB) = $20\log(p/p_o)$ re p_o Pa
The reference level used in audiology is 20 uPa (0dB SPL = normal hearing threshold).

NB. American custom is followed in this thesis and the word 'intensity' is used when Sound Pressure Level is intended.

CHAPTER ONE - INTRODUCTION

The computer based audiometer moves away from the classic audiometer of dials and buttons and allows the audiologist to concentrate more on the task than on the instrument.

With this in mind, the aim of the thesis was to develop a personal computer based audiometer: not to simulate the classic audiometer on the screen of a computer, nor to perform the tests automatically, but to provide the audiologist with a computer interface to perform audiometric testing. The computer program applies recently developed user-interface enhancements (menu driven, overlaid windows, on-line context sensitive help and extended interface facilities) to an audiometric environment.

In order for the computer program to have control of the test signals and signal levels, an audiometric prototype card was developed. The prototype card plugs into an IBM compatible Personal Computer (PC) and generates the signals and masking noise necessary for the audiometric tests. In addition, the prototype card contains amplifiers to drive the headphones and bone conduction transducer used in audiometric evaluation. Included on the card is an infra-red remote-control receiver that allows communication between a hand-held remote-control unit and the audiometric program.

Chapter Two provides an overview of current methods of hearing evaluation - specifically the Basic Test Battery approach, and describes the role of the audiologist, patient and audiometer in assessing the patient's hearing. The standardisation of test symbols is also discussed.

The literature survey in Chapter Three reviews methods of controlling the flow of a program with examples of existing software, entering data into and producing hard copies from the PC, and representing data on the computer screen. Visual coding techniques such as color, shape and size are also discussed.

The design of the hardware is covered in Chapter Four. This includes the audiometric prototyping card and the hand-held remote-control. Specifications for the test signals, masking noise and amplifiers are given.

The audiometric computer program comprises a number of software units. These units are described in Chapter Five. Flow diagrams for test procedures are compared to the flow diagrams of the computer program. The computer screen formats for the tests and pop-up menus are presented.

The audiometric prototype card was calibrated to meet international audiometric standards. The method employed to calibrate and evaluate the prototype card, together with test results, is described in Chapter Six.

Chapter Seven includes conclusions and recommendations for clinical trials and further work that could possibly expand on the audiometric computer based environment.

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CHAPTER TWO - A PERSPECTIVE ON AUDIOLOGIC ASSESSMENT

2.1 INTRODUCTION

The current approach to audiologic assesment is to employ a sequence of tests to assess the hearing ability of a patient. The selection of appropriate tests follows a rational course based on sequential clinical findings. These tests are grouped into **quantitative** and **differential diagnostic** tests.

2.2 EVALUATION METHODS

In assessing the hearing ability of the patient, the audiologist is pursuing a double purpose:

i) **Quantitative assessment**

A patient's hearing ability can be assessed quantitatively by using a **basic test battery.** The basic test battery results provide a measure of hearing sensitivity as a function of frequency, and contribute to the estimation of probable handicap. The audiologist uses this battery to establish how the patient subjectively perceives calibrated, standard signals of different frequencies corresponding to the range encountered in normal speech (125 Hz to 8000 Hz). The sound pressure levels at a set of frequencies are measured on a numeric scale and can be compared to standard hearing levels. The unit used in hearing evaluation is the decibel (dB). Hearing loss is defined as $20 \log_{10}(P_1/P_o)$ where P_1 is the lowest sound pressure which the patient is able to perceive. (NB. Although not technically correct, SPL is subsequenctly referred to as 'intensity').

The basic test battery includes the following tests:

* Pure Tone Air and Bone Conduction.
* Speech Reception Threshold (SRT).
* Most Comfortable Loudness level (MCL).
* Threshold of Discomfort (TD).
* Speech Discrimination.

(Refer to appendix I.3 - Test Procedures, for a description of the basic test battery.)

ii) **Differential Diagnostic Testing**

Once a hearing impairment has been established by use of the basic test battery, further diagnostic tests can be employed to establish the site of the lesion causing the hearing disorder. These tests are called the **special** or **site of lesion tests.** The site of lesion test results suggest which portion of the auditory system is the probable cause of the hearing loss.

The audiologist interprets these results together with the medical history and the clinical findings in an effort to diagnose the site of the lesion, arrive at a possible prognosis and plan treatment. The site of lesion tests include :

* Short Increment Sensitivity Index (SISI).
* Tone Decay (TD).
* Sensorineural Acuity Level (SAL).
* Alternate Binaural Loudness Balance (ABLB).
* Alternate Monaural Loudness Balance (MLB).
* Bekesy
* Masking Level Difference (MLD).

(Refer to appendix I.3 - Test Procedures, for a description of the site of lesion tests.)

This thesis concerns itself only with the basic test battery.

2.3 THE BASIC TEST BATTERY

The basic test battery can be divided into **pure tone** and **speech** tests.

2.3.1 Pure Tone Audiometry

By using the pure tone tests it is possible for the audiologist to describe the degree and type of hearing loss ie. conductive, sensorineural or mixed hearing loss. (Refer to Appendix 1.2 - Types of Hearing Loss.)

Pure tone tests include pure tone air conduction and pure tone bone conduction tests. Hearing levels are plotted on an audiogram shown in figure 2.1 and pure tone averages for left and right ears are calculated. The pure tone average for each ear is the average of the hearing thresholds (measured in dB) at 500 Hz, 1000 Hz and 2000 Hz in that ear.

The audiologist endeavours to determine when patient responses are accurate and to predict when sound may have crossed over to the non-test ear. This occurs if one ear is significantly better than the other ($\pm$40 dB SPL), when it may become necessary to "mask" the non-test ear. If this is the case, tests are conducted with narrow band masking noise in the better, non-test ear. The masking noise centre frequency is the same as the pure tone test frequency, with a band width of approximately one tenth of the test frequency. This band width is called the **critical band width**. The effect of the masking noise in the better ear is to raise the hearing threshold of that ear. (Refer to Appendix I.4 - Masking.)

2.3.2 Speech Threshold and Speech Discrimination Tests

Speech threshold tests are a direct measure of overall speech sensitivity and also provide a comparative measure for confirming pure tone thresholds. The purpose of **speech discrimination testing** is to measure how well the listener can differentiate monosyllabic words, as a function of increasing intensity.

Masking is necessary in speech testing (threshold and discrimination) under the same circumstances as pure tone tests. The masking noise used for speech tests, however, is speech noise. (The band width of speech noise is from approximately 100 Hz to 4000 Hz.)

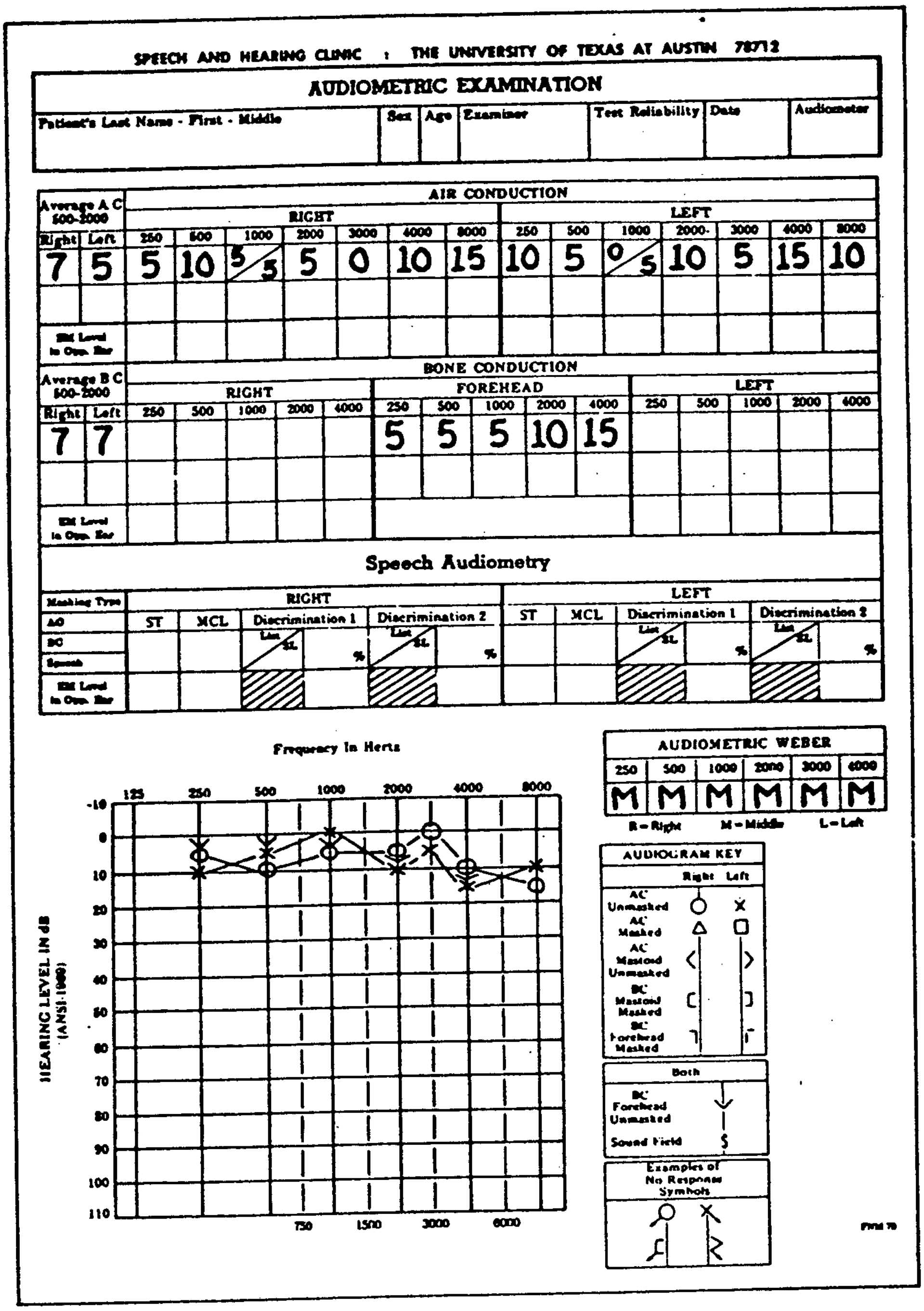
Speech threshold testing includes measurement of the patient's :

* Speech Reception Threshold (SRT).
* Most Comfortable Loudness Level (MCL).
* Threshold of Discomfort (TD).
* Dynamic Range (DR). (TD - SRT = DR)

Potential problems relating to a patient's home or preferred language are inherent to speech discrimination testing. Unfamiliar words in word lists used in speech discrimination tests may lead to an incorrect response by the patient.
A greater problem, is that the speech signal may be delivered live or via a pre-recorded tape. If the former is used, care must be taken to monitor the speech with a VU meter.

Speech threshold and discrimination results are tabulated together with pure tone results in figure 2.1

Figure 2.1 **Typical Audiologic Assessment Sheet For Pure Tone and Speech Tests.**



SPEECH AND HEARING CLINIC : THE UNIVERSITY OF TEXAS AT AUSTIN 78712

AUDIOMETRIC EXAMINATION

Patient's Last Name - First - Middle	Sex	Age	Examiner	Test Reliability	Date	Audiometer

AIR CONDUCTION

Average A C 500-2000 Right	Left	RIGHT 250	500	1000	2000	3000	4000	8000	LEFT 250	500	1000	2000	3000	4000	8000
7	5	5	10	5/5	5	0	10	15	10	5	0/5	10	5	15	10
EM Level in Opp. Ear															

BONE CONDUCTION

Average B C 500-2000 Right	Left	RIGHT 250	500	1000	2000	4000	FOREHEAD 250	500	1000	2000	4000	LEFT 250	500	1000	2000	4000
7	7						5	5	5	10	15					
EM Level in Opp. Ear																

Speech Audiometry

Masking Type	RIGHT ST	MCL	Discrimination 1 List SL %	Discrimination 2 List SL %	LEFT ST	MCL	Discrimination 1 List SL %	Discrimination 2 List SL %
AC								
BC								
Speech								
EM Level in Opp. Ear								

AUDIOMETRIC WEBER

250	500	1000	2000	3000	4000
M	M	M	M	M	M

R = Right M = Middle L = Left

AUDIOGRAM KEY

	Right	Left
AC Unmasked	O	X
AC Masked	Δ	□
AC Mastoid Unmasked	<	>
BC Mastoid Masked	[	]
BC Forehead Masked		

Both

BC Forehead Unmasked

Sound Field S

Examples of No Response Symbols

2.3.3 Further Assessment

After the basic test battery and site of lesion tests are completed, the audiologist is able to assess the hearing impairment and comment on the reliability of the tests. Further audiologic evaluation using additional objective diagnostic equipment can now be performed if the audiologist feels the results are equivocal. These tests include acoustic immittance measurement, acoustic reflex and auditory brain stem evoked response tests.

2.4 AUDIOMETRIC ENVIRONMENT

The environment in which audiometric tests are conducted should be sufficiently sound proof so that ambient noise is below normal hearing thresholds.

Hearing tests are normally conducted in a sound proof booth (typical ambient noise levels for sound proof booths are ±30dB above normal hearing thresholds). Booths are designed for either one-room or two-room use. In one-room booths the audiologist and equipment are outside the booth, and the patient is in the booth. In a two-room arrangement, the audiologist and equipment are in one room and the patient is seated in the examining room. In both cases the audiologist has visual communication with the patient via a window. Electrical connections provide a link between the audiometer and the headphones, loudspeakers, patient response button and intercoms.

2.4.1 Role of the Audiologist

The audiologist's role is to explain the test procedure and make the patient aware of his or her task in the test. Secondly the audiologist controls the signals generated by the audiometer and performs the test by making choices based on sequential findings. The audiologist is responsible for correct tabulation, diagnosis and filing of the results and reports.

2.4.2 Role of the Patient

Because of the large variation in patient cooperation (due to age, language ability, intelligence, culture, education and willingness), the testing approach will be different for each individual.

Procedures are often confusing for unsophisticated patients or patients whose home language differs from that of the audiologist.

The patient must respond when a tone is heard during pure tone audiometry, must ignore masking noise in masked tests, and repeat or identify words in speech tests.

Without the patient's cooperation, the basic test battery and site of lesion test results are impossible to obtain.

2.4.3 Role of the Audiometer

The audiometer is a calibrated instrument that can produce single or frequency-modulated pure tones, which may be continuous or pulsed, as well as speech and narrow band masking noise. These signals can be produced at frequencies, intensities and durations selected by the audiologist.

An audiologist-to-patient (talk forward) microphone and a patient-to-audiologist (talk back) microphone are provided with the audiometer so that the audiologist can communicate with the patient. The audiometer also contains a multiplexing system that enables the test signal to be routed to either left or right headphone, left or right loudspeaker or the bone conduction transducer. More sophisticated audiometers are able to interface with computers to provide a patient data base. The most recent audiometers have been integrated with personal computers and provide the audiologist with the ability to control the test procedures from the personal computer.

2.5 STANDARDISATION OF TEST SIGNALS AND SYMBOLS

At least three autonomous authorities have been involved in specifying test signals for audiometers. These are the American National Standards Institute (ANSI), the American Speech and Hearing Association (ASHA) and the International Standardisation Organisation (ISO). (Refer to Appendix I.6 - Audiometric Signal and Symbol Standards.)

ASHA has specified symbols to be used on audiologic result forms. It can be seen from the symbol keys in Appendix I.5 - Sample Audiogram Results - that certain symbols have not been internationally standardised.

These differences are not as obvious in non-computerised audiology as in computer based audiology. In non-computerised audiology the audiologist is able to draw any shape symbol on the audiogram, as long as a key on the audiogram defines that symbol.

In computer based audiology the audiologist is limited to the symbols displayed on the computer screen. This leads to problems when trying to fulfill a general need with an audiometric software program.

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CHAPTER THREE - LITERATURE SURVEY

3.1 INTRODUCTION

A distinction between computer-controlled and computer-based audiometers should first be made.

A **computer-controlled** audiometer contains the classic audiometer user-interface and an integrated microprocessor. The microprocessor is capable of controlling the signal generation, monitoring the audiometer panel switches, displaying the information in a digital form on a numerical display board and setting up the audiometer in pre-configured test arrangements (Grason Stadler GSI-10 and GSI-16 Service Manuals, 1985; Maico MA-24 Operating Instructions, 1986; Tracor RA400 Users Guide, 1987). The presence of the microprocessor in the audiometer does not alter the user-interface, and the audiologist still uses a similar method for selecting and presenting test signals. The microprocessor does, however, make it simple for the audiologist to select test procedure configurations and also provides the communication ability to interface the classic audiometer with an external computer for data transfer and graphic results. When a computer is coupled to the audiometer, the operator still performs the audiometric tests from the audiometer and not from the computer.

A **computer-based** audiometer on the other hand provides the audiologist with a screen-based user-interface (Madsen IGO-HAT System Users Guide, 1987). The audiologist interfaces with a computer screen and keyboard instead of the myriad of switches found on classic audiometers. Data is displayed in a manner intuitive to the audiologist rather than presenting the audiologist with the equipment and having the audiologist adapt to it. During pure tone evaluation, for example, the audiogram on the screen is updated while the test is being conducted, and the audiologist need not transfer data manually from a numerical read-out to a graph (CA540 Hortmann Computer Audiometer User Information, 1988).

It is important to note that the method of execution of the audiometric tests on a computer-based audiometer differs from that of a computer-controlled audiometer (a screen and a keyboard as opposed to buttons and dials respectively). Comparative testing of test speed and accuracy of hearing thresholds was performed by Jervall et al. (1983). These results showed no significant differences in speed of testing and hearing thresholds using the same test procedures on computer-based and computer-controlled audiometers.

Both computer-based and computer-controlled audiometers have the ability to interface with a computer data base for mass storage of patient information. The computer-controlled audiometer, however, does not have a keyboard interface, and can only transfer raw test data from the audiometer to the computer. Patient information must then be entered from the computer side.

3.2 <u>CONTROL OF TEST ENVIRONMENT</u>

If an audiometric test environment is to be computer-based, it must replace all conventional audiometric instruments and accessories - such as the audiometer, recording pens, audiogram forms and word lists - with a computer, screen, keyboard and printer.

Corell et al. (1983) describe reactions by both audiologist and patient to a computer-based hearing aid fitting program within the first six months of use. Results showed that there is little resistance on both sides to computer-controlled audiometric environments. The inclusion of a computer-based audiometer should also offer little resistance.

Huckle (1981), Spilman and Wong (1989), Lynch-Freshner et al. (1989) and Crow (1989) all agree that the most important aspect of computer acceptance is the ease with which the user is able to perform a task - whether this task is searching for information in a data base, seeking assistance, entering data or saving a file.

The audiometric test environment is becoming more and more computer oriented (Harris JD, 1980; Harris S, 1981; Grisanti G et al. 1986; Grogan JB, 1987; Gatehouse S, 1986; Sakabe N, 1981; Mason SM, 1984; Chan FH et al. 1984) and also more instrument fragmented, ie. more instruments are being developed, each one computer-based and performing a certain audiometric test. The only link between the various pieces of equipment is the audiologist who has to perform the tests and maintain the data base. If the audiometric environment is to be optimal as far as equipment and control are concerned, it appears that the instruments should link to a common data base in the audiometric environment thus alleviating some of the data base management responsibilities from the audiologist.

3.2.1 Program Control

Program control refers to the method employed by the system (computer) in allowing the operator to make and execute choices within a computer environment. Program control may be seen as the system / operator interface. The more intuitive and natural this process, the less the operator need be aware of the computer and the greater the attention that can be given to the task (Showman, 1989).

Program control is not related to hardware input devices, ie. keyboard, mouse, touch screen or wand, but only to the way the software is controlled by the user. (Input devices are discussed in section 3.2.2). Seven types of Program Control have been identified. These are described with reference to commercial software packages based on the different types.

i) **Line Command Sets.**

When a function is required, the menu key is pressed, and a command set appears in the top or bottom row of the screen (LOTUS 123, 1983; Microsoft-WORD, 1984; CHIWRITER 1989). The operator may then select a function from the set of functions presented. This is done by typing the first letter of the function or positioning the highlight bar with cursor movement keys over the function required and pressing the ENTER key. Once a function is selected, the command set will change to a sub command set related to this function. For example, if the operator selects FILE from the main command set, the sub command set might include functions such as SAVE, LOAD and DELETE. The level of sub command sets is determined by the functions that are necessary in that subset and may be as deep as seven levels (Microsoft-WORD, 1984).

A drawback with this method is that the operator is presented with only one command line at a time, and cannot recall previous functions or command sets without first exiting from the present subset. There are usually no short cuts to functions and the operator has to traverse all levels to execute the desired function.

ii) **Interactive Pointer.**

Most Computer Aided Design (CAD) packages employ a screen pointer to point to marked off positions on the screen (EGA-PAINT, 1985; TANGO release 1, 1986; ORCAD release 1, 1986; PCAD, 1983; Hewlett Packard NEW WAVE environment, 1989; Microsoft WINDOWS, 1988). The screen usually has a static command set at one of its borders. The command set is often made up of iconic representative graphics (graphic pictures that represent the function, eg. a paint brush for painting or a box for drawing rectangles). When the pointer is positioned over a function, and the required key is pressed, that function is executed. Pointer parameters - such as shape or colour - may change to indicate which function has been selected. These programs have been designed to work with extended input devices such as the mouse, tracker ball or cursor keys to move the pointer around on the screen.

iii) **Hidden Program Execution.**

Program flow is hidden from the operator and choices are made solely by the software program. Terminate and stay resident (TSR) programs such as PIZZAZZ, 1986, GRAFPLUS, 1984 and TURBO EMS Extended Memory Manager, 1986 only allow the operator to install the program. From then on, the program will monitor the relevant keys and status and will execute when necessary. The operator is usually unaware of this execution.

programs implement 90 Wordstar-type functions while NORTON COMMANDER implements only 20).

Wordstar-type flow control provides a quick means of selecting a function, but suffers from the drawback that there are many special keys and that the key sequence does not necessarily reflect the function. For example, CONTROL Y deletes a line of text in the Wordstar text editor and Borland software.

vi) **Function Key Flow Control.**

Along the top or on the left hand side of most personal computer keyboards is a set of function keys. The number of function keys found on keyboards varies from eight to twenty.

Programs employing function key flow control allow the operator to choose functions by pressing one of the function keys. WORDSTAR-type programs usually allow the operator to select functions from function keys as a means of reducing the number of key strokes.

The action of a function key is usually defined by the program, although recent releases of programs allow the operator to define their own function keys (WORDSTAR Pro, 1987 and Borland SIDEKICK, 1987). A static information bar across either the top or bottom of the screen defines the functions of the function keys.

The number of functions that can be defined depends on the number of function keys on the keyboard. This is usually limited to ten with further functions provided for by using CONTROL or ALTERNATE buttons in conjunction with the function keys. The operator still has to learn what the function keys do, and when to press CONTROL or ALTERNATE keys.

vii) **Overlaid Menu Flow Control.**

A menu is a list of command options currently available to the computer operator - some stay onscreen, while "pop up" and "pull down" menus are activated by the operator.

To provide the operator with maximal screen workspace, menus are displayed on the screen only when a certain function key is pressed. The menu is displayed over existing screen data ("overlaid"). Once the operator has made a choice from the menu by pressing a highlighted letter of a menu item or positioning the menu highlight bar over the required item with cursor movement keys and pressing ENTER, the menu either disappears or a second level menu is displayed.

The operator may exit from a menu by pressing the ESCAPE key instead of making a menu choice. This returns the operator to the menu one level up. The operator is able to manoeuvre between menus with only four buttons (up key, down key, Enter and Escape) and is also able to see all previous menus without having to leave the current level (Borland version 4 and 5 compilers for Pascal and C, 1988; PC-TOOLS deluxe, 1989; Microsoft Windows, 1988).

Of the seven flow control methods described in this section, the overlaid menu method is the most recent for graphic intensive programs (CAD programs and other programs where the screen has pictorial information). Perry and Voelcker (1989) describe the Graphic User-Interface (GUI) - a combination of windowing, menus, icons and a mouse - as being the cornerstone of modern personal computer and work station applications.

3.2.2 Input Methods and Devices.

"Input devices are peripheral hardware products that attach to the computer and provide the operator an alternative means of computer communication" (Huckle 1981).

More specifically, an input device is added to a computer to provide the operator with an effective method of controlling program flow, retrieving information, entering data and managing the user environment. The input device is often a graphics pointing device. There is usually an arrow, crosshairs or highlighting at the position on the screen selected by the device.

Five types of input devices are discussed. These are the keyboard, the mouse, the tracker ball, the light wand and the touch screen.

i) **The Keyboard.**

Although the computer keyboard is seen as an integral part of the computer, it is still an input device. It is not however a graphics pointing device, but a means by which alpha-numeric data can be entered into a computer. The keyboard may be divided into blocks of keys depending on their position on the keyboard and their function. These blocks typically include 58 QWERTY keys, 17 calculator keys, 4 cursor keys, 12 function keys and 6 scroll keys. The number of keys in each block may vary with cost and functionality.

QWERTY keys provide a means of input for alpha numeric-data. In conjunction with the CONTROL and ALTERNATE keys the QWERTY keyboard can double as a flow control interface (in Wordstar, for example, all functions can be controlled with the QWERTY keys).

The calculator key block resembles a calculator keyboard. This block is provided to make numeric entry into the computer easier.

The cursor keys allow the operator to move a cursor around the screen. Usually four keys (up, down, left, right) make up this set but diagonal cursor keys may also be included. The cursor keys are sometimes incorporated into the calculator key block.

Scroll keys perform on-screen formatting such as deleting a letter, back spacing, moving the cursor to the top or bottom of the screen (Home and End) or moving between pages (page down, page up).

Function keys may be configured by the program to control program flow or any other function. There is no standard configuration for the function keys.

ii) **The Mouse.**

This is one of the most widely used input devices. The mouse is a hand-held device whose motion across a smooth surface causes an on-screen cursor to move proportionately. Current devices contain a roller-ball and one, two or three select buttons.

Research conducted by Apple Computers and Sun Microsystems in the late 1970's (Perry and Voelker 1989) into the number of buttons necessary on the mouse had the effect of polarizing mouse interfaces. Apple Computers opted for a one button mouse system with a double click action for selecting menus or menu items and Sun Microsystems developed devices with two buttons - (one to accept an item, and one to reject an item) and three button systems (the centre button actuates an assistance screen).

iii) **The Tracker Ball.**

The tracker ball consists of a housing with a large ball placed in a hole on the top of the housing. The ball moves freely within the hole, and the relative motion of the ball causes the position of the cursor on the screen to move proportionately.

With a mouse, the operator has to physically move the mouse around a flat surface, whereas the tracker ball housing is stationary, with the advantage that it requires less physical desk space to operate. The tracker ball may also have one, two or three select buttons.

iv) **The Light Wand.**

The light wand is a pointer that translates the physical position of the wand on the screen into an X and Y coordinate location for the computer program.

A pressure sensitive switch at the end of the wand indicates when the wand is pressed against the screen. When this happens, a light sensitive transducer at the tip of the wand detects when the screen raster passes the point beneath the light wand. A reference time is recorded and translated into an X and Y screen coordinate.

v) **The Touch Screen.**

One method of implementing a touch screen is to place a pressure sensitive transparent sheet on the computer screen. This allows the operator to select functions by pressing appropriate areas on the screen which are transformed into X and Y coordinates that the computer program can interpret.

The screen outlay has to be designed specifically for touch screens. If touch areas designed on the screen are too small or too close together, the chances of an operator touching two adjacent areas exist and ambiguous or incorrect operation may result. As with the wand, arm lifting becomes annoying and physically tiring if the selection task must be performed regularly.

3.2.3 Output Methods and Devices.

Although patient data is stored in the computer database, it is also essential to have a hard copy of test results. Copies are presented to the referring doctor and filed together with the doctor's report.

There are at present three devices used for obtaining hardcopies in the audiometric environment:

i) **Dot Matrix Printers.**

The print head of a dot matrix printer is made up of a column of print pins (usually 8, 9 or 24). These pins are programmed to strike the printer ribbon against the paper and so create a permanent image on the paper. Most dot matrix printers conform to a standard protocol of data transfer (Centronics Interface Standard) for text or ASCII characters. These characters are stored in the printer's memory and not in the computer's memory. Control of the print head is done by the printer as the computer only instructs the printer as to which character to print.

Unfortunately graphic hardcopies on dot matrix printers are not as simple. The computer must send a stream of pixel information (bit mapped graphics) to the printer instead of just a single character (Star Gemini Printer Users Manual, 1986). This process is time consuming, noisy and difficult to program efficiently.

The printer ribbon may be of any colour or multi-coloured, allowing colour graphic hard copies.

ii) **Laser Jet Printers.**

The laser jet printer works on the same principle as a photo copying machine. An image is scanned onto a rotating drum with a laser light. The drum rotates through a carbon powder which is electrostatically attracted to the areas where the laser light has scanned. Paper is rolled over the drum, and the carbon is burned onto the paper forming a permanent image.

New series of laser printers have a standard printing protocol. Printer control languages (PCL) have been developed and adopted by laser printer manufacturers. Hewlett Packard's Printer Control Language (1989) allows the computer to load the laser printer with both graphic and text information before printing starts. Graphic capabilities are similar to those of printers, except that the method of generating the graphic symbols is different (because of PCL). The advantages of laser printers over dot matrix printers are the speed involved in producing graphic hardcopies, the resolution of the image and the reduction of noise due to the absence of a print head. For practical purposes, the laser printer can be considered to print in one colour only.

iii) **Ink Jet Printers.**

The print head of the ink jet printer is able to squirt a fine jet of ink at a controlled rate onto the moving paper. At present ink jet printers are able to print in colour but are relatively slow for graphic hard copies (Hewlett Packard Desk Jet Printer Users manual 1989).

tone testing, for example, the screen represents the audiogram, and a cursor represents the test parameters - the cursor's vertical position on the audiogram represents the intensity of the signal, its distance from a margin represents the frequency of the signal, and the colour of the cursor represents the test ear (normally red for right ear and blue for left ear - ASHA Standard).

This allows the audiologist to test the patient without having to turn dials or push buttons for different channels, and then interrupt the test procedure to plot the frequency and amplitude on the audiogram. It also provides the audiologist with a dynamic graphic representation of the patient's hearing threshold.

This seems very well, but the tests are not as simple as that, and if the screen is to be effective it should provide the audiologist with the same facilities that are presently available on non-computerised audiometers. One example is the ability to make notes at any time during the test. If this facility is not available, the audiologist will use a note pad, and the effectiveness of the screen is reduced (the data base cannot interpret a note stuck to the screen !).

In order to provide full audiometric control, the program must control the screen in such a way as to make results as clear as possible to the audiologist. The use of colour and other information coding techniques together with relevant program control and input methods must be employed to avoid screen clutter and ensure optimal flow control (Huckle, 1981).

3.3.1 Colour as an Aid in Data Presentation.

Teichner (1979) describes certain applications where the use of colour is beneficial for information coding. These applications include :

i) **Capturing Operator Attention.**

Selective attention is used by the operator to concentrate on a certain area on the screen. Colour helps the operator de-clutter screen information.

Greenstein and Flemming (1984) also showed that warning or error messages should normally be displayed in a colour different to the background colour. This information represents a change in status of some process.

ii) **Categorisation of Data.**

Large numbers of objects displayed on a screen can be effectively separated into smaller groups by colour coding the objects. This can significantly aid operator performance (Kopala, 1979).

iii) **Standardisation of Status.**

In an environment where operators do not necessarily use the same display all the time, it is possible to give the operator a common reference for information retrieval; for example, all the operator has to remember is that green means safe and red means danger.

The selection and conceptual mapping of colours in these applications are important in providing relevant coded information. The technology to display 650 000 colours on the screen simultaneously is available. This does not imply that using the full spectrum of colours will aid the operator.

Irrelevant colours will not aid performance but will clutter the screen. Not more than four important colours should be displayed on the screen at once (Matthews, 1987).

Colours must also be mapped conceptually with objects being displayed, for example, green does not fit into the colour concept of heat, but blue and red do. The use of multiple colours on a screen does give aesthetic appeal. It does not imply that it will result in better performance by the operator (Riesling and Emerson,1985).

The advantages of using colour over monochrome include:

- On briefly exposed displays, an operator's memory for item location is enhanced by colour (Wedell and Alden,1972). Thus colour may facilitate the subsequent recollection and reassociation of tasks which have been performed.

- Colour combines well with other types of visual coding systems. Colour can be used in conjunction with geometric shapes, sizes, position or brightness. Human performance has been shown to improve with colour codes when used in conjunction with other coding methods (Narborough-Hall, 1985).

- Colour is generally a consistent advantage in visual search tasks, but only if the colour of the target is known in advance (Teichner, 1977). Thus colour is helpful in indicating to the operator which area of the screen to look at.

- Colour coding reduces the number of items through which the operator must search.

3.3.2 Alternative Forms of Visual Coding.

Huckle (1981) and Narborough-Hall (1985) have identified four other types of visual information coding. These are brightness, symbols (shapes), symbol size and flashing displays. Generally only two brightness levels are considered, where the brighter level is normally used to introduce items of greater importance. Brightness levels may also be used to attract the operator's attention to a specific area of the screen.

Symbol coding should be meaningful or the symbols will be confused with one another. Symbols can be combined satisfactorily with other forms of coding, such as size, colour or flashing.

Size of the symbols as a form of visual coding has not been used satisfactorily. Size is relative to the screen size, and can easily be confused on different screens.

An efficient way of attracting the attention of an operator is to allow an item to flash. It has however been observed that excessive use of flashing for trivial reasons can become counter-productive and annoying (Huckle, 1981). Flashing can be used to indicate an alarm condition or to force the operator to attend to a certain item. For example, in a menu driven program, there may be up to seven levels of menus on the screen at one time. To attract the operator's attention to a certain menu, the highlight bar in the active menu can be made to flash.

3.3.3 Coding Redundancy.

A question that has not been successfully answered is whether a form of coding should be redundant or non - redundant. In the case of colour for example, if the same information was presented in monochrome, would the distinction that colour makes be lost? Luder (1982) showed that for search tasks (looking for an item on a screen among a group of items) colour was the most effective form of coding, redundant or non - redundant. But for identification tasks (identifying an item on a screen such as an active menu) non - redundant coding was superior to most other coding methods. It therefore appears that whether coding is redundant or not depends on the nature of the task involved.

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CHAPTER FOUR - HARDWARE DESIGN

4.1 INTRODUCTION

The IBM Personal Computer (PC) was chosen as a base for the development of the hardware for the following reasons:

- Eight hardware interface slots are available. A range of IBM feature cards that plug into these slots allow the PC to perform functions not available on the standard PC. The slots have a standard interface format (Micro-Channel) which implies that any feature card may plug into any compatible PC (Eggebrecht, 1987). Commercial feature cards include a printer interface, a real time clock, hard disk drive controller, floppy disk drive controller and a prototyping card.

  The prototyping card allows designers to develop their own feature cards using the IBM PC Micro-Channel interface.

- Program development language compilers are available for the IBM PC. These include Basic, Pascal, C, Fortran and Prolog.

- Upgrading the system (eg. more memory, higher resolution monitor, faster processing speeds) is not dependent on the prototype card design.

- The Mirco-Channel interface is supported by most computer manufacturers and the portability of the development card between computers is therefore guaranteed.

The circuitry necessary for pure tone and speech evaluation was developed on a commercially available IBM PC prototyping card.

The final product was calibrated and tested using a standard head set (Telephonics TDH 39) and bone conductor (Radioear B-71-A), and a Bruel & Kjaer 2230 precision sound pressure level meter. The evaluation and calibration of the hardware is discussed in chapter 6 and the circuit diagrams and prototype layout are presented in appendix II.

4.2 <u>SPECIFICATION</u>

The basic test battery procedures require the following circuitry (Hodgson, 1984) :

i) **Sinusoidal Signal Generator.**

The band width of the sinusoidal pure tone signal generator must exceed the frequency range of the pure tone tests. ASHA specify a frequency range from 125 Hz to 8000 Hz in one half octave steps. (An octave is the range from a specified frequency to a value of twice that frequency eg. 250 Hz to 500 Hz or 2000 Hz to 4000 Hz). Frequency modulated pure tone test signals (warble tones) are also required. The rate of modulation is suggested as 20 Hz with a frequency deviation of no more than $\pm$10% of the frequency of the pure tone signal (Hodgson, 1984).
Specifications for the pure tone test signals are given in appendix I.6 - Audiometric Signal and Symbol Standards.

ii) **Masking Noise Generator.**

The masking noise generator must produce band width limited white noise dependent on the test tone. During pure tone testing, the band width of the masking noise must approach that of the critical band (Refer to appendix I.4 - Masking, for critical band masking values) and during speech tests the band width must approach that of speech noise (approximately 100 Hz to 4000 Hz).

iii) **Input and Output Selector.**

A method of selecting different input signals is required. These signals include the pure tone test signal, masking noise, microphone, record player or cassette deck. The selected input signal is routed to one of the output channels by means of an output channel selector. Output channels include left and right headphones, left and right loudspeakers and the bone transducer.

iv) **Amplifiers.**

An amplifier is required for each output channel. In addition a patient-audiologist monitor system is required. The monitor system allows the audiologist to hear what the patient is saying.

Maximum amplification levels are frequency dependent and must reach maximum intensities comparable to those specified by ASHA (refer to appendix I.6 - Audiometric Signal and Symbol Standards).

In addition to these circuits, interface circuitry between the PC and the prototype card is necessary. This includes address decoding and data latching for the above circuits.

A remote-control unit is necessary in situations where the audiologist wishes to elicit a tone while in the patient booth. (Applications for remote-control audiology also exist in paedo-audiology).

Circuitry for the remote-control receiver is necessary on the prototype board. The remote-control transmitter consists of six buttons and transmitter circuitry in a hand-held unit.

4.3 IMPLEMENTATION

4.3.1 Pure Tone Generation

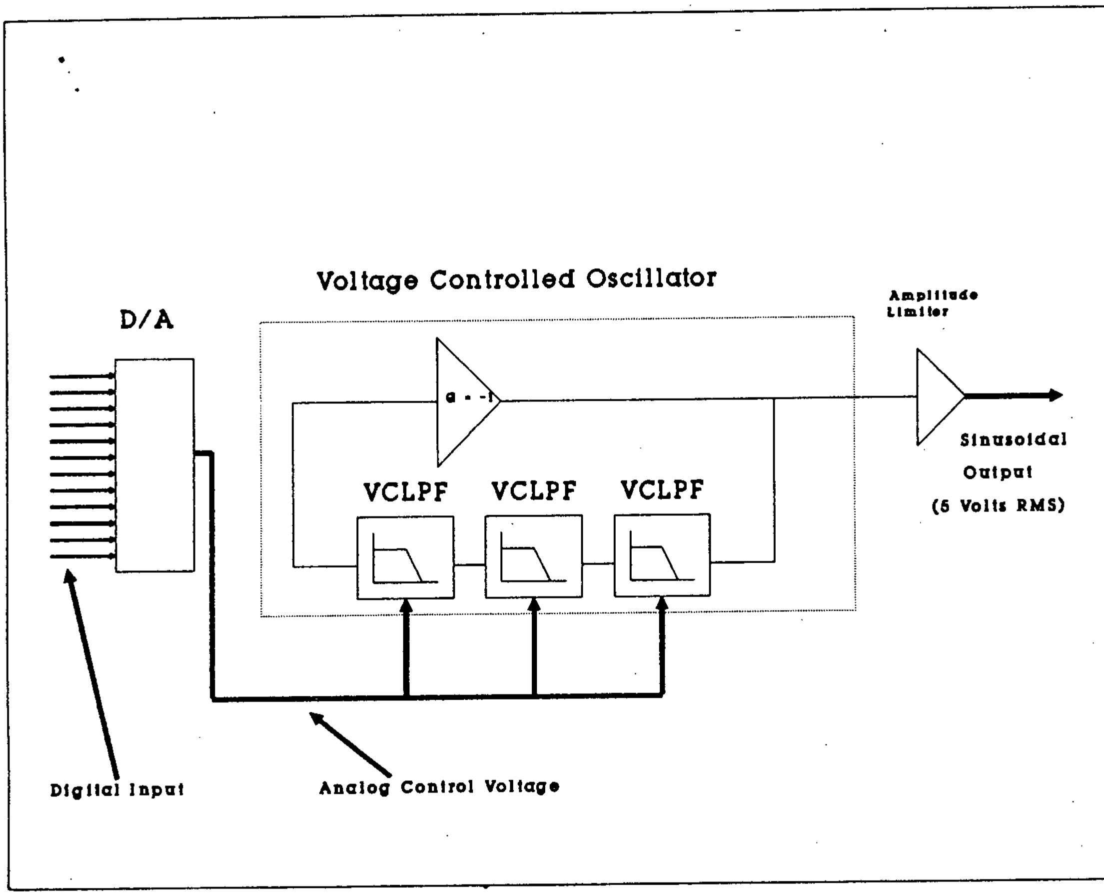
A number of techniques for generating sinusoidal signals are described by Bogart (1986) and Boylestad and Nashelsky (1982). There is little difference in complexity or accuracy of the different circuits.

The method chosen for this design is an analogue **Voltage Controlled Phase Shift Oscillator** (VCO) - the controlling voltage coming from a Digital to Analog (D/A) converter as shown in figure 4.1.

The sinusoidal generator works as follows:
The inverting amplifier causes a 180 degree phase shift in the signal passing through it, and the purpose of the cascaded first-order **voltage controlled low pass filters** (VCLPF) is to introduce an additional 180 degrees at some frequency. (The output of a single VCLPF leads its input by a phase angle that depends on the signal frequency). When the signal passes through all three VCLPFs, there will be some frequency at which the cumulative phase shift is 180 degrees. When the signal having that frequency is fed back to the inverting amplifier, the total phase shift around the loop will equal 180 degrees + 180 degrees = 360 degrees and oscillation will occur at that frequency, provided the loop gain is 1.

The frequency of the oscillating circuit can be changed by varying the frequency response of the VCLPFs with a control voltage.

FIGURE 4.1. Block diagram of Digital Voltage Controlled Oscillator.



The VCLPFs and the inverter were built using a National Semiconductor transconductance operational amplifier LM13600 (refer to circuit diagram and data sheets in appendix II.1 - Pure Tone Generation). This chip contains two **transconductance operational amplifiers** (TOA) in one package, as well as linearising diodes and buffer circuitry for each device.

Only two LM13600 packages are required : three TOA's are configured as VCLPFs and the fourth as an inverter.

The VCLPFs were designed to have unity gain in the passband and a roll off of 6dB per octave above the 3dB cut off point. The control voltage sets the 3dB cut off point for all three filters - thus producing a voltage controlled filter configuration.

The control voltage for the VCO is supplied by a DAC1008 D/A converter. The 12 bit digital control word to the D/A converter is driven via decoding logic circuitry from a port location within the IBM PC.

The 12 bit control word gives a resolution of 1 in 4096 steps at the output of the D/A converter. If the reference voltage to the D/A converter is 10.0 volts, then the smallest change in output voltage is

$$\begin{aligned} V_{out} &= 10 \text{ volts} / 4096 \text{ steps} \\ &= 2.4 \text{ mV per step.} \end{aligned}$$

The VCO is designed to work on a voltage range from 0 volts to +10 volts, giving a 50Hz sinusoidal wave at 0 volts and a 10 kHz sinusoidal wave at +10 volts. Assuming the sinusoidal VCO is linear over the specified input voltage range, a 2.4 mV change in control voltage will give a change in output frequency of

$$\begin{aligned} F_{out} &= (2.4 \text{ mV} \times 10 \text{ KHz}) / 10 \text{ V} \\ &= 2.4 \text{ Hz.} \end{aligned}$$

Signal frequency is specified by ASHA, IEC and SABS to be within 5% of the nominal test value. The lowest test frequency (and therefore the greatest area of error) is 125 Hz. This implies that the test signal must be in the range 125 $\pm$ 6,2Hz. The tolerance allowed is more than twice the smallest VCO step (2.4 Hz). The accuracy of the generated test signal is therefore within ASHA specifications.

A National Semiconductor LM356 operational amplifier limits the sinusoidal output voltage to 5 volts peak to peak.

4.3.2 Masking Noise Generation

The source used to generate the white noise is the P-N junction of a "noisy" N-P-N transistor. The P-N junction noise is amplified by a single transistor amplifier (common emitter configuration). The signal is then filtered using switched high pass and low pass filters. This design provides enough scope to produce any band width of noise across the audible spectrum.

Maxim Electronics MAX260 switched capacitor filter integrated circuits are used for the low pass and high pass filter design. (Refer to circuit diagram and data sheets in Appendix II.2 - Masking Noise Generation.) Each integrated circuit (IC) contains two second-order filter stages. These may be cascaded to provide one fourth-order filter per IC. The filter cut off points are software selectable and can be programmed to be either high pass, low pass or band pass.

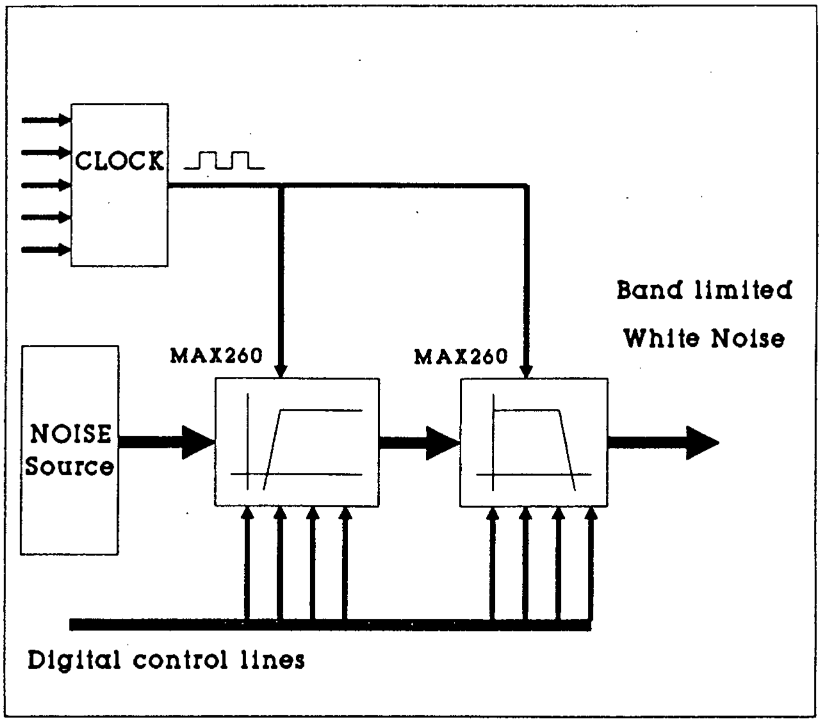
Requirements for critical band limited noise suggest that a fourth-order Butterworth filter is adequate for audiometric test masking applications (ASHA and Bogart, 1986).

Two MAX260 ICs are therefore required - one for high pass filtering and one for low pass filtering. Control data for the MAX260s is latched via decoding logic from a port within the IBM PC. The MAX260s require a clock input that is ten times higher than the filter centre frequency. This implies that a programmable clock generator circuit be implemented together with the filter design. The maximum clock signal needed is

8000 Hz (maximum test frequency) X 10 = 80 000 Hz.

An LM356 operational amplifier limits the output of the noise generator to 5 volts RMS.

FIGURE 4.2 **Block diagram of Digital Controlled Band Limited Noise Generator.**



4.3.3 <u>Signal Selection</u>

The signal selection circuitry is a four by five input/output multiplexing matrix. Any one of the four input signals can be routed to any one of the five output channels.

The input signals include :

- sinusoidal test signal for pure tone tests
- masking noise for pure tone and speech masking
- microphone for audiologist-patient communication and speech tests
- Auxiliary input for tape recorder or record player used in speech tests.

The output channels include :

- left headphone
- right headphone
- left loudspeaker
- right loudspeaker
- bone transducer

More than one route may be selected at a time so that, for example, the sinusoidal signal is routed to one channel while masking noise is routed to another channel.

National CD4052 dual audio multiplexer chips are used to implement the switching circuitry. (Refer to circuit diagram and data sheets in Appendix II.3 - Signal Selection and Amplification.)

A separate channel is available for patient-audiologist communication.

4.3.4 Amplification

The IC chosen for the amplification of each channel was a National Semiconductor LM388 audio amplifier. This IC offers high signal to noise ratio and high supply noise immunity. (Refer to circuit diagram and data sheets in appendix II.3 - Signal Selection and Amplification.) This IC is commonly used in commercial audio circuits.

Signal amplification is internally set to 50 dB in the LM388, and the output is stable over the operating temperature and frequency. The output level of the amplifier is therefore fixed by the input signal level. The input level is attenuated using a National Semiconductor DAC1008 D/A converter, and is then amplified by the LM388 to produce the required output level.

As discussed in sections 4.3.1 and 4.3.2 the outputs of the pure tone signal and noise generators are set to 5 volts peak to peak and 5 volts RMS respectively. The input to the D/A converter (when one of these signals is selected) is therefore constant, and can be attenuated by the D/A converter and amplified by the LM388 to give the required output level.

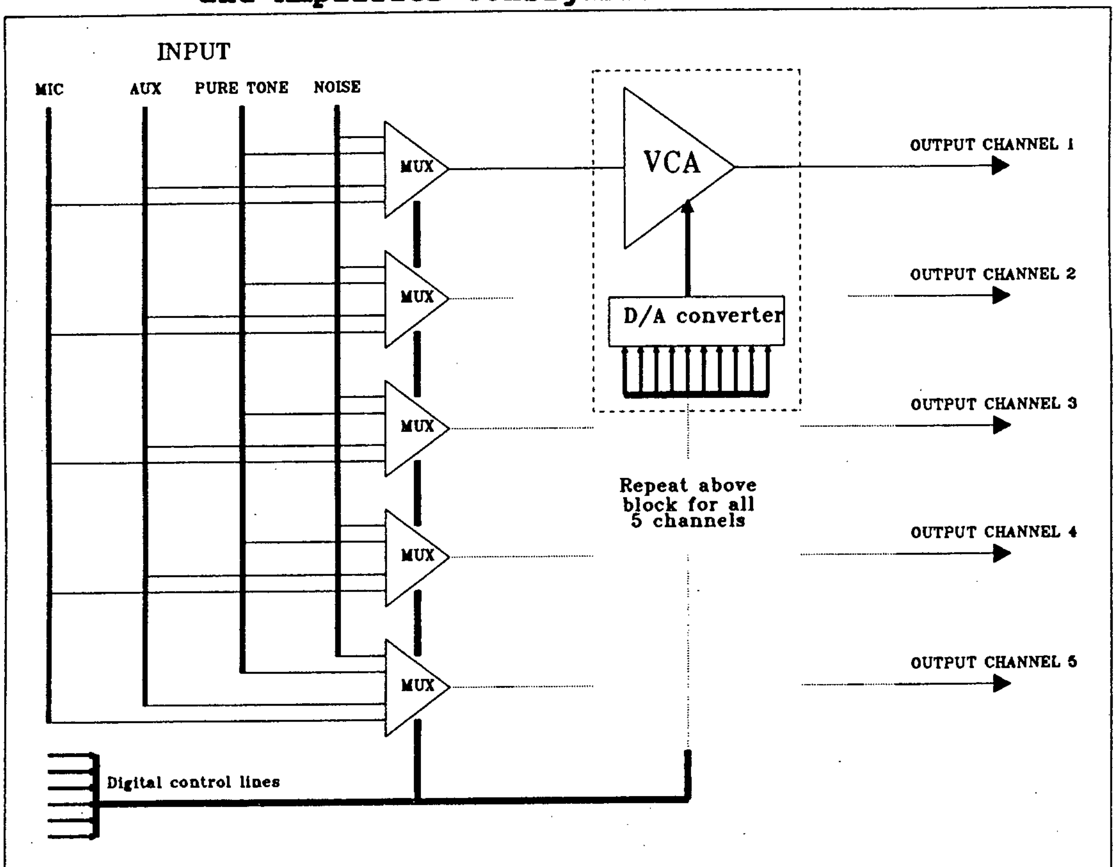
The **audiologist-patient microphone** amplification circuitry requires pre-conditioning. The pre-conditioning involves setting the gain control of the audiologist's microphone pre-amplifier. This is important because the audiologist uses monitored live voice to conduct speech tests. The accuracy of the speech tests is related to the deviation of the SPL of the test speech. In the classic audiometer the sound level of the audiologist's voice is displayed on an SPL meter on the front panel of the audiometer. The audiologist controls the level of the speech by talking louder or softer.

The microphone input stage of the proposed design consists of a pre-amplifier with level limiting and boosting circuitry. This allows the microphone input SPL to vary over a range of ± 40 dB SPL but limits the output to a fixed amplitude (± 5 volts). As long as the audiologist speaks into the microphone with an SPL of over ± 60 dB SPL, the output of the pre-amplifier will be kept at a constant level. The signal can then be attenuated by the D/A converter and amplified by the LM388 to achieve the required output SPL.

The circuitry used to limit the input signal is a Plessey Electronics VOGAD IC (refer to data sheet in appendix II.3 - Signal Selection and Amplification) used in commercial and military two way radios.

The **auxiliary input** is intended for a tape recorder, record player or other audio signal input. This input is buffered and pre-amplified to a fixed level ($\pm$5 volts). The input signal voltage to the auxiliary input is assumed to be a commercial audio input level ($\pm$200mV).

FIGURE 4.3 Block diagram of Input Multiplexer and Amplifier Configuration.



4.3.5 Remote Control

The remote control provides the audiologist with the ability to carry out tests while in the patient's booth, and is intended to assist the audiologist when testing becomes difficult or awkward, rather than to replace the computer interface.

The remote control circuit couples a hand held transmitter to the PC via an infra red light beam. The software has been designed so that all test functions can be accessed by the remote control.

Transmitter

The transmitter has six keys: four direction keys (up, down, left, right) and two response keys (Enter and Escape). The remote control simulates the same six keys on the computer's keyboard.

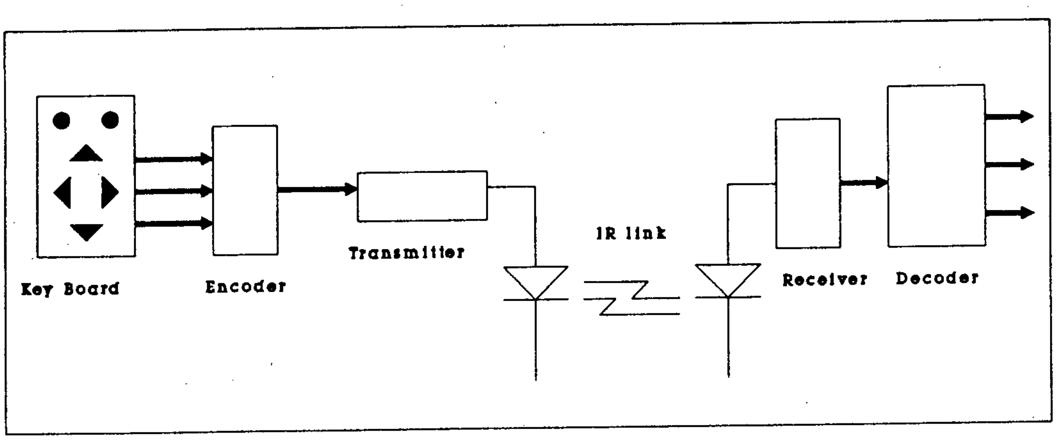
The transmitter circuit comprises a keyboard encoder (National Semiconductor 74HC923), encoder/transmitter (Motorola MC145026) and an output transistor amplifier (refer to Appendix II.4 - Remote Control, for circuit diagram and data sheets). The amplifier drives an infra-red Light Emitting Diode (LED).

When a key on the transmitter is pressed , the encoder ICs convert the information into a serial data stream. The data stream is transmitted via the amplifier and LED to the receiver circuitry in the PC.

Receiver

The signal is detected by an infra red light detector diode (PIN diode) and amplified by a Siemens TDE4061 infra red receiver / amplifier. The signal is decoded by means of a Motorola MC145027 decoder/receiver IC and can be read into the PC via a port.

FIGURE 4.4 **Block diagram of Infra Red Remote Control Link.**



4.4 CONSTRUCTION AND TESTING OF PROTOTYPE

Each module described was subject to a six stage development cycle:

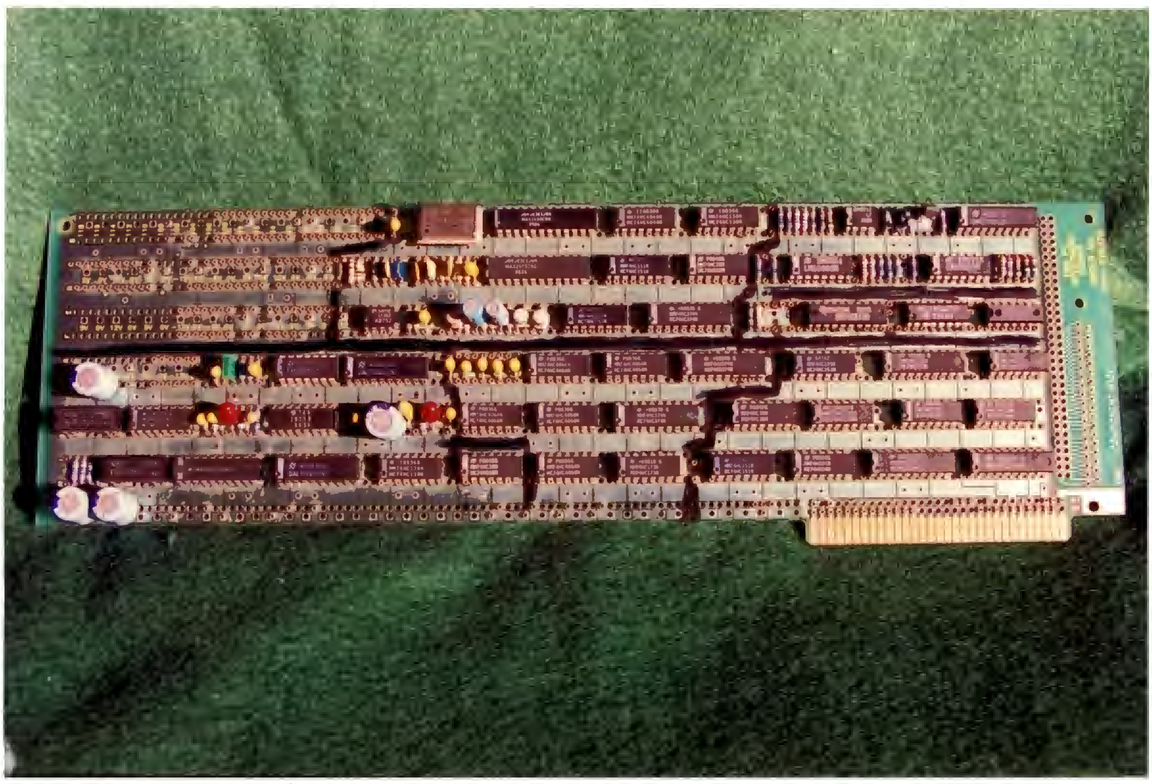
i) circuit design
ii) initial circuit construction and fault finding
iii) component outlay on prototyping card
iv) circuit construction on prototyping card
v) circuit testing in PC environment
vi) fault correction and modifications.

The circuits were initially built on bread board with supply voltages coming direct from the PC. Once the circuit was working correctly, the circuit was constructed on the prototyping card and tested inside the PC environment (refer to figure 4.5).

The Vero Card (Vero Bicc International Pty. Ltd) for IBM PC prototyping provides easy construction due to Vero's "Speed Wire" method for component connections. Supply voltages of +12V, +5V, 0V, -5V and -12V are present at points along the card. The board also facilitates fault recovery as components can be disconnected quickly and neatly.

FIGURE 4.5 Photograph of completed prototyping card.

Demarcated areas on the card show - top left - masking noise generator, top right - sine-wave generator, bottom right - bone and insert amplifiers, bottom centre - audio multiplexers, bottom left - IBM PC interface logic and decoding, centre right - infra red receiver circuitry.



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CHAPTER FIVE - OPERATOR INTERFACE AND SOFTWARE DESIGN

5.1 INTRODUCTION

The aim of this thesis is to provide the audiologist with an audiometric program that will replace the classic audiometer interface with a concept that is already familiar to the audiologist, ie. performing the test **and** plotting the audiogram, without having to worry about how the "audiometer" is configured or where the data should go. The program aims at providing the audiologist with a transparent instrument interface, allowing the audiologist to concentrate on the patient and obviating the need for assessment pads and coloured pens.

Any user resistance to computer-based audiology will imply that the audiologist is for some reason not satisfied with the way the program is structured. This might possibly be the case in computer based audiometers that try to **simulate** classic audiometers by reconstructing the audiometer on the screen. Buttons, knobs and dials are replaced by a screen icon that the audiologist has to press (touch screen) or select (moving a mouse or light wand around the screen) to find the correct switch, or levels of menus that are so deep that by the time the audiologist has found the function, the patient has fallen asleep and the audiologist is so frustrated that the test is performed with a tuning fork instead.

Another possible reason for user resistance is the method by which the coupling of the computer and the audiometer is accomplished.

If the audiometer can transmit raw data to the computer only, then the audiologist must indicate to the computer what type of data is being transmitted. (The GSI 10 audiometer has a transmit key that must be pressed every time data is transmitted to the computer).

Not only does the audiologist now have to perform the audiometric test on the audiometer, but the audiologist also has to operate the computer to plot results, input data and retrieve previous results. The computer is not seen as an aid, but as another instrument that has to be mastered and incorporated into the test environment.

When conducting speech SRT, MCL, TD and discrimination tests the audiologist has to maintain a constant SPL. To do this on a classic audiometer, a SPL meter is provided and the audiologist has to watch the needle to maintain the correct SPL. The computer based audiometer on the other hand, can automatically monitor the audiologist's speech and adjust the intensity accordingly, leaving the audiologist free for other tasks. Also, the computer based audiometer is able to provide the audiologist with a **dynamic audiogram.** Instead of numerical information, the audiologist is presented with a blank audiogram on the screen at the beginning of the test, and can perform the test and fill out the audiogram at the same time and on the same screen.

5.2 <u>SPECIFICATION</u>

Specifications for the user interface include :

i) **Methods of data representation**

- The use of **colours** specified by ASHA (1979) to designate the right and left ear (red for right ear and blue for left ear) and to enhance test results and graphic information.

- Internationally accepted audiometric **symbols** (ASHA 1978) to differentiate test results on the screen and on the report.

- A **flashing menu highlight bar** to signify an active menu when more than one menu is present on the screen. The highlight bar will only flash for an initial period.

- **Error messages** displayed in the centre of the screen in the highest background-contrasting colour.

ii) **Graphic and tabular representation of test data**

When pure tone and speech discrimination tests are conducted, symbols will be plotted on the screen in the correct place **during** the test. Tabular information regarding completed tests (pure tone average, SRT, MCL, TD and DR) will be displayed on the screen during all tests.

iii) **Pop-up menus**

The audiologist need not leave the present task in order to perform another task, but the present task should be overlaid by the new task selected from pop-up menus.

iv) **Pop-up text editor**

The pop-up text editor allows the audiologist to make notes during the tests. Simple text editing functions such as inserting or deleting a letter, word or line and reformatting a paragraph should be available. These notes are printed with the relevant tests when the report is printed.

v) **Special short-circuit keys**

The implementation of short-circuit keys allows the audiologist to move from one task to another without having to traverse levels of menus. The <ALTERNATE> key is used to signify that the next key pressed is a short-circuit choice (eg. to actuate the pop-up text editor, the audiologist presses <ALTERNATE> <C>).

vi) **Context sensitive help**

Context sensitive help screens that describe the audiometric environment and the present audiometric test being conducted will be available at all times. The help screens replace the need to refer to a user's manual for information regarding test procedures, printer and screen configuration, or patient data retrieval and storage.

vii) **Data base management**

The audiometric environment interfaces with a patient data base. The data base manages all data storage, retrieval and search functions.

viii) **Printer interface**

Audiometric report generation is configured for any printer with Hewlett Packard's Printer Control Language (PCL) protocol. This includes certain ink jet, laser jet and dot matrix printers.

5.3 IMPLEMENTATION

In order to best represent process control in the program, decision flow diagrams of the sequence of events during different tests were applied to the structure design of the program. This involved the development of modular structures or **units** that represent events during the test: filing procedures, entry of patient information, test selection, printing functions and the display of previous test results.

The units are separate entities that pass data between one another and the host program. This allows the program to follow an object or task oriented structure. All tasks related to patient information entry, for example, are located in a unit that is able to manage the screen during its lifetime (while information is being entered), communicate with other units to retrieve data from the data base, and pass the new patient information back to the host program. Although this technique is not true Object Oriented Programming (OOP), it does give the program an object or task oriented structure.

The units were implemented using a high level language, Turbo Pascal Version 5.5 (Borland International, 1989). This structured language allows the development of units that can be compiled into one program.

The following units were developed to control the audiometric environment:

i) **Keyboard unit.**

The keyboard is monitored for data or control key entry. If a key is pressed, this unit responds by reading the keyboard, translating the code into relevant information and passing this information to the host program. The mouse interface is also controlled by this unit.

ii) **Text menu driver unit.**

This unit contains code to generate overlaid text menus, to control selection of menu items, and to replace the overlaid screen when an item has either been chosen or the menu disregarded. A highlight bar controlled by keyboard responses scrolls through menu choices. Menus may be either in a vertical or horizontal format.

iii) **Graphic menu driver unit.**

Graphic or iconic representation menus are employed for certain functions where the representation of choices is better justified by symbol shape rather than textual description. In pure tone audiometric testing, ASHA symbols are used to plot the hearing response of a patient to certain stimuli. A blue cross and a red circle are used to represent pure tone thresholds in left and right ears, respectively. It therefore makes sense to allow the audiologist to choose a pure tone test by symbol identification instead of textual information. This unit controls the design and capture of the icons, the display of the graphic menu and the restoration of the overlaid screen. The menu item is selected in the same manner as text menus.

iv) **Input box driver unit.**

Data entered into the computer is done in the form of an input box. The box provides a means of prompting the operator with a statement and waiting for a correct response. Entered data may be edited using normal editing keys (<INSERT>, <DELETE>, <BACK-SPACE>, <HOME> and <END>). The input box passes the correct type of data back to the host program. The input box unit maintains the overlaid screen when an input box is displayed.

v) **Printer interface unit.**

The printer interface unit allows the audiometric program to print high quality (up to 300 dots per inch) multi-font text and graphics to any PCL printer.

vi) **Help unit.**

The help unit provides assistance when the help key is pressed. The help key may be specified by the end user. (Most programs referred to in this thesis employ the <F1> key as the default help key.) The help unit interprets the position of the highlight bar in the active menu and displays an overlaid help window in which assistance to the specific function is given.

vii) **ata base management unit.**

Access and storage of patient records is maintained through the Turbo Pascal Data Base Unit. This unit provides high level routines for inserting, searching and removing records from the data base. The unit has been streamlined for the audiometric program. All file access errors are trapped by the unit and reported to the audiometric program.

5.3.1 Menu Driver Routines

i) **Screen manipulation**

The screen of the IBM PC can be formatted for either text or graphics applications. The **text screen** format is limited in that it can only display characters from a set of 256 defined ASCII characters. No graphic capabilities are available in the text format. The **graphics screen** format allows each point on the screen to be accessed individually. Text that is displayed on a graphics screen has to be configured in a special way because of the graphic point address format. The menu routines for the audiometric program make use of the graphics screen format. This implies that text menus can be displayed on the screen at the same time as graphics.

No commercial programs are available at this time that can implement menus on a graphics formatted screen. The menu routines therefore had to be written and **debugged** before the audiometric program could be implemented.

The Enhanced Graphics Adapter (EGA) screen used for development is a graphics format bit-mapped display. Each picture element (pixel) on the screen is represented by four bits in memory. The bits represent the colour of the pixel on the screen - from black (0000) through a spectrum of 16 colours to white (1111). The EGA screen is 640 pixels wide by 350 pixels high. This implies that the memory requirement to store a full screen of graphics is

$$350 \times 640 \times 4 \text{ bits} = 896\ 000 \text{ bits or } 112\ 000 \text{ bytes}$$

where a byte represents 8 bits or 2 pixels.

The basic IBM PC is limited to 640 000 bytes of application Random Access Memory (RAM) (This is the limit specified by the Disk Operating System (MS-DOS)). Memory must therefore be used as conservatively as possible to avoid memory overflow errors.
The menu driver units make effective use of computer memory that resides above the program (the heap) in RAM. Menus are created and disposed dynamically, freeing the heap for other uses when not displayed.

When a menu routine is called, the program searches a linked list (the freelist) that resides at the top of the heap in memory and calculates if there is enough memory space to store a copy of the screen area where the menu is to be created. This copy is needed when the menu is made to "disappear" from the screen. The menu is not removed, but covered with a copy of what was initially under the menu, giving the illusion of the menu disappearing.

ii) **Highlight bar manipulation**

The height of the highlight bar is calculated by dividing the total height of the menu by the number of menu choices. The width of the highlight bar varies with different widths of menu choices. This is more obvious in horizontal menus than vertical menus, where the spacing between menu choices has to be calculated before the menu is created. The highlight bar is created dynamically in the heap.

iii) **Menu choice - control and commands**

After a menu has been created and the underlying screen area saved, control is passed to the keyboard unit together with a list of active keys. The active keys are the first letters of each menu choice, and the relevant arrow keys (<UP> and <DOWN> arrow keys for vertical menus, <LEFT> and <RIGHT> arrow keys for horizontal menus). The keyboard procedures monitor data entry from the keyboard until either an active key is pressed, the <ENTER> key is pressed, or the <ESCAPE> key is pressed. Control is then passed back to the menu procedure together with the valid keyboard response. This is either an integer between 1 and the number of choices in the menu, an arrow key code, or the <ESCAPE> key code. The menu procedure is then able to move the highlight bar from its present menu choice to the next choice or

to terminate menu selection by making the menu disappear.
The text, graphic and input box menu routine headings available for the creation, management and deletion of menus on a graphics screen are presented together with examples of each in appendix III.1 - Menu Driver Procedures and Functions.

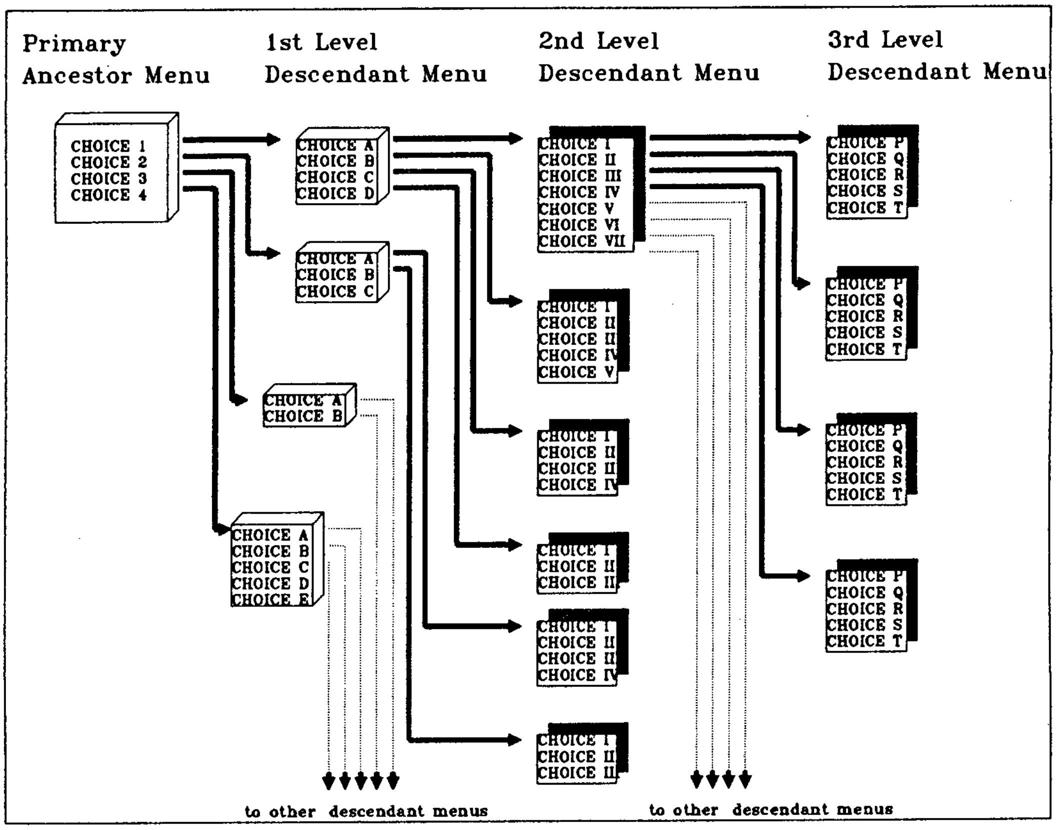
With the Text Menu, Graphic Menu and Input Box Driver units, implementation of a menu driven graphics program becomes simpler - the host program need not control the screen or data entry, only process flow. Also, modifications to the program can be made with minimal adjustment to process flow. The program is easily adapted to other forms of graphic displays with different colour or size specifications.

5.3.2 Flow Diagrams for Computer Program and Audiometric Tests

i) Menu Flow Control

When discussing the flow of control from one menu to another, it is clearer to refer to the level of the menu as its **inheritance level.** A menu may have only one **ancestor** but more than one **descendant.** A **primary ancestor** menu is defined as having no ancestors.
Figure 5.1 illustrates the use of inheritance in a menu based environment.

Figure 5.1 Illustration of the Menu Inheritance

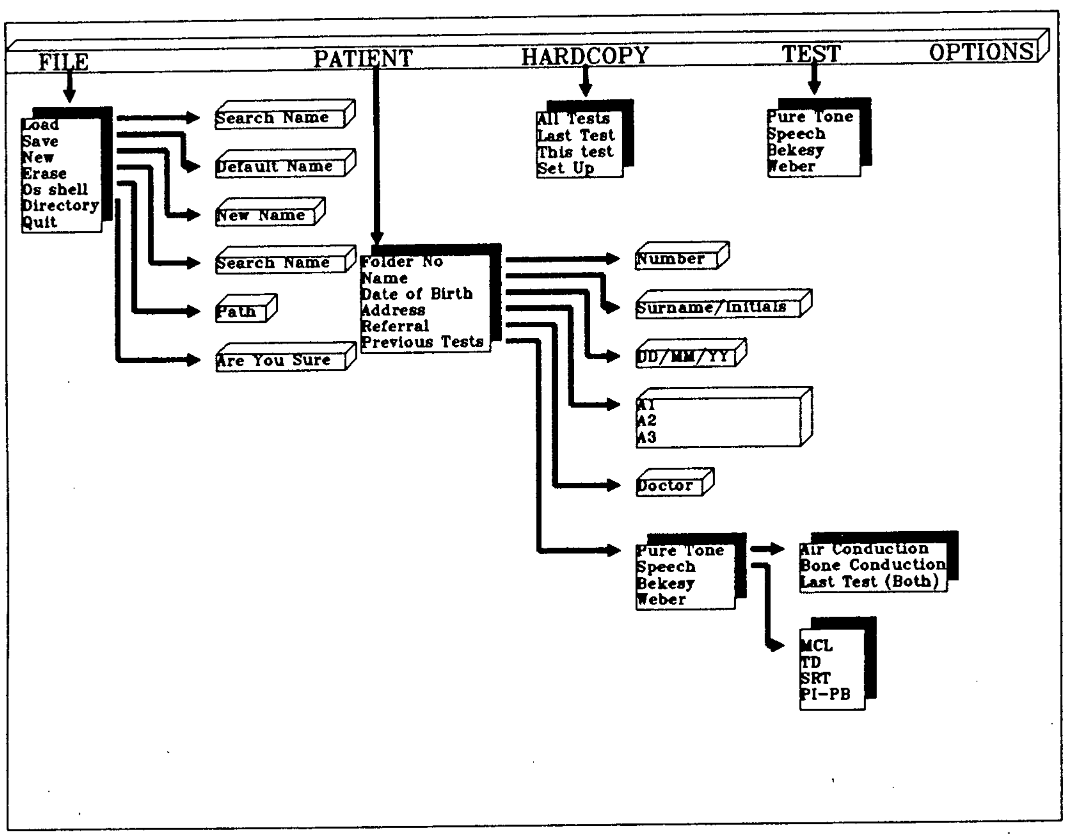


- Inheritance levels are not greater than four.
- A primary ancestor menu is always present at the top of the screen.
- Menus always appear in the same place.
- A menu does not disappear until all descendant menus have disappeared.
- Menus are removed in reverse order of placement.
- Ancestor menus can be accessed by activating an alternate key (see section 5.3.3. keyboard Interface).

Figure 5.2 shows the audiometric menus and input boxes that may be accessed from the primary ancestor. Shadowed menus are descendant menus and 3-D boxes are input boxes for data. The horizontal menu at the top of the figure represents the primary ancestor.

The program is divided into five primary functions - File, Patient, Hard Copy, Test and Options. The primary ancestor or main menu contains these functions as menu choices.
File functions include erasing, saving and retrieval of patient data, directory manipulation and exit commands.
Patient functions allows entry of the patient's personal particulars.
Hard copy functions control what data is to be printed.
Audiometric tests are selected from the Test menu and an **Options menu** is provided for initial program set-up parameters (directory of program, help level, screen colours).

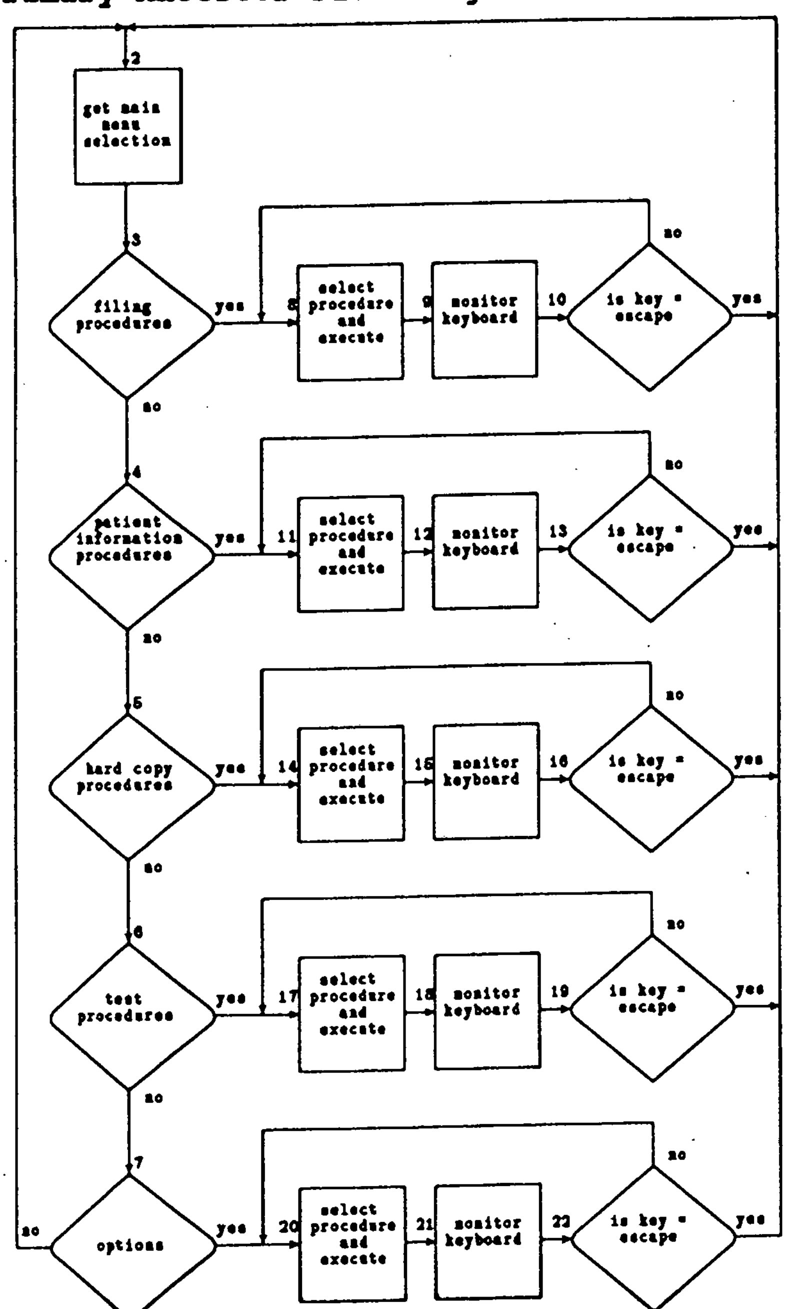
Figure 5.2 Menus and Input Boxes Available From Primary Ancestor.



ii) **Access to Primary Ancestor**

The primary ancestor flow process is represented by the flow diagram in figure 5.3. The primary ancestor is displayed when the program is initially run. The program then waits until a choice is made (step 2 in figure 5.3). If a valid choice has been made, the appropriate primary function (step 3 to 7 in figure 5.3) is selected and executed. After execution, program control is passed back to the primary ancestor. Steps 8 to 19 are control structures for the primary functions.

Figure 5.3 Primary Ancestor Flow Diagram.



The primary ancestor may be accessed at any point in the program by typing an alternate key <ALT M>. At this stage, the highlight bar in the present active menu is turned off, and the highlight bar in the primary ancestor is turned on. The operator is therefore made aware that a change in active menus has occurred.

Functions available in the primary ancestor may now be selected. Once the operator has completed a primary function, the highlight bar in the primary ancestor will be turned off and the last active menu before <ALT M> was pressed will become active again.

Once in a primary function, the user is able to perform all primary function tasks available in that menu. The selection process for the primary functions is the same for all primary functions with only the function changing, not the selection process.

Descendant menus such as patient information, filing or hard copy selection are straight forward and can be followed from Figures 5.2 and 5.3.

iii) **Pure Tone Test**

Initially, the audiologist must choose the type of pure tone test to perform. This includes which ear to test, the method of conduction (air or bone) and whether the non-test ear is to be masked. (Refer to appendix I.3 - Test Procedures.) Because each test has an ASHA defined symbol, the program presents the audiologist with an icon menu of the possible symbols. The symbols are coded in terms of shape and colour (right ear symbols are red, left ear symbols are blue following ASHA standards). The central menu in figure 5.11 is the symbol menu. If the audiologist has selected the pure tone test without masking, the flow diagram is that of figure 5.4. Otherwise, masking has been selected, and the flow diagram is that of figure 5.5.

The pure tone unmasked test flow diagram in figure 5.4 is described below :

After the audiologist has selected the initial test frequency and intensity (1), the audiologist presents the test tone to the patient in the selected test ear (2). If the patient responds (3) the intensity of the test tone is decreased (4), otherwise the intensity is raised (5). The level by which the audiologist increases or decreases the test tone depends on the test method employed. Carhart and Jerger (1959) prescribe an increase of 5 dB and a decrease of 10 dB. If a level has been crossed at least three times (6) then this is the patient's hearing threshold (9). If the test intensity has reached the limit of the audiometer (7) and the patient has still not responded, then a 'No Response' is recorded (8) for that frequency. Once the patient's hearing threshold has been found, the audiologist selects another frequency and the test is repeated.

With masking (refer to figure 5.5) the initial selection also includes the masking level presented in the non-test ear (2). After the tone is presented (3) and the patient does not respond (4), the audiologist raises the level of test tone (5) and presents the tone again (Hood method). This process is repeated until the patient responds (6). The masking level is then raised (7). If cross over occurs (8) overmasking has taken place (12): if not, and the patient responds for at least three levels of masking (10), the plateau has been reached (11). (Refer to appendix I.4 - Masking, for a description of the masking procedures.)

Figure 5.4 Flow Diagram of Pure Tone Test Without Masking

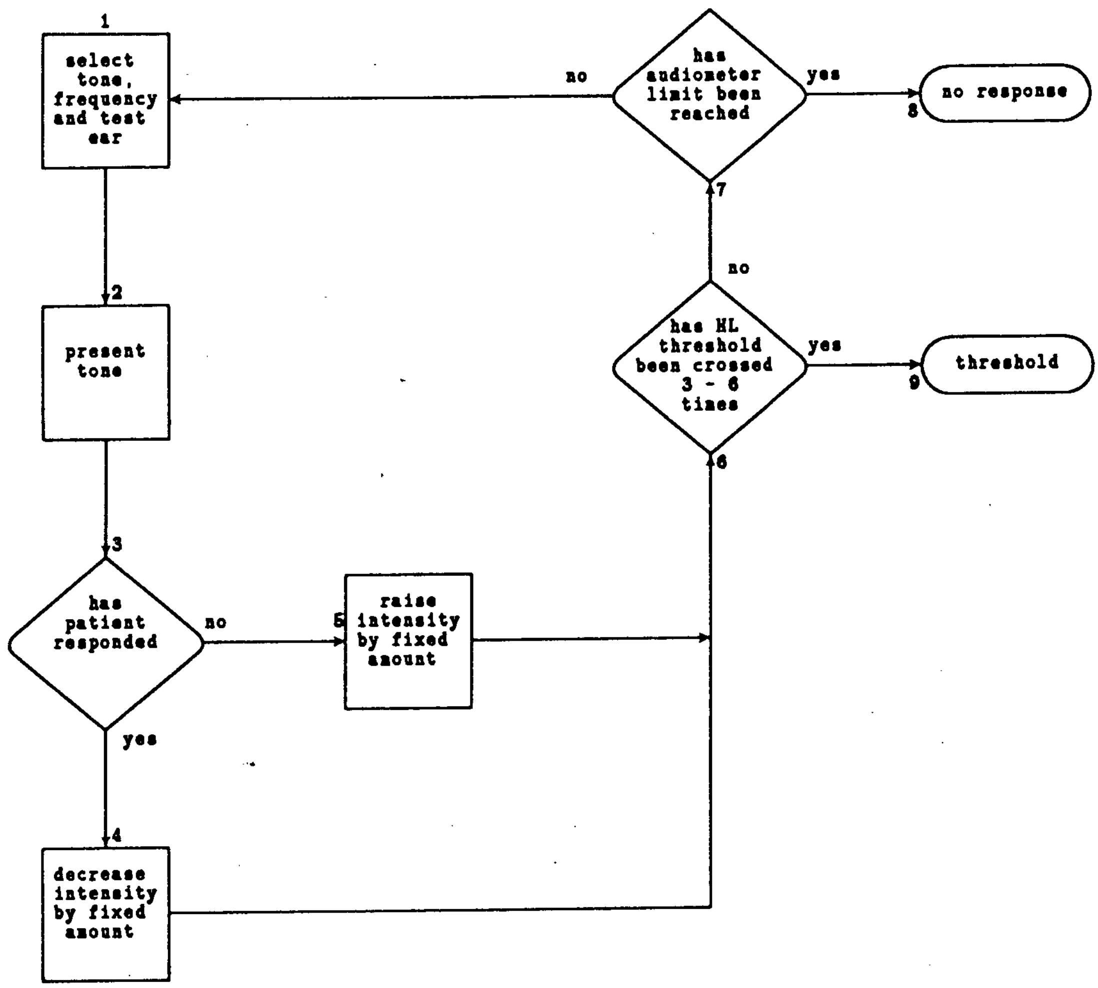


Figure 5.5 Flow Diagram of Pure Tone Test With Masking

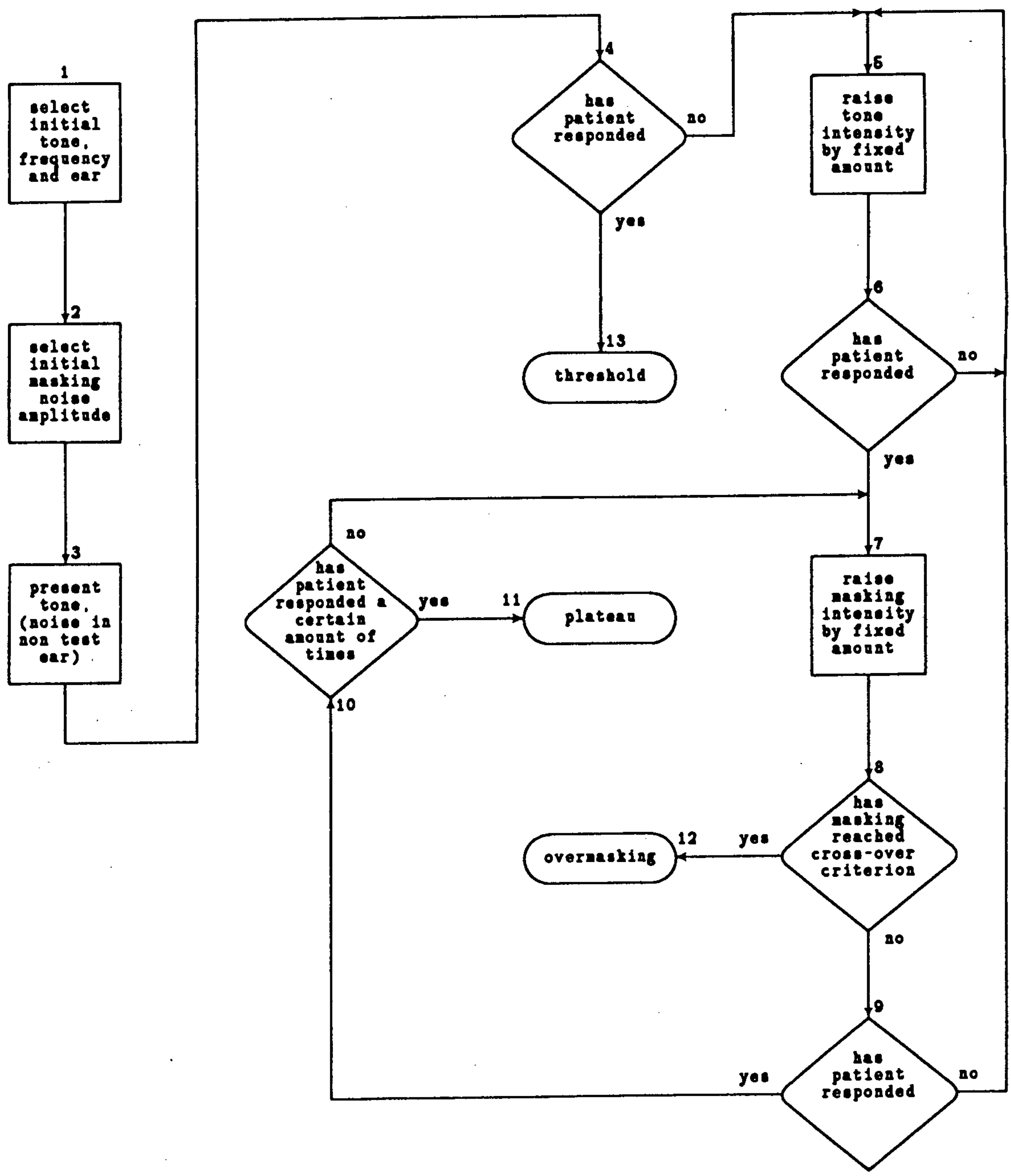
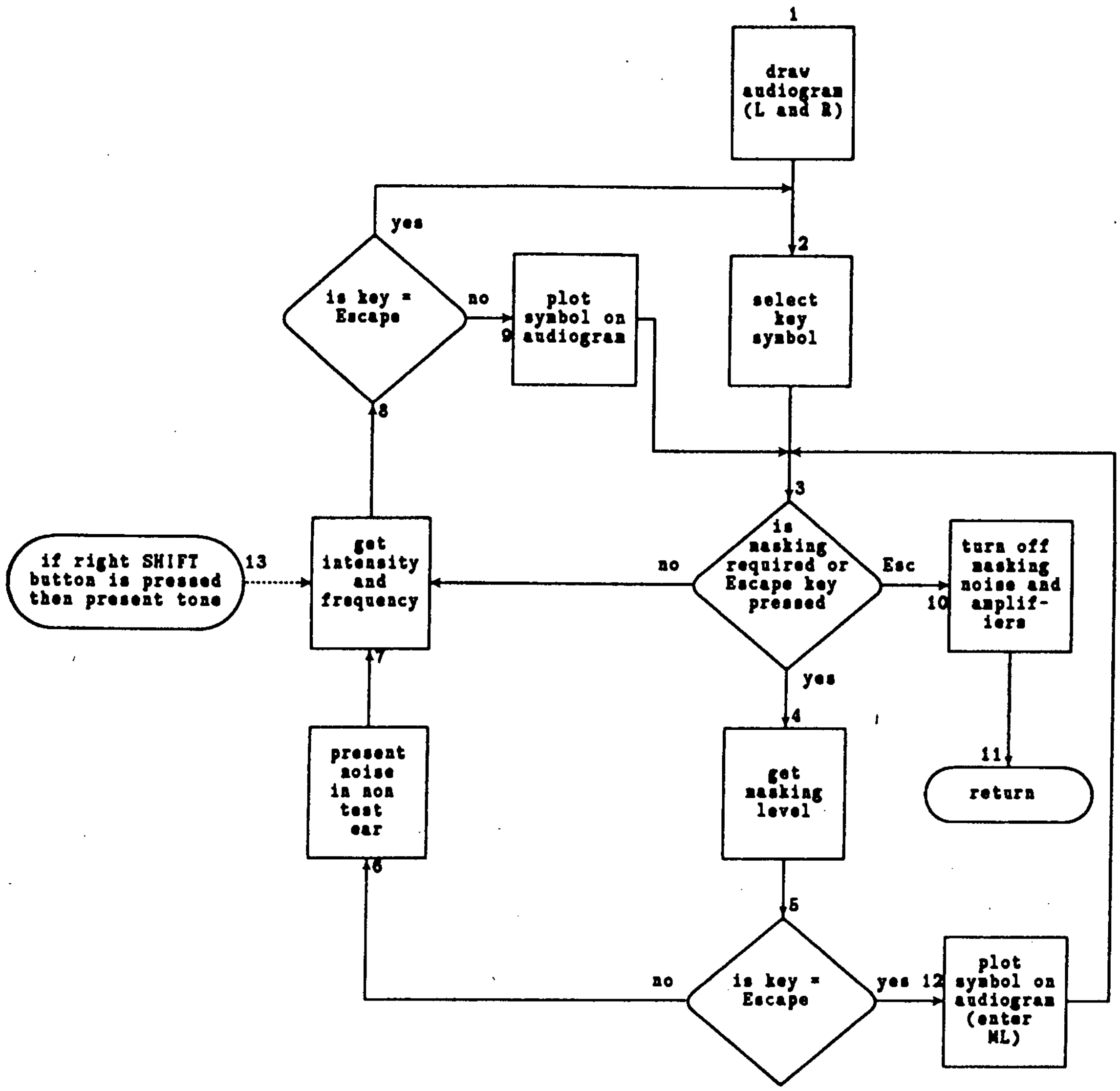


Figure 5.6 Program Flow Diagram



From these two **audiologist flow diagrams** (figures 5.4 and 5.5) it is obvious that the audiologist is faced with choices throughout the test. The direction chosen by the audiologist depends on the response of the patient. Patient response can only be assessed by the audiologist. The **program flow diagram** in figure 5.6, is different to the audiologist flow diagrams. Whereas the audiologist flow diagram is dependent on patient response, the program flow diagram is dependent on menu choice. Steps 2, 4, and 7 of figure 5.6 require the audiologist to make a menu choice regarding the test ear, the masking level and the test signal frequency and intensity.
(See section 5.4 for screen formats.)

iv) **Speech Test**

The ascending method of **SRT testing** with spondaic words as proposed by ASHA (1979) is represented in figure 5.7. The audiologist follows the test procedure and makes judgements regarding patient response in steps 3, 6 and 7.
MCL and TD tests require free speech instead of spondaic words. The audiologist adjusts the speech intensity until the patient indicates that the intensity is at its most comfortable (MCL) level and that the intensity is discomforting (TD). It is not necessary to represent the MCL and TD tests with flow diagrams.

Speech discrimination requires the audiologist to choose a word list, read the words one at a time and monitor how many words the patient is able to repeat correctly. This is repeated for at least three different intensities.

The combined **program flow diagram** of the SRT, MCL, TD and Discrimination tests is shown in figure 5.8. In all these tests

Figure 5.7 Flow Diagram of Speech Reception Threshold Test

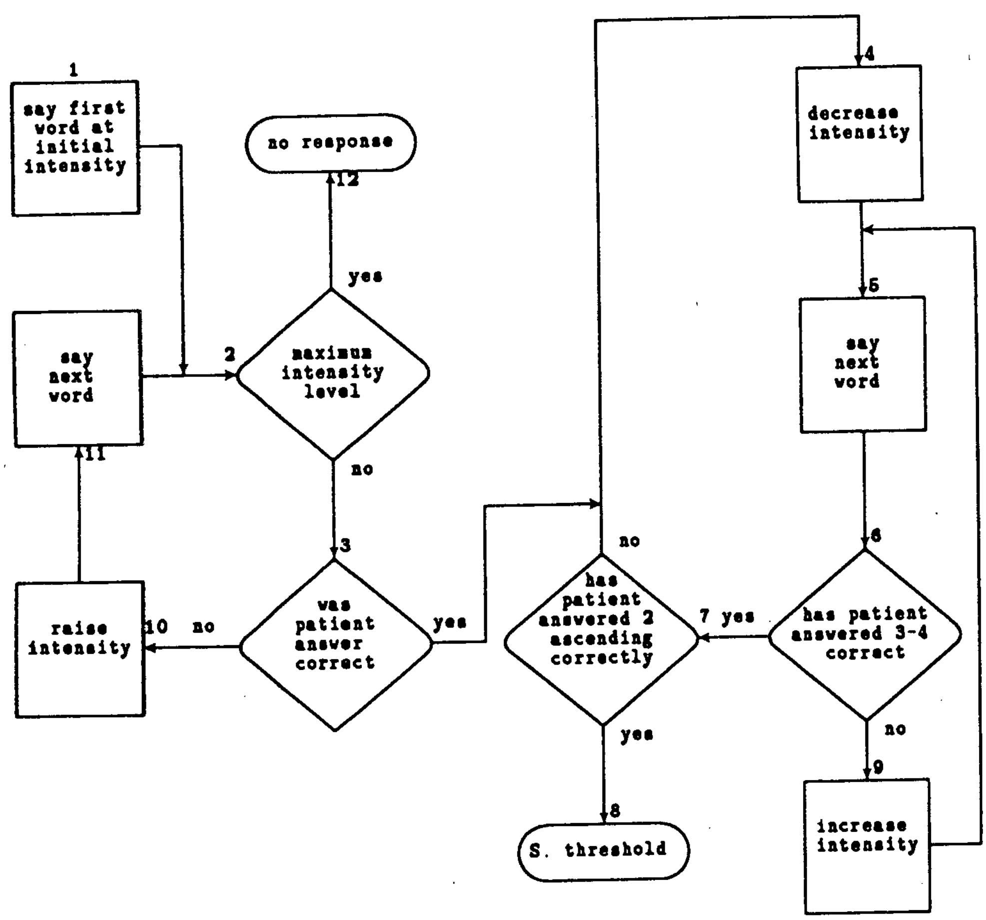
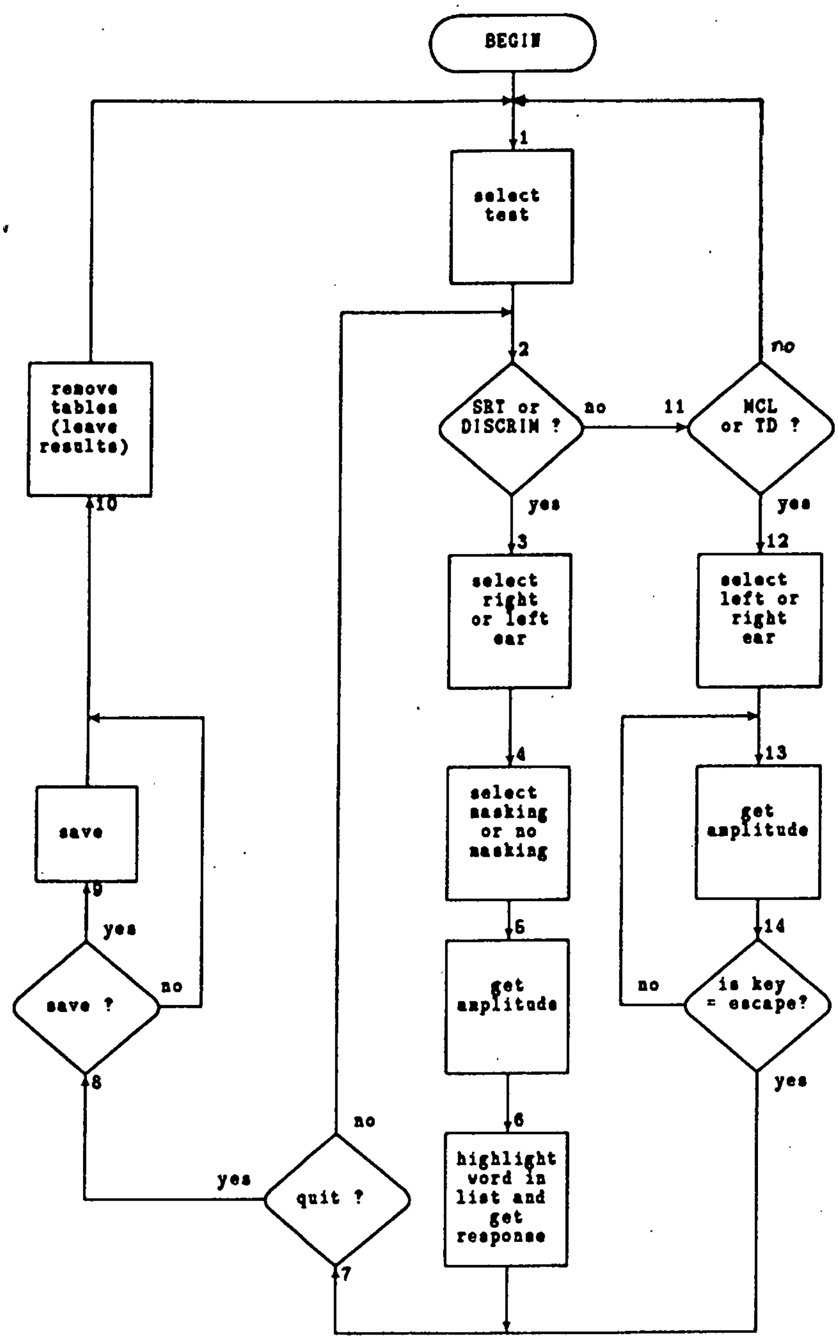


Figure 5.8 Combimed Program Flow Diagram of Speech Reception Threshold, Most comfortable Level, Threshold of Discomfort and Speech Discrimination Tests



the audiologist has to select which ear to test, whether masking is to be applied to the non-test ear, and the intensity of the signal. With Speech Discrimination and SRT tests the audiologist also has to select a list of words to be read to the patient.

Results obtained from the tests are entered into a table together with the calculated Dynamic Range (DR) of the patient. The DR is calculated by subtracting the patient's SRT value from the TD.

When results for the above tests are obtained, the relevant area in the results table is filled. If the test is repeated, the value is updated. The audiologist is able to see all speech results on the screen at the same time. If pure tone testing has already been conducted, the pure tone average is also displayed on the screen.

When a test involving word lists is selected, an area of the screen is configured to display this word list. A cursor in this area indicates to the audiologist the last word read. All words read are marked with either a tick (√) for correct responses or a cross(X) for incorrect responses. (see section 5.4 for screen formats.)

5.3.3 Keyboard Interface

The keyboard interface is controlled by the keyboard interface unit. Valid keys are passed to this unit, and the unit scans the keyboard until a valid key is pressed. This may include selected 'qwerty' keys, numeric keys, function keys or a combination of these. Appropriate action is taken once a valid key is pressed.

Selection for all menus is made by moving a highlight bar in the menu over the required choice and pressing the <ENTER> key or by pressing the first letter of the menu choice. If no choice is required (the menu was selected by mistake) then the <ESCAPE> key removes the menu from the screen and returns an escape value to the calling program.

Alternate keys may be used at any stage in the program to select primary ancestor functions. Holding down the <ALTERNATE> key while pushing the <F> key, for example, will allow access to the file functions of the primary ancestor. The selected function is overlaid on the existing screen. Program control is passed to this function, together with a set of legal keys (viz. the first letters of the various functions). The keyboard unit will now monitor the keyboard for these legal keys. To access the file function for example, the alternate key and the letter F are pressed. Control is then passed to that function.

The remote control circuitry is interfaced to the keyboard unit. The remote control keys are configured to emulate the arrow keys, the <ENTER> key and the <ESCAPE> key. When performing the test remotely, the audiologist may manoeuvre between menus with the remote control unit.

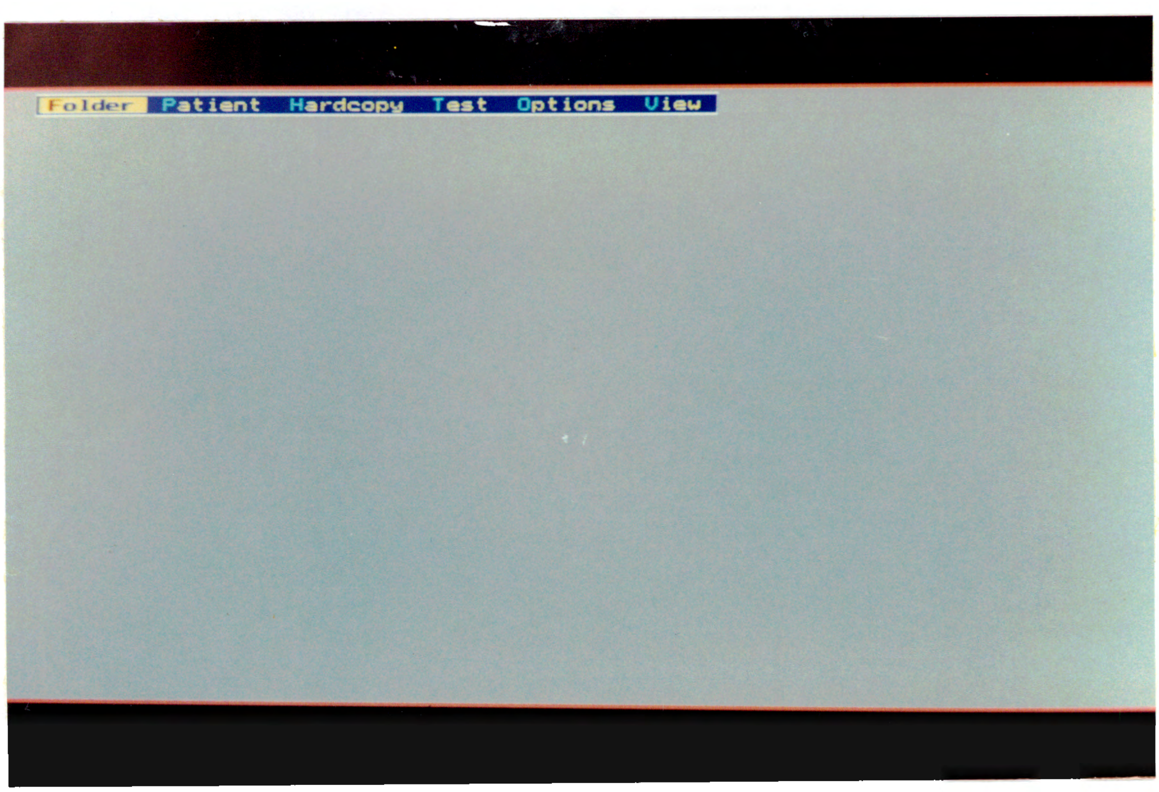
5.4 **SCREEN FORMAT**

The layout of the screen is explained under three headings. These are the primary ancestor and descendant menu format, the pure tone test format and the speech tests formats.

5.4.1 Primary Ancestor and Descendant Menu Format.

The primary ancestor menu (see figure 5.9) is displayed along the top of the screen. A highlight bar moves horizontally along the menu by the action of the left and right arrow keys. Once a choice has been made, a vertical menu is displayed under the primary function. A highlight bar in the vertical menu can now be scrolled through the descendant menu choices.

Figure 5.9 Primary Ancestor Menu Format



Once a choice in a descendant menu has been made, an input box is displayed (except in the case of patient previous tests) alongside the descendant menu choice. The user is also able to scroll through the input boxes with the arrow keys instead of accessing the input boxes via the ancestor menu. This allows for fewer key strokes and shorter access time when entering patient information, for example.

5.4.2 Pure Tone Format.

i) **Unmasked**

Figure 5.10 to figure 5.15 show the screen format for the pure tone test without masking.

The initial test selection menu is shown in figure 5.10. Once the pure tone test has been selected, the menu disappears and is replaced with the screen of figure 5.11 showing the active audiograms and pure tone test symbol menu. The audiologist may now select one of the eight ASHA symbols from the pure tone test symbol menu. Upon selection, the symbol menu disappears and a highlight bar together with a ladder menu is displayed at 1000Hz and 30dB in the test ear audiogram (see figure 5.12). The audiologist is able to select the intensity of the test signal by moving the highlight bar up or down the ladder in increments of 5 dB (SPL), and the test frequency in half octave increments, by moving the ladder right or left with the arrow keys. The test signal is delivered as long as the right <SHIFT> key is pressed.

If the audiologist wishes to indicate a positive response from the patient to a test signal, the <ENTER> key is pressed and the relevant pure tone symbol is placed on the audiogram at the correct point (see figure 5.13). When two or more pure tone symbols are displayed on the audiogram, they are joined by a line. When an incorrect symbol has been displayed, no response is recorded or the audiologist wishes to exit the test, the <ESCAPE> key is pressed and the menu in figure 5.14 is displayed.

If the first choice - Remove last entry - is selected, the audiogram is updated by removing the symbol in the column of the ladder. The lines joining the remaining symbols are then generated. A Standard symbol for 'no response' does not exist. However it seems common to use 'CNE' for no response. If this choice is selected, the respective letters are displayed at the correct position on the audiogram.

This menu also contains a 'Comment' choice. Although the 'Comment' function is activated by the <ALTERNATE> <C> key at any stage, it is added as a menu choice to prompt the audiologist to make a note at this stage. (Figure 5.18 shows the overlaid text editor used for comments.)

When the audiologist has completed the pure tone test and exits from the active audiogram, the menu in figure 5.15 is displayed, prompting the audiologist to enter the test reliability.

ii) **Masked**

Standard ASHA symbols for masked pure tone audiometry are depicted as the lower four symbols in the symbol menu of figure 5.11. Once the audiologist has selected a masked pure tone test, the symbol menu disappears and the screen is replaced with the screen of figure 5.16. The active audiogram is displayed as in the unmasked pure tone test. A masking level menu is displayed in the area of the audiogram of the non-test ear, and in the colour of the non-test ear. In the lower section of the screen a table of effective masking levels in displayed. Once the audiologist has determined the hearing level at a specific frequency, the masking level is entered into the table automatically. If the presence of a plateau at a certain frequency is to be indicated, the audiologist may select the first entry in the masking menu (Plateau) of figure 5.16. The plateau levels are displayed in the conventional way in the masking table (lower limit, slash, upper limit).

If a threshold is indicated, the audiologist presses <ENTER> to place the masked symbol on the audiogram, and the program automatically switches to the masking level menu. The audiologist may now either change the masking level with the arrow keys, or return to the audiogram with the <ENTER> key.

If the <ESCAPE> key is pressed, the exit menu in figure 5.14 is displayed. When the test is exited, the masking level menu and the masking table disappear, and the symbol menu of figure 5.11 is displayed.

At any stage in the pure tone test, the audiologist may examine or compare results from a previous test. This is accomplished as shown in figures 5.17, 5.18 and 5.19. The primary ancestor function - Patient - is accessed (by typing <ALTERNATE> <P>) and the patient menu is displayed. From the patient menu, 'previous tests' and 'puretone' are subsequently selected. The screen displays the previous tests and overlays the present test so that the audiologist may compare results. The program will wait for a key to be pressed and then remove the previous tests and return to the pure tone test.

Figure 5.10 Initial Test selection Menu

Figure 5.11 Dynamic Audiogram and Pure Tone Test Symbol Menu

Figure 5.12 Ladder Menu at 1000Hz on Dynamic Audiogram of Right Ear

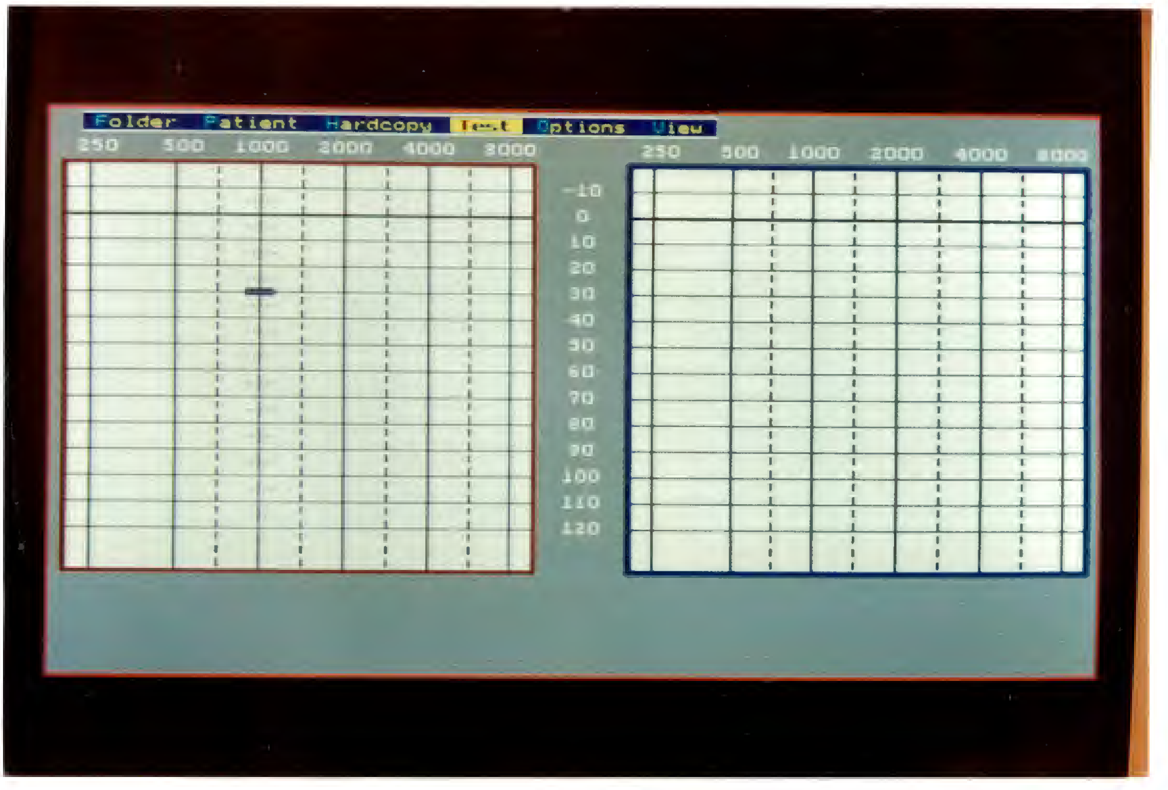


Figure 5.13 Symbols Displayed on Dynamic Audiogram

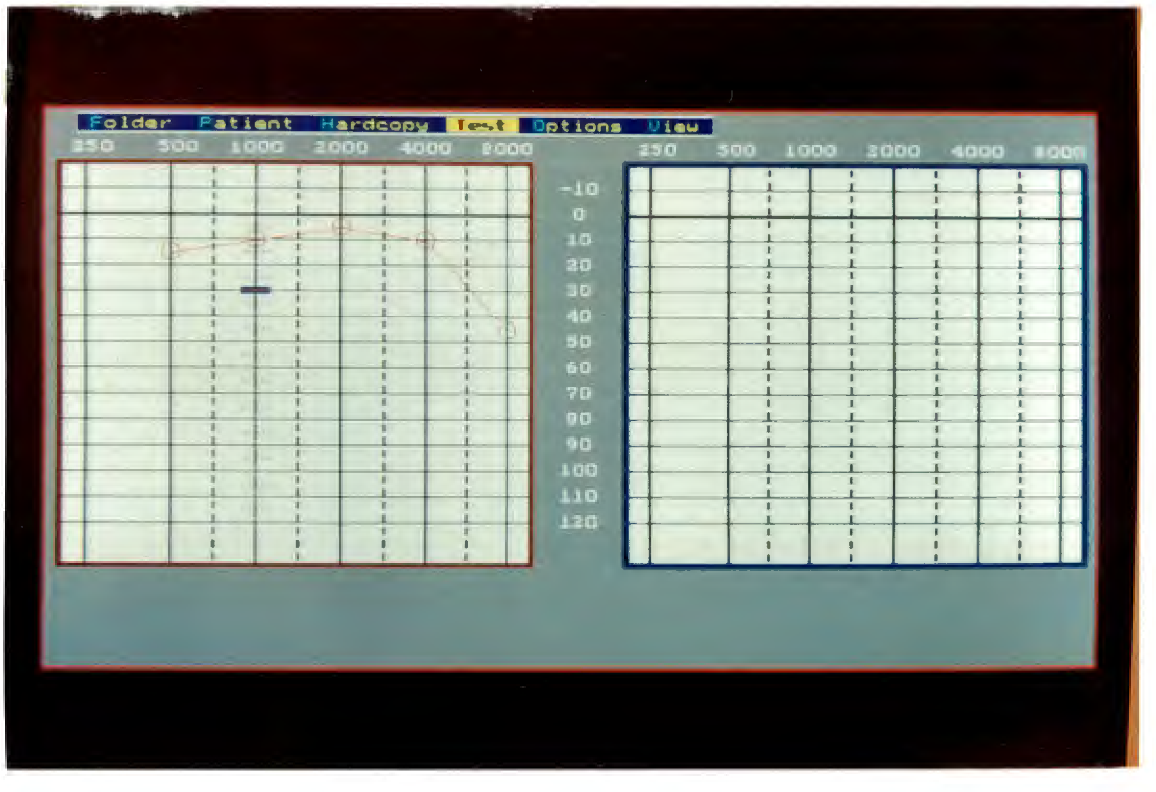


Figure 5.14 Escape Menu Displayed over Dynamic Audiogram

Figure 5.15 Test Reliability Displayed over Dynamic Audiogram

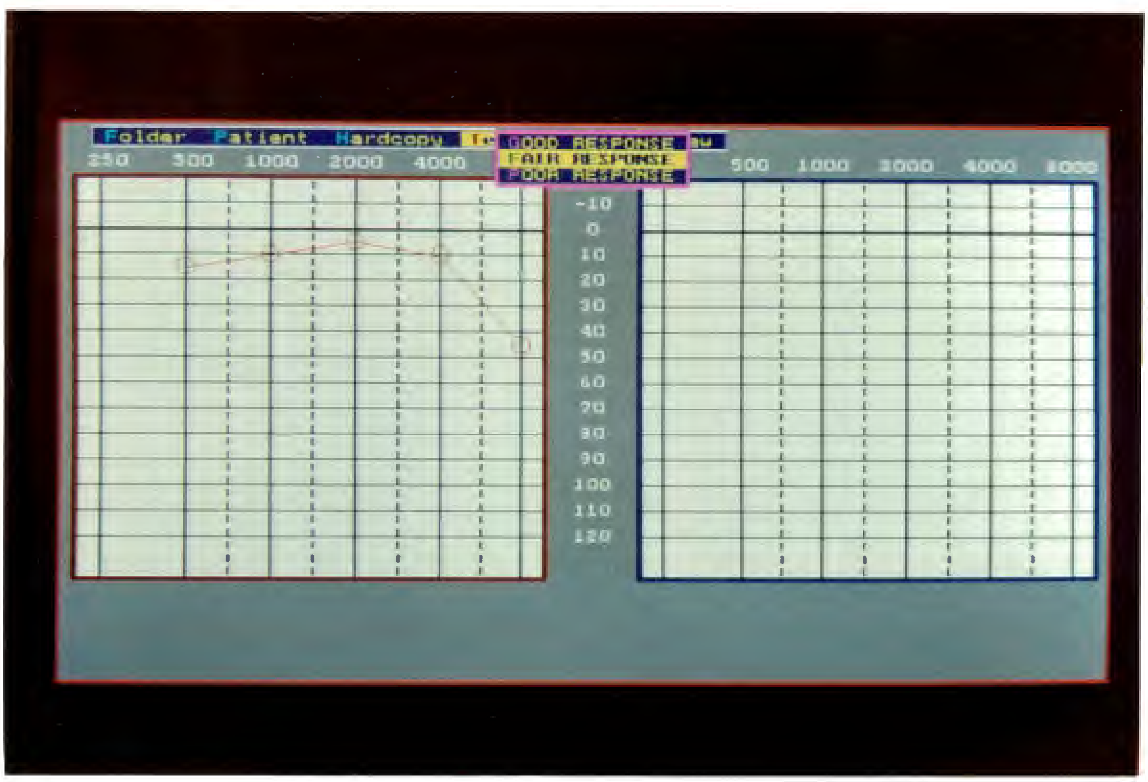


Figure 5.16 Dynamic Audiogram for Pure Tone Masked Test

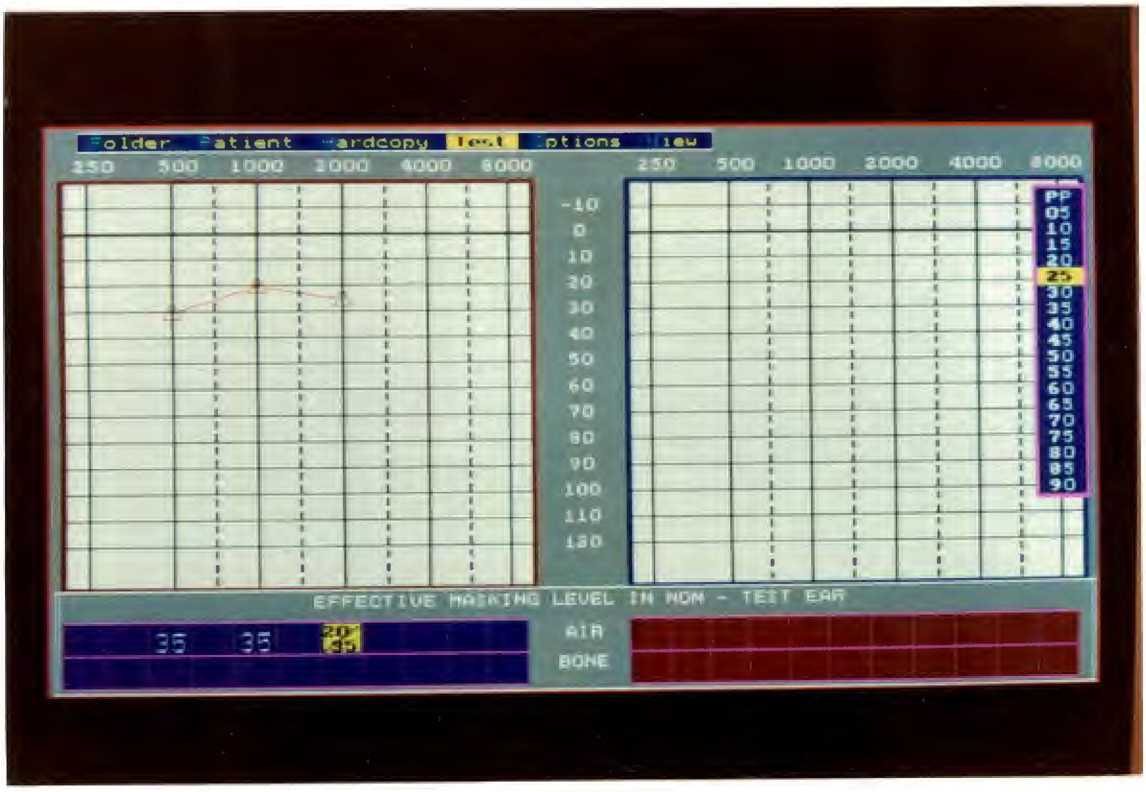


Figure 5.17 Previous Tests Menu

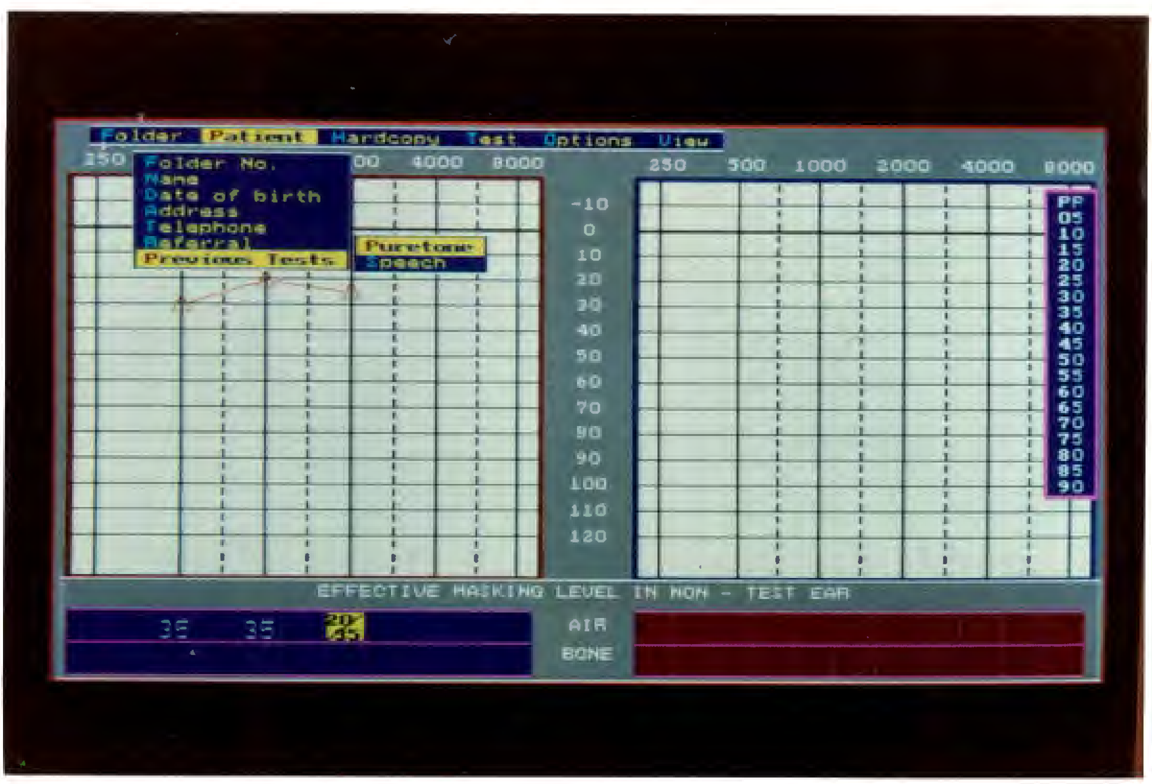


Figure 5.18 Previous Tests Screen for Pure Tone Results

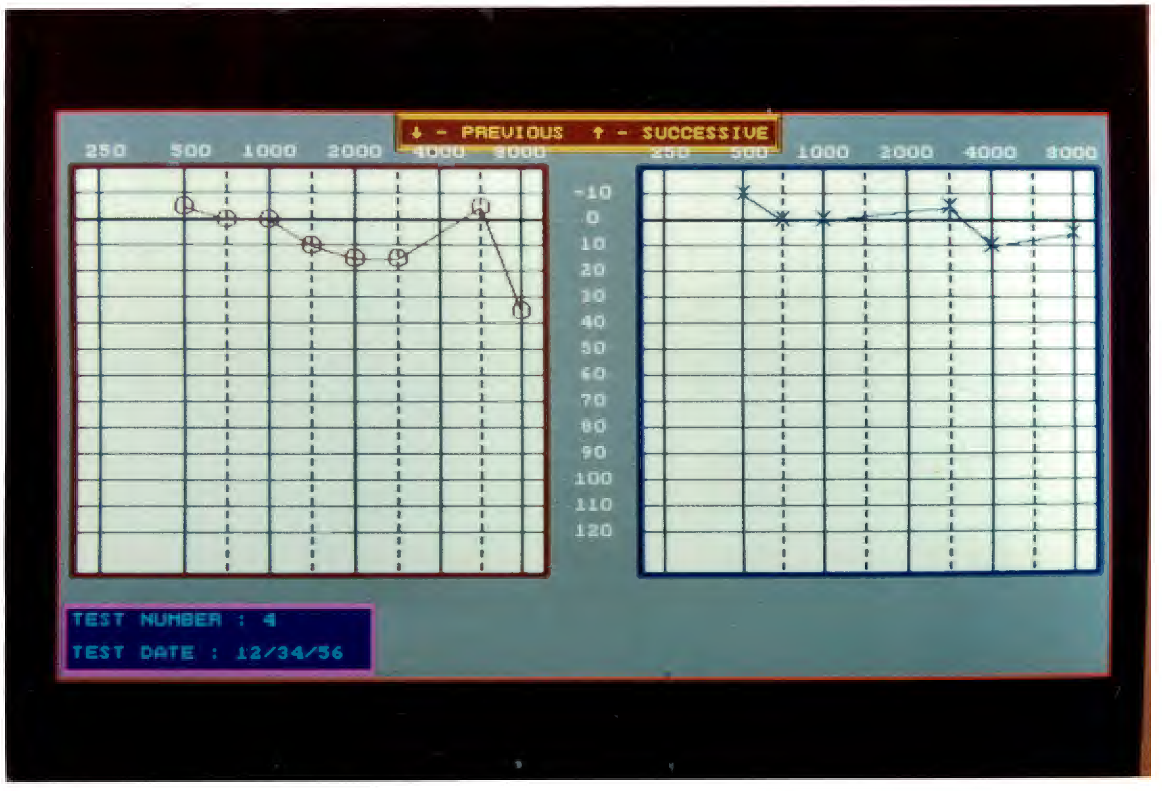


Figure 5.19 Previous Tests Screen for Speech Results

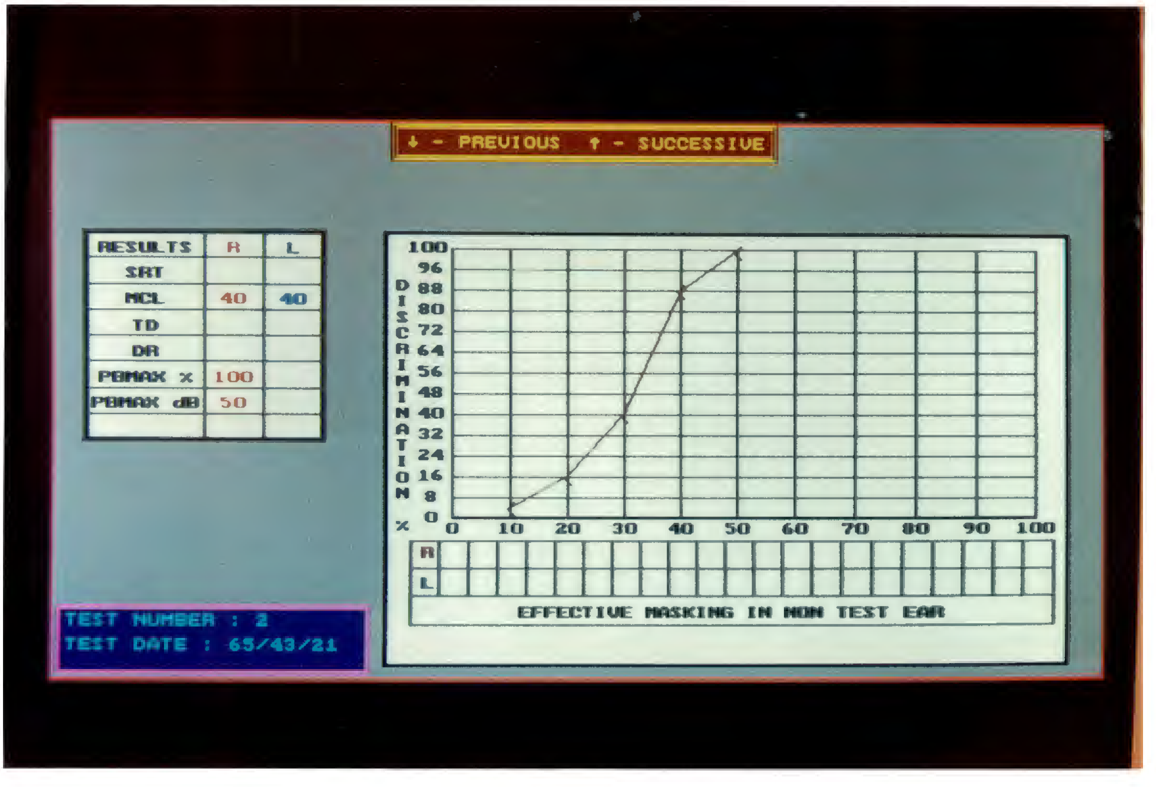


Figure 5.20 Overlaid Text Editor Used for Comments

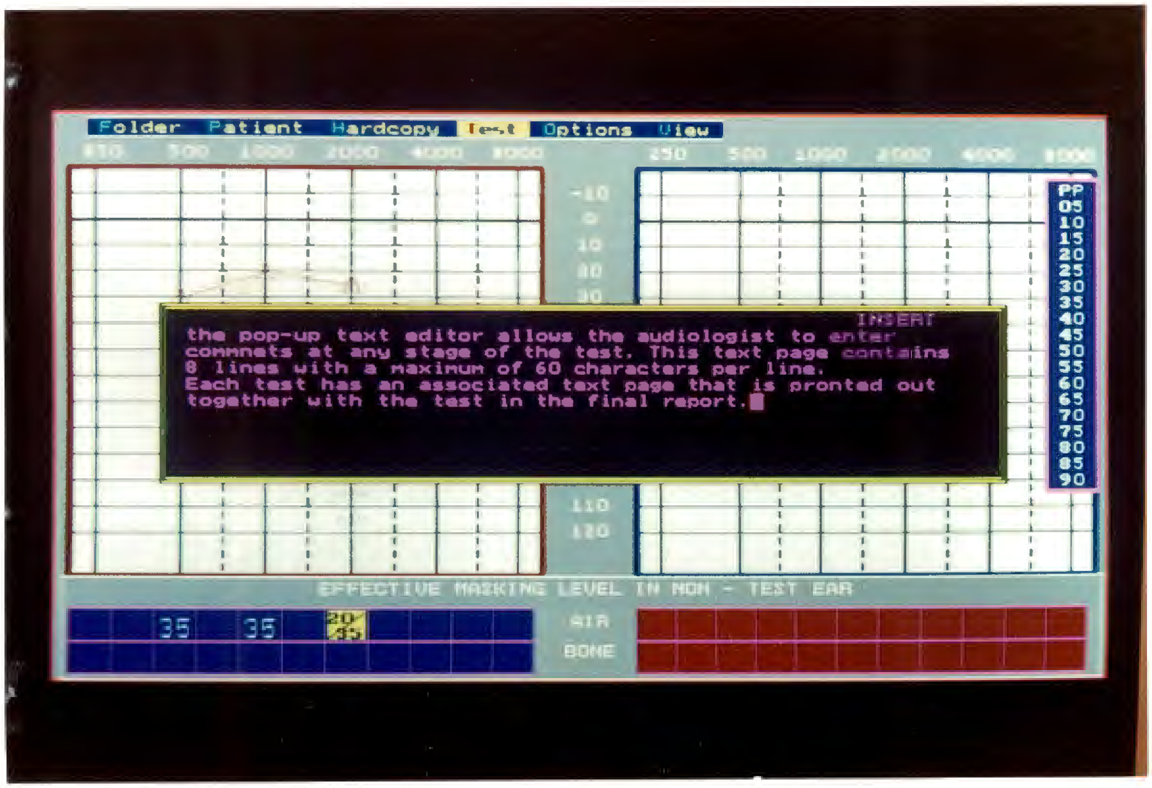


Figure 5.21 Speech Tests Menu

5.4.3 Speech Tests Format.

If the 'speech tests' function is selected from the test menu of figure 5.10, the test menu disappears and is replaced by the speech test menu of figure 5.21. The audiologist may choose a speech test by placing the highlight bar over the speech test and pressing <ENTER>, or pressing the first letter of the speech test.

Figure 5.22 shows the next three menus that are displayed sequentially after the speech test has been selected. These select the ear to test, masking noise in the non-test ear, and the language to be used.

i) **SRT, MCL and TD Tests**

Once the audiologist has selected the above parameters, these menus disappear and the screen displays a speech level menu, a masking menu, a word list and a results table. The position of the speech level menu and the masking level menu depend on the choice of ear, and whether masking is required. (If masking is not required, the masking menu is not displayed. The speech level menu is displayed on either the left or the right side of the screen, depending on whether the right or left ear has been selected, respectively.) Figure 5.23 represents the SRT test in the right ear with masking noise in the left ear. The number of correct responses out of a possible six responses is recorded next to the level being tested. (Appendix I.3 - Test Procedures, describes the speech tests.) Correct and incorrect words are marked with a tick (√) or a cross (X) respectively, as the patient responds to the words.

The Audiologist may move between menus with either the <ENTER> or the <ESCAPE> key. The <ENTER> key moves the active highlight bar from the masking level menu to the speech level menu and then to the word list. The active menu is indicated with a flashing cursor. The <ESCAPE> key allows the audiologist to re-select from a previous menu.

When a total of six words have been marked either correct or incorrect, the audiologist must change the masking level or the speech level. At this point the highlight bar of the speech level menu is automatically incremented 10 dB(SPL), and the last result is displayed next to the last speech level tested. At any stage the audiologist can therefore see the previous results. Figure 5.24 represents the same test as figure 5.23, but the audiologist has already tested six words at 30 dB(SPL) and six words at 40 dB(SPL).

Pressing the <ESCAPE> key displays the quit menu shown in figure 5.25. When the audiologist selects 'Exit and Save', from the menu, the final result for the SRT, MCL or TD test is displayed in the appropriate position in the results table as shown in figure 5.25. If the audiologist selects 'Dont save and Exit' the results table is not updated and the test is ended.

ii) **Speech Discrimination Test**

'Speech Discrimination' is selected from the speech tests menu in figure 5.21. The audiologist initially selects the test ear, whether masking is needed for the non-test ear, and the language of the test. These choices are made from the menus shown in figure 5.22.

The screen then displays a masking level menu (if masking is selected), a speech discrimination gram and a word list as shown in figure 5.26. The screen positions of the masking level menu and the word list depend on the ear being tested. If the test ear is the right ear, the word list is displayed on the left of the screen and the masking level menu on the right of the screen. If the test ear is the left ear, the positions of the masking level menu and the word list are reversed.

A flashing vertical arrow on the horizontal axis of the discrimination gram (speech level) indicates the speech level being tested. The audiologist selects the speech level to test by moving the arrow to the required level with the <LEFT> or <RIGHT> arrow keys and then pressing <ENTER>. The relevant ASHA symbol is displayed at the selected speech level in the discrimination gram and the audiologist may now mark correct and incorrect words in the word list with the <LEFT> and <RIGHT> arrows (<LEFT> arrow implies correct, <RIGHT> arrow implies incorrect).

The speech discrimination gram (percent correct) vertical axis represents the percentage of words that the patient has correctly answered. A maximum of one hundred percent is recorded when the patient is able to correctly repeat all twenty five words. Once all twenty five words have been presented, the audiologist may change the speech level or the masking level. If a speech level is changed, the words in the word list are changed. If more than one speech level has been tested, the symbols on the speech discrimination gram are connected with a solid line as in figure 5.27.

The maximum percentage words correct (PBMAX) that the patient scored in the speech discrimination test is displayed in the results table when the test is ended (the audiologist presses <ESCAPE> and selects one of the exit choices from the exit menu in figure 5.25).

Pressing <ALT V> will bring up the view screen as shown in figure 5.28, with a summary of the patient information, tests done and a graphic representation of pure tone and speech discrimination tests.

This allows the audiologist to view the results before they are saved or while comments are made.

Figure 5.22 Speech Test Configuration Menus

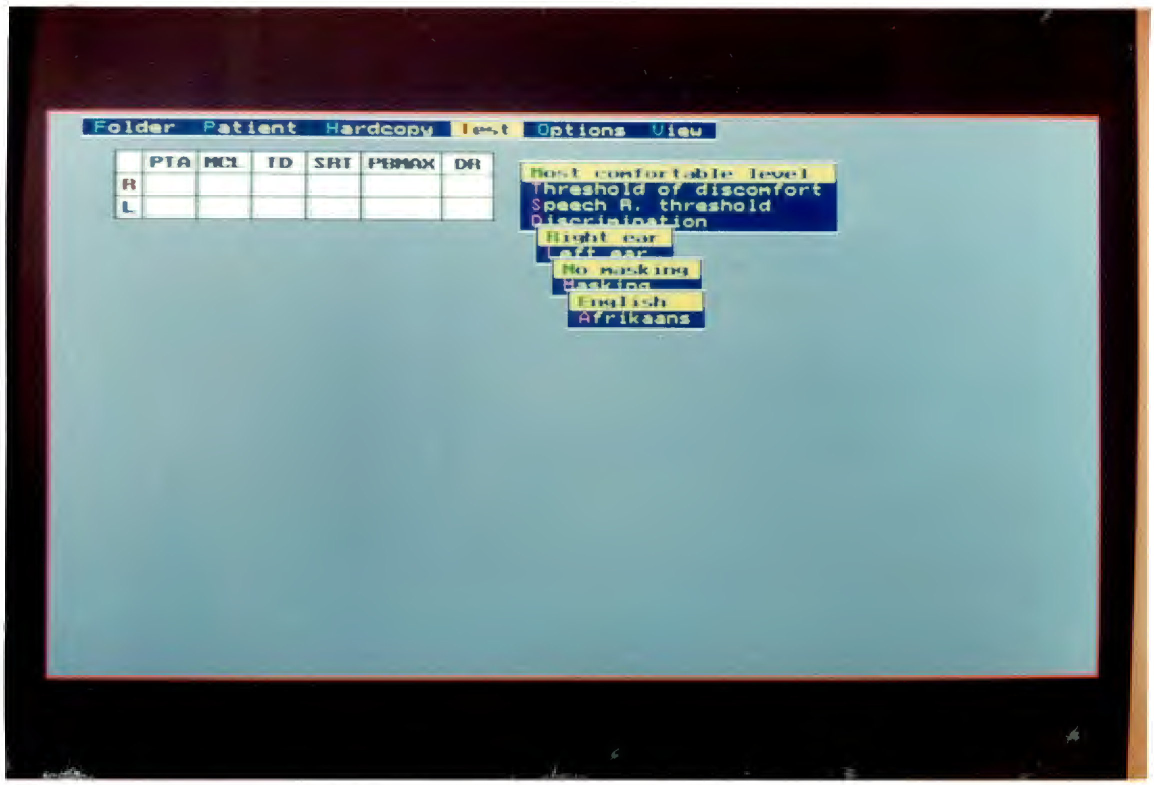


Figure 5.23 Speech Reception Test in Right Ear

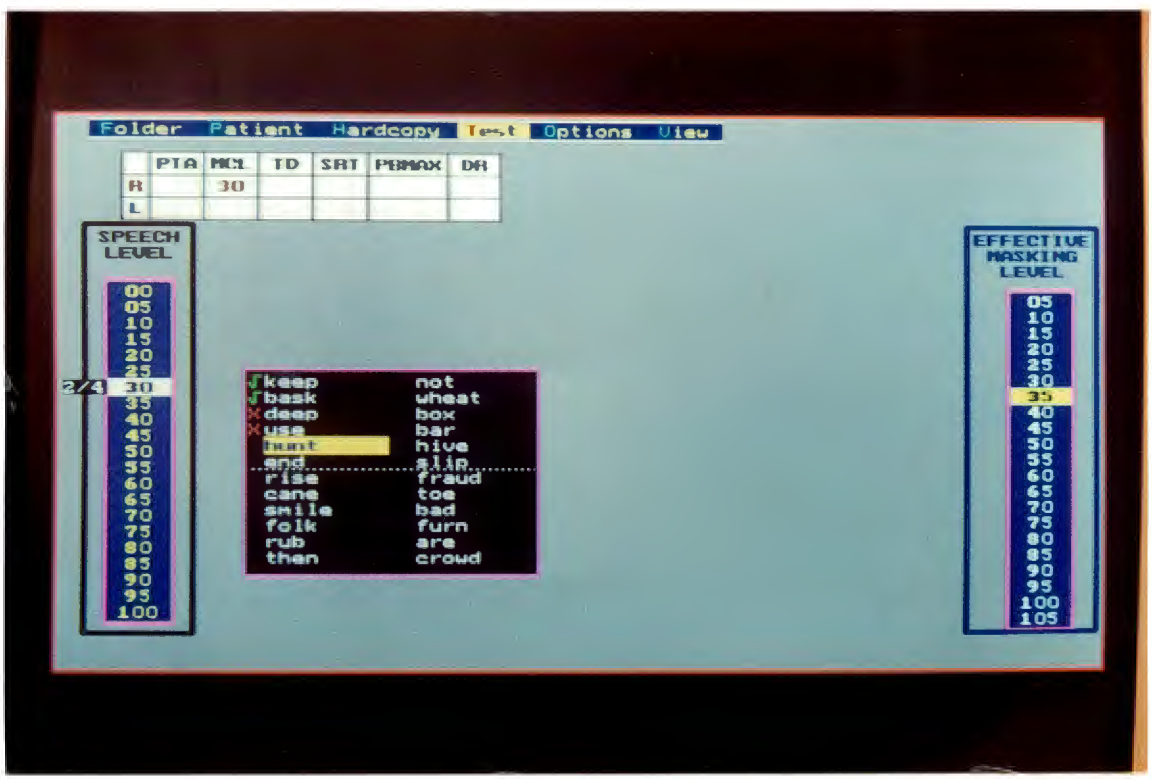


Figure 5.24 Speech Reception Test with Ticked off Words

Figure 5.25 Escape Menu Displayed

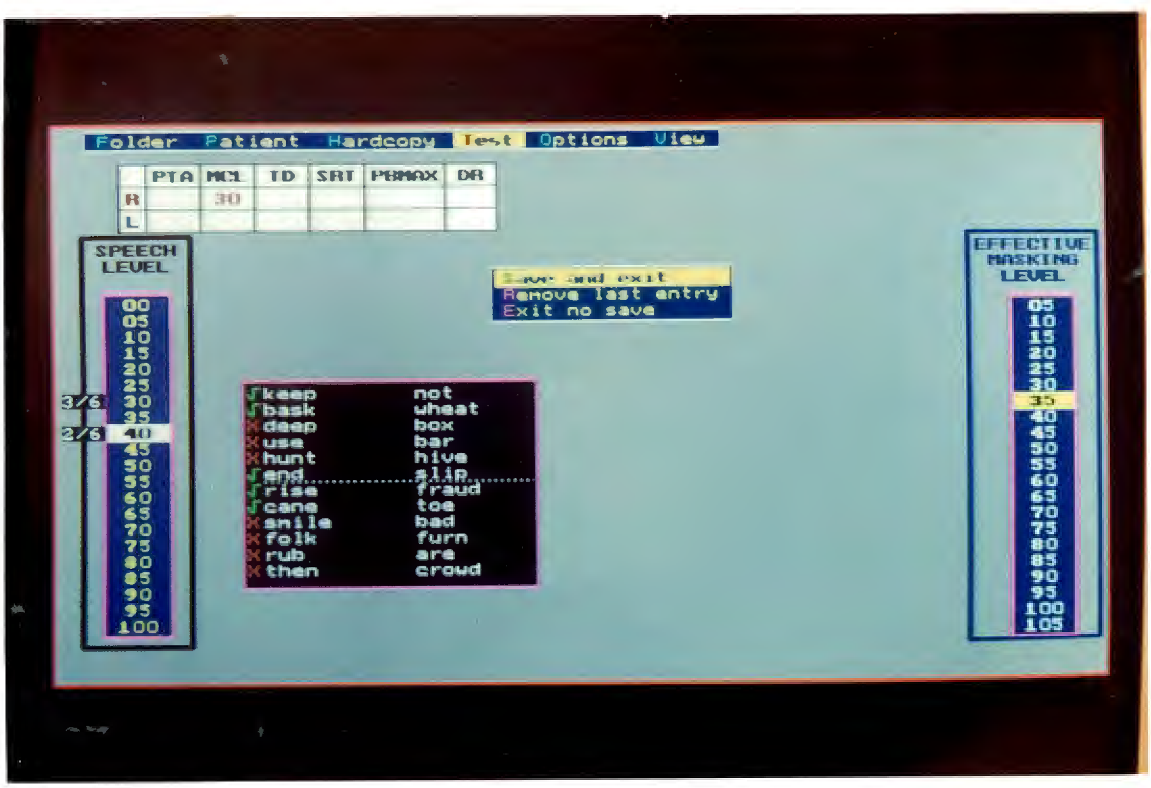


Figure 5.26 Speech Discrimination Test Screen

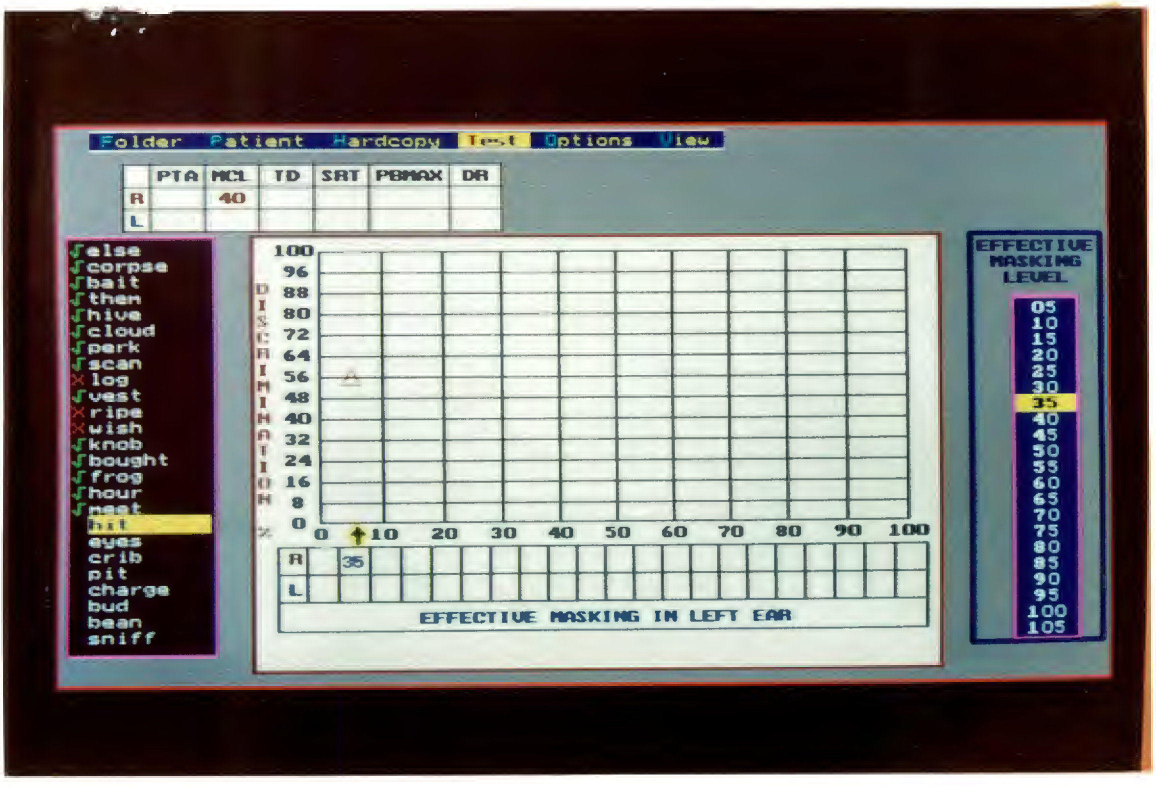


Figure 5.27 Symbols Placed on Dynamic Discrimination - Gram

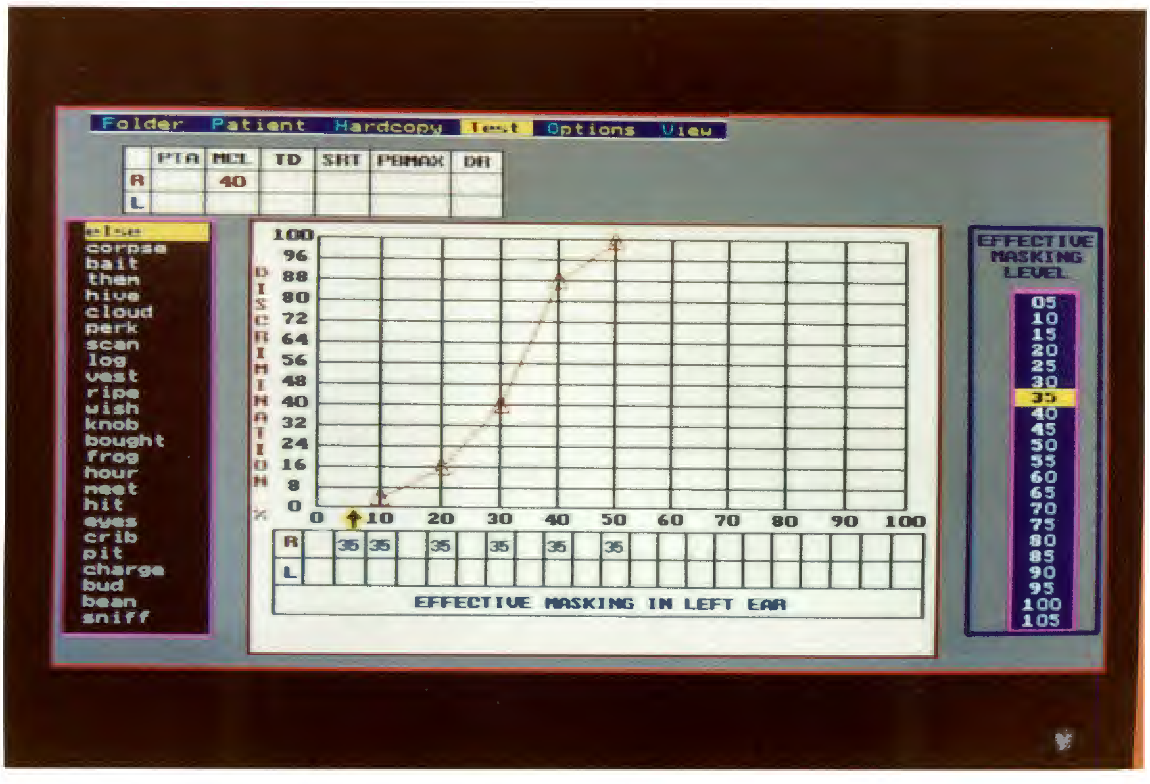
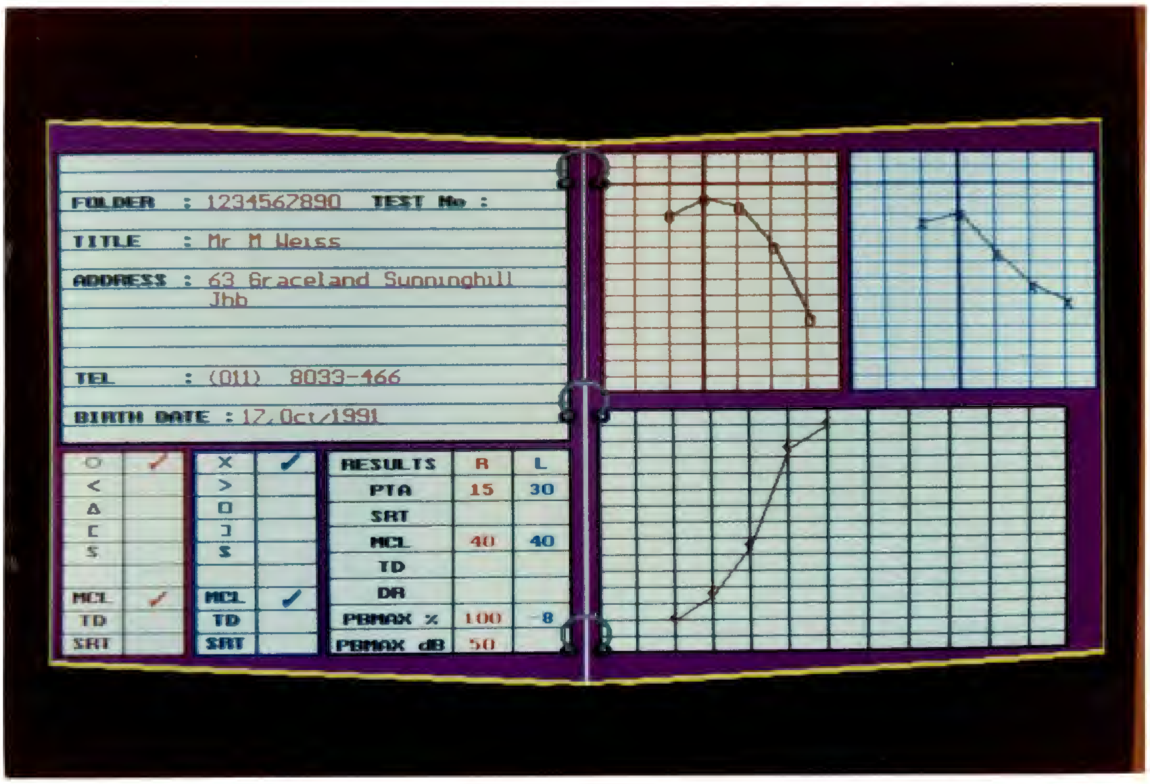


Figure 5.28 View Screen



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CHAPTER SIX - DISCUSSION

6.1 LITERATURE SURVEY

The theory of audiologic assessment and evaluation is well documented and researched. Unfortunately, this is not the case for computer based audiologic assessment and evaluation. The literature survey does not reveal significant information in this respect.

The most important clues for the design of the user interface are found by observation of audiologists performing audiological tests in their clinical environment on manual and computer based audiometers. These observations reveal procedural knowledge not found in the literature survey.

Both the pure tone and speech tests are performed correctly, but nuances in techniques indicate that a simple model of the audiometer transposed onto the computer screen will not meet user requirements.

A survey of commercial audiometer brochures and user manuals also aided in the design of the user interface. From the brochures it is clear that the user interface for computer based audiometers was sadly lacking. Designers have not used the computer to aid the audiologist, but merely to replace the existing non - computer based audiometers.

6.2 **SOFTWARE EVALUATION**

Preliminary software trials in the user environment have revealed several points that need to be added into subsequent versions of the software.

These include:

i) a prompt bar on the bottom row of the screen that indicates the current active keys;

ii) the feed back monitor volume must be reduced during speech tests so that acoustic feedback does not occur;

iii) a positive audio and visual indication of patient response. The response triggering is sometimes lost due to the PC performing other tasks (eg. eliciting a tone) while the response button is pressed.

6.3 **HARDWARE EVALUATION**

Although signal specifications meet IEC645 approval, the method of generating the pure tone and narrow-band noise signals can be improved. Digital techniques employing a Digtal Signal Processing (DSP) integrated circuit can be employed to generate the signals, monitor the frequencies and correct any distortions that may appear. The digital techniques will also increase system stability over long periods of time, and allow software configurable data to modify noise, pure tone and speech characteristics.

A third output channel is neccesary for an ear insert device. The device replaces the headphones in patients who find them uncomfortable. This channel must be selectable from software and have the same characteristics as the headphone channels.

6.4 COMPUTER MEMORY REQUIREMENTS

Of the available 640 Kilo bytes of memory space on the PC, the Disk Operating System (DOS) requires about 80 Kilo bytes, the audiometric program requires just over 200 Kilo bytes and the graphic storage area about 300 Kilo bytes. The total amount of memory used by the system is almost equal to the total memory available. This implies that further expansion is limited.

Another factor that limits the expansion of the present system is the restrictions placed on the data segment of programs written for the Intel 80 series of microprocessors (the PC is based on the 80X86 microprocessor). The data segment is limited to only 64 Kilo bytes.

With the audiometric program, each channel, frequency setting and intensity setting has a table of constants that set the digitally controlled amplifiers.

These constants together with other constants that describe the screen sizes, colours and position of menus fills up a large portion of the data segment, even before program constants are taken into account. The new version of MicroSoft DOS (version 5.5) promise to alleviate this problem by moving the DOS into memory areas above 640 Kilo bytes and allowing the data segment to grow outside the 64 Kilo byte boundary.

6.5 ACOUSTIC INTERFERENCE

The IBM PC is not an ideal environment for the development of low noise acoustic or measurement electronics. The power supply is not designed for this type of application, and the various internal motors produce unwanted electrical and electromagnetic noise. As a result, the test signals produced by the audiometric prototype card contain interference (continuous) and artifact (transient) noise.

Most of these electrical noise problems have been overcome by power supply regulation and supply voltage smoothing on the prototype card, and a physical separation and decoupling of analogue and digital circuitry.

For future development it is advisable to develop the audiometric analogue circuitry in an external chassis with its own power supply, and only the digital control signals from the PC controlling the audiometric card. See appendix IV - Acoustic interference and Calibration, for a complete discussion of the acoustic noise problem.

6.6 CALIBRATION

Calibration of the system involved placing the Unit Under Test (UUT) in a sound proof booth and calibrating each frequency and amplitude to SABS 0154 specifications. See appendix IV - Acoustic interference and Calibration, for a complete discussion of the Calibration Procedures.

A problem that arises with calibration at very low intensities -

such as testing at attenuated output levels - is that the ambient noise is greater than the signal to be tested. It is therefore impossible to calibrate at these intensities acoustically, and either an extrapolation method must be used, or the signal must be measured electrically.

Because of the almost perfect linearity of the digital to analogue converters that control the amplifiers, it was possible to predict the digital settings for these very low intensities.

The UUT was able to meet and better SABS 0154 and IEC645 specifications (clinical diagnostic type 1 audiometers) in both frequency distortion and intensity accuracy over all channels.

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CHAPTER SEVEN - CONCLUSIONS AND RECOMMENDATIONS

7.1 CONCLUSIONS

The prototyping card and software allow the audiologist to perform pure tone and speech audiometry. These tests may be conducted in the presence of narrow-band or speech noise, respectively. Speech test word intensities are controlled by a voice-controlled gain-adjusting device, eliminating the need for an SPL meter.

The system attempts to imitate the audiologist's view of the audiometric environment by:

- simulating the results sheet in colour on the screen, with a dynamic audiogram and speech discrimination-gram during the audiometric evaluation,
- allowing the audiologist to choose procedures from graphic menus, where menu items are standard audiometric symbols,
- reducing the amount of key strokes necessary to perform a test, search for a file or save a file,
- providing a pop-up text editor that will allow the audiologist to add notes or comments to a specific test at any stage of the audiometric evaluation.

It is felt that this audiometric environment will provide the basis for the development of a fully computerised audiometric environment.

7.2 **RECOMENDATIONS**

Although the screen formats have been designed in cooperation with practicing audiologists, the user-interface has not been tested for acceptability in a working environment.

7.2.1 Clinical Trials

It is therefore recommended that the system be implemented on a trial basis in an audiometric department and that pure tone and speech tests be performed on patients.

The system will only accept pure tone and speech results and not special tests results. Pure tone and speech tests should therefore be used to evaluate the system, and NOT the patient.

7.2.2 Future Work

The circuit implemented on the prototyping card needs to be implemented on a printed circuit board.

The following need to be studied and incorporated into the computerised audiometric environment:

- Audiometric tests not included yet - the special tests, SISI, WIPI, SSI and Bekesy;
- A system for evaluating acoustic immittance (consisting of pressure measurement and applicable software);
- A hearing aid evaluation and selection program;

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APPENDIX I - INTRODUCTION TO AUDIOMETRY

I.1 THE ACOUSTIC PATHWAY

The acoustic pathway is divided into three regions: the outer ear, the middle ear and the inner ear.

The outer ear is the visible part of the pathway and consists of the pinna or earlobe and the external ear canal. Functionally the pinna acts like a funnel by channeling sound waves into the external ear canal. The external ear canal is terminated about 2 cm into the temporal bone of the cranium by the tympanic membrane.

The middle ear is a small air-filled bony cavity that extends from the tympanic membrane on its anterior wall to the membrane over the oval window on its medial wall.

Suspended by tiny ligaments within this cavity are a series of three small bones or ossicles. Sound waves impinge on the tympanic membrane and cause it to vibrate. The vibration is transferred to the base of the first ossicle (malleus). The malleus is attached to the second ossicle (incus) which in turn is attached to the third ossicle (stapes). The footplate of the stapes is attached to the oval window. This ossicular chain acts as an impedance transformer between air borne energy of sound waves in the external ear canal, to fluid borne energy within the inner ear.

Air pressure inside the inner ear is equalized by the Eustachian tube. The Eustachian tube passes to the posterior wall of the nasopharynx. The tube in the region of the nasopharynx usually opens on swallowing through the action of the tensor palatini muscle.

A second membrane-covered window, the round window exists on the inferior anterior part of the middle ear. The round window permits the displacement of inner ear fluid when the footplate of the stapes moves against the oval window.

The inner ear, or labyrinth, contains sections important to balance and to hearing. The part of the inner ear concerned with hearing is the membranous cochlea. This spiral-shaped space in the temporal bone converts fluid borne vibrations into electrical energy. This conversion takes place in the following way: within the scala media (a fluid filled cavity within the membranous cochlea) is a structure that traverses the entire length of the membranous cochlea. This structure (the organ of Corti) contains thousands of sensory cells that convert physical energy from the surrounding fluid into electro-chemical energy in the form of nerve impulses.

Fibres of the cochlear portion of the VIIIth cranial nerve (auditory nerve) are distributed to these sensory cells. These fibres travel from the organ of Corti with the facial nerve through the internal auditory canal into the brain stem. The cochlear fibres terminate in the cochlear nucleus in the brain stem.

I.2 TYPES OF HEARING LOSS

The auditory pathway may be divided into a conductive region (external and middle ear), a sensorineural region (inner ear) and a central region (central auditory pathways within the brain). The type of hearing loss is based on the location of the disorder causing the problem.

Conductive hearing disorders arise from problems associated with the outer or middle ear. These disorders reduce the sound conducting ability of the pathway. The inner ear remains normal, but a hearing loss is present because of the reduction of sound energy reaching the inner ear.

Sensorineural hearing disorders arise from problems associated with the cochlea or the cochlear nerve. The combined term sensorineural disorder is used rather than sensory disorder or neural disorder because the audiologic test battery cannot separate sensory and neural pathology.

Mixed hearing loss consists of a conductive hearing loss components, and a sensorineural loss component coexisting in the same ear.

The final region where a hearing loss may exist is in the neural pathways within the brain. These **Central auditory disorders** are not usually as a result of auditory pathway pathology, but may be as a result of other neural disorders. The basic audiologic test battery is usually not sensitive to central auditory disorders.

Atypical hearing perception

If a continuous tone is presented to a normal ear, the perception of loudness decreases during the first few minutes of stimulation. Because this decrease is gradual, one is not consciously aware of this effect (Langenbeck 1965). This is a temporary threshold shift. An abnormal threshold shift occurs in some ears when the perceived loudness of a tone initially well above threshold fades completely after a few seconds or minutes. This effect variously known as **auditory adaptation, auditory fatigue or tone decay** has diagnostic value.

Recruitment is considered to be the abnormal physiological response of the hair cells of the cochlea. It is an abnormal growth in perceived loudness. Therefore recruitment can be used as a diagnostic tool for differentiating the type of loss. There are various degrees of **recruitment**: partial recruitment, hyper-recruitment and decruitment (Coles 1974, Goodman 1966, Fowler 1936).

I.3 TEST PROCEDURES

I.3.1 Pure Tone Audiology.

A number of tests utilize pure tone stimuli for threshold measurements. These are obtained by **air conduction testing** and **bone conduction testing**. Different protocols have been devised to test pure tone threshold. Carhart and Jerger (1959) investigated different pure tone procedures and found that test results obtained with different methods showed no real differences in results. A procedure based on their research has been widely adopted.

Air conduction testing

The purpose of air conduction testing is to test the patient's hearing sensitivity at different frequencies by delivering the stimuli to the external ear. If a hearing loss is present, air conduction test results can specify the degree of loss but cannot indicate the type of loss, ie. conductive or sensori - neural.

A **pure tone average** (PTA) is calculated using the following frequencies: 500Hz, 1000Hz, 2000Hz. Goodman (1965) determined a scale to describe the hearing loss handicap based on the pure tone average. The **pure tone average** is also a useful correlate of the speech threshold.

As the thresholds are obtained for each frequency, they must be plotted graphically on an **audiogram** using symbols to depict the threshold intensity at specific frequencies.

Bone conduction testing

The purpose of bone conduction is to determine the patient's cochlear thresholds. The stimulus is delivered directly to the cochlea via a transducer. A bone conduction oscillator is applied over the forehead or behind either ear on the mastoid bone. The sound transmission in bone is high, resulting in the sound vibrating the skull equally at all positions. Bone conduction occurs by inertia or compression of the cochlea and middle ear structures. If there is a difference between the air conduction and bone conduction thresholds a conductive hearing loss is present. In pure conductive hearing losses bone conduction is within normal limits. This fact may be used to determine the site of the lesion.

A problem with bone conduction measurement occurs when the nontest ear is masked. The ear is occluded by headphones and the ear canal forms a resonating chamber. The vibrations resonate at an intensity higher than those produced by direct vibration. This is the **occlusion effect** and may improve bone conduction thresholds artificially. It is rarely seen in patients with conductive hearing losses because the increase in intensity created in the canal is attenuated by the conductive hearing loss.

The levels at which the bone conduction thresholds are obtained, are plotted on the audiogram with specific symbols indicating certain parameters (eg left or right ear, masking used or no masking used).

Pure Tone audiometry measures the peripheral auditory mechanism and is used primarily to determine air conduction and bone conduction thresholds of hearing. These thresholds are necessary for a diagnosis of the degree and type of hearing loss.

The **plateau method** described by Hood (1960) has been adopted by the majority of audiologists. The plateau effect is described in the following paragraphs:

Undermasking occurs when the tone continues to be heard in the masked ear despite the masking, since the tone level is below threshold of the test ear.

Plateau The tone has reached the threshold of the test ear. Raising the masking level in the masked ear does not shift the threshold of the tone.

Overmasking The masking level is so intense that it crosses to the test ear, resulting in continuous shifts in the threshold of the tone with increases in the masking noise.

The plateau indicates a level where the masking noise can be increased several times without affecting the level of the tone that is being tested. The incidence of a plateau and the intensity level at which it is obtained has to be recorded on the audiometric examination sheet.

I.3.2 Differential Pure Tone Tests.

Special applications of pure tone procedures may be used to determine site of lesions.

i) **Weber test**

The bone transducer is placed on the skull and a pure tone is used to vibrate the transducer. The patient is then asked in which ear the sound is present. It is expected that tones referred to the poorer hearing ear signify a conductive loss, and tones referred to the better ear signify a sensori - neural loss.

The Weber test is based on the Stenger principle (1956) - if a signal is presented to both ears at the same intensity, the better ear only will hear the signal.

ii) **SAL test**

Results obtained from bone conduction audiometry contain inaccuracies due to the positioning of the bone transducer, the pressure of the transducer on the skull and the calibration of the transducer (Wood, Wittich and Mahaffey 1973).

The sensorineural acuity level (SAL) test yields the same information as bone conduction audiometry but without these sources of error. The SAL test therefore measures the threshold shift of an air conducted pure tone signal produced by a bone conducted broadband noise signal.

The thresholds are frequency dependent and the SAL values are calculated as per tested frequency. Because the hearing test is performed with both ears occluded, the occlusion effect of the headphones is overcome.

The test is performed by first measuring the air conduction threshold. Then the air conduction thresholds are re-tested while a broad band noise stimulus is delivered to the bone conductor placed on the forehead.

iii) **Bekesy audiometry**

An automatic test method that allows the patient control of the test is another diagnostic site of lesion procedure. The Bekesy audiometer has a variable frequency generator that is able to sweep through the audio test spectrum (100Hz to 10kHz). The tones are either automatically pulsed or presented continuously. The audiogram traced by the Bekesy audiometer represents the patient's hearing threshold over a continuous frequency range. The patient is told to respond differentially to the presence or absence of the stimulus. This causes the Bekesy audiometer to automatically adjust the intensity of the tone in the following way: when the button is pressed, the intensity decreases, and when the button is released, the intensity increases.

Bekesy audiometry may be carried out on either ear, with or without masking. The sweep frequency may be forwards or reversed and the sweep rate may be varied.

Five types of Bekesy tracings have been identified (Jerger 1960) for sweep and fixed Bekesy audiograms.

Type I tracings are observed in patients with normal hearing and those with conductive hearing losses. Patients with inner ear lesions are also typical of Type I tracings.

Patients with inner ear disorders present with **Type II** tracings, indicating cochlear pathology associated with recruitment ie. atypical hearing perception.

Type III tracings are associated with lesions beyond the inner ear, indicating retrocochlear pathology associated with adaptation ie. demonstrate atypical hearing perception.

Type IV tracings are similar to type II except that the frequency at which the separation of continuous and pulsed tracings occurs is at 500Hz. Type IV tracings may be typical of lesions of the cochlea or may indicate retrocochlear pathology.

Type V tracings are typical of patients who are believed to be exaggerating their hearing thresholds.

Variations of the Bekesy procedure have been developed (Palva 1970; Jerger 1972; Rose 1962). These variations include:

change in sweep direction - 10kHz down to 100Hz instead of 100Hz up to 10kHz;

change in intensity direction - instead of increasing the intensity from below threshold, the test is begun from suprathreshold levels;

change in stimulus duration - patients with normal hearing or conductive losses show lower thresholds for longer tones, while patients with inner ear lesions do not show the need for higher intensities to achieve threshold (Wright 1978);

most comfortable tracking levels - Jerger (1974) found that sensitivity of the Bekesy audiogram increases when the patient is asked to track at the most comfortable intensity;

lengthened off time - used specifically for testing a patient who presents initially with a type V Bekesy.

iv) **Short Increment Sensitivity Index (SISI).**
Harford (1959) suggested that patients with disorders of the inner ear appear to be able to discriminate transients in a steady state signal. People with normal hearing or hearing disorders elsewhere, do not have this ability. The **SISI** procedure is designed to test the ability of the patient to discriminate 1dB increments superimposed on a pure tone.
The test requires that the audiometer be able to produce a steady pure tone and superimpose a 1dB intensity increment onto the signal. Masking may also be used with the **SISI** procedure.
Practice must include the test using a 5dB and 2dB increment.
Because of the high incidence of patients with recruitment showing high SISI scores, Bartels and Rupp (1969) suggested that SISI may be used as an indirect test of recruitment.
Variations on the SISI procedure include:

- presenting the tone at 20dB SL,
- presenting the tone at 75dB HL.

v) **Alternate Binaural Loudness Balance Test (ABLB).**
For the administration of this test, the patient must have normal hearing in one ear. Hearing thresholds in the poorer ear must be less than 20dB than those of the good ear. The ABLB test is the only absolute measure of recruitment.

The test is executed as follows : the pure tone signal is alternated between ears. The intensity of the signal to each ear must be controllable. The patient compares the loudness of the two signals. The audiologist then adjusts the intensity of the signal delivered to the poorer ear until the loudness is perceived to be equal in both ears.

vi) **Monaural Loudness Balance Test.**

MLB is a variation of the ABLB test. This test is applied when there is a symmetrical hearing loss, and a difference of at least 20dB between adjacent frequencies..

vii) **Tone Decay Tests.**

Several **tone decay** tests have been developed (Carhart 1957, Rosenberg 1969, Green 1963, Olsen and Noffsinger 1974).

Tone decay may be described as the inability of the ear to sustain a constant frequency. The ear fatigues and is unable to detect the frequency for sustained periods. The method of testing is straight forward. A tone of a constant intensity is presented to a patient and the time interval recorded until the patient can no longer hear the tone. If the level of the tone is increased, the patient may then hear the tone again, but it will again fade into silence. This procedure is repeated over a 30dB SL range until the tone is perceived for sixty seconds. If this does not occur, adaptation or fatigue is present.

One form of tone decay test, the **Suprathreshold Adaptation Test (STAT)** has been developed because it appeared to be more sensitive for identifying lesions in areas beyond the inner ear (Jerger 1975).

viii) **STAT.**

The STAT test is performed with pure tone presented at 110dB Sound Pressure Level (SPL) at 500Hz, 1kHz and 2kHz. The test is concluded after one minute or after the patient signals that the tone is inaudible.

STAT results are positive if complete adaptation occurs within one minute. If a discrepancy in the test is evident, the test should be repeated using a different procedure. The audiologist may use the procedure of preference for the initial tone decay test.

I.3.3 Speech Audiology.

Speech audiology measures the function of the entire auditory nervous system. Speech reception thresholds and speech discrimination scores are obtained by speech audiometry to assist diagnosis. Speech tests are also used to evaluate the performance of hearing aids.

Determination of the **threshold of hearing** is emphasized in both subfields. The clinical procedure used to determine these thresholds employs the **method of limits**. The listener is first given a signal at supra-threshold intensity. The intensity of succeeding signals is progressively lowered until the listener fails to respond. The signal intensity is then lowered by a given value and raised again in specified steps until the listener responds. By repeating this procedure two or three times the hearing threshold is determined.

Speech audiometry is used to measure various aspects of receptive speech. These aspects include :

- speech reception threshold (SRT)
- most comfortable loudness level (MCL)
- threshold of discomfort (TD)
- discrimination of speech as a function of increasing intensity levels.

Difficulties experienced in **speech audiology** include:

- The lack of understanding of the test by the patient
- The lack of knowledge of the words used in the test (applies particularly to "third world" situations)
- Speech or language problems associated with hearing problems
- Distortion found in feedback amplifiers of audiometers
- Observation of the audiologist's face during testing (the patient may be able to lip read and thus defeat the hearing test).

Speech threshold testing may be of two kinds: **Speech Detection** and **Speech Reception** thresholds.

Speech Detection Threshold (SDT) is defined as the lowest level at which the patient can **detect** the presence of speech and **identify** it as speech (Martin 1978). Here understanding is not a criterion but just the identification that the signal **is** speech.

i) **Speech Reception Threshold (SRT).**

SRT is defined as the hearing level (HL) yielding 50% correct performance for spondaic words. **SRT** values prove to be more useful than **SDT** values. SRT should correlate with the pure tone averages by ± 6dB as an index of test reliability.

The audiologist tests a patient's **SRT** level by reading a list of spondee words and finding the loudness level at which the patient can repeat half of the words. Spondee words are equally stressed double syllable words. 20dB HL above pure tone average is the starting level for the SRT procedure.

Masking must be used to eliminate the nontest ear to determine the true threshold of the test ear. This is done in the same way as for pure tone air conduction audiometry.

Since speech is a broad band signal, the masking signal must also be broad band. Band limited white noise with 12dB per octave rolloff above 1000Hz has been preferred because of its greater low frequency energy. Speech noise is the term applied to this noise (Bailey 1963).

ii) **Most Comfortable Loudness Level.**

The patient is asked to indicate the level at which the speech presented to him is at the most comfortable loudness. The **MCL** is determined by approaching the most comfortable level from above and below, and may be determined monoaurally (one ear) or binaurally (both ears) under headphones or free field.

iii) **Threshold of Discomfort.**

Under certain conditions it is of diagnostic and rehabilitative importance to determine the level at which the patient experiences discomfort due to an excessive intensity level. The threshold of discomfort (TD) may be tested with speech.

In addition there are speech test procedures utilizing competing stimuli simultaneously. The competing stimulus may be speech or noise, and the signal to noise ratio variable. The speech signal

itself may be altered by filtering. The speech stimuli may be presented **monotically, diotically** or **dichotically.**
The use of these tests further assists site of lesion diagnosis since the use of speech under the prescribed conditions indicates the presence of a central auditory processing disorder.

These tests are:

* Synthetic Sentence Index (SSI)
* Speech Performance in Noise (SPN)
* Filtered Speech
* Rapidly Alternating Speech (RASP)
* Staggered Spondaic Word (SSW)

I.4 <u>**MASKING**</u>

When a test signal is presented to a patient's poorer ear, the intensity may be at such a level that it is transmitted through the skull and actually perceived in the better ear. Therefore one must eliminate the possibility of response from the better, non-test ear. This is achieved by the use of a masking noise in the non-test ear. If two different sounds (eg. pure tone and speech noise) are introduced into the ear simultaneously, the threshold for one or both may be higher than when heard separately.

The masking may be carried out with a number of different sound stimuli. The most common is band limited white noise with the centre frequency of the noise at the frequency being tested.

Several masking methods are used to eliminate the non-test ear from the test (Martin and Forbis 1978). Equations for determining the minimum and maximum effective masking levels developed by Linden, Nilsson and Anderson (1964) are used to limit the masking stimulus delivered to the patient.

The minimum masking level required to obtain a test result is known as the **effective masking level.** The effective masking level is the intensity level at which the noise in the good (non-test) ear masks the test signal delivered to the test ear. The possibility of the patient responding to the test signal in the non-test ear by bone conduction is thus eliminated.

The band width of the white noise is important in ensuring that no additional noise is added to the masking noise. In other words, the effect of having too wide a spectrum of masking noise is to increase the level of the noise in the non-test ear, without effectively masking out the test signal. The masking noise should therefore be centered around the test frequency, and the band width be such that narrowing the bandwidth produces under-masking effects, and increasing the bandwidth produces over-masking effects (Durlach, 1972, Price, 1976 and Sanders 1972).

I.5 SAMPLE AUDIOGRAM RESULT SHEETS

i) Pure tone Audiometry

Results obtained from pure tone audiometry are plotted on a Hearing Level Threshold versus Frequency graph for left and right ears. Masking levels (if any) are noted and the level at which the masking plateau occurred is indicated. The pure tone average is calculated for each ear.

If any of the special tests (Weber, Stat, SISI, Tone decay, Bekesy) are conducted, the results are tabulated and percentages (SISI) calculated.

ii) **Speech Audiometry**

Results obtained from speech audiometry may include **SRT, MCL, TD** and **DR**.

These results are tabulated together with the masking (if any) levels, type of masking and which ear was the non-test ear. Speech discrimination results are tabulated together with the test list number.

iii) **Patient information**

Patient information includes :

* name, address, date of birth
* refering doctor
* examiner
* examination number.

This information is acquired from the transferral forms or directly from the patient.

In addition to this information, the audiologist may wish to make recommendations to the referring doctor and give additional information not indicated in the test results.

If future testing is required, a future date must be indicated for both the patient's and the audiologist's information.

The audiologist must keep a record of the information, the hospital must have a record of the information and the referring party must be presented with the information.

SAMPLE RESULTS

H F VERWOERD HOSPITAL

DEPARTMENT : SPEECH AND HEARING

TEL : : (012) 21-3211

AUDIOLOGICAL ASSESSMENT

PATIENT PARTICULARS

FOLDER : ____________________

SURNAME : ____________________ **INITIALS :** __________ **TITLE :** __________

DATE OF BIRTH ____________________ **EXAMINER :** ____________________

ADDRESS : ____________________ **TEST DATE :** 20 Aug 1991

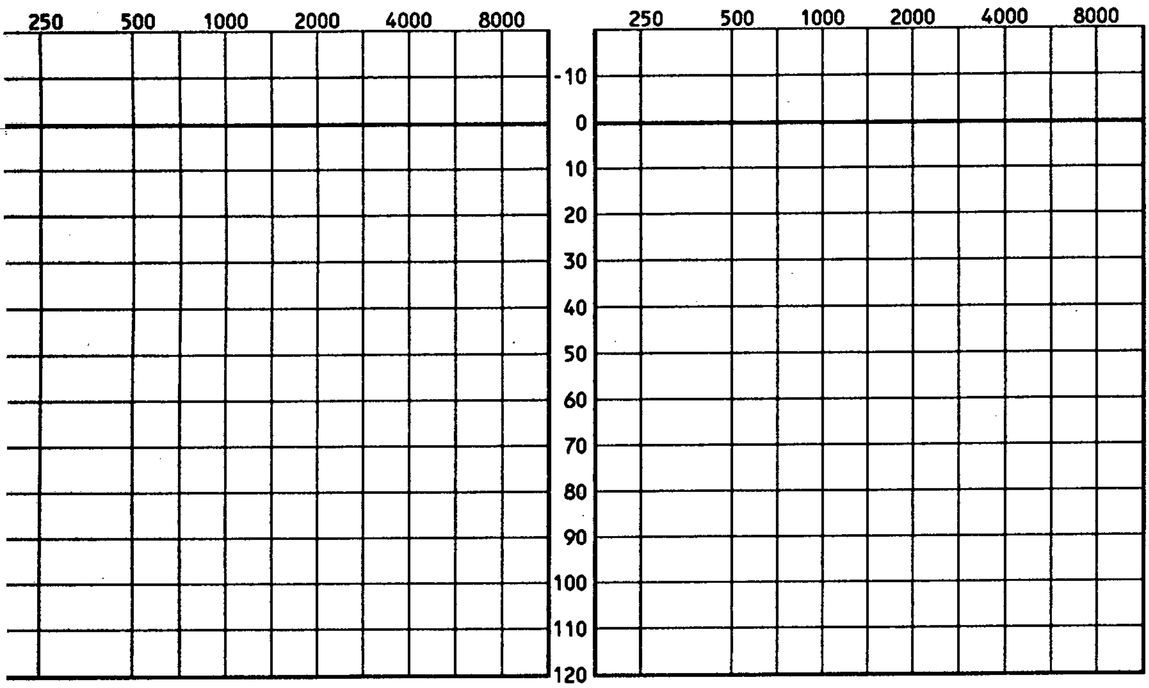
____________________ **TEST No. :** ____________________

____________________ **REFERRAL :** ____________________

TEL : ()

PURE TONE AUDIOGRAM

RIGHT EAR **LEFT EAR**



KEY	RIGHT	LEFT
Air Cond	O	X
Bone Cond	<	>
A/C Masked	△	□
B/C Masked	[	]
Free Field	S	S

P.T. AVG		

TEST RELIABILITY

							Air Cond					
							Bone Cond					
							A/B Gap					

MASKING LEVEL IN NON-TEST EAR

PURE TONE TEST COMMENTS

H F VERWOERD HOSPITAL

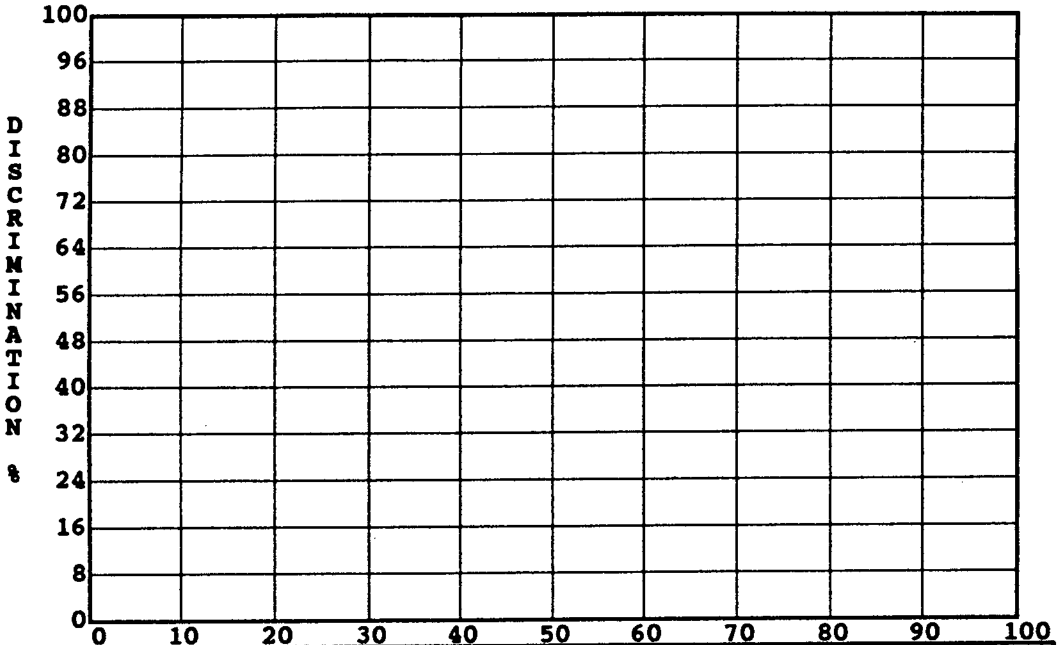
DEPARTMENT : SPEECH AND HEARING

TEL : : (012) 21-3211

AUDIOLOGICAL ASSESSMENT

SPEECH TEST RESULTS

	MCL	TD	SRT	MASK	PB-MAX dB	%	DR
R							
L							



R																					
L																					
EFFECTIVE MASKING IN NON-TEST EAR																					

SPEECH TESTS COMMENTS

I.6 AUDIOMETRIC SIGNAL AND SYMBOL STANDARDS

Table of IEC 645 Specification for Pure Tone frequency tolerances taken fron SABS 0145 (1983).

1	2	3	4
Frekwensie-stelling*, Hz	Toelaatbare frekwensie-bestek, Hz	Nominale klank-drukpeil by 'n stelling van 70 dB, dB	Toelaatbare klank-drukpeilbestek by 'n stelling van 70 dB, dB
125	121- 129	115,0	112,0-118,0
160	155- 165	108,5	105,5-111,5
200	194- 206	102,5	99,5-105,5
250	242- 258	97,0	94,0-100,0
315	305- 325	92,0	89,0- 95,0
400	388- 412	87,0	84,0- 90,0
500	485- 515	83,5	80,5- 86,5
630	611- 649	80,5	77,5- 83,5
750	727- 773	79,0	76,0- 82,0
800	776- 824	78,5	75,5- 81,5
1 000	970-1 030	77,5	74,5- 80,5
1 250	1 212-1 288	77,5	74,5- 80,5
1 500	1 455-1 545	77,5	74,5- 80,5
1 600	1 552-1 648	78,0	75,0- 81,0
2 000	1 940-2 060	79,0	76,0- 82,0
2 500	2 425-2 575	80,5	77,5- 83,5
3 000	2 910-3 090	81,5	78,5- 84,5
3 150	3 055-3 245	81,5	78,5- 84,5
4 000	3 880-4 120	82,0	79,0- 85,0
5 000	4 850-5 150	81,0	78,0- 84,0
6 000	5 820-6 180	86,0	81,0- 91,0
6 300	6 111-6 489	+91,0	+86,0- 96,0
8 000	7 760-8 240	85,5	80,5- 90,5

TUV REIHNLAND Specifications for Air and Bone Conduction Amplitudes in Pure Tone Audiometry.

	Frequenz Hz	Klasse 1 Hörpegel dB		Klasse 2 Hörpegel dB		Klasse 3 Hörpegel dB		Klasse 4 Hörpegel dB	Klasse 5 1) Hörpegel dB
		Luft-leitung	Knochen-leitung	Luft-leitung	Knochen-leitung	Luft-leitung	Knochen-leitung	nur Luft-leitung	nur Luft-leitung
Obere Grenze des Hörpegel-bereiches	125	70		70					
	250	90	45	90	40	90	40		
	500	120	60	110	60	100	60	90	
	750 2)	120	60	110	60				
	1000	120	70	110	70	100	70	90	
	1500	120	70	110	70	100	70		
	2000	120	70	110	70	100	70	90	
	3000	120	70	110	70	100	70	90	
	4000	120	60	110	60	100	60	90	
	6000	110	50	100	50	90	50	90	
	8000	100	50	90	50	80	50		
ıtere Grenze	alle Fre-quenzen	−10	−10	−10	−10	−10	−10	−10	

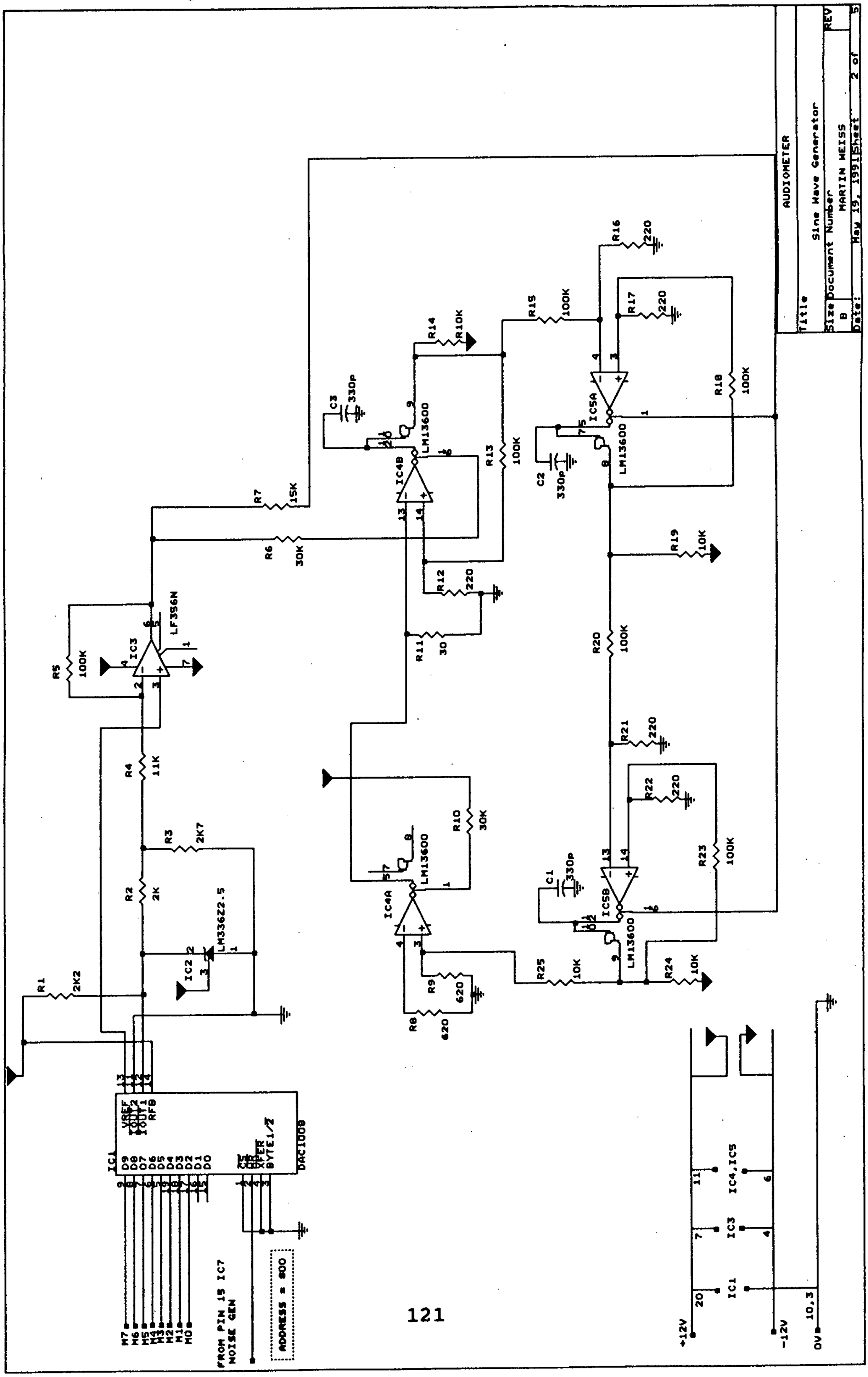
1) Keine Mindestanforderungen festgelegt.
2) Äquivalente Bezugsschwellenschalldruckpegel für 750 Hz sind in ISO 389 DAD 2 „Acoustics – Standard reference zero for the calibration of pure-tone air conduction audiometers, addendum 2" festgelegt.

ASHA Pure Tone Audiometry Symbol Standardisation (1969)

KEY	RIGHT	LEFT
Air Cond	O	X
Bone Cond	<	>
A/C Masked	△	□
B/C Masked	[	]
Free Field	S	S

APPENDIX II - HARDWARE DESIGN - CIRCUIT AND COMPONENT DETAILS

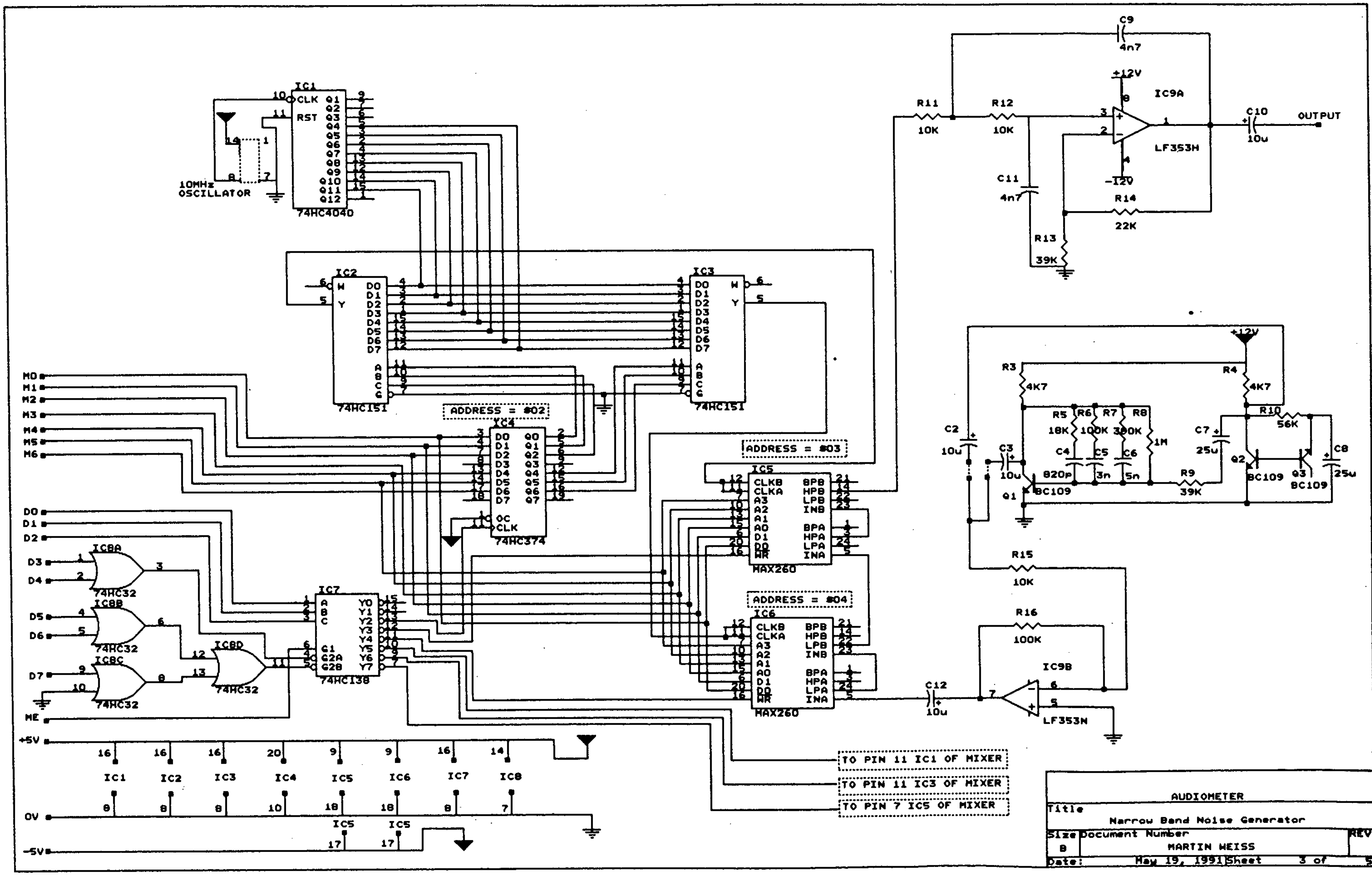
II.1 Circuit Diagram - Pure Tone Generation



Component List - Pure Tone Generation

Item	Quantity	Reference	Part
1	1	R1	2K2
2	1	R2	2K
3	1	R3	2K7
4	1	R4	11K
5	6	R5,R13,R15,R18,R20,R23	100K
6	1	IC3	LF356N
7	2	R6,R10	30K
8	1	R7	15K
9	2	R8,R9	620
10	1	R11	30
11	5	R12,R16,R17,R21,R22	220
12	1	R14	R10K
13	3	R19,R24,R25	10K
14	3	C1,C2,C3	330p
15	1	IC1	DAC1008
16	2	IC4,IC5	LM13600
17	1	IC2	LM336Z2.5

II.2 Circuit Diagram - Masking Noise Generation

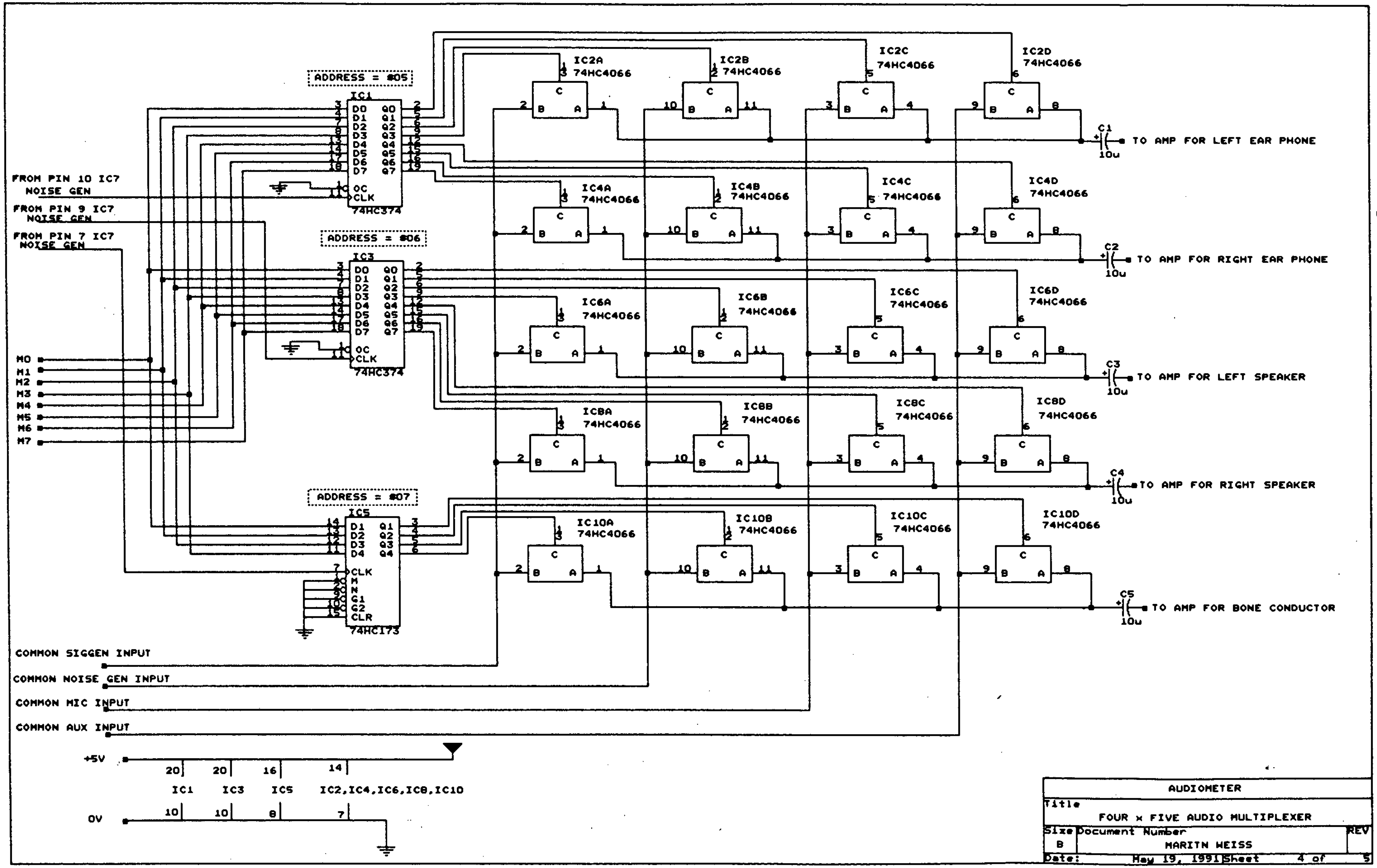


Component List - Masking Noise Generation

Item	Quantity	Reference	Part
1	2	IC5,IC6	MAX260
2	2	IC5,IC4	74HC374
3	3	Q3,Q1,Q2	BC109
4	2	R3,R4	4K7
5	1	R5	18K
6	2	R6,R16	100K
7	1	R7	390K
8	1	C4	820p
9	1	C5	3n
10	1	C6	5n
11	1	R8	1M
12	2	R9,R13	39K
13	2	C7,C8	25u
14	1	R10	56K
15	1	IC1	74HC4040
16	2	IC2,IC3	74HC151
17	1	IC7	74HC138
18	1	IC8	74HC32
19	4	C2,C3,C10,C12	10u
20	2	C9,C11	4n7
21	3	R11,R12,R15	10K
22	1	R14	22K
23	1	IC9	LF353H
24	1	IC9	LF353N

II.3 Signal Selection And Amplification

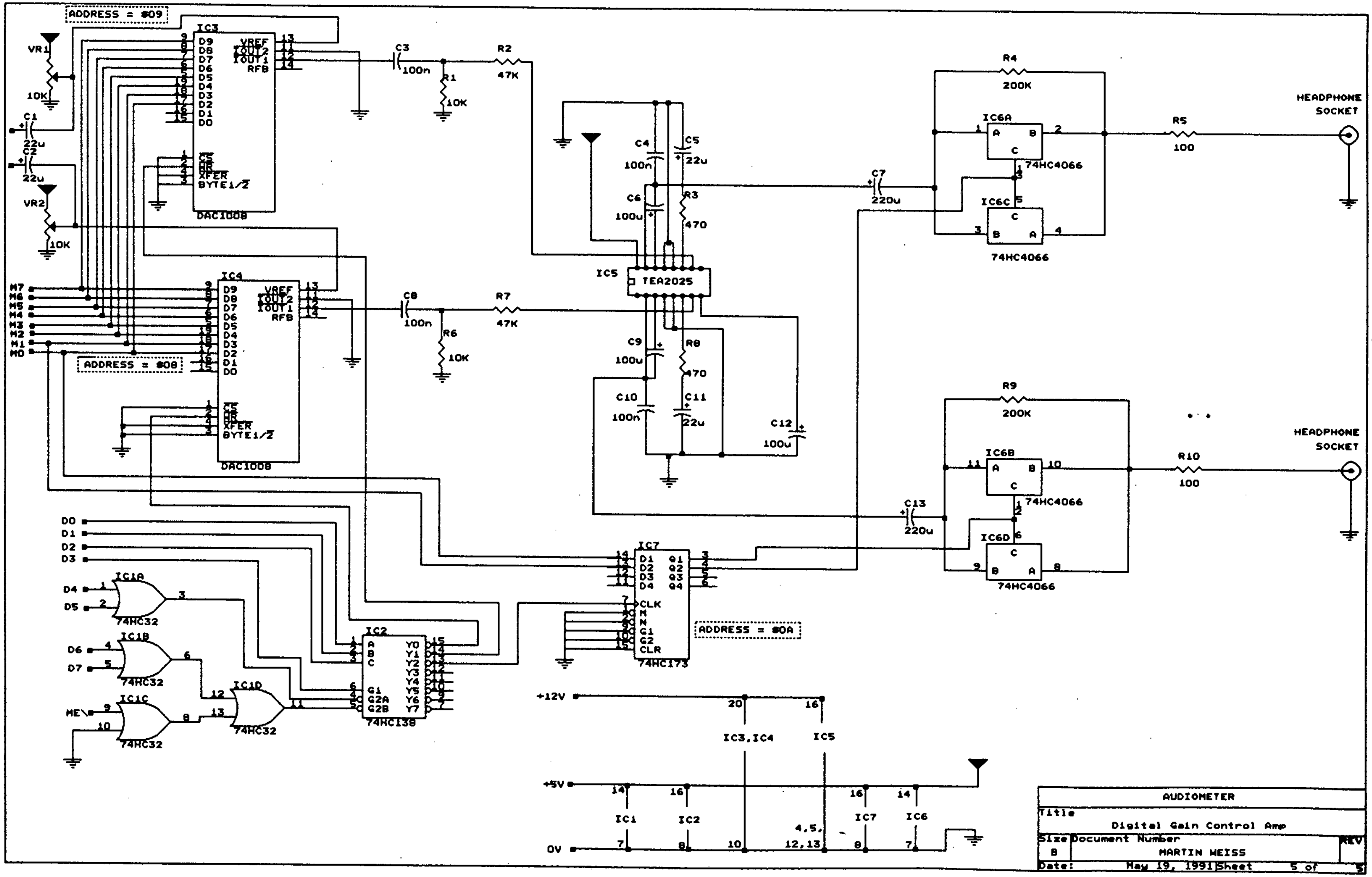
Circuit Diagram - Four by Five Multiplexer



Component List - Four by Five Multiplexer

Item	Quantity	Reference	Part
1	5	IC2,IC4,IC6,IC8,IC10	74HC4066
2	2	IC1,IC3	74HC374
3	1	IC5	74HC173
4	5	C2,C1,C3,C4,C5	10u

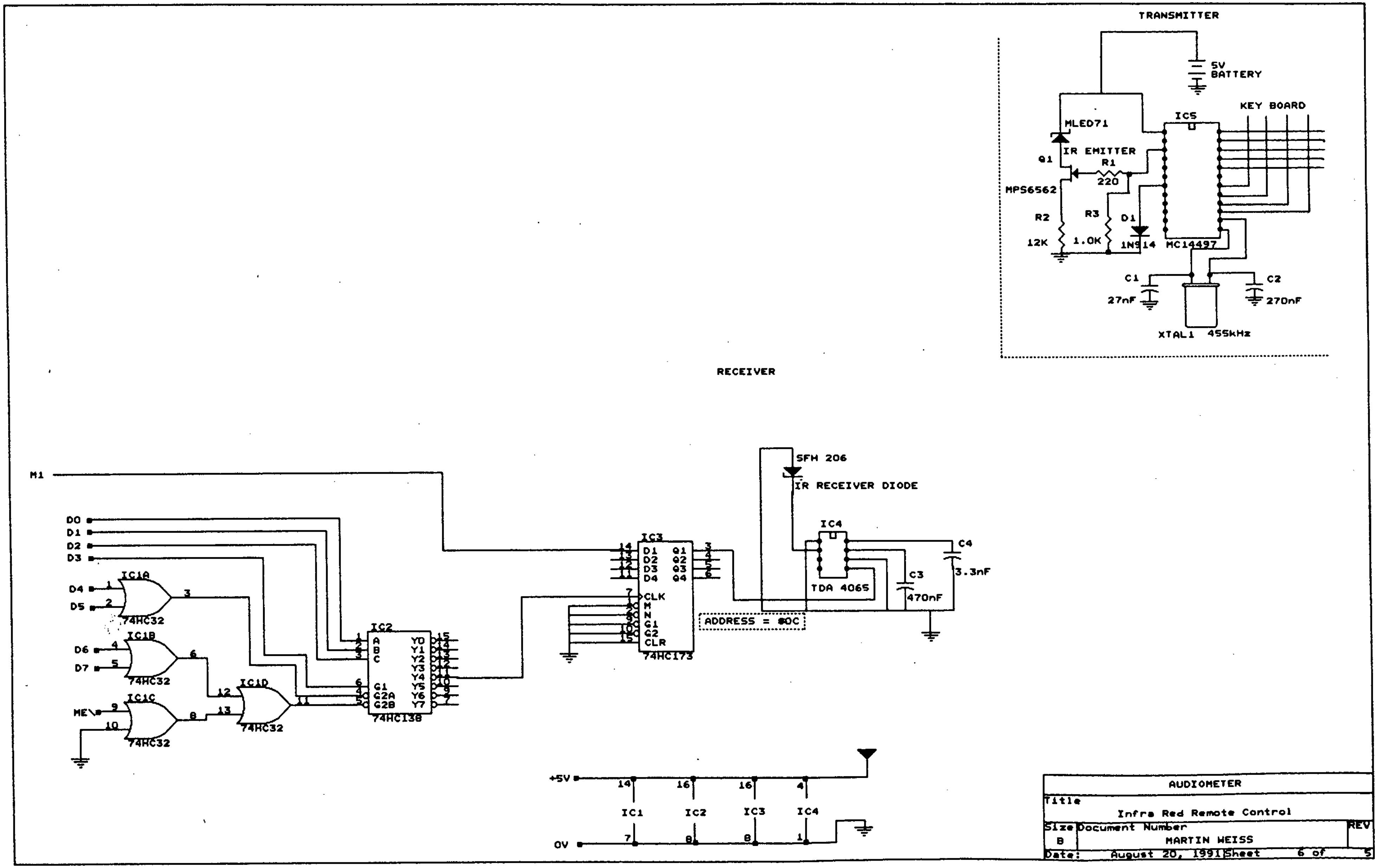
Circuit Diagram - Digital Gain Control Amplifier



Component List - Digital Gain Control Amplifier

Item	Quantity	Reference	Part
1	1	IC1	74HC32
2	1	IC2	74HC138
3	1	IC6	74HC4066
4	1	IC5	TEA2025
5	4	C3,C4,C8,C10	100n
6	4	R1,VR1,VR2,R6	10K
7	2	R2,R7	47K
8	4	C5,C1,C2,C11	22u
9	3	C6,C9,C12	100u
10	2	R3,R8	470
11	2	C7,C13	220u
12	2	R4,R9	200K
13	2	R5,R10	100
14	1	HEADPHONE SOCKET	4mm STEREO
15	1	IC7	74HC173
16	2	IC3,IC4	DAC1008

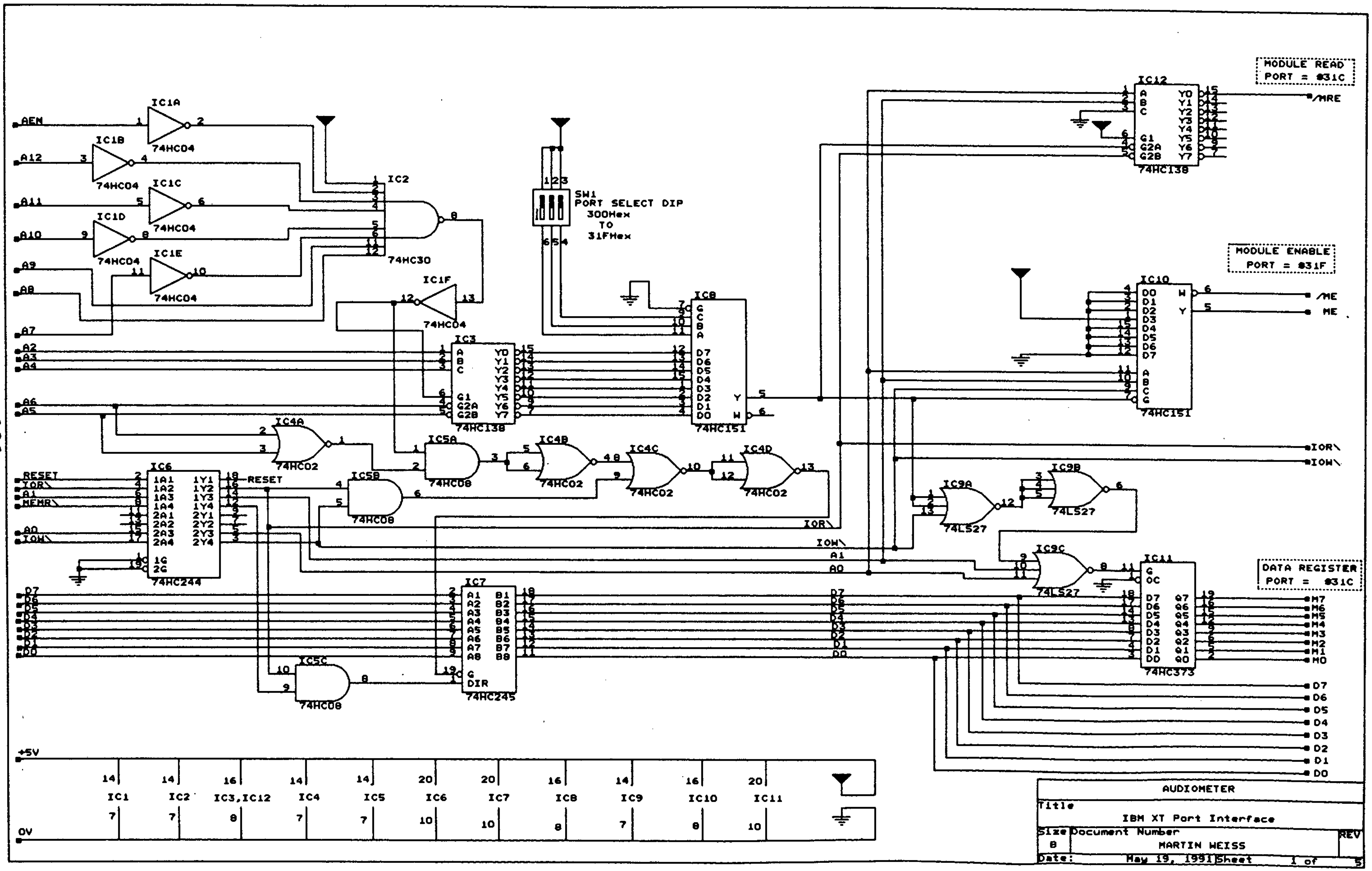
II.4 Circuit Diagram - Remote Control



Component LIst - Remote Control

Item	Quantity	Reference	Part
1	1	IC1	74HC32
2	1	IC2	74HC138
3	1	IC3	74HC173
4	1	TDA 4065	IC4
5	1	IC5	MC14497
6	1	D1	1N914
7	1	SFH 206	IR RECEIVER DIODE
8	1	MLED71	IR EMITTER
9	1	Q1	MPS6562
10	1	R1	220
11	1	R2	12K
12	1	R3	1.0K
13	1	C1	27nF
14	1	C2	270nF
15	1	XTAL1	455kHz
16	1	5V	BATTERY
17	1	C3	470nF
18	1	C4	3.3nF

II.5 Circuit Diagram - Computer Interface



Component List - Computer Interface

Item	Quantity	Reference	Part
1	1	IC1	74HC04
2	1	IC2	74HC30
3	2	IC3,IC12	74HC138
4	1	IC4	74HC02
5	1	IC5	74HC08
6	1	IC6	74HC244
7	1	IC7	74HC245
8	2	IC8,IC10	74HC151
9	1	SW1	PORT SELECT DIP
10	1	IC9	74LS27
11	1	IC11	74HC373

APPENDIX III - SOFTWARE DESIGN - PROCEDURE AND FUNCTION DETAILS

III.1 Menu Drivers and Procedures

Text Menu Routines

```
function  get_control_key : integer;

procedure set_menu_text(str : string; entry : integer;
                   var textrecord : textmenutype);

procedure num_text_entries(number : integer; var textrecord :
                           textmenutype);

procedure set_menu_colors(writingcolor,backcolor,lettercolor :
                          integer;var textrecord : textmenutype);

procedure set_border_style(width,color : integer;
                          var textrecord : textmenutype);

procedure set_menu_style(font,charsize,textjustify,direction :
                          integer;var textrecord : textmenutype);

procedure Put_Text_Menu(X,Y : integer; var TextRecord : TextMenuType;
                          bitblt : integer);

procedure Remove_Text_Menu(var TextRecord : TextMenuType);

procedure Set_Hilite_Style(FillStyle,Color,DefaultPosition : integer;
                       var TextRecord : TextMenuType);

procedure Move_Hilite_To(Position : integer; var TextRecord : TextMenuT

function  Get_Hilite_Val(TextRecord : TextMenuType) : integer;

function  Get_Hilite_Str(TextRecord : TextMenuType) : string;

procedure Set_Hilite_On(var TextRecord : TextMenuType);

procedure Set_Hilite_Off(var TextRecord : TextMenuType);

function  Get_Text_Menu( var TextRecord : TextMenuType) : integer;
```

example :

The following Turbo Pascal Source code defines the parameters to draw a horizontal menu along the top of the screen.

Lines 1 to 5 input the menu choices and their respective places in the menu.

Line 7 defines the text, background, and first letter colours.

Lines 8 and 9 define the colour and thickness of the high - light bar and the border around the menu.

Line 10 defines the text font and size, the text justification and the direction of the menu.

```
1   set_menu_text('File',1,mainmenu);
2   set_menu_text('Patient',2,mainmenu);
3   set_menu_text('Hardcopy',3,mainmenu);
4   set_menu_text('Test',4,mainmenu);
5   set_menu_text('Options',5,mainmenu);
6   num_text_entries(5,mainmenu);
7   set_menu_colors(yellow,blue,lightmagenta,mainmenu);
8   set_hilite_style(solidfill,egawhite,1,mainmenu);
9   set_border_style(normwidth,egawhite,mainmenu);
10  set_menu_style(defaultfont,1,lefttext,horizdir,mainmenu);
11 put_text_menu(10,10,main_menu,horizdir,normal_put);
12  repeat
13    choice := get_text_menu(mainmenu);
14  until (choice in [..5]);
```

The procedure Put_Text_Menu(x,y,Text_Tecord) places the named menu (Text_Record) at coordinate (x,y) on the screen, keeping the integrity of the underlying graphics.

Procedure Remove_Text_Menu(Text_Record) removes the menu from the screen and replaces the underlying graphics.

Icon menu routines

```
procedure Init_Gr_Size(X1,Y1,X2,Y2 : integer; var GrRecord : GrMenuType

procedure num_gr_entries(number : integer; var grrecord : grmenutype);

procedure Get_Menu_Image(X1,Y1,X2,Y2,EntryNumber : integer;
                         var GrRecord : GrMenuType);

procedure Set_Gr_Hilite_Style(FillStyle,Color,DefaultPosition: integer;
                              var GrRecord : GrMenuType);

procedure set_gr_border_style(width,color : integer;
                              var grrecord : grmenutype);

procedure Put_Gr_Menu(X,Y : integer; var GrRecord : GrMenuType;
                      direction,bitblt : integer);

procedure Remove_Gr_Menu(var GrRecord : GrMenuType);

procedure Move_Gr_Hilite_To(Position : integer; var GrRecord : GrMenuTy

procedure Set_Gr_Hilite_On(var GrRecord : GrMenuType);

procedure Set_Gr_Hilite_Off(var GrRecord : GrMenuType);

procedure Put_Menu_Image(X1,Y1,EntryNumber : integer;
                       var GrRecord : GrMenuType; bitblt : integer);

procedure free_gr_images(var grrecord : grmenutype);

function Get_Gr_Menu(var GrRecord : GrMenuType) : integer;

function Get_Gr_Hilite_Val(GrRecord : GrMenuType) : integer;
```

The following example draws four images 20 by 20 pixels on the screen one at a time. When the image is drawn, that image is captured and stored as an icon.

Line 1 initialises the size of the icon image.
Lines 2 to 6 draws one of four images and then captures the image as a menu image or icon with its respective menu position (count).

Lines 8 and 9 define the high - light bar and border thickness and colour.

```
1    Init_Gr_Size(0,0,20,20,Gr_Menu);
2    for count := 1 to 4 do
3    begin
4      Draw_Image(count,0,0,lightblue);
5      Get_Menu_Image(0,0,20,20,count,Gr_Menu);
6    end;
7    Num_Gr_Entries(count,Gr_Menu);
8    Set_Gr_Hilite_Style(solidfill,lightblue,1,Gr_Menu);
9    Set_Gr_Border_Style(normwidth,blue,Gr_Menu);
10   put_Gr_menu(10,10,Gr_menu,horizdir,normal_put);
11   repeat
12     choice := get_Gr_menu(Gr_menu);
13   until (choice in [..5]);
```

The procedure Put_Gr_Menu(x,y,Direction,Gr_Record) places the graphic menu (Gr_Menu) at coordinate (x,y) on the screen in either a vertical column or a horizontal row (defined by Direction), saving the integrity of the underlying screen. Procedure Remove_Gr_Menu(Gr_Record) removes the graphics menu from the screen and replaces the underlying graphics.

Input Box routines

```
procedure set_box_prompt(promptstring : PromptStrType; ReturnLength,
                      direction : integer; var boxname : inputboxtype)

procedure Set_box_text_style(font,size,color : integer;
                      var BoxName : InputBoxType);

procedure Set_Return_color(returncolor,cursorcolor : integer;
                      var BoxName : InputBoxType);

procedure Set_Box_Fill_Style(FillStyle,FillColor,backcolor : integer;
                      var BoxName : InputBoxType);

procedure Set_Box_Border_Style(borderwidth,bordercolor : integer;
                      var BoxName : InputBoxType);

procedure Set_Return_Default(defaultstr : ReturnStrType;
                      var BoxName : InputBoxType);

procedure put_input_box(x,y : integer; putdefault : boolean;
                      var boxname : inputboxtype);

procedure remove_input_box(var boxname : inputboxtype);

function get_input_box(var boxname : inputboxtype) : integer;
```

function get_input_box_str(boxname : inputboxtype) : returnstrtype;

The input box routines allow data to be typed into an area of the screen and formatted in a preconfigures way.

The following example prompts the user for an initial and a surname.

```
1    set_box_prompt('TYPE IN INITIAL AND SURNAME',30,
                 vertdir,input_box);
2    Set_return_color(lightblue,white,input_box);
3    Set_box_fillstyle(solidfill,black,input_box);
4    Set_return_default('JJ EXAMPLE',input_box);
5    put_input_box(10,10,true,input_box);
6    repeat
7      value := get_input_box(input_box);
     until (value = return) or (value = escape);
8    name_string := get_input_box_str(input_box);
```

Data Base Routines

function get_num_of_prev_tests

Retrieves the total number of tests a patient has had, and returns that number to the calling procedure.

function retrieve_folder

Retrieves a specific folder based on the patient's name and date of birth.

function remove_folder

The complete record of a specified patient is removed from the data base. This includes the data and the patient's uniuqe index.

function save_folder

The data collected during the tests is saved in the data base and the appropriate index is generated and stored in an index file.

function retrieve_pred_folder

The previous test is retrieved from the data base and stored in a buffer for later viewing.

function retrieve_succ_folder

The successive test is retrieved if it exists and is stored in a buffer for later viewing.

procedure rebuild

If the data base or index file is in anyway corrupted, this routine first tries to reconstruct the data base, and then re-index the data base.

procedure cleanup

Data stored in volatile memory is dumped into the data base.

A floppy disk containing the sample program is attached to the back cover of this thesis.

Specifications for running this software are as follows:

1) AT IBM computer or 100% compatible machine,
2) EGA or VGA graphics adaptor,
3) Hewlett Packard LaserJet Series II printer or 100% compatible printer. The printer is only neccessary if sample printouts are required.

III.2 Flow Control Procedures and Functions

The following Turbo Pascal source extract defines the main control loop for program execution. The procedure names describe the functionality of the procedures or functions without having to list the entire program.

```
put_text_menu(20,4,main_menu,normal_put);
current_help_name := 'main';
selection := get_text_menu(main_menu);
repeat
  present_test := no_test;
  case selection of
       uparrow      : begin
                        sound(6000);
                        delay(10);
                        nosound;
                        selection := escape;
                      end;

       downarrow    : begin
                        case get_hilite_val(main_menu) of
```

```
                          1   : selection := file;
                          2   : selection := patient;
                          3   : selection := hardcopy;
                          4   : selection := test;
                          5   : selection := options;
                          6   : selection := view;
                         else selection := escape;
                        end;{case}
                      end;
        escape      :  begin
                        sound(4000);
                        delay(20);
                        nosound;
                        selection := escape;
                      end;


           1   : selection := file;
           2   : selection := patient;
           3   : selection := hardcopy;
           4   : selection := test;
           5   : selection := options;
           6   : selection := view;

         alt_f : begin
                   move_hilite_to(1,main_menu);
                   selection := do_alt_keys(alt_f);
                 end;
         alt_p : begin
                   move_hiliteto(2,main_menu);
                   selection := do_alt_keys(alt_p);
                 end;
         alt_h : begin
                   move_hilite_to(3,main_menu);
                   selection := do_alt_keys(alt_h);
                 end;
         alt_t : begin
                   move_hilite_to(4,main_menu);
                   selection := do_alt_keys(alt_t);
                 end;
         alt_o : begin
                   move_hilite_to(5,main_menu);
                   selection := do_alt_keys(alt_o);
                 end;
         alt_c : begin
                   selection := do_alt_keys(alt_c);
                 end;
         alt_v : begin
                   selection := do_alt_keys(alt_v);
                 end;

     else selection := escape;
   end;{case}
until (Quit_string = 'Y');
clean_up;
remove_text_menu(main_menu);
closegraph;
```

APPENDIX IV - ACOUSTIC INTERFERENCE AND CALIBRATION

IV.1 Acoustic Interference

During hearing threshold measurement, the test signal elicited to the patient's ear must contain as little acoustic interference as possible. If interference is present, the hearing threshold of the patient is raised by at least the level of the interference.

It is therefore important that the level of the interference be so low as to be able to test patients with normal hearing thresholds. (The normal hearing threshold specified by ASHA, 1978, is 0 dB SPL).

Two types of acoustic interference may be present:

- **ambient acoustic interference** from surrounding equipment, ventilation and motor vehicle vibrations;
- **electrical interference** in the test signal from the electrical system of the PC or the power supply.

IV.1.1 Ambient Acoustic Interference

Ambient acoustic interference is reduced by conducting the tests in a sound-proof room. The primary function of the sound proof-room is to isolate it acoustically from the building in which it is housed.

Further acoustic isolation is achieved in the use of earphone enclosures. The enclosures contain the audiometer earphone and cushion and help to achieve a tight seal against the head.

IV.1.2 **Electrical Interference**

The origin of electrical interference in the PC environment may be due to the PC's switched mode power supply, the cooling fan in the rear of the PC and the motors used in the disk drives (Eggebrecht, 1987).

High frequency interference (considered to be above 25 KHz and therefore above the audible range) is filtered out by the natural response of the earphones.

Low frequency interference (below 25 KHz), caused primarily by the motors inside the PC, is reduced in the following ways:

- analog and digital ground references are isolated from each other, and only one central analog ground point on the prototyping card is used to prevent ground loops;
- the power supply for the analog circuitry is smoothed and filtered on the prototyping card;
- a ground plane on the wire-side of the prototyping card reduces interference;
- the disk drives are not accessed while a test is being performed;
- smoothing capacitors are placed across the power supply pins on all integrated circuits on the board;
- shielded cabling is used for connecting the PC prototyping card to the headphones.

IV.2 **Hardware Calibration**

The circuitry that require calibration are the sine-wave and white noise generators, the headphones and the bone-conduction transducer.

This circuitry is digitally controlled and calibration involves the generation of a look-up table for each of the circuits.

These look-up tables are stored on disk and in memory and referenced whenever a signal is elicited.

The look-up tables include:

- a frequency table for the sine-wave generator;
- band-width and centre frequency tables for the white noise generator,
- an amplification table for the headphones (left and right ears) and bone-conduction transducer.

The calibration equipment includes:

- a Bruel & Kjaer Sound Pressure Level Meter (SPL meter),
- a Bruel & Kjaer Artificial Mastoid,
- a Bruel & Kjaer Artificial Ear,
- a Fluke 87 Digital Multitester,
- a Radioear B-70-A Bone Conduction Transducer,
- and a Maico TDH 39 insert.

IV.2.1 **Sine-Wave Frequency Calibration**

The frequency of the sine-wave generator is measured with the Fluke multitester in 'frequency mode'. A look-up table is generated by first selecting the lowest digital value (0) and measuring the corresponding frequency. The digital value is then increased and the frequency noted. The procedure is repeated until the test frequency range has been covered.

IV.2.2 **Narrow-Band White-Noise Calibration**

The program PZ, supplied by Maxim Integrated Products with the programmable integrated circuit filters (MAX 260), generates code that determines the bandwidth and centre frequency of the filters. Codes generated for the nine narrow-band filters and the broad-band speech noise filter are stored in a look-up table.

IV.2.3 **Insert and Bone Transducer Calibration**

The calibration procedure for the insert is as follows:

The artificial ear is coupled to the SPL meter (refer to figure IV.1). The insert is attached to the artificial ear and a signal of a certain frequency is generated by the circuitry. The intensity of the signal is increased from the lowest digital value (0) to the highest digital value (1023). The SPL meter is used to determine the corresponding SPL of the digital value. These values are tabulated and stored.

This procedure is used to determine the look-up table for the bone-conduction transducer as well, except that the artificial ear and insert are replaced with an artificial mastoid and bone transducer (refer to figure IV.2).

Figure IV.1 Photograph of typical calibration setup for artificial ear

Figure IV.2 Photograph of typical calibration setup for artificial mastoid