Computer Controlled Audiometry

by

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To my family.

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Contents

Preface

The design and construction of a computer controlled audiometer at the University of Glasgow started in 1985 and the work is still continuing. This thesis describes work in the period October 1988 to April 1991.

The existing pure-tone audiometer known as the ASRA 2000 Series (Automatic Self-Recording Audiometer) is commercially manufactured and marketed by Mercury Electronics (Scotland) Ltd., Newton Mearns. The author made a substantial contribution to the software distributed with this version of the audiometer. The software was written in the 'C' programming language using a Borland TURBO C compiler.

Chapter 1 gives an introduction to the subject of audiology.

Chapter 2 covers the development of the standard pure-tone audiometer which is now manufactured. The hardware had been designed and built before the work of this thesis began. The work of the author involved some major modifications, additions and debugging of the existing 'C' code written by Dr. A.M. MacLeod.

Chapter 3 describes the use, by the author, of a random test facility in a clinical environment. Several forms of analysis were undertaken to determine whether the test could be used to distinguish truthful subjects from those exaggerating their thresholds for the purposes of obtaining industrial hearing loss benefit. The use of discriminant analysis was suggested by Dr. A. Bowman from the statistics department of the University of Glasgow, and this analysis was done using the MINITAB statistical software package for the PC.

Chapter 4 describes the implementation by the author of several supra-threshold tests.

Chapter 5 details the speech test again implemented by the author.

Chapter 6 develops the work of chapter 5 into a test which will give some measure of speech processing ability. This work was undertaken entirely by the author and followed a brief discussion with Dr. A. Corcoran, Audiological Scientist, Bournemouth. The MINITAB package was again used to collate frequency information from individual phonemes.

Chapter 7 explains the construction by the author of a multi-frequency tympanometer; some introductory thoughts on the probe design were supplied by Dr. M. Lutman, Institute of Hearing Research, Nottingham.

Finally chapter 8 gives some brief conclusions of the work.

(Note: In Chapter 7 help was received in the designing and debugging of the hardware from Mr Alex Blackley and Dr A. M. MacLeod.)

Summary

Pure Tone audiometry is performed routinely in hospital hearing clinics throughout the world. The test machines in general use are manually operated and the tester controls the frequency and intensity of the stimulus. Although the advent of computers has brought a certain amount of computer control into the field, the computers have been used for storing data which has been accumulated using manual machines, or for performing completely automated screening tests which allow the operator no intervention.

The work of this thesis tries to bridge this gap by using an automated audiometer which can be interrupted and overruled by the operator at any stage but which still affords all the benefits of the computer in data storage and management. It also investigates and uses aspects of computer control which allow tests to be carried out which would otherwise be too complicated or time-consuming.

Chapter 1 offers an introduction to audiology, both from a medical and a scientific point of view and describes some of the work which has been done previously in this field.

Chapter 2 explains the implementation under computer control of the conventional pure-tone threshold tests (recommended by the British Society of Audiology). This includes an entirely automated test which can be paused, stopped, restarted or overruled by the operator, if and when required. The system also incorporates the ability to perform a completely manually controlled test.

Chapter 3 describes a 'random test' which has been designed and implemented, and which is unique to this machine. It takes a block of points around the patient's thresholds and randomises the presentation of tones in ear, in frequency and in intensity. This random test is

then investigated to determine whether it can be used to distinguish subjects who are suspected of exaggerating their hearing loss in order to qualify for industrial hearing loss benefit, from the subjects who are being truthful about their hearing thresholds.

A number of normal hearing subjects were asked to simulate a hearing loss and then asked to respond truthfully. Several quantities were measured and the two sets of results are compared. Using various types of analysis, it was found possible to distinguish the truthful set of results from the non-organic hearing loss set.

Chapter 4 describes the implementation of computer control to several supra-threshold tests, namely the loudness balance tests, and the tone decay tests. There are many benefits of using a computer to control these tests.

Chapter 5 and chapter 6 deal with a different form of audiometry which uses speech as the stimulus instead of pure-tones. Chapter 5 describes the implementation of a conventional speech test. This has the advantage that the words spoken can be displayed on the PC screen by using a coding system for the second channel of a stereo audio tape. Chapter 6 is a modification of this test which, using a mechanism of scoring the phonemes individually, allows a chart of phoneme versus hearing level to be displayed, giving speech processing information.

Chapter 7 describes the design and fabrication of a multi-frequency tympanometer.

A brief conclusion summarising the work achieved is contained in chapter 8. It describes the benefits afforded by computer control and in particular deals with those tests which would be virtually impossible to undertake manually.

Appendix A gives a brief outline of clinical decision analysis, one of the forms of analysis used in chapter 3. Appendix B gives a complete list of the Boothroyd wordlists along with their phonetic components. Appendix C gives a test program written in 'C' which fits a cubic spline curve to the points on a speech audiogram. Appendix D gives the results of a frequency analysis done of each individual phoneme of the words in the Boothroyd wordlists. Appendix E shows two MC68000 assembly code programs used to control the tympanometer.

This thesis shows several uses of computers in audiology which until now have remained untapped.

Table of abbreviations

AC	-	Air Conduction (in pure-tone testing).
BC	-	Bone Conduction (in pure-tone testing).
ASRA	-	Automatic Self Recording Audiometer.
BSA	-	British Society of Audiology.
SPL	-	Sound Pressure Level.
HL	-	Hearing Level.
DAC	-	Digital to Analogue Converter.
ADC	-	Analogue to Digital Converter.
NOHL	-	Non-Organic Hearing Loss.
dev ; d	-	deviation (measured by random test).
diff ; df	-	difference (measured by random test).
resp ; r	-	response time (measured by random test).
stime ; s	-	standard deviation of response time (measured by random test).
HT	-	Hit Rate.
FA	-	False Alarm Rate.
ROC	-	Receiver Operator Characteristic.
AGC	-	Automatic Gain Control.

CHAPTER 1: Introduction

1.1 The History of Hearing Tests.

Audiology is the generic name for the study of the ear and of hearing disorders. More specifically, audiometry is the measurement of these hearing disorders and defects.

The field of audiology encompasses many diverse subjects. Newby and Popelka (1985) describe the complete audiologist as a combination of 'speech pathologist, otorhinolaryngologist, pediatrician, gerontologist, psychiatrist, psychologist, physicist, electronics engineer, and educator', and it is important to realise that all these subjects hold importance in audiology.

The need to measure a subject's hearing disorder is obvious; it is only by measuring the defect that any idea can be gained of what can be done to improve it.

The most basic measure of hearing disorder is to determine whether a subject can hear words spoken in a soft whisper, or requires them to be shouted. Next, is to put some kind of external loudness measure on the whisper and shout, to determine more accurately what the subject can and cannot hear.

Audiologists over the years have wanted to develop better and more accurate ways of measuring a subject's hearing and have developed a test that measures the hearing over a range of frequencies.

All these types of test, however, require the patient to give a subjective response and it is obvious that the next stage in the development of tests would be to measure the hearing objectively. These objective tests are becoming more widespread.

The field is still a long way away from sitting a subject down in a clinic, asking them to do nothing, diagnosing both the type and extent of disorder, and any more serious associated problems, and then prescribing a hearing aid to suit their precise needs. Perhaps the audiology clinics of the 21st century will be like this.

1.2 The Ear.

The ear is a remarkably complicated piece of natural engineering. A more detailed investigation of some of the physical properties underlying the transmission of sound in the ear will be discussed in section 1.3, however here, the basic biology and physiology of the human hearing mechanism will be discussed. The human ear has, on average, a range from 20Hz to 20kHz and, with the body's nervous system, can be a very efficient frequency analyser.

The ear is divided into 3 main sections called, for convenience, the outer, middle and inner ears. These are shown with the main mechanism of the ear in figure 1.1.

1.2.1 The Outer ear.

The part of the ear which is visible, the pinna, acts as a horn that collects the sound and channels it into the auditory canal. This is a fairly ineffective part of the human hearing system, but in other animals can be extremely developed, e.g. in bats.

The auditory canal is an approximately straight, air filled tube with a diameter of about 7mm and length 25mm. It is closed at the inner end by the eardrum or tympanic membrane which is roughly circular and has a flattened cone shape. The auditory canal has a very broad resonance curve and the gain between the pinna and the tympanic membrane can be of the order of 10 or 20 dB at frequencies between 2 and 6 kHz.

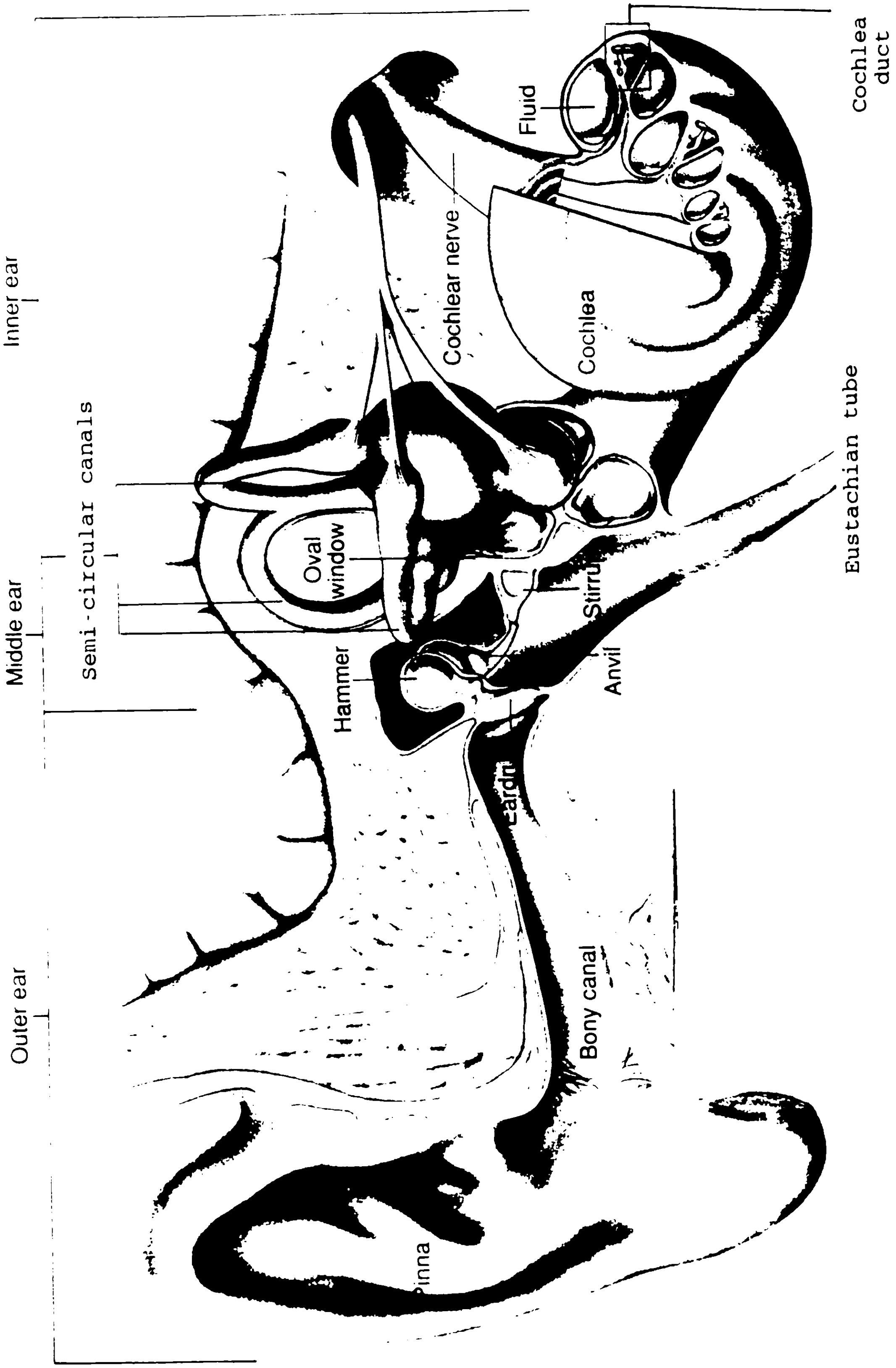

Figure 1.1 The ear (adapted from The Funday Times Jan. 21st 1990 - **Section of the Sunday Times).**

1.2.2. The Middle ear.

The middle ear is also air filled and has a volume of approximately 2 cm^3. It is connected to the back of the throat via the eustachian tube which is used as a pressure equaliser. There are three small bones within the middle ear and sound vibrations are transmitted through the tympanic membrane, to the malleus (hammer) then the incus (anvil) and finally to the stapes (stirrup). From there they enter the inner ear via the oval window. These small bones are collectively called the ossicles.

The middle ear acts as an impedance match between the low impedance of the air filled outer ear and the inner ear which is fluid filled, and has high impedance. The impedance is however dependent on intensity. If the sound has high intensity then the muscles surrounding the middle ear change tension to reduce the amplitude of motion of the stapes, thus protecting the inner ear. This is called the acoustic reflex. Since this reflex takes about half a second to become effective, short, loud sounds such as a gunshot can still damage the hearing mechanism.

1.2.3. The Inner ear.

The inner ear has three distinct parts, the vestibule, the semi-circular canals and the cochlea. It is surrounded by bone, except for the two entrances to the cochlea.

a) The vestibule connects with the middle ear through the round and oval windows. These windows are sealed to avoid leakage of the inner ear fluid into the air filled cavity of the middle ear.

b) The semi-circular canals have nothing to do with the hearing mechanism, but are involved in the subject's sense of balance.

c) The cochlea is perhaps the most important part of the inner ear, from the point of view of hearing. It is an approximately circular tube, coiled rather like a snail shell, making about two and a half turns. If it were straightened out it would have a total length of about 3.5cm. The cross-sectional area of the tube decreases irregularly from the base to the apex, but the total volume is about 0.05 cm^3. The cochlea will be looked at in more detail.

1.2.4 The Cochlea.

The cochlea tube is divided longitudinally, by the membraneous cochlea partition (which houses the cochlea duct), into the scala vestibuli (or upper gallery) and the scala tympani (or lower gallery). Figure 1.2 shows a schematic diagram of the uncoiled cochlea.
Figure 1.3 on the other hand shows a cross-sectional view,

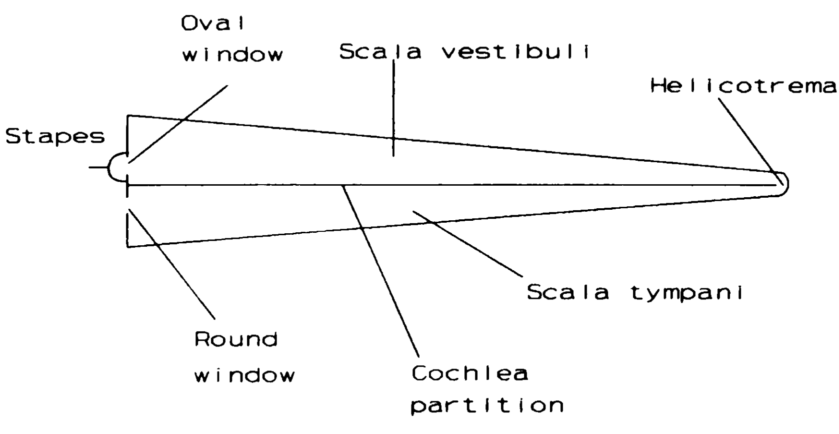

Figure 1.2 Schematic Diagram of the uncoiled cochlea.

and figure 1.4 shows the cochlea duct in more detail.

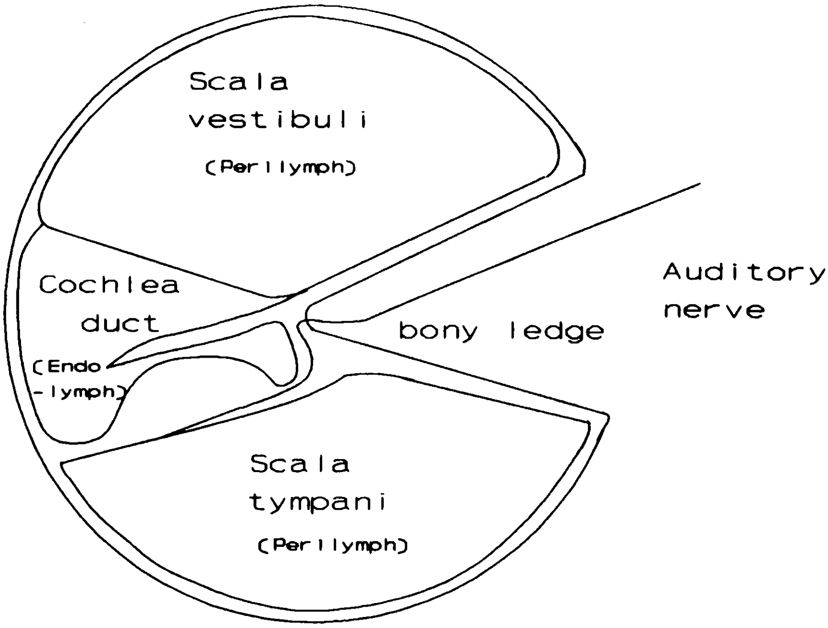

Figure 1.3 Cross-sectional view through the cochlea.

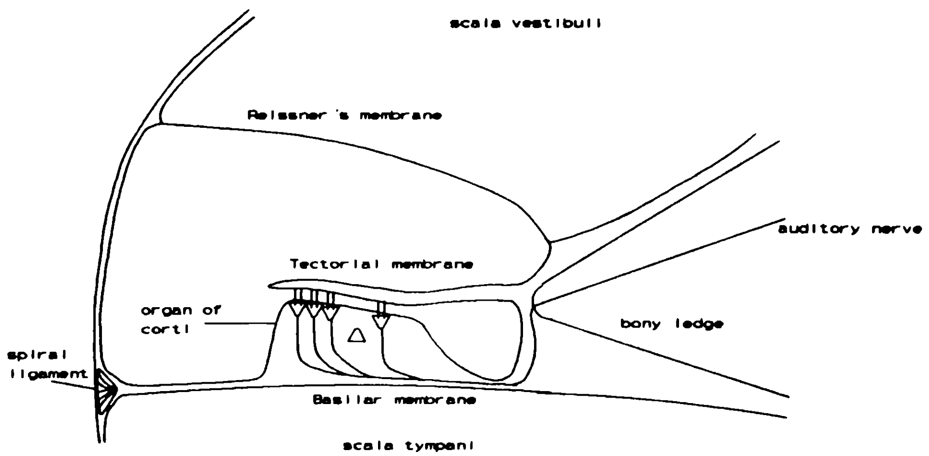

Figure 1.4 The cochlea duct.

The bony ledge shown projects from the central portion of the cochlea and carries the auditory nerve. After this bony ledge the nerve fibres enter the Basilar membrane which continues to the outside of the cochlea where it is attached to the spiral ligament. The tectorial membrane lies above the Basilar membrane and is attached at one side to the bony ledge. Also attached to this bony ledge is Reissner's membrane which runs diagonally across the cochlea to the opposite wall. The scala vestibuli lies above Reissner's membrane and the scala tympani below the Basilar membrane; the pie-shaped section in between is known as the cochlea duct.

On the top of the Basilar membrane is the organ of Corti which has 4 rows of hair cells, spanning the entire length of the cochlea. Several dozen small hairs extend from each hair cell to the upper surface of the tectorial membrane. There are about 30000 hair cells in all.

When the ear is exposed to a sound, for example a pure-tone, the motion of the eardrum is transmitted by the bones of the middle ear to the oval window. This creates a fluid disturbance which travels down the scala vestibuli, through the helicotrema and back up the scala tympani to the round window which acts as a pressure release terminator.

Since the organ of Corti is attached to the Basilar membrane, and the Tectorial membrane is attached to the bony ledge, relative motion between them flexes the hairs, thus exciting the nerve endings attached to hair cells into producing electrical impulses which can be interpreted by the brain.

1.3 Some Physical Properties in Hearing.

There are many basic principles of physics which are used by the human ear to allow it to detect and analyse the sounds of the environment. This section includes some of the most important and most interesting.

1.3.1 Perception of loudness.

The equation for the subjective perception of how loud a sound is comes from a statement of the Weber-Fechner law. It states :

> "The increase in stimulus necessary to produce a given increase in sensation is proportional to the pre-existing stimulus".

The law was originally associated with giving additional small weights to a subject already holding a larger weight. In this form it could be written as -

$$dS=k\frac{dW}{W},$$

where dS = the noticeable difference in stimulus,
dW = the addition of a small weight,
and W = the existing weight.
The equation can then be written as -

$$S=k \log_e W.$$

Applying this principle to a subjective assessment of the loudness perceived by a subject, it can be said that -

$$Loudness=k \log(Intensity)$$
$$i.e.\ L=k \log I.$$

In the above equations k is taken as unity, by definition, and the logarithms are taken to the base 10.

Experimentally, it can be shown that at 1kHz, the average root mean square pressure threshold has the value $2\text{x}10^{-5}$ Nm^{-2} and this is designated p_o. The associated intensity and loudness are I_o and L_o respectively.
i.e $L_o = log_{10}\ I_o$, and the difference in loudness is given by -

$$Z = \frac{Force}{velocity} = \frac{pA}{u}$$

Note: The more usual definition of impedance in physics is

$$Z = \frac{Force}{velocity} = \frac{pA}{u}$$

$$L-L_o=\log_{10}\frac{I}{I_o}\ \ bels$$
$$=10\ \log_{10}\frac{I}{I_o}\ \ dB.$$

If 0dB is defined to correspond to a root mean square pressure p_o of $2x10^{-5}$ Nm^{-2} at 1kHz then -

$$L=10\ \log_{10}\frac{p^2}{p_o^{\ 2}}\ \ dB,$$

since Intensity, I, is proportional to the square of pressure p^2. This quantity is known as the sound pressure level or SPL.

$$SPL=20\ \log_{10}\frac{p}{p_o}\ \ dB.$$

1.3.2 Acoustic Impedance.

The definition of acoustic impedance Z_{ac} is -

$$Z_{ac}=\frac{p}{\boldsymbol{v}},$$

where p = pressure
and $\boldsymbol{v}$ = volume velocity.

Volume velocity is equal to the average velocity u, of each particle, multiplied by the area through which the particles travel. So

$$Z_{ac}=\frac{p}{uA}.$$

There is another important quantity defined as specific acoustic impedance Z_s and -

$$Z_s=\frac{p}{u}.$$

1.3.3. Sound Transmission from outer to inner ear.

An interesting problem to look at is the transmission of a sound wave from the auditory canal through the middle ear and then on to the oval window of the cochlea.

In the first instance, the transmission from the ear canal to the tympanic membrane must be considered. (See figure 1.5.)

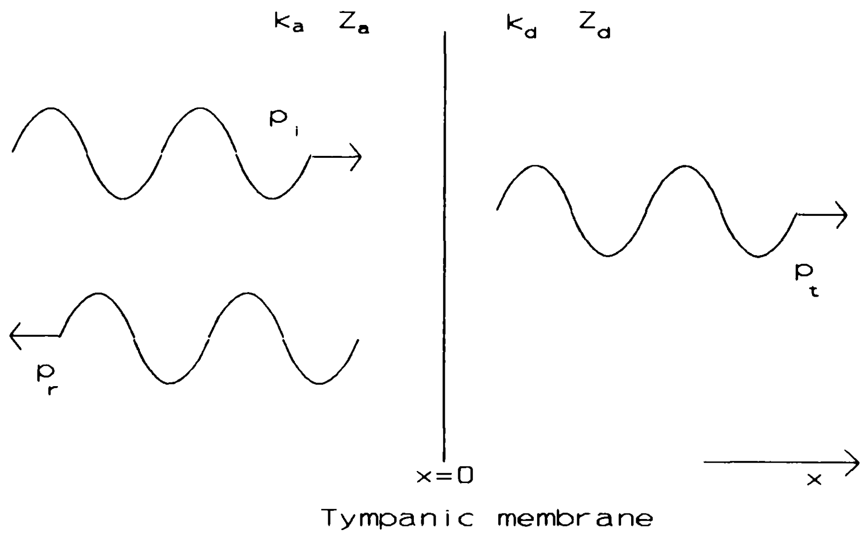

Figure 1.5 Waves at the eardrum. p_i, p_r, *and* p_t. k_a *and* Z_a, *wavenumber and acoustic impedance of the air canal,* k_d *and* Z_d *of the eardrum.*

In the air canal the incident wave has pressure p_i, and the reflected wave p_r while at the drum the transmitted wave has pressure p_t.

$$p_i = A_i \exp^{i(\omega t - k_a x)}$$

$$p_r = A_r \exp^{i(\omega t + k_a x)}$$

$$p_t = A_t \exp^{i(\omega t - k_d x)}$$

There are two important boundary conditions which apply at the eardrum.

i) The pressures must be continuous at the drum, and ii) the velocity must be continuous at the drum.

$$p_i+p_r=p_t \quad {}_{x=0}$$

$$u_i-u_r=u_t \quad {}_{x=0}$$

Taking the first of these, at x=0 we have -

$$A_i\exp^{i\omega t}+A_r\exp^{i\omega t}=A_t\exp^{i\omega t}$$

$$i.e. \quad A_i+A_r=A_t,$$

and since

$$u_i=\frac{p_i A}{Z_a}; \; u_r=\frac{p_r A}{Z_a}; \; u_t=\frac{p_t A}{Z_d},$$

the second boundary condition becomes -

$$\frac{p_i}{Z_a}-\frac{p_r}{Z_a}=\frac{p_t}{Z_d}.$$

Substituting from the first condition we get -

$$\frac{1}{Z_a}(p_i-p_r)=\frac{1}{Z_d}(p_i+p_r),$$

and multiplying through the whole equation by $(Z_aZ_d)/p_i$ gives -

$$Z_d-Z_d\frac{p_r}{p_i}=Z_a+Z_a\frac{p_r}{p_i}$$

$$i.e. \quad \frac{p_r}{p_i}=\frac{Z_d-Z_a}{Z_a+Z_d}.$$

Since I is proportional to p^2, the ratio of reflected intensity to incident intensity, sometimes known as the refection coefficient R becomes -

$$\frac{I_r}{I_i}=R=\frac{(Z_d-Z_a)^2}{(Z_a+Z_d)^2} \quad ,$$

and so the transmission coefficient is -

$$T=1-R$$

$$=1-\frac{(Z_d-Z_a)^2}{(Z_d+Z_a)^2}$$

$$\rightarrow \quad T=\frac{4Z_aZ_d}{(Z_a+Z_d)^2}.$$

Z_a can be calculated using the fact that -

$$Z_{ac_a}=\frac{\rho_o c}{A}.$$

Since the average cross sectional area of a human ear is about 4.4×10^{-5} m^2 , ρ_o = 1.21 Kgm^{-3} for air, and the speed of sound in air is 343ms^{-1}, Z_{ac_a}= 94×10^5 Pa s m^{-3}.
Zwislocki (1975) measured the acoustic impedance of the cochlea to be about Z_{ac_c} = 3.5×10^{10} Pa s m^{-3}. If it is assumed that the impedance at the eardrum is equal to this cochlear impedance, the transmission from canal to eardrum can be calculated from

$$T=\frac{4Z_aZ_c}{(Z_a+Z_c)^2}.$$

Z_a = 0·028 Pa m s
Z_c = 105 9 Pa m s

Substituting in the above numerical values gives

$$T = 0.001.$$

That is, the transmission is 0.1%. This seems improbable and very energy inefficient.

In fact, what happens is that the middle ear system acts as a transformer, matching the lower impedance of the air canal with the higher impedance of the cochlea. The middle ear utilises a lever action between the malleus and the incus, and the ratio of the effective surface area of the tympanic membrane to that of the oval window at the entrance to the cochlea. (See figure 1.6.)

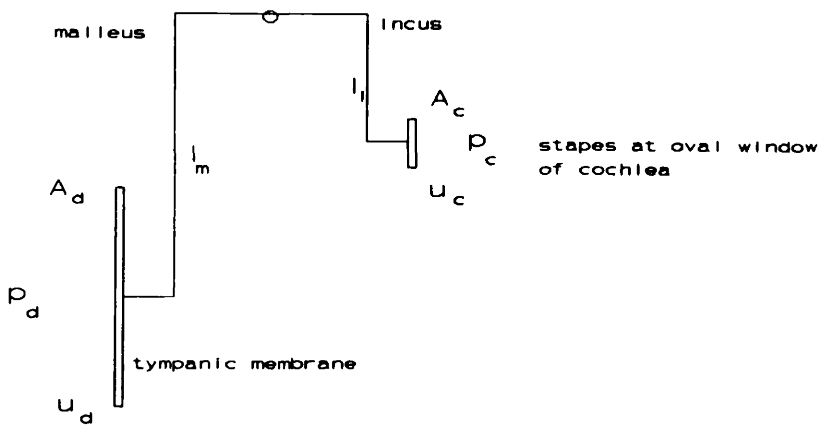

Figure 1.6 A Schematic diagram of the middle ear transformer action. *A,p,u correspond to area, pressure and average velocity; l=length; d=drum, c=cochlea; m=malleus and i=incus.*

$A_{c(\text{oval window})}$ =

In the human ear, on average, A_d = 5.5x10^{-5} m^2

$A_{c(\text{oval window})}$ = 3.2x10^{-6} m^2

and l_m/l_i = 1.3.

For equilibrium -

$$F_d l_m = F_c l_i$$

$$A_d p_d l_m = A_c p_c l_i$$

$$\frac{p_c}{p_d} = \frac{A_d l_m}{A_c l_i}.$$

Where F_d and F_c are the forces at the drum and cochlea respectively. Substituting in the values gives p_c/p_d =22.3. This is equivalent to a sound pressure level of 27dB, using the previously defined equation for SPL. This means that the sound at the cochlea is 27dB louder than that at the eardrum.

In the literature, the impedance quoted is the acoustic impedance and not Force/velocity. Although the acoustic impedance is technically not the one that should be used in the transmission equation, substitution of these acoustic impedances into the equation yield identical values for T. This is because the values of area cancel top and bottom.

$$\frac{u_d}{u_c}=\frac{l_m}{l_i},$$

$$Z_c = \frac{p_c A_c}{u_c} \quad ; \quad Z_d = \frac{p_d A_d}{u_d}$$

and so

$$\frac{Z_c}{Z_d}=\frac{l_m p_c A_c}{l_i p_d A_d}$$

Substituting in from above for p_c/p_d gives-

$$\frac{Z_c}{Z_d} = (\frac{l_m}{l_i})^2$$

Since Z_c=0.3584 Pa m s, substituting in the values gives Z_d=0.212 Pa m s, and the transmission equation gives T=0.41.

This means a transmission of about 40% which is of course much more reasonable.

1.3.4 Electrical Analogue.

A technique often used to model the response of the ear is to produce its electrical analogue. It is well known that mechanical systems have electrical analogues and that the behaviour of a mechanical system can be investigated by considering its electrical counterpart. The same is true of acoustical or mechano-acoustical systems. In this case the electrical current is equivalent to the volume velocity, and the voltage represents the pressure.

A simple acoustic/electrical analogue which can be considered is the Helmholtz resonator. (See figure 1.7).

A Helmholtz resonator is a rigid walled container of volume V which has a neck of length L and cross-sectional area S.

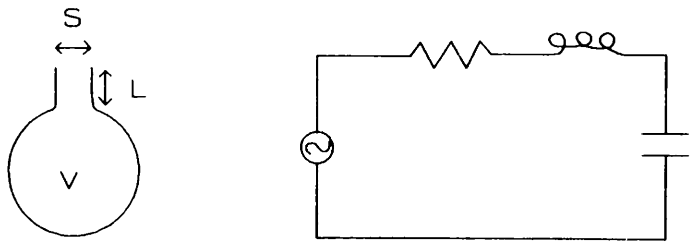

Figure 1.7 The Helmholz Resonator and its electrical analogue.

In the long wave approximation:

i) If $\lambda >> L$, the fluid in the neck moves as a unit providing a mass element.

ii) If $\lambda >> V^{1/3}$, the acoustic pressure in the cavity leads to a stiffness element.

iii) If $\lambda >> S^{1/2}$, the opening provides a resistive element.

i) The fluid in the neck has a total effective mass m,

$$m=\rho_o S L',$$

where ρ_o = density of air,

L' = effective length of the neck - the detail of this is not important, but because of radiation and mass loading, the effective length might be longer than L.

ii) We assume that the neck is fitted with an air tight piston, which is pushed by a distance ξ so that the volume changes by ΔV where -

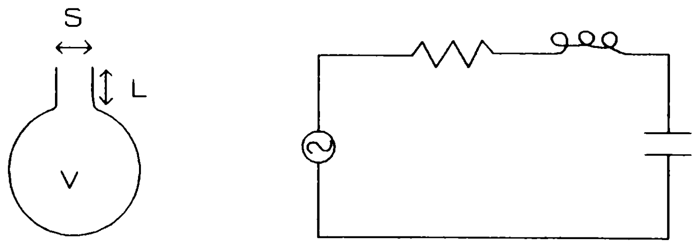

Figure 1.7 The Helmholz Resonator and its electrical analogue.

In the long wave approximation:

i) If $\lambda \gg L$, the fluid in the neck moves as a unit providing a mass element.

ii) If $\lambda \gg V^{1/3}$, the acoustic pressure in the cavity leads to a stiffness element.

iii) If $\lambda \gg S^{1/2}$, the opening provides a resistive element.

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where ρ_o = density of air,

L' = effective length of the neck - the detail of this is not important, but because of radiation and mass loading, the effective length might be longer than L.

ii) We assume that the neck is fitted with an air tight piston, which is pushed by a distance ξ so that the volume changes by ΔV where -

$$\Delta V=-S\xi .$$

This results in a condensation -

$$\frac{\Delta\rho}{\rho}=-\frac{\Delta V}{V}=\frac{S\xi}{V},$$

and so the pressure increases by

$$p=\rho_o c^2\frac{\Delta\rho}{\rho}=\frac{\rho_o c^2 S}{V}\xi .$$

If a force F, is required to maintain the displacement then -

$$F=pS=\frac{\rho_o c^2 S^2}{V}\xi ,$$

and so the effective stiffness, s -

$$s=\frac{\rho_o c^2 S^2}{V} .$$

iii) It is assumed that the moving air in the neck radiates sound into its surroundings in the same manner as an open ended pipe and so -

$$R_r=\frac{\rho_o c k^2 S^2}{2\pi} .$$

Kinsler and Frey (1962)

The equation of the Helmholz resonator can be written as follows :

$$Sp=m\frac{d^2\xi}{dt^2}+R_r\frac{d\xi}{dt}+s\xi ,$$

now since -

$$Z=\frac{p}{v}\quad ;v=\frac{d\xi}{dt}S,$$

we can write -

$$Z=\frac{p}{\mathbf{v}}=\frac{m}{S^2}\frac{d\mathbf{v}}{dt}+\frac{R_r}{S^2}\mathbf{v}+\frac{s}{S^2}\int\mathbf{v}dt$$

$$=i\omega\frac{m}{S^2}+\frac{R_r}{S^2}+\frac{s}{S^2}\frac{1}{i\omega}.$$

$$\text{That is}\quad Z=R+i(\omega M-\frac{1}{\omega C}),$$

$$\text{where}\quad R=\frac{R_r}{S^2}\ ;M=\frac{m}{S^2}\ ;C=\frac{S^2}{s},$$

and so

$$R=\frac{\rho_o ck^2}{2\pi}\ ;\ M=\frac{\rho_o L'}{S}\ ;\ C=\frac{V}{\rho_o c^2}\quad .$$

This equation for C is often used for calibrating acoustic immittance devices (see section 1.6).

Using this technique, the ear, or part of the ear can be analysed. The ear as a whole, is a very complicated system, but as an example, an ear with a missing incus bone will be considered, as this is much simpler. (See figures 1.8 and 1.9).

The first three elements, L_d, R_d and C_d are equivalent to the acoustic inertance, resistance and compliance of the diaphragm. The shunt capacitance C_{ad} represents the acoustic compliance of the air in the space behind (to the right of) the diaphragm. L_r and R_r symbolize the acoustic inertance and resistance of the constriction and C_a is the acoustic compliance of the second volume of air.

It can be shown, by comparing the elements shown in figure 1.9 to the values of acoustic immittance* of a real ear with a missing incus bone, that figure 1.9 is a very good approximation to the real ear. Without going into mathematical detail it can be seen that this type of analysis can give physical information about the ear. This electrical analogue technique is often used in conjunction with measurements of the acoustic impedance of the middle

* See section 7.1.

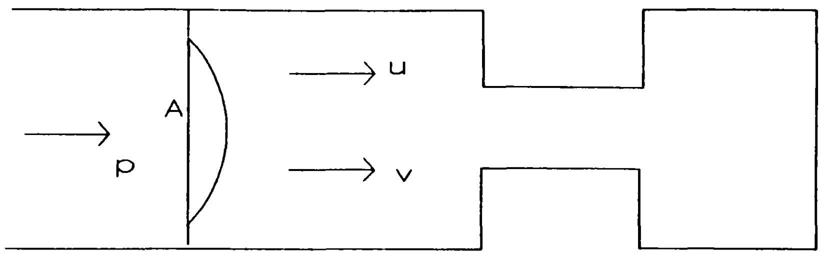


Figure 1.8 The mechano-acoustic system for an ear with a missing incus.

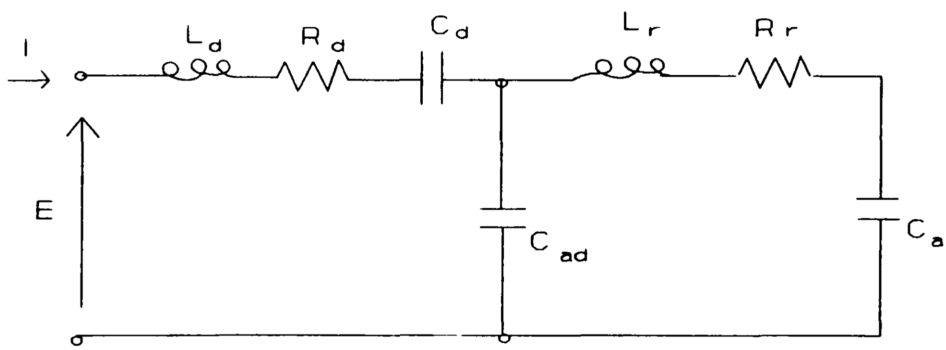


Figure 1.9 The electrical analogue of figure 1.8.

ear system (see chapter 7 for acoustic impedance techniques).

1.3.5 Sound Transmission in the Cochlea.

Most of what is known about the properties of the cochlea come from the work of Bekesy (1960) who received the Nobel Prize for his work. He studied models of the cochlea to simulate the behaviour of the cochlea partition and subsequently investigated the behaviour of the human cochlea. He made several important observations which give some understanding of how sound is transmitted through the cochlea.

The most important of these observations are listed :

a) The acoustic compliance of the cochlea partition increases exponentially from the base of the cochlea to the apex.

b) The travelling waves in the cochlea always decrease in velocity from base to apex.

c) The waves reach a maximum amplitude somewhere along the membrane and then fall away rapidly.

d) As the frequency of the wave increases, the site of maximum amplitude moves systematically from apex to base.

e) As the wave progresses there is an increase in the phase lag between the Basilar membrane displacement and the displacement of the stapes.

f) The partition vibrations are heavily (but not critically) damped.

As a consequence of Bekesy's findings, many theories have been proposed to explain the workings of the cochlea. There is still some controversy over which theory is correct and here the author simply wishes to give an outline of the most important of these theories.

Bekesy himself proposed the travelling wave theory, which states that sound travels through the cochlea by setting up travelling waves in the Basilar membrane. He believed that the maximum amplitude of these waves occurs at a point on the Basilar membrane corresponding to the frequency of excitation and it was that frequency which was perceived by the brain. This does not, however, account for the fact that the human ear can detect very small changes in frequency and so Bekesy believed that this travelling wave idea was enhanced by neural processing of the nerve signals from the hair cells of the cochlea.

Some researchers have proposed a frequency theory, which explains the perception of frequency by the rate of electrical impulses in the auditory nerve. Unfortunately, modern techniques of measuring nerve impulses have shown that the maximum rate at which a nerve can 'fire' is 1kHz. Even if we consider that several fibres could act together and discharge at different synchronised times, it is really only possible to accept that this explains frequency perception up to 5kHz.

Another theory known as the volley theory, uses this nerve impulse theory to explain frequency perception up to about 5kHz and Bekesy's travelling wave theory, dependent on the place of maximum amplitude, to explain frequencies above 5kHz.

There have been several attempts to mathematically model Bekesy's travelling wave theory, most notably by Ranke(1950), Zwislocki(1950) and more recently Siebert(1974). The physical basis of these models is reasonably simple. The model considers the cochlea to be a box, split in two, lengthways, by a partition whose mechanical properties are expressed in terms of its admittance. The mathematical details of the model are very complicated and not really important here, but figure 1.10 shows a simplified diagram of the model.

The oval and round windows are represented by sinusoidal velocity sources, as are the reflections at the end of the box corresponding to the apex of the cochlea. The depth of the two canals is d, and the partition has admittance Y(x), which is a function of the distance from

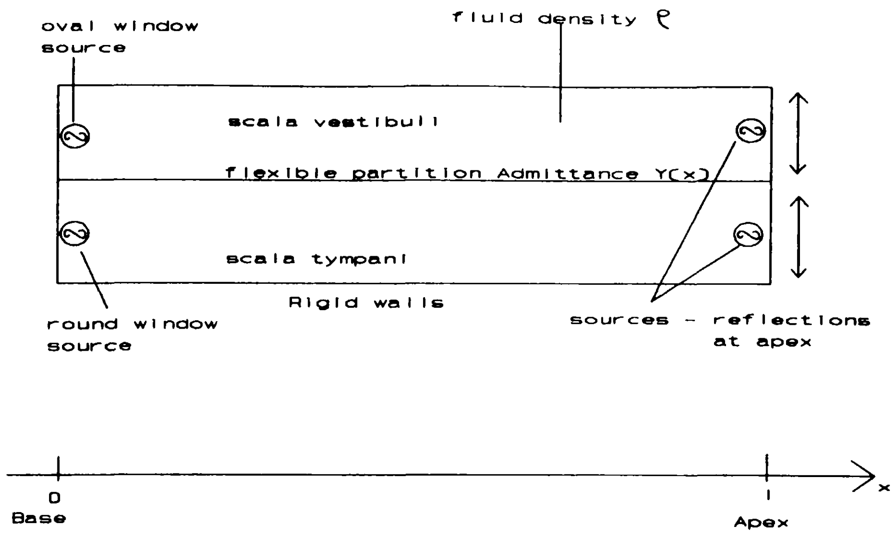


Figure 1.10 Model of the cochlea.

the stapes, x.

Seibert (1974) considered a short wave approximation of waves in this type of box, while Zwislocki (1950) considered a long wave approximation, and it was this work that led Zwislocki in 1975 to calculate the impedance at the oval window - which was used in section 1.3.3.

These models do, however, show that this type of representation explains Bekesy's findings quite accurately.

1.4 Some Useful Properties of the ear.

What the human ear actually perceives, given a set stimulus, is not always straightforward and there are some interesting effects which must be considered when measuring hearing and hearing loss.

1.4.1 Equal Loudness.

The response of the ear is not flat over frequency. Experiments have been done to determine when two tones of different frequency are perceived as equally loud. For example, Robinson and Dadson (1956), who measured equal loudness contours for a human listener as shown in figure 1.11. The curves are labelled with the sound pressure levels associated with the corresponding tone of 1kHz. It can be seen that lower and higher frequencies require much greater sound pressure levels to be perceived by an observer as the same loudness as the corresponding 1kHz tone.

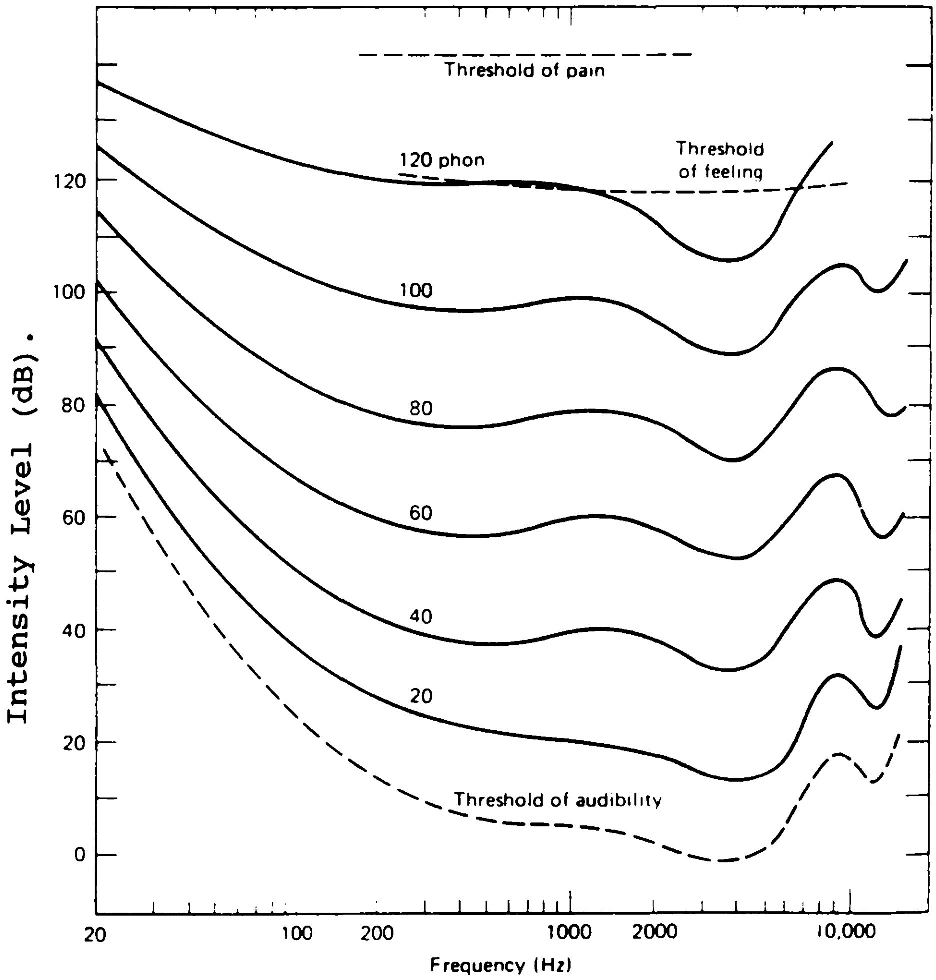


Figure 1.11 Equal Loudness Contours (after Robinson and Dadson).

1.4.2 Critical Bandwidths.

The ear can conveniently be thought of as a series of filters whose pass-bands are known as critical bands. Fletcher and Munsen (1937) showed that the threshold of hearing for a tone in the presence of background noise increases as the bandwidth of the noise is increased. At a critical value (called the critical bandwidth), further increases in the noise bandwidth have no effect on the pure-tone thresholds.

An alternative technique, used to show the same effect, is to measure the pure-tone thresholds in the presence of a uniform wide-band noise. This leads to a measure of what are known as the critical ratios.

1.4.3 Masking.

Masking is the process of making one sound inaudible by the introduction of another. The most straightforward type of masking, is the masking of one pure-tone by a second pure-tone. In the main, however, in audiometry, noise is used to mask pure-tone signals. From section 1.4.2 it can be seen that masking noise, with the appropriate critical bandwidth, is the optimal method of masking a pure-tone with a frequency equivalent to the centre of the band.

1.4.4 Non-Linearity of the Cochlea.

There is much evidence of non-linearity in the cochlea, but as an example, we take the phenomenon of beats. If two tones of similar frequency f_1 and f_2 are presented with equal loudness to one ear, the ear perceives a tone of single frequency $f_c = (f_1+f_2)/2$ whose intensity oscillates at a beat frequency of f_1-f_2. The ear will only resolve the frequencies when their difference is increased to about the magnitude of the critical bandwidth.

This phenomena does not occur if f_1 and f_2 are presented to different ears, implying that it is a cochlea effect.

1.4.5 Non-Linearity in Processing.

If two or more harmonics of a tone are presented to the ear, it is possible for the ear to detect the fundamental of these two harmonics. For example, if 1000Hz and 1200Hz are presented to an ear, the ear may detect 200Hz. Even if this fundamental frequency is masked and therefore 'undetectable', it can be perceived. This happens even if the tones are presented to different ears, and is therefore a neural and not a cochlea effect.

1.5 Hearing Defects.

If a loss in hearing is due to a defect situated in the outer or middle ears, it is termed a conductive loss, whereas, if it is situated in the cochlea or neural passages to the brain, it is termed sensori-neural.

1.5.1 Conductive Hearing Defects.

As the name implies, a conductive hearing loss means a loss in the conduction of the sound from the outer ear to the tympanic membrane and through the bone system of the middle ear. The loss may be caused by a number of factors, the simplest of these being an obstruction in the outer ear canal.

A detailed investigation of all forms of conductive loss is not important here, but some of the most common defects are listed.

(i) A blockage in the outer ear, e.g. small marble.

(ii) A perforation of the eardrum (or tympanic membrane).

(iii) Infections can sometimes lead to scarring of the middle ear.

(iv) The presence of fluid in the middle ear, rather than air. In children, this is known as 'glue ear'.

(v) Otosclerosis is a disease where extra bone grows sporadically in the middle ear, usually near the oval window.

The types of loss described in (i)-(iii) affect, in the main, only lower frequencies, and the higher frequencies remain virtually undamaged. This happens because the higher frequencies are transmitted more readily through the skull and stimulate the cochlea directly. The losses described in (iv) and (v) are more likely to affect frequencies uniformly.

1.5.2. Sensori-Neural Defects.

This type of loss means that the loss occurs either in the cochlea (sensori-) or in the neural passages to the brain (neural or retrocochlear). There are again many reasons for a loss of this type and only the most common shall be looked at here.

(i) Presbycusis, is the most common form of deafness in elderly patients. It is caused by the deterioration of the cochlea with age, and begins with the highest frequencies.

(ii) Infections, congenital defects and injury to the cochlea.

(iii) Meniere's disease. Unlike other problems of the inner ear, this disease is episodic and the sufferer can be free from symptoms for long intervals. The symptoms include vertigo, deafness and tinnitus (which will be explained in section 1.5.3).

(iv) Tumours of the VIIIth nerve. The most common of these is the acoustic neuroma. This can grow very large if undetected, and compress the surrounding vessels which may cause severe problems to develop.

Usually, but not exclusively, sensori-neural losses are greatest at high frequencies, and they are regularly accompanied by poor speech discrimination and recruitment. Recruitment is a rapid increase in the sensation of loudness of a tone without a corresponding increase in the actual loudness of it. This phenomena will be discussed in more detail in chapter 4.

1.5.3. Tinnitus.

This is a condition where subjects hear sounds which have no external origins. These are often heard as high pitched tones or noise and have a subjective intensity of around threshold. Even though these noises are quiet, their persistence can make tinnitus a distressing condition.

Tinnitus usually accompanies sensori-neural losses but can also be found in otosclerosis.

1.6 Audiometry.

Audiometry is the generalised name for the field of measuring hearing loss. There are many types of audiometry which use a variety of test materials and techniques, and only some of these types are dealt with in this thesis. Each of these will be explained in more detail in the relevant chapter, but here a general overview of the most common hearing tests is provided.

The most widely used form of hearing test is that of pure-tone audiometry. This, as the name implies, uses pure-tones as the stimulus. These tones are presented at various intensities and frequencies to the subject who must decide whether or not they hear the tone, usually by raising their arm or pressing a button when they hear it. This is a good way of testing to give an overall view of the subjects thresholds across frequency. The test procedure and protocol for this type of test have been well established over the many years it has been used. The tones can be presented on either earphones, bone conductors or free field, and masking* of the non-test ear can be employed to avoid the possibility of cross-hearing,

* See section 2.3.2.

that is, the tone actually being heard in the ear which is not being tested. To gain as much information as possible in as short a time as possible about the subjects hearing, the pure-tone method is the easiest and best as it covers many frequencies.

This pure-tone test may suggest, or the audiological technician may suspect some other type of hearing defect, for example, Meniere's disease or recruitment. For a full diagnosis, other more specialised tests may be required. There is an extensive battery of these tests which will be used depending upon the defect suspected. In particular there is a range of tests known collectively as supra-threshold tests, because they are all carried out above the subject's threshold. These can be used to confirm the presence of such disorders as recruitment.

There are, however, a few subjects who find it difficult to respond to pure-tones and it is clear that in every day life it is not crucial that a subject can hear pure-tones. The most important facility for any subject is that they can communicate. They must be able to hear and understand speech.

It is an obvious progression from employing pure-tones as the stimulus for measuring hearing, to move on to speech materials. It is equally obvious that there is an almost infinite range of speech materials which can be used. This type of testing will be discussed in more detail in chapter 5 but involves the subject being read words or phrases and then repeating these words or phrases to determine whether they have heard them correctly.

All the forms of test mentioned so far involve some sort of subjective decision on the part of the subject. This is not ideal in many cases. For example, some patients are completely incapable of giving a subjective response. Objective testing would then be a more accurate and reproducible way of finding a subject's hearing loss.

Several methods of objective testing have been developed over a number of years. The most common form of objective testing available to most hearing departments is that of acoustic immittance measurements. This is a mechanism where the acoustic impedance of the middle ear is measured by comparing a probe tone with its reflection from the tympanic membrane. These measures can be of great benefit in diagnosing conditions of the middle ear such as glue ear and otosclerosis.

Other methods of objective testing measure some of the various bio-electric potentials in and around the brain and how they react to auditory stimulus. There are a number of bio-electric potentials associated with the hearing mechanism and only a few of these are of any use from a diagnostic point of view. There are three main types of test used in this field:

1. Electrocochleogram (ECochG).

This picks up potentials originating in the cochlea, called endocochlea potentials. They are measured using a transtympanic needle electrode which rests on the niche of the round window.

It is important for this type of measurement, that the sound stimuli have very sharp onset times. Very often a wideband click or a high frequency tone burst is used. Between 100 and 500 stimuli are necessary for all the background noise to be eliminated using an averaging technique.

In adults this test is useful to differentiate between Meniere's disease and an acoustic neuroma.

2. Brainstem Electric Response (BSER).

The waves picked up in this type of measurement have an amplitude of less than 1 μV and are measured by placing an electrode on the vertex and one on the earlobe. Again clicks and high frequency tone bursts can be used and averaging must be employed.

Examination of the waves, paying particular attention to their latencies (that is, the time between stimulus and response) often leads to clarification of a neurological diagnosis because in many diseases the latencies are prolonged.

3. Cortical Evoked Potentials (V Potentials).

'Cortical' means that the potentials are produced in the grey matter of the brain. In this case, three electrodes are used, one on the vertex, a reference on the earlobe and an earth on the forehead. This was the original objective test which measured bio-electric potentials and was known as Evoked or Electric Response Audiometry ERA. A pure-tone audiogram can be constructed from this test but it is usually reserved only for those patients whose pure-tone audiograms were unreliable or suspect since the averaging means that the construction of an audiogram may take about an hour.

These tests are very expensive in both money and time, since the electrodes used are very specialised and the averaging procedure means they take a long time to execute. For these reasons they are reserved for a small number of subjects. At the moment no way has been found round the averaging since the signal to noise ratio is very low.

1.7 Computers in the field of Audiometry.

The potential for the use of computers in audiology has been recognised for many years (Levitt 1966), and computers have been used in clinical audiology practices since about 1974. Their use, however, has, in the main, been restricted to the measurement of auditory evoked potentials. In recent years many immitance testing devices have used up to date computer technology and there are now many such devices available which run automatic tympanometry under microprocessor control.

Since the cost of this technology is plummeting, it is expected that there will be an ever increasing use of computers in the field of audiology, allowing easier, more sophisticated and more efficient tests.

Yanz and Siegel (1989) discuss the 'Computerized Audiology Clinic' and their article suggests many of the ways in which computers can aid the audiologist. They write :

"The major advantage of a computerized instrument does not depend on automation but on the ability to increase accuracy, speed and power of audiologic testing under the direct control of an experienced tester".

They do not suggest that computers will replace the trained and experienced audiologist. On the contrary, these computers are only really powerful in the hands of one such trained operator.

Some of the main points discussed under the question 'What can computers accomplish?' are listed here. They are of great interest since they encompass many of the aims of this thesis.

i) The computer saves time doing boring, repetitive tasks.

ii) The computer increases the accuracy and consistency of the test setup. That is, where options can be used, the computer ensures that the same setup is used for every patient.

iii) The computer can offer automation of routine tests. The authors are at pains to stress the difference between computerization, which may mean a test which is either automatic or manual, and automation, which is often used in industrial screeners and where the test is completely inflexible.

iv) The computer allows the incorporation of more sophisticated tests, and those tests which are difficult to perform manually. Many new test designs are easy to implement since they require changes only in the software.

v) There is no better way of collecting and managing large quantities of data than a computer.

Yanz and Siegel stress, however, that there may be inertia encountered when trying to introduce these types of machine, since they are so different from the currently used manual machines. This means that the human-computer interface is of crucial importance.

Many authors have compared the threshold results obtained using a computer controlled, or automatic audiometer, to those obtained using manual audiometry, in order to determine whether these computerized machines give valid threshold results. For example, D. Harris (1979) compared the AC thresholds obtained using Bekesy* self-recording audiometry to those obtained using a microprocessor controlled test simulating the manual audiometric test. He found only slight differences in the thresholds obtained, and concluded that the microprocessor was superior in two ways, a) it required less analysis and b) it could be used either automatically or manually. J. Harris (1980) compared four different computerized audiometry methods in an industrial hearing conservation program and found a method of pulsed Bekesy which he called NSMRL Mark I to give marginally better thresholds than the other systems tested. Jerlvall et al. (1983) performed tests on ten normal hearing subjects, comparing their thresholds measured manually and by computer control. The results showed that the thresholds were very similar using the two techniques but that the computerized thresholds were more reliable when a test-retest measurement was calculated. They stress, however, that the subjects tested were young and normal-hearing and that tests done on hearing impaired subjects need not give the same good agreement. Cook and Creech (1983) discussed the reliability and validity of computer hearing tests and again showed no significant difference between the thresholds obtained using a manual test and those measured under computer control.

Computers have been used simply to manipulate audiometric data after a test has been done, for example,

*** See section 2.9.1.**

Grabowsky and Zalewski (1990). Many recent manual instruments allow test results to be downloaded to this type of data-base.

Some specific computer controlled systems have been implemented and tested. Many of these use what is now outdated technology but it is interesting to study them to appreciate the progress which has been made. In 1972, Sparks used a Beltone 15-C audiometer and a Digital PDP-8 computer. He investigated the results obtained for AC and BC thresholds, with masking if required, using this instrument and compared them to results measured manually. He found that the computer gave accurate and reliable results since it was much more stringent and persistent than a manual operator would have been. The system, however, used a manual computer/audiometer link, which is not ideal.

Wood et al. (1973) reported very similar results with a comparable system. Campbell (1974) reported the use of a random test, selecting ear and frequency randomly on each trial. He reported that the results of this random test gave marginally lower thresholds than those obtained manually.

Sakabe et al. (1975) and (1978) utilised a microcomputer to measure pure-tone AC thresholds following the manual test recommended by the Japan Audiological Society. (It should be noted that this is a very different procedure from that recommended by the BSA). This produced a set of patient instructions on a character display screen, and showed the responses on an LED matrix representing an audiogram. The responses of the subject were monitored and if unusual patterns occurred an error code was produced, halting the test. Compared with modern technology this system is very slow and cumbersome.

Another, more modern system, used a computer to control two manual audiometers. The work was done by Picard et al. and named BOBCAT (Battery of Basic Audiometric Tests).

This system was found to be slower than a manual test but it did allow multiple patient testing. Again the results of manual and computer controlled tests agree very well.

The final use of computers in audiology which should be discussed is their role in training. Computers can be used to simulate closely the responses of a patient, to allow audiological trainees to accomplish life-like tests without the presence of an actual patient. Gatehouse (1986) discussed the general characteristics which a realistic simulator should have, and Haughton (1990) described his application of some of these principles.

Finally, let us return to the article by Yanz and Siegel (1989) who describe the ultimate goals of a computer controlled audiometer as :-

a) a fully integrated system which
b) stores and retrieves patient information,
c) gets test data,
d) directs patient rehabilitation,
and e) deals with patient billing.

1.8 The work of this thesis.

It would appear from the literature that computer controlled audiometry has been around in some form or other for many years and, more importantly, that thresholds measured on these computer controlled machines agree very well with those measured using manual methods. It is valid therefore to use computer controlled machines in clinical tests. There is a worry expressed by many of the authors, that there are some patients who find it impossible to carry out an automatic test, since they require reinstruction and encouragement during the actual running of the test, which is of course not possible if the test is completely automatic.

The work of this thesis involved the development of a computer controlled audiometer which in fact meets the ultimate goals described by Yanz and Siegel.

The work implements a fully automated audiometric test to measure AC and BC thresholds, with masking if required, and displays the results on the screen of the computer. Although the test is automated, extensive provision has been made to meet the needs of many patients. The test can be stopped, to allow the operator to talk to the patient, or parameters can be changed, allowing many more patients to be accommodated.

The work implements other tests and integrates these into the system (goal (a)). Patient information can be stored, retrieved and printed out along with the relevant test data (goals (b) and (c)). The system has also been used for a direct referral scheme, in a hospital hearing clinic which went some way along the road of meeting goal (d). Goal (e) would be an easy programming job if billing were required in the UK.

Computer control of audiometric tests allows the use of many other facilities and opens up the possibility of other tests which would be virtually impossible to carry out manually. The work of this thesis also encompasses some new ideas and modifications to existing tests, utilising the power of a computer in audiology.

CHAPTER 2: Pure Tone Threshold Audiometry

2.1 Introduction

Hearing disorders in Britain are investigated, in the first instance, by the determination of a pure-tone audiogram. These audiograms are graphs of frequency against hearing level which are obtained by presenting pure-tone signals of varying frequencies and intensities to the patient who responds depending on whether or not they hear the tone. A Pure-tone audiogram will be carried out on every patient seen in a clinic. Other subsequent tests may then be performed. However, these are dependent on the pure-tone results.

In the U.K. the standardising body for audiometry is the British Society of Audiology (BSA) and pure-tone audiograms are carried out following a procedure laid out in the BSA's recommended procedures. These procedures will be discussed in more detail in section 2.3.

Pure tones are delivered to the patient in two ways:

i) through earphones into the outer ear

and ii) through a bone vibrator placed on the bones of the skull. The most commonly used bone is the mastoid bone behind the ear (thus channelling the sound directly into the inner ear).

The combination of these two, allows clinicians to differentiate between conductive and sensori-neural pathologies. The former would produce reduced hearing via the earphones but virtually normal hearing via the bone conductor, thus indicating some kind of blockage between inner and outer ear. Such a loss in conduction of sound to the inner ear could be due to, for example,a foreign body in the ear canal or a problem with the ossicle chain. In most cases a loss of this type can be rectified.

A sensori-neural loss means that the fault lies in the inner ear. That is, the cochlear or the neural pathways to the brain are damaged. These faults are usually not rectifiable although recent cochlear implant technology may help some limited types of disorder. This surgery does not restore "normal" hearing as we know it, but does provide help for patients willing to learn to hear again.

Hearing aids will be prescribed in most cases on the basis of a pure-tone audiogram alone, unless other pathologies such as Meniere's disease are indicated. Hearing aids are given only to patients who:

a) have a big enough loss to mean that hearing is markedly impaired,

and b) have enough residual hearing to make it of any benefit.

2.2 Audiometer.

The audiometer used for the work in this thesis was a Mercury ASRA 2000 Series audiometer.

ASRA - Automatic Self Recording Audiometer.

The ASRA comes in two parts:

a) The computer (an IBM PC compatible)

and b) The audiometer.

The basic hardware had been developed already but some modifications and developments were made during the course of the work to meet specialised requirements not previously catered for. These applications are discussed later.

Some control software to implement threshold audiometry on the PC had already been written before the work of this thesis began. The author, however, participated in substantial development of this software particularly making a large contribution in the implementation of the masking procedures described later in this chapter.

The two parts of the ASRA machine are linked via two cables, one cable from the parallel port of the computer which drives the colour graphics printer and the other from the computer's serial port which is linked through an RS232 chip to the microprocessor in the audiometer hardware (a Motorola MC68701). This microprocessor is the "brain" of the audiometer - it allows frequencies to be set, tone signals to be sent to left or right ear, and the intensity of the signal to be set via digital attenuators.

Apart from the power supplies and printer circuitry the audiometer contains only two circuit boards - one digital and the other analogue.

a.) The digital board contains: a crystal oscillator to define the rate at which a table of sine values, held in EPROM, is clocked through a digital to analogue converter (DAC). The patient response button circuitry is on this board and allows the microprocessor to detect when a response has been made by the patient.

b.) The analogue board contains: the DAC used to create the sine wave from the digital words produced on the other board. This sine wave is directed into an envelope generator to produce tone bursts with the correct rise and decay times. Alternatively a narrow band noise signal can be produced. There are three outputs from this board, two for driving the earphones and/or an insert masker and the third to drive the bone conductor. The decision on which one of these should be used is obtained from decoded signals on the digital board.

2.3 BSA Recommended Procedures.

The recommended procedures for pure tone audiometry as described by the BSA are contained in BSA (1981), BSA (1985), BSA (1986) and BSA (1989).

There are several criteria used in selecting the procedures, including the test's duration, its repeatability and the ease with which someone can learn to test.

2.3.1. Rules for Pure Tone Audiometry.

The pure tone test implemented follows the procedure for method A as described in BSA (1981) and BSA (1985).

The procedure states that the threshold should be taken as the lowest intensity at which the patient hears the tone for at least 50% of the presentations. The advised duration of the tone is between 1 and 3 seconds, but should not be less than 0.5 seconds. Rhythmic presentations are to be avoided as these can lead to 'lock in' conditions.

The test starts with a presentation at 1kHz of 60dB. On other frequencies the first presentation is made 30dB above the threshold on the nearest adjacent frequency. Thereafter, if the patient does not respond to that first tone, the level is increased in steps of 20dB until a response is obtained. The test then proceeds, the tones are presented, each subsequent step 10dB quieter until the patient no longer responds, they are then increased in intensity in steps of 5dB until a response is achieved. The patient's threshold is defined arbitrarily as the lowest level at which responses occur in at least half of the ascending trials (as long as there are at least two responses). Once the set of frequencies has been tested, a check should be made at 1kHz, and if this differs markedly from the previously found threshold another check should be made at 2kHz and so on until the thresholds agree again.

In the ASRA audiometer this checking is not installed automatically, but can be done easily using manual mode.

The unmasked bone conduction (BC) test is done in exactly the same way as the air conduction (AC) described above with the obvious replacement of the earphones by the bone conductor.

2.3.2 Masking Rules.

Where there is a hearing loss in only one ear or where one ear has very much worse hearing than the other, there is a high possibility that sound not heard in the bad ear will in fact, via interaural attenuation, be heard in the good ear. Masking of the good ear is required to prevent this happening.

The amount of masking necessary is quite critical for the detection of the true threshold. If it is too small, the cross-hearing will NOT be eliminated and, if it is too great, the test ear threshold will be raised. To find the correct level, masking charts are used, as described below.

There are three main rules used to determine when and where masking should be applied.

Rule 1 : In AC tests masking is required when the difference between the unmasked left and right AC thresholds is greater than the interaural attenuation for that frequency. The worse ear would then be used as the test ear and the better ear would be masked.

Rule 2 : In BC tests, masking is needed at any frequency where the unmasked BC threshold is better by 10dB or more, than the worse AC threshold. Again the better ear would be masked and the test ear would be the worse ear.

If the masked BC threshold does not shift by more than 10dB, it may also be necessary to mask the worse ear and test on the good ear.

Rule 3 : This rule again refers to AC tests. This rule is only applied where rule 1 has NOT been applied and where the masked BC threshold is better by the interaural attenuation or more, than the non-masked AC threshold attributed to the worse ear.

Before masking is attempted the threshold of masking noise, M, must be determined. Normally it would be expected that this mask threshold M would be approximately equal to the pure tone threshold of the ear used for masking. When the masking noise is presented through earphones, it may be that M is equal to the pure tone threshold on that frequency. This would be dependent on the calibration of the masking noise and should be checked. If insert maskers are used (i.e. the masking noise is presented to the ear via a tiny earpiece which fits in the ear rather like a hearing aid) the threshold of masking could easily be very different from that obtained for pure-tones in the earphones. A similar method to that used to measure the pure-tone thresholds was adopted when measuring M.

Once M had been determined, a masking chart was drawn to compute the masking function. A masking chart is shown in figure 2.1.

Several steps should be followed:

The tone should be presented again, this reminds patients what they are listening for. Masking noise is then introduced at a level M+10dB and the tone threshold re-established in the presence of this masking noise. This threshold value can then be plotted on the chart.

This procedure is continued, increasing the masking by 10dB each time until

either

i) three consecutive masking levels 10dB apart yield the same threshold to within 5dB.

or ii) the level has become uncomfortable to the patient.

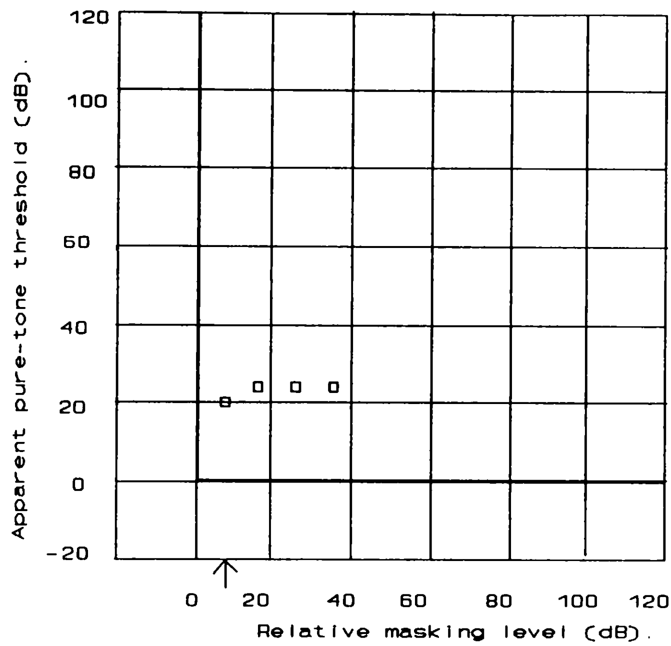


Figure 2.1 Plateau Masking Chart. The arrow represents the threshold of masking noise. In this case M=9dB.

or iii) the maximum output of the audiometer has been reached.

If no clear plateau is apparent on the chart, further thresholds should be determined at intermediate mask levels.

2.3.3 Interpretation of the masking function.

Once a masking chart has been plotted, a best fit masking function should be drawn for that chart, based on some idealised masking functions.

(i) No cross hearing present - this is when the original unmasked threshold is equal to the true threshold even though there was a risk of cross-hearing. See Figure 2.1.

(ii) Cross-hearing present - cross-hearing occurs when the non-masked threshold in the test ear actually originates from the better hearing of the non-test ear. It is sometimes called a "shadow point" of the non-test ear.

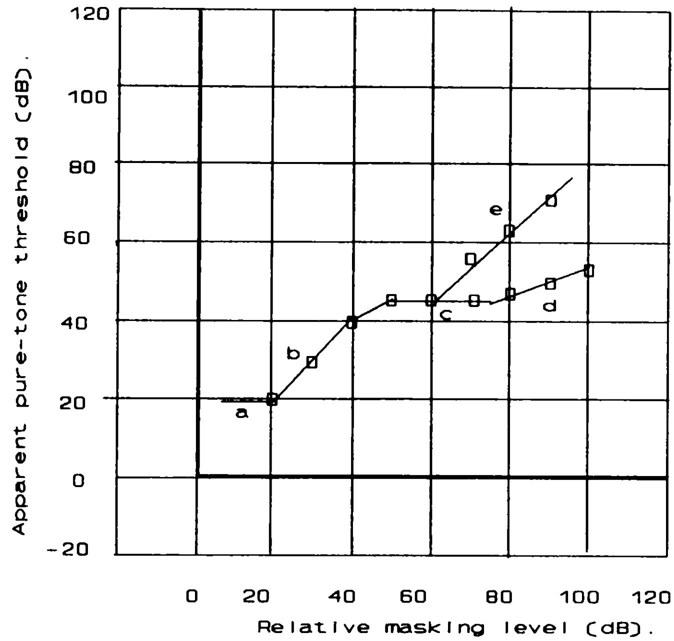


Figure 2.2 Masking Functions.
(a) region where tone and mask are heard, (b) peripheral masking region, (c) true threshold, (d) central masking effect and (e) peripheral masking.

Figure 2.2 shows typical masking functions and the following sections (a)-(e) describe the 5 different parts of the figure.

(a) Represents the conditions where both tone and masking noise are heard in the non-test ear, that is, the masking is not at a high enough intensity. This usually only happens at masking levels up to M+20dB.

(b) Represents the threshold of the non-test ear being raised by the presence of the masking noise. Again both tone and noise are heard by the non-test ear. This slope will be at an angle of 45° and this is called peripheral masking.

(c) This part of the masking function represents the true threshold of the test ear. The threshold of the non-test ear has been raised such that the tone is only heard in the test ear. It is this part of the function which gives rise to the name, plateau masking.

There are then two paths which the masking chart might take:

(d) This is caused by a central masking effect. The slope of this curve is less than 45° (usually 5° - 30°). The reason for this type of rise is an inability of the brain to distinguish sounds of very different intensities.

(e) A rise of 45° following a reduced or non-existent plateau (peripheral masking again). This happens when the masking becomes sufficiently loud to provide a masking effect in the test cochlea itself. A true threshold cannot therefore be found. This is known as cross-masking.

2.4 Calibration.

2.4.1 Why is calibration required?

Calibration is necessary to ensure that an audiometer produces a pure tone at a specified level and frequency. Calibration means that audiograms produced in two or three different hospitals each with its own calibrated audiometer are directly comparable. When an instrument is being calibrated it is normal to check that the signal is present only in the transducer to which it is directed and that the signal is free from distortion and unwanted noise.

The calibration of an audiometer must examine several features. Earphones, bone vibrators and loudspeakers have different characteristics and must be calibrated separately. Earphones for example are very specific and changing them affects the whole calibration of the machine. In general they are selected to give good long term stability and a flat frequency response.

The threshold in a human ear is not uniform. The ear is very much more sensitive at frequencies around 2kHz and insensitive at low and very high frequencies (see section 1.4.1). To measure the perceived intensity of a given sound, the response of the ear must be taken into account.

The calibration of the intensity of an audiometer is dependent on audiometric zero, that is, the sound pressure level at each frequency corresponding to the threshold of hearing for a pure-tone played through the earphones to be used. Audiometric zero is measured by taking a large sample of otologically normal subjects and measuring their threshold of hearing for the frequency and earphone concerned. The sound pressure level (SPL) from the earphone in a real ear is not measured, but the voltage applied to the earphone in order to produce that sound pressure is. From these voltages, equivalent to the thresholds of hearing, the modal value is taken and applied to the earphone when placed on the acoustic coupler. The sound pressure developed in the acoustic coupler is known as the Reference Equivalent Threshold SPL, that is RETSPL. The sound pressure level which is required for a subject to actually hear a certain hearing level is in fact this particular hearing level (HL) added to the corresponding RETSPL.

i.e. $SPL_{required} = (RETSPL)_{frequency} + dB\ HL_{required}$.

Table 2.1 shows RETSPL values for the TDH-39 headphone which was the one used in the work of this thesis. The coupler used was a 9A standard.

Bone conductors are very similar - but a mechanical coupler is required to measure the output of a bone vibrator. This coupler is sometimes called an artificial mastoid.

Harmonic distortion is another important test which must be made. Bone vibrators have particularly poor harmonic distortion characteristics and the implication for the listener is that, depending on his hearing loss, he may not hear the fundamental frequency but may hear a higher harmonic thus appearing to have better hearing than he actually has, at the test frequency. Table 2.2 shows the percentage harmonic distortion of the test tones in a typical ASRA machine.

Table 2.1 RETSPL for TDH-39 earphones.

Frequency (Hz)	RETSPL reference to 20μPa (dB)
125	45
250	25.5
500	11.5
1000	7
1500	6.5
2000	9
3000	10
4000	9.5
6000	15.5
8000	13

The intensity range -10dB to 120dB must be entirely 'linear' (i.e. it must follow the logarithmic scale accurately) otherwise hearing levels cannot be measured to any degree of accuracy using the currently employed techniques of measurement. In the ASRA machine the intensity scale is linear to within 0.5dB.

Finally the rise and decay times of the test tone are significant as they too influence detectability. If the rise or decay times are too short there is a subjective impression of a click. If these times are too long there may be frequency changes and distortion of the tone.

Table 2.2 Percentage Harmonic Distortion of the test tones in a typical ASRA audiometer.

Frequency (Hz)	Hearing Level (dB)	Percentage Distortion
125	70	0.2%
250	90	<0.1%
500	110	0.1%
750	110	0.2%
1000	110	0.1%
1500	110	0.1%
2000	110	0.1%
3000	110	<0.1%
4000	110	<0.1%
6000	110	0.1%
8000	110	0.2%

2.4.2 Calibration using the computer.

In standard manual audiometers, calibration is done using a large number of potentiometers, which must be individually adjusted to give the correct response on all frequencies to be tested. The advantage of computer control is that in the ASRA audiometer, the hardware response is made absolutely flat, that is, the pure tone signals are made uniform over all frequencies, and the calibration is done in software. Calibration constants are stored on a floppy disc file and loaded when the program is started.

This method has other advantages:

a) If the clinic has several pairs of earphones it will have a calibration file for each set of earphones. This avoids a time consuming re-calibration of the audiometer every time the earphones are changed.

b) A separate calibration file can be available for free field audiometry. This is a form of audiometry where loudspeakers are used instead of earphones. It is very often used in hearing aid evaluation as patients can be tested while wearing their aids. Usually if this is done regularly in clinics a separate audiometer is kept calibrated solely for this type of testing. With the ASRA machine however, it is simply another calibration data file which is required.

2.4.3 Masking Implementation.

A separate program is provided for calibration of the audiometer. The earphones must be placed on an artificial ear coupled to a sound level meter, as under normal calibration procedure. In the case of the bone conductor, it should be placed on an artificial mastoid.

The program produces a continuous tone, the intensity and frequency of which can be altered. The calibration constant is set by adjusting the up/down keys until the sound level meter is correct for that particular frequency and intensity. These levels are described in section 2.4.1.

As an example of this procedure we take the calibration of an earphone for 1kHz at 70dB.

-- 70dB hearing level is displayed on the screen of the calibration program. RETSPL tables for the human ear state that at 1kHz, the sound level from the earphones must be 77dB to produce a sensation of 70dB in the ear (see Table 2.1). The calibration constant must be adjusted until a reading of 77dB is obtained on the sound level meter. Once all the calibration constants have been obtained they must be saved to floppy disc. Saving the file of calibration constants creates a file with suffix '.cal', whose format is correct for the audiometer program to read.

2.5 Implementation.

The implementation of the basic test must follow the procedure laid out in the BSA recommended procedures (see section 2.3) and decisions on masking etc. will be made according to the rules laid out in the appropriate articles.

At every stage where one part of the test is over and another part is about to begin, the system will offer the operator a decision on what it thinks should be tested next. The operator may, at any stage intervene or disagree with the computer's decision. In most cases, however, the test will proceed unhindered. These options are included so that the operator is in FULL control ALL of the time.

A fully automated test was implemented following the BSA procedures. The test pauses at several prompts which allow the operator for example to replace the headphones with a bone vibrator or instruct the patient on masking procedures.

See figures 2.3,2.4,2.5 and 2.6.

Figure 2.3 shows an unmasked air conduction (AC) audiogram for a patient with a good hearing right ear and a left ear where the hearing deteriorates rapidly at high frequencies. This type of loss is commonly seen in older people.

Figure 2.4 shows the addition of the unmasked bone conduction (BC) thresholds. These BC thresholds follow the AC thresholds of the right ear quite closely. There are two possible reasons for this:

i) There is some blockage in the left ear which means the sound is not reaching the inner ear. This

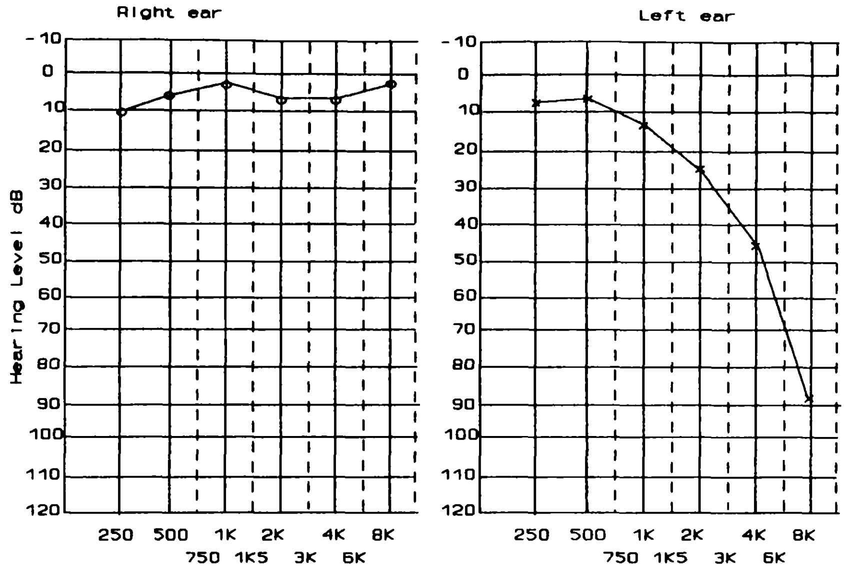


Figure 2.3 Audiogram.
Unmasked air conduction, left ear showing decline in hearing at high frequencies, right ear showing normal hearing.

is unlikely to occur only at high frequencies.

ii) These BC thresholds are due to the hearing in the right ear.

Figure 2.5 shows the intermediate process of masking for the 4kHz BC threshold. The initial mask value M is 6dB (the same as the pure-tone right AC threshold) and at this level the BC threshold is also about 6 - 7dB. As M is increased, the BC threshold is pushed up until it reaches 42dB where further increases in M cause no change to it. This is the true BC threshold. This can be seen clearly on the insert masking chart of figure 2.5. Figure 2.6 shows the finished audiogram.

To allow the operator full control, a manual mode was also implemented. The operator uses keys to select frequency and intensity and to present the tone to the earphones, in a similar way to a manual audiometer.

Within this manual mode, a reduced automatic facility is available. Once frequency and starting intensity have been selected an automatic threshold determination can be

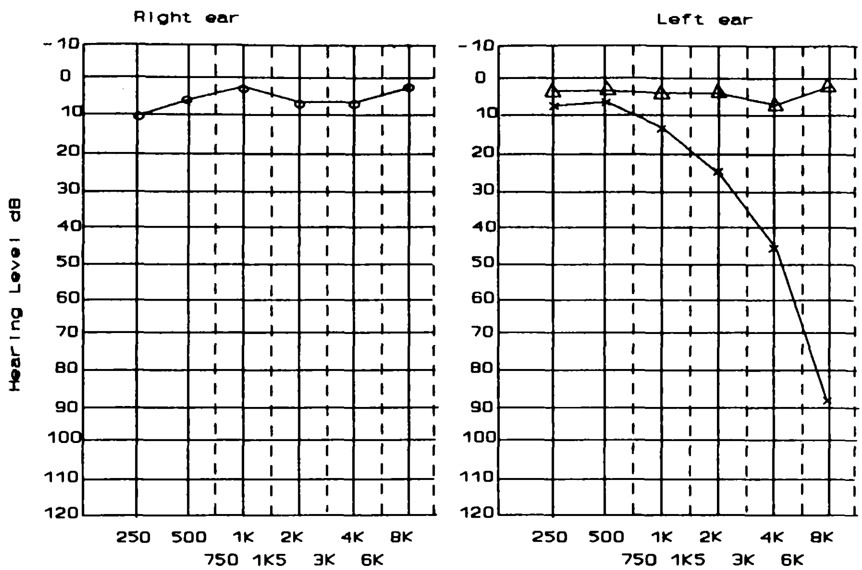


Figure 2.4 Audiogram - Unmasked Air and Bone Conduction. *The unmasked bone conduction thresholds may be associated with the thresholds of the right ear.*

conducted for the selected frequency.

Similarly a reduced manual facility is available in the automatic test for use with patients who get into difficulties.

2.5.1. Setup Menu.

Before a test can begin, several parameters have to be set up. In the main these values will not be changed from day to day save in exceptional circumstances. The default values of these parameters, therefore, are loaded from a floppy disc file menu1.mnu. It is possible to alter these parameters within the running of the program if necessary. Normally, however, operators will alter menu1.mnu to their own specifications and then load this unchanged every day, only thinking about alterations in unusual cases.

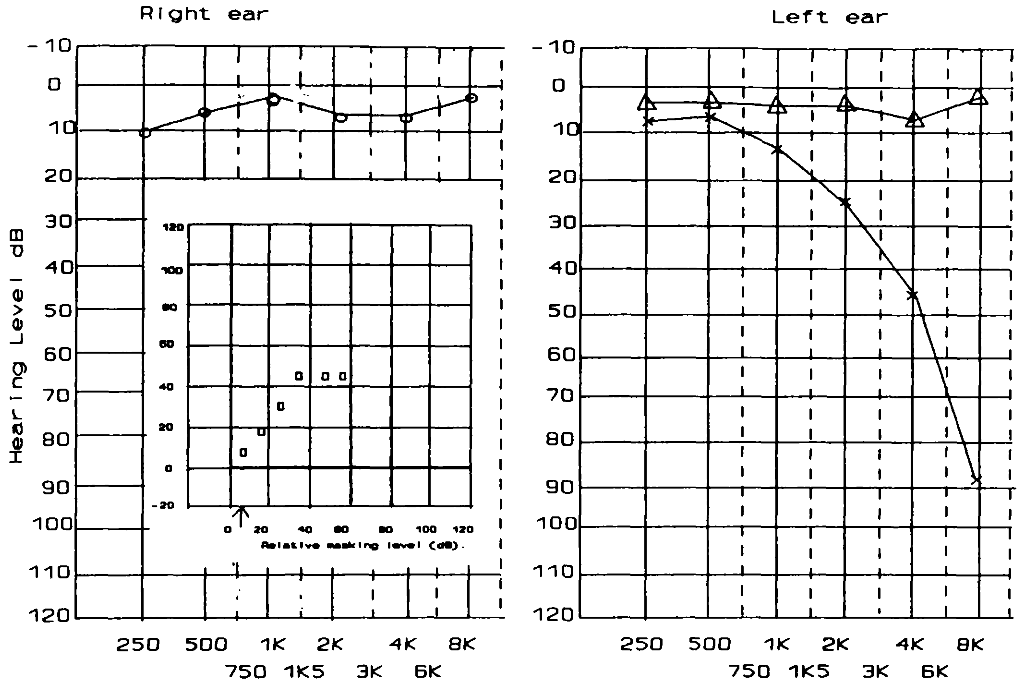


Figure 2.5 Bone Conduction Masking. *Plateau masking of bone conduction at 4kHz. The true bone conduction threshold at 4kHz on the right ear is 42dB. Insert masking chart like that shown in figure 2.1.*

There are twenty two of these parameters but not all are worth mentioning in detail:

a) Default response time. If there is no response from the patient within a certain time, the computer decides that the patient has not heard that particular tone. This parameter is set at a default value of 1500ms but has a range of 0 - 4000ms.

As a precaution, the possibility exists to increase or decrease this value during the test so that if a patient is clearly responding but always too late, the response time can be increased easily in steps of 100ms without having to come back to the setup menu.

Tone presentations are made at regular intervals, with minimum delay to allow the test to proceed as fast as possible. If, however, the operator wishes to avoid these rhythmic tone presentations, fearing that the patient may

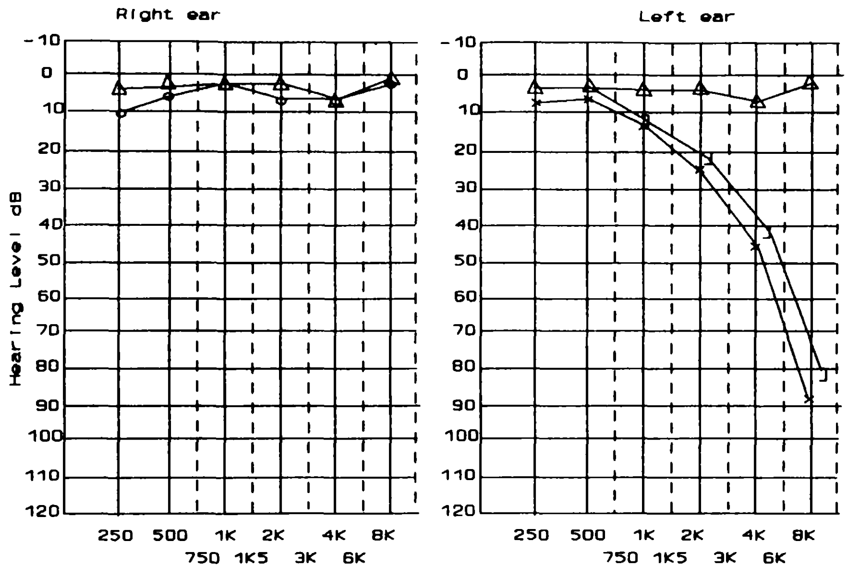


Figure 2.6 Finished Audiogram *showing a subject suffering from high frequency sensori-neural hearing loss in the left ear.*

lock in to the tones, an additional random delay of 0 to 2s can be introduced between tones. This facility is activated by using an odd number in the response time parameter. For example 1501 instead of 1500. This is a successful technique but it does slow the test down considerably.

b) Tone length. It has default of 1000ms and range 0-4000ms. As in (a) random tone lengths may be introduced by entering an odd value for this parameter and again it may be changed during testing.

c) The frequency order for testing AC and BC can be changed if necessary. In most hospitals the order of testing is :

AC : 1K, 2K, 4K, 8K, 500, 250 Hz

BC : 1K, 2K, 4K, 500, 250 Hz.

However this system allows easy change to other standard test orders, or the user may define his/her own order for the test.

d) The step size for both the scanning and random tests are two other variables. The default value is 5dB but any value in the range 1-10dB for the scanning test, and 1-20dB for the random test, can be chosen. This means that the thresholds can be measured more accurately than to 5dB if this is of benefit.

e) The number of scans which are done before accepting the threshold value in the scanning test is important. Again by changing this, a much more rigorous condition can be set to gain a more exactly defined threshold.

f) The system was implemented so that the scanning test (left and right AC) would continue uninterrupted if required. In some cases the operator may wish to reinstruct the patient, for example, to clarify that they will be testing the other ear. A pause is incorporated between these two steps to allow for this. The operator can set the length of this pause at any time between 1s and 30s.

g) The shift in BC on masking, below which the other ear will be tested, can be set from 5-10dB. It is normally set to 10dB to agree with BSA procedures.

h) The size and number of mask increments on a plateau masking chart can be defined.

i) The width of the noise spectrum can be chosen. Altering it however will require recalibration and should only be done if it is absolutely necessary.

j) A parameter can be added to the interaural attenuation during testing to determine if masking is necessary. Normally this will be zero but it is possible to add up to +/- 30dB.

k) Optional printing of the grid is available, when pre-printed grids are not in use.

l) The 'options' facility can be enabled or disabled from this menu as well as in other parts of the test. These options allow loading of an alternative calibration file, loading old threshold files and using either threshold or discomfort level symbols.

m) Different print colours for the grid, characters and symbols can be selected.

n) The masking noise can be changed from narrow band masking to white masking noise or to silence.

o) Thresholds can be taken on a scan upwards or a scan downwards (although only the scan downwards agrees with BSA procedures).

p) A display spot can be enabled to show in which ear the patient has heard the tone. This is done by detecting which of two response buttons was pressed and this can be of use in diagnosing certain pathologies.

2.6 Additional features during test.

Other useful facilities exist, for example being able to type in patient details which can then be stored on disk. The interesting feature of this is that these details (which can include up to 80 characters) will be on any subsequent printout of an audiogram.

Other facilities on offer during the actual running of the automatic test include -

a) the ability to move to either the next or the previous frequency in a test list,

b) to stop the operation of the test completely while the operator talks to the patient (perhaps encourages or reinstructs), and

c) to quit the current part of the test completely.

2.7 Implementation of masking charts.

The BSA recommended procedures for masking are followed in the implementation of computer control. The noise threshold is determined in the same way as the pure tone threshold. There is an option which allows the test to proceed without calculating the mask threshold and taking the pure-tone threshold as the mask threshold. This is not advised unless several other frequencies have been tested and there has been no difference between tone and mask thresholds.

A masking chart is then displayed on the screen (see figure 2.5) and the procedures followed, increasing mask level until either 3 consecutive thresholds agree within 5dB, or the maximum output of the audiometer is reached. The system will then prompt, and the operator may agree to finish or may continue with further mask levels.

As an optional short cut the mask level may be stepped up by 30dB from the mask threshold. The idea is that if the threshold has not changed then there is no need to go on as the plateau has been found. Great care must be taken when using this not to miss vital information.

2.8 Additional Features.

Other features which have also been made available include:-

a) A colour graphics printer which allows printouts. The printer used is an Epson BA-3740 and produces an audiogram similar to that shown in figure 2.3. This function will print whatever is on the screen at the time of calling. This could be an audiogram just accumulated or one loaded from floppy disc.

b) Audiograms can be saved and retrieved from floppy disk - this is particularly useful if a patient does part of an audiogram and at a future date, perhaps after the patient has had time to rest, the remainder of the audiogram is required. Pure-tone threshold files are saved with the suffix '.thr' to avoid confusion with other files which will be described later.

c) At any stage the audiogram can be redrawn on the screen.

d) Perhaps the most useful of all the functions is the facility to display and printout up to five audiograms on the same graph. This is done by assigning each audiogram with a number 1-5 corresponding to a block of memory. When that particular number is typed the corresponding audiogram will be displayed. When zero is typed all the stored audiograms 1-5 will be displayed simultaneously.

This has particular use when

i) comparing results over several monthly or yearly intervals.

ii) comparing results before and after surgery.

iii) looking for non organic hearing loss (NOHL) inconsistencies in several threshold results (see chapter 3).

iv) comparing Hewson Westlake results with Bekesy (see 2.9.1).

v) comparing results of free field audiometry with patients using different hearing aids.

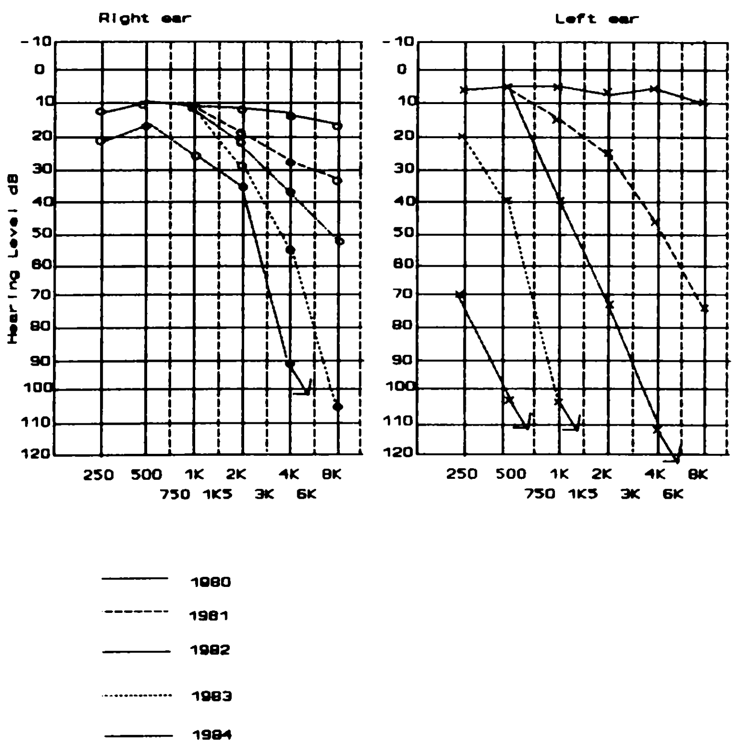


Figure 2.7 Decline in hearing over 5 years.
Shows the air conduction thresholds of a subject from 1980 to 1984.

An example is shown in Figure 2.7., displaying a marked decline in hearing over a period of 5 years.

2.9 Other Types of Audiometry.

The audiometry described so far has been standard Hewson-Westlake pure-tone audiometry, as recommended by BSA, but other methods of threshold audiometry are available.

2.9.1 Bekesy Audiometry.

Historically Bekesy audiometry was a technique which involved sweeping continuously from low frequency to high frequency. The patient was asked to hold a response button down while they heard the tone and release it when they stopped hearing the tone. In this way the frequency was swept slowly by the audiometer (in the horizontal plane of the audiogram) while the intensity was swept at a much faster rate by the patient (in the vertical plane of the audiogram). The Bekesy audiometer was in fact the first 'self-recording' audiometer available.

To allow maximum pathological determination, other slight variations on conventional Bekesy audiometry were introduced. The option to have an interrupted tone and the option of fixed frequency Bekesy were both introduced. The latter, as the name implies, is a Bekesy type test, not done over continuously swept frequencies but instead over several discrete frequencies. Again either continuous or interrupted signals can be used.

Bekesy identified four types of result and associated these results with certain pathologies. The details of these are not important but can be found in any textbook on audiometry e.g. Newby and Popelka 1985.

Bekesy audiometry has now been largely superseded although the price of Bekesy machines still makes them attractive in many industrial and screening applications.

The nature of tone production in the ASRA machine means that continuously swept tones are not feasible, at least not without altering the hardware for specific implementation of continuous frequency Bekesy. It was felt that there was not sufficient need for continuous frequency Bekesy to warrant this. Fixed frequency Bekesy, with either continuous or interrupted tones, was however introduced as this could be done without any hardware alterations. This type of Bekesy is used in testing for Non-Organic Hearing Loss (NOHL). By a comparison of the pure tone thresholds and the fixed frequency Bekesy levels differences may arise by virtue of the fact that they involve two different test strategies.

Figure 2.8 shows a fixed frequency Bekesy audiogram. Bekesy files are stored on floppy disk with the suffix '.bek'.

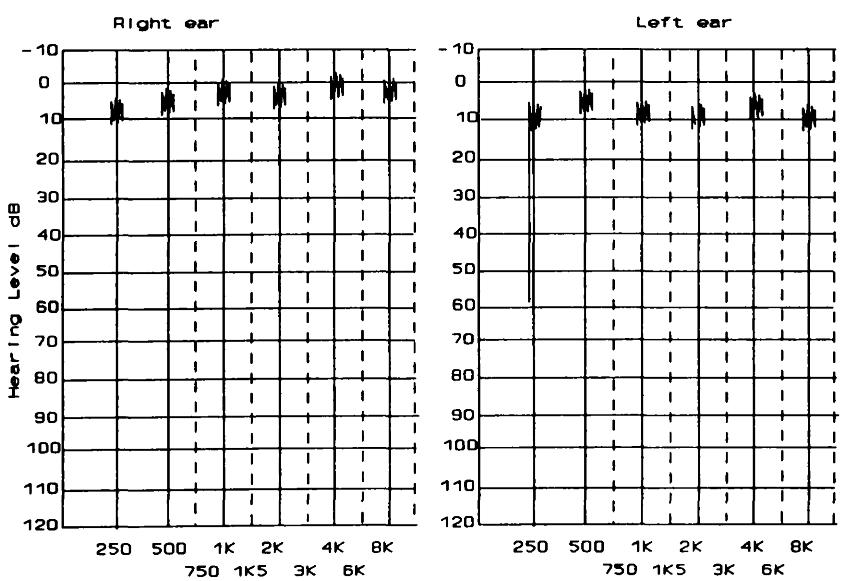


Figure 2.8 Fixed frequency Bekesy Audiogram. *Showing Normal hearing.*

(Note: Each of the Bekesy tests are done at one fixed frequency. The spread in frequency shown is to show the variation in intensity during each test.)

2.9.2 Random Testing.

As mentioned above, the type of Bekesy audiometry which has been implemented here is used predominantly for the determination of NOHL. The Random test was also implemented for use in detection of NOHL. This test will be described in much more detail in chapter 3 but, suffice to say, it tests a range of 3 frequencies and 9 intensities around the thresholds found by pure-tone audiometry. In this block of points, the tones are presented randomly in ear, in frequency and in intensity, and a measure is made of how good an agreement there is between the pure-tone and the random thresholds. This test is powerful, and is particularly suitable for a computer controlled audiometer since it would be very difficult for a manual operator to chart the results quickly enough.

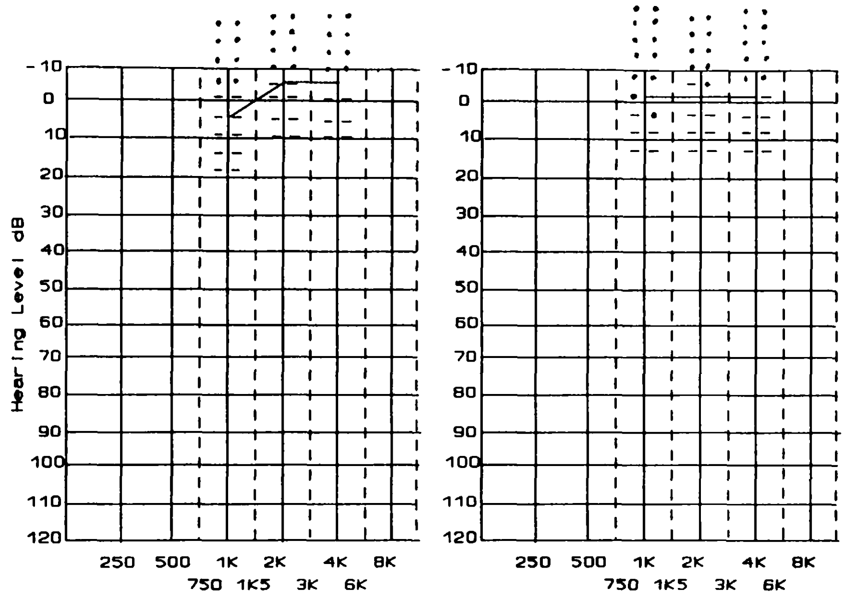
Test results will be discussed in the next chapter. These look hopeful, but it would be very interesting to see the results of a clinical trial comparing directly, the fixed frequency Bekesy results on NOHL, to those of the Random test, since Bekesy is accepted in legal courts when used in Industrial hearing loss benefit cases.

Figure 2.9 shows a random test audiogram. Files saved to disk have the suffix '.ran'.

2.9.3 Stenger Test - Randomised.

This test is another used in the detection of NOHL. It is very powerful but is only of use when the patient is displaying monaural hearing loss. The principle behind it is that when both ears are stimulated by a tone of the same frequency but different intensity, patients will hear a tone only in the ear in which it is perceived to be loudest (see section 1.4.3).

Let us consider a patient who, at 1 kHz, has a threshold of 5dB in the right ear and one of 50dB in the left ear.



Difference	-5	0	5	8	-2	0
Deviation	0	0	0	2	2	0

Figure 2.9 Random Audiogram. Shows good agreement between scanned (joined by solid lines) and random thresholds of a normal hearing subject. [- = tone has been heard; . = tones has **not** been heard.]
(Note: The dots above the graph do not signify intensity levels quieter than -10dB but show the number of presentations at -10dB.)

The operator has reason to believe that the threshold in the left ear is better than 50dB and that the hearing loss is being exaggerated. Suppose the true threshold on the left ear is 25dB.

Firstly, a tone should be presented above threshold on the right ear (i.e. about 15dB). The patient has already admitted a threshold of 5dB in that ear and so responds. Next, keeping the tone at 15dB in the right ear, another tone of the same frequency is added to the left ear, firstly at low intensity. The patient hears the tone only in the right ear, and responds. The intensity of this second tone is now gradually increased until it is above 35dB (25+10 dB). The patient will now hear this in precedence to the 15dB and hears it as below the 50dB threshold which he/she admitted. The patient will not respond while the tone is heard in the left ear, until it reaches the 50dB level admitted.

Because the tone is still being presented supra-threshold in the good ear the patient MUST hear something in that ear, so their failure to respond suggests NOHL.

In the ASRA machine the Stenger test was implemented, with the addition of randomisation as in the random test. Tones are presented either singly to each ear or two tones simultaneously to both ears. The patient therefore has no idea what combination is being presented.

To allow the test to be performed in a reasonable length of time, the randomised Stenger test is only done at 3 frequencies 1k, 2k, and 3k, or 1k, 2k and 4k, or 3 frequencies defined by the operator.

A pure-tone audiogram must have been done before the Stenger test can be carried out and if the frequencies to be tested have no threshold values, the computer will NOT allow the test to proceed. The default value of step size is 5dB but any value between 1dB and 20dB may be chosen.

Tones are presented at the following intensities:

i) On the better ear, at intensity steps ranging around threshold (for each of the three frequencies) from -2 steps to +3 steps.

ii) On the worse ear at intensity steps around threshold of -2 to +1.

and iii) On the worse ear at intensity steps ranging around the thresholds on the worse ear of -3 steps to -10 steps (50dB below threshold) in conjunction with a tone of +2 steps at the same frequency in the better ear.

The test sequence is totally random in frequency, intensity and ear.

Figures 2.10 and 2.11 show one Stenger test where the patient is telling the truth and one where the patient has failed the test because he is exaggerating hearing loss. Stenger files are stored on floppy with the suffix '.ste'.

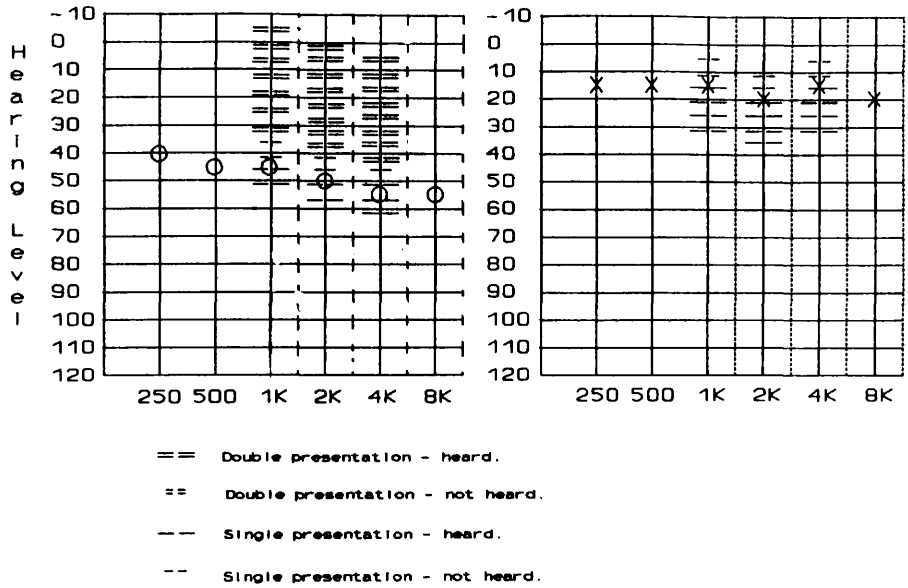


Figure 2.10 A Stenger Test. The patient has been truthful about his/her thresholds.

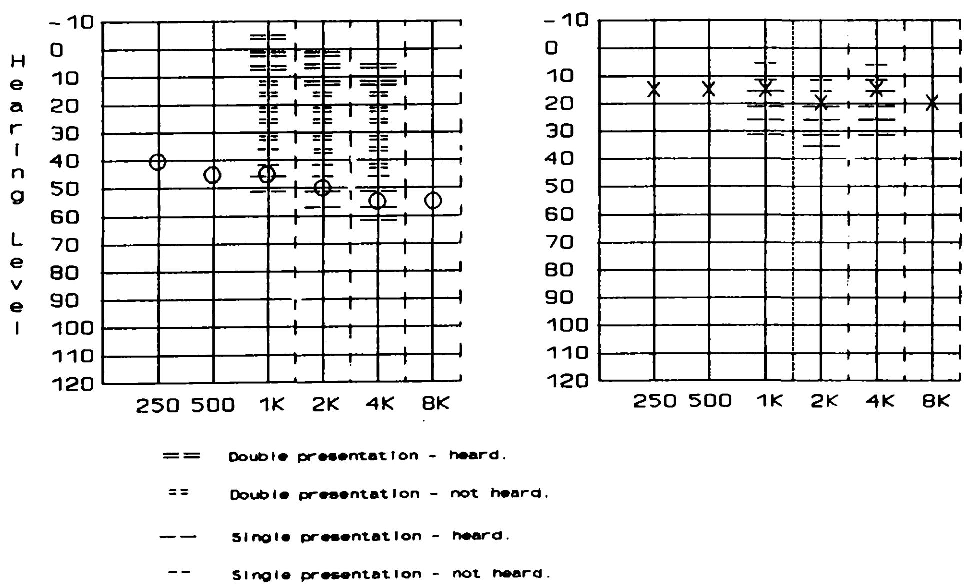


Figure 2.11 A Stenger test. The patient has exaggerated his/her thresholds in the right ear.

In figure 2.10 the patient hears all the combined presentations because they include a tone which is above threshold in the good (left) ear. In figure 2.11 the patient admits to hearing the combined presentations when they appear to come only from the left ear. When the presentations sound as if they are from the right ear, (i.e. the tone presented to the right ear is subjectively louder than that presented to the left ear) he does not admit to hearing either tone and is caught out since he must at least hear the presentations on the left ear which are above threshold.

2.10 Conclusions.

The standard pure-tone clinical test has been implemented satisfactorily on the ASRA computer controlled audiometer. Certain advantages exist. In particular the complicated decisions on what masking procedure should be followed are taken by the computer limiting operator error. The operator, however, is always consulted before important decisions are carried out - allowing total flexibility.

One of the major disadvantages encountered was the time the test took. In clinical trials, it was felt that a good manual operator could perform a test faster than the computer. Because of this several modifications were made to speed up the computer test and make it comparable with a manual test.

The major change in the implementation was to make the randomisation in time optional. By removing this randomisation in time, the test is considerably speeded up. The worry about this is that the patient will lock in to the tones and create better thresholds than are true. There is no evidence to suggest that this happens; however, by making it an option each operator can decide for him/her self.

In every other way the computer system at least matches a manual test and the computer can offer very much more: storing and displaying old files alongside new ones for comparison, maximum flexibility, and allowing the operator to observe the test as it continues, perhaps picking out small discrepancies he/she might otherwise have missed.

The other great advantage to this system above a manual machine is the random tests for use in the detection of NOHL. These tests can be done on the currently used manual machines but the charting of them makes them almost impossible for one operator to attempt.

CHAPTER 3: Use of the Random Test in the detection of Non Organic Hearing Loss

3.1 Introduction.

One problem in hearing clinics is that of detecting non-organic hearing loss (NOHL) among patients intent on being awarded Industrial Hearing Loss benefit. Although non-organic hearing loss can be a psychogenic complaint, in this context we are referring specifically to subjects who are exaggerating their hearing loss in order to deceive. Until now non-organic hearing loss has been detected using the skill and expertise of the audiological technician in his/her detection of small discrepancies during the hearing test and any subsequent test (Aplin and Kane 1985). The definitive test used at the moment for the detection and quantification of non-organic hearing loss is Electric Response Audiometry, however this is time consuming and not always readily available.

With the introduction of Computer Controlled Audiometry it has become possible to measure quantities otherwise inaccessible to the technician and to use these to add further weight to any suspicion they might have regarding Non Organic Hearing Loss (NOHL).

Response time is something which technicians are generally unable to measure accurately, but with computer control, where the on/off state of the stimulus is microprocessor controlled, it is easy to measure the time accurately from the onset of the stimulus until the patient presses the response button.

Another feature which computer control can offer is that of testing using random tone presentations. In this application, as mentioned in Chapter 2, tone presentation is randomised in ear, in intensity and in frequency, thus allowing an accurate measure of the level of agreement between a normal scanned audiogram, (following BSA Recommended procedure) and a 'Random audiogram'. This agreement may be significantly worse in the presence of non-organic hearing loss. Further it would be expected that the random threshold would be less well defined in the presence of non-organic hearing loss.

In this case these random tests were carried out on a number of relatively normal-hearing subjects first displaying non-organic hearing loss and then responding truthfully. Several quantities were measured and the results then compared to find the quantity best used in the detection of non-organic hearing loss.

3.2 Methods.

3.2.1 Subjects.

Nineteen subjects were chosen having a mean age of 28, ranging from 20 to 54 years. Most individuals had normal or near-normal hearing and all found it easy to understand and carry out the instructions.

Tests were also carried out on eighteen patients during hearing aid clinics at Gartnavel General Hospital, Glasgow. These patients ranged in hearing loss, age and ability and none was suspected of non-organic hearing loss.

These subjects were used to determine how successful the tests would have been in detecting NOHL among a clinical population.

3.2.2 Random Test.

The random test is somewhat misnamed, in this context, if one considers the truly mathematical definition of random. For the purposes here, however, the test can be thought of as random.

The thresholds of each of the subjects must first be measured, using the standard puretone audiogram determined with the Hughson-Westlake procedures as recommended by BSA (see Chapter 2). Once these are known, tone presentations are made at 5 dB intervals over the range from threshold+15dB to threshold-25dB for 3 frequencies on each of the ears. The three frequencies used in this application were 1, 2 and 4 kHz.

Using the random number generator in the computer, points in the above range are selected for test. However, once two tests have been performed on any one point, no further tests are permitted on that point. In this way, tone presentations are continued until 2 tests have been performed on each of the 54 points in the above range. The random test thus gives 108 tone presentations of random ear, frequency and intensity.

3.2.3 Instruction to subjects and Test procedure.

Each subject undertook two distinct tests, each test having two parts:

a) Firstly, the subject was given the standard BSA recommended instructions for pure tone audiometry. In addition, they were asked to attempt to qualify for industrial hearing loss benefit. They were informed that in order to do this they must have a hearing loss of at least 50dB at 1, 2 and 3kHz. They were further informed that the first test would start at 1kHz in the left ear. Each subject was warned against making the tone so loud that the test became uncomfortable. Further, the operator would not allow the system to present a sound level above 100dB.

A standard Hughson-Westlake pure tone audiogram was run on each subject who was then told that the tone presentations would now be randomised in ear, frequency and intensity and that they were still trying to qualify for benefit. A random test was then performed.

These constituted the set of data with non-organic hearing loss.

b) Secondly the same procedure was followed, but this time the subjects were asked to be as truthful as possible.

These constituted the true set of data.

This technique, using normal hearing subjects to simulate hearing loss, has been shown to give valid information (for example, Aplin and Kane, 1985).

Obviously only part (b) was used with the patients from Gartnavel.

A full test - i.e. a scanned threshold test followed by a random test, for a reasonably competent patient takes of the order of, 5-7 minutes.

3.2.4 Equipment.

The subjects were seated in a quiet room. All tones were presented through a Mercury ASRA 2000 Series audiometer.

The tones were presented to the subject using TDH-39P earphones and the subject was given a response push button.

All results were stored on floppy disc from an IBM PC compatible computer (AMSTRAD 1512) for later analysis.

3.2.5 Measurements taken.

Four of the measurements were used in the subsequent analysis:

i) **Deviation.**

As the subject responds to the random test a picture is built up of how consistently the test is being carried out.

If the responses show significant inconsistencies, for example, the subject hears 1 tone presentation at say 30dB, none at 25dB and 2 tone presentations at 20dB, then this quantity will be high.

The *'deviation'* will be zero if all the tone presentations louder than and including a certain level are heard but none quieter is heard.

In fact *'deviation'* is calculated by taking the standard deviation of the distribution formed from the differences in responses between consecutive intensity levels.

It would seem probable then that subjects who are displaying non-organic hearing loss will give higher values of deviation. They are less consistent.

This test will be referred to in the text as *'deviation'* and in figures and tables as *'dev'* or simply *'d'*.

ii) **Difference.**

If a subject is being truthful about his/her thresholds then the random test should reproduce the same pure-tone thresholds as the scanned test (see section 3.3). This applies, of course, to normal subjects with no other relevant medical problem such as dementia.

The value of *'difference'* is the shift in threshold between the Hughson-Westlake scanned threshold and the threshold measured by the random test. It is expected that this value will be larger in the presence of non-organic hearing loss, and approach zero for subjects who respond truthfully.

In the text this test is known as *'difference'*, also known as *'diff'* or simply *'df'*.

iii) **Average response time**

'Response time' is measured for each press of the push-button, indicating that a tone has been heard. For each frequency an average response time is calculated.

Since the same subjects are used to produce both sets of data, it is of interest to see if there is a marked difference in response time for the two cases.

This test is known as *'response time'*, *'resp'* or *'r'*.

iv) **Standard deviation (time)**

Here a standard deviation of response time is calculated for each frequency. Again if there is a wide range of response times the deviation will be large. However, if in general, response times are very similar, then deviation will be small.

This test is referred to as *'standard deviation (time)'*, *'stime'* or *'s'*.

Since the same subjects were used to collect both non-organic hearing loss data and true data, it is possible to compare all four sets of data and see which ones allow maximum differentiation between true and falsified results.

It has been well documented that response time, in general, increases as threshold is approached (Wright et al. 1981) and it is of interest to see if this trend is also the case in the presence of NOHL. The results showed that some patients appeared to show less increase near their exaggerated thresholds but there was no obvious trend.

3.3 Results.

As described in section 3.2.1 the patients used as the reference in this work were, in the main, normal hearing. They were asked to feign a degree of hearing loss in order to qualify for Industrial Hearing Loss Benefit. It should be noted that, in general, patients who exaggerate their thresholds in this way do have some degree of hearing loss already and so may only have to exaggerate by about 20dB to reach the 50dB limit.

Table 3.1 shows the average of the truthful thresholds of hearing for each patient and the corresponding average of the exaggerated thresholds thus giving the degree to which each patient exaggerated. It can be seen that, in the main, this 'amount of exaggeration' is relatively high and this should be taken into account when drawing conclusions.

Table 3.1 Amount of Exaggeration in Thresholds.

Average Truthful Threshold (dB)	Average Exaggerated Threshold (dB)	Amount of Exaggeration (dB)
0.5	88	88
10	52	42
9	56	47
-2	73	75
3	85	82
5	83	78
5	59	54
-1	77	78
-2	81	83
5	80	75
-2	69	71
1	41	40
34	83	49
7	62	55
-4	90	94
1	36	35
14	51	37
10	62	52
0	95	95

For a "good" test the probability distribution curves of the two sets of data should show a high degree of separation.

In this case true data and NOHL data should be well separated for best results. See figures 3.1, 3.2, 3.3 and 3.4 for these probability distribution curves corresponding to *'deviation'*, *'difference'*, *'average response time'* and *'standard deviation (time)'* respectively.

Figures 3.1 and 3.3 show the best separation. Figure 3.1 indicates that if the *'deviation'* is greater than or equal to 5dB it is very probable that the subject is displaying non-organic hearing loss. In figure 3.3 the area of *'response time'* greater than or equal to 550ms contains most of the non-organic data and the area less than 550ms contains most of the true data.

Figures 3.2 and 3.4 do not show such marked separation of the groups. In the case of figure 3.2 it would seem that most of the true data has a *'difference'* which has a zero or small negative value, while the non-organic data spreads over a wide range, with a tendency to positive values. This, in fact, is quite reasonable since a value greater than zero means that a subject has made his/her threshold louder. It is possible that in the non-organic population the random test encourages these subjects to exaggerate their threshold so as not to be caught out. It would also seem to be the case that because of the way the scanned (Hughson-Westlake) threshold is done, a patient is not sure if he/she has heard the tone if it is very close to threshold, so does not respond. They will respond to the next tone because it is 5dB louder but their Hughson-Westlake threshold will be at this louder level. In the random test because patients have no idea where the next tone will be, they concentrate more, and respond to tones which they might otherwise have missed. This may be why a substantial number of the truthful subjects have a *'difference'* of between -2 and -5dB.

It should be noted that the normal-hearing patients tested here did reproduce their Hughson-Westlake thresholds from the random test to within less than 5dB. This means that the random test is a valid test of threshold for this type of subject.

In figure 3.4 most of the non-organic data has *'standard deviation (time)'* greater than or equal to 150, while most of the true data has a value less than 150. This graph, however, of the four clearly shows least separation between NOHL and truthful data.

The results from Gartnavel hospital were used as a control set of data to check that typical hospital patients could perform the random test. Of the patients tested, 80% had no real problem with the test. Those who did were either very elderly or had difficulty in concentrating. Of these, none was in a category which would have suggested non-organic hearing loss.

There is some concern that the age of the patient may itself have a significant bearing on the measurements of *'average response time'* and *'standard deviation (time)'*. It might be expected that both of these quantities would increase with an increase in the age of the patient.

Using the data presented in this chapter, investigations show a fairly even spread of 'average response time' and 'standard deviation (time)' with age (see figures 3.5 and 3.6). A wider spread of ages may have been of benefit but it would seem that there is much less of a correlation here than was anticipated.

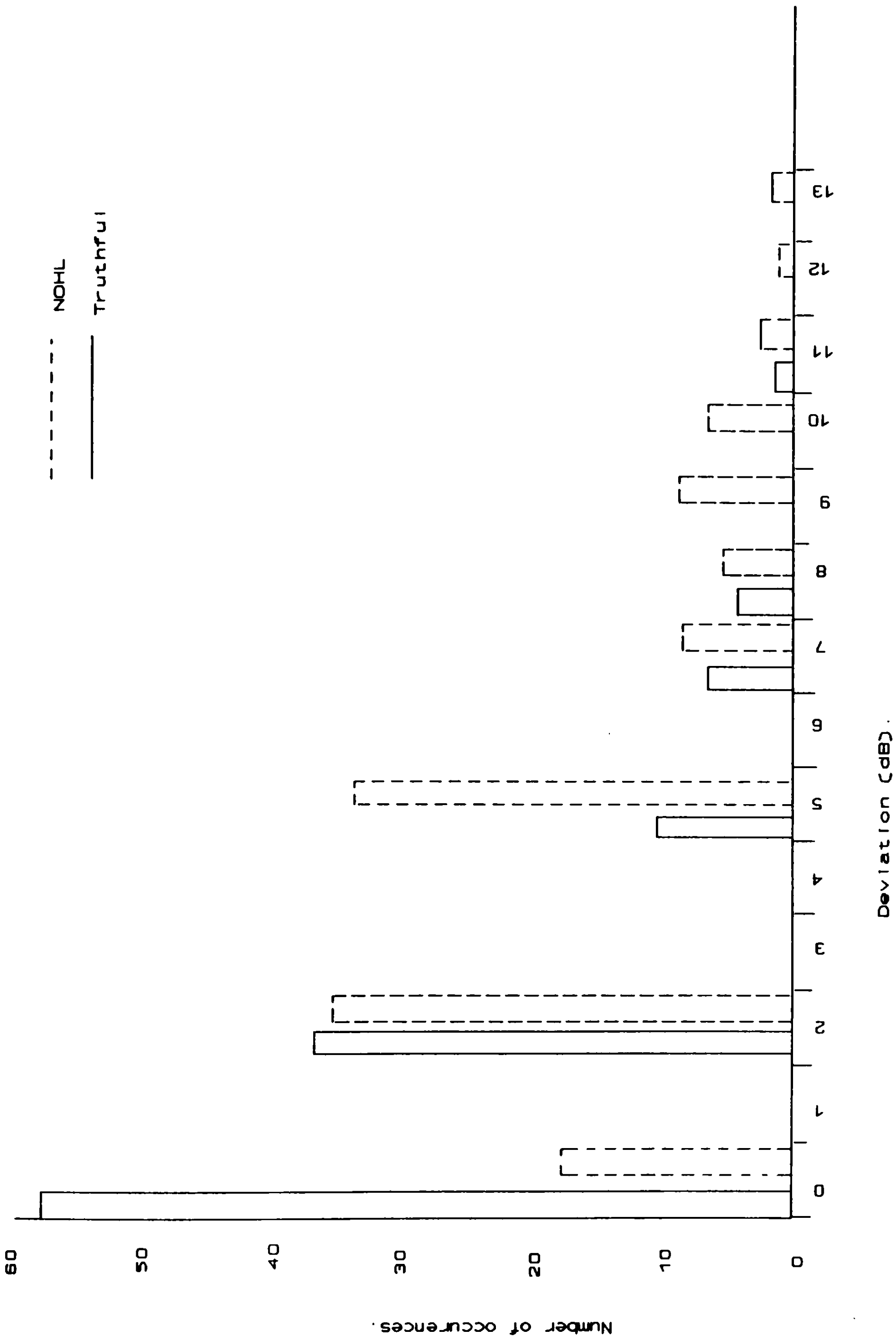


Figure 3.1 Deviation.
A measure of how self-consistent the random thresholds are.

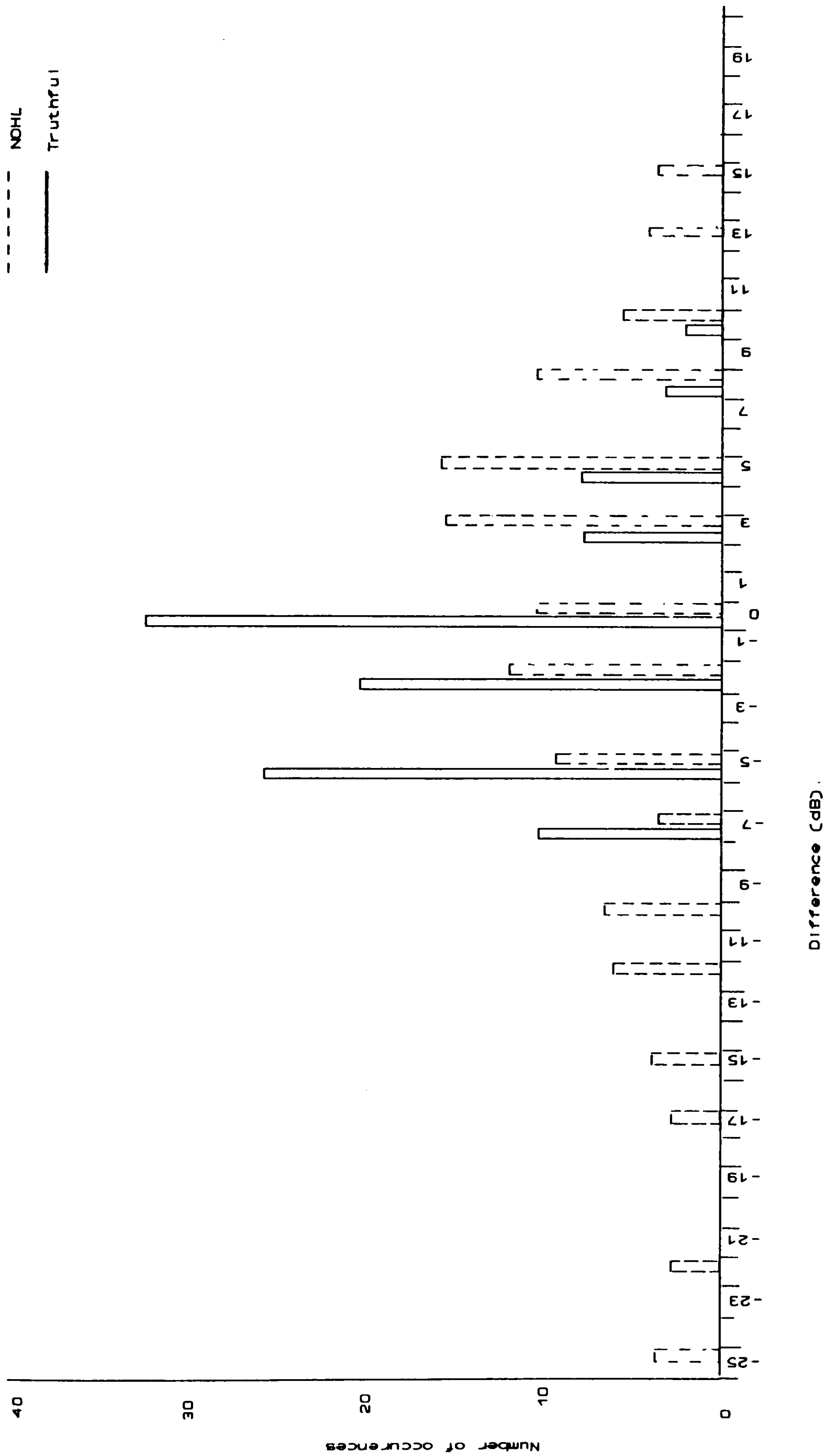


Figure 3.2 Difference.
The shift in threshold between random and scanned tests.

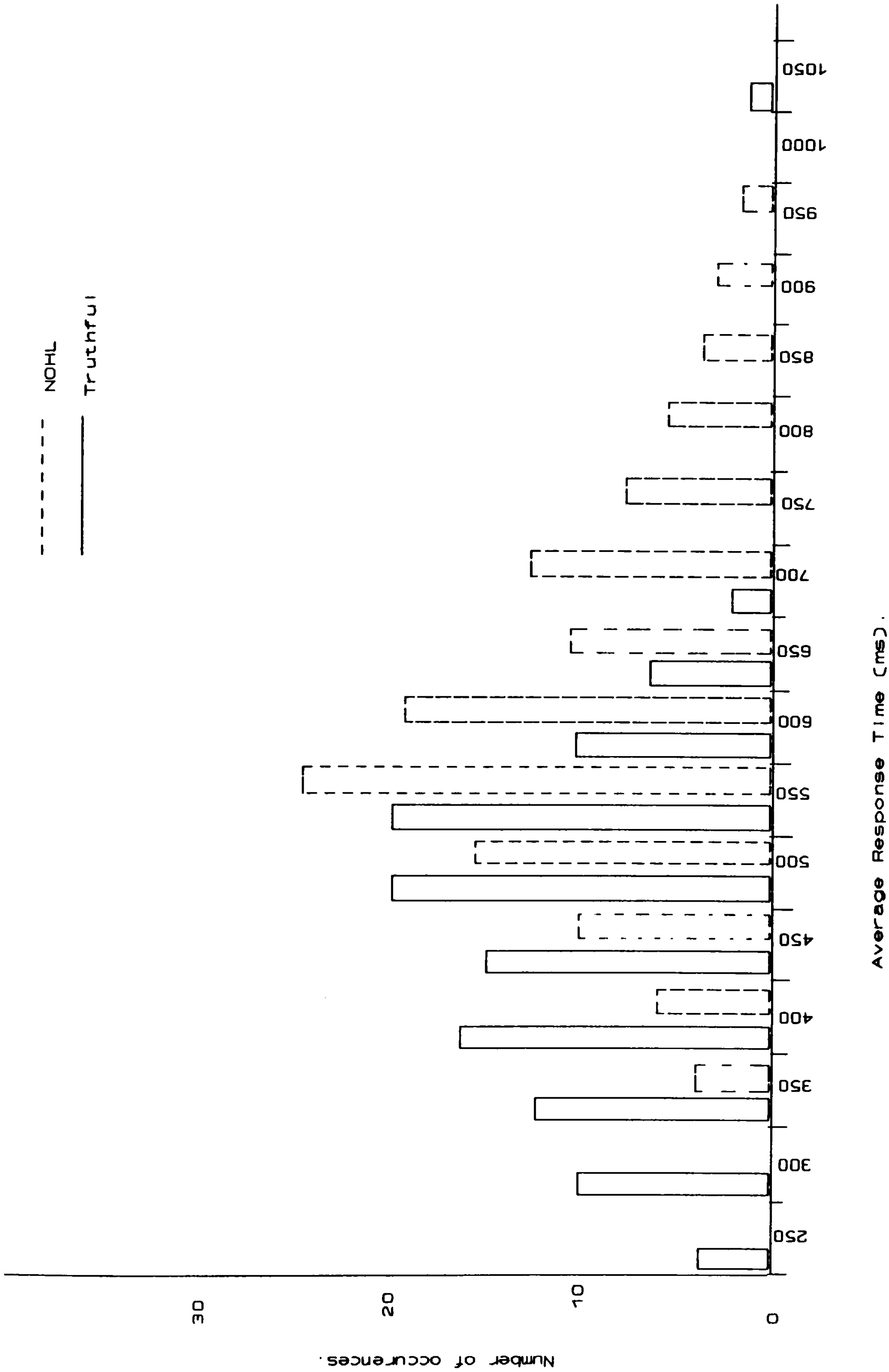


Figure 3.3 Average Response Time.

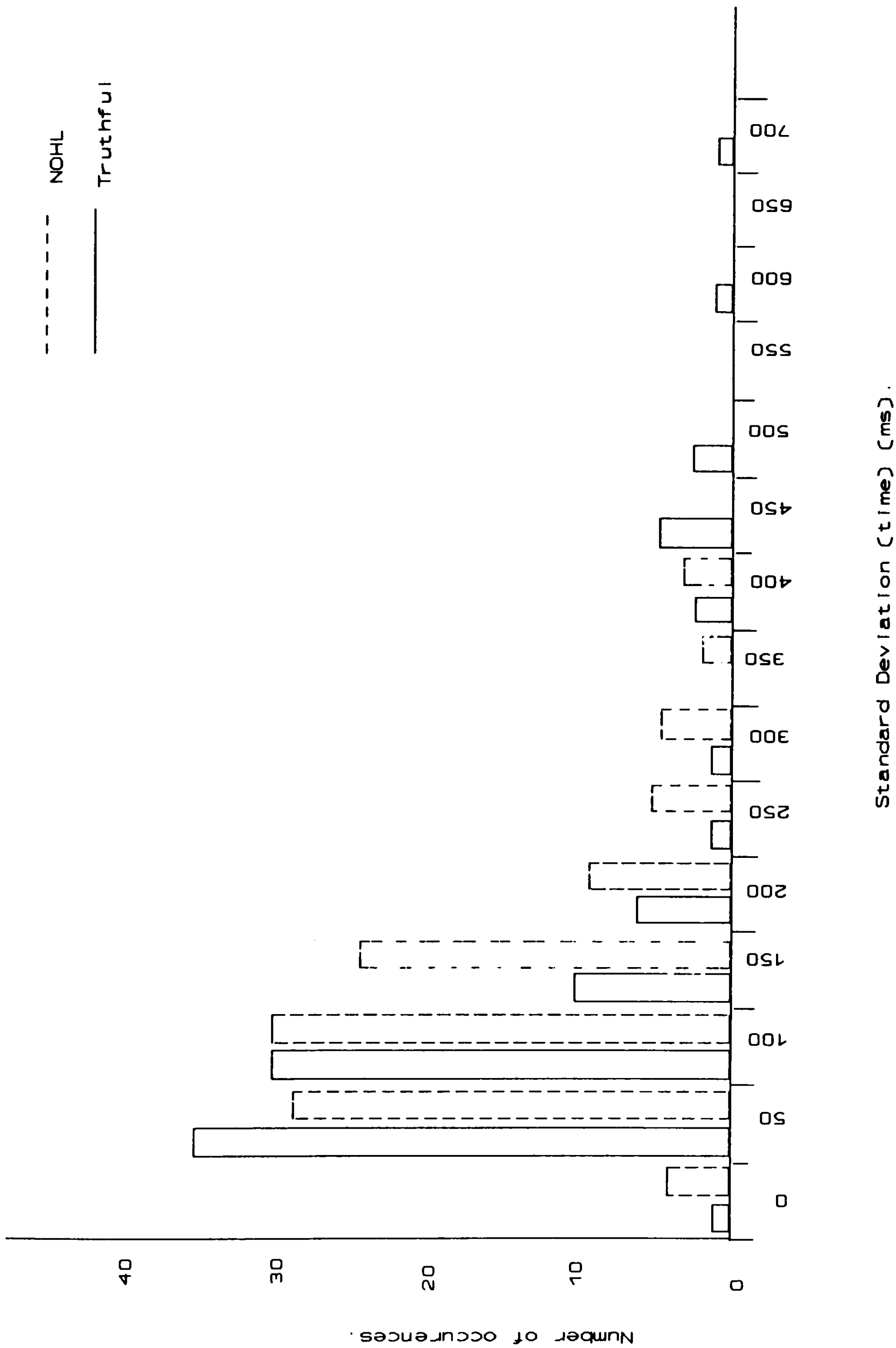


Figure 3.4 Standard Deviation in the response time.

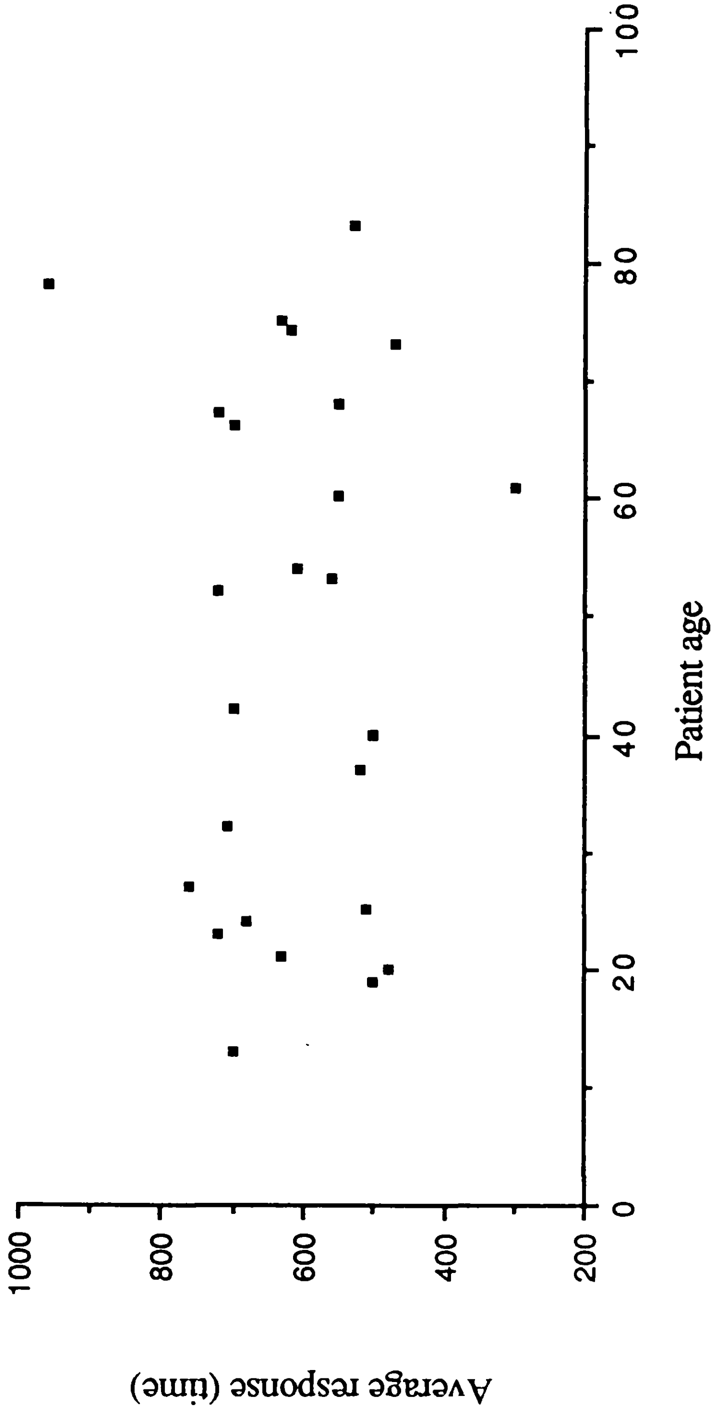


Figure 3.5 Age versus Average Response Time.

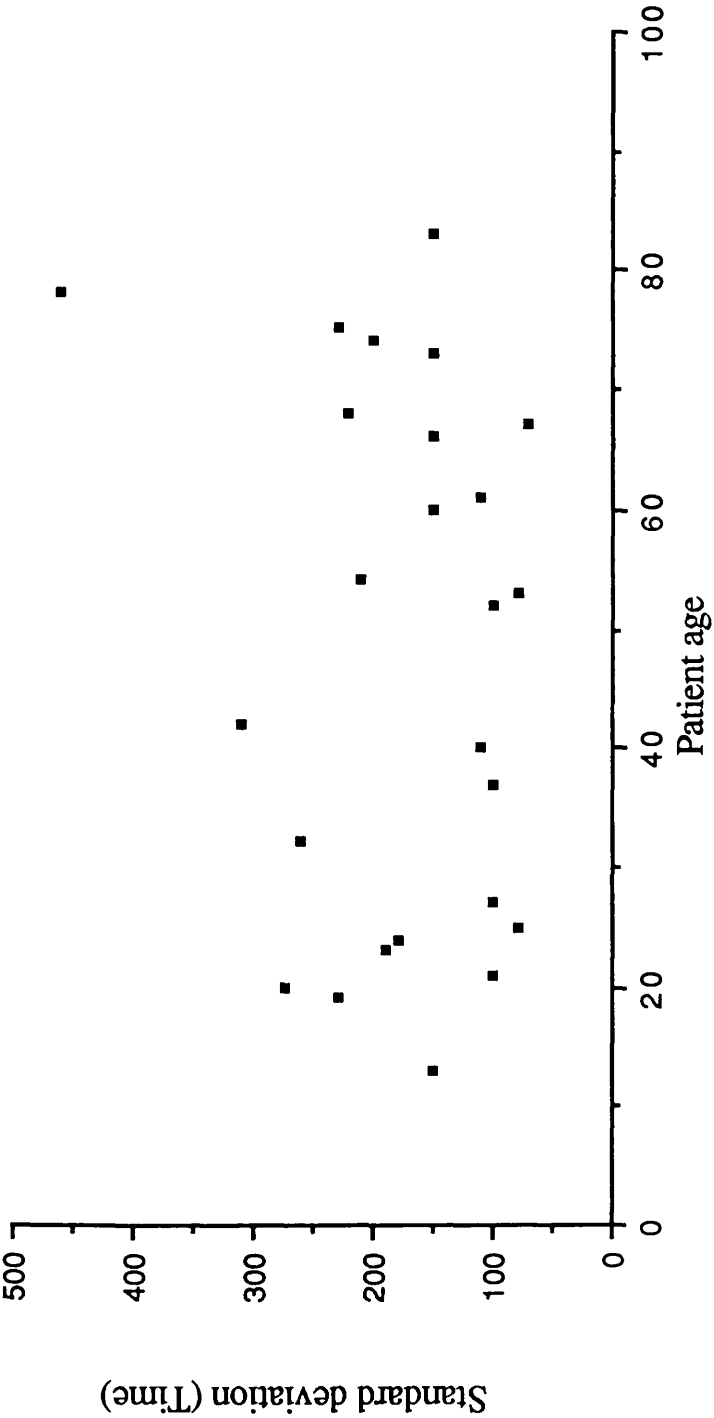


Figure 3.6 Age versus Standard Deviation (time).

3.4 Analysis.

There are several ways of considering the data and its significance.

Turner et al. (1984) in their series of articles on clinical decision making analysis used several indices in their detection of how "good" an audiological test was.

In particular Turner & Neilsen (1984) discuss the use of probability distribution curves used to determine Receiver Operator Characteristic (ROC) curves and an index referred to as hit rate/false alarm rate, or HT/FA formed from a 2*2 matrix. The ideal test, would have HT=100% and FA=0% and so any test aims to maximise HT and simultaneously minimise FA.

Other indices of value are sensitivity - another name for hit rate, and specificity - or true negative rate. These two looked at together give the value of HT/FA. Predictive value can give useful information and is equal to HT/(HT+FA). Finally the index d' gives probably the best overall performance score of a test, as it takes into account HT and FA and their relative values (Swets 1964).

Appendix A gives more detail about these techniques and indices and from where they are derived.

It was decided that the ROC curves and the 5 indices mentioned above, would give the most information and were used in this application. These were used on the four individual measures described. It is true, however, that higher test results can be obtained by combining the results of two or more tests - this is sometimes referred to as using test protocols.

Another statistical method of separating groups of data is discriminant analysis.

All these methods of analysis were used and their results compared.

3.4.1 Individual Tests.

Using these probability distribution curves figures 3.1 - 3.4, 2*2 decision matrices were formed for all four tests under different conditions (see appendix A). For example, in figure 3.1 the break off point, as described, is at *dev*=5. A decision matrix was calculated for this cut-off and for *dev*=2, thus allowing comparison.

Table 3.2 shows the results of these decision matrices.

Table 3.2 Individual Test Results.

Test & Conditions	HT/FA	Sens.	Spec.	Pred. Value.	d'
Deviation					
dev>=5	31/5	31%	94%	86%	1.14
dev>=2	84/31	84%	68%	73%	1.50
Difference					
diff>0	73/31	73%	68%	70%	1.12
Average Response Time					
resp>=500	78/47	78%	52%	62%	0.84
resp>=550	68/21	68%	78%	76%	1.28
resp>=600	52/10	52%	89%	83%	1.33
Standard Deviation					
stime>=150	57/36	57%	63%	61%	0.54
stime>=100	94/68	94%	31%	58%	1.08

The 19 points considered in Table 3.2 are derived by taking the average of the 6 points obtained from each patient (2 ears x 3 frequencies). It can be seen that, by reducing the condition from *dev>=5* to *dev>=2*, HT has been increased. However inevitably FA has also been increased and the resulting value of d' remains relatively unchanged.

Table 3.2 shows that *'deviation'* is the best of the tests, with *'response time'* and *'difference'* close behind. *'Standard deviation (time)'* is the worst test as could have been predicted from the probability distribution curves (i.e. least separation).

The best value in Table 3.2 is the *'deviation'* with *dev>=2*, which gives a result of d'=1.50.

It is the case, however, that a test is not really considered worthwhile unless d' is at least 2 and preferably greater than 3. Figure 3.7 shows the ROC curves for these individual tests, and displays clearly that the tests on their own are poor. Consideration must be given to how to improve these results.

3.4.2 Test Protocols.

Turner, Frazer et al. (1984) describe a method of combining tests into test protocols. The aim of these protocols is to improve HT and/or FA, that is to increase HT and decrease FA.

Turner et al. describe two ways of combining tests into protocols - parallel and series. In this application only series-positive and series-negative protocols will be considered since these give the two extremes. Series positive, effectively means that all tests must be positive for the protocol to be positive (see figure 3.8), while series-negative means that only one test need be positive for the test protocol to be positive (see figure 3.9). For the purposes of this data, series positive can be thought of as an "AND" operation and series-negative as an "OR" operation.

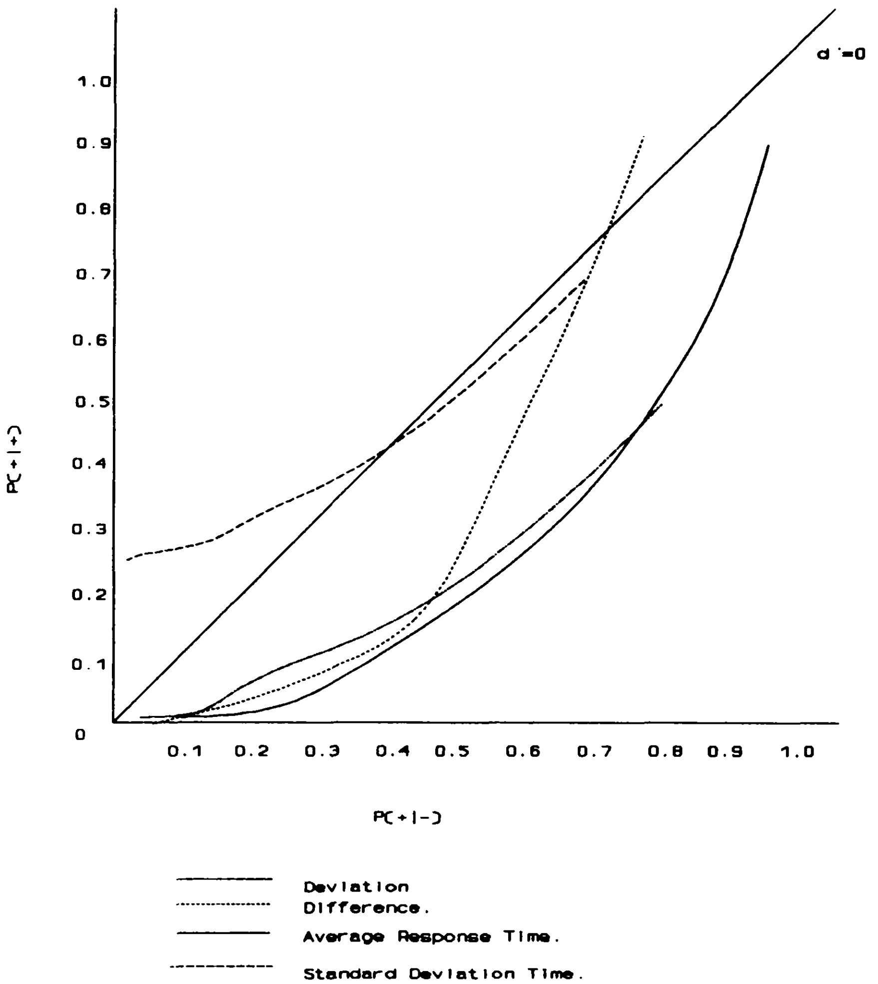


Figure 3.7 ROC Curves.
Showing the ROC curves for all four quantities considered with respect to d'=0.

Using the results from the individual tests, all combinations of protocols were considered. The different conditions referred to in table 3.2 were used. This gives 126 combinations of 2, 3 and 4 test protocols, using both "AND" and "OR" combinations.

Table 3.3 shows the best of these protocol results for the 19 points.

The conditions are given under the columns d,df,r and s. If x is thought of as the corresponding number in the table then d is equivalent to dev>=x , df to diff>x , r to resp>=x and s to stime>=x.

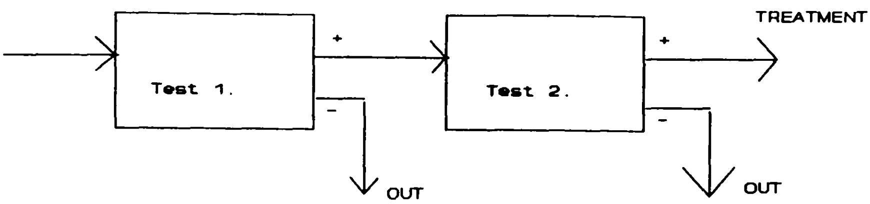


Figure 3.8 Series Positive Test Protocol.

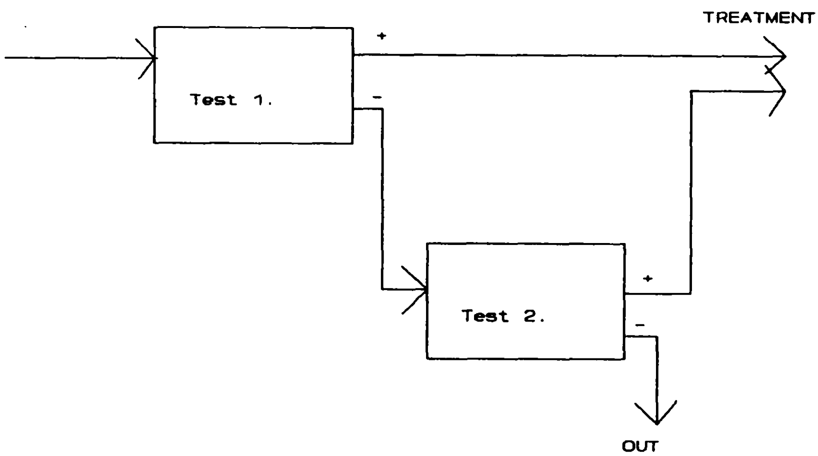


Figure 3.9 Series Negative Test Protocol.

This time the best protocol has d'=2.27 which still has some room for improvement.

Table 3.3 Best Test Protocols.

TEST	d	df	r	s	HT/FA	Pred V.	d'
d OR df	2	0	500	150	100/52	65%	2.27
d OR df OR r	2	0	600	150	100/57	63%	2.14
df AND r AND s	5	0	500	100	68/5	93%	2.11
d OR df OR r OR s	2	0	600	150	100/63	61%	1.99
d OR df OR s	2	0	500	150	100/63	61%	1.99
d OR r	2	0	600	150	94/36	72%	1.91
d OR r	2	0	500	150	100/68	59%	1.85
d OR df OR r	2	0	550	150	100/68	59%	1.85
df AND r	5	0	550	150	57/5	91%	1.82
df AND r AND s	5	0	550	100	57/5	91%	1.82
d AND r	2	0	550	150	57/5	91%	1.82
d AND r AND s	2	0	550	100	57/5	91%	1.82

3.4.3 Discriminant Analysis.

The method of test protocols has certain drawbacks in that it combines tests in a digital or discrete fashion. The general statistical way to discriminate two or more sets of data is to use a form of multivariate analysis called discriminant analysis.

The concept underlying this analysis is fairly simple. Linear combinations of independent variables are formed and serve as a basis for classifying cases into one or other of the groups.

In discriminant analysis, the emphasis is on analyzing the variables together. By considering the variables simultaneously, important information can be incorporated about their relationship.

In linear discriminant analysis, a linear combination of independent variables is formed using a weighted average technique. The weights are estimated so that they result in the BEST separation between groups.

That is, if we define

$$D = B_0 + B_1X_1 + B_2X_2 + \ldots\ldots + B_nX_n.$$

where the X's are the variables, and B's are the coefficients then the B's are chosen so that the D's differ as much as possible for the two groups.

The underlying assumption of linear discriminant analysis is that the distributions are "normal". If figures 3.1-3.4 are again considered it can be seen that figures 3.1 and 3.3 do look approximately normal while 3.2 and 3.4 do not.

If the distributions are not normal, then the optimum discrimination function is no longer the linear function. The optimum function in this case is quadratic and quadratic discriminant analysis should be performed. Table 3.4 shows the results of linear discriminant analysis for the 19 points. Table 3.5 shows the results of quadratic discriminant analysis.

The discriminant analysis was done using a statistical package called MINITAB - Release 7.

This allowed both linear and quadratic discriminant analysis to be carried out, and permitted cross-validation of the results. The package also allowed the possibility of predicting the outcome of a second set of data using the discriminant function of the first.

If tables 3.4 and 3.5 are looked at, it is interesting to note that d' for *dev* and *resp* is not improved by quadratic analysis, since they have approximately normal distributions and hence the linear discrimination is optimal. *'Difference'* on the other hand, which definitely does not have a normal distribution, is significantly better under quadratic analysis.

Table 3.4 Linear Discriminant Analysis.

TEST	HT/FA	Sens.	Spec.	PredV	d'
dev	78/15	78%	85%	83%	1.81
resp	68/31	68%	69%	68%	0.98
diff	73/31	73%	69%	70%	1.12
stime	47/31	47%	69%	60%	0.43
dev,resp	84/10	84%	90%	89%	2.27
dev,diff	73/10	73%	90%	87%	1.89
dev,stime	68/5	68%	95%	93%	2.11
resp,diff	68/26	68%	74%	72%	1.11
resp,stime	68/26	68%	74%	72%	1.11
diff,stime	73/26	73%	74%	73%	1.25
dev,resp,diff	78/10	78%	90%	78%	2.05
dev,diff,stime	73/5	73%	95%	93%	2.25
resp,diff,stime	68/26	68%	74%	72%	1.11
dev,resp,stime	78/5	78%	95%	93%	2.41
dev,resp,diff,stime	78/5	78%	95%	93%	2.41

Table 3.5 Quadratic Discriminant Analysis.

TEST	HT/FA	Sens.	Spec.	PredV	d'
dev	78/15	78%	85%	83%	1.81
resp	68/31	68%	69%	68%	0.98
diff	57/10	57%	90%	85%	1.46
stime	84/52	84%	48%	61%	0.94
dev,resp	84/10	84%	90%	89%	2.27
dev,diff	78/10	78%	90%	88%	2.05
dev,stime	68/0	68%	100%	100%	2.79
resp,diff	78/15	78%	85%	83%	1.81
resp,stime	73/26	73%	74%	73%	1.25
diff,stime	63/15	63%	85%	80%	1.37
dev,resp,diff	89/5	89%	95%	94%	2.87
dev,diff,stime	94/0	94%	100%	100%	3.87
resp,diff,stime	78/15	78%	85%	83%	3.09
dev,resp,stime	73/10	73%	90%	87%	1.89
dev,resp,diff,stime	94/0	94%	100%	100%	3.87

It can be seen that the linear discrimination results of table 3.4 are no better than the corresponding protocol results of table 3.3.

The quadratic analysis of table 3.5, however, is significantly better than table 3.3.

Table 3.5, in fact, gives d' values of 3.87 which corresponds to a very feasible test. Indeed the tables used to evaluate d' (Swets 1964) did not give a value for d' corresponding to FA value 0, so the FA value of 1 was used instead. This is well within the bounds of the possible statistical fluctuation on FA, but it does also suggest that the value d'=3.87 is likely to be an underestimate.

3.4.4 Checking the Validity of the Analysis.

In any sort of statistical analysis, nineteen is a very small number of subjects to use. It is important to consider the possibility that the results may be due to some idiosyncrasy of the particular set of nineteen subjects chosen. It should be stressed, however, that these were a random selection of volunteers and not picked for any particular reason.

Ideally another completely separate nineteen subjects should be tested and the results of the two compared. This, however, was not practical from the point of view of time and equipment available. Another technique used to check statistical results is to divide the larger sample into several smaller groups and do the analysis on each group independently, to allow a comparison to be made.

In the set of data here it was decided to divide the 19 into a group of 9 and a group of 10. The results show identical trends in the analysis and this confirms the presence of an increase in d' when using discriminant analysis. One of the sets gave slightly better results than the full set and the other slightly worse. The advantage of this grouping is that it gives an approximation of the errors present. The error in d' values were measured and found to be +/- 0.4 in the case of the individual results, +/-0.8 for the test protocols and +/-0.4 for both sets of discriminant analysis. This means that the value given in section 3.4.3 is

$$d' = 3.9 \pm 0.4$$

and even in the worst case it is above the d' > 3 limit. This confirms that the test is good and gives valid NOHL information.

3.4.5 Predictions for Gartnavel Hospital Data.

The Gartnavel Hospital Data was accumulated during the everyday running of the hearing aid clinic. As mentioned previously, 20% of the patients were either very elderly or had difficulty concentrating for the duration of the test. These patients were eliminated from this part of the analysis since they were patients who would not have been included in a category of NOHL subjects.

The quadratic discrimination calculated in section 3.4.3 was used to produce the criteria and the 'predict' function used to predict the categories of the Gartnavel data. These patients were all assumed to be truthful.

Table 3.6 shows the percentage correct obtained for the different combinations of tests used for the quadratic analysis. It can be seen that, at best, the agreement in the 2nd column is 78%. This is associated with discriminant analysis of *'deviation'* and *'standard deviation (time)'*. The results in this column are not as good as might have been expected and this may be due to the fact that the sample population was not suspected of NOHL and so included patients who simply would not have been considered for this type of analysis. By removing 4 more patients from the sample, again not in the category which would indicate NOHL, the column 3 results were obtained and these show much better agreement. Again, the best results come from the discriminant analysis of *'deviation'* and *'standard deviation (time)'* and this gives 100% agreement. However, it is always dangerous to remove too many data sets from one sample simply to get better agreement.

In the data from the normal hearing subjects it was clear that most subjects showed an increase in their response time when exaggerating their thresholds. With this set of data, both values of response time were measured and thus a comparison could be made.

Table 3.6 Predictions of Gartnavel Hospital Data.

Discriminant Analysis	% Correct (18 patients)	% Correct (14 patients)
dev	71%	90%
resp	14%	20%
diff	57%	60%
stime	35%	40%
dev,resp	64%	80%
dev,diff	71%	90%
dev,stime	78%	100%
resp,diff	57%	70%
resp,stime	21%	20%
diff,stime	50%	60%
dev,resp,diff	64%	80%
dev,diff,stime	71%	90%
resp,diff,stime	50%	60%
dev,resp,stime	64%	80%
dev,resp,diff,stime	64%	80%

With any other set of data, however, there is no way of telling whether the measured response time is greater than the patient's normal response time. This has a bearing in the response time data of the Gartnavel Hospital tests.

Since the sample used to determine the quadratic discrimination was made up of predominantly young people, their response times were reasonably fast. Any patient, especially very elderly patients, displaying a large response time will almost immediately be classed as NOHL. This accounts for the low percentage correct in the response time test in table 3.6. Possible solutions to this might be to have a much larger sample of 'test' subjects displaying both truthful and NOHL thresholds or to have some other way of measuring the patient's true response time to pure-tones.

It can be seen from table 3.6 that the Gartnavel data gives best agreement with discriminant analysis of *'deviation'* and *'standard deviation (time)'*. Any of the tests which include *'response time'* have the disadvantage discussed above and will therefore be limited in their use. The standard deviation of response time, however, does not have this disadvantage, since a subject being truthful will have a fairly constant response time - increasing only near threshold (Wright et. al. 1981). A subject displaying NOHL, on the other hand, is much more likely to show vastly differing response time and hence a higher standard deviation.

Similarly, with *'deviation'*, it is much more probable, that a truthful subject will hear two identical intensity of tones near threshold than a subject displaying NOHL will agree with two tones of equal intensity near their admitted threshold. It is possible that these NOHL subjects will be out by 5 or 10 dB, thus increasing the value of *'deviation'*.

3.5 Discussions and Conclusions.

The indices measured in this work show considerable significance in the detection of non-organic hearing loss.

i) The value of *'deviation'* has on average a higher value for subjects displaying non-organic hearing loss.

ii) *'Difference'* is much more likely to be less than or equal to zero for truthful subjects.

iii) *'Response time'* is in general higher for non-organic subjects. This is presumably because they have to think whether or not they hear each tone.

iv) *'Standard deviation (time)'* was the poorest of all the quantities measured, from the point of view of detecting non-organic hearing loss.

Using all the combinations of tests it would seem that the best method to separate the two groups was that of quadratic discriminant analysis.

The results of the tests are slightly ambiguous because the non-organic subjects were not actually going to receive industrial hearing loss benefit and therefore did not have the same motivation as patients who would receive benefit. The other problem with this set of patients is the large amount of exaggeration necessary to allow them to qualify for benefit. The advantage of these subjects, however, is that they can be classed as definitely non-organic.

This notwithstanding, the results of some subjects were excellent from the point of view of agreement on thresholds and from this standpoint, may not have been picked up in a clinic.

The results do show a trend in variables not normally available to the technician. The 'Random Test' does allow comparisons to be made which are not otherwise possible and it does seem that these are useful. The Gartnavel hospital results show that it is possible to perform this random test on most patients.

It would be worth carrying out a more detailed study using a large number of truthful subjects from a hospital clinic, who were in the same sort of age range as a similar set of subjects seriously attempting non-organic hearing loss.

Another possibility is to use a similar set of patients but to ask them to exaggerate their thresholds by only 10-15dB; however, this means that they would not reach the 50dB limit set for Industrial Hearing Loss Benefit.

In conclusion, the pilot experiments show that this 'random test' technique has much to offer in the detection of non-organic hearing loss among a large proportion of the population. They also show that quadratic discriminant analysis is the optimum way to combine the data obtained.

CHAPTER 4: Supra-Threshold Tests

4.1 Introduction.

The types of audiometry discussed have been primarily concerned with measuring the patient's threshold of hearing. These thresholds allow the consultant to quantify the patient's hearing loss and, if necessary, form the basis for a hearing aid prescription.

There are certain hearing defects, however, where the patient notes very little loss of hearing, or exhibits a typical sensori-neural loss, but where other symptoms suggest a more serious problem associated with the hearing system.

The acoustic neuroma or tumour is one such problem. Most acoustic neuromas cause some kind of hearing loss which will show up in reduced thresholds during pure-tone testing. There are, however, a number of cases which have been reported, where tumours reached fairly large proportions before they significantly affected the hearing thresholds.

There are a number of tests carried out above the patient's threshold (supra-threshold) which give some indication of various defects such as the presence of an acoustic neuroma or of Menieres disease. The tests, which will be discussed here, are predominantly tests for recruitment and abnormal adaption. They are, alternate binaural loudness balance, simultaneous binaural loudness balance and monaural loudness balance for recruitment and for abnormal adaption, tone decay and tone change.

(i) Recruitment:

This is the rapid increase in the sensation of loudness once the threshold of hearing has been crossed. The situation with some patients who have a sensori-neural impairment is that once a sound is intense enough to be heard, an increase in the intensity results in a disproportionate increase in the sensation of that loudness. Thus, a patient with a sensori-neural loss of 40dB can just barely detect the presence of sound at 40dB. A sound 5dB greater may, however, be perceived by the patient with a loudness greater than that heard by a normal hearing person at 5dB above threshold. Further increases in the intensity of the stimulus would result in more rapid increases in the patient's sensation of loudness. See figure 4.1.

Recruitment of loudness is characteristic of sensori-neural impairment because of the cochlear involvement - it is thought not to exist in cases of VIIIth nerve lesions although some authors have disputed this, for example Priede and Coles (1974), who show that incomplete recruitment can be indicative of VIIIth nerve pathology. Very often, in cases of acoustic neuromas, the size of the tumour means that it interferes with the blood supply to the cochlea and although the primary site of lesion is neural, there is a secondary cochlear site.

(ii) Adaption:

Adaption is any temporary change in auditory perception. Tone decay, specifically, is a patient's inability to maintain the audibility of a continuous tone. It is observed in many patients with Menieres disease and VIIIth nerve lesions.

Although the supra-threshold tests described in this chapter have largely been superseded by impedance audiometry (see chapter 7) it was felt that there was benefit in applying computer control to them. In many cases computer control reduces the time and complications involved in performing these tests.

It is believed that the cochlea has, along its length, detectors which determine with what frequency and intensity the subject hears a tone. If only one part of the cochlea is damaged then only one or two frequencies

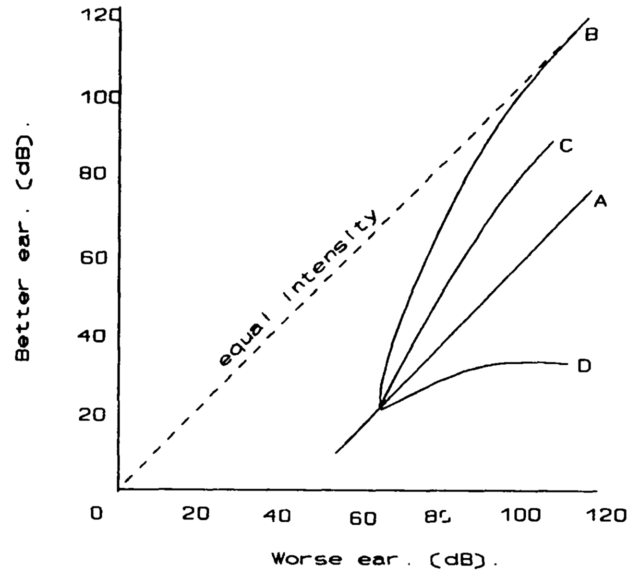


Figure 4.1 Recruitment. *A - No recruitment; B - Complete recruitment; C - Incomplete recruitment ; D - Loudness Reversal.*

may be lost. It is these sites of loss which loudness balance testing and tone decay aim to find.

4.2 Hardware Implementation.

The firmware in the audiometer (which contains a MC68701 microprocessor) had already been set up to allow for the use of two push buttons. In the loudness balance tests, one push button was used to increase the intensity of the test tone and the other to decrease it.

In the tone decay test the patient was required to hold the button down as long as he/she heard the tone and release the button when the tone became inaudible. To implement this easily, another button was constructed which was wired such that the button registered as the patient released it, when the tone became inaudible.

The MC68701 has, on board, a timer which was used to measure the time elapsed from the beginning of a job. An interrupt was generated every 1ms and the number of interrupts counted to give a value of time to the nearest millisecond. The use of this timer in the tone decay test is of great benefit since this allows an automatic measure of the time since the start of the tone.

4.3 Features.

All the usual features are implemented in these supra-threshold tests. The operator can quit or continue, as in the pure-tone threshold test. The test can be temporarily halted so that the operator can speak to the patient. Test results can be saved to floppy disk and printed out on the colour graphics printer. Old files can be reloaded for inspection or printing and a new threshold file can be loaded if required. Help is always available to the operator.

4.4 Loudness Balance Tests.

Loudness balance testing was originally suggested by Fowler (1936). Early studies showed that the presence of recruitment indicated a cochlear disorder, while no recruitment suggested retrocochlear. Work has been done over the years, to determine the degree of recruitment and what can be deduced from this. Priede and Coles (1974) suggest the use of standard criteria curves and a set of rules associated with these curves, which allow the audiologist to determine the site of lesion as cochlear or neural.

There are two distinct types of loudness balance testing, alternate binaural loudness balance and monaural loudness balance.

4.4.1 Alternate Binaural Loudness Balance (ABLB).

As the name implies, ABLB compares the hearing levels at which two signals sound equally loud in both ears. It is really only useful when the patient presents a picture of one relatively normal ear and one ear with some degree of sensori-neural hearing loss. Tones of the same frequency, but different hearing levels are presented to each ear alternately.

Another form of ABLB presents the tones simultaneously to both ears rather than alternately. In this application this form of testing is called Simultaneous Binaural Loudness Balance - SBLB. The patient is asked to achieve median plane localisation of the tone, that is, to find where the tone appears to be placed in the middle of the head.

Comparisons of ABLB and SBLB test results can be used to differentiate between cochlear and retrocochlear disorders. For recruitment testing, however, ABLB must be used since it has been shown that ABLB and SBLB do NOT yield equivalent results and that equal loudness cannot be inferred from median plane localisation. (Jerger and Harford (1960)).

4.4.2 Monaural Loudness Balance (MLB).

In the case of a bi-lateral sensori-neural loss, where the higher frequencies are more severely damaged than the lower ones, monaural loudness balance can be performed. In this test, the loudness of the tones in the impaired region are compared with the loudness of the tones in the normal region at and above threshold on the one ear. The test works on the same principle as ABLB, and is useful in that a 'good' ear is not required. There must, however, be relatively normal hearing associated with at least one frequency which can be used as the reference.

Many patients find it difficult to match the loudness of two different frequencies and the greater the separation of the frequencies, the harder the test becomes. The results of this test should be treated with care unless the patient has had some practice in the test.

4.5 Loudness Balance Implementation.

The system has been implemented so that several decisions can be made before the test begins.

In both ABLB and MLB there are two different ways of displaying the results and either can be chosen from an initial menu:

(i) A graph as shown in figure 4.2; this graphs better ear/frequency against worse ear/frequency and for a normal hearing person should yield a diagonal line at 45°.

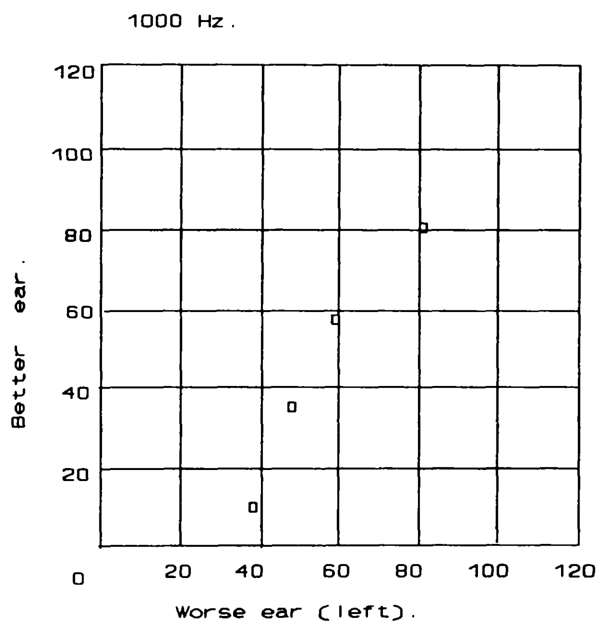


Figure 4.2 Loudness Balance Testing - Graph.

(ii) A laddergram as shown in figure 4.3; this displays the results by connecting the points of perceived equal intensity. For a normal hearing person the laddergram would show horizontal lines.

There are different schools of thought on whether the worse or better ear/frequency should be varied, and again

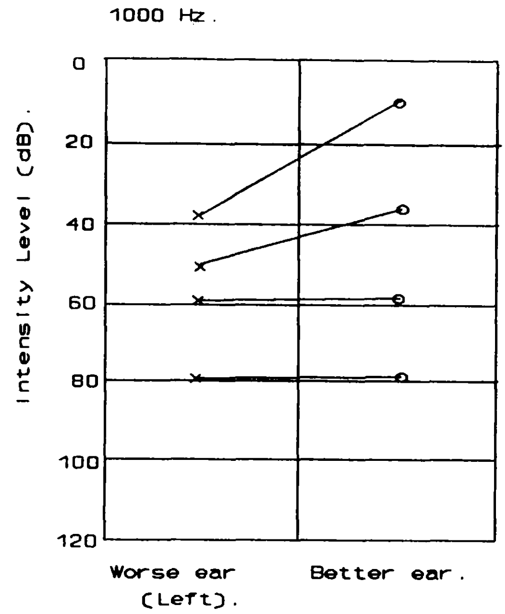


Figure 4.3 Loudness Balance Testing - Laddergram.

this decision can be made in the initial menu. A facility has been included which allows the ear/frequency being varied to be swapped round during the running of the test if required.

Auto or manual advance is available. Auto-advance means, if the patient stops balancing the signals, the system will wait for N presentations and then proceed automatically to the next test level (again N is set in the menu). In manual advance on the other hand, N is set to a large number (e.g. 100) and the operator advances the test to the next level when he/she is satisfied that the patient has balanced the current tones.

The menu also allows a decision to be made on the delay between pulse pairs and the number of supra-threshold intensity levels to be tested at each frequency.

The tones, of 500ms duration, are presented to the patient, who has two push buttons. Button A increases the intensity of one or other of the tones in steps of 2dB, while button B decreases its intensity in 2dB steps.

Alternatively the operator can increase and decrease the tones intensity directly from the keyboard.

4.5.1 Alternate Binaural Loudness Balance and Simultaneous Binaural Loudness Balance.

Before an ABLB test begins, the system examines the threshold data, looking for frequencies at which there is a difference of greater than 20dB between left and right ears. Up to 4 of these frequencies will be offered for testing. If the operator wishes to test at other frequencies, a list is offered sequentially of all the frequencies for which there is a valid threshold available and the operator may pick up to 4.

In the case of SBLB the test is similar, but the pulses are presented simultaneously using either continuous or pulsed tones. A key can be used to set the intensity of the test tone to zero during the test and a second key will return it to its previous intensity - this allows the operator to remind the patient which tone is the reference.

Results of ABLB and SBLB tests may be stored on floppy disk and have suffix '.alb'. Printing these results allows a choice of laddergrams or graphs to be made independently of that displayed on the screen.

4.5.2 Monaural Loudness Balance.

In this case, one ear is selected and the intensities of two different frequencies balanced by the patient or operator, as in ABLB.

The system checks at 500Hz and 1kHz for possible use of these frequencies as references. It then seeks comparisons with higher frequencies which have hearing losses. Otherwise frequencies are selected by the operator.

Results are stored to files with suffix '.mlb'.

4.6 Tone Decay Tests.

Various researchers have noted that some patients have difficulty maintaining audibility of a long continuous tone presented at threshold (Hood (1950)). This feature is known most commonly as tone decay, but is also known as abnormal adaption.

4.6.1 Tone Decay.

The standard tone decay test involves presenting a tone at an audible level for up to 60 seconds.

A patient suffering from tone decay will lose audibility of the tone before the 60 seconds has expired, the intensity of the tone is then increased in 5dB steps until the patient maintains audibility for the whole 60 seconds. The amount of tone decay is expressed as the dB change from the original threshold to the final hearing level required to meet the 60 second criterion. This is the test as described by Carhart (1958).

Some modifications have been suggested by different authors e.g. Rosenberg (1958) who suggested that the entire duration of the test at each frequency should be 60 seconds.

The patient is asked to hold the button down while they hear the tone and release it when audibility is lost. The tone decay test implemented in this case follows the Carhart test. Stephens and Hinchcliffe (1968) show this form of the test to give the most sensitive measure of abnormal adaption.

4.6.2 Tone Change.

In this case the patient is asked to press the button when the tone becomes inaudible, this will increase the intensity until the tone is again audible. The patient must maintain the sensation level of the tone. That is, the tone must be kept at what is heard as a constant intensity level.

4.7 Tone Decay Implementation.

As with the loudness balance tests, an initial menu of parameters is offered to the operator. Initial and final sensation levels can be chosen, and a decision on whether or not to pause between each intensity level is made.

4.7.1 Tone Decay.

The operator can choose the ear and frequency to be tested. The patient's threshold at that frequency is displayed by a horizontal dotted line at the appropriate intensity on the tone decay chart. Thereafter, the solid lines represent the length of time for which the patient maintained audibility. See figure 4.4.

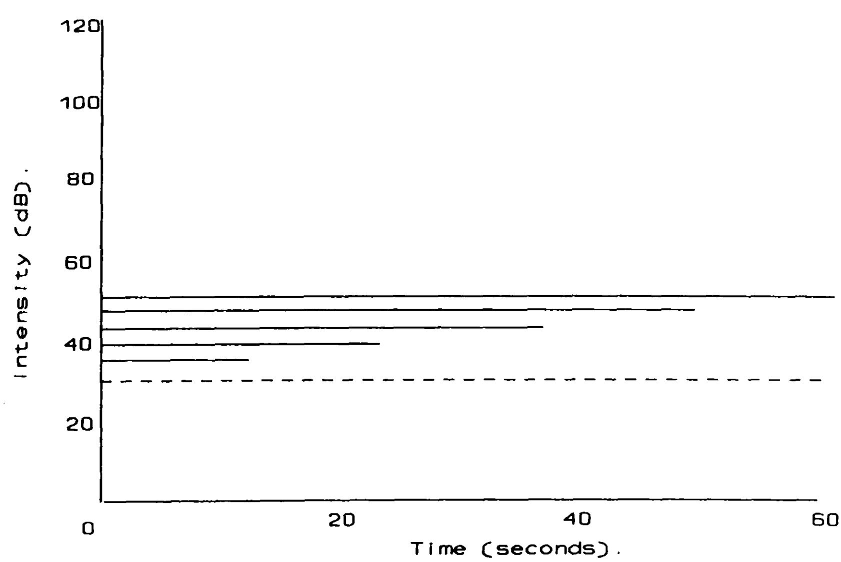


Figure 4.4 Tone Decay Test.

These files are stored to floppy disk with the suffix '.dec'.

4.7.2 Tone Change.

Again ear and frequency are selected by the operator. The test will be done at 3 initial intensity levels 5dB apart. The patient is asked to maintain the sensation level of the tone by pressing the push button when the tone's intensity appeared to decrease. Each time the button is pressed the tone intensity is increased by 2dB. If a patient is suffering from tone decay, for the sensation level to stay constant the intensity level must increase dramatically (see figure 4.5).

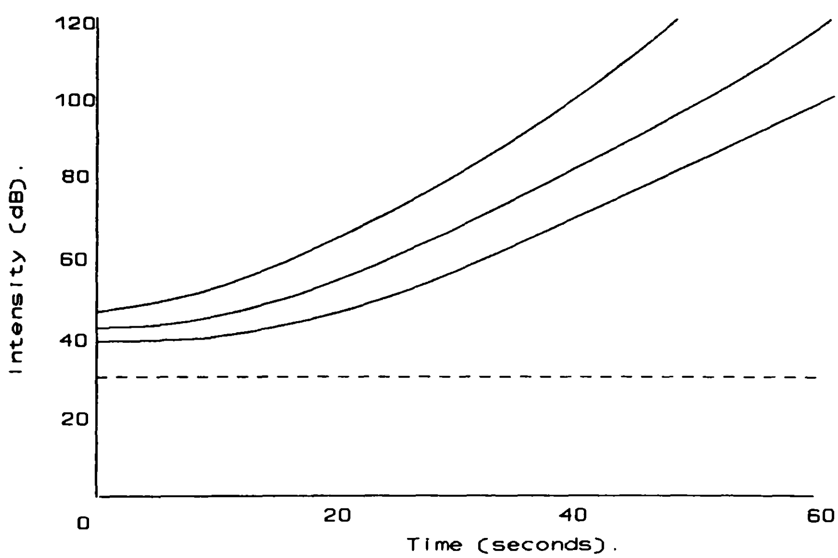


Figure 4.5 Tone Change Test.

These files are stored to floppy disk with suffix '.inc'.

4.8 Conclusions.

The supra-threshold tests adapt well to computer control.

In the loudness balance tests, the frequencies at which the tests could be executed can be suggested by the system. This can save the operator's valuable time but still allows the opportunity to override these suggestions, if required. On a manual audiometer it can be quite difficult to alternate the ear to which the tone is being presented, whereas the computer does this automatically and with ease which saves the operator continually having to be alert for the next switch change.

In section 4.4, the Priede and Coles criteria curves were mentioned. These were in fact added to the working program at a later stage and it is now possible to display each ABLB curve superimposed on the correct set of criteria curves. A further development might include the Priede and Coles rules being used, thus allowing the system to make the decision between cochlear and neural site of lesion.

In the tone decay test the timing is done automatically, saving a tedious process of starting, stopping and resetting a stopwatch.

Finally, of course, the charting of the tests is all done automatically which again is of immense benefit to the operator.

The adaption, then, of these supra-threshold tests to computer control allows the tests to be carried out with the minimum of effort on the part of the operator. In all cases the efficiency of the tests is remarkably increased under computer control.

CHAPTER 5: Speech Audiometry

5.1 Introduction.

The audiometry discussed in the preceding chapters has used pure-tones as its stimulus. It is clear, however, that human culture has developed to such an extent that the main use humans have for hearing is to comprehend speech. It is important, therefore, to measure, not only which pure-tone frequencies a patient can hear, but also what they can hear and comprehend when presented with speech. This is important since speech is the fastest known way of transferring information from one human to another.

On the simplest level, the audiologist may have a conversation with the subject, and judge from their answers to simple questions what they are able to hear. Speech audiometry is a development of this which involves assigning some kind of standardisation to the speech materials used, and to the loudness at which these are presented to the subject.

There are many different versions of speech audiometry and, in particular, America and the UK each have their own most commonly used procedures.

In the UK, the most standard form of speech audiometry uses monosyllabic words and presents the results on a speech audiogram. It is this type of test which will be discussed here and which has been implemented. The implementation is, however, flexible and allows other types of speech audiometry to be performed with only minor modifications.

5.2 Implementation of the basic speech test.

There are many variations on the speech audiometry test, involving differing types of test material, but all involve the presentation of speech and a reasonably consistent way of scoring the patient's response to it. In Britain the test material which is most regularly used consists of monosyllabic words usually in phonetically balanced lists. In these tests the words contain 3 distinct parts or phonemes and are scored accordingly out of 3. For example, if the word were 'CAT' then the scores corresponding to the given responses are shown in table 5.1.

Table 5.1 Response scores to the word 'CAT'

Response Word	Score
CAT	3
MAT	2
CAP	2
MAP	1
DOG	0

The sum of scores for 10 words can then be graphed against the hearing level of the speech on a speech audiogram.

5.2.1 Monosyllabic Words and Phonetic Balance.

There are several factors involved in the selection of the speech material to use. These are discussed by Lyregaard (1987).

a) Redundancies.

In this context, phonemes are the least redundant, and sentences the most redundant type of test material. Sentences need fewer acoustic cues for the patient to recognise the stimulus, and sentence tests measure not only hearing deficiency at a peripheral level, but linguistic competence and general ability to understand, which have nothing to do with the hearing mechanism.

b) Scoring of Responses.

This is a difficult problem. Phoneme tests make it awkward for the patient to explain to the operator what they have heard, while the meaning of a sentence may be understood even though it was not heard perfectly.

c) Relation to 'Everyday' Speech.

Obviously sentences and words rate much higher in this category than individual phonemes.

d) Duration of Test.

Sentences are much longer and therefore inefficient compared to other items.

The more complicated the item, the more factors like linguistic competence and intelligence affect the result. When the items are more specific, the test results look more like the pure-tone audiogram.

Weighing up all the above factors, a choice of MONOSYLLABIC meaningful words as a test material is a compromise which best satisfies all the conditions.

Another way of making a test more akin to everyday speech is to insist that it has a phonemic composition equivalent to that of everyday speech. Different phonemes should appear in the test material with the same relative frequency as they do in everyday speech. This is known as phonetic balance and can be considered as a weighting S.

$$S = W_1S_1 + W_2S_2 + \ldots + W_iS_i + \ldots + W_NS_N$$

where S = Total score

W_i = Weighting factor for i^{th} phoneme

S_i = Score obtained for phoneme i

N = Total number of phonemes

Although the procedure which has been implemented here can be used with many speech audiometric tape recordings, it was designed with particular reference to the AB(S) wordlists.

These contain 15 short isophonemic lists of monosyllabic words and are named after their developer Arthur Boothroyd (1968).

Each of the lists present 10 test words, (30 phonemes) at 4 second intervals and at the beginning of each tape there is a 1 minute long 1kHz tone which can be used to calibrate the machine for speech level. These AB(S) wordlists are shown in appendix B along with their phonetic components - these phonemes will be discussed in chapter 6.

5.2.2. Basic Test Implementation.

The test implemented here, essentially follows the procedure suggested by the Institute of Sound and Vibration Research (I.S.V.R) in Southampton.

The first wordlist is presented at a level Ds(1) of

$$Ds(1) = BTA + Cs + 5$$

where BTA is the average of the best two air conduction thresholds at 250, 500, 1000, 2000 and 4000Hz from the pure-tone audiogram.

Cs is a calibration constant and will be explained in section 5.3.1.

Each word is scored out of three, as described in section 5.2. The overall score therefore for each wordlist (10 words) is out of 30 and is usually expressed as a percentage. If this score, for the first wordlist, is less than 40%, the intensity for the next wordlist is increased by 10dB, otherwise it is reduced in steps of 10dB until a score of less than 40% is obtained. The intensity is then set to Ds(1) + 10dB and increased in steps of 10dB until the score becomes greater than 95%. It is further increased by 20dB for either two steps, or until 100dB is reached. In this way a speech audiogram can be accumulated as shown in figure 5.1.

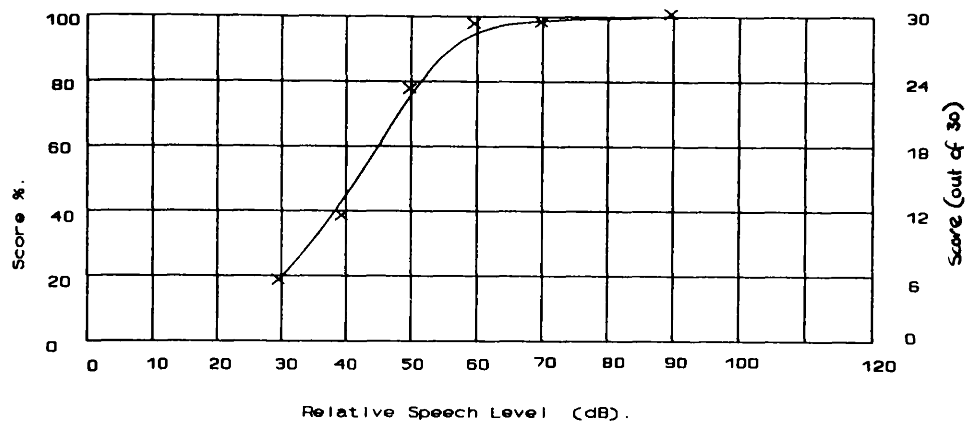


Figure 5.1 Speech Audiogram.

There are several advantages of using a computer for these tests:

(i) The level at which the next wordlist should be presented, will be set automatically by the system and saves the operator performing these calculations.

(ii) Although the operator must type the individual word scores into the computer, the subsequent tally of scores, is kept by the system. This again saves the operator having to calculate these at the end of the test and plot them on the speech audiogram.

(iii) A best fit curve can be fitted to the points by the system which is much more accurate and consistent than one drawn by hand (see section 5.2.4).

There are also several measurements made which are used in determining the patient's ability to hear speech. These, of course, can be automatically calculated by the system. They are, Optimal Discrimination Score (ODS), Half Peak Level (HPL) and Half Peak Level Elevation (HPLE); see figure 5.2.

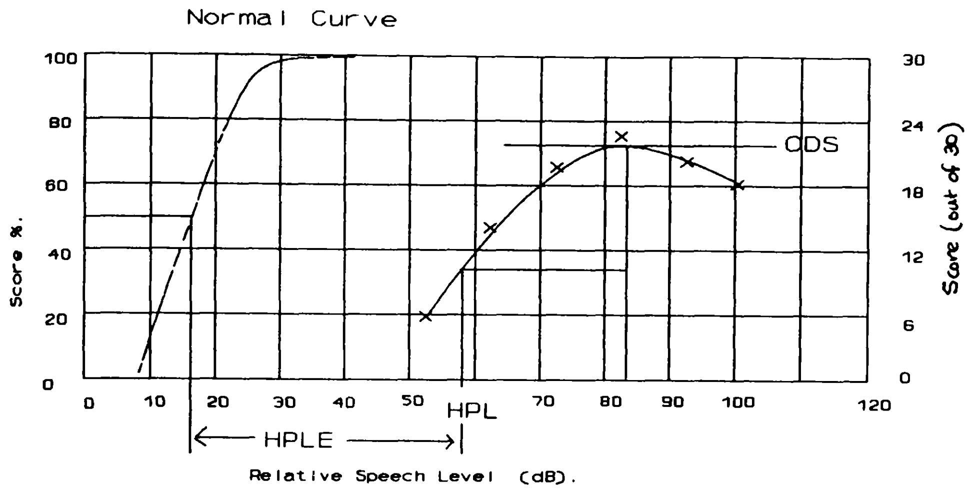


Figure 5.2 Definitions of ODS, HPL and HPLE from the speech audiogram.

Optimal Discrimination score (ODS), sometimes called PB_{max} is the maximum discrimination score which the patient can achieve during the test.

In cases of normal hearing, or where a purely conductive loss is present, the subject should obtain an ODS of 100%. In cases of sensori-neural loss, however, the ODS obtained is liable to be less than 100%.

It can be shown that in many cases the sound level at which the subject can hear about half the words is comparable with their average pure-tone threshold.

In Britain, however, what is measured is not the intensity level at which a 50% score is achieved, but the level at which a score of ODS/2 is attained. This is known as the half-peak level, HPL, and is a measure in arbitrary units. A more general measure is the difference between this HPL and an average normal HPL (i.e. the HPL obtained by averaging the results of several normal hearing subjects). This is known as the HPLE and this value, measured on different machines can be directly compared.

5.2.3. Masking.

As in pure-tone audiometry, it may be necessary to mask the non-test ear, and the mask level M is given by the equation:

$$M = Ds + maxABG_{nt} - 40 + Em$$

where Ds is the speech level

$maxABG_{nt}$ is the maximum air bone gap in the non-test ear at any one of the frequencies in the range 250 - 4000Hz.

Em is a calibration constant and will be discussed in section 5.3.2.

5.2.4 Curve Fit - Cubic Spline.

The cubic spline, sometimes known as a B-spline, is a standard mathematical technique to fit a smooth curve to a series of points. It takes two successive points in a sequence and constructs a cubic curve between them using these two points and the two adjacent. It produces a very smooth curve which is continuous even in the second derivative. This smoothness is at the expense of the curve not going through every point exactly, but it is an ideal type of curve for this application.

If we consider two points P_i and P_{i+1}, then the B-spline curve segments between these two successive points are obtained by computing X(t) and Y(t) where t grows from 0 to 1. Before starting the calculation, however, a check should be made that the points are ordered in ascending x values.

Now

$$X(t) = ((a_3t+a_2)t+a_1)t+a_0$$
$$Y(t) = ((b_3t+b_2)t+b_1)t+b_0$$

where

$$a_3 = \frac{(-x_{i-1}+3x_i-3x_{i+1}+x_{i+2})}{6}$$

$$a_2 = \frac{(x_{i-1}-2x_i+x_{i+1})}{2}$$

$$a_1 = \frac{(-x_{i-1}+x_{i+1})}{2}$$

$$a_0 = \frac{(x_{i-1}+4x_i+x_{i+1})}{6}$$

and the b coefficients are identical, simply replacing y for x. It can be shown easily that

$$x \quad , \frac{dx}{dt}, \quad \frac{d^2x}{dt^2}$$

and the corresponding y values are continuous.

A program was written in the system to fit a cubic spline curve to the audiometric points, and therefore allow ODS, HPL and HPLE to be calculated. The algorithm used is based on one taken from Ammeraal (1986) and appendix C shows a program written to test and use this algorithm.

The arrays valx[] and valy[] contain the audiometric data points. The array valx[] contains numbers which are equal to 120-(speech level in dB) and valy[] contains the scores out of 30. The arrays calvallx[] and calvally[] are the corresponding values for a curve obtained from the average of the results of a number of normal hearing subjects.

The main program simply gives these arrays sensible 'test' values so that the subroutines used to calculate the spline may be examined.

The function drawgris() draws a speech audiogram grid while calclate() draws the curve and calculates the corresponding ODS, HPL and HPLE values.

Turning firstly to the function cubic(), it is the function used to calculate all the coefficients a_0, a_1, a_2, a_3, b_0, b_1, b_2 and b_3. These are declared so that their values will be passed back to calclate() which is the calling function.

Because the spline program fits a cubic to the points i and i+1 by using points i-1 and i+2, the first and last points are not joined by the curve. In this application, however, the first and last points are duplicated, to allow the curve to be fitted to all the data points.

The parameter i is passed to the function cubic() to allow xA, xB, xC, xD and the corresponding yA, yB, yC and yD to be calculated. The values of xA and yA correspond to the coordinates of the point at i-1, xB and yB to the point at i, xC and yC the point at i+1 and xD and yD the point at y+2. The parameter k is not used in this test program, but tells the function whether to use the valx[] arrays or valx1[] corresponding to right and left ears.

Turning back to the function calclate(), the values in the arrays valx[] and valy[] are ordered in ascending x values, this is so that the curve is drawn in a sensible order. A value for N is calculated, where N is the number which will determine how many values of t between 0 and 1 are used. In this application, N is taken to be the difference between the two x values which are currently being considered.

X and Y, corresponding to X(t) and Y(t) discussed previously, are calculated according to the equations given and the curve is plotted for each of the t values between 0 and 1. At the same time the value of ODS is calculated as the maximum value of Y attained on the curve. Once ODS is known a similar procedure is required to measure HPL. This time the curve is not plotted, but calculated until a Y value of greater than ODS/2 is achieved, and at this point the HPL value is equal to X.

The only thing remaining to do is to measure HPLE. In most cases this is calculated by subtracting an average normal HPL from the subject's HPL. Since the values in calvallx[] and calvally[] correspond to the points of an average normal curve (this will be discussed in more detail in section 5.3.1), these will be used to calculate the normal HPL. This value is HPL1 in calclate() and is evaluated by calculating the X value on this normal curve corresponding to a Y of 15 (i.e. 15/30 or 50%).

In a few hospitals it is the practice to define the HPLE as the difference between the patient's HPL and the X value on the average normal curve at that particular HPL score. This can be achieved by replacing the 15 discussed above by ODS/2.

These values of ODS, HPL and HPLE will then be printed on the speech audiogram, which saves the operator performing these calculations.

5.2.5 Additional Features.

All the usual advantages of computer control exist in this type of test. The results can be saved (with suffix '.spe'), retrieved and printed out. This allows ODS, HPL and HPLE to be saved and printed out along with the speech audiogram.

During the running of the test the operator can quit or continue to the next intensity level by simply pressing the appropriate key. If at any stage the operator wishes to change the level at which the test is being performed, simply typing 'n' will allow the operator to input the value required. The system will calculate the corresponding mask level, but the operators may disagree and insert their own choice of level.

5.3 Calibration of Speech.

In section 5.2, reference was made to an average normal curve, and to calibration constants Cs and Em. These come from the test method adopted by the ISVR, and their definitions can be obtained from ISVR internal circulars.

The precise measurements of the constants Cs and Em, are explained in detail in the following sections 5.3.1 and 5.3.2.

It is important that the correct size of signal is presented to the audiometer from the particular tape recorder to be used. If the signal is too small, the full range of intensities will not be available, whereas, if the signal is too large, the peak intensities will saturate, disturbing the signal waveform and distorting the word. To check the size of this signal, a 'link amplifier' was built. It allows the maximum amplitude to be set to just below the level at which the audiometer will saturate. In the first instance, calibration is achieved using the 1kHz calibration tone at the start of the tape. The RMS level of this calibration tone is equal to the average RMS level of the wordlists and experimentally it was shown that the peak level of the wordlists is never more than 5 times the peak level of the calibration tone. The link amplifier contains an LED which lights if the signal going to the audiometer is liable to saturate. There are two settings of the amplifier, one for use with the calibration tone (entered by pressing a push button) and the other for use with speech (the normal setting).

In the first case, the calibration tone was presented to the amplifier with the push-button pressed, and the gain setting on the amplifier set to just below the level at which the LED lit.

The push-button was then released and the calibration tone replaced by speech signals. Releasing the push-button reduces the signal by a factor of 5, which should mean, from the experimental evidence mentioned, that the peaks of the speech will NOT saturate (i.e. the LED will not light).

It is possible, however, that this ratio of 5 is an overestimate making the level set by the calibration tone an underestimate. To reach the optimum setting, the push-button is released and the speech signal set as high as possible making sure that the LED never lights. It should be noted that when using speech to calculate this setting, two or three words are not enough. For accuracy, a large cross-section of words should be used.

5.3.1. Normal Hearing Curve and Cs.

The constant Cs is known as the calibration correction factor for the speech audiometer. It is a constant which must be measured using the particular equipment which will be used for testing, the specific recording of the wordlists, the calibration tone and the instructions to be given to the subjects.

Cs is the average intensity level at which a 50% intelligibility score is attained by a group of normal hearing subjects. Cs is calculated by averaging the speech audiometry curves for several normal hearing subjects (to give an average normal curve), and then taking the intensity level corresponding to 50% score.

Because of the variation in intensity in any one word of speech, it is not possible to say that normal hearing subjects should hear speech at 0dB, even though they may be able to hear a pure-tone at this level. A word presented at an average intensity level of 0dB may have part of the word at -10dB and/or +10dB (see figure 6.4).

Figure 6.4 shows the waveform of the word 'cheek'. Because of the length of the word, the waveform is broken up and displayed on several lines; these lines are compressed to minimise space. The first eight lines correspond to the phoneme 'ch' which has a small intensity and is high frequency. The next eight lines are of lower frequency and have a much larger amplitude; these correspond to the phoneme 'ee'. There follows a period of silence, then the consonant 'k', again of high frequency and lower amplitude. This period of silence is characteristic of the family of consonants like 'k' known as stop consonants.

Adding the value of Cs to the speech intensity level means that the speech always starts at a level which should be heard by the patient.

5.3.2. Em - Masking calibration constant.

The equation given by the ISVR for masking includes a calibration constant Em. This is calculated by:

i) Measuring the speech level, SdB, at which normal hearing listeners obtain a score of over 95%.

ii) Speech is then presented, at this level, to the same normal hearing subjects, this time with noise added on the same earphone.

iii) The noise is increased to a level NdB until the score obtained by the subject is less than 10%.

Em is defined as the difference between the 95% speech level, S, and the speech and noise 10% level, N.

i.e. $Em = (N-S)$ dB. (see figure 5.4).

5.3.3. Implementation.

Since the calibration of speech audiometry relies on an average normal speech curve and the derived constants Cs and Em, a program was written to calculate this average curve and compute Cs and Em. A hardware switch is provided to allow speech and noise to be presented to the same earphone. This switching is done just before the patient earphones. A speech audiogram is obtained, with no masking, for a normal hearing subject, either by testing or loading a previously accumulated file from disk.

The hearing level at which the system calculates the score to be 95% is displayed. The operator may, however,

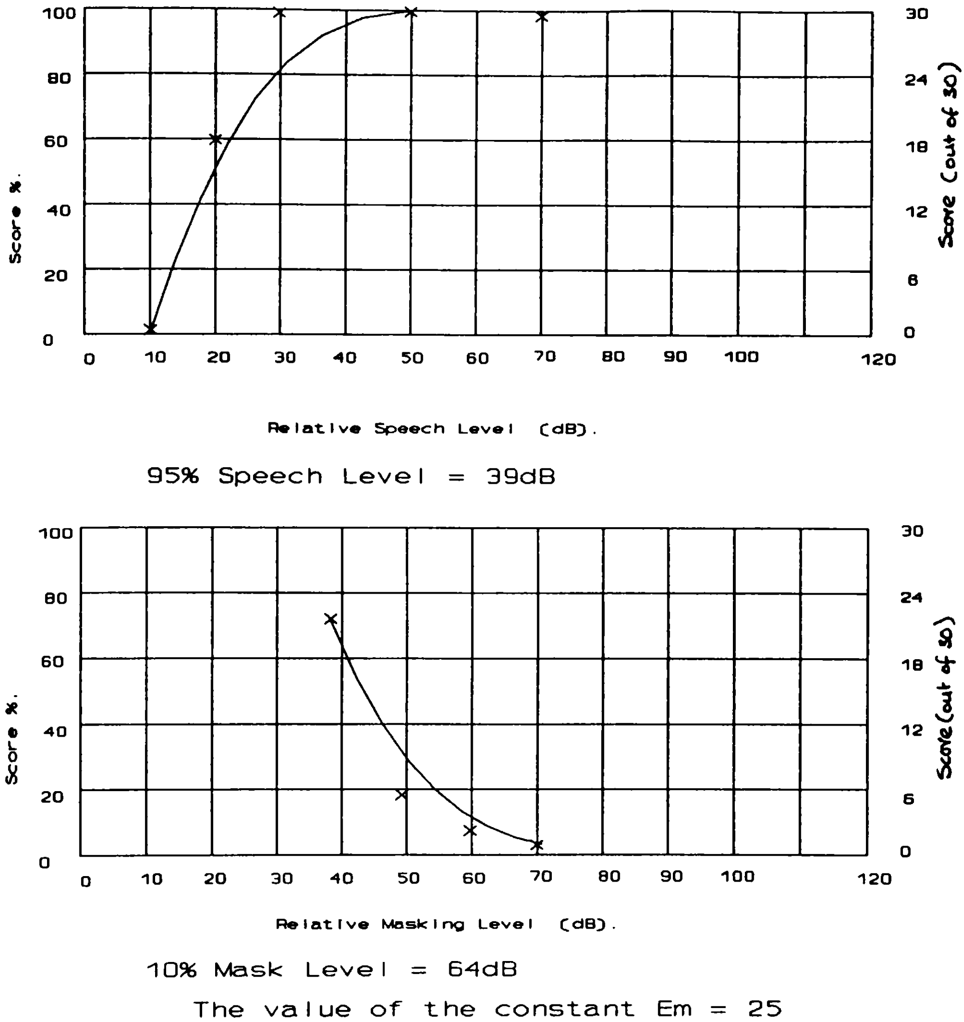


Figure 5.4 Determination of Em.

choose to disagree with this value and enter their own estimate.

Another audiogram is then drawn, this time the x-axis displays the masking level. The speech level is held constant at the 95% level determined above, while the mask level is increased until the speech discrimination score has fallen to less than 10% (see figure 5.4).

If the operator agrees with the level measured as having a 10% score, the value of Em is calculated and displayed.

Once these curves have been accumulated, they should be stored in one of the memory blocks which have been labelled A - O. The system will offer to store it in the next vacant memory block, but any block can be chosen. Using this technique allows 15 pairs of curves to be built up in successive memory blocks A,...,O. The memory blocks can be cleared, or all the blocks currently loaded can be displayed. Saving and printing of the individual curves is also available.

Once all the normal audiograms required for the calculation of the average have been accumulated, an average can be computed from them. On entering the averaging routine, all the blocks are displayed as shown in figure 5.5.

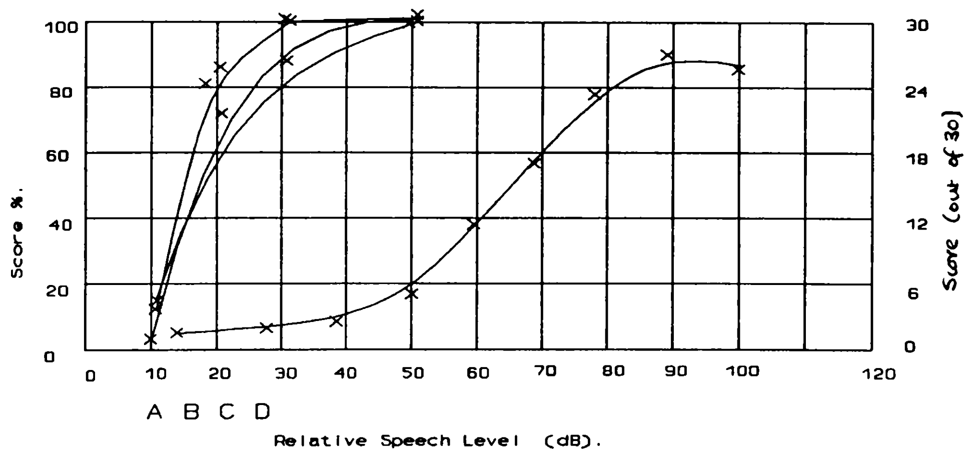


Figure 5.5 Determination of the normal curve and Cs.

On the PC screen, these are displayed in colour, making it easy to distinguish between blocks. At this stage, the operator must decide which curves should be used for the computation of the average. In the case of figure 5.5, it was decided not to use the curve stored in block D. Simply typing '-D' removes this curve from the screen, and from any subsequent calculation of the average. If, at a later stage, curve D is again required, simply pressing 'D' reinstates it. Using this technique it is possible to remove spurious results before calculating the average.

The average of these curves and the corresponding mask curves is calculated along with the related Cs and Em values and stored in a file with suffix '.ave' which will be used for calibration in the main speech audiometer program.

Values of the normal curve and a Cs value can also be calculated for bone conduction, and these will be stored in the same file.

5.4 Coded Messages.

In the past, speech audiometry, of the type described above, has required the operator to listen, on a reference earphone, to the words as they are spoken on the tape. The scoring is then done by comparing what the operator heard, with what was repeated by the subject. This introduces errors, since the operator is now dependant on what they hear and may therefore subsequently mishear. The reference earphone channels are very often not of such high quality as the patient channels, and the earphones used as references are usually of lower calibre. It is also an inconvenient way of scoring since there are many procedures to be considered simultaneously. It was felt important to find some way of improving on this 'listening' technique of scoring and making it more convenient and less erroneous. Several methods were considered and some of these will be discussed in section 5.5, but the most convenient of these is discussed here.

Codings for the words are stored (using a mechanism discussed in section 5.4.1) on one channel of a stereo tape, while the audio signals for the words are simultaneously spoken on the other channel of the tape. When the tape is played back, the coded messages are sent to the PC, where they can be displayed on the screen for the operator to see. Meanwhile, the audio signals are directed as usual to the patient earphones. This reduces the error in operator listening since the words are now visually presented, and it is then up to the operator to score each word on the PC, which will keep a tally of the overall score.

5.4.1. Frequency-shift-keying via a Modem.

The most commonly used way of coding messages in an audio form is to use frequency shift keying. This involves using frequencies f_1 and f_2 as carrier signals which represent digital 0 and 1. It is this type of coding mechanism which a MODEM uses and for this reason a commercially available MODEM chip, the Am7910 FSK MODEM, made by American Micro Devices Inc., was used to produce and read the coded messages. This uses frequencies 1070Hz and 1270Hz to represent digital 0 and 1 respectively.

Using the mode of operation which is required, this MODEM chip must operate at 300 baud. This is inconvenient since the serial port of the PC has already been set up to run at 9600 baud. This means that after the coded messages are read from the tape, they must be sent to the serial port of the audiometer at 9600 baud. There must therefore be some mechanism introduced to increase the baud rate from 300 to 9600, after the codes have been read from the tape, and before they are sent to the PC.

To do this we use a UART (Universal Asynchroneous Receiver/Transmitter), in this case an RS UART 6402. These devices are, in the main, used to convert serial data to parallel and vice-versa. In this case, however, the data is clocked in serially at 300 baud, and the parallel output immediately recycled to the parallel input. The resulting serial output is then clocked out at 9600 baud. This gives the 300 baud serial to 9600 baud serial transfer required.

Figure 5.6 shows a schematic diagram of the setup.

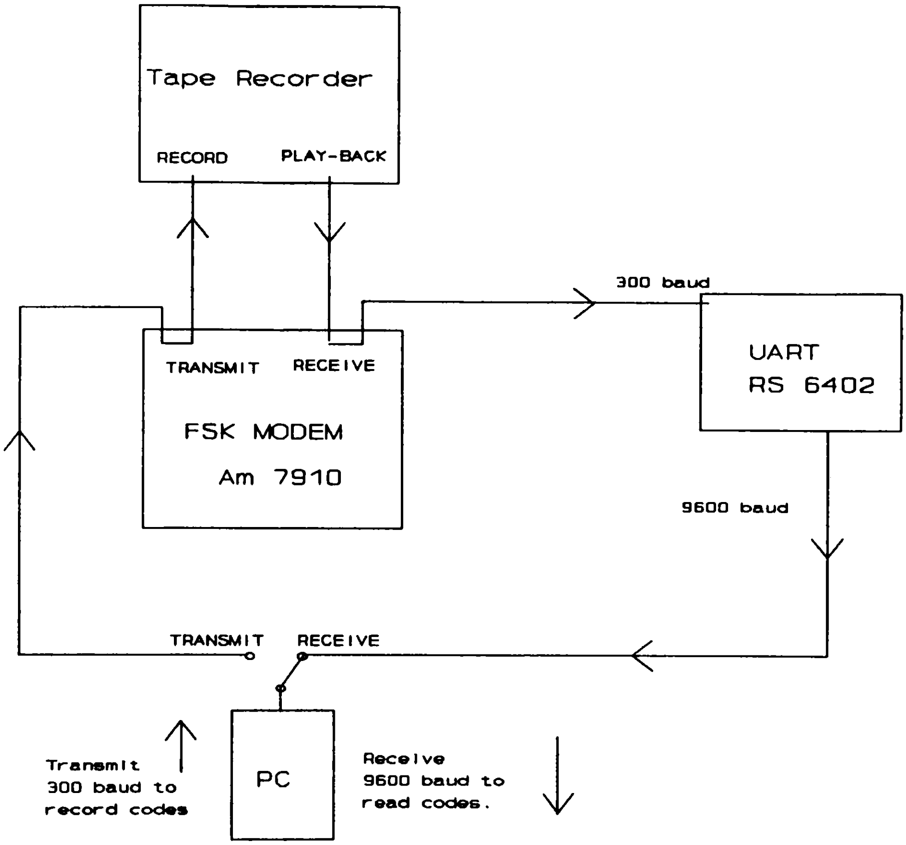


Figure 5.6 Schematic Setup of ASCII coding Mechanism.

In the first instance the tape is produced by setting the switch to transmit, running the tape on record, setting the PC to output on its serial port at 300 baud and sending the codes down the serial line from a program written in the PC. The codes are then received by changing the switch to receive, thus hooking in the UART, and with the recorder set to playback, the PC receives the codes at 9600 baud from its serial port.

5.4.2. Coded Messages - ASCII Characters.

Each of the letters which make up the words spoken in the test has a printable ASCII code i.e. the ASCII codes corresponding to the letters a-z, which are decimal 97-122 or hexadecimal 61-7A which are equivalent to binary 1100001 through to 1111010.

The simplest coding system to use then, is the ASCII character codes. This would mean sending a start bit, 7 bits of data with even parity and 2 stop bits to the tape. In this way, when the codes are received, they can immediately be displayed on the screen of the PC, as the letters which they represent.

Once this mechanism has been set up it is easy to send other messages to the tape, so that these will be displayed on the screen of the PC as the audiometer program is executed.

In this application, which used the Boothroyd wordlists, each wordlist was preceded by a coded message WORDLIST X, and the two trial words in each list were followed by a message '* trial word - don't score *'. This trial message was then used in the program, so that a score was not expected for these words.

This technique can be adapted to use any of the sets of wordlists which are available. It is a simple matter of producing a new tape, and there is a PC program available to do this. It reads a file previously constructed by the operator and sends it's contents to the tape every time the space bar is pressed. Other specialised messages, like those mentioned above, require specialised key-presses.

Care must be taken to ensure tape channel separation. This separation is not always consistent from machine to machine. For best results it is advisable to use the same machine both to record and play back, as use of another machine may cause cross-talk between the channels.

This facility greatly enhances the speech audiometry test, making it a much more convenient and elegant test to carry out.

5.5 Other Methods of Speech Production.

As was mentioned in section 5.4, other ways of producing the words were considered. Some of these are worth consideration in the future, others were tried and found to be lacking in some way.

5.5.1. Digital Storage of Words on Floppy Disk.

By using an analogue to digital converter (ADC), the words on the audio tape can be digitized and these digits stored on a floppy disk or hard disk. Work on this type of mechanism has been done by other authors, e.g. Kamm et al. (1980) who digitized words and investigated several properties. They looked at a random selection of word order for the tests, and undertook some manipulation of the digital numbers which led to different word presentations.

It was originally thought that the digitized words could be loaded into the computer's memory at the start of a test, or loaded a wordlist at a time, and then any order of words could be chosen for the test. Consideration must be given, however, to the phonetic balance of the lists.

The ADC used in this application was on a card for a PC extension slot. It was made by Flight Electronics Ltd. and called a PC-30 card. It contains 4 DAC's (Digital to Analogue converters) and an ADC which may be set to accept either uni- or bi- polar input.

It was estimated that the highest speech frequency is about 10kHz, and so to digitize without losing information, we must sample at at least 20kHz, the Nyquist frequency. In fact, to give good quality, the sampling was done at 40kHz, and 24576 integer samples were taken. Each integer corresponds to 2 bytes of data, that is 48kbytes of data was digitized. This was limited to 48K because, in large model on the TURBO C compiler, each C file or module allowed a maximum of only 64kbytes of data and for other reasons, explained later, the number of integers had to be a multiple of 8192.

It was found that using a program adapted from the driver programs for the PC-30, the data could be captured and stored in the 24576 integer array. This was done for each word, with the exception of some long words, which were stored in two blocks. This facility was in fact used for another purpose which will be described in chapter 6.

Once the words were obtained in digital form, they were stored in a disk file which could be retrieved at any time, fed through the DAC on the PC-30 card, and presented to the patient earphones. Because of the way the digitized words were stored to disk, each word took up 99840 bytes of memory. This meant that only three words would fit on one floppy, and the 180 words of the Boothroyd wordlists would take up about 60 floppy disks or 18Mbytes of memory.

The words from the floppy disks take a long time to load and it is inconvenient to have to keep changing floppy. A hard disk has a fan and most of the audiometers are run from floppy-only machines to avoid the fan noise - which can increase thresholds by up to 10dB. It therefore seemed completely impractical to use this technique from floppy disk. Also, 18Mbytes on a hard disk is a substantial amount of space. It is, however, available for anyone who wishes to use it. New technology like compact discs and digital audio tapes means that there are other ways available for storing digital data, these are compact, fast and easy to read (see section 5.5.3).

5.5.2 Synthetic Speech.

Another method of speech production for audiometric testing which was considered was the use of synthetic speech. An RS chip the speech synthesis i.c. 263 was purchased and an M68000 assembly code program was written to allow this chip to be controlled. This contained seven subroutines, entered by typing 1-7, to control the inflection, slope of inflection, rate of speech, extension and range of pitch, rate of articulation, amplitude, and filter frequency range. Each allowed the quantity concerned to be taken up or down, and restricted it to its limits. The chip contained 64 phonemes which can be linked together to produce words. Although some words were made to sound quite reasonable, using this technique, the phonemes given, were American English and so some words sounded particularly American which of course is useless for a British speech audiometry test.

5.5.3. Audio Tape or Compact Disc.

Some authors have, very recently, discussed the use of the compact disc in material for speech audiometry, e.g. Wilson et al. (1990) who list its advantages for speech audiometry as including :-

a) random and not necessarily sequential track selection, b) display of the current track and its time to run, and c) variable output level which facilitates connection to an audiometer.

Other more specific advantages are that the speech is of high quality and does not degrade as audio tapes do.

In general compact disc players digitize at 44kHz and so there is no problem with lost information; see section 5.5.1.

With reference to section 5.4.2, although the ASCII coding technique is very effective, it suffers from a cross-talk problem between the two channels of an audio tape. The same ASCII codes could be used with compact disc recordings, but, because of the digital nature of the CD medium, there is no such problem of cross-talk.

Unfortunately, neither the equipment nor the funding to produce a CD was available, but it would be easy to record one if the equipment became accessible.

Similarly Digital Audio Tapes (DAT) could be used.

The disadvantage of using these systems is that hospitals would be forced into buying new and sometimes expensive equipment.

5.6 Conclusions.

The speech audiometry test normally carried out under manual control adapts well to computer control and affords many useful advantages.

The calculation of the average normal curve can be done easily in the system, saving the operator much work.

The ASCII coding allows the operator to do everything from the PC rather than having to listen to the tapes themselves as the test proceeds. The great advantage of this coding is that it is adaptable to other storage media, for example CD and floppy or hard disks, making it the favoured technique to provide for hospitals at the present moment, especially since the committee set up at NPL to look into speech materials are considering producing it on CD.

CHAPTER 6: Phonemic Speech Analysis

6.1 Introduction.

The previous chapter described the conversion of the standard speech audiometry test to computer control. The complexity of the human neural system means that it has the ability to make optimum use of the extensive information which it receives. A human listener extracts the meaning of a sentence (or other acoustic cue) by gathering and processing, not only acoustic, but also linguistic information. Because of this, an impaired auditory system often copes very well in an environment where the speech presented is of high quality, and the listening conditions are good. A hearing loss in only part of the auditory frequency range may go relatively undetected in an informal speech test.

A reliable, controlled speech test, however, can give good predictions of the hearing thresholds in the mid-frequency region, and may provide important 'site of lesion' information. That is, normal, conductive and sensori-neural losses can be discriminated.

Speech, however, is a complicated acoustic signal, and although a subject may 'hear' the speech cue at a peripheral level, they may encounter difficulties in processing it at a neural level.

The identification and differentiation of central neural disorders cannot be readily achieved using the conventional speech test. Other more complicated tests are available, but these are often time consuming.

This chapter describes a modification of the conventional test which gives some speech processing information.

6.2 Measuring Speech Processing.

Many experimental studies of speech processing use isolated phonemes or nonsense syllables as their stimuli. The advantage of phonemes is their minimal semantic content, which allow any differences in the education or vocabulary of the subject to be eliminated. Tests done with phonemes, however, are difficult to score and awkward to carry out. A good compromise, as discussed in chapter 5, is to use monosyllabic words.

6.2.1 Consonant Confusion.

Much research has been done into the confusion of consonants in the English language, and the reasons for this confusion. For example, Miller and Nicely (1955) investigated sixteen heavily used consonants. Their work was undertaken in America, but the ideas are applicable to UK - English.

The phonemes (or consonants) used were spoken before the vowel 'a' [as in father] and a 16 X 16 confusion matrix of the consonant spoken against the consonant heard was examined. The 16 consonants used were p t k f θ s ʃ b d g v ð z ʒ m n. These symbols come from the international phonetic alphabet (see appendix B).

The authors discuss some interesting results which depend on the definitions of five articulatory classes of phonemes, and their acoustical characteristics. These classes are :-

1) Voicing: The vocal cords do not vibrate in the production of the consonants p t k f θ s ʃ and these are termed voiceless. They have noisy and aperiodic waveforms. The vocal cords do, however, vibrate in the production of b d g v ð z ʒ m n and hence, these are called voiced. Their waveforms are periodic.

2) Nasality: m and n are nasal consonants, they are created with lips closed, and the pressure being released

through the nose, giving a nasal resonance which is periodic and contains no noise.

3) Affrication: When the articulators are brought together and air forced between them, a turbulence is generated. This means that the turbulent sounds of f θ s ʃ v ð z ʒ can be distinguished from the stop consonants, p t b d g k (characterised by their preceding period of silence) and the nasal sounds m and n.

4) Duration: The extra duration of s ʃ z ʒ, and their intense, high frequency noise characterises the difference between these and the other consonants.

5) Place of Articulation: This can be front, back or middle, referring to the place in the mouth where the major constriction of the vocal passage occurs - p b f v m are front, t d θ s ð z n are middle, and k g ʃ ʒ are back. The acoustic information contained in these, however, is complex. The consonants b g d rely on the second formant frequency of the 'a'. If it rises, the consonant is b whereas if it falls, the consonant is g or d. On the other hand, p t k rely on the intense plosive sound which has high frequency in the case of t, and low frequency in the cases of p and k. Phonemes s z can be distinguished from ʃ ʒ because the former pair have high frequency energy, while f v and θ ð are distinguished by listening to the transition to the following vowel. It should be noted that these are difficult to distinguish, and depend a great deal on context, rather than acoustic information.

Using these classifications, Miller and Nicely report that voicing and nasality are much less affected by masking noise than the other classes. It is easier to distinguish consonants using the classes of affrication and duration than it is to distinguish them by place of articulation. In fact, the different places of articulation are the hardest class to distinguish and therefore the most easily confused. An interesting point is that although 'place' is difficult to hear, it is the easiest to see on the lips, which is of value to lip-readers.

6.2.2 Specialist Audiometric Tests of Speech Processing.

Several audiometric tests are available to test consonant confusion, phonemic error patterns and phonetic differentiation. One of the oldest of these is the rhyme test described by Fairbanks (1958). This uses a vocabulary of 250 monosyllabic words in 50 sets of 5 rhyming words e.g. hot, got, not, pot, lot. One word from each set is picked randomly, and presented to the patient, who fills in the first consonant on a score sheet. The score sheet for the above example set would read '__ot'. The advantage of this test is that the response word can be compared to the word which was presented, giving information on error type. The disadvantage is that the test only examines initial consonants.

A modification of the rhyme test, suggested by House et al. (1963), uses a multiple choice scoring method, allowing the subject to choose one word from 6 CVC (Consonant Vowel Consonant) words. This means that both initial and final consonants can be investigated.

A similar test has been proposed by Foster and Haggard (1979) called the FAAF (Four Alternative Auditory Feature) test, which uses a vocabulary of 80 CVC words in 20 sets of 4, e.g. bad, bag, bat, back. One of these words is presented to the subject who must then decide which one of the 4 they heard. It, also, is a forced choice test and therefore allows analysis of error type, in this case with the help of a microcomputer.

Results of this test show that, in general, the contrasts of <u>g</u>ay-<u>b</u>ay-<u>d</u>ay are sensitive to even a small amount of hearing loss, while <u>t</u>ick-<u>l</u>ick-<u>p</u>ick and <u>b</u>ay-<u>p</u>ay are easier to distinguish even in the presence of an impairment.

A development of this FAAF test uses a video presentation system to test, in addition to the above, the subject's lipreading skills.

The major disadvantage of all these types of test is that their execution times tend to prohibit their use for plotting the discrimination functions described in chapter 5.

6.3 Testing Speech Processing from Conventional Speech Audiometry.

The scoring mechanism used with the conventional speech test prohibits its use in gathering information on individual phonemes, since it scores on the basis of whole words. It is, however, a simple matter to score on an individual phoneme basis and hence gain information on the particular phonemes which the patient has difficulty hearing. The work discussed here is a modification of the conventional speech test designed, implemented and tested by the author to give speech processing information. As an example, consider the word 'CAT' again. Table 6.1 shows the scores using this amended phoneme scoring system.

Table 6.1 Phoneme Scoring System.

WORD	PHONEME SCORE	WORD SCORE
CAT	1 1 1	3
MAT	0 1 1	2
CAP	1 1 0	2
MAP	0 1 0	1
DOG	0 0 0	0

Each phoneme is scored 0-wrong or 1-correct, and the sum of scores for each of the three phonemes gives the word score.

6.4 Modifications to the ASCII Coded Messages.

To facilitate the kind of scoring mechanism described in section 6.3, the system must keep a record of which phoneme has been spoken and what score was given to it. To keep a tally of this manually is tedious, and graphing the results would be very complicated. It was decided to modify the ASCII coding system described in section 5.4 to monitor which phonemes were being spoken and therefore scored. Appendix B shows the Boothroyd wordlists and their phonetic components. The phonemes are taken from the word definitions in the Collins English Dictionary and are the symbols of the International Phonetic Alphabet. Many of the symbols used are English letters, and therefore have ordinary ASCII codes. The more unusual symbols are explained at the end of appendix B and do not have obvious ASCII codes.

To allow these characters to be printed on the PC screen, the font was edited so that some of the unprintable ASCII character codes were used. For example, the symbol ʃ was drawn in the font at ASCII code 212 so that when ASCII 212 is written to the screen, the symbol ʃ appears. This was done for all the unusual symbols shown in appendix B.

Unfortunately, these unprintable codes are greater than 128 and cannot therefore be sent directly to the stereo tape, since only 7 bits are sent. To get round this the ASCII code 126 is sent, followed by the required symbol's ASCII code. When the message is received, if it is 126, then 128 is added to the next ASCII character code received from the tape.

For example, ASCII 212 is equivalent to 11010100.

The code 126=1111110 is sent, followed by 212 which will be sent as 1010100.

When 126 is received, this signifies that 10000000 must be added to the next character received, which is 1010100.

This gives 11010100 equivalent to 212 which will be printed as ʃ.

In this way, the coded messages are adapted so that the word appears on the screen followed by its phonetic components, as in appendix B.

6.5 Scoring and Presentation of Results.

The same procedure as the test described in chapter 5 is adopted, but instead of scoring 0, 1, 2 or 3, the operator must score some combination of three 0's and/or 1's (see table 6.1). These are summed and the score for the conventional speech audiogram calculated and plotted as before.

Meanwhile two arrays are accumulated to allow the results of the phoneme analysis to be plotted at the conclusion of the test. One of these is a two dimensional array called phoneme and has dimensions 180 by 4. The other is a one dimensional array of 180 elements called phosco. The arrays have dimensions of 180 to allow an element for each word in the AB(s) wordlists (12 X 15). The first word spoken is element 0 in the array, ... , the seventh word spoken is element six and so on, and the two arrays are interlinked.

As an example, let us postulate that the fifth word spoken is the word 'SHIP', that it is presented to the subject's right ear at 50dB and that the patient's response to this word is 'THICK' giving a score 0 1 0. The array phoneme[180][4] keeps a record of the three phonemes spoken, and the speech level and ear to which they were presented, while, phosco[180] records the corresponding score.

The relevant elements of the arrays for the above example are as follows:

phoneme[4][0] = 212 = The ASCII code for phoneme ʃ
phoneme[4][1] = 199 = The ASCII code for phoneme I
phoneme[4][2] = 112 = The ASCII code for phoneme p
phoneme[4][3] = 150 = 50 (dB of hearing level)
+ (100 X left/right).
where left/right = 1 for right
= 0 for left
phosco[4] = 10 = (1^{st} phoneme score) +
(10 X 2^{nd} phoneme score) +
(100 X 3^{rd} phoneme score).

Once the conventional test has been completed and ODS, HPL and HPLE calculated, a phoneme chart can be displayed using the above two arrays. Figure 6.1 shows a phoneme chart. It graphs phoneme spoken along the x-axis and hearing level on the y-axis. The difference between a positive response and a negative one is indicated by the colours of the corresponding boxes on the chart.

On the PC screen, the phonemes not spoken are coloured the background colour of blue, those heard correctly are coloured yellow, while those heard incorrectly are coloured red. If a phoneme is spoken more than once at one particular hearing level, all the scores are taken into account. For example, if one phoneme is presented twice at 50dB, and heard correctly on one occasion and not on the other, half the corresponding box will be coloured red and the other half yellow.

6.6. Frequency Content of Phonemes.

Work has been done on the determination of the frequency content of individual phonemes, for example Fletcher (1929). This work, however, averages over a range of speakers since each speaker's voice is at a slightly different frequency.

In the case of the work undertaken by the author, the tape of the AB(s) wordlists is used in the test and it is

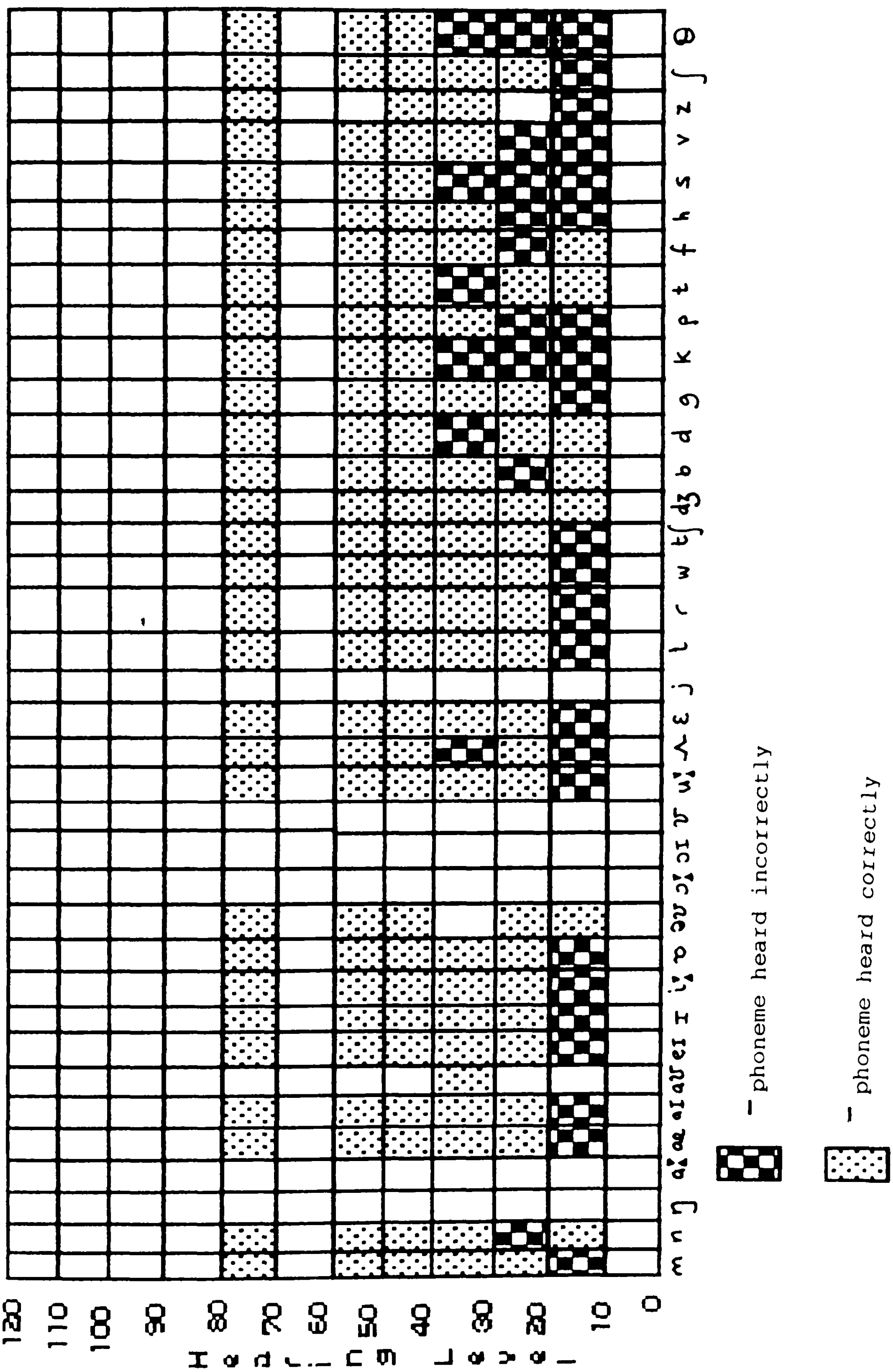


Figure 6.1 Phoneme Chart for a Normal Hearing Subject.

obvious that this means that the test is done with the same speaker no matter where it is played. It is interesting to consider the specific frequency content of each of the phonemes on this tape. It may then be possible to use this information in conjunction with the phoneme chart to give particulars on specific hearing loss. Although the work was done with the AB(s) tape, it is easily adapted to other wordlist recordings.

6.6.1. Measurement of Frequency Content.

A technique of digitization using an ADC was described in section 5.5.1 and each of the 180 words were digitized using this method and are stored on floppy disk.

Mathematically, the frequency content of signals is calculated by performing a Fast Fourier Transform (FFT). There are many algorithms available but the one used in this application can be found in Press et al. (1988). A program was written to accept the digitized word data from the floppy disk, store it in a 24576 element (3×2^{13}) integer array and display it on the PC screen (see figure 6.4). Any portion of this displayed waveform can then be selected and an FFT performed to give its frequency content.

To illustrate the FFT, consider figure 6.2 which shows the waveform of a 1kHz sine wave from a signal generator.

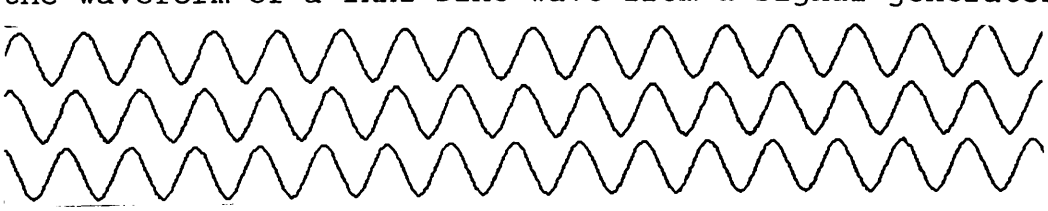


Figure 6.2 The digitized waveform of a 1kHz sine wave.

The corresponding FFT is shown in figure 6.3.

It shows that the entire frequency content of the waveform

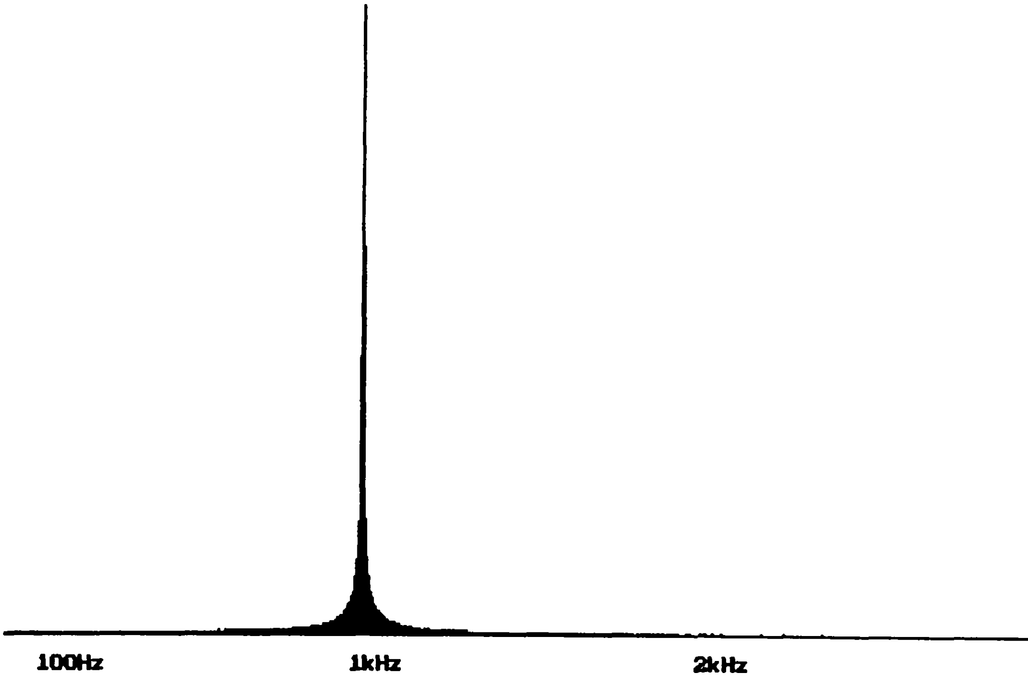


Figure 6.3 The frequency content of the 1kHz sine wave of figure 6.2.

is at 1kHz, as would be expected. In the ideal situation, the FFT should be zero in all channels except the one at 1kHz. It can be seen however, in figure 6.3 that the channels immediately adjacent to 1kHz have some content. This occurs because of the ending conditions of the wave used, that is, the portion selected for calculation of the FFT does not necessarily begin and end at a zero point on the waveform.

Unfortunately, the algorithm of Press et al. used with the TURBO C compiler allows the FFT to be calculated on a maximum number of elements of only 8192. This is because of a 64k limit which the C compiler puts on any one array.

By splitting the 24576 elements into three groups of 8192, carrying out the FFT on each of the three and adding the results together, a measure is gained of the frequency content of the whole word.

Let us again consider the word 'cheek' (see figure 6.4) which is made up of the phonemes tʃ - iː - k.

The distinction between these phonemes is clear from the waveform as shown, tʃ has high frequency, iː much lower frequency and greater amplitude and k is again high frequency and follows a period of silence.

Figure 6.5, 6.6 and 6.7 show the FFT's for the three phonemes.

Figure 6.5 shows the high frequency component of tʃ. It also shows clearly how noisy and aperiodic the waveform is, as is expected from section 6.2.1.

Figure 6.6 has the appearance of a low frequency pure-tone, although there is a small component at around 2kHz.

Figure 6.7 is again very noisy and contains much higher frequencies than iː. There is a large component around 1.8kHz.

The FFT's shown are all displayed in such a way that in each case the maximum intensity present is set at the maximum height on the screen. They are all accompanied by a multiplying factor so that their relative strengths can be compared accurately. Another way, however, of 'seeing' this is to look at the FFT for the whole word (see figure 6.8). It can be seen that the 5kHz signal which is a significant part of the tʃ phoneme is dwarfed by the intensity of the vowel, iː. It is therefore not surprising that patients with any high frequency loss, in particular the elderly, have difficulty in hearing these high pitched consonants like tʃ and k.

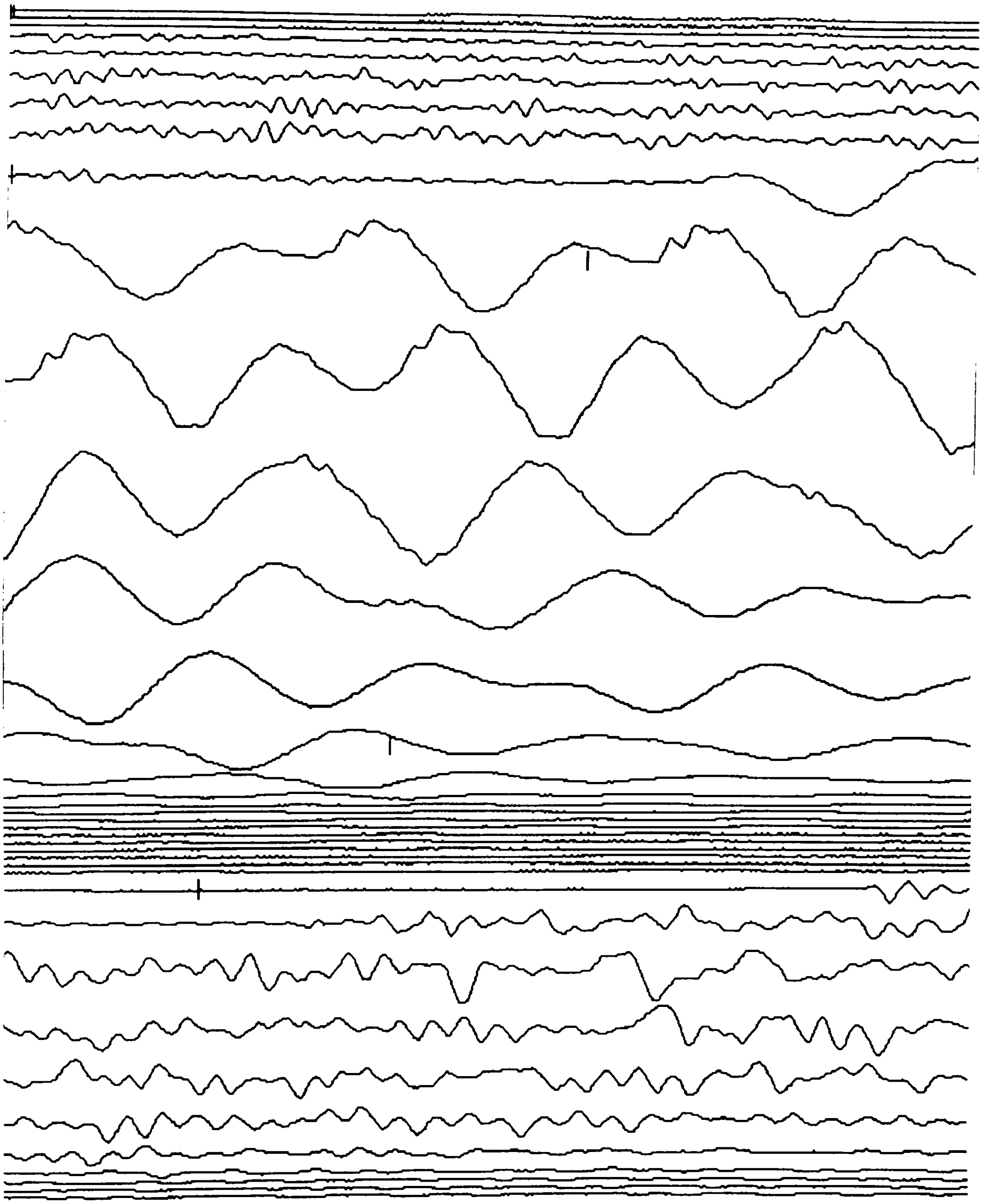


Figure 6.4 The waveform of the word 'cheek' - made up of the phonemes tʃ - iː - k. To give greater clarity the waveform is displayed on many lines.

SCALES:

0.016 seconds.

962mV.

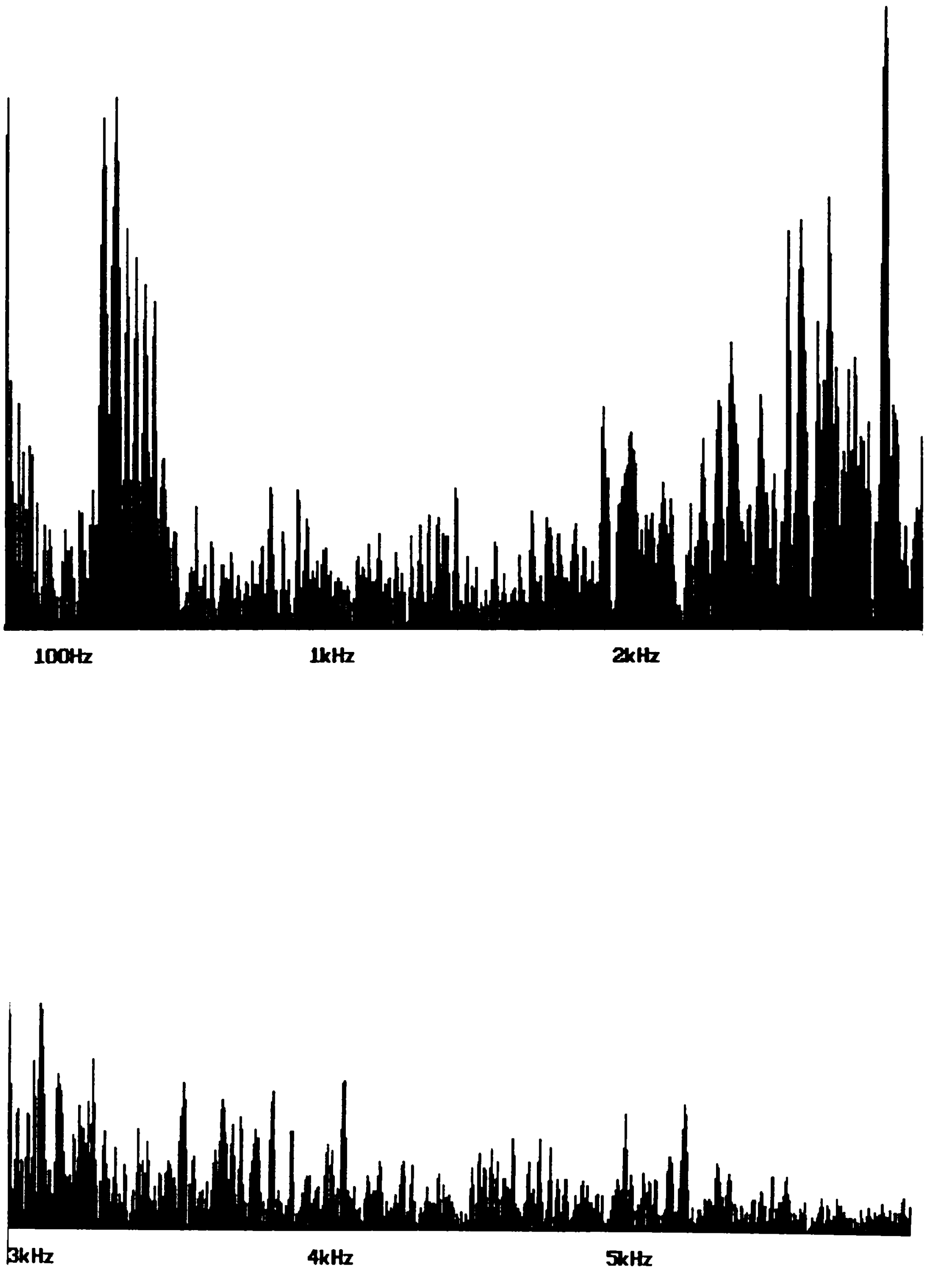


Figure 6.5 The frequency content of the phoneme tʃ.

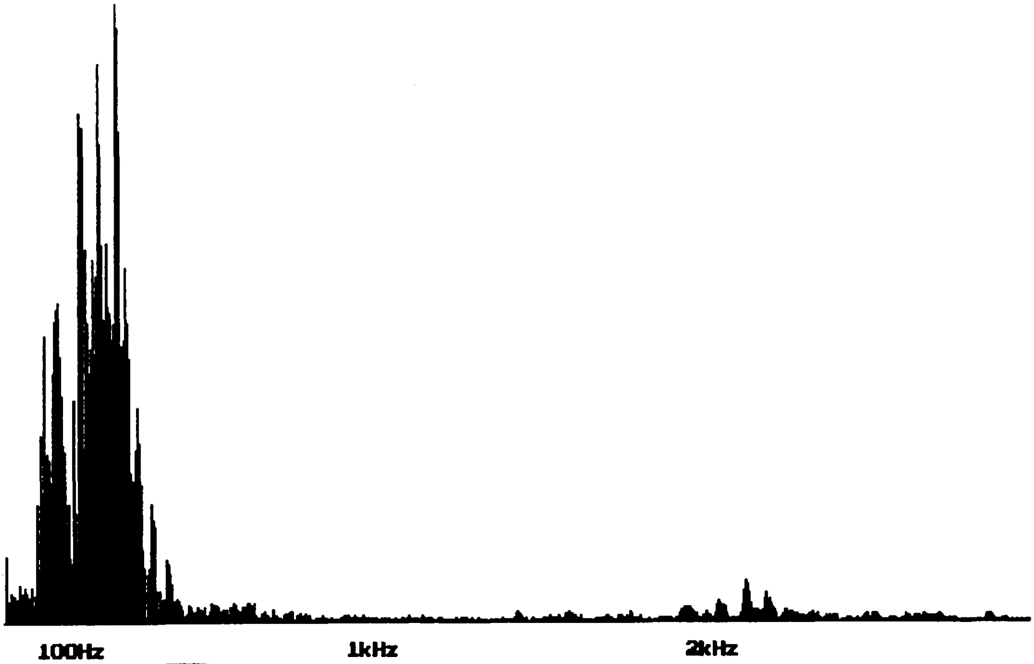


Figure 6.6 The frequency content of the phoneme iI.

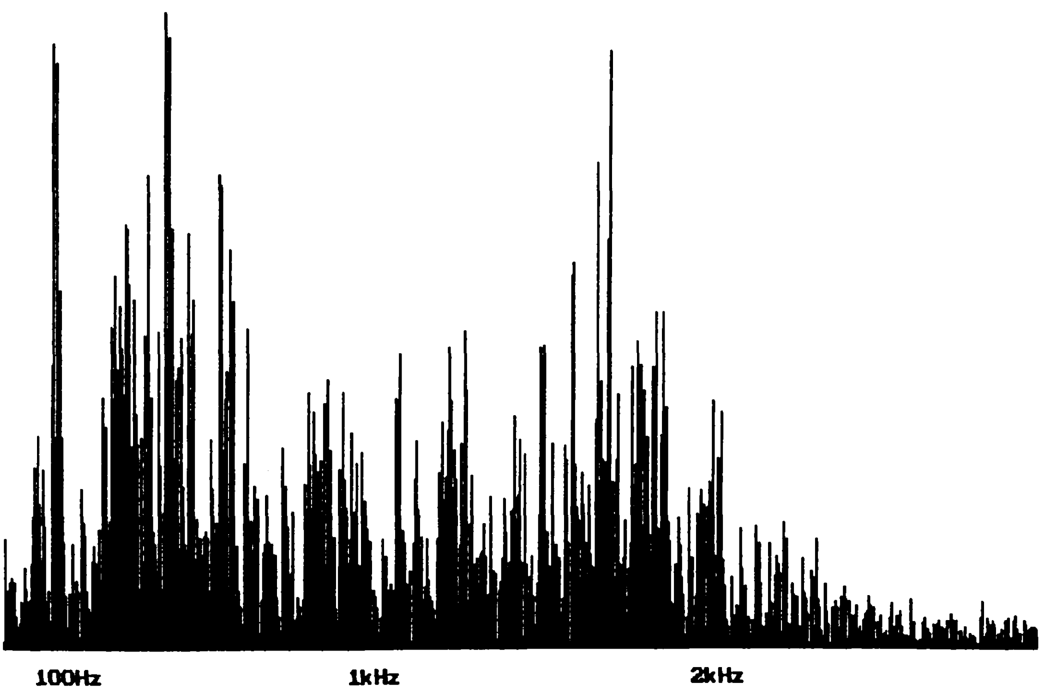


Figure 6.7 The frequency content of the phoneme k.

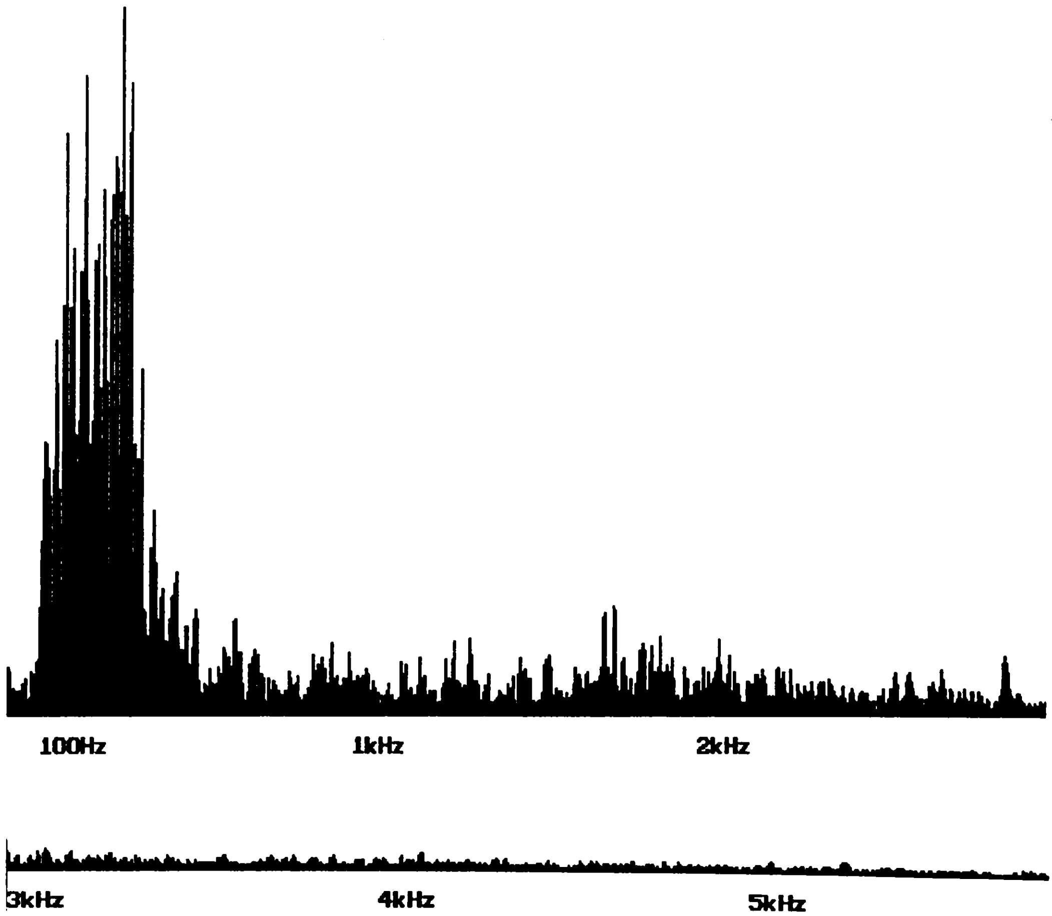


Figure 6.8 The frequency content of the word 'cheek'.

It is interesting to compare the frequency content of consonants which are often confused and which sound reasonably alike. Take as a first example, t and tʃ. Figure 6.5 shows the FFT of tʃ, while figure 6.9 shows the FFT for t. The phoneme tʃ has its largest contributions at 2.8kHz and 400Hz, whereas, from figure 6.9 it can be seen that 2.8kHz does not feature significantly in the FFT of t. It has it s main features at 200Hz and around 1kHz. Both graphs do, however, have a contribution just below 4kHz, but careful inspection of the FFT's show the differences in these. One of the most significant similarities is the broad spread of noise.

Another interesting comparison is that of the phonemes k, t and p which again are often confused and sound remarkably similar at the end of words, for example, sit,

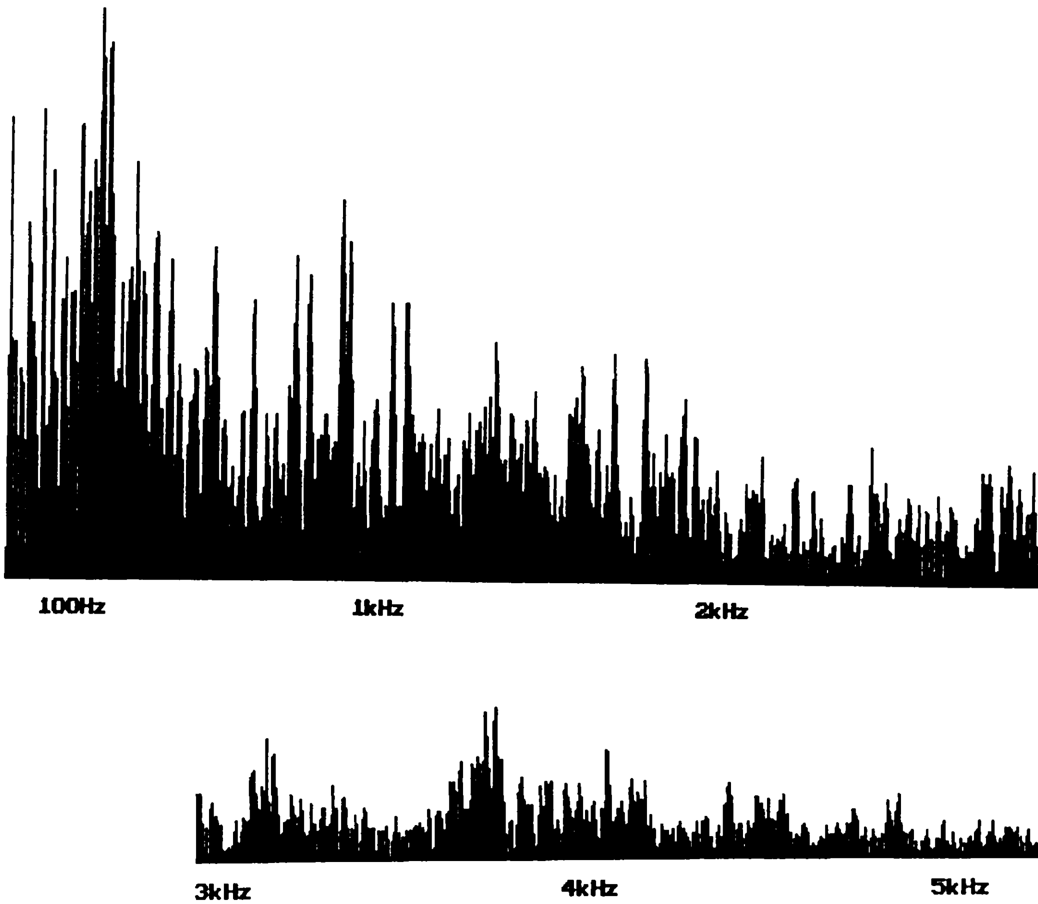


Figure 6.9 The frequency content of the phoneme t.

sip, sick. The frequency content of k and t have already been shown in figures 6.7 and 6.9 respectively and figure 6.10 shows the FFT graph for p.

The phoneme k has its main frequencies at about 400Hz and 1.8kHz and almost no significant components above this. t, on the other hand, has its most significant components around 200Hz, 1.9kHz and 3.8kHz, while p features frequencies of 90Hz, 1kHz, 1.3kHz and 4kHz. Each, has a frequency content which is just slightly different from the others and allows a 'fingerprint' to be made for each phoneme. It should be noted that this 'fingerprint' will be dependent on the speaker, and this is why it is so beneficial to look particularly at each phoneme on the Boothroyd tape. This tape is a standard from hospital to hospital and hearing test to hearing test, and so the frequency content results can be used directly with the phoneme hearing test results.

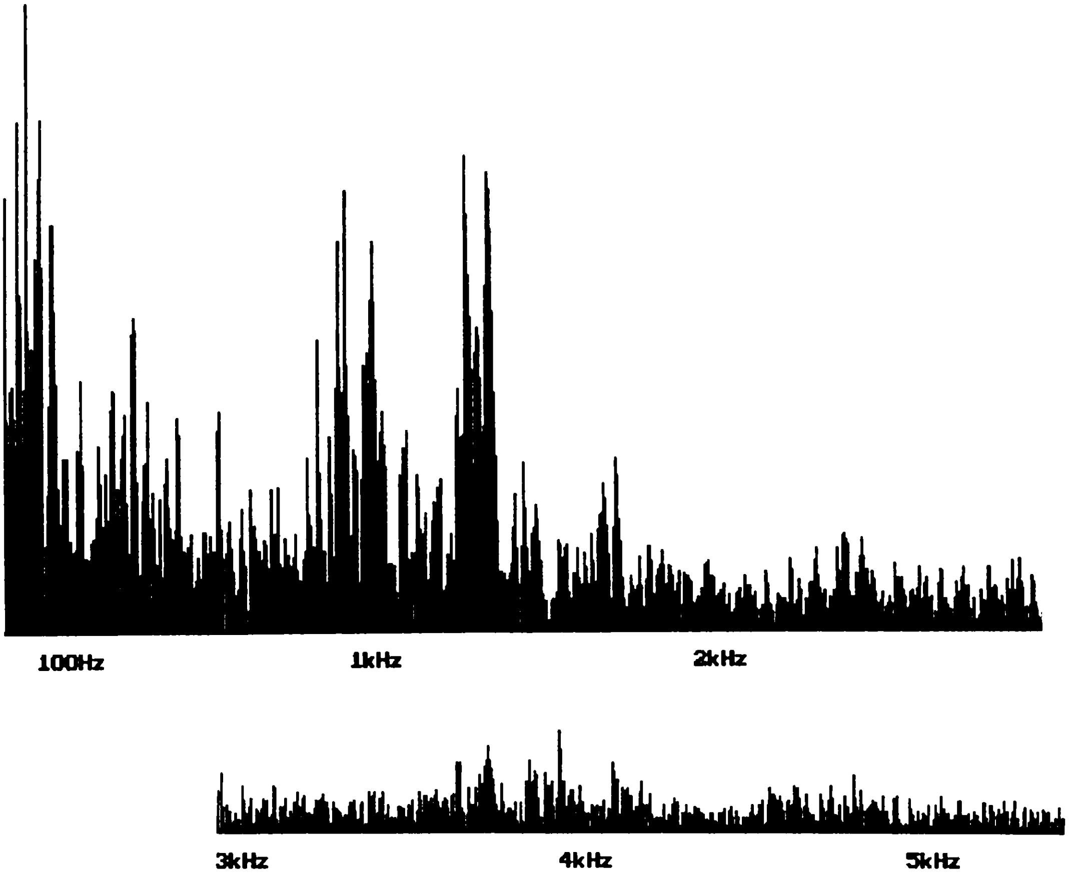


Figure 6.10 The frequency content of the phoneme p.

6.6.2. Phoneme Frequency Analysis - Results.

The analysis in section 6.6.1 was done for all the words and phonemes in the AB(s) wordlists. The accurate frequency analysis is therefore available, and the frequencies can be measured from the FFT graphs to an accuracy of about +/- 10Hz.

The statistical software package MINITAB was used. The frequencies present in each phoneme were measured from the FFT graphs and about 8 or 9 of the most predominant frequencies stored in MINITAB, along with their corresponding strengths. These values can then be used for further analysis. They are shown in Appendix D.

Although to gain an accurate picture of the frequencies of each phoneme, the FFT graph should be consulted, it is impossible to contain all these 540 graphs in this thesis. The values in Appendix D are the author's interpretation of the most important features of these graphs.

It is, however, interesting to group all the occurrences of each phoneme and check that the same frequencies appear each time. Using MINITAB, a dotplot or barchart was plotted for each phoneme. These charts 'bin' the frequencies present in each phoneme. It is clear then that the bins with the biggest content are the most predominant frequencies for each phoneme. The size of each bin was 120Hz, chosen because of the maximum frequency present in any one phoneme (about 5kHz).

Table 6.2 shows these most predominant frequencies. The bins are labelled by their maximum value. The error on these frequency values is therefore +/- 60Hz, which seems high especially at the lower frequencies. It is, however, valid to consider these since it is believed that the kinds of defect found in the human hearing mechanism mean that the slope of the hearing loss is not high, and 60Hz is therefore not significant.

Obviously, if more accurate frequency readings are required, the operator can refer back to the FFT results.

Another interesting feature to consider is the overall impression of whether a phoneme has a high or low average frequency. For this we measure a weighted average frequency which takes into account, not only the frequency of each phoneme, but also its intensity:

$$\frac{\sum(freq\ X\ height)}{\sum(height)}$$

These results are shown in table 6.3 and are in ascending order of this average frequency, with 'w' as the lowest and 'tʃ' the highest. It should be noted, however, that although these values give a good indication of the frequencies present in each phoneme, the frequencies listed in table 6.3 do not actually appear in the frequency spectrum of the corresponding phoneme. That is, 187Hz may not be present in the frequency spectrum of the phoneme 'w'.

Table 6.2 Most predominant frequency 'bins'.

Phoneme	Most predominant 'bins' (in Hz).
b	360 480 1200
d	240 360 480 1200
f	120 960 1800 3600
g	240 600 1080
h	120 2040
j	300 600 2400
k	120 600 4080
l	240 360 480 600
m	360 1200
n	440 1200
p	240 960 4000
r	360 600 840
s	120 360 4200 5280
t	600 2040 4000
v	240 600 1200
w	240 360 480
z	240 360 5000
aɪ	920 1200

æ	360 1200
aɪ	1120 480 240
aʊ	240 720 1440
eɪ	360 840 1800
ɪ	480 1800
iː	480 720 2400
ɐ	960 1200
əʊ	420 600
ɔː	840 600
ɔɪ	880 360
ʊ	480 1080
uː	480 600 1200
ʌ	840 1080
ʃ	2160 5280
tʃ	2280 5640
dʒ	120 600 5000
θ	120 4200
ŋ	240 360 600
ɛ	400 1920

Table 6.3 Weighted Average Frequencies.

Phoneme	Weighted Average Freq. (in Hz)
w	187
r	209

aʊ	232
v	248
n	267
m	288
ŋ	355
uː	392
iː	393
ɛ	410
j	414
ɪ	445
d	501
eɪ	511
b	523
f	527
ɔː	543
l	544
əʊ	555
ʊ	569
g	587
z	593
ɐ	593
ɔɪ	630
æ	673
θ	693
aɪ	699

aɪ	728
ʌ	763
dʒ	905
h	1027
s	1100
p	1149
t	1372
k	1753
ʃ	2301
tʃ	2688

Care must therefore be taken not to give these values more significance than they deserve. They are simply a useful guide and should be used only to gain a superficial idea of the differences between phonemes. A more accurate picture should then be obtained from table 6.2 and appendix D.

6.7 Experimental Results - Phoneme Charts.

The best way to examine this type of testing is to undertake clinical trials on a wide range of subjects suffering from varying types of hearing loss, particularly sensori-neural type losses. The results of these trials can then be analysed and compared with existing procedures for obtaining speech processing information. Unfortunately, the facilities to undertake clinical trials were not available and these could not, therefore, be carried out. It was decided, however, to conduct very limited trials on a handful of subjects with some sort of hearing loss.

Five subjects were tested. For each, a pure-tone audiogram was accumulated, followed by a speech test which also produced a phoneme chart like that shown in figure 6.1. One of the subjects had normal hearing, two had conductive type losses (one very mild and the other quite severe at low frequencies), and two had sensori-neural type losses, in both cases only in one ear and again one was a mild loss with an ODS of 90%, while the other was reasonably severe and had an ODS of 73%.

In the case of the normal hearer, as would be expected, at low intensities there is an even spread of the phonemes which are and are not heard, while at high intensities all the phonemes are heard (see figure 6.1). This same pattern is displayed by the subject with the mild conductive loss. The subject with severe low frequency conductive loss shows a similar pattern, although at high intensities there are a few phonemes which were not heard - n, ʌ, ɛ, d, f. From table 6.3 it can be seen that these phonemes are all around 200-500Hz, which is the area of greatest loss on the pure-tone audiogram. If a pure-tone audiogram had been unfeasible, e.g. the patient had found it impossible to respond to pure-tone stimulus, the speech phoneme chart would have indicated low frequency loss.

The sensori-neural losses are interesting, as many low intensity phonemes are not heard, those of medium intensity are, and at high intensity many are not heard again. This, of course, leads to the characteristic speech audiogram shape for a sensori-neural loss.

The subject with mild sensori-neural loss misheard the phonemes m, n, r, b, d, g, f, v, θ at high intensities. Of more interest, however, are the results of the subject with severe sensori-neural loss, and these will be discussed in more detail.

Figure 6.11 shows the pure-tone audiogram for this subject. This graph shows that the left ear has reasonably good hearing (except for a small dip at 4kHz), while the right ear thresholds tail away quite steeply. This right

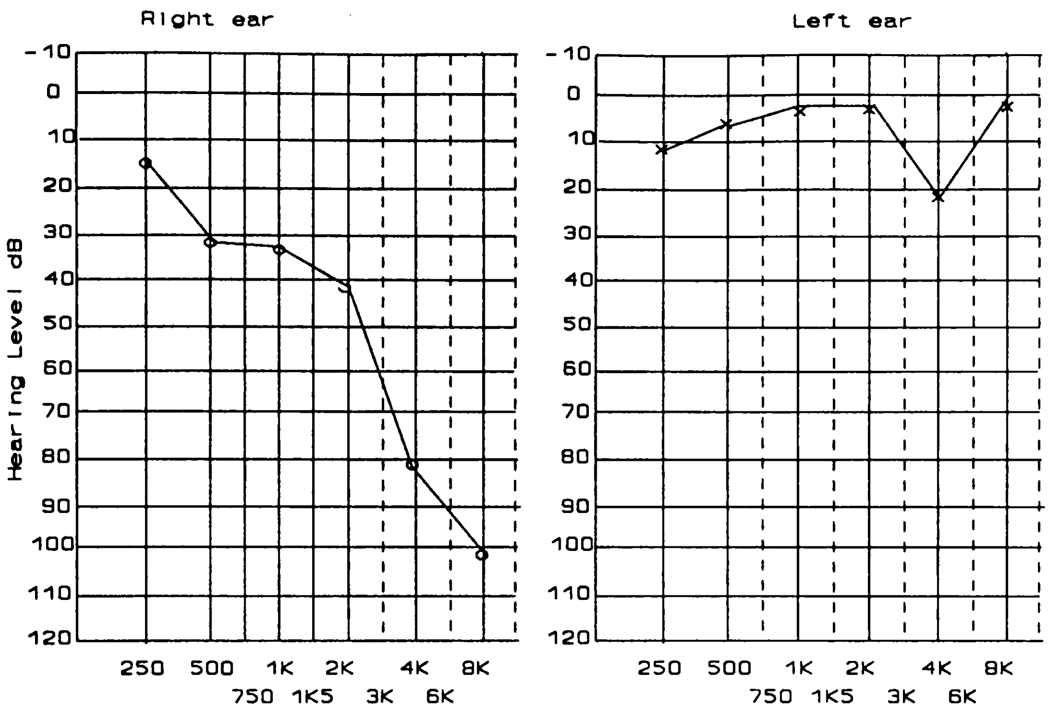


Figure 6.11 Pure Tone audiogram - subject has a sensori-neural loss in the right ear.

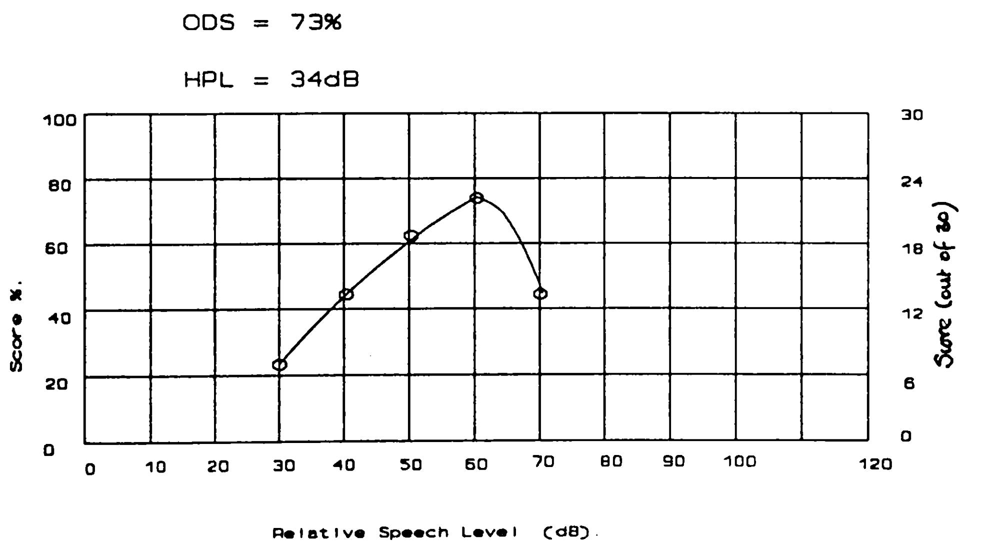


Figure 6.12 Speech audiogram of same subject as figure 6.11 (right ear).

ear is characteristic of presbycusis or old age and shall be looked at more closely.

Figure 6.12 shows the corresponding speech audiogram which demonstrates the characteristic high frequency ski-slope curve of a sensori-neural loss. Figure 6.13 shows

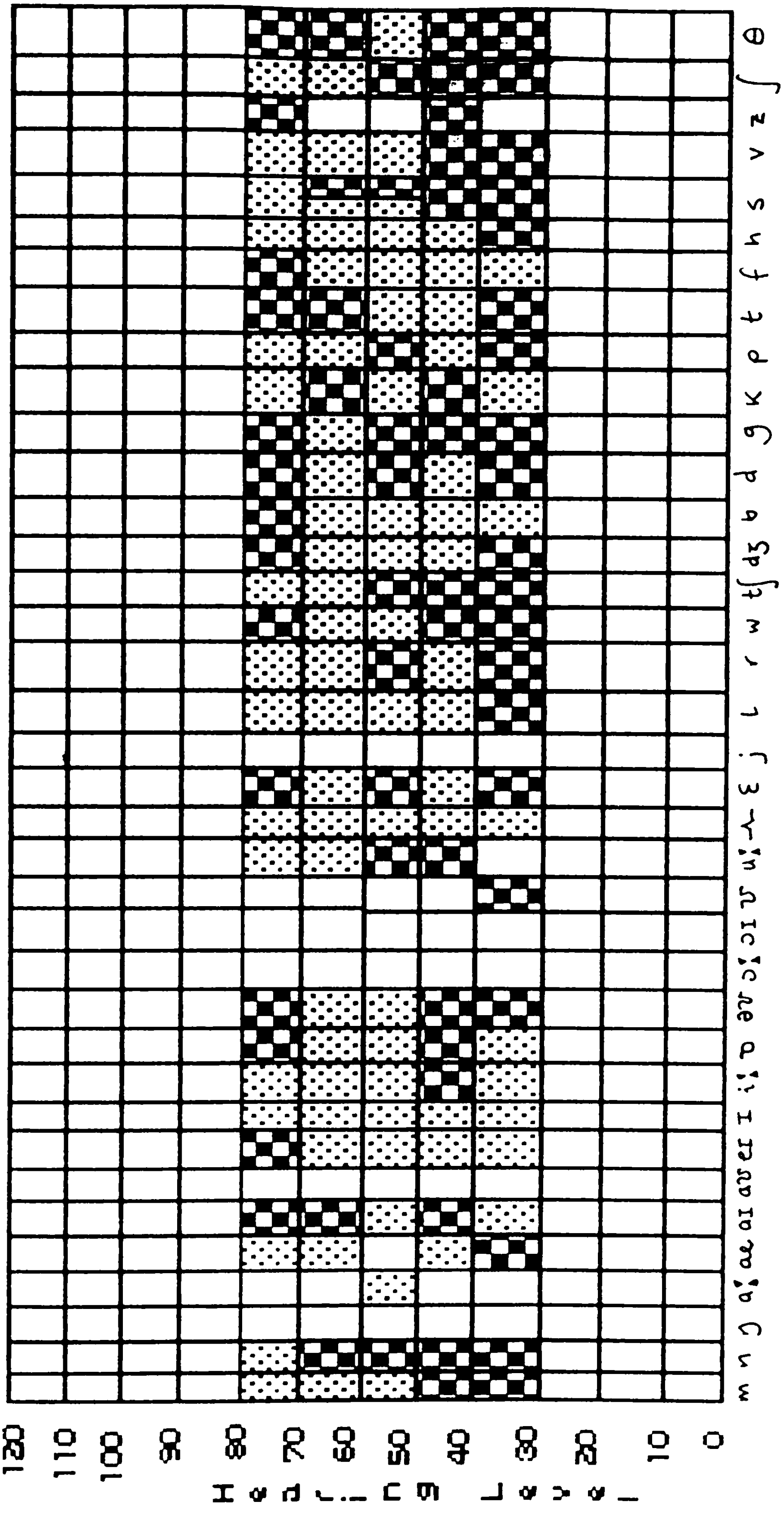


Figure 6.13 Phoneme chart (right ear) for subject of figure 6.11 and 6.12.

the phoneme chart. The phonemes that were not heard can then be investigated using table 6.2 and appendix D.

It can be seen from table 6.2 that the phonemes t, f, z, θ, and dʒ all have a frequency content in the range 3.6-5 kHz and, of course, this is the frequency range at which the hearing starts to fall off rapidly. It is clear therefore why the subject mishears these phonemes. The consonants w, b, d, and g are also missed. These have their major components at low frequencies, but all have a small component at about 1-1.2 kHz. Although the subject has only a small loss around this frequency, it is possible that, because this is important for distinguishing these phonemes, and because they have such small intensities, the subject misses them.

Finally, the subject mishears some vowels, eʊ, aI and eI. It is harder to decide why these are not heard, since these FFT's are swamped by low frequency sounds which the subject should hear. Careful inspection of appendix D, however, shows some low intensity components at 2-2.5 kHz. Again, since the hearing curve of figure 6.11 starts to fall off at about these frequencies, the subject may not hear them and hence mishear the vowel.

Although the left ear does not have as much hearing loss, it is very interesting to note the slight dip at 4kHz. The phoneme chart shows that most phonemes are heard, but there is evidence that the subject has difficulty with s and θ, both of which (from table 6.2) have a component at 4.2kHz.

It would be of interest to undertake a large clinical trial and determine whether pure-tone audiogram, and phoneme chart results can be matched up, so that, in the case of difficult to test patients who cannot respond to pure-tones, the information of the pure-tone audiogram could be obtained accurately using speech.

It is interesting to compare these consonants not heard by both the sensori-neural loss subjects, with the results of these consonants on a consonant confusion matrix such as that described by Miller and Nicely (1955). It is found that they are all reasonably high scoring on the matrices and so would be expected to cause confusion.

It would appear then that this type of test does give some indication of speech perception as well as giving speech discrimination information.

6.8 Conclusions.

This type of phoneme testing shows promise. It does give some speech processing information although its disadvantage is that it simply measures what was not heard. There is no attempt made to take into account what was heard in its place. This would allow confusion matrices to be calculated. It is, however, very difficult to score this type of test without insisting on a closed set of responses like, for example, the rhyme test.

The test does, however, offer the advantage that it takes the form of a normal speech test, uses the available test material and measures values for ODS, HPL, and HPLE, in addition to giving speech processing information.

As a byproduct of this test, extensive frequency content information has been gained for the Boothroyd wordlist recordings and can be used in conjunction with the phoneme results. The technique of gaining this information can be applied to any of the wordlist recordings available; it is simply a matter of digitizing the waveform of each word.

It would be very interesting to see extensive clinical trials of this test as compared to other tests designed to give speech processing information, but the initial results look encouraging since they do give information consistent with the pure-tone audiogram. It is also of great benefit that this speech processing information is obtained by doing a conventional speech test and not an extra test.

CHAPTER 7: Tympanometry

7.1 Introduction.

The audiometry discussed up to this point has been subjective, that is, it has required some feedback from the subject to determine whether or not they have heard the stimulus. This is not ideal. It would be superior to measure the hearing disability objectively without consulting the subject.

One form of objective testing is tympanometry which measures the acoustic impedance of the eardrum or tympanic membrane. This can give information on many types of hearing loss, but particularly those concerned with the middle ear. It involves directing the sound into the ear canal towards the eardrum and measuring the reflected signal to determine the amount of sound transmitted. This, in turn, leads to a measure of the acoustic impedance of the ear canal and drum, and because the impedance of the canal is reasonably constant from patient to patient, a compensated impedance can be calculated for the impedance of the eardrum.

Tympanometry is in fact more specific than this. It is the name given to a measure of the acoustic impedance of the outer and middle ear systems as the pressure in the ear canal is swept from positive to negative or vice versa. A graph of acoustic impedance against pressure is known as a tympanogram.

A third investigation that is done in this field is the acoustic reflex test. This graphs the acoustic impedance of the system as a function of time and looks at the recovery of the ear to the presentation, to either ear, of a loud impulsive stimulus. When the ear is presented with a loud sound, the muscles of the face contract to tighten the tympanic membrane. This is the ear's 'safety valve' to avoid damage.

Collectively these tests are known as acoustic immittance tests. Immittance is a term used in this field to describe either impedance or its reciprocal admittance.

7.2 Requirements for a Tympanometer.

Tympanometry has been performed routinely in clinics for over 20 years. Historically it was executed using a 220Hz probe tone which was sounded continuously in the ear canal and the corresponding reflected signal measured as the pressure was swept up or down. Later tests incorporated another probe tone of frequency 660Hz.

When tympanometry is performed at low probe tone frequencies, the tympanogram reflects predominantly the stiffness controlled components of the admittance and much less information is gained on the mass controlled elements (e.g. the ossicles) which would dominate the graphs at high frequency probe tones. Although the 660Hz probe tone gives more information on ossicular conditions, recent research has shown the benefit of using even higher frequencies, and in particular of performing multi-frequency tympanometry, e.g. Margolis et al. (1985).

Van Camp et al. (1983) stressed that high frequency probe tones used with machines capable of measuring conductance and susceptance (the real and imaginary parts of admittance) were of great value in detecting ossicular discontinuity. In particular, high frequency tympanograms were found to be distinctly abnormal in cases of incudostapedial joint pathologies.

This work involved the accumulation of many conventional tympanograms at a variety of frequencies. Other research by Funsaka et al. (1988) has held the pressure constant and swept the frequency.

Although work has been done on multi-frequency probe tones and their clinical use has been proven, there are few commercially available instruments which allow this type of test to be done.

It was decided to design a tympanometer which would use the PC and therefore allow a variety of tests to be carried out and displayed. By interfacing this to the existing ASRA audiometer, the ASRA can provide acoustic reflex stimuli.

Firstly the general requirements of any audiometer are discussed. These are taken from the recommendations of ASHA (1988).

Immittance instruments must have the following five components:

1) A probe which can be sealed in the ear canal via a probe tip and which contains:

a) a loudspeaker to produce the probe signal,
b) a microphone which monitors the sound pressure in the ear canal,
and c) a port for a pneumatic system.

2) A pneumatic (or pressure) system used to vary the air pressure in the sealed ear canal.

3) An acoustic immittance measurement system which converts the relation between the loudspeaker and microphone voltages into immittance units.

4) A method of delivering acoustic reflex stimuli, either pure-tone or noise. There must also be some way of delivering these stimuli to both ears. The ear which does not have the probe in it is known as the contralateral while the one with the probe is the ipsilateral.

5) A recording device e.g. a meter, an XY plotter, or a computer.

Tympanometers are then classified into 'types' depending on the particular features which they offer.

Type 1: must allow measurement-plane and compensated static immittance measures (measurement-plane results include the immittance of both ear canal and drum, while compensated values adjust to give only the value at the drum). Tympanometry must be possible under manual **and** automatic control of pressure, and contra- and ipsilateral measures of acoustic reflex must be attainable. The acoustic reflex stimuli can be either pure-tone or noise.

Type 2: allows compensated static immittance and tympanometry under manual **or** automatic control of pressure. Also ipsi- or contra- lateral reflex with pure-tone stimuli must be possible.

Type 3: allows static immittance, tympanometry and the monitoring of an acoustic reflex at one specified stimulus level.

Type 4: are special purpose machines and allow one, or a combination of the above measures to be made.

The instruments are still further classified by the type of measurement made:

a) impedance meters,

b) admittance meters,

c) admittance and phase meters,

d) conductance and susceptance meters,

or e) instruments with non-sinusoidal probe signals.

The work done here uses a measure of admittance but also monitors phase. Because admittance Y = (1/impedance) = 1/Z, it is directly proportional to a constant peak volume velocity and inversely proportional to the corresponding sound pressure. That is

$$Z = \frac{p}{\mathbf{v}}, \quad \textit{while} \quad Y = \frac{1}{Z} = \frac{\mathbf{v}}{p}.$$

This means that the peak volume velocity is directly proportional to the rectified loudspeaker voltage and therefore gives a measure of the magnitude of the admittance at the probe tip. In this instrument the peak sound pressure is maintained at a constant level so that the SPL in the ear canal is held fixed. This is done via a feedback circuit between the microphone and loudspeaker which adjusts the voltage delivered to the loudspeaker to maintain a constant voltage at the microphone. This type of feedback is known as automatic gain control (AGC).

The voltage at the loudspeaker then gives a measure of admittance Y. Since the phase difference Φ between microphone and loudspeaker is also monitored and because Y=G+iB, we can calculate conductance G, and susceptance B, from:

$$Y = \sqrt{B^2+G^2} \qquad \Phi = \tan^{-1}\frac{B}{G}$$

The instrument will undertake multi-frequency tympanometry and allow Y, B, G and/or Φ to be graphed.

7.3 Hardware Development.

The instrument that was designed and built had to incorporate many of the features described in section 7.2. It allows measurement of admittance Y, and phase Φ, which permits the calculation of G and B. It produces the frequencies that are output to the loudspeaker in the probe and it measures the sound received at the microphone. It controls the generation of pressure in the sealed ear canal and measures this pressure.

The prototype system contains several individual, but interlinked elements, as shown in figure 7.1.

The tympanometer board incorporates the automatic gain control (AGC) circuit which controls the sounds generated at the probe. This board will be discussed in more detail in section 7.3.1, but the most important function of the AGC circuit is that it generates a digital number proportional to the acoustic admittance at the probe tip.

The interface between the PC and the tympanometer board is achieved via a MC68008 microprocessor and a number of related devices. The prototype uses a Motorola EXORmacs system to assemble and link an assembly code control program which can be downloaded via a single board 68000 development module, the KDM board. In the final version,

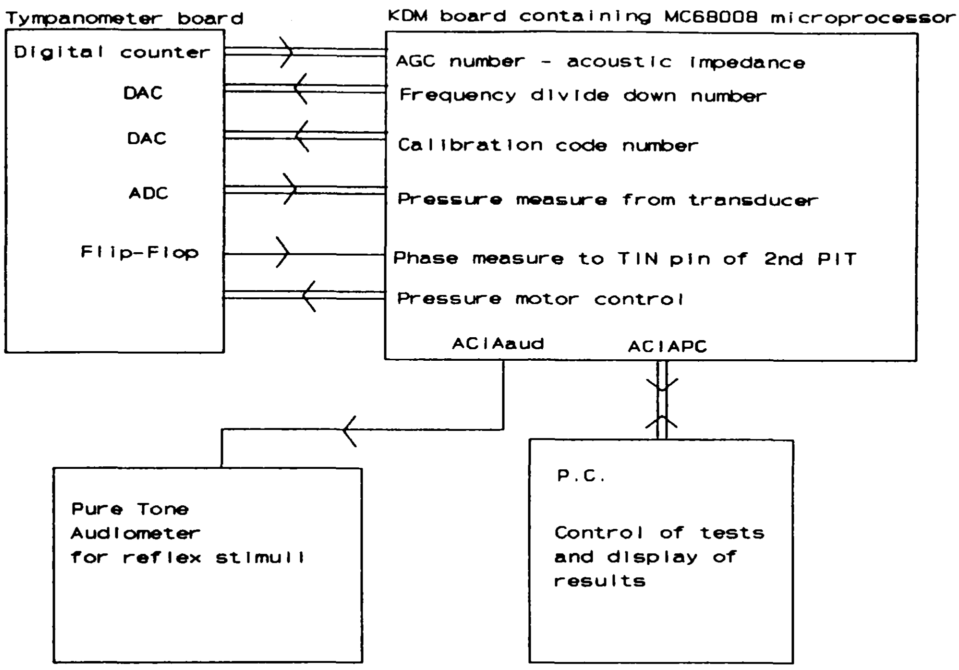


Figure 7.1 Schematic layout of Prototype System.

the assembled and linked program will be run from EPROM. This program allows the 68008 microprocessor to interact with a number of latches which permit communication with the tympanometry board via a 50-way connector. The processor uses a Peripheral Interface/ Timer (PI/T) chip (MC68230) to generate interrupts every 1ms which allow the values generated on the tympanometer board to be read and sent to the PC. There are two Asynchronous Communications Interface Adapters (ACIA MC6850) on this KDM board. One allows communication with the serial port of the PC, while the other communicates with the ASRA audiometer which can then be used to generate acoustic reflex stimuli. For convenience, the term 'KDM board' will be used to describe, in addition to the KDM module and processor, the board containing the latches, PI/T's and ACIA's.

7.3.1 Tympanometer board.

Figure 7.2 shows a block diagram of the completed tympanometer board. The detailed electronics are not shown to avoid complication.

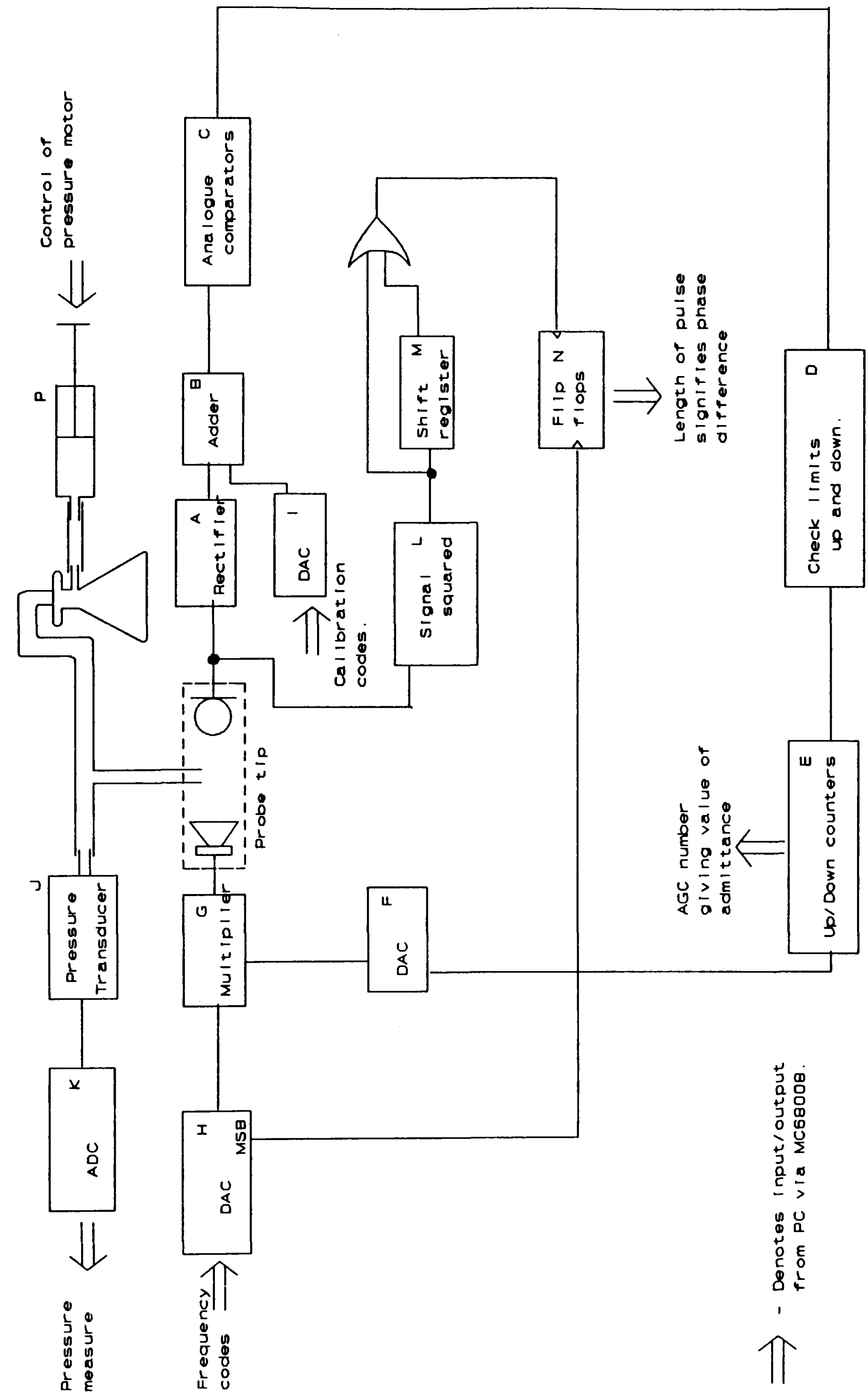


Figure 7.2 Block Diagram of Tympanometer Board.

NOTE: The need for the calibration voltage comes from the variation in response of the ear canal, eardrum and microphone with frequency. The voltage measured at the microphone (to maintain 85dB SPL in the ear canal) is not constant over frequency and it follows that the voltage required at the comparator for equilibrium will not be constant over frequency.

The calibration voltage ensures that if the SPL in the ear canal is 85dB then the voltage at C will be 2.45V and the AGC circuit will therefore be in equilibrium.

The circuitry represented in figure 7.2 is designed to interface to the KDM board and subsequently to the PC. The wide arrows on the diagram signify this communication.

The loudspeaker produces a sound that is received at the microphone (both loudspeaker and microphone are housed in the probe). The aim of the circuit is to maintain this microphone signal at a constant voltage level corresponding to 85dB SPL in the ear canal. This microphone signal is AC coupled, amplified and rectified (at A) to produce a DC ripple voltage level proportional to the signal at the microphone. This voltage level is compared (at C) to two voltage levels, one of 2.3V and the other of 2.6V. The circuit operates because, if the voltage at the microphone is too large, the voltage at C will be greater than 2.6V and the comparator will set the next part of the circuit to count down and therefore reduce the loudspeaker voltage. At equilibrium then, the voltage at C should be 2.45V (midway between 2.3 and 2.6) and the DAC (at I) and adder (at B) add a calibration voltage to the standard equilibrium voltage measured at A, for each frequency, to ensure 2.45V at C. These calibration numbers are measured using a 2cc coupler in conjunction with the probe tip and a sound level meter. The calibration number at I is varied until the voltage at C is 2.45V, making sure that the SPL in the 2cc coupler is 85dB. The output of comparator C then instructs the digital up/down counter(E) whether to count up or down. This will increase or decrease the loudspeaker voltage. To ensure that the counter does not overflow at either hexadecimal 0 or hexadecimal FF, these limits are checked at D.

At E there is a digital number which is proportional to the loudspeaker voltage. This digital number is also proportional to the admittance. Using the DAC (F), a voltage proportional to this digital number is used as the multiplying voltage for the analogue multiplier at G. The sine wave for this multiplier is obtained from the KDM board, and originates in a table of sine values which will be explained in detail in section 7.3.2. The multiplied voltage at G is sent to the loudspeaker closing the loop and giving AGC. The circuit at L, M and N is used to calculate the phase difference and will be explained in section 7.6.

The pressure is produced via a syringe at P which is driven by a small motor. The reservoir is for safety, to avoid very large pressures, or sharp changes in pressure being presented to the ear. The pressure in the ear is monitored by a differential pressure transducer (an LX1601D) at J, whose output is digitized by the ADC (K). The reason for using this transducer was that it was small, allowing it to be placed in the probe.

7.3.2 MC68008 and KDM board.

An assembly code program was written to interface the tympanometer board and the PC.

The sine waves are produced via a ROM which contains 64 steps of a sine table. The different frequencies of sine waves are produced by clocking the table of sine values at varying speeds. The crystal on this board has a frequency of 3MHz which was carefully chosen so that the frequencies 220Hz and 660Hz were possible.

220Hz is produced by clocking at 3MHz/64/**213**, and

660Hz is produced by clocking at 3MHz/64/**71**.

The important quantity is the last number in each case which can be set by the assembly code. Multiple frequencies are available and are produced by changing this last divide number. For example 3MHz/64/**312** gives 150Hz, while 3MHz/64/**23** gives 2038Hz.

Appendix E shows the assembly code programs TYMP5.SA and TYMPAUD.SA. TYMP5 is the program used to control a multi-frequency scan of admittance versus pressure and also measures phase. TYMPAUD on the other hand is a test routine to generate an acoustic reflex stimulus from the ASRA audiometer and will be discussed in section 7.8.

The MC68008 allows interface with the tympanometer board via a number of latches and the communications with the PC and audiometer are obtained via two serial ACIA chips, named ACIAPC and ACIAUD respectively.

Turning to TYMP5, the main program starts at the label RESTART. The preceding code simply sets up the input/output facilities and ports. The main loop obtains the frequency code from a table and outputs it to the corresponding latch which will send it to the DAC on the tympanometer board. The frequencies start at 150Hz and increase in $1/6^{th}$ of an octave steps to about 2kHz. Similarly the calibration number is sent to its corresponding DAC. The program loops continuously to START (or RESTART if the end of the frequency table has been reached). The frequency divide numbers are contained in TABL, while the calibration numbers are in TABL2. The main program remains in this infinite loop writing the frequency and calibration numbers to their respective latches and hence DAC's. Meanwhile, there are two interrupts possible, a timer interrupt from the 1st PI/T at level 2, and a port interrupt from the 2nd PI/T at level 1. The level 1 interrupt allows the phase calculation to be computed and will be discussed in detail in section 7.6.

The interrupt service routine TIMEINT services the level 2 interrupt which sends all the data measured to the PC, via ACIAPC and the PC's serial port. This data includes 8 bits which are the AGC number proportional to the admittance, 10 bits that are a measure of the pressure, and 8 bits for a phase number. The values sent to the PC are the ASCII codes for the hexadecimal values measured; for example if the AGC number obtained was hexadecimal 3A, then $33 and $41 would be sent (where $ signifies hexadecimal). These would be read at the PC as character 3 and character A. The subroutine ASCII does this translation to ASCII and sends the characters to ACIAPC. There must be a delay between these characters to allow the PC time to receive them.

The character string sent is

$$'a\ x_{agc}^{1}x_{agc}^{2}\ p\ x_{p}^{1}x_{p}^{2}x_{p}^{3}\ b\ x_{ph}^{1}x_{ph}^{2}\ c'$$

The a signifies the start of the AGC data and x_{agc}^1 and x_{agc}^2 are the hexadecimal ASCII codes representing the eight bits of the AGC number, p signifies the start of the pressure 10 bits x_p^1, x_p^2 and x_p^3, b precedes the phase data which is 8 bits x_{ph}^1 and x_{ph}^2 and c terminates this set of data. In this way the data can be sorted out by the PC once it has been received.

7.4 Tympanograms.

A program was written for the PC to accept the data accumulated on the tympanometer board and display it on the screen.

The tympanic membrane or eardrum can be thought of as a small drum which is less than a centimetre in diameter. It is stretched in such a way that it allows maximum transmission of sound to the middle and inner ears. It is easy to imagine that if this membrane is put under pressure of some sort, it will stretch and bow. That is, it will become tighter and therefore reflect more sound back into the ear canal and transmit less to the middle ear (see figure 7.3).

If there is some problem in the ear, for example, the eustachian tube is not functioning correctly or the ossicle bones are jammed in some way, then the eardrum will not behave normally under pressure and it is these abnormalities that tympanometry is designed to detect.

Figure 7.4 shows the tympanogram of a normal ear carried out with a probe tone of frequency 660Hz.

The admittance values of the y-axis are not calibrated and are given simply as the values of the corresponding AGC

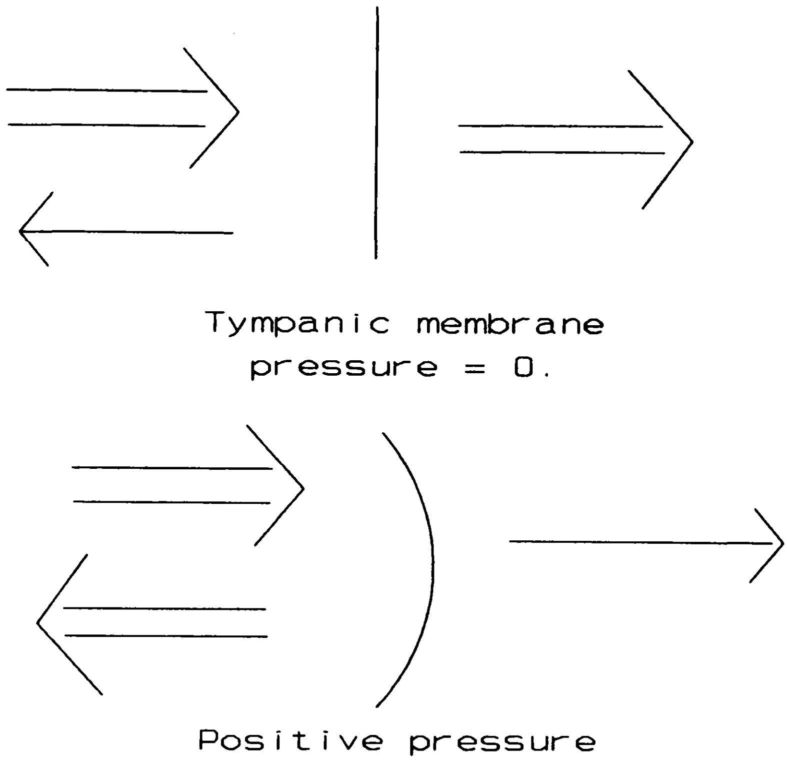


Figure 7.3 Behaviour of the tympanic membrane under pressure.

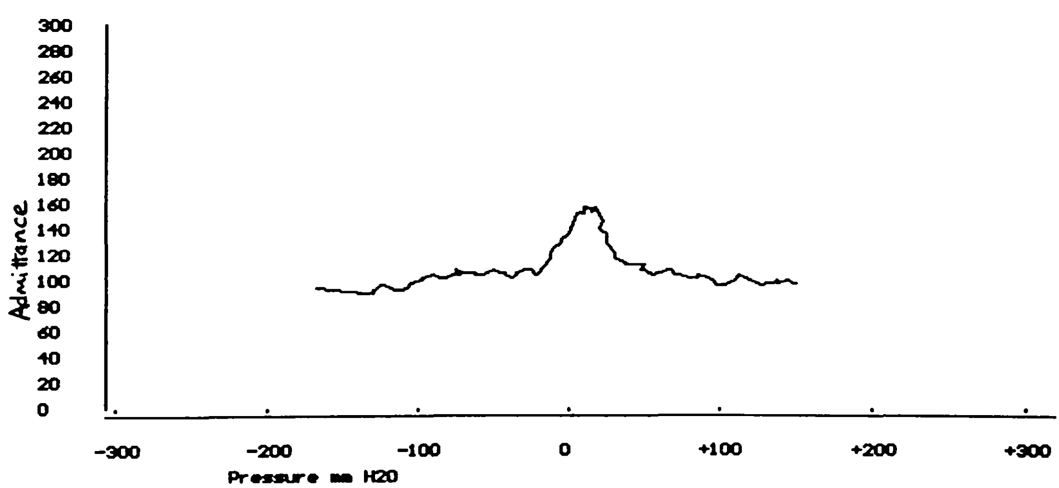


Figure 7.4 Tympanogram of normal ear at 660Hz probe tone.

numbers.

Note that at high positive or negative pressures the AGC number is lower than at 0 pressure. This signifies a lower admittance and hence a higher impedance. Less is transmitted and more reflected, as would be expected from figure 7.3.

In the case of a perforated eardrum there would be a flat tympanogram with no pressure peak. A tympanogram with low amplitude may suggest ossicular fixation, that is the impedance is high no matter what the pressure, since it is due to the ossicle bones and reasonably independent of the eardrum. Ossicular discontinuity on the other hand may result in increased tympanogram amplitude, since at zero pressure there is low contribution to the impedance from the eardrum but virtually no contribution from the ossicle bones because they have become detached.

7.4.1 Multi-Frequency Tympanometry.

When running multi-frequency tympanometry, two types of graph can be plotted. A conventional two-dimensional tympanogram of admittance versus pressure can be plotted for each of the frequencies being tested (like the one shown in figure 7.4). Alternatively, a three dimensional plot can be made of admittance, pressure and frequency. One of these three dimensional plots for a normal hearing ear is shown in figure 7.5

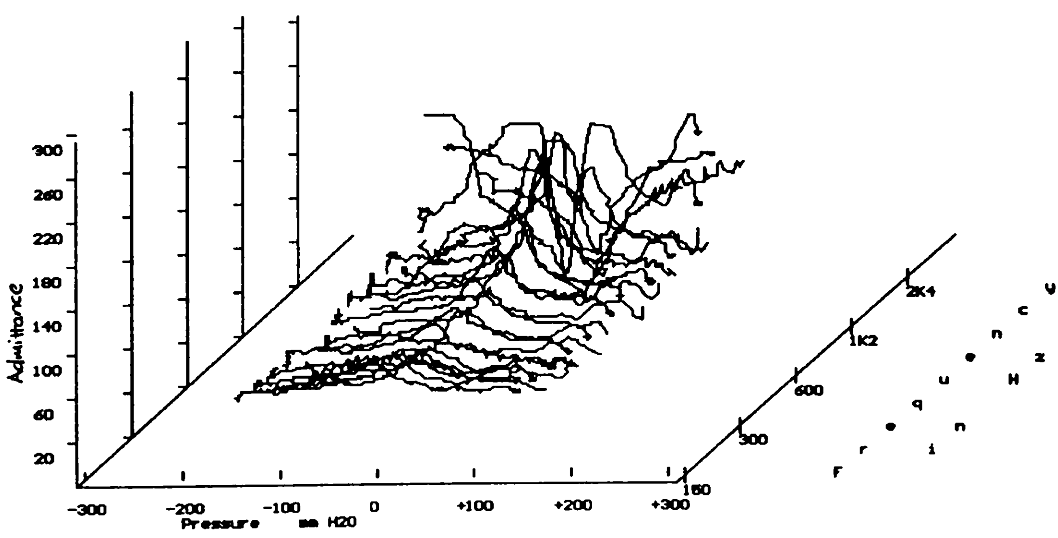


Figure 7.5 A multi-frequency tympanogram of a normal hearing ear.

This diagram is somewhat difficult to interpret here, but on the PC screen, when each frequency can be highlighted in turn by a different colour, it can be quite clear. Another problem is that this data was accumulated using manual pressure control and so tends not to be smooth. What is interesting, is that the normal small positive peak of the tympanogram becomes bigger as the probe frequency is increased and then changes to a negative peak, often with notches. This is in agreement with the results of Margolis et al. (1985) who use frequencies of up to 910KHz. Their results show B and G curves and demonstrate that the transitions occur first in the B graph, then, as the probe tone frequency is increased, in the G graph and finally in the admittance curve.

Figure 7.6 shows the two-dimensional plot of one frequency below the transition and figure 7.7 shows a frequency above the transition.

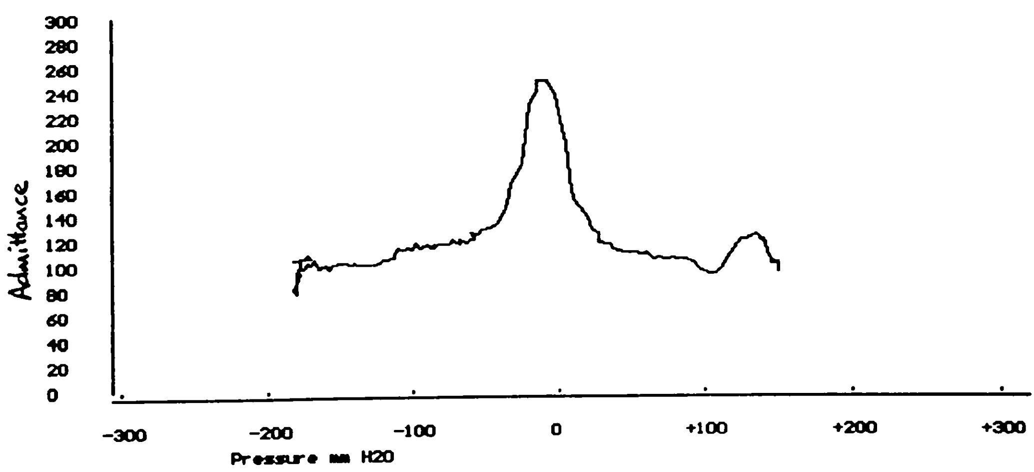


Figure 7.6 Normal tympanogram below transition frequency.

Margolis et al. explain that these transitions follow the results of a model known as the Vanhuyse model and are due to the acoustic reactance shifting from large negative values, when it is mainly compliance controlled, to

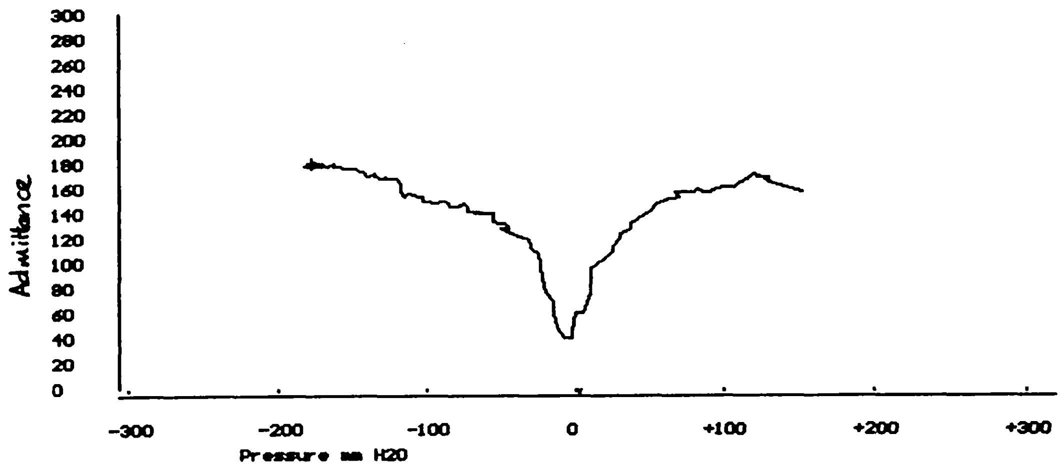


Figure 7.7 Normal tympanogram above transition frequency.

positive values when it is mass controlled.

Similar results were also given by Shanks et al. (1987) and agree closely with the results of figure 7.5.

It should be noted that although B, G, and Φ graphs are not shown here, it is possible to accumulate these with this system.

7.5 Probe Assembly.

A probe was designed and built which allowed a pressure seal to be made in the ear canal and sound to be produced and detected. It contained three tubes, one to conduct the sounds from the loudspeaker into the canal, one to receive the reflected sounds at the microphone and one to allow the pressure to be varied. Figure 7.8 shows a diagram of the probe.

The diameter of the tubes (1mm) was determined by the dimensions of the loudspeaker and microphone. These tubes were set in epoxy, which was tapered so that they could be sealed in the ear with a probe tip. The tip of this epoxy was coated in araldite to avoid pressure leakage. The pressure tube is made out of steel (again to avoid loss of pressure) while the microphone and loudspeaker tubes are

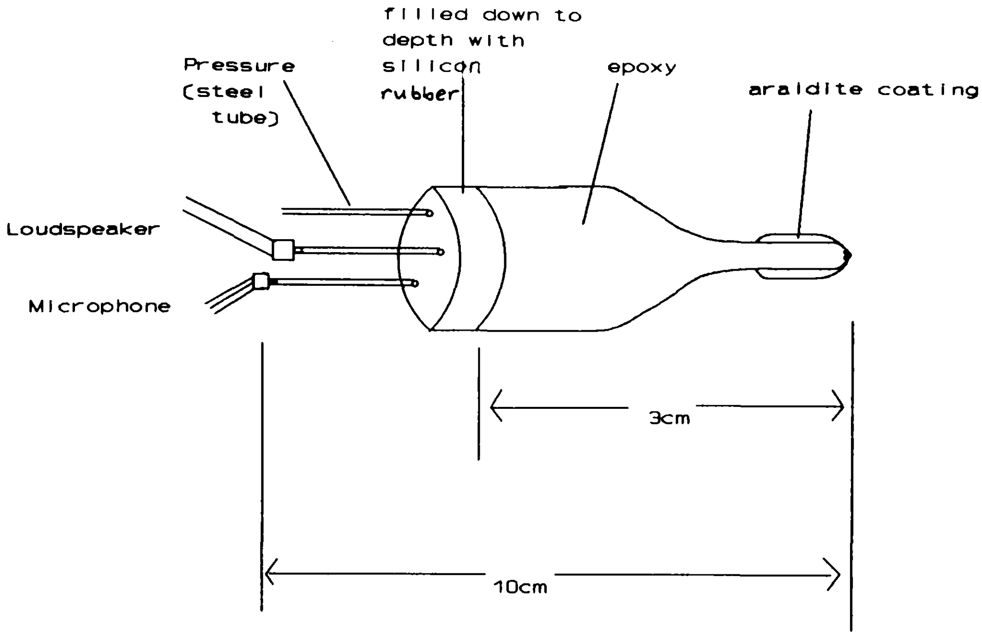


Figure 7.8 Probe manufacture.

made out of flexible plastic.

7.5.1 Probe Response.

The response curves of the loudspeaker and microphone are flat up to about 3 or 4 kHz. To check that the responses are still flat once installed in the above probe, a graph was drawn of microphone response at constant loudspeaker voltage (see figure 7.9).

This shows that the responses are certainly not flat and implies that the tubes of the probe set up a resonance. The first resonance occurs at about 700Hz which is equivalent to a wavelength of 40cm and since the tubes are about 10cm long (i.e. $\lambda/4$) resonance is confirmed.

By cutting the tubes so that they are in line with the top of the epoxy, and casing and embedding the loudspeaker and microphone in silicon rubber, the length of the tubes is reduced to 3cm. This means that the first resonance is shifted to about 2.5kHz. This is more acceptable, but since probe frequencies of up to 2kHz are used it is still not ideal. The silicon rubber also helps to avoid loss of

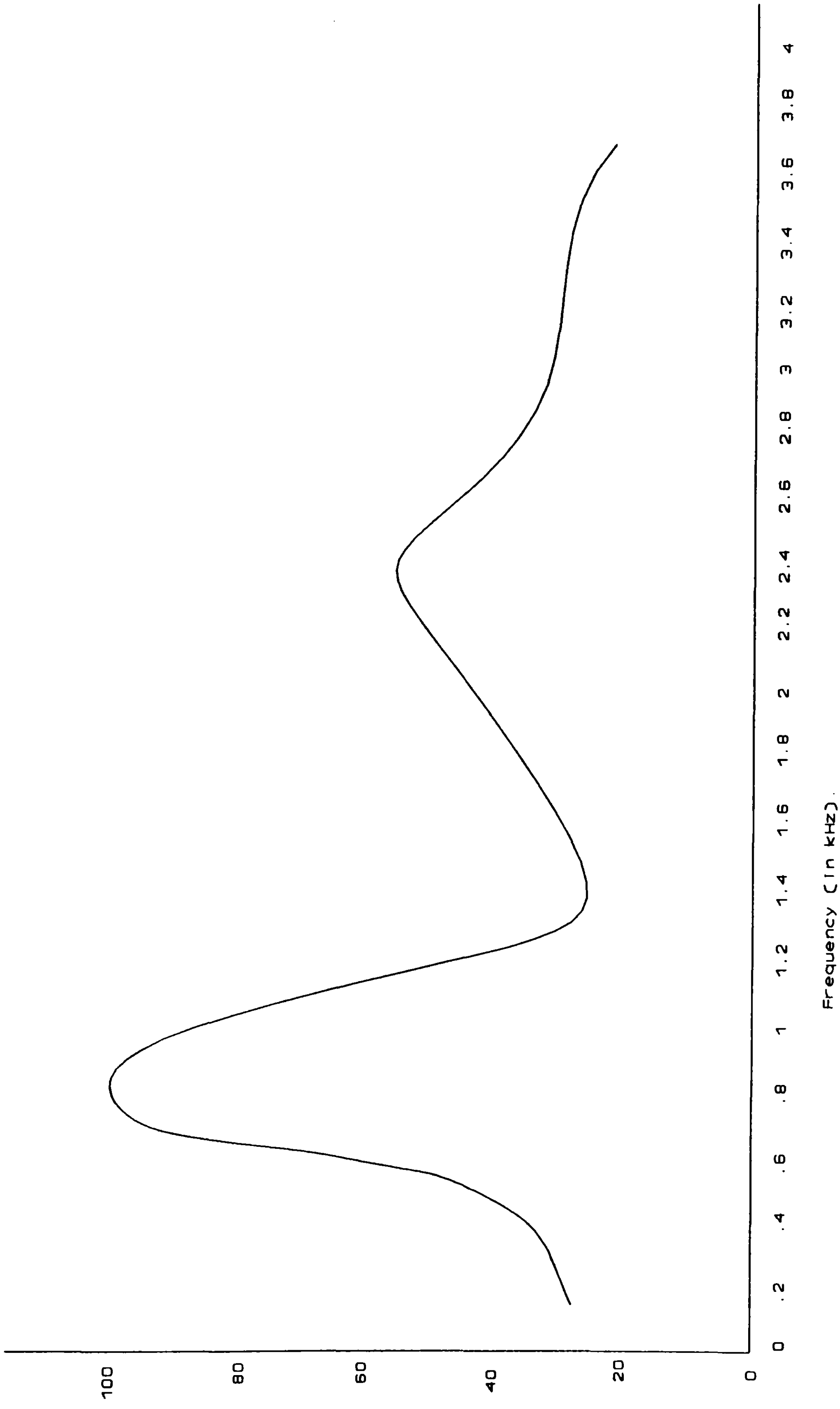


Figure 7.9 Microphone Response in Probe.

signal.

A new probe tip was designed, this time with length 1cm, moving the first resonance to 4.5kHz which is preferable. This probe follows the same sort of design as the one shown in figure 7.8 but is obviously not as long. This was the design adopted for the tympanometry tests.

7.5.2. Artificial Ear.

It is not satisfactory to continuously test the tympanometer on a human ear. While the machine is a prototype and still in the development stage it may be possible to damage the ear. It was necessary to find something which would mimic the ear's behaviour accurately.

A device was used which had roughly the dimensions of the ear canal and drum. Different materials were clamped across the circular diameter to mimic the human tympanic membrane. Several thicknesses of rubber were investigated but none of these was elastic enough. It was discovered, however, that a very thin sheet of mylar resembled the eardrum with enough accuracy and this was therefore used for testing purposes.

7.6 Phase Difference Measurements.

Turning back to figure 7.2, the elements at L, M and N are those used to calculate the phase difference between the loudspeaker and microphone. The block N is the important phase measurement and is shown in detail in figure 7.10.

Consider an initial state in which flip-flop B is cleared making Q_b low. Thus CLR_a is also low making $\bar{Q}_a$ high. Since flip-flop A is held cleared it will be insensitive to any edge at CK_a. Let us consider a positive edge from the loudspeaker signal at CK_b. Because $\bar{Q}_a$ is high, so is CLR_b and this positive edge will cause Q_b to go high (since

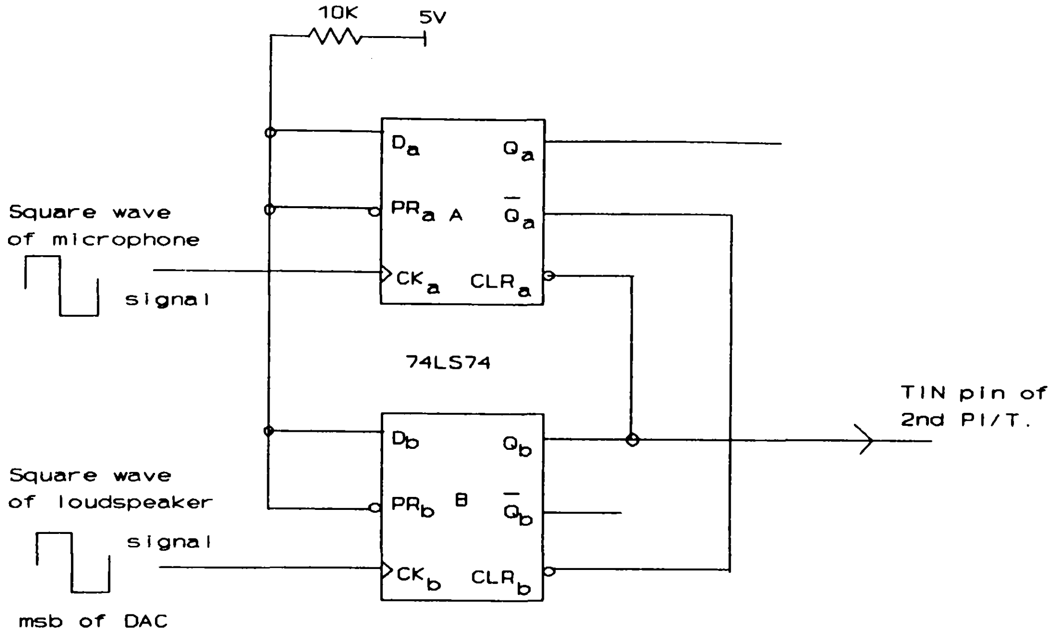


Figure 7.10 Flip-flop circuitry for phase measurement.

D_b is high). This removes the low from CLR_a and so when a positive edge from the microphone subsequently occurs on CK_a, $\bar{Q}_a$ goes low clearing flip-flop B and causes Q_b to return low which in turn clears flip-flop A and removes the low at CLR_b.

The length of time for which Q_b is high gives a measure of the phase difference. This is done via the TIN pin of the second PI/T allowing the time for which the pin was high to be measured. There are two scenarios possible shown in figure 7.11.

The case on the left is when $X < (T/2)$ while the case on the right takes effect when $X > (T/2)$. Since the frequency is known, the period T can be calculated easily.

Because the square wave from the loudspeaker is obtained from the MSB of the DAC it has very 'clean' edges. The squaring of the microphone signal, however, is done via a comparator and the edges contain 'jitter'. To avoid this stopping the PI/T counting too soon, the shift register and OR gate are inserted at M in figure 7.2.

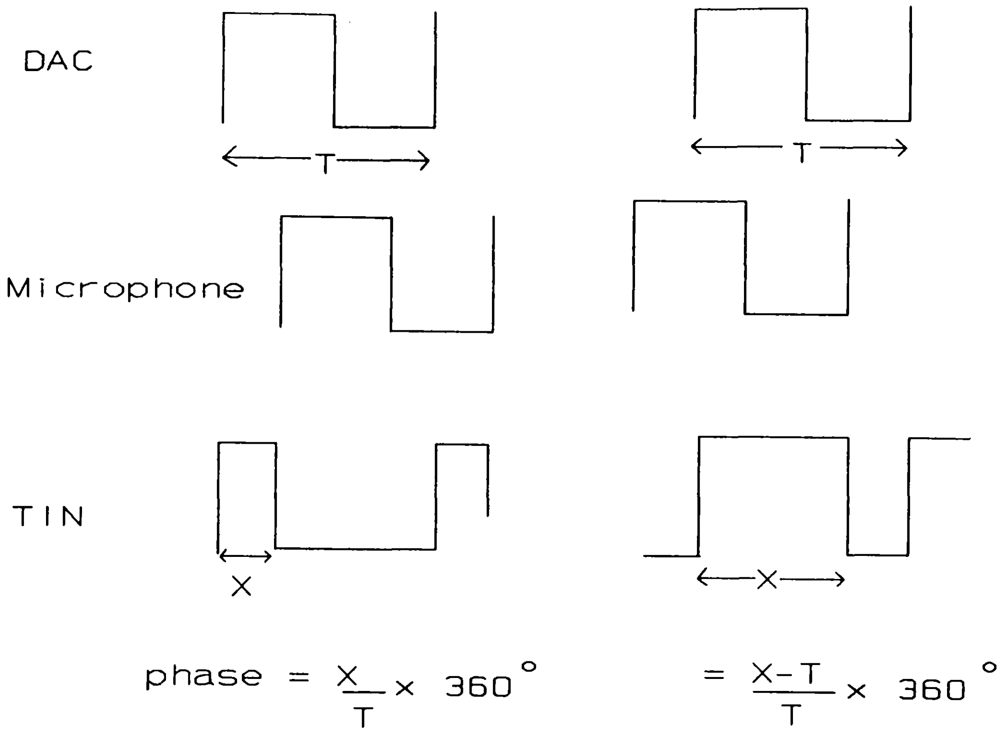


Figure 7.11 Phase measurements made at TIN of PIT.

In this way the PI/T measures a value X (shown in figure 7.11) and this is sent to the PC where it can be translated into a phase difference which can then be used to calculate conductance and susceptance allowing graphs of pressure versus B and G to be plotted.

7.7 Computer Control of Pressure.

The motor to drive the pressure syringe was an RS stepping linear actuator and this motor was itself driven from a stepper motor drive board (unipolar 2A). Although from the RS specifications the miniature actuator was sufficient (from the point of view of power) it was found experimentally that it was impossible to ensure the motor and syringe were exactly in line and that the motor was having to drive in only one direction. It was necessary to use the standard motor (i.e. the standard linear actuator) which was found to be more than adequate.

A short program was written to control this motor. It can be stopped or started and its direction can be changed - all done via latches on the KDM board. Two microswitches were placed at either end of the syringe. When these are depressed, a signal is sent back to the KDM board and the operator can decide either to change direction, or to stop the syringe.

To allow a maximum sweep of pressure from positive to negative or vice versa, the ear seal should be made when the syringe is at its central position. A photodiode was used and when the centre of the syringe is reached, another signal is sent to the KDM board. In this way the motor can set the syringe at its centre before the pressure seal is attempted.

7.8 Acoustic Reflex.

The acoustic reflex test has not yet been implemented fully. However, several checks have been made to ensure that the hardware operates correctly, at least for a contralateral test. This leaves only the PC software to be written which is a straightforward task.

The assembly code program TYMPAUD in appendix E is designed to operate the ASRA audiometer through the KDM board. In normal operation, the audiometer produces sounds by decoding a number of characters sent from the PC via its serial port. In tympanometry mode, the processor waits for the character 'r' and, once received, it directs the characters which follow, straight to the audiometer via ACIAUD until the character 's' is received. By enclosing the normal audiometer codes between the characters 'r' and 's', the audiometer will operate normally via the tympanometer.

This was tested and found satisfactorily to produce contralateral stimuli for an acoustic reflex test. It is then a matter of measuring the admittance as described before, and plotting it against time, which can be measured using the fact that the 1st PI/T interrupts every millisecond.

Ipsilateral reflexes are more complicated. The audiometer can still be used to generate the stimulus but there must be some mechanism for sending this stimulus to the probe loudspeaker, or to a second loudspeaker in the probe. It is also necessary to protect the microphone from being damaged by the very loud sound which will be present immediately after the stimulus. Two methods of protection are commonly used:

(i) filtering,

and (ii) multiplexing - so that the microphone is switched off for a period of time around the stimulus and then on again to take the measurements.

This is something to look to in the future.

7.9 Conclusions.

The work shows that all types of tympanometric testing required for a type 1 admittance, phase, conductance and susceptance multi-frequency meter are possible using this machine, with the exception of the ipsilateral acoustic reflex test.

Software was written by the author to prove the design and to allow the measurement of single and multiple frequency tympanograms; little emphasis was, however, placed on the software front-end.

An information technology student (Hacking (1991)) has subsequently continued this work and using the techniques tested by the author, has produced a more 'user-friendly' system. This work was written up for an MSc project and incorporates a main menu, the ability to obtain single, double and multi-frequency tympanograms displaying Y, Φ, G and/or B versus pressure graphs and a graph of admittance versus time, for a contralateral acoustic reflex. Great care must be taken to make the system safe for use with patients and some safety was built in with this work.

Future work must include the implementation of the ipsilateral acoustic reflex test and a proper calibration routine.

CHAPTER 8: Conclusions

This thesis shows how the use of a computer can give more flexibility to audiometric tests. A conventional (BSA recommended) pure-tone test has been implemented which allows completely automated testing, or completely manual testing if required. What is particularly valuable about this system is that these two extremes are not rigid. It is quite possible to operate the test automatically, use manual control to check something and then switch back to the automatic test. This allows maximum flexibility during the test, while offering all the usual data handling and management benefits of a computer.

Computer control also offers benefit when applied to other audiometric tests, such as supra-threshold and speech audiometry. Tone decay particularly has the advantage that the timing is done by the PC, and in speech audiometry the total speech score can be kept by the computer avoiding extra paper work.

Perhaps of more importance are the extra tests which can be accomplished, many of which would otherwise have been unthinkable. Firstly, the use of random testing. By randomly presenting tones (random in ear, frequency and intensity) and carefully noting the subjects responses, an indication can be gained of how consistent the subject is about these particular thresholds. This test is then applied to the field of non-organic hearing loss. By testing subjects who are exaggerating their threshold and comparing the results with those gained when the same subjects are responding truthfully, analysis has shown that these two groups can be distinguished. Although the group of subjects tested is small and not as motivated to deceive as those who would actually receive industrial hearing loss benefit, the results show that a proper clinical trial would be worthwhile.

Unfortunately, the facilities to undertake such a clinical trial were not available to the author; perhaps in the future this work will be used to attempt such a study, using a large group of subjects including some who are actually suspected of non-organic hearing loss.

Another facility offered is that of phonemic testing in speech audiometry. By using only a minor modification in the scoring of a conventional speech audiogram, a graph of phoneme versus hearing level gives information on which phonemes the subject finds particularly difficult. Again a proper clinical trial would be worthwhile since the preliminary results from a handful of subjects tested by the author look favourable. It would be of interest to obtain a large set of data showing pure-tone audiograms and the corresponding phoneme charts so that the two could be linked. This might allow phoneme charts to give valuable information about pure-tone thresholds.

The frequency analysis which was done on the individual phonemes of the Boothroyd wordlists gives a great deal of information. The most important feature of it is that this information is specific to this AB(S) tape which of course can be compared directly from hospital to hospital. The other advantage is that although the work here was implemented with the AB(S) wordlists, it can be easily applied to any other wordlist sets or tapes.

The work has shown the flexibility which can be offered when using a computer to control hearing tests. It has also shown some interesting results of novel tests which suggest further clinical investigations would be profitable.

APPENDIX A: Clinical Decision Analysis

A.1 The Probability Distribution Curve.

Probabilities are usually expressed in one of two ways:

(i) as a percentage,

or (ii) as a number less than 1 which is equal to the percentage divided by 100.

For example, the value of probability expressed as 50% can also be expressed as 0.50.

Figures A.1 and A.2 show the probability distribution curves (PDC) for a hypothetical experiment. Figure A.1 shows the results expressed as discrete numbers in the form of barcharts, while figure A.2 shows the results shown as continuous curves.

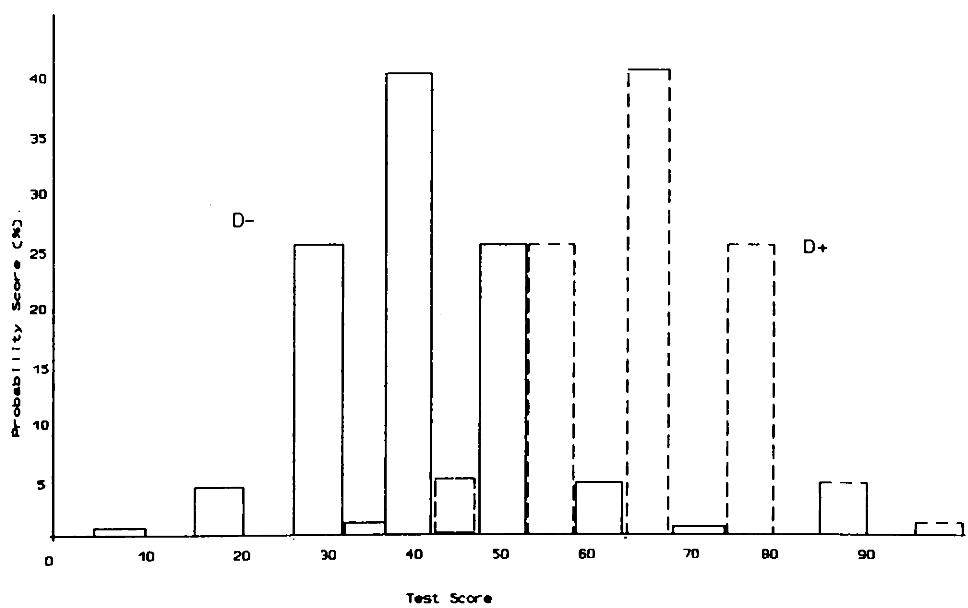


Figure A.1 Probability Distribution Curves - Barcharts. *(D+,D- correspond to patient's with and without the disease repectively).*

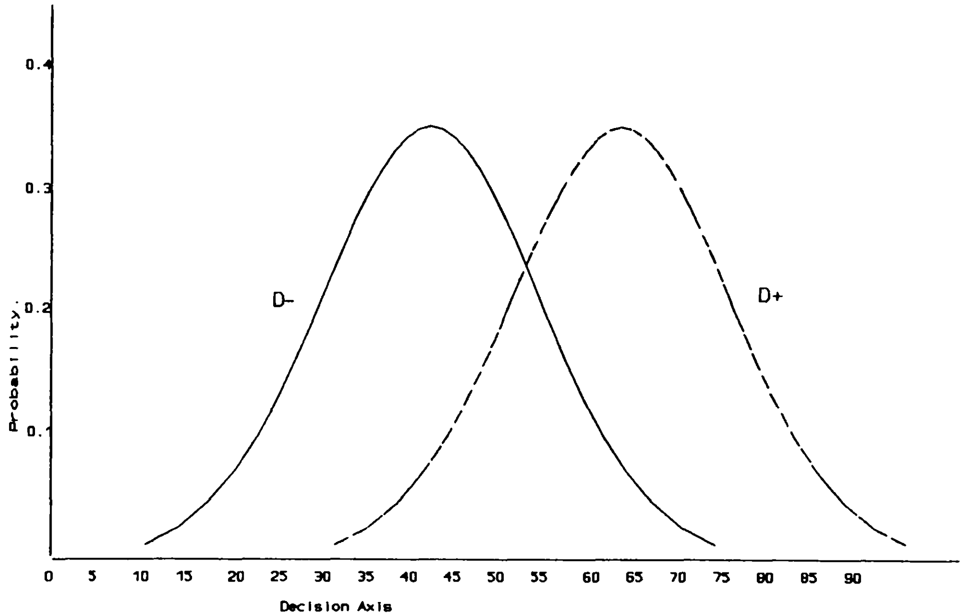


Figure A.2 Probability Distribution Curves.
Containing the same data as figure A.1 displayed as continuous curves.

The solid lines and bars represent the population of patients who do not have a certain disease (D-) and the dotted lines and bars represent the population of those with that disease (D+). A vertical line ('decision line') can be drawn on these curves and used to determine the disease state of a patient. This line of course should maximise the number of correct decisions both positive and negative.

In the examples given in figure A.1 and A.2 an obvious place to draw this 'decision line' would be at a test score of 50. Very often, however, less obvious factors or 'costs' come into the choice of position for this line.

In the application of clinical decision analysis in this thesis, the data has discrete values and is therefore presented in the form of figure A.1. In this figure, once the 'decision line' has been drawn, all the patients tested with a test score to the right of this line are said to have a positive test score and are classed as T+, while those to the left have a negative test score and are classed T-.

Conditional probabilities can then be used to describe different groups of data that are present:

P(T+|D+) is the measured probability that the test result will be positive, given that the patient has the disease. This is also known as Hit Rate or HT.

P(T-|D+) is the measured probability that the test result will be negative, given that the patient has the disease. Also known as a Miss Rate or MS.

P(T+|D-) is the measured probability that the test result will be positive, given that the patient does not have the disease. Also known as False Alarm rate or FA.

P(T-|D-) is the measured probability that the test result will be negative, given that the patient does not have the disease. Also known as the correct rejection rate or CR.

These can be drawn up into a 2x2 decision matrix:

	T+	T-
D+	P(T+|D+) HT	P(T-|D+) MS
D-	P(T+|D-) FA	P(T-|D-) CR

In the case of the continuous curves of figure A.2

$$P(T+|D+) = \int_{c}^{\infty} P(z|D+)\,dz.$$

From figure A.1 it is easy to form the decision matrix.

	T+	T-
D+	95/100	5/100
D-	5/100	95/100

It is clear that $P(T+|D+) + P(T-|D+) = 1.0$ and $P(T+|D-) + P(T-|D-) = 1.0$. This means that two out of four of these conditional probabilities are enough to calculate the other two. e.g. HT and FA are enough to determine MS and CR.

Another name given to $P(T+|D+)$ is sensitivity and $P(T-|D-)$ is also known as specificity.

Something that is very often important is the prevalence of a certain disease. A hit rate of 100% and a miss rate of 0% does not reveal much if only one person in the sample population actually had the disease. This prevalence or 'a priori' probability is the value $P(D+)$. $P(D+)$ will be the fraction of the sample population who actually have the disease.

Another useful index is the ratio of the number of patients giving a positive test result correctly to the total number of patients who give a positive test result. This is known as predictive value.

$$\textit{Predictive value}=\frac{HT}{HT+FA}=\frac{P(T+|D+)}{P(T+|D+)+P(T+|D-)}.$$

A.2 The Index d'.

The theory of signal detection plays an important role in clinical decision analysis since an important index d' is derived from it (Swets 1964). Consider figure A.2 again. This time the solid line represents the probability density of a purely noise signal being detected and the dotted line represents the probability density of a signal in noise being detected. The distributions are, for simplicity, normal and with equal variance. The conditional probabilities $P(Y|s)$ and $P(Y|n)$ can then be considered. These correspond to the probability of measuring Y given that the signal presented is signal in noise and the probability of measuring Y given that the signal presented was only noise respectively.

Assuming the decision line is drawn at a value on the decision axis of c, the probabilities are:

$$P(Y|s) = \int_c^\infty P(z|s)\,dz$$

$$P(Y|n) = \int_c^\infty P(z|n)\,dz$$

There is no loss of generality if the noise spectrum is shifted on the axis so that its mean is at zero. If the variances of the two spectra are equal and have value σ^2 the above equations (assuming a Gaussian Distribution of noise) become:

$$P(Y|s) = \frac{1}{\sqrt{2\pi}\sigma}\int_c^\infty \exp\left[-\frac{(z-d)^2}{2\sigma^2}\right] dz$$

$$P(Y|n) = \frac{1}{\sqrt{2\pi}\sigma}\int_c^\infty \exp\left[-\frac{z^2}{2\sigma^2}\right] dz$$

By definition, d′ is the difference between the means of the two density functions divided by their standard deviation and so $d' = d/\sigma$.

It can be seen that d′ gives a measure of the separation between the probability curves D- and D+. The higher the d′ value therefore the better the separation and the better the test.

A.3 Receiver Operator Characteristics (ROC) Curves.

Basically an ROC curve is a plot of P(T+|D+) or Hit rate against P(T+|D-) or false alarm rate for varying criteria. To plot the ROC curve corresponding to figure A.1 the decision line should be swept from left to right across the decision axis and the corresponding set of values (P(T+|D+),P(T+|D-)) plotted each time. Figure A.3 shows this type of curve and the line marked d'=2 is the line associated with figure A.1.

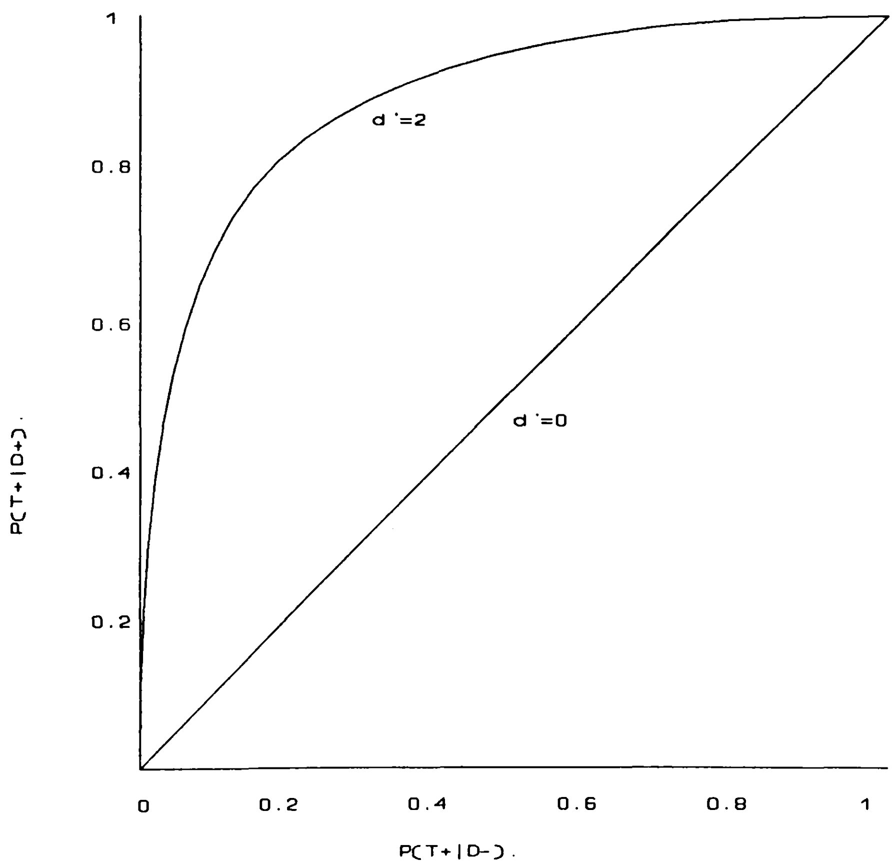


Figure A.3 ROC Curve Corresponding to Figure A.1. *with respect to d'=0.*

ROC curves are very often plotted on double probability coordinates which allow distributions which are normal and have equal variance to yield straight lines with unity slope. These are shown in figure A.4.

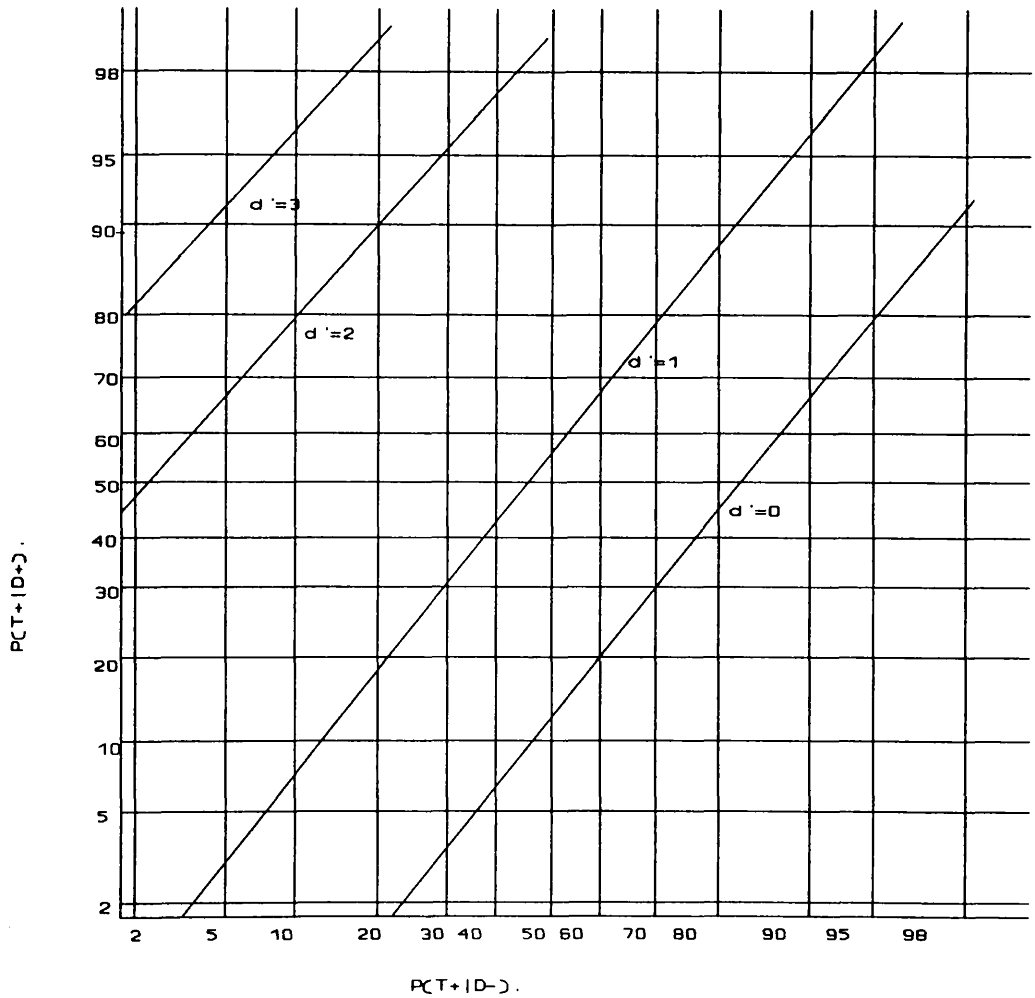


Figure A.4 Double Probability Coordinates.
Used to plot ROC curves.

The data in chapter 3, however, does not give normal distributions with equal variance. It is possible though, from the probability curves, to calculate P(T+|D+) and P(T+|D-) for each possible criteria, and therefore an ROC curve can be generated. The difference between this set of data and that shown in figure A.4 is that it does not yield straight lines when plotted on double probability coordinates.

d′ can be found from the corresponding probability distribution curves, but it can also be found from the index HT/FA using published tables (Swets. 1964). When the distributions are normal and have equal variance, all points on the corresponding ROC curve lie on a straight line when plotted on double probability coordinates and therefore for any particular test d′ is a constant. That is, d′ is independent of test criteria. If the curves, however, are non-normal, d′ will vary with test criteria. In this case the test which gives the maximum d′ must be quoted along with the criterion used to give that value of d′ as d′ is meaningless when quoted independently.

APPENDIX B: Boothroyd Wordlists - Phonetic Components

Wordlist 1.

good - g-ʊ-d

sing - s-I-ŋ

ship - ʃ-I-p

rug - r-ʌ-g

fan - f-æ-n

cheek- tʃ-iː-k

haze - h-eI-z

dice - d-aI-s

both - b-ə ʊ-θ

well - w-ɛ-l

jot - dʒ ɒ -t

move - m-uː-v

Wordlist 2.

tail - t-eI-l

men - m-ɛ-n

fish - f-I-ʃ

duck - d-ʌ-k

gap - g-æ-p

cheese- tʃ-iː-z

rail - r-eI-l

hive - h-aI-v

bone - b-əʊ-n

wedge- w-ɛ-dʒ

moss - m-ɒ-s

tooth- t-uː-θ

Wordlist 3.

dig - d-I-g

pull - p-ʊ-l

thud - θ-ʌ-d

witch- w-I-tʃ

wrap - r-æ-p

jail - dʒ-eI-l

keys - k-iː-s

vice - v-aI-s

get - g-ɛ-t

shown- ʃ-əʊ-n

hoof - h-uː-f

bomb - b-ɒ-m

Wordlist 4.

tap - t-æ-p

boys - b-ɔ ɪ-s

fun - f-ʌ-n

will - w-ɪ-l

vat - v-æ-t

shape- ʃ-eɪ-p

wreath- r-iː-θ

hide - h-aɪ-d

guess- g-ɛ-s

comb - k-ə ʊ-m

choose- tʃ-uː-z

job - dʒ ɒ -b

Wordlist 5.

new - n-j-uː

bull - b-ʊ-l

fib - f-ɪ-b

thatch- θ-æ-tʃ

sum - s-ʌ-m

heel - h-iː-l

wide - w-aɪ-d

rake - r-eɪ-k

goes - g-ə ʊ-s

shop - ʃ - ɒ -p

vet - v-ɛ-t

june - dʒ-uː-n

Wordlist 6.

sack - s-æ-k

weep - w-iː-p

fill - f-I-l

catch- k-æ-tʃ

thumb- θ-ʌ-m

heap - h-iː-p

wise - w-aI-z

rave - r-eI-v

goat - g-əʊ-t

shone- ʃ-ɒ-n

bed - b-ɛ-d

juice- dʒ-uː-s

Wordlist 7.

book - b-ʊ-k

shop - ʃ-ɒ-p

badge- b-æ-dʒ

hutch- h-ʌ-tʃ

kill - k-I-l

thighs- θ-aI-s

wave - w-eI-v

reap - r-iː-p

foam - f-əʊ-m

goose- g-uː-s

not - n-ɒ-t

shed - ʃ-ɛ-d

Wordlist 8.

oil - ɔI-l

good - g-ʊ-d

bath - b-aː-θ

hum - h-ʌ-m

dip - d-I-p

five - f-aI-v

ways - w-aI-s

reach- r-iː-tʃ

joke - dʒ ə ʊ-k

noose- n-uː-s

got - g- ɐ -t

shell- ʃ -ɛ-l

Wordlist 9.

five - f-aI-v

yes - j-ɛ-s

hush - h-ʌ-ʃ

gas - g-æ-s

thin - θ-I-n

fake - f-eI-k

chime- tʃ -aI-m

weave- w-iː-v

jet - dʒ-ɛ-t

rob - r- ɐ -b

dope - d- ə ʊ-p

lose - l-uː-z

Wordlist 10.

paw - p- ɔː

hood - h-ʊ-d

jug - dʒ-ʌ-g

match- m-æ-tʃ

whip - w-I-p

faith- f-eI-θ

sign - s-aI-n

bees - b-iː-s

hell - h-ε-l

rod - r-ɐ-d

vote - v-əʊ-t

shook- ʃ-ʊ-k

Wordlist 11.

dale - d-eI-l

witch- w-I-tʃ

man - m-æ-n

hip - h-I-p

thug - θ-ʌ-g

ride - r-aI-d

siege- s-iː-dʒ

veil - v-eI-l

chose- tʃəʊ-z

shoot- ʃ-uː-t

web - w-ε-b

cough- k-ɐ-f

Wordlist 12.

rush - r-ʌ-ʃ

bark - b-aː-k

have - h-æ-v

whizz- w-I-z

buff - b-ʌ-f

mice - m-aI-s

teeth- t-iː-θ

gauge- g-eI-dʒ

poach- p-ə ʊ-tʃ

rule - r-uː-l

den - d-ɛ-n

cosh - k-ɒ -ʃ

Wordlist 13.

coil - k- ɔI-l

put - p-ʊ-t

kiss - k-I-s

buzz - b-ʌ-z

hash - h-æ-ʃ

thieve- θ-iː-v

gate - g-eI-t

wife - w-aI-f

pole - p-ə ʊ-l

wretch- r-ɛ-tʃ

dodge- d-ɒ -dʒ

moon - m-uː-n

Wordlist 14.

joy - dʒ-ɔɪ

bead - b-iː-d

wish - w-ɪ-ʃ

dutch- d-ʌ-tʃ

jam - dʒ-æ-m

heath- h-iː-θ

laze - l-eɪ-z

bike - b-aɪ-k

rove - r-ᵊʊ-v

pet - p-ɛ-t

fog - f-ɐ-g

soon - s-uː-n

Wordlist 15.

mess - m-ɛ-s

paw - p-ɔː

hug - h-ʌ-g

dish - d-ɪ-ʃ

ban - b-æ-n

rage - r-eɪ-dʒ

chief- tʃ-iː-f

pies - p-aɪ-s

wet - w-ɛ-t

cove - k-ᵊʊ-v

loose- l-uː-s

moth - m-ɐ-θ

International Phonetic Alphabet.

aː - as in father

æ - as in act

aɪ - as in dive

aʊ - as in out

eɪ - as in paid

ɪ - as in pretty

iː - as in see

ɒ - as in pot

əʊ - as in note

ɔː - as in thaw

ɔɪ - as in void

ʊ - as in pull

uː - as in zoo

ʌ - as in cut

ʃ - as in ship

tʃ - as in chew

dʒ - as in jaw

θ - as in thin

ŋ - as in sing

ɛ - as in bet

ð - as in the

APPENDIX C: Cubic Spline Curve

```
#include "stdio.h"
#include "graphics.h"
int valx[10],valy[10],valx1[10],valy1[10],
    n,n1,x1,x2,mult;
int calvallx[24],calvally[24],calvalrx[24],calvalry[24],
    nl,nr;
static int horposs[]={-4,12,32,52,72,92,112,132,
                      152,172,190,228};
static char
*horscls[]={"0","10","20","30","40","50","60","70",
           "80","90","100","120"};
main()
     {int i,j,k;
     int gmode,gdriver,gerror,size,tempmen1,tempmen3;
     detectgraph(&gdriver,&gmode);
     if (gmode==EGAHI) gmode=EGALO;
     initgraph(&gdriver,&gmode,"");
     gerror=graphresult();
     if (gerror<0)
          {printf("initgraph error:%s.\n");
          grapherrormsg(gerror);
          exit(1);
          }
     settextstyle(2,0,0); setcolor(14);
     setbkcolor(9); cleardevice();
     outtextxy(10,100,"Program to test Cubic Spline");
     mult=2;
     for (i=0;i<10;i++)
          {valx[i]=valy[i]=valx1[i]=valy1[i]=0;
          calvallx[i]=calvally[i]=
              calvalrx[i]=calvalry[i]=0;
          }
     n=5;
     n1=0;
     valx[0]=83; valy[0]=5;
     valx[1]=73; valy[1]=10;            /* Test points */
     valx[2]=63; valy[2]=20;
     valx[3]=53; valy[3]=22;
     valx[4]=33; valy[4]=20;
     nl=5;
     nr=0;
     calvallx[0]=110; calvally[0]=0;
     calvallx[1]=100; calvally[1]=23;   /* Normal
     calvallx[2]=90; calvally[2]=30;       reference
     calvallx[3]=70; calvally[3]=30;       curve */
     calvallx[4]=50; calvally[4]=30;
     drawgris(100,70);
     calclate(100,70,0);
     getch();
```

```
     closegraph();
     }
calclate(xf,yf,mask)
int xf,yf,mask;
     {int i,j,k,first,N,yy,tempval,tempvaly,x,
          minx,maxx,miny,maxy,flg,flgm,fg,hpl;
     int tempx,tempy,multit,tmpx[10],tmpy[10],nn,hpl1;
     float X,Y,t,ods;
     float a0,a1,a2,a3,b0,b1,b2,b3;
     char *tempstr="      ",str[2];
     fg=flg=flgm=0;
     minx=0; maxx=130;
     for (i=0;i<n;i++)
          {if (valx[i]>minx)
               {minx=valx[i]; miny=valy[i]; }
          if (valx[i]<maxx)
               {maxx=valx[i]; maxy=valy[i]; }
          }
     ods=miny;
     k=1;
     while (k<n)
          {for (i=0;i<(n-1);i++)
               {if (valx[i] < valx[i+1])
                    {tempval=valx[i];
                    tempvaly=valy[i];           /* Order
                    valx[i]=valx[i+1];             in x */
                    valy[i]=valy[i+1];
                    valx[i+1]=tempval;
                    valy[i+1]=tempvaly;
                    }
               }
          k++;
          }
     first=1;
     for (i=0;i<(n-1);i++)
          {cubic(i,&a0,&a1,&a2,&a3,&b0,&b1,&b2,&b3,0);
          N=valx[i]-valx[i+1];
          if (N!=0)                /* Plotting curve */
               {for (j=0;j<=N;j++)
                    {t=(float)j/(float)N;
                    X=((a3*t+a2)*t+a1)*t+a0;
                    Y=((b3*t+b2)*t+b1)*t+b0;
                    if (Y>ods) ods=Y;
                    if (!flg && Y>=28) {flg=1; x1=X; }
                    if (!flgm && Y<=3) {flgm=1; x2=X; }
                    yy=yf-2*Y;
                    if (yy>yf) yy=yf;
                    if (yy<(yf-60)) yy=yf-60;
                    x=xf+(240-2*X)*mult;
                    if (first) moveto(x,yy);
                    else lineto(x,yy);
                    first=0;
                    }
               }
          }
     if (ods>30.0) ods=30.0;
```

```
     for (i=0;i<(n-1);i++)
          {cubic(i,&a0,&a1,&a2,&a3,&b0,&b1,&b2,&b3,0);
          N=valx[i]-valx[i+1];
          if (N!=0)          /* Finding HPL */
               {for (j=0;j<=N;j++)
                    {t=(float)j/(float)N;
                    X=((a3*t+a2)*t+a1)*t+a0;
                    Y=((b3*t+b2)*t+b1)*t+b0;
                    if (Y>=(ods/2))
                         {hpl=X; fg=1; break;}
                    }
               if (fg) break;
               }
          }
     fg=0;
     if (yf<80) nn=nl;
     else nn=nr;
     for (i=0;i<(nn-1);i++)
          {tmpx[i]=valx[i]; tmpy[i]=valy[i];
          if (yf<80)
               {valx[i]=calvallx[i];
               valy[i]=calvally[i];
               }
          else
               {valx[i]=calvalrx[i];
               valy[i]=calvalry[i];
               }
          cubic(i,&a0,&a1,&a2,&a3,&b0,&b1,&b2,&b3,0);
          valx[i]=tmpx[i]; valy[i]=tmpy[i];
          if (yf<80) N=calvallx[i]-calvallx[i+1];
          else N=calvalrx[i]-calvalrx[i+1];
          if (N!=0)      /* Finding normal HPL */
               {for (j=0;j<=N;j++)
                    {t=(float)j/(float)N;
                    X=((a3*t+a2)*t+a1)*t+a0;
                    Y=((b3*t+b2)*t+b1)*t+b0;
                    if (Y>=(15)) {hpl1=X; fg=1; break;}
       /* This 15 may be changed to ods/2 if required */
                    }
               if (fg) break;
               }
          }
if (mult==2) multit=150; else multit=0;
     if (hpl>80) moveto((xf+160)+multit,yf-46);
     else moveto((xf+2)*mult,yf-46);
     if (!mask)      /* Calculating ODS, HPL, HPLE */
          {x=ods*100/30; itoa(x,tempstr,10);
          outtext("ODS = "); outtext(tempstr);
          str[0]='%'; str[1]=0; outtext(str);
          if (hpl>80)  moveto((xf+160)+multit,yf-36);
          else moveto((xf+2)*mult,yf-36);
          itoa(120-hpl,tempstr,10);
          outtext("HPL = "); outtext(tempstr);
          outtext("dB");
          if (hpl>80)  moveto((xf+160)+multit,yf-26);
          else moveto((xf+2)*mult,yf-26);
```

```
        itoa(hpl1-hpl,tempstr,10);
        outtext("HPLE = "); outtext(tempstr);
        outtext("dB");
        }
    }
                                /* Calculating coefficients */
cubic(i,a0,a1,a2,a3,b0,b1,b2,b3,k)
int i,k;
float *a0,*a1,*a2,*a3,*b0,*b1,*b2,*b3;
    {float xA,xB,xC,xD,yA,yB,yC,yD;
    if (!k)
        {if (i==0)
            {xA=valx[i]; xB=valx[i];
            xC=valx[i+1];  xD=valx[i+2];
            yA=valy[i]; yB=valy[i];
            yC=valy[i+1]; yD=valy[i+2];
            }
        else
            {if (i==(n-2))
                {xA=valx[i-1]; xB=valx[i];
                xC=valx[i+1]; xD=valx[i+1];
                yA=valy[i-1]; yB=valy[i];
                yC=valy[i+1]; yD=valy[i+1];
                }
            else
                {xA=valx[i-1]; xB=valx[i];
                xC=valx[i+1]; xD=valx[i+2];
                yA=valy[i-1]; yB=valy[i];
                yC=valy[i+1]; yD=valy[i+2];
                }
            }
        }
    else
        {if (i==0)
            {xA=valx1[i]; xB=valx1[i];
            xC=valx1[i+1]; xD=valx1[i+2];
            yA=valy1[i]; yB=valy1[i];
            yC=valy1[i+1]; yD=valy1[i+2];
            }
        else
            {if (i==(n1-2))
                {xA=valx1[i-1]; xB=valx1[i];
                xC=valx1[i+1]; xD=valx1[i+1];
                yA=valy1[i-1]; yB=valy1[i];
                yC=valy1[i+1]; yD=valy1[i+1];
                }
            else
                {xA=valx1[i-1]; xB=valx1[i];
                xC=valx1[i+1]; xD=valx1[i+2];
                yA=valy1[i-1]; yB=valy1[i];
                yC=valy1[i+1]; yD=valy1[i+2];
                }
            }
        }
    *a3=(-xA+3*(xB-xC)+xD)/6.0;
                                *b3=(-yA+3*(yB-yC)+yD)/6.0;
```

```
     *a2=(xA-2*xB+xC)/2.0;          *b2=(yA-2*yB+yC)/2.0;
     *a1=(xC-xA)/2.0;               *b1=(yC-yA)/2.0;
     *a0=(xA+4*xB+xC)/6.0;          *b0=(yA+4*yB+yC)/6.0;
     }
drawgris(xf,yf)                 /* Drawing Grid */
int xf,yf;
     {int i,j,k;
     char *tempmsg="     ";
     setcolor(5);
     for  (i=0;i<61;i+=12)     /*horiz lines*/
          {moveto(xf,yf-i);
          linerel(240*mult,0);
          }
     for  (i=0;i<241*mult;i+=20*mult)    /*vert lines*/
          {moveto(xf+i,yf);
          linerel(0,-60);
          }
     k=0;                           /*vert scale*/
     setcolor(3);
     for  (i=100;i>=0;i-=20)
          {j=yf-60+k; if  (j<0) j=0; if  (j!=0) j-=3;
          moveto((xf-26),j); itoa(i,tempmsg,10);
          outtext(tempmsg);
          moveto(xf+242*mult,j); itoa(i*3/10,tempmsg,10);
          outtext(tempmsg);
          k+=12;
          }
     for  (i=0;i<12;i++)
          {moveto(xf+horposs[i]*mult,yf+2);
          outtext(horscls[i]);
          }
     moveto((xf-40),yf-50);
     settextstyle(2,1,0);
     outtext("Score %");
     settextstyle(2,0,0);
     setcolor(15);
     }
```

APPENDIX D: Frequency Components of Phonemes

This appendix contains a printout of the output from MINITAB and has several columns. The first of these represents the row in the MINITAB matrix on which this phoneme appears. There are 540 of these rows corresponding to 3 (phonemes) X 180 (words). The next column is the ASCII code of the phoneme and a key to these codes is given below. The columns are then labelled f1, s1, f2, s2 etc. and contain the frequencies present and their corresponding strengths. In general, the column f1 contains the frequency with the highest strength which is shown in s1, while f9 contains the frequency with lowest strength which is in s9.

Key to character ASCII codes.

Character	ASCII Code.
b	98
d	100
f	102
g	103
h	104
j	106
k	107
l	108
m	109
n	110
p	112
r	114
s	115
t	116
v	118
w	119

z	122
aː	192
æ	193
aɪ	194
aʊ	196
eɪ	197
ɪ	199
iː	200
ɒ	202
əʊ	203
ɔː	204
ɔɪ	205
ʊ	206
uː	207
ʌ	211
ʃ	212
tʃ	214
dʒ	215
θ	216
ŋ	218
ɛ	221

Output from MINITAB....

print c1 c7-c12

OW	phoneme	f1	s1	f2	s2	f3	s3	f4	s4	f5	s5	f6	s6	f7	s7	f8	s8	f9	s9
1	103	1190	1.0167	1240.0	0.8379	1000	0.7374	490	0.4916	350	0.44691	50	0.89381	130	0.82677	1320	0.42456	4220	0.06703
2	206	400	12.3859	500.0	9.6637	200	7.3499	1000	6.3971	290	6.30543	950	3.40271	*	*	*	*	*	
3	100	110	0.5750	190.0	0.2654	10	0.2338	340	0.1074	2690	0.05055	1590	0.04423	1680	0.04423	*	*	*	
4	115	50	0.2226	4890.0	0.0849	540	0.0762	1800	0.0747	4100	0.06444	5200	0.06151	5350	0.05859	*	*	*	
5	199	320	7.4865	190.0	5.3475	500	4.1957	1850	1.2340	2000	1.31630	2660	0.82269	800	1.15177	*	*	*	
6	218	95	0.5707	220.0	0.5707	180	0.5079	400	0.3135	1120	0.09406	490	0.13796	*	*	*	*	*	
7	212	10	0.3287	2300.0	0.3106	2700	0.2781	490	0.2384	1700	0.19144	5250	0.10114	4210	0.08308	*	*	*	
8	199	400	4.4581	300.0	3.5763	200	3.4293	500	3.0374	1850	0.63687	2320	0.29394	*	*	*	*	*	
9	112	100	0.3961	950.0	0.1958	1360	0.1697	900	0.1741	1100	0.14797	1900	0.05658	2400	0.05658	3670	0.04787	*	
10	114	290	5.2419	200.0	3.8594	100	3.9170	500	0.8064	700	0.40322	800	0.34562	*	*	*	*	*	
11	211	200	3.9869	500.0	3.9869	900	3.9869	1210	3.9650	1190	3.81167	300	3.32974	620	3.11067	*	*	*	
12	103	160	1.3674	310.0	0.5560	490	0.3756	1280	0.3907	1400	0.37565	650	0.18031	*	*	*	*	*	
13	102	*	*	*	*	*	*	*	*	*	*	*	*	*	*	*	*	*	
14	193	200	2.5938	800.0	2.5938	300	2.3943	810	2.3658	1600	0.91210	1710	0.62707	*	*	*	*	*	
15	110	400	0.3841	160.0	0.3461	210	0.2954	1320	0.0675	990	0.05909	2200	0.02954	*	*	*	*	*	
16	214	2900	0.5732	390.0	0.5118	10	0.4845	2710	0.4162	2600	0.44354	4160	0.14330	5300	0.12283	*	*	*	
17	200	310	7.8942	290.0	7.8075	250	7.1135	200	4.5110	2200	0.60725	450	1.64825	*	*	*	*	*	
18	107	200	0.6990	500.0	0.6990	1800	0.6990	680	0.5684	1400	0.38404	2100	0.28419	4400	0.05377	*	*	*	
19	104	10	0.3429	200.0	0.2977	680	0.1507	1000	0.1545	810	0.13944	1900	0.12813	3700	0.10175	3950	0.08291	*	
20	197	210	6.0279	390.0	6.0279	500	4.4381	2120	1.0599	2000	0.86113	700	2.71587	*	*	*	*	*	
21	122	350	0.5029	150.0	0.4200	240	0.3703	100	0.3316	1400	0.02763	2500	0.02763	*	*	*	*	*	
22	100	10	0.8899	100.0	0.3699	500	0.1849	850	0.1088	1200	0.07615	2900	0.07615	3900	0.06527	*	*	*	
23	194	700	4.8776	300.0	4.1272	200	3.7520	1150	3.1088	1300	3.00158	1380	1.60799	2500	0.16080	*	*	*	
24	115	100	0.1898	15.0	0.1898	200	0.0772	700	0.0438	2200	0.02919	4310	0.03754	5400	0.02294	*	*	*	
25	98	250	0.6094	10.0	0.5290	500	0.2344	650	0.1674	1250	0.08706	1550	0.06697	2250	0.03348	3350	0.03348	*	
26	203	390	7.7334	510.0	6.7136	200	6.6287	700	4.3341	1310	2.63447	1190	1.27474	2500	0.33993	*	*	*	
27	216	100	0.2686	150.0	0.2686	500	0.1122	1250	0.0443	2260	0.02952	3200	0.02657	4250	0.02362	5200	0.02362	*	
28	119	100	3.1231	200.0	3.1231	300	0.7550	10	0.4805	510	0.27456	400	0.20592	*	*	*	*	*	
29	221	200	11.7069	320.0	11.7069	650	11.7069	500	4.7600	1100	3.73077	1400	2.70159	1250	2.57295	2450	0.64324	*	
30	108	190	0.7771	450.0	0.6149	300	0.4014	100	0.6832	510	0.38428	780	0.15371	*	*	*	*	*	
31	215	10	1.6959	2300.0	0.4286	300	0.4286	150	0.4100	2900	0.26090	4800	0.18636	5250	0.20499	*	*	*	
32	202	290	3.4508	690.0	3.4508	190	3.0716	610	2.9958	750	2.95783	900	2.27525	1000	2.25629	1300	1.13763	*	
33	116	600	0.1072	1350.0	0.1072	1000	0.0954	100	0.0977	1700	0.06242	3300	0.07302	1050	0.03180	6200	0.02355	*	
34	109	290	1.6124	210.0	1.4884	10	1.1163	390	1.0631	1110	0.35438	1020	0.23035	2320	0.10631	*	*	*	
35	207	300	12.5000	190.0	6.3187	470	4.8077	1100	1.0989	2300	0.41209	900	0.68681	*	*	*	*	*	
36	118	150	2.4252	300.0	2.0521	400	2.2653	220	1.7590	990	0.23986	20	0.63962	*	*	*	*	*	
37	116	300	0.2872	10.0	0.2496	1000	0.2393	1400	0.1402	1850	0.13677	2500	0.08548	3850	0.08206	4500	0.04445	*	
38	197	280	7.1389	460.0	6.2760	700	5.6484	1900	2.3535	860	2.11815	2700	1.01985	*	*	*	*	*	
39	108	100	0.9305	320.0	0.7771	500	0.7976	900	0.3988	220	0.36812	800	0.26587	*	*	*	*	*	
40	109	400	6.8905	290.0	6.5119	160	6.0576	1000	0.6058	800	0.45432	1300	0.22716	*	*	*	*	*	
41	221	260	7.0793	440.0	7.0793	790	4.6677	600	3.3452	850	1.63369	2750	0.46677	*	*	*	*	*	
42	110	320	0.8150	200.0	0.7165	100	0.7165	450	0.2687	1400	0.31426	1250	0.08956	*	*	*	*	*	

ROW	phoneme	f1	s1	f2	s2	f3	s3	f4	s4	f5	s5	f6	s6	f7	s7	f8	s8	f9	s9
43	102	*	*	*	*	*	*	*	*	*	*	*	*	*	*	*	*	*	
44	199	300	7.1208	440.0	7.1208	210	3.9908	550	1.5650	1820	1.09551	2000	0.86076	2500	0.39125	*	*	*	
45	212	2300	0.2323	90.0	0.2323	1910	0.1889	500	0.1481	2850	0.08170	3800	0.05362	5100	0.05106	*	*	*	
46	100	60	1.1792	150.0	0.7645	300	0.4147	850	0.1685	2100	0.12958	2600	0.12958	4000	0.10367	*	*	*	
47	211	800	4.4153	210.0	4.3183	310	3.8816	700	4.1727	1300	3.25084	1000	2.62008	*	*	*	*	*	
48	107	1600	0.9318	500.0	0.6656	1200	0.5734	850	0.4403	1700	0.30720	2270	0.16384	*	*	*	*	*	
49	103	100	0.5521	290.0	0.2427	850	0.2124	2410	0.1578	3900	0.06674	2050	0.10922	*	*	*	*	*	
50	193	800	3.7106	180.0	3.7106	300	3.3029	1200	1.3048	1400	0.97862	1550	0.89707	2500	0.28543	*	*	*	
51	112	100	0.2076	1400.0	0.1734	1040	0.1597	1800	0.0548	4060	0.03879	3850	0.03194	*	*	*	*	*	
52	214	2900	0.3878	2500.0	0.3126	90	0.3473	2100	0.1910	3800	0.15049	4700	0.08682	5450	0.10419	*	*	*	
53	200	400	7.4247	350.0	7.4247	200	4.4058	2300	1.0607	290	5.30333	700	0.40795	*	*	*	*	*	
54	122	110	2.4447	220.0	2.4447	300	2.2297	460	0.4970	2350	0.10746	350	1.34321	*	*	*	*	*	
55	114	190	1.1212	390.0	1.1212	270	1.0103	460	0.2218	920	0.17249	650	0.11088	*	*	*	*	*	
56	197	210	7.4758	400.0	6.0792	700	4.6826	500	3.0396	1000	1.15012	1700	1.23227	1550	0.90367	*	*	*	
57	108	500	0.8699	200.0	0.7647	410	0.6691	350	0.6405	270	0.53532	210	0.51620	900	0.21030	*	*	*	
58	104	1100	0.3086	10.0	0.3086	800	0.2917	700	0.2035	1350	0.17637	200	0.15602	2750	0.11532	2900	0.09497	3800	0.05087
59	194	210	4.4318	700.0	4.4318	1010	4.4318	1090	4.4318	1160	3.70131	2950	0.24351	*	*	*	*	*	
60	118	410	1.3037	210.0	1.2034	330	0.9025	110	0.8882	2500	0.08596	2050	0.07163	*	*	*	*	*	
61	98	90	0.5642	150.0	0.2232	280	0.2170	1100	0.0744	770	0.06820	1780	0.04340	*	*	*	*	*	
62	196	150	11.4688	300.0	11.4688	500	8.9482	650	5.1673	1300	2.14253	1410	1.89046	*	*	*	*	*	
63	110	200	2.3135	100.0	1.8813	350	1.7033	260	1.1186	1250	0.15254	1000	0.12711	*	*	*	*	*	
64	119	250	5.2136	150.0	4.2397	400	2.6928	620	1.1459	750	0.97398	1000	0.40105	*	*	*	*	*	
65	221	160	6.3808	300.0	6.1704	690	6.1704	980	1.4024	1900	0.98166	250	1.54261	*	*	*	*	*	
66	215	2350	0.1732	2150.0	0.1466	140	0.1218	3050	0.0800	4950	0.09329	5500	0.05711	*	*	*	*	*	
67	109	110	2.5047	220.0	2.5047	350	2.2570	10	1.0734	1000	0.27524	1100	0.24772	*	*	*	*	*	
68	202	680	6.0370	930.0	6.0370	1000	5.7717	200	5.5063	310	4.71021	1180	1.32682	1650	0.33170	*	*	*	
69	115	50	0.2297	200.0	0.0858	100	0.0833	3950	0.0555	4900	0.02398	2350	0.02524	*	*	*	*	*	
70	116	510	0.3262	90.0	0.2043	210	0.2043	1620	0.1434	3700	0.08245	2050	0.08603	1000	0.10754	*	*	*	
71	207	390	9.1265	290.0	5.0146	200	3.6105	410	4.3125	700	0.60175	1300	0.40117	*	*	*	*	*	
72	216	40	0.2828	110.0	0.2051	250	0.1119	400	0.0963	1400	0.05282	1300	0.04350	1760	0.03107	3050	0.02796	*	
73	100	20	1.0623	200.0	0.3385	480	0.3152	1200	0.1401	3800	0.10506	2150	0.08172	*	*	*	*	*	
74	199	460	13.8817	500.0	6.1019	310	5.0340	220	4.8815	2250	1.22037	2070	0.99155	*	*	*	*	*	
75	103	100	0.2433	420.0	0.2246	1810	0.1524	1270	0.0855	1550	0.08020	760	0.07485	*	*	*	*	*	
76	112	900	2.3035	990.0	2.1263	30	1.6201	510	1.2657	1100	0.83535	260	0.78472	1500	0.35439	3800	0.25314	*	
77	206	410	5.0557	220.0	4.3890	900	4.1668	510	3.5556	1110	1.00002	1800	0.44445	*	*	*	*	*	
78	108	110	0.7025	200.0	0.2162	880	0.1390	390	0.1390	1000	0.13124	1250	0.05404	*	*	*	*	*	
79	216	110	0.1035	410.0	0.0944	1150	0.0409	3200	0.0330	4350	0.02047	5700	0.02047	*	*	*	*	*	
80	211	1210	17.9368	1100.0	12.0235	800	10.8409	710	11.0380	310	9.46115	190	8.27850	1400	6.50454	*	*	*	
81	100	200	0.1876	290.0	0.1505	400	0.0701	460	0.0598	1100	0.01443	1750	0.01031	*	*	*	*	*	
82	119	100	3.1791	220.0	3.1442	350	1.3974	500	0.5240	610	0.45416	760	0.27948	*	*	*	*	*	
83	199	310	9.3751	500.0	5.4602	160	5.0482	700	1.5454	1390	1.23628	2450	0.51512	*	*	*	*	*	
84	214	2200	0.4773	2300.0	0.4773	2490	0.4668	3000	0.3409	400	0.25175	4000	0.14685	5050	0.12588	5700	0.11014	*	
85	114	150	8.1998	250.0	8.1998	100	4.9559	420	2.3428	900	1.52194	700	0.90108	*	*	*	*	*	
86	193	200	3.6376	350.0	3.6376	950	3.6376	410	2.8781	1000	2.63829	1320	1.27917	1600	0.83946	2200	0.27982	*	
87	112	120	1.0034	1000.0	0.4852	500	0.3639	400	0.2757	1500	0.23156	2250	0.09924	3950	0.09924	*	*	*	
88	215	190	15.7375	300.0	14.6998	80	9.1658	450	8.1281	2100	1.55645	2900	1.38351	*	*	*	*	*	

Row	phoneme	f1	s1	f2	s2	f3	s3	f4	s4	f5	s5	f6	s6	f7	s7	f8	s8	f9	s9
89	197	110	6.9625	300.0	5.9678	500	3.902	680	2.9839	1000	0.34162	1900	0.99464	*	*	*	*	*	*
90	108	310	0.0673	400.0	0.0643	200	0.046	500	0.0481	100	0.03401	900	0.02810	1060	0.01479	*	*	*	*
91	107	100	1.4465	450.0	0.9696	3100	0.953	3000	0.8533	4000	0.22253	2100	0.28611	*	*	*	*	*	*
92	200	210	3.7024	300.0	5.8335	390	4.781	2500	0.6694	700	0.66942	3000	0.38252	*	*	*	*	*	*
93	115	200	0.6789	220.0	0.2760	50	0.186	360	0.1194	4230	0.04476	5200	0.03730	*	*	*	*	*	*
94	118	100	2.9391	220.0	1.5503	90	1.001	300	1.1304	450	0.45217	1200	0.12919	*	*	*	*	*	*
95	194	1100	2.6346	800.0	2.1135	310	1.852	200	1.7371	500	0.86856	2150	0.14476	*	*	*	*	*	*
96	115	100	0.3685	190.0	0.1944	1900	0.048	2200	0.0445	5300	0.03847	4200	0.03240	*	*	*	*	*	*
97	103	100	0.1436	290.0	0.1436	900	0.088	2250	0.0552	2900	0.05523	3300	0.04102	4100	0.02525	*	*	*	*
98	221	310	4.6716	700.0	4.6716	200	3.285	500	2.6181	750	1.33473	1950	0.71870	*	*	*	*	*	*
99	116	50	0.8244	600.0	0.4711	4000	0.407	1800	0.3443	3700	0.32614	700	0.45297	5200	0.13589	*	*	*	*
100	212	2310	0.4731	2610.0	0.3743	2060	0.259	500	0.1196	90	0.20794	4150	0.13516	5200	0.13516	4900	0.12996	*	*
101	203	250	3.2105	440.0	2.9988	620	3.0341	800	2.0462	1500	0.70560	2505	0.35280	*	*	*	*	*	*
102	110	400	0.9904	480.0	0.9904	210	0.6204	150	0.5115	950	0.13061	1200	0.10884	*	*	*	*	*	*
103	104	1300	0.1116	100.0	0.1116	1200	0.1116	200	0.0797	460	0.06376	2090	0.02452	4300	0.01594	*	*	*	*
104	207	400	11.8191	250.0	11.1697	450	8.4422	1050	2.0781	650	0.77928	190	1.68844	*	*	*	*	*	*
105	102	400	0.1696	10.0	0.1491	250	0.1062	1300	0.0839	1850	0.05592	4000	0.02609	5000	0.02609	*	*	*	*
106	98	10	1.2670	250.0	1.2113	500	0.8354	950	0.5987	1400	0.15316	1800	0.09746	*	*	*	*	*	*
107	202	1090	6.0548	750.0	5.9883	250	5.0567	400	4.9902	900	3.72602	1160	2.32876	*	*	*	*	*	*
108	109	190	1.0712	100.0	0.6827	300	0.3884	250	0.3178	950	0.10594	1100	0.08240	*	*	*	*	*	*
109	116	10	1.0518	500.0	0.7860	390	0.5201	1200	0.3352	720	0.34675	2000	0.27740	4150	0.16182	*	*	*	*
110	193	200	3.0511	850.0	3.0511	1000	3.0511	1100	2.8499	400	2.14582	1800	0.83821	*	*	*	*	*	*
111	112	100	0.2850	1000.0	0.1253	700	0.1096	1350	0.0940	3800	0.03132	2900	0.02819	*	*	*	*	*	*
112	98	10	0.4226	300.0	0.3390	200	0.2787	850	0.0790	1160	0.06038	1900	0.02787	*	*	*	*	*	*
113	205	500	4.6253	600.0	4.6253	900	4.2695	300	3.8629	1200	1.77897	2200	0.30497	*	*	*	*	*	*
114	115	10	0.9694	200.0	0.9588	100	0.8949	350	0.8203	1800	0.10653	5250	0.05327	*	*	*	*	*	*
115	102	190	0.2856	300.0	0.2511	850	0.1067	1500	0.0628	2050	0.04394	3250	0.01569	*	*	*	*	*	*
116	211	700	11.3133	1290.0	8.9512	1180	8.2052	200	8.0809	350	4.35127	2100	1.24322	*	*	*	*	*	*
117	110	150	1.2073	300.0	0.9950	100	0.9287	270	0.5705	1400	0.10614	1160	0.07960	*	*	*	*	*	*
118	119	190	1.2353	280.0	1.1538	310	0.6923	400	0.2172	650	0.13574	800	0.10860	*	*	*	*	*	*
119	199	400	10.7557	210.0	6.6189	660	5.4369	1210	1.5365	1700	0.70917	300	1.53652	*	*	*	*	*	*
120	108	200	0.6042	310.0	0.5976	100	0.5312	500	0.5644	780	0.21248	900	0.11952	*	*	*	*	*	*
121	118	110	1.9961	220.0	1.2065	20	1.0748	320	0.4387	450	0.10968	2450	0.08774	*	*	*	*	*	*
122	193	890	5.3182	1050.0	4.2078	780	3.7987	200	4.6754	350	3.33121	1500	1.16884	1800	0.93508	*	*	*	*
123	116	10	0.3161	520.0	0.2466	1110	0.2223	2100	0.2015	4000	0.19104	3050	0.13894	5250	0.09378	*	*	*	*
124	212	10	0.5410	2790.0	0.4875	2450	0.4042	2150	0.3864	4050	0.21401	450	0.16051	4950	0.16051	*	*	*	*
125	197	420	5.5129	550.0	5.5129	220	4.1801	1050	0.7876	2150	0.72698	1900	0.72698	*	*	*	*	*	*
126	112	20	0.8264	120.0	0.5358	1020	0.2543	1140	0.2543	1500	0.21795	1900	0.12714	4000	0.06357	*	*	*	*
127	114	110	1.8391	310.0	1.8391	220	1.8189	500	0.5457	1050	0.30315	710	0.22231	*	*	*	*	*	*
128	200	400	13.3126	220.0	10.3868	1250	1.0240	2350	0.7315	750	0.73146	450	5.41283	*	*	*	*	*	*
129	216	110	0.3179	210.0	0.2026	400	0.1956	2090	0.0524	2390	0.04891	1100	0.04192	3200	0.03843	*	*	*	*
130	104	200	2.5105	1090.0	1.6277	1250	0.9380	2740	0.6621	3050	0.49659	3820	0.22071	*	*	*	*	*	*
131	194	1120	5.1675	750.0	4.3157	200	3.0664	330	2.6121	1350	1.98749	2550	0.39750	*	*	*	*	*	*
132	100	200	0.3704	400.0	0.3704	100	0.3460	500	0.2280	1080	0.04885	3150	0.02849	*	*	*	*	*	*
133	103	100	0.2258	400.0	0.1638	800	0.0992	1120	0.0868	2000	0.03684	3350	0.07195	4550	0.03225	*	*	*	*
134	221	390	10.3114	700.0	8.8383	200	6.9120	750	4.985	1900	1.58637	2150	1.35974	*	*	*	*	*	*

Row	phoneme	f1	s1	f2	s2	f3	s3	f4	s4	f5	s5	f6	s6	f7	s7	f8	s8	f9	s9
135	115	50	0.0621	110.0	0.0593	680	0.0436	500	0.0409	1800	0.02387	5500	0.03069	4150	0.02864	*	*	*	*
136	107	10	1.6093	1000.0	1.6093	200	1.4501	620	1.2379	500	1.07876	1750	0.83117	1600	0.76044	2100	0.63664	3600	0.176846
137	203	250	4.9612	510.0	1.4705	180	4.1979	400	3.9253	1400	1.19940	1100	1.19940	*	*	*	*	*	*
138	109	220	2.2336	460.0	0.4418	1310	0.3927	1150	0.2700	760	0.26999	1500	0.31908	*	*	*	*	*	*
139	214	2050	0.4462	2400.0	0.4462	10	0.3922	500	0.2697	2800	0.28436	3900	0.11277	5150	0.11277	*	*	*	*
140	207	300	3.8906	220.0	3.6768	400	2.9928	1100	0.3420	800	0.34203	1300	0.21377	*	*	*	*	*	*
141	122	150	0.3818	1600.0	0.0671	800	0.0671	3900	0.0545	4150	0.05035	10	0.13427	*	*	*	*	*	*
142	215	150	0.4021	300.0	0.4021	400	0.2386	1050	0.1105	2400	0.11488	3000	0.10604	4800	0.05302	*	*	*	*
143	202	1110	8.4661	700.0	8.2800	320	7.1636	1000	6.4193	1290	4.27956	500	3.81439	*	*	*	*	*	*
144	98	100	1.8201	200.0	0.8000	250	0.4800	350	0.4400	450	0.32001	900	0.12000	1250	0.12000	*	*	*	*
145	110	280	3.2212	400.0	3.2212	200	2.7257	40	1.0619	450	1.30973	1150	0.35398	*	*	*	*	*	*
146	106	210	8.5458	320.0	8.5458	450	7.0433	850	1.3147	700	1.50256	2450	0.65737	*	*	*	*	*	*
147	207	400	1.3849	210.0	0.8523	120	0.7914	500	0.5783	1050	0.21306	900	0.16741	*	*	*	*	*	*
148	98	290	0.5347	410.0	0.3525	40	0.2997	950	0.1175	1200	0.08813	1700	0.02938	*	*	*	*	*	*
149	206	450	8.3770	590.0	8.1929	250	6.4439	950	3.9584	1020	4.05043	800	2.85371	*	*	*	*	*	*
150	108	110	0.5120	400.0	0.4783	210	0.3207	500	0.3151	830	0.13504	1100	0.08440	*	*	*	*	*	*
151	102	300	0.2905	400.0	0.2905	270	0.2905	50	0.2586	810	0.07024	1800	0.05747	*	*	*	*	*	*
152	199	400	7.8551	600.0	7.8551	210	4.9202	2050	1.5537	2200	1.20847	2300	0.94951	*	*	*	*	*	*
153	98	100	1.0464	300.0	0.2990	370	0.2300	860	0.1150	1300	0.09199	470	0.20698	*	*	*	*	*	*
154	216	10	1.6658	200.0	0.6773	500	0.2380	1200	0.1831	1700	0.16475	3300	0.14644	*	*	*	*	*	*
155	193	1000	3.2639	820.0	2.4748	250	1.6140	1300	1.2195	1600	0.71733	500	0.68147	*	*	*	*	*	*
156	214	2600	0.4957	2300.0	0.4848	2800	0.3704	3800	0.1961	1300	0.16887	500	0.15797	30	0.20155	5200	0.14163	6100	0.108946
157	115	100	1.7477	400.0	0.3841	550	0.3841	700	0.3841	2150	0.19206	3900	0.19206	5250	0.19206	*	*	*	*
158	211	720	4.2181	900.0	4.2181	1300	4.2181	200	3.7546	350	3.38377	1000	3.24471	*	*	*	*	*	*
159	109	100	0.7438	850.0	0.3515	1350	0.2534	380	0.2616	50	0.31879	1500	0.17983	*	*	*	*	*	*
160	104	200	0.3088	50.0	0.2884	150	0.2884	400	0.2138	950	0.11197	2250	0.07465	2900	0.07125	3150	0.06786	*	*
161	200	400	8.5411	200.0	7.5086	380	6.7578	420	6.6639	2450	0.56315	3000	0.46929	*	*	*	*	*	*
162	108	150	1.2439	250.0	1.2439	500	1.2439	450	0.6698	920	0.32805	1000	0.27338	*	*	*	*	*	*
163	119	200	1.7129	260.0	0.9976	350	0.5270	750	0.1318	420	0.16940	50	0.18823	*	*	*	*	*	*
164	194	1050	9.7084	1100.0	9.7084	1200	7.3613	750	5.4409	400	6.40111	250	4.90752	2350	0.74680	*	*	*	*
165	100	450	2.3960	200.0	1.9220	110	1.3165	350	1.1322	290	1.05317	2400	0.13165	*	*	*	*	*	*
166	114	450	15.9866	310.0	13.1758	200	7.9055	260	3.5135	1300	2.63516	1400	1.75677	*	*	*	*	*	*
167	197	400	4.0352	490.0	3.4144	210	2.9266	700	2.3058	2550	0.44343	1170	0.39909	*	*	*	*	*	*
168	107	2150	0.6355	2190.0	0.6355	1950	0.6076	400	0.5727	500	0.50284	1500	0.37015	2600	0.25840	*	*	*	*
169	103	10	1.1153	400.0	0.5760	450	0.5515	750	0.3922	1950	0.36769	1900	0.36769	4050	0.12256	*	*	*	*
170	203	400	7.2454	580.0	6.6880	210	6.3696	750	5.1753	1250	1.19429	1550	1.03505	*	*	*	*	*	*
171	115	100	0.1213	4100.0	0.0680	500	0.0493	4800	0.0387	1400	0.02533	5200	0.03200	*	*	*	*	*	*
172	212	2200	0.4510	2500.0	0.4212	4200	0.1189	4400	0.1536	5400	0.14867	1300	0.12389	*	*	*	*	*	*
173	202	800	6.8898	400.0	5.5270	1150	5.6784	1300	5.0727	600	3.10421	2500	0.52999	*	*	*	*	*	*
174	112	100	0.3204	1000.0	0.2253	230	0.2218	450	0.1725	1450	0.13377	3750	0.06337	2950	0.05281	*	*	*	*
175	118	150	3.8401	250.0	2.7429	320	1.3925	500	0.7174	600	0.29539	50	1.60354	*	*	*	*	*	*
176	221	700	6.1157	800.0	6.1157	210	5.1748	400	5.7125	900	2.88984	2100	1.00808	*	*	*	*	*	*
177	116	10	0.2446	3950.0	0.1666	450	0.1720	990	0.1586	1700	0.13438	2000	0.13169	5750	0.04838	*	*	*	*
178	215	100	1.4821	300.0	1.4821	2350	1.2378	2700	1.0587	3100	0.76549	1200	0.55376	4700	0.40717	*	*	*	*
179	207	410	16.1538	350.0	10.1183	210	7.9882	600	3.3728	720	1.24260	400	7.98816	*	*	*	*	*	*
180	110	150	0.8519	100.0	0.7489	320	0.5617	400	0.3370	800	0.28085	1200	0.24340	*	*	*	*	*	*

Rad	phoneme	f1	s1	f2	s2	f3	s3	f4	s4	f5	s5	f6	s6	f7	s7	f8	s8	f9	s9
181	115	10	0.1463	420.0	0.0608	1100	0.0513	4800	0.0361	5200	0.03230	6300	0.02850	3500	0.02850	*	*	*	*
182	193	160	3.6473	990.0	3.6473	300	3.1262	1100	3.0060	700	2.56511	1550	1.36271	800	2.80558	*	*	*	*
183	107	550	0.8225	180.0	0.7231	1720	0.5875	2000	0.4971	1100	0.51522	2200	0.24405	*	*	*	*	*	*
184	119	250	2.5998	150.0	2.3427	400	1.3428	20	0.5428	500	0.59996	600	0.34284	*	*	*	*	*	*
185	200	320	9.7113	480.0	9.6045	220	5.7627	500	3.9485	800	0.74702	2350	0.53359	*	*	*	*	*	*
186	112	100	0.2473	10.0	0.2002	1550	0.1236	300	0.1325	1900	0.07065	2650	0.04415	*	*	*	*	*	*
187	102	10	0.9670	250.0	0.5951	400	0.3826	500	0.2763	800	0.11689	1750	0.12752	4600	0.09564	*	*	*	*
188	199	320	6.2273	460.0	5.4746	350	5.3377	210	3.4216	310	4.10594	1150	1.09492	*	*	*	*	*	*
189	108	100	0.3545	360.0	0.3272	150	0.2766	450	0.1402	1050	0.09739	820	0.08571	*	*	*	*	*	*
190	107	500	0.3332	20.0	0.3295	120	0.3295	1000	0.3039	2080	0.19039	2600	0.16476	3850	0.08421	*	*	*	*
191	193	200	4.2506	1100.0	3.3164	1200	2.8493	300	2.8493	900	2.38221	1550	1.54143	2150	0.98091	*	*	*	*
192	214	2300	0.4429	2600.0	0.4429	2500	0.3602	3100	0.1703	1300	0.12654	400	0.11194	4850	0.08274	*	*	*	*
193	216	300	4.7881	50.0	4.4724	500	2.6834	750	1.4206	1150	0.89447	2150	0.47354	*	*	*	*	*	*
194	211	700	5.5097	800.0	5.5097	1200	5.5097	200	3.5117	350	3.33006	1000	3.63280	*	*	*	*	*	*
195	109	110	0.4076	310.0	0.3493	210	0.2956	850	0.2413	1150	0.11644	1400	0.06718	*	*	.	*	*	*
196	104	10	1.4320	400.0	0.3147	800	0.2360	2800	0.2046	3000	0.17310	4250	0.15736	*	*	*	*	*	*
197	200	300	7.4785	280.0	7.3142	210	4.9309	400	4.1091	180	3.12290	10	0.82182	*	*	*	*	*	*
198	112	100	0.4609	10.0	0.2836	380	0.1013	550	0.0760	1050	0.06585	1720	0.05572	*	*	*	*	*	*
199	119	140	2.6821	230.0	2.4463	20	1.2084	320	1.5916	500	0.44210	700	0.23579	*	*	*	*	*	*
200	194	1180	12.1463	1090.0	12.0129	380	5.7395	1320	5.0721	750	4.67167	200	3.73734	*	*	*	*	*	*
201	122	200	0.0980	400.0	0.0560	4450	0.0452	4900	0.0409	6000	0.02153	1500	0.02045	2000	0.01830	*	*	*	*
202	114	200	4.0175	300.0	4.0175	100	3.3553	970	0.9271	10	0.83882	550	0.44149	*	*	*	*	*	*
203	197	500	6.4194	470.0	6.1372	200	6.0666	710	2.2574	2050	0.77597	1750	0.63488	*	*	*	*	*	*
204	118	120	1.2020	220.0	1.0699	400	0.6340	600	0.4491	1300	0.33023	1120	0.27739	*	*	*	*	*	*
205	103	100	0.3073	50.0	0.3039	1820	0.2803	400	0.2533	680	0.17899	1100	0.16885	3100	0.04728	4050	0.05066	*	*
206	203	320	7.0755	510.0	7.0755	200	5.7537	450	4.6651	700	1.78830	125	0.77752	*	*	*	*	*	*
207	116	10	0.1976	1750.0	0.1581	450	0.1556	1100	0.1408	2000	0.13341	3400	0.11365	4200	0.05188	*	*	*	*
208	212	2200	0.3975	2390.0	0.3931	2500	0.3844	2000	0.2970	3900	0.13540	4600	0.12230	5250	0.10919	*	*	*	*
209	202	750	9.7430	1090.0	8.0300	200	6.9593	1300	4.6039	400	4.81798	2400	0.74946	*	*	*	*	*	*
210	110	280	1.9446	410.0	1.9446	200	1.7950	500	0.8120	1250	0.42738	1350	0.29916	*	*	*	*	*	*
211	98	10	0.5156	350.0	0.3229	260	0.3060	500	0.1870	700	0.11332	1100	0.07932	*	*	*	*	*	*
212	221	200	5.0999	390.0	4.5395	650	3.5867	750	2.0175	2000	0.84064	2200	1.00877	*	*	*	*	*	*
213	100	150	1.1895	280.0	0.3529	400	0.4314	20	0.1961	2450	0.05229	3850	0.03922	*	*	*	*	*	*
214	215	180	1.0861	290.0	1.0503	400	0.4655	2050	0.1074	2600	0.13129	3000	0.08355	*	*	*	*	*	*
215	207	320	25.3987	200.0	9.2105	420	6.6985	500	2.7911	900	1.67464	2250	0.83732	*	*	*	*	*	*
216	115	160	0.2748	100.0	0.2446	400	0.1691	1650	0.0393	4300	0.03322	2610	0.03020	*	*	*	*	*	*
217	98	100	1.9757	220.0	0.7164	320	0.5211	10	0.3039	420	0.10855	1150	0.06513	*	*	*	*	*	*
218	206	410	10.0313	1000.0	6.9448	600	6.3936	300	6.1731	200	3.85820	1400	1.10234	*	*	*	*	*	*
219	107	1300	0.6397	1400.0	0.4921	920	0.3374	450	0.2460	720	0.22495	2500	0.05624	*	*	*	*	*	*
220	212	10	0.2934	2200.0	0.2743	2300	0.2591	400	0.2134	1600	0.11430	2850	0.15240	5200	0.11430	*	*	*	*
221	202	800	9.0548	700.0	8.6568	1000	6.0697	350	5.0747	200	4.17913	1130	5.17417	2100	0.39553	*	*	*	*
222	112	110	0.3957	910.0	0.1652	510	0.1261	1050	0.1218	1400	0.10871	3000	0.04348	3750	0.03913	*	*	*	*
223	98	200	7.8326	260.0	7.7465	350	5.7668	610	4.9922	500	4.90612	10	4.82005	1500	0.36072	*	*	*	*
224	193	800	3.3893	200.0	3.2775	960	2.8679	1050	2.7169	320	2.49540	1210	1.82500	1950	0.63316	*	*	*	*
225	215	2500	0.2884	2600.0	0.2820	1860	0.2535	10	0.2725	500	0.17430	3400	0.15845	4100	0.10458	4800	0.12676	5250	0.120424
226	104	10	0.8813	1120.0	0.3486	1340	0.3196	700	0.2131	1750	0.13558	2100	0.09684	3300	0.09684	*	*	*	*

Row	phoneme	f1	s1	f2	s2	f3	s3	f4	s4	f5	s5	f6	s6	f7	s7	f8	s8	f9	s9
227	211	180	3.3028	1100.0	3.3028	1000	2.9036	1220	2.6495	400	2.03249	2450	0.25406	*	*	*	*	*	*
228	214	1990	0.3183	2500.0	0.3183	2200	0.3148	10	0.2624	500	0.20989	3610	0.15742	4700	0.12594	5700	0.09795	*	*
229	107	400	0.5285	500.0	0.5285	10	0.3717	1180	0.2207	2810	0.26136	2350	0.18586	4300	0.08131	*	*	*	*
230	199	500	11.4811	580.0	9.4625	400	9.4625	250	7.6961	900	1.76633	2000	1.89250	*	*	*	*	*	*
231	108	400	0.5181	110.0	0.3701	450	0.4441	300	0.2391	1000	0.11387	250	0.26190	*	*	*	*	*	*
232	216	180	13.5092	320.0	10.0948	810	7.8680	1000	5.9381	500	5.19585	1250	4.30513	*	*	*	*	*	*
233	194	200	4.8863	1100.0	4.8863	1250	4.8863	940	3.9198	350	2.89958	500	2.52371	2650	0.42957	*	*	*	*
234	115	100	0.2719	10.0	0.1285	4100	0.0329	4000	0.0299	4780	0.02988	5250	0.02689	*	*	*	*	*	*
235	119	110	4.0678	320.0	1.3857	10	1.1175	230	1.2069	450	0.35761	500	0.22350	*	*	*	*	*	*
236	197	150	8.5739	300.0	8.5739	480	6.2185	610	5.7474	1350	0.80086	850	1.03641	*	*	*	*	*	*
237	118	200	3.5760	10.0	1.1789	380	1.1396	410	0.4323	210	0.74663	1250	0.11789	*	*	*	*	*	*
238	114	120	4.3369	230.0	4.3369	350	4.3369	410	1.7634	1020	0.81019	1100	0.71487	*	*	*	*	*	*
239	200	220	10.5922	400.0	7.5658	500	4.1903	750	0.3148	2120	0.31478	2450	0.31478	*	*	*	*	*	*
240	112	10	0.2089	100.0	0.2066	400	0.1331	1100	0.1263	1420	0.12396	2500	0.05280	3400	0.03214	*	*	*	*
241	102	200	0.2668	10.0	0.1730	500	0.1055	1050	0.0674	800	0.05277	4300	0.03225	*	*	*	*	*	*
242	203	200	5.5815	500.0	5.5202	300	5.2135	630	2.2081	1300	1.53338	1150	1.22671	*	*	*	*	*	*
243	109	250	10.4163	500.0	3.4339	1080	1.3736	1200	1.2591	1380	1.14465	550	2.06037	*	*	*	*	*	*
244	103	210	3.8072	500.0	3.8072	110	2.8449	750	1.0041	1500	0.92041	1700	0.83674	*	*	*	*	*	*
245	207	320	13.8641	210.0	7.3129	450	5.0276	190	4.7229	1100	0.76176	550	1.06647	*	*	*	*	*	*
246	115	10	0.1445	1550.0	0.0843	3810	0.0723	4300	0.0795	1900	0.06745	700	0.06022	5500	0.03613	*	*	*	*
247	110	220	1.8500	310.0	1.8500	150	1.4637	410	0.3538	1200	0.18296	2400	0.08132	*	*	*	*	*	*
248	202	700	6.3606	800.0	6.2907	1090	6.2208	200	3.6346	1200	3.14533	2100	0.55917	*	*	*	*	*	*
249	116	500	0.1581	10.0	0.1499	700	0.1417	1400	0.1036	3900	0.08176	3300	0.06813	5500	0.03543	*	*	*	*
250	212	2600	0.2715	2150.0	0.2339	2000	0.2297	10	0.2047	500	0.10442	3750	0.09607	4500	0.08354	5250	0.08771	*	*
251	221	400	5.9020	210.0	5.7723	620	3.9563	720	2.8537	2100	1.42685	1050	1.16742	*	*	*	*	*	*
252	100	100	0.5332	150.0	0.2109	450	0.1523	820	0.0879	1050	0.08202	3550	0.05273	*	*	*	*	*	*
253	205	800	0.7601	600.0	0.7267	1000	0.6181	450	0.5847	210	0.29233	10	0.28398	1200	0.20046	*	*	*	*
254	108	700	7.5295	900.0	6.4538	200	5.5437	320	4.6335	1050	4.38529	1250	3.06143	1500	1.24112	*	*	*	*
255	*	100	0.6686	10.0	0.6686	400	0.4702	900	0.2425	500	0.39674	2400	0.04408	*	*	*	*	*	*
256	103	350	3.5913	1200.0	2.5652	10	1.8154	750	1.5786	990	1.57858	1600	0.43411	*	*	*	*	*	*
257	206	280	12.9329	480.0	10.9432	120	6.6796	1000	5.3269	700	2.27392	1150	2.84240	*	*	*	*	*	*
258	100	150	0.6156	300.0	0.1827	50	0.1962	520	0.1150	1900	0.04735	3200	0.03382	*	*	*	*	*	*
259	98	120	3.0091	220.0	1.1243	10	0.8928	330	0.5952	400	0.33067	500	0.16533	*	*	*	*	*	*
260	192	1100	4.8240	1220.0	4.8240	820	4.8240	320	4.4530	200	4.29392	500	2.49153	1350	2.17347	2450	0.42409	*	*
261	216	100	0.4125	10.0	0.2357	200	0.1587	400	0.0635	850	0.05893	1400	0.04987	*	*	*	*	*	*
262	104	1000	0.1983	1090.0	0.1983	1400	0.1525	800	0.1111	100	0.10024	1650	0.06319	2750	0.04576	4050	0.03269	*	*
263	211	210	11.8787	800.0	11.8787	400	9.1375	1150	7.9627	1300	5.87409	1450	3.52446	*	*	*	*	*	*
264	109	190	1.0040	320.0	1.0040	250	0.7282	1200	0.2317	1020	0.20963	10	0.55167	*	*	*	*	*	*
265	100	100	0.2992	10.0	0.1808	200	0.2236	500	0.1414	1100	0.05260	1600	0.04603	3780	0.02959	*	*	*	*
266	199	320	11.8048	200.0	8.3023	500	10.3778	600	3.1133	1910	1.42695	2120	0.90806	*	*	*	*	*	*
267	112	100	0.7467	10.0	0.6072	350	0.2051	1100	0.1641	1500	0.13129	1650	0.05744	2300	0.10667	*	*	*	*
268	102	10	1.0348	150.0	0.4549	500	0.1819	750	0.2615	1400	0.21605	2200	0.10234	*	*	*	*	*	*
269	194	1200	3.5394	1300.0	3.2283	800	2.9560	680	2.7226	310	2.76153	180	2.72264	1450	1.20574	2600	0.22337	*	*
270	118	100	0.3421	300.0	0.3421	400	0.3421	180	0.3120	450	0.25184	2000	0.04135		*	*	*	*	*
271	119	100	4.1466	210.0	3.1441	310	1.5948	10	2.0050	400	0.54681	350	0.81897	*	*	*	*	*	*
272	197	200	5.4490	380.0	4.6706	580	3.9520	700	2.5149	990	1.25746	1750	0.89819	*	*	*	*	*	*

row	phoneme	f1	s1	f2	s2	f3	s3	f4	s4	f5	s5	f6	s6	f7	s7	f8	s8	f9	s9
273	115	10	0.1771	110.0	0.1771	190	0.1051	1010	0.0331	1450	0.03503	4100	0.03503	5250	0.02919	*	*	*	*
274	114	320	8.7407	250.0	6.8197	110	3.8421	10	2.1131	450	2.01709	900	0.86447	*	*	*	*	*	*
275	200	310	16.0208	200.0	9.6829	550	2.8168	10	2.2887	2200	0.88026	2450	0.88026	*	*	*	*	*	*
276	214	2400	0.3885	2250.0	0.3159	30	0.2946	480	0.1580	780	0.11099	4200	0.11526	5250	0.10246	*	*	*	*
277	215	220	0.2014	10.0	0.1979	2500	0.1519	2100	0.1095	2800	0.10247	2450	0.07420	*	*	*	*	*	*
278	203	320	8.3647	190.0	6.5263	550	4.2283	700	2.9414	1500	1.47072	1720	1.28688	*	*	*	*	*	*
279	107	1800	0.7319	450.0	0.7319	820	0.6434	280	0.5952	1050	0.48256	2300	0.20911	2750	0.14477	3250	0.12064	4320	0.080426
280	110	120	1.0503	220.0	1.0503	330	1.0503	10	0.3116	410	0.28854	1150	0.08079	*	*	*	*	*	*
281	207	230	9.3843	400.0	7.9406	320	7.4250	800	1.0312	1220	0.72187	1000	0.61875	*	*	*	*	*	*
282	115	10	0.8281	350.0	0.1183	4550	0.0910	3950	0.0819	4860	0.07735	4280	0.07280	*	*	*	*	*	*
283	103	1260	0.2832	1400.0	0.2116	10	0.2116	480	0.1680	180	0.15560	800	0.10581	200	0.03734	4300	0.02490	*	*
284	202	200	6.2031	350.0	6.2031	660	6.2031	800	6.2031	1050	6.20308	500	3.47645	1780	0.47716	*	*	*	*
285	116	10	0.3165	500.0	0.1530	1600	0.1496	3250	0.1496	2200	0.12173	5100	0.06608	*	*	*	*	*	*
286	212	2150	0.3599	2600.0	0.2382	1450	0.2329	10	0.2435	480	0.12704	3900	0.16939	4250	0.09528	5410	0.07411	*	*
287	221	700	6.7460	800.0	4.0031	320	3.4842	200	2.9653	520	2.00156	900	1.48263	*	*	*	*	*	*
288	108	500	0.3766	400.0	0.3062	150	0.2690	900	0.2235	750	0.12001	1920	0.03311	*	*	*	*	*	*
289	102	10	0.1312	300.0	0.0840	620	0.0787	2000	0.0595	1700	0.05422	1350	0.03148	2450	0.03148	4050	0.02274	*	*
290	194	1200	7.8458	700.0	7.0699	800	6.7250	1000	5.9490	200	3.96603	1350	3.70737	*	*	*	*	*	*
291	118	180	0.6208	350.0	0.5048	100	0.3684	550	0.3752	1900	0.06481	2000	0.04775	*	*	*	*	*	*
292	106	100	1.7350	210.0	1.4109	310	0.8199	450	0.2097	20	0.18113	2250	0.05720	*	*	*	*	*	*
293	221	310	5.0810	500.0	4.6902	650	4.4110	150	4.0202	1900	1.34006	2150	0.94921	*	*	*	*	*	*
294	115	10	0.1265	180.0	0.0890	450	0.0570	1420	0.0334	4250	0.03476	5600	0.02781	4500	0.03337	*	*	*	*
295	104	10	1.0079	1100.0	0.4984	1400	0.3323	200	0.3434	800	0.10936	2900	0.11076	4250	0.09968	*	*	*	*
296	211	700	5.9529	1250.0	5.9529	820	5.9529	950	5.4950	300	3.79415	200	2.94373	1500	0.78500	100	3.92498	*	*
297	212	2200	0.0853	2400.0	0.0853	2300	0.0853	1900	0.0732	500	0.05627	4200	0.03189	4800	0.03001	5300	0.02813	*	*
298	103	100	0.1651	10.0	0.1542	200	0.1361	500	0.0726	650	0.05806	2400	0.05625	2700	0.04173	*	*	*	*
299	193	720	4.5371	310.0	3.8889	820	3.7393	200	2.4929	500	2.39317	1650	1.54559	*	*	*	*	*	*
300	115	10	0.1845	900.0	0.0598	680	0.0549	4350	0.0574	4150	0.03740	4650	0.03740	*	*	*	*	*	*
301	216	20	0.7343	100.0	0.2017	300	0.1614	700	0.0807	2200	0.07666	4200	0.04842	*	*	*	*	*	*
302	199	500	7.3571	250.0	5.2551	520	2.9913	800	1.2127	2200	0.97017	1950	0.80847	*	*	*	*	*	*
303	110	360	0.5263	200.0	0.4511	100	0.4164	10	0.2487	900	0.05205	1400	0.05205	*	*	*	*	*	*
304	102	10	0.2882	120.0	0.0855	410	0.0823	1080	0.0697	1400	0.05383	4300	0.03800	4800	0.03483	*	*	*	*
305	197	410	4.8201	280.0	3.2841	500	3.4430	150	2.0658	720	2.11875	2020	0.95344	*	*	*	*	*	*
306	107	1900	0.4361	780.0	0.3163	10	0.2971	380	0.2013	1380	0.16294	4300	0.05272	*	*	*	*	*	*
307	214	10	0.3149	480.0	0.1618	1900	0.1881	2700	0.1618	3300	0.10498	5200	0.09623	2300	0.14435	*	*	*	*
308	194	720	8.3992	820.0	6.8301	1300	5.4457	1100	4.1535	200	3.87657	320	3.96887	*	*	*	*	*	*
309	109	300	3.1791	200.0	2.2708	280	2.3756	320	2.3407	1200	0.20961	1100	0.17468	*	*	*	*	*	*
310	119	100	2.5243	250.0	1.9695	400	1.1096	10	0.8322	500	0.52705	650	0.38835	*	*	*	*	*	*
311	200	350	6.4557	200.0	5.9591	220	0.7804	2320	0.4257	2500	0.42565	2400	0.56753	*	*	*	*	*	*
312	118	200	1.0737	10.0	0.4955	350	0.2124	220	0.2242	2880	0.04719	950	0.04719	*	*	*	*	*	*
313	215	10	1.1747	200.0	1.1360	2400	0.6196	2780	0.5293	3020	0.42601	4800	0.25819	*	*	*	*	*	*
314	221	220	8.6713	400.0	8.6713	720	8.6713	550	6.0985	1950	2.66809	2000	2.09635	*	*	*	*	*	*
315	116	10	0.1552	410.0	0.0921	650	0.0921	2060	0.0751	3700	0.04947	4100	0.04264	5000	0.02900	*	*	*	*
316	114	300	3.7409	200.0	3.4942	100	2.8776	320	2.0554	750	0.69884	950	0.36997	*	*	*	*	*	*
317	202	800	5.4084	350.0	5.2301	200	4.6357	1000	4.6357	650	3.56596	500	1.78298	*	*	*	*	*	*
318	98	180	1.1118	300.0	0.6353	250	0.2932	450	0.2688	1050	0.08552	580	0.08552	*	*	*	*	*	*

Row	phoneme	f1	s1	f2	s2	f3	s3	f4	s4	f5	s5	f6	s6	f7	s7	f8	s8	f9	s9
319	100	10	0.5618	120.0	0.2222	500	0.1420	310	0.1235	1000	0.09877	2300	0.06173	3250	0.05865	*	*	*	*
320	203	550	9.5060	400.0	8.3569	200	5.3275	700	2.4026	500	2.50707	1300	1.77584	1500	1.67138	*	*	*	*
321	112	120	0.2303	10.0	0.1771	1000	0.0961	1400	0.0835	300	0.06832	1700	0.04554	2000	0.04048	*	*	*	*
322	108	300	0.9980	420.0	0.9322	200	0.5045	50	0.1974	800	0.08774	1400	0.08774	1500	0.07677	*	*	*	*
323	207	300	4.6125	1.6	2.4330	400	2.3823	1150	0.4055	1250	0.35481	1300	0.30412	*	*	*	*	*	*
324	122	10	0.1379	110.0	0.1379	3900	0.0424	1450	0.0333	4210	0.03485	5400	0.01970	*	*	*	*	*	*
325	112	20	0.7589	130.0	0.7589	880	0.7339	1000	0.5087	350	0.42533	1250	0.20850	2000	0.10842	1000	0.06672	*	*
326	204	900	5.3865	680.0	4.7354	560	4.1435	450	4.0251	220	3.07801	1100	2.84124	*	*	*	*	*	*
327	*	10	0.2244	660.0	0.2244	920	0.2244	1000	0.1800	1060	0.08136	260	0.11341	*	*	*	*	*	*
328	104	10	0.2793	110.0	0.2271	450	0.2056	1200	0.1535	1000	0.12584	2000	0.04604	*	*	*	*	*	*
329	206	350	5.1546	450.0	4.8147	1000	4.0217	200	3.6252	750	1.81261	1280	0.62309	*	*	*	*	*	*
330	100	110	0.5659	10.0	0.1928	200	0.3358	390	0.1244	480	0.11193	150	0.04975	*	*	*	*	*	*
331	215	10	0.9113	3100.0	0.2604	2620	0.2003	2250	0.1602	1100	0.16023	230	0.18026	5400	0.14020	*	*	*	*
332	211	180	5.8238	310.0	5.8238	1300	5.1198	800	4.7359	500	3.71189	1000	3.19990	2100	0.57598	*	*	*	*
333	103	100	0.5870	200.0	0.1935	300	0.1742	1600	0.1548	1000	0.10965	700	0.09030	*	*	*	*	*	*
334	109	250	2.1125	150.0	1.5090	10	0.8822	250	1.8108	480	0.55716	1270	0.30179	*	*	*	*	*	*
335	193	300	4.0847	750.0	4.0847	890	4.0847	170	3.2319	1100	2.87277	1200	2.42390	1600	1.16706	*	*	*	*
336	214	2300	0.2713	2500.0	0.2683	2050	0.1670	20	0.2385	3100	0.07752	500	0.08050	5300	0.05963	*	*	*	*
337	119	110	1.1837	220.0	1.1837	350	0.7805	460	0.2732	600	0.11707	700	0.10406	*	*	*	*	*	*
338	199	300	8.5260	180.0	4.8720	460	4.6846	520	2.9982	1000	1.21800	1100	1.12431	*	*	*	*	*	*
339	112	120	0.4125	400.0	0.1224	1400	0.1133	2150	0.0544	2400	0.04986	900	0.07252	*	*	*	*	*	*
340	102	10	1.1675	270.0	0.6286	150	0.5517	320	0.6286	800	0.19244	1600	0.12829	*	*	*	*	*	*
341	197	500	3.7269	310.0	3.1535	200	2.8668	700	2.0477	900	0.61432	1800	0.49145	2150	0.45050	*	*	*	*
342	216	100	0.1237	10.0	0.1156	460	0.0476	1020	0.0286	280	0.04079	2250	0.02040	*	*	*	*	*	*
343	115	200	0.2271	300.0	0.0524	700	0.0374	4350	0.0349	5000	0.03244	4700	0.02994	*	*	*	*	*	*
344	194	1200	4.8702	850.0	4.4956	200	4.0674	360	2.9435	520	1.28446	1500	0.80278	*	*	*	*	*	*
345	110	150	0.8192	80.0	0.5491	290	0.5401	400	0.4051	500	0.36009	560	0.21606	*	*	*	*	*	*
346	98	100	1.5630	300.0	1.5630	410	1.5630	150	0.5668	750	0.20611	1250	0.10306	*	*	*	*	*	*
347	200	400	6.4063	300.0	5.9839	200	3.8719	410	2.1324	2310	0.31680	2450	0.21120	*	*	*	*	*	*
348	115	100	0.1951	10.0	0.1724	4700	0.0520	5250	0.0488	4200	0.04553	4400	0.04878	*	*	*	*	*	*
349	104	180	1.0516	10.0	1.0516	780	0.4391	620	0.3120	1050	0.32356	2910	0.16178	*	*	*	*	*	*
350	221	200	4.0511	750.0	4.0511	400	3.7395	1100	1.7807	1280	1.69166	1400	1.64715	*	*	*	*	*	*
351	108	520	0.1586	400.0	0.1481	300	0.1342	100	0.1289	1300	0.03136	1700	0.02265	*	*	*	*	*	*
352	114	200	2.1047	300.0	2.1047	100	1.7115	10	0.9252	500	0.39319	700	0.27755	*	*	*	*	*	*
353	202	800	6.1189	700.0	4.3706	310	3.7654	200	3.4965	980	2.55513	1200	1.54652	*	*	*	*	*	*
354	100	160	0.6057	280.0	0.1864	50	0.1664	450	0.1265	3000	0.08653	2200	0.05991	800	0.09984	*	*	*	*
355	118	120	2.2890	10.0	1.0062	300	0.8552	200	0.7798	380	0.27670	500	0.15093	*	*	*	*	*	*
356	203	400	6.4291	210.0	5.5107	520	4.2390	350	1.9075	1250	0.63585	900	0.49455	*	*	*	*	*	*
357	116	180	0.1917	750.0	0.1917	1800	0.1846	3400	0.1349	1000	0.13845	400	0.15975	*	*	*	*	*	*
358	212	10	0.1987	2000.0	0.1987	2200	0.1921	380	0.0808	2700	0.07424	3000	0.06987	3900	0.05895	*	*	*	*
359	206	310	4.2994	620.0	4.2994	180	3.6852	1100	3.5435	1000	2.36231	1150	2.55129	*	*	*	*	*	*
360	107	1400	0.6684	1380.0	0.6023	10	0.3158	280	0.1983	1500	0.24238	2250	0.06610	4080	0.04407	*	*	*	*
361	100	10	0.2878	230.0	0.1478	500	0.0583	2010	0.0428	3000	0.03500	3650	0.03500	*	*	*	*	*	*
362	197	310	6.0780	190.0	5.7441	500	5.6773	680	5.2097	1900	1.33[illegible]	1000	1.00187	*	*	*	*	*	*
363	108	450	0.5649	300.0	0.2669	20	0.1738	1800	0.1117	750	0.10[illegible]	1100	0.08690	*	*	*	*	*	*
364	119	210	2.7765	320.0	2.7765	450	1.0984	60	5.9459	560	0.610[illegible]	100	0.33563	800	0.33563	*	*	*	*

Raw	phoneme	f1	s1	f2	s2	f3	s3	f4	s4	f5	s5	f6	s6	f7	s7	f8	s8	f9	s9
365	199	300	8.4863	490.0	6.9942	190	4.3830	1120	1.9584	1800	0.55954	1000	0.83930	*	*	*	*	*	*
366	214	2400	0.6196	1900.0	0.4017	10	0.3064	500	0.1975	1800	0.26554	5200	0.16341	2850	0.21107	*	*	*	*
367	109	250	1.5744	350.0	1.1592	110	0.9689	10	0.2941	800	0.10381	1320	0.10381	*	*	*	*	*	*
368	193	190	6.3464	820.0	5.9279	700	4.4634	310	4.1844	1000	2.44091	1200	1.74350	1500	1.25532	*	*	*	*
369	110	200	1.2846	100.0	0.7623	310	0.5082	610	0.2117	750	0.14116	10	0.33879	*	*	*	*	*	*
370	104	10	0.3096	170.0	0.0968	2250	0.0774	2100	0.0581	800	0.05321	3600	0.04354	*	*	*	*	*	*
371	199	510	6.1175	320.0	6.0502	200	5.5124	600	2.4201	1800	1.34449	2400	0.30670	*	*	*	*	*	*
372	112	100	0.2543	1000.0	0.0755	300	0.0671	680	0.0587	1400	0.05589	1800	0.04192	2300	0.03633	*	*	*	*
373	216	10	0.2243	100.0	0.1217	500	0.1057	1100	0.0609	3500	0.04165	2200	0.04165	*	*	*	*	*	*
374	211	1200	5.8401	820.0	5.1341	1100	5.0058	300	4.2998	180	3.01631	1000	3.59390	1500	0.57759	*	*	*	*
375	103	100	0.2403	1500.0	0.1452	500	0.0951	1250	0.0845	1010	0.07658	2000	0.04225	2700	0.03169	*	*	*	*
376	114	310	2.9104	220.0	2.8784	140	2.6225	10	1.4072	450	1.40721	1000	0.54369	*	*	*	*	*	*
377	194	1100	5.3680	800.0	4.9550	200	4.6011	310	4.5421	980	4.12921	1250	3.24438	2300	0.41292	*	*	*	*
378	100	110	0.1574	200.0	0.1557	90	0.1522	260	0.0969	400	0.06573	800	0.03460	3700	0.03287	1750	0.03287	*	*
379	115	10	0.9583	190.0	0.4739	300	0.3475	500	0.2106	1750	0.16850	1600	0.12637	*	*	*	*	*	*
380	200	350	9.6317	200.0	3.8104	500	1.4818	530	0.9526	700	0.52922	2400	0.42337	*	*	*	*	*	*
381	215	100	0.4446	10.0	0.3273	200	0.1856	2480	0.1710	2200	0.15633	3450	0.06351	4950	0.05374	*	*	*	*
382	118	110	3.1354	220.0	2.1017	10	1.1370	400	0.6891	500	0.55127	1300	0.20673	*	*	*	*	*	*
383	197	700	8.4390	200.0	7.1407	360	5.1932	500	3.9876	800	2.31840	2000	1.20557	*	*	*	*	*	*
384	108	300	0.5124	510.0	0.5124	450	0.4279	100	0.4054	800	0.18017	1000	0.12387	*	*	*	*	*	*
385	214	2300	1.1313	20.0	1.1313	1950	1.1064	2410	0.9945	400	0.03401	2450	0.31079	5000	0.26106	*	*	*	*
386	203	200	3.7185	400.0	3.7185	550	2.9830	750	1.1850	1500	0.89898	2280	0.65380	*	*	*	*	*	*
387	122	110	0.2323	290.0	0.0664	3600	0.0485	3850	0.0383	460	0.03574	5200	0.02553	*	*	*	*	*	*
388	212	2200	0.4225	2000.0	0.3482	2300	0.3528	400	0.1625	2800	0.14391	3600	0.12998	4750	0.11606	*	*	*	*
389	207	400	11.2494	300.0	6.1810	200	3.5850	450	3.0905	520	0.74172	730	0.61810	*	*	*	*	*	*
390	116	10	0.2372	200.0	0.1616	510	0.1251	910	0.0730	2200	0.07820	3400	0.09384	3520	0.09124	*	*	*	*
391	119	220	5.4161	520.0	5.0590	100	4.2853	710	2.6783	410	2.20215	1000	1.01180	1900	0.89276	*	*	*	*
392	221	310	7.2215	620.0	6.7453	200	5.7930	780	3.5711	1000	1.26971	1750	1.11100	1900	1.11100	*	*	*	*
393	98	200	2.7508	310.0	2.3578	10	1.2091	470	0.3464	300	0.45343	900	0.33252	*	*	*	*	*	*
394	107	10	0.5203	1250.0	0.5203	1340	0.5203	940	0.4059	600	0.37161	200	0.33731	1700	0.11434	3500	0.04002	*	*
395	202	1080	8.7058	780.0	5.9314	900	5.8357	1150	3.6354	200	3.77437	300	3.58304	*	*	*	*	*	*
396	102	10	0.0974	400.0	0.0600	1000	0.0480	1200	0.0465	2250	0.03597	3300	0.02998	4000	0.02248	4500	0.02248	*	*
397	114	220	2.6562	100.0	2.2475	380	2.1600	810	0.5254	10	0.49621	1000	0.37946	*	*	*	*	*	*
398	211	1200	7.7459	750.0	5.6179	420	5.1072	220	4.7667	900	4.25598	1300	3.40478	*	*	*	*	*	*
399	212	10	0.1531	2300.0	0.1531	2200	0.1077	500	0.0639	3500	0.04207	5150	0.05048	*	*	*	*	*	*
400	98	10	1.3816	230.0	1.2601	410	1.1235	1050	0.7136	1800	0.10627	2300	0.09109	*	*	*	*	*	*
401	192	700	5.3019	1010.0	5.3019	300	2.0975	150	1.9809	1180	2.21399	1300	0.99047	*	*	*	*	*	*
402	107	1400	0.3150	1600.0	0.3150	1850	0.3116	1400	0.2804	1120	0.24580	140	0.20772	400	0.20079	4100	0.04847	*	*
403	104	50	0.1972	150.0	0.1972	250	0.1257	870	0.1279	1300	0.08668	1950	0.10335	2300	0.06935	3700	0.03901	*	*
404	193	200	3.5333	750.0	3.5333	380	2.6403	1100	2.1744	1500	1.28132	1300	0.66007	*	*	*	*	*	*
405	118	100	0.2689	300.0	0.2689	700	0.2689	510	0.1711	900	0.13591	1500	0.04727	2200	0.04136	*	*	*	*
406	119	100	2.5892	220.0	1.5364	10	1.0527	800	0.9958	410	0.39834	550	0.22762	*	*	*	*	*	*
407	199	310	10.7350	450.0	5.3085	200	4.0109	720	1.1737	1100	0.94374	2250	5.70780	*	*	*	*	*	*
408	122	120	0.1965	10.0	0.0929	480	0.0410	1450	0.0281	5100	0.02376	2400	0.01944	*	*	*	*	*	*
409	98	220	0.1969	10.0	0.1536	480	0.1514	1150	0.0541	900	0.03894	1950	0.02330	*	*	*	*	*	*
410	211	700	7.7225	1050.0	5.0918	750	5.0069	550	3.8188	200	3.64909	600	2.46101	*	*	*	*	*	*

Row	phoneme	f1	s1	f2	s2	f3	s3	f4	s4	f5	s5	f6	s6	f7	s7	f8	s8	f9	s9
411	102	80	0.3723	10.0	0.2700	200	0.1023	900	0.0982	1200	0.06547	1800	0.04501	2300	0.04092	*	*	*	*
412	109	300	7.0653	180.0	5.5902	400	2.5622	1060	2.0963	1190	1.94103	510	0.77641	*	*	*	*	*	*
413	194	800	3.4251	650.0	2.6347	1300	2.5594	1620	2.4842	1500	0.63986	350	1.84429	200	2.10776	*	*	*	*
414	115	10	0.0983	280.0	0.0540	5000	0.0389	5510	0.0378	720	0.03350	1800	0.03350	2900	0.02053	*	*	*	*
415	116	50	0.2207	220.0	0.1795	390	0.1552	600	0.1310	750	0.10914	2200	0.07276	4800	0.04365	*	*	*	*
416	200	330	6.7890	200.0	4.3271	10	1.3429	510	1.1191	2220	0.59684	2450	0.59684	*	*	*	*	*	*
417	216	10	0.2301	250.0	0.1062	380	0.0657	2300	0.0354	1850	0.03287	4200	0.02022	*	*	*	*	*	*
418	103	500	18.4621	280.0	15.6218	150	10.9556	10	10.5498	750	6.28930	1000	4.05761	1800	2.02881	*	*	*	*
419	197	600	4.6631	280.0	4.6631	150	4.3044	1900	1.2298	2100	1.17859	1000	0.71740	*	*	*	*	*	*
420	215	2200	0.2252	2050.0	0.2252	2400	0.2104	10	0.2054	450	0.06188	3100	0.07425	5250	0.06188	3300	0.06930	*	*
421	112	10	0.8048	1000.0	0.6280	100	0.3803	400	0.2830	1350	0.21227	1700	0.15920	2300	0.07960	*	*	*	*
422	203	400	4.2728	520.0	4.2728	200	3.6155	700	2.1129	1350	1.40862	1020	0.84517	*	*	*	*	*	*
423	214	10	0.2455	2300.0	0.2371	450	0.1227	1900	0.1534	3400	0.10601	4850	0.09206	5250	0.07811	*	*	*	*
424	114	100	0.4190	250.0	0.3407	1050	0.1842	10	0.2164	1010	0.12893	750	0.07828	*	*	*	*	*	*
425	207	300	9.0403	200.0	6.2587	480	5.2652	560	2.8810	300	1.98688	1000	1.58950	*	*	*	*	*	*
426	108	100	0.4975	410.0	0.4975	200	0.3991	250	0.2296	850	0.10934	950	0.07107	*	*	*	*	*	*
427	100	10	0.1507	100.0	0.1507	200	0.1507	300	0.0745	750	0.05301	2100	0.02982	3350	0.02485	*	*	*	*
428	221	150	5.5690	290.0	5.5690	450	4.0391	600	4.4063	750	3.30471	900	1.52996	1900	1.22397	*	*	*	*
429	110	300	5.8263	150.0	5.7622	350	0.9604	500	0.5762	300	0.32012	10	0.44817	*	*	*	*	*	*
430	107	1300	0.2942	10.0	0.2502	520	0.2232	800	0.2164	1500	0.16909	2200	0.06763	4100	0.04734	*	*	*	*
431	202	800	7.3308	1000.0	6.0418	700	4.6724	300	3.2223	200	2.73897	1350	1.12781	*	*	*	*	*	*
432	212	10	0.1861	2350.0	0.1227	2150	0.0982	500	0.0532	3500	0.04908	5200	0.04295	*	*	*	*	*	*
433	107	1200	0.5492	1300.0	0.5492	700	0.2716	520	0.2716	10	0.21725	110	0.20518	350	0.19915	*	*	*	*
434	205	680	7.7192	1000.0	7.7192	800	6.7013	220	4.7503	400	4.07168	460	3.90202	*	*	*	*	*	*
435	108	300	0.6482	500.0	0.6482	180	0.5058	10	0.4986	1000	0.37042	1100	0.28494	700	0.25644	3100	0.05699	*	*
436	112	20	0.3069	300.0	0.3069	1000	0.2023	800	0.1821	1300	0.11803	1800	0.06070	2100	0.05058	*	*	*	*
437	206	300	3.0825	520.0	3.0825	180	2.9470	1020	2.7777	440	2.54055	750	1.42271	10	1.76145	*	*	*	*
438	116	10	0.1505	490.0	0.1264	3590	0.1114	3310	0.1083	1500	0.09029	1050	0.07524	*	*	*	*	*	*
439	107	10	1.2747	450.0	0.4202	610	0.3782	1010	0.2381	2300	0.23813	3510	0.22413	3000	0.21012	*	*	*	*
440	199	410	4.7688	680.0	4.0876	210	3.2491	2120	1.5721	1900	1.04809	750	0.94328	*	*	*	*	*	*
441	115	50	0.0397	500.0	0.0397	200	0.0397	5200	0.0235	4100	0.02006	1000	0.02093	5750	0.01744	*	*	*	*
442	98	400	0.2517	50.0	0.1995	1000	0.0860	1500	0.0522	2500	0.05219	2100	0.04298	*	*	*	*	*	*
443	211	300	6.2908	750.0	6.2908	300	4.8391	1310	4.9082	1200	4.56256	180	4.14778	600	1.93563	*	*	*	*
444	122	100	1.0735	200.0	0.2949	280	0.2949	350	0.1180	5100	0.04719	1200	0.04719	*	*	*	*	*	*
445	104	1900	1.2204	10.0	1.2204	1180	1.0192	1500	0.5633	2350	0.49620	3510	0.33527	*	*	*	*	*	*
446	193	150	3.4612	290.0	3.4612	800	3.4612	910	3.4612	620	2.73850	1500	1.40728	1700	0.95087	2520	0.45642	*	*
447	212	2300	0.2222	2200.0	0.1954	2060	0.1832	520	0.0659	3700	0.05861	1300	0.04640	5250	0.04884	*	*	*	*
448	216	10	0.1872	90.0	0.1872	300	0.1316	480	0.1296	700	0.04525	1100	0.03702	3200	0.03085	*	*	*	*
449	200	400	11.8163	320.0	9.8685	200	8.3103	550	0.9089	650	0.90894	2300	0.77309	*	*	*	*	*	*
450	118	200	0.7949	100.0	0.6989	310	0.6639	250	0.3582	10	0.27954	450	0.14051	2300	0.05241	*	*	*	*
451	103	100	0.1701	410.0	0.1474	2400	0.1043	2050	0.0885	800	0.06805	1000	0.06125	3550	0.02949	*	*	*	*
452	197	500	6.5551	300.0	5.4026	150	4.8263	610	4.0339	2090	1.65679	2280	1.22458	*	*	*	*	*	*
453	116	10	0.2829	1000.0	0.1119	1950	0.1150	3780	0.0808	700	0.09637	1400	0.06830	5000	0.04974	*	*	*	*
454	119	100	0.8615	200.0	0.5396	300	0.3314	10	0.1420	800	0.06627	500	0.05660	*	*	*	*	*	*
455	194	1000	9.3443	700.0	8.9336	380	7.2906	200	5.9557	520	2.66981	1400	2.66981	*	*	*	*	*	*
456	102	100	0.6755	10.0	0.2598	300	0.1485	450	0.0891	2000	0.06631	1560	0.06661	*	*	*	*	*	*

Row	phoneme	f1	s1	f2	s2	f3	s3	f4	s4	f5	s5	f6	s6	f7	s7	f8	s8	f9	s9
457	112	10	0.2484	1000.0	0.1829	170	0.1338	400	0.1228	1100	0.10920	1300	0.09009	3750	0.03549	*	*	*	*
458	203	700	8.9678	1050.0	8.4751	250	7.5881	450	5.4201	830	5.02591	1250	1.97095	*	*	*	*	*	*
459	108	120	0.5296	310.0	0.5238	450	0.5121	280	0.3492	900	0.19787	700	0.12803	*	*	*	*	*	*
460	114	250	6.9800	350.0	6.9800	150	3.2983	1000	1.6875	1100	1.07385	10	1.30396	*	*	*	*	*	*
461	221	310	6.4967	770.0	6.4967	200	4.3549	500	2.7843	1000	1.85620	1550	1.28506	*	*	*	*	*	*
462	214	2100	0.5172	2400.0	0.5172	1950	0.4433	10	0.2557	3600	0.14776	5400	0.14208	400	0.14208	*	*	*	*
463	100	10	0.7113	180.0	0.3205	500	0.2189	3000	0.2345	2100	0.14069	1700	0.16414	*	*	*	*	*	*
464	202	800	6.8436	700.0	5.3395	1000	4.3619	330	3.4594	200	2.55695	1250	2.10572	*	*	*	*	*	*
465	215	10	0.1738	100.0	0.1738	2020	0.1528	2400	0.1547	400	0.11077	3200	0.07639	4920	0.07066	*	*	*	*
466	109	100	1.0343	200.0	1.0343	300	0.9093	10	0.1819	1100	0.17049	580	0.10229	*	*	*	*	*	*
467	207	300	18.4772	200.0	9.9493	500	4.4670	1100	2.2335	700	1.21828	1000	1.21828	*	*	*	*	*	*
468	110	200	1.4362	360.0	1.3257	110	1.1363	10	0.4735	1150	0.12626	1500	0.09469	*	*	*	*	*	*
469	215	50	0.7427	200.0	0.7346	350	0.2775	400	0.2857	2200	0.24486	800	0.19589	3850	0.14691	2800	0.17956	*	*
470	205	920	6.1435	400.0	4.7933	200	4.3882	610	4.7258	1050	4.59075	1400	0.74262	*	*	*	*	*	*
471	*	20	0.1824	190.0	0.1824	310	0.1824	590	0.1824	1900	0.07017	2050	0.06816	1400	0.04210	3420	0.02807	*	*
472	98	250	4.8888	10.0	2.1489	450	2.1489	850	0.8596	1850	0.69839	1250	0.53723	*	*	*	*	*	*
473	200	300	16.6902	250.0	6.2359	190	5.5022	500	1.5590	520	1.46727	2320	0.73363	*	*	*	*	*	*
474	100	100	0.3809	200.0	0.3641	250	0.3139	350	0.1130	1300	0.04604	1800	0.03348	3600	0.03348	*	*	*	*
475	119	280	3.3882	170.0	1.6755	420	0.7447	950	0.2979	300	0.24202	600	0.26063	*	*	*	*	*	*
476	199	320	11.4698	200.0	8.1927	500	8.0667	1200	1.5125	1400	1.51251	1000	1.38647	*	*	*	*	*	*
477	212	2300	0.2765	2350.0	0.2765	2000	0.2127	450	0.1671	10	0.23091	1300	0.06076	*	*	*	*	*	*
478	100	10	0.1656	250.0	0.0910	300	0.0546	4020	0.0291	3520	0.02547	2200	0.02184	*	*	*	*	*	*
479	211	800	6.9165	1250.0	6.1564	200	3.8763	330	3.2682	1420	2.35615	2550	0.53203	*	*	*	*	*	*
480	214	2300	0.3348	2400.0	0.3348	2600	0.2244	10	0.2134	180	0.11036	1850	0.11404	3350	0.09932	5300	0.08829	*	*
481	215	10	1.0094	100.0	0.9207	2400	0.743[illegible]	300	0.6877	1000	0.39933	3800	0.26622	1900	0.32168	*	*	*	*
482	193	200	8.7697	320.0	5.8786	820	5.78[illegible]	700	4.8185	1000	4.14392	500	3.08385	1450	1.54192	*	*	*	*
483	109	200	0.5556	310.0	0.5556	100	0.3236	250	0.2992	1200	0.09159	1000	0.06106	*	*	*	*	*	*
484	104	180	0.1831	10.0	0.1553	100	0.1159	480	0.1090	620	0.10200	2200	0.06723	2500	0.05332	*	*	*	*
485	200	320	14.0489	280.0	10.0349	190	6.0210	10	1.5438	2320	0.77192	2400	0.77192	*	*	*	*	*	*
486	216	10	0.1843	300.0	0.1114	400	0.0749	1860	0.0486	2300	0.03847	4250	0.02632	*	*	*	*	*	*
487	108	300	3.9535	410.0	3.3019	200	2.0419	10	0.3124	500	0.52135	520	0.26067	*	*	*	*	*	*
488	197	600	11.7112	150.0	10.1668	300	8.2364	450	6.9495	800	3.73213	1800	1.80172	1900	1.41563	*	*	*	*
489	122	100	0.4515	200.0	0.2679	10	0.2233	310	0.1984	420	0.18356	1400	0.03969	4300	0.02977	5300	0.03473	*	*
490	98	10	0.3490	110.0	0.2723	300	0.2685	400	0.1649	820	0.10740	1250	0.05370	*	*	*	*	*	*
491	194	800	5.1800	900.0	5.1800	1050	4.3262	300	2.9600	1400	2.44770	180	1.99232	1600	0.74000	*	*	*	*
492	107	510	0.5448	1720.0	0.5448	1950	0.5448	10	0.5029	200	0.34725	1400	0.33528	2320	0.22751	*	*	*	*
493	114	450	10.6901	310.0	10.3377	190	6.6960	1100	1.5272	1200	1.52716	1800	0.70484	*	*	*	*	*	*
494	203	200	6.3699	310.0	5.4599	500	4.6899	650	3.7800	1250	2.58997	1320	2.44997	*	*	*	*	*	*
495	118	100	0.2055	200.0	0.2055	350	0.2055	410	0.1310	250	0.08583	1000	0.04066	1380	0.02710	*	*	*	*
496	112	10	0.7827	100.0	0.6709	980	0.4042	1950	0.2666	500	0.28383	3750	0.13761	*	*	*	*	*	*
497	221	750	4.6224	200.0	3.5049	150	3.3525	250	2.7938	400	1.82864	900	1.42228	1870	0.71114	*	*	*	*
498	116	10	0.1599	120.0	0.1599	1000	0.0668	420	0.0615	3600	0.06327	3900	0.04921	2100	0.04218	*	*	*	*
499	102	200	0.1984	30.0	0.1643	750	0.1085	1450	0.0713	1000	0.05581	2200	0.04651	*	*	*	*	*	*
500	202	1020	8.1455	710.0	6.4448	400	5.7287	210	4.8336	550	3.84899	850	2.77485	1200	0.89511	*	*	*	*
501	103	100	0.4345	1250.0	0.2722	1400	0.2674	220	0.2006	500	0.14803	2850	0.04238	*	*	*	*	*	*
502	115	10	0.5520	2100.0	0.0667	4000	0.0667	4400	0.0546	950	0.06066	5250	0.04246	*	*	*	*	*	*

Row	phoneme	f1	s1	f2	s2	f3	s3	f4	s4	f5	s5	f6	s6	f7	s7	f8	s8	f9	s9
503	207	360	7.1741	200.0	4.2571	240	2.9958	700	0.9460	1250	0.70952	1030	0.63069	*	*	*	*	*	*
504	110	400	0.5592	200.0	0.4302	100	0.3441	1220	0.1106	1000	0.07989	1300	0.10447	*	*	*	*	*	*
505	109	300	3.8921	200.0	2.3096	400	1.6680	10	0.9409	1300	0.51324	1170	0.29939	*	*	*	*	*	*
506	221	220	5.9276	400.0	5.7322	800	5.4065	700	5.0808	1800	1.62846	950	1.49818	*	*	*	*	*	*
507	115	50	0.1782	150.0	0.1782	260	0.0588	650	0.0529	4500	0.03721	4200	0.03525	5300	0.02938	*	*	*	*
508	112	110	0.2179	870.0	0.2059	1010	0.1963	530	0.1843	300	0.12928	1300	0.08379	1450	0.05746	1750	0.04788	4000	0.028728
509	204	610	6.1486	90.0	5.4729	900	5.4729	800	5.0675	400	4.05403	200	3.37836	*	*	*	*	*	*
510	*	900	0.5327	600.0	0.5210	700	0.5093	120	0.4566	550	0.26929	390	0.25758	250	0.24587	2200	0.03512	*	*
511	104	1100	0.2301	990.0	0.1564	120	0.1623	710	0.1033	2250	0.06490	4000	0.05605	4100	0.05310	*	*	*	*
512	211	750	4.7049	1150.0	4.7049	190	2.5851	310	2.5851	900	3.20555	1090	3.41236	*	*	*	*	*	*
513	103	100	0.2930	190.0	0.2254	430	0.2028	280	0.1835	1600	0.13845	1200	0.11269	2600	0.03542	*	*	*	*
514	100	10	0.1879	200.0	0.1059	330	0.0688	570	0.0609	1120	0.03970	2050	0.04234	3850	0.03970	4500	0.03705	*	*
515	199	350	7.3218	510.0	4.5862	200	3.2184	600	1.0460	700	0.96551	2050	0.72413	*	*	*	*	*	*
516	212	2300	0.3729	2450.0	0.2581	10	0.2745	500	0.1147	1900	0.10654	5650	0.06146	5250	0.05327	*	*	*	*
517	98	30	2.1529	250.0	1.8217	500	0.7571	800	0.3312	2350	0.26024	1600	0.16561	*	*	*	*	*	*
518	193	120	3.9083	250.0	3.9083	700	3.9083	810	3.9083	1400	2.31922	400	1.41730	1600	1.84678	2650	0.34359	*	*
519	110	110	0.4703	200.0	0.4703	310	0.4703	400	0.3618	1310	0.05169	300	0.04910	*	*	*	*	*	*
520	114	350	4.3131	250.0	4.0287	120	2.2750	10	1.0427	900	0.56876	700	0.33177	*	*	*	*	*	*
521	197	220	8.0976	400.0	6.0509	550	5.7840	710	4.7162	320	1.15680	2200	0.97883	1820	0.88984	*	*	*	*
522	215	2280	0.2612	2100.0	0.1435	10	0.1866	100	0.1263	500	0.07750	2700	0.09472	3100	0.06028	5300	0.05454	*	*
523	214	2300	0.7300	2690.0	0.6578	2150	0.5535	420	0.4492	10	0.41715	1000	0.16847	3850	0.19253	5350	0.14440	*	*
524	200	320	13.8979	250.0	5.3453	200	4.4290	2300	1.9854	700	0.76362	2350	0.76362	*	*	*	*	*	*
525	102	20	0.1325	110.0	0.1150	400	0.0670	900	0.0408	1950	0.04223	2700	0.02912	1310	0.03640	3250	0.02134	*	*
526	112	1010	0.4497	90.0	0.3904	650	0.2026	210	0.2520	400	0.21741	2310	0.06918	4100	0.05929	3750	0.05435	*	*
527	194	1000	4.3028	1200.0	4.3028	800	4.1609	200	3.5935	360	3.12069	500	1.56035	1300	3.54624	*	*	*	*
528	115	50	0.1773	100.0	0.1520	380	0.0994	250	0.0682	500	0.05456	830	0.03118	4300	0.02533	5200	0.02533	*	*
529	119	110	1.6598	290.0	1.5138	10	0.8025	350	0.6931	500	0.32830	700	0.21887	750	0.20063	*	*	*	*
530	221	680	5.2838	190.0	4.4129	500	3.9484	310	3.7161	1200	2.03225	1000	1.85805	*	*	*	*	*	*
531	116	1000	0.1205	4010.0	0.1173	650	0.1173	10	0.1110	400	0.10782	2000	0.09830	2300	0.06501	4300	0.09672	*	*
532	107	500	0.3699	10.0	0.2398	1850	0.1829	1210	0.1545	200	0.19511	1550	0.10569	*	*	*	*	*	*
533	203	260	5.8536	480.0	5.4676	610	3.8595	1350	1.9941	1200	1.47948	2450	0.51460	*	*	*	*	*	*
534	118	100	0.3311	300.0	0.2547	400	0.2365	180	0.2256	10	0.16011	1010	0.04003	1200	0.03639	*	*	*	*
535	108	200	1.1074	100.0	0.9736	300	0.6815	400	0.3407	10	0.17037	720	0.09736	*	*	*	*	*	*
536	207	300	10.4331	400.0	9.5159	350	4.7006	180	4.3567	1200	1.14650	1000	0.30255	*	*	*	*	*	*
537	115	10	0.1473	250.0	0.0906	4280	0.0502	1380	0.0437	3900	0.04208	5500	0.02589	*	*	*	*	*	*
538	109	150	2.7147	250.0	2.7147	320	1.4021	10	0.5071	1100	0.41765	1000	0.29832	*	*	*	*	*	*
539	202	700	3.6772	1000.0	7.8029	900	5.7400	280	4.7535	150	3.35659	1150	2.95971	400	2.78033	*	*	*	*
540	216	10	0.1928	100.0	0.1928	200	0.1207	700	0.0932	850	0.08473	1200	0.03813	3100	0.02118	*	*	*	*

APPENDIX E: Tympanometry related assembly code programs

This contains two M68000 assembly code programs TYMP5.SA and TYMPAUD.SA.

TYMP5 is the program which controls the tympanometer hardware as is accumulates a multifrequency tympanogram. The two levels of interrupts used are initialised at the beginning of TYMP5. The main program simply sends the corresponding frequency and calibration codes to the hardware. This main part loops continuously which allows all 25 frequencies to be produced.

TIMEINT is the interrupt service routine for the level 2 timer interrupt which occurs every millisecond. This starts the pressure ADC, then reads and sends the information on pressure, phase, and AGC number(admittance) to the PC. PHASEINT is the service routine for the level 1 interrupt and measures the phase difference using the second PI/T.

TYMPAUD tests the use of the ASRA for the production of reflex stimuli. It searches for the character 'r' sent from the PC and redirects all subsequent characters to the ASRA until the character 's' is received.

```
TYMP5      IDNT    1,1
*
           DC.L    $C0800                Tympanometer Stack Address
           DC.L    $800                  Tympanometer Start Address
*
           ORG     $100
           DC.L    TIMEINT,0,PHASEINT Interrupt Service Routines Address
*
           ORG     $800                  Program Start Address
PIT        EQU     $88000                PI/T Address
PITPHA     EQU     $89000                PI/T for phase measurement
FRLATCH    EQU     $90008                Address of Freq select latch
ACIAPC     EQU     $84400                Address of PC Acia
MEM        EQU     $C0000                memory storage
DAC        EQU     $9000B                dac for multi freq calibration
*
*
           MOVE.B #3,ACIAPC              Reset
           MOVE.B #$01,ACIAPC            Sets up control word
*                                          disables interupt
*                                          7 data bits even parity
*                                          2 stop bits CLK/16=9600 baud
           MOVE.B #64,PIT+17             Vector Number
           MOVE.L #2500,PIT+18            Count for 1 ms
           MOVE.B #1,PIT+26              Clear Zero Detect Status Reg.
           MOVE.B #$A0,PIT+16            Set up Timer Control Register
*
           MOVE.B #65,PITPHA+5           Vector number
           MOVE.B #$20,PITPHA            port general control register
*                                           H34 interrupts enabled on going low
           MOVE.B #$18,PITPHA+1          Port service request register
           MOVE.B #0,PITPHA+3            input direction for port
           MOVE.B #0,PITPHA+7            port control register
           MOVE.L #$FFFFFF,PITPHA+18 counter preload register
           MOVE.B #$B2,PITPHA+16         control register
*
           MOVE    #$2000,SR             Set Interrupt Level
           BSET    #0,PIT+16             Start Timer (Enable INT)
           MOVE.B #$2,PITPHA+7           enable H3 interrupts
           MOVE.B #$B3,PITPHA+16
           CLR     D6
*
RESTART    MOVE    #0,D5
START      LEA     TABL,A0               Initially point A0 to Table
           LEA     TABL2,A2              Initially point A2 to cal table
           MOVE    (A0,D5),D0            Get freq code
           MOVE.L #0,MEM
           CMP     #$4,D0                Check for end of table
           BEQ.S   RESTART
           MOVE    D0,FRLATCH            and output it
           MOVE    (A2,D5),D2            get calibration number
           MOVE.B D2,DAC                 output it
           BRA.S   START
*
```

```
*
*          Subroutine ASCII
ASCII      CMP     #$0A,D1             Check if < or > 10
           BLT.S   TEN
           ADD     #$07,D1             change to ASCII
TEN        ADD     #$30,D1
           BSR.S   SEND
           RTS
*
*          Subroutine SEND
SEND       BTST    #1,ACIAPC           Ready to transmit?
           BEQ.S   SEND                  No---back to send
           MOVE.B D1,ACIAPC+1            Yes---- Transmit!!!
           BSR.S DELAY
           RTS
*
*          Subroutine Delay
DELAY      MOVE    #$200,D4
LLDEL      SUB     #1,D4
           BNE.S   LLDEL
           RTS
*
*          Frequency Code Table
*
TABL       DC.W    $C8,$E9,$109,$123
           DC.W    $13B,$151,$164
           DC.W    $175,$184,$191
           DC.W    $19E,$1A9,$1B2
           DC.W    $1BA,$1C2,$1C9
           DC.W    $1CF,$1D4,$1D9
           DC.W    $1DD,$1E1,$1E4
           DC.W    $1E7,$1EA,$1EC
           DC.W    $4                  Termination Code
*
*
*          Calibration code table
*
TABL2      DC.W    $76,$76,$76,$76
           DC.W    $76,$76,$76
           DC.W    $76,$76,$76
           DC.W    $76,$76,$76
           DC.W    $76,$76,$76
           DC.W    $76,$76,$76
           DC.W    $76,$7A,$7A
           DC.W    $7A,$7A,$7A
*
*
*          ORG     $7C00               Interrupt Service Routine
AGCNO      EQU     $90012              Address of AGC number latch
PRESSNO    EQU     $90010              Address of pressure reading from
*                                           pressure transducer
*
```

```
*
*
TIMEINT   AND      #$01FF,D0         reproduce pressure number
          ADD      #$8000,D0         bit set for adc convert pulse
          MOVE     D0,FRLATCH        send pulse
          BCLR     #15,D0            clear pulse again
          MOVE     D0,FRLATCH
*
          LEA      AGCNO,A1          Point to AGC number with A1
          CLR      D1
          LEA      PRESSNO,A2        Points A2 to pres. trans. no.
          MOVE     #$61,D1           Send a to start agc no.
          BSR      SEND
          MOVE     (A1),D1           Get word AGC no. into D1
          MOVE     D1,D2
          AND.W    #$00F0,D1         Get LSB`s
          LSR      #4,D1
          BSR      ASCII             Change to ASCII and send
          MOVE     D2,D1
          AND.W    #$000F,D1         GetMSB`s
          BSR      ASCII
          MOVE     #$70,D1           Send p for start of pressure data
          BSR      SEND
READPRES  MOVE     (A2),D1           Get pressure reading
          MOVE     D1,D2
          MOVE     D2,D3
          MOVE     D3,D4
          AND      #$4000,D4         Check for Data ready
          CMP      #$4000,D4
          BEQ      READPRES          read again if not ready
          AND      #$0300,D1         Get MSB`s  upper 2 bits
          LSR      #8,D1
          BSR      ASCII
          MOVE     D2,D1
          AND      #$00F0,D1         Get next sig bits
          LSR      #4,D1
          BSR      ASCII
          MOVE     D3,D1
          AND      #$000F,D1         Get LSB`s
          BSR      ASCII
          MOVE     #$62,D1           send out b to terminate pressure data
          BSR      SEND
          MOVE.L MEM,D1              read stored phase
          AND      #$F0,D1           get upper phase no.
          LSR      #4,D1
          BSR      ASCII             convert and send
          MOVE.L MEM,D1
          AND      #$0F,D1           get lower phase number
          BSR      ASCII             convert and send
          MOVE.L #0,MEM              zero counter in case phase=0
          MOVE     #$63,D1           send c to indicate end of data
          BSR      SEND
```

```
REC       BTST    #0,ACIAPC          check for received character
          BEQ.S   FINISH
          MOVE.B  ACIAPC+1,D2        receive character
          CMP.B   #$66,D2            check to see if f is received
          BNE     FINISH
          ADD     #2,D5              increment counter for freq table
FINISH    BSET    #0,PIT+26          Clear Status Bit
          RTE                        Return
*
*         ORG     $7F00
PHASEINT  MOVE.L  PITPHA+22,D6       read counter number
          MOVE.L  #$FFFFFF,D7
          SUB.L   D6,D7              subtact to get time
          MOVE.L  D7,MEM
          MOVE.B  PITPHA+9,D6        read to clear interrupt
          RTE
*
          END
```

```
TYMPAUD   IDNT    1,1
*
          DC.L    $C8000              Stack Address
          DC.L    $800                Start Address
*
          ORG     $800
ACIAPC    EQU     $84400              Address of PC acia
ACIAUD    EQU     $84800              Address of audiometer acia
*
          MOVE.B #3,ACIAPC            reset
          MOVE.B #$01,ACIAPC          9600,7,e,2
          MOVE.B #3,ACIAUD            reset
          MOVE.B #$01,ACIAUD          9600,7,e,2
*
          CLR     D1
BEGIN     BSR.S   REC
          CMP     #$72,D1             Check is it r?
          BNE.S   BEGIN                 No---look again
AUDATA    BSR.S   REC                   Yes--- Get another character
          CMP     #$73,D1             Check is it s?
          BEQ.S   BEGIN                 YES--look again for r
          BSR.S   SEND                  NO---send
          BRA.S   AUDATA              look for next data
*
*
*         Subroutine to receive data
REC       BTST    #0,ACIAPC
          BEQ.S   REC                 wait till ready to receive
          MOVE.B ACIAPC+1,D1          receive
          RTS
*
*         Subroutine to send
SEND      BTST    #1,ACIAUD
          BEQ.S   SEND                wait till ready to send
          MOVE.B D1,ACIAUD+1          send
          RTS
*
          END
```

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