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AN INVESTIGATION INTO THE APPLICATIONS OF REAL-TIME COMPUTING TECHNIQUES IN INDUSTRIAL AUDIOMETRY

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A thesis submitted to the University of Warwick
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SUMMARY

The past decade has witnessed an unprecedented public concern about the effects of high noise levels. The concern has mostly resulted from the increased noise levels of aircraft and the annoyance which that noise can cause to an individual exposed to it regularly. An effect of high level noise which has also resulted in enormous concern, but mostly out of the public-eye, has been that of noise induced hearing loss. Throughout industry in the U.K. alone there is an estimated 2 million people subjected to noise levels sufficiently high to endanger their hearing if regular exposure continues over a period of years. In order that these people may be protected from eventual partial or complete deafness their hearing acuity must be measured at regular intervals. The general title of the techniques used for making this measurement is audiometry. In industry there is a growing requirement for accurate and reliable but fast and simple-to-use audiometric equipment to cope with the large numbers of workers requiring regular examination. As a result of recent advances in the design and performance of digital computers intended for control applications the decision was made to investigate their possible use in routine audiometry and in particular in an industrial audiometric unit.

Initially in this thesis, an extensive review of the existing audiometric techniques as used in hearing conservation programmes is given. In addition, deficiencies in the methods presently used are highlighted and discussed to reveal possible suitable areas for the application of computer techniques. As a result of this work a new concept of a screening audiometer is evolved in the form of

an adaptive screening instrument capable of adjusting its measuring technique to produce optimum results from each subject and of performing much of the record keeping and result-scanning presently done by hand.

To substantiate this theoretical work the proposed audiometer system was built on a computer situated in the University. A series of examinations were performed using the system and the results compared with others obtained from the same people by a conventional method. The two sets of results agreed to within acceptable limits and the degree of personal attention required to administer the test was greatly reduced.

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LIST OF ABBREVIATIONS

DIS	decreasing intensity series
ER(n)	error number
HL_f	hearing loss at frequency f
H_{10}	hearing loss after 10 years of noise exposure
IIS	increasing intensity series
L_{A2}	sound pressure level in dB(A) exceeded for 2% of the time
LC	test level counter in audiometer programme
MAF	minimum audible field
NF	number of files(programme variable)
NIPTS	noise induced permanent threshold shift
NR	number of records(programme variable)
RAD	rapid access data unit(Sigma-5)
SPL	sound pressure level
SS(n)	silent period response sample(n=1-10)
T	noise exposure duration reckoned on the basis of 8 hours per day,5 days per week.The unit is 1 callendar month or year.
T_o	A reference duration, equal to the unit in which T is reckoned(month or year).
TS(n)	tonal period response sample(n=1-10)
T_i	time at which test tone is initiated
T_s	time at which test tone settles to steady state value.
T_c	time at which test tone ceases
T_p	time at which subject responds
TTS	temporary threshold shift
$TTS_{2\ min}$	TTS as measured 2 minutes after an 8 hour noise exposure

U_p	A parametr in the equation for H,related to the values of the centile of noise exposed population,reckoned from the most susceptible(0%) to the most resistent (100%),through the proporties of the Gaussian distribution.
VDU	visual display unit(Sigma-5 system)
WN	works number or patient reference number
x	a pseudo-random number(audiometer programme)
λ_i	a frequency dependent parameter related to the A-weighted noise immission level.(i denotes the audiometric frequency)
τ_{RA}	actual inter-tonal period
τ_{RD}	defined inter-tonal period

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Finally, he wishes to thank Miss E. Flockhart for the typing of this thesis, and his wife for her encouragement and patience.

DECLARATION

Use has been made of publications relevant to this work and references to such cases have appropriately been made.

Use has also been made of information contained in a thesis submitted by the author in partial fulfillment of the requirements for the degree of Master of Science to the University of Warwick, Inter-University Institute of Engineering Control, September, 1971.

All other work and conclusions are due to research performed by the author during the period from September, 1971 to September, 1973.

........................Candidate

CHAPTER 1

1.1 INTRODUCTION

Audiometry is the technique of detecting the threshold of hearing and measuring the extent of the deviation from 'normal'. As far as human beings are concerned, audiometry is used principally as a diagnostic tool for physiological disorders but, in recent years, has come to be used increasingly for the detection of noise-induced hearing loss. The work described in this thesis is concerned with the application of computer control techniques to all aspects of audiometry. In particular, the thesis is directed towards the specific problems of industrial audiometry.

For many years, loss of hearing, in various degrees, was taken for granted as being a part of growing old. It was widely recognised that extremes of noise, such as that due to artillery, could cause a sudden loss or rapid degradation in hearing ability. The fact that employees in industries such as drop-forging, weaving or boiler-making, suffered on average, a more severe loss and at an earlier age than employees in other industries was accepted as an unfortunate occupational hazard. The extremely high noise levels inside a boiler during manufacture made this perhaps the most obvious industry liable to cause a loss of hearing and led to the term 'boilermaker's deafness' being used to describe noise-induced hearing loss for many years. In 1947 a large number of industrial compensation claims for alleged occupational noise-induced hearing loss were filed in the USA, notably in the states of New York, New Jersey and Wisconsin. At that time the USA legally recognised sudden loss of hearing due to industrial accident as a compensable disability. However, the gradual loss due to many years of noise exposure presented a new field of legal battle. Not only was it necessary to prove that the hearing loss was due to noise exposure but also, whether that exposure had been experienced during the present or a previous employment. The most likely answer was that the loss had accumulated slowly throughout the working life with several employments possibly including service in the armed forces. It was as a result of the inability of compensation claims, such as these, to reach a just settlement that industry began to take an active interest in what is today called 'hearing-conservation'. With the post-war industrial expansion and resulting general increase in noise levels, an accurate means of assessing the hearing ability of employees in noisy areas became essential if the growing number of compensation claims were to be judged in a fair manner. There were economic advantages to both the employee and employer from reliable assessment methods. The employee could actually prove his loss of hearing in a quantitive manner and the employer could protect himself from falsified claims which may otherwise have been difficult to disprove.

The advent of quantitive methods of threshold assessment have allowed employers to keep records of individual employee's hearing acuity from their commencement of employment. The employer who wishes to protect both his employees from loss of hearing and himself from monetary claims can therefore keep accurate acuity records and have a system, agreed with his employees, to enable him to take remedial action where necessary. Such action can take a number of forms. An individual displaying symptoms of excessive noise exposure may be removed to a less noisy environment or he may be supplied with hearing-protective devices to reduce his exposure to the noise. Reduction of the noise at its source can also be considered. Frequently, machinery may be redesigned or de-rated to reduce its noise output. The physical location of a machine may increase its effective noise level. Relocation or the use of suitably designed screens or hoods can often reduce this level. Systems intended to achieve such action are generally referred to as 'Hearing Conservation Programmes'.

In the U.K. a Government recommendation contained within a Code of Practice (1), recommends that no unprotected ear shall be exposed to a noise level of greater than 90 dBA for a period of eight hours a day. The implication is that if these conditions are exceeded, with respect to either duration or level, then there exists a risk of hearing loss. According to figures published recently, R.P. Grime (2) there are a half a million workers actually exposed to this risk and over two million in potentially damaging situations. The distribution of these people at risk within the country's labour force and within any single factory, is not uniform. Most factories have one or two personnel in noisy areas, perhaps a boilerhouse, but some have several hundred. Regular examination of large numbers of employees, once or twice a year, and screening of the results is a major undertaking, placing great demands on the time of the medical officer and his staff. For this reason, accurate audiometers and rapid procedures are essential to the operation of any hearing conservation programme.

The 'ideal' industrial audiometer should be capable of accurately assessing the hearing threshold, recording the results and analysing them in any required manner. Implicit in the accurate assessment of the threshold is the ability to detect and counteract errors or falsifications in the test responses of the subjects. Current audiometers fall short of this ideal because they rely upon the skills of their operator to produce reliably accurate results. Many of the procedures followed, and decisions made by the operator could be performed by a suitably programmed computer, releasing the skilled audiometrician for less tedious and repetitive tasks.

It is the aim of this thesis to investigate the possible uses and limitations of computers in industrial audiometry.

1.2 The Ear and the Hearing Mechanism

The acousto-mechanical operation of the peripheral ear has been thoroughly explained by the experiments and work of von BEKESEY (3). However, understanding of the operation of the inner ear and its conversion of mechanical motion into neural activity is still relatively incomplete. Still less is known about the neural transmission of information and the final mechanism of perception. A schematic diagram of the human ear is given in Fig. 1.1. It is conventionally divided into three regions, namely the outer, the middle and the inner ears.

1.2.1 The Outer Ear

The outer ear is responsible for matching the impedance of the ear drum to that of the external air. It consists of the external convoluted appendage on the side of the ear called the 'pinna', the external ear canal or 'auditory meatus' and the ear drum or 'tympanic membrane'. The pinna serves little purpose other than protection for the canal, its directional reception properties evident in many mammals being almost non-existent in man. The auditory meatus has the form of a reasonably uniform pipe, open at one end and sealed at the other by the tympanic membrane. It has been shown by Wiener and Ross (4) that a sound pressure increase of between 5 and 10 dB occurs between the pinna and membrane. The membrane is a relatively stiff, inwardly-directed cone.

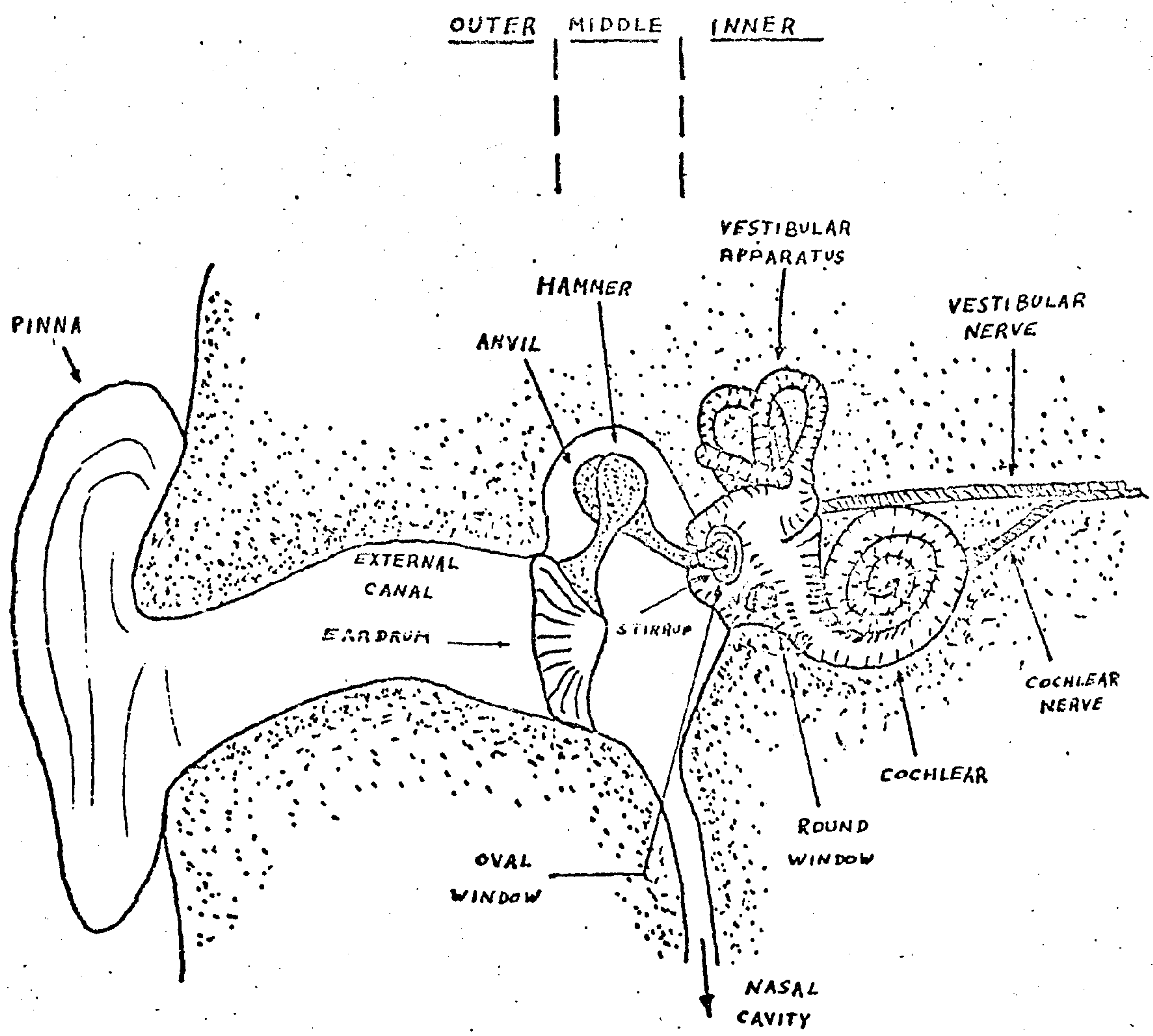


Fig.1.1. Schematic diagram of the human ear. This is not to scale, the middle and inner ears are enlarged.

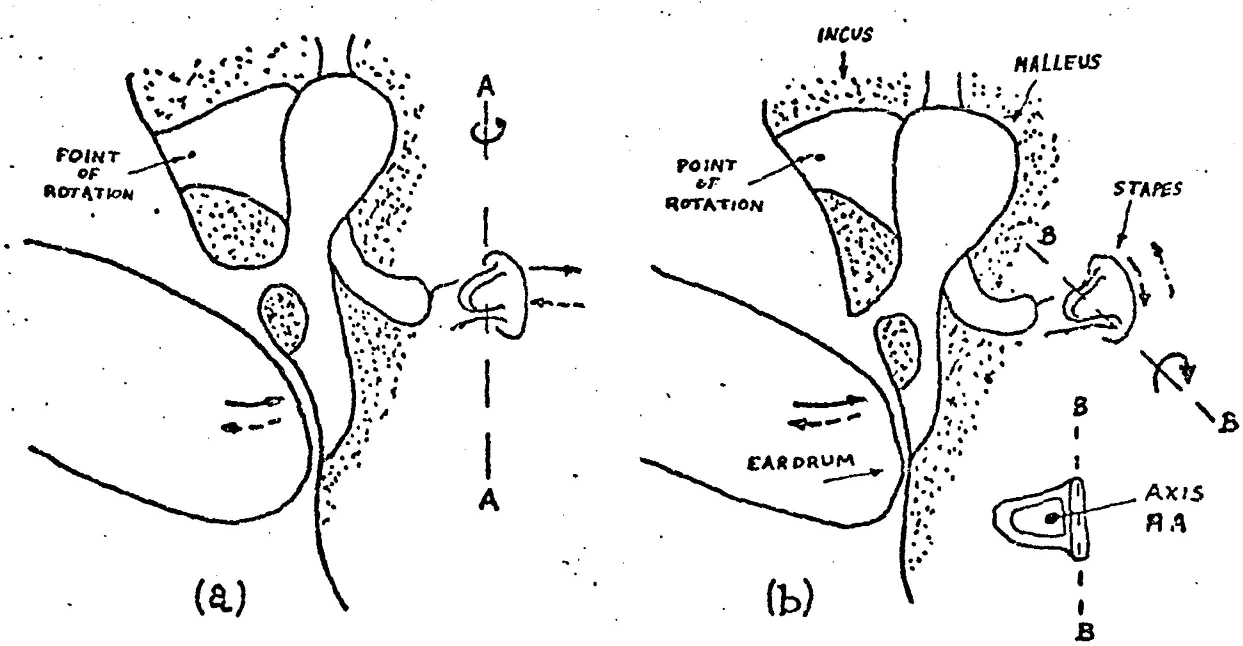


Fig.1.2. Vibration modes of the ossicles (a) below the threshold of feeling (b) above the threshold of feeling (after Békésy, 1960)

1.2.2 The Middle Ear

The middle ear is the air-filled cavity immediately interior to the tympanic membrane. Within the cavity are the three ear bones or ossicles: the malleus, the incus and the stapes; and two small muscles: the stapedius, which is attached to the stapes and the tensor tympani, which inserts at the handle of the malleus. The cavity presents five openings: the one covered by the tympanic membrane, the auditory tube leading to the nasal cavity, the opening into the mastoid cavity, and the oval and round windows. The flattened, bony footplate of the stapes fits the oval window and acts as a hinge allowing mechanical motion to be transmitted to the inner ear. The round window is very small and covered by a thin membrane.

The middle ear has three functions:

a) It transmits energy from the airborne sound in the air column of the external auditory canal, by means of the tympanic membrane and the ossicular chain, into the fluid contained within the cochlea. The air vibrations alone, without the effective amplification of the middle ear mechanism, cannot exert sufficient force on the oval window to activate the cochlea.

b) It protects the inner ear from the shock of intense sounds of low frequencies by the reflexive action of the middle ear muscles. These muscles tense the tympanic membrane and ossicular chain, thus reducing the amplitude of large vibrations. This protective role of the middle ear is still the subject of active research. Recent studies have revealed that we do not know exactly how it operates.

c) The opening to the nasal cavity acts as a pressure equalisation tube between the two surfaces of the tympanic membrane.

1.2.3 The Inner Ear

As illustrated in Fig. 1.1 the inner ear is composed of the cochlear and the auditory nerve terminations. Attached to the cochlear, but apparently not a part of the hearing mechanism, is the vestibular apparatus its purpose being to sense spatial orientation.

The cochlear is the ear's mechanical to neural transducer. If it could be uncoiled from its snail-like form it would have the appearance shown in Fig. 1.4. The action of the cochlea is such that the stapes footplate acts as a piston on the oval window and displaces the perilymph fluid in the scala vestibuli. At low frequencies the perilymph and the cochlea may be considered to be incompressible and rigid respectively, thus the pressure must be relieved by some means. This need is satisfied by the round-window at the middle ear end of the scala tympani. The round window is a membrane-covered opening, compliant to the pressure fluctuations in the perilymph. At frequencies below 20 to 30 Hz any displacement of the stapes results in a flow of fluid around the helicotrema to the round window with no other effect upon the inner ear at all. However, at higher frequencies the pressure fluctuations are transmitted from the scala vestibuli to the scala tympani by the yielding of the cochlear partition. The position along the partition at which the yielding takes place is a function of the frequency of the stimulus. This is shown in Fig. 1.4. The cochlear cross section is shown in Fig. 1.3 and can be seen to have two membranes between the two fluid channels. The Reissner's membrane is sufficiently flexible to allow it to be considered, from a mechanical point of view to be non-existent and the scala media and scala vestibuli to be one channel. The fluid contained within the scala media, effectively adds to the vibrating mass of the basilar membrane. The basilar membrane supports the organ of Corti which contains some 30,000 sensory cells, on which the endings of the auditory nerve terminate. Owing to its tapering dimension along its length the basilar membrane is stiffer and less massive at the narrow, basal end and more compliant and massive at the broad, apical end. How the organ of Corti acts as a transducer to neural activity is a matter best left to the various research texts upon the subject (5)(6). In outline, the flexing of the basilar membrane causes stressing of embedded hair cells which give rise to electrical potentials in the scala media. These potentials cause the nerves to be excited and so pass information along the VIIIth nerve to the brain.

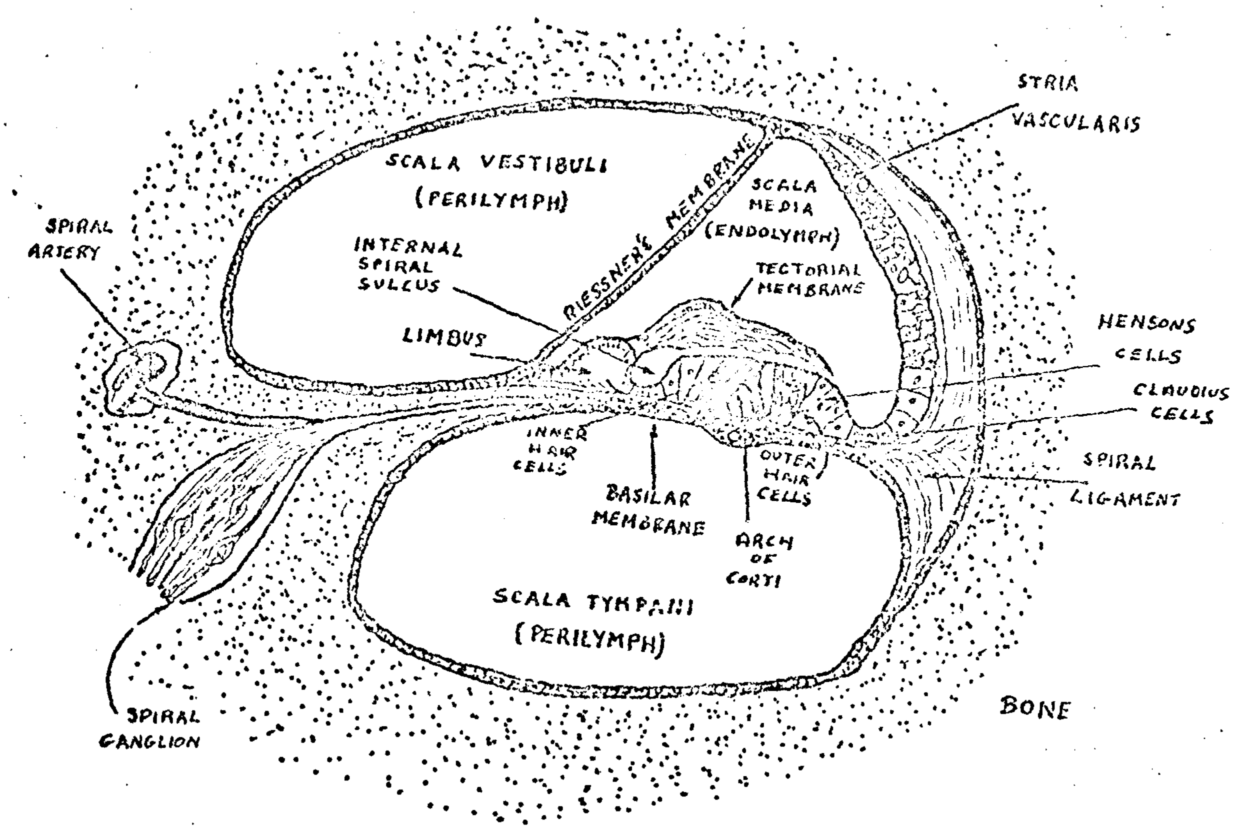


Fig.1.3 Cross sectional diagram of the cochlearshowing the organ of Corti

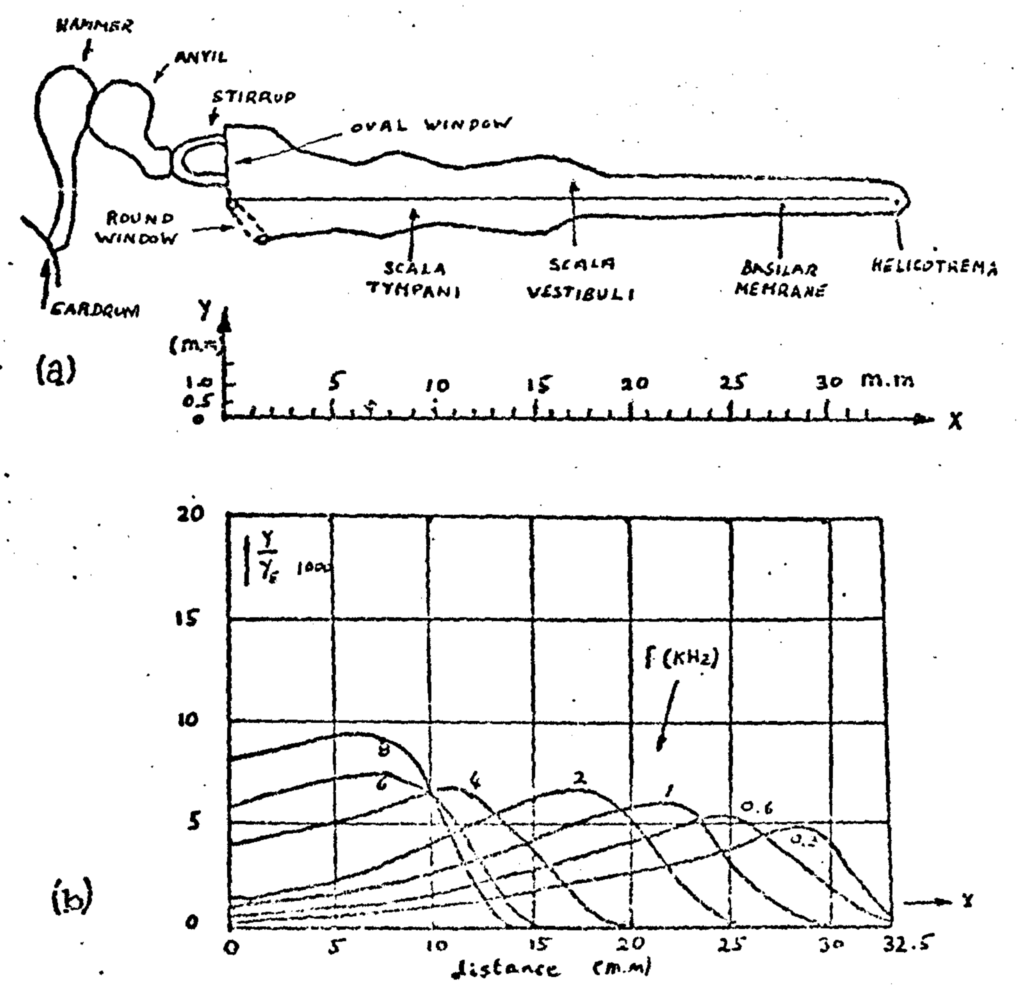


Fig.1.4 a) A sectional view of the straightened cochlear

b) Graph showing frequency sensitivity with distance.

1.2.4 The Loudness Sensitivity of the Ear

The loudness sensitivity of the ear at different frequencies is usually shown in the form of the 'Normal Equal Loudness Contours'. Standard contours have been defined as a result of widespread research and examination over many years. BS 3383 defines these contours for binaural, free-field (not earphones) listening conditions and for 'otologically normal listeners, aged 18 to 25 years'. An otologically normal person is someone in a normal state of health who is free from all signs or symptoms of ear disease and from wax in the ear canal, and from previous history of ear disease or blows to the head. The curves may be seen in Fig. 1.5. The lowest curve shown is the minimum audible field or 'normal threshold of hearing'. Modified curves exist to correct the contours for use with given models of earphone; details may be found in BS 2497. The normal threshold of hearing is used as a reference against which to measure hearing defects. Audiologists draw the threshold curve as a straight line when plotting graphs of hearing characteristics and plot points relative to it in dB HL (Hearing Level in dB relative to the normal threshold). From the normal contours it may be seen that the ear is at its most sensitive at and around 4 kHz, a fact which will be emphasised later in the chapter.

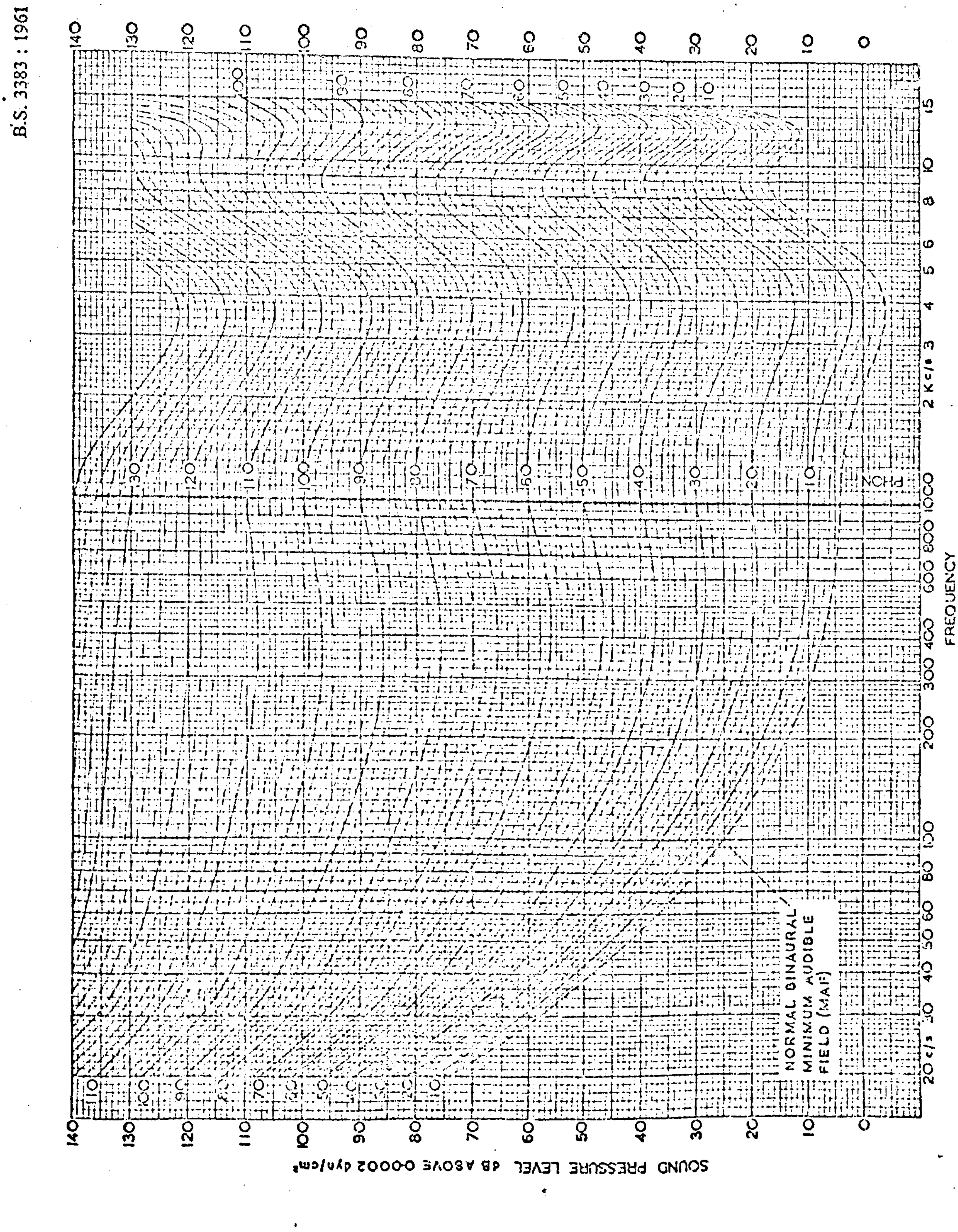


Fig. 1.5 Normal equal loudness contours(free-field).

1.3 Noise Induced Hearing Impairment

The delicate mechanism of the ear is susceptible to permanent damage if exposed to prolonged high level noise. This is an effect of noise which has been investigated in great detail throughout the world. In 1970, the British Government, Department of Health and Social Security published their report "Hearing and Noise in Industry". As a result, proven hearing impairment due to particular employment is now a compensatable ~~comparsable~~ disability under the Industrial Injuries Act. Permanent hearing damage of this type is known as Noise Induced Permanent Threshold Shift (NIPTS). The term 'permanent' threshold shift is used to differentiate it from a related condition known as Temporary Threshold Shift (TTS), which needs to be discussed before further considering NIPTS. only in [illegible]

1.3.1 Temporary Threshold Shift

TTS is a condition familiar to many people as an apparent 'muffling' of the hearing after exposure to high level noises. It is known that after prolonged exposure to high-level noise, say well over 80 dBA, the hair cells on the basilar membrane become 'tired', and no longer signal the degree of motion to the brain. A detailed explanation of why this occurs is not possible as the condition is the subject of a great deal of research, (40,41). After a period of time in less noisy surroundings the apparent loss of hearing disappears, hence the name temporary threshold shift. The length of time taken for the hearing threshold to return to its original condition may be a matter of hours or days depending upon the exposure received and on unknown attributes of the individual. Providing that the shift in the threshold has returned to normal before further exposure occurs its effects are of no concern. However, if an individual requires twenty-four hours to recover from the effects of an eight hour working day, and is exposed for five days in every seven, then a cumulative effect takes place which may not disappear during the two days of 'quiet', Fig. 1.6. From this description it could be inferred that a prolonged period of non-exposure would allow the shift to return to normal. The extent of TTS varies with frequency, in particular high frequency. It may be noted from Fig. 1.6 that the degree of TTS at 4 kHz is greater than at the surrounding frequencies. This is a peculiarity of both temporary and permanent noise induced hearing loss. The maximum loss always occurs at 4 kHz whatever the frequency characteristics of the noise source that cause it. The 'dip' in the hearing threshold at this frequency is often referred to as the C^5 or 4 kHz dip. Attempts have been made to explain the 4 kHz dip, Lehnhardt (35), Ward (43). Research with small mammals (7) has shown that damage occurs to the cells of the organ of Corti, Fig. 1.3, at and near where the maximum displacements would have been expected to take place, corresponding to the frequencies of the applied noise. The extent of the damage is increased by exposing the ear to noise whilst it is still in its 'tired' state.

Considerable research has been undertaken to determine

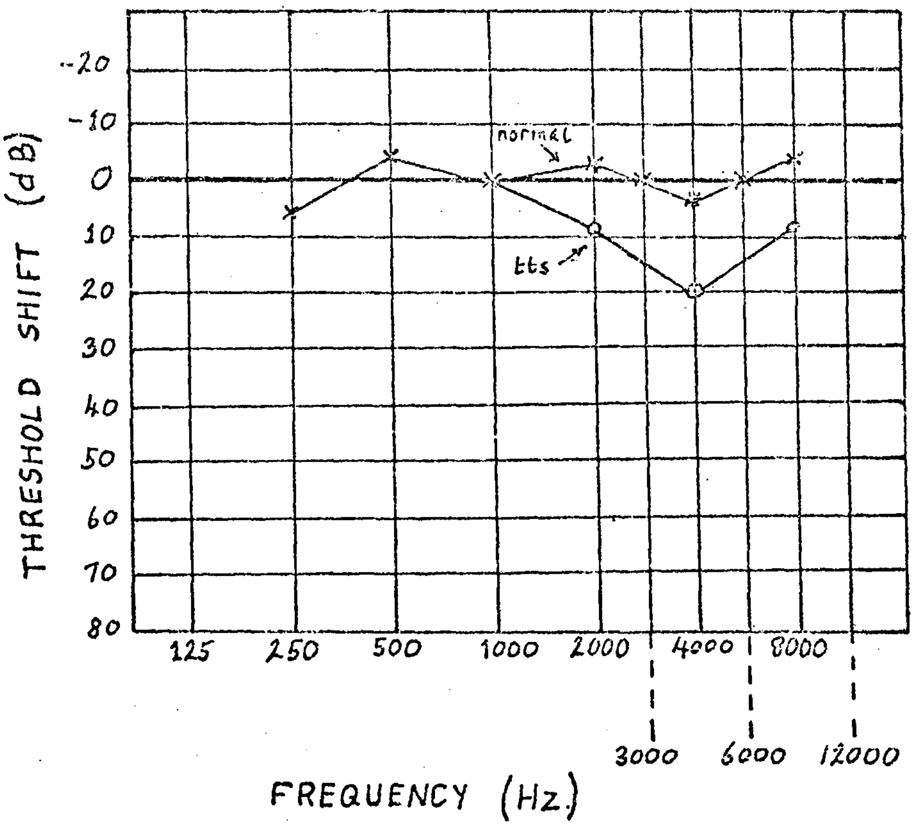


Fig. 1.6 Example of temporary threshold shift (TTS)

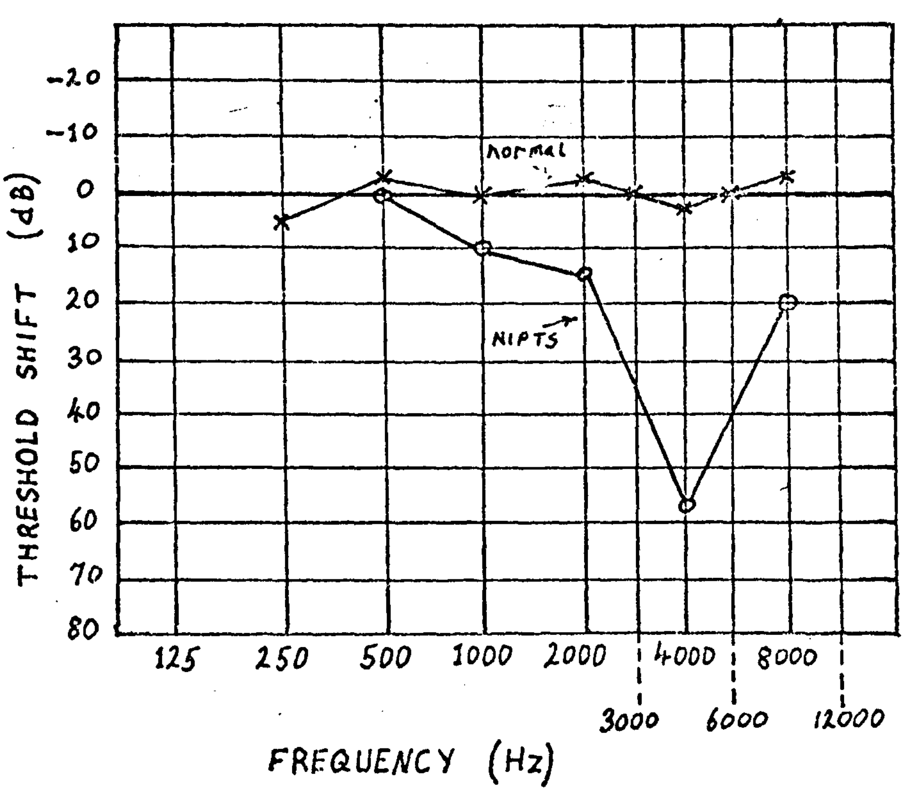


Fig. 1.7 Example of permanent hearing loss (NIPTS)

a reliable relationship between TTS and NIPTS. Such a relationship could be used as a susceptibility test of an individual to long term hearing loss.

It has been suggested (8) that the TTS at 1 kHz, recorded two minutes after completing an eight hour period of exposure, say the working day, is approximately equal to the permanent loss to be expected from working in the same environment for ten years, that is:

$$TTS_{2\text{ minutes}} \quad = \quad H_{10\text{ years}}$$

Unfortunately, if a working day really were to be used the results would probably be erratic as a result of the equipment generating the noise being switched off at varying time intervals from the official end of the working day. Thus, a measurement of TTS made two minutes after the working day may well be made up to half an hour after the last high-level noise exposure.

This test is one of many to be found in the literature. Ward (8) tabulates 19 publications by various authors, describing tests involving TTS. The assumption throughout all the tests described is that TTS may be used to indicate the degree of NIPTS an individual may expect to suffer. The conclusion reached by Ward after study of the tests is that it may be possible to determine the susceptibility to NIPTS from a particular noise from knowledge of the susceptibility to TTS from that same noise.

A remarkable observation made by Tota and Bossi (9) is that a correlation exists between the colour of a person's eyes and his susceptibility to TTS and hence possibly NIPTS. This correlation has been confirmed by Hood, Poole and Freedman (10). The common factor between the eyes and the ears is a substance named melanin. Melanin is the substance responsible for iris pigmentation but is also present in the cochlear. The amount present appears to affect the ear's adaption to high level noise.

Fig. 1.7 shows an example of a well established case of NIPTS accumulated over a number of years. The practical effects of the loss are often not noticed until the condition is extremely advanced, when speech intelligibility begins to be lost. This occurs in a progressive manner as the acuity deteriorates in the speech frequencies,(500 Hz to 4 kHz). Firstly, the higher frequency consonants such at 't' or 's' begin to disappear, then, as the lower frequencies start to lose their acuity, speech eventually becomes a mumble. An unfortunate characteristic of NIPTS is that it occurs initially and principally at the same frequencies as the hearing loss associated with age, presbycusis. The two losses are thought to add directly to each other. Fig. 1.8, derived from equations given in BS 3383, shows values of hearing loss due to presbycusis that might be expected in people of the ages indicated. The similarity between the symptoms of presbycusis and NIPTS has been largely responsible for working people taking early hearing loss for granted for so long. Only with the observation that people in certain occupations tended to suffer greater losses than others did

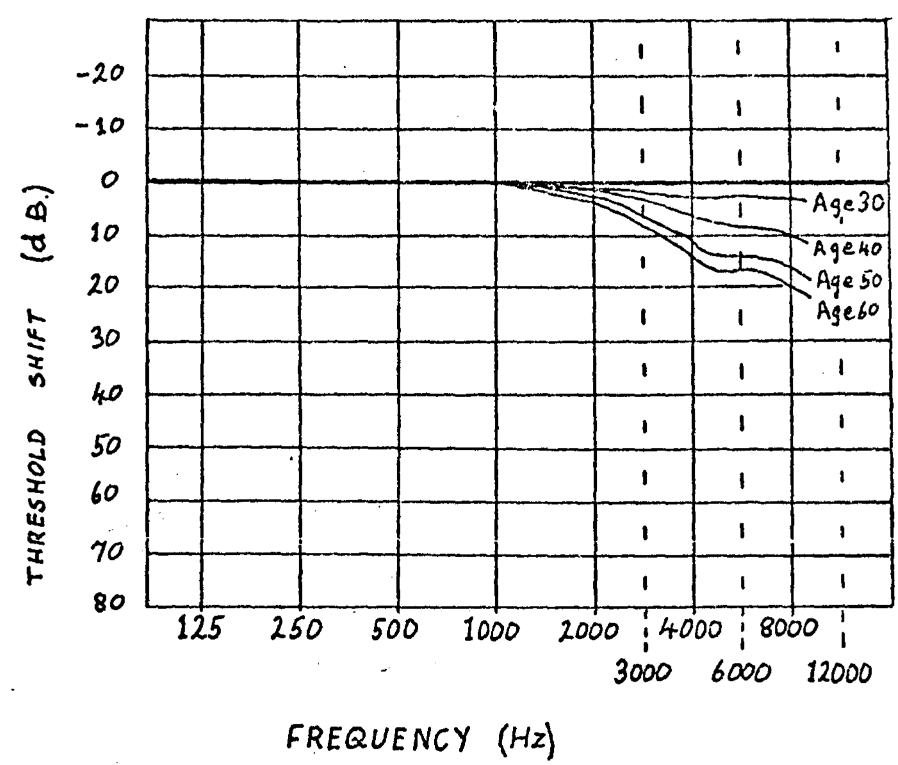


Fig. 1.8 Approximate values of hearing loss due to presbycusis

it become apparent that the loss was at least partially due to the noisy environment.

1.3.2 Noise Induced Permanent Threshold Shift

In the publication 'Hearing and Noise in Industry' (11), is stated an expression for the hearing loss to be expected from a given exposure over a given period. The expression was derived as a result of measurements and observations made over a considerable period of time upon a large number of people exposed to industrial noise. The hearing loss, H, to be expected in a given percentage of a normal population subject to habitual daily exposure to a noise level L_{A2}, over a period of time T, is:

$$H = 27.5 \left[1 + \tanh(L_{A2} + 10 \log(T/To) + Up - \lambda_i)/15 \right] + Up$$

where λ_i is a constant, dependent on audiometric frequency. To is reference duration and Up is a constant depending upon the selected percentage p. This equation is valid for the entire audio-frequency range and takes into account the C^5 (or 4 kHz) dip. Values of H calculated in this way need to be corrected for the sex of the individual concerned. For a male 1.5 dB should be added and for a female the same amount should be subtracted. This is because observations (36) of the degree of TTS and NIPTS suffered by men and women in the same environments has shown that women, on average, suffer less loss than men. An additional factor, not taken into account in the above equation is that of body vibration. Yamamoto and Fujii (12) reported in 1966 that noise and vibration jointly may cause a higher eventual hearing loss than noise alone. The additional loss due to the vibration with the noise being greater than the loss due to the vibration alone.

Currently, NIPTS is an irreversible condition. In the absence of a proven reliable prognostic test to determine the susceptibility of individuals to NIPTS one is led to the possibility that the most satisfactory method of reducing the risk of it occurring is to prevent the noise from entering the ear. This can be achieved in two principal ways; reduce the noise at its source or protect the ears themselves. It is not the intention of this thesis to discuss in detail noise reduction or prevention techniques.

1.4 Audiometry Outlined

Audiometry, as it originated, was the determination of the intensity levels of a number of pure tones (i.e. sounds of one frequency) that could just be heard by an attentive listener in an otherwise 'silent' environment. These levels form the auditory threshold, as described in section 1.2.4, which varies greatly between individuals. An accurate determination of the threshold is of great interest to the audiologist and it is a major tool in the diagnosis of hearing disorders, and in monitoring an individual's hearing condition. In the days before electronic equipment was readily available pure tones were generated by various means, all centred around a tuning fork interrupting magnetic circuits and causing sympathetic sounds to be generated in telephone earpieces. This type of equipment tended to be clumsy and difficult to use. The advent of the electronic valve and subsequent oscillator with variable tone output at the turn of a knob greatly simplified the generation problem. Also, the accurate measurement and ease of presentation of varying intensity signals allowed far superior estimation of the actual threshold level to be made. The path was then wide open to greater accuracy and deeper investigation into the why's and wherefore's of human hearing and it's defects. Indeed, there are now many specialised test procedures capable of pinpointing hearing defects to specific regions of the ear (13).

The use of audiometry as a diagnostic tool is restricted to anatomical localisation of the fault and is not adequate for complete diagnosis. The diagnosis can only be made after a study of the case history and after physical examination. Current diagnostic test procedures are far too numerous to mention and are best left to the specialist texts for description (13). Of the everyday test procedures three will be outlined here and references made to variations upon them.

1.4.1 Pure Tone Audiometry

Pure tones are uniquely suited for making threshold measurements for the following reasons:

a) A pure tone is the simplest type of auditory stimulus. It is relatively easy to generate and present to the ear a tone of accurately known frequency and intensity. The patient has only to indicate a yes or no response, he is not required to describe or repeat what he hears.

b) Some patients may respond normally to speech but have high frequency defects possibly due to occupational noise that pass unnoticed. The majority of essential speech sounds are within the range 300 to 3000 Hz. Specific frequency testing can determine such cases before possible intensive damage is caused at the higher frequencies and before it spreads to the lower range with time.

c) Threshold sensitivity may be precisely determined using pure tones. This is of great value in assessing the relative merits of certain types of middle ear surgery.

A disadvantage of pure tone tests is their inadequacy to predict speech perception as compared with direct speech testing, a fact which will be discussed in section 2.1.1.

The test should be performed with the patient in a quiet surrounding. Six or eight tones are normally used, these being comprised of a selection from 125, 250, 500, 1000, 2000, 3000, 4000, 6000, 8000 Hz. A typical selection might be the octave-spaced values 250, 500, 1000, 2000, 4000 and 8000 Hz. One ear is tested at a time. The subject is requested to respond in the manner prescribed by the examiner during the entire period for which he can hear a signal in his earphones, thus allowing the examiner to tell tone responses from those due to spurious clicks and other noises. The examiner usually sits in such a position that he can observe the subject's behaviour during test. Before placing the earphones on the subject's head the tone should be 'OFF'. The operator may switch the tone applied to the earphones on or off by means of a key referred to as the interrupt key or interrupter. The following description is of the 'Method of Limits' technique widely used in manual examinations. Once the earphones are in place the frequency is set to 1 kHz and the level to 40 dB HL. The interrupt key is set to 'ON' for one or two seconds. If the response from the subject is positive the level is reduced by 10 dB with tone OFF and then presented again. This procedure is repeated until a negative response is achieved. Upon a negative response the level is raised by 5 dB. A positive response at this raised level is followed by a reduction by 5 dB. The threshold is taken as the lowest intensity at which the subject responds two or more times out of four. A similar technique is used if a negative reply is received at the initial 40 dB, except that the 10 dB steps would have increased the level until the tone was audible.

The duration of the tones and the 'silent' intervals between them are normally varied to avoid rhythmical responses. Tones may be presented for ½ to 2 seconds, inter-tone periods may be ½ to 5 seconds. Once the threshold at 1 kHz has been determined the test is repeated at the higher frequencies in ascending order. The 1 kHz is then repeated for comparison and the low frequencies tested in descending order. There are two other techniques of threshold determination as well as the Increasing Intensity Series just described.

a) Decreasing intensity series

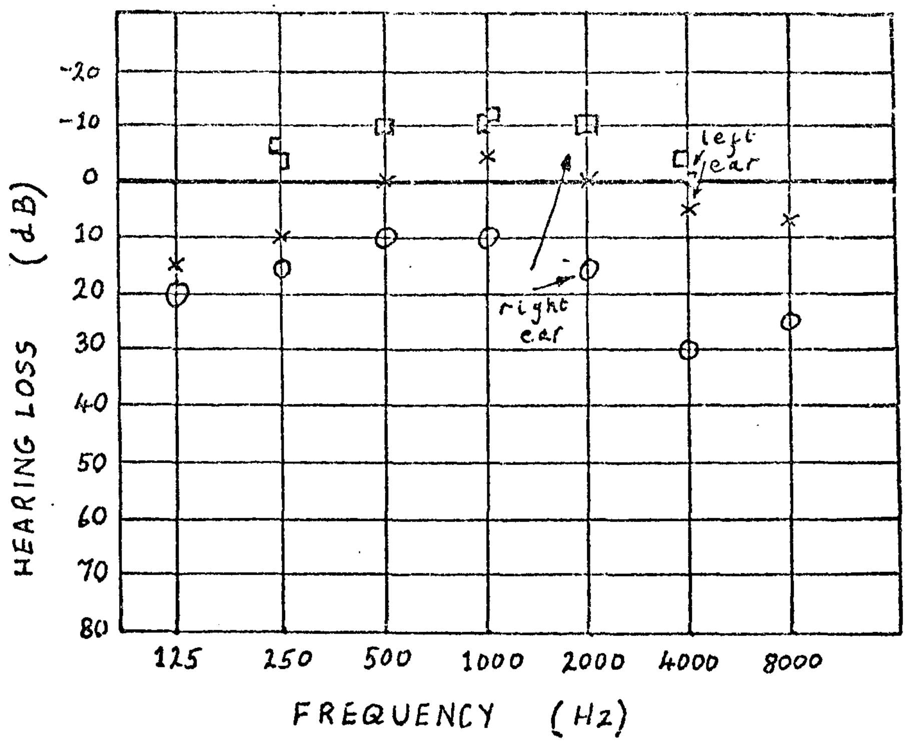
A level is determined above the threshold and then decreased by 5 dB steps until a negative response occurs. The threshold is the lowest level responded to, two or more times out of four.

b) Average of decreasing and increasing series

In this system a level above threshold is found and the intensity reduced by 5 dB until a negative response is obtained and then increased by 5 dB until a positive response occurs. The threshold is the average of the increasing and decreasing series levels so determined. At least four threshold crossings are determined and averaged in this system.

The above description of threshold determination methods is general and makes no mention of special adaptions for non-co-operative or below average intelligence subjects. In cases of this type usage of specially developed techniques is required (13).

The test described is an air conduction test. The same procedure may be repeated using a bone-conductor in place of the earphone. An earphone vibrates the air next to the eardrum and so ultimately the fluid in the cochlea, the bone of the skull and the cochlear duct being stationary. With a bone-conductor the skull is vibrated so causing the cochlear duct to move relative to the fluid in it, the result being perception of the sound. Persons with imperfections in the ossicles or the eardrum may not hear well by air conduction but normally by bone conduction. Typical air and bone conduction audiograms are given in Fig. 1.9. The air and bone conduction audiograms in cases of NIPTS should normally be identical as the condition only affects the cochlear, not the outer or middle ear.



air conduction : o , x ; bone conduction : [,]

Fig. 1.9 Typical pure-tone audiogram

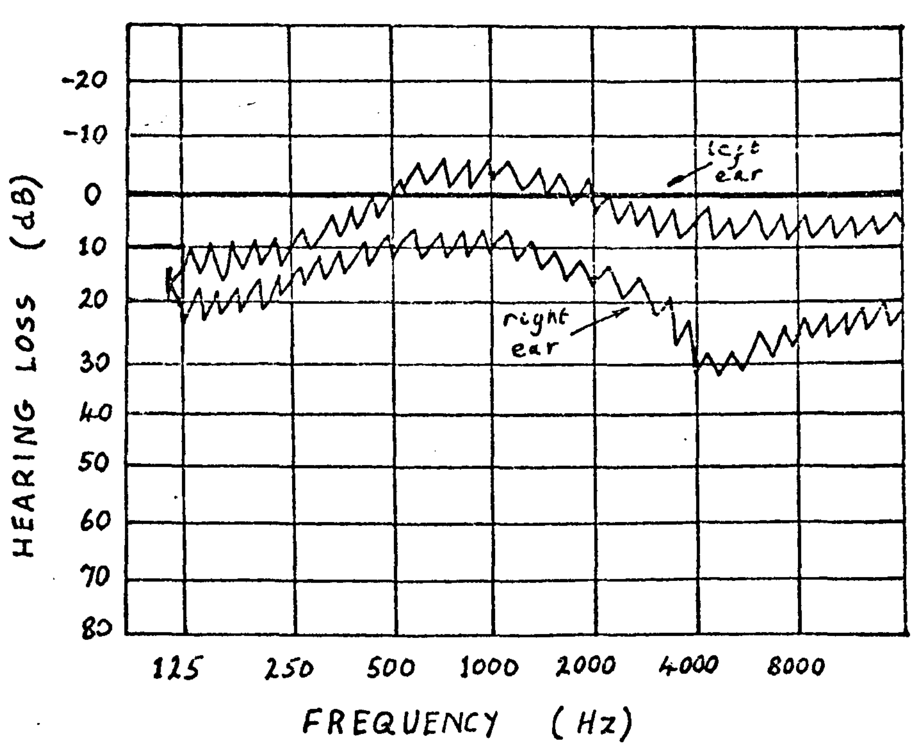


Fig. 1.10 Typical Békésey audiogram

1.4.2 Békésy Audiometry and Self-Recording Audiometry

This is a derivation of the pure-tone test (14). The tone is continuously swept from low frequency to high at a rate of approximately ½ octave/minute, by a motor driven oscillator. Throughout the test the patient has a response button which he is required to depress when he hears a tone and to release when he does not. The button is linked to a motor driven attenuator which reduces the tonal intensity with the button depressed and increases it with the button released. The frequency and intensity are constantly recorded by a pen recorder, with a constant left to right sweep linked to the oscillator sweep-motor, and a directionally varying vertical sweep linked to the attenuator-motor. The subject's responses upon the button produce a 'zig-zag' curve, as shown in Fig. 1.10, which brackets his threshold.

A variation upon Békésy audiometry replaces the constantly increasing frequency with six or eight fixed tones, each lasting one sixth or eighth of the test time respectively. This system produces six or eight separate 'zig-zag' recordings, each corresponding to a fixed frequency.

1.4.3 Speech Audiometry

The view has generally been taken that the ability to receive and understand conversational speech is an almost universal,

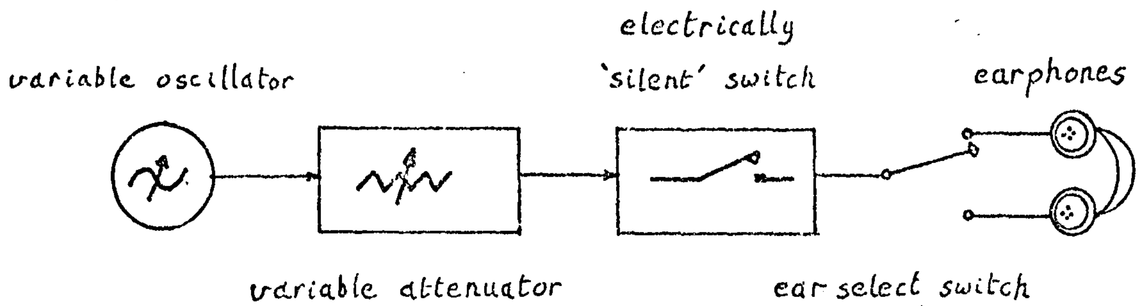


Fig. 1.11 Manual pure-tone audiometer

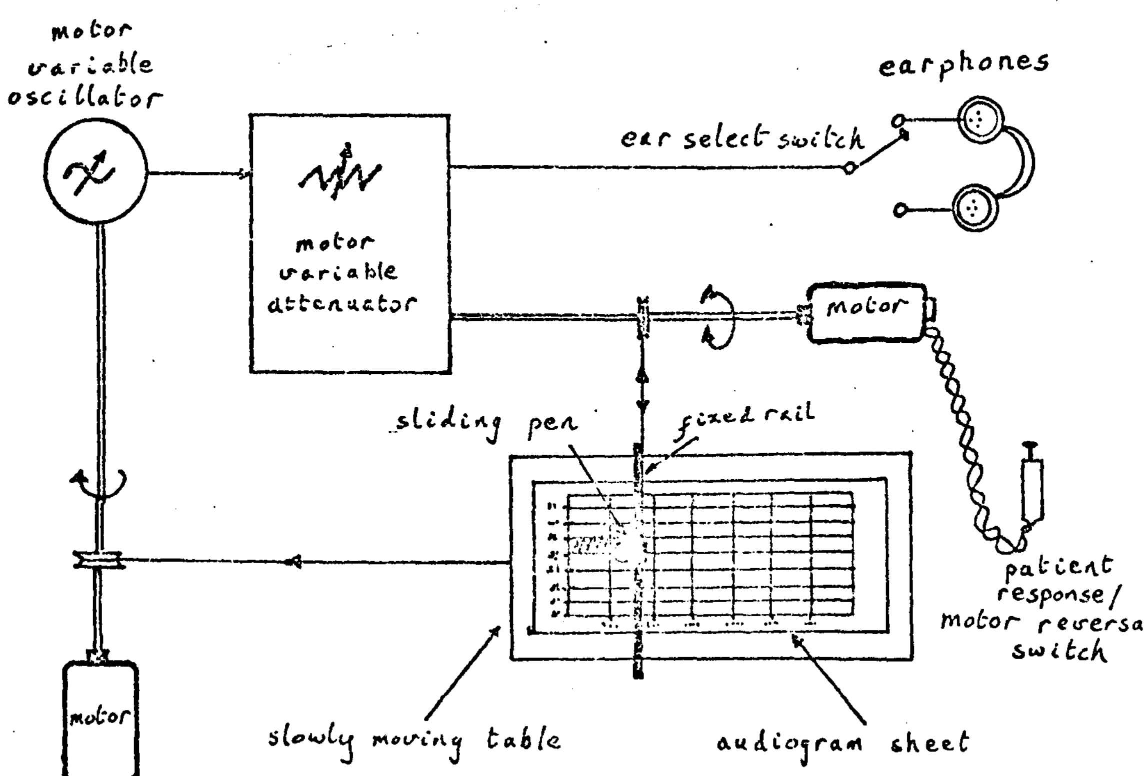


Fig. 1.12 Békésey audiometer system

and hence, more practical, criterion of unimpaired hearing than one based upon the retention of full hearing capacity over the entire audible frequency range. A further reason for this is that the acuity of hearing progressively diminishes with age (presbycusis) beginning with the highest frequencies and ultimately extending downwards. For the population as a whole it would be unrealistic to describe hearing impairments in terms of the whole frequency range audible to the young. Speech audiometry has been developed to enable an assessment of an individual's hearing adequacy in everyday communication to be obtained.

The field of speech audiometry is in need of standardisation and efforts to accomplish this are now beginning internationally. The practices in use can be divided into two groups, the first is concerned with the assessment of the 'speech reception threshold', and the second is a test upon 'speech discrimination'. The speech reception threshold is obtained by presenting spoken words at predetermined decibel levels and requesting the subject to repeat them. The threshold level, is the level in decibels at which the subject obtains a 50% correct score, relative to standards established on normal ears. The speech discrimination test is designed to determine a measure of speech intelligibility with the presentation level at a predetermined audible level. The results of each of these tests is used to determine the 'social adequacy index' (15).

Speech audiometry is a complex subject best studied from the relevant literature (13)(16)(17).

1.4.4 Screening Tests

Not every test needs to give an accurate assessment of an individual's hearing. It is often necessary only to check for deviations from normal in order that they may be checked thoroughly at a later date. Such tests are used by schools and industry where speed and low cost are a requirement in screening large numbers of people. Such large scale examinations may be carried out with a group of subjects at the same time or by brief examinations individually.

These routines do not normally cover the entire audible frequency range but only a part of it (32), often only 500 Hz to 3000 Hz, the frequency range necessary for the reception of intelligible speech. With industrial screening exposure to 4000 Hz only is not uncommon as this frequency produces very enlightening results, as described in Section 1.3.1. Noise induced hearing losses tend to be highest at and around this frequency. This test takes around 30 secs. per person, whereas a standard pure tone test may last 20 minutes. In a typical group test, the assembled listeners, each having a pair of matched earphones, is requested to write yes or no depending on whether they can or cannot hear a tone each time a light flashes to indicate a reply is necessary. The written replies may be quickly checked at the end to determine which individuals need further testing. Up to 150 persons may be checked in an hour by methods such as this.

1.5 Computers in Audiometry

Computers are widely regarded as either calculating machines or highly efficient filing systems. However, modern advances in computer technology, particularly with respect to digital computers, have opened the way to applications far more complex than these basic functions. The field of on-line control is a first class example of this advancement. Systems may be monitored for their current condition by means of analogue or digital output transducers, and the information rapidly processed by the computer. Any necessary correction or other action may then be fed back to the system in a form suitable to control its operation correctly. In a pure-tone audiometric test the audiometrician and patient may be equated to control system components. The patient forms the plant, or controlled part of the system, the audiometrician the controlling network and source of input. A tone is fed to the system at a given frequency and sound pressure level. The controlling network then sets about the task of detecting the minimum level that the plant will respond to regularly at that frequency. This process is repeated with a number of different tones and the resulting minima detected represent the audiogram of the plant. Thus it may be seen that the procedure could be simplified very easily upon a computer if the patient were to be considered merely as a "black-box" upon which we hang transducers. However, as any audiometrician or audiologist would correctly point out, there is far more to the investigation of an individual's hearing characteristics than machine-like measurements of the threshold. The patient is not a piece of machinery to have transducers placed in critical positions for the purpose of measurement, he is human with both psychological and physiological characteristics. Exceptions to this do exist in objective audiometry where the patient may have electronic probes positioned for the duration of the examination and needs to make no conscious reaction as the probes detect all the necessary information. However, for everyday investigation the audiometrician must rely upon the patient's conscious 'yes' or 'no' reply however it may be transmitted from the patient to the operator. The patient may be a very indecisive person, quite incapable of making a definite decision upon whether he can or cannot hear. The only means of discovering the reactions of the patient during testing is to carry out the procedure and watch him closely. Does he dither, is he concentrating, does he really know how he is supposed to react if he hears a tone? These factors influence the results of the examination and as a result require the skill of the operator to observe them and estimate their effects. If a computer is to be used to control a hearing test it is not sufficient that it should merely be capable of imitating the basic operations of the test, it must also be able to detect subject errors, obvious cases of falsification of responses and lack of concentration. It is possible, as will be discussed in chapter 3, for a computer to detect at least some of the likely sources of error by careful analysis of replies from the subject. An uncertainty of how to respond is particularly straightforward to detect as responses tend to be inconsistent to repeated tone presentations. An audiometer capable of producing an audiogram as simply as a self-recording unit but with an increased accuracy and built-in record-keeping and screening would be of great value to industrial audiologists.

Industrial health officials are usually overworked from the outset. The added burden of testing the hearing of thousands of employees regularly, perhaps every nine months or a year is a load often too large for them to cope with. They have neither the time nor the staff for the task of checking the records even if they have the time to test everyone. At present a few of the large industrial organisations have specialised hearing conservation units employed with the sole task of investigating the hearing of employees. In most cases however the testing, recording and checking is done manually. Boredom is the great enemy of repeated manual tasks and with it comes the problem of mistakes made as a result. Computers, once programmed correctly, rarely make mistakes and do not suffer from boredom and its associated lack of concentration. Thus, programme a computer to administer the screening tests and check the results and mistakes are virtually eliminated. Perhaps more important is the fact that a skilled operator is no longer required since the computer is now programmed to perform many of the observational tasks required by analysis of the subject's responses. Given the correct instructions and a clear operating console a moderately intelligent person can supervise the whole procedure with a minimum of training. However, as with the self-recording methods of audiometry, the subject is required to understand his role in the test and perform it accordingly. This requirement places restrictions upon the types of people that may be tested by the system and upon the types of establishment that may use it. Industrial employees may in general be assumed to be of a sufficient level of intelligence to be readily examined in this manner. The same could not be said in a hospital where subjects of widely varying mental ages and physical abilities are examined regularly.

An apparent disadvantage of a computer-controlled audiometer is its size. Until 1972, the involvement of a computer in any measurement system made it very bulky indeed. In that year an electronic, semiconductor component, known as the microprocessor, was introduced. As its name implies, it is a very small computer processing unit designed to be incorporated into equipment requiring the ability to perform computer-like tasks. The pocket-calculator is an example of its usage. The microprocessor device introduced at that time has been greatly improved by many manufacturers in the years since then but the size of the component has been almost constant. Typically, the device is less than eight square centimetres in area and approximately one-half of a centimetre in thickness. When coupled to its associated devices to form a complete computer system it usually requires a circuit board area of less than three hundred square centimetres. As a result of this development, computing techniques may be incorporated into audiometres without increasing their present physical size by more than 10% and their cost by even less. Thus, the two problems common to all computer controlled audiometric units developed to date, of size and cost, may be dismissed. As a result of these problems computers have frequently been employed audiometrically only as calculators or record-keepers. They have been used to analyse the complex waveforms detected from the nervous system in electroencephalography (20) and in a small number of cases to control simple pure-tone audiometers (21), (37), (38), but not on a widespread basis.

In conclusion to this section it may be said that in time saving, and increased accuracy, computers have a great deal to offer both industrial health workers and the hospital service.

1.6 INDUSTRIAL AUDIOMETRY

1.6.1 The Hearing Conservation Programme

A heightened awareness of noise induced hearing loss and its causes, together with its recognition as a compensable industrial injury, has led industry to consider means by which they may reduce, if not eradicate, its incidence. The methods involved in reducing the risk of noise induced hearing loss include noise reduction of machinery, the issue and use of personal hearing protection equipment, co-operative agreements between management and employees and regular audiometric examinations of noise exposed employees to detect whether or not a loss of hearing acuity is occurring. Methods such as these, together with any others considered necessary and helpful, have in the past two decades become widespread throughout industry. They are generally organised into a single, co-ordinated programme commonly referred to as a 'Hearing Conservation Programme'. Audiometry is a vital constituent of a hearing conservation programme because, the hearing threshold measurements obtained during the course of an individual's employment in a given environment, provide an important part of the information required to either prove or disprove the occurrence of a hearing loss due to presence in that environment. Hearing conservation programmes frequently involve several hundreds of noise exposed personnel. As a result, a method of measuring, recording and analysing the hearing threshold which is efficient both in time and operating staff, and is also accurate, is required.

A successful industrial audiometric system requires great care in its initial planning to ensure that it will be able to function efficiently once in operation. In the past, numerous companies have purchased audiometers and merely presented them to nurses or safety personnel with instructions to 'make hearing tests' (13). Frequently, no provisions are made for medical supervision of the hearing test programme or professional training of the audiometer operator. It has also been a common practice to perform such examinations in environments insufficiently quiet to ensure accurate results. Practices such as these produce inaccurate results which if required for legal purposes would be rejected as unacceptable. To ensure the protection of both the employee's hearing and the employer's legal interests, industrial audiometric systems must be operated professionally and under expert medical supervision.

1.6.2 Who Should Receive Hearing Tests?

All employees should receive a pre-employment audiometric examination. The audiogram produced may then be used as a base for the employee's hearing threshold against which any future change in hearing acuity can be compared. The audiogram may serve as an important consideration in job placement when interpreted, in conjunction with the possible future working environment, by the medical supervisor. Also, pre-employment audiograms are useful for medical-legal purposes. It may be necessary for an employer to prove that a large part of a hearing loss existed prior to present employment, the pre-employment audiogram provides the necessary information. The recording of hearing thresholds for all employees is similar to the common practice in industry of making sight and blood tests prior to employment. However, as with other types of medical examination, subsequent examinations need only be performed, on a regular basis, on employees known to be working in environments which will put some aspect of their health at risk, or an employee who frequently passes through noise hazard areas as a part of their job. Although ideally all employees should be examined audiologically on a regular basis, in many cases the testing unit concerned could not handle the large number of extra examinations, this would cause and maintain a reasonable regularity.

1.6.3 The Testing Environment

Industrial plants, because of their high noise levels, require special areas or rooms for the performance of hearing tests. These special provisions for noise-controlled areas are necessary to prevent ambient noise from 'masking' the audiometric test-tones. Masking is the phenomenon whereby one sound reduces the apparent intensity of another. If two sounds are introduced to the ear simultaneously, the threshold for one or both may be higher than when heard separately. The number of dB by which the threshold is raised is the amount of masking. It is important to realise that if the masking sound is below the threshold of the listener then it has no masking effect for that individual. Masking is a complex subject best studied from the literature (18)(19). The most critical case for low levels of background noise occurs in the measurement of hearing of young persons, notably in pre-employment testing, when hearing levels down to -10 dB HL (10 dB below the 'normal threshold of hearing' as in BS 3383) or lower can be expected in some cases. It is extremely unusual to find a situation with noise levels approaching those of the 'normal' threshold of hearing unless they have been specifically designed for the purpose. In the majority of cases, the purchase of a commercially available audiometric booth is the least expensive source of a suitable environment. Such booths have the advantage of being accurately characterised, in their sound insulation properties, by their manufacturer. Thus, given the noise characteristics of the room in which the booth is to be situated, the internal noise levels may be assessed with reasonable accuracy. Quantitive information concerning acceptable noise levels is available in the American Standard Criteria for Background Noise in Audiometer Rooms, S 3.1 - 1961. The acceptable levels will vary with the type of earphone used. Some earphones have circumaural cases, providing extra sound insulation beyond that of the booth, whilst others are mounted on rubber cushions which, although presenting sound insulation from outside noise, do so to a different extent. I.E.C. Recommendation 178, concerning pure tone screening audiometers, suggests that a suitable test to ensure a sufficiently 'quiet' environment is for a person of 'normal' hearing to sit in the booth, wearing the disconnected earphones, and decide whether he can detect any noise present. If the decision is that no noise is present then the booth is suitable for performing examinations in. This subjective assessment of the test

environment is adequate where the location of the booth allows such low noise levels to be achieved, however it is important to realise that it is not necessary for the environment to be completely silent to the normal ear, only that the noise present does not cause test-tone masking to a degree where it affects the required results.
In industrial audiometry, where hearing thresholds are often not measured below 0 dB HL, a low level of background noise is acceptable, provided that it has been previously analysed to ensure that it will not cause masking of the test tones.

1.6.4 Industrial Audiometric Equipment

For industrial audiometry, a simple, but properly calibrated and maintained, discrete frequency, air-conduction audiometer is all that is essential. Self-recording audiometry has become popular for industrial hearing-testing programmes, its purpose being to save time and personnel. This type of audiometry allows the subject to control the test instrument and make his own hearing-threshold determinations, as described in section 1.4.3. In practice, the time required to complete an audiogram is approximately the same as that for a manually controlled examination and supervision is still required as with a manual audiometer. However, the supervisor may oversee more than one audiometer at a time. This has the advantage of allowing a larger number of examinations in a given time, a most important feature in large industrial plants. Unfortunately, the lack of personal attention to each subject throughout his test can allow inaccuracies such as a loss of concentration to pass unnoticed.

1.6.5 Personnel Requirements

A minimum of two people is required to operate an industrial audiometric unit. The first is the audiologist whose task it is to establish the testing programme, ensure its efficient running and evaluate and make the necessary decisions upon all audiograms produced as a part of the programme. The second is the audiometrician, the person who actually administers the examinations. Ideally, the audiometrician should be experienced, and highly trained specifically for this task. However, frequently it is the company nurse, or even an employee, who has been trained to perform the basic tasks necessary. The use of self-recording audiometers has encouraged the use of less highly trained staff as the task of operator has been reduced to loading and removing the audiogram sheet from the recorder and explaining the procedure to the subject. The personal skills required to perform a test manually are no longer required, but nevertheless it is desirable for the audiometrician to understand more than the operation of an audiometer. To perform effectively his task as a part of a hearing conservation programme he should be aware, and able to explain, the reasons for any measures being undertaken as a part of the programme. For this a basic understanding of the anatomy and physiology of the ear, fundamentals of sound, and the problems of industrial noise is necessary.

1.6.6 Audiometric Testing Procedures

a) Audiometric technique

As previously described in section 1.6.5 self-recording audiometry is becoming the standard technique for industrial audiometry. Although such audiometers do not necessarily produce such precise results as a professionally operated manual audiometer, self-recording units are consistent in their technique. Variations in technique can result in measured differences in hearing levels and so should be avoided. The important frequencies to be used as test-tones are those necessary for the understanding of speech; namely 500, 1000 and 2000 Hz. It is usual to use at least three other frequencies, often 4000, 6000 and 8000 Hz, as it is at these frequencies that the effects of NIPTS are displayed before they intrude upon the speech frequencies. The lower threshold limit to which measurements are made is usually -10 dB HL or 0 dB HL, the audiologist supervising decides which is the most useful for his purpose.

b) Record keeping

The operation of a hearing conservation programme requires more information upon a subject than his pre-employment, and any subsequent audiograms. It is necessary for the audiologist to have a knowledge of the person's working environment, post medical history where it is relevant to his hearing, such as head injuries, family history of early deafness and frequent earaches, any pastimes involving high level noise exposure, and his previous employment. A questionnaire is usually used at the pre-employment examination to record such information as is required. An example of such a questionnaire is shown in Fig. 1.13, the pre-employment audiogram may be seen in Section C. The information obtained in this manner, together with results from subsequent audiometric examinations are carefully filed either in written form or on a computer.

c) <u>Periodic follow-up testing</u>

The initial pre-employment audiogram is used as a reference with which later audiograms can be compared. Periodic follow-up examinations are necessary where the worker is exposed to potentially hazardous noise levels. The first follow-up test is usually performed 9 to 12 months after initial employment, unless the worker is employed for long periods in particularly high level noise of perhaps greater than 100 dB(A), when an examination is often made at 6 months. Follow-up tests should preferably be made as long after a period of noise exposure as possible to allow the effects of temporary threshold shift (TTS) to recover. This is often a cause of problems because complete recovery may take longer than overnight. It is not always possible to arrange for an employee to work in a quiet environment in order to recover. Neither is it possible to make all examinations on a Monday morning before commencement of work. It is the responsibility of the supervising audiologist to adopt techniques to minimise errors due to TTS. Each time a new audiogram is produced, for a given employee, it is compared with the reference stored on the

HEARING CONSERVATION DATA CARD NO.________

TYPE OF AUDIOGRAM: REFERENCE AND/OR PRE-EMPLOYMENT ☐ RECHECK ☐ OTHER________

A. IDENTIFICATION

LAST NAME	MIDDLE	FIRST	SEX	DATE OF BIRTH		
			MALE FEMALE	DAY	MO.	YR.

SOCIAL SECURITY NUMBER	COMPANY NUMBER

B. CURRENT NOISE-EXPOSURE

JOB TITLE OR NUMBER	DEPARTMENT OR LOCATION	TIME IN JOB		
		NONE	MOS.	YRS.

NOISE-EXPOSURE

STEADY NOISE	IMPULSE NOISE	PERCENT TIME NOISE ON	EMPLOYEES ESTIMATE OF OWN HEARING
CONTINUOUS ☐	CONTINUOUS ☐	10 20 30 40 50	GOOD
INTERMITTENT ☐	INTERMITTENT ☐	60 70 80 90 100	FAIR
			POOR

C. AUDIOGRAM

TIME SINCE MOST RECENT NOISE-EXPOSURE	DURATION OF MOST RECENT NOISE-EXPOSURE
0-20 MIN 1 HR 4-7 HRS 1 DA	0-20 MIN 1 HR 4-7 HRS
21-50 MIN 2-3 HRS 8-16 HRS 2-3 DAS	21-50 MIN 2-3 HRS 7+ HRS
4+ DAS	

AGE	DATE OF AUDIOGRAM	DAY OF WEEK	TIME OF DAY	EAR PROTECTION WAS EAR PROTECTION WORN? YES NO

RIGHT EAR								LEFT EAR							
250	500	1000	1500	2000	3000	4000	6000	250	500	1000	1500	2000	3000	4000	6000

D. PREVIOUS NOISE-EXPOSURE AND MEDICAL HISTORY

PREVIOUS EMPLOYMENT (LAST 3 JOBS)

TYPE OF WORK	FOR WHOM	HOW LONG

HISTORY

HEAD INJURY (WITH UNCONSCIOUSNESS) ☐
HEARING LOSS IN FAMILY (BEFORE AGE 50) ☐
TINNITUS FOLLOWING NOISE-EXPOSURE R L

STATUS

PERFORATIONS OF DRUMHEAD R L
DRAINAGE FROM EAR R L
MALFORMATION OF EAR R L

RECORD ANY COMMENTS SUBJECT MAKES ABOUT HEARING

TECHNICIAN________
PHYSICIAN________

AUDIOGRAM

TIME SINCE MOST RECENT NOISE-EXPOSURE	DURATION OF MOST RECENT NOISE-EXPOSURE
0-20 MIN 1 HR 4-7 HRS 1 DA	0-20 MIN 1 HR 4-7 HRS
21-50 MIN 2-3 HRS 8-16 HRS 2-3 DAS	21-50 MIN 2-3 HRS 7+ HRS
4+ DAS	

AGE	DATE OF AUDIOGRAM	DAY OF WEEK	TIME OF DAY	EAR PROTECTION WAS EAR PROTECTION WORN? YES NO

RIGHT EAR								LEFT EAR							
250	500	1000	1500	2000	3000	4000	6000	250	500	1000	1500	2000	3000	4000	6000

AUDIOGRAM

TIME SINCE MOST RECENT NOISE-EXPOSURE	DURATION OF MOST RECENT NOISE-EXPOSURE
0-20 MIN 1 HR 4-7 HRS 1 DA	0-20 MIN 1 HR 4-7 HPS
21-50 MIN 2-3 HRS 8-16 HRS 2-3 DAS	21-50 MIN 2-3 HRS 7+ HRS
4+ DAS	

AGE	DATE OF AUDIOGRAM	DAY OF WEEK	TIME OF DAY	EAR PROTECTION WAS EAR PROTECTION WORN? YES NO

RIGHT EAR								LEFT EAR							
250	500	1000	1500	2000	3000	4000	6000	250	500	1000	1500	2000	3000	4000	6000

AUDIOGRAM

TIME SINCE MOST RECENT NOISE-EXPOSURE	DURATION OF MOST RECENT NOISE-EXPOSURE
0-20 MIN 1 HR 4-7 HRS 1 DA	0-20 MIN 1 HR 4-7 HRS
21-50 MIN 2-3 HRS 8-16 HRS 2-3 DAS	21-50 MIN 2-3 HRS 7+ HRS
4+ DAS	

AGE	DATE OF AUDIOGRAM	DAY OF WEEK	TIME OF DAY	EAR PROTECTION WAS EAR PROTECTION WORN? YES NO

RIGHT EAR								LEFT EAR							
250	500	1000	1500	2000	3000	4000	6000	250	500	1000	1500	2000	3000	4000	6000

AUDIOGRAM

TIME SINCE MOST RECENT NOISE-EXPOSURE	DURATION OF MOST RECENT NOISE-EXPOSURE
0-20 MIN 1 HR 4-7 HRS 1 DA	0-20 MIN 1 HR 4-7 HRS
21-50 MIN 2-3 HRS 8-16 HRS 2-3 DAS	21-50 MIN 2-3 HRS 7+ HRS
4+ DAS	

AGE	DATE OF AUDIOGRAM	DAY OF WEEK	TIME OF DAY	EAR PROTECTION WAS EAR PROTECTION WORN? YES NO

RIGHT EAR								LEFT EAR							
250	500	1000	1500	2000	3000	4000	6000	250	500	1000	1500	2000	3000	4000	6000

Fig. 1.13 Typical industrial audiometric questionnaire and record sheet. ("Audiometry", A. Glorig.)

initial questionnaire. A deterioration in any part of the audiogram of more than 10 dB from the reference is usually considered sufficient to warrant referral of the subject for a detailed clinical examination to determine the exact extent of the loss. Should the loss be proved to exist, measures are usually taken to protect the employee from further loss.

CHAPTER 2

An Analysis of Industrial Audiometric Procedures

2.1 Audiometric Test Techniques

The purpose of industrial audiometry is to determine, by whatever means are available, the condition of every employee's hearing acuity upon a regular basis in order that any loss of hearing which may be attributable to the conditions of employment be detected as soon as possible and suitable remedial action taken. Provided that the test technique used will supply reliable, repeatable results which if recorded at intervals over a period of time may be used to detect a developing hearing loss then the nature of the test is not important. The choice of test to be used in any particular industrial situation is governed by conditions common to all such audiometric installations: a) The need for brevity of examination duration; b) The need to operate under a minimum of personal supervision. The following paragraphs discuss the suitability of two types of audiometric test; speech and pure tone audiometry.

2.1.1 Speech Audiometry

The tests adopted by industry have been largely directed towards the determination of the hearing threshold rather than towards measuring the communication disability of the subject as in speech audiometry. Historically, many valid reasons exist for this choice of measurement. Only with recent advances in methods for presenting speech sounds has speech audiometry become sufficiently precise for the results to be used in legal arguments concerning claimed hearing loss. However, as Robinson (22) states, speech audiometry is still in need of standardisation and efforts in this direction are only now beginning internationally. Speech audiometry is not the only technique for deriving a measure of speech perception. During the period prior to speech audiometry being refined into an accurate measurement, Fletcher (16) developed a system to assess the ability to hear speech from a knowledge of hearing acuity for pure tones. Fletcher's system computed the loss of hearing for speech (speech loss) from the following equation, where HL represents pure tone hearing loss at the frequency indicated by the subscript.

$$\text{Speech Loss (dB)} = 5 + \left(\frac{HL_{500} + HL_{1000} + HL_{2000}}{3}\right) \qquad 2.1.1$$

This system was later modified by Quiggle, Glorig, Delk and Summerfield (23). The predicted speech loss is computed from a knowledge of the pure tone hearing loss at three frequencies; 500, 1000 and 1500 or 2000 Hz. The three losses are independently weighted, added together and then added to a constant. The equations produced are given below.

$$\text{Speech Loss} = 6.9 + 0.22\ HL_{500} + 0.35\ HL_{1000} + 0.21\ HL_{1500} \qquad 2.1.2$$

$$\text{Speech Loss} = 6.9 + 0.22\ HL_{500} + 0.47\ HL_{1000} + 0.09\ HL_{2000} \qquad 2.1.3$$

The low weighting of the 2000 Hz hearing loss indicates its small contribution to the computation. The reason for the derivation of equation 2.1.3 was that many commercially available pure tone audiometers have a 2000 Hz source but not a 1500 Hz source. Further research has been undertaken to improve the reliability of these equations by J.T. Graham (24). However, it must be noted that the formulae are not always applicable when cochlear or central nervous disorders exist, or when there are unusual language backgrounds. Also, they apply only to the intensity of speech needed to furnish 50% correct judgments. As a result of these limitations, the prediction of speech loss by these methods is rapidly being replaced by modern speech audiometry.

From the preceding paragraph it may be understood why industrial audiologists have not used speech audiometry or attempted to estimate speech loss from pure tones. The results so obtained would be sufficiently open to argument that they could not be used in a legal situation. Until such time as speech audiometry techniques are standardised, and a measure of speech loss is internationally agreed, they are likely to remain clinical in their application.

2.1.2 Pure-tone audiometry

Industrial audiometry is required to be simple in operation and as brief as possible in duration. The equipment necessary should be as inexpensive as the accuracy limits set by the supervising audiologist will allow. For these reasons, self-recording audiometers have become widely used in industry. The use of self-recording instruments is not necessarily a result of their being exactly what is required in industry, but of their being the nearest solution to the measurement problem that the audiometer industry has produced. In order to support this statement an analysis of what is required of an industrial audiometer is necessary.

An industrial audiometer is required to perform two different types of examination. The first is the pre- and post-employment examination, where an accurate, clinically acceptable audiogram of the full audiometric frequency range is required. The second is the follow-up examination where we wish to detect changes in the hearing threshold relative to the pre-employment audiogram. In particular, we wish to detect changes that might have occurred due to noise exposure, in order that action may be taken to prevent further possible deterioration in hearing acuity. Each of these measurements could be made using a clinical audiometer. However, the level of skill required from the operator and the length of time to perform each examination, are both contrary to the requirements of an industrial audiometric unit, as already stated. These requirements are frequently overruled for the pre-employment audiogram, where a high level of reliability is required, but for follow-up examinations, every effort is made to adhere to them. The accuracy required from a follow-up test may be relaxed because any apparent hearing loss observed as a result will cause a referral for an accurate examination. The standard of accuracy should not be allowed to become so low that the number of referrals outweighs the time advantage of the technique, but nevertheless the follow-up tests, acting as screening examinations of the noise exposed population of a factory, may be performed on equipment requiring less attention during examinations. Self-recording audiometers have virtually become the standard instruments for this purpose. The term screening is normally used to indicate that a measurement or instrument may be used to detect variations from a fixed specification, such as the 'normal' threshold of hearing for 18-25 year olds. An example of such a screening test is the individual pure tone 'sweep-check test' (13), (39). The audiometer is set to a pre-determined, fixed level, perhaps 10 dB HL. The subject is requested to respond in an agreed manner if he hears a tone in his earphones. Tone bursts are then presented at the test frequencies of 1000, 2000, 3000 and/or 4000, and 6000 Hz, in that order. If the subject fails to respond to the tones in a predetermined manner, then he is referred for a complete threshold examination. When wishing to detect school children with defective hearing, screening techniques such as this may be used. However, in the industrial environment, each employee supplies his own 'normal' against which to be screened and thus the term, screening audiometry does not have its accepted meaning. Screening audiometers with adaptive screening criteria do not currently exist commercially. Self-recording audiometers, previously described as having become the standard industrial instrument, measure only an approximation to the absolute threshold of hearing of the subject, comparisons with his pre-employment audiogram must be made manually. An instrument required to adapt its comparison criteria to suit the subject under test must have a store of these criteria and a means of relating them to the individual as required. The feasibility of such an instrument will be discussed in chapter 3.

The duration of each examination is required to be as short as possible to allow a maximum number to take place in any given period of time. Manual tests are automatically adaptive in duration as the operator makes the decision as to whether or not he has determined the threshold of the subject at any given frequency. Once determined at one frequency the operator moves on to the next. Self-recording audiometers present each frequency either continuously or in a pulsed on-off manner for a fixed period of time, usually approximately 30 seconds. It is assumed that the graph plotted during that time will contain sufficient information for a threshold figure to be derived. In the majority of cases 30 seconds is more than sufficient time but occasionally it is not. What is really required is a decision mechanism capable of recognising that a threshold figure may be reliably computed from the responses up to that point in time and able to terminate the test at that frequency. Thus, in many cases the measurement would be made in a shorter time and only occasionally in longer.

Current self-recording audiometers are of two types. The first presents a number of tones continuously for 30 seconds each and sweeps up and down in level as directed by the subject's response button. The other also presents tones for 30 seconds but as a series of 'beeps' 250 mS in duration every 250 mS. The latter technique has one distinct advantage, particularly at high frequencies; 1 kHz and higher. The advantage derives from the fact that a silent period between tone bursts enables the subject to contrast the test-tone with background sounds. Although the test environment should not be subject to background noise, self-generated noise such as that of tinnitus, 'ringing in the ears', cannot be controlled by the operator. Thus, whereas in a continuous swept-tone test the tone and the tinnitus may cause the subject to become confused, a pulsed tone will at least assist him to differentiate between them.

2.2 Sources of Error in Pure Tone Audiometry

2.2.1 Calibration

Let us now examine some of the major causes of error in pure tone audiometric measurements. The first of these is an error present wherever electronic instrumentation is used; faulty calibration. The calibration of pure tone audiometer systems, including their associated earphones, is specified in both the British Standard, BS 2497 (parts 1-4) and in the International Standards Organisation document, ISO 389. These documents specify the information necessary for a laboratory or workshop to calibrate accurately an audiometer at manufacture or maintenance time using precision sound level meters, microphones and artificial ears, but are rarely of use to the operator in a clinic who, in all probability, has no access to, or understanding of, the equipment necessary. Routine, fine adjustment of the overall calibration of a particular instrument is usually described in the audiometer manufacturer's handbook. It is the responsibility of the operator to ensure that the instructions are performed correctly and at the required regularity. The procedures are usually straightforward, often entailing no further action than adjustment of a 'calibration' knob until a meter needle is in a pre-determined position. However, the meter needle is normally used to monitor the applied voltage level, at a given frequency and sound pressure level, across the earphones. Such a check is valid only on the assumption that the earphones are in good condition, and the meter circuit is operating correctly. Both meter circuits and earphones may be constructed to high degrees of reliability, but the possibility of them being a source of error is always present. As already stated, this type of check is often performed only at a preset frequency and sound pressure level, it does not guarantee the performance of the audiometer at any other setting. It is the responsibility of the manufacturer to ensure that his recommended check maximises the probability of the instrument functioning correctly over its full range of operation. A further source of error concerning the earphones is the impracticality of ensuring that they are situated on the head in such a way that the sound pressure level (SPL) they present to the eardrum is that which should be present. Earphones are calibrated whilst situated on one of a number of standard artificial ears. The dimensions of the ear canal and the airtight seal between the earphone surround and the head are both critical to achieving the correct SPL at the eardrum. Any variation in these from the standard can give rise to errors as high as 3 dB. For this reason spectacles should normally be removed and hair should not be allowed to remain between the earphone and the head. Differing shapes of pinna cause earphones to sit differently positioned relative to the ear canal. To overcome this problem, and to provide a better barrier against outside noise, circumaural earphone housings have been developed. With this type of earphone the pinna is untouched inside a cavity and the earphone diaphragm relatively constant in position relative to the ear canal, from one person to another. Unfortunately, the increased distance from the eardrum causes inconsistencies of results at high frequencies. The problem of earphones is one which has received much attention (22) and is likely to remain a discussion point for some considerable time to come.

2.2.2 Operator error

The next source of error to be considered is that of the operator himself. The types of error that can be made are highly dependent upon the equipment being used. Manual audiometers offer a far greater possibility for an operator to make a mistake than do self-recording models with which once the test has been initiated it continues automatically. With manual units every operation may be a source of error, mistaken level or frequency settings, incorrect recording of results, and even reversal of the left and right ear readings due to inadvertantly placing the earphones on the subject's head the wrong way round. With the exception of the last stated example all these errors can be avoided by the use of a self-recording audiometer. However, self-recording units produce their results in graphical form which the operator has to translate into numerics for record keeping. Not only is the action of transferring data from one place to another, a possible cause of error but also the reading of a self-recording 'zig-zag' graph is itself a subject upon which qualified audiologists may disagree in cases of extreme excursions of the pen marks. Included in the possible sources of error due to an operator is the previously discussed lack of calibration.

2.2.3 Errors introduced by the subject

A third class of error which may be caused by the operator, or by the subject under examination, is that of faulty understanding by the subject of what he is expected to do. It is most important that before an examination the operator should explain clearly, and ensure understanding by the subject of the task he is expected to perform. Unfortunately, many subjects, even in industry, are unable to remember what is expected of them for more than a few minutes. Consequently, they cease to respond to tones in the required manner and thus invalidate the results. With a manual audiometer, the operator will detect such people as soon as they start to falter, but with a self-recording model, unless the operator is looking at the graph as it is plotted, the examination may provide a meaningless result. Lack of understanding from the subject is more often a source of wasted time than a source of error. In order to ensure a reasonable understanding of what is required during an examination, many self-recording audiometers have practice phases. Typically, the subject will be tested at 1 kHz prior to the main test taking place. During this practice phase the operator must observe whether the subject is responding correctly and rectify any misunderstandings. The practice phase recording is often part of the final result sheet and can in fact be used to compare with the later 1 kHz recording made in the actual examination.

Although the pre-employment audiogram may be made a special-case and be performed with all the care and attention available, including ensureing that the employee has not been exposed to high level noise for two or more days prior to the examination, the subsequent re-tests must usually be made as and when possible. Thus, the probability of TTS existing to some unknown degree is quite high. This error will vary with the length of time the subject has been exposed to the particular noise on the day of the examination and whether any TTS existed at the start of the day's work. At the end of an eight hour work period, having been exposed to noise levels in excess of 90 dBA, the degree of TTS at 4 kHz may be as high as 20 dB. At 4 kHz a greater loss than at any other frequency is normally displayed with the loss at 2 kHz being perhaps only half that figure. The degree of TTS suffered by individuals during a given period of noise exposure varies greatly and thus cannot be accurately compensated for. The only reliable means of removing TTS from audiograms is to ensure that the subject has not suffered noise exposure for at least 36 hours previous to the examination.

TTS is not the only temporary source of error outside the control of both the subject and the operator. Head-colds tend to block the eustachian tube causing a pressure difference between the middle and outer ear. This in turn makes the eardrum less compliant and so effectively reduces the hearing acuity. Before the commencement of any examination the external ear canal should always be checked for excessive wax which may cause a partial blockage of the passage and so increase its impedance to sound pressure-waves. Physical exertion can cause an effect often described as 'the sound of blood rushing around in the head'. If after climbing stairs, for example, an individual sits down in a quiet environment he may hear his heart beat and a hissing sound. Were an examination to take place at that time the sound would act as a mask and thus modify the audiogram. Another effect is that of mental adjustment after strenuous activity. If one sits down immediately after such activity it takes a few minutes for the mind and body to relax and become adjusted to the quiet, relaxed environment. This relaxation is essential to ensure the maximum concentration from the subject.

Examination of self-recorded audiograms will often show singular large excursions amidst otherwise regular responses. These are frequently the result of a momentary laps of concentration from the subject, although another far more serious explanation may exist in the form of malingering.

2.2.4 Malingering and Psychogenic Hearing Loss

Malingering is the wilful imitation of illness or its severity, for purposes of personal gain, usually monetary, or avoidance of duty. Compensation payments to employees able to prove noise induced hearing loss make attempts at feigning such a loss an attractive proposition. In the USA, in 1959, it was estimated that of all industrial claims for damage, 90% were possible malingerers and 20% were obvious malingerers (25). The patient interview is extremely important, for an experienced observer can frequently expose a malingerer at this time. Obvious discrepancies in auditory behaviour that do not correlate with the test performance may be noticed. The patient's attitude toward his hearing, as revealed in the interview, supplies the examiner with an important clue. The person with a true hearing loss is usually worried and shows it. The malingerer, however, displays a more nonchalant attitude and appears quite unconcerned. It is in the interview also that motivation for simulation may be revealed, perhaps accidentally. Hardly any individual malingers without a motive, and when the motive is discovered and removed, the problem often disappears. A major problem exists in that malingering is not confined to someone intent upon fraud and who consciously feigns illness, it is also exhibited by individuals with psychogenic hearing losses. In cases of psychogenic hearing loss the simulation of loss is subsconscious and therefore cannot be classed as deliberate. As a result, extreme care must be taken before accusing a subject of malingering, as it is a charge which can seldom be proved without an admission from the individual of an intent to defraud. It should be stated that as a source of erroneous results, malingering is not unique to the pure tone techniques under discussion. However, self-recording audiometers allow examinations to take place without operator observation of the subject, thus making attempts of malingering a little less daunting. Malingering is an excellent example of why the subject under test should never be able to see the audiogram as it is being plotted during the examination. Without the visual feedback that the graph would provide, the subject must rely upon his own concentration and judgement which may lapse to provide important information to the operator upon eventual scrutiny of the audiogram. A technique developed for pure tone audiometers, to assist in the detection of malingerers, is the '+20 dB' button. The purpose of the button is to elevate the current sound pressure level output by 20 dB for as long as the button is depressed. A sudden change in level of such magnitude will amost certainly cause a subject to lose concentration. Anyone responding honestly to the examination will attempt to return the level to his threshold, either on a manual audiometer by a positive response or on a self-recording model by pressing a reply button. A malingerer will similarly respond in a positive manner but will have the problem of recognising his previously selected, feigned threshold for that frequency. In the case of a self-recording audiometer this may be displayed as a series of excursions, possibly down to his true threshold and back up again. A comparison of the settled excursions, before and after the +20 dB was applied, may show a definite threshold change caused by the inability of the malingerer to recognise his would-be threshold upon return to it. Unfortunately, the +20 dB button requires an observation and an action from the operator. Thus, although most self-recording audiometers are fitted with them, only those units that are observed by the operator may make use of them. For this reason it is important that an operator should not have so many units in operation at the same time that he is rendered incapable of at least partially observing each examination. A malingerer is attempting to create an apparent threshold within his hearing ability. A less common, but very real, situation is the individual who wishes to create an apparent hearing acuity superior to that which he actually possesses. His reason may be to prevent his removal from a particular task, or to avoid the need to wear hearing protection. By his actions he is endangering himself and ultimately, possibly his employers if he ever decides to claim for the resultant hearing loss. Should such an occurrence take place at a pre-employment examination, and succeed, then any subsequent hearing loss will appear greater in comparison with the original than it really is. Thus, valid reasons exist to suspect feigned thresholds in either direction.

An examination of the pure-tone testing techniques reveals a characteristic of them that greatly contributes to the ability of an individual to feign a hearing threshold. The characteristic responsible is predictability. This criticism is most valid in self-recording audiometry. In swept level self-recording tests the level sweeps at a constant rate, thus by careful timing a regular 'yes-no' response pattern may be recorded at any signal level, above or below true threshold. Timing may be effected by use of a wrist watch or by the musical technique of foot-tapping. With pulsed-tone self-recording audiometers the time between tones is often regular, thus allowing an individual to recognise a rhythm and respond in a 'yes-no' manner even below his threshold. Manual instruments are less susceptible to the below threshold problem as the operator controls the regularity of the tones. However, above threshold, where the subject can hear every tone, however randomly distributed in time, he need only respond 'yes-no' two or three times in succession to apparently indicate his threshold level at the tonal frequency. Two distinct problems may be seen from this discussion. Above threshold, the subject can hear when the tone is present, thus by regular responses may imitate a threshold. Below threshold, the subject cannot hear when tones are presented but may determine a rhythm whilst the tone is decreasing above threshold, such that anticipation of when the tone is present is possible. From the knowledge that tones change level in regular increments or at constant rates, a subject may, with sufficient concentration, approximate his feigned loss to a predetermined level below his known threshold, assuming he has access to such details.

2.2.5 Audiogram assessment and referral procedures

Once an audiogram has been obtained, by whatever method, the result must be compared with the pre-employment record to determine whether or not a sufficient loss has occurred in the period since it was taken to require the subject to be referred for a more detailed examination. For this purpose, the pre-employment results must be assumed to be correct and not subject to errors beyond those normally attributable to the technique by which they were obtained. At this point let us examine the possible errors which can occur in producing these records.

Pre-employment records may take the form of a graph, either manually plotted or of self-recording type or be numeric. The latter form is more usual where personnel records are kept on a computer system. Where numeric records are kept, the figures stored are usually the result of an interpretation from an audiogram. Such interpretations, particularly from swept-level recordings, are subject to variations in judgement. These variations can be as high as 10 dB in extreme cases of self-recording audiograms where excursions are large and the turning points very variable in level. Under more usual conditions a 3-5 dB difference can be expected. The decision to interpret a given pattern as a certain level of hearing loss, is governed by the training and experience of the person reading it. It is therefore desirable, but not always possible, for the same person to interpret each audiogram. In any particular audiometric unit a standard procedure for interpretation should be adopted and agreed to by a qualified medical practicioner. In many cases the audiogram graphs are only interpreted by the medical supervisor and not the audiometrician, in order to maximise the amount of meaningful information that is extracted from the result and effectively standardise the reading technique. In practice it is necessary to apply an error margin of $\pm$ 5 dB to any audiogram gained by a self-recording technique. Manual audiograms are also subject to operator variation. Unlike self-recorded audiograms the manual type show relatively precise hearing levels on the graph, marked by crosses and circles. However, the precision is only as accurate as the operator's judgement at the time of completing the test. Once again experience has a large effect on the recorded loss and as a result a similar error margin must be allowed. The error margins discussed in this paragraph are only meant to compensate for variations in judgement by an operator, they do not allow for errors due to a subject suffering from TTS or any other temporary impairment at the time of the examination.

These same errors occur in the production of subsequent re-test audiograms with the added source of error due to a probable lesser degree of operator participation as already discussed. Although it has been stated that test standards may be relaxed at re-test as cases displaying extraordinary characteristics may be referred for an accurate examination, it is desirable to minimise the number of such referrals as they are both time consuming and inconvenient to all concerned. A significant improvement could be made to self-recording audiometers if a means were found of retaining the information contained in the excursion pattern, extra to the threshold, but presenting the threshold figures in numeric form derived from the pattern in a standard manner. A major problem with such a system could be to standardise a decision criterion that would be acceptable to all users.

The comparison of the re-test and pre-employment audiograms must also take into account the period of time that has elapsed between them. As shown by BS 3383 the normal threshold of hearing changes with age, particularly above 1 kHz. Thus, the region of the hearing threshold most susceptible to noise damage is also that affected by age. Despite the BS 3383 curves, as many people must be above an average as below and the spread can be very large indeed, as much as $\pm$ 20 dB may be expected. From this it is difficult to imagine how any correction for age can be derived on a personal basis and thus how presbycusis can be separated from NIPTS in individual cases. This same argument is valid for any permanent impairment of the hearing, whatever its cause. It is only in a legal claim that the exact degree of loss due to noise exposure is important. However, the purpose of the hearing conservation programme is to prevent employee's hearing deteriorating beyond given limits. It does not matter what the cause of the loss is, the important objective is to prevent deterioration beyond the point where it will cause personal difficulties. Should an individual exceed the limits of threshold shift set by the employer then he is normally referred to a hospital or clinic for detailed examination.

CHAPTER 3

Applications of Computing Techniques to Pure Tone Audiometry

3.1 An Analysis of the Responses to Test Tones

As was discussed in the previous chapter there are two commonly used techniques in self-recording audiometry; swept-level and pulsed-level. Pulsed tones have the advantage of reducing interference from tinnitus which, at high frequencies, can cause difficulties for the subject if he is not provided with a definite contrast between the stimulus and the silent period. For that reason the pulsed tone technique was adopted as the preferred system for this realisation. The decision is otherwise arbitrary as the computer realisation of both is equally possible. Fig. 3.1 shows the logical sequence of events in a pulsed-tone examination. This sequence is valid for both pulsed, self-recording and manual audiometry. The diagram is generalised in form but shows each of the major decisions and operations which will be discussed in the following paragraphs.

An assumption which must be stated before any further discussion is undertaken is concerned with the mental and physical ability of the subject. It is assumed that the subjects upon which the examination is to be performed are of a mental and physical condition commensurate with their presence in an industrial occupation and not, as is frequently the case in hospital audiometric clinics, disabled in either respect such as to require continual assistance in order to partake in the examination. The above assumption greatly simplifies the task of automating the pure-tone procedure as the majority of

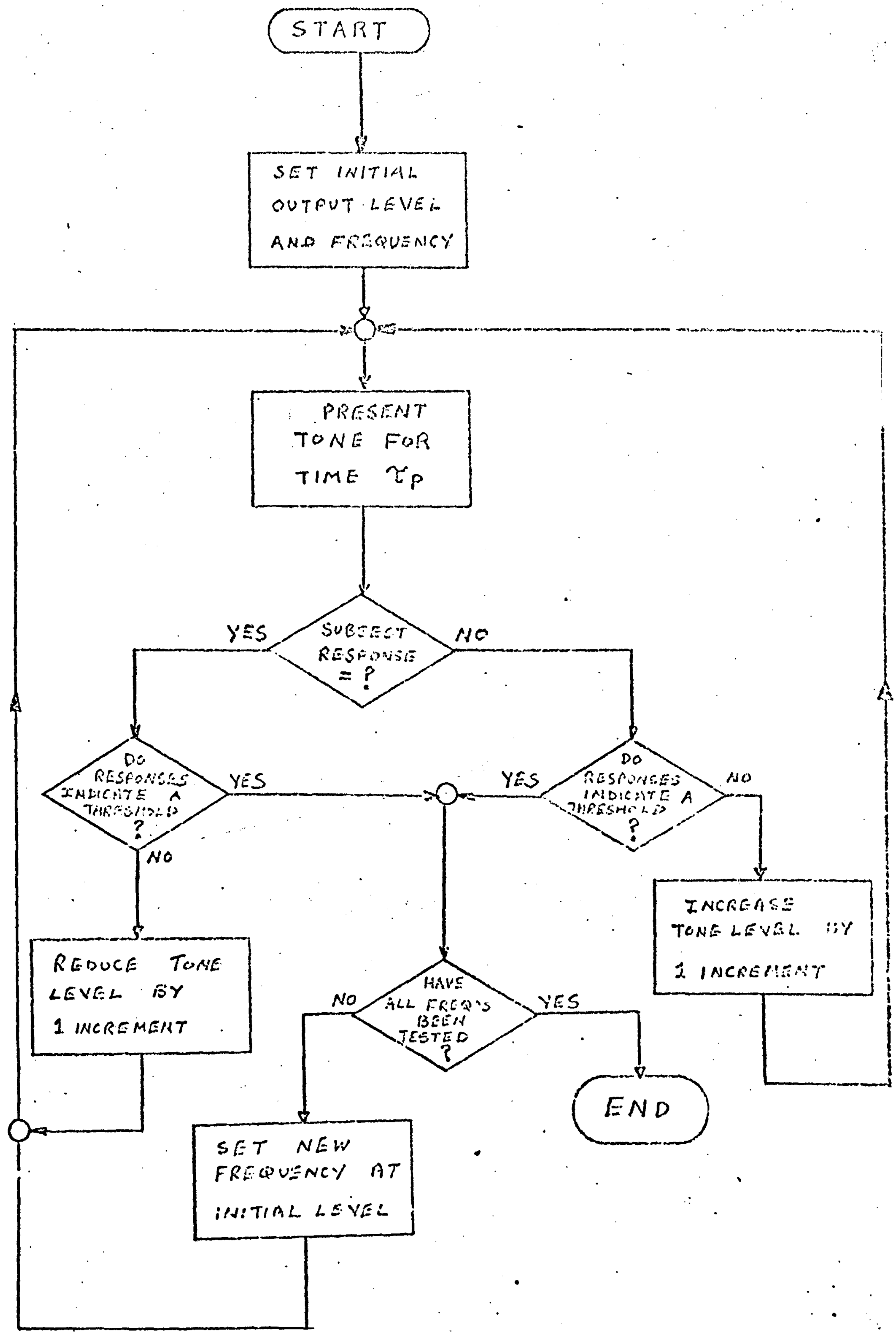


Fig. 3.1 Basic sequence of manual pure tone test

subjects may be assumed to understand what is expected from them and why they are being examined.

3.1.1 Information Contained in the Timing of Individual Responses

The responses made by a subject to test-tones presented to him during a manual pure-tone examination are influenced by many factors in addition to the condition of his hearing. These include a lack of understanding of what he should be doing, anxiety about the outcome of the examination, general discomfort in the testing environment and a possible wish to produce a false result. The experienced examiner learns to recognise many of these factors from the subject's movements, appearance and of course, his response pattern. The advent of self-recording audiometers resulted in many examinations being at least partially non-observed, but careful visual analysis of the resultant graphs can provide evidence of inaccuracies or even provide diagnostic information as in the case of recruitment (26). The following paragraphs will discuss means by which a computer controlled audiometer, using a tone presentation technique, similar to that used in manual pure tone audiometry, may derive useful information from the responses it receives.

The recording of simple YES-NO replies to the presentation of test-tones provides no information concerning the nature of the subject's responses, such as decisiveness, uncertainty and delay.

The manner of each response provides information concerning the certainty of the YES or NO associated with it. It is possible, according to statistical decision theory, to disentangle two distinct components of a response, Swets (27). One represents the subject's decision criterion: whether he is more willing to risk saying yes when he might be wrong than to be sure he is right, or vice-versa. The other is the inherent detectability of the signal in the particular circumstances, based upon a statistical model. The testing time required to isolate the two components, and hence eliminate the effects of the subject's decision criterion, is unfortunately very long and cannot be practised on a regular basis. As a result it must be hoped that a subject will respond in the manner suggested to him by the operator with a reasonable level of consistency. It is the intention of the following paragraphs to discuss means by which responses may be assessed and used to improve the accuracy of the overall test results even though a precise criteria, such as that suggested by Swets is not practical.

Fig. 3.2 shows the sequence of events from the time that the audiometer instigates a tone presentation until its removal. For the purpose of the discussion the final tone level is assumed to be audible to the responding subject, and the subject is assumed to understand the test procedure. Referring to Fig. 3.2, at an unknown time after Ti, but before Ts, the tone level will reach the point at which it becomes audible to the subject. At this point he will make the mental decision to depress the reply button to indicate a YES. As successive test tones are presented at levels approaching the subject's

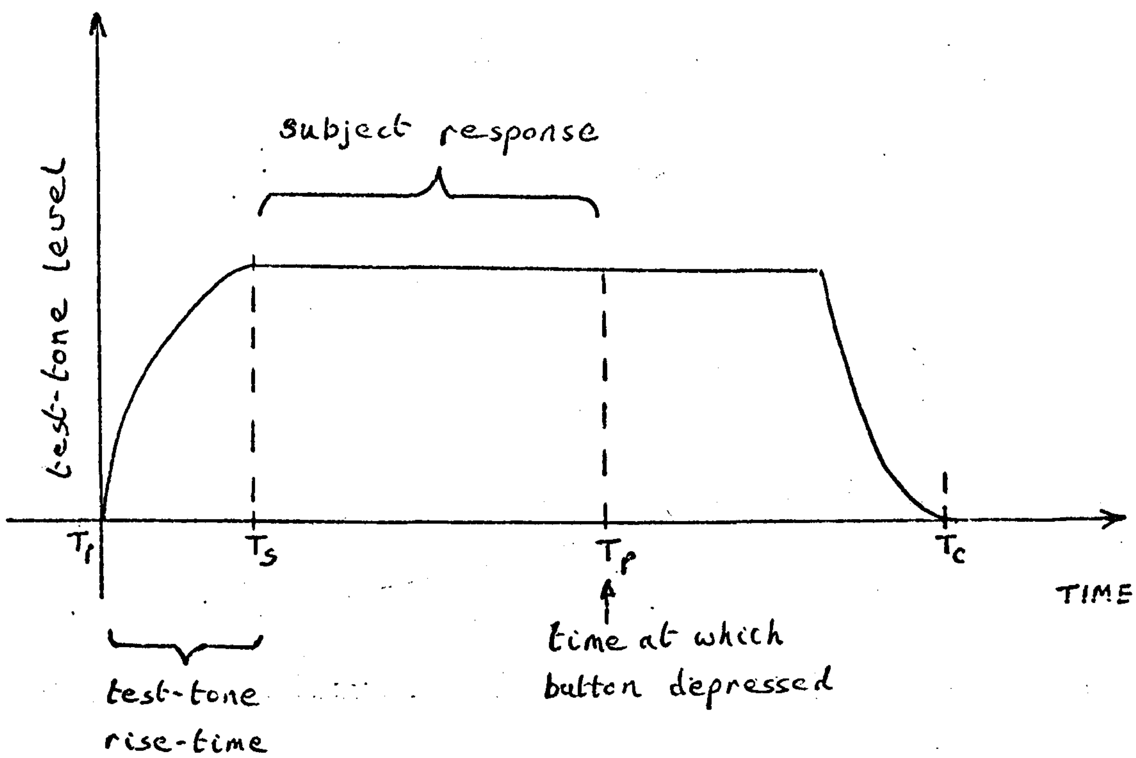


Fig. 3.2 Test-tone response timing.

threshold, his degree of certainty of hearing will also decrease (44), causing increased delays in response time. The length of time taken to make a decision is highly dependent upon the character of the individual concerned. A decisive person may depress the button almost immediately upon mentally detecting the tone, whereas someone more thoughtful may delay before depressing it. Each of these responses is made with a similar degree of certainty by the subject, but in different styles. From this it is clear that it is not universally possible to assess certainty of response by means of an absolute measurement of time from Ti to Tp. It is not even possible to assume a consistency of response from an individual. Throughout an examination the level of concentration and motivation of a subject may vary such that responses to the same test-tone at the same level may appear immediate in one case and delayed and uncertain in another. Thus, comparison of the rates of response does not appear to offer a reliable means of accuracy enhancement. There is one situation in which the rate of response may indicate an erroneous reply. If the response button is pressed so soon after Ti that it is unreasonable to expect that the subject could have responded to the test-tone in that time, then the response must be considered faulty. Another means of considering the nature of a response is by its position in time relative to the removal of the test-tone. If we assume that all test-tones are of sufficient duration to allow any positive response to be made before cessation of the tone (Tc), then any response initiated after that time has a high probability of being indicative of an error in the system. This may also be said of any response initiated before Tc but continued into the silent period by more than a margin to allow for realisation and release of the button. Indeed, should either of these incorrect responses be maintained until the presentation of the next tone, then they may definitely be considered faulty.

Let us now consider what might cause such erroneous responses. If the subject has failed to understand that he should only depress the button whilst he can hear a tone he may keep it depressed throughout the descent in levels to his threshold. This will lead to inaccurate results and thus it is useful to know that it is happening. An alternative may be the onset of tinnitus in the subject or of unwanted background noises. In both cases, the pulsed nature of the test-tone should attract their attention to the correct sounds to be listened for but will not remove the inaccuracy so once again a vital warning has been given. Lastly, a response during the silent period may be the result of someone attempting to produce an audiogram with a lower threshold than his own. Once below the true threshold he can only respond by attempting to keep the test-tone repetition rhythm in his mind and respond at the relevant times. If his timing loses synchronism with the test, which it undoubtedly will, he may respond during the silent periods. Thus, observations of the YES-NO response patterns are as important during the silent period as they are during the test-tone. A third method of considering the timing responses is to study the number of responses made relative to the number of tones. If a subject responds to each tone in a series then the responses will be regularly spaced and synchronous with the test-tones. However, should responses occur more frequently than test-tones there is either an intermittent contact on the response switch or the subject is responding in a manner that means there is doubt about the validity of the replies. If all the responses occur during tonal presentation periods then an explanation may be an uncertainty, resulting in a jittery response. Responses during the silent periods are indicative of possibly erroneous results and as such their detection represents useful information.

The decisions that may be made, recognising the points in the preceding discussion, as a result of sampling the response button ten times during the tone period and a further ten during the silent period are illustrated in Fig. 3.3. The possible errors that can be detected are classified by ER numbers. The twenty samples are stored in a YES or NO form in the computer, labelled TS1 - 10 and SS 1 - 10 as shown in the diagram. Immediately that sample SS 10 has been taken the computer inspects TS 1. If TS 1 is a YES then either the button was already pressed, ER 1, or the subject responded before he could reasonably be expected to have heard the tone, ER 2. In each case it is uncertain what the result should be and so it should be discounted and the tone repeated. If TS 1 is a NO then TS 2 - 10 may be either all NO's or a combination of NO's and YES's. If TS 2 - 10 is all NO's and SS 1 - 10 is the same then a definite NO reply may be accepted. If SS 1 - 10 contains any YES's then it is not certain which of the possibilities they represent, a delayed YES to the tone, a response to some unknown background noise, or an out-of-time attempt to generate a low threshold curve, ER 3. Using the argument that a response delayed by a sufficient length of time to be in the silent

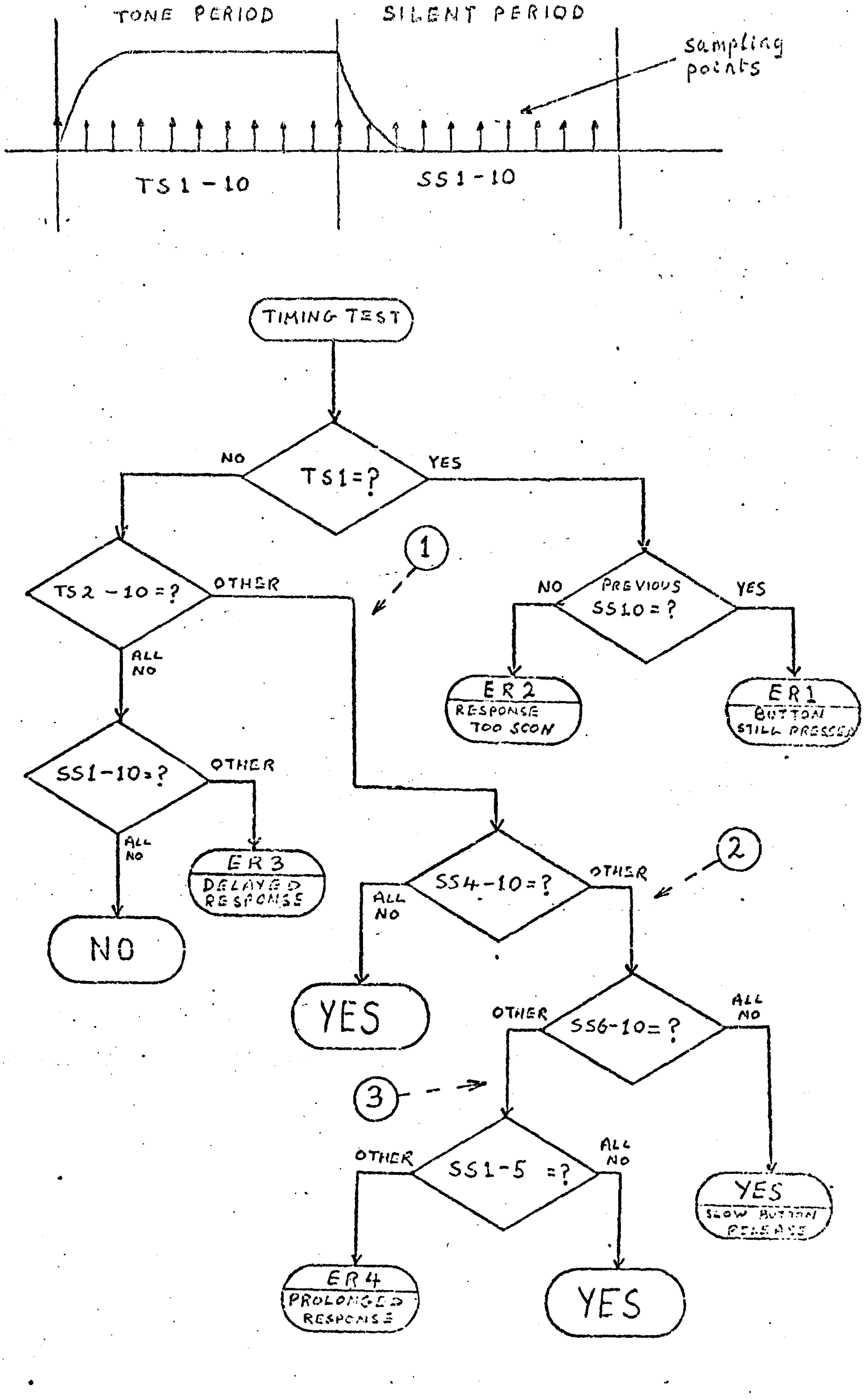


Fig. 3.3 Summary of Response-Timing Analysis

period must be a marginal decision, the result may be recorded as a NO. If either of the second or third possibilities are the case then a NO is a justifiable result on the grounds that it is correcting for background noise or making an attempt at a fraudulent response more difficult.

If the samples TS 2 - 10 contain any YES's, point 1 on Fig. 3.3, then we need to decide whether or not they are valid. To do so we inspect SS 4 - 10. The reason for ignoring SS 1 - 3 in this instance is that a slow release of the button may leave them as YES's. Now if SS 4 - 10 are all NO's then the result of TS 2 - 10 may be recorded as a YES. If, however, any YES's occurred, point 2 in Fig. 3.3 , then the validity of the YES response is questionable. It may mean that the button was released very slowly in which case the YES is correct, but it could indicate a continuously depressed button or responses to unwanted sounds. An analysis of the SS 1 - 10 samples can be made to determine how the result should be recorded. If we consider SS 1 - 10 as two parts SS 1 - 5 and 6 - 10, it is assumed that any delay in releasing the button will have passed before SS 6. Thus, if SS 6 - 10 are all NO's, then the last result may be accepted as a YES with a delayed button release. If SS 6 - 10 are not all NO's, point 3 on Fig. 3.3 , then a further decision must be taken. If all the SS 1 - 5 replies are NO's then we may accept the response made during TS 2 - 10, a YES in this case, but if SS 1 - 5 contain YES's as well as SS 6 - 10 then an error, ER 4, has occurred and no result can be recorded. The cause of ER 4 is most likely to be a continuous depression of the button, either by mistake or due to tinnitus.

In chapter 2 it was explained that regular presentation of test-tones can enable a subject to predict when one is to be presented to him simply by observing the rhythm whilst they are audible and tapping his foot in synchronism. In order to overcome this possibility the test-tone presentations may be made irregular, both in duration and presentation. A simple procedure to achieve this is to define minimum on and off periods for the tone presentations and then add pseudo-random numbers, varying between one and two times the minimum, for example, to each of them. Thus, an off-time of T_{OF} may become a duration between T_{OF} and $3.T_{OF}$, and similarly, the on-time T_{ON} may vary between T_{ON} and $3.T_{ON}$. In this way the computer may avoid the possibility of producing regular, rhythmic presentations.

3.1.2 The Pattern of Responses

The currently accepted methods used in manual audiometry for recognising an individual's threshold at a particular frequency, were described in section 1.4.1. These methods have an advantage over those of self-recording audiometry in the flexibility of their duration. Self-recording units continue testing at each frequency for a preset period of time. At the end of that period there may or may not be sufficient graphical information upon the audiogram to determine a threshold level, whereas in a manual test an audiometrician will continue testing at a given frequency until he is satisfied that he has assessed the threshold level according to the rules laid down for his particular technique. Unfortunately, not all subjects respond in a consistent manner, causing the examination to become protracted while the examiner attempts to observe a reliable pattern or trend. On the other hand, in many cases the period of time taken to determine a threshold at a particular frequency will be less than the fixed period taken by the self-recording unit. It is the intention of the following paragraphs to discuss alternative methods for determining a threshold level at any given frequency, which are suitable for use with a digital computer.

The accepted techniques for clinical determination of the hearing threshold are straightforward and their sequence of events can be readily simulated by a digital computer. The two most popular techniques are those known as the increasing intensity series (IIS) and the decreasing intensity series (DIS), both of which have the advantage of speed and clinically-acceptable accuracy. The term 'series' is the accepted description of a sequence of tone presentations, at the same frequency, changing in level by equal amounts in a single direction. Thus, the increasing intensity series method commences at a level below the threshold of the subject and increments it until he indicates that it is audible (an increasing series). Several increasing intensity series are normally used and the indicated minimum audible levels noted for each. It is usual to continue presenting increasing series until the noted minimum audible levels of more than 50% of four or more consecutive series are consistent, the exact number being decided upon by the examiner. The consistent minimum level is recorded as the threshold. The decreasing intensity series method is similar in all but direction and also uses the minimum audible level as the threshold. A third method used is an adaption of the laboratory 'method of limits' (13). Initially, the tone is presented above the threshold and then decreased in intensity from audibility to inaudibility in 5 dB steps until it can no longer be detected by the subject (descending series). The minimum intensity setting of the audiometer at which the subject could perceive the tone is noted by the examiner. The intensity is then increased from inaudibility to audibility in 5 dB steps until the tone is perceived by the subject (increasing series). The minimum intensity at which the subject could perceive the tone in the IIS is also noted. An equal number of increasing and decreasing series are presented to the subject, the exact number being determined in advance by the supervising audiologist. The threshold of the subject is obtained by averaging the minimum intensity levels at which the tone is heard on the increasing and decreasing intensity series and then calculating the mean of the two results. The difference between the method described and the 'method of limits' is that the latter uses 1 dB steps.

In practice it is often found that the IIS average is as much as 6 dB greater than the DIS average. Had a simple increasing or decreasing system been used, where the average minimum level to which a response is obtained is accepted as the threshold,in a case such as that just described, an error of 3 dB would have been made, assuming that the true threshold is midway between the averages for the two series. Thus, the optimum system would seem to require that minimum hearing level be determined for each direction.

In clinical tests 5 dB is usually the smallest step size used. Considering that in the laboratory technique, 6 dB difference may be found between the two averages, the use of a 5 dB step size may prevent the detection of a difference between the averages. In that case a step size of 3 dB would be necessary. As a result of the smaller step size required to perform the averaging technique and the need to determine the threshold at each frequency by both an increasing and a decreasing intensity series, the duration of the examination will be at least 75% greater than for a single directional series although the results should be to within $\pm$ 2 dB instead of $\pm$ 5 dB. This increased length of test-time could certainly be justified for pre- or post-employment audiograms where the extra precision at the expense of time is a valid exchange. However, in routine re-tests the length of time must be kept to a minimum and a single directional, 5 dB step size, method adopted.

In all manually controlled examinations the examiner must make the decision that the pattern of responses he has obtained conveys sufficient information to indicate the threshold accurately. For such techniques to be implemented by a computer a decision-making system must be devised to simulate that of the examiner. It must also be capable of recognising and identifying irregularities in response patterns which are caused by the subject.

Using a computer to supervise the test methods described above, there are two possible means of determining the point at which to terminate the examination at any given frequency. The first employs a system of fixed test length. A pre-determined number of increasing or decreasing intensity series is made at each frequency and the mean minimum audible level is calculated for each and used as the threshold. The second method requires that the examination is not terminated until at least 50% of consecutive series, in the same direction, display minimum audible levels which are consistent. This method has the advantage of not terminating until a satisfactory result has been obtained, whereas the first may or may not return an accurate threshold, depending upon whether or not the responses were consistent. In any system where the optimisation of accuracy is important the second method is to be preferred.

Let us consider the simplified 'method of limits' system described earlier in this chapter. In this method both increasing and decreasing series of levels are presented alternately. Because minimum audible levels tend to be higher using an increasing intensity series than on decreasing series, a hysterisis effect in the response is evident in many cases. This can be understood by considering the following example. Assuming that the first inaudible level after reducing in 10 dB steps from 30 dB HL is found to be 0 dB HL, and that 5 dB steps are now being used. Fig. 3.4 illustrates the sequence of presented levels and responses clearly showing the hysterisis effect about the 5 dB HL. The form of hysterisis in the results, displayed by the 5 dB HL level, is particularly evident when using step sizes

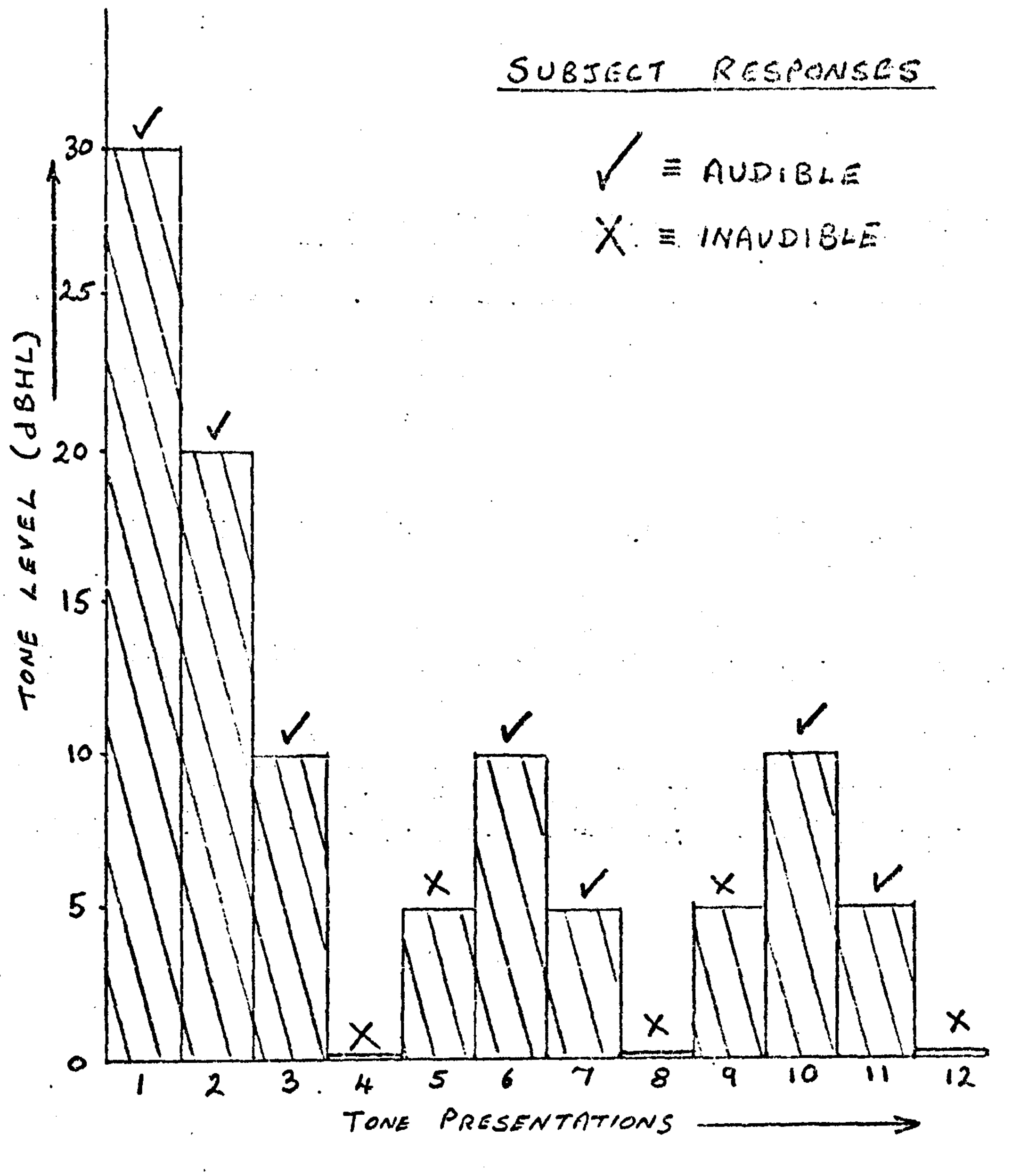


Fig. 3.4. Response pattern displaying hysterisis.

of 5 dB or less. Over a 10 dB increment the apparent loudness difference between two successive tones is sufficient to render the two levels clearly audible or inaudible except when one of the 10 dB stepped levels is situated approximately at the threshold for that frequency. In such a case the subject may display hysterisis over a 20 dB range. When an audiometrician observes the responses during an examination he is constantly looking back to ensure that the apparent threshold is not drifting or that replies are not contradictory. In order to make such assessments he must establish a reference amongst the responses against which to compare the others. This reference is frequently the first indicated non-audible level. Having determined this level the threshold will normally be within $\pm$ 10 dB of this and frequently within $\pm$ 5 dB. If the threshold shows a tendency to be outside those limits, the responses may be considered too erratic to be acceptable unless the tendency is substantiated by consistency. For the same reasons a reference against which to compare the progress of the examination is required by the computer. The system described whereby the result should be within $\pm$ 10 dB of the first negative reply level could be used but instead let us consider the following adaptive reference approach which is shown diagrammatically in Fig. 3.5.

The test commences at 30 dB HL which should be audible. Assuming it is, then the level is reduced in 10 dB steps until the first negative reply at, for example, 0 dB HL occurs. Next, instead of changing to 5 dB steps, go back to 10 dB HL, if audible reduce to 0 dB HL, if not, step to 20 dB HL. Whichever the case, the intention

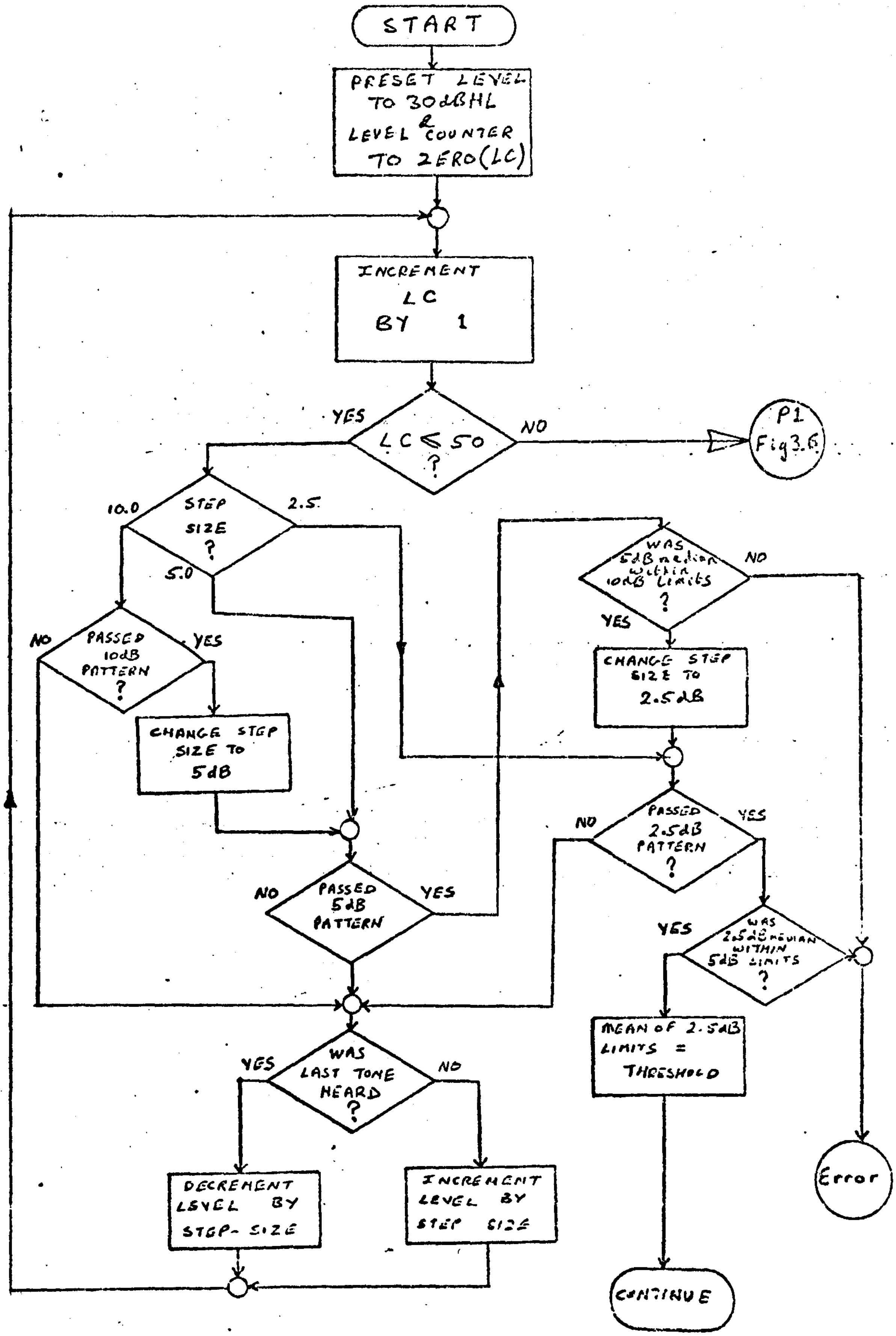


Fig. 3.5 The basic decision system

is to produce a pattern of replies such that ten consecutive test-levels do not vary by more than two step sizes. In the case of 10 dB steps, this requires that for ten consecutive tone presentations the tone-levels remain within 20 dB of each other. The choice of two step sizes is made to ensure consistency of responses whilst allowing a margin for variation. It is quite possible that a YES-NO-YES-NO response varying by no more than 10 dB might be achieved in many cases using a 10 dB step size, but the possibility of hysterisis must not be overlooked as it displays an equal level of consistency.

The number of consecutive levels required to remain within the 2 step limit is chosen to allow a subject displaying a hysterisis level, the opportunity to complete two full response cycles and show evidence of commencing a third. For example, the test-levels, in dB HL, might be similar to those shown in Fig. 3.4, showing two cycles and the start of a possible third in ten levels. Once this acceptance pattern has been detected, then it may be said with reasonable confidence that the threshold is between the level limits of the pattern. Taking the lower limit as a starting point the steps are changed to 5 dB. The same pattern of responses is looked for and when detected the step-size modified again to 2.5 dB based upon the new lower limit. At the detection of the pattern at both the 5 and 2.5 dB step sizes, the median of the pattern level limits is calculated to ensure that it is within the limits set by the extremes of the preceding cycle. The threshold is taken to be the median of the 2.5 dB cycle. Thus, if the extremes of the 5 dB cycle are 10 and 15 dB HL and the median of the 2.5 dB cycle equals 15 dB this means that although the 2.5 dB cycle must have been between 12.5 and 17.5 dB HL, displaying hysterisis at 15 dB HL, the median level is within limits and as such, acceptable. The above example is an extreme case and it is very unlikely that a threshold would eventually be found on a limit of a previous cycle as that limit would probably have been a hysterisis point as a result of it's marginally audible level. However, the purpose of the example is to demonstrate both the general operation of the system and the fact that only the median of the new limits need be within those of the pattern preceding it. The system described is straightforward to implement on a digital computer and is sensitive to drifts of the apparent threshold. The system also has the advantage of ignoring random fluctuations of the test level due to temporary lack of concentration. Should a consistent drift of threshold be detected, of such a magnitude as to be outside the probability of it being a learning trend then the computer can indicate the possibility of malingering. Adaptations to this technique for the purpose of improving its immunity to malingering will be discussed in a later paragraph.

A disadvantage of this technique is its inability to produce a threshold figure unless the acceptance pattern is detected. Subjects suffering from tinnitus or of an indecisive nature may produce erratic response patterns. The effect of insisting upon the detection of this pattern would be to prolong many examinations until it occurred, possibly only by chance. For cases such as these it is necessary to include a mechanism which allows them to be identified and to make optimum use of their responses to investigate whether a result may be computed from them, see Fig. 3.6. In the event of a set of responses becoming random to the extent that no sensible result can be extracted then a warning must be raised to call the attention of an operator. The identification of a subject not conforming to the acceptance pattern may be achieved by imposing a test-tone sound upon that section of the programme responsible for determining test levels. If a count is kept of the number of test-tones presented at each frequency, the examination may continue as normal until the count exceeds a figure considered to be reasonable for an average subject to have achieved both the 5 and 2.5 dB pattern. The count limit is a figure requiring experimental determination. Too large a number may unnecessarily elongate the test and too small a number cause more subjects to be rejected as random then necessary. When the count limit is exceeded, then the responses received up to that time are analysed to determine their usefulness. If the acceptance pattern was achieved at 10 dB steps then the limits may be used as a reference by which to assess the later points unless the 10 dB pattern wa snot achieved until within four test levels before the count limit. In such a case its median is the only estimate of the threshold available and the result cannot be considered reliable. The figure of four test levels before the count limit is used as this is the least number of levels capable of producing a response pattern with evidence of consistency within 10 dB. When more than four levels occur after the 10 dB acceptance pattern then either all the remaining levels will be at 5 dB steps or possibly 2.5 dB steps if the 5 dB pattern was also achieved. It is necessary to determine which step size the test had advanced to when terminated in order that the appropriate action may be taken as described below.

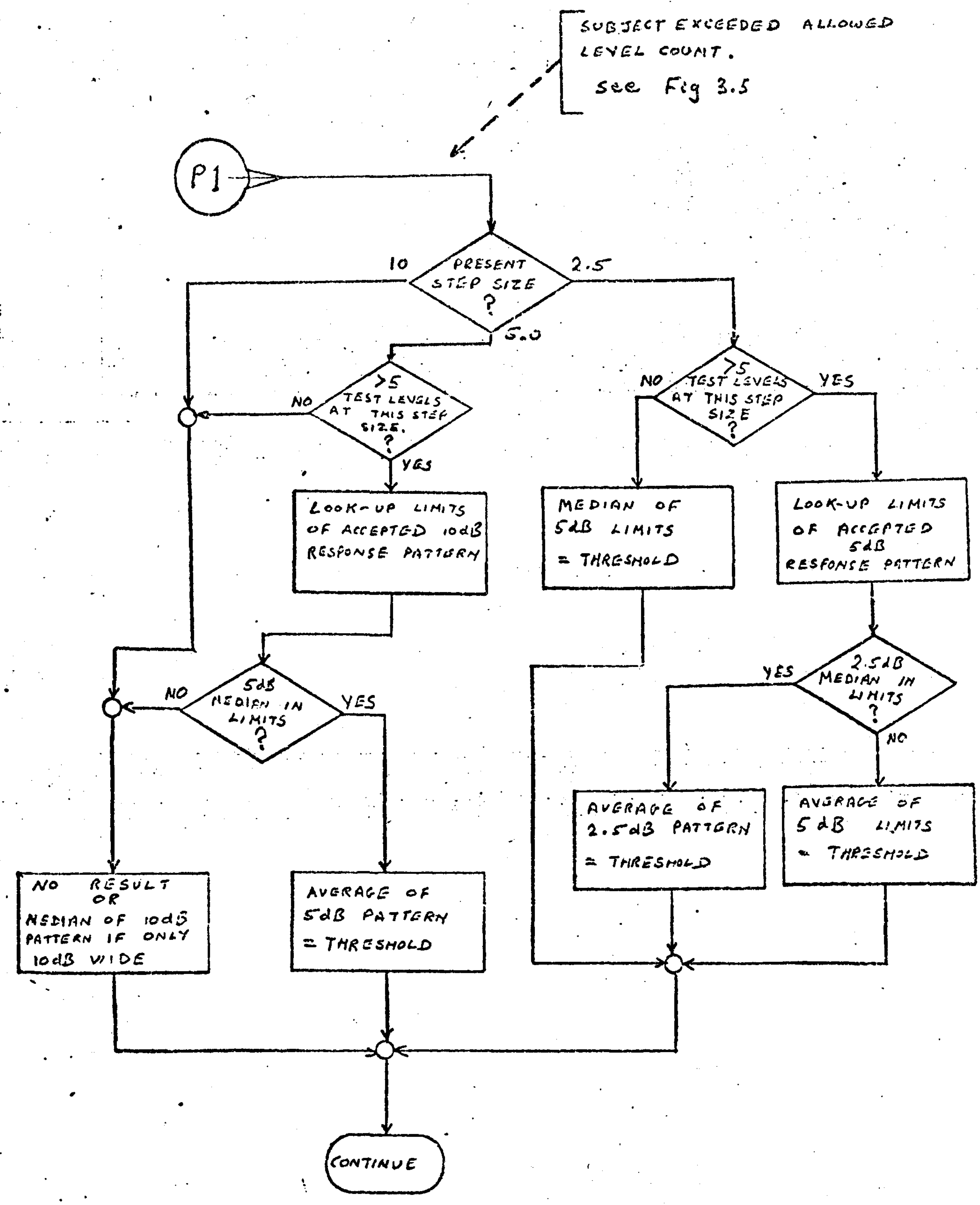


Fig.3.6 Procedure when level-count limit is exceeded.

a) If the step size is 5 dB then obtain the limits of the 10 dB pattern, ensure that all 5 dB levels are within those limits and if they are take the mean 5 dB figure as the threshold. It would also be possible to allow 10 or 20% of the 5 dB levels to be outside the limits, particularly if the limits were only 10 dB apart. The smaller this allowance is made the more precise will be the result but also the number of non-decisive tests will increase. If the 5 dB levels are outside the limits set by the 10 dB steps and do not satisfy any percentage allowance either then the test must be declared void. At best if the 10 dB pattern limits were only 10 dB apart an estimate of the mean level could be made. The inclusion of such a decision in a system would be at the discretion of the audiologist planning it.

b) Where the step size is 2.5 dB, a similar test may be made on the 2.5 dB levels relative to the 5 dB pattern limits. In this case, a failure of the 2.5 dB levels to satisfy the limits criteria may be counteracted by using the mean 5 dB pattern level as the threshold.

In both case a) and b) accuracy relative to a successfully completed examination is sacrificed so that a result may be produced that is at least within $\pm$ 5 dB.

The basic system of response pattern identification has now been derived. The following paragraphs are concerned with the possibility of adding extra pattern criteria for the identification of irregularities in the test procedure. It has already been explained in the preceding text that an erratic response pattern will cause the level counter to bring into action a series of operations designed to extract any valid information from the responses obtained. Unlike the erratic response which is identified by its lack of a pattern, several irregularities may be recognised by distinctive patterns. The subject who has not understood, or has forgotten, that he is required to press the response button whenever he can hear a tone in his earphones will ultimately receive the maximum output level from the audiometer as it increments the level by 10 dB at each presentation. At this condition the computer must decide whether the subject has very seriously impaired hearing or is simply failing to respond. Two alternatives exist. Either the computer can call the attention of an operator or it can change the test frequency and repeat the test. If impaired hearing is the true cause then a different frequency may substantiate this by causing the subject to demonstrate that he knows how to respond, unless the threshold at that frequency is also outside the range of the audiometer. In such a case, calling the attention of the operator is probably the most advisable action as lack of understanding is a more likely cause. A similar situation may occur at the lowest level of the audiometer with a subject possessing an exceptionally low threshold or with one responding to tinnitus, or someone permanently depressing the button. Once again, calling the operator's attention would seem to be the optimum solution.

In section 2.2.4 it was described how an individual attempting to malinger may lose his concentration and show an apparent threshold shift at a particular frequency whilst under examination. It would therefore appear reasonable to be able to detect such people by a suitable analysis of the test responses. Unfortunately, the detection of individuals attempting to malinger is a difficult task, for although an indicative response pattern may be expected from them if they lose their concentration, it may not if they maintain it. It is possible for a computer to make special purpose tests to detect malingerers but only by extending the examination duration. Two straightforward techniques may be used which are standard audiometric practice. The first is to elevate the test tone to 20 or 30 dB above what appears to be the threshold for that tone. If the subject's responses indicate that the original threshold was correct then either he is not malingering or he is and is well practiced. The second technique involves repeating either the entire test or at least a half of it and comparing the two sets of results. Changes in threshold of more than 10 dB at a particular frequency should be treated with suspicion and the subject referred for a retest. Each of these techniques may be performed with a computer system but would add to the duration of the examination. A means by which the extra test time could be avoided in many cases is as follows. During the conversation between the operator and the subject prior to an examination, it may be evident to the operator that the individual is not consistent in his behaviour. He may hear clearly what is being said to him quietly at one instant and not at another. The person with a true hearing loss will usually be concerned and show it, whereas a malingerer is often more nonchalant and may even say something that will reveal a motive for his wishing to simulate a hearing loss. If an operator suspects for any reason that a particular person is a malingerer then he may put into action routines which make a more extensive examination, paying particular attention to the possibility of malingering, provided of course, that such routines have been incorporated into the system. If however, no indication is given prior to the test, then the only possible opportunity for detection is a lapse of concentration. A consideration of the response patterns may reveal such a lapse. In the method described earlier in this section for determining the threshold level at a frequency by recognising patterns at 10, 5 and 2.5 dB step sizes the median of each pattern was compared with the limits of that previous to it. If the median was within the limits then the response pattern was accepted as consistent. The method determined limits about the threshold three times with different step sizes, a more detailed analysis of the patterns may reveal a tendency for the threshold to shift during the test period for a particular tone. Firstly, consider the 10 dB step size pattern, this will be either 10 or 20 dB between limits. If it is 10 dB then it may only be estimated that the threshold is somewhere between the two levels forming the pattern and probably near the middle. However, if the threshold were near one of the levels, for example, within 3 dB, then quite possibly a three level pattern would result, due to the response at the middle level being dependant upon whether it was approached from above or below the threshold being near the middle hysterisis level. It is most likely that in a case such as this the true threshold is below the middle level. Thus, if the median of the 5 dB step size pattern is found to be more than 5 dB away from this level, in either direction, a threshold shift may be suspected. A similar comparison between the median of the 2.5 dB step pattern and both the 10 and 5 dB patterns may confirm this suspicion. The apparent movement of a threshold is not conclusive evidence of malingering. It is known that a learning effect does exist such that the threshold effectively becomes lower as a test continues, but nevertheless, the comparisons suggested above could be used to cause the computer to perform the malingerer-detection tests described previously.

3.1.3 The Influence of Response Analysis upon Test Duration

The techniques described above are all intended to improve the overall accuracy and reliability of pure tone examination. Unfortunately, the inclusion of many of the suggested procedures implies that the duration of an individual examination may be considerably longer than the six minutes attributed to self-recording audiometer. However, it is most unusual for a self-recorded examination to last much less than ten minutes. The reason for this time difference is the necessity for the operator to explain the procedure to the subject and allow him to practice with the equipment before commencing the true test. Also, if the operator realises that the subject is failing to respond correctly during a test then he must reset the audiometer to whichever frequency the subject began to falter at. Thus the true duration of a self-recording audiometric examination may be greatly extended by the need for operator intervention. The duration of a computer controlled test, using the techniques described previously, may similarly be extended but by computer intervention. For example, if the computer is programmed to attempt to recognise malingerers and modify its procedure if a subject causes it to suspect him, then the test time will be extended due to the extra tests the computer will perform. Similarly, if an individual repeatedly loses concentration then the computer will repeatedly call the operator to attend. However, each of the elongations mentioned should result in reliable measurements at the end of the examination. Using the techniques described, the shortest period of time that an examination upon both ears can be fully performed in, is approximately three minutes. This may be achieved with the basic test time duration and inter-tone duration set to 0.5 seconds. Thus both times will vary between 0.5 and 1.5 seconds. The subject is also required to respond in a consistent YES-NO manner as described in section 3.1.2. However, the 3 minutes does assume that the 2.5 dB increments are used, which if discorded reduces the time to 2½ minutes. It is quite possible for a subject to respond in such a consistent manner although even allowing for inconsistencies of response, it may be seen that a test time of 6 minutes is quite reasonable to expect. It is not known how short a test may be made using these methods. The durations of the tone and silent periods are the ultimate limiting factors. How short these two periods may be made before the timing analysis becomes invalid requires experiment although it is unlikely that the above times can be improved upon by more than a few seconds. With tone to tone times as low as one second, button depression response times can become comparable in magnitude resulting in regular responses appearing in the silent period and possibly lasting into the next tone presentation. It is necessary for the system to detect subjects with slow responses, using the methods described in Fig. 3.2, and either request the operator to modify the basic tone and inter-tone periods or modify them itself until the responses may be clearly understood in relation to their stimulating tones. In this way the audiometer could optimise its tone presentation period to suit the individual.

3.2 Computer Assisted Calibration

The correct calibration of an audiometer, at least as far as the manufacturer's instructions and design will allow, is the responsibility of the operator. But precise calibration can be performed only by competent people using specialised equipment and so, it is not normally recommended that an operator should be responsible for it. It is becoming increasingly common practice in the design of complex measuring instruments such as those used in the telecommunications field, to incorporate automatic self-calibration, using what are effectively small computer circuits. For example, if an instrument is to be used to measure the frequency of electrical signals, then one or a number of precise reference frequencies will be generated internally and the instrument will measure them at regular intervals, compare its results with stored correct answers and if an error is present, adjust its measurement circuitry until it is able to measure the standard correctly. If an error is detected of such magnitude that a correction cannot be made then an 'out-of-calibration' warning will be given to the user. In this way, the accuracy of the instrument may be maximised without any requirement for the operator to perform a calibration routine. Once in six months or a year the reference frequencies must be compared with standards by a qualified person but that should be the only manual calibration necessary. In a similar manner the electronics circuitry of an audiometer may be self-calibrated to ensure its accuracy of operation.

There are two variables of particular importance in the calibration of an audiometer, the frequency of the test tones and the sound pressure level presented to the eardrum. To understand how calibration of these two variables may be automated an explanation of the conventional technique and its equipment is necessary. The method used is not complicated but it requires careful use of precision measuring equipment: an audio-frequency signal analyser capable of receiving an input signal from a microphone, normally a one-inch diameter condensor microphone, and an earphone coupler (often referred to as an artificial ear), which complies with BS 4469 and BS 2497, mounted in a test-jig designed to correctly interface with the earphone type in use. The earphones are connected to the audiometer output in the usual manner. The earphones are tested by placing each of them in turn upon the earphone coupler, without removing the cushion, and applying a force of 4-5 Newtons (0.4 - 0.5 kg_f) in a direction coaxial to both the coupler and the phone such that the earphone is held firmly in place against the coupler. Test-jigs are normally either preset to apply this force or have an adjustable system with a scale to set the required force. Once in place the combination represents the standardised acoustic equivalent of an earphone correctly mounted upon an ear under test, and where the microphone diaphragm is approximately at the location at which the eardrum would be. There are many different patterns of earphones each having different dimensions. Thus when coupled to the same standard coupler the enclosed volume of air will vary in size and shape such that identical excitations of the air column at the earphone will cause different sound pressure levels at the microphone, the variations being frequency dependent. To compensate for this variation in earphone pattern BS 2497 parts 1 and 2 contain tables of sound pressure levels (SPL's) which are the equivalent of the reference equivalent threshold SPL's when measured at the microphone in the standard coupler. Eight of the most common earphone types are covered by the two parts of BS 2497. Having determined the earphone pattern in use the audiometer is set to each of the frequencies listed in the standard at an output level of either 0 dB HL, to be directly equivalent to the figures of the standard, or at any other fixed level. The sound pressure levels are measured by audio frequency signal analyser and have observed for both the accuracy of their frequency and of their level when compared with the BS 2497 tables. In this way each of the earphones is tested and the audiometer adjusted internally to correct for any detected lack of accuracy. It may be noted from the above description that the apparatus used for the calibration is all designed to be general purpose and thus of use with a large number of different combinations and types of audiometer and earphone. If this apparatus were to be designed in a dedicated manner to suit only one audiometer and earphone pattern then its size and complexity could be greatly reduced to the extend that it would be feasible to include it within the audiometer itself. A computer controlled autiometer is ideally suited to operating this type of measuring system. Let us now examine how a computer may simplify and control an audiometer calibration system suitable for maintaining the instrument within predetermined accuracy limits with a minimum of operator effort. Fig. 3.7 shows an outline diagram of such a system. The operator is required to place correctly both of the earphones in turn over a coupler which may be fitted with a simple clamping mechanism to hold the earphone in place at the required pressure. Two buttons are supplied for the operator to press, the first to indicate that the right earphone is under test and the second to indicate the left. Depression of either button sets the computer into an automatic measuring cycle. Firstly, the oscillator is set to its lowest frequency and the accuracy of that frequency checked as follows. The oscillator sine wave output is fed into a high gain amplifier such that its output is approximately a square wave, or to be more correct, a heavily clipped sine wave. This output is digital in form and may be recognised as a string of '1''s and '0''s by a computer digital input. The highest frequency output of the audiometer is likely to be 10,000 Hz which represents a '1' input to the computer every 100 microseconds. Modern microcomputers are quite capable of reading digital inputs at many times that rate, thus by programming the computer to read the input for a pre-determined period of time, for example one second, a count of the number of '1''s may be kept and the frequency calculated to within the accuracy of the computer timing network which may be crystal controlled and so better than .01%. This percentage represents an error of 1 Hz in 10 kHz, the longer the count period the greater the accuracy. The frequency measured in this way is compared with the demanded frequency and if different by an amount greater than pre-determined limits may either give a warning to the operator or store a correction factor in non-volatile memory in the computer in

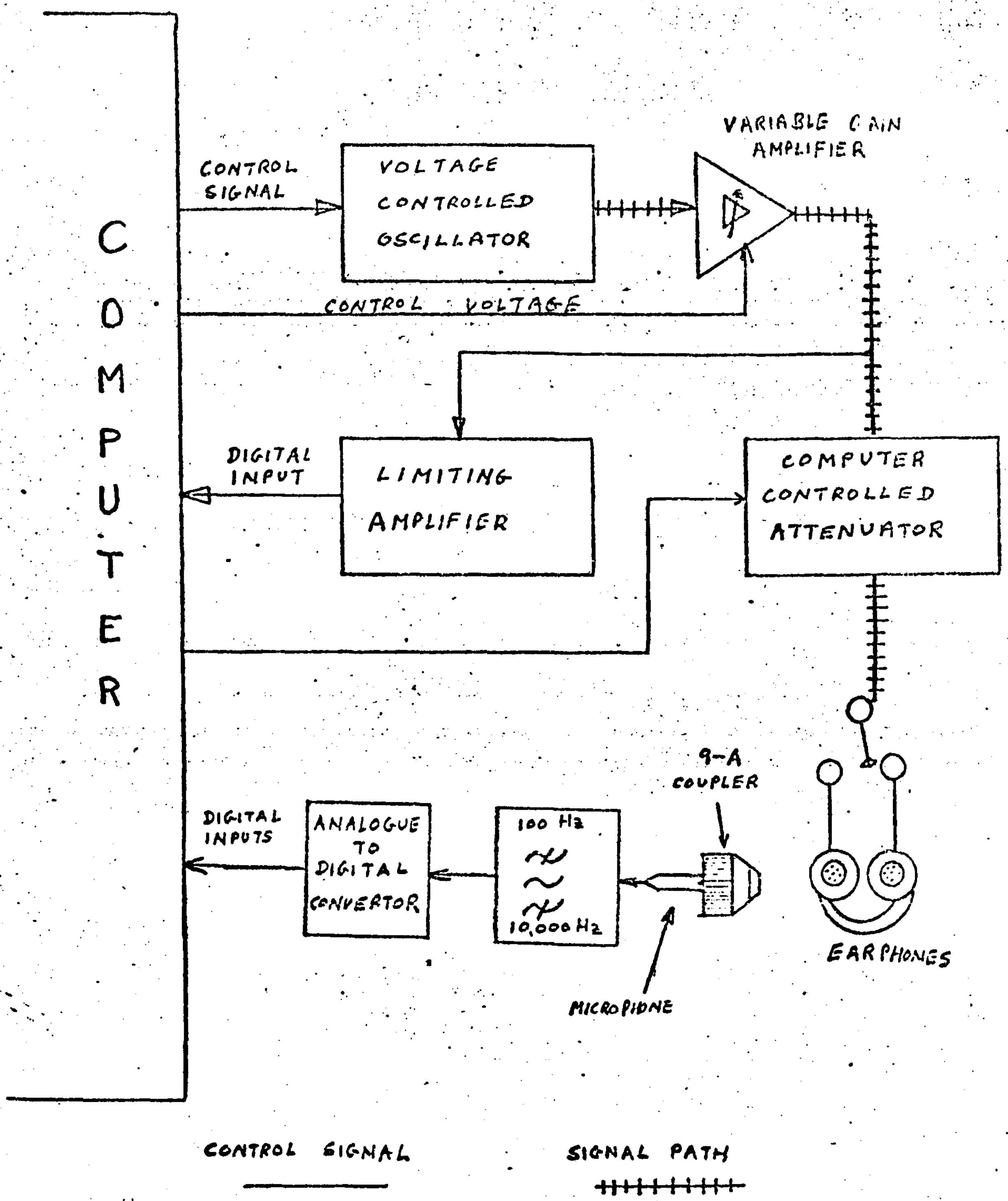


Fig. 3.7 Automatic calibration system.

order to prevent it being lost even if the computer is turned off and then on again. A frequency check of this type could be made without the coupler and earphone being involved, perhaps as an automatic check at the start of each test. Once the accuracy of the frequency has been established then the sound pressure level may be measured. A microphone with known amplitude versus frequency characteristics receives the acoustic signal from the earphone and passes it through an amplifier and filter capable of passing signals from 100 Hz to 10 kHz but excluding all signals outside those limits. The output of the filter is passed through an analogue to digital convertor which transforms the analogue level into a digital number. The convertor is arranged such that a pre-determined output level at each frequency will give a middle of range output, thus any received variation from that digital number will reflect the degree of error. The computer has a table giving an output level for each frequency that is equivalent in the sound pressure level it will produce. The characteristic of the microphone is included in the table to compensate for its non-linearities. In this way the need for an expensive 'flat' response microphone is removed although the lower cost unit would probably be less sensitive and require to be used at higher levels than 0 dB HL for the test. Errors in the received level may be due to several factors such as the initial oscillator output varying or the earphones changing in characteristic as a result of being dropped. Whatever the reason for the error, it may be corrected by modifying the output level from the oscillator. Thus, if a variable gain amplifier, for example, from 0.8 to 1.2, is placed after the oscillator, then for each frequency an error may be stored internally and a correction applied each time that frequency is used. A test such as this could be based upon the voltage applied to the earphones instead of the sound pressure level and run automatically before each test together with the frequency test. A measurement of this type removes the possibility of detecting irregularities in the mechano-acoustic coupling of the earphones but nevertheless, if the test were performed automatically and regularly without operator intervention of any sort, the reliability of the accuracy of the audiometer would be greatly improved.

It is possible to envisage a computer controlled audiometer overcoming the problems of differing earphone patterns by using a pressure probe tube suitably fitted to the earphones such that the computer could monitor the true sound pressure level as the test is performed. However, the extra accuracy to be obtained from such a system could only be justified in a laboratory environment where extreme precision is required.

3.3 Record Handling, Storage and Analysis

The preceding sections have shown that a computer may be used to improve the reliability of the results obtained in a minimally-supervised pure-tone examination. However, another large problem which an industrial audiometric unit has to handle is that of using the results obtained to detect progressive loss of hearing during the period between succeeding audiograms. This problem is in turn linked to another, the storage of the new and past test results. Briefly, the task which the computer is required to perform in an ideal situation is as follows.

At the time of initial employment, store the precise audiogram obtained using the precision audiometer system. At each follow-up test thereafter, compare the new results with those of the pre-employment test and check that they are within pre-determined limits of them. Store the new results together with the pre-employment set with the date of the examination and, if the level at any frequency is found to be out of limits, inform the operator in order that a referral may be arranged. If the audiometer is controlled by a large computer system capable of storing past records of many hundreds of people, then the pre-employment audiogram will automatically be present and available for recall in order that the comparison may be made immediately. In practice the audiometer would probably be controlled by its own small computer and the results stored permanently either on a large computer or on magnetic tape. In order that the audiometer can make immediate comparisons upon a subject's test results, the pre-employment reference audiogram must be available in the instrument. In the absence of the large memory mentioned above the following system may be used. At the start of a day a list of all employees to be examined is fed into the large computer and the past records of all of them written onto a magnetic tape cassette indexed by the employee's works numbers. This cassette can then be loaded onto a suitable reader on the audiometer to allow access to the data, the employee's works number having been typed into the audiometer at the start of the test. At the end of each test the new results are written onto the tape and at the end of the day all the results fed back into the storage computer. The audiometer should also be capable of commencing a new file on the cassette for the results of pre-employment examinations where no previous file exists.

Where sufficient past test results are stored, it is possible to produce trend curves. In this way tendencies towards a hearing loss of greater than the allowable limits set, at which a referral takes place, may be observed and action taken upon them if considered necessary. An example of this type of trend prediction is the exponentially weighted moving average (33). This technique weighs each result used to compute the moving average such that the latest is given the most importance and the oldest, the least; it also effectively smoothes out variations in the results to lessen the effects of random fluctuations. This smoothing is particularly important as an individual may perform an examination twice in the same day and produce apparently correct results 10 dB apart. To use this technique at least six past records are necessary (34). There are many other methods of predicting trends which could be used in a similar manner, but all require a reasonable number of past records which will not be available for any individual until he has been examined over a period of years.

A pattern which can be recognised from looking back at the past records is that of a sudden apparent deterioration or improvement in the hearing such as might possibly be caused by a malingerer. The loss would cause the subject to be indicated for a referral under the usual mechanism, but the added information of the possibility of malingering would assist the referral examiner.

CHAPTER 4

A DESCRIPTION OF THE COMPUTER AUDIOMETER

4.1 INTRODUCTION

In order to evaluate the techniques proposed for improvements in audiometric measurements, a computer-controlled audiometer was constructed. The computer programme and its associated peripheral equipment represents a very large part of the time and effort expended upon this research. However, the system in itself does not directly contribute to this thesis as it is a means to an end rather than an end in itself. As a result, it will be described in outline only and no attempt made to analyse it in detail except where necessary to the understanding of its operation. At the time of commencement of this research the Inter-University Institute of Engineering Control (IUIEC) had recently installed a Rank Xerox Sigma 5 computer in the Department for use in automatic control experiments and was thus ideally suited for the purpose. The standard programming languages were FORTRAN 1V-H for high level and SYMBOL for low level. Data storage was possible by means of both magnetic tape and a fixed Rapid Access Disc (RAD) system. A high-speed lineprinter was provided for printed output. A block diagram showing the basic computer system may be seen in Appendix A. The computer was linked to remote laboratories with three cables. The first contained data lines for linking a standard teletype or visual display unit (VDU) to the computer for the purpose of program control and general interaction. The second contained ten twisted-pairs of wires from a patchboard which enabled any pair to be connected to any of the computer input or output lines. The input and output lines were of five types; analogue input and output, digital input and output and interrupt input. Analogue to digital and digital to analogue converters together with interrupt prioritisers were situated in a Sigma 5 peripheral cabinet. The third cable provided an intercom system between the computer room and the remote laboratory.

4.2 A Description of the System Operation

The computer-controlled audiometer system is shown in simplified form in Fig. 4.1. It may be thought of as three interdependent systems, one internal to the computer in the form of the controlling programme or 'software', secondly the record-keeping system comprised of the magnetic-tape and disc-files, and thirdly the external equipment and circuitry or 'hardware'. The following description is of the overall system operation as initially constructed before any modifications were made for the purpose of experimentation. The description will also serve as a general explanation of the programme operation. All information transferred between the operator and the computer is performed by means of a Tektronix, graphics, visual display unit.

Each examination commences with the writing of brief instructions to the operator on the VDU. These are:

a) How to terminate replies to questions from the computer at the keyboard.

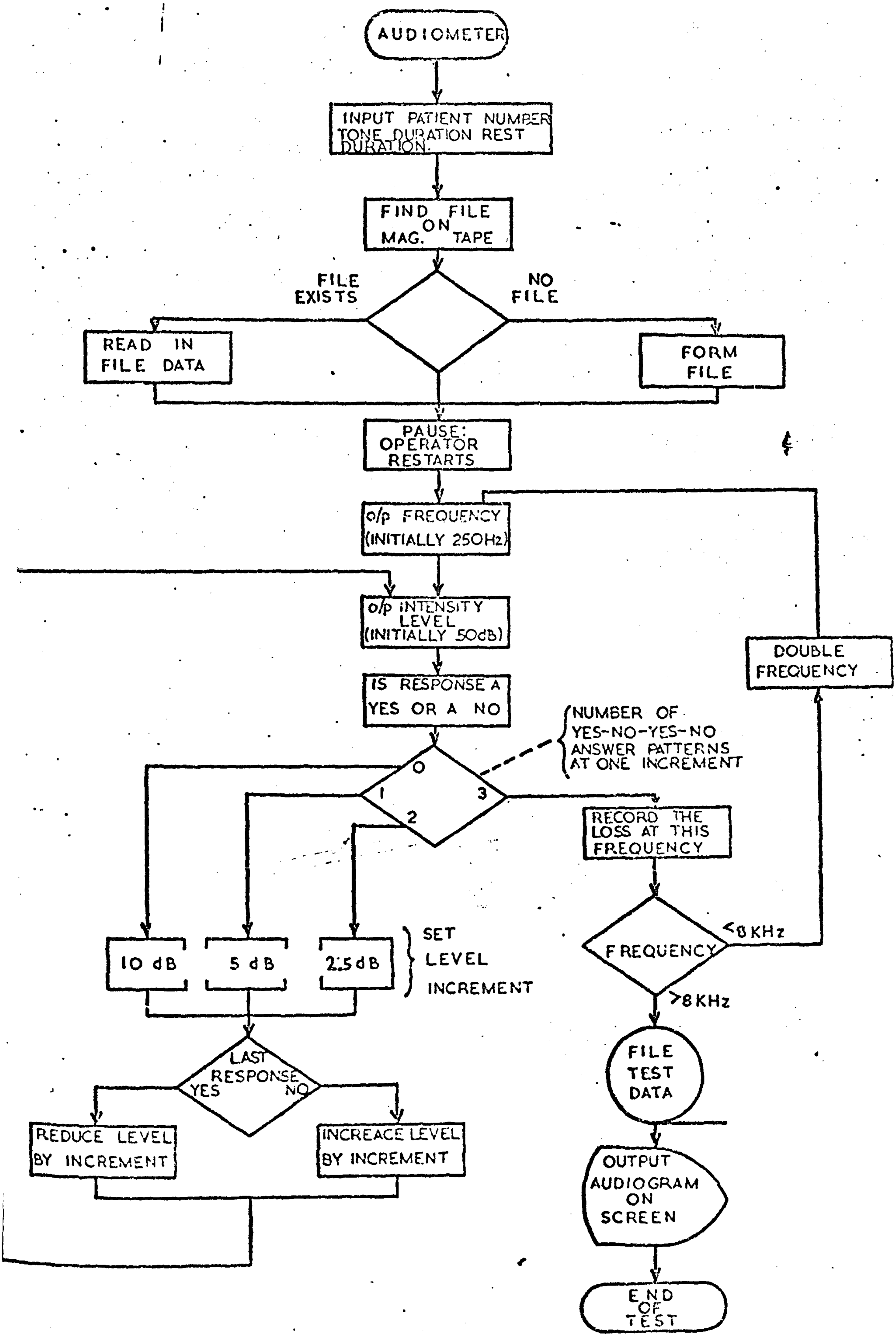


Fig. 4.1 Basic sequence flow-chart of computer audiometer.

b) A reminder to check all system interconnection cables.

c) A list of calibration data. The data includes a set of six sound-pressure-levels, one for each of six listed frequencies, which should be measured on a sound level meter, and a list of voltage levels used to select the six listed frequencies from the programmable oscillator.

d) The time of day and the date.

e) A request to type the reference number of the next examinee on the keyboard.

Next, the operator is requested to indicate whether a manually overridden or fully automatic examination is required. For the purposes of this description it is assumed that an automatic examination is selected. The final request is for the tonal and intertonal time periods to be defined in milliseconds. The intertonal period is modified each time it is used as shown in the equation below

$$\tau_{RA} = \tau_{RD} + x.\tau_{RD}$$

where,

τ_{RA} = actual inter-tonal period

τ_{RD} = defined inter-tonal period

x = pseudo randomly generated number between 0.1 and 1.0.

In this way a rhythmic presentation of the tones is prevented so making the task of malingering more difficult.

The program then looks up the subject reference number in a table, stored at the beginning of the filing tape, to determine whether a file already exists or not. If a file does exist then the test continues, if not then one is formed and the table updated. The formation of a new file is indicated on the VDU screen to the operator, together with a request to type GO to enable the test to commence or NO to restart from the beginning. The right ear is examined first. The initial frequency, 250 Hz, is selected and the level adjusted by the controlled attenuator. The 'silent' switch selects the tone for the pre-determined interval. The initial level is set at 50 dB HL. This should normally be above the subject's threshold. The sound pressure level is written on the screen. During the presentation of the tone the subject's response button is sampled ten times. On this system, only the last response is considered. The intention of the system is to analyse the ten for characteristics such as uncertainty or purpose falsification. A positive response causes a reduction in the level by 10 dB and the tone is presented again. A negative response would cause an increase of 10 dB. The level continues to reduce by 10 dB until a negative reply is received. The level then increases by 10 dB until a positive reply. This process continues until a YES-NO-YES-NO pattern is achieved. At this point the level commences to increment or decrement by 5 dB each time and the process is repeated. A check is made that the average of the 5 dB excursions is within the limits of the 10 dB excursions. Once a regular 5 dB pattern is achieved the same sequence of events is repeated at 3 dB increments. Finally, the

3 dB excursion average is taken as the threshold, written on the screen and filed on magnetic tape. If a subject fails to produce a threshold figure after 50 tone presentations, then that frequency is noted on the VDU and the test continues at the next tone. The frequency is then altered to 500 Hz and the procedure repeated. In the same manner, the test continues up to 8 kHz. Finally, all twelve threshold figures are stored together with the date of the examination on magnetic tape. The end of the examination is indicated to the operator by a computer message on the VDU. At any time during the examination the operator can abort the test by pressing a button. Similarly, the test may be restarted by means of another button. At the end of the examination a third button enables the operator to call for the audiogram to be plotted on the VDU screen. Four examples of audiograms produced after examination on the system may be seen in Fig. 4.2.

Should the operator select Manual Override instead of Automatic at the start of the test the procedure is modified as follows. Previous to the request to the operator to type GO, the program requests the frequency to be output and ear to be tested. Once the frequency is input to the computer and the system instructed to GO, the automatic procedure takes place but only at that frequency. Thus, the operator has control over the order of the tones though not of the choice of levels.

Throughout the automatic and manual tests each test-tone level is written in turn upon the VDU. This produces a record of the

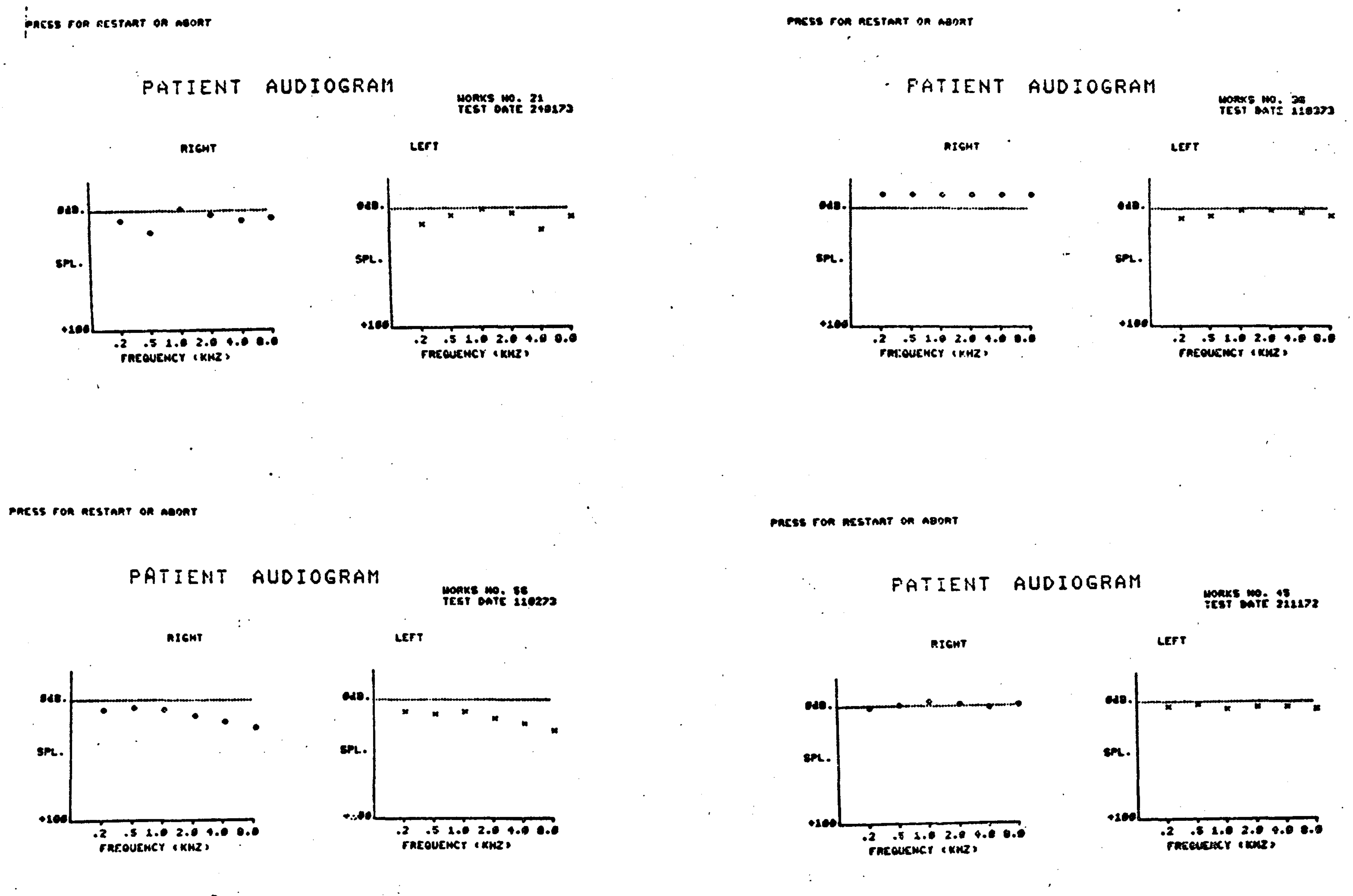


Fig. 4.2 Four samples of computer produced audiograms

test and the responses produced. Should a permanent record be required, the VDU may be replaced by a teletype which produces the record directly onto paper, or even paper-tape for computer storage.

Interrogation of individual magnetic-tape files is possible using a separate program. The files are selected by typing in the required works numbers at the VDU and the threshold figures, together with their relevant frequencies and the dates of the examinations, are reproduced on the line-printer.

4.3 The Computer Programme

The computer programme developed to control the audiometer system is basically straightforward in its structure and operation. However, the detail of how each component part of the programme functions is far more complex and largely irrelevant to the understanding of this thesis. Thus, the programme will be described in a general manner presenting detail only where considered necessary. A printed copy of the programme may be found in Appendix B, together with a set of basic programme flow diagrams.

The programme is written in Rank-Xerox Extended Fortran IV-H language with small, embedded sections written in Macro-Symbol, the assembly language used on the Sigma-5 computer. Assembly language is used for sections of programme required to perform input or output functions at high speed, such as the generation of a pulse-train to drive the programmable attenuator. The programme represents 12206 words of computer memory. This figure considerably exceeds the memory area available for running a programme, or 'job space', on the Sigma-5, as configured at the University of Warwick at that time. To overcome this problem a technique known as 'overlaying' is used, whereby only those sections of the programme that are currently required to operate are loaded into the computer job-space and moved out again when their purpose is complete. In this way, the overall programme size may be many times greater than the job-space available. A map of the overlay structure is shown in Appendix B. The programme was developed using punched-cards as the instruction source, this was continued until the system was completed when the entire card-deck was compiled onto a disc-file. All test-records are stored on magnetic tape.

4.3.1 Interrupts and Timers

In many parts of the programme, events are required to occur at specific times relative to each other. For example, the start and the end of the test-tone pulses should be a given duration in milliseconds apart. To achieve this the program is required to operate in conjunction with a clock and perform preset tasks at the time of the selected clock signals. This is known as operating in 'real-time' as opposed to 'computer-time', where events occur as a result of their relative position in the sequence of events defined by the programme. The clock signals may also be externally generated, such as the restart button, indicating a requirement for the computer to perform a specific action immediately. It is not usual to use a computer such that it is idle until a signal indicates it is required, more often the computer will be busy with another problem or part of a control system when the signal arrives. These signals require that the computer stops what it is doing and perform the task associated with them and then, when completed the original task may be continued. These signals are referred to as interrupts. Interrupt structures form an important part of control oriented computers such as the Sigma-5, where up to sixteen separate interrupts are permissable. In the eventuality that more than one interrupt occurs at a time, or that one occurs whilst another is being serviced, a priority system operates. Each interrupt signal is allotted a specific level in a priority table, such that if two or more occur together or whilst each other are operational the highest priority task will take place before any of the others and then the next and so on until all interrupts have been serviced. The more important the task the higher it is usually placed in the table. The computer audiometer system is designed around such an interrupt system with the ABORT and RESTART buttons at the top of the table. Internally, a computer clock is linked to another priority level to control the duration of the test tones and the inter-tonal period.

4.3.2 Record Keeping

This system is designed to operate fully automatically from the main audiometer system. The only input required from the operator is the works number of the patient. Patient reference numbers, or works numbers, will be denoted by WN.

Immediately after compilation of the program the operator is requested to input at the VDU the WN of the patient. Upon reading this data the computer calls a subroutine named DIRECTORY. This subroutine is arranged such that it positions the magnetic tape reader upon its start and then reads in the first 1000 words of information. Each of these words corresponds to a value of WN in ascending order from 1 to 999, the first word being a record of the total number of patients having records in store. Thus, should the WN be 98, the computer looks at the 99th word. The value of the 99th word or more generally, the (WN+1)th word is denoted as NF. NF is the position indicator of the patient's record on the magnetic tape files. Upon the determination of this number the computer skips NF records on the magnetic tape and reads in the one on which it ends, the (NF+1)th, the first being the directory record. The data acquired from this record consists of the past test results of the same patient with the number of tests the patient has had stored as the first word in the record. Once input to memory the data is stored into the necessary order for later use by a routine called ORDERIN. Had the value of NF been a zero, this would have indicated that the patient had not been tested before and so no record exists. The resulting action by the program is to call FILEGO, a record initiation routine. This reads the value of NR, the total number of records in store, and updates this by one. The new value is then written in place of the old on the first record, the directory. Thus, in future, any access to this value of NR, which in this case would be the value of NF, would cause the computer to read in the (NF+1)th record. Having updated the directory, the routine next outputs a blank record, large enough for ten years of twice yearly tests, immediately after the last record currently on the magnetic tape. This is the (NF+1)th record. In doing so it places an end-of-record marker at the end of the blanks. This marker is renewed every time new data is read into one of the records.

Having completed the above processes, the program is ready for testing to commence. As soon as testing is complete the acquired data is suitably ordered to be fed, along with any previous data, back into the relevant record. The first word of the record, the number of tests, is updated at this stage also. Throughout the filing system all information read off or written onto magnetic tape was transferred by means of BUFFER IN/OUT routine calls.

4.4 Descriptions of the Peripheral Equipment

The hardware system components are individually described in the following paragraphs and their overall specification as an audiometer presented at the end. Individual specifications are not given as they are largely superfluous to the purpose of this thesis. A photograph of the peripheral equipment may be seen in Fig. 4.3.

4.4.1 Patient's Response Button

This is a single-pole biassed push-button switch, mounted in a small tube suitable for holding comfortably in one hand. The button protrudes from one end of the tube in such a way that is is possible to depress it with the thumb of the holding hand. The switch contacts are fed through the opposite end of the tube by means of a screened cable. A direct-current voltage is placed in series with the switch such that closure of the contacts causes a voltage to appear at the far end of the cable where it is detected by a computer digital-input. The computer programme inspects the digital input at regular intervals to determine the patient's response.

The switch itself has two particular properties which are most important for the accurate execution of the examination.

i) There is no mechanical 'click' from the switching action as this could prove distracting to the subject in what is an otherwise quiet environment.

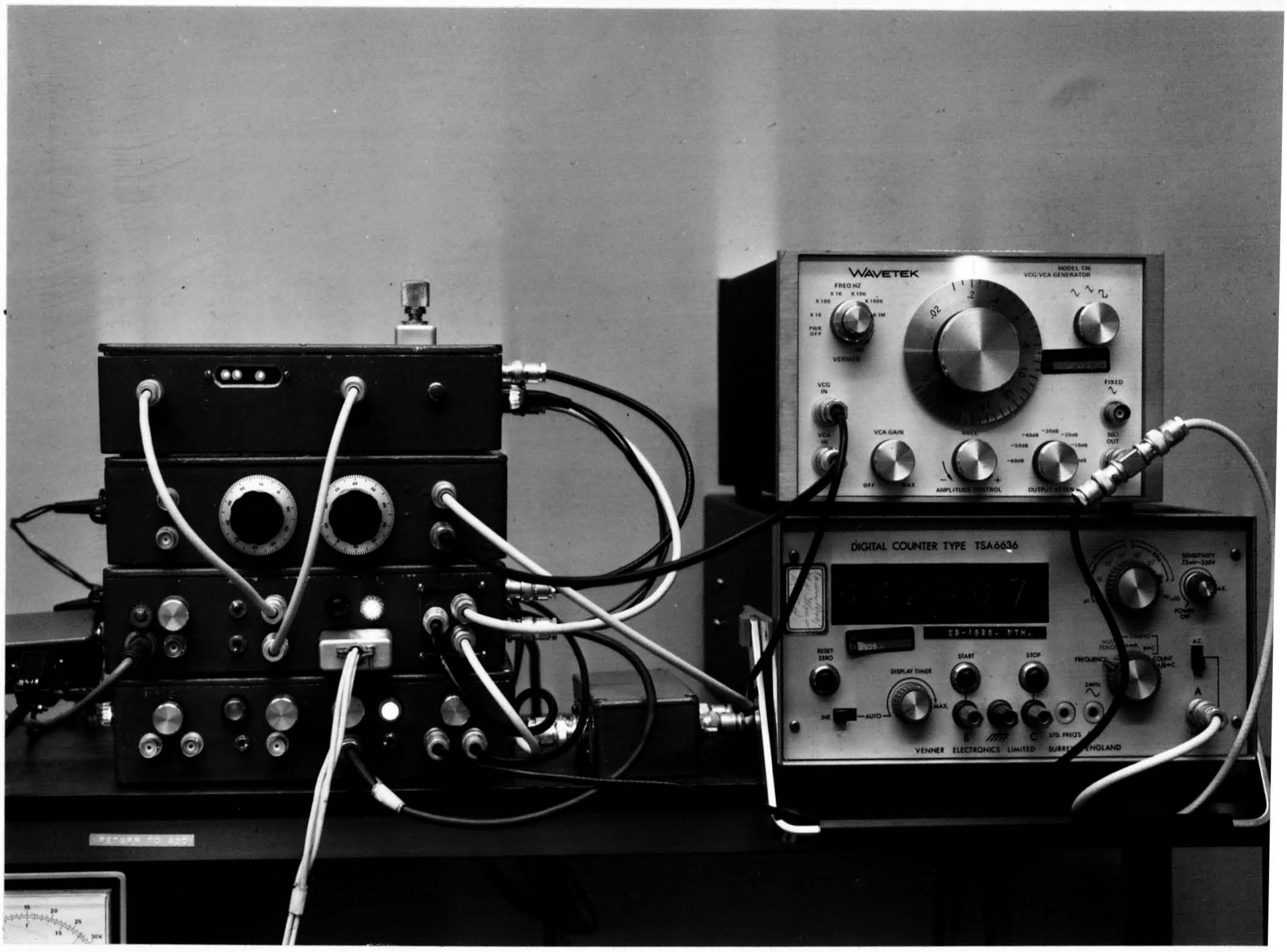


Fig. 4.3 The computer audiometer hardware.

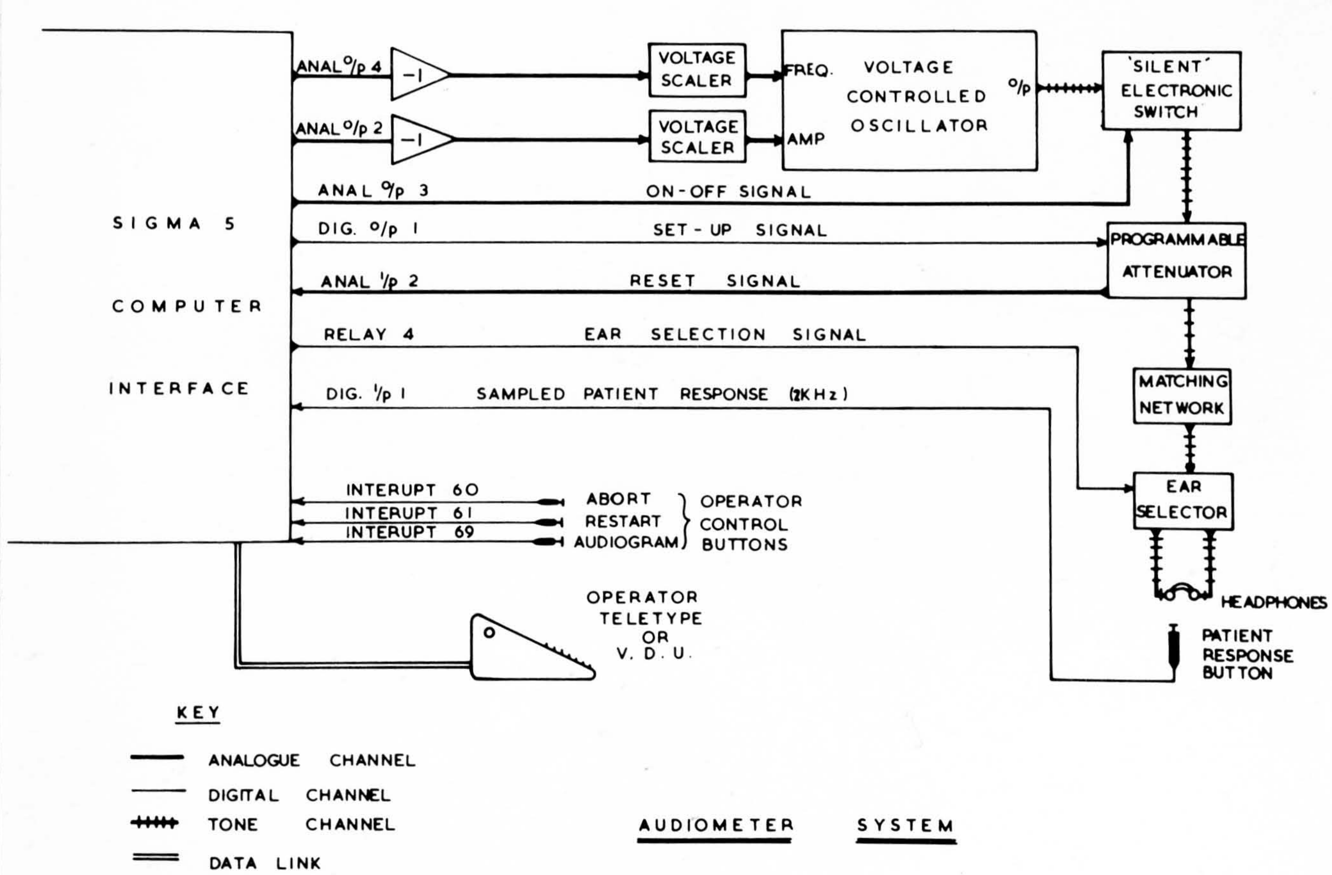


Fig. 4.4 System diagram of audiometer hardware.

ii) The button has a sufficiently light action to ensure that it's depression does not require a conscious effort. Conversely, it is not so light as to allow it to be pressed accidentally by the action of resting a thumb lightly upon it. The switch finally used was selected by presenting two men and a woman with three available alternatives mounted in tubes and asking them to indicate which they preferred.

4.4.2 Operator Control Buttons

The operator has two groups of controls with which to administer the examination. The first is the keyboard of the VDU, and the second a set of three biassed push-buttons similar to the response button. The three are mounted on a single panel and placed adjacent to the VDU. The three buttons represent:

i) Abort the system

ii) Return to start of the test

iii) Draw audiogram on VDU.

The three buttons are connected to different interrupt inputs of the computer.

4.4.3 The Programmable Oscillator

An audio frequency, voltage programmable oscillator is used as a tone source. The oscillator is programmable in both amplitude and frequency by two DC voltages, V_A and V_f respectively, a variation of from 0-2.0V being needed to achieve programmability over the full range. Both control voltages are transmitted from the computer via the analogue outputs and fed through differential amplifiers to the oscillator. In order to utilise the full 0-10V output voltage range of the computer, the transmitted voltages are attenuated by a factor of 5 by scaling networks after passing through the differential amplifiers. The values of V_A and V_f are preset in the programme and written on the VDU at turn-on time of the system to allow checks to be made upon the received values for calibration purposes. A digital read-out frequency meter is permanently connected to the oscillator output to allow the operator to check tone accuracy at all times. In practice the tone was found to be constant to within $\pm$ 2%, BS 2980 (29) and IEC 178 (28) require $\pm$ 3% as the worst case.

As was shown in Section 1.2.4, the sound pressure level necessary at a 'normal' ear, to be just heard, is not constant with changing frequency. BS 2497, Part 2 lists calibration sound pressure levels (SPL) to be measured with various common earphone types when coupled to a standard artificial ear (type 9-A). To compensate for this, the output level of the oscillator is adjusted for each frequency to present 67.5 dB above the threshold SPL. In this way, the need for a shaping filter is avoided. IEC 178 requires that all SPL's should be within ± 5 dB of the indicated values. The oscillator output was monitored on several occasions and found to be better than ± 2 dB of the required values. The ability to reduce the oscillator output to a nominally zero level by applying a programming voltage equal to zero volts is utilised when removing the tone from the headphones. However, IEC 178 suggests the SPL applied to the ear should be 60 dB lower when the tone is in an OFF state than when it is in the ON state. The oscillator used cannot achieve this difference, particularly at the relatively low output levels required at 1 and 2 kHz. To overcome this, the oscillator is followed by a switch capable of much greater signal reduction.

4.4.4 The 'Silent Switch'

It is most important that the subject does not hear an electrical 'click' in the earphones at the point in time that the tone is presented or removed. This could cause the subject to imagine a tone when in fact it is below his threshold. To avoid this problem a switch was designed such that the signal is 'faded' out or in, as instructed, over a pre-determined period of time. IEC recommendation 178 suggests that the sound pressure level produced by the earphone should reach -1 dB relative to its final steady state is no longer than 0.2 seconds from the instant of the tone being switched 'ON'. The rate of increase of the sound pressure level should not exceed 500 dB/second in the region -20 dB to -1 dB relative to its final steady state. When switched 'OFF' the sound pressure level should decay from -1 dB to -60 dB relative to the initial steady state value in not more than 0.2 seconds. The rate of decay should not exceed 500 dB/second between -1 dB and -20 dB relative to the 'ON' level. The switch constructed and used in the system has five selectable time periods from 50 mS to 150 mS, each of which is within the above limits. The rise and fall times always being equal. The 'silent switch' presents 70 dB of attenuation to signals from 200 Hz to 12 kHz. A circuit diagram of the switch may be seen in Fig. 4.5.

The switching action is performed by passing the signal through an analogue multiplier circuit, where the multiplication factor is determined by a d.c. voltage applied to a control input. In operation as a switch the circuit is used in one of only two multiplication modes, x/ and X0. The signal required to select the two is fed from a computer digital output, through a differential amplifier to the switch. Upon entry into the switch, the signal is passed through a simple resistance-capacitance (RC) network to generate the necessary controlled rise and fall times from the digital outputs 'instant' switching time. The rise and fall time is adjustable by means of a selection of switched capacitors.

A lamp is situated on the front panel of the box containing the silent switch to indicate to the operator whether the switch is in the ON or OFF state. A photograph of the silent switch may be seen in Fig. 4.6. The small toggle switch immediately beneath the lamp is a manual test switch. The signal leaving the silent switch

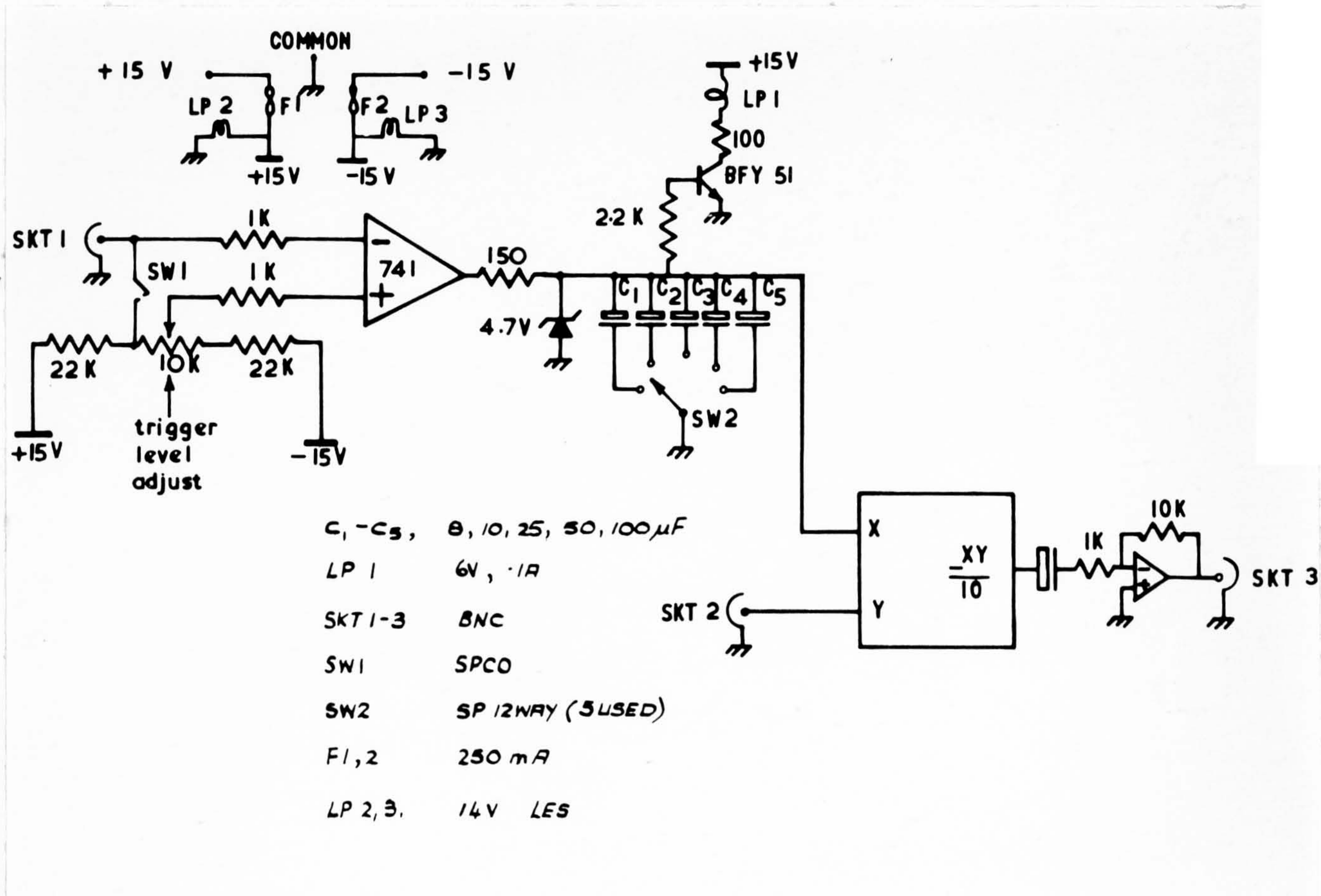


Fig. 4.5 Diagram of the 'silent-switch'.

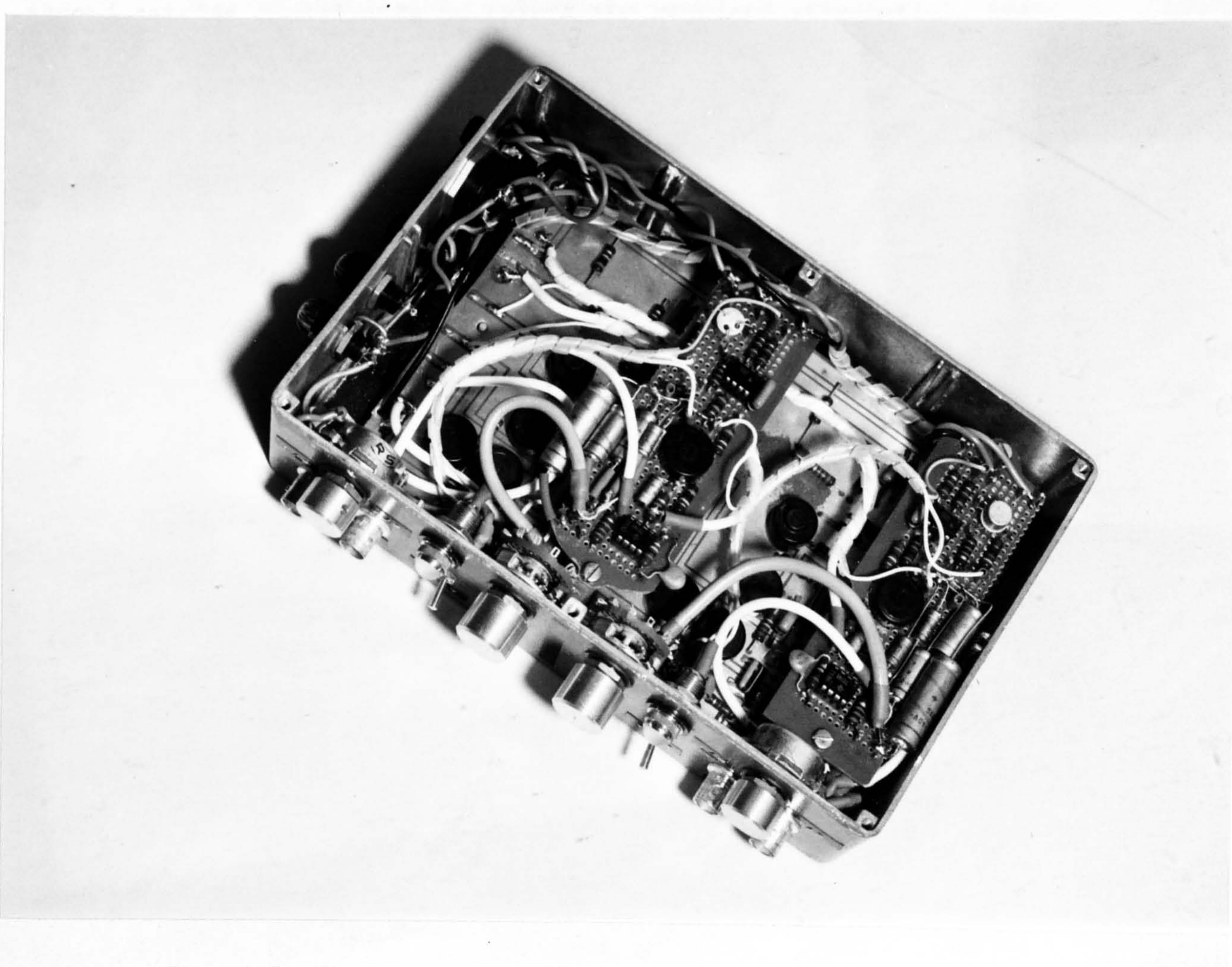


Fig. 4.6 The electronic'silent-switch'.

is fed to a programmable attenuator.

4.4.5 The Programmable Attenuator

The level of the tone presented to the earphones is determined by a programmable attenuator. This unit has an attenuation range of 77.5 dB, in steps of 2.5 dB.

The unit is computer controlled by means of an analogue input and a digital output. The first transfers a pulse to the set computer to indicate that the attenuation is set to zero, and the second carries serial data to select the required attenuation, see Fig. 4.4. The principle of operation may be seen from Fig. 4.7. A series of five bistable integrated circuits are fed by the serial data from the computer to form a five-bit binary counter. Each data pulse from the computer causes the binary counter to advance by one. The output signal from each bistable is fed through a transistor driver circuit to a relay drive coil. The relays are arranged such that the one connected to the lowest order binary digit selects a 2.5 dB resistive attenuator, the second a 5 dB attenuator, the third a 10 dB and so on up to 40 dB. The relay contacts are connected such that when selected they place the attenuators in series with each other thus causing a cumulative attenuation in the signal path. Unselected attenuators place a short-circuit in the signal path. The value of attenuation in circuit is indicated to the operator by means of a set of five binary indicator lamps on the front panel of the attenuator case. For test purposes, a manual 'advance by 2.5 dB' button is situated on the front panel. With this instrument it is only possible to count upwards. To reduce the attenuation by any amount requires a reset to zero and count up to that point. To allow for the possibility of different source and termination impedances the attenuators were mounted in 'plug-in' modules. Three different modules were constructed, 50 ohms, 600 ohms and 1000 ohms. The attenuator unit may be seen in Fig. 4.8.

4.4.6 The Control and Interface Box

With the exception of the VDU and intercom., all cables to and from the computer pass through this box. It is primarily a convenient means of distributing the various inputs and outputs to their respective parts of the system. The only other function performed by the box is the selection of the right and left earphones. An analogue output from the computer drives a reed-relay which directs the levelled signal to the required ear. The ear selected is indicated on two lamps on the case exterior. The d.c. supply necessary for the patient's response button is also situated in this box. The box may be seen in Fig. 4.3 where it is the second from bottom unit in the stack of four.

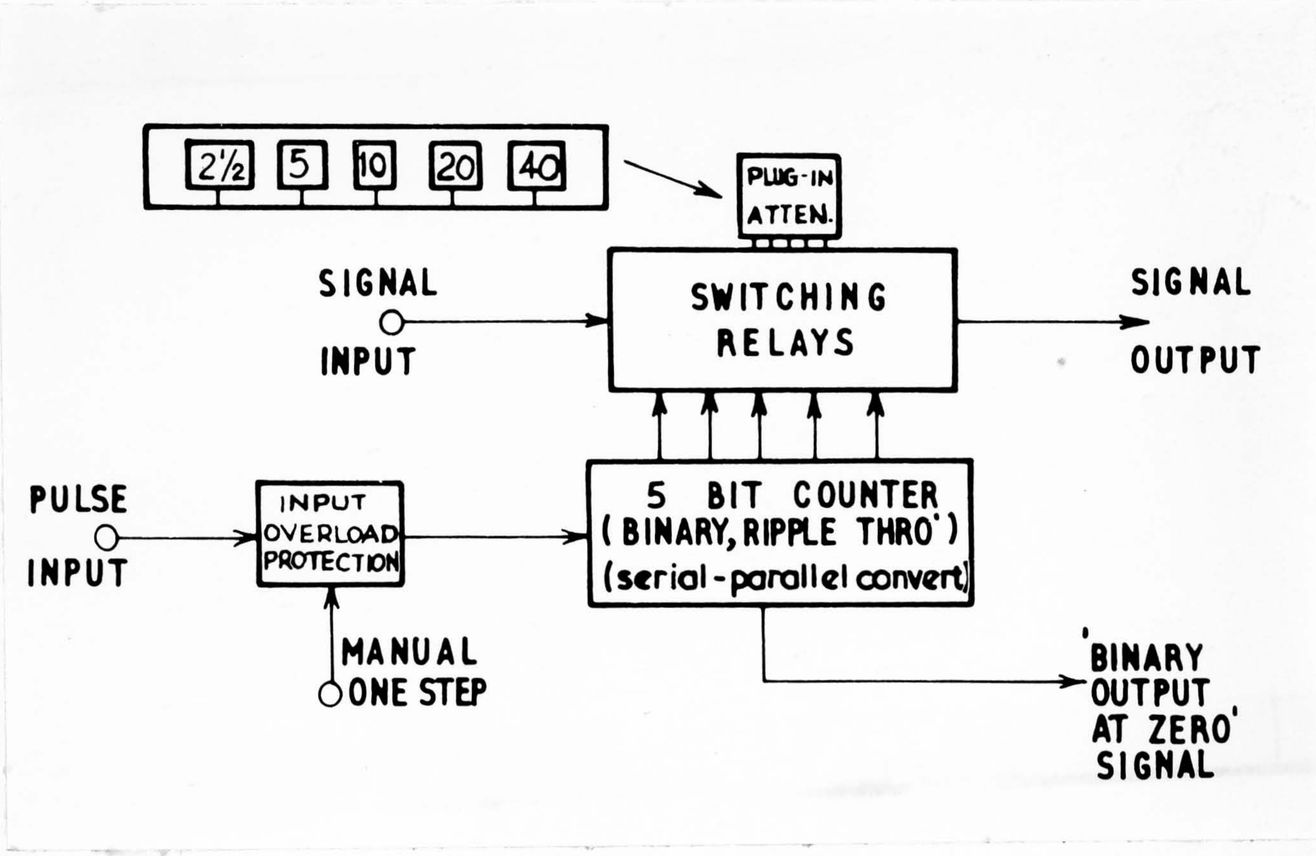


Fig. 4.7 Diagram of the programmable attenuator.

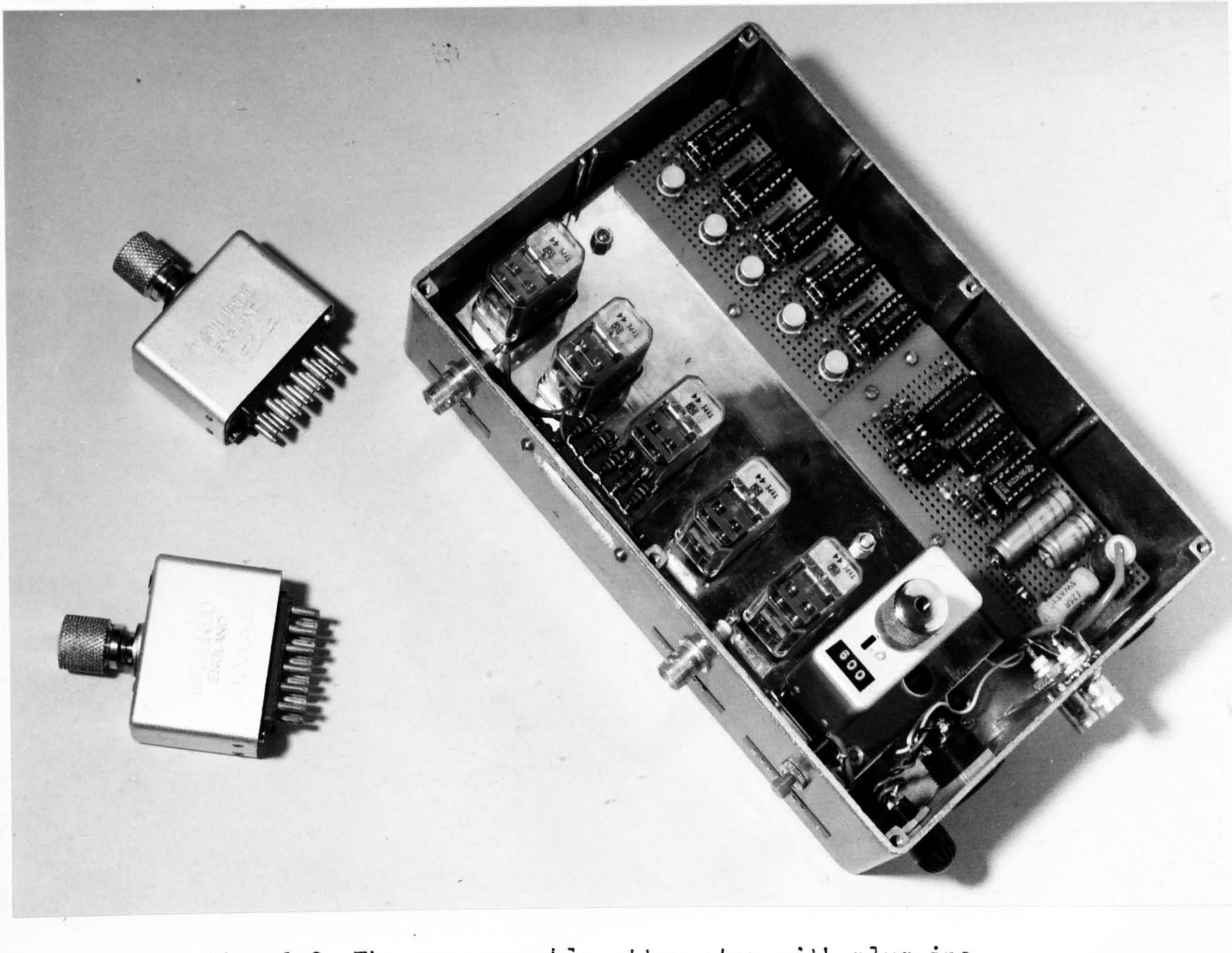


Fig. 4.8 The programmable attenuator with plug-ins.

4.4.7 The Earphones

The earphones used are of the Telephonics TDH-29 type with MX41/AR cushions. The earphones together with the headband used with them were originally supplied with an Alfred Peters clinical audiometer, Model AP6. The computer audiometer was calibrated to allow them to be used with it and the resultant sound pressure levels compared with those from the Peters model for similar frequencies and levels. The two sets of results agreed to within ± 0.5%. The calibration was performed using an artificial ear to BS 2497, mounted in a variable pressure calibration jig designed and manufactured in the University workshops, see Fig. 4.9 and Fig. 4.10. The SPL measurements were made using a Bruel & Kjœr, type 2107 Frequency Analyser with a 1" condensor microphone of similar manufacture. The analyser and microphone were initially calibrated using a Bruel & Kjœr pistonphone.

4.4.8 Earphone Calibration Jig

The jig is solidly constructed on a base of one-half inch thick aluminium mounted on soft rubber feet, see Fig. 4.10. A standard Model 9-A, artificial ear is located and clamped in place in a vertical block of aluminium such that the microphone-opening faces outwards from the jig. The earphone is held in place over the conical face of the artificial ear by means of a specially contoured cup pressed against the back of the earphone by a spring loaded piston. The spring pressure may

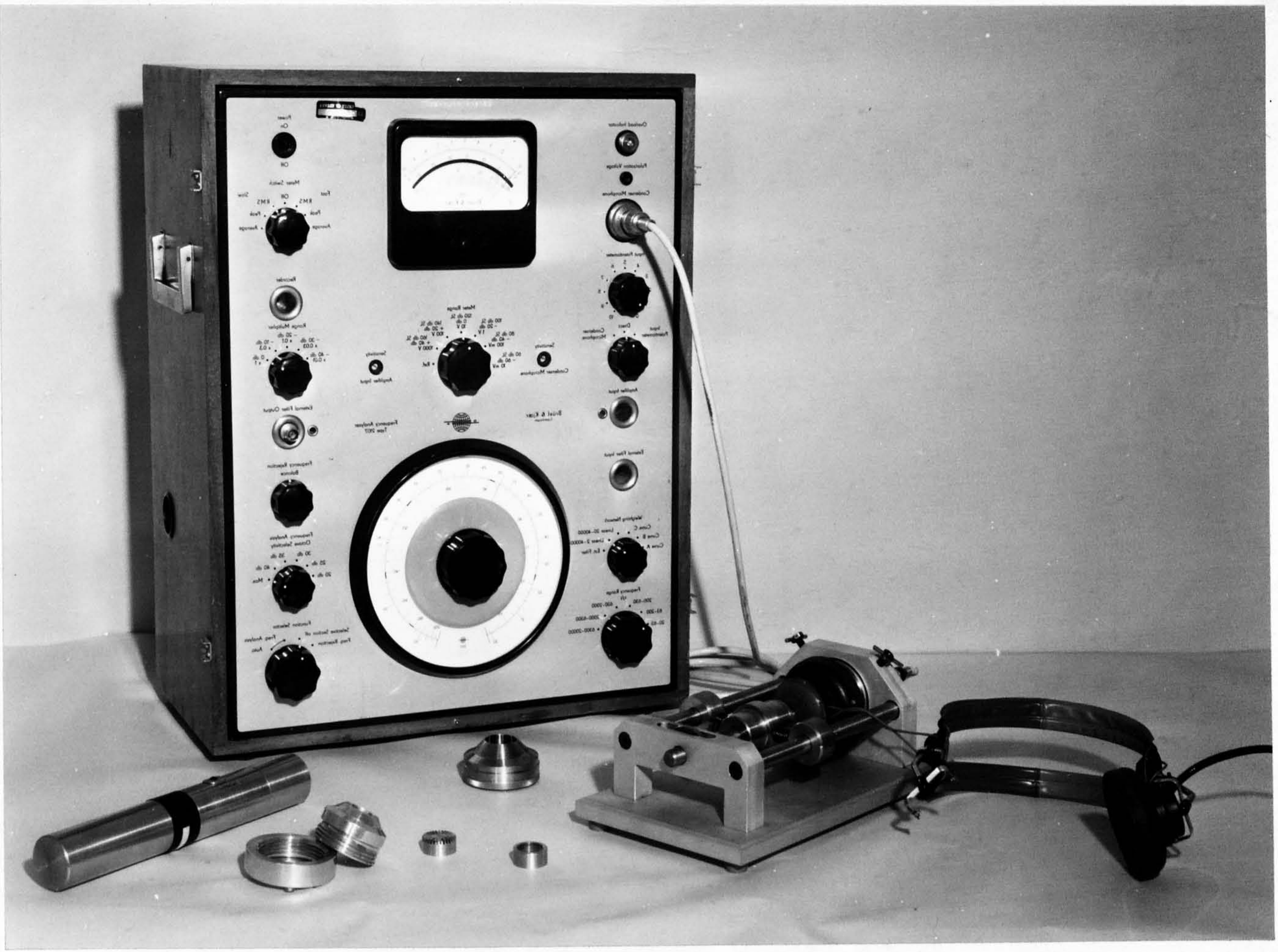


Fig. 4.9 Audiometer calibration equipment.

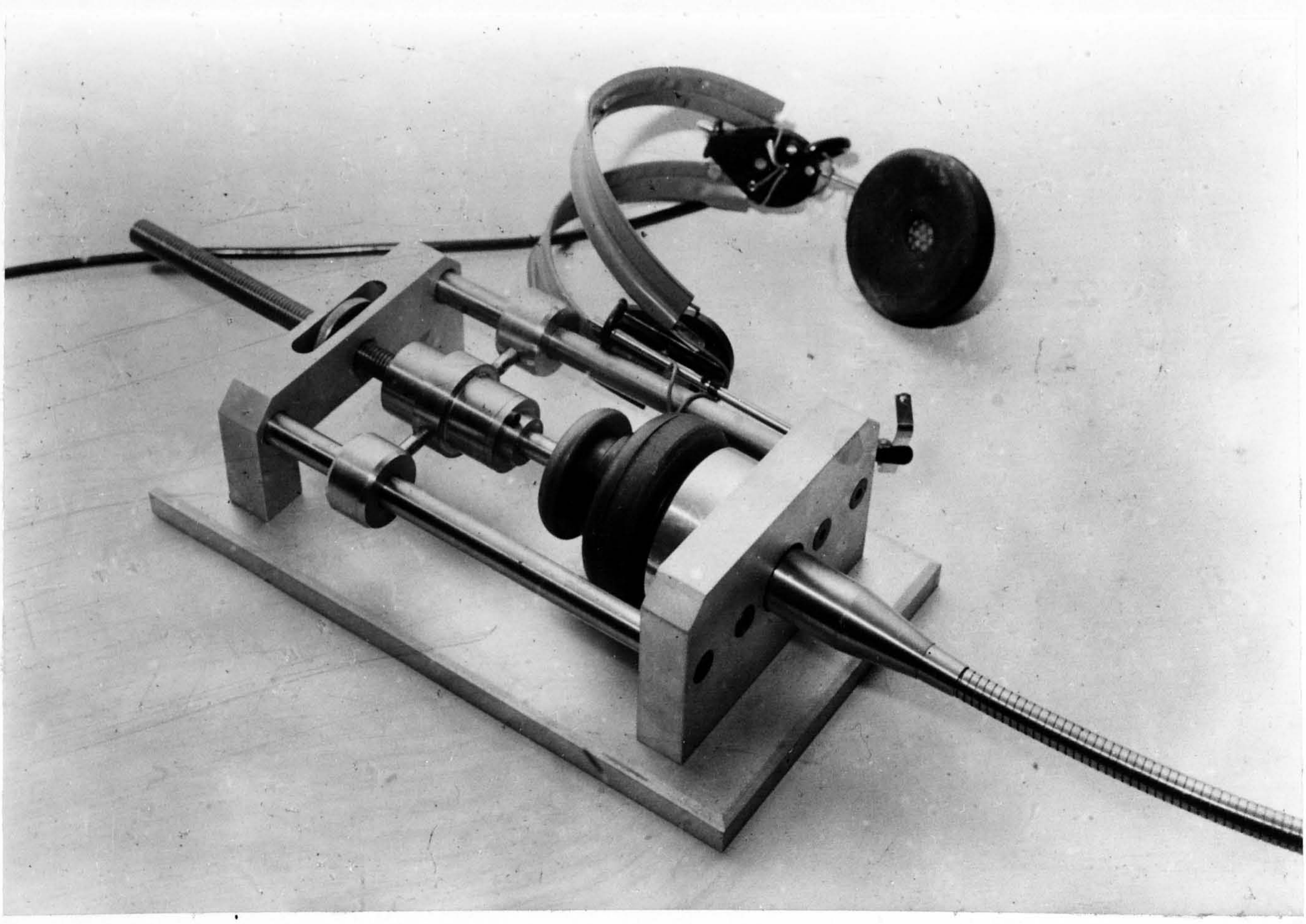
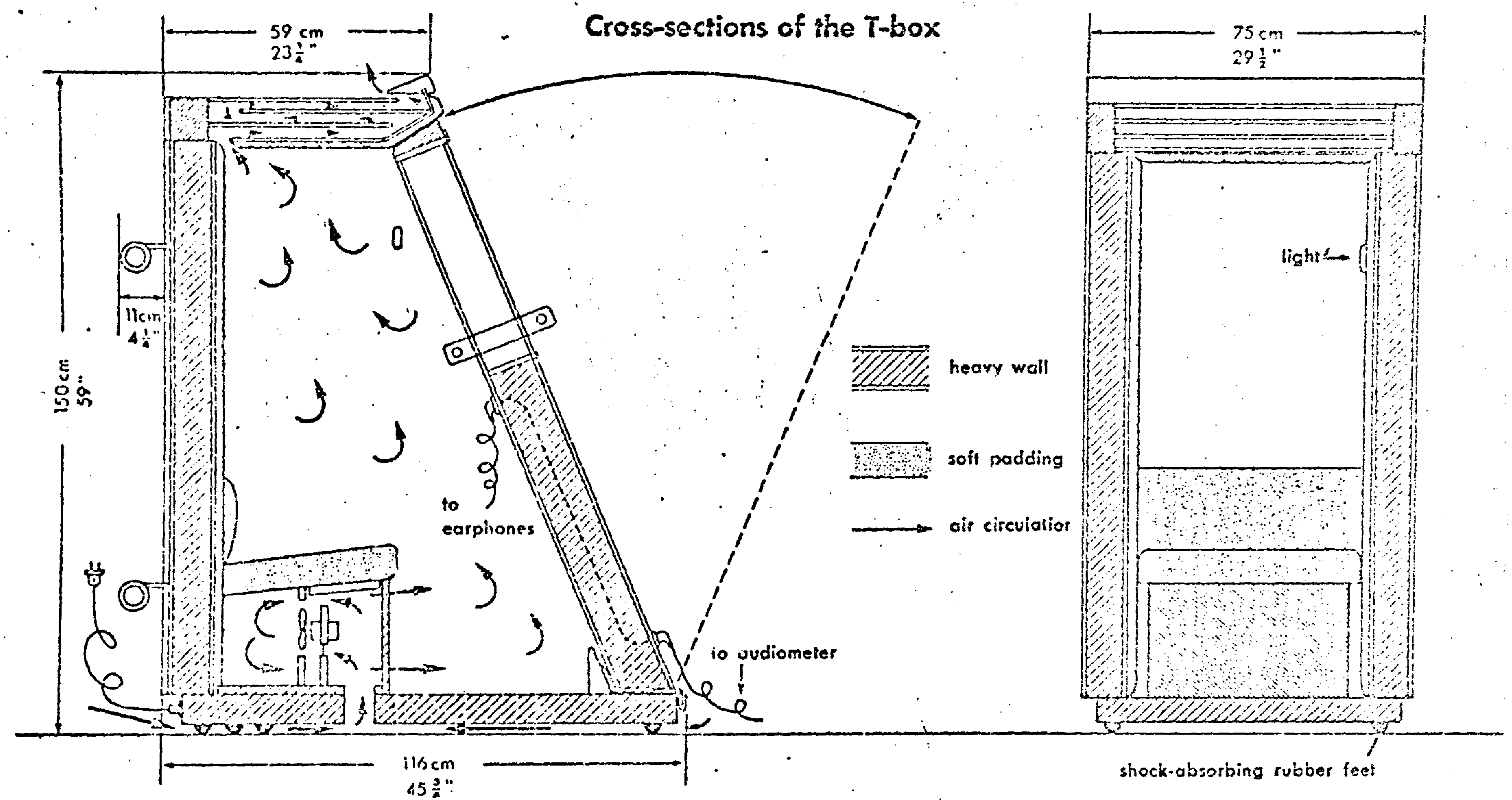


Fig. 4.10 Mounting-jig for 9-A coupler.

be adjusted by means of a thumbwheel mounted in a second vertical block, positioned opposite the conical face of the ear. Through the block passes a threaded rod rigidly connected to a cylinder in which the spring and piston are situated. The piston pressure may be read off a scale engraved upon the piston shaft. The scale indicator being the cylinder collar where the shaft enters the cylinder. The cylinder is supported by two arms, one protruding horizontally from each side of the cylinder, and running along two stainless-steel guide rods clamped between the artificial ear block and the thumbwheel block. A sliding fit brass ring on each rod is connected to the cylinder arms thus supporting the cylinder in both the horizontal and vertical planes. The spring pressure was calibrated using a series of weights and marking off the deflections on the cylinder wall.

4.4.9 The 'Sound-Proof' Environment

The equipment was situated in a completely enclosed room with no windows, surrounded by relatively quiet offices on three sides and a corridor on the fourth. The room dimensions were approximately 12' x 9' x 8' in height. In a corner of the room was situated a commercially available 'sound-proof' booth. The booth is manufactured in Sweden and known as a 'T-box'. The manufacturer's sound reduction figures are shown in Fig. 4.13 . Constructional details of the T-box may be seen in Fig. 4.11 . The ventilation fan was found to be too clearly audible to allow its use during an examination but the natural ventilation through the fan ducts proved to be sufficient in all but

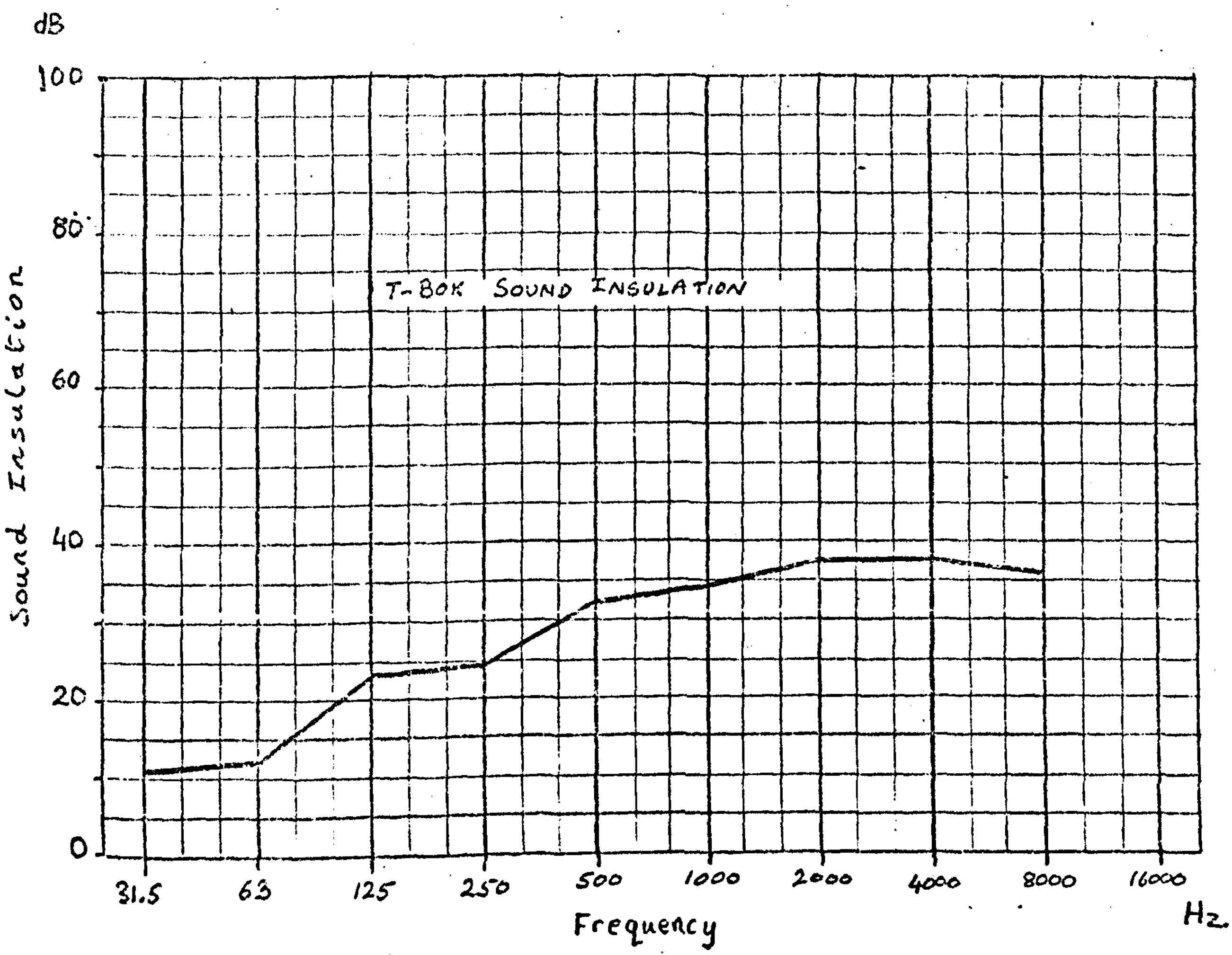


Construction of the T-box.

the hottest weather. The door does not have any form of catch, it remains closed by the tightness of its rubber seals and gravity. This has an advantage with claustrophobic subjects as they are aware that they can open the door by simply pushing. Although small in size, the T-box proved to be both comfortable and efficient for its purpose. Cable sockets were provided on the door of the box. The sockets were connected to others on the inside by cables passed through an insulating labyrinth. The noise level inside the room and the booth is shown in Fig. 4.13.

The general layout of the room may be seen in Fig. 4.14. The subject was situated approximately seven feet from the operator with the majority of the peripheral equipment out of his direct line of sight. There were no indicators or dials of any type visible to the subject. Because the computer was in another room the subject was not necessarily aware of its participation in the examination and thus any lack of concentration due to an interest in its presence was minimised.

Fig. 4.12 The T-box.



a) Manufacturers sound-insulation figures.

$\frac{1}{3}$-octave band centre frequency (Hz)	sound pressure level (dB re. $2 \times 10^{-5} N/m^2$)		
	In room	In T-box	NPL -10dB Hearing Level
63	60.0	55.0	62
125	51.0	40.0	48
250	47.0	24.0	36
500	30.0	<0	14
1000	22.0	<0	16
2000	12.0	<0	29
4000	12.0	<0	37
8000	12.0	<0	32

b) Noise levels inside and outside the T-box. as measured in the audiometry room at the University of Warwick.

Fig.4.13 Sound insulation of the T-box.

Fig. 4.14 A general view of the audiometric room.

4.5 Audiometer Performance Specifications

The audiometer is designed such that its performance is within the requirements of the International Electrotechnical Commission's recommendation number 178, which is specifically concerned with pure-tone screening audiometers.

a) Frequencies

Six tones are programmed for use: 250, 500, 1000, 2000, 4000 and 8000 Hz. An alternative selection of any other six tones may be made from the frequency range 250 to 8000 Hz by simple programme changes. The stability of all tones is better than ± 2%.

b) Purity of the tones

The sound pressure level of each harmonic of each tone at each setting of the attenuator is at least 40 dB below that of the fundamental component when measured with the earphone applied to a 9-A coupler.

c) Attenuator

The sound pressure level of each tone is adjustable in 2.5 dB steps. Due to the oscillator output being adjusted at each frequency to 67.5 dB above the standard reference equivalent threshold sound pressure level, one of the attenuator settings automatically corresponds to it.

d) <u>Tone Switch</u>

With the tone in an 'OFF' condition, the sound pressure level produced by the earphone in a 9-A coupler is at least 80 dB below the level present when in the 'ON' condition.

The build-up and decay times of the tones may be selected, by means of a front panel switch, from: 50, 75, 100, 125 and 150 milliseconds. These times represent the time taken to rise or decay between 0.9 and 0.1 of the two steady-state levels.

e) <u>Sound Source</u>

The earphones are of the Telephonics model TDH-39 type, fitted with MX-41 rubber, detachable earcaps. The headband is of similar manufacture allowing the earphones to be positioned upon an individual's ears with a force of at least 4 Newtons.

f) <u>Range of Sound Pressure Level</u>

For all test tone frequencies the range of sound pressure levels is 77.5 dB. This extends from 10 dB below the standard reference equivalent sound pressure level to 67.5 dB above. The IEC recommendation requires 70 dB above 0 dB HL. The programme is written to allow the oscillator output to be adjusted such that its level is up to 72.5 dB above 0 dB HL, thus allowing the attenuator to control it over a range from 0 dB HL to 72.5 dB HL.

g) <u>Accuracy of Sound Pressure Level</u>

The sound pressure levels produced at the earphones, when measured in a 9-A coupler, are within $\pm$ 2 dB of the absolute values requested by the programme at all frequencies.

h) Unwanted Sound in the Earphones

There are no sounds in the earphones during the tone 'OFF' state which are discernable to an otologically normal subject. During tone presentations no clicks, mains-hum or other unwanted electrical noise can be detected by a normal listener.

4.6 An Analysis of the Performance of the Computer Controlled Audiometer

In order that the hearing threshold results obtained from the computer audiometer could be compared with those of a conventional manual audiometer it was decided that a group of people selected randomly from amongst the university staff and students should be subjected to tests by both methods and the results compared. Each subject was tested using a manual audiometer and then the computer system. Immediately prior to a group of examinations the frequency and level outputs of both the manual audiometer (Alfred Peters AP6) and the computer audiometer were compared using the same earphones to be used in the examination, mounted upon a 9-A coupler. Any discrepancies in output from the computer audiometer when compared with the AP6 were adjusted until zero.

Thus, for any given setting of either audiometer the sound pressure level at the ear of the subject should be identical within the limits of the measuring apparatus used to compare them. The AP6 audiometer was calibrated by the manufacturer six months prior to the tests and it's output checked in the laboratory at the initial commencement of them, thus the output could be trusted to be within the limits required by I.S.O. 178(28) and BS 2980(29). The same six frequencies were used in each test; 250, 500, 1000, 2000, 4000 and 8000 Hz. The operator performing the manual tests was not a qualified audiometrician but was a research assistant in the noise and vibration department with a thorough knowledge of the purpose and principles of the test she was performing. Prior to each test she explained the examination procedure to each subject and ensured that they understood what they were to do. In cases where subjects wore spectacles or hair over their ears, the earphones were positioned with each of these obstructions removed or repositioned. The subject was then requested to sit in the soundproof box and to close the door themselves. In this way they realised that they could always open it if necessary, a particularly important feature with a claustrophobic subject. The test would then commence at the 250 Hz tone and progress up to 8000 Hz on each ear in turn. The test was performed in this increasing frequency manner as that was the order in which the computer was programmed to present the tones on the initial system. The computer system utilised for the test used a simple YES-NO-YES-NO pattern for threshold recognition at 10, 5 and 2.5 dB step sizes. Failure to produce a threshold after 50 level presentations at a given frequency resulted in an 'INDETERMINATE' message and the system continued at the next frequency. The basic tone duration was 1 second as was the inter-tonal period. No decisions were made based upon the timing of individual responses. All thresholds were rounded up to the nearest whole number of decibels.

The original intention was to examine 50 individuals with the basic system described, unfortunately only 20 were tested due to time limitations. However, 20 people tested at six frequencies on two ears provides 240 result comparisons, a number sufficiently large to allow them to be used for a simple statistical study. Of the 240 results 7 were indeterminate and 18 were lost due to a computer fault, leaving 215 results. The differences in decibels between the manual results and computer results are shown in Fig. 4.15. From this diagram it may be seen that the majority of the computer results were within ± 6 dB of the manual results, the mean error is -1.41 decibels. However, a more meaningful figure is the standard deviation, which for these results is 5.75 dB. This means that approximately 68% of the results are within ± 5.75 dB of the manual results and that 95% are within ± 11.5 dB of them (33). At first examination the above error limits appear to be quite large, it should however be considered that the repeatability of a measurement using any of the current pure-tone techniques is often only to within ± 10 dB (13) even when the two measurements are made within 10 minutes of each other. Thus the differences of up to ± 10 dB are within the normal variations to be expected from repeated tests. These limits include 192 or 86.5% of the measured thresholds. Of the 31 thresholds outside these limits 7 are declared indeterminate which leaves 23 or 10.36% of the total which are out of limits. Amongst these 23 there are three times as many negative differences (seventeen) as there are positive (six). A similar pattern exists amongst the results as a whole. Frequently a second measurement of a particular threshold displays a lower result than the first. The apparent lowering of the threshold is partially explained by the effect of the subject learning the test routine and becoming more aware of what to listen for. This learning effect could possibly account for the fact that 61.7% of the differences were negative as compared with 34% which were positive. In order to verify this possibility it would be necessary to repeat the tests with the computer controlled examination preceding the manual test. Unfortunately, this is not possible and so it is necessary to assume that a learning effect did influence the results to an unknown degree. It should, however, be considered that an overall movement of +1 dB in the differences would have meant that 50 or 22.5% would have agreed, 35% would have been negative and 39% positive. Thus it is reasonable to say that the results are relatively evenly distributed about the zero difference, with no evidence of the computer thresholds being lower than those of the manual audiometer. However, outside the ± 10 dB limits described above, there were 3 times as many negative differences as there were positive. A larger sample size of subjects may have removed this imbalance although if the fact that the magnitude of the average learning effect is approximately -10 dB is considered then this possibly accounts for some of them. Also, a close investigation of the operator's result sheets revealed that all six of the differences greater than +10 dB were generated by the same subject, whereas the negative differences of similar magnitude were the result of nine different subjects. It would therefore seem that the negative differences may be partially explained by the learning effect and probably also partially by operator error in performing the manual tests. It must also be concluded, however, that the computer audiometer system itself caused a negative offset of the thresholds for reasons not obvious from the evidence available.

CHAPTER 5

CONCLUSIONS

1. The implementation of a computer-controlled audiometer has been effected. The instrument constructed produces test results of an accuracy comparable with those of current screening audiometers.

2. It has been shown how the use of digital computing techniques solves many of the problems associated with industrial audiometry and simplifies the audiometric procedure.

3. To ensure that the computer-controlled audiometer does not become merely a machine-minded instrument the study has looked beyond the audiometric measurement problem so that audiogram assessment, patient referral and record-keeping can be encompassed. The instrument has thus become an integrated audiometric screening system.

4. The system devised during this research offers many advantages over conventional instruments:

i. each individual response of the patient is analysed for its temporal relationship to the stimulating tone and for its place in the sequence of responses.

ii. an adaptive test procedure maximises the suitability of the test for the individual under examination, greatly increasing the reliability of the results.

iii. the duration of each examination is minimised by terminating the test at a pre-determined reliability level.

iv. the degree of operator monitoring is greatly reduced because the system is capable of recognising the more common subject errors and taking remedial action.

v. malingerers, not uncommon in industrial practice, are readily detected.

vi. the need to 'manually' examine all test results and compare them with pre-employment audiograms is removed by the data storage and manipulation parts of the system.

5. Following considerable research into the requirements, background and current mode of operation, the system devised has been so structured as:

i. to be most useful in industrial audiometric screening.

ii. to take full account of the variability of manner of approach of the patient to the test.

iii. to provide the test supervisor with the greatest flexibility within the need to reduce test-time and his involvement in strictly repetitive tasks.

5.1 Proposals for Future Work

There are two proposals for future work arising directly out of the work of this thesis.

a) A study is required of pattern recognition techniques to improve upon the basic systems suggested. It is possible that the use of 'glide-tones' rather than discrete-level pulsed-tones, could improve accuracy by allowing the pattern analysis programme a greater resolution of tone levels. In this way it may be possible to perform basic diagnostic operations within the analysis programme.

b) The temporal analysis of individual responses remains unproven by practical experiment. Extensive trials are necessary before it may be stated with any confidence that it can, in practice, offer a significant improvement in accuracy and reliability of test results.

c) The major disadvantage of the computer-controlled audiometer is its size. Computers are bulky and usually static. However, microprocessor devices would allow a system such as that described in this thesis to be constructed in an instrument only slightly larger than a conventional clinical audiometer and at a cost similar to that of current units.

d) The major disadvantage of all subjective tests is the requirement for the examinee to actively participate. Potentially, the accuracy of pure-tone audiometric examination can be greatly improved and made independent of the subject by means of electroencephalographic (EEG) techniques. The use of a computer to control an EEG audiometer is quite practical, the major problem would be to develop a suitable programme to statistically analyse the waveforms detected from the nervous system. A current problem with this technique is the fact that the waveforms used to recognise that a tone has been heard are surrounded to such a high degree by apparently random noise, that they become irrecoverable from it at test tone levels near to the ear's threshold. However, for the purposes of a screening audiometer the approach would seem to offer a genuine possibility.

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APPENDIX A

THE SIGMA-5 ON-LINE COMPUTER

The system installed is as shown in Fig. A.1. An experiment or controlled apparatus is linked to the interface equipment by means of a cable laid through the building, terminating in a set of ten input/output sockets mounted on a wall panel. A teletype and visual display unit link is also provided to each terminal. The input/output sockets are connected to the various computer functions, (e.g. D/A convertors, external interrupts) by means of a patchboard. An intercommunications unit exists to allow verbal communication between the computer room and a remote laboratory.

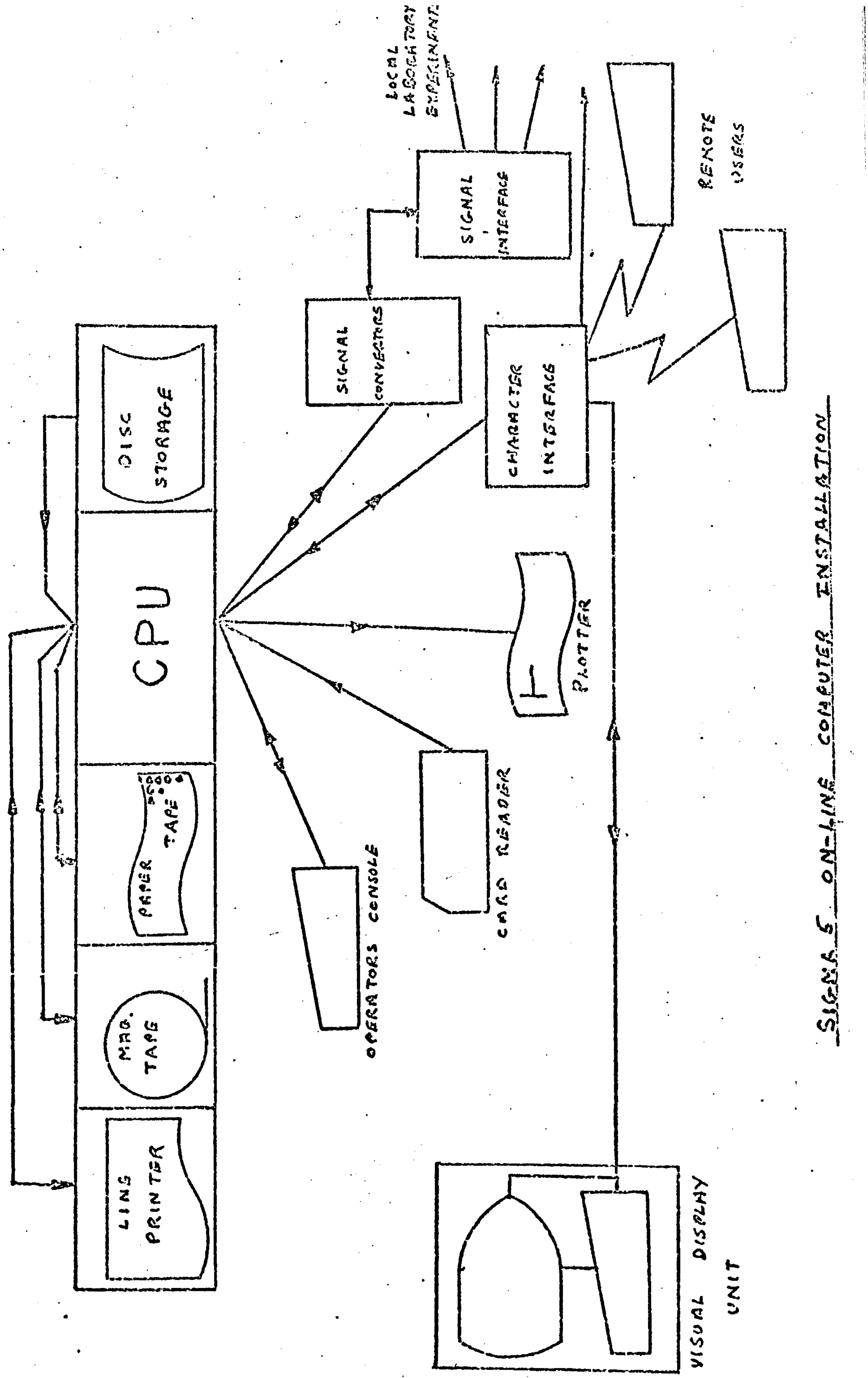

LOCAL LABORATORY EXPERIMENTS
SIGNAL INTERFACE
REMOTE USERS
SIGNAL CONVERTERS
CHARACTER INTERFACE
DISC STORAGE
CPU
PLOTTER
PAPER TAPE
CARD READER
OPERATORS CONSOLE
MAG. TAPE
LINE PRINTER
VISUAL DISPLAY UNIT
SIGMA 5 ON-LINE COMPUTER INSTALLATION

Fig. A.1

APPENDIX B

THE COMPUTER PROGRAMME

The following diagrams 1-8 show the outline logical flow of the audiometer programme. The overlay structure of these routines is shown in Fig. B9. The instruction listing which follows is that which was used to perform the tests reported in Chapter 4.

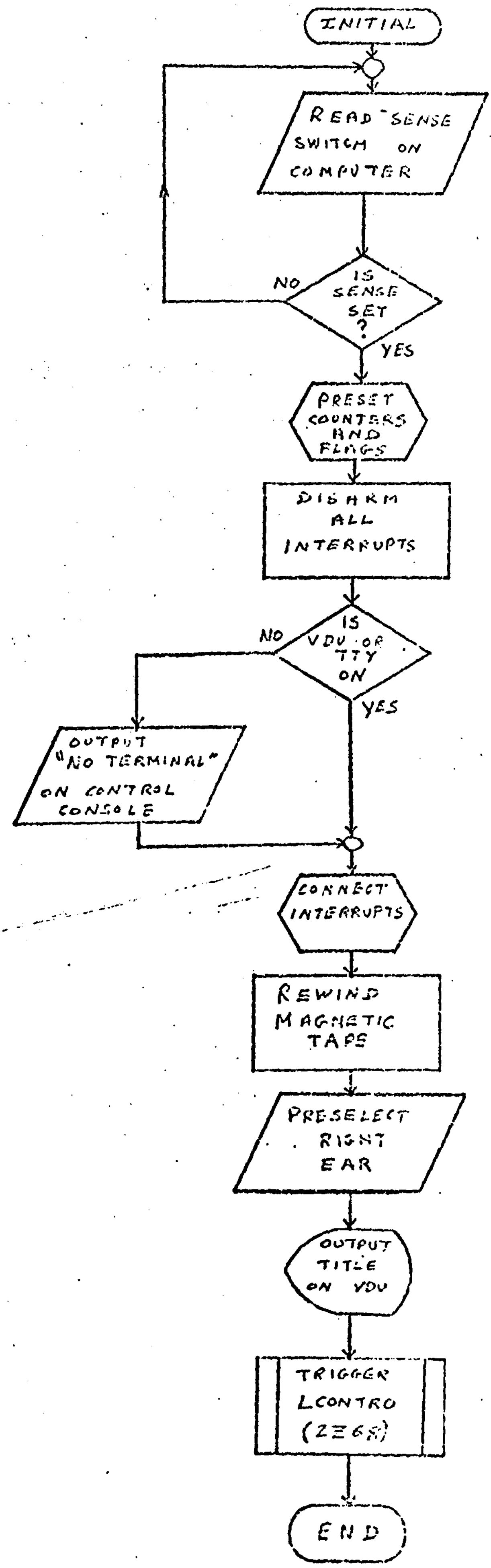
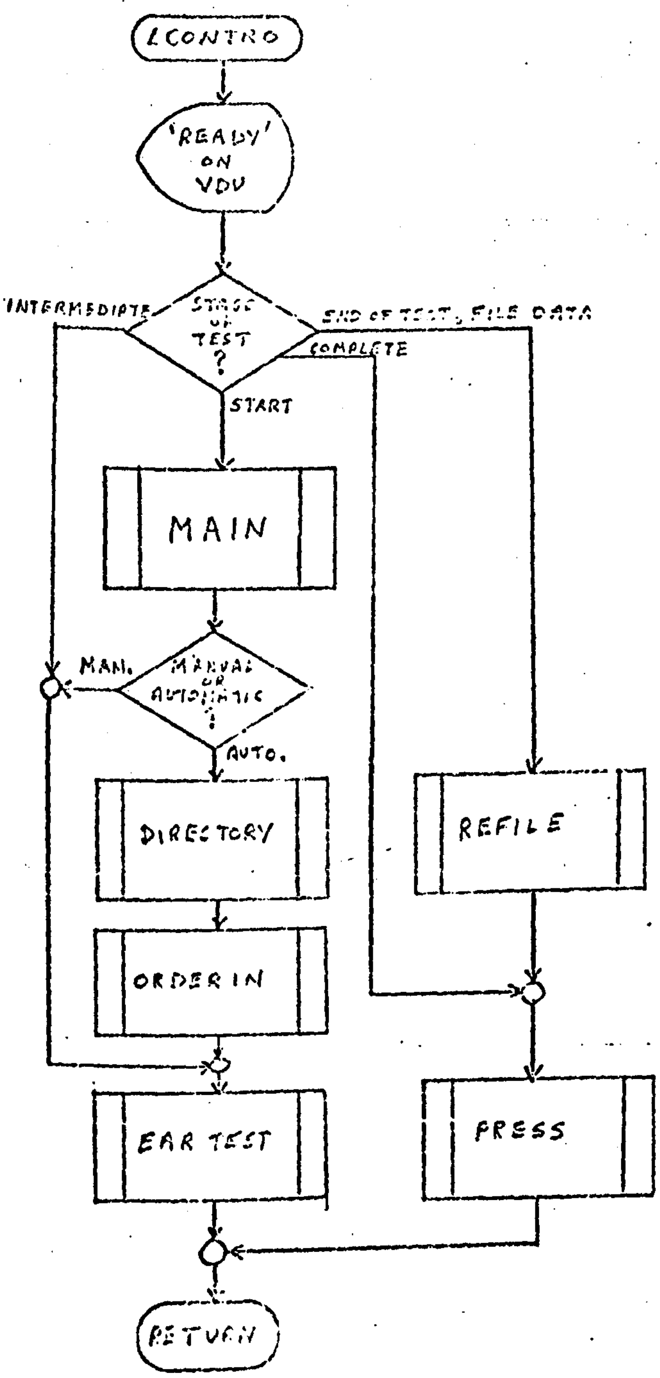

INITIAL
READ SENSE SWITCH ON COMPUTER
IS SENSE SET ?
NO
YES
PRESET COUNTERS AND FLAGS
DISARM ALL INTERRUPTS
IS VDU OR TTY ON
NO
YES
OUTPUT "NO TERMINAL" ON CONTROL CONSOLE
CONNECT INTERRUPTS
REWIND MAGNETIC TAPE
PRESELECT RIGHT EAR
OUTPUT TITLE ON VDU
TRIGGER LCONTRO (2E68)
END

Fig. B.1


LCONTRO
'READY' ON VDU
STAGE OF TEST ?
INTERMEDIATE
END OF TEST, FILE DATA
COMPLETE
START
MAIN
MANUAL OR AUTOMATIC ?
MAN.
AUTO.
DIRECTORY
ORDER IN
EAR TEST
REFILE
PRESS
RETURN

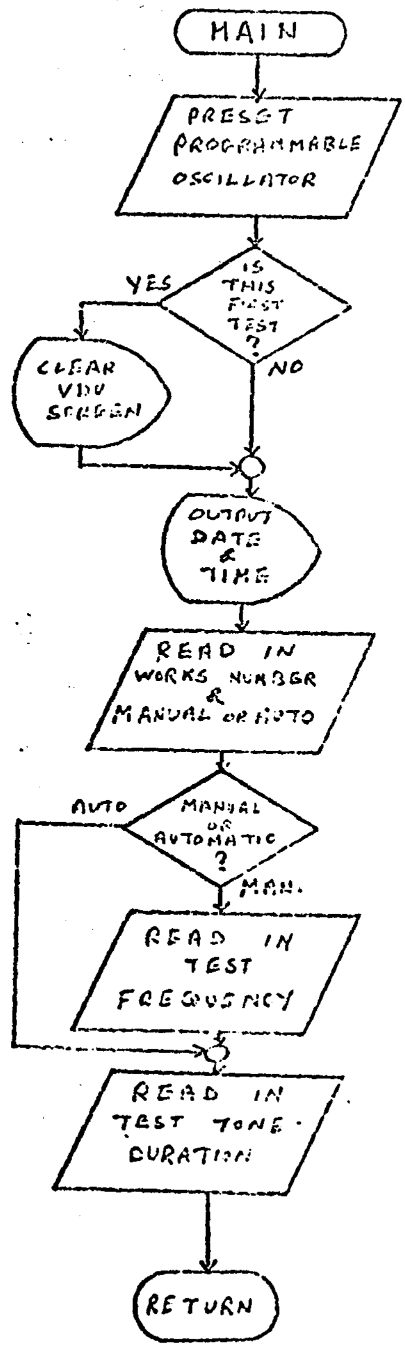

MAIN
PRESET PROGRAMMABLE OSCILLATOR
IS THIS FIRST TEST ?
YES
NO
CLEAR VDU SCREEN
OUTPUT DATE & TIME
READ IN WORKS NUMBER & MANUAL OR AUTO
MANUAL OR AUTOMATIC ?
AUTO
MAN.
READ IN TEST FREQUENCY
READ IN TEST TONE DURATION
RETURN

Fig. B.2

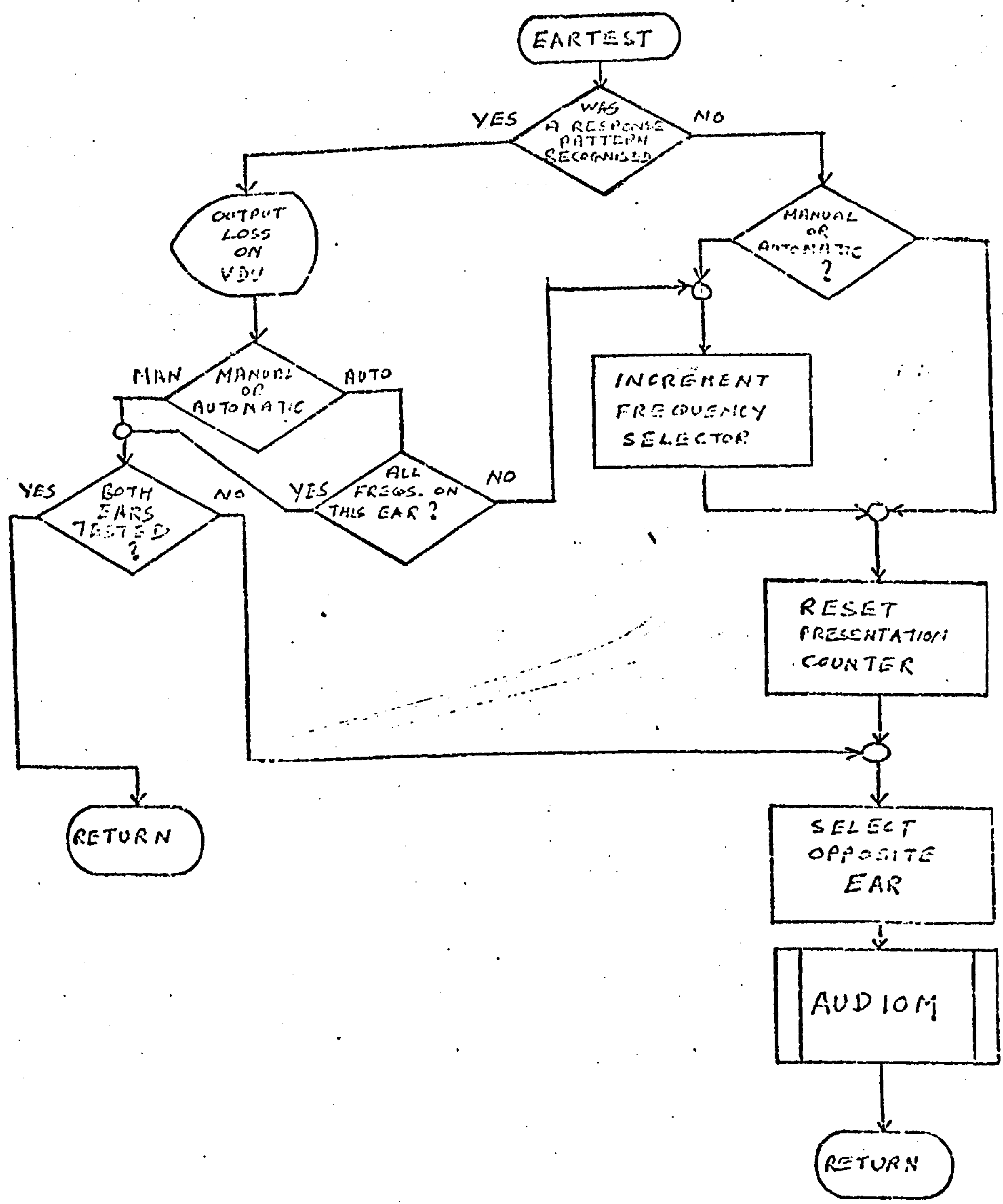
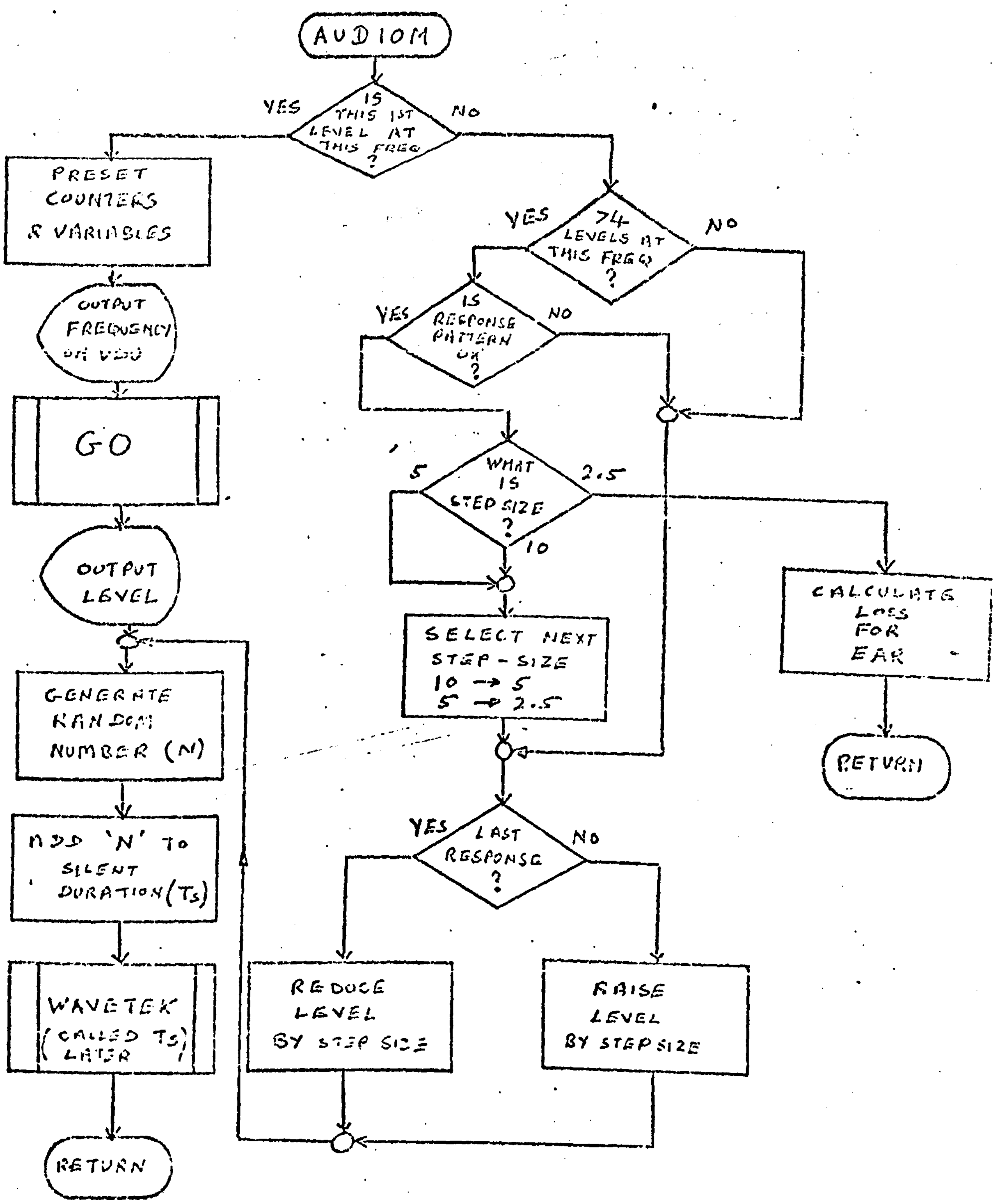

EARTEST
WAS A RESPONSE PATTERN RECOGNISED
YES
NO
OUTPUT LOSS ON VDU
MANUAL OR AUTOMATIC ?
MAN
MANUAL OR AUTOMATIC
AUTO
INCREMENT FREQUENCY SELECTOR
BOTH EARS TESTED ?
YES
NO
ALL FREQS. ON THIS EAR ?
YES
NO
RESET PRESENTATION COUNTER
RETURN
SELECT OPPOSITE EAR
AUDIOM
RETURN

Fig. B.3



AUDIOM
IS THIS 1ST LEVEL AT THIS FREQ ?
YES
NO
PRESET COUNTERS & VARIABLES
OUTPUT FREQUENCY ON VDU
GO
OUTPUT LEVEL
GENERATE RANDOM NUMBER (N)
ADD 'N' TO SILENT DURATION(Ts)
WAVETEK (CALLED Ts LATER)
RETURN
74 LEVELS AT THIS FREQ ?
IS RESPONSE PATTERN OK ?
WHAT IS STEP SIZE ?
5
2.5
10
SELECT NEXT STEP-SIZE
10 → 5
5 → 2.5
CALCULATE LOSS FOR EAR
RETURN
LAST RESPONSE ?
REDUCE LEVEL BY STEP SIZE
RAISE LEVEL BY STEP SIZE

Fig. B.4

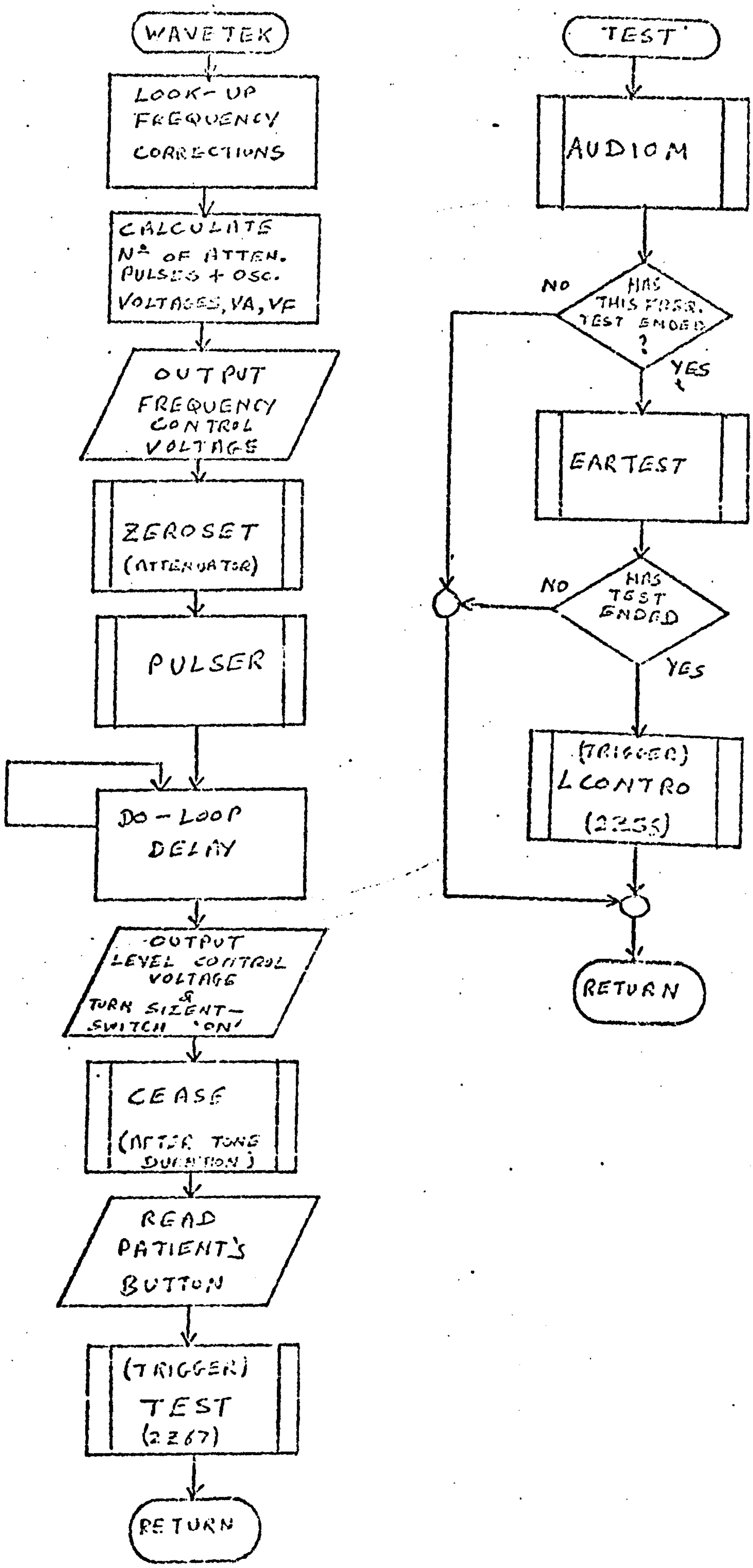

WAVETEK
LOOK-UP FREQUENCY CORRECTIONS
CALCULATE Nº OF ATTEN. PULSES + OSC. VOLTAGES, VA, VF
OUTPUT FREQUENCY CONTROL VOLTAGE
ZEROSET (ATTENUATOR)
PULSER
DO-LOOP DELAY
OUTPUT LEVEL CONTROL VOLTAGE & TURN SILENT-SWITCH 'ON'
CEASE (AFTER TONE DURATION)
READ PATIENT'S BUTTON
(TRIGGER) TEST (2267)
RETURN
TEST
AUDIOM
HAS THIS FREQ. TEST ENDED ?
NO
YES
EARTEST
HAS TEST ENDED
NO
YES
(TRIGGER) LCONTRO (2255)
RETURN

Fig. B.5

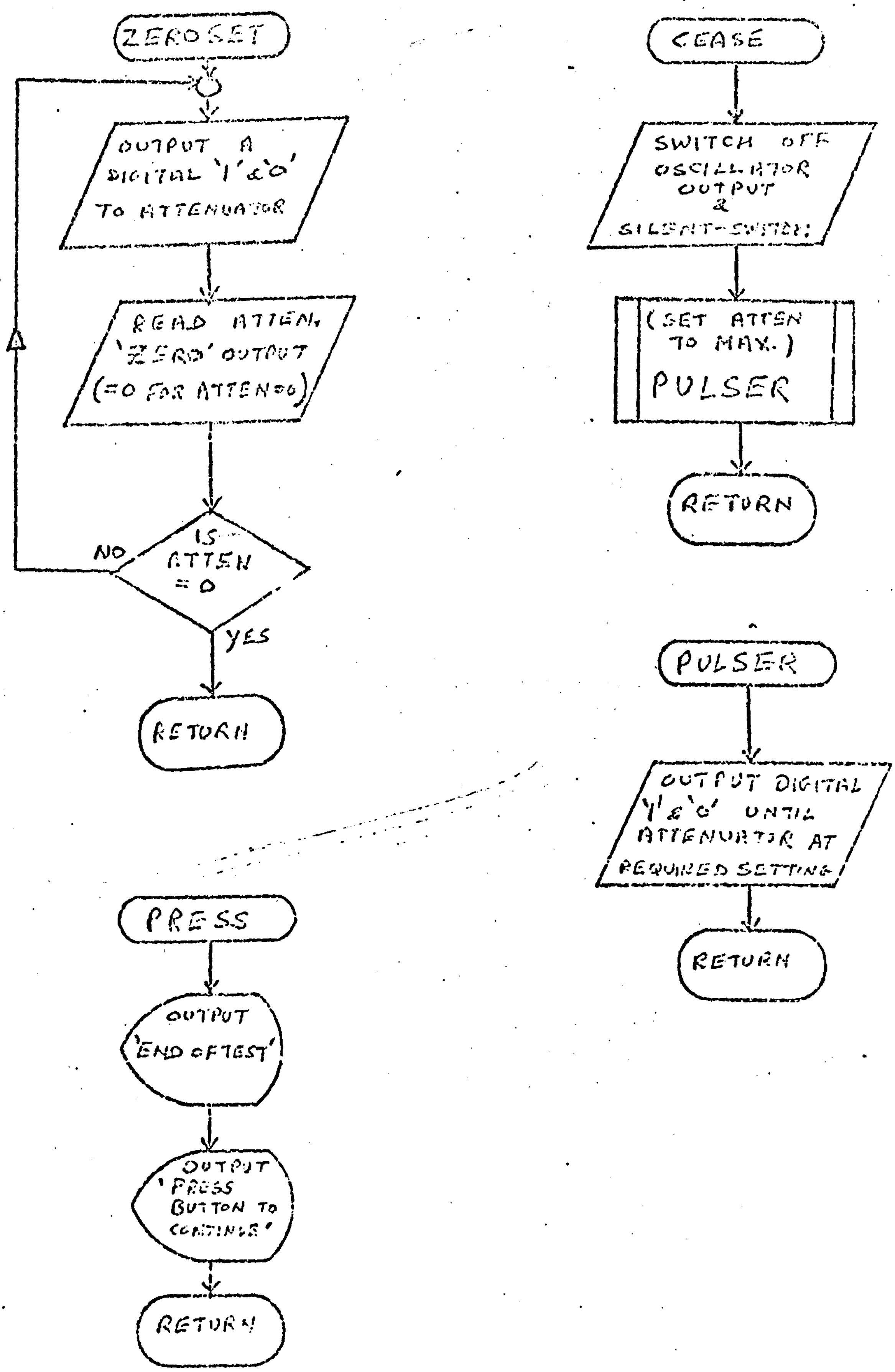

ZEROSET
OUTPUT A DIGITAL '1' & '0' TO ATTENUATOR
READ ATTEN. 'ZERO' OUTPUT (=0 FOR ATTEN=0)
IS ATTEN = 0
NO
YES
RETURN
CEASE
SWITCH OFF OSCILLATOR OUTPUT & SILENT-SWITCH;
(SET ATTEN TO MAX.)
PULSER
RETURN
PULSER
OUTPUT DIGITAL '1' & '0' UNTIL ATTENUATOR AT REQUIRED SETTING
RETURN
PRESS
OUTPUT 'END OF TEST'
OUTPUT 'PRESS BUTTON TO CONTINUE'
RETURN

Fig. B.6

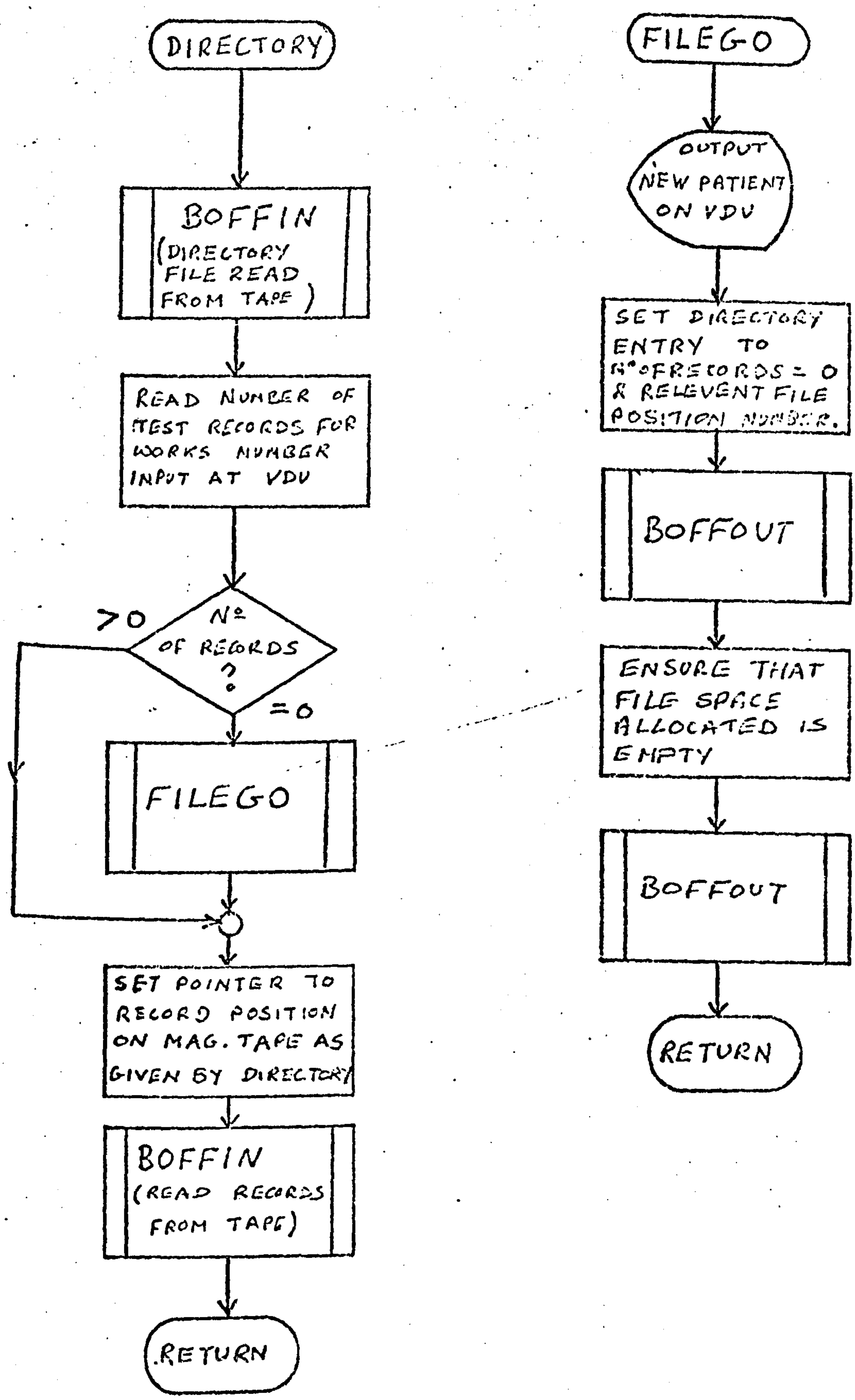

DIRECTORY
BOFFIN (DIRECTORY FILE READ FROM TAPE)
READ NUMBER OF TEST RECORDS FOR WORKS NUMBER INPUT AT VDU
>0
No OF RECORDS ?
=0
FILEGO
SET POINTER TO RECORD POSITION ON MAG. TAPE AS GIVEN BY DIRECTORY
BOFFIN (READ RECORDS FROM TAPE)
RETURN
FILEGO
OUTPUT NEW PATIENT ON VDU
SET DIRECTORY ENTRY TO No OF RECORDS = 0 & RELEVENT FILE POSITION NUMBER.
BOFFOUT
ENSURE THAT FILE SPACE ALLOCATED IS EMPTY
BOFFOUT
RETURN

Fig. B.7

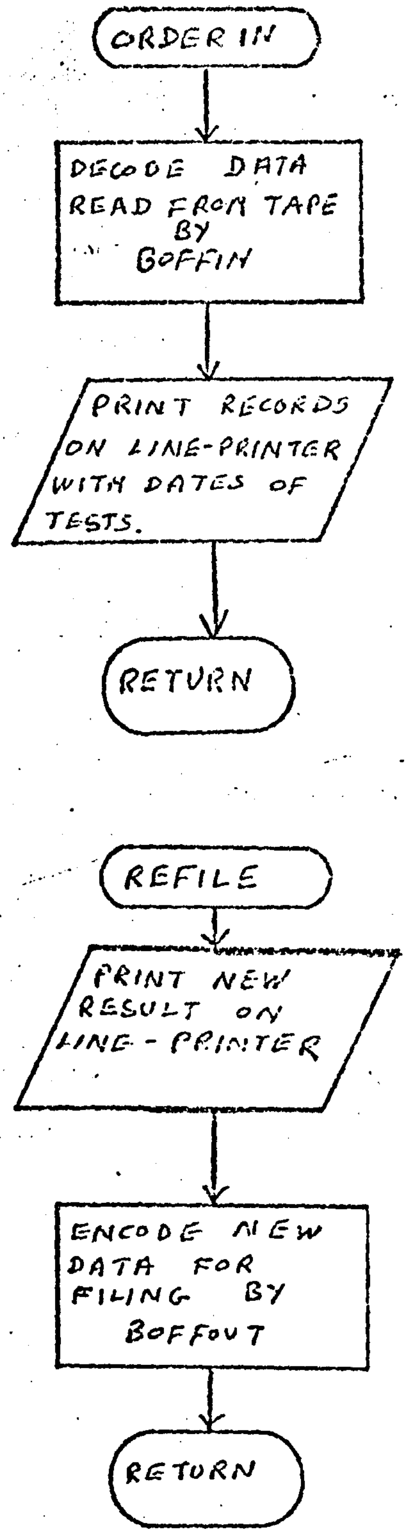

ORDER IN
DECODE DATA READ FROM TAPE BY BOFFIN
PRINT RECORDS ON LINE-PRINTER WITH DATES OF TESTS.
RETURN
REFILE
PRINT NEW RESULT ON LINE-PRINTER
ENCODE NEW DATA FOR FILING BY BOFFOUT
RETURN

Fig. B.8

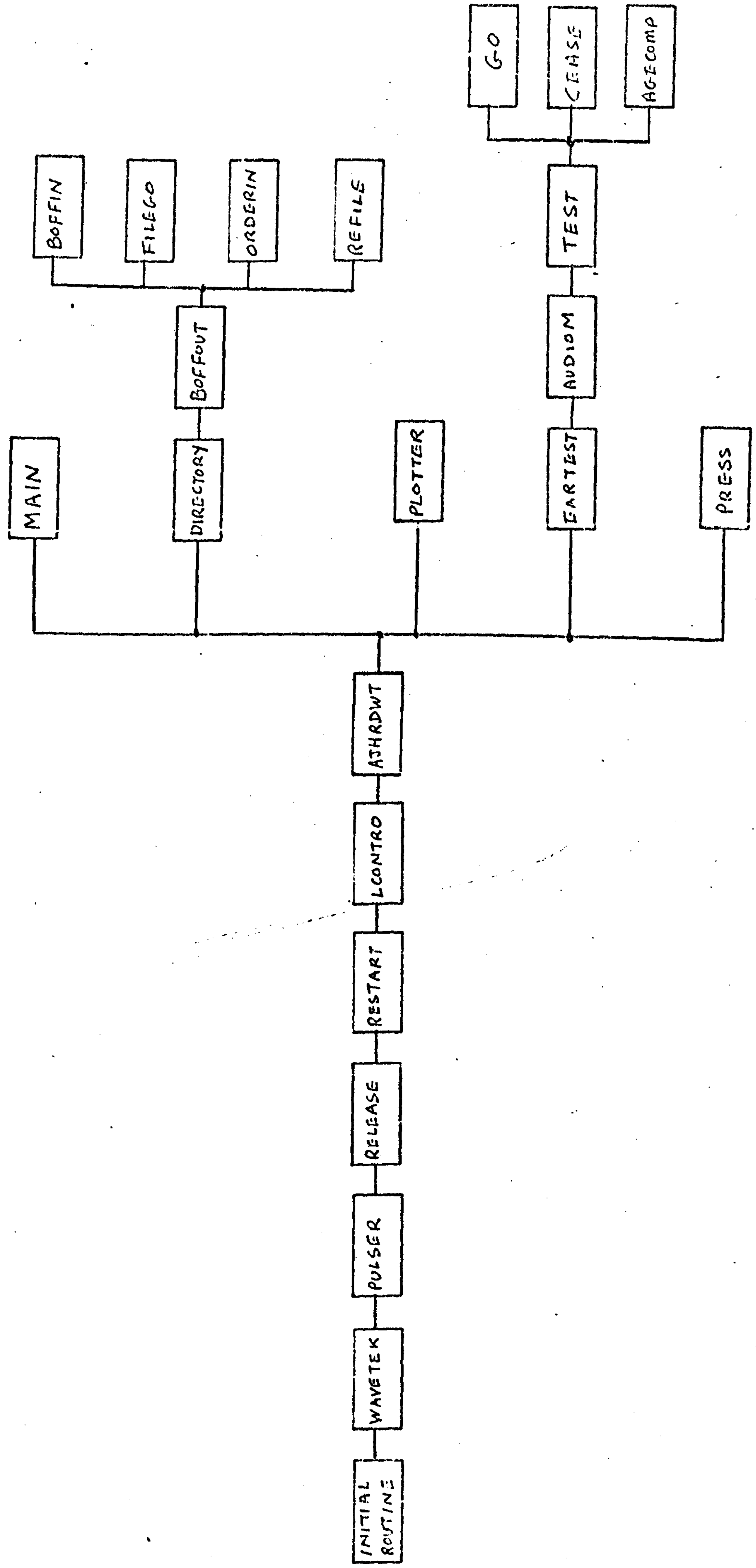


Fig. B.9 AUDIOMETER PROGRAMME OVERLAY STRUCTURE

```
SU2EAR T=00003 IS ON CR00002 USING 00024 BLKS R=0000

!JOB LLOWE,AUDIOMETER
!ATTEND
!FORTRANH RT,GO,LS,S
C
C
C
C
C
C
C
C
C     THIS PROGRAM IS INTENDED TO CONTROL THE COMPUTER
C     CONTROLLED AUDIOMETER.THE PROGRAM IS WRITTEN IN
C     FORTRAN-IV H AND EMBEDDED MACRO-SYMBOL.IT IS
C     INTENDED TO BE RUN ON A RANK XEROX SIGMA 5 COMPUTER.
C
C
C
C
C
C
C
      DIMENSION FVOLT(6),INIT(10)
      DATA INIT/'C O M P U T E R ',' A U D I O M E T',' R I C S'/
      COMMON/B/L,I,K,ILIN
      COMMON/CGO/JE
      COMMON/ORD/NF,NT,THREPR(10,6),THREPL(10,6),DATE(10)
      COMMON/DT/IDATE
      COMMON/MAYN/ITEST,JF
      COMMON/TIM/ITIME(10)
      COMMON/GEN/NA,NB,TAPE,JRS,NR,WN
      COMMON/M/M,IRST
      COMMON/PHASE/IP
      COMMON/ER/IM
      COMMON/WAVE/IDELAY,IAMP,ICOUNT
      COMMON/NOT/NOTE
      COMMON/JOT/JO
C     SENSE WILL BE 1FOR TELETYPE RUNNING AND 2FOR VDU
      NOTE=1
  103 CALL SENSE(ILINE)
      IF(ILINE.EQ.1.OR.ILINE.EQ.2)GO TO 104
      PAUSE'SET SENSE TO 1 FOR TTY OR 2 FOR VDU'
      GO TO 103
  104 L=ILINE
      ILIN=L+2
      JO=ILIN
      IM=0
      ISTAT=0
      TAPE=1
      RAD=2
      NA=1
      NB=1
C  DISARMS ANY INTERUPTS LEFT CONNECTED BY THE PREVIOUS USER
S         WD,10        X'1102'
S         LI,10       X'FFE0'
  101 CALL CSTART(L,ISTAT)
      IF(ISTAT.NE.1.AND.ISTAT.NE.4)GO TO 102
      PAUSE'VDU NOT ON'
```

```
      GO TO 101
  102 CONTINUE
      CALL CSET(L,132,0)
      CALL ALPHA
      CALL HOME
      CALL ERASE
      CONNECT(2Z60,RELEASE)
      CONNECT(2Z61,RESTART)
      CONNECT(2Z64,CEASE)
      CONNECT(2Z65,WAVETEK)
      CONNECT(2Z67,TEST)
      CONNECT(2Z68,LCONTRO)
      CONNECT (2Z69,PLOTTER)
      REWIND TAPE
C
C        ENSURES THAT RELAY 4 IS CLOSED
S        LW,9       =8Z00800000
S        WD,9       X'C224'
C
      M=0
      CALL CWRITE(L,8Z19404040,0,4,ISTAT)
      CALL CWRITE(L,INIT,0,40,ISTAT)
      CALL CWRITE(L,8Z15150707,0,4,ISTAT)
C
      CALL CWRITE(L,8Z15154040,0,4,ISTAT,2Z68)
C
      STOP
      END
C
C
C
      SUBROUTINE WAVETEK
C THIS SUBROUTINE CONTROLS THE WAVETEK 136 SIGNAL GENERATOR
C
      DIMENSION COR(6)
      DATA COR/10.0,23.0,27.5,24.5,25.5,0.0/
      COMMON/B/L,I,K,ILIN
      COMMON/C/F(6),THRENC(6),TNCWA(6)
      COMMON/WTEK/DB
      COMMON/WAVE/IDELAY,IAMP,ICOUNT
      COMMON/TIM/ITIME(10)
      COMMON/CES/IPUL
      CALL DISAB (2Z65)
      IF(IDELAY.NE.-1)CALL ENAB(2Z64)
      ICOUNT=0
      ATT=52.5-DB
      IPUL=INT(ATT/2.5+0.5)
      VA=(10.0**(-COR(I)/20.0)*20.0-10.0)
      VF=-(10.0*(F(I)/8000.0)+0.04)
C
C        PRESENTS FREQUENCY PROGRAMMING VOLTAGE FOR WTK
      CALL RDAC(4,VF)
C
C        SETS ATTENUATOR TO ZERO
      CALL ZEROSET
C
C         SETS ATTENUATOR TO REQUIRED VALUE
      CALL PULSER
C
      DO 100 JD=1,10000
```

```
 100   CONTINUE
C        PRESENTS PROGRAMMING VOLYAGE FOR COMPENSATED
C        WAVETEK OUTPUT.
      CALL RDAC(2,VA)
C       SWITCHES ELECTRONIC SWITCH ON
      CALL RDAC(3,8.0)
      CALL TIMER(ITIME(1),2Z64)
 10   CONTINUE
S        RD,9       X'C208'
S        STW,9      IVAR
      IF(IVAR.LT.0)ICOUNT=ICOUNT-1
      IF(IDELAY.EQ.-1)GO TO 11
      GO TO 10
 11   CONTINUE
      IDELAY=0
      CALL TIMER(100000000,2Z64)
      CALL TRIGR(2Z67)
      RETURN
      END
C
C
C
!FORTRANH LS,GO,S
C
      SUBROUTINE PULSER
C
      COMMON/CES/IPUL
      IF(IPUL.EQ.0)GO TO 10
      DO 10 I=1,IPUL
S        LW,9       =8Z80000000
S        WD,9       X'C218'
S        LW,9       =8Z00000000
S        WD,9       X'C218'
 10   CONTINUE
      RETURN
C
      ENTRY ZEROSET
C
 20   CONTINUE
S        LW,9       =8Z80000000
S        WD,9       X'C218'
      DO 30 J=1,1000
 30   CONTINUE
S        LW,9       =8Z00000000
S        WD,9       X'C218'
      DO 40 J=1,1000
 40   CONTINUE
      CALL RADCS(1,V,10.0,1)
      IF(V.LT.3.8)GO TO 20
      RETURN
      END
C
C
C
      SUBROUTINE RELEASE
      COMMON/WAVE/IDELAY,IAMP,ICOUNT
C        ABORT ROUTINE
      ISTAT=1
      IDELAY=-1
      CALL DISAB(2Z60,2Z61,2Z64,2Z65,2Z67,2Z68,2Z69)
```

```
      CALL WAVETEK
S                CAL1,9    8
      RETURN
      END
C
C
C
      SUBROUTINE RESTART
C RESTARTS THE PROGRAM FROM THE VERY BEGINNING
C REWINDING THE RAD FILE  "LLEARS".
      COMMON/M/M,IRST
      COMMON/WAVE/IDELAY,IAMP,ICOUNT
      M=0
      IDELAY=-1
      IRST=1
      CALL DISAB(2Z67)
       CALL ENAB(2Z68)
      CALL TRIGR(2Z68)
      RETURN
      END
C
C
C
      SUBROUTINE LCONTRO
C THIS SUBROUTINE HOLDS CONTROL OF THE OVERALL ACTION
C  OF THE AUDIOMETER.
      COMMON/C/F(6),THRENC(6),TNCWA(6)
      COMMON/M/M,IRST
      COMMON/MAYN/ITEST,JF
      DATA F/250.,500.,1000.,2000.,4000.,8000./
      CALL COCINIT(L)
      IRST=0
      GO TO (12,10,14),M
      COMMON/B/L,I,K,ILIN
      CALL WTNL(25,'AUDIOMETER READY FOR TEST')
      CALL WTNL(22,'TYPE IN DATA REQUESTED')
      CALL SEGLOD(16)
      CALL MAIN
      IF(ITEST.EQ.2) GO TO 12
      CALL WTNL(16,'LOOKING FOR FILE')
      CALL SEGLOD ( 1)
      CALL DIRECTORY
      CALL SEGLOD (5)
      CALL ORDERIN
 12   CONTINUE
      CALL SEGLOD (7)
      CALL EARTEST
      CALL SEGLOD(11)
      RETURN
 10   CONTINUE
      CALL SEGLOD(2)
      CALL SEGLOD(3)
      CALL SEGLOD (13)
      CALL REFILE
 14   CONTINUE
      CALL SEGLOD(15)
      CALL PRESS
      CALL SEGLOD(14)
      RETURN
      END
```

```
C
C
C
!ASSIGN (M:CI,DI,AJHRDWT)
!MACRSYM CI,GO
!FORTRANH GO,S,LS
C
C
      SUBROUTINE MAIN
C   SUBROUTINE TO INPUT THE INITIAL DATA NECESSARY
C   TO PERFORM THE TEST.(WN,DATE,ETC.)
      DIMENSION IDT(2)
      INTEGER WN
      INTEGER TIME(4)
      COMMON/B/L,I,K,ILIN
      COMMON/GEN/NA,NB,TAPE,JRS,NR,WN
      COMMON/CGO/JE
      COMMON/ORD/NF,NT,THREPR(10,6),THREPL(10,6),DATE(10)
      COMMON/DT/IDATE
      COMMON/ER/IM
      COMMON/MAYN/ITEST,JF
      COMMON/TIM/ITIME(10)
      INTEGER IOB(6),PERIOD(4,2),ITIM(10),FREQ(4)
      IM=IM+1
      CALL RDAC(2,0,0,4,-10,0)
      ITEST=0
      ISTAT=0
      IF(IM.EQ.1)GO TO 31
      CALL ERASE
      CALL HOME
 31    CONTINUE
C
C  OUTPUTS THE DATE AND TIME ON THE OPERATING
C  COMMUNICATIONS UNIT.
S        LI,9      TIME
S        AW,9      =8Z10000000
S        CAL1,8    9
      CALL CWRITE(L,2Z15,3,1,ISTAT)
      CALL CWRITE(L,TIME,0,16,ISTAT)
C
C    READS IN WORKS NUMBER
      CALL WTNL
      CALL RDNL(15,'TYPE WORKS NO. ',WN)
      CALL CWRITE(L,8Z15150707,0,4,ISTAT)
C
C    READS IN THE DATE
      CALL CWRITE(L,8Z07404007,0,4,ISTAT)
      CALL RDNL(10,'TYPE DATE ',IDATE)
C
C  ASKS WHETHER OPERATION IS TO BE AUTO. OR MANUAL.
      CALL CWRITE(L,8Z07404007,0,4,ISTAT)
 33   CONTINUE
      CALL CHOOSE(16,'TYPE MAN OR AUTO',1,'AUTO,MANUAL;',ITEST)
      IF(ITEST.EQ.0)GO TO 33
      CALL WTNL
C     ITEST=1 FOR AUTOMATIC RUNNING AND 2 FOR MANUAL RUNNING
      IF(ITEST.EQ.1)GO TO 70
C
C    READS IN THE VALUE OF FREQUENCY REQUIRED FOR THE TEST
      JF=0
```

```
 32   CONTINUE
      CALL CHOOSE(39,'TEST FREQ.= 250,500,1000,2000,4000,8000',2,'250,50
     &0,1000,2000,4000,8000;',JF)
      IF(JF.EQ.0)GO TO 32
 70   CONTINUE
C
C         TIME ITIME(2) SETS THE SILENT PERIOD
C         ITIME(1) SETS THE DURATION OF THE TONE
      CALL CWRITE(L,8Z15150707,0,4,ISTAT)
      CALL RDNL(11,'TONE DUR.= ',ITIM(1))
      CALL RDNL(11,'REST DUR.= ',ITIM(2))
      DO 72 ICR=1,2
 72   ITIME(ICR)=ITIM(ICR)*2
      RETURN
      END
C
C
C
      SUBROUTINE DIRECTORY
C  LOOKS AT THE DIRECTORY ON MAG.TAPE,DECIDES IF A RECORD
C  EXISTS FOR THE WORKS NO. IF "YES" IT FINDS AND READS
C  IT,IF"NO" IT CALLS FILEGO.
C
C  NR=NO. OF RECORDS       NF=POSITION OF RECORD ON TAPE.
      COMMON/GEN/NA,NB,TAPE,JRS,NR,WN
      INTEGER WN
      COMMON/FILE/A(250)
      COMMON/ORD/NF,NT,THREPR(10,6),THREPL(10,6),DATE(10)
      JRS=0
      CALL SEGLOD(2)
      CALL SEGLOD(3)
      CALL BOFFIN
C READS FIRST RECORD OFF MAG.TAPE.FIRST WORD IS THE
C NUMBER OF RECORDS.
      NR=A(1)
C   WORKS NUMBER RECORD POSITIONS ARE GIVEN BY THE (WN+1)TH
C    MEMBER OF THE ARRAY.(E.G WN=45 WILL BE IN POSITION 46).
      NF=A(WN+1)
      IF(NF.NE.0)GO TO 2
      CALL SEGLOD (3)
      CALL SEGLOD(4)
      CALL FILEGO
 2    CONTINUE
      JRS=NF
      CALL SEGLOD(2)
      CALL BOFFIN
      RETURN
      END
C
C
C
      SUBROUTINE BOFFOUT
C CALLS RECSKIP TO THE REQUIRED RECORD BUFFERS IT OUT
C AND REWINDS THE MAG. TAPE.
      COMMON/GEN/NA,NB,TAPE,JRS,NR,WN
      INTEGER WN
      COMMON/FILE/A(250)
      IF(JRS.EQ.0)GO TO 12
      JRS2=2*JRS
      CALL RECSKIP(JRS2)
```

```
 12   CONTINUE
      CALL BUFFER OUT(NA,NB,A,250,I)
 10   GO TO (10,11),I
 11    CONTINUE
      REWIND TAPE
      RETURN
      END
C
C
C
      SUBROUTINE BOFFIN
C CALLS RECKSKIP TO THE REQUIRED RECORD,BUFFERS
C IT IN AND REWINDS THE MAG. TAPE.
C
C BECAUSE EACH RECORD IS SEPERATED FROM THE NEXT BY A
C BLANK FILE,ALL SKIPPING MUST BE TO 2*NF.
      COMMON/GEN/NA,NB,TAPE,JRS,NR,WN
      INTEGER WN
      COMMON/FILE/A(250)
      JRS2=2*JRS
      CALL RECSKIP(JRS2)
      CALL BUFFER IN(NA,NB,A,250,I)
 10   GO TO (10,11),I
 11   CONTINUE
      REWIND TAPE
      RETURN
      END
C
C
C
      SUBROUTINE FILEGO
C UPDATES THE DIRECTORY FILE AND CALLS "BOFFOUT" WITH
C A BLANK FILE FOR WN.
      COMMON/B/L,I,K,ILIN
      COMMON/GEN/NA,NB,TAPE,JRS,NR,WN
      INTEGER WN
      COMMON/FILE/A(250)
      COMMON/ORD/NF,NT,THREPR(10,6),THREPL(10,6),DATE(10)
      CALL CWRITE(L,5Z15150707,0,4,ISTAT)
      CALL WTNL(11,'NEW PATIENT')
      NR=NR+1
      A(1)=NR
      A(WN+1)=NR
      NF=NR
      CALL BOFFOUT
      JRS=NR
      DO 10 I=1,250
 10    A(I)=0.
      CALL BOFFOUT
      RETURN
      END
C
C
C
      SUBROUTINE ORDERIN
C        ORDERS THREPR,THREPL,DATE FROM THR ARRAY A(I)
      COMMON/GEN/NA,NB,TAPE,JRS,NR,WN
      INTEGER WN,DATE
      COMMON/FILE/A(250)
      COMMON/ORD/NF,NT,THREPR(10,6),THREPL(10,6),DATE(10)
```

```
      NT=A(1)
      WRITE(108,103)NT
      I=1
      IF(NT.EQ.0)GO TO 501
      DO 300 J1=1,NT
      DO 300 K1=1,6
      I=I+1
      THREPR(J1,K1)=A(I)
 300   CONTINUE
      DO 200 J2=1,NT
      DO 200 K2=1,6
      I=I+1
      THREPL(J2,K2)=A(I)
 200   CONTINUE
      DO 400 J3=1,NT
      I=I+1
      DATE(J3)=A(I)
 400  CONTINUE
C
C  THIS SECTION SORTS THE DATA INTO LOGICAL TEST RESULT
C  ORDER FOR OUTPUTTING TO THE VDU OR LINE-PRINTER.
      K=1
      WRITE(108,100)WN
      WRITE(108,201)
 201  FORMAT(//,25X,6H RIGHT,52X,5H LEFT,25X,5H DATE,/)
      DO 40 L=1,NT
      JL=K+1
      K=JL+5
      JR=K+1+(NT-1)*6
      JR2=JR+5
      WRITE(108,202)(A(I),I=JL,K),(A(I),I=JR,JR2),(A(NT*12+L+1))
 202  FORMAT(1X,6F8.1,8X,6F8.1,8X,I6)
 40   CONTINUE
      WRITE(108,105)
 105  FORMAT(/,52X,14H *************  )
 100  FORMAT(/////,8X,20H WORKS RECORD NUMBER   ,2X,I)
C
 103  FORMAT( 'NUMBER OF TESTS RECORDED = ',I)
  501 RETURN
      END
C
C
C
      SUBROUTINE REFILE
C  ORDERS THE ARRAY B(J) FROM THE VALUES OF THREPR,THREPL,DATE
      COMMON/GEN/NA,NB,TAPE,JRS,NR,WN
      COMMON/FILE/A(250)
      COMMON/ORD/NF,NT,THREPR(10,6),THREPL(10,6),DATE(10)
      COMMON/DT/IDATE
      INTEGER WN,DATE
      A(1)=NT
      J=1
      DO 10 I1=1,NT
       DO 10 K1=1,6
      J=J+1
      A(J)=THREPR(I1,K1)
 10   CONTINUE
      DO 20 I2=1,NT
      DO 20 K2=1,6
       J=J+1
```

```
      A(J)=THREPL(I2,K2)
   20 CONTINUE
      DATE(NT)=IDATE
       DO 30 I3=1,NT
       J=J+1
      A(J)=DATE(I3)
   30 CONTINUE
      JRS=NF
      CALL BOFFOUT
C
C     THIS SECTION SORTS THE DATA INTO LOGICAL TEST RESULT
C     ORDER FOR OUTPUTTING TO THE MAG.TAPE.
      K=1
      WRITE(108,100)WN
      WRITE(108,201)
  201 FORMAT(//,25X,6H RIGHT,52X,5H LEFT,25X,5H DATE,/)
      DO 40 L=1,NT
      JL=K+1
      K=JL+5
      JR=K+1+(NT-1)*6
      JR2=JR+5
      WRITE(108,200)(A(I),I=JL,K),(A(I),I=JR,JR2),(A(NT*12+L+1))
  200 FORMAT(1X,6F8.1,8X,6F8.1,8X,I6)
   40 CONTINUE
      WRITE(108,105)
  105 FORMAT(/,52X,14H ************* )
  100 FORMAT(/////,8X,20H WORKS RECORD NUMBER  ,2X,I)
C
       RETURN
       END
C
C
C
      SUBROUTINE PLOTTER
C    "PLOTTER" IS ON 2Z69,PRODUCES AUDIOGRAMS FOR BOTH EARS
C    ON THE VDU.
      DIMENSION FO(6),ITF(4),IF(10)
      INTEGER TR(2)/8ZD9C9C7C8,8ZE3404040/
      INTEGER IPA(9)/8Z19D740C1,8Z40E340C9,8Z40C540D5,8Z40E34040,8Z4040
     &C140,8ZE440C440,8ZC940D640,8ZC7400940,8ZC140D41F/
      COMMON/JOT/ILIN
      INTEGER WN
      COMMON/DT/IDATE
      COMMON/GEN/NA,NB,TAPE,JRS,NR,WN
      COMMON/ORD/NF,NT,THREPR(10,6),THREPL(
      DATA ITF/'FREQUENCY (KHZ) '/
      DATA FO/0.2,0.5,1.0,2.0,4.0,8.0/
      L=ILIN-2
      CALL CSTART(L,ISTAT)
      CALL CSET(L,132,0)
      CALL ERASE
      ISTAT=0
C
C     PEN TO START POSITION FOR LHS GRAPH
      CALL TPLOT(0,100,400)
C
C PLOTS 60 DOTS ACCROSS SCREEN TO SHO THRESHOLD
C NORMAL LHS GRAPH.
      IXD=100
      IYD=400
```

```
      DO 220 JD=1,60
      CALL TPLOT(1,IXD,IYD)
      IXD=IXD+5
  220 CALL TPLOT(0,IXD,IYD)
C
C     DRAWS VERTICAL AXIS  LHS GRAPH
      CALL TPLOT(0,100,450)
      CALL TPLOT(-1,100,200)
C
C     DRAWS HORIZONTAL AXIS WITH MARKER BLIPS
      IYO=195
      IX=100
      DO 110 JX=1,6
      IX=IX+50
      CALL TPLOT(-1,IX,200)
      CALL TPLOT(-1,IX,IYO)
  110 CALL TPLOT(0,IX,200)
C
C     PLOTS 1ST POINT WITH AN 0
      IY=(400-2*THREPR(NT,1))
      IX=50*1+100
      CALL TPLOT(0,IX,IY)
      CALL ALPHA
      CALL CWRITE(L,2Z96,3,1,ISTAT)
C
C     PLOTS NEXT FIVE WITH AN 0
      DO 100 JP=2,6
      IX=50*JP+100
      IY=(400-2*THREPR(NT,JP))
      CALL TPLOT(0,IX,IY)
      CALL ALPHA
      CALL CWRITE(L,2Z96,3,1,ISTAT)
  100 CONTINUE
C
C     PEN TO START POSITION FOR RHS GRAPH
      CALL TPLOT(0,600,400)
C
C     PLOTS 60 DOTS FOR RHS GRAPH
      IXD=600
      DO 320 JD=1,60
      CALL TPLOT(1,IXD,IYD)
      IXD=IXD+5
  320 CALL TPLOT(0,IXD,IYD)
C
C     DRAWS VERT AXIS  RHS GR...
      CALL TPLOT(0,600,450)
      CALL TPLOT(-1,600,200)
C
C     DRAWS HORIZ AXIS WITH BLIPS  RHS GRAPH
      IX=600
      DO 210 JXL=1,6
      IX=IX+50
      CALL TPLOT(-1,IX,200)
      CALL TPLOT(-1,IX,IYO)
  210 CALL TPLOT(0,IX,200)
C
C     PLOTS 1ST POINT WITH AN X
      IY=(400-2*THREPL(NT,1))
      IX=50*1+600
      CALL TPLOT(0,IX,IY)
```

```
      CALL ALPHA
      CALL CWRITE(L,2ZA7,3,1,ISTAT)
C
C     PLOTS NEXT FIVE WITH AN X
      DO 200 JPL=2,6
      IY=(400-2*THREPL(NT,JPL))
      IX=50*JPL+600
      CALL TPLOT(0,IX,IY)
      CALL ALPHA
      CALL CWRITE(L,2ZA7,3,1,ISTAT)
 200  CONTINUE
C
C     OUTPUTS  LEFT  OVER RHS GRAPH
      CALL TPLOT(0,640,500)
      CALL ALPHA
      CALL CWRITE(L,8ZD3C5C6E3,0,4,ISTAT)
C
C     OUTPUTS RIGHT OVER LHS GRAPH
      CALL TPLOT(0,200,500)
      CALL ALPHA
      CALL CWRITE(L,TR,0,8,ISTAT)
C
C     OUTPUTS HEADIND  PATIENT AUDIOGRAM
      CALL TPLOT(0,200,600)
      CALL ALPHA
      CALL CWRITE(L,IPA,0,36,ISTAT)
C
      CALL WTNL
      CALL WTNL(11,'WORKS NO.  ',WN)
      CALL WTNL(11,'TEST DATE  ',IDATE)
C
C     O/P +100 ON RHS GRAPH
      CALL TPLOT(0,550,200)
      CALL ALPHA
      CALL CWRITE(L,8Z4EF1F0F0,0,4,ISTAT)
C
C     O/P +100 ON LHS GRAPH
      CALL TPLOT(0,50,200)
      CALL ALPHA
      CALL CWRITE(L,8Z4EF1F0F0,0,4,ISTAT)
C
C     O/P 0DB ON LHS GRAPH
      CALL TPLOT(0,50,400)
      CALL ALPHA
      CALL CWRITE(L,8ZF084C24B,0,4,ISTAT)
C
C     O/P 0DB ON RHS GRAPH
      CALL TPLOT(0,550,400)
      CALL ALPHA
      CALL CWRITE(L,8ZF084C24B,0,4,ISTAT)
C
C     O/P FREQ AXIS LABELLING  LHS THEN RHS
      IXF=125
 250   CONTINUE
      DO 240 MF=1,6
      CALL TPLOT(0,IXF,175)
      CALL ALPHA
      IXF=IXF+50
      ENCODE(10,101,IF)FQ(MF)
 101  FORMAT(F3.1)
```

```
 240   CALL CWRITE(L,IF,0,10,ISTAT)
       IF(IXF.GT.700)GO TO 260
       IXF=625
       GO TO 250
C
C      O/P LABEL  FREQUENCY (KHZ)ON LHS  THEN  RHS
 260   CONTINUE
       CALL TPLOT(0,150,150)
       CALL ALPHA
       CALL CWRITE(L,ITF,0,16,ISTAT)
       CALL TPLOT(0,650,150)
       CALL ALPHA
       CALL CWRITE(L,ITF,0,16,ISTAT)
C
C      O/P  SPL  LHS  THEN  RHS
       CALL TPLOT(0,45,310)
       CALL ALPHA
       CALL CWRITE(L,8ZE2070348,0,4,ISTAT)
       CALL TPLOT(0,545,310)
       CALL ALPHA
       CALL CWRITE(L,8ZE2070348,0,4,ISTAT)
C
       CALL HOME
       CALL WTNL(26,'PRESS FOR RESTART OR ABORT')
       RETURN
       END
C
C
C
       SUBROUTINE EARTEST
       COMMON/M/M,IRST
       COMMON/HL/HLOSS
       COMMON/B/L,I,K,ILIN
       COMMON/C/F(6),THRENC(6),TNCWA(6)
       COMMON/BLOG/J
       COMMON/CGO/JE
       COMMON/ORD/NF,NT,THREPR(10,6),THREPL(10,6),DATE(10)
       COMMON/WAVE/IDELAY,IAMP,ICOUNT
       COMMON/MAYN/ITEST,JF
       COMMON/NOT/NOTE
C  J=1 ONLY IF ENTERING FROM SUB. TEST HAVING
C   SAMPLED A RESPONSE.
       IF(J.EQ.1)GO TO 20
       NT=NT+1
       IWRT=0
 4     I=0
       IF(ITEST.EQ.2)GO TO 61
 60    CONTINUE
       I=I+1
 61    IF(ITEST.EQ.2)I=JF
       JSCC=0
       JC=0
       JE=0
 7     CONTINUE
       JE=JE+1
       IF(JE.EQ.2)GO TO 10
C
C         CLUSES RELAY 4 FOR RIGHT EAR
S         LW,9       =8Z00820080
S         WD,9       X'C224'
```

```
C
           GO TO 11
 10     CONTINUE
C
C       OPENS RELAY 4 FOR LEFT EAR
S          LW,9        =8Z00000000
S          WD,9        X'C224'
C
 11        CONTINUE
C          RIGHT OR LEFT EAR DECISION
C JE=1 MEANS THE RIGHT EAR IS UNDER TEST, JE=2 LEFT EAR
 6      CONTINUE
        IF(NOTE.EQ.1)GO TO 8
        CALL SEGLOD (12)
        CALL AGECOMP
 8      CONTINUE
        J=0
        CALL SEGLOD(8)
        CALL AUDIOM
        RETURN
 20     CONTINUE
C  WRITES ON THE VDU THE LOSS ,COMPENSATIONAND CALLS SUB'GO'
        IF(NOTE.EQ.1)GO TO 9
        CALL WTNL
        CALL WTNL(33,'THRESHOLD COMPENSATION WITH AGE  ',TNCWA(I))
 9      CONTINUE
        IF(NOTE.EQ.1)CALL WTNL(19,' THRESHOLD LEVEL = ',HLOSS)
        IF(NOTE.EQ.2)CALL WTNL(16,' HEARING LOSS = ',HLOSS)
C CHECKS THAT BOTH EARS HAVE BEEN TESTED IN THE MANUAL MODE
        IF(ITEST.EQ.2.AND.JE.EQ.2)GO TO 2
        IF(I.EQ.6.AND.JE.EQ.2) GO TO 2
        M=0
        GO TO (7,60),JE
 2      CONTINUE
        M=2
 3      CONTINUE
        RETURN
        END
C
C
C
!FORTRANH RT,LS,GO,S
C
C
C
        SUBROUTINE AUDIOM
        COMMON/B/L,I,K,ILIN
        COMMON/WTEK/DB
        COMMON/HL/HLOSS
        COMMON/C/F(6),THRENC(6),TNCWA(6)
        COMMON/M/M,TRST
        COMMON/BLOG/J
        COMMON/CGO/JE
        COMMON/ORD/NF,NT,THREPR(10,6),THREPL(10,6),DATE(10)
        COMMON/WAVE/IDELAY,IAMP,ICOUNT
        COMMON/MAYN/ITEST,JF
        COMMON/TIM/ITIME(10)
        COMMON/NOT/NOTE
        DIMENSION TII(50)
C     J=1 ONLY IF ENTERING FROM "TEST" HAVING SAMPLED
```

```
C      A RESPONSE.
        IF(J.EQ.1)GO TO 121
        JSCC=0
        TINC=10.
        TII(1)=30.
        IY=2E9
        JC=0
        IWRT=0
        CALL WTNL(17,'TEST FREQUENCY = ',F(I))
        CALL SEGLOD (9)
        IF(ITEST.EQ.2)CALL GO
        IF(JE.EQ.1.AND.I.EQ.1.AND.ITEST.EQ.1)CALL GO
        IF(L.EQ.1)GO TO 102
        CALL WTNL(25,'SOUND PRESSURE LEVEL (DB)')
  102   CALL CHECK(L,IMODE,ISTAT,IBC)
        IF(ISTAT.EQ.2)GO TO 102
  111   CONTINUE
C       THIS SECTION PREVENTS THE TEST STEP ARRAY OVERFLOWING.
        IF((JC+1).LT.50)GO TO 103
        CALL WTNL(10,'INDECISIVE')
        HLOSS=999.9
        GO TO 104
  103   CONTINUE
         J=0
        JC=JC+1
        ISTAT=0
        IWRT=IWRT+1
C          OUTPUTS THE INTENSITY VALUE ON THE VDU
        IF(L.EQ.1)GO TO 106
  107   CALL WT(TII(JC))
  106   CONTINUE
        DB=TII(JC)
C
C     THIS SECTION ENSURES THAT "TEST" DOES NOT OCCUR
C     DURING A RESTART.
        IF(IRST.NE.1)GO TO 140
        IDELAY=-1
        GO TO 141
  140   CONTINUE
        CALL SEGLOD (10)
        CALL ENAB(2Z67)
  141   CONTINUE
C
        ICOUNT=0
        CALL ENAB (2Z65)
C
C     THIS SECTION GENERATES A RANDOM NUMBER TO ADD TO
C     THE REST DURATION.
        IX=IY
        IY=IX*65539
        IF(IY)5,6,6
  5     IY=IY+2147483647+1
  6     YFL=IY
        YFL=YFL*.4656613E-9
C         YFL IS A RANDOM NUMBER BETWEEN 0 AND 10.
C
        ITR =YFL*ITIME(2)*1.5+ITIME(2)
C     THIS TIMER SETS THE DELAY BETWEEN THIS STATEMENT
C     AND THE PRESENTATION OF THE TONE.
         CALL TIMER(ITR,2Z65)
```

```
      RETURN
C
 121  CONTINUE
      IF(JC.LT.4)GO TO 110
      IF(TII(JC).NE.TII(JC-2).OR .TII(JC-1).NE.TII(JC-3))GO TO 113
C   DECIDES IF THE PATIENT HAS SHOWN A YES-NO YES-NO
C   PATTERN OF RESPONSE.IF ANSWER IS YES THEN TII IS
C   REDUCED,NO THEN IT IS INCREASED.
C        DECIDES IF THE PATIENT HAS SHOWN A YES NO YES NO PATTERN OF RESP
C        IF ANSWER ABOVE WASA YES THEN THE VALUE OF TII IS REDUCED,NO INC
      JSCC=JSCC+1
      IF(JSCC.EQ.1)TINC=5.0
      IF(JSCC.EQ.2)TINC=2.5
      IF(JSCC.EQ.3)GO TO 105
 113  IF(IWRT.NE.5.OR.L.EQ.1)GO TO 110
C IWRT ENSURES THAT ONLY 5 VALUES OF INTENSITY FORM
C    ONE ROW ON THE VDU.
      CALL CWRITE(L,8Z0D151540,0,3,ISTAT)
      IWRT=0
 110  IF(ICOUNT.LT.0)GO TO 100
      GO TO 101
 100  TII(JC+1)=TII(JC)-TINC
      IF(TII(JC+1).LT.-19.0)GO TO 105
      GO TO 111
C   THESE CONDITIONS SET THE UPPER AND LOWER LIMITS
C   OF THE INTENSITY RANGE.
 101  TII(JC+1)=TII(JC)+TINC
      IF(TII(JC+1).GT.52.5)GO TO 105
      GO TO 111
 105  CONTINUE
      IF(JE.GT.1) GO TO 114
C        RIGHT EAR LOSS CALCULATION
      IF(ITEST.EQ.2)NT=1
      IF(NOTE.EQ.1)THREPR(NT,I)=(TII(JC)+TII(JC-1))/2.
        IF(NOTE.EQ.2)THREPR(NT,I)=(TII(JC)+TII(JC-1))/2.-THRENC(2)
      HLOSS=THREPR(NT,I)
 104  CONTINUE
      M=1
      RETURN
 114  CONTINUE
C        LEFT EAR LOSS CALCULATION
      IF(NOTE.EQ.1)THREPL(NT,I)=(TII(JC)+TII(JC-1))/2.
        IF(NOTE.EQ.2)THREPL(NT,I)=(TII(JC)+TII(JC-1))/2.-THRENC(2)
      HLOSS=THREPL(NT.I)
      M=1
      RETURN
      END
C
C
C
      SUBROUTINE
      COMMON/M/M,IRST
      COMMON/BLOG/J
      COMMON/WAVE/IDELAY,IAMP,ICOUNT
      COMMON/MAYN/ITEST,JF
      IF(IRST.EQ.1)RETURN
      J=J+1
C   DECIDE WHAT TO CALL AS A RSULT OF A FINISHED
C   SAMPLING SESSION.
      CALL AUDIOM
```

```
C          M=1 ONLY IF A LOSS HAS BEEN CALCULATED
        IF(M.NE.1)RETURN
        CALL EARTEST
C          M=2 ONLY IF THE ENTIRE TEST IS COMPLETE
        IF(M.NE.2)RETURN
        IF(ITEST.EQ.2)M=3
         CALL ENAB(2Z66)
        CALL TRIGR(2Z68)
        RETURN
        END
C
C
C
        SUBROUTINE GO
C     TYPE"GO" TO ALLOW THE PROGRAM TO CONTINUE,
C     "NO" TO ABORT IT.
        COMMON/CGO/JE
          DATA IOK/'OK'/,INO/'NO'/,IGO/'GO'/
 11     CONTINUE
        CALL WTNL
        IF(JE.EQ.1)GO TO 12
        IF(JE.EQ.2)GO TO 15
 12     CALL WTNL(9,'RIGHT EAR')
        GO TO 14
 15     CALL WTNL(8,'LEFT EAR')
 14     CONTINUE
        IBC=-2
        CALL RDNL(31,'TYPE GO FOR TONE OR NO TO ABORT',IBC,IG)
        IF(IG.NE.IGO.AND.IG.NE.INO)GO TO 11
        IF(IG.EQ.INO)CALL RELEASE
        CALL ERASE
        CALL HOME
        RETURN
        END
C
C
C
        SUBROUTINE CEASE
        COMMON/CES/IPUL
C     SUBROUTINE USED TO CEASE THE OUTPUT OF THE WAVETEK
C     ON INTERRUPT LEVEL 2Z64.
        COMMON/WAVE/IDELAY,IAMP,ICOUNT
C
C          REDUCES WTK OUTPUT TO MINIMUM
        CALL RDAC(2,-10.0)
C        SWITCHES ELECTRONIC SWITCH OFF
        CALL RDAC(3,0.0)
        IPUL=31-IPUL
C
C          SETS ATTENUATOR TO MAXIMUM
        CALL PULSER
        IDELAY=-1
        RETURN
        END
C
C
C
        SUBROUTINE AGECOMP
        DIMENSION GAMMA(60),THREN(6),BETA(6)
C     THRESHOLD COMPENSATION WITH AGE CALCULATION AS
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C    PER BS...... FIGURES.
      COMMON/B/L,I,K,ILIN
      COMMON/C/F(6),THRENC(6),TNCWA(6)
      DATA THREN/11.2,6.0,4.2,1.0,-3.9,15.3/
      DATA GAMMA/0.,0.,0.,0.,0.,0.,0.,0.,0.,0.,0.,0.,0.,0.,0.,0.,0.,0.,0
     &.,0.,0.,0.,0.,0.,0.,3.2,3.7,4.4,5.,5.9,6.7,7.6,8.5,9.7,10.8,12.2,1
     &3.8,15.7,17.7,19.9,21.7,23.8,25.2,26.9,27.9,29.2,29.6,31.1,31.8,32
     &.7,33.3,34.0,34.5,35.3,35.7,36.4,36.8,37.4,37.8,38.3/
      DATA BETA/0.0,0.0,0.0,0.06,0.32,0.43/
      IAGE=20
      K=IAGE
      IF(I.GT.1) GO TO 101
      CALL WTNL(18,'AGE OF PATIENT IS ',IAGE)
 101    CONTINUE
       TNCWA(I)=BETA(I)*GAMMA(K)
       THRENC(I)=THREN(I)+TNCWA(I)
      RETURN
      END
      SUBROUTINE PRESS
C ASKS THE OPERATOR TO PRESS ONE OF THE EXTERNAL
C INTERRUPT BUTTONS TO CALL FOR RESTART,RELEASE OR AUDIOGRAM.
      COMMON/JOT/ILIN
      L=ILIN-2
      CALL CWRITE(L,8215151507,0,4,ISTAT)
      CALL WTNL(11,'END OF TEST')
      IF(L.EQ.1)CALL WTNL(26,'PRESS FOR RESTART OR ABORT')
      IF(L.EQ.2)CALL WTNL(35,'PRESS FOR RESTART,ABORT OR AUDIOGRAM')
      RETURN
      END
!ALLOBT (FILE,X2),(FSI,50),SAVE
!OLOAD (LIB,USER,SYSTEM),(MAP,ALL);
:(FORE,2000),(PUBLIB,COCLIB),(TASKS,10),(TEMP,1000),(UDCB,5)
:ROOT (FILE,BT,GO,7)
:SEG (LINK,16)
:SEG (LINK,1)
:SEG (LINK,3,ONTO,1)
:SEG (LINK,2,ONTO,3)
:SEG (LINK,4,ONTO,3)
:SEG (LINK,5,ONTO,3)
:SEG (LINK,13,ONTO,3)
:SEG (LINK,14)
:SEG (LINK,7)
:SEG (LINK,8,ONTO,7)
:SEG (LINK,11,ONTO,8)
:SEG (LINK,9,ONTO,11)
:SEG (LINK,10,ONTO,11)
:SEG (LINK,12,ONTO,11)
:SEG (LINK,15)
:ASSIGN (F:1,9TA80)
:ASSIGN (F:2,D2,LLEARS)
:ASSIGN (F:3,TY),VFC
:ASSIGN (F:4,TK),VFC
!SFIL 9TA80,1
!RADEDIT
:COPY (FILE,BT,OV),(OUT,9TA80)
!WEOF 9TA80
!REWIND 9TA80
!PAUSE KEY IN FGC
!RUV
!FIN
```