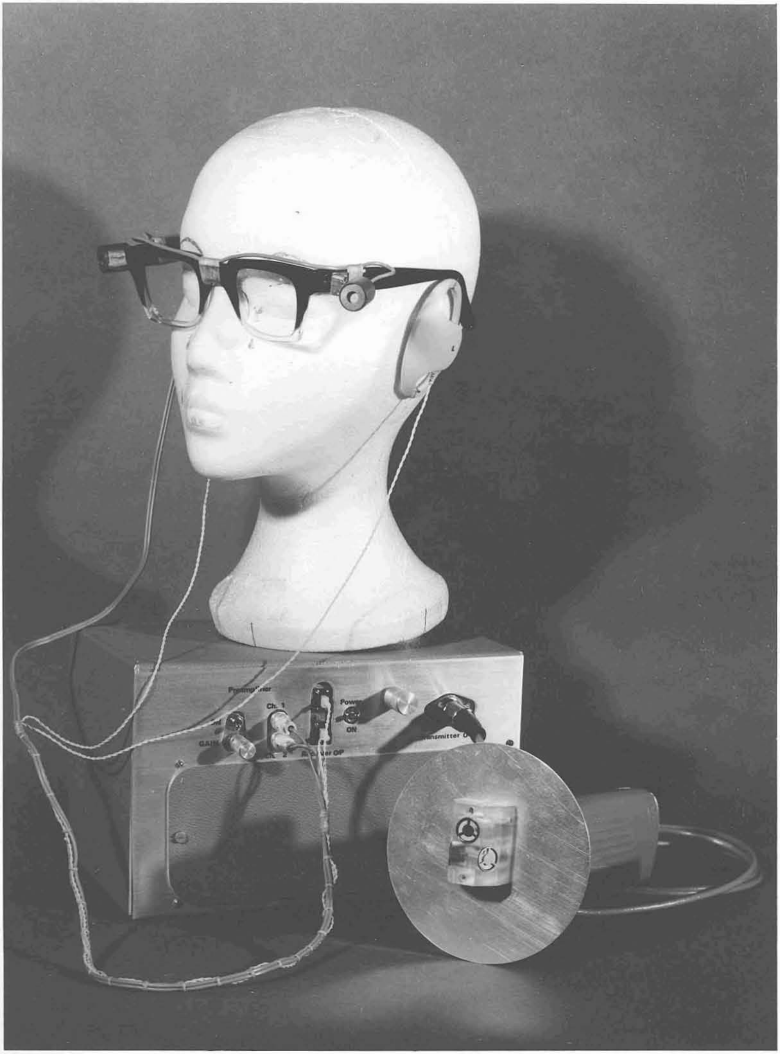

Frontispiece: An experimental pulsed ultrasonic binaural aid for the blind

ULTRASONIC

ECHO REINFORCEMENT

FOR THE BLIND

by

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SUMMARY

The ability that many blind people develop of detecting near objects by means of echo-location is well known. This investigation is concerned with the development of an aid that will enhance this ability by transmitting ultrasonic pulses into the environment and receiving the echoes returning from surrounding objects. The echoes are detected and heard as clicks by the user, who is able to determine the direction from which the sound is coming by using his natural localisation ability. Further, the clicks are "coded" by switching the receivers on and off, so that the rate of clicking indicates the distance of an object.

A key component of the system is the ultrasonic transducer. Piezoelectric bimorph transducers have been selected for use as both transmitters and receivers, although the characteristics of the standard type have been readjusted to meet the requirements of the aid. In particular, the transducer bandwidth has been considerably extended to ensure a sharp sounding click.

In order to provide accurate directional information it has been necessary to closely simulate naturally occurring interaural differences. Interaural time differences are readily provided by spacing the receiving transducers at the inter-ear distance apart, whereas the provision of accurate interaural intensity differences has necessitated the shaping of the directional response of the receiver to match that of the ear.

A series of psychophysical tests has been performed to measure the ability of subjects using the device to localise objects in the horizontal plane. Considerable accuracy was shown with the best

results being obtained using interaural time information alone or interaural intensity information alone, rather than a combination of both.

Elementary outdoor trials with the device have shown the range and directional information to be effective with the indication of the shoreline being particularly clear. A progressive refinement of the aid should lead it to a stage where its usefulness to blind people can be evaluated.

CONTENTS

CHAPTER 1

INTRODUCTION

A severe restriction imposed by blindness is the difficulty in moving around by oneself. This impairment of mobility has fostered much research work, the aim of which is to enable a blind person to travel on foot, safely, independently, without undue stress, and without inconvenience to others. In order to achieve this aim, it is necessary to determine what kind of information the blind person requires from the environment in order to travel effectively within it. Unfortunately, the mobility requirements of blind people have not yet been fully investigated, although research in this field over the past twenty years has led to the formulation of many useful concepts.[1,2]

One may begin to define the required information by listing the categories of information supplied by vision for the purpose of sighted mobility. There are four main ones: object detection, orientation, posture, and terrain changes. The problem, therefore, can be seen as presenting non-visually the information which is normally available through vision.

However, it seems reasonable to assume that it is unnecessary to present all the information non-visually which is available through vision. We know, for example, that some congenitally blind people move about very well without any aids at all. On the other hand, a person who is sighted and under blindfold is totally incapacitated as far as effective mobility is concerned. The difference between the two conditions is a measure of the extent to which it is possible to achieve effective mobility by non-visual means. Although the type of blind person just described is not able to move about as well as the sighted person, it is almost certain that he manages on very much less information.

Consideration must be given not only to the information that is required, but also to the way in which it is gathered and subsequently displayed. Leonard[1] stressed how important it is for the display of an aid to achieve the best transfer of information from the environment to the blind person so that appropriate action can be carried out. He referred to the problem as one of compatibility between the display of an aid and the control of its user over his actions; the higher the compatibility, the easier it is to translate a change in the display into the appropriate action.

It is useful to consider existing non-visual displays which are known to produce a measure of compatibility in the mobility situation. Only two of the remaining sensory channels available to a blind person appear to be capable of providing a high compatibility; those of hearing and touch.

Tactual stimuli can be sensed by the whole of the body surface and more particularly by the extremities. Contact with the environment gives a fairly direct signal which, of course, is very limited in range and therefore requires some extension. The displays provided through canes and guide dogs are evidently helpful and can achieve high levels of compatibility with the blind traveller. Their use clearly extends the limits and speeds of progress, once their manipulation has been learnt.

On the auditory side, the extensive work of Dallenbach[3] and his colleagues finally settled the controversy that had existed in the earlier part of the century over the reason why some blind people have the ability to detect obstacles at a distance. This ability has been sometimes called "facial vision" or "the obstacle sense of the blind". Dallenbach determined that hearing is the necessary and sufficient sense for this ability. Muffling the footsteps of a listener as he approaches an obstacle interferes with his ability to detect it, and plugging his ears prevents the detection. The moving listener can be replaced by a microphone and loud-speaker mounted on a trolley, while the subject sits in another room listening through headphones. If the loudspeaker emits continuous high frequency tones or broadband noise, the listener can detect the obstacle almost as well as when he is there in person. Kohler,[4] in another set of experiments, used seated listeners who held a source radiating broad-band noise by their chests. On average, they were able to detect the approach of a circular cardboard disc 50 cm in diameter at a distance of about 1 metre.

It was stated earlier that the comparison of the mobility performance of a blind person having good "obstacle sense" with that of a blindfolded sighted person would give a measure of the mobility that can be achieved by non-visual means. Kellogg[5] made this type of comparison although he performed his experiments in a static situation rather than the dynamic one encountered in mobility. He tested the ability of blind and blind-folded sighted listeners to distinguish between circular discs of different diameters (6 to 12 inches) at different ranges (1 to 2 ft.). The listeners were not provided with artificial sound generators, but were allowed to make any sound they wished. The blind listeners could distinguish the differences in size and distance and even differences in the hardness of the material. The performance of blindfolded sighted listeners in the same test was almost never above the level of pure chance.

Thus Kellogg demonstrated the wealth of useful information that his blind subjects could gather using only echo-location. Moreover, his subjects were able to interpret this information easily, which suggests a high compatibility between the display provided by natural echo-location and its user; this is the criterion stressed by Leonard as being particularly important. In view of these findings, therefore, an aid based on enhancing natural echo-location could well prove of use in aiding the mobility of blind people.

Kohler[4] made a particularly detailed study of orientation by aural cues. He concurred with the findings of Dallenbach et al[3] that sightless subjects can orientate themselves by listening to echoes from surrounding obstacles. Moreover, he stated that their ability to detect these obstacles was substantially increased if they were allowed to generate a sound or noise by themselves. This led him to an investigation of the characteristics of the sound source which would be optimal for echo-location. He determined that impulsive sounds are more favourable than continuous sounds. His test subjects, as well as considering these more agreeable, found that short impulsive sounds enabled them to hear a distinct echo. This clue was lost when continuous sounds were used. A disadvantage of impulsive sounds, however, is that as the interval between impulse and impulse echo decreases, the two amalgamate into one. This is a limitation of the human hearing system, which is unable to separate two sound pulses which follow each other in quick succession.

Kohler, however, observed that the detection of obstacles using clicks is not primarily based on listening to the actual echo, but rather on the perception of a modification in the sound made by the click. The modification is heard as a change in pitch, an effect which was also found to occur with other guide sounds. Kohler, therefore, felt that it was most important to choose a guide sound which produced modifications in the perceived sound that the ear could easily detect. He considered that click sounds are such that the ear can readily differentiate between the signal modifications occurring in echo-location.

Beurle[6] also carried out an investigation to determine the form of sound most suitable for human echo-location. A series of tests were performed in which a number of blind people were asked to detect certain objects with the assistance of a variety of different sounds. A similar conclusion was reached in that a succession of clicks was found to be the most useful form of acoustic stimulus. Although clicks were not the best under all circumstances, they did give the greatest amount of assistance under a wide range of different conditions. This led Beurle to construct a guiding device in which a source emitting audible clicks was placed near the focus of a parabolic reflector to produce a fairly wide beam with a concentration of sound towards the centre of the beam. The hand-held "Clicker", as the device became called, enabled the user to listen directly to the sound reflected from any obstacle.

The device was initially received with enthusiasm by blind subjects and it proved useful as a training aid for blind persons undergoing mobility exercises. However, there was a tendency to discard it soon after the user began to gain confidence in his ability to detect obstacles, and to orientate himself by using natural random sounds. Furthermore, the Clicker had two serious disadvantages when considered as a guiding aid: firstly, the sound emitted by the device was easily drowned by heavy traffic noise, thus rendering it useless under these conditions. Secondly, blind people were reluctant to use it since the audible output attracted attention to their handicap. These problems could, of course, be eliminated by constructing an ultrasonic version of the device, although the use of ultrasound has certain drawbacks. In particular, ultrasonic energy is considerably attenuated as it passes through air, and furthermore, reflections from smooth surfaces are specular so that little energy is returned to the user unless the reflecting surface is at right angles to the transmitted beam. Several experimental ultrasonic versions were constructed, despite the lack of efficient ultrasonic transducers and the limitations of using valves. The devices turned out to be clumsy, and did not become popular since many blind people found they could manage equally well without their assistance. Also several of the devices necessitated the wearing of headphones, which interfered with the ability of the blind person to hear useful environmental sounds.

As a result of his work on the Clicker and other similar guiding devices Beurle concluded in 1951 "...neither the Clicker device nor any other device which can be envisaged at the present time is likely to become popular among blind people".[7]

Ten years later, however, interest in ultrasonic guiding aids for the blind was reawakened; the stimulus had come from two different directions. Firstly, the advent of transistors, microminiaturisation techniques and improvements in ultrasonic transducers promised to considerably simplify the technological side. Secondly, a number of studies into the ability of bats to navigate using echo-location gave fresh information which was considered to be of possible use in the design of guiding aids.

It was already known that the type of sounds emitted by bats include frequency-modulated and constant frequency pulses. Several new theories, however, were put forward as to how the bats interpret the echoes from their sounds in order to orientate themselves. Nordmark[8] proposed that the principal method by which a bat locates objects in space is to generate a stream of pulses and subsequently to listen to the tone formed by the interval between the outgoing and reflected pulses. Pye,[9] on the other hand, suggested that bats emitting a swept-frequency pulse obtained range and directional information from the beat notes formed by the interaction between the echo and the emitted pulse.

Kay[10] concurred with Pye's conclusion after carrying out a series of experiments to determine the optimum form of auditory presentation of signals received with ultrasonic echo-location guidance aids. He compared a pulse system with a frequency modulation system using ultrasonic transducers capable of operating over a frequency band of 20 kHz centred on 50 kHz. The pulse system employed pulses of 50 kHz and the signals presented to the ear were either 3 kHz tone pulses or DC pulses. In some tests the direct transmission could not be heard, and observers found no indication of the range from which the signal was being received. In other tests the transmitted pulse could be heard at equal volume to the received pulse. When using audio frequency pulses it was not found possible to estimate distance on the basis of the time interval between clicks. There was a faint sensation of hearing a note with the click, the pitch of which varied with the distance to the transmitter, but the effect was very weak. Using DC pulses, the pitch of this note, heard together with the clicks, was much more marked, and could be judged sufficiently well to give a rough indication of the distance. The click, however, was the predominant sound.

The frequency-modulation system used a transmission which varied linearly from 60 kHz to 40 kHz at a rate which produced a difference frequency of 3 kHz between transmission and an echo from 15 ft. Thus the subject experienced signals covering a frequency range of 0–3 kHz depending on the distance to the object. Using this system there was no difficulty in determining the distance to an object, which depended on the ability of a subject to estimate the frequency. Kay concluded that the most efficient guidance system, using ultrasonic transmission, is in the form of a frequency-modulated wave similar to that used by bats.

On the basis of this conclusion, Kay developed a portable guidance aid with an auditory display which was based on an FM system to give range indication. An evaluation of this device gave favourable results[11] and it was later redesigned in a binaural form in which directional information was obtained from interaural amplitude differences. This binaural device, which became known as the "Sonic Glasses", underwent an extensive development programme and a prototype version was recently subjected to a major evaluation. The conclusion reached was that the aid appears to be of significant value to some blind people.[12]

This conclusion indicates that the display offered by the "Sonic Glasses" (i.e. presentation of range information in terms of pitch and direction information as an interaural amplitude difference) is to some measure compatible with the mobility requirements of the user. However, we can only speculate as to whether this form of display in an ultra-sonic echo-location device is better than any other, since we have none to compare it with. When considering Kay's conclusion that a frequency-modulation system is superior to a pulsed system for an ultrasonic blind aid, we should perhaps accept that it is only superior to the pulse systems he examined. There may be other pulse systems not investigated by him which might be as good as or better than the frequency-modulation system. In fact, Kay has stated recently that if a pulsed echo-location system with an auditory display were to be developed, then it would require some complex form of processing.[12]

The complexity of such processing has already been investigated by Crawford et al.[13] who found that the coding of pulse echoes into a number of distance ranges is readily achieved with modern techniques in logical circuitry. It seems relevant, therefore, in view of Kohler and Beurle's conclusion that click sounds are the most useful form of signal to blind people using a sonic echo-location device, that a pulsed ultrasonic echo-location device should be developed having a binaural display emitting clicks. The requirements of this device are to enable a user, by reinforcing his natural echo-location ability, to sense his environment in a fairly wide beam over a range of several metres. An evaluation of a pulsed ultrasonic binaural device, whose development is described in this thesis, will reveal whether Kohler and Beurle's conclusion for a sonic device is equally valid for an ultrasonic version.

CHAPTER 2

A PULSED BINAURAL ULTRASONIC AID

2.1 The Proposed Device

The concept of the device is essentially simple. It involves transmitting ultrasonic pulses in a wide beam and receiving the echoes returning from surrounding objects. The echoes are detected and presented to the ears of the user as clicks which sound to him as if they are coming from the object causing the echoes. Thus, the ability that many blind people develop of detecting near objects by means of naturally occurring sounds is reinforced by these artificial sounds. The direction from which the sound is coming is determined binaurally by the auditory system of the user which processes the difference in the sound in one ear relative to the other. Further, the clicks are "coded" by switching the receivers on and off, so that the rate of clicking indicates the distance of an object.

2.2 Description of the Experimental Device

The device, which is shown in block diagram form in Fig. 2.1, comprises a transmitter which issues ultrasonic pulses of energy at a pre-determined rate, together with matched ultrasonic receivers, and earphones through which the returning echoes are heard after detection. The receiving amplifiers are switched in accordance with a pre-determined coding arrangement, so as to indicate the range of an object from which the echo has returned. The code is arranged so that the click rate heard by the user varies from a low rate for distant objects to a high rate for close objects, in five discrete steps.

To achieve this coding arrangement, both channels of the amplifier connected to the ultrasonic receiving transducers are switched in unison by means of a waveform synchronised with the transmitted pulses. Fig. 2.2 shows the relationship between the amplifier switching waveform and the transmitted pulses. The operation

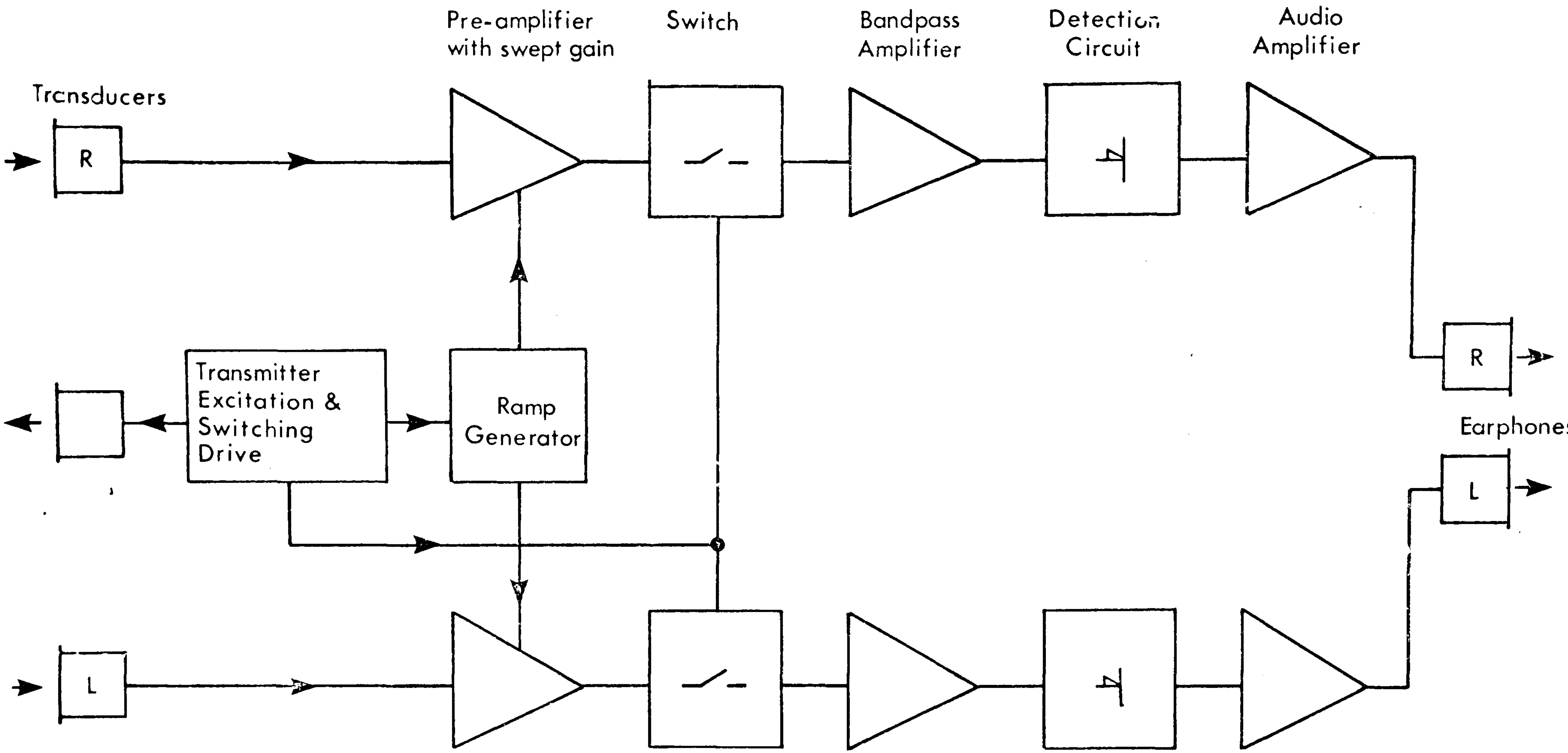


Figure 2.1 Block diagram of experimental device

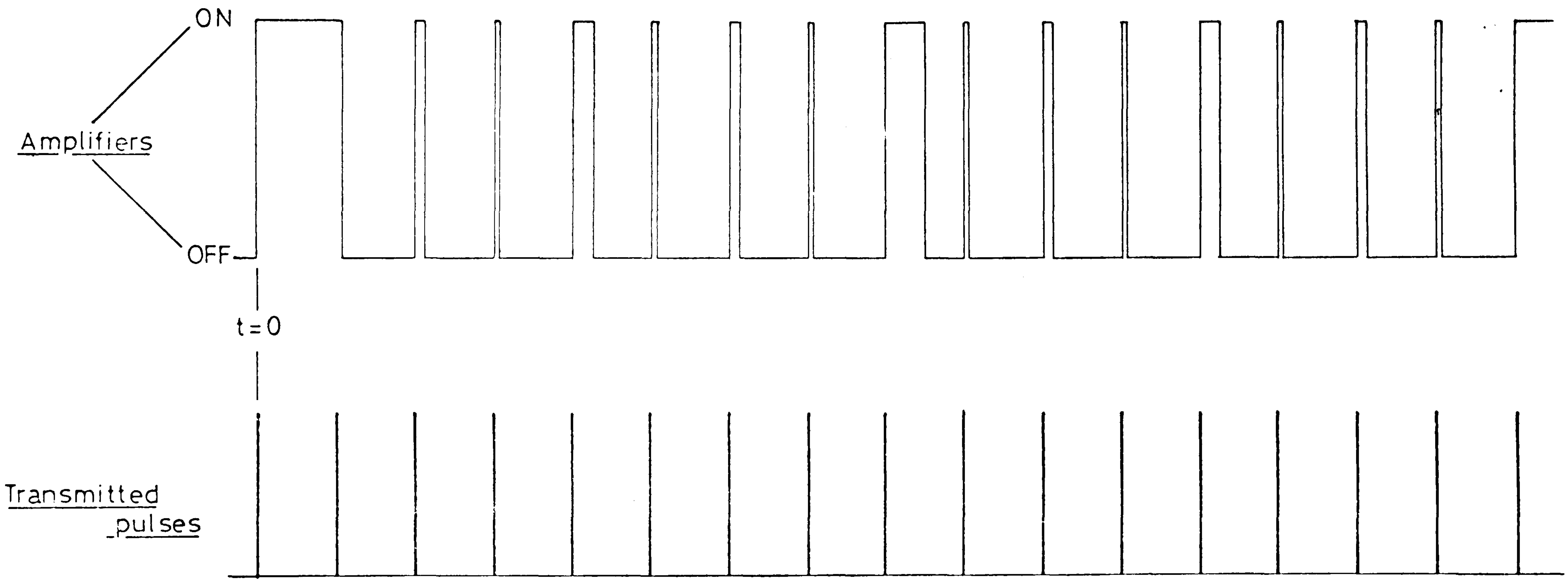


Fig. 2.2 Amplifier switching waveform

of the switching waveform has been elegantly described by Crawford et al.[13] with reference to a modified form of Fig. 2.2, shown as Fig. 2.3. Here, the "on" periods of the switching waveform have been expanded along a time scale, and the waveforms following successive transmitted pulses are shown on separate horizontal lines one below the other. Thus, line one shows the waveform following the first pulse, line two the waveform following the second pulse and so on. The presence of a horizontal line indicates that the amplifiers are switched on, while the horizontal length of line denotes the length of the "on" period. Starting at t = 0, the amplifiers are switched on for a period of 64 ms after a pulse is transmitted. Echoes from the outward travelling pulse may then be received up to the end of this period. In Fig. 2.3 this period of 64 ms is marked at intervals of 4, 8, 16 and 32 milliseconds from the transmission of the pulse. As sound travels at approximately 1ft./ms, the transmitted pulse would take 1 ms to reach an object situated 1 ft. from the device, and the resultant echo would take 1 ms to return, making 2 ms in all. It is thus possible to convert the time scale to a distance scale. This is shown above the time scale in Fig. 2.3. So in 64 ms, echoes from objects up to 32 ft. away will be received and heard, since the amplifiers are switched on for the whole of this period.

At t = 64 ms, which is shown at the beginning of the next line, another pulse has been transmitted. This time, however, the amplifiers are switched on for only 4 ms, so that echoes from objects 2 ft. away or less are received and heard, while echoes from more distant objects are suppressed. The amplifiers are off for the next 60 ms, when the third pulse is transmitted. The cycle continues as shown until 16 pulses have been transmitted, by which time a period of approximately 1 second has elapsed (16 x 64 ms). After this the cycle repeats itself.

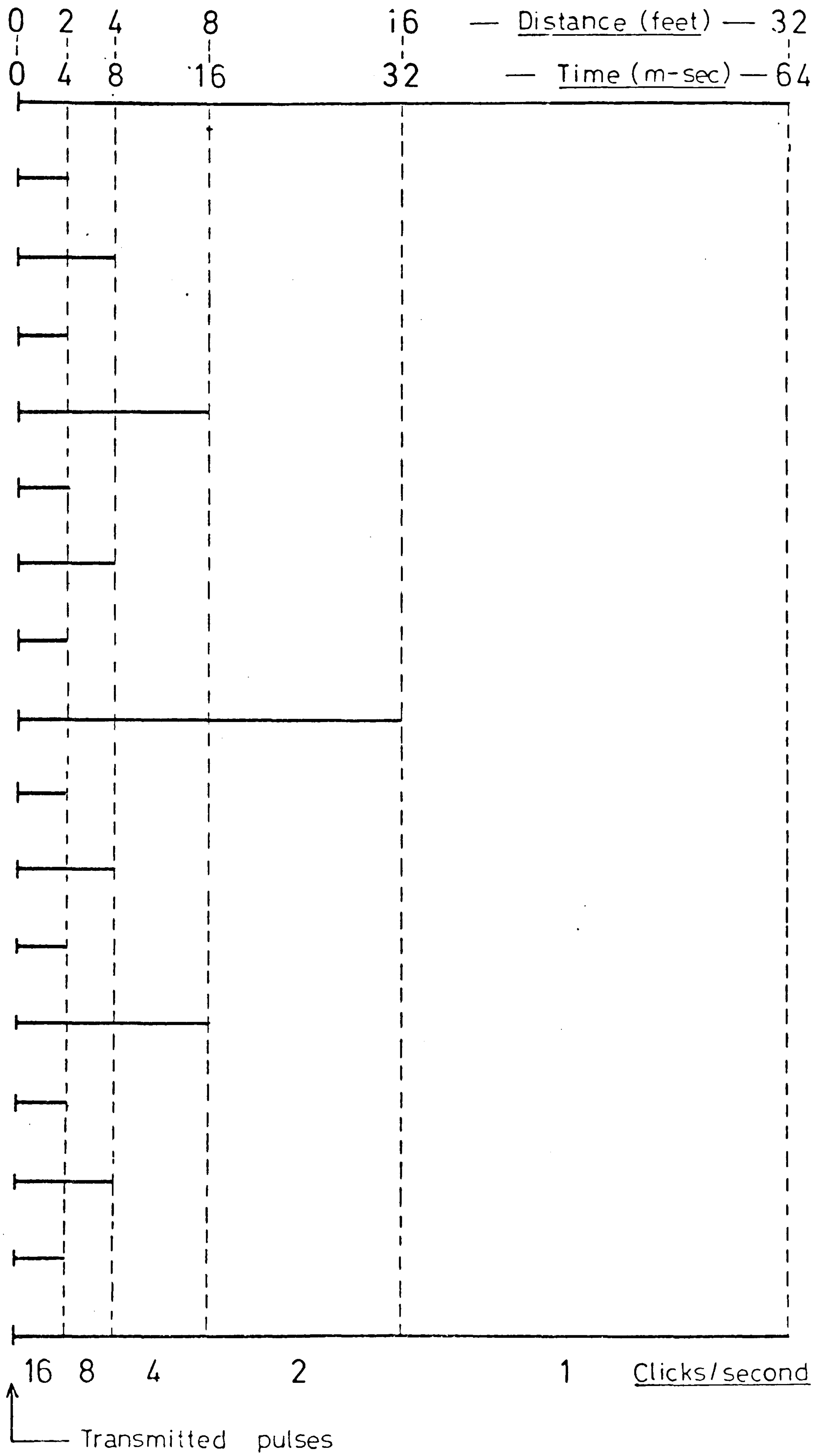


Fig. 2.3 Switching sequence of receiving amplifiers

Thus, if an object is in the range 0–2 ft. from the device, then 16 clicks per second will be heard and so on, falling to 1 click per second for objects in the range 16–32 ft. These rates are marked for the appropriate distance ranges at the bottom of the diagram.

Clearly, this particular switching sequence is only one of many possible sequences that could have been chosen. The reason for its adoption and maintained use is based on two main factors.

Firstly, it is important that the user can easily distinguish between clicking rates, so that he can quickly learn to use them in judging the range of an object. A series of psycho-physical experiments carried out by Crawford and Rudlin[13] indicated that for minimum confusion between adjacent rates it is preferable to use as large a ratio between these rates as possible. They added, however, that practical considerations dictate that this ratio should not exceed approximately 2 to 1.

Secondly, the maximum clicking rate is governed by the maximum range of the device, since a pulse cannot be transmitted until the echo from objects at the maximum range has been received. Although the range will not be finally decided upon until a detailed evaluation of the device has been carried out, the present range of about 32 ft. with the clicking rate set to the maximum of 16 per second has proved reasonably adequate in the limited trials performed so far.

2.2.1 Ultrasonic Transducers

Piezoelectric bimorph transducers are used in both the transmitter and the receivers.

An arrangement of three transducers is used in the transmitter in order to provide a wide beam in the horizontal plane. The beamwidth is sufficient to radiate some energy sideways so as to produce strong reflections from walls and fences adjacent to the user, thereby facilitating his ability to shoreline. In the vertical plane a beamwidth of about 60 degrees provides protection for the head and body.

The receiving transducers have a directional response which is shaped to correspond as closely as possible to the directional response of the human ear. Thus, when the transducers are splayed apart at a pre-determined angle, the amplitude differences between channels produced by objects in various azimuth directions are similar to the interaural amplitude difference cues used in natural localisation. Interaural time difference cues are obtained by spacing the transducers by the inter-ear distance. Recent psycho-physical experiments have revived the question of whether one or other of the cues should be used as opposed to a combination of both, the latter arrangement being the one previously thought to be optimal. The recurrence of this problem and a proposed solution is discussed in Chapter 8.

2.2.2 The Transmitter

The transmitter circuit, originally designed by Crawford,[14] is shown in Fig. 2.4. An astable multivibrator provides a 256 Hz square wave at point X for clocking the switching logic circuits. At point Y a 16 Hz square wave returning from the logic triggers a monostable whose unstable mode is set at approximately 3 ms. This allows a pulse to be transmitted shortly before the receiver is switched on. This monostable, in turn, triggers a second monostable whose unstable mode is set at 250 μs, a value chosen to correspond with the width of the click from the original sonic torch.[7] Thus the output of this monostable consists of pulses at a 16 Hz rate and 250 μs pulse width. This output is used to enable or inhibit a 40 kHz square wave coming from a second astable multivibrator. The method used is to connect the

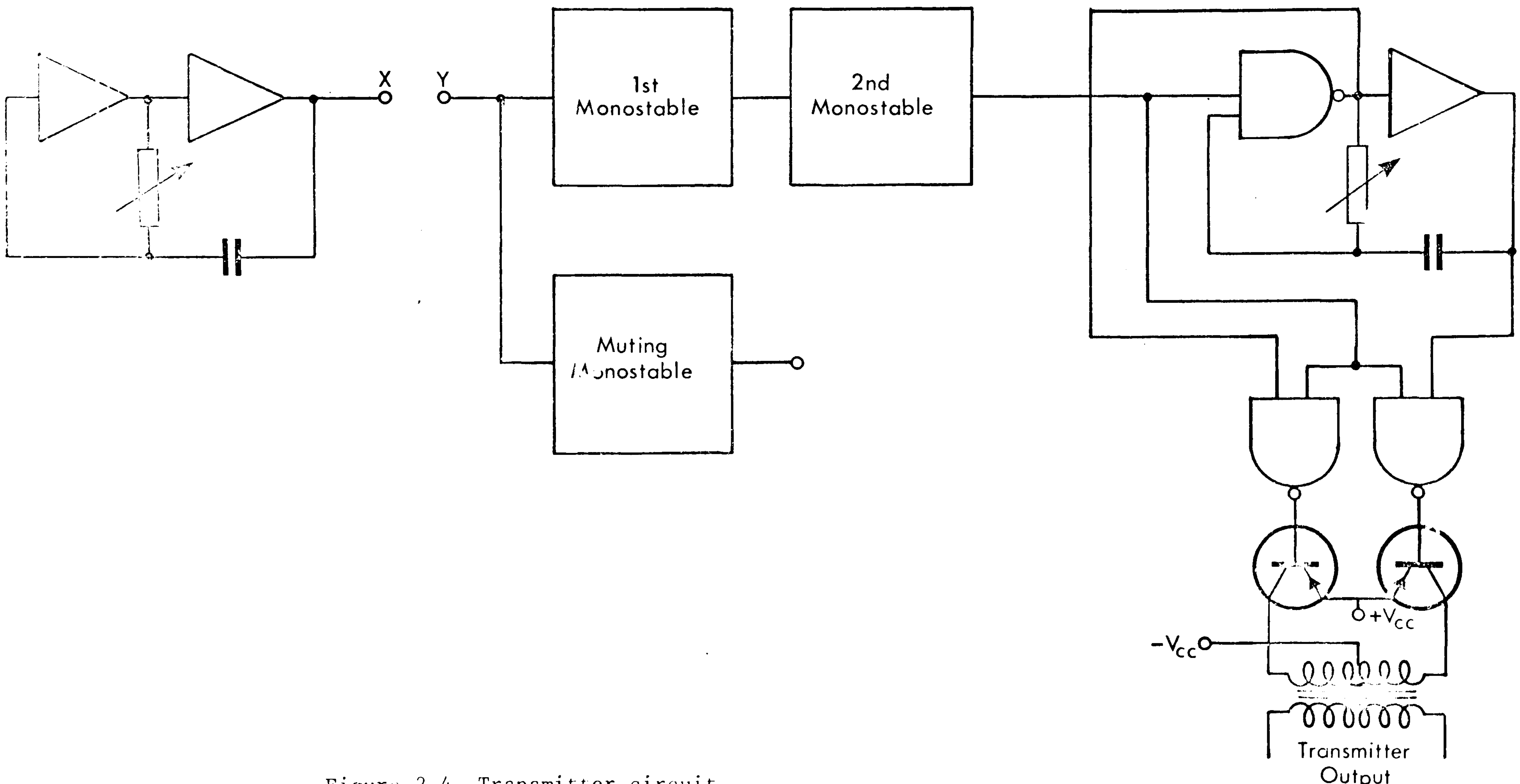


Figure 2.4 Transmitter circuit

40 kHz square wave signal to one input of a NAND gate, the other coming from the second monostable. Two NAND gates, in fact, are used with their 40 kHz input signals out of phase. This means that the outputs of the two NAND gates can be used in a push-pull amplification stage.

It is necessary to mute the receiver for a short period as each pulse is radiated, to prevent direct pick-up of the transmitted ultrasonic signal by the receiving transducers. The period is set at approximately 3 ms to enable a hand held transmitter to be spaced from the receivers by up to about 3 ft. The circuitry used to accomplish the muting again uses a monostable which is triggered by the 16 Hz square wave returning from the logic.

2.2.3 Switching Logic

The logical connections required to give the desired switching waveform have been evaluated by considering the switching sequence shown in Fig. 2.3.[14] This diagram enables all the points at which the state of the receiving amplifiers is to be specified to be determined.

The practical circuitry which accomplishes the switching logic at present requires the use of nine logical elements although with the rapid progress in the manufacture of versatile logic chips it may be possible to reduce this number.

2.2.4 Receivers

The pre-amplifiers, designed by Jones,[15] are matched in gain and frequency response and include a low noise input stage; one channel is shown in Fig. 2.5. The input impedance is low owing to the requirements of the piezoelectric bimorph transducers (see Section 4.3), the appropriate value being provided by shunt negative feedback.

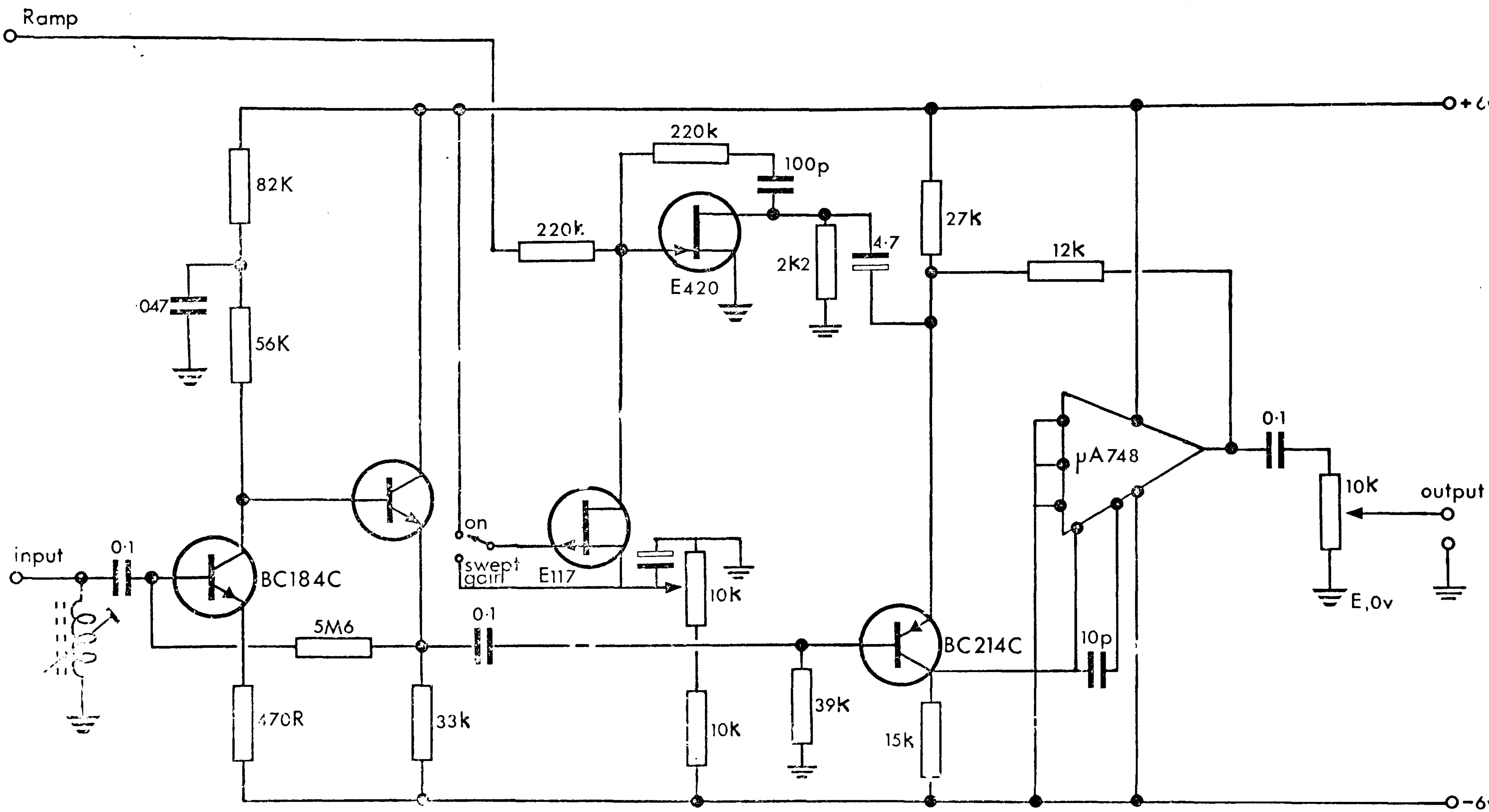


Figure 2.5 Circuit diagram of pre-amplifier with swept gain

To reduce the variation in amplitude of echo signals from different distances, the gain of the second stage of the pre-amplifier is steadily increased after the transmission of each pulse. Distant echoes are thus amplified more than those emanating from nearby objects.

The signals passing from the volume control of the pre-amplifier are filtered by a bandpass filter with a centre frequency of 40 kHz and a pass-band extending from approximately 30 kHz to 50 kHz; the design is shown in Fig. 2.6. This filtering reduces the front end noise, which although low was audible as an intermittent hissing sound after being gated by the switching logic.

The switching is accomplished using a monolithic f.e.t. device, this being followed by diode detection and audio amplification as shown in Fig. 2.7. Small earphones, mounted in earshells which hang on the pinna, direct the signal towards the entrance of the ear canal or meatus. This arrangement, which spaces the transducer about one centimetre from the meatus, does not appear to impede the user's natural ability to hear and locate ambient sounds.

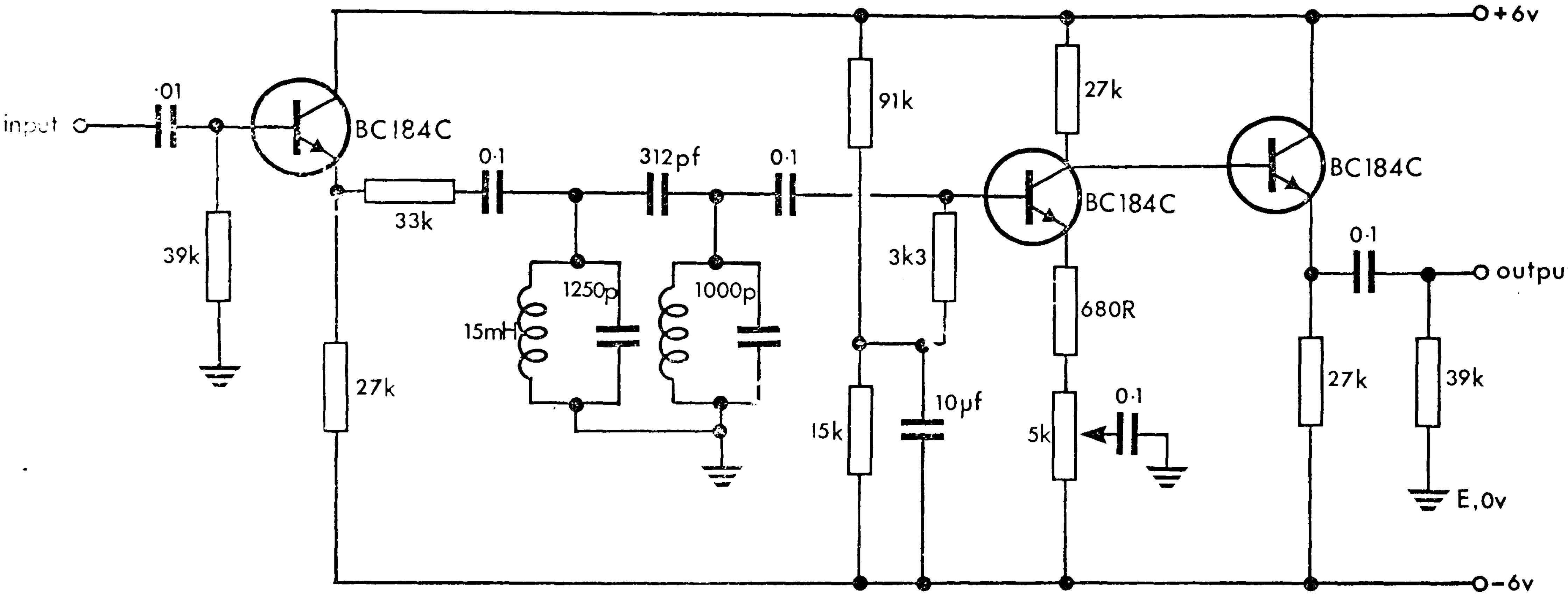


Figure 2.6 Circuit diagram of filter/amplifier

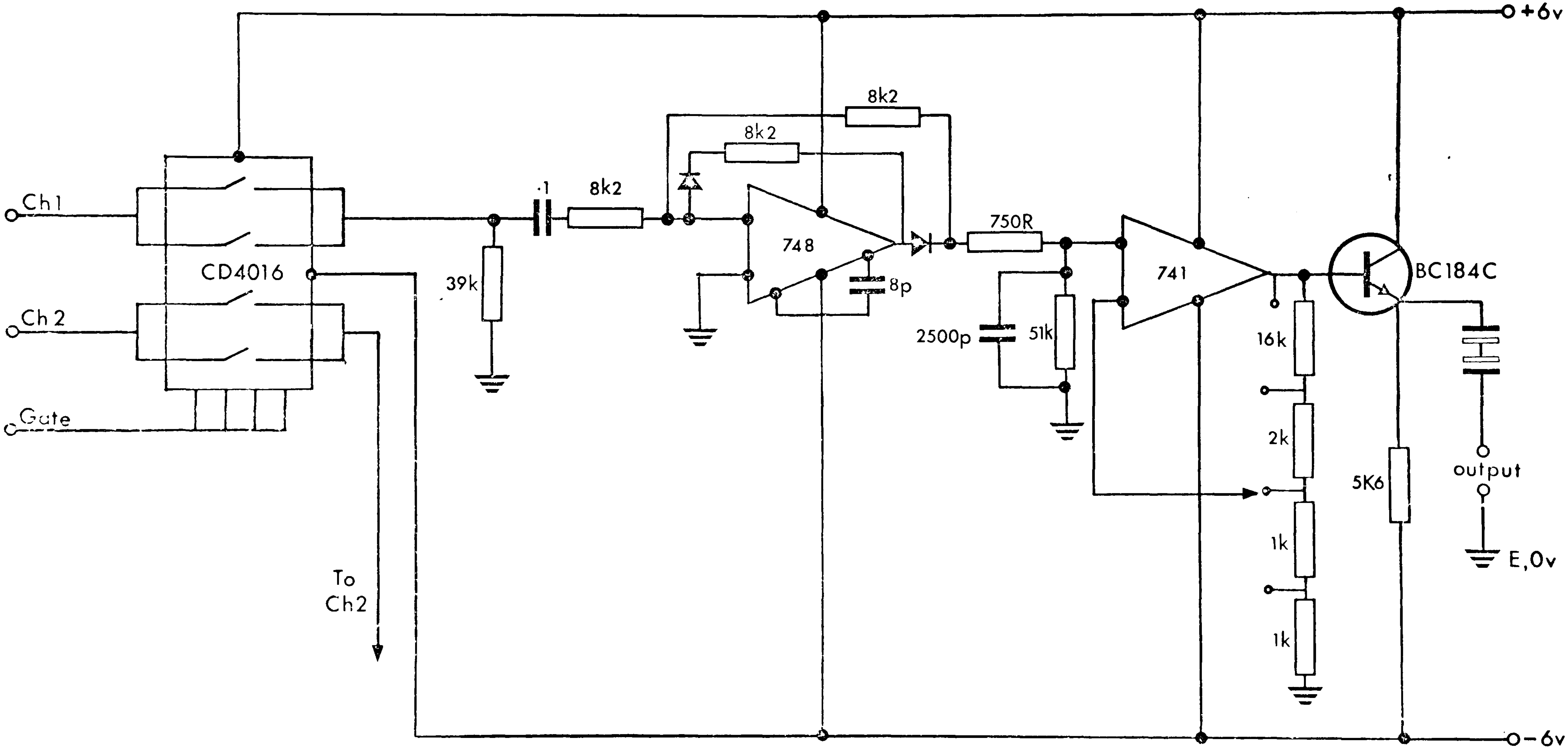


Figure 2.7 Circuit diagram of switch and detector

CHAPTER 3

ULTRASONIC TRANSDUCERS

3.1 Introduction

The pulsed ultrasonic signals of the mobility aid are generated and detected by ultrasonic transmitting and receiving transducers, each type having its own particular requirements. To select the best type for each application a preliminary investigation was carried out by Rudlin[16] into the various transducers which exist. This investigation has been extended and the results have enabled the most appropriate transducers to be chosen.

During an investigation of this nature it is often necessary to test transducers under anechoic conditions. An anechoic room was therefore constructed with approximate dimensions of 15 x 8 x 8 ft. The inside walls were lined with polyurethane foam wedges 1 ft. in length; the arrangement of these is illustrated in Fig. 3.1. The room has been extensively used for testing transducers in the frequency range 1-50 kHz and the anechoic conditions have on all occasions proved adequate.

3.2 Transducer Characteristics

3.2.1 Transmitting Transducer

This is required to generate a high acoustic power in each pulse and to radiate this power uniformly over a wide angle.

Firstly, the optimum frequency for operating the aid must be selected. This choice must take into account two conflicting factors: if the wavelength of the signal is to be comparable to the granular size of the reflecting surface, so minimising the specularity of the reflected signal, then the frequency of the sound must be high. However, to counterbalance this, the absorption of ultrasound in air increases rapidly with increase in frequency. For practical purposes, this attenuation can be quoted as a half-value distance, i.e. the distance

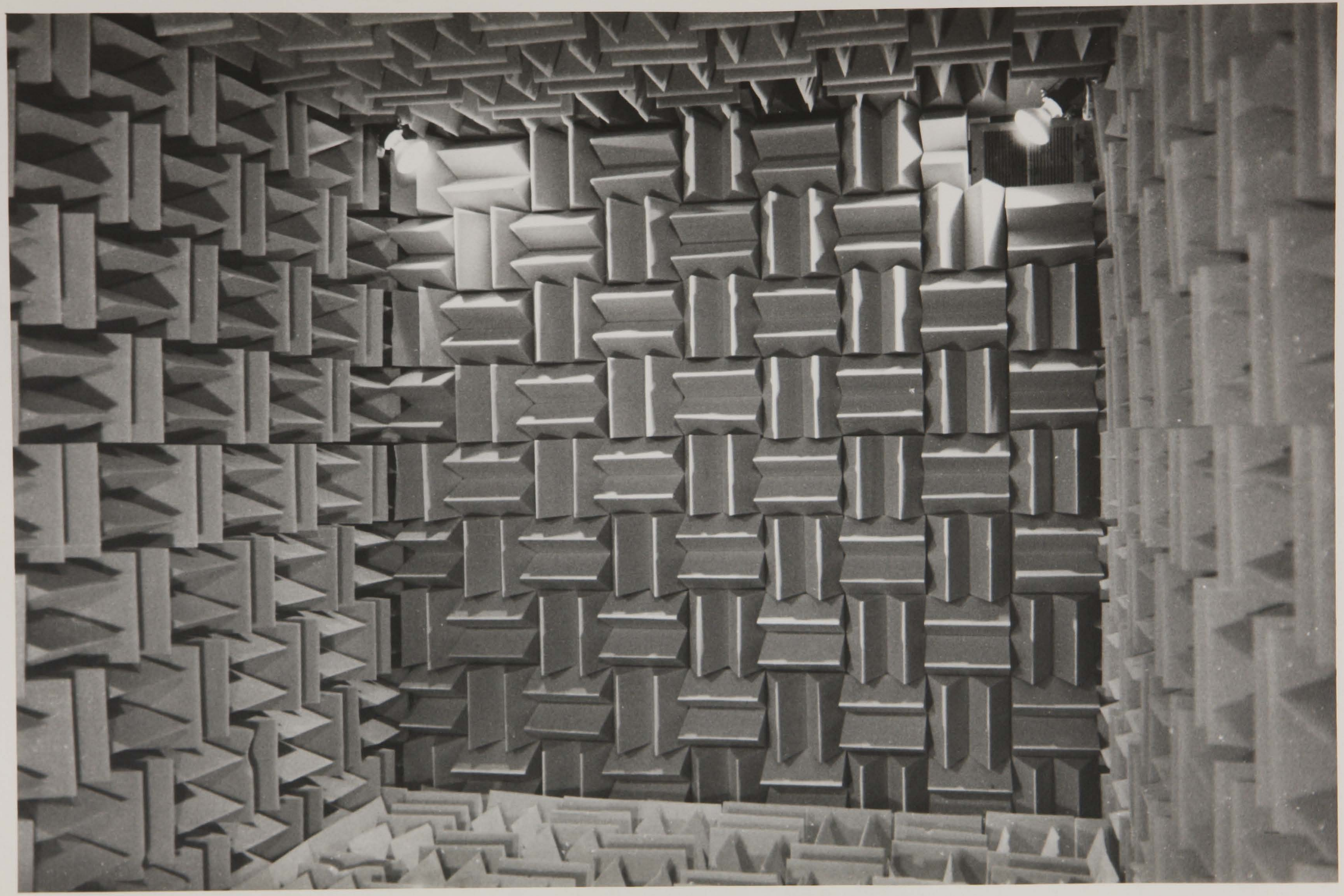


Figure 3.1 Interior view of anechoic room showing lining of foam wedges

at which the sound pressure has dropped by 50%. Table 3.1 gives half-value distances over a frequency range relevant to the design of the aid.

Table 3.1 Half-value distances for attenuation of ultrasound in air

frequency kHz	half-value distance (m)
20	10
50	3
100	2

In view of the fact that the maximum distance travelled by the signals from the aid is approximately 20 metres, the use of frequencies approaching 100 kHz would result in substantial signal attenuation. However, the use of frequencies as low as 20 kHz would undesirably increase the specularity of the echoes. The compromise, therefore, appears to rest in the choice of a frequency band around 50 kHz. It should also be noted that at higher frequencies the dimensions of a transducer tend to be smaller, thus resulting in a reduction in sensitivity.

3.2.2 Receiving Transducers

The receiving transducers are required to be sensitive and to possess a wide frequency response. The latter characteristic is necessary to ensure a faithful reproduction of the received acoustic echo. The directional characteristics of the transducers in the ultrasonic range should be similar to those of the ear at audio frequencies so that accurate directional information can be provided. The accuracy of this information relies on close matching between the two receiving transducers, both in directional characteristics and frequency response. Finally, as the receiving transducers are to be worn on the head, a small physical size is desirable so that a cosmetically acceptable arrangement can be obtained.

3.3 Ultrasonic Sources and Detectors

Ultrasonic vibrations can be produced in many ways and can be detected by many different methods. Table 3.2 gives a list of the more common sources and detectors, each item being followed by a figure which indicates the approximate upper frequency of operation.

Table 3.2 Sources and Detectors of Ultrasound

Sources	Reversible Devices (i.e. capable of operation both as sources and as detectors)
Explosions (up to 10^8 Hz)	Electrostatic (variable capacity) (up to 10^7 Hz)
Sparks (up to 10^{12} Hz)	Electrodynamic (moving coil and moving iron) (up to 10^7 Hz)
Whistles and Sirens (up to 10^5 Hz)	Piezoelectric (and electrostrictive) (up to 10^{11} Hz)
Impact (up to 10^7 Hz)	Piezomagnetic (and magnetostrictive) (up to 10^8 Hz)

In his preliminary investigation, Rudlin examined the majority of these methods for producing and detecting ultrasound. His results led to the conclusion that, of the various types examined, three could prove suitable for use in the mobility aid. These are the electrostatic, the piezo-electric and the impact types of device. The electrostatic device was the first to be investigated in detail.

3.3.1 The Electrostatic Transducer

The design of this type of device, which is capable of being used as both a transmitter and a receiver, has been described in a paper by Kuhl et al.[17] Fig. 3.2 shows the construction, which consists of a metalised plastic diaphragm lightly stretched, and resting with its non-metalised surface on a circular metal backplate which has concentric grooves. The dielectric thus consists of a thin insulating diaphragm and a layer of air whose thickness varies from zero at points of contact between the diaphragm and backplate to fractions of a millimetre over the grooves. This type of design, with its small interelectrode distances, enables a high sensitivity to be obtained.

Crawford[14] and Rudlin[16] investigated the performance of an electrostatic transducer and determined the optimum polarisation voltage and frequency of operation for a typical device. Two identical transducers were set to face each other at a fixed distance apart and the output produced by applying a constant amplitude signal of variable frequency to one transducer was monitored by the second transducer. A series of curves was thus constructed showing the relationship between polarisation voltage and frequency of signal. The curves, shown in Fig. 3.3, indicate that the optimum operating frequency is 80 kHz, since above this frequency the required polarising voltage for maximum response becomes too close to the puncturing voltage to allow a large alternating signal to be applied. The puncturing voltage lies between 300 and 350 volts.

The optimum polarising voltage was thus considered to be 180 volts which allows a peak signal of approximately 100 volts. The output pressure level under these conditions was measured as 129 dB (re 2×10^{-5} N/m^2) at a distance of 10 cm. The receiving sensitivity was evaluated for a range of incident sound pressures and the average result was approximately 0.15 mV/μbar.

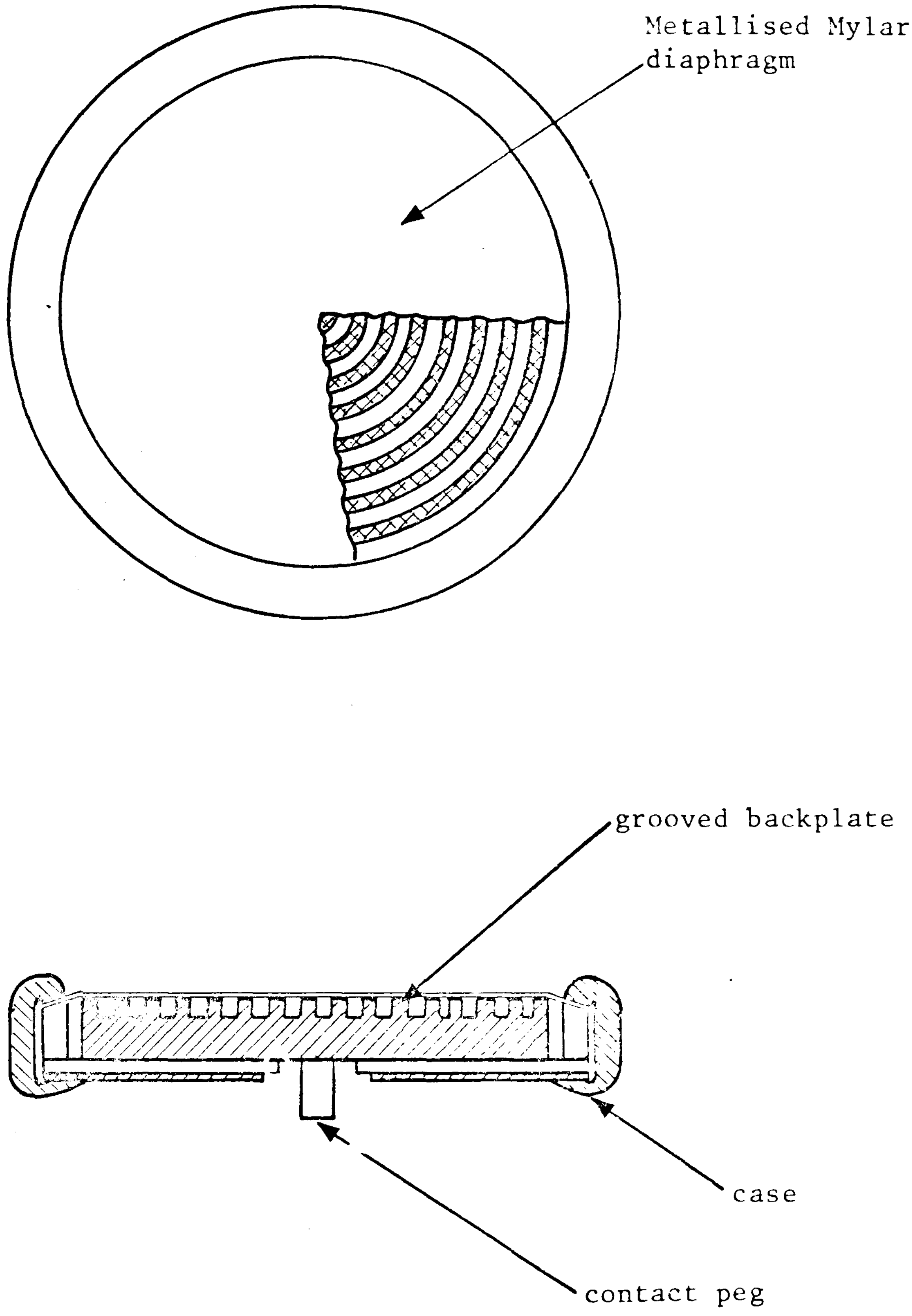


Figure 3.2 Construction of electrostatic transducer (after Kay, ref. 12)

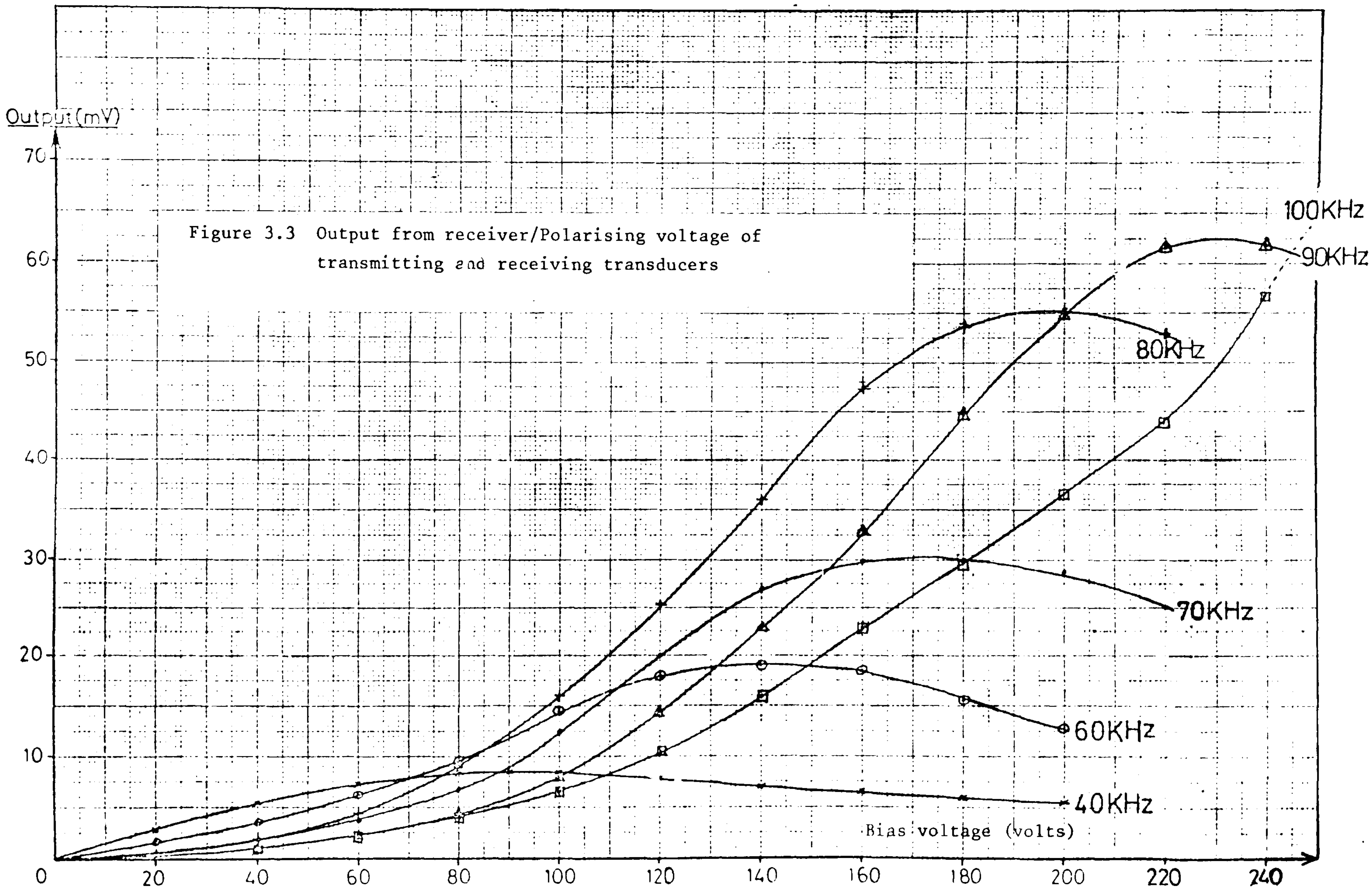


Figure 3.3 Output from receiver/Polarising voltage of transmitting and receiving transducers

The directional responses of the transducer at 80 kHz and 40 kHz are drawn in Fig. 3.4. Both show somewhat narrow main lobes of about 20 degrees and 30 degrees respectively. The frequency response, given in Fig. 3.5, shows that the electrostatic transducer has a wide bandwidth of approximately 40 kHz which will ensure a good pulse response.

An intensive development programme on the electrostatic transducer has been carried out by Kay et al. for use in the ultrasonic spectacles.[12] Despite this, however, it is still not possible to produce these transducers with reasonable tolerances. The beamwidth is not easily controllable and the frequency response particularly is variable. Each transducer must be individually made and tested, and the rate of rejection is high. An indication of the variation in transducer sensitivity as a transmitter is shown in Fig. 3.5. This spread in characteristics renders the transducers difficult to use as receivers, since pairs are required that are matched closely in polar and frequency responses. For use as a transmitter, the transducer beamwidth is too narrow and would require considerable widening. This could be accomplished by employing reflectors, although their use would be accompanied by a reduction in the sensitivity. Another possibility is the use of more than one transmitting transducer, but the difficulty in matching them complicates this approach.

3.3.2 Impact Device

The requirements of the transmitting transducer are rather stringent in that ultrasonic pulses of high intensity are to be radiated over a wide angle into air. Ultrasonic generation in air is beset by the difficulty of matching the characteristic impedance of the transducer to the low characteristic impedance of air. A possible method of improving the impedance match is the use of an extensive plate surface, which can

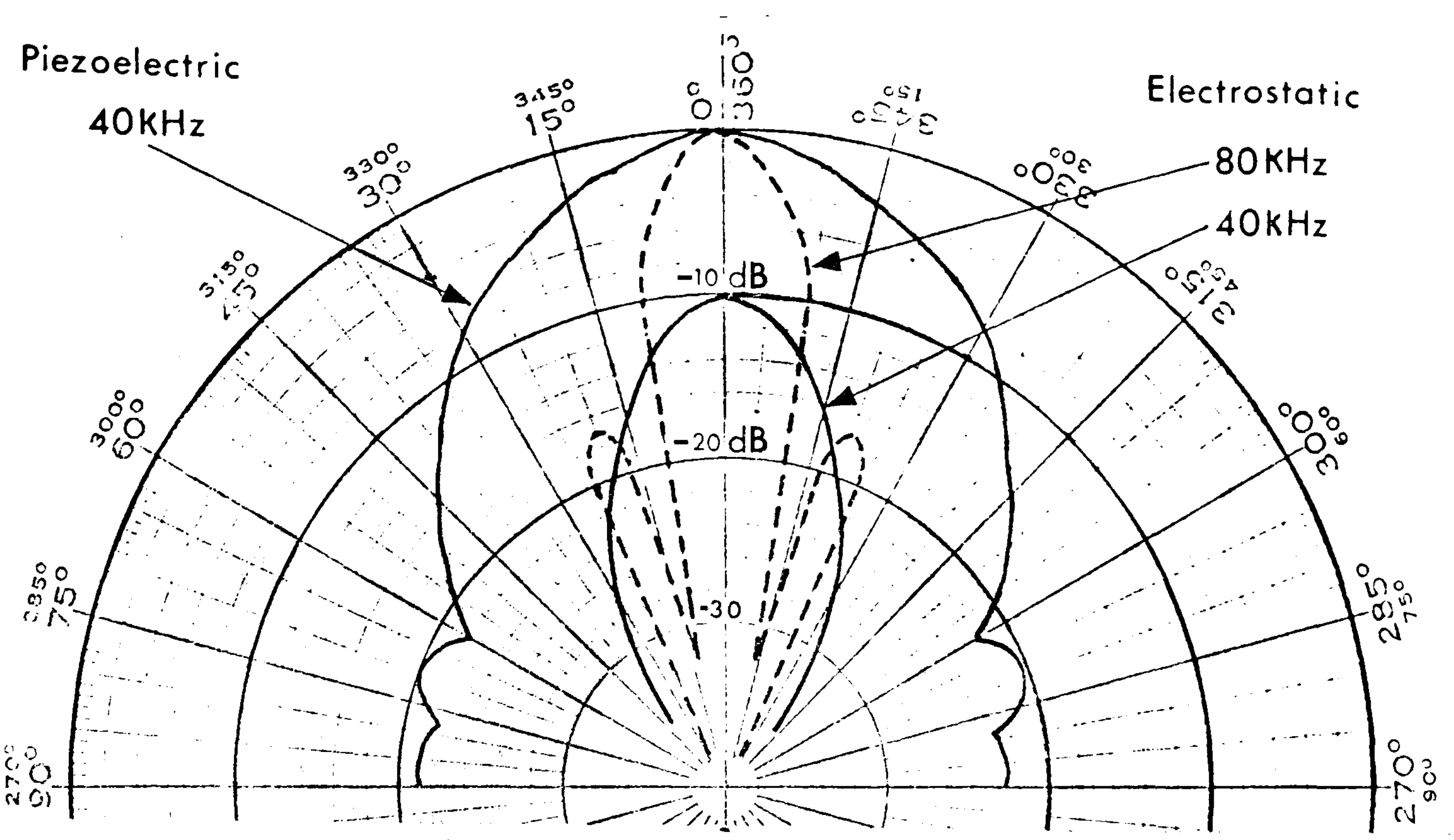


Figure 3.4 Directional responses of electrostatic and piezoelectric transducers

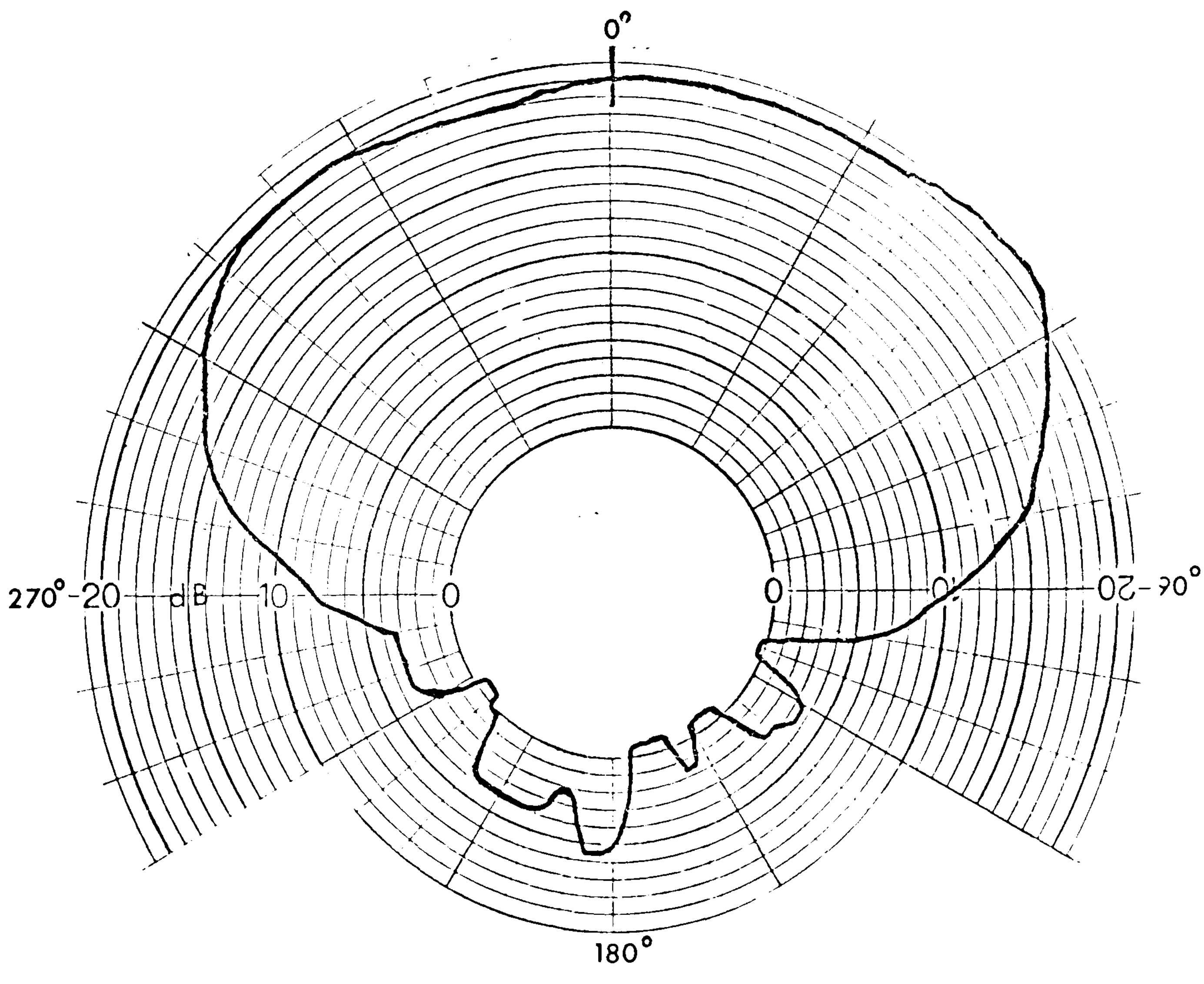


Figure 3.6 Typical directional response of a cone for use in an impact transducer

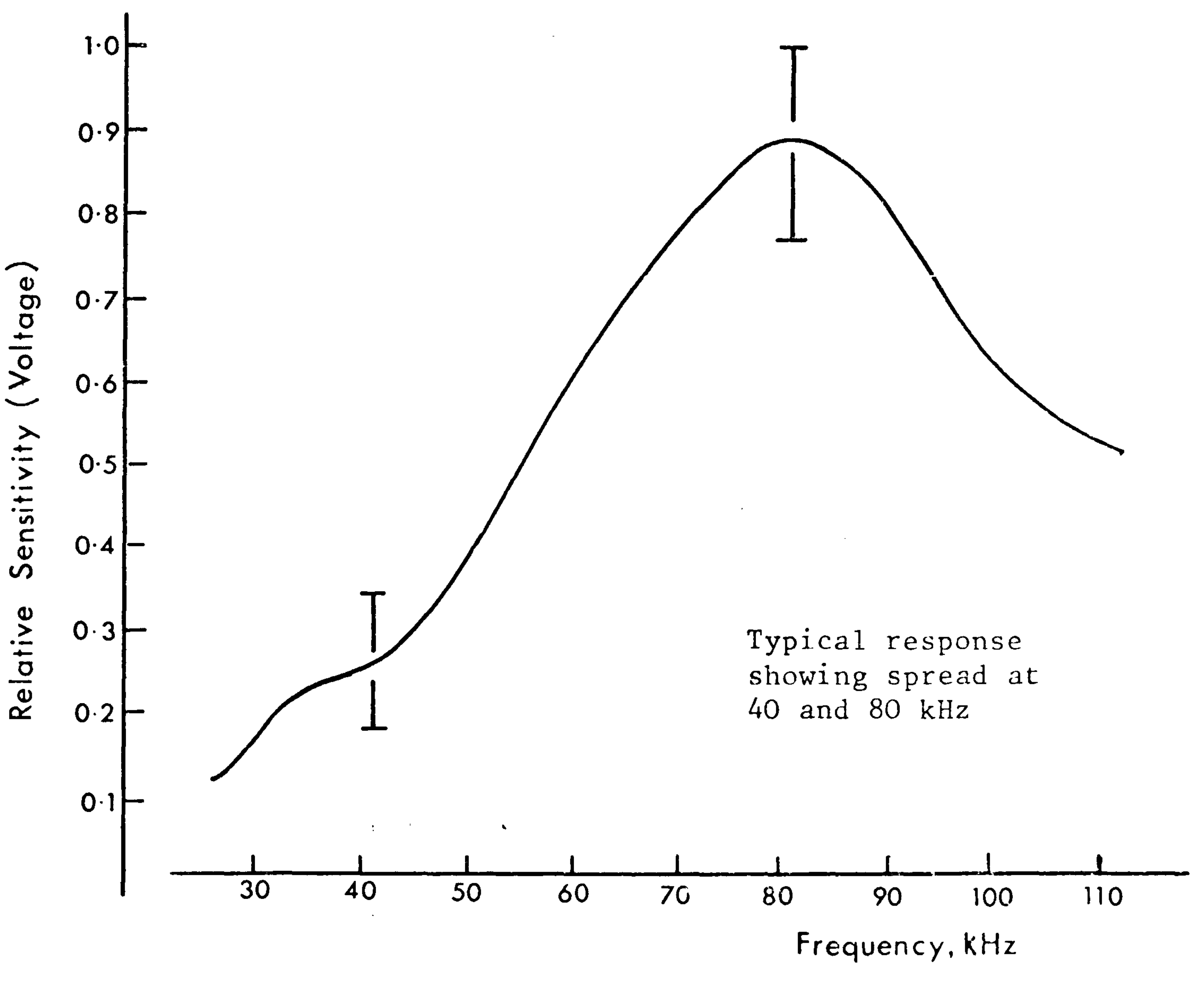


Figure 3.5 Sensitivity of an electrostatic transducer (transmitting) as a function of frequency (After Kay, ref. 12)

be used to produce powerful ultrasonic pulses by employing transient excitation; for example the plate could be subjected to successive blows of a hammer. In order to determine the usefulness of this approach in the design of a transmitting transducer for the mobility aid, an investigation into this type of device was carried out at audio frequencies, since in this range the transducer dimensions enable the modes of vibration to be studied more easily. If the results proved successful the system could be subsequently translated into the ultrasonic region.

To find a suitable shape for a metal plate an investigation was carried out by Goulbourne into the vibrational characteristics of discs and cones.[18] His results indicated that a high output level can be produced over a wide beam by the use of a thin aluminium cone driven at its centre. A typical measurement of the directional characteristic of a cone driven in this manner is shown in Fig. 3.6. Several thin cones were constructed for testing with typical examples being shown in Fig. 3.7. The attached stem enables the driving force to be transmitted to the centre of the cone.

The transient excitation was provided by an electrodynamic vibrator which is capable of generating large forces at the low repetition rates of interest, these being typically around 20 Hz. The hammer-head of the vibrator was arranged to strike the end of the stem to which the cone was attached and the resulting acoustic waveform was displayed on an oscilloscope after reception by a capacitor microphone placed in the far field. A typical oscilloscope trace, shown in Fig. 3.8, shows that after impact the transducer exhibits a main frequency of oscillation and the radiated acoustic pulse has a fast rise and slow decay time. The acoustic output of a cone increases as the thickness decreases so that the most suitable cones for use in a

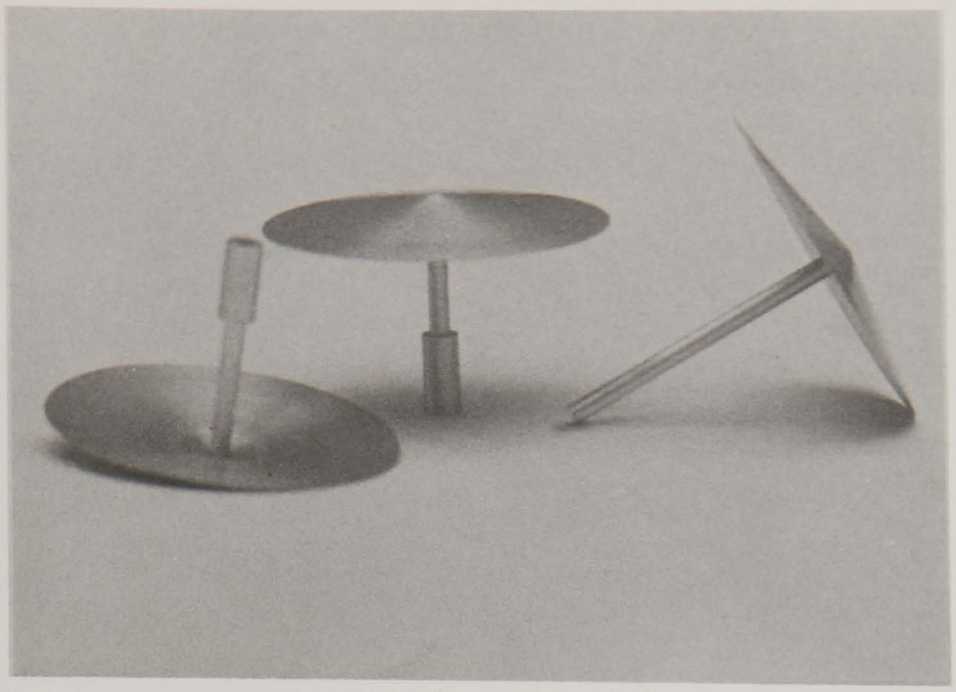


Figure 3.7 Thin cones for use in impact transducer

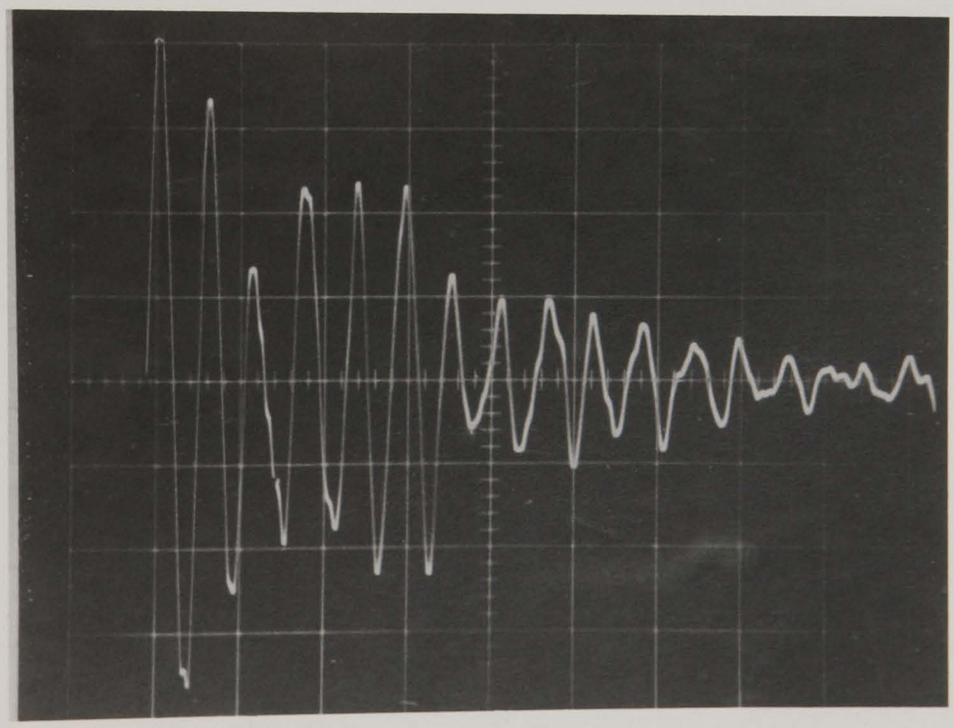


Figure 3.8 Acoustic pulse produced by impact transducer (0.5 ms/div sweep rate)

transducer are those that are very thin but still maintain mechanical rigidity. The thinnest cones that could be manufactured to this specification were 1.8×10^{-2} cm thick. When these were subjected to successive impacts, the acoustic output varied according to the contact made between hammer-head and stem, but the sound pressure level was predominantly in excess of 120 dB at a distance of 1 metre. This system, therefore, appears to be capable of producing high output powers over a wide beam. The resultant acoustic pulse has a rapid build-up time, but tends to have a slow decay which is dependent on the mechanical Q.

The main vibrational frequency of a transducer was found to be influenced by three main factors: the cone thickness, the apical angle and the mouth diameter. The main frequency increases with increase in cone thickness but as stated earlier the thickness is to remain as small as possible in order to maintain a large acoustic output. The influence of the apical angle can be seen with reference to the data shown in Table 3.3.

Table 3.3 Variation in main frequency with apical angle

Mouth Diameter cm	Thickness cm	Apical Angle degrees	Main Frequency kHz
6.5	10^{-1}	180	1.27
"	"	160	3.20
"	"	140	4.10
"	"	120	4.15

Initially the rise in frequency with reduction in apical angle is quite rapid from 180 degrees to 140 degrees but beyond this the rise is curbed.

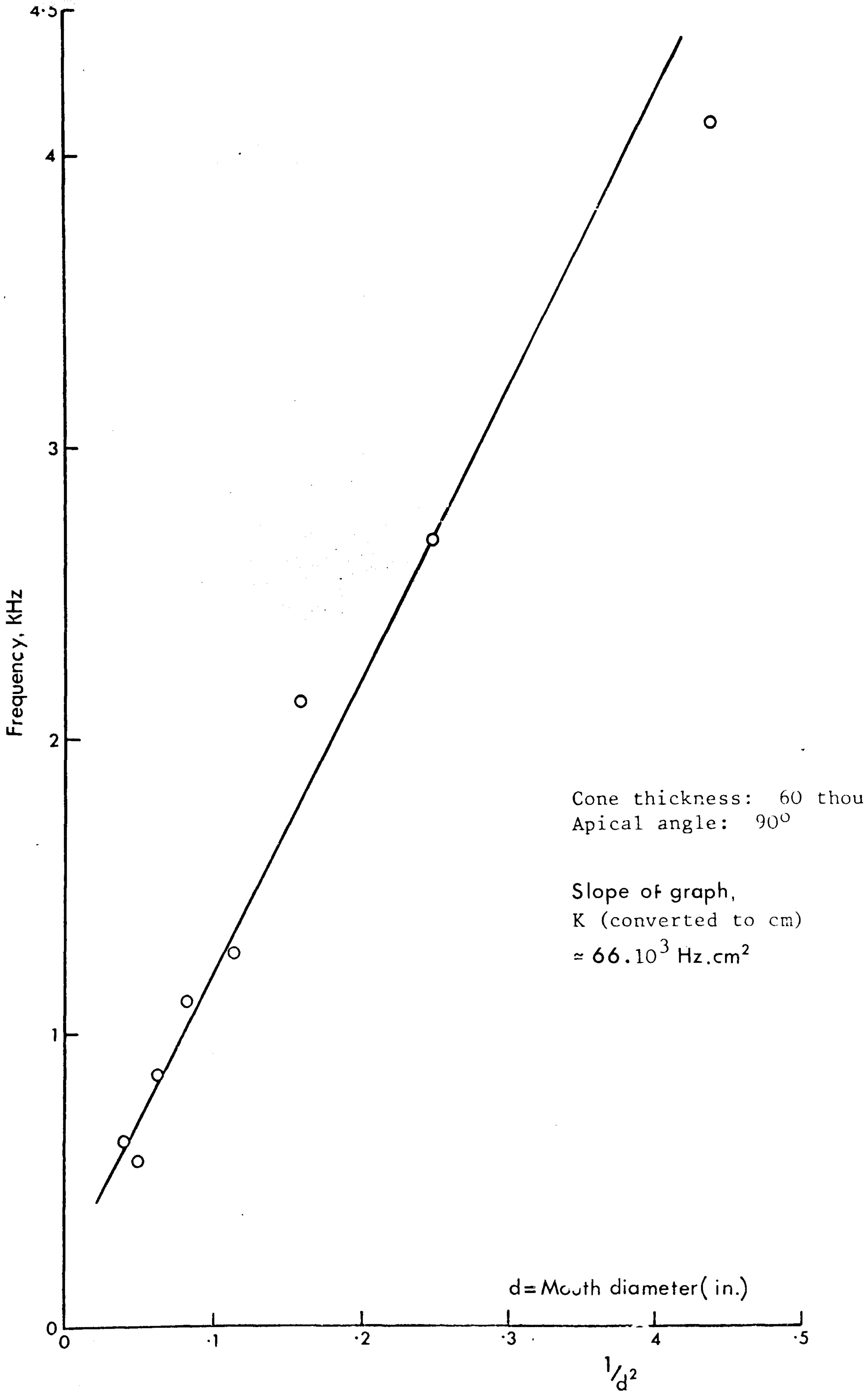


Figure 3.9 Influence of mouth diameter on main vibrational frequency of cone

The influence of the mouth diameter on the main frequency of vibration is that the frequency obviously increases with decrease in mouth diameter. The frequency and mouth diameter appear to be related by an inverse square relationship since a plot of frequency against the inverse square of the mouth diameter, Fig. 3.9, was linear. The constant of proportionality for the cones measured is approximately 66.10^3 which means that a cone of the same apical angle and thickness but operating at say 40 kHz will have a mouth diameter of approximately 1.3 cm.

A transducer with these dimensions no longer provides an extensive surface to the air and consequently cannot provide the impedance match that was hoped for. Furthermore, other problems indicated that it would be difficult to adopt this transducer for use in the aid. In particular, the design of a small impact system with sufficient force to give the required output whilst consuming little power was a major obstacle. Also, it was difficult to envisage a simple method for eliminating the undesirable audible output produced at each hammer stroke. For the present, therefore, it has been decided to forego further development in this direction.

3.3.3 The Piezoelectric Transducer

Ceramic piezoelectric materials such as lead zirconate titanate are capable of generating high acoustic powers at a high efficiency.[19] However, the high characteristic impedance of these materials causes considerable difficulty in obtaining good energy transfer to and from air. The coupling to air can be increased by the use of a bilammelar (or 'bimorph') transducer. A typical transducer illustrating the principle of this arrangement is shown in Fig. 3.10. The transducer consists of two flat ceramic plates oppositely poled and cemented

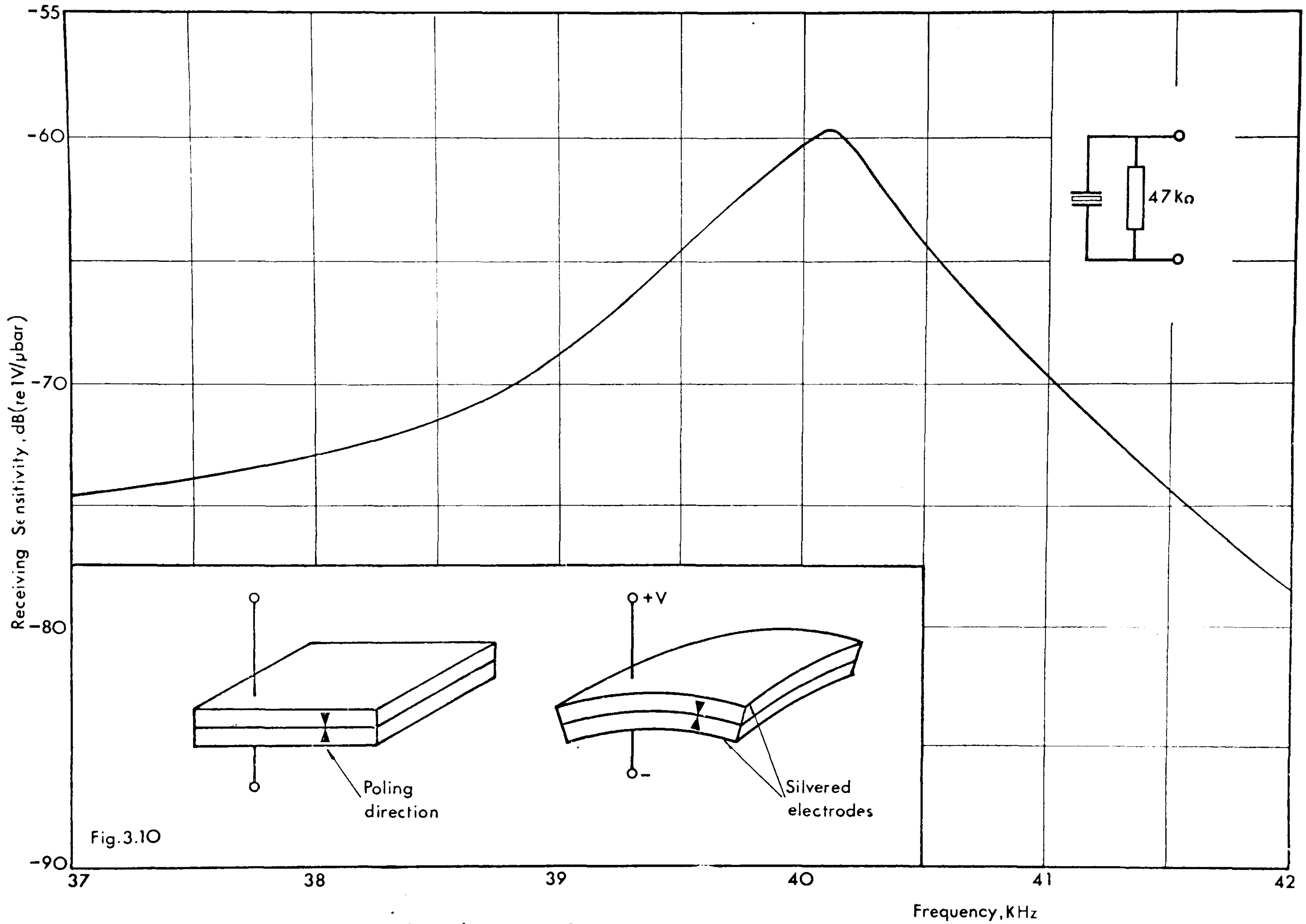


Figure 3.10 Piezoelectric bimorph plate

Figure 3.11 Frequency response of a piezoelectric transducer (receiving) together with electrodes formed on the top and bottom surfaces. When electrically energised between these electrodes, one plate expands while the other contracts. This differential strain causes a considerable displacement, so improving the coupling. However, as the central area vibrates in counterphase with the periphery, shielding arrangements are employed to avoid cancellation of the radiated ultrasound.

Measurements have been carried out on piezoelectric bimorph transducers to determine their sensitivity, frequency response and directional response. Fig. 3.11 shows the frequency response of a typical transducer in the receiving mode. The curve shows a resonant peak at approximately 40 kHz at which the sensitivity is 1 mV/μbar. An evaluation of the bandwidth gives a figure of 0.7 kHz.

In the transmitting mode, the transducer exhibits a similar frequency response curve. The variation of acoustic output with input voltage, given in Fig. 3.12, shows that initially the output rises linearly with increase in the drive voltage but levels off as the voltage reaches 60 volts peak to peak. The limiting effect is thought to be due to temperature rise within the transducing element. The sound pressure at this drive level was measured as 112 dB (re: 2×10^{-5} N/m^2) at a distance of 1 metre.

The directional characteristic, shown in Fig. 3.4, shows a smooth response in the forward direction with a 6 dB beamwidth of 55 degrees. All the transducers tested had very similar polar diagrams which could be readily matched to within 1 dB in the sector 30 degrees to either side of the straight ahead position. Transducers could also be closely matched in frequency response by choosing similar resonant frequencies.

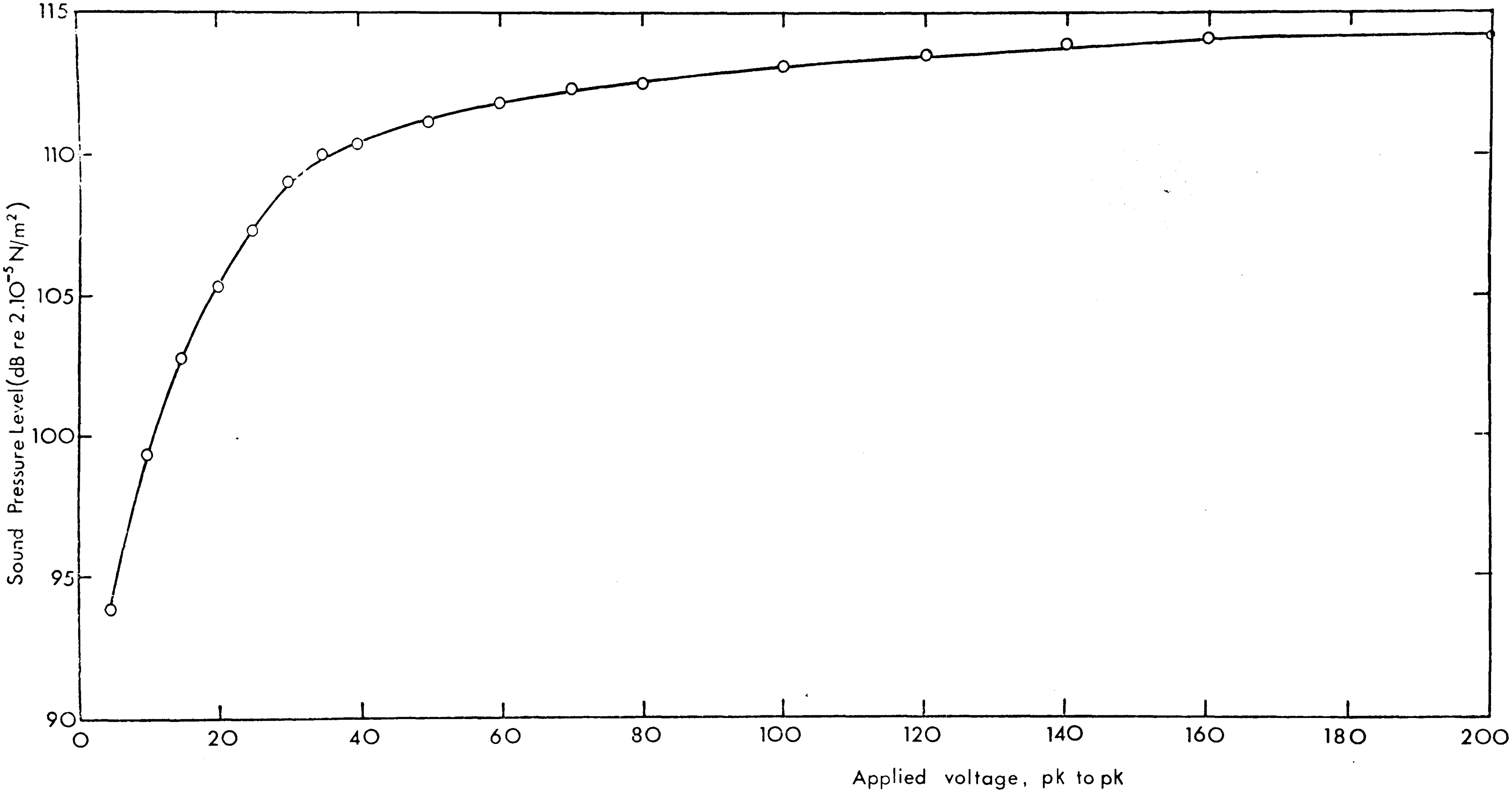


Figure 3.12 Acoustic output of the piezoelectric transducer as a function of applied voltage

3.4 Discussion

The characteristics of three types of transducer examined are compared in Table 3.4.

Table 3.4 Comparison of Transducer Characteristics

	Unit	Electrostatic Device	Piezoelectric Device	Impact Device
Transmitting Sensitivity	dB re 2×10^{-5} N/m^2	109	112	>120
Receiving Sensitivity	mV/μbar	0.15	1.0	-
Frequency of Operation	kHz	80	40	~3
Bandwidth	kHz	40	0.7	-
Beamwidth	degrees	20	55	150

The choice between the three types can be reduced to two since the impact device was not made in an ultrasonic version and therefore cannot be considered in the final selection. The sentivities of the piezoelectric and electrostatic devices are of the same order of magnitude in both the transmitting and receiving modes; although the piezoelectric device appears to be slightly more sensitive. The remaining characteristics of the two types of transducer differ widely. The frequency of 80 kHz at which the electrostatic device operates most effectively means that the half-value distance for the attenuation of ultrasound in air is only slightly over 2 metres. Clearly, this is a low value when compared to 20 metres; the maximum distance travelled by the signals of the aid. The half-value distance at 40 kHz, which is the operating frequency of the piezoelectric device, is approximately 4 metres. Use of piezoelectric transducers at 40 kHz will therefore result in considerably less signal attenuation due to air absorption than use of the electrostatic device at 80 kHz. The latter device may of course be used at 40 kHz but its operation under these conditions is less efficient.

The directional characteristics of both devices exhibit a smooth response in the forward direction. The 6 dB beamwidth of the piezoelectric transducer at 40 kHz, however, is greater than that of the electrostatic by a factor of three at 80 kHz and a factor of two at 40 kHz. The 55 degree beamwidth of the piezoelectric device is narrow when compared to the wide directional response of the human ear (shown in Chapter 5), but clearly it is the more favourable of the two for use as a receiver. The choice is the same for the transmitter, where a wide beam is also desirable.

The frequency responses of the two devices differ widely in character. The piezoelectric device exhibits a strong resonant peak giving a bandwidth of 0.7 kHz, whilst the electrostatic device has a less pronounced peak and a bandwidth of approximately 40 kHz. The wide bandwidth of the electrostatic device will ensure a good pulse response whereas that of the piezoelectric device will result in considerable distortion of the pulse shape.

On the basis of these comparisons it is necessary to draw a conclusion as to which type of device is more suitable for use in the mobility aid. The decision appears to rest on whether the more favourable directional characteristics, optimum operating frequency and smaller inter-transducer variations of the piezoelectric device are sufficient to offset the substantially larger bandwidth provided by the electrostatic device. Certainly the difficulty in matching the electrostatic transducers renders them particularly difficult to use as receivers, but on the other hand the narrow bandwidth of the piezoelectric transducers will give a poor pulse response in a transmitting/receiving system.

The problem was resolved in favour of the piezoelectric device since the literature indicated[20,21] that the bandwidth of a piezoelectric transducer can be extended up to about 20 kHz by connection of a suitable tuning inductor. This extended bandwidth will considerably improve the pulse response in a transmitting/receiving system using piezoelectric transducers. They were therefore adopted in an experimental device with a view to extending their bandwidth in the manner described.

CHAPTER 4

OPTIMISATION OF TRANSDUCER CHARACTERISTICS

4.1 Introduction

It is the intention, in the design of the mobility device, to enhance a blind person's ability to use echoes as a source of information about his physical surroundings. Kohler[4] and Rice[22] have indicated that subjects, when performing echo perception experiments, usually give the best results when using a click sound which contains frequencies throughout the audible range. It is desirable, therefore, to closely simulate this form of signal.

An examination of the auditory display of an experimental aid using piezoelectric bimorph transducers, however, revealed that the clicks emanating from the earpieces had very little frequency content above 1 kHz; indeed, they would be more aptly termed 'thuds'. Also, it was possible to perceive part of the transmitted pulse being detected directly by the receiver, despite the latter being muted for 3 ms after each transmission.

These undesirable effects are mainly attributable to the bandwidth limitations of the transmitting, receiving and audio transducers. The bimorph flexure elements in the transmitting and receiving transducers are resonant devices with a high Q-factor; typically about 60 in the electrically unloaded condition, which means a 3 dB bandwidth of about 0.7 kHz. The ultrasonic output pulses from such a narrow-band system necessarily have slow rise and fall times as illustrated in Fig. 4.9(a). The envelopes of these pulses contain only low audio frequencies and decay slowly. Thus the signal produced by direct crosstalk between transmitter and receiver can persist long enough to extend past the receiver muting period, so causing the direct reception. Further, the slow rise time coupled with a short driving pulse means that the transmitting elements do not have sufficient time to reach their peak output, resulting in a waveform almost triangular in shape.

Ideally the acoustic output presented to the ear of the user should consist of rectangular pulses whose energy is spread throughout the audible range. However, it is not easy to obtain the requisite extended ultrasonic bandwidth with piezoelectric bimorph transducers. An investigation was carried out, therefore, to see if the transducers could be modified to produce a response containing sufficient spectral information to closely simulate natural echo-location. Modifications which would unduly degrade the overall sensitivity of the system were not permissible.

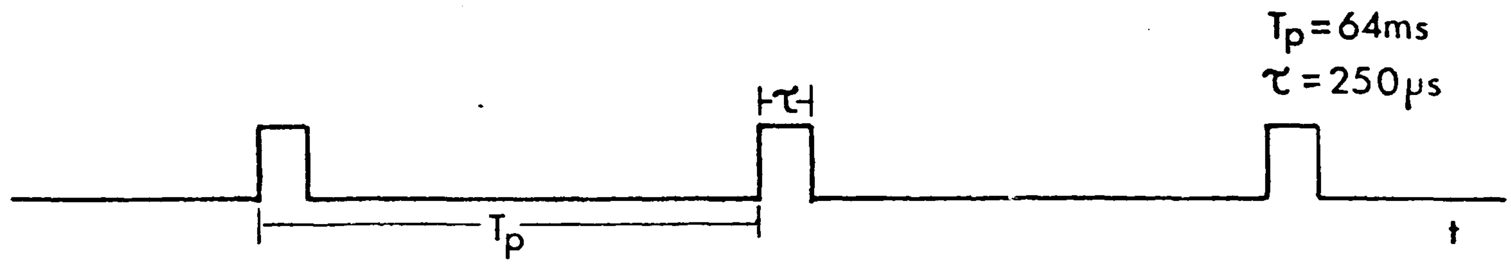
4.2 Signal Considerations

The signal to be transmitted, i.e. the modulating signal, consists of rectangular pulses, of 250 μsec. duration occurring 16 times per second. These pulses are used to amplitude modulate a carrier of nominal frequency 40 kHz. Diagrams of the modulating signal and the modulated wave are shown in Fig. 4.1(a).

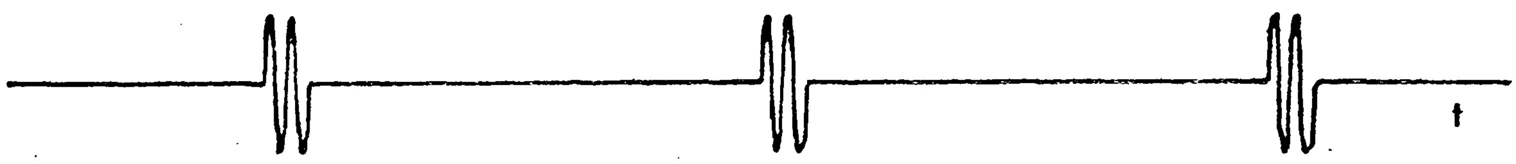
The frequency spectrum of the modulating function, shown in Fig. 4.1(b), follows a ${}^{\sin x}/x$ envelope and is made up of discrete frequencies spaced ${}^{1}/T_p$ apart, where T_p is the period. When this function modulates the carrier frequency, each spectral component gives rise to two sets of sidebands with a spectral distribution as shown in Fig. 4.1(c), in which much of the energy lies within the range 40 ± 4 kHz.

It is clear that the pass band required for the transmitting and receiving transducers is, at the minimum from 36 kHz to 44 kHz, i.e. 8 kHz. However, this will only pass components below the first zero of Fig. 4.1(b) and will distort the pulse shape considerably. A more faithful representation of the pulse form would be obtained if the pass band extended from 32 kHz to 48 kHz. This requires a bandwidth of 16 kHz, and it can be deduced[23] that this would enable the pulses to build up and decay in about 88 μsec. or 3 oscillations. This seems a reasonable criterion to aim towards, if a good pulse shape is to be preserved.

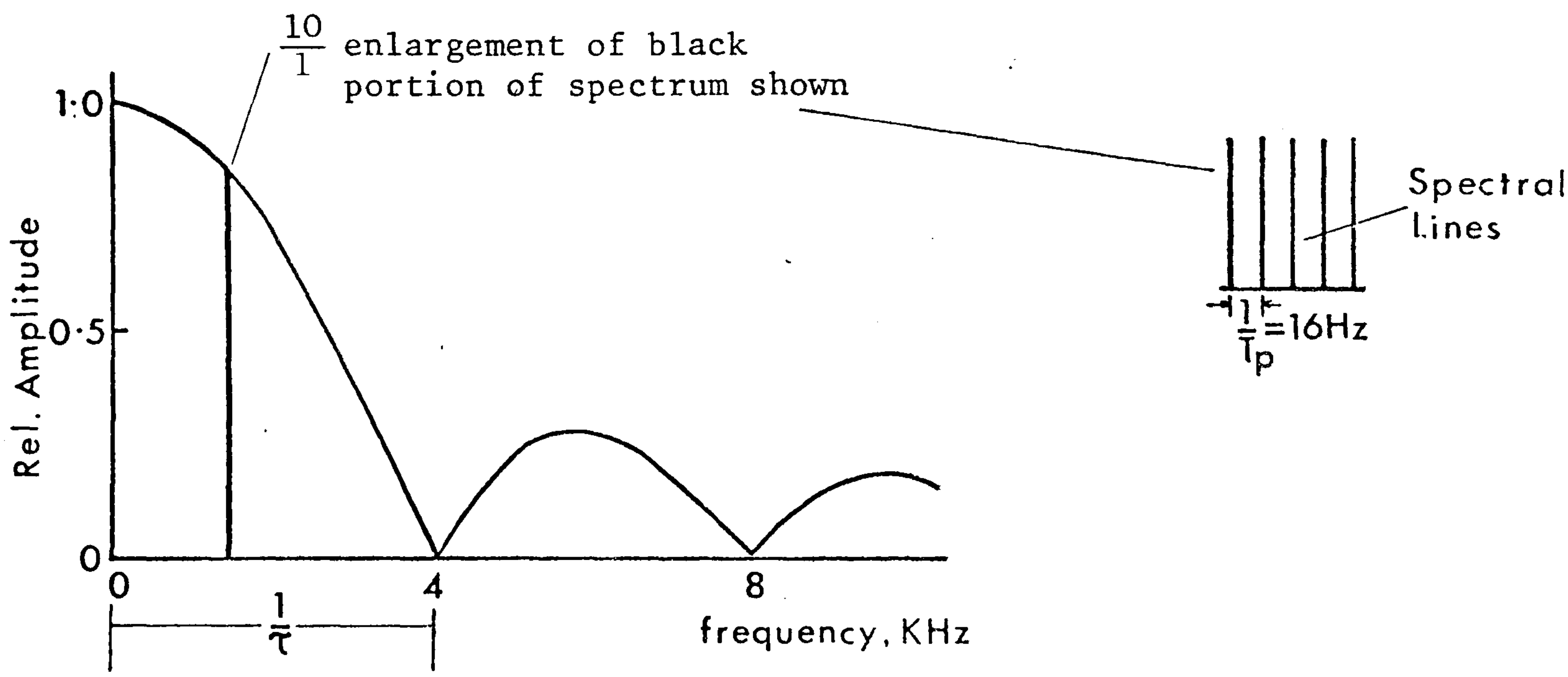
(a) A rectangular pulse modulating signal



A pulse modulated wave



(b) The frequency spectrum of a rectangular pulse



(c) The frequency spectrum of a pulse-modulated wave

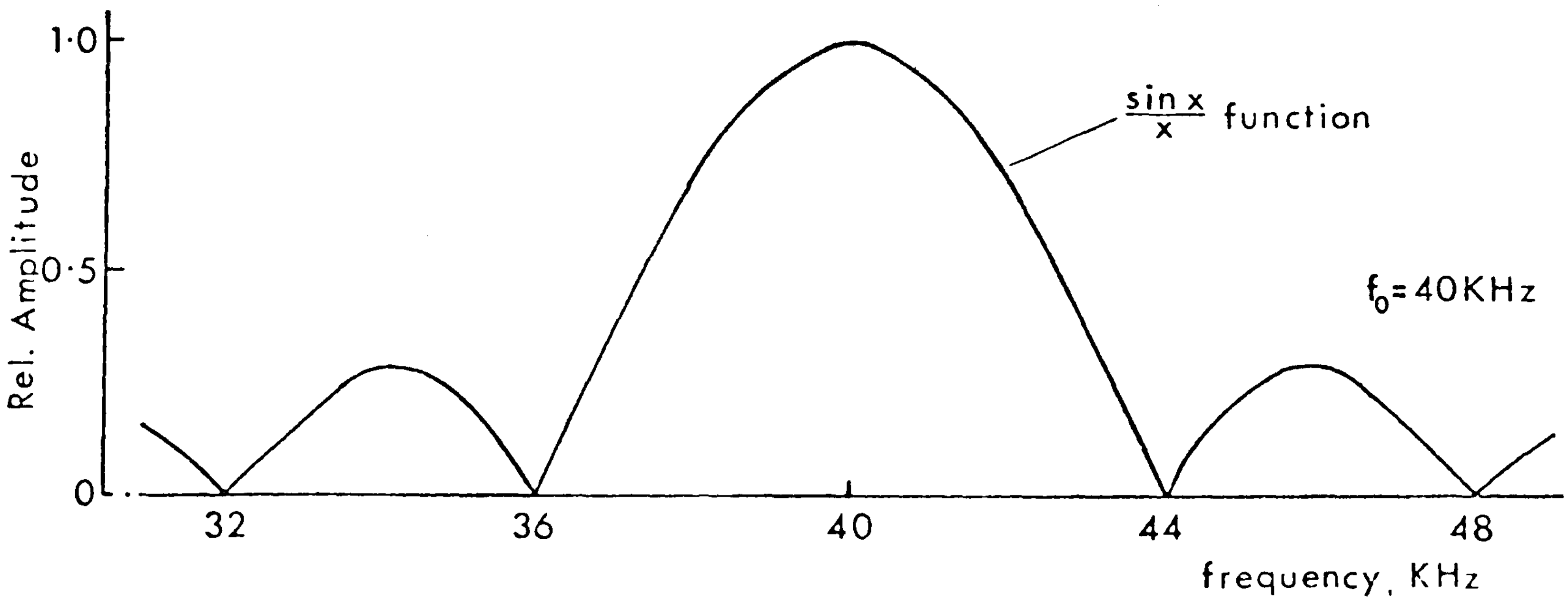


Figure 4.1 Signal characteristics

It can be seen from Fig. 4.1(b) that such a bandwidth would give an output signal after detection containing frequencies up to 8 kHz. It must be ensured, therefore, that the earpieces selected have an adequate frequency response, in order that the signals may be passed without further appreciable distortion.

It is considered that a device with a bandwidth giving an audio output extending to 8 kHz should reasonably adequately simulate the natural echo perception process, since many people do not in any case have good hearing above 8 kHz.

4.3 Characteristics of the Piezoelectric Transducer

The best insight into the piezoelectric ceramic transducers can be found by analysing mathematically the equivalent circuit given in the I.R.E. standards on Piezoelectric Crystals[24]. The circuit is shown in Fig. 4.2.

In the electromechanical system represented by this equivalent circuit, the crystal itself has a vibrating mass that appears to the circuit to be a motional inductance, L_1. The mechanical losses of the vibrating element, which include the molecular friction and acoustic loading by the ambient air, appear as an equivalent resistance, R_1. The elasticity of the crystal appears to the circuit to be a small motional capacitance, C_1. Capacitance C_o is the static capacitance between the electrodes plated on the element, together with stray lead capacitances.

A piezoelectric crystal can operate in the series-resonant mode or in the parallel anti-resonant mode, or at some frequency between these points.

At a certain frequency, f_s, where the reactance of L_1 and capacitance C_1 are equal, given by $f_s = \dfrac{1}{2\pi\sqrt{L_1 C_1}}$, the net reactance is zero and

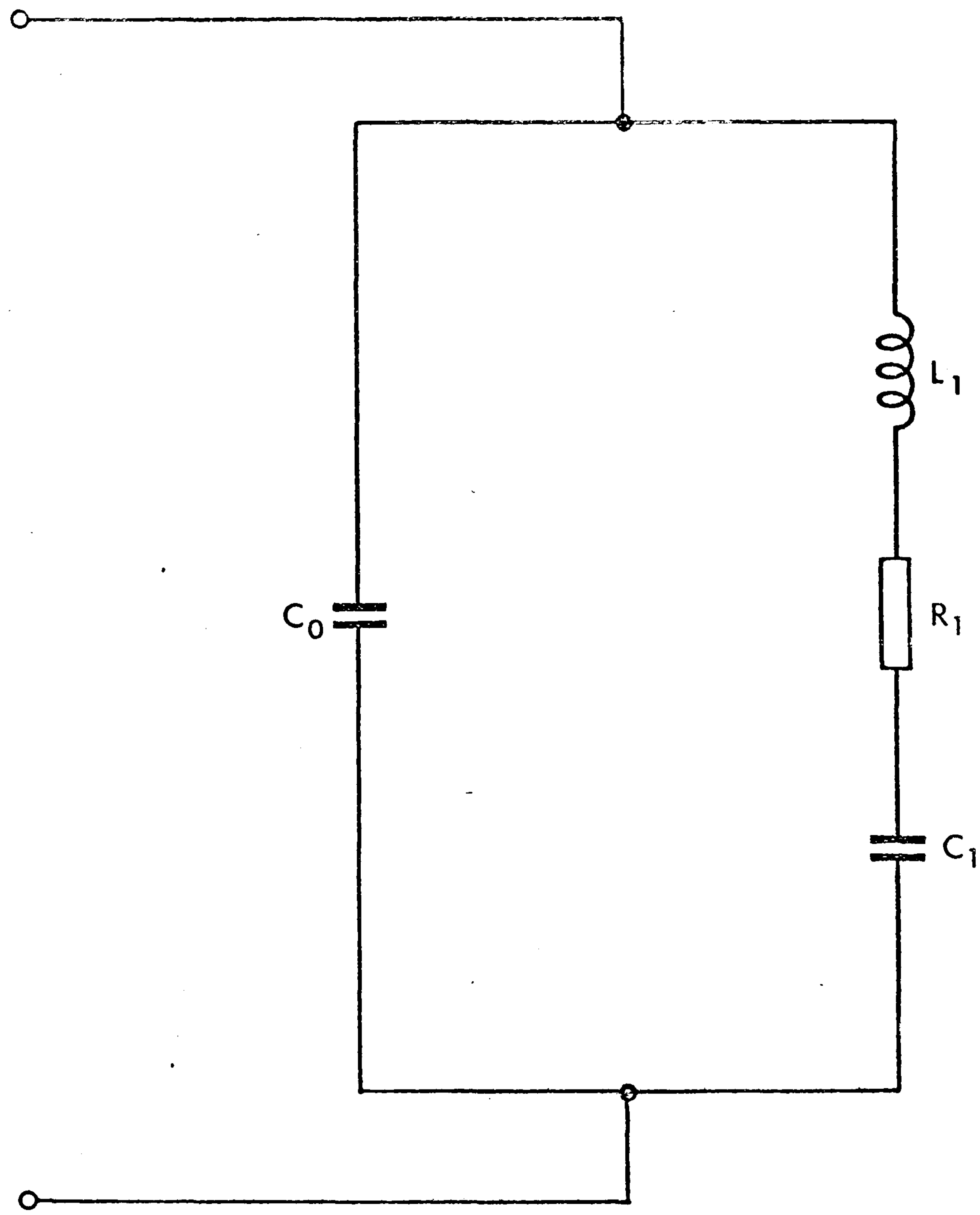


Figure 4.2 Equivalent circuit of a piezoelectric transducer

the series resonant circuit becomes the equivalent of R_1 connected in parallel with C_o. The resistance R_1 is small compared with the reactance of C_o, and series resonance will cause the impedance to drop to a minimum.

As the frequency is increased slightly to above f_s, the inductive reactance increases and the capacitive reactance decreases which produces a rapid increase in the net inductive reactance. Capacity C_o is now a significant part of the network. When $X_{L_1} - X_{C_1} = X_{C_o}$, at a frequency f_p, parallel anti-resonance occurs and the combined impedance is a maximum. The frequency separation between the series and parallel resonance points is a function of the ratio ${}^{C_1}/C_o$. This impedance variation, as a function of frequency for the bimorphs used is illustrated in Fig. 4.3.

Owing to the high electromechanical coupling factor of the piezoelectric ceramic material, the transducer characteristics are highly dependent on the electrical load to which they are connected. This could be the input resistance of a receiver amplifier, or the internal resistance of a transmitter generator, since the characteristics of the crystal under these two operating conditions are similar.

Fig. 4.4 shows the results of measurements on how the maximum response frequency and bandwidth vary with load resistance for a receiver. The maximum response frequency is that frequency at which the receiver sensitivity reaches its maximum. Fig. 4.5 clearly shows the displacement of maximum response frequency from f_s to f_p as the load resistance is increased, whilst Fig. 4.6 illustrates that a maximum bandwidth of about 3 kHz can be obtained. This bandwidth is still too small for the transmission of sharp pulses.

A more detailed analysis of the equivalent circuit of the transducer indicated that a much wider bandwidth can be obtained by tuning

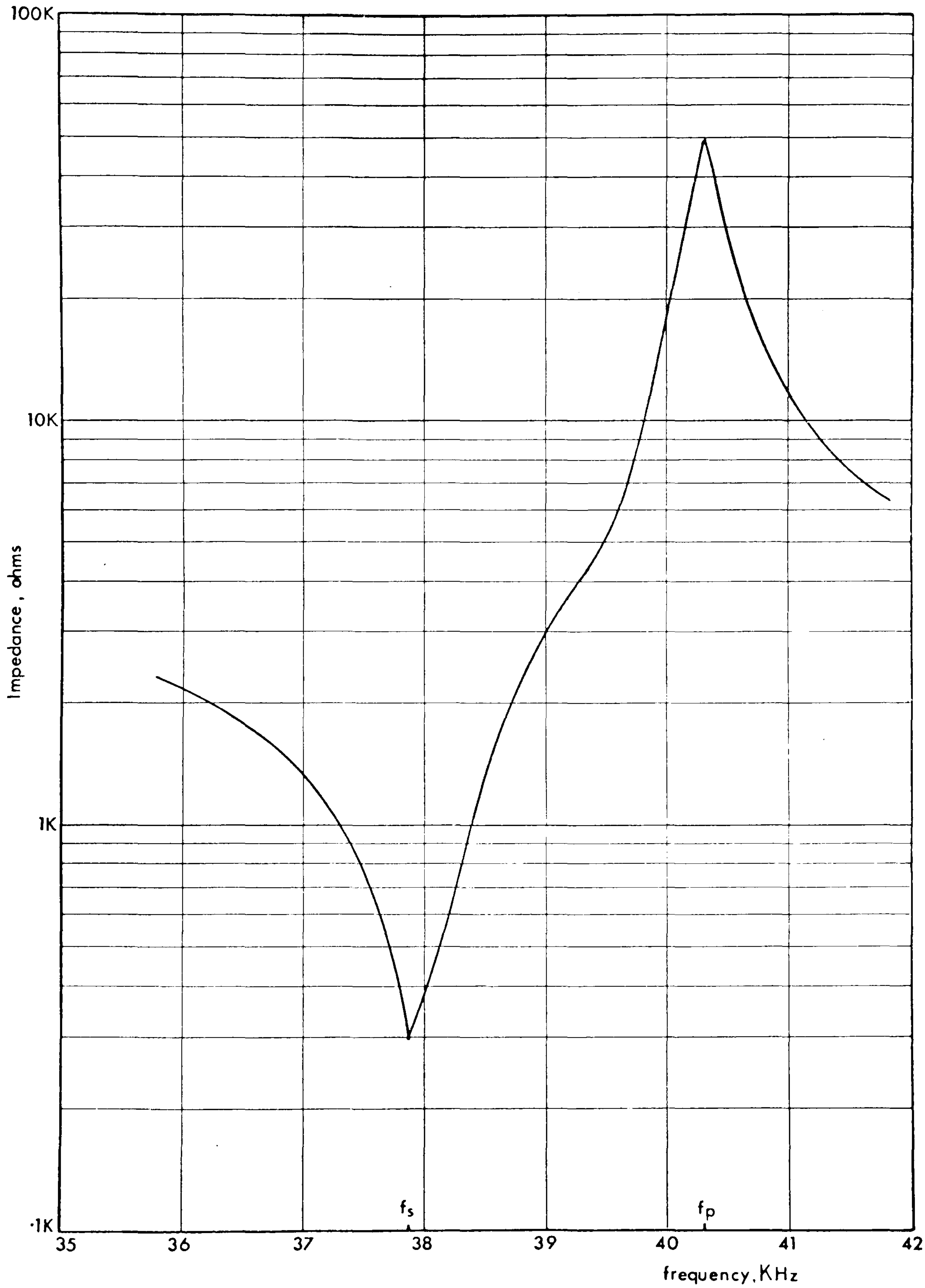


Figure 4.3 Transducer impedance as a function of frequency

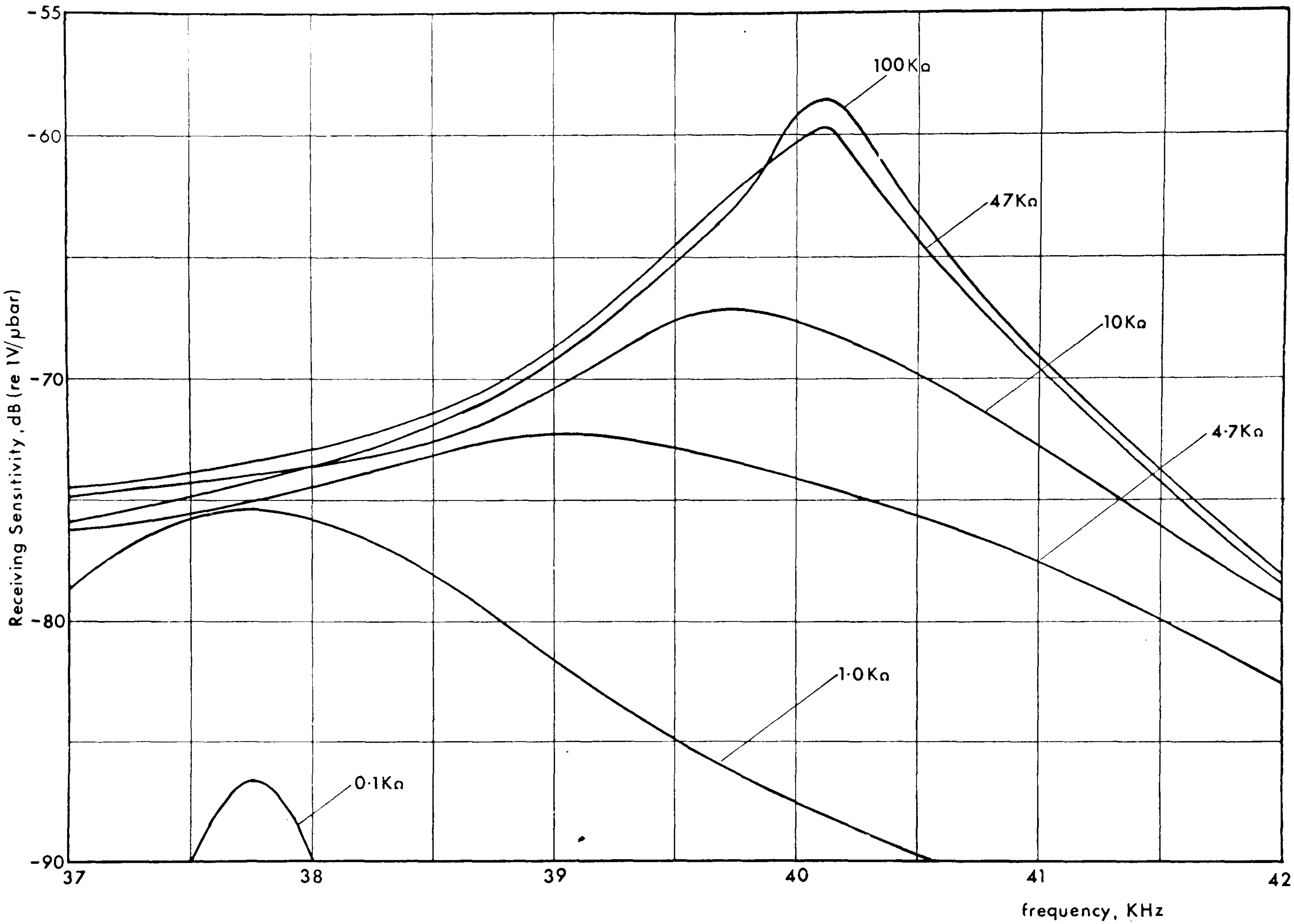


Figure 4.4 Effect of shunt resistance on the receiving response of the transducer

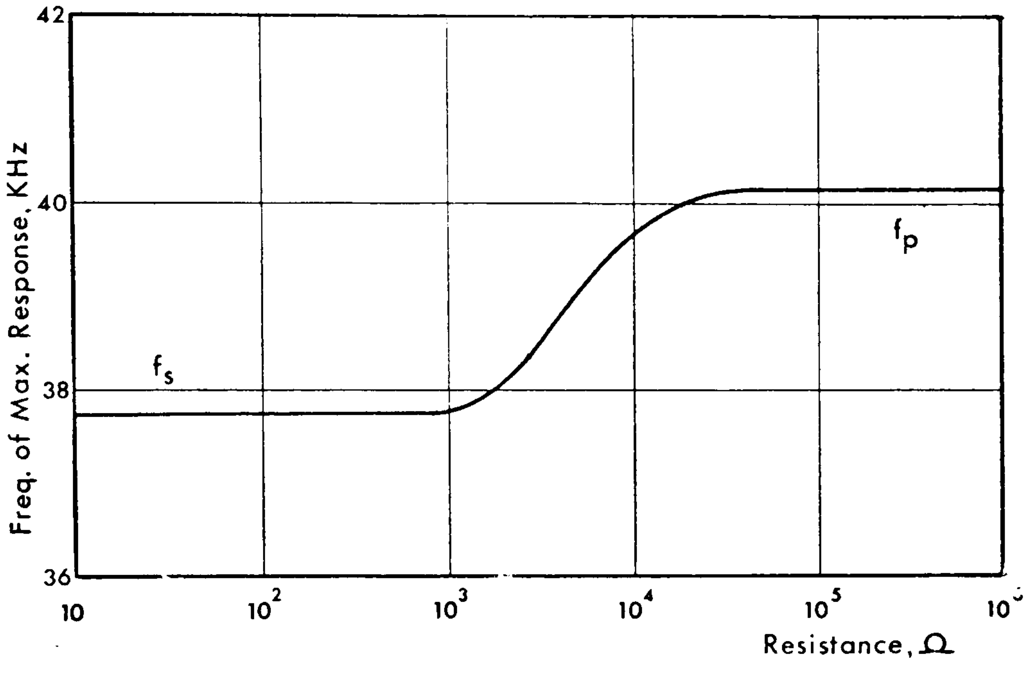


Figure 4.5 Frequency of maximum response of the transducer as a function of shunt resistance

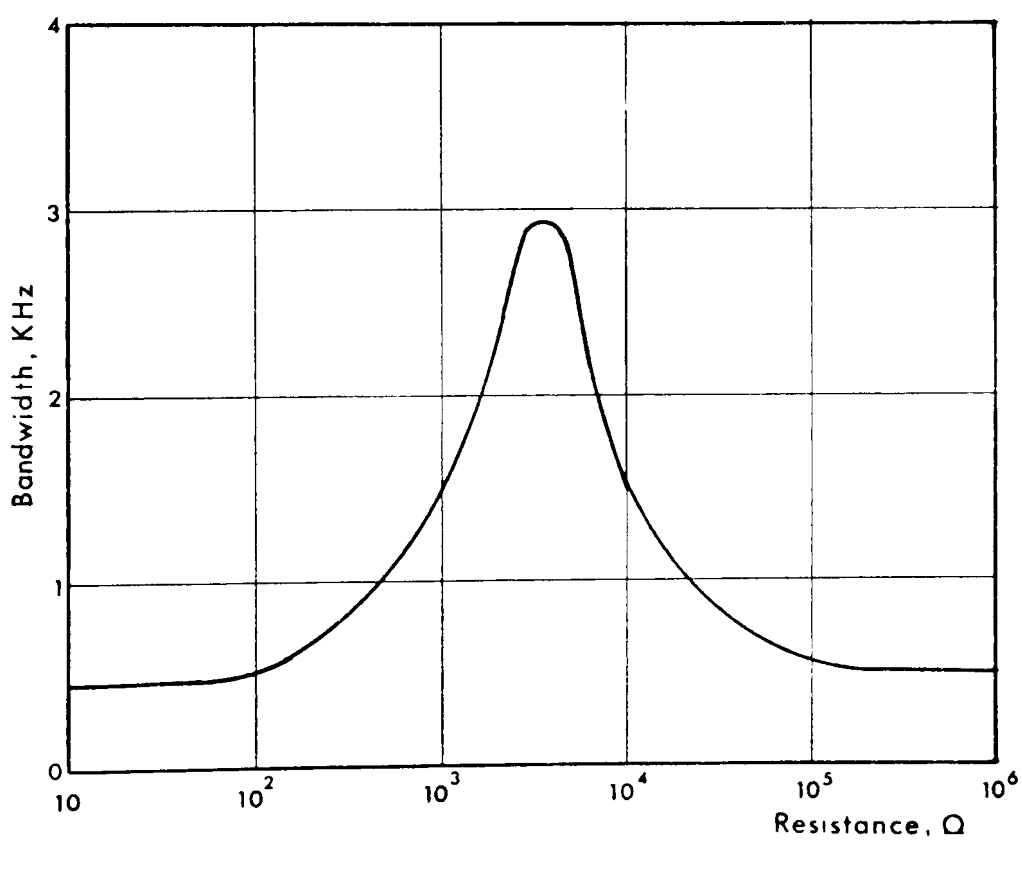


Figure 4.6 Bandwidth of the transducer as a function of shunt resistance

out the parallel capacitance, C_o, by a parallel inductance at the series resonant frequency, and damping this parallel circuit with an appropriate resistance (see Appendix 1). The results of the analysis agree with the results of similar analyses by other workers.[20,25]

4.4 Bandwidth Experiments

The theory, which has been given in detail in Appendix 1, may be summarised as follows: the transducer with the appropriate valued shunt inductance and damping resistance exhibits a transfer characteristic as a function of frequency which is a fourth power band-pass characteristic. The 3 dB bandwidth, which has been evaluated using typical values of transducer parameters, is approximately 18 kHz. The impedance as a function of frequency in the undamped condition is a double high peaked curve where the peaks have been estimated to occur at approximately 34 kHz and 47 kHz.

The bandwidth obtainable, as suggested by the theory, meets the requirements initially laid down for the transmission of sharp pulses. Detailed experimental work was carried out to show whether the theoretical predictions would be realised in practice.

A plot of impedance as a function of frequency with a parallel inductance connected across the transducer yielded the expected double high peaked curve. The optimum value of inductance was found to be 16 mH; at this value the geometric mean of the two frequencies at the peaks is equal to the series resonant frequency of the equivalent circuit. The curve is shown in Fig. 4.7 and indicates that the occurrence of the two peaks at 32 kHz and 45 kHz is in good agreement with the theory.

An experimental evaluation of the transfer function using the same value of inductance should give a similar double high peaked curve which can be suitably damped with a resistor to give a fourth power band pass characteristic. The transducer was used in the receiving mode

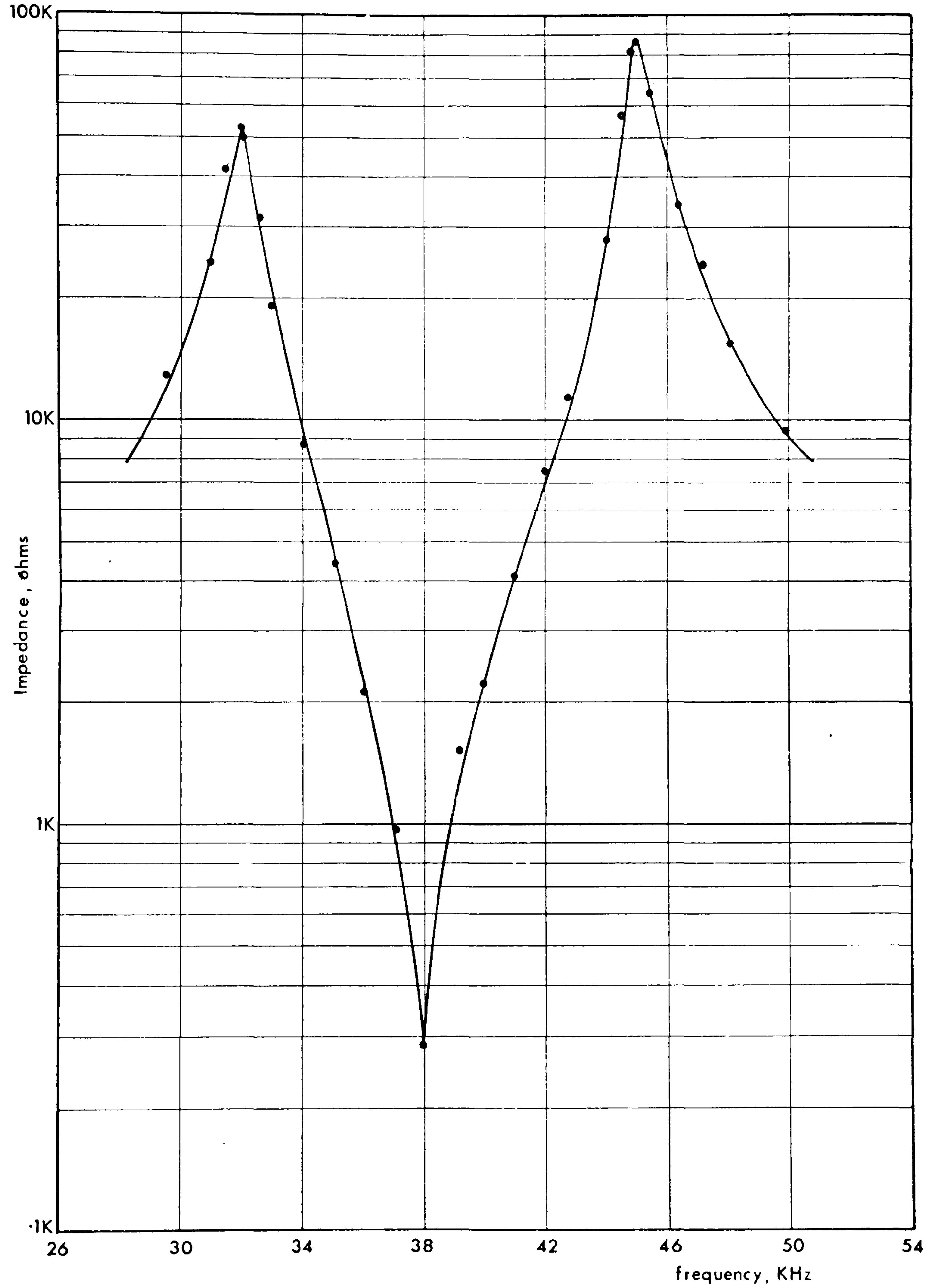


Figure 4.7 Transducer impedance as a function of frequency with a shunt tuning coil

and placed in a sound field of constant pressure and variable frequency. This was achieved by driving a high frequency loudspeaker from a signal generator and monitoring the acoustic output with a capacitor microphone placed adjacent to the receiver. The output voltage from the transducer was measured after amplification by an amplifier with a high input impedance and of known gain. Fig. 4.8 shows the family of curves plotted, which range from the double peaked undamped condition through to the single peaked overdamped condition. A damping resistance of 22 kΩ gave the flattest curve with a 3 dB bandwidth of about 16 kHz. When the loudspeaker was fed with pulsed sine waves the output of the receiving transducer consisted of pulses which built up and decayed in 80μsec. or 3 to 4 oscillations. These pulses, shown in Fig. 4.9(b), have a far better shape than those originally obtained without the tuning inductor and are considered to be of adequate sharpness for use in the aid.

The receiving transducers, however, have their cases modified in order to obtain a wider beam (see Chapter 5) and it is necessary to ensure that the transfer characteristic after modification is not appreciably different from that of the original transducer. Fig. 4.10 gives typical curves of an optimally damped modified and unmodified transducer. A comparison shows that the modified casing gives a characteristic which is largely similar to the unmodified condition. When subjected to acoustic pulses, the output of the modified transducers was almost identical to that obtained previously. This can be seen by comparing Fig. 4.9(c) with Fig. 4.9(b).

The sensitivity of the receiving transducers is not decreased by extending the bandwidth in the manner described but the modification of the casing causes a loss of 8 dB; the resulting sensitivity being -80 dB relative to 1 volt per microbar. The correspondence between the frequency responses of different receivers after modification is close; it was

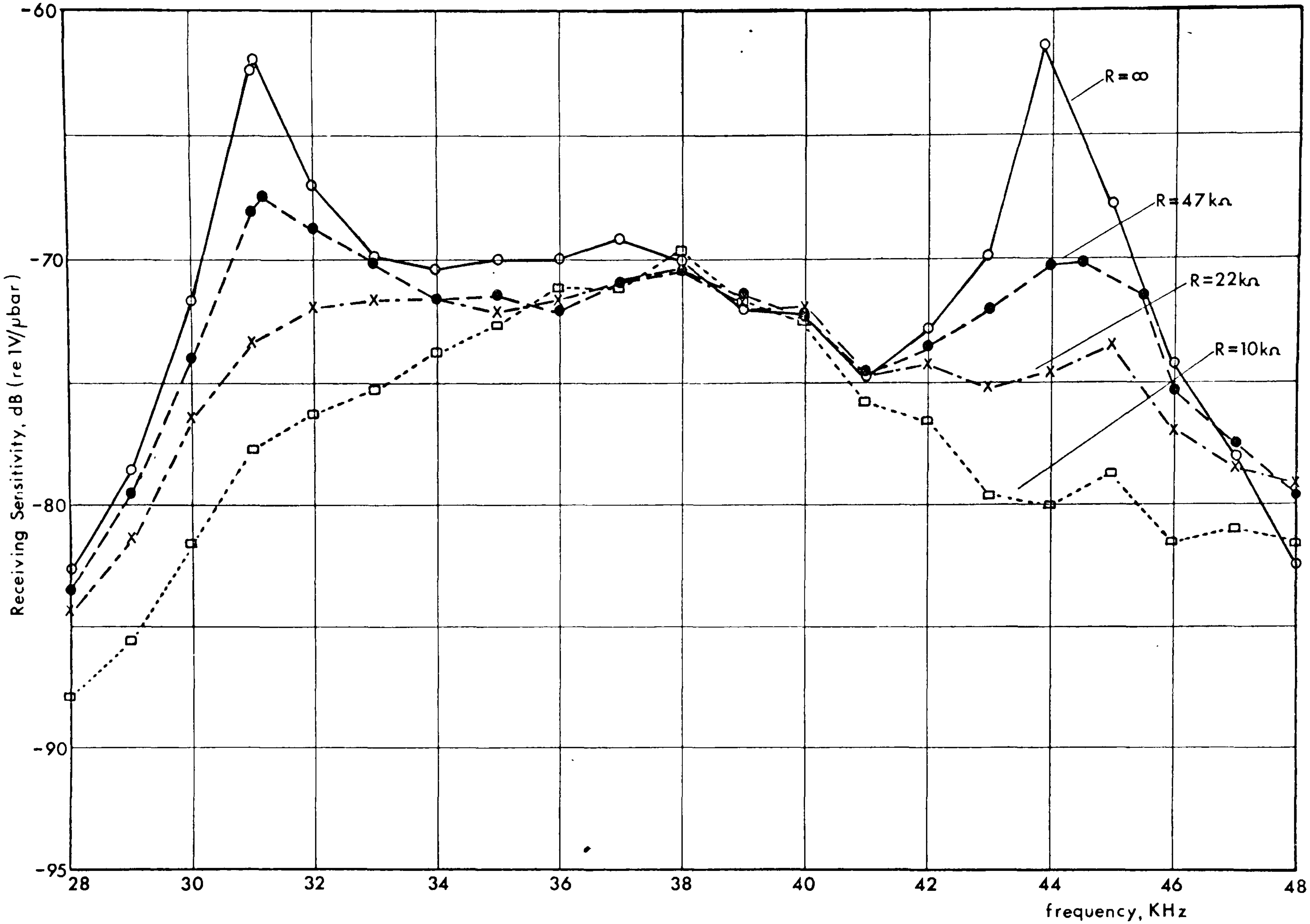
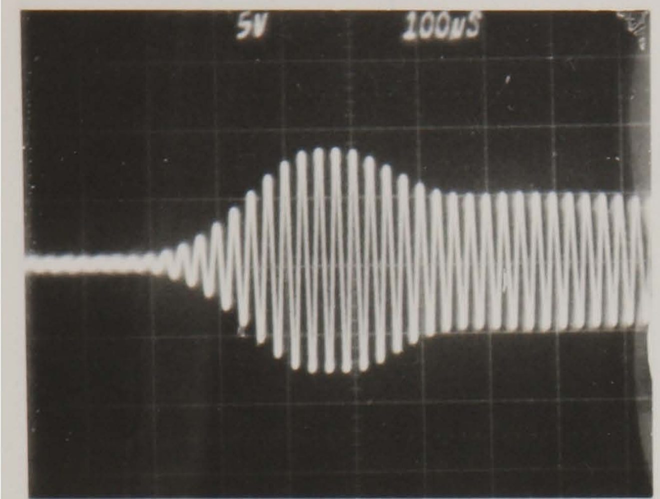
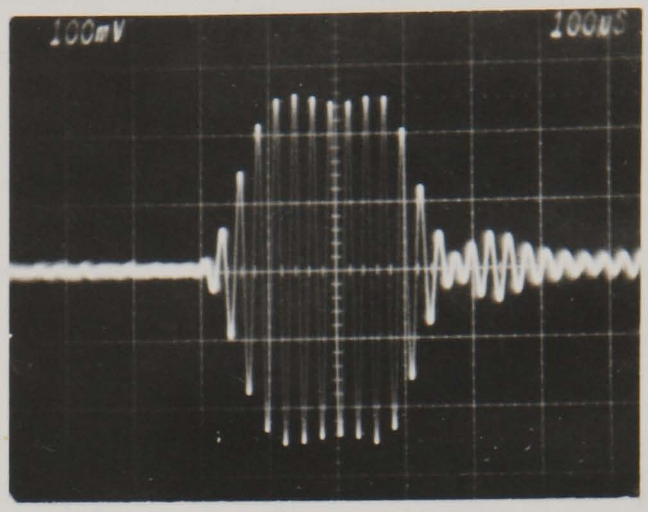


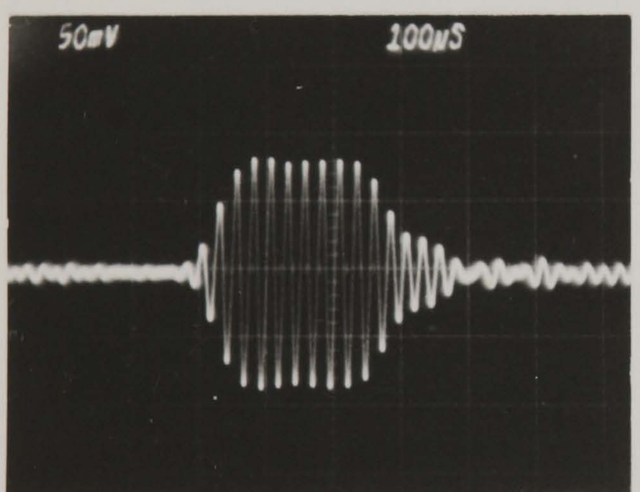
Figure 4.8 Effect of shunt resistance on the receiving response of the transducer with a shunt tuning coil



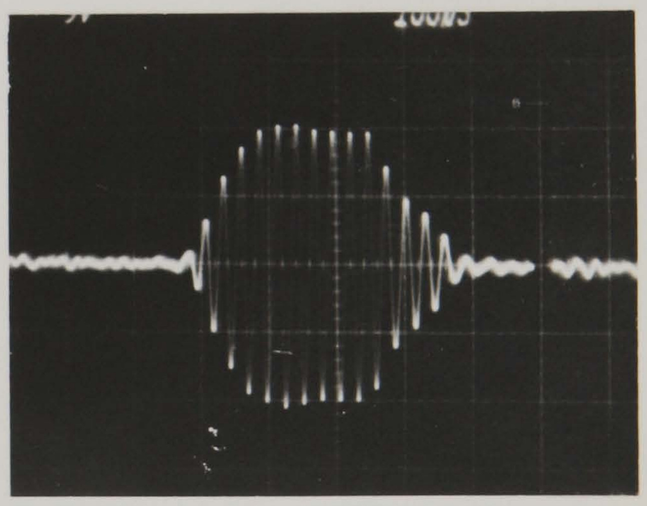
(a) Unmodified transducer
(receiving)



(b) Unmodified transducer
with inductive tuning and
resistive damping (receiving)



(c) Modified transducer
with inductive tuning
and resistive damping
(receiving)



(d) Unmodified transducer
with inductive tuning
and resistive damping
(transmitting)

Figure 4.9 Comparison of Transducer Pulse Responses

(100 μs/div sweep rate)

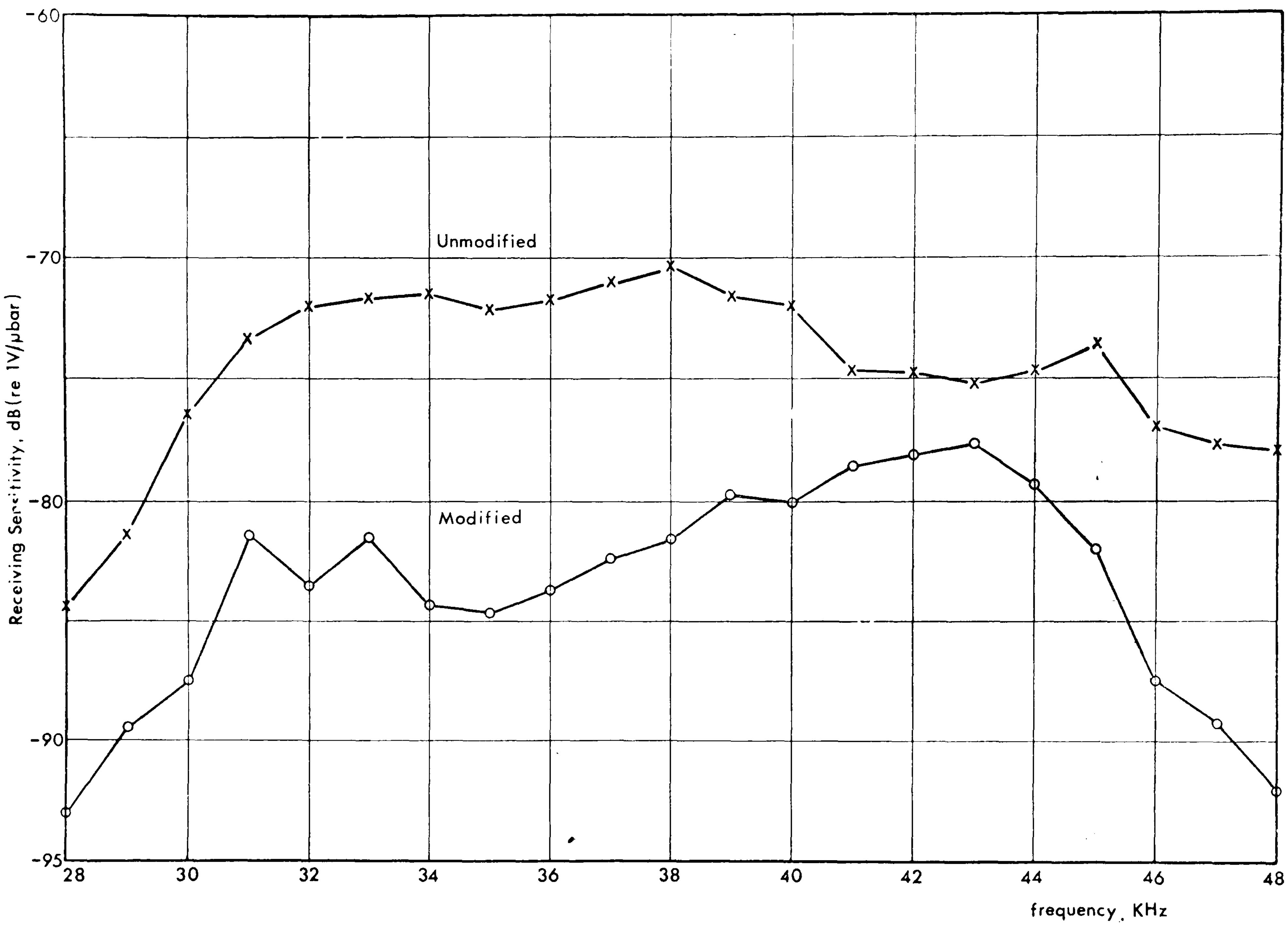


Figure 4.10 Comparison of receiving responses of a modified and unmodified transducer with inductive tuning and optimal resistive damping

found that by selecting transducers for similar values of series resonant frequency, responses could be matched to within 2 dB in the pass band. In the receiver system design the negative feedback in the first stage of the pre-amplifier is arranged so that the input impedance provides the appropriate damping for the receiving transducer. This arrangement enables the optimum signal-to-noise ratio to be obtained (see Appendix 1).

It was necessary to confirm that the transducer in the transmitting mode would yield similar results to those obtained in the receiving mode. The transfer function in the undamped condition was obtained by connecting the appropriate value of inductance in parallel with the transducer and driving the latter at 10 volts rms through a high valued series resistance. The output was measured by placing a capacitor microphone at a distance of 1 metre. Again, the expected double peaked curve was obtained which, when damped with a 30 kΩ series resistance, gave a similar transfer characteristic to that obtained with a receiver. Fig. 4.11 is a typical curve which shows that the bandwidth is marginally narrower than that of an unmodified receiver. When the transmitter was driven with pulsed sine waves it exhibited output pulses with a rise and fall time of 100 μsec. Fig. 4.9(d) shows that the rise time is very slightly longer than that obtained with a receiver.

In order to examine the effects of the modifications in a complete system, a preamplifier was designed with an input impedance of 22 kΩ, thus providing the optimum damping conditions for a modified receiving transducer plus parallel inductor. The receiving system was subjected to acoustic pulses produced by the transmitting system described above. The output pulses from the preamplifier were observed to build up and decay in 100 μsec. and looked identical to those shown in Fig. 4.9(d). The modifications to the transmitter and receiver, therefore, give a system of the required bandwidth which is thus capable of producing sharp pulses.

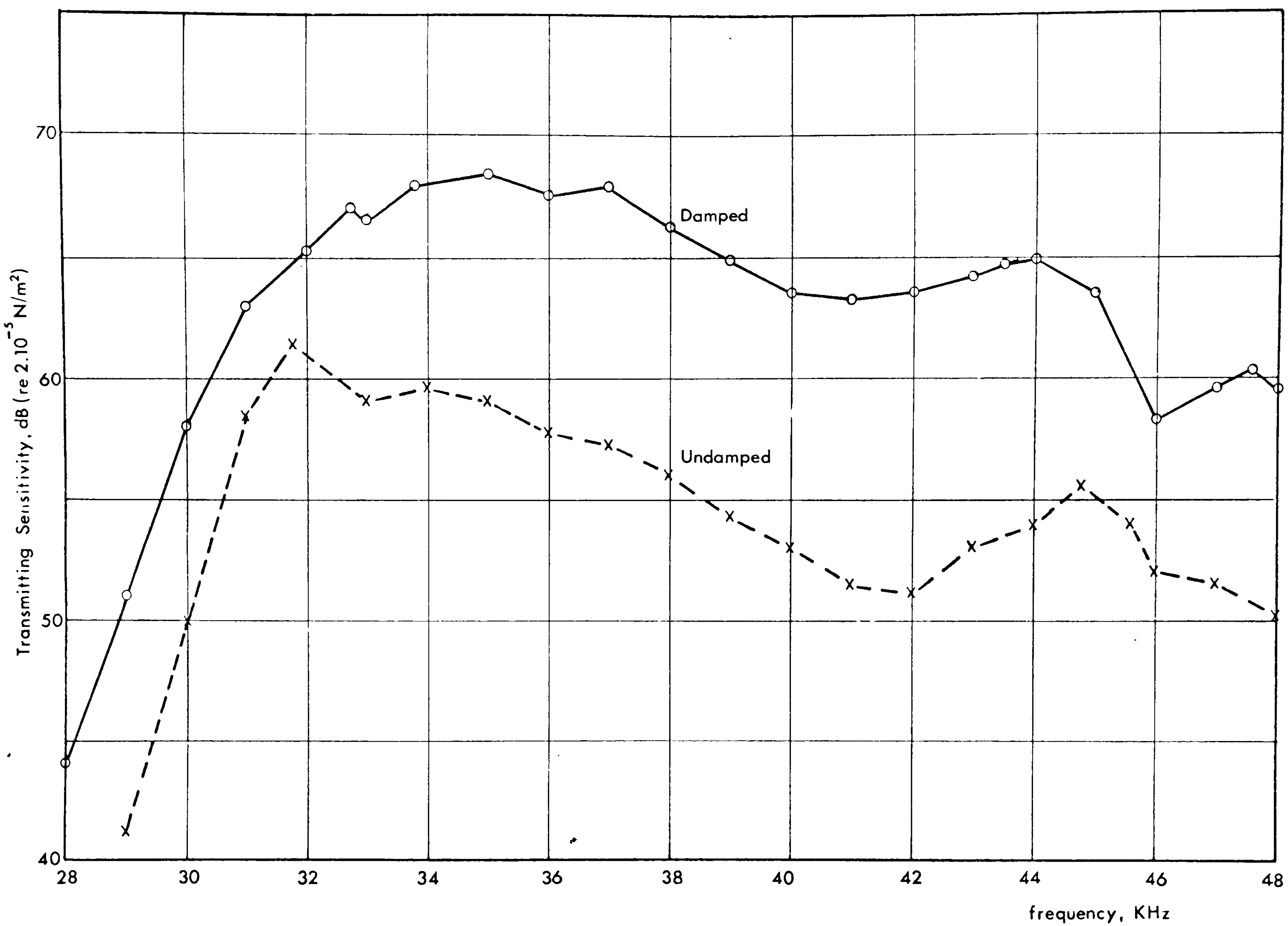


Figure 4.11 Transmitting response of the transducer with inductive tuning showing the undamped and optimally damped conditions

4.5 Power Limits of the Transmitting Transducer

To obtain sufficient output from the transmitter it is necessary to drive it at a level which approaches the power producing limits of the active material.

The useful acoustical output power of radiating piezoelectric ceramic transducers is limited by two main factors, one mechanical and one electrical. The mechanical factor is the maximum permissible alternating stress in the surface of the ceramic material before the dynamic strength is exceeded. The electrical factor is the maximum permissible alternating field in the piezoelectric layers before depolarisation occurs.[26] An electric field sufficient to cause depolarisation produces extremely high dielectric losses and therefore very low efficiency, so it is generally unnecessary to directly consider this factor. A further limitation is depolarisation due to temperature rise but, as the transducers of concern transmit signals having a low duty cycle, a limitation of this nature is unlikely.

There is an optimum value of mechanical quality factor (Q_M) of the ceramic element at which the limitations due to electrical losses and mechanical strength are equal. Figures available for pulsed sonar transducers manufactured from lead titanate zirconate, from which the ceramic element is made, indicate that this optimum value is between $Q_M = 2$ and $Q_M = 3$.[27] For $Q_M < Q_{optimum}$ the limiting factor is due to dielectric losses and for $Q_M > Q_{optimum}$ the limiting factor is the dynamic strength. In our case the transducer has a mechanical quality factor of about $Q_M = 60$ and so it is likely that the dynamic strength is the dominant limiting factor.

To test the power limits practically, a transmitter circuit with a pulsed sine wave output was constructed to be capable of driving a transducer at up to 400 volts peak to peak. Using this circuit a transducer, with its associated inductor, was driven at its mechanical resonant frequency, the output being monitored by a capacitor microphone. The results are shown graphically in Fig. 4.12. As the voltage across the transducer was increased from zero, the acoustic output rose linearly until, at 175 volts peak to peak, it dropped and showed little sign of increasing as the voltage was further raised. As the voltage was lowered the output decreased further and showed no tendency to revert to its original high level. When the test was repeated with the same transducer only the low output curve was followed. It appeared that the transducing element had sustained permanent damage, most probably due to the ceramic material being overstressed. Subsequent tests with new transducers indicated that they could be operated at 100 volts peak to peak without incurring permanent damage, although one disadvantage is that a faint audible click accompanies each ultrasonic pulse.

It was not possible to use the drive circuitry described above in an experimental aid due to its bulkiness and excessive power consumption. A smaller version was constructed using a push-pull arrangement which is capable of driving the transducer at 50 volts peak to peak. This system enables the transducer to produce a sound pressure level in each pulse of 101 dB (re: 2×10^{-5} N/m^2) at a distance of 1 metre, with a rise and decay time of 100 μsec.

It must be borne in mind, however, that the acoustic output power can at least be quadrupled and the realisation of compact electronic circuitry of low power consumption which can produce the required drive merits further investigation.

4.6 Earphones

The main requirement of the earphones is that they should provide the listener with a faithful acoustic reproduction of the electrical output of the mobility device. This must be achieved without impairing

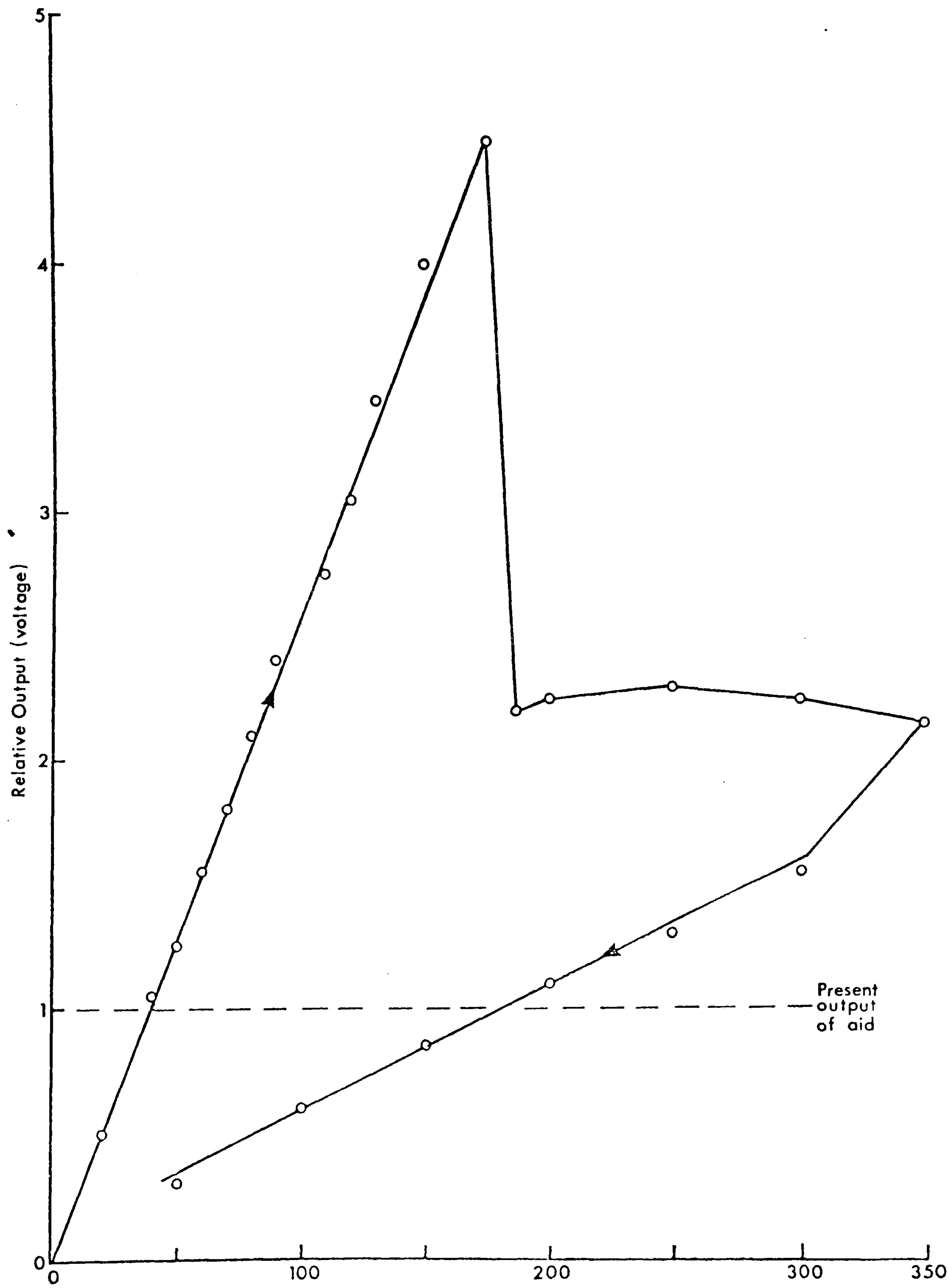


Figure 4.12 Transducer output as a function of applied voltage. (Measured as a voltage output from a capacitor microphone)

normal hearing ability, as the blind traveller derives many useful auditory cues from the environment in addition to those provided by the aid. It is intended, therefore, to use miniature transducers which are remotely mounted from the ear, with short pieces of plastic tubing leading the sound to the entrance of the ear canal or meatus. The meatus is thus left sufficiently unobstructed to allow the perception of natural sounds. This technique is similar to the one utilised in the "CROS" system of hearing aids.[28]

By referring to the previous sections we note that the ultrasonic transducers fulfil the bandwidth specification laid down in the introduction. Consequently, in order to preserve a good transient response, the bandwidth requirement for the earphone system is 8 kHz. A further requirement is a smooth response curve as the elimination of peaks minimises ringing or overshoots of the transient acoustic signal. A detailed study of commercially available earphones and earphone/coupling tube arrangements was carried out by Harris.[29]

The output of earphones which are normally coupled directly into the auditory meatus by means of a moulded insert is usually measured using an artificial ear which simulates the acoustic loading of a typical real ear. In our case the earphone is much more loosely coupled into the meatus, which considerably modifies the acoustic performance. The main effect is to progressively attenuate the lower frequencies owing to the difficulty in sustaining the condensation or rarefaction of the air in the coupler. In order to carry out frequency response tests the loose coupling was simulated by spacing the earphone under investigation, with or without its coupling tube, 1 centimetre from a capacitor microphone in an anechoic enclosure. Response curves were plotted using a Bruel and Kjaer trace plotting system. Fig. 4.13 shows a selection of the response curves obtained without coupling tubes; each is offset

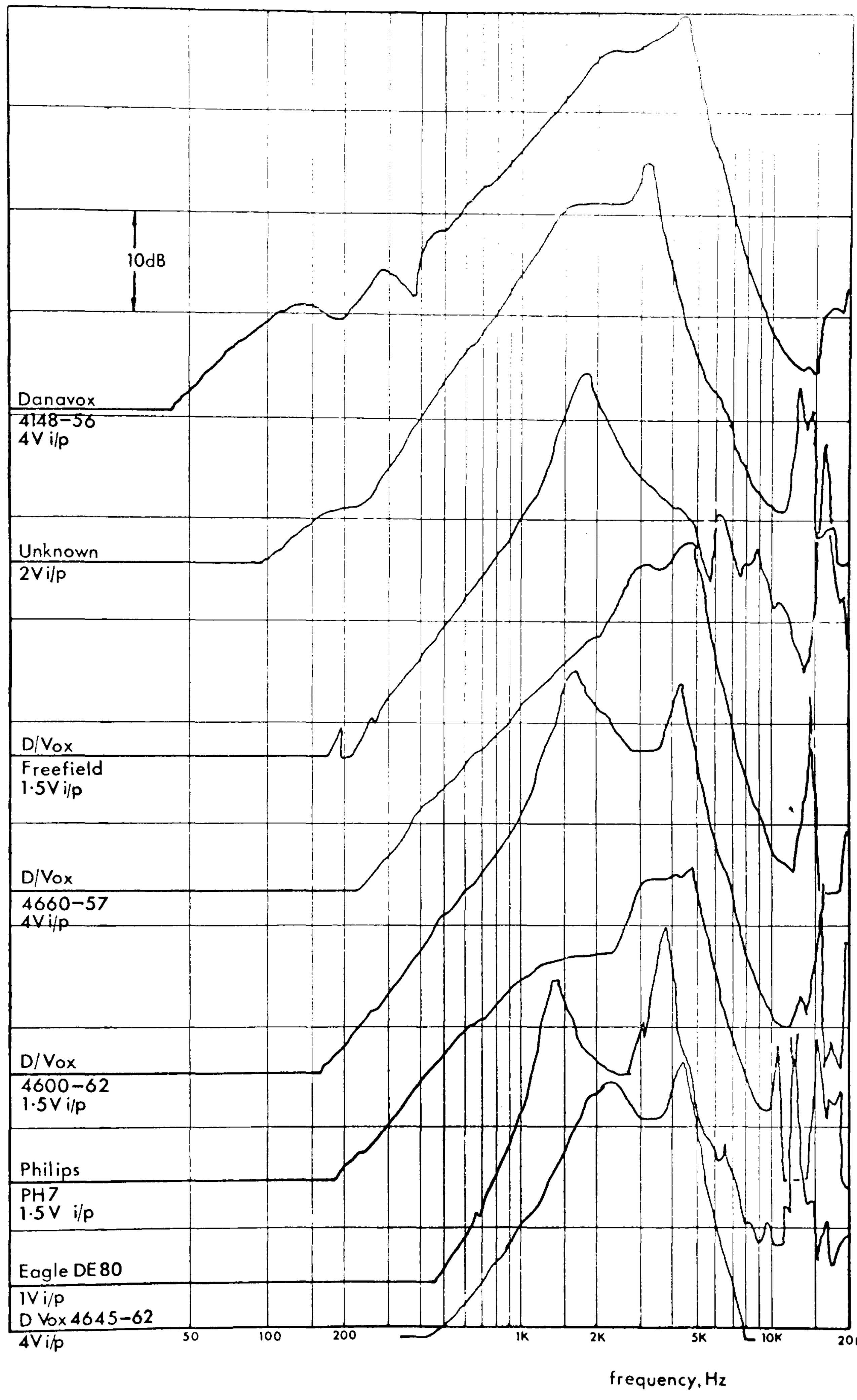


Figure 4.13 Frequency responses of various earpieces

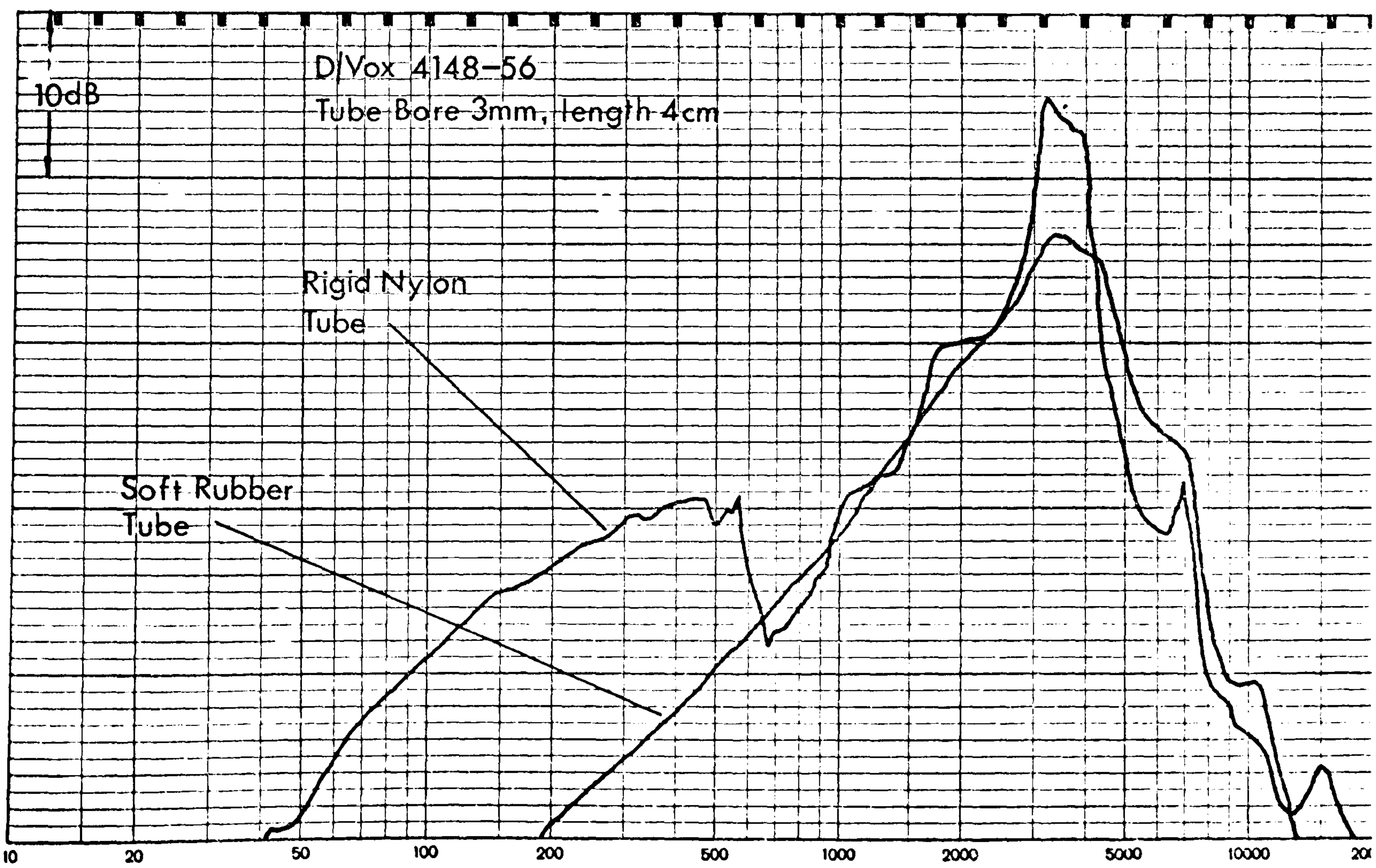
vertically to enable a direct comparison to be made. Most responses exhibit a strong peak between 3 kHz and 5 kHz and some have an additional peak at around 1.5 kHz. Several show a distinctly better low or high frequency response than others.

The most promising earphones were selected for further tests with coupling tubes. Frequency responses were plotted using various tube lengths, tube diameters and tube materials. Fig. 4.14(a) shows the effect on the response to changes in tube length, the diameter remaining constant. As the tube length is increased the low frequency response deteriorates and additional peaks arise due to resonances of the tube itself. Rigid plastic tubes were found to give a peaky curve while those using softer materials dampen the peaks giving a smoother response although there is some attenuation of the signal level; typical examples of both are given in Fig. 4.14(b). This diagram also shows that the response with the softer material closely resembles the "no-tube" condition except for some increased loss at low frequencies. Minor changes in tube diameter did not cause large changes. When the diameter was reduced from 3 mm to 1.5mm, however, the output was greatly attenuated due to the increased acoustic resistance of the smaller tubing.

Several other tests were made on the selected earphones to ensure an adequate performance for inclusion in the device. Fig. 4.15 indicates that either constant voltage or constant current drive is satisfactory whilst Fig. 4.16 shows the impedance variation as a function of frequency. This rises above the nominal value at high frequencies owing to the inductive nature of the earphone.

As the aid is a binaural device, accurate matching of earphone sensitivities is essential to prevent the occurrence of interaural amplitude distortion. Several of the most promising earphones were tested and the results showed that characteristics of each type could

(a) Effect of tube length on response



(b) Typical effect of tube material on response

Figure 4.14

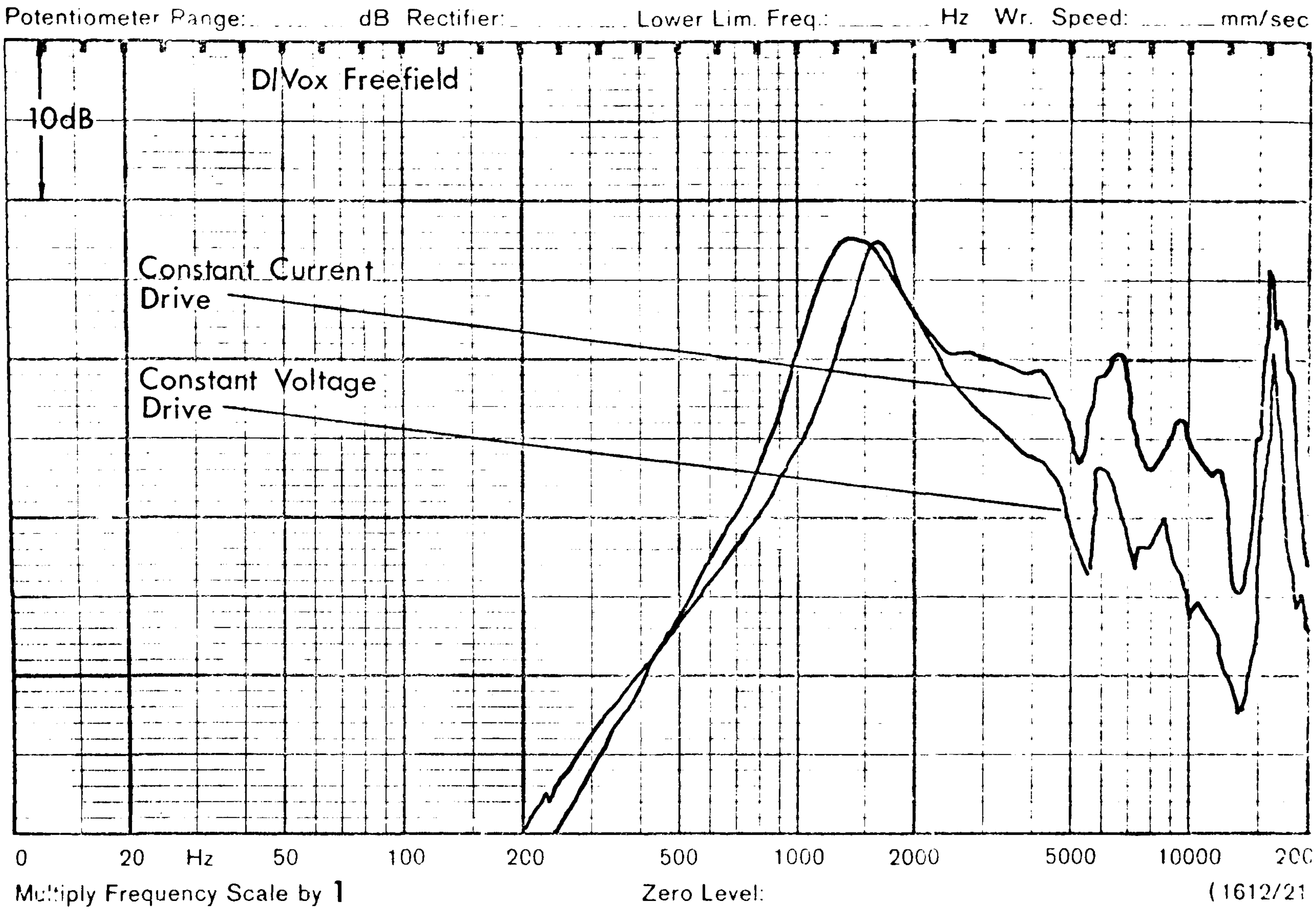


Figure 4.15 Comparison of responses with constant current and constant voltage drive

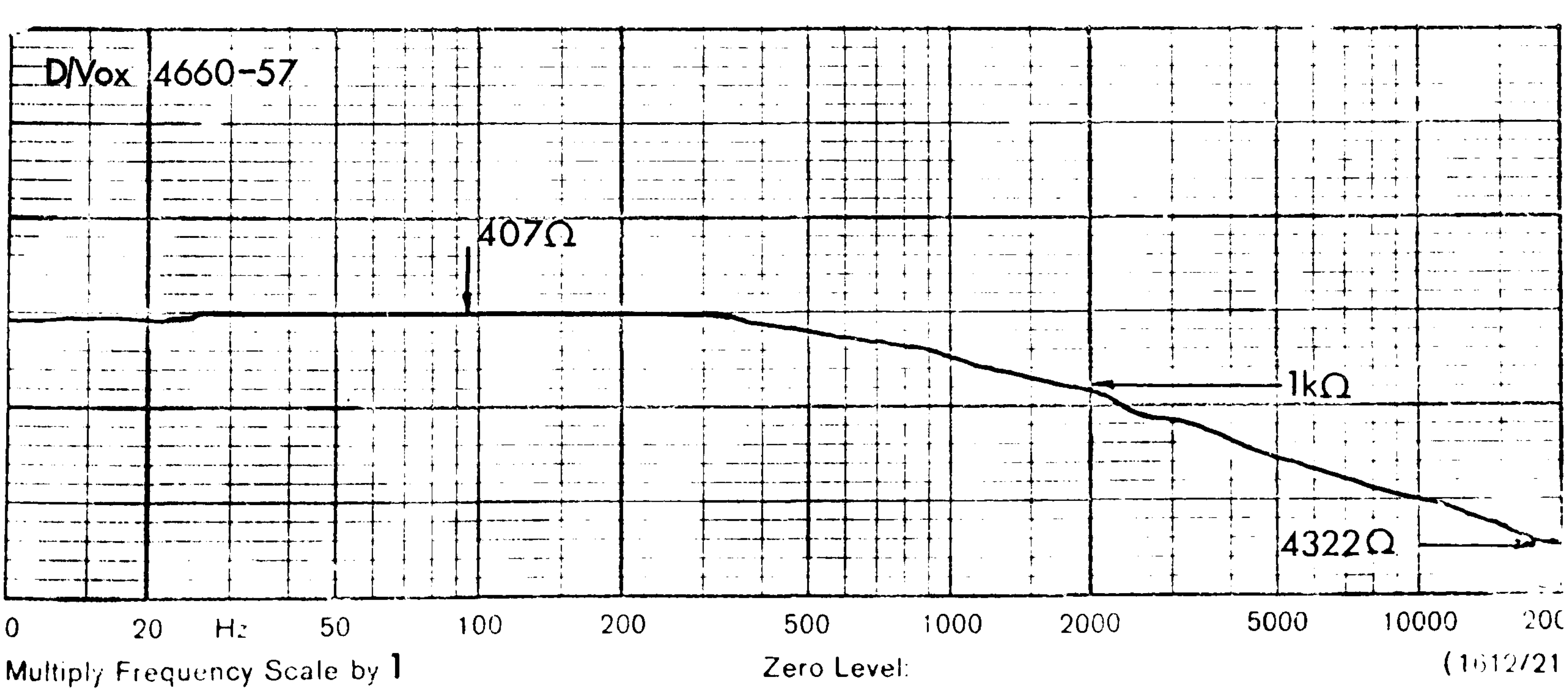


Figure 4.16 Impedance characteristic of a typical earphone

be readily matched to within 1 dB SPL.

The two earphone systems finally selected were of different types. One is a small version of the type commonly fitted to hearing aids. Its response curve using a soft rubber coupling tube of length 4 cm and diameter 3 mm is shown in Fig. 4.14(b). This gave the best bandwidth whilst providing adequate damping of the peaks. The other type is called a "freefield" earphone, which is designed to be mounted in an earshell, hanging on the pinna; this spaces the transducer 1 centimetre from the entrance to the ear canal. This type is unsuitable for use with tubes as there are three output apertures. The response curve is shown in Fig. 4.13.

Both types were tested subjectively by driving them with pulses derived from the output of an experimental aid using the improved, wider band, ultrasonic transducers. With the earphone attached to a coupling tube, the exact position of the end of the tube in the auditory meatus was not finalised, the position giving maximum signal level and minimum frequency distortion being best found by trial and error. Several listeners were invited to compare the outputs of the two new earphones with each other, as well as with those that had been in use previously. All stated that the clicks produced by both the new earphones were very similar and in all cases much sharper than those of earlier types. The type for use with a coupling tube, Danavox 4148-56, has the advantage of smaller size which will enable it to be neatly mounted on a pair of spectacle frames. The "freefield" type, however, Danavox 4301-56, has much greater flexibility in experimental work and was adopted for present use.

4.7 Conclusions

In an attempt to simulate the signals often used in human echo perception it was found that a bandwidth of at least 16 kHz was required for the ultrasonic transducers. The piezoelectric transducers, when operated in their normal mode, could only achieve a maximum bandwidth of 3 kHz. A theoretical analysis of the equivalent circuit of the transducer indicated that the connection of an appropriate value of inductance and damping resistance across the transducer should extend the bandwidth up to about 18 kHz. The results of practical investigations were in close agreement with the theoretical predictions, a bandwidth of about 16 kHz being obtained with the transducers in both the transmitting and receiving modes. When a transmitting and receiving system was set up for pulse operation, it was found that the output pulses from the receiver amplifier built up and decayed in 100 μsec. This is considered satisfactory for use in the aid.

The extension of the bandwidth of a receiving element does not reduce its sensitivity but modification of the casing causes a loss of 8 dB. It is thought that this loss can be compensated by an increase in transmitter power. An investigation of the power limits of transmitting transducers using pulsed operation revealed that, as the voltage across the element was increased, the acoustic output rose linearly until, at a voltage of 175 volts peak to peak across the element, a breakdown in the output occurred. The limiting factor here is considered to be the dynamic strength of the ceramic material. As the transmitting transducers are at present driven at a level substantially below this limit, a considerable increase in acoustic output could be gained by suitably redesigning the transmitting circuitry to increase the drive level.

An evaluation of commercially available earphones yielded two models, both of which seemed superior to the rest in the attempt to reproduce signals up to the required 8 kHz. Each type avoids obstructing the auditory meatus by a different method; one type utilises a coupling tube whilst the other is arranged to be mounted close to the entrance of the ear canal. The latter type was found to be more convenient for experimental use.

An experimental device was constructed which embodied all the modifications to improve the bandwidth of the system. Subjective tests indicated that the acoustic clicks emanating from the earphones sounded sharp and that the overall sensitivity of the system was adequate. Other faults which were due to the narrow bandwidth in the original aid, such as the direct pick up from the transmitter, had been partially cured as a result of the modifications.

CHAPTER 5

DIRECTIONAL RESPONSES OF THE TRANSMITTING AND RECEIVING TRANSDUCERS

5.1 Transmitting Transducers

5.1.1 Introduction

As the mobility device is intended to give localisation information in the horizontal plane in front of the user, the transmitter is required to irradiate a large angle in this plane. In addition, it is useful to radiate some energy sideways as this produces strong reflections from adjacent walls and fences, enabling a user to follow these more easily; this technique is known as shorelining. In the vertical plane a beamwidth of about 60 degrees should prove sufficient. For a hand held transmitter this would give protection for the whole body, and for a head mounted version protection would be given from the waist upwards.

5.1.2 Practical Investigations

In order to produce the required beam pattern it was decided to use three piezoelectric bimorph transducers. It was thought that one pointing directly ahead would give a sufficiently wide beam in the forward direction, whilst the other two pointing sideways would provide the radiation towards the shoreline. An investigation was carried out into the directional characteristics of this arrangement by driving each of the three transducers with the same 40 kHz signal and recording the resulting polar diagrams. The measuring equipment is shown in Fig. 5.1.

It was found best to remove the cases from the transducers and cluster them together as closely as possible, since in this configuration the least destructive interference occurred in the main beam. This meant that the transducers were mounted in a pyramidal-type structure with the two sideways pointing transducers in one plane and the forward pointing one on top. An example of this arrangement is shown in Fig. 5.2. When the angle between the side transducer axes was

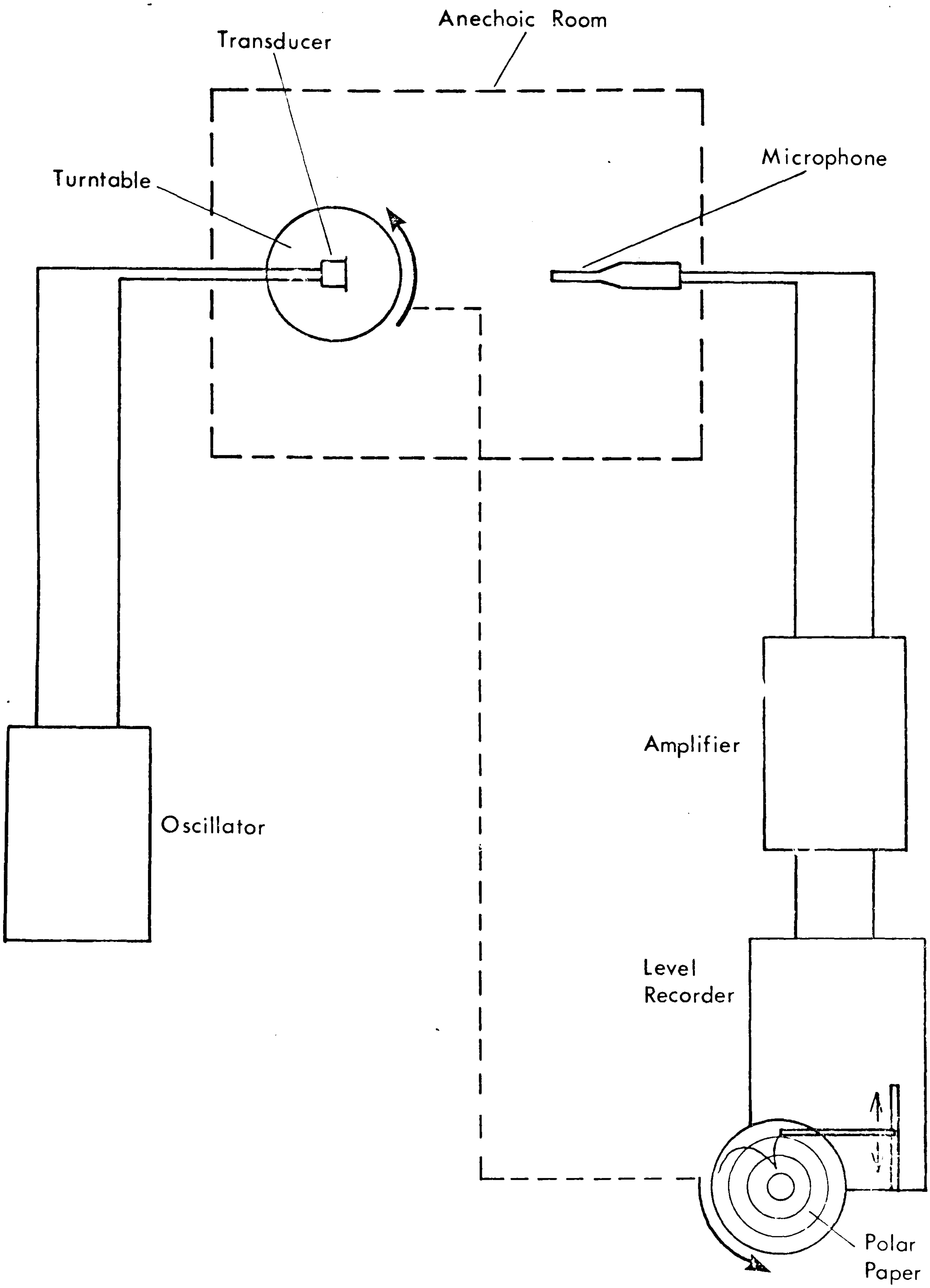


Figure 5.1 Measuring arrangement for recording directional characteristics of transducers

Figure 5.2 Arrangement of three transducers in transmitter

increased to 120 degrees or more, the polar diagram began to resemble the required response. A typical plot with a splay angle between the side transducers of 150 degrees is shown in Fig. 5.3. From this it can be seen that the level of radiation in certain directions is several dB 's greater than the level on the axis itself, a feature which is not compatible with the original requirements. In an attempt to overcome this effect the side transducers were driven at a reduced level. This was achieved by using a transformer with a tapped secondary winding. The voltage level finally selected to drive the side transducers was one third of the voltage across the forward facing transducer. The resulting polar diagrams with this arrangement were promising: a plot with the splay angles between the side transducers set at 150 degrees is shown in Fig. 5.4.

This transducer configuration was incorporated into an experimental version of the mobility aid and some elementary outdoor tests were carried out. These showed that echoes received from adjacent walls and fences were strong, thus making the task of shorelining relatively easy. It was noted, however, that echoes from objects at a certain angle on either side of the transmitter axis became very weak and sometimes disappeared altogether. It was found that this angle corresponded to the angle at which minima occurred in the polar diagram. It was clearly necessary to eliminate these minima and at this point it was thought worthwhile to reconsider the requirements for the beam pattern. Rather than having a strong forward beam with additional energy radiated sideways it might be easier to generate a very wide beam in the horizontal plane with a constant response over the whole of this sector.

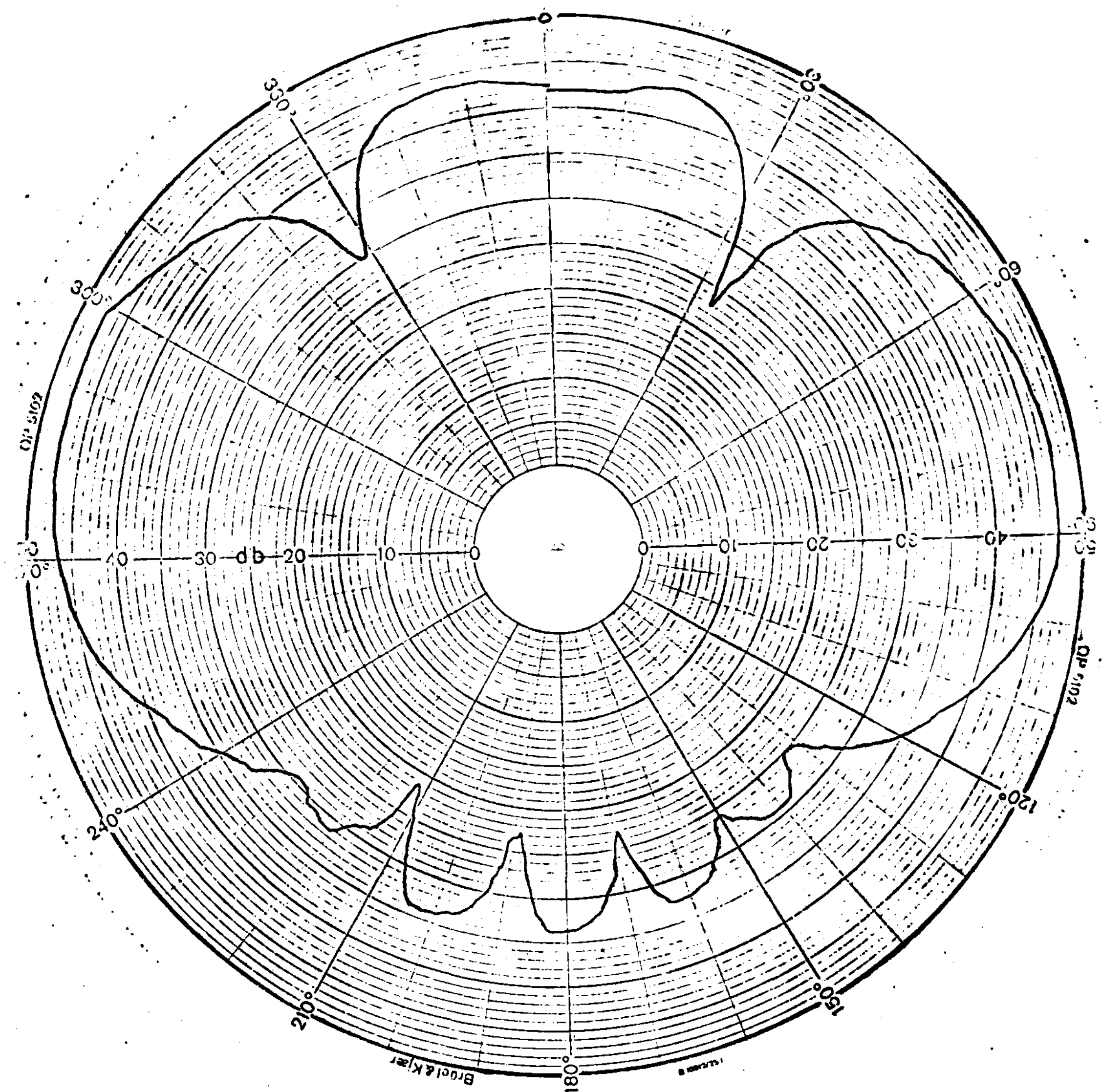


Figure 5.3 Directional response of transmitter using three transducers, all driven at the same level

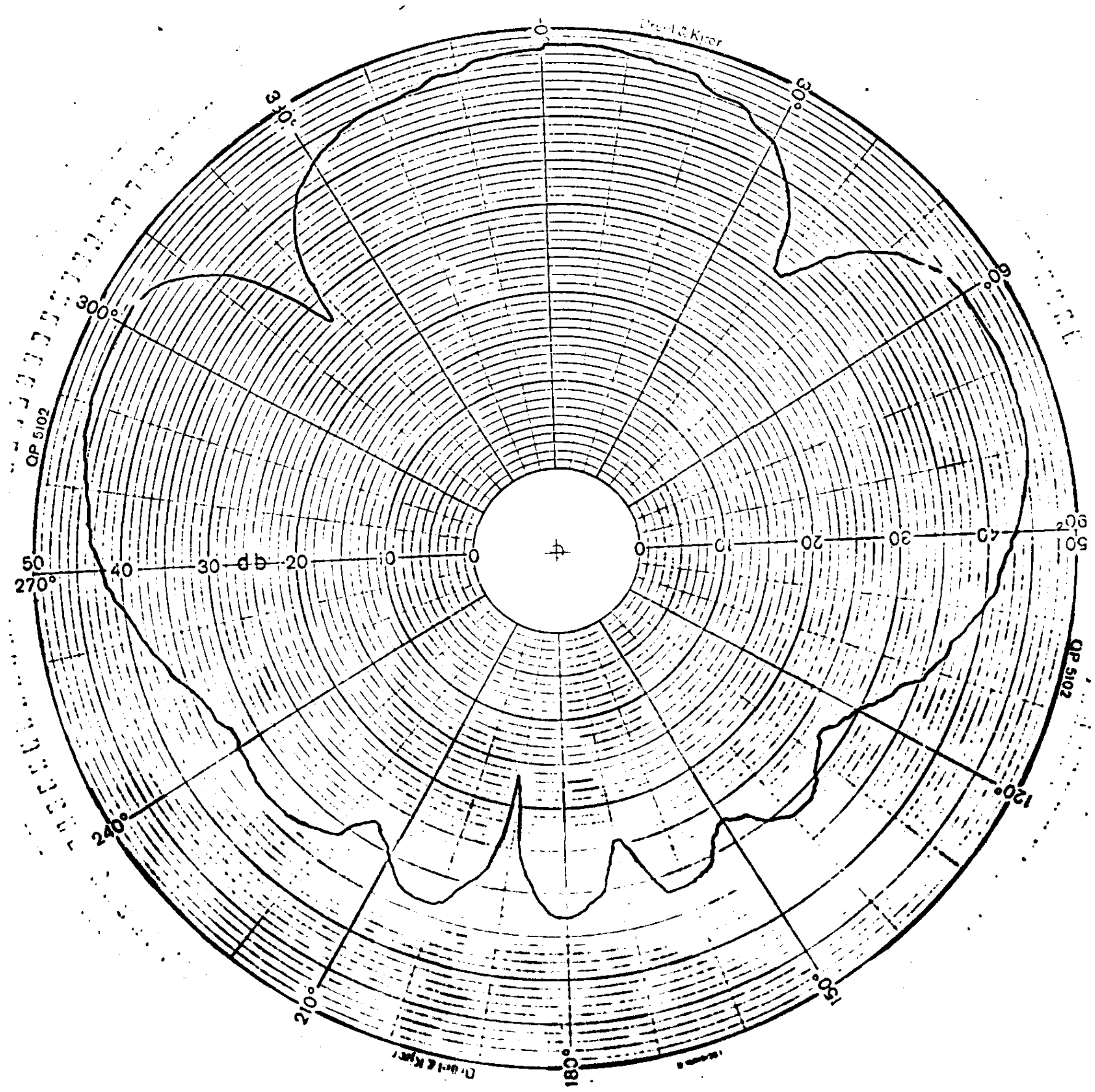


Figure 5.4 Directional response of transmitter using three transducers with side transducers driven at reduced level

5.1.3 Theoretical Considerations

When considering the design of a wide beam pattern we have the general case that a radially vibrating sphere radiates sound uniformly outwards in all directions. A corollary of this is that a portion of a spherical surface, large compared to the wavelength and vibrating radially, emits uniform sound radiation over a solid angle subtended by the surface at the centre of curvature.[30] To obtain uniform sound distribution over a certain solid angle, the radial air motion must have the same phase and amplitude over the spherical surface intercepted by the angle having its centre of curvature at the vertex, and the dimensions of the surface must be large compared with the wavelength. When these conditions are satisfied for all frequencies the response characteristic is independent of the position within the solid angle.

One approach to the transmitter design which would satisfy the abovementioned conditions would be to have a piezoelectric element in the shape of a portion of a spherical surface which subtends an angle of 180 degrees in the horizontal plane and 60 degrees in the vertical plane, the dimensions of the element being large compared with the wavelength. Ideal though this design may be it is unsatisfactory due to the difficulty in obtaining a cheap element to this specification.

An alternative approach is to approximate the spherical surface using individual circular transducers arranged in the arc of a circle. We can consider the arc to be broken up into a number of equal chords, each chord representing one transducer. If we further consider the intensity of radiation over each chord to be uniform and the phase of all the chords to be the same, then the directional characteristics in the plane of the arc have been shown[31] to be given by

$$R_\alpha = \frac{1}{2m+1} \left| \sum_{k=-m}^{k=m} \cos\left\{\frac{2\pi R}{\lambda} \cos(\alpha + k\theta)\right\} \frac{\sin\left[\frac{\pi d}{\lambda} \sin(\alpha + k\theta)\right]}{\frac{\pi d}{\lambda} \sin(\alpha + k\theta)} \right.$$

$$\left. + j \sum_{k=-m}^{k=m} \sin\left\{\frac{2\pi R}{\lambda} \cos(\alpha + k\theta)\right\} \frac{\sin\left[\frac{\pi d}{\lambda} \sin(\alpha + k\theta)\right]}{\frac{\pi d}{\lambda} \sin(\alpha + k\theta)} \right|$$

where R_α = ratio of the pressure for an angle α to the pressure for an angle $\alpha = 0$,

λ = wavelength

k = variable,

R = radius of the arc,

2m+1 = number of chords,

θ = angle subtended by any of the chords at the centre of the circumscribing circle, and

d = length of one of the chords

In the practical situation it is not convenient to have a cluster of more than three transducers, so that the number of chords, 2m+1, is fixed; the wavelength, λ, remains constant and the length of chord, d, is fixed by the diameter of a transducer. We are in a position, therefore, to evaluate directional characteristics for different values of θ. It must be borne in mind, however, that as it is only possible to use three transducers, the approximation to an arc of a circle is poor. For a good approximation the greatest distance from the chord to the corresponding point on the arc should be small compared to the wavelength of sound being radiated. Whereas a ratio of at least 1 : 20 is desirable, in our case the ratio is only 1 : 5. Further, the assumption that the intensity over each chord is uniform might not be accurate for the transducers in question. The use of the formula, however, to obtain a result which can be compared with a directional characteristic measured practically should indicate the viability of this approach.

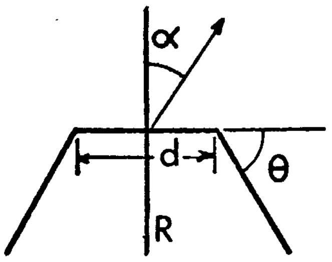
The directional characteristics evaluated for various values of θ are shown in Fig. 5.5. It can be seen that the calculated characteristic for θ = 75 degrees resembles the measured characteristic in Fig. 5.3, both these plots being for the same configuration. This suggests that use of the formula might give a good indication as to the angular orientation of the three transducers to give the required beam pattern. If this is the case, the characteristic calculated at θ = 60 degrees would seem to give the most desirable result.

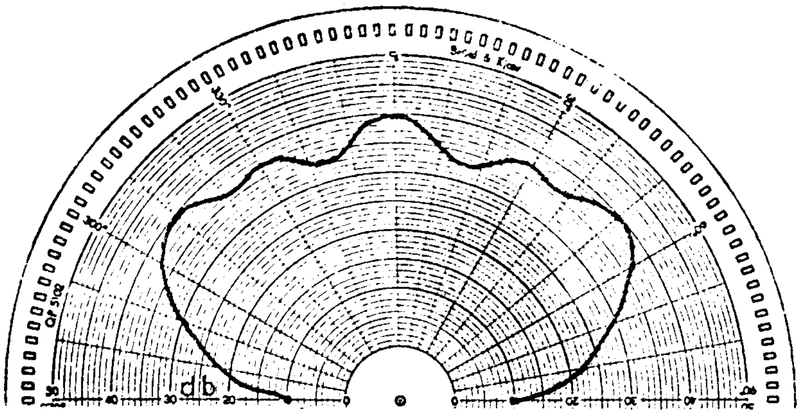
5.1.4 Further Measurements

Measurements were carried out with the three transducers arranged in the orientation where θ = 60 degrees, after a visual inspection was made to ensure that the front edges of the transducers formed the chords of a circle. The resulting polar diagram, shown in Fig. 5.6, reveals a very smooth response over a wide angle and shows a marked similarity to its calculated counterpart. In the actual response, as in the calculated one, there is a slight increase in sensitivity over a range of angles off the main axis. In the vertical plane the response resembles that of a single transducer, i.e. with a 6 dB beamwidth of 60 degrees.

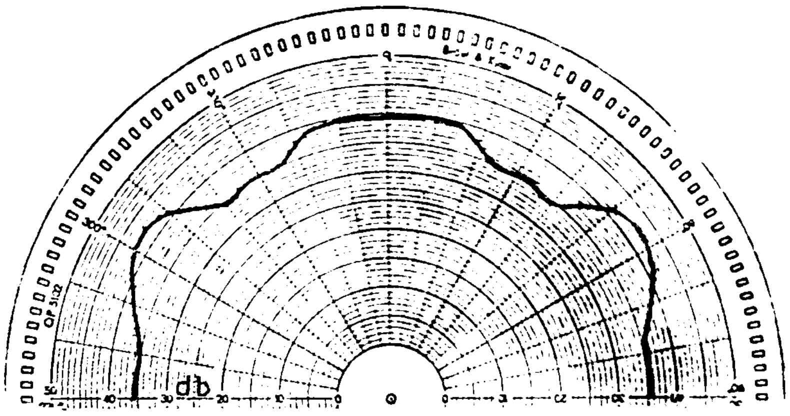
This transducer arrangement, then, replaced the previous one in the experimental aid and further simple mobility tests were carried out. It was found that the task of shorelining could be performed very easily, and there did not appear to be any directions off the axis of the main beam at which echoes became weak. A further discussion of the use of this beam pattern in a device is found in the next section.

Another effect which was found in all the tests using three transducers was that the small amount of energy radiated in a backwards direction produced obtrusive echoes from the body of the user. Experiments showed that these could be reduced by mounting a metal disc

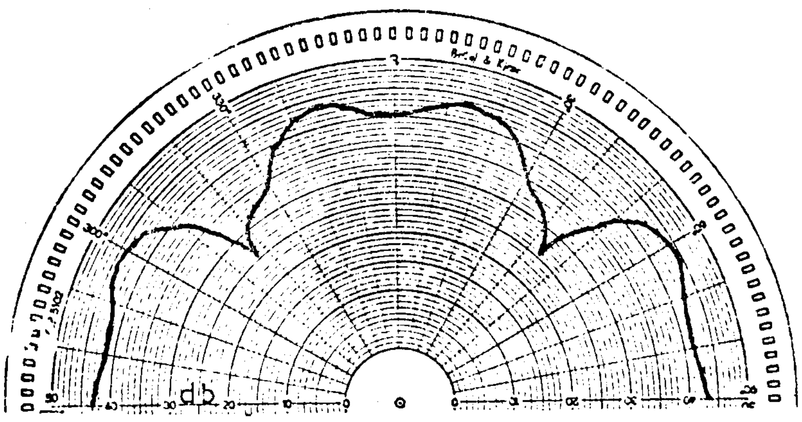




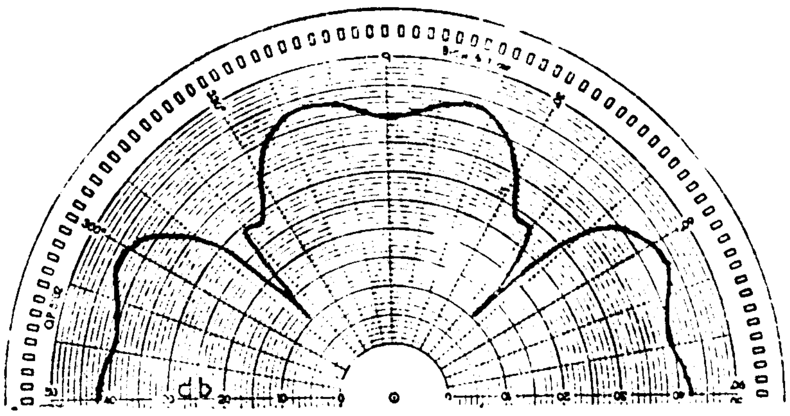
$\theta = 50^{o}$



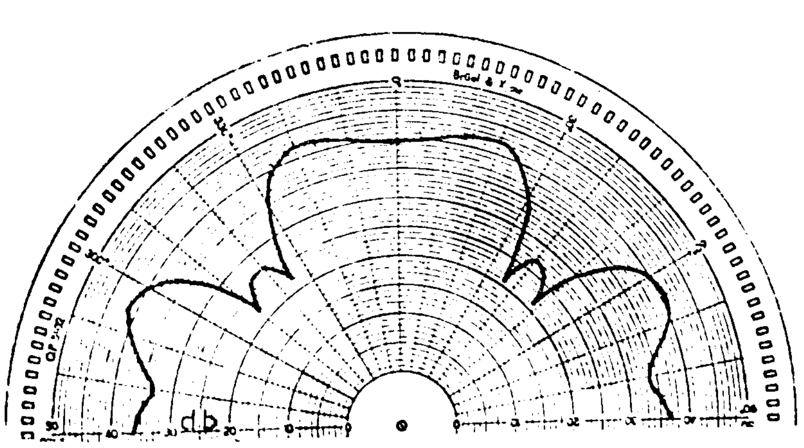
$\theta = 60^{o}$



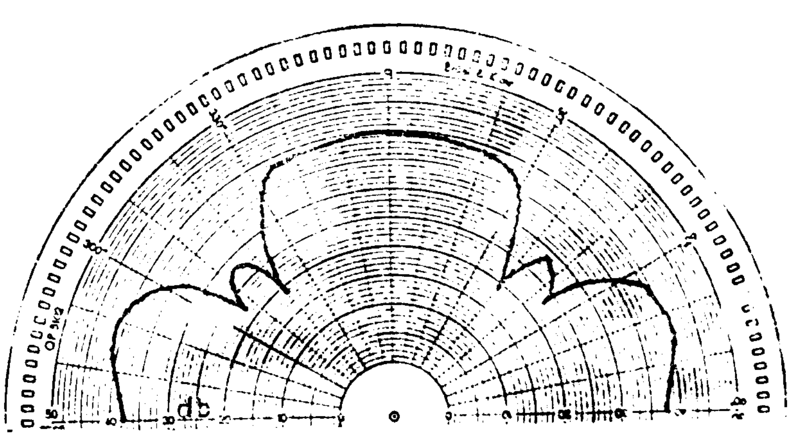
$\theta = 70^{o}$



$\theta = 75^{o}$



$\theta = 80^{o}$



$\theta = 90^{o}$

Figure 5.5 Theoretical evaluation of directional characteristics

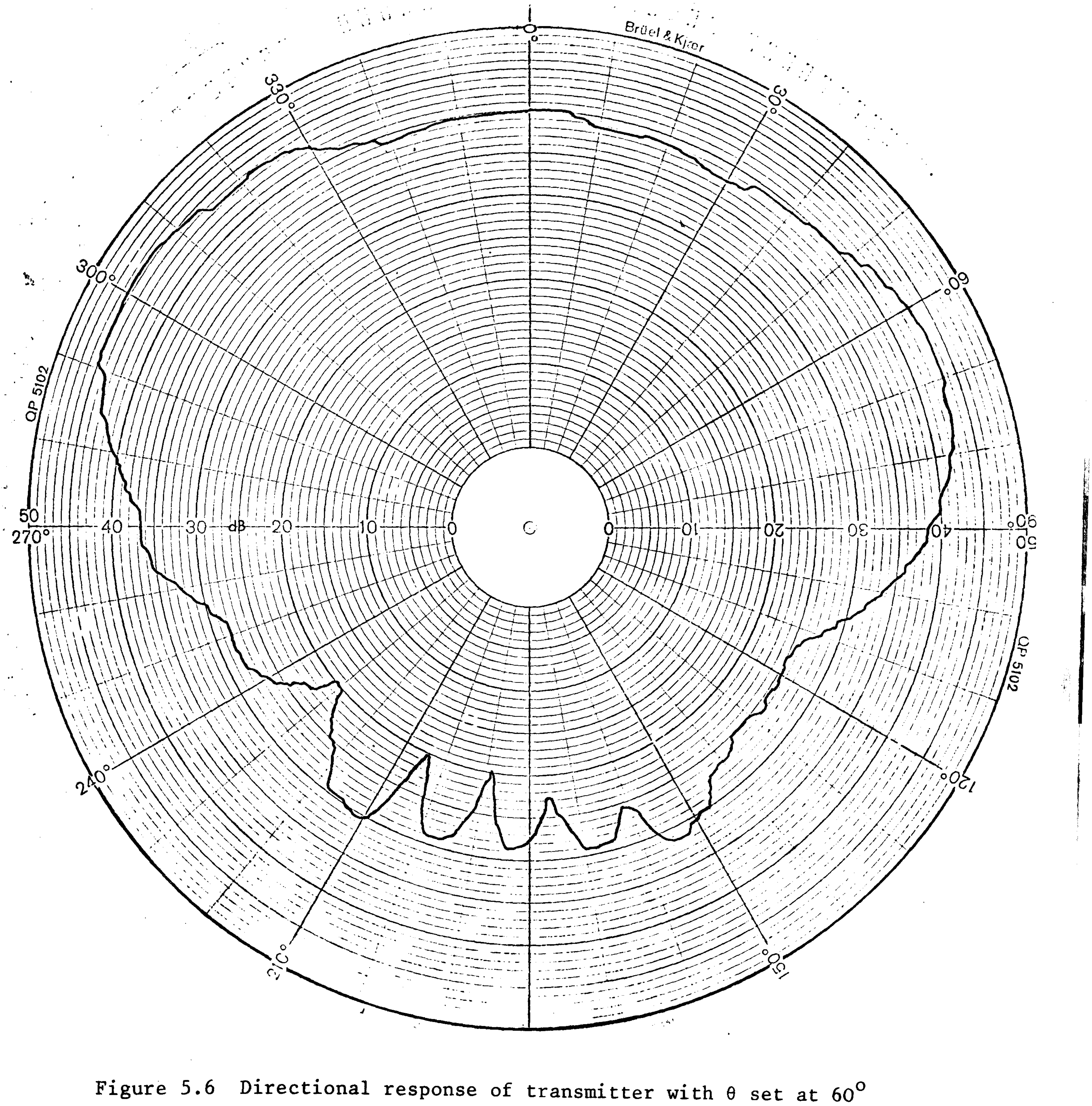


Figure 5.6 Directional response of transmitter with θ set at 60°

behind the transducer configuration, as illustrated in Fig. 5.2.

5.1.5 Discussion

In the design of an acoustic guiding aid which emitted audible clicks, Beurle[7] stated that it was useful to generate a wide beam with a concentration of sound on the centre line. This concentration in the centre of the beam enabled the user to pick out objects such as trees, lampposts, etc., some way ahead in the line of advance in the presence of larger objects close by. He added that the presence of some energy towards the sides facilitated the task of shorelining.

This line of thought was continued in the present device, where it was found that the most practical way of achieving the desired beam pattern was to arrange three transducers in a pyramidal form. In the system where the sideways pointing transducers are driven at reduced power the polar response revealed that almost as much energy was directed sideways as straight ahead. In the mobility tests carried out with this strong radiation to the side, the user was capable of following a shoreline very easily and irregularities in the shoreline such as open doorways were readily detected. It is unfortunate that with this directional pattern minima occurred at a particular angle on either side of the main axis, thus giving weak echoes from objects at this angular position.

The usefulness of radiating a substantial amount of energy sideways coupled with the necessity to eliminate the minima indicated the desirability of producing a very wide beam with a constant response over the whole of this sector. The theoretical calculations considering the three transducers to be arranged on the arc of a circle gave good agreement with the practical results despite the assumptions made.

The most favourable response recorded, that with θ set at 60 degrees, eliminated the unwanted minima but radiated slightly more energy sideways than straight ahead. This is perhaps at variance with Beurle's requirement of having a concentration of sound in the centre of the beam. This problem was initially overcome rather crudely by incorporating a facility which enabled the sideways pointing transducers to be switched off. Thus distant objects could be pinpointed with a narrow beam whilst progress along the shoreline could be checked using the wider beam pattern. A more elegant solution to the problem is now being developed. This entails sweeping the gain of the receiving amplifiers each time a pulse is transmitted so that echoes from a nearby shoreline are amplified less than those from a more distant lamppost or tree. This should prevent the masking of weak echoes from a distance by strong echoes from closer objects. The user can thus safely follow a shoreline without the fear of receiving inadequate warning of an obstacle or missing a useful landmark. Investigations are being carried out at present on the receiving amplifiers to find the optimum rate for sweeping the gain.

5.2 Receiving Transducers

5.2.1 Introduction

Several outdoor trials were carried out with an early experimental device using piezoelectric bimorph transducers as receivers. During these it was noticed that echoes from adjacent walls and fences were weak thus making it difficult to shoreline. Also, as the wearer of the head mounted receivers rotated his head from side to side, he perceived that objects did not remain stationary in space. In fact, they appeared to move away from the direction in which the head was turned. This meant that either the interaural time difference cues or the interaural intensity difference cues, or both, were inconsistent with the cues used by the natural auditory system of the user. It was felt that as the receivers are spaced at approximately the inter-ear distance, then the interaural time difference cue should be reasonably close to the natural one. The interaural intensity difference cue depends on the directional response of the transducers and the angle at which they are splayed outwards from the median plane. The transducers used have a 6 dB beamwidth of 60 degrees at 40 kHz, which is a narrow directional response when compared to the data available for the directional response of the human ear at audio frequencies[32] (see section 5.2.2). This narrow response tends to give a narrow perceptual field when the transducers are splayed by only a small angle from the median plane. However, when the angle is increased to give a broader field the interaural amplitude differences become much greater than the natural ones, thus resulting in the poor perceptual constancy found in the outdoor trials.

One may consider at this point two possible alternatives in the design of the binaural output of the device: the first is that the auditory sensation of space provided by the binaural output of the device should approximate as closely as possible to the natural auditory space of the user. For example, if the user is facing a "bleeping" traffic light at a pelican crossing, then the device should indicate the presence of the traffic light in the same direction as that from which the bleeping sound emanates. Thus, information from the mobility device and the natural perception of space would be matched.

The alternative is for the device to have a binaural output which is not matched to the auditory system of the user. The user would then have to learn a new localising function and use it to interpret signals from the device whilst still using his natural response when listening to ambient sounds. It is not known whether, in such a system, there would be interaction between the two localising functions causing confused localisation. Experiments on auditory reorientation carried out by Held[33] showed that listeners could perceive two images from one sound source after exposure to an atypical set of localisation cues.

A choice between these two options arose in the development of the Kay binaural spectacles,[12] which use interaural amplitude differences alone to give localisation cues. Investigations carried out by Rowell[34] indicated that due to wide variations between localisation functions of individuals the display of the device should be matched to each individual, a difficult task even under laboratory conditions. Subsequent experiments were conducted to see if a subject, after being taught to use a direction cue stimulus which was not 'matched' to his hearing, could adapt to the new stimulus. The results were inconclusive. Kay finally stated that there was no more evidence in favour of one opinion than the other, and he went on to adopt a localisation function for the device based on successful user experience.

It is thought by the writer that the localisation function of the device should, in the first instance, be matched as closely as possible to the average natural auditory localisation function. It is realised that a perfect match will be difficult to achieve, and it is felt that discrepancies which arise between the artificial and natural system will be overcome by the user learning to respond accurately to the new stimulus. Thus, in fact, the user may well have to respond to two localisation functions but it is thought that the closer the artificial stimulus is to the natural one, then the more rapid will be the learning process with the least possibility of confusion.

As a consequence of this conclusion, information was sought on the directional response of the human ear in the horizontal plane, so that the receiving transducers could be modified to match this response as closely as possible.

5.2.2 The directional response of the human ear

Shaw[32] has recently brought together in a common framework measurements from twelve studies on the transformation of sound pressure level from the free field to the eardrum in the horizontal plane. The pool of data covers 100 subjects, the majority male, measured in five countries over a 40 year period. Data are presented showing the azimuthal dependence of the sound pressure transformation at the eardrum and the interaural level difference at the eardrum as functions of azimuth at 24 discrete frequencies. Fig. 5.7. shows examples of the presentation of the data at three frequencies of interest. Fig. 5.8 shows the azimuthal dependence data plotted out in polar form.

The curves presented are average curves but one may infer from Shaw's discussion that there is not too large a deviation from the

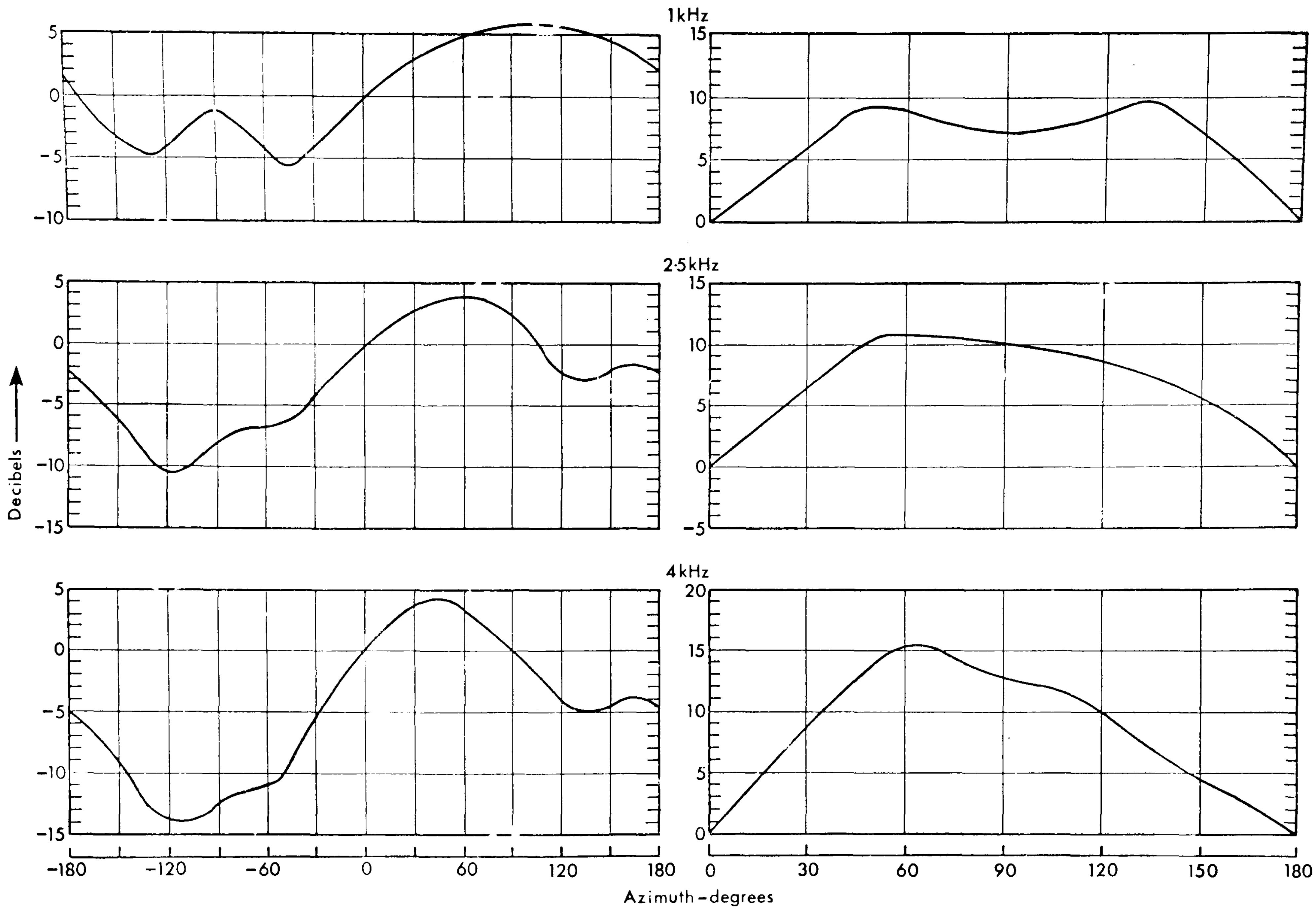


Figure 5.7 Interaural amplitude difference (right) and azimuthal dependence (left) as functions of azimuth at three frequencies (After Shaw, 1974)

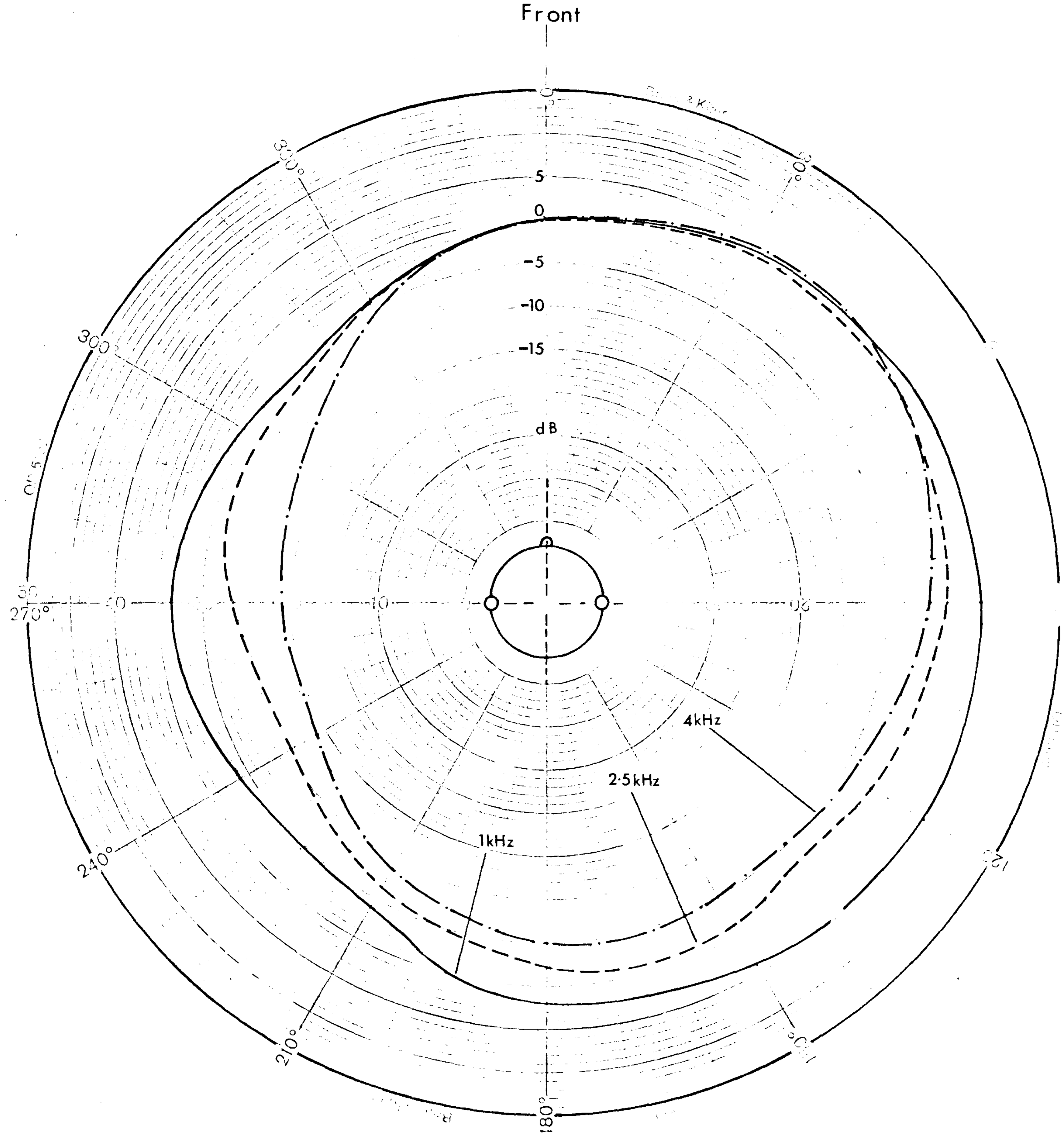


Figure 5.8 Azimuthal dependence of the sound pressure transformation at the eardrum (right ear)

curves of an individual subject, although some of the finer structure may have been smoothed out. We may take these curves, then, as an ideal to aim for when modifying the receiving transducers to broaden their polar response. The curves, however, are frequency dependent, and as it would be very difficult to simulate this frequency dependence in the polar response of an ultrasonic receiver, a compromise must be reached. The audio output of the mobility aid nominally contains frequencies up to 8 kHz; however, the low frequency response is poor due to the loose coupling into the ear, and also little energy is contained in the higher frequencies. We may assume, then, that most of the audio energy is contained in the frequency band 1 - 4 kHz. This was confirmed by examining the frequency spectrum of the audio output of the device. We must therefore aim for the receiving transducers to have a polar response at 40 kHz which matches that of the ear in the frequency range 1 - 4 kHz.

5.2.3 Investigations on the directional response of a receiver

Preliminary Measurements

The directional response of a receiving transducer was recorded by driving it as a transmitter. This was more convenient, and is valid because, by reciprocity, the transmitting and receiving patterns of linear electroacoustic transducers are identical. The resulting polar diagram, shown in Fig. 5.9, reveals a smooth response in the forward direction with a 6 dB beamwidth of 55 degrees.

It can be seen that this polar diagram deviates appreciably from that of the ear (c.f. Fig. 5.8). Consequently, an investigation was made into the possibility of broadening the directional response and matching it as closely as possible to that of the ear.

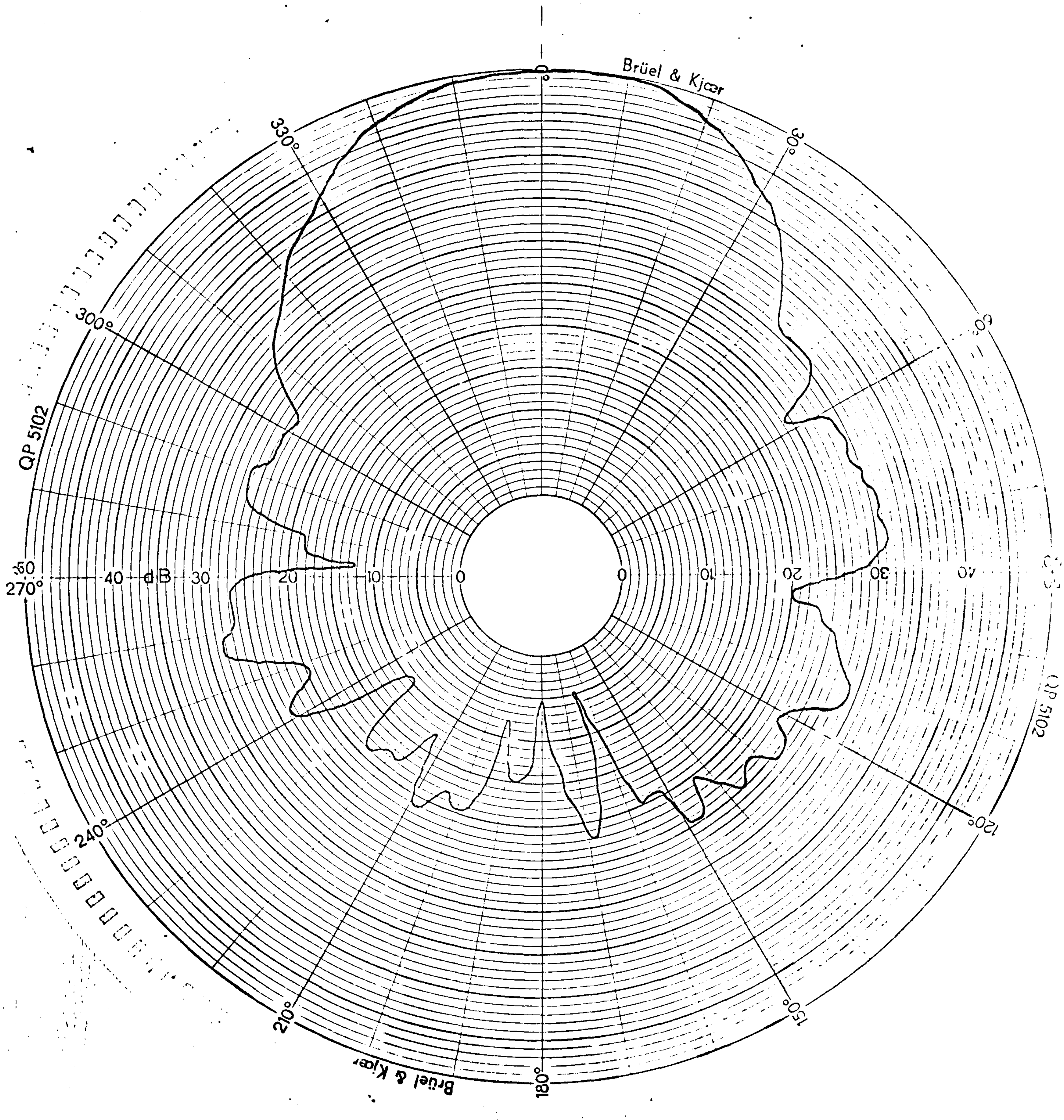


Figure 5.9 Directional response of standard transducer

Initial experiments were carried out using curved reflectors. These gave a wider beam with the expected loss in sensitivity; values for the 6 dB beamwidth approaching 90 degrees were obtained with a drop in peak sensitivity of 5 dB. Outside the most sensitive region, however, the response tended to fall off rapidly. It was thought that a wider and smoother response could be obtained by a more fundamental approach.

Theoretical Considerations

The fundamental relationship in considering directionality is the Fourier transform which, in the case of a receiver, relates the sensitivity over the face of the transducer to the distribution of directional response in space. The directional response is given by the directional function D(K) which, for a symmetrical transducer, is related to the sensitivity or taper function T(r) across the face of the transducer by the cosine Fourier transform. We may write this in the form

$$D(K) = \int_{-\infty}^{\infty} T(r).\cos Kr.dr.$$

where $K = \frac{2\pi}{\lambda}\sin\theta$; λ being the wavelength in the medium and θ being the angle of a particular signal direction to the axis of the transducer.

As we are concerned with circular transducers it is useful to consider the beam pattern of a circular piston source with uniform sensitivity over the surface. It can be shown,[35] using the above relationship, that the directional function is given by

$$D(K) = \frac{2J_1(Ka)}{Ka}$$

where a is the radius of the transducer and $J_1(Ka)$ is the Bessel function of the first kind and first order and of argument Ka. A graph of this directional pattern is shown in Fig. 5.10. The function

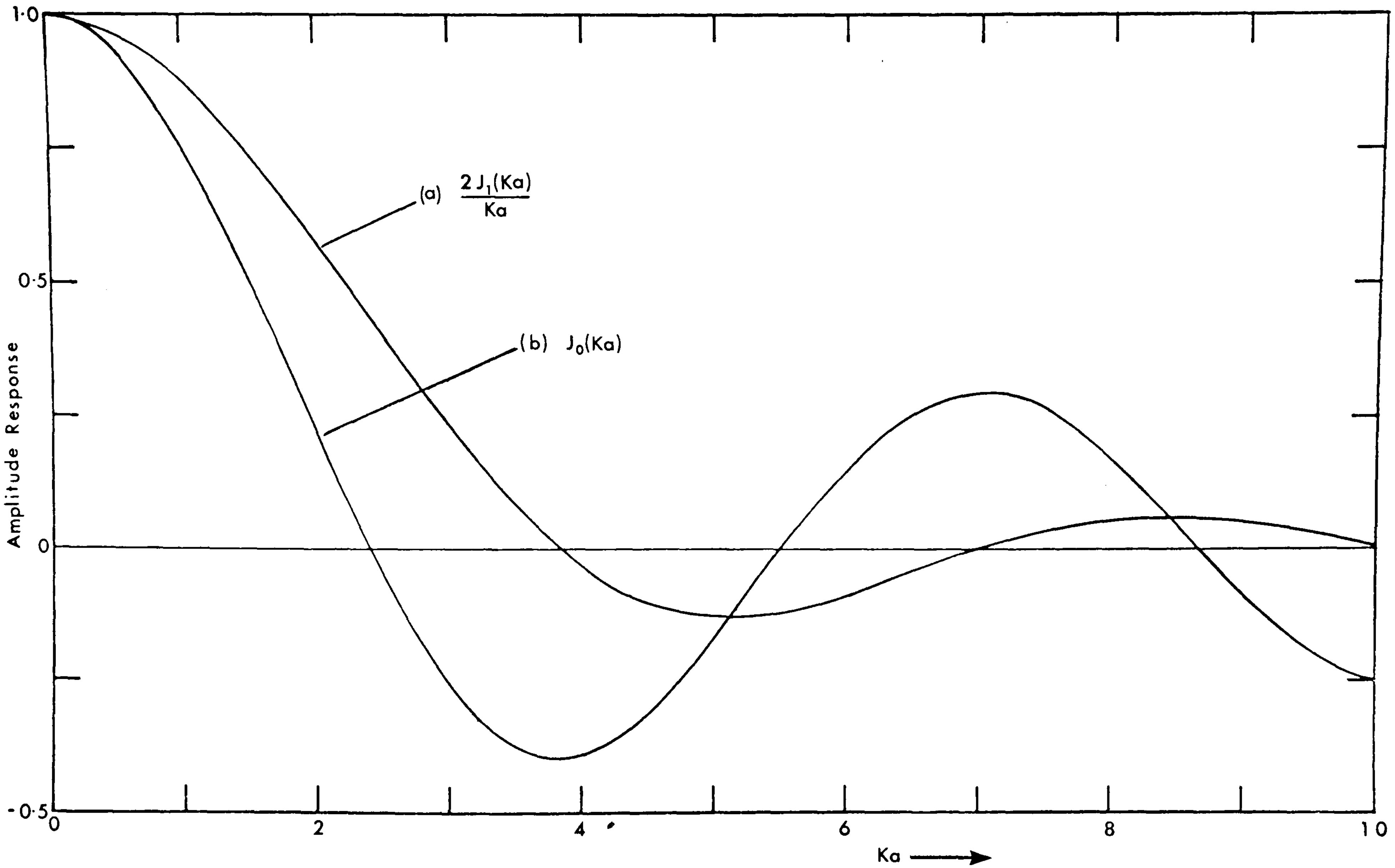


Figure 5.10 Directional pattern for (a) circular transducer and (b) circular ring source

becomes zero for arguments equal to 3.83, 7.02, 10.15 etc. The main lobe, therefore, is confined between the angles $\pm\theta$ which are determined by

$$\sin\theta = \frac{3.83}{\frac{2\pi}{\lambda}\cdot a} = 0.61 \cdot \frac{\lambda}{a}$$

This is the Fraunhofer formula for determining the sharpness of the main beam produced by a circular piston source. Clearly, to obtain a wide beam it is necessary to make the ratio $^{\lambda}/a$ as large as possible.

In order to investigate the directional behaviour of the piezoelectric bimorph transducers it is necessary to examine their construction; this is sketched in Fig. 5.11. The central area of the piezoelectric element vibrates in counterphase with its edges so that, without any precaution, there would be partial cancellation of the radiated ultrasound, resulting in very low efficiency. To overcome this, the central area of the element is shielded by an aluminium disc which prevents direct radiation from this region. A secondary effect is that phase conversion of the laterally directed sound takes place at the same time, so that the radiation from the circumference of the element is reinforced. The $^{\lambda}/a$ ratio for the element is 1.67 which gives a beamwidth of 180 degrees by the Fraunhofer equation, or 73 degrees to the 6 dB points.

As this beamwidth is greater than the one actually measured, it seemed likely that a closer result would be obtained by considering the transducer as a circular ring source. This has a directivity function given by

$$D(K) = J_o(Ka)$$

where J_o is the Bessel function of zero order and of argument Ka. From a graph of this directional pattern, shown in Fig. 5.10, we

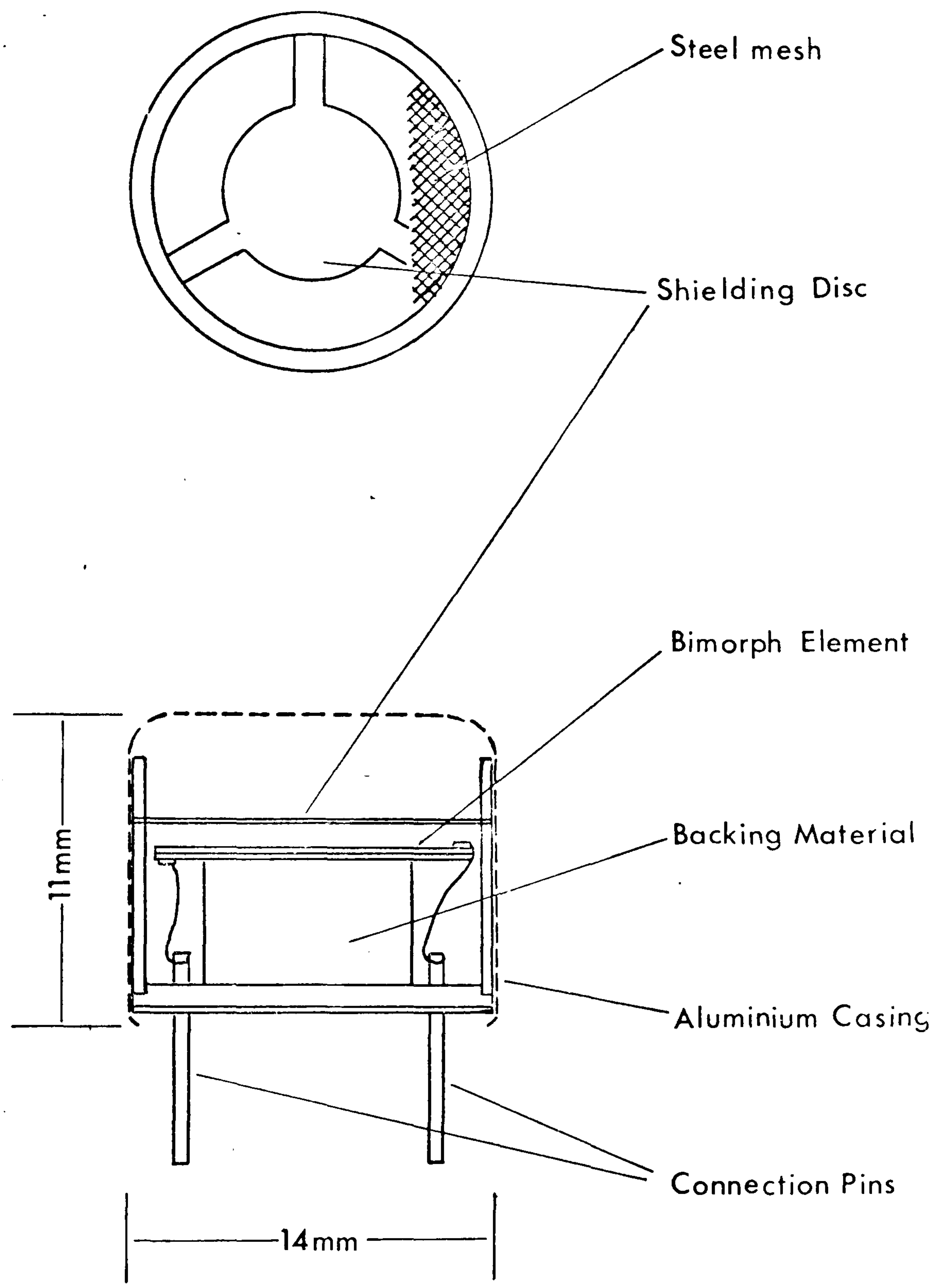


Figure 5.11 Construction of Piezoelectric Bimorph Transducer

can evaluate that the 6 dB beamwidth is 47 degrees, a result agreeing more closely with the measured value. We may consider the piezoelectric bimorph transducer, then, to approximate to a circular ring source whose $^{\lambda}/a$ ratio we need to increase in order to obtain a wider beam pattern.

One method of increasing the $^{\lambda}/a$ ratio is by reversing the shielding arrangements in the transducer design. Thus the area at the centre of the element would be utilised, with the edges masked off. This arrangement almost doubles the $^{\lambda}/a$ ratio to 3 and, if we consider the exposed central area as a circular piston source, we obtain a value for the 6 dB beamwidth of 180 degrees. This beam pattern would approximate to the directional response of the ear much more closely, and so investigations were carried out to see if it could be realised in practice.

<u>Practical Investigations</u>

Initial experiments were carried out by using a masking annulus which reversed the original shielding arrangements; thus the previously exposed circumference was covered whilst the centre was exposed. The annulus was mounted at the same distance above the element as its predecessor. The polar response revealed a much wider beam, the 6 dB beamwidth being 115 degrees, but with a loss in axial sensitivity of 11 dB. Several other masking annuli were tried with different sized central holes; the results revealed little difference between the responses, although the original annulus seemed to give the widest beam with the least loss of sensitivity. An illustration of the conversion of a standard transducer to one with a modified masking arrangement is shown in Fig. 5.12.

Figure 5.12 (Top) Standard Transducer, (Centre) Casing removed for transmitter, (Bottom) Masking arrangement modified for receiver

It is thought that with the modified masking arrangement, as in the original one, the radiation from the masked area seems to reinforce that from the unmasked area. In an endeavour to optimise this effect, the masking annulus was spaced at different distances above the surface of the element to find the position at which the laterally directed sound gave maximum reinforcement. This was achieved by placing different lengths of cylindrical spacer on the transducer base and surmounting this by the masking annulus. Fig. 5.13 shows on the same axes a plot of spacer length against axial sensitivity, and spacer length against frontal area encompassed by the polar response. Both rise to a maximum with a spacer of length 0.287 inches, indicating that this separation gives the widest beam and the maximum sensitivity. The beamwidths, in fact, were found to vary little throughout this experiment, a 6 dB beamwidth of 120 degrees being obtained with the 0.287 inch spacer.

This length of spacer, then, was chosen to make several receiving transducers of the modified form. The process consisted of centring and glueing the steel spacer to the transducer base and then centring and glueing the masking annulus onto the top. The polar response of the modified receiver, shown as the solid line in Fig. 5.14, indicates a 6 dB beamwidth of 120 degrees at 38 kHz with a loss in axial sensitivity, as a result of the modification, of 8 dB. This diagram also illustrates the change in polar response from an unmodified transducer (dotted line) to a modified one. Polar responses taken over the frequency range 34-42 kHz exhibited negligible differences between each other with change in frequency.

Further polar plots were carried out to discover how well the modified transducers could be matched to each other. The polar plots

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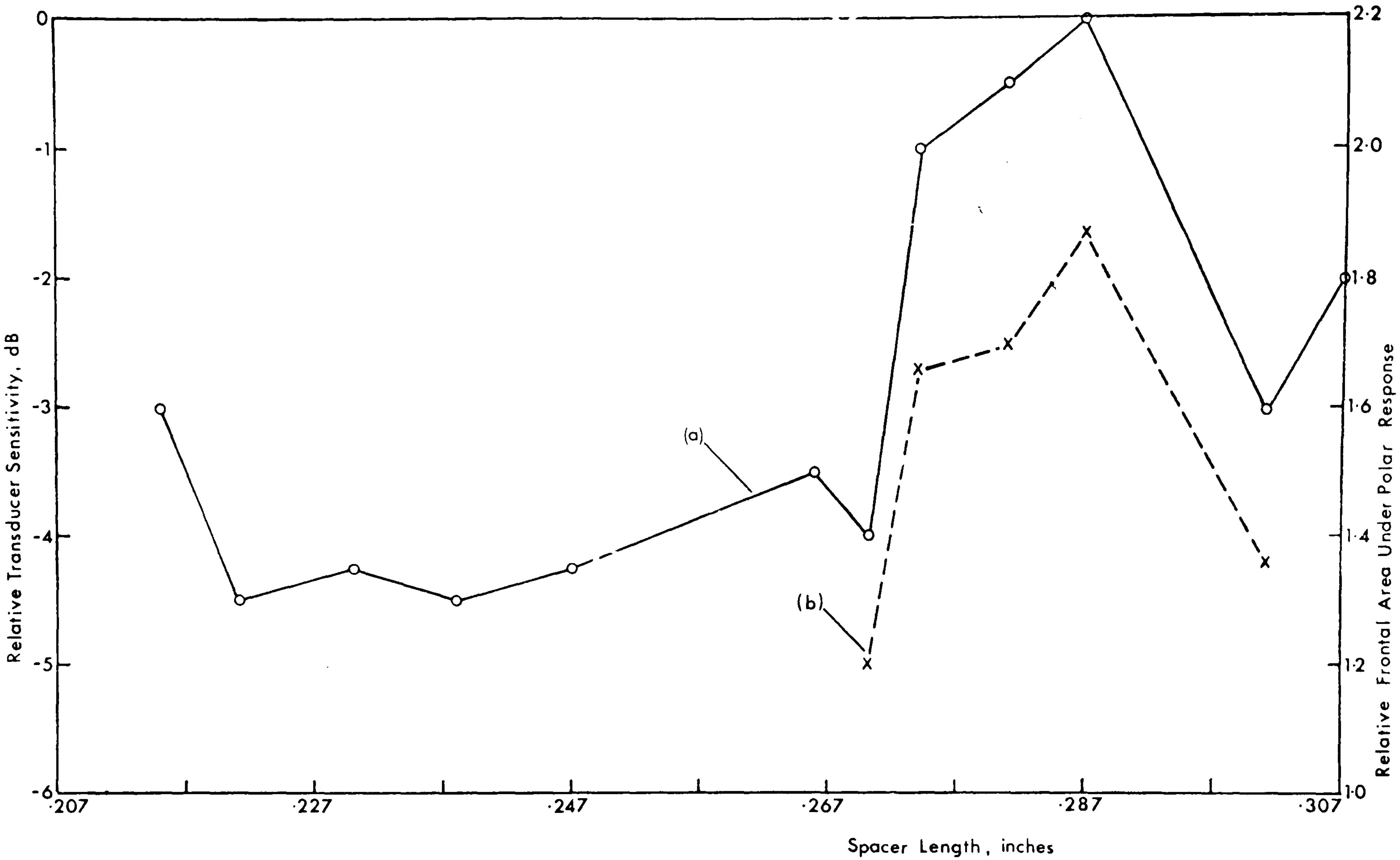


Figure 5.13 (a) Relative transducer sensitivity and (b) Relative frontal area under polar response as a function of spacer length

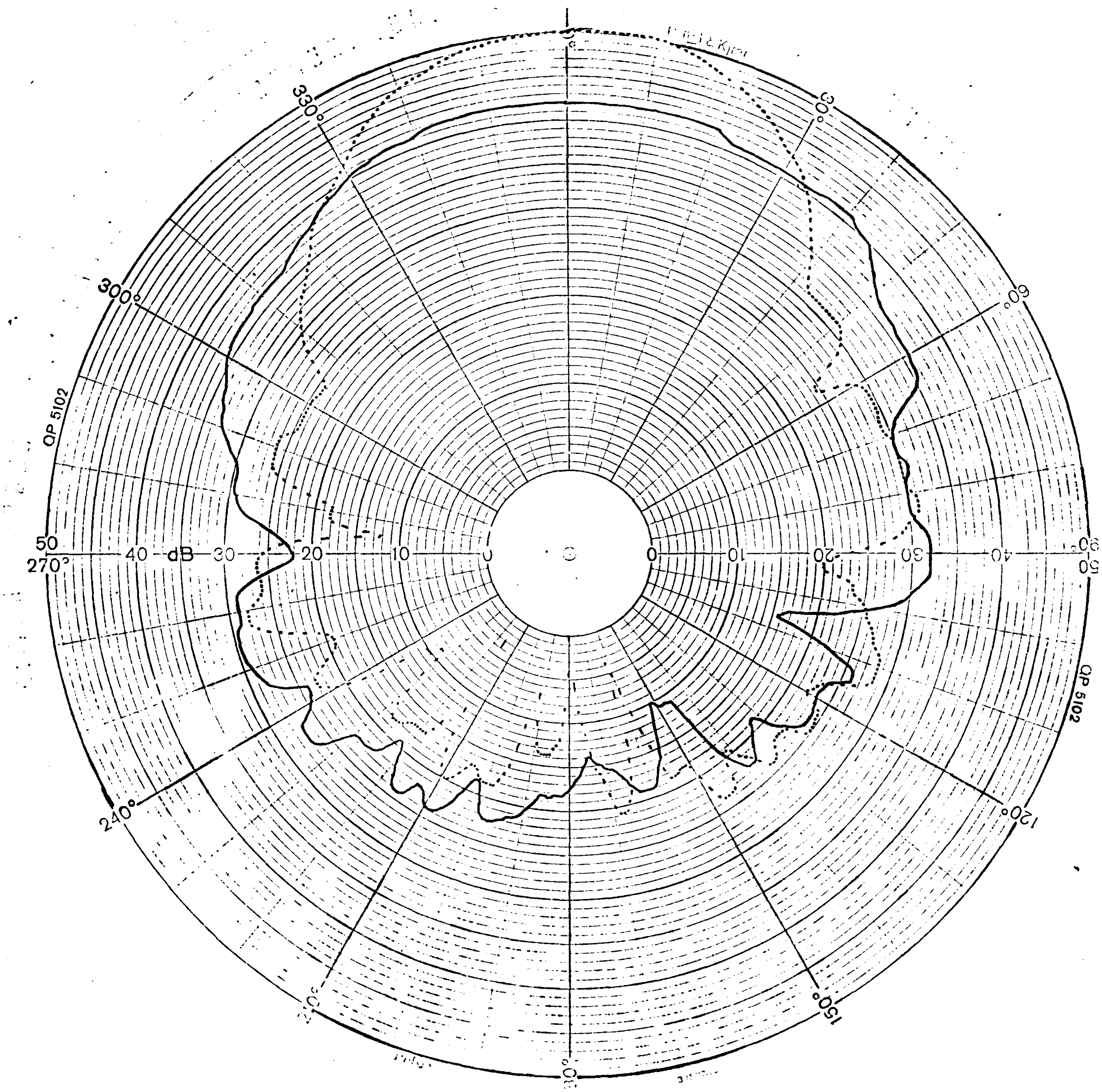


Figure 5.14 Comparison of the directional response of a modified receiver (solid line) with an unmodified one (dotted line)

were found to be very similar and showed that matching to within 2 dB. in the forward direction could easily be achieved. Fig. 5.15 shows the directional response of the human ear at 2.5 kHz, together with that of the modified receiver. It can be seen that the resemblance is close.

5.2.4 Discussion

From the theory we may infer that an approximate doubling of the λ/a ratio for a standard transducer should yield a beam pattern similar to that of the ear in the frequency range 1-4 kHz. Reversal of the normal masking arrangements which almost doubled the λ/a ratio gave an increase in beamwidth from 55 degrees to 120 degrees. This did not quite match up to the theoretical prediction of a beamwidth of 180 degrees, probably due to the inaccuracy of considering the exposed area of the modified transducer as a circular piston source. A possible reason for this discrepancy is that the laterally directed sound gives an effective diameter larger than the actual one. Nevertheless, the theoretical considerations are thought to have given a useful indication of the beamwidth.

The loss in peak sensitivity of 8 dB is as expected since, theoretically, the broadening of the beam from 55 degrees to 120 degrees should give a loss of 7 dB. Any loss in sensitivity, however, is undesirable and it is thought that the original overall sensitivity of the system can be regained by an increase in transmitter power.

A comparison between the directional response of a modified transducer and the response of the ear at 2.5 kHz, Fig. 5.15, shows that, for optimum matching, the transducer axis should be moved about 30 degrees from the median plane of the head. Having done this, the match between the two responses is very close in a 120 degree sector

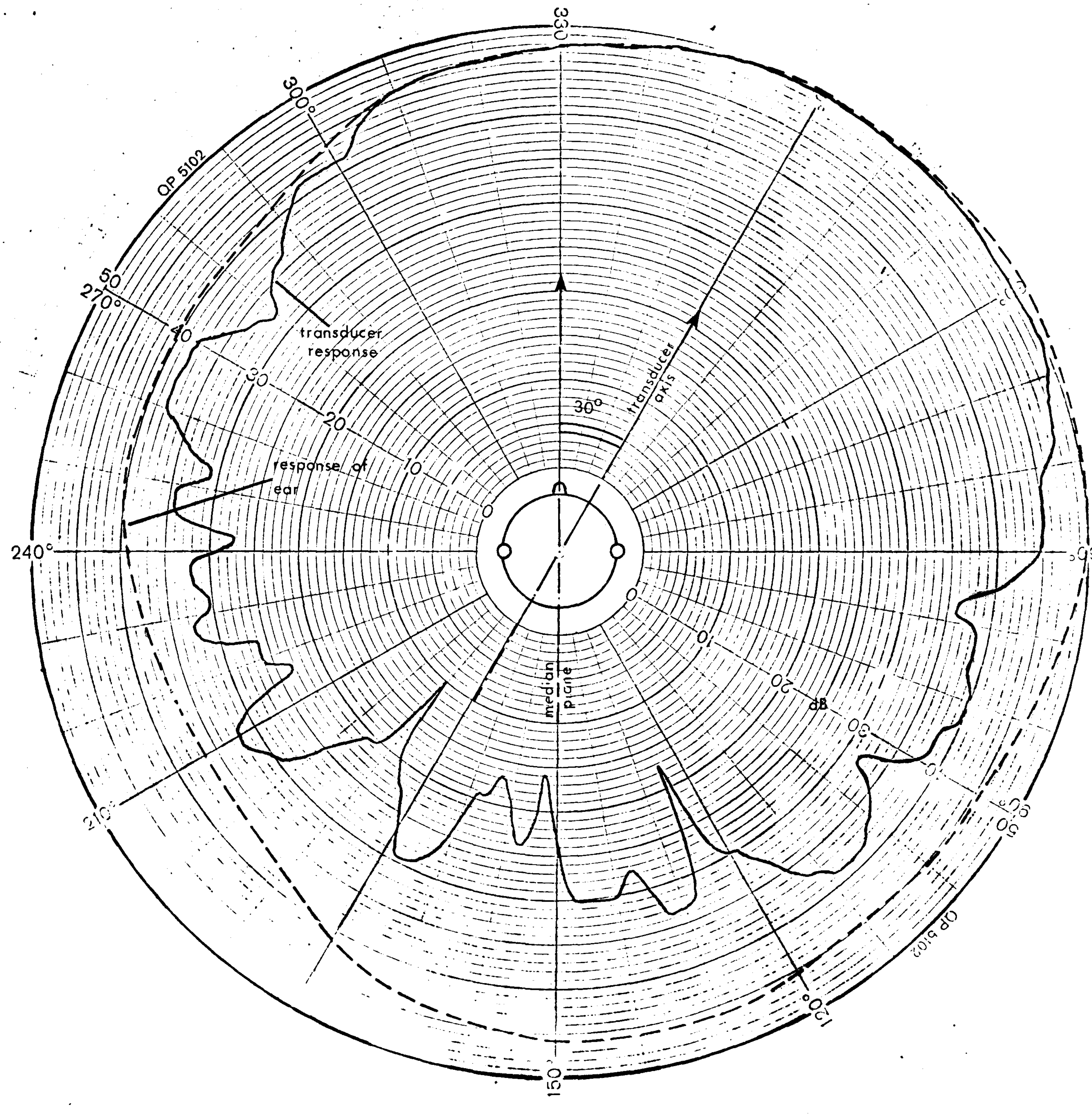


Figure 5.15 Comparison of the directional response of the human ear at 2.5 kHz with a modified receiving transducer.

in front of the head. If the modified transducers were incorporated into a device and oriented in this way then, in the frontal sector, the interaural amplitude differences produced by objects in various azimuth directions would be very similar to those used in natural localisation. A comparison between the two sets of interaural amplitude differences is made in Fig. 5.16; the natural amplitude differences are obtained from Shaw's data, whilst those produced by the device have been evaluated from a typical pair of matched transducers whose axes are splayed 30 degrees outwards from the median plane. This data is regarded as being representative since, as referred to earlier, all modified transducers exhibited a similar response and pairs could be matched to within close tolerances.

It can be seen from Fig. 5.16 that for natural localisation the slope of the graph increases from just over 0.2 dB/degree to 0.3 dB/degree in the range 1-4 kHz. With the modified transducers splayed outwards by 30 degrees, a smooth curve is obtained from 0-40 degrees with a slope of 0.26 dB/degree. At angles greater than 40 degrees the curve becomes rather erratic due to irregularities in the directional response.

Thus with this arrangement incorporated into a device, it is expected that users will receive accurate interaural amplitude difference cues in the sector ± 40 degrees to either side of the median plane, but that the cues outside this area will be rather poor. It would be desirable, of course, for the users to receive accurate cues over a wider area, but an improvement cannot be envisaged without further extensive work on the beam shaping of the receiving transducer.

We may consider, therefore, that the matching of the receiver response to that of the ear appears sufficiently good not to justify further work at present. In the 80 degree sector in front of the user, the interaural amplitude difference cues from the device are so close

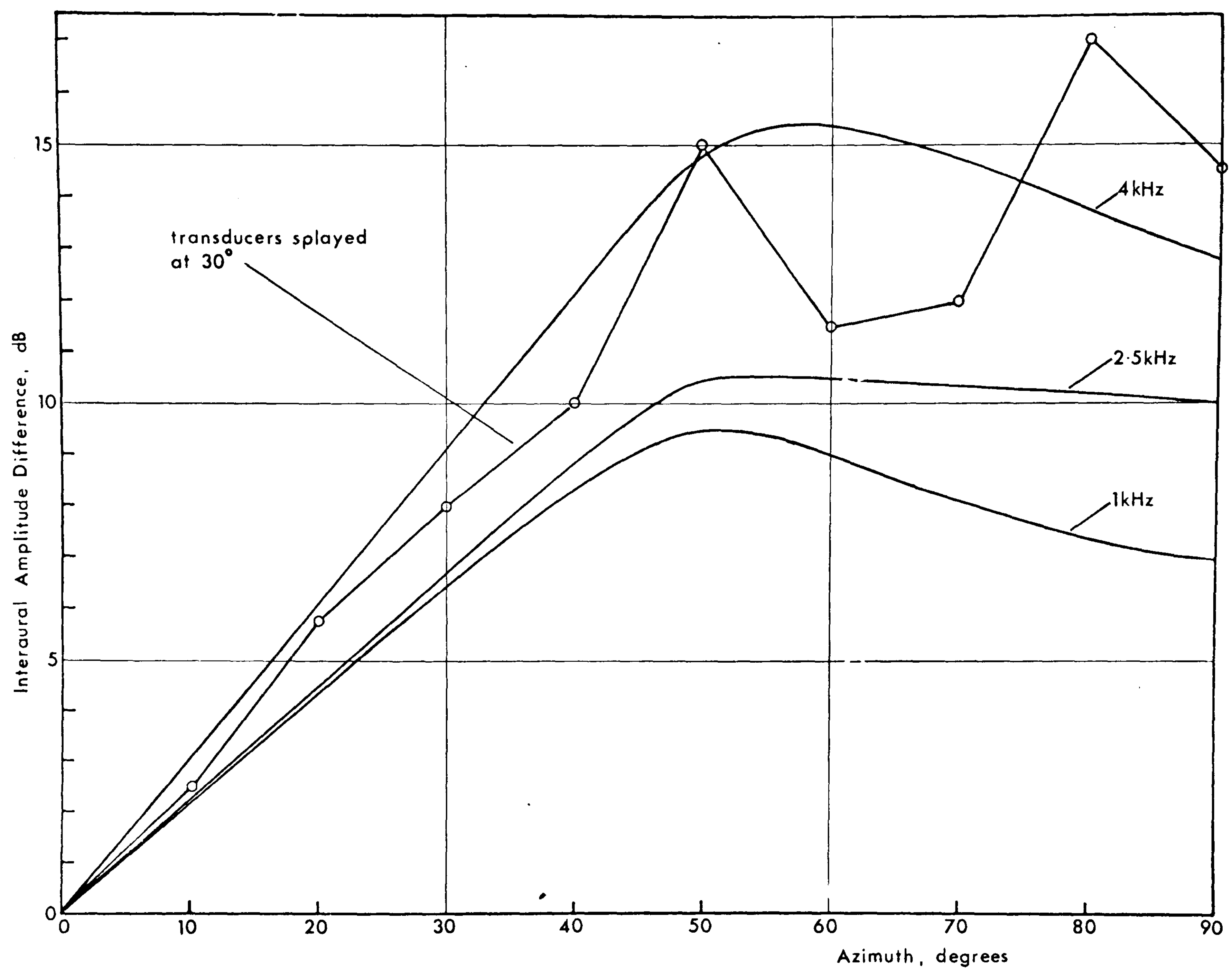


Figure 5.16 A comparison between the interaural amplitude differences as a function of azimuth provided by the modified receiving transducers splayed at 30^{o} with those occurring naturally at three frequencies

to the natural ones that it is hoped the user will need little time to adapt to them in order to localise accurately. Outside this area, however, more time may have to be spent in becoming familiar with the new stimuli.

The modified receiving transducers were incorporated into an experimental version of the device. Outdoor tests showed firstly that the perceptual constancy was greatly improved, i.e. that objects remained stationary as the head was rotated, and secondly that the task of shorelining was much easier. More rigorous localisation experiments to test quantitatively the effectiveness of the modified receivers were carried out at a later stage.

CHAPTER 6

AUDITORY LOCALISATION

6.1 Introduction

The preceding chapters have described the work carried out in optimising the transmitting and receiving transducers for inclusion in a prototype blind mobility device. It was felt, at this stage, that the development of the system had reached a sufficiently high level to justify the performance of objective psychoacoustic tests. The results of these should hopefully yield useful pointers as to the direction that further development should take. As an introduction to this work a study has been made of the present day theories on auditory localisation. Some of the information has been drawn from several excellent reviews that exist on this topic.[36,37]

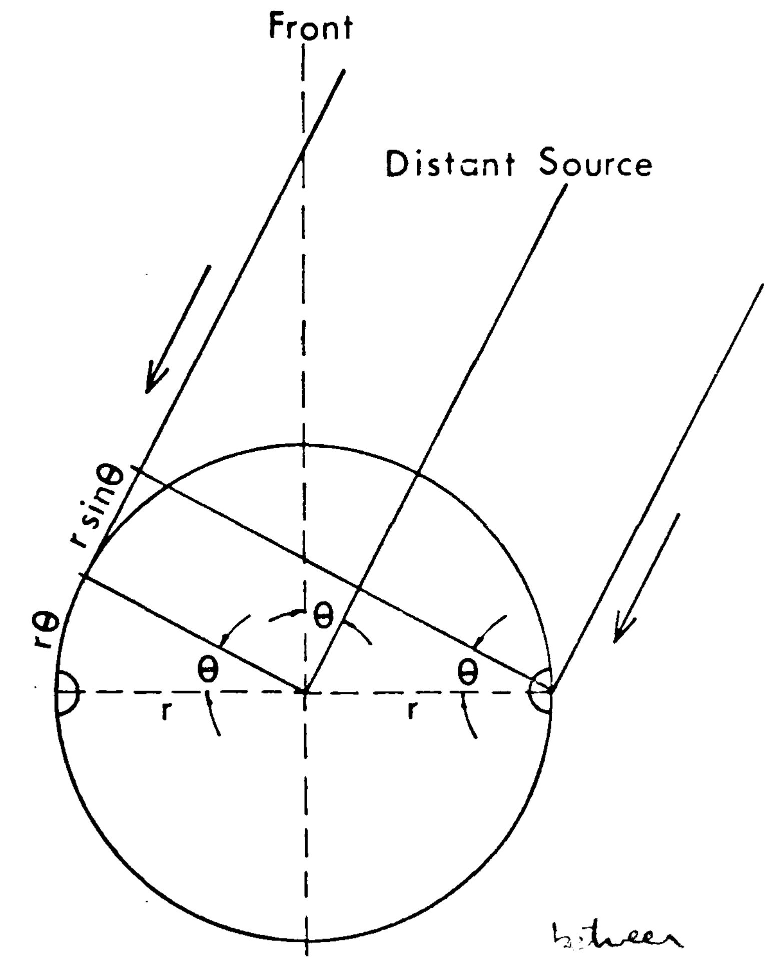
6.2 Auditory Localisation Phenomena

Human beings have the capability to say in what direction a sound source lies relative to them in their normal environment. Early investigators into auditory localisation, realising the importance of having an ear on either side of the head, concentrated on two measurements: interaural differences in the intensity of the sound and interaural differences in the time at which it reaches the ears.

If the ears are considered as a pair of holes separated by a sphere, then a distant sound source lying in front of the interaural axis and producing a plane wave front will provide a difference between the shortest distance to the ears, Δd, given by

$$\Delta d = r(\theta + \sin\theta)$$

This is illustrated in Fig. 6.1 where the angle θ is measured in radians. If the spherical head is given a radius of 8.75 cm. and the velocity of sound is taken to be 343 metres per sec., then the



between

Figure 6.1 ~~Differences~~ Difference in ~~between~~ the distances ~~of~~ the ears and ~~from~~ a source of sound sufficiently ~~distant~~ to produce a nearly plane wave front. far from the head

Path lengths from the ears to a

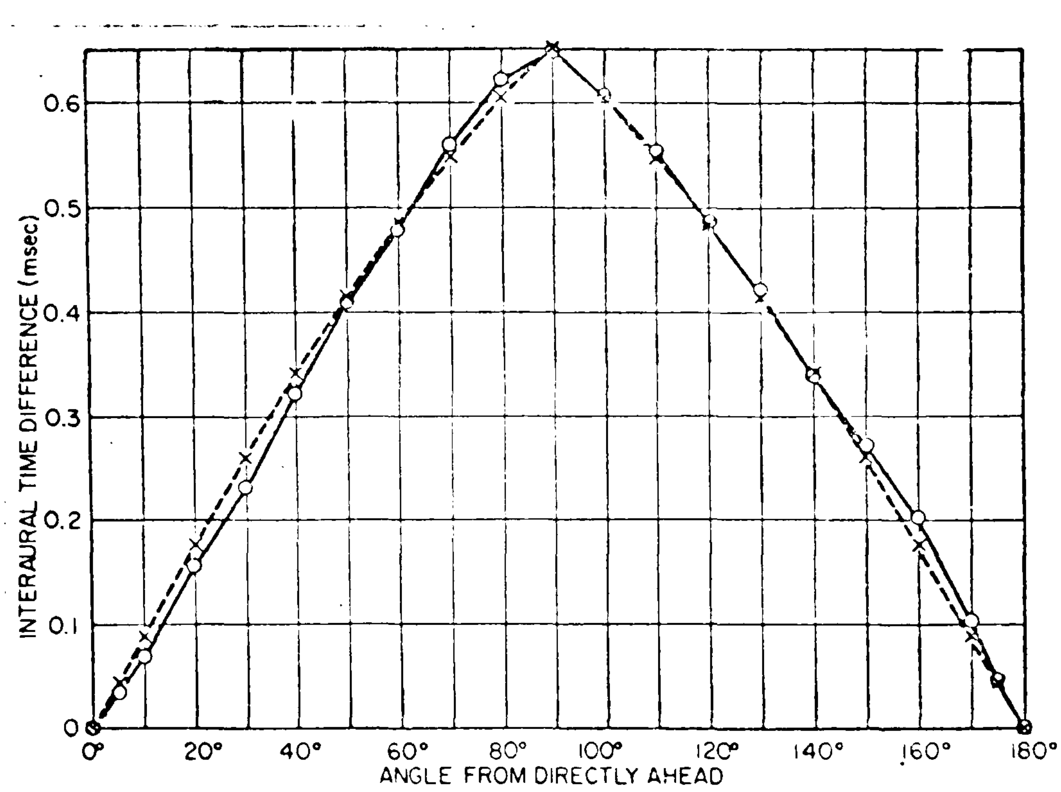


Figure 6.2 Interaural time differences as a function of the direction of the source of clicks (—— : measured values for five subjects; - - - : values computed from sphere) (Feddersen et al, ref. 38).

interaural time difference, t, is given by

$$\Delta t_{\mu sec} = \frac{\Delta d}{c} = 255\ (\theta + \sin\theta)$$

This formula gives results which are in good agreement with actual measurements of interaural time differences carried out by Feddersen et al.[38] This is shown in Fig. 6.2.

Interaural difference in intensity, illustrated in Fig. 6.3 after Shaw,[32] does not show as logical a pattern. The far ear lies in a sound shadow created by the head, the depth of the shadow depending on the wavelength of the sound and the direction of the source. The head can be regarded as a low pass filter for the far ear whose characteristics are not a simple function of either the frequency or the direction of the sound.

Stevens and Newman,[39] using tone bursts from a source movable in the horizontal plane, showed that listeners practically never confused a sound on one side of the head with one on the other, although a low frequency sound was often indistinguishable from its mirror image behind. Their objective measurements indicated that localisation is more accurate at low or high frequencies than in the middle of the acoustic spectrum.

Sandel et al,[40] using pure tones showed that the frequency at which localisation is most uncertain is about 1500 Hz. Further experimentation led them to say that the localisation of pure tones is determined by interaural differences in time or phase for frequencies below about 1500 Hz, and by interaural differences in intensity for higher frequencies.

Mills[41] measured the resolution of auditory localisation, the minimum audible angle, by having a listener indicate whether two successive sounds came from the same or different directions.

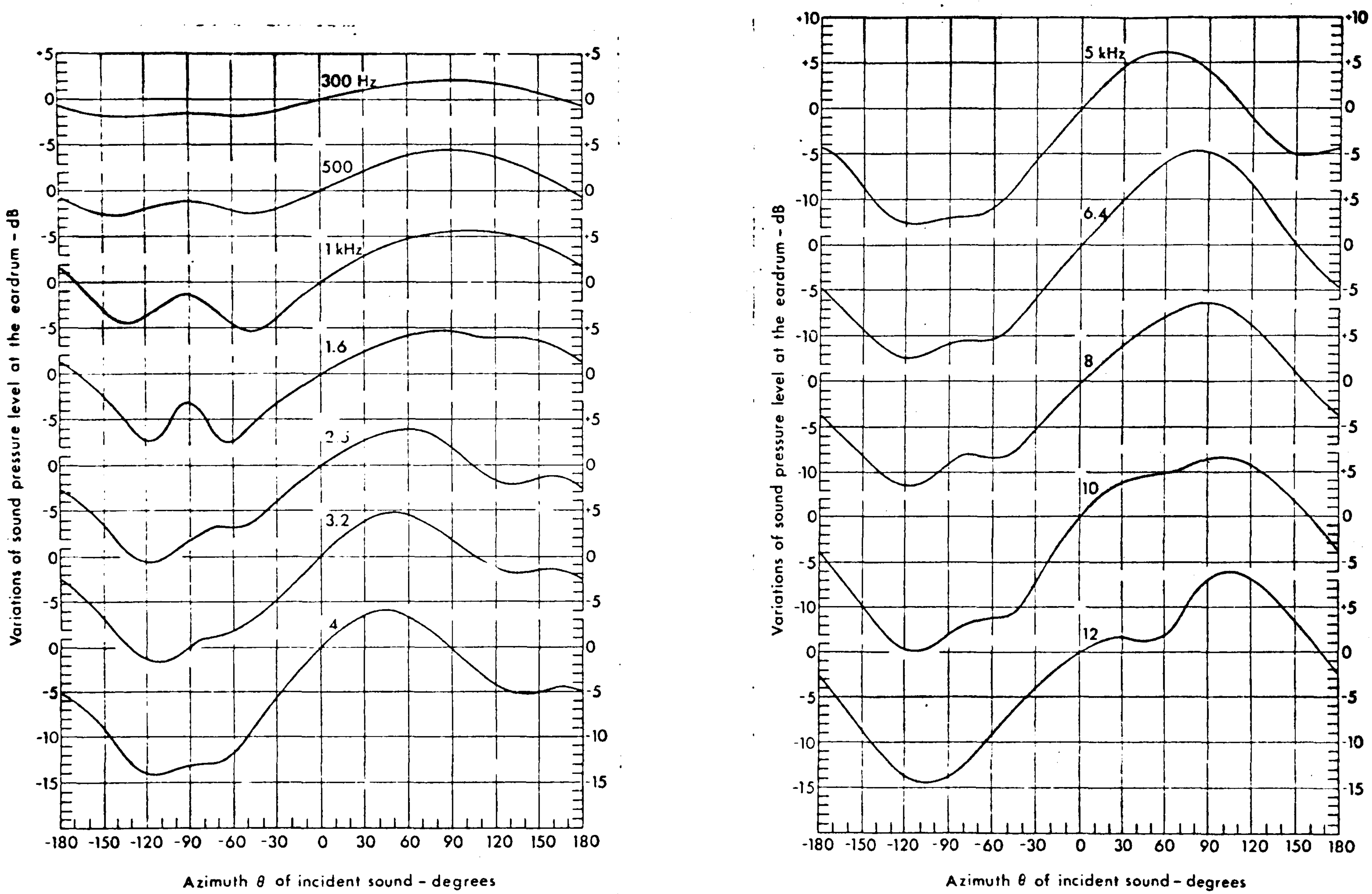


Figure 6.3 Variations of average sound pressure level at the human eardrum as a function of azimuth θ of incident sound for various frequencies. (From Shaw, ref. 32) These curves permit the evaluation of interaural intensity differences.

His results show that the minimum audible angle is only a few degrees for low or high frequencies and large for frequencies between 1500 and 2000 Hz. It is small for sources straight ahead and large for sources to either side. For tone sources between 1500 and 2000 Hz at azimuths of more than 45 degrees, the minimum audible angle is indeterminately large.

The minimum audible angle has been used to investigate the effect of interaural time and intensity differences on localisation by delivering sounds separately to each ear. These dichotic, earphone presented sounds are heard as if inside the head instead of out in the environment, but they still possess a subjective laterality that depends on the interaural differences.

Zwislocki and Feldman[42] measured the interaural phase difference that is just noticeable when equal amplitude pulses of tone are presented dichotically. Their results are given by the heavy dashed line in Fig. 6.4. The light dashed line represents the interaural phase differences produced by moving an actual source out of the median plane by one minimum audible angle. The two functions lie close together below about 1500 Hz and separate abruptly above this frequency. This confirms that the localisation of tone pulses is determined by the interaural phase of the tones below about 1500 Hz.

The interaural intensity difference that is just noticeable when cophasic pulses of tone are presented dichotically is represented by the heavy continuous line in Fig. 6.4. The light continuous line represents the interaural intensity difference that is produced by moving an actual source out of the median plane by one minimum audible angle. There is a wide separation between the two functions at low frequencies, the interaural intensity difference produced by an actual

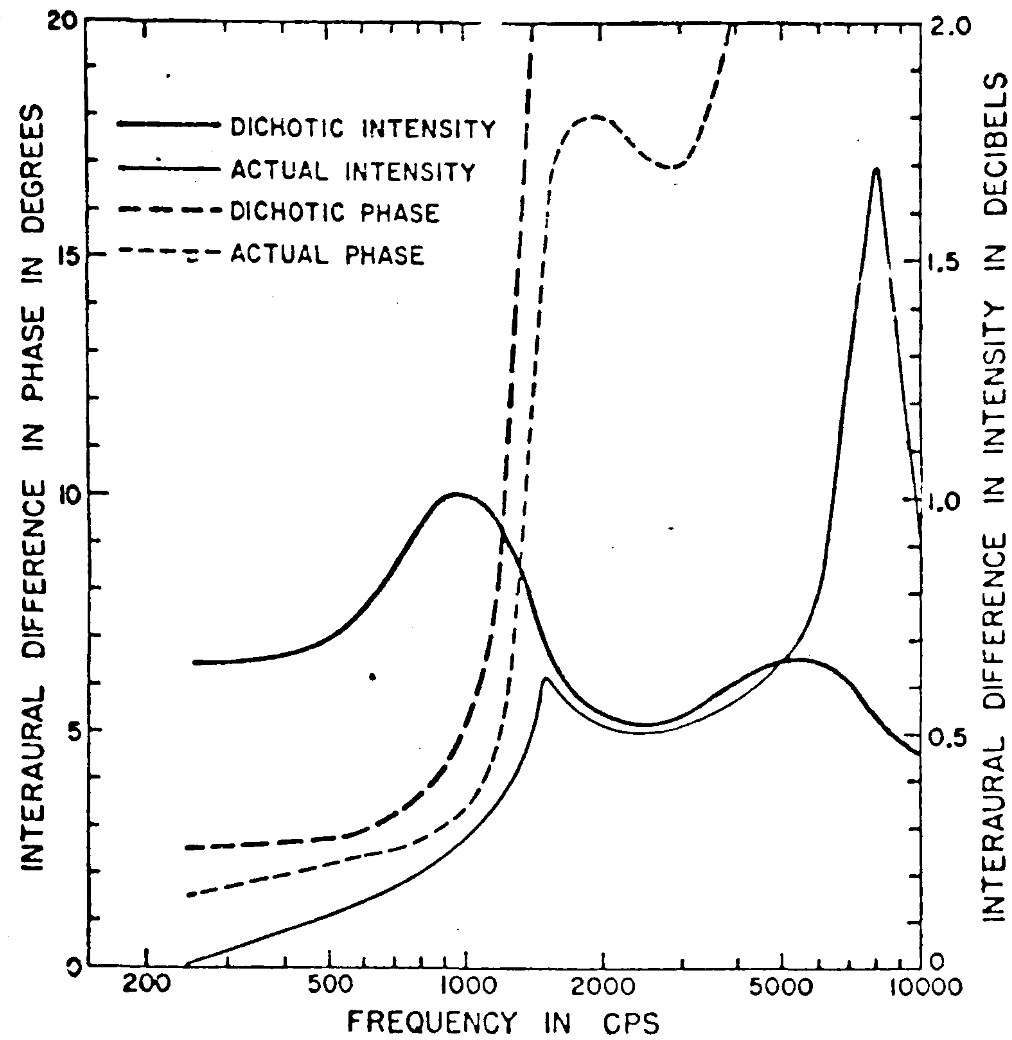


Figure 6.4 Comparison of just noticeable dichotic interaural differences with the differences induced when an actual sound source is just noticeably displaced from the median plane (After Mills, 1960).

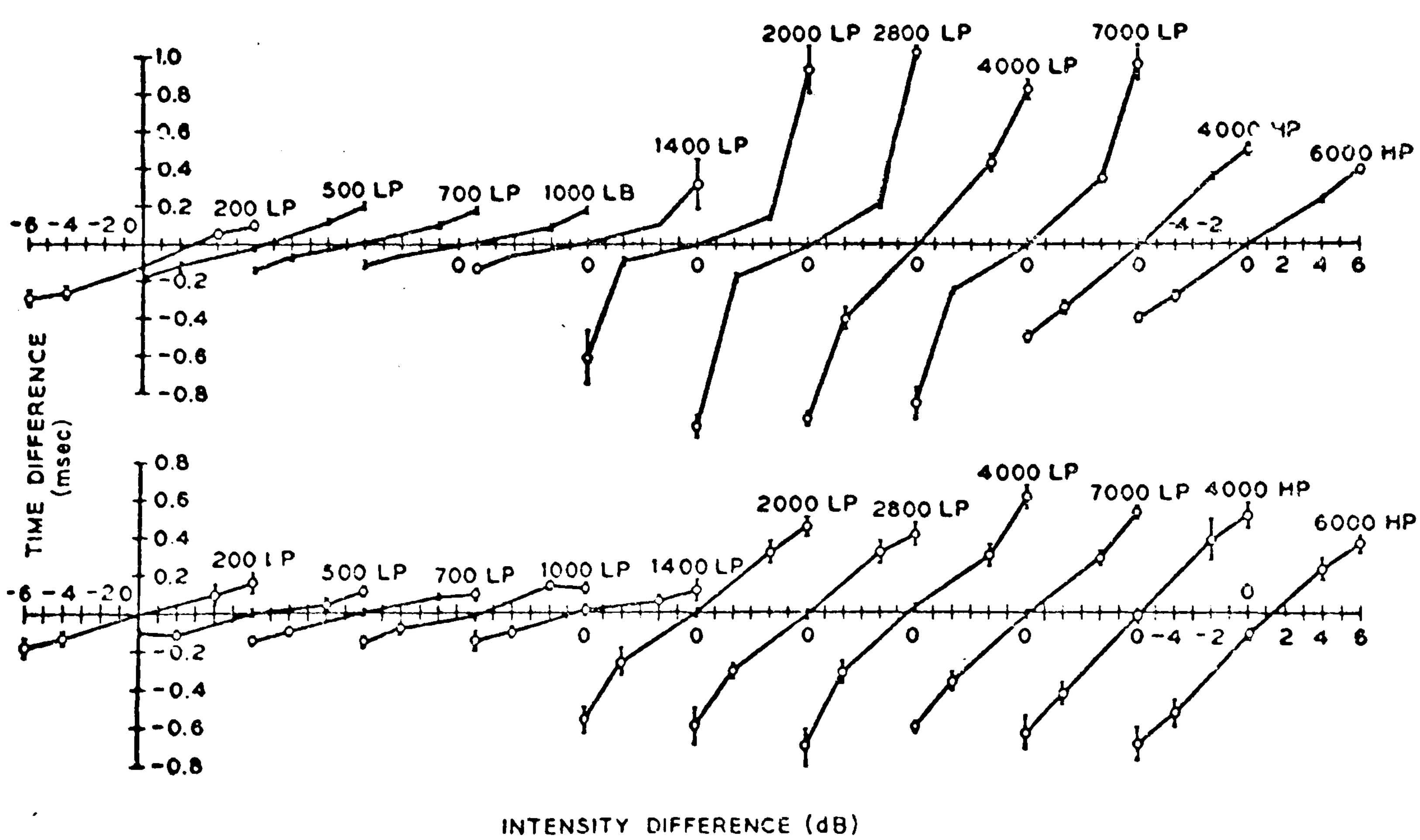


Figure 6.5 The interaural time difference required to null a given interaural intensity difference as a function of the intensity difference. The parameter is the cut off frequency of the filtered click. The data are plotted on a common abscissa, but the origin is different for each cut off frequency. The data are for two subjects at a 20 dB sensation level (Harris, ref. 43).

source being much smaller than that of a dichotic source. At about 1500 Hz the two functions converge on the same interaural intensity difference and remain together up to about 6000 Hz. From the equality of the actual and dichotic interaural intensity differences it appears that the minimum audible angle in this frequency range depends on the detection of interaural differences in intensity.

Fig. 6.4 also shows that there is a sharp divergence of the two intensity functions above 6000 Hz. The minimum audible angles for dichotic measurements at these high frequencies are much smaller than those actually observed. This discrepancy may indicate a third mechanism of auditory localisation at very high frequencies.

The results of the preceding experiments indicate that in the localisation of pure tones there is a shift from a temporal to an intensive mechanism at about 1500 Hz. One possible explanation for this is that at frequencies above about 1500 Hz the acoustical path between the ears is greater than one half wavelength. Any phase difference above this frequency will give possible source positions on both sides of the head, thus resulting in an ambiguous cue. Another explanation is that individual fibres of the auditory nerve cannot fire in synchrony with the stimulating tone unless it is below about 1500 Hz.

Interaural differences in the time of onset or time of arrival of the first wave of a tone pulse form a different type of temporal cue from ongoing interaural time differences. It is not subject to the phase ambiguities of steady tones and is effective at all frequencies since it is not limited by the size of the head or by the refractory periods of the auditory neurons. The information in an onset disparity lasts only until the sound has reached both ears. Interaural time differences giving rise to ongoing disparities, on the other hand, provide temporal cues as long as the sound lasts.

6.3 Time-Intensity Trading

In the localisation of actual sound sources, ongoing, transient, and intensive interaural disparities have different effects depending on the spectrum of the sound. Time-intensity trading experiments try to find out how the disparities interact by presenting them dichotically, and by varying one kind of disparity independent of the other.

One method of evaluating the relative effects on lateralisation of interaural time and intensity differences is to pit them against one another, trading the effect of an interaural intensity difference for the equal and opposite effect of an interaural time difference. A dichotic sound that is lateralised near one ear because it is louder there can be brought back to the centre of the head by delaying it relative to the sound in the other ear.

Harris[43] used this centring technique to investigate differences in time-intensity combinations at low and high frequencies. The stimuli were low-pass or high-pass filtered clicks. His results are shown graphically in Fig. 6.5; each point indicates the interaural time difference that is just sufficient to offset an opposite interaural intensity difference. At frequencies below about 1500 Hz, interaural time differences are more potent than at higher frequencies. The slope of these trading functions ($^{\Delta t}/\Delta I$) change at about that frequency.

The trading relation between time and intensity depends on the overall intensity of the two stimuli. As the sensation level is increased the importance of time difference increases relative to intensity difference which results in a systematic decrease of the trading constant. David et al[44] investigated this effect of the overall level on the trading constant using bursts of noise high-passed at

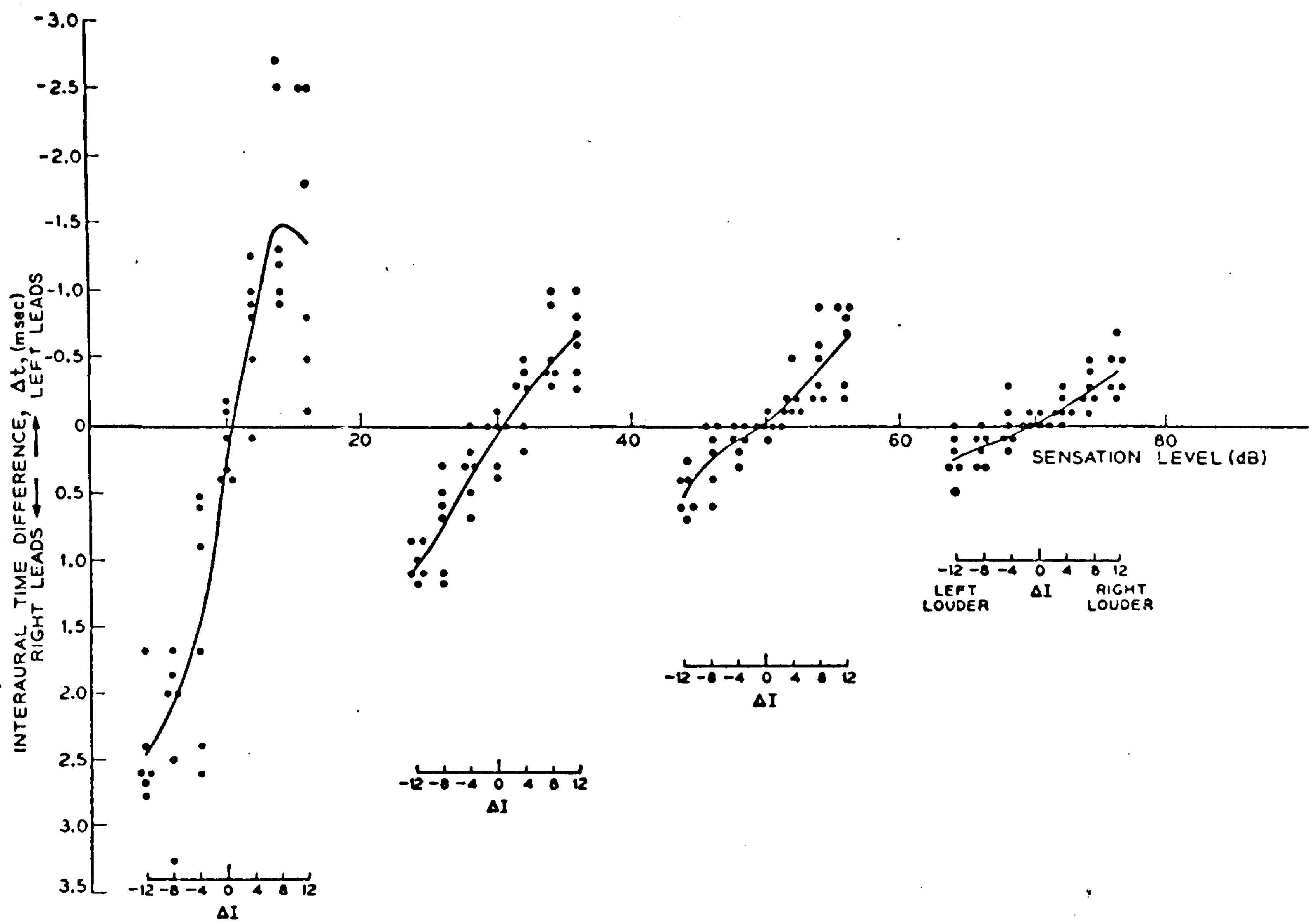


Figure 6.6 Interaural time-intensity trading functions for one listener at each of four levels of intensity. The interaural intensity differences (ΔI) are indicated on the sub-abscissas, and the interaural time difference that produces a centred sound image is shown on the ordinate for each ΔI (David et al, ref. 44).

2000 Hz; their results are shown in Fig. 6.6. The slopes of these trading functions vary from about 200μsec. per dB at 10dB S.L. to about 25μsec. per dB at 70 dB S.L.

The centring experiments described above are null measurements which require lateralisation only in the straight ahead position. To study the interaction of time and intensity differences more generally it is necessary to define a scale of subjective laterality.

Békésy[45] measured auditory laterality by requiring subjects to position an air jet producing a tactual sensation on the forehead such that the perceived direction of the tactual sensation matched that of the perceived dichotic sound. The results showed that generally the laterality of the image increases linearly with the interaural time difference. Mickunas[36] repeated Békésy's measurements using the method of constant stimuli and also found that, with small interaural time differences such as are produced by actual sound sources, the laterality of the image of a click varies linearly. The slope of the function obtained from a given listener is the same for equally intense dichotic clicks at 30 or 60 dB S.L., but varies considerably from listener to listener.

Moushegian and Jeffress[46] used a different method for measuring lateralisation based on a comparison within the auditory modality: a judgement of equilaterality. They used a broadband burst of noise as an auditory "pointer", this was presented dichotically at equal intensities but with various interaural time differences. In time-intensity trading experiments using the centring method, the direction of the interaural time difference is opposite or antilateral to the interaural intensity difference. A single actual source of sound produces combinations of interaural time and intensity differences that work in the same or colateral direction.

It would be interesting to know whether an interaural intensity difference combines colaterally with an interaural time difference to add the same amount of laterality to the sound image that it subtracts when it is combined antilaterally.

The Moushegian and Jeffress experiment included both antilateral and colateral combinations of time and intensity differences between dichotic tones. The interaural time differences used were similar to those occurring naturally but the interaural intensity differences were larger than life. The results were expressed as the interaural time difference of an equilateral pointer. For two listeners, the effects of antilateral intensity differences were opposite but not equal to the effects of the same intensity differences acting colaterally. When the interaural intensity difference was acting in the same direction as the interaural time difference, it had a greater effect on lateralisation than when time and intensity were opposed.

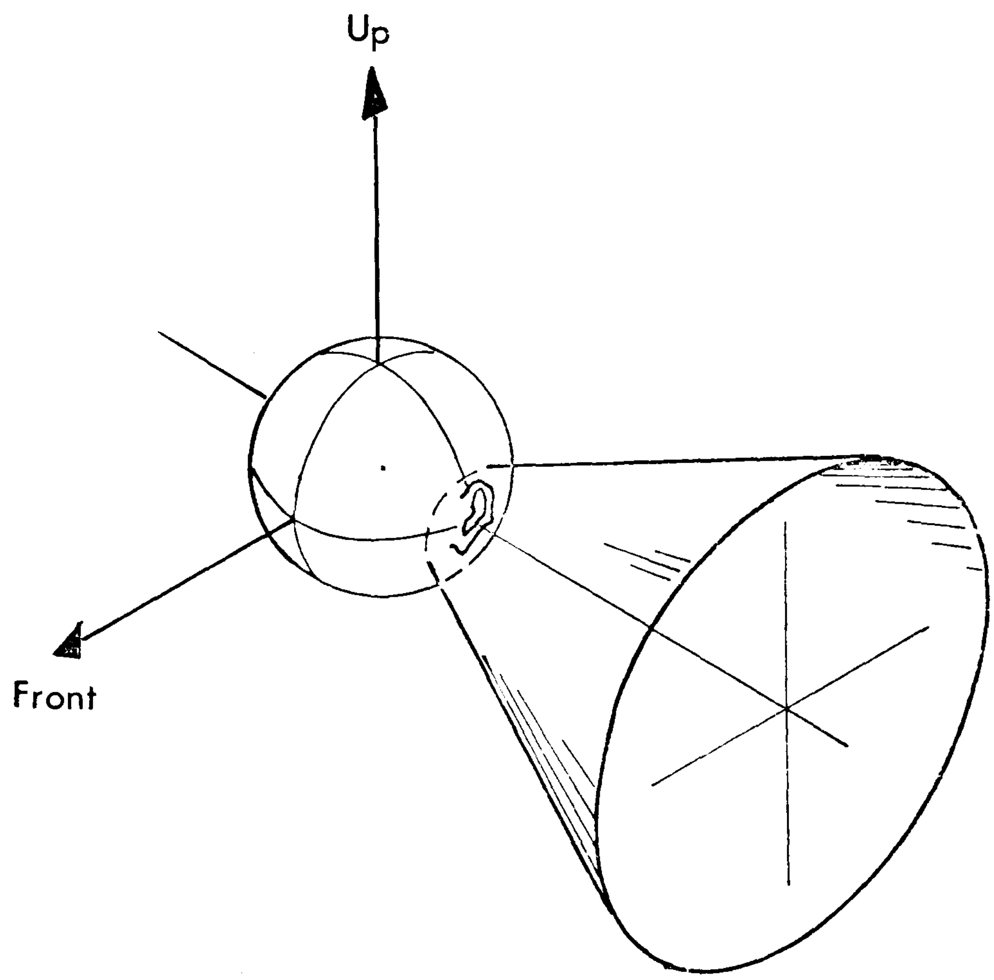
Their third listener indicated shifts of lateralisation for colateral and antilateral intensity differences that were nearly symmetrical. The listener also gave a time-intensity trading function which was more constant and less steep than the functions of the other listeners. It is commonly considered that the results of this experiment suggest that it is not possible to combine time and intensity by means of a single constant even at one frequency although Whitworth and Jeffress[47] dispute this for listeners with normal hearing. Mills[36] has summed up by saying that if it is appropriate to relate interaural differences in time and intensity by linear weighing constants, then there are at least two, and perhaps three constants: colateral, antilateral, and equilateral.

These experiments with dichotic signals have given an indication of how interaural time and intensity differences interact in lateralisation, but sufficient data is not available to form quantitative predictions of the localisation of actual sources.

6.4 The Outer Ear

A pair of stationary holes separated by a sphere is not sufficient to uniquely define the position of an actual sound source unless head movement is allowed. The concept of a stationary sphere contains spatial ambiguity commonly called the "cone of confusion". This is illustrated in Fig. 6.7. A sound source that produces a given interaural time difference may lie anywhere on the conical surface shown; the intensity difference indicating only on which side of the head the source is located. Thus, without head movement, the ambiguity is unresolved. However, it has been clearly shown[48] that a complex sound is localised without head movement and can be located monaurally.[49] This means that another mechanism must exist to provide the resolution, and to explain other observations such as monaural localisation.

Batteau[50] has shown that the pinnae alone are sufficient to distinguish between sounds from the front and the back, when these sounds contain high frequency components. He fitted a pair of microphones with artificial pinnae without attempting to simulate a head in between them. His subject, seated in an adjoining room listening through electrostatic earphones, reported the apparent position of a maraca shaken at various azimuth positions. With the pinnae present the listener made fairly accurate judgements of azimuth; with bare microphones, however, his results were unsure and erratic. The elevation was judged almost as accurately as the azimuth when pinnae were present. For these effects to occur, the sounds presented to



Illustrating the

Figure 6.7 A cone of confusion for a spherical head. The surface of the cone is the locus of sources producing the same interaural time difference.

the listener must contain components of sufficiently high frequency, and thus short wavelength, to interact strongly with the pinna.

When the microphones are fitted with artificial pinnae, listeners localise the sound image at a distance from the head, instead of inside the head as dichotic sounds are normally perceived. This impression persists even if one microphone is disconnected. With this single pinna-microphone system a moving source of high frequency transients is perceived as an external source that moves roughly in the same directions as the actual source. This effect is not produced with a bare microphone. The pinna evidently transforms the sound entering the ear canal in a way that indicates the direction of the source and, to some extent, its proximity. It would seem that this transformation is in the intensity-frequency domain. This still does not fully explain, however, how it is possible to localise an unfamiliar sound monaurally without head movement.

Batteau proposed an additional action of the pinna: a transformation of the incoming signal in the amplitude-time domain by means of echoes from the surface of the pinna. These echoes would vary in relative delay as the angle of incidence of the wavefront varies.

This theory may be able to explain the erratic changes found in the minimum audible angle at high frequencies. Above about 8000 Hz, the period of a tone approaches the length of the azimuth delay postulated by Batteau, and the phase of the delayed wave specifies not a single delay, but a set of equivalent delays. In measuring the minimum audible angle at these high frequencies, a listener makes judgements that are consistently wrong, as might be expected if the reflected wave led by $\pi + \phi$ and thus also lagged by $\pi - \phi$.

An objection raised to Batteau's theory expressed doubt that such short time delays could be interpretable by humans. This was recently resolved by Wright et al.,[51] whose experiments showed that humans have sufficient ability to detect the very short time delays in question.

6.5 Perception of Multiple Objects

In a free-field listening situation, where there are usually many complex sounds occurring simultaneously, the listener is able to switch his attention at will to any one of them. This is often called the "cocktail party problem" and arises when a person selectively listens to one voice among many without becoming confused.[52] The task is most easily performed when two ears are used and the voices to be isolated are separated both spacially and temporally.

However, in an environment of multiple echoes, such as a reverberant enclosure, the listener's perception of direction is pre-empted by the sound that reaches the ears first, i.e. by the most direct path. The secondary echoes can be perceived as a change in quality of the sound but their influence on localisation is largely suppressed. This is known as the precedence effect and has been demonstrated by Wallach et al[53] under both dichotic and free-field listening conditions. With dichotic stimuli two successive binaural clicks are heard as a unit if the second pair follows within a few milliseconds of the first; the azimuthal image direction is dominated by the interaural differences in the first binaural click. In a field of multiple sound sources only the nearest will be perceived, and the perception of direction will be dominated by the nearest source.

The precedence effect suggests that the use of pulse signals in the display of an echolocation device might result in only the nearest object being perceived since only the first echo would be taken into account. However, with the particular range coding employed in the proposed device this problem has not arisen.

In experiments designed to select the most discriminable set of clicking rates for the range coding of the device, Crawford and Rudlin[13] exposed listeners to two different pulse rates simultaneously, each fed from a separate equidistant loudspeaker. In later experiments with this arrangement the listeners were trained to recognise the five different rates that were available. The results of these tests indicated that the subjects could selectively listen to either of the two rates, describe which particular two were present, and attribute them to the appropriate loudspeaker. Thus, the use of this type of display in an echolocation device appears to be suitable for the perception of multiple objects. There are occasions, however, when the precedence effect may occur, but this was deliberately prevented in these tests by interposing a suitable time delay between the channels.

In earlier tests the subjects had not been trained to recognise the different rates, and it was clear that they were far less successful in achieving correct results. This means that they could give a much better performance after they had acquired a prior knowledge of the temporal patterns of the different sounds. This type of prior knowledge is thought to be essential in selective listening tasks such as in the "cocktail party problem".[52] Crawford et al. also found during the experiments that if an error was made in judging one of the rates and the other rate was subsequently muted, the misjudged rate was easily corrected.

Although the experiments only presented two rates from two different directions, the results gave a good indication of what is experienced with the display of a real device. Here, an experienced listener is able to switch his attention from one rate of interest to another and also recognise each particular rate in question. It is possible that differences in amplitude existing between pulse trains from different objects makes the discrimination easier, as Crawford's experiment suggests. There may also be differences between the echoes from objects due to the different reflection characteristics of various surfaces. The existence of these differences may possibly prevent the precedence effect from occurring on those occasions where it might normally operate. The differences may also hopefully facilitate the discrimination of multiple objects for a listener who has learned the different available clicking rates.

6.6 Externalisation

It has been noted earlier that sounds presented to a listener via earphones are heard as if inside the head instead of "out there" in the environment. The sounds produce an internal image which can be located to a point along a left-right line connecting the ears; this task is often referred to as "lateralisation".

Jeffress and Taylor[54] investigated this effect by studying the accuracy with which subjects could assign an azimuth position to a sound coming to them over earphones. Interaural time differences were used to simulate the various azimuth positions. Their subjects did about as well initially as Stevens and Newman's[39] subjects did with an external source, and they showed a small amount of improvement with practice. The conclusion reached was that the task of localising sounds coming to the ears via earphones is essentially the same as localising an external sound. Occasionally some of the subjects managed to externalise the sound, but this was not generally the case and did not seem to improve with training. These infrequent externalisations have been recently disputed by Plenge,[55] who has pointed out that their later research indicates that outside-head location did not occur.

Sayers and Cherry[56] have examined the physical differences that exist at the ear drums in binaural listening. They noted that, as well as interaural time and intensity differences, there are interaural (short-term) spectral differences due mainly to (a) diffraction by the head, (b) reflective properties of the environment, and (c) different angles of incidence of wave fronts impinging on the ear orifice and consequent impedance mismatching. They added that further clues arise from head movement and an acquired knowledge of acoustic properties of "typical situations" as well as the whole sensory integration faculty. They stressed that interaural time and intensity differences, by themselves, are not sufficient to externalise a sound, other factors are necessary to provide outside-head localisation.

It has been shown by the experiments of Kock,[57] amongst many others, that clues arising from head movements enable a sound image to be perceived externally provided that the head movements are combined with concomitant changes in the physical differences at the eardrums. This, however, does not fully solve the problem. If these concomitant changes due to head movements are prevented whilst listening to a distant sound source in free space, a listener should perceive a sound image inside the head. The experiments of Toole[58] and others have shown that this does not occur.

Batteau's experiment, noted earlier, clearly showed that a sound presented via earphones may be externalised if it is initially coded by pinnae. This result was more recently reinforced by Plenge[55] who derived signals from microphones situated in carefully constructed dummy heads. Plenge called these externalised signals "ear-adequate" and investigated the possibility of converting a "non-ear-adequate" signal to an "ear-adequate" signal by adding the extra clues outlined by Sayers and Cherry. A positive result in this experiment, he claimed, would solve the problem of the difference between localisation and lateralisation.

Plenge, using electronic techniques, was not able to completely simulate an "ear-adequate" signal from a "non-ear-adequate" monophonic signal but managed to always maintain outside-head localisation. He concluded that it is not important whether a sound is conveyed by earphones or not, but that the only condition necessary for localisation is that the sound signals have to be such as would originate from an external source; otherwise, lateralisation occurs. He argued that this supported the presumption of Sayers and Cherry that true external projection may be set up by interaural (short-term) spectral differences.

In other experiments with dummy heads Plenge found that when listening to a sound source in the median plane, the sound image is initially located inside the head and, after a short time of exposure, it becomes located outside the head. Plenge explained this by proposing an adaptation period where knowledge is acquired of the acoustic properties of typical situations. This corresponds to the second presumption of Sayers and Cherry. He finally concluded that in the localisation of a sound source, the subject does not merely use the sound signals perceived at a given moment, but also makes a comparison with stored stimulus patterns.

In a model of localisation, Plenge postulated the existence of a long-term store filled with data about sound events learned since the head reached full size. The data is recalled automatically whenever a hearing event occurs. To clear this long-term storage and slowly fill it again may take several days as was shown in an experiment by Held.[33]

The long-term store may not be sufficient for correct localisation. This particularly applies to distance perception which depends on familiarity with the volume and timbre of the source. A short-term storage of such knowledge of sound sources would be useful. Lateralisation, then, is likely to occur if (1) the short-term storage has missing or deficient knowledge of sound sources and sound field, and/or if (2) the signals cannot be related to any of the stimulus patterns stored in the long-term storage.

The bearing of this discussion on the auditory display of the echolocation device is that it is desirable for the signals to give a natural sounding representation of the outside world. The echoes should preferably sound like natural audible frequency echoes and, ideally, objects should appear to be emitting the sounds themselves. For this impression to occur, externalisation of the sounds is essential. However, Rudlin[16] concluded, as a result of his work on pinna transformations, that if it is necessary to impress pinna coding on the echo to effect externalisation, then the practical realisation of this at ultrasonic frequencies would not be straightforward.

Rudlin looked for an alternative method to achieve externalisation by investigating whether certain frequency components in sounds were more important than others in producing the effect. He concluded that it is useful for sounds to have a high frequency content but added that externalisation is a personal sensation which varies considerably between subjects and, indeed, for one subject at different times. This latter part of his conclusion appears to be at variance with the observations of Plenge, who stated that once a signal was "ear-adequate" all his subjects localised it outside the head. The two statements, however, are not necessarily incompatible since the sounds that Rudlin was using were not particularly "ear-adequate".

The author carried out simple tests with an experimental mobility device incorporating the "free-field" earphones and modified, wide-beam, receiving transducers described earlier. He found that inexperienced listeners could externalise echoes from a nearby wall providing large head movements were allowed. The listener was initially asked to face the wall and when the device was switched on he perceived an image inside the head. When he rotated his head the image became externalised and remained so even when the head was returned to the straight-ahead position.

This effect is not the complete answer to externalisation, as mentioned earlier, but it is interesting to note the congruence of this result with the observations of Kock and others. If we consider that in the simple tests the initial effect was an in-head localisation, we may parallel this situation with that obtained by Plenge when listening to a median plane source via a dummy head. In his case the image became externalised after a short adaptation period, whereas in ours it did not until head movement was allowed. If we are to concur with Plenge's storage hypothesis, then this effect might suggest that our long-term storage is deficient in the stimulus patterns to which we are listening, thus producing in-head localisation.

It may be that if we fill our long-term storage with a knowledge of the new stimuli, then we will be able to achieve externalisation. The short-term storage, which is largely concerned with distance perception, is not as important since the mobility device incorporates its own distance coding. However, the preceding suggestion has only considered a sound on the median plane. For other azimuth positions, the receiving transducers of the mobility device are unable to impress the pinna transformations, described earlier, on the incoming echo. It remains to be seen if listeners, who have been given a sufficiently long adaptation period, are able to acquire sufficient knowledge in their long-term store to override this shortcoming and thus always achieve outside-head localisation.

CHAPTER 7

A COMPARISON BETWEEN TWO METHODS OF RESPONSE FOR AUDITORY LOCALISATION IN THE AZIMUTH PLANE

7.1 Introduction

When many of the system designs described in earlier chapters had been incorporated into an experimental device, a need arose to measure objectively the accuracy with which a user could localise objects in the azimuth place. The results of such measurements would indicate the effectiveness of the designs used, and would enable further adjustments to be made if systematic errors occurred. The preceding chapter has described the various methods of indication used by experimenters in auditory localisation. Since we are concerned with an objective measurement of the azimuthal image direction, methods such as centring and matching are unsuitable. Further, as blind subjects may be used at some stage, methods which depend on vision, such as the use of light as a 'pointer', are also unsuitable.

Two methods which do commend themselves are the one described by von Békésy[45] and straightforward pointing. Von Békésy's apparatus, shown in Fig. 7.1, consists of a head band carrying a protractor and an air-pipe; subjects experience a tactual sensation from a jet of air on the forehead and are required to position the air jet such that the perceived direction of the tactual sensation matches that of the perceived sound. The use of pointing, on the other hand, is more straightforward but has been objected to since it involves an intermodal transfer of information from auditory to kinesthetic space. This objection may be particularly valid in the case of some blind people, especially those congenitally blinded, who have no means of knowing where their arms are pointing to. However, as both methods have been used in the design of binaural information displays for the blind[34,59] it was thought useful to carry out a comparison study. Thus an experiment was designed in which pointing and, let us say, blowing, were compared as methods of indicating direction. The experiment was initially carried out using free field sounds, and was then repeated for sounds presented through earphones. This is the stimulus originally

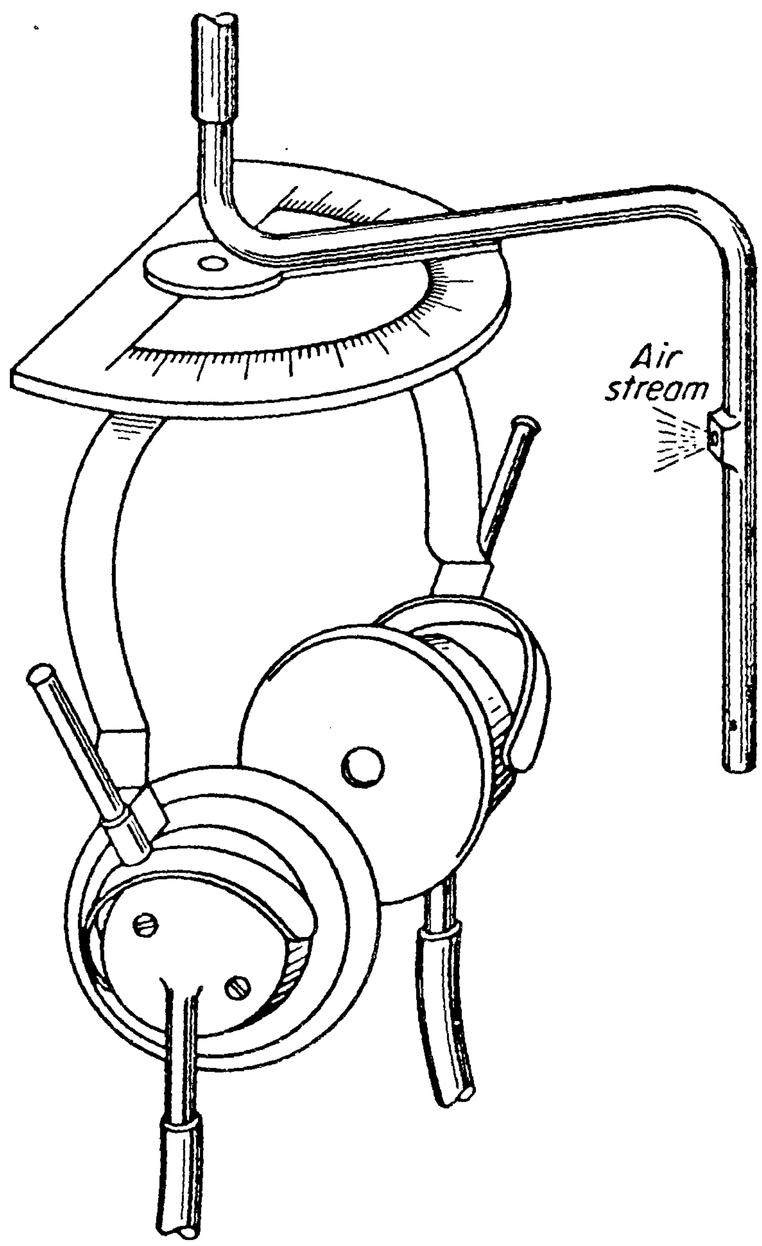


Figure 7.1 Von Békésy blowing method of indication (from von Békésy, 1960).

used by Békésy, and also occurs in the auditory display of the mobility device.

In a pilot study the results obtained with von Békésy's method were radically different from those obtained using pointing. Consequently the study was prolonged so that this anomaly could be investigated and reported.[60]

7.2 Experiment A. Sounds presented in the free field

7.2.1 Method

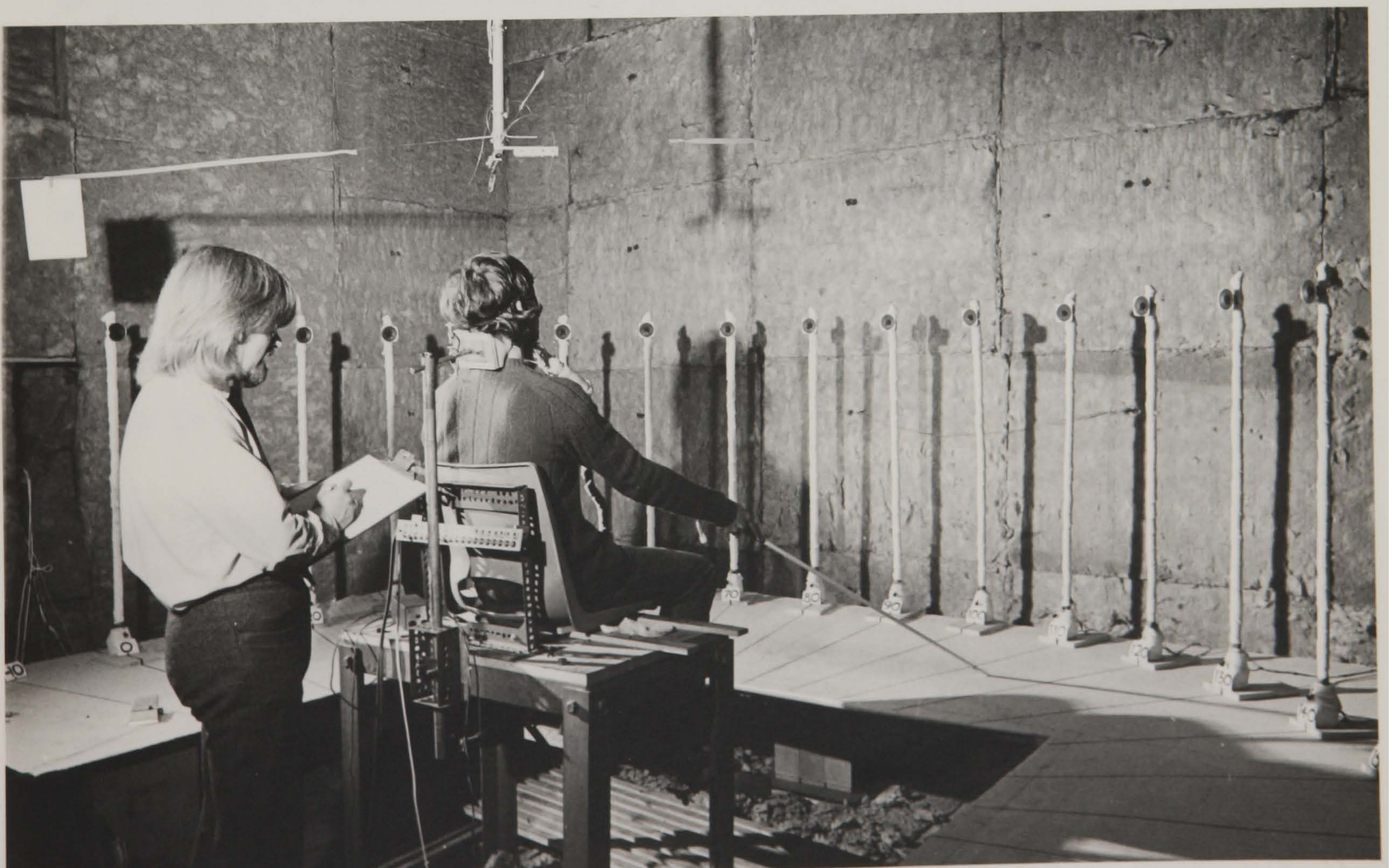
The subjects, who were all sighted volunteers with normal hearing, were blindfolded and seated with their heads clamped into position at the centre of an arc of loudspeakers in an anechoic chamber. The position of the hairline on the subjects restricted the blowing method to responses 60 degrees to either side of the medial plane, and because of this the whole experiment was confined to this region. Thirteen matched loudspeakers were arranged at 10 degree intervals to make up the arc, the distance from the subject to the speakers being 2 metres. Typical experimental arrangements for both methods of indication are shown in Figs. 7.2(a) and (b).

For the pointing experiment subjects were provided with a long cane, the tip of which rested on a low table on which was drawn a protractor. They were instructed to point using the cane as an extension of their arm and to choose to point with the left or right arm depending upon the side from which the stimulus was perceived.

The stimuli consisted of eight second bursts of clicks, the clicking rate being sixteen per second. The loudspeakers were addressed in random order, there being a five second gap between stimuli. An experimental session consisted of the randomised presentation of each of the thirteen loudspeakers four times, making a total of 52 stimuli.

(a) The von Békésy Method

Figure 7.2 Two methods of Response for Auditory Localisation in the Azimuth plane



(b) Straightforward Pointing

An experimental design was chosen in which each subject performed the task twice, once using pointing as the method of indication and once using blowing. Four subjects were used, two pointing first and two blowing first. A short preliminary session was given prior to the first test session in order to familiarise the subject with the experimental conditions.

In the blowing experiment, von Békésy's specifications were closely followed in that the measurements were made by centring the protractor, not at the centre of the interaural line, but rather at the physical centre of the head. Von Békésy had deduced the position of this centre to be 45.0 ± 10 mm forward of the mid-point of the interaural line.

Subjects' data was collapsed and the results for the two methods of indication are shown in Figs. 7.3(a) and (b). Each point on the graph represents the mean of 16 estimations of the direction of each of the speakers in the azimuth, viz. 4 presentations to each of the 4 subjects; the ideal response is included for comparison. The error bar, of length equal to the standard deviation of the mean, presents a measure of consistency. An overall measure of performance was calculated by collapsing the 16 response errors at each of the 13 angles and calculating an overall root mean square error value (R.M.S.) which is also shown on the graph.

7.2.2 Discussion of Experiment A

Fig. 7.3(a) shows pointing to be a good method of indication for free field sounds; accuracy and consistency are both high. Fig. 7.3(b) shows that blowing also offers a method of indication having a high consistency, but accuracy appears to be very low. However, the graph 7.3(b) is approximately a straight line passing through the origin. A similar graph but with a different slope would

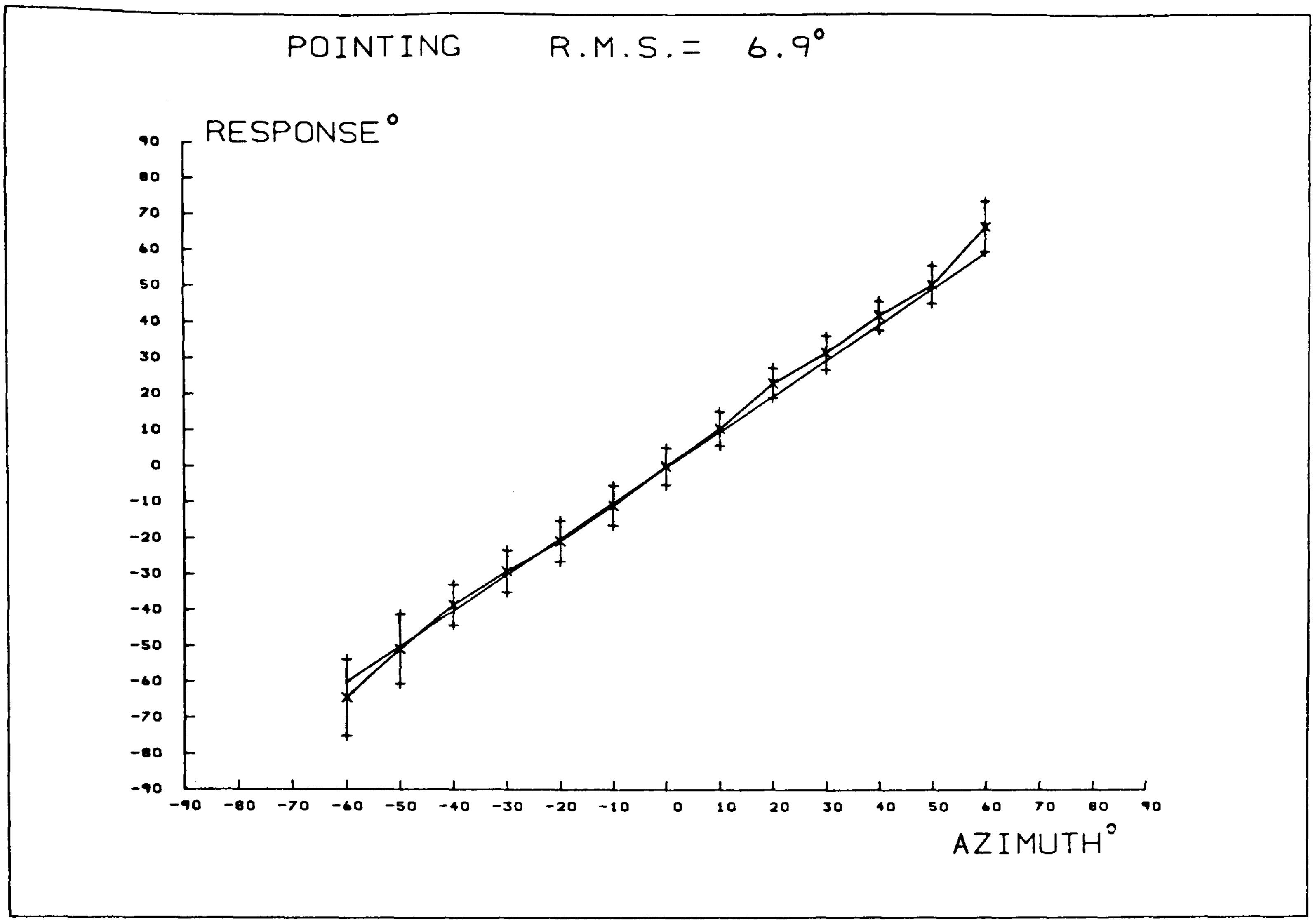


Figure 7.3(a) The localisation of free field sounds using pointing

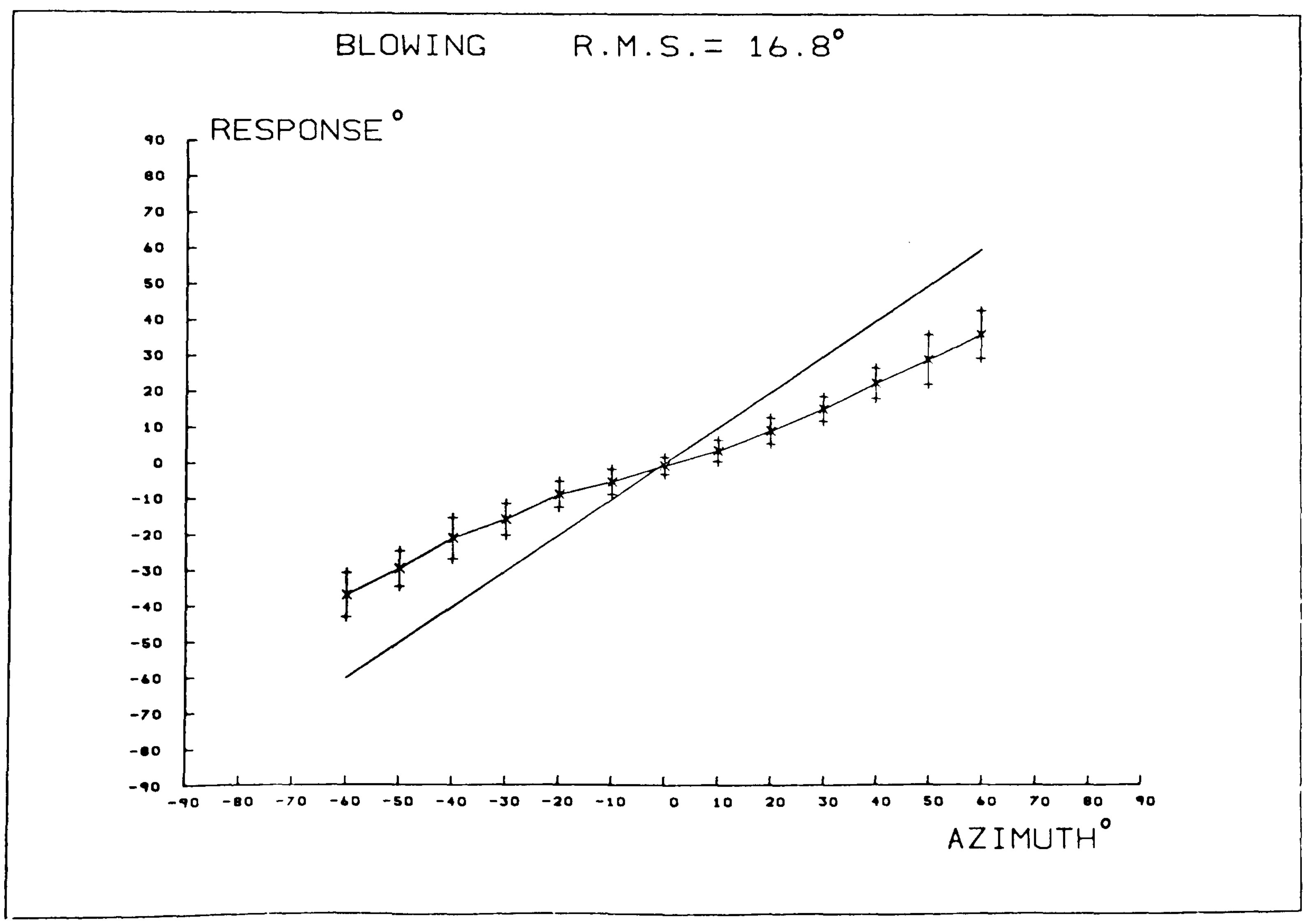


Figure 7.3(b) The localisation of free field sounds using the blowing method of von Békésy

have been obtained if another centre of measurement had been used. For instance, a forward shift of the centre would result in an increase in the slope of the line.

Fig. 7.4 shows O as the centre of measurement (the physical centre of the head chosen by von Békésy), with N as a new centre some distance in front of O. P is the point on the forehead where the stimulus is felt. Θ_1 is an angle recorded in the experiment and Θ_2 is a new angle obtained by calculating the effects of a centre shift. The 13 mean values of Θ_1 obtained in the experiment were used together with the known stimulus directions, Θ_2, to calculate a population of values for x from which a mean value was obtained equal to 45.6 ± 15 mm. The value used for r, the head radius, was 90 mm. The results from the blowing experiment, when recalculated using this value of x, are in good agreement with those obtained using pointing.

Using modified apparatus, with the pivot for the air pipe and the centre of the protractor mounted 45 mm forward of the physical centre of the head (i.e. 90 mm forward of the interaural line), the experiment was repeated with six new subjects. Fig. 7.5 shows the results obtained for the blowing experiment using the modified apparatus. The relocation of the centre point appears to be justified.

7.3 Experiment B. Sounds presented through earphones

7.3.1 Method

A high quality stereophonic tape recorder (Revox, Stereo Mod. G 36) was used, in conjunction with a pair of capacitor microphones (Brüel and Kjaer ½") mounted in the ear canals of an artificial head with pinnae, to record stimuli similar to those used in Experiment A. The artificial head was placed at the centre of the arc of loudspeakers during the recording to enable the experimenters to know the direction

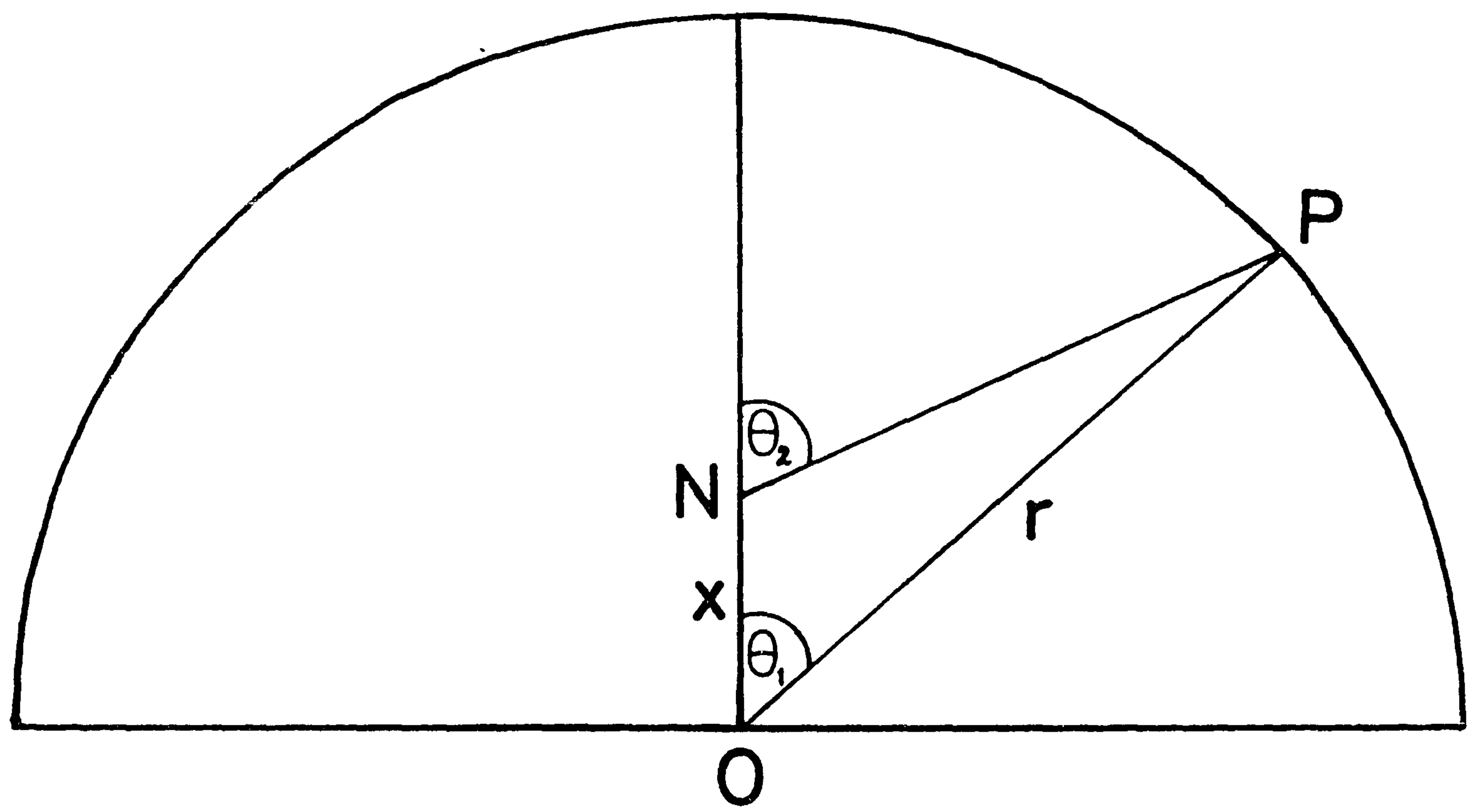


Figure 7.4 Showing how the azimuth deviation of the stimulus P is a function of the centre of measurement.

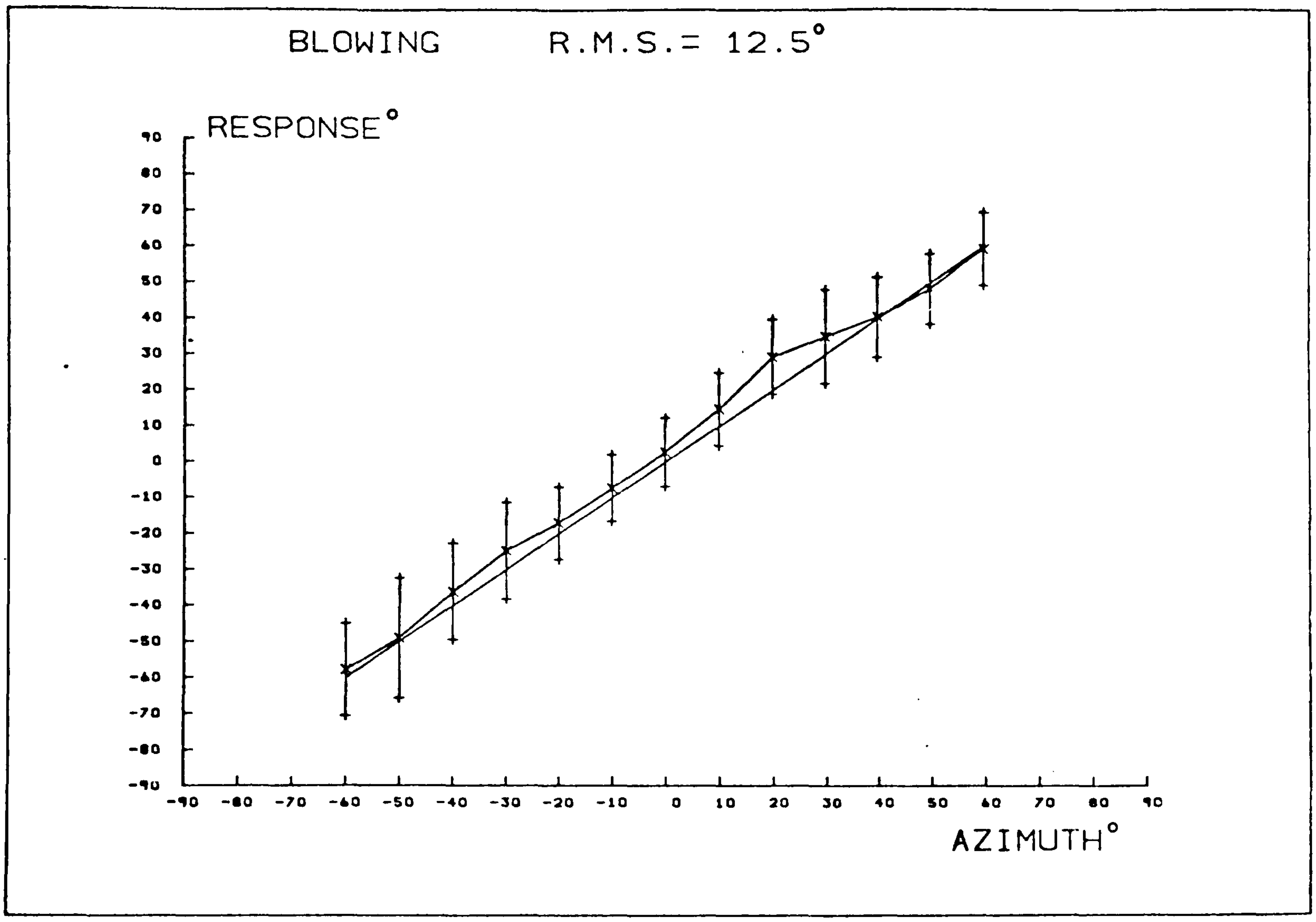


Figure 7.5 The results of the repeated free field localisation experiment using the new centre of measurement

of correct response of the recorded stimuli for earphone presentation. The recordings were subsequently played back, using electrostatic stereophonic earphones (Stax type SRD-5) to the same six subjects who had performed the repeat of Experiment A.

The subjects were required to respond by either pointing or blowing exactly in the same manner as previously. The pivot centre for the air pipe, and the centre of the protractor, were again concurrent and at 90 mm forward of the middle of the interaural line (45 mm forward of the physical centre of the head).

7.3.2 Results

The stimulus response curves for the two types of response, shown in Figs. 7.6(a) and 7.6(b) are similar to the ones obtained under free field conditions. Once again the use of the modified apparatus resulted in good agreement between the two methods of indication.

During the experiment the subjects perceived the stimulus as being out in the environment; thus outside-head localisation occurred rather than lateralisation.

7.4 General Discussion

The graphs show that, provided the corrected centre of reference is used in the blowing experiment, the results obtained are equivalent to those using pointing. Indeed data from both methods compares favourably with the results of other workers. For example, Stevens and Newman[39] reported an average error of localisation of 8.0 degrees when using free field click sounds. The pointing and blowing results for free field sounds in our experiments gave average errors of 3.7 and 6.8 degrees respectively. For earphone presented sounds, Jeffress and Taylor,[54] who used lamps as indicators of position, obtained a

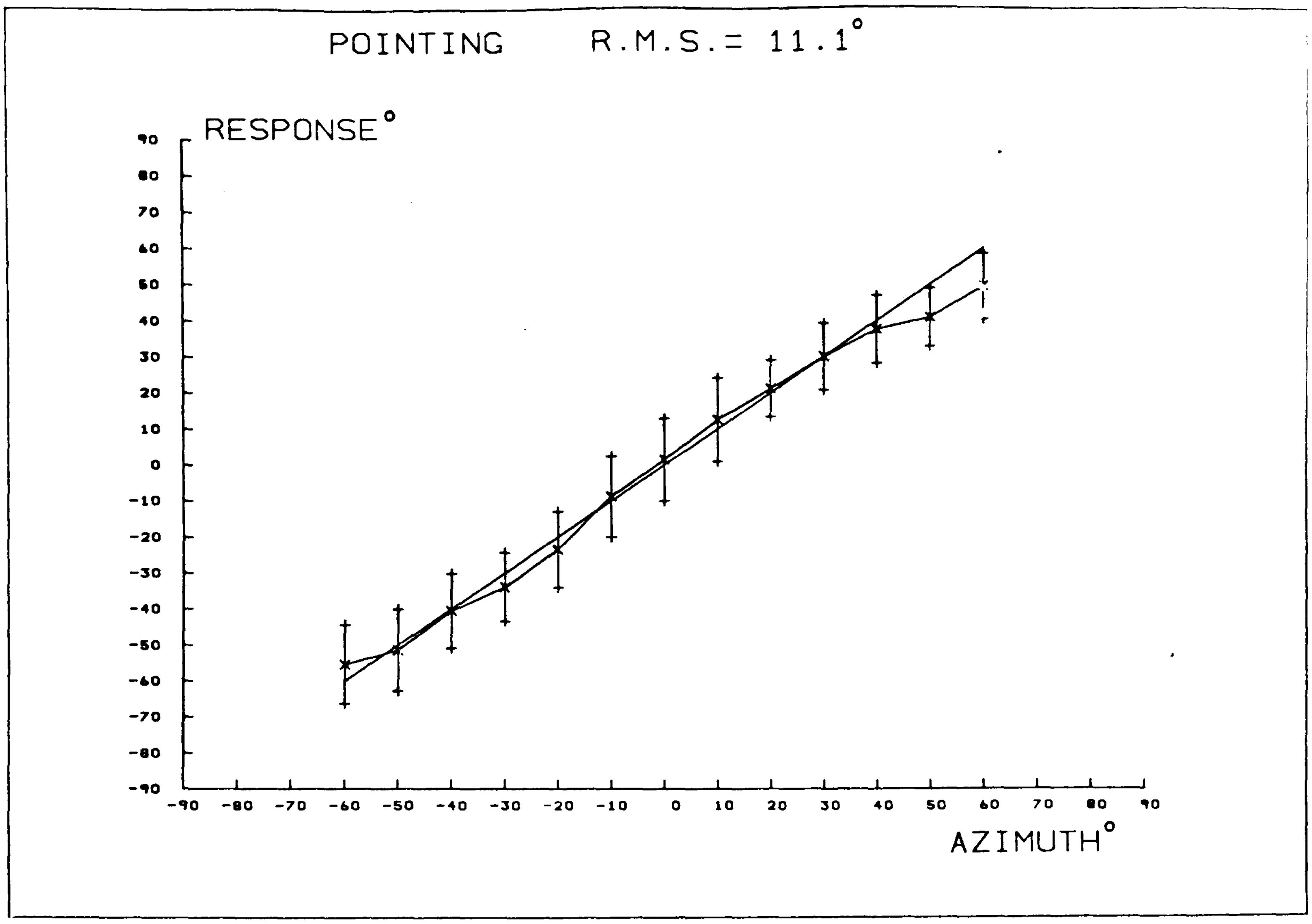


Figure 7.6(a) The localisation of dichotic sounds using pointing

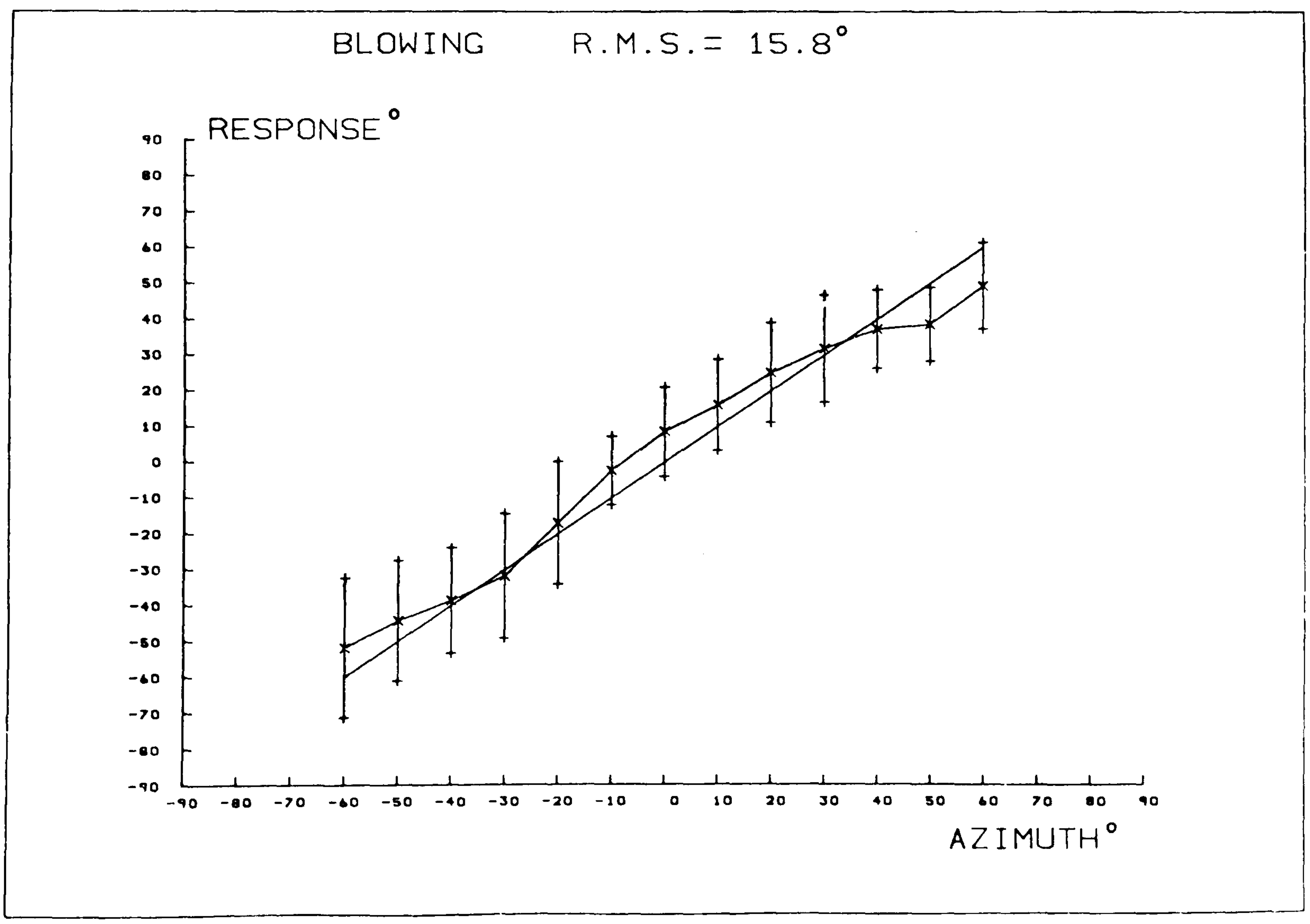


Figure 7.6(b) The localisation of dichotic sounds using the blowing method with a new centre of measurement

value of 12.8 degrees for the standard error (the r.m.s. error computed around the "correct" response). Our results using the pointing and blowing methods gave standard errors of 11.1 and 15.8 degrees respectively. Furthermore, the equivalence of the results between free field and earphone presented sounds, and the fact that the centre for measurement for blowing remains the same in both cases, reinforces Jeffress and Taylor's conclusions that the two types of stimuli can be judged relative to azimuth rather similarly.

Mickunas,[36] who also carried out experiments using von Békésy's method, found that the slope of the function he obtained varied considerably from listener to listener. However, in this series of experiments, the inter-subject variability was found to be low.

In comparing the two methods of indication in terms of both absolute error and consistency, as measured by the standard deviations, the pointing method is significantly superior. (Wilcoxon T test < 0.001). Nevertheless, the blowing method would be a reasonable choice if the experiment were to be performed in a confined space. Furthermore, it might prove essential to use it in the case of blind people who show a poor ability to point.

One difficulty, however, was experienced with the blowing method. Namely, the use of a pivot for the air pipe which was so far removed from the physical centre of the head (45 mm) meant that the nozzle of the jet was in general not perpendicular to the forehead. This gave a rather diffuse tactual sensation. This could easily be avoided if, instead of making the pivot for the air pipe concurrent with the centre of measurement, the pivot and protractor are centred at the physical centre of the head and each measurement is transposed to the appropriate centre. The formula for such a transposition would simply be:

$$\Theta_{new} = \tan^{-1}\left[\frac{r \sin \Theta_{old}}{r \cos \Theta_{old} - x}\right]$$

Where Θ_{new} and Θ_{old} are the response angles, r is the radius of the head and x the distance from the point of measurement to the new reference point.

A further point worth noting relates to the impression of outside-head location that the subjects had during the sessions with earphone presentations. These results agree closely with the findings of Plenge[55] who also made a stereophonic tape recording of sounds derived from microphones located in a dummy head with pinnae. He found that the sounds could always be perceived extracranially when replayed to a listener through earphones. These observations add further weight to the idea that pinna coding is very important in achieving outside-head localisation.

Finally, this series of experiments has given an indication of how accurately we can position actual sources of sound and additionally, how accurately we can localise sounds derived from a carefully constructed artificial system. We may use this information as a yardstick in measuring the localisation information afforded by an experimental mobility device.

7.5 Conclusions

For sighted subjects under blindfold, both pointing and the method devised by von Békésy using an air jet on the forehead are suitable non-visual methods of indication in auditory localisation experiments in the azimuth plane. However, the point chosen by von Békésy as the centre of reference for his method is not correct. Rather, the measurement should be referred to a point 90 mm forward of the mid-point on the interaural line. For some blind subjects only the method of von Békésy may be appropriate.

CHAPTER 8

AUDITORY LOCALISATION USING AN EXPERIMENTAL MOBILITY AID

8.1 Introduction

The experiments described in this chapter were designed to measure quantitatively the accuracy with which a listener, using an experimental device, could localise objects in the azimuth plane. The positioning of the receiving transducers in the device was variable, thus enabling different combinations of interaural information to be obtained. It was intended to find a receiving arrangement which gave the most accurate results, and also to compare these results with those obtained in other auditory localisation experiments.

Initially, it was thought that the greatest accuracy of localisation would be obtained by selecting a combination of interaural time and intensity information to resemble, as closely as possible, that occurring in the human auditory system. Thus a main feature of the original experimental design was the inclusion of different combinations of interaural time and intensity differences, in an attempt to find the most appropriate mixture.

The interaural time information was obtained by spacing the receiving transducers 18 cm apart. This separation gives propagation path length differences which can be computed by referring to the geometry in Fig. 8.1. The path length difference, Δd, arising from a distant object at azimuth Θ is given by

$$\Delta d = 2r \sin \Theta$$

where 2r is the transducer separation and Θ is the angle measured from the median plane of the head. By taking the velocity of sound, c, as 343 metres per sec., the interaural time difference, t, is given by

$$\Delta t_{\mu sec} = \frac{\Delta d}{c} = 526 \sin \Theta$$

which is in close agreement up to Θ = 60 degrees with actual measurements of interaural time differences made by Feddersen et al.[38] A comparison

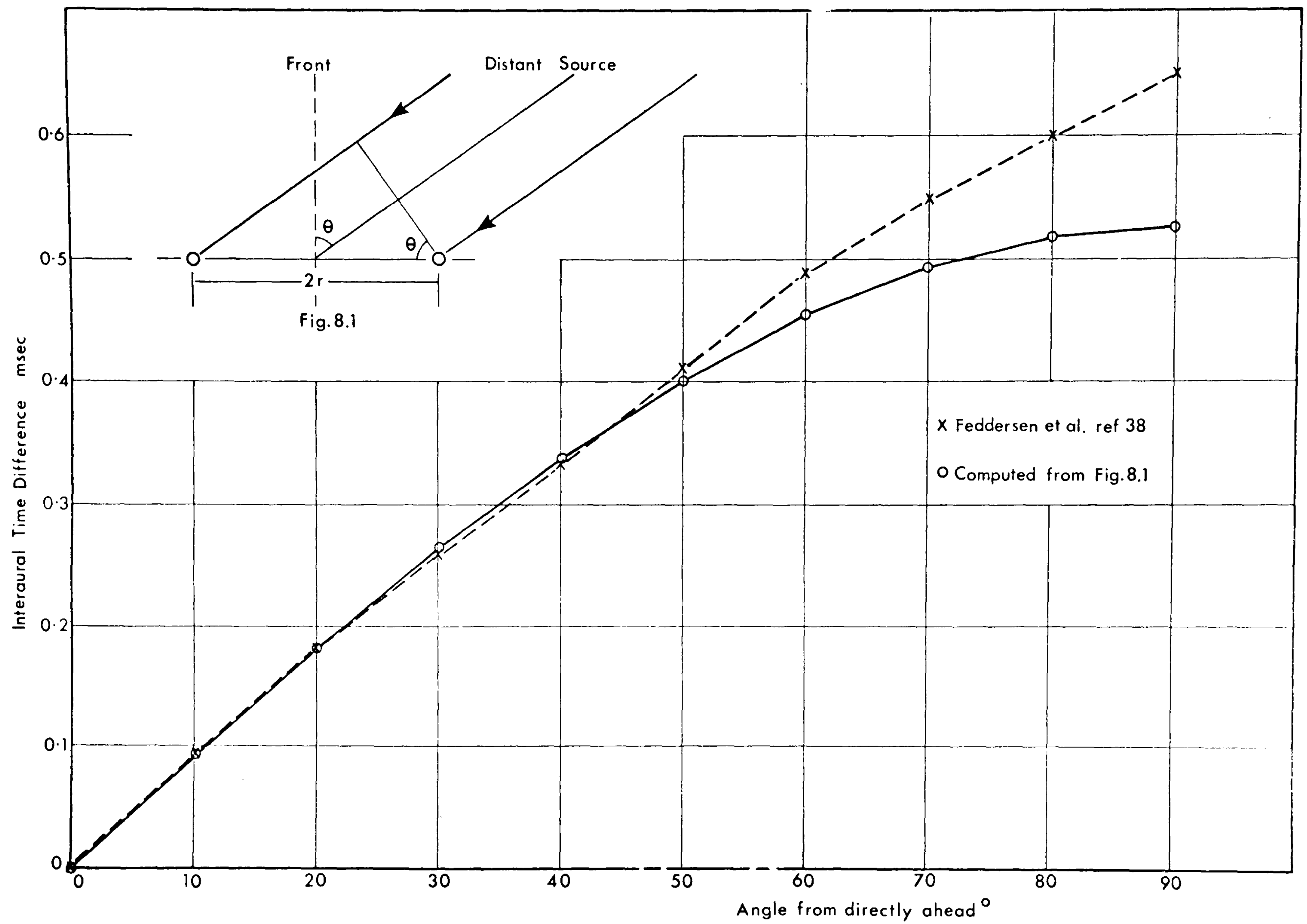


Figure 8.1 Differences between the distances of the receiving transducers from a distant sound source

Figure 8.2 Interaural time differences as a function of the direction of the source of clicks is made in Fig. 8.2. Owing to this close correspondence it was thought best to leave the time information fixed, since there seemed little scope for improving the match.

The interaural intensity information depends upon the directional response of the receiving transducers and the angle at which they are splayed outwards from the median plane. It was suggested in Chapter 5 that the modified broad-beam receiving transducers should be splayed outwards by 20 to 30 degrees in order to obtain interaural amplitude differences that resemble those occurring in the natural auditory system in the frequency range 2.5 - 4 kHz. Furthermore, it was predicted that at a splay angle of 30 degrees the interaural amplitude information would be most accurate in a sector 40 degrees to either side of the median plane, the information outside this area being somewhat less accurate. The initial experiments, therefore, were designed to test whether the predicted optimal arrangement of the receiving transducers did give the best results.

The assumption that localisation with the mobility device is most accurate when both time and intensity information are present was based on conclusions reached by Rudlin.[16] He had carried out a series of experiments in which he assessed localisation ability with time information alone, intensity information alone and also a combination of time and intensity information. The experimental arrangement consisted of a pair of matched microphones placed in an anechoic duct, at the far end of which a loudspeaker emitted a series of clicks. The microphones were chosen to have directional characteristics which resembled those of the human ear. The relative and absolute orientations of the microphones were adjustable, so as to isolate the required auditory parameter, and to enable the apparent source of sound to emanate from any desired angle. The outputs of the microphones were fed, via a stereophonic amplifier, to headphones worn by the subject. The subject's task was to indicate the direction from which the sound appeared to be coming.

The results showed that the mean angle estimation had a standard deviation of 9.2 degrees for intensity differences alone, 7.8 degrees for time differences alone, and 7.3 degrees for time and intensity differences. These results were taken to indicate that the combination of time and intensity information gave a better idea of direction than either time or intensity alone. Additionally, a comparison was made of the subjects' estimated azimuth of the sound source with its actual orientation. The graphs for the three conditions are shown in Figs. 8.3(a) - (c). For intensity information alone (Fig. 8.3(a)) it is clear that the estimated azimuth does not correspond closely to the true angle and the wide scattering of points confirms the large standard deviation obtained. The time information graph (Fig. 8.3(b)) approximates much more closely to the ideal 45 degree line, but still exhibits considerable scattering. Fig. 8.3(c) for time and intensity information shows less variation between estimates and a better approximation to a linear relationship, but the slope tends to be rather steeper than the ideal slope. Finally, it was stated that the subjects had found the task of localisation to be easier when both time and intensity information were present.

In the present experiments, however, it became evident that results were being obtained which were at variance with Rudlin's conclusions. The original experimental design, therefore, was expanded to include an investigation into the localisation information contained in arrangements using time differences alone and intensity differences alone as well as combinations of time and intensity differences.

Intensity Difference

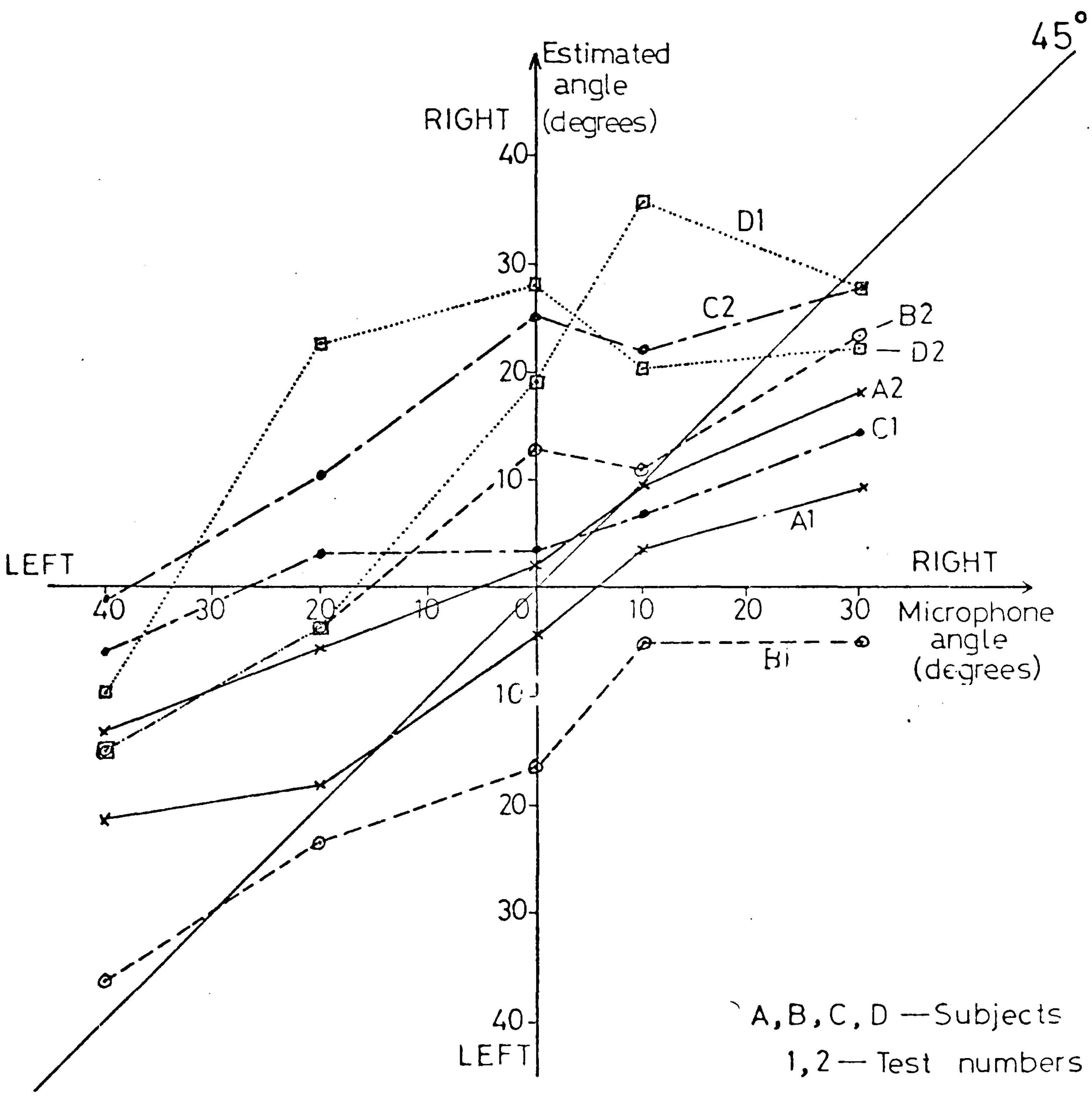


Figure 8.3(a) Mean estimated azimuths of sound source (1) (After Rudlin)

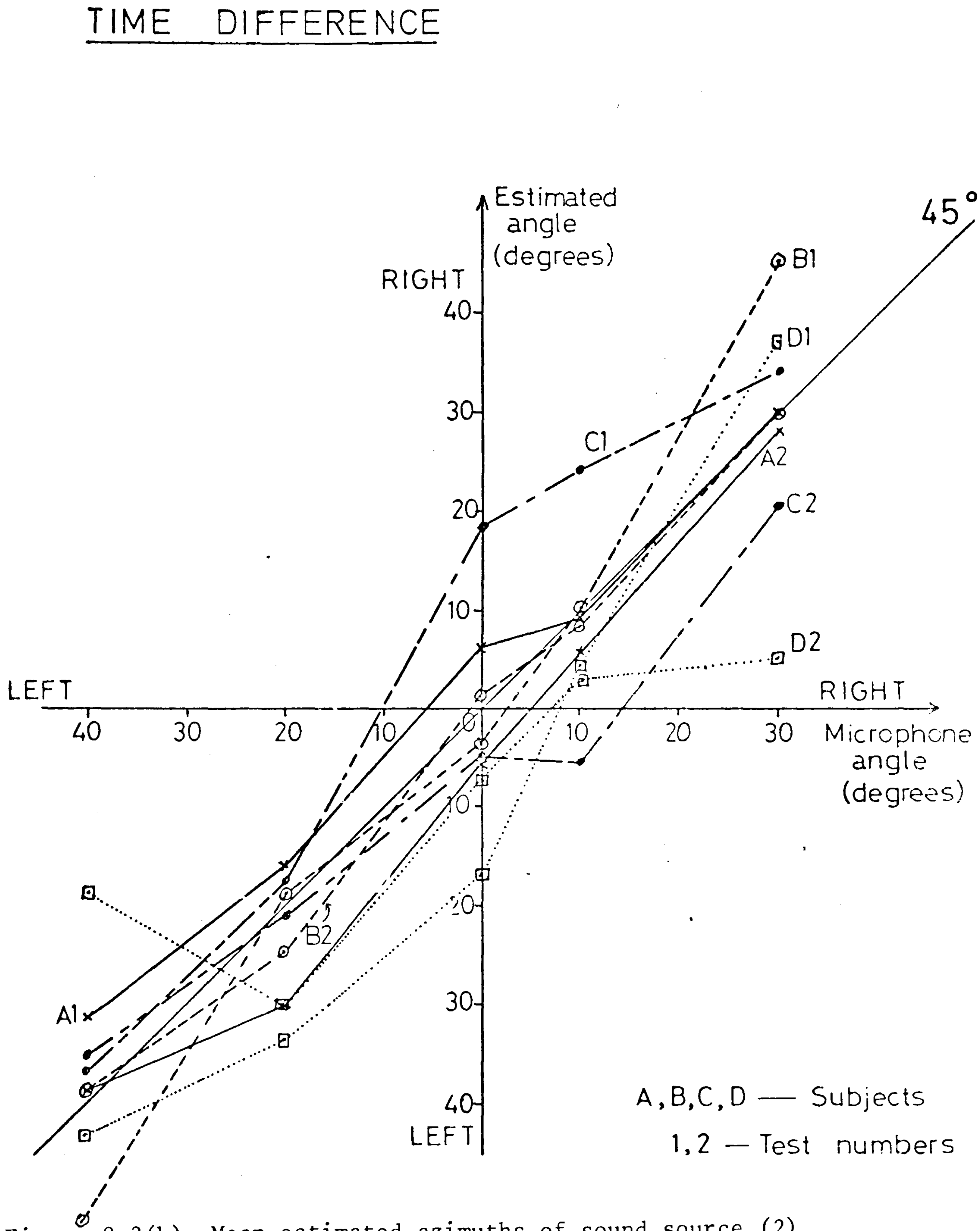


Figure 8.3(b) Mean estimated azimuths of sound source (2)
(After Rudlin)

Time and intensity differences

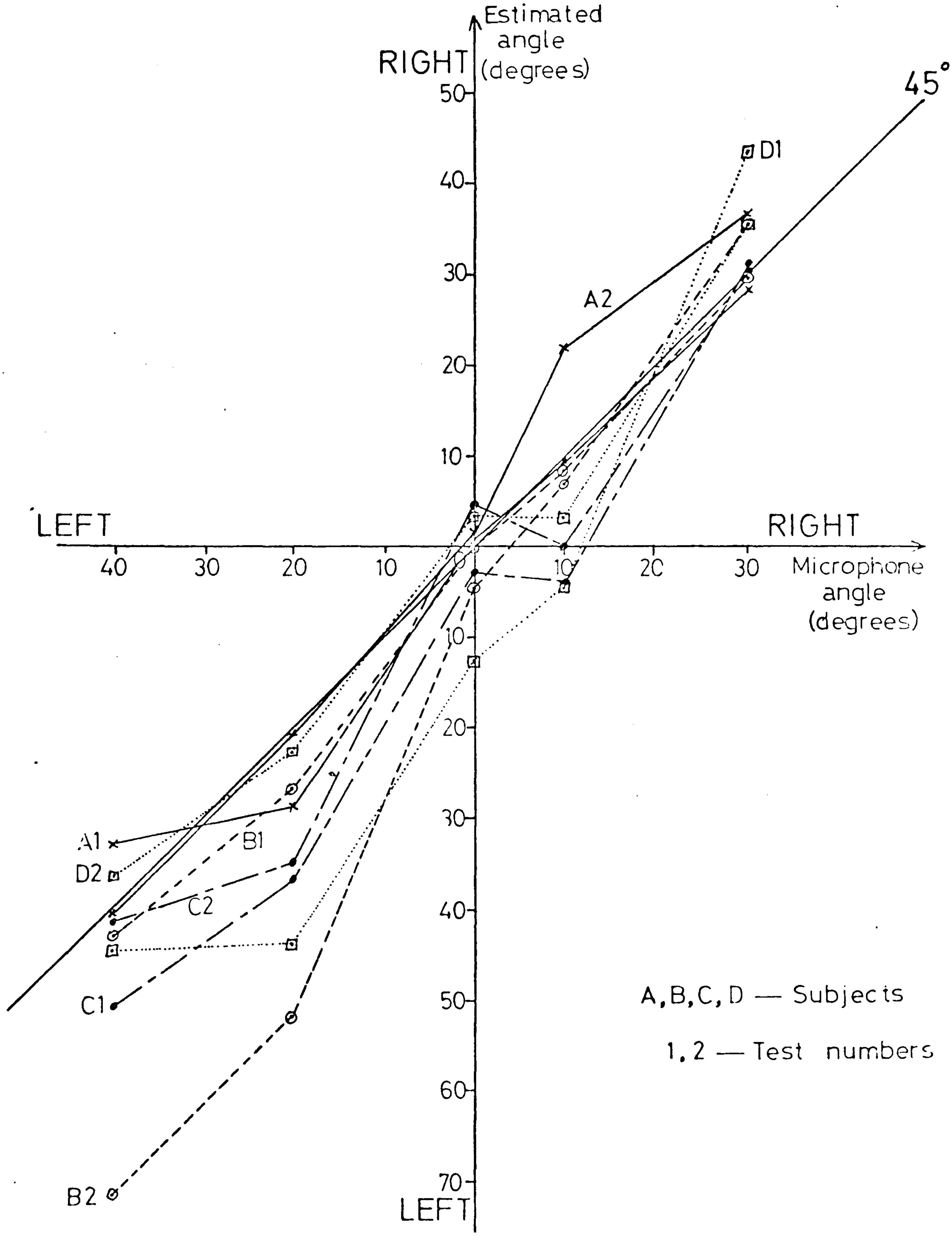


Figure 8.3(c) Mean estimated azimuths of sound source (3) (After Rudlin)

8.2 Apparatus

The previous chapter indicated that pointing and the method of von Békésy are both suitable methods of indication in auditory localisation experiments with sighted subjects under blindfold. In the experiments described in this chapter the pointing method was adopted, since it is rather more straightforward in use.

All the experiments were performed out-of-doors in order to minimise unwanted reflections. A typical experimental arrangement is shown in Fig. 8.4. The subjects, who were all normally hearing sighted volunteers, were blindfolded and seated with their heads fixed into position at the centre of a large protractor marked out on a square of canvas. The object to be detected was chosen to be a long cylindrical pole, 8.5 cm in diameter, since this is the type of obstacle frequently encountered by blind travellers. The pole was moved along the circumference of the protractor, a distance of about 1½ metres from the subject, and placed at any of the 10 degree intervals marked out.

The spectacle frames and earphones of the device were fitted onto a subject's head and the remainder of the electronic apparatus, including an "on-off" switch, was placed at his side. The transmitting torch was positioned above the head and slightly forward in order to decrease echoes coming from the body; as an additional precaution a piece of absorbent foam material was placed behind the transmitting transducers.

8.3 Procedure

The subject was provided with a cane and instructed to point using the cane as an extension of his arm and to choose to point with the left or right arm depending upon the side from which the stimulus was perceived. At the beginning of an experimental session the subject

Figure 8.4 Auditory localisation using an experimental mobility aid

was asked to point straight ahead. Then, with the pole in the straight ahead position, the device was switched on and the subject was asked to point to the perceived image. The two indications were normally coincident but when they were not the error could be corrected by minor adjustments in the positioning of the apparatus.

An experimental session consisted of placing the pole at each of the thirteen positions four times in a random order thus making 52 positionings in all. At each positioning the subject was asked to switch on the device and point in the direction of the perceived image. He was subsequently asked to switch off the device and return the pointing cane to the straight ahead position. A source of white noise was switched on from a cassette tape recorder to distract the subject's attention while the pole was moved to a new position. Each test session was preceded by a short preliminary session in order to familiarise the subjects with the experimental conditions.

8.4 Results

Four subjects were used in each group of sessions, there being five groups in all. A group consisted of providing the subjects with either interaural time information, interaural intensity information, or a mixture of interaural time and intensity information; the different conditions being achieved by various configurations of the receiving transducers. In the first group a combination of both time and intensity information was obtained by using the arrangement shown in Fig. 8.5; the receiving transducers are spaced at interaural distance apart and splayed outwards from the median plane by 30 degrees. The results obtained are shown in Fig. 8.6. Each point on the graph represents the mean of 16 estimations and the length of the error bar is equal to the standard deviation of the mean. An overall measure of

Figure 8.5 Configuration of receiving transducers to provide interaural time and intensity information.

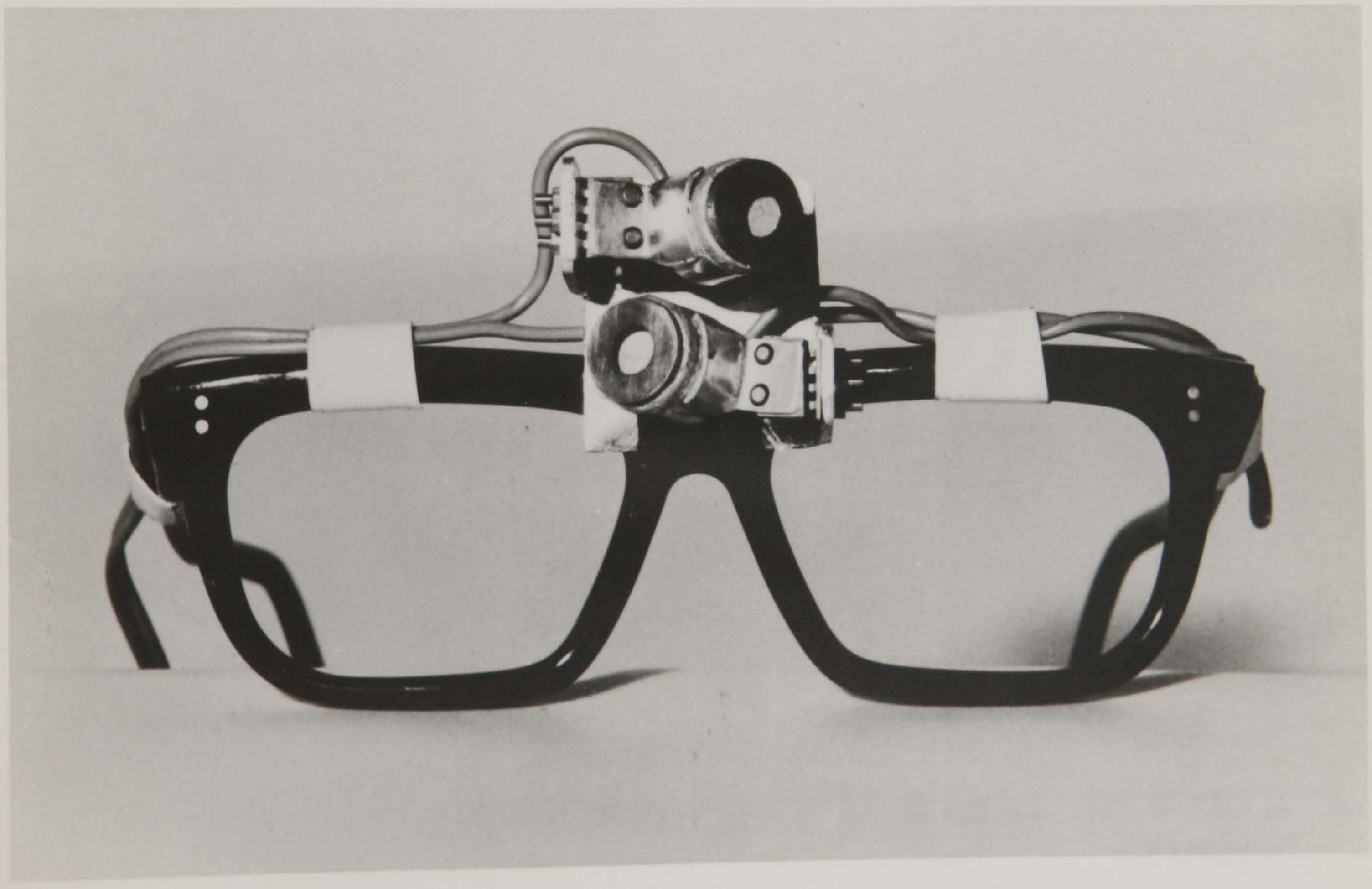


Figure 8.9 Configuration of receiving transducers to provide only interaural intensity information.

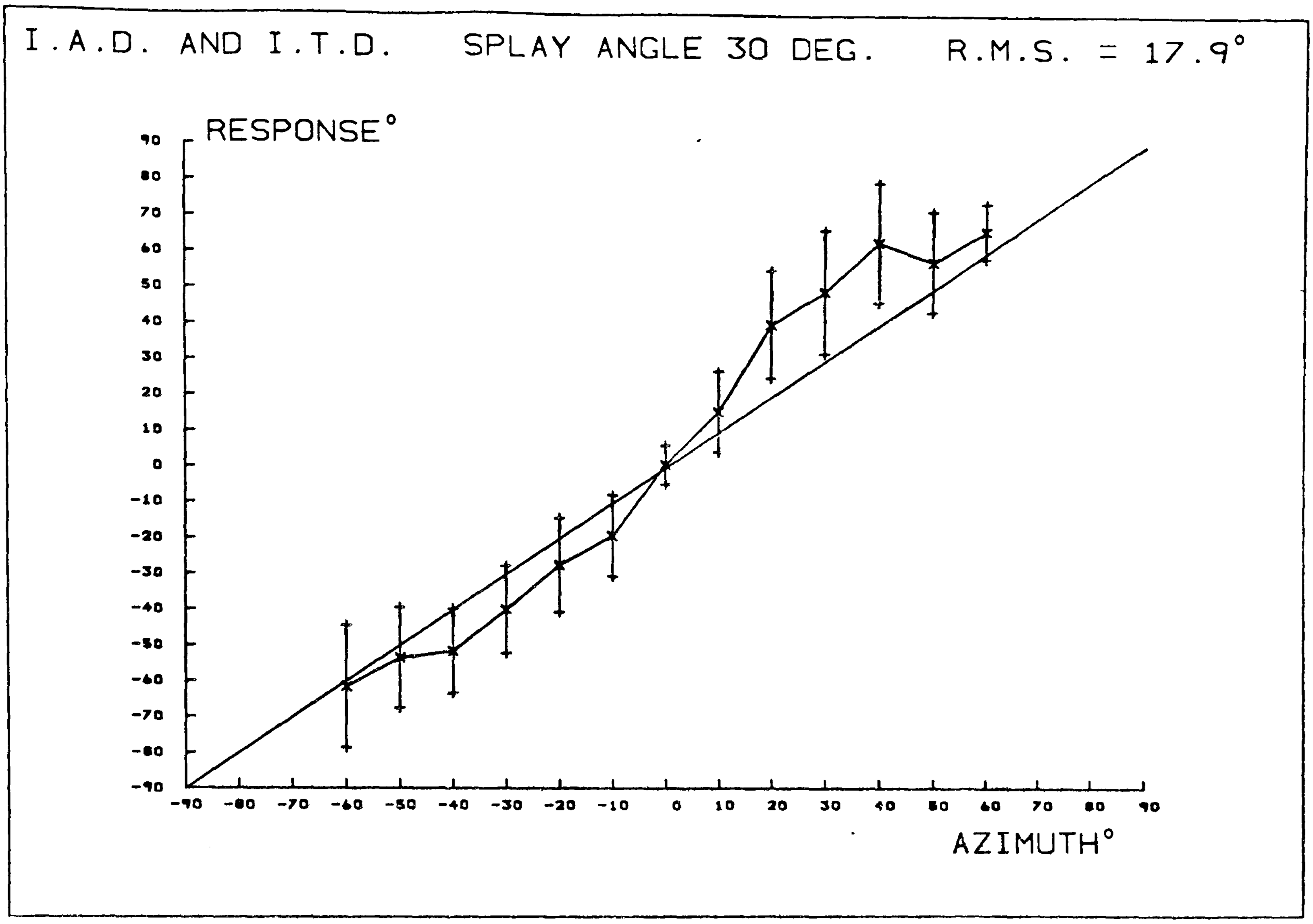


Figure 8.6

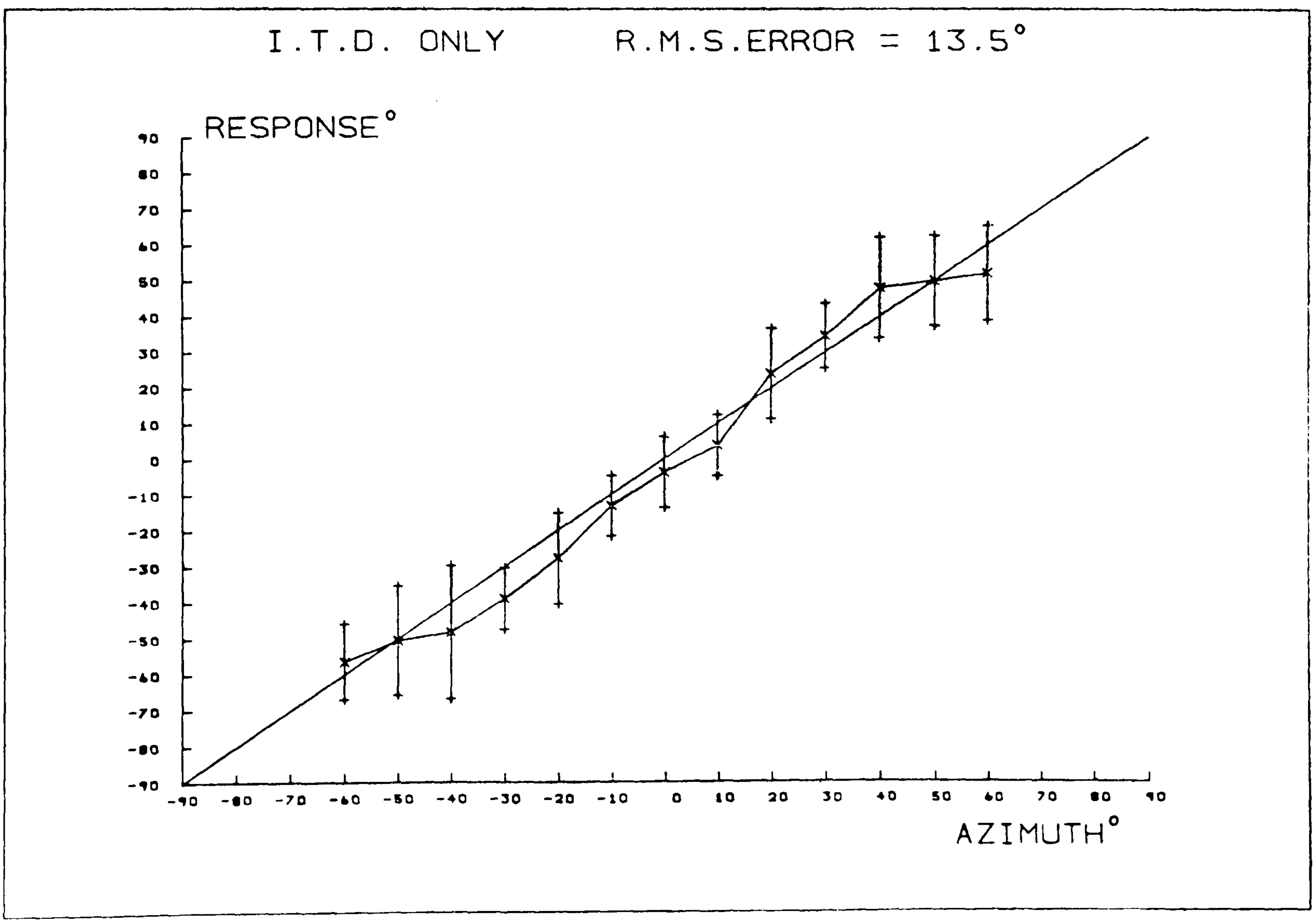


Figure 8.7

Localisation using an experimental mobility aid

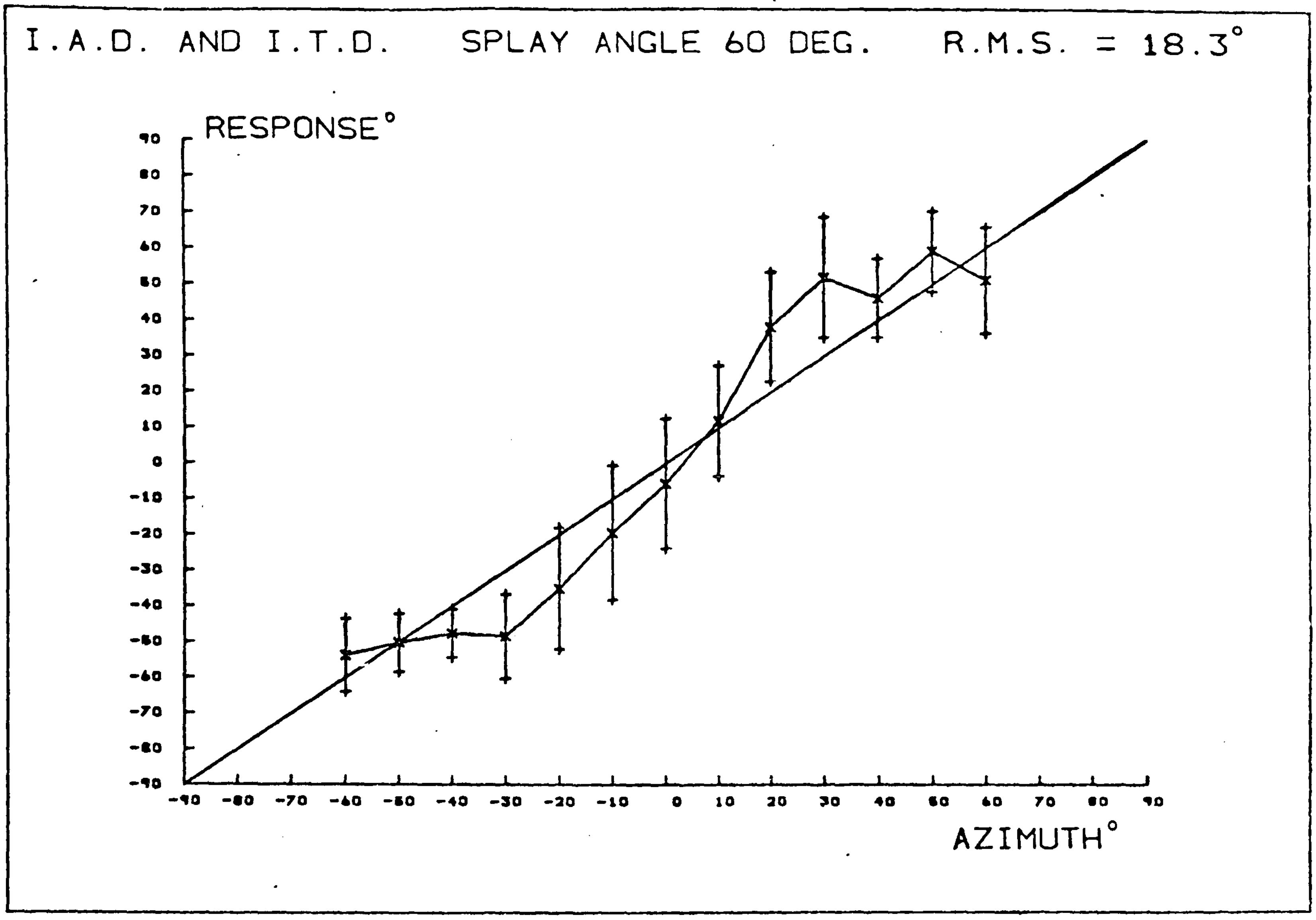


Figure 8.8

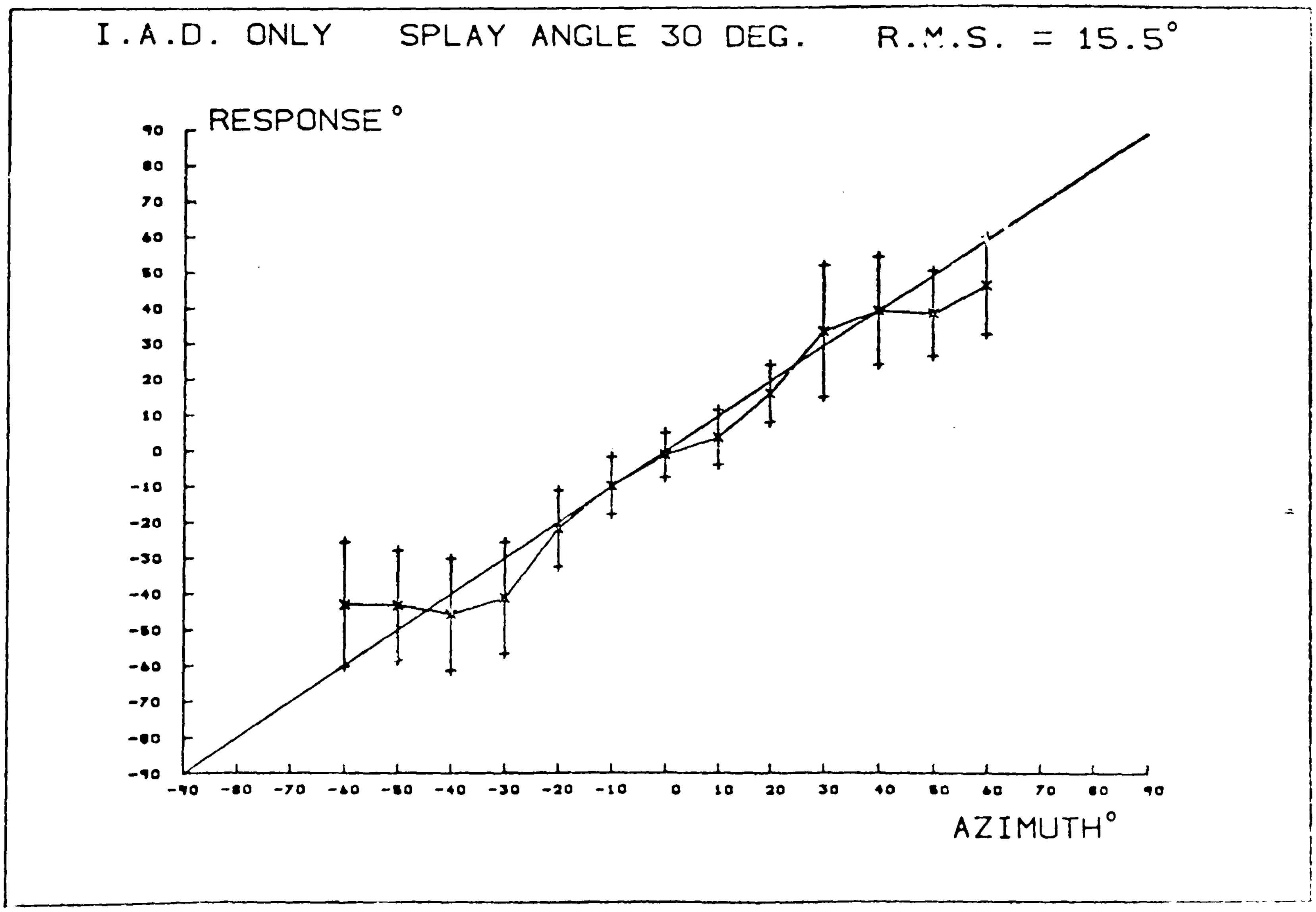


Figure 8.10

Localisation using an experimental mobility aid

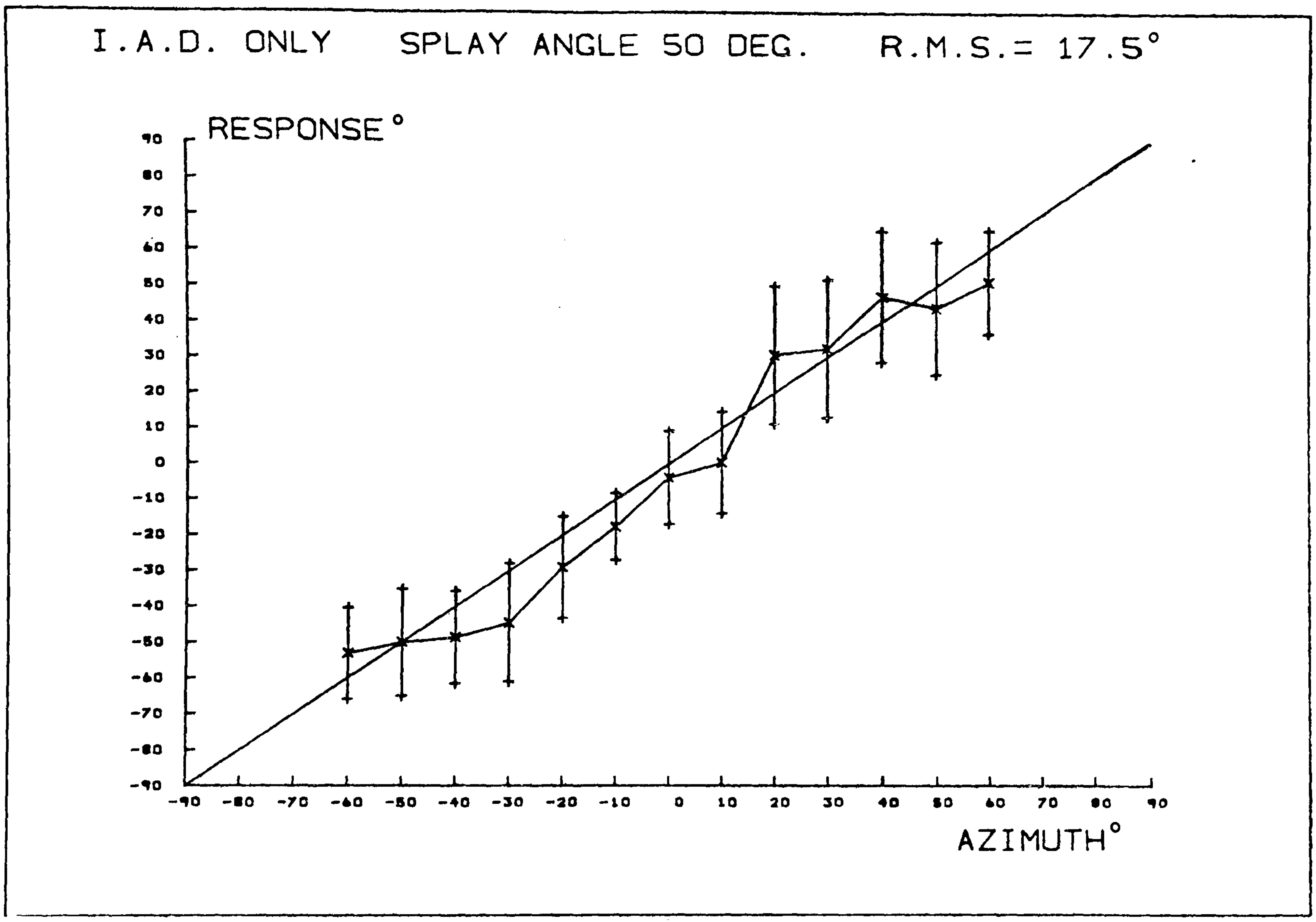


Figure 8.11

Localisation using an experimental mobility aid

performance, shown at the top of the graph, was obtained by collapsing the 16 response errors at each of the 13 angles and calculating an overall root mean square error value (R.M.S.). A tabulated presentation of this data is included in Appendix 2.

In the second group of sessions only interaural time information was presented to the subjects. The interaural intensity information was eliminated by pointing both receiving transducers in a straight ahead direction so that the splay angle was zero. The results are shown in Fig. 8.7.

Fig. 8.8 shows the results of a group of sessions with a return to the use of both time and intensity information. On this occasion, however, different intensity information was presented by changing the splay angle of the receiving transducers to 60 degrees outwards from the median plane.

The final two groups of sessions presented only interaural intensity information to the subjects. The interaural time information was eliminated by mounting the receiving transducers centrally, one above the other; the arrangement is shown in Fig. 8.9. The interaural intensity information was varied between the two groups by changing the splay angle of the receiving transducers from 30 degrees to 50 degrees. The results are shown in Figs. 8.10 and 8.11 respectively. The splay angles were presented in a random order to the subjects in an attempt to prevent learning effects from biasing the results.

Finally, some sessions were attempted still using the arrangement in Fig. 8.9 but with the receiving transducers pointing straight ahead, i.e. no interaural time or intensity information. As expected, the subjects generally perceived a central image which never varied whatever the position of the pole.

After each experimental session the subjects were questioned about their subjective impressions whilst performing the task. All the subjects reported that they experienced no real difficulty during the experiments but mentioned that the signals were not always of the same loudness. This was to be expected, of course, since the acoustic output of the transmitting torch varies slightly with azimuth in the sector concerned. One subject reported that he found the experiments using only interaural intensity differences slightly easier than those using a combination of interaural time and intensity differences, whereas another subject reported the exact opposite impression. The remaining two felt there was little difference in performing the task between any of the sessions.

8.5 Discussion

By an inspection of the graphical representation of the results it is clear that the level of localisation information provided by the mobility device is high. None of the azimuth/response curves departs by very much from the ideal. What is of more specific interest is that the most accurate results have been obtained when time information alone or intensity information alone have been present rather than a combination of both. This disputes Rudlin's conclusions and raises fundamental questions about the nature of the laws governing the synthesis of interaural time and intensity differences into subjectively perceived images.

A comparison between Figs. 8.6 and 8.8, which both represent combinations of time and intensity information, was expected to show that greater accuracy can be achieved at a splay angle of 30 degrees than at a splay angle of 60 degrees. This expectation was based on the conclusions reached in Chapter 5; namely, that at a splay angle of

30 degrees the interaural intensity information provided by the device resembles most closely that occurring in the natural auditory system in the frequency range 2.5 - 4 kHz. The two curves, however, are very similar, both tending to exhibit a slope that is slightly steeper than the slope of the ideal line. There is little difference between the overall R.M.S. error values, that of 17.9 for the 30 degree splay angle is only marginally better than that of 18.3 for the 60 degree splay angle. The corresponding differences in the mean standard deviations of 12.9 and 13.5 are also slight.

When the interaural intensity information is eliminated so that only time information remains, the results are much better. These are shown in Fig. 8.7. The slope of the curve now follows the slope of the ideal line and the overall R.M.S. error of 13.5 and mean standard deviation of 11.9 are considerably lower than those mentioned above. In fact, these results show an accuracy approaching that obtained using the dummy-head techniques described in the previous chapter. There the overall R.M.S. error and mean standard deviation were 11.1 and 10.0 degrees respectively. These results, in turn, compare reasonably with those obtained with the natural auditory system where the figures were 6.9 and 6.0 degrees respectively.

It is perhaps not surprising that accurate results have been obtained using time information alone since it was shown in Fig. 8.2 that the particular spacing of receiving transducers employed gives interaural time differences which are almost identical to those obtained naturally in the sector concerned. We know from the localisation experiments of Jeffress and Taylor [54] that good accuracy is achieved when naturally occurring interaural time differences are simulated. Their results gave an R.M.S. error value of 12.8 and an overall standard deviation of 11.4.

Figs. 8.10 and 8.11, which represent the results for interaural intensity information alone, show curves whose slopes also tend to follow the slope of the ideal line and whose accuracy, in both cases, is better than that obtained with combinations of time and intensity information, although it is slightly less good than that with time information alone. On this occasion, considerably better accuracy was obtained using a splay angle of 30 degrees than with a splay angle of 50 degrees. An interesting feature of the curve for the splay angle of 30 degrees is that the results obtained for the angles 20 degrees to either side of the 0 degree position are very accurate and remain reasonably accurate up to 40 degrees to either side of the 0 degree position. This agrees largely with the predictions made in Chapter 5 which were based on the similarity between the interaural intensity information provided by the device and that occurring naturally in the frequency range 2.5 - 4 kHz. The natural interaural intensity information in this region, therefore, appears to be a yardstick against which the accuracy of the interaural intensity cues provided by the device can be measured.

We have a situation, therefore, where we are able to obtain relatively accurate interaural time and interaural intensity information from the device. Yet, when both cues are combined, the judgements of azimuth by listeners are measurably worse than those obtained when the individual cues are used.

The expectation that the most accurate judgements should be made with an appropriate combination of time and intensity information is partly based on the conclusions reached by Rudlin. In his experiments, he was concerned with a smaller sector, 30 degrees to one side of the median plane and 40 degrees to the other. When considering his results for interaural intensity differences alone, Fig. 8.3(a), it can be seen that although the slopes of the curves tend to be the same they are widely spaced on the y-axis. In certain cases the intra-subject variability as well as the inter-subject variability is high. This is somewhat unexpected at the microphone angle of 0 degrees, where the signals presented to the two ears of the listener should nominally be identical. This type of result might suggest that the gains of the two channels did not remain equal between experimental sessions.

A comparison between the mean standard deviations for time information alone and time and intensity information combined shows a difference of only 0.5 degrees. This is not a very significant amount and must be borne in mind when considering the conclusions. Perhaps the strongest piece of evidence Rudlin obtained in favour of a combination of time and intensity information is that the subjects reported that they found the task easier when both cues were present. It is of additional interest to note that his curve for a combination of time and intensity information, Fig. 8.3(c), tends to exhibit a steeper slope than the slope of the ideal line, a result similar to the one obtained in the present series of experiments.*

However, despite the reservations in accepting Rudlin's conclusions, we know from the dummy-head experiments, described in the previous chapter, that an artificial combination of time and intensity differences can produce accurate results.

It is useful to examine the results of other workers who have performed experiments in this field to see if any comparisons can be drawn. Whitworth and Jeffress[47] carried out experiments where signals with various combinations of interaural time and intensity differences were presented to subjects via earphones. The subjects were asked to match the lateral position of one tone, the "signal", with another, the "pointer", by adjusting the interaural time difference for the pointer

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* A further discussion of Rudlin's experiments appears in Appendix 3.

until it seemed to be in the same lateral position as the signal. The subjects with normal hearing were able to perceive the signal in two places during the experiment and were able to match either signal image at will with the tonal pointer. One signal image, which the experimenters called the "time" image, depended almost wholly upon the interaural time difference for its location. The other, called the "intensity" image, depended upon both the time difference and the intensity difference. A typical set of results for a normally hearing listener is shown in Fig. 8.12. The upper graph, representing "intensity" responses, consists of lines with a slope of about 20μsec. per decibel difference of level. The lower graph, for "time" judgements, shows a slope of only 0.36μsec/dB. Subjects who had a high frequency hearing loss were unable to give a "time" response as they could not hear a second signal tone. Their "intensity" response was similar to the one described above. In summarising their results Whitworth and Jeffress stated that, of the two signal tones concerned, one moved considerably in its lateral position with changes in the difference of level at the two ears whilst the other was affected very slightly by level differences. They added, taking into account physiological inferences drawn from their results,* that the mechanism involved in the lateralisation of the "intensity" signal resembled in its trading ratio the mechanism responsible for the lateralisation of clicks.

Monshegian and Jeffress,[46] who performed a similar experiment, reported a finding which did not occur in the experiment described above. Their results, shown in Fig. 8.13, indicate a change in slope with a change in the interaural time difference for the signal. They found this with subjects who had high frequency hearing losses. They took this to mean that when time and intensity are in opposition, interaural time differences have less effect than where no intensity difference exists,

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* They suggested the existence of two peripheral auditory mechanisms.

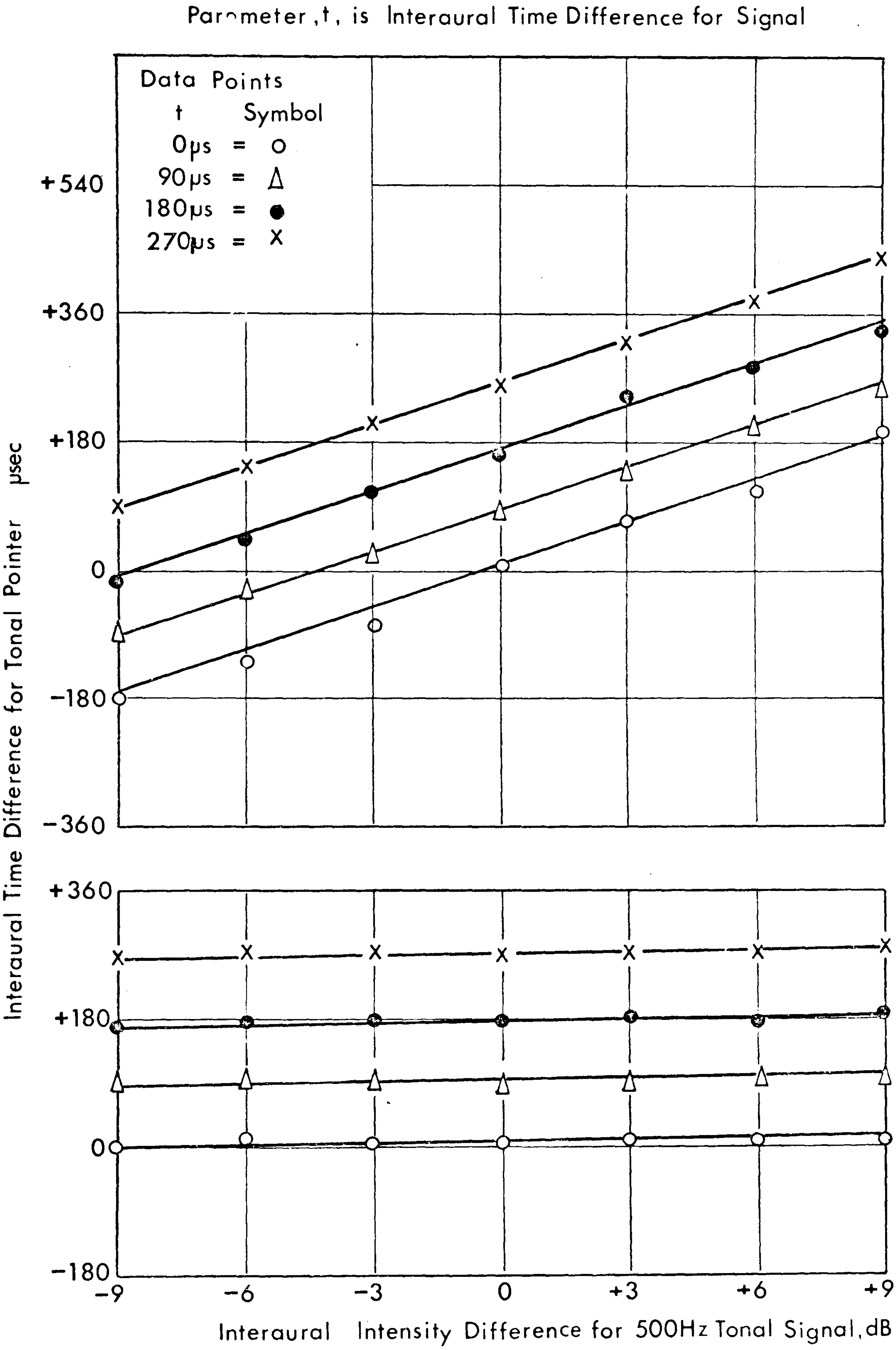


Figure 8.12 "Intensity" results (upper) and "time" results (lower for one subject. (After Whitworth and Jeffress, ref. 47)

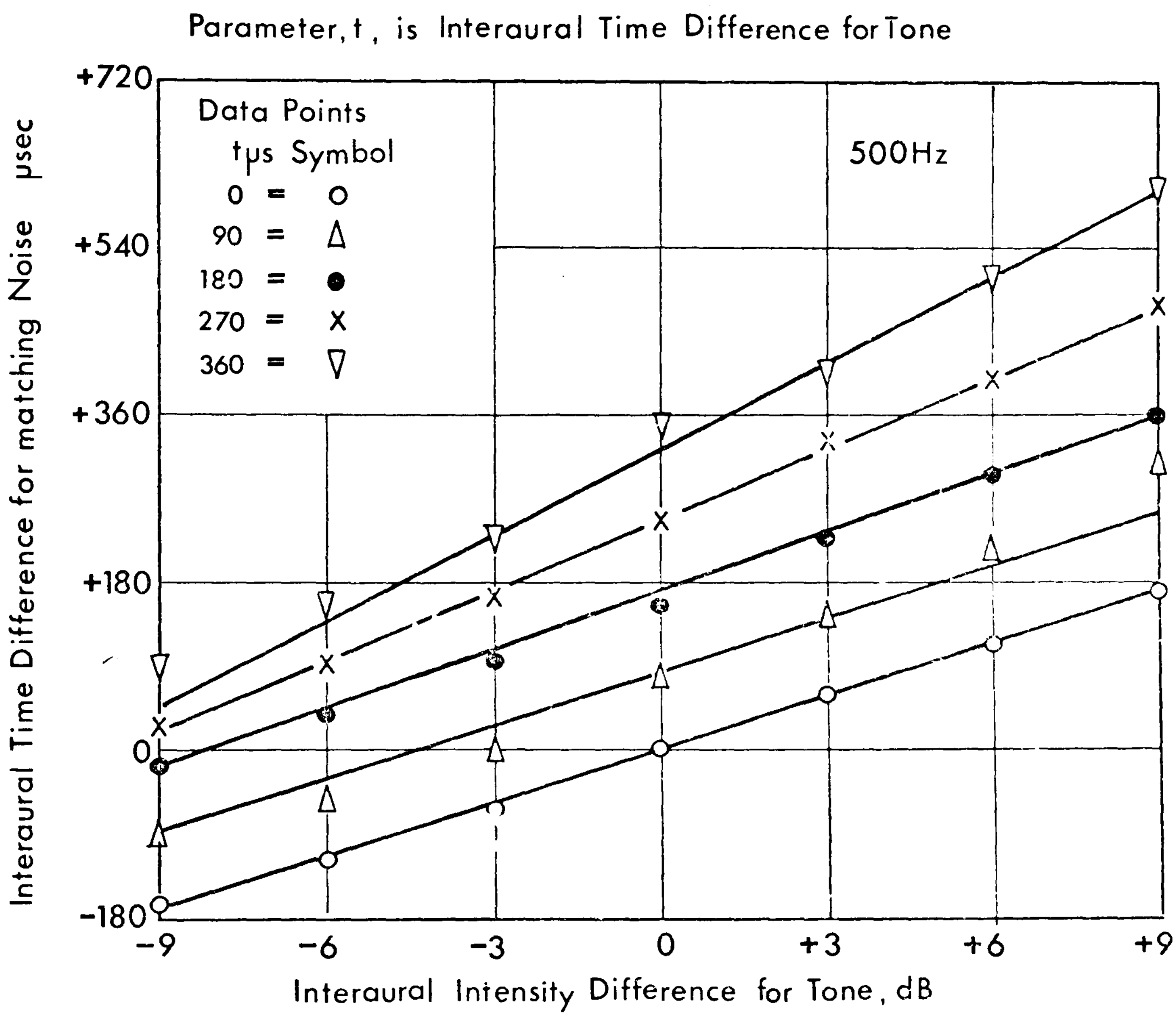


Figure 8.13 Results for one subject in time and intensity binaural lateralisation (After Moushegian and Jeffress, ref. 46)

and that when time and intensity are in concert, interaural time differences have a greater effect.

Although there is some disparity between certain results of these workers, it might not be unreasonable to draw some comparisons between their results and our own.

Jeffress et al. suggest that their results indicate the existence of an auditory mechanism, resembling in its trading ratio the mechanism responsible for the lateralisation of clicks, which produces an image for a given interaural time difference which seems to be shifted to a more extreme lateral position when a colateral interaural intensity difference is added.

In another experiment Jeffress and Taylor[54] showed, as indeed our own results have shown, that an interaural time difference alone gives rise to an image which is lateralised to a position that corresponds to the position of an actual source of sound that produces the same interaural time disparity.

Thus, if we were to consider that the mechanism described by Whitworth and Jeffress is in operation in listening to the click signals emanating from the mobility device, then the addition of an interaural intensity difference to an interaural time difference would produce an image which would always be located to a more extreme lateral position than an actual source of sound producing the same interaural differences. This is the trend our results have indicated with combinations of interaural time and intensity differences.

However, although it might be possible to compare our results with those of Whitworth and Jeffress when considering the results obtained with the mobility device, those obtained with combinations of interaural time and intensity differences in the dummy-head experiment did not show this trend. They indicated that a high accuracy of localisation can be achieved with this system. One of the main differences between the dummy-head experiment and the present one is that pinnae were present in the dummy-head arrangement. The experiments of Batteau[50] using a pair of microphones fitted with artificial pinnae showed that listeners were able to localise reasonably accurately when pinnae were present, but that their judgements became unsure and erratic when the pinnae were removed. Could it be possible, then, that the absence of information provided by the pinnae in combinations of time and intensity information leads listeners to make consistently inaccurate judgements of azimuth?

In the mobility device the directional response of the receiving transducers has been shaped to approximate the directional response of the pinna. As the simulation of this function of the pinna has been reasonably successful it is considered that some other function of the pinna must be lacking. Batteau and Rudlin have proposed that echoes from the surface of the pinna are important in localisation. It would be useful to investigate whether the addition of this factor would improve localisation with combinations of time and intensity information. This would necessitate further localisation experiments with the mobility device, where some kind of reflector system is added to the receiving transducers to simulate pinna reflections.

8.6 Conclusions

Auditory localisation experiments with an experimental version of the mobility device showed that accurate judgements of azimuth can be made. The results indicate that the judgements are more accurate when using interaural time information alone or interaural intensity information alone rather than a combination of the two. This raises the fundamental question as to whether one or other of the cues should be adopted in a prototype device, rather than the original concept of retaining a combination of both.

Before this question is answered it is suggested that the localisation experiments are repeated, but under slightly modified conditions. When an echo is used as the received signal, its intensity varies slightly with azimuth owing to the slight variation of the acoustic output of the transmitting torch with azimuth. To overcome this the torch itself could be substituted for the reflecting obstacle thus giving a direct acoustic signal which would be heard by the listener at the same loudness for each azimuth position. This would provide slightly more rigorous conditions by eliminating any effects of level differences,[44] although it is not thought that the small differences involved in the present experiment will have altered the trend of the results.

If the results concur with the present findings a further experiment should be attempted where pinna echoes are simulated by reflectors attached to the receivers. This should indicate whether a simulation of this pinna function will enable better localisation judgements to be made for combinations of time and intensity information.

With this additional data it will be possible to make a firm decision as to the type of interaural information most suitable for the device.

CHAPTER 9

CONCLUSIONS AND FURTHER WORK

9.1 Personal experience in the use of an experimental version of the aid

Although the benefits that the aid might provide for blind people cannot be adduced without an extensive evaluation programme, for which the present experimental version is not yet ready, personal use of the aid in various environments gives a useful indication of its potential, and the areas where further work is required. Most of the experience in the use of the aid has been obtained around a built-up part of the University campus and also in a quiet residential area.

When, in an outdoor location, the receiver gain is set to its maximum value and both the transmitting and receiving transducers are pointed into free space, the inherent noise of the system can be heard through the earphones as a faint clicking sound which is not obtrusive. With the transducers returned to their normal orientation, an appreciable echo can be heard from the ground. This could provide useful information when travelling along irregular terrain; for example, it was noticed that the loudness of the echo increased when the path in front rose up steeply.

The five different clicking rates which provide the range indication can all be easily discerned, although the assessment of the distance of objects is not always very accurate. It is felt, however, that this accuracy could be substantially improved with training. The directional information can also be readily appreciated and used for localisation. The perception of objects is such that they tend to remain stationary in space as the head is rotated to left and right, i.e. perceptual constancy appears to be good.

In attempting to walk along the pavement in a built-up area, it is possible to follow a shoreline without much difficulty with doorways and gaps between buildings being readily discernable. One quickly realises the large differences between the strengths of echoes that various surfaces give; for instance, brick walls give a very strong echo whilst railings give a weak one. The sound quality of the acoustic clicks also appears to vary for different surfaces, although it is thought that a long period of use would be required before these variations could be interpreted with any accuracy. The perception of more than one object at a given time is not infrequent, and on occasions one may be aware of as many as three. Moving objects can also be detected as they traverse the auditory space.

The swept gain in the receiving stage seems to be a very valuable facility. Whilst shorelining, it accentuates more distant echoes and attenuates relatively strong ones from the nearby shoreline. Thus an object on the road side of the pavement is rendered more distinct than when the normal gain setting is used. Furthermore, when crossing a road, the facility improves the ability to locate a landmark on the opposite side, so that the latter can be used as a navigation "beacon".

The task of listening to the acoustic clicks produced by the device does not interfere with the perception of ambient sounds, so the natural clues present in these sounds can be used to enhance orientation. For example, the awareness of traffic stopping and starting often indicates a road intersection ahead.

An obvious disadvantage of the aid is that no protection is given against drop offs and low obstacles. This problem can be alleviated by using the aid in conjunction with a long cane. A difficulty that arises here, however, is that the use of one hand to hold the cane and the other to hold the transmitting torch would result in a cumbersome arrangement. A version of the aid, therefore, with a head-mounted or chest-mounted transmitter should be considered.

The general impression after using the aid is that the information provided may well be adequate, after a suitable period of training, to perform the skills thought to be associated with a device of this nature. These may be listed as localisation, shorelining, recognition of relationships between click rate and distance, the ability to identify the environment through auditory signals, and the ability to make use of natural clues.

A further impression is that some of the information provided by the device would be more effective if it were not limited by the physical behaviour of ultrasound. In particular, the vast differences in reflection coefficients of various surfaces means that echoes from a strong reflector nearby will tend to dominate the attention, thus making it difficult to perceive echoes from poorer reflectors. Although the swept gain facility alleviates this problem to a certain extent, it might be advantageous to add some form of automatic gain control to further reduce the large variation in loudness of perceived echoes.

Another shortcoming of using ultrasound is the specular nature of echoes from smooth surfaces. These specular reflections can give rise to dangerous situations especially when two smooth surfaces meet at a sharp external corner. The blind person approaching such an obstacle will find difficulty in detecting it, since most of the acoustic energy is reflected away from him.

Thus, although the aid has disadvantages, some of which are inherent in the use of ultrasound, the overall feeling is that the information it provides may well be of use in aiding the mobility of blind people. However, before a full evaluation can be carried out, there is a considerable amount of further work to be done.

9.2 Further Work

Additional development of the drive circuitry for the transmitting transducers is required. At present it has been found convenient to mount this rather bulky circuitry together with the transmitting transducers in a torch handle. With the possibility of having to mount the transducers on the head or chest in order to facilitate the additional use of a long cane, an optimisation of the transmitter electronics is essential so as to provide a compact circuit which maintains sufficient acoustic output at a low power consumption. The work already done in this area indicates that the acoustic output of the transducers can be substantially increased. Thus it may be possible to obtain sufficient acoustic output from the transmitter by substituting the present three narrow beam transducers by one modified wide beam transducer. This, of course, would considerably simplify the whole transmitting system.

The earphones at present in use in the mobility aid were chosen because of their flexibility in use during experimental work. Although their performance is satisfactory, it is worth investigating a new type of earphone which uses a piezoelectric foil membrane. This earphone can be made smaller and cheaper than the electrostatic or dynamic membrane earphone, and does not require either the electrode and polarising arrangements of the former, or the magnets and pole pieces of the latter. The wide frequency response provided should ensure a faithful reproduction of the electrical output of the aid, and thus a sharp acoustic click.

Owing to the usefulness that swept gain in the receiving amplifiers has shown so far, it is considered that this facility should be optimised in order that the maximum benefit may be derived from it. The optimisation might involve carrying out a specific mobility task under blindfold conditions, where the sweep of the gain is varied between trials. Thus the most suitable sweep could be found and adopted for future use. The complexity of constructing automatic gain control should be studied to see if this course of action is worth embarking upon.

Finally, it is necessary to resolve which of the three types of interaural information available should be adopted for use in a prototype aid. This may involve, firstly, a repeat of the localisation experiments with the aid under the slightly modified conditions referred to earlier, and secondly, an investigation into the possibility of using artificial pinnae in conjunction with the receiving transducers. If it proves useful to include artificial pinnae to improve localisation, an added advantage might be an improvement in the ability to perceive sounds externally.

When the developments outlined above have been completed, it is suggested that before the aid is released to blind people extensive trials should be carried out. This will enable the performance of the device to be adjusted and modified until further improvements only marginally improve mobility.

After this stage, consideration must be given to an evaluation of the usefulness of the aid to blind people. The timing at which this is done is critical since this step carries with it a substantial risk. A realistic evaluation must involve professional teachers of blind mobility; and a rejection of the device by them could prove severely detrimental to the aid's future development. Nevertheless, the evaluation is essential if a conclusion is to be reached as to whether the aid can provide the blind user with useful environmental information which improves his orientation and mobility.

APPENDICES

APPENDIX 1

An Analysis of the Equivalent Circuit of a Piezoelectric Transducer for Ultrasonic Pulse Generation and Detection

The piezoelectric transducers for the mobility aid are required to have a wide band pass characteristic in order to provide a good pulse response. The analysis has been carried out according to the conventional equivalent circuit of a piezoelectric transducer (given in Fig. 4.2). It examines the effect on the bandwidth of tuning out the parallel capacitance of the transducer by a parallel inductance at the series resonant frequency and damping this parallel circuit by a resistance.

The analysis has been subdivided as follows:

(a) Numerical Data

(b) Transfer Function as a function of frequency

  (i) with inductor

  (ii) without inductor

(c) Impedance as a function of frequency

(d) Noise as a function of frequency

(a) Numerical Data

(Supplied by Manufacturers)

For a series resonant circuit ℓ,c,r, $\omega_o = \frac{1}{\sqrt{\ell c}}$

$$Q_s = \frac{\omega_o \ell}{r} = \frac{1}{\omega_o c r} = \sqrt{\frac{\ell}{c}} \cdot \frac{1}{r}$$

Numerical data: $Q_s = 70$

$r = 200$

Parallel capacitance: $C = 1400$ pf

$\omega_o = 2\pi\ 40$ kHz

therefore

$$c = \frac{1}{\omega_o Q_s r} = \frac{1}{2\pi.40.10^3.70.200} \doteqdot 284 \text{ pf}$$

therefore

$$\frac{C}{c} = \frac{1400}{284} = 4.93 \simeq 5$$

For a parallel resonant circuit L,C,R, $\omega_o = \dfrac{1}{\sqrt{LC}}$

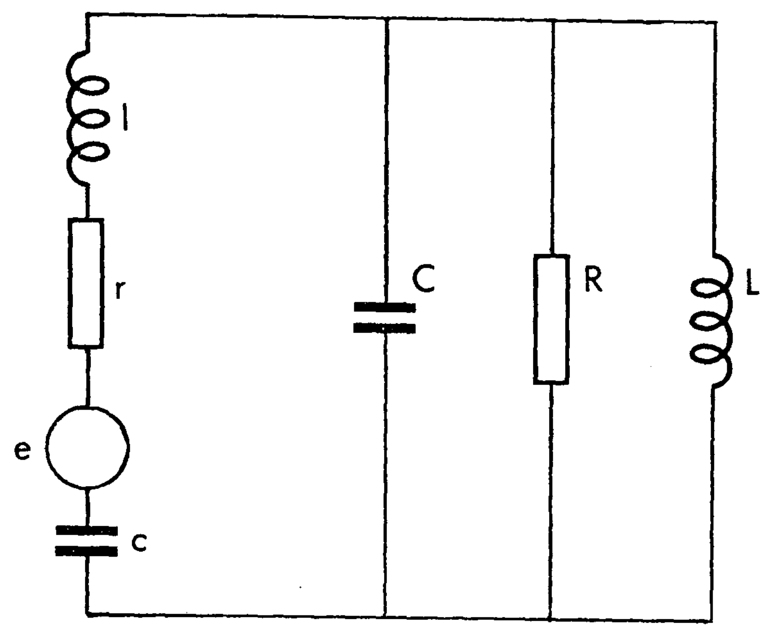
$$Q_p = \frac{R}{\omega_o L} = \omega_o CR = \sqrt{\frac{C}{L}}\, R$$

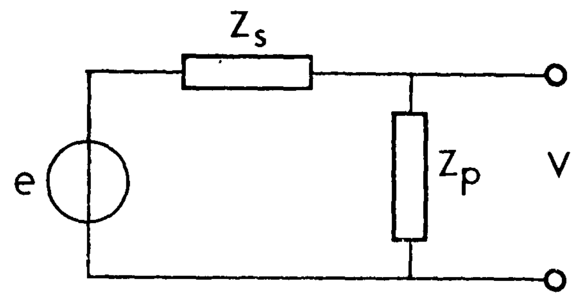
For $\omega_o = 2\pi 40.10^3$, $C = 1400$ pf

$$L = \frac{10^{12}}{(2\pi 40.10^3)^2.1400} = 10 \text{ mH}$$

(b) Transfer Function

(i) With Inductor





$$\frac{V}{e} = \frac{Z_p}{Z_p + Z_s}$$

$$\frac{V}{e} = \frac{\dfrac{j\omega LR \cdot \dfrac{1}{j\omega C}}{j\omega LR + \dfrac{R}{j\omega C} + \dfrac{L}{C}}}{\dfrac{\dfrac{LR}{C}}{j\omega LR + \dfrac{R}{j\omega C} + \dfrac{L}{C}} + j\omega\ell + r + \dfrac{1}{j\omega c}}$$

$$= \frac{LR}{LR - \omega^2 LRC\ell + j\omega LRCr + \dfrac{LRC}{c} + R\ell + \dfrac{Rr}{j\omega} - \dfrac{R}{\omega^2 c} + j\omega\ell L + rL + \dfrac{L}{j\omega c}}$$

$$= \frac{R}{R + \dfrac{RC}{c} + \dfrac{R\ell}{L} + r - \omega^2 RC\ell - \dfrac{R}{\omega^2 cL} + j\omega(RCr + \ell) + \dfrac{1}{j\omega}\left(\dfrac{Rr}{L} + \dfrac{1}{c}\right)}$$

$$= \frac{R}{R\left[\frac{\ell}{L} + \frac{C}{c} + 1\right] + r - R\left[\omega^2 C\ell + \frac{1}{\omega^2 cL}\right] + jr\left[\omega CR + \frac{\omega\ell}{r}\right] + \frac{r}{j}\left[\frac{R}{\omega L} + \frac{1}{r\omega c}\right]}$$

If $\ell c = \frac{1}{\omega_o^2} = LC$

$$\frac{V}{e} = \frac{R}{R\left[\frac{\ell}{L} + \frac{C}{c} + 1\right] + r - R\left[\frac{\omega^2}{\omega_o^2} \cdot \omega_o^2 C\ell + \frac{\omega_o^2}{\omega^2} \cdot \frac{1}{\omega_o^2 cL}\right] + \frac{j\omega r}{\omega_o}\underbrace{\left[\omega_o CR + \frac{\omega_o \ell}{r}\right]}_{Q_p + Q_s} + \frac{r}{\frac{j\omega}{\omega_o}}\underbrace{\left[\frac{R}{\omega_o L} + \frac{1}{r\omega_o c}\right]}_{Q_p + Q_s}}$$

$$= \frac{\frac{R}{r}}{\frac{R}{r}\left[\frac{\ell}{L} + \frac{C}{c} + 1 - \left(\frac{\omega}{\omega_o}\right)^2 \frac{\ell}{L} - \left(\frac{\omega_o}{\omega}\right)^2 \frac{C}{c} + \frac{r}{R}\right] + j\left[\frac{\omega}{\omega_o} - \frac{\omega_o}{\omega}\right](Q_p + Q_s)}$$

$$= \frac{1}{\left\{1 - \left(\frac{\omega}{\omega_o}\right)^2\right\}\frac{\ell}{L} + \left\{1 - \left(\frac{\omega_o}{\omega}\right)^2\right\}\frac{C}{c} + 1 + \frac{r}{R} - j\,\frac{\omega_o}{\omega}\left[1 - \left(\frac{\omega}{\omega_o}\right)^2\right]\frac{r}{R}(Q_p + Q_s)}$$

$$= \frac{1}{1 + \frac{r}{R} + \left\{1 - \left(\frac{\omega}{\omega_o}\right)^2\right\}\frac{\ell}{L} + \left\{1 - \left(\frac{\omega_o}{\omega}\right)^2\right\}\frac{C}{c} - j\,\frac{\omega_o}{\omega}\left[1 - \left(\frac{\omega}{\omega_o}\right)^2\right]\frac{r}{R}(Q_p + Q_s)}$$

At the resonant frequency $\frac{\omega_o}{2\pi}$, $\frac{V}{e} = \frac{R}{R+r}$ ($\simeq 1$ if $R >> r$)

There will be peaks in the ratio $\frac{V}{e}$ approximately at the two frequencies at which the real component of the denominator is zero.

i.e. when $1 + \frac{r}{R} + \left[1 - (\frac{\omega}{\omega_o})^2\right] \frac{\ell}{L} + \left[1 - (\frac{\omega_o}{\omega})^2\right] \frac{C}{c} = 0$

or, since $\ell c = LC$ and $\frac{\ell}{L} = \frac{C}{c}$

when $(1 + \frac{r}{R}) \frac{c}{C} - (\frac{\omega}{\omega_o})^2 + 2 - (\frac{\omega_o}{\omega})^2 = 0$

$$(\frac{\omega}{\omega_o})^2 - 2 + (\frac{\omega_o}{\omega})^2 = (1 + \frac{r}{R}) \frac{c}{C}$$

$$(\frac{\omega}{\omega_o} - \frac{\omega_o}{\omega})^2 = (1 + \frac{r}{R}) \frac{c}{C}$$

$$\frac{\omega}{\omega_o} - \frac{\omega_o}{\omega} = \pm \sqrt{(1 + \frac{r}{R}) \frac{c}{C}}$$

$$(\frac{\omega}{\omega_o})^2 \mp \sqrt{(1 + \frac{r}{R}) \frac{c}{C}} \cdot \frac{\omega}{\omega_o} - 1 = 0$$

$$\frac{\omega}{\omega_o} = \frac{\mp \sqrt{(1 + \frac{r}{R}) \frac{c}{C}} \pm \sqrt{(1 + \frac{r}{R}) \frac{c}{C} + 4}}{2}$$

If $\frac{r}{R} << 1$ and $\frac{c}{C} << 1$

$$\frac{\omega}{\omega_o} \doteqdot \frac{\pm \sqrt{(1 + \frac{r}{R}) \frac{c}{C}} \pm 2 \left[1 + \frac{1}{8} (1 + \frac{r}{R}) \frac{c}{C}\right]}{2}$$

$$\doteqdot 1 \pm \frac{1}{2} \sqrt{(1 + \frac{r}{R}) \frac{c}{C}}$$

Numerically, if $\frac{c}{C} = \frac{1}{5}$ $\qquad f = 40\ (1 \pm \frac{\sqrt{\frac{1}{5}}}{2})$ kHz

$= 40\ (1.224)$ or $40(.776)$ kHz

$= 48.94$ kHz or 31.06 kHz

If bandwidth is $2\delta\omega$, then $\frac{2\delta\omega}{\omega_o} = 2 \times \frac{1}{2} \sqrt{\frac{c}{C}}$

The peaks of $\frac{V}{e}$ will not be at precisely these frequencies because of the effect of the varying magnitude of the imaginary component. The exact position is found by examining the magnitude of $\frac{V}{e}$.

<u>Magnitude of $\frac{V}{e}$</u>

(Assuming $CL = \frac{1}{\omega_o^2} = c\ell$)

$$\frac{V}{e} = \frac{\dfrac{1}{\left(\frac{1}{R} + j(\omega C - \frac{1}{\omega L})\right)}}{\dfrac{1}{\frac{1}{R} + \underbrace{j(\omega C - \frac{1}{\omega L})}_{\boxed{j\sqrt{\frac{C}{L}}\left(\frac{\omega}{\omega_o} - \frac{\omega_o}{\omega}\right)}}} + r + \underbrace{j\ (\omega\ell - \frac{1}{\omega c})}_{\boxed{j\sqrt{\frac{\ell}{c}}\left(\frac{\omega}{\omega_o} - \frac{\omega_o}{\omega}\right)}}}$$

$$= \frac{1}{1 + \frac{r}{R} - \underbrace{\sqrt{\frac{C\ell}{Lc}}}_{\boxed{\frac{r}{R} Q_p Q_s}} \left(\frac{\omega}{\omega_o} - \frac{\omega_o}{\omega}\right)^2 + \underbrace{j\left(r\sqrt{\frac{C}{L}} + \frac{1}{R}\sqrt{\frac{\ell}{c}}\right)\left(\frac{\omega}{\omega_o} - \frac{\omega_o}{\omega}\right)}_{\boxed{j\frac{r}{R}(Q_p + Q_s)\left(\frac{\omega}{\omega_o} - \frac{\omega_o}{\omega}\right)}}}$$

(Magnitude)^2 of Denominator is

$$\left[1 + \frac{r}{R} - \frac{r}{R} Q_p Q_s \left(\frac{\omega}{\omega_o} - \frac{\omega_o}{\omega}\right)^2\right]^2 + \frac{r^2}{R^2} (Q_p + Q_s)^2 \left(\frac{\omega}{\omega_o} - \frac{\omega_o}{\omega}\right)^2$$

$$= \left(1 + \frac{r}{R}\right)^2 - 2\left(1 + \frac{r}{R}\right) \frac{r}{R} Q_p Q_s \left(\frac{\omega}{\omega_o} - \frac{\omega_o}{\omega}\right)^2 + \frac{r^2}{R^2} Q_p^{\,2} Q_s^{\,2} \left(\frac{\omega}{\omega_o} - \frac{\omega_o}{\omega}\right)^4$$

$$+ \frac{r^2}{R^2} (Q_p + Q_s)^2 \left(\frac{\omega}{\omega_o} - \frac{\omega_o}{\omega}\right)^2$$

$$= \left(1 + \frac{r}{R}\right)^2 + \frac{r^2}{R^2} \left[(Q_p + Q_s)^2 - 2\left(\frac{R}{r} + 1\right) Q_p Q_s\right] \left(\frac{\omega}{\omega_o} - \frac{\omega_o}{\omega}\right)^2$$

$$+ \frac{r^2}{R^2} Q_p^{\,2} Q_s^{\,2} \left(\frac{\omega}{\omega_o} - \frac{\omega_o}{\omega}\right)^4$$

If $(Q_p + Q_s)^2 - 2(\frac{R}{r} + 1) Q_p Q_s = 0$ there is only a fourth power of $(\frac{\omega}{\omega_o} - \frac{\omega_o}{\omega})$, i.e. a band pass characteristic, remaining.

For this to be so either $Q_p = Q_s$ and $\frac{R}{r} = 1$ or,

if $Q_s >> Q_p$

$$Q_s = 2\left(\frac{R}{r} + 1\right) Q_p \quad \text{and} \quad \frac{Q_s}{Q_p} = 2\left(\frac{R}{r} + 1\right)$$

But $Q_p = \omega_o CR$

Therefore

$$Q_s = 2\left(\frac{R}{r} + 1\right) \omega_o CR = 2\omega_o CR + 2\omega_o C \frac{R^2}{r}$$

$$\text{or} \quad R^2 + rR - \frac{Q_s}{2\omega_o C} r = 0$$

$$\text{or } R = \frac{-r \pm r\sqrt{1 + 2\frac{Q_s}{\omega_o Cr}}}{2}$$

$$\text{or } \frac{R}{r} = \frac{-1 \pm \sqrt{1 + 2Q_s^2 \frac{c}{C}}}{2}$$

$$\text{or } \frac{R}{r} \simeq Q_s\sqrt{\frac{c}{2C}} \qquad \text{say } 70\sqrt{\cdot 1} \simeq 22$$

$$\left[\text{or } R \simeq \sqrt{\frac{\ell}{2C}}\right]$$

$$\text{or } R \simeq 200 \times 22 = 4400$$

But if $R = 4400$,

$$Q_p = \omega_o CR$$

$$= 2\pi\ 40.10^3.1400.10^{-12}.4400$$

$$= 1.4$$

Thus $Q_s >> Q_p$ as assumed.

<u>Fourth Power Band Pass Characteristic</u>

Down 3 dB when

$$(1 + \frac{r}{R})^2 = (\frac{r}{R})^2\ Q_p^2 Q_s^2\ (\frac{\omega}{\omega_o} - \frac{\omega_o}{\omega})^4$$

$$\text{i.e. } (\frac{\omega}{\omega_o} - \frac{\omega_o}{\omega})^4 = \frac{R^2(1 + \frac{r}{R})^2}{r^2 Q_p^2 Q_s^2} \simeq \frac{22^2}{(70 \times 1.4)^2} \simeq \frac{22^2}{10^4}$$

$$(\frac{\omega}{\omega_o} - \frac{\omega_o}{\omega})^2 \simeq \pm\ 0.20$$

$$\frac{\omega}{\omega_o} - \frac{\omega_o}{\omega} \approx 0.45$$

$$\left(\frac{\omega}{\omega_o}\right)^2 \mp .45\frac{\omega}{\omega_o} - 1 = 0$$

$$\frac{\omega}{\omega_o} = \frac{\pm .45 \pm \sqrt{.20+4}}{2}$$

$$= \frac{\pm .45 \pm 2.05}{2} = \frac{2.5 \text{ or } 1.60}{2}$$

$$= 1.25 \text{ or } 0.8$$

i.e. the pass band extends from 32 kHz to 50 kHz.

The bandwidth is therefore 18 kHz.

(ii) Without inductor ($L = \infty$)

$$\frac{V}{e} = \frac{1}{\frac{C}{c} + 1 + \frac{r}{R} - \omega^2 C\ell + \frac{j\omega}{\omega_o}\cdot\frac{r}{R}\left[\underbrace{\omega_o CR}_{Q \text{ of } CR} + \underbrace{\frac{\omega_o \ell}{r}}_{Q_s}\right] + \frac{r}{\frac{j\omega R}{\omega_o}}\left[\underbrace{\frac{1}{r\omega_o c}}_{Q_s}\right]}$$

$$= \frac{1}{1 + \frac{r}{R} + \left(1 - \left(\frac{\omega}{\omega_o}\right)^2\right)\frac{C}{c} - j\frac{r}{R}\cdot\frac{\omega_o}{\omega}\left[\left(1 - \left(\frac{\omega}{\omega_o}\right)^2\right) Q_s - \left(\frac{\omega}{\omega_o}\right)^2 Q_{CR}\right]}$$

There is now no zero for the real component of the denominator but only a minimum when $\omega = \omega_o$. Minimum of imaginary component gives a peak of $\frac{V}{e}$ and occurs at a frequency given by:-

$$Q_s = \left(\frac{\omega}{\omega_o}\right)^2 (Q_s + Q_{CR})$$

$$\text{or } \left(\frac{\omega}{\omega_o}\right)^2 = \frac{Q_s}{Q_s + Q_{CR}}$$

$$\text{Magnitude at this } f = \frac{1}{1 + \frac{r}{R}} = \frac{R}{R + r}$$

[e.g. for R = 4400, r = 200, magnitude is $\frac{4.4}{4.6} = 0.96$]

Zero of real component given by

$$1 + \frac{r}{R} + \left(1 - \left(\frac{\omega}{\omega_o}\right)^2\right) \frac{C}{c} = 0$$

$$\left(\frac{\omega}{\omega_o}\right)^2 = \frac{c}{C}\left(\frac{C}{c} + 1 + \frac{r}{R}\right)$$

Numerically,

$$\left(\frac{\omega}{\omega_o}\right)^2 = \frac{1}{5}\left(5 + 1 + \frac{200}{4400}\right)$$

$$= 1.21$$

$$\text{or} \quad f = 40.10^3\sqrt{1.21}$$

$$= 44 \text{ kHz}$$

Magnitude at this frequency

$$= \frac{1}{-\frac{r}{R} \cdot \frac{\omega_o}{\omega} \left[(1 - 1.21)70 - 1.21 \cdot \frac{1}{2}\right]}$$

$$= \frac{1.1\ R}{r\ (14.7 + 0.61)}$$

$$= 0.072 \frac{R}{r} \qquad \text{i.e.} \propto R$$

$$= 0.072 \cdot \frac{4400}{200} \quad \text{for } R = 4400, \quad r = 200$$

$$= 1.58$$

(c) Impedance of the parallel arrangement of two resonant circuits

$$Z = \frac{Z_p Z_s}{Z_p + Z_s}$$

$$= \frac{\frac{1}{\frac{1}{R} + j(\omega C - \frac{1}{\omega L})} \cdot (j\omega \ell + r + \frac{1}{j\omega c})}{\frac{1}{\frac{1}{R} + j(\omega C - \frac{1}{\omega L})} + (j\omega \ell + r + \frac{1}{j\omega c})}$$

$$= \frac{j\omega \ell + r + \frac{1}{j\omega c}}{1 + \frac{r}{R} - \frac{r}{R} Q_p Q_s \left(\frac{\omega}{\omega_o} - \frac{\omega_o}{\omega}\right)^2 + j\frac{r}{R}(Q_p + Q_s)\left(\frac{\omega}{\omega_o} - \frac{\omega_o}{\omega}\right)}$$

$$|Z|^2 = \frac{r^2 + \frac{\ell}{c}\left(\frac{\omega}{\omega_o} - \frac{\omega_o}{\omega}\right)^2}{(1 + \frac{r}{R})^2 + \frac{r^2}{R^2}\left[(Q_p + Q_s)^2 - 2(\frac{R}{r} + 1)Q_p Q_s\right]\left(\frac{\omega}{\omega_o} - \frac{\omega_o}{\omega}\right)^2 + \frac{r^2}{R^2} Q_p{}^2 Q_s{}^2 \left(\frac{\omega}{\omega_o} - \frac{\omega_o}{\omega}\right)^4}$$

[equation (i)]

When R is very low, the negative term in the denominator is small and the real part is never zero. There is only one peak in the reciprocal of the denominator.

When R is very large, the real part is zero when

$$(1 + \frac{r}{R})^2 - 2\frac{r^2}{R^2}(\frac{R}{r} + 1) Q_p Q_s \left(\frac{\omega}{\omega_o} - \frac{\omega_o}{\omega}\right)^2 \doteqdot 0$$

or $$\left(\frac{\omega}{\omega_o} - \frac{\omega_o}{\omega}\right)^2 = \frac{(1 + \frac{r}{R})^2 \frac{R^2}{r^2}}{2(\frac{R}{r} + 1)Q_p Q_s}$$

thus giving two peaks of impedance.

If R is very high

$$R \simeq \frac{Q_p}{\omega_o C} = \frac{Q_p . 10^{12}}{2\pi \ 40.10^3 . 1400} \simeq 2{\cdot}84.10^3 Q_p$$

and $$\left(\frac{\omega}{\omega_o} - \frac{\omega_o}{\omega}\right)^2 = \frac{2{\cdot}84.10^3}{2 \times 200 \times Q_s} = \frac{7.1}{70} = 0.1014$$

$$\frac{\omega}{\omega_o} - \frac{\omega_o}{\omega} = \pm \ 0.317$$

$$\left(\frac{\omega}{\omega_o}\right)^2 \mp 0.317 \ \left(\frac{\omega}{\omega_o}\right) - 1 = 0$$

$$\frac{\omega}{\omega_o} = \frac{\pm \ 0.317 \pm \sqrt{{\cdot}1014 + 4}}{2} = \frac{\pm \ 0.317 \pm 2.025}{2}$$

$$= \frac{2.342 \text{ or } 1.708}{2} = 1.171 \text{ or } 0.854$$

i.e. $f = 46.8$ kHz or 34.1 kHz

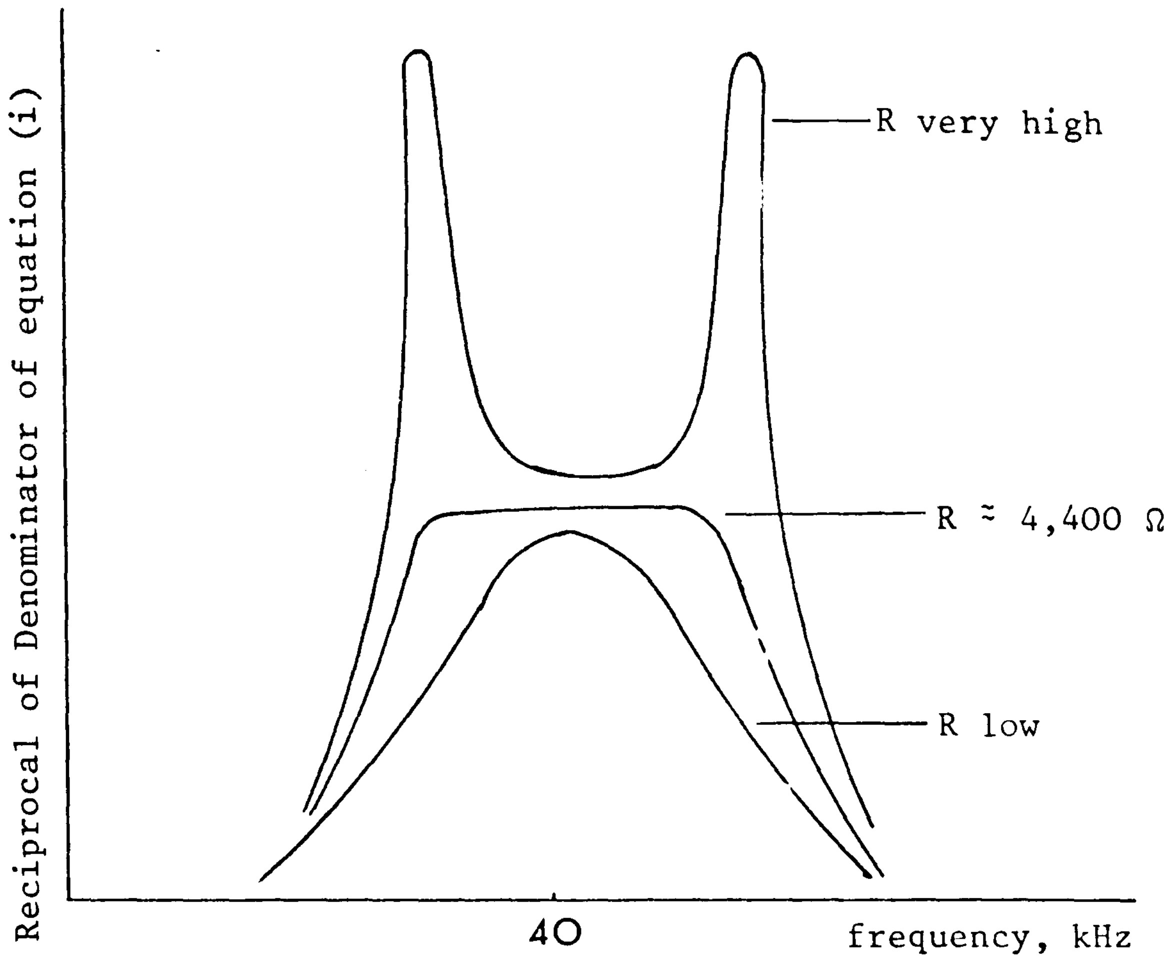


Figure (i) Reciprocal of denominator of equation (i) as a function of frequency with parallel tuning.

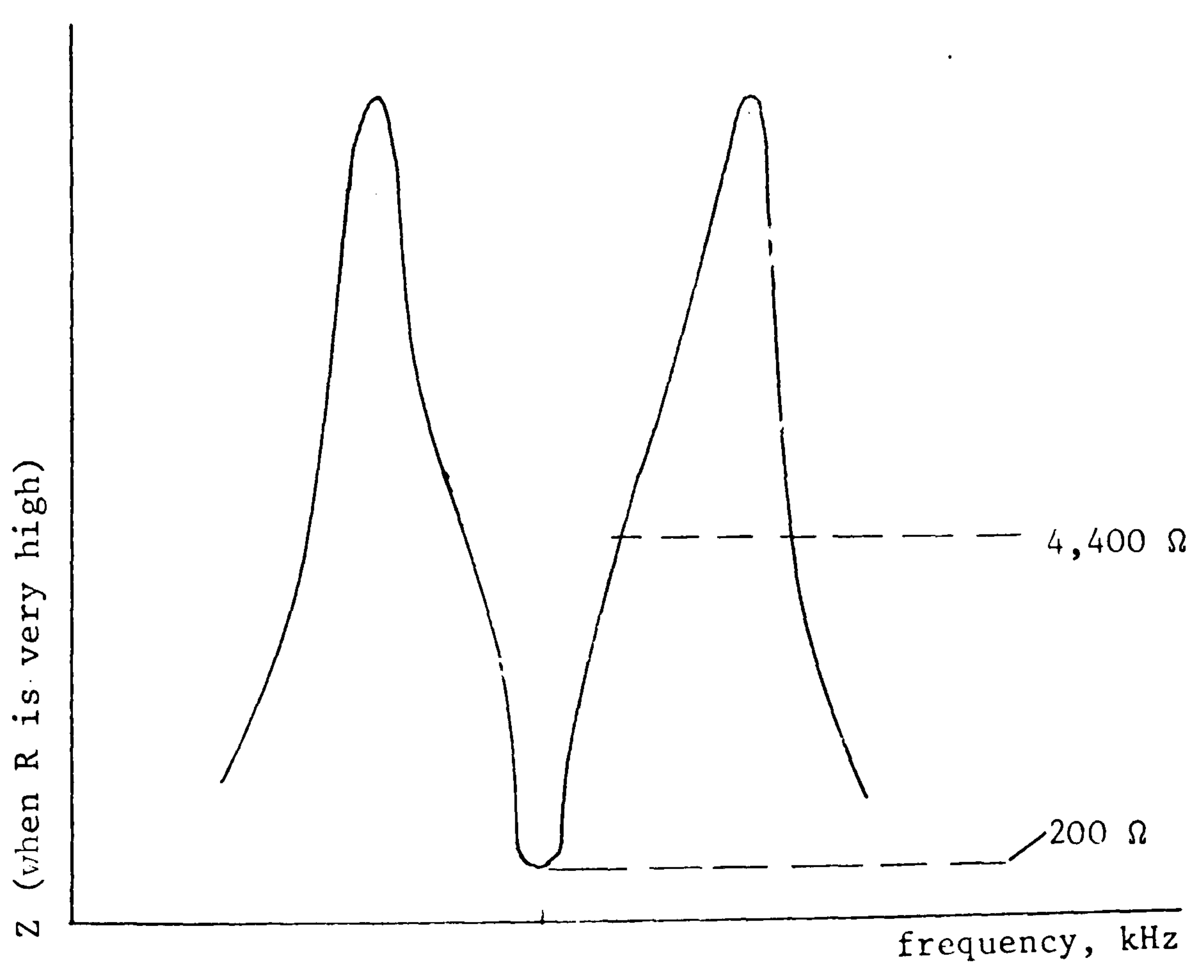


Figure (ii) Impedance as a function of frequency at parallel tuning

Maximum bandwidth condition

The optimum value of R eliminates the term $(\frac{\omega}{\omega_o} - \frac{\omega_o}{\omega})^2$ in the denominator and thus there are no longer any values of ω for which the denominator is zero. This smooths the two high peaks and gives a band pass shaped curve due to the term $(\frac{\omega}{\omega_o} - \frac{\omega_o}{\omega})^4$ in the denominator.

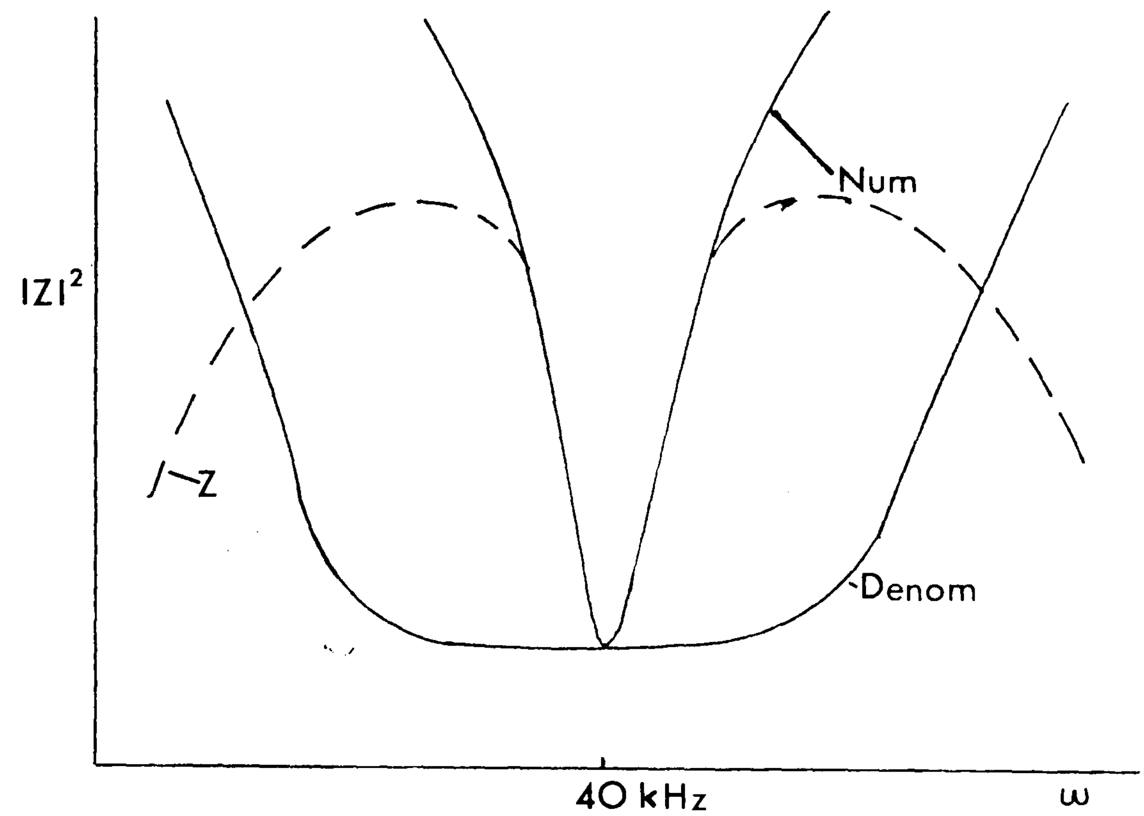
Impedance seen by the amplifier in the maximum BW condition

$$Z = \frac{Z_p Z_s}{Z_p + Z_s}$$

$$= \frac{\dfrac{1}{\frac{1}{R} + j(\omega C - \frac{1}{\omega L})} \quad (j\omega\ell + r + \frac{1}{j\omega c})}{\dfrac{1}{\frac{1}{R} + j(\omega C - \frac{1}{\omega L})} + (j\omega\ell + r + \frac{1}{j\omega c})}$$

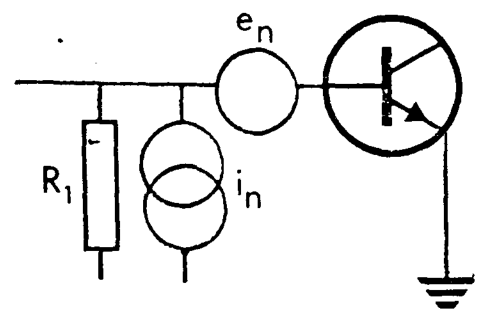
$$= \frac{j\omega\ell + r + \frac{1}{j\omega c}}{1 + \frac{r}{R} - \frac{r}{R} Q_p Q_s \ (\frac{\omega}{\omega_o} - \frac{\omega_o}{\omega})^2 + j\,\frac{r}{R}\,(Q_p + Q_s)(\frac{\omega}{\omega_o} - \frac{\omega_o}{\omega})}$$

$$|Z|^2 = \frac{r^2 + \frac{\ell}{c}\,(\frac{\omega}{\omega_o} - \frac{\omega_o}{\omega})^2}{(1 + \frac{r}{R})^2 + \frac{r^2}{R^2}\,Q_p^{\,2} Q_s^{\,2}\ (\frac{\omega}{\omega_o} - \frac{\omega_o}{\omega})^4}$$



From a noise point of view, the impedance R is best provided by negative feedback within the amplifier. The source impedance then presented to the amplifier (with its own low input impedance due to feedback) will then be the original undamped high peaked curve of impedance shown in Fig. (ii).

(d) Noise Performance



Assuming

(a) An emf of a $nV/\sqrt{Hz}$

(b) A current of b $pA/\sqrt{Hz}$ or b $pA/\sqrt{Hz}$ per 1000Ω of source impedance

(c) A current of 4 $pA/\sqrt{Hz}$ for a 1000Ω resistor R but $\propto \frac{1}{\sqrt{R}}$

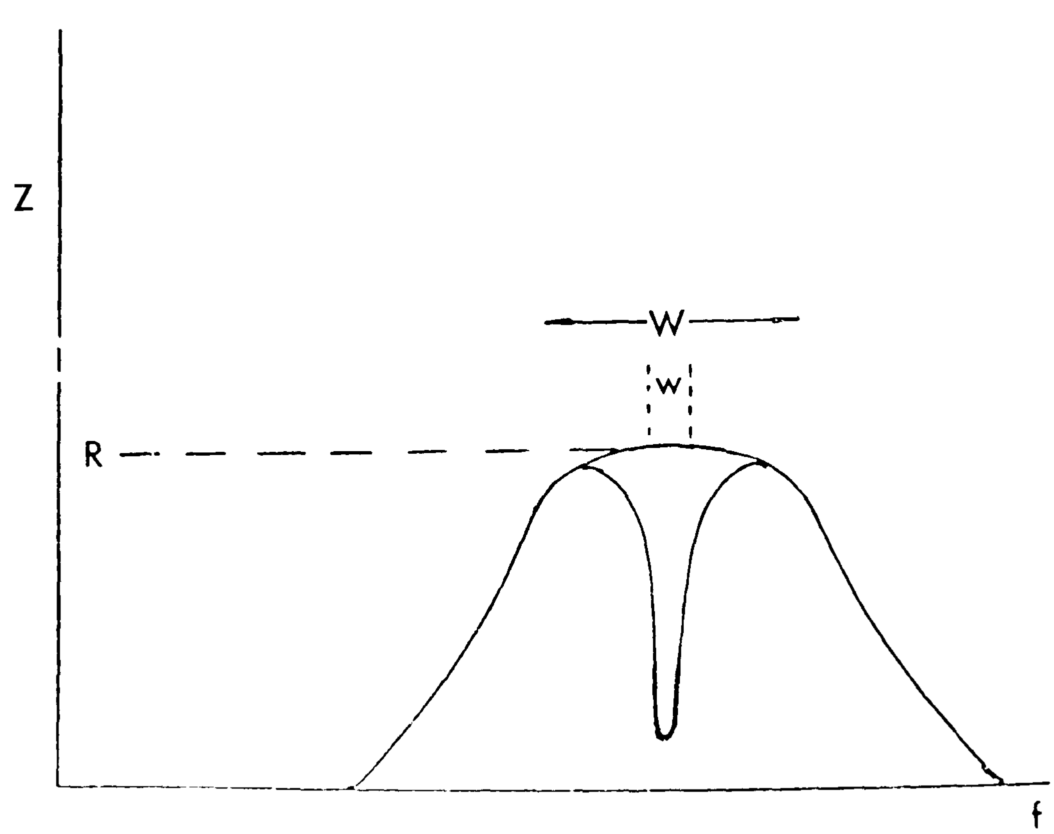
Thus provided $b > 4\sqrt{\frac{1000}{R_1}}$ we can neglect (c)

(a) gives a uniform noise $(\text{voltage})^2$ per Hz and its effect can only be reduced by increasing the signal.

(b) gives a noise $(\text{voltage})^2$ per Hz given by

$$b^2\ Z_1^{\ 2}$$

For a low Q parallel resonant circuit in parallel with a high Q series resonant circuit the impedance as a function of frequency for the maximum bandwidth condition is:



But this is not the impedance Z_1 we must consider (see later).

The $(\text{noise})^2$ per unit bandwidth will follow the square of Z_1 for a circuit without feedback and will thus concentrate in the sidebands outside the resonance of the series resonant circuit ($w \doteqdot \frac{40 \text{ kHz}}{70} \doteqdot 600$ Hz).

The signal, on the other hand, for a given induced emf, will be uniform over the wider band W of the band pass characteristic of the two circuits in parallel. If, for response time, such a wide bandwidth is required, the impedance of the circuit will determine the noise. We must however consider separately the influence on the impedance of the parallel circuit and its effective parallel resistance R; and of the parallel impedance R_{FB} that may be introduced by feedback.

We have signal $(\text{voltage})^2$

$$\left(\frac{V}{e}\right)^2 = \left(\frac{Z_p}{Z_p + Z_s}\right)^2$$

where Z_p includes R but may not provide the ideal maximum bandwidth condition.

We have noise $(\text{voltage})^2$

$$V_n^{\,2} = b^2 \left(\frac{Z_p Z_s}{Z_p + Z_s}\right)^2 + a^2$$

Thus $\left(\frac{S}{N}\right)^2$ ratio is

$$\frac{e^2}{b^2 Z_s^{\,2} + a^2 \left(\frac{Z_p + Z_s}{Z_p}\right)^2}$$

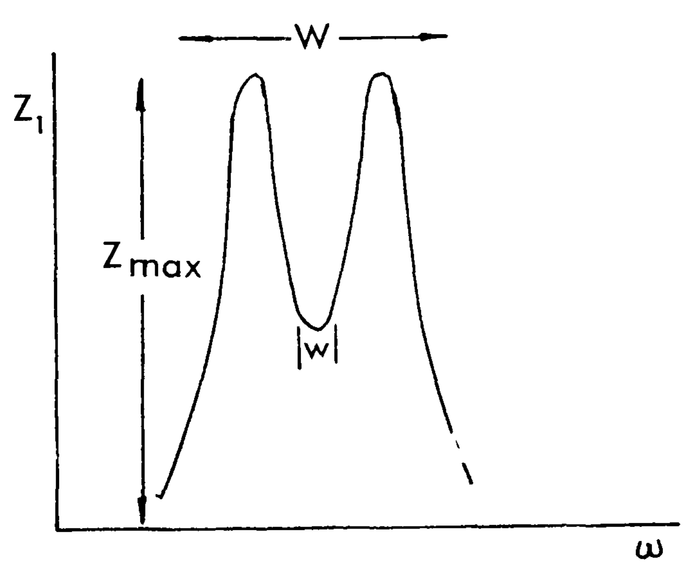
and cannot be improved except by choosing the parameters of the input stage to optimise the denominator. Note that Z_p should be as high as possible, i.e. a high Q parallel circuit damped only by negative feedback which will not change the $\frac{S}{N}$ ratio. The $\frac{S}{N}$ ratio is however greatly helped when "a" is important if there is a resonance between Z_p and Z_s as in the antiresonant peak of the crystal itself. The denominator we wish to make minimal can be rewritten

$$b^2 Z_s^{\,2} + a^2 Z_s^{\,2} \left(\frac{Z_p + Z_s}{Z_p Z_s}\right)^2$$

Since Z_s is given, it is now clear that we wish to minimise

$$b^2 + \frac{a^2}{Z_1}$$

The impedance Z_1 of the combined circuit (before it has been damped by the addition of R_{FB} to give the maximum bandwidth) should therefore be as high as possible at all frequencies and "a" and "b" should be optimised with regard to this high impedance.



If we say the frequency band w corresponds to low modulation frequencies and is less important it will be the Q_p of the undamped parallel circuit that will largely determine Z_{max}.

Suppose, for example, the parallel circuit has a Q of 20 and C = 1400 pf. Its peak impedance will be

$$\frac{Q}{\omega C} = \frac{20.10^{12}}{2\pi\ 40.10^3.1400} \doteqdot 50\ k\Omega$$

The peak impedance of the combined circuit (merely two peaks due to interaction instead of the original one) will have a similar impedance and are located at approximately $\pm$ 8 kHz. The impedance will thus vary from 4400Ω for the sidebands corresponding to middle frequencies to 50 kΩ or so for sidebands corresponding to higher modulation frequencies around 8 kHz. If a and b are optimised for the higher impedance, the noise at medium modulation frequencies will be high. If optimised for the medium frequency and impedance range, the noise will not be as good as it could be at high modulation frequencies.

APPENDIX 2

Tabulated Results of Auditory Localisation Experiments with the mobility aid

I.A.D. and I.T.D. Splay Angle 30 degrees (Fig. 8.6)

ANGLE	MEAN	ABS.E	E	RMS	S.D
-60	-61.5	14.1	-1.5	17.2	17.1
-50	-53.5	10.4	-3.5	14.5	14.1
-40	-51.6	13.8	-11.6	16.6	11.9
-30	-39.9	12.9	-9.9	15.8	12.2
-20	-27.4	13.4	-7.4	15.1	13.2
-10	-19.1	11.1	-9.1	14.6	11.4
0	0.9	4.5	0.9	5.5	5.5
10	16.0	10.1	6.0	12.8	11.3
20	40.4	20.4	20.4	25.3	15.0
30	49.5	20.6	19.5	26.0	17.2
40	63.2	24.2	23.2	28.5	16.5
50	57.7	12.2	7.7	16.0	14.0
60	66.2	7.3	6.2	9.9	7.7

Overall RMS 17.9   Mean SD 12.9

$$\text{ABS. Error} = \frac{\Sigma(|\text{Indication-Target}|)}{\underset{(4)}{\text{No. of Subjects}} \times \underset{(4)}{\text{No. of Times}}} \qquad \text{Error} = \frac{\Sigma(\text{Indication-Target})}{\text{No. of Subs.} \times \text{No. of Times}}$$

$$\text{RMS} = \sqrt{\left[\frac{\Sigma(\text{IND-TAR})^2}{\text{No. of Subs.} \times \text{No. of Times}}\right]} \qquad \text{SD} = \sqrt{\left[\frac{\Sigma(\text{IND-MEAN})^2}{\text{No. of Subs.} \times \text{No. of Times}}\right]}$$

I.T.D. Only (Fig. 8.7)

ANGLE	MEAN	ABS.E	E	RMS	S.D.
-60	-56.4	8.2	3.6	11.2	10.6
-50	-50.5	13.1	-0.5	15.2	15.2
-40	-48.2	14.6	-8.2	20.4	18.6
-30	-39.0	9.5	-9.0	12.4	8.6
-20	-27.9	12.6	-7.9	15.0	12.7
-10	-13.2	7.4	-3.2	9.0	8.4
0	-3.9	8.0	-3.9	10.7	10.0
10	3.7	8.2	-6.3	10.6	8.5
20	23.9	10.5	3.9	13.3	12.7
30	34.4	8.2	4.4	10.2	9.2
40	47.9	12.1	7.9	16.1	14.1
50	49.8	8.7	-0.2	12.6	12.6
60	51.8	12.8	-8.2	15.6	13.2

Overall RMS 13.5 Mean S.D. 11.9

I.A.D. and I.T.D. Splay Angle 60 degrees (Fig. 8.8)

ANGLE	MEAN	ABS.E	E	RMS	S.D.
-60	-53.7	8.9	6.3	12.1	10.3
-50	-50.2	6.4	-0.2	8.1	8.1
-40	-47.6	8.0	-7.6	10.2	6.8
-30	-48.4	18.9	-18.4	21.8	11.8
-20	-35.0	17.5	-15.0	22.7	17.0
-10	-19.4	17.6	-9.4	21.0	18.8
0	-5.6	14.2	-5.6	19.1	18.2
10	12.0	11.6	2.0	15.6	15.5
20	38.2	18.4	18.2	23.8	15.2
30	52.0	25.4	22.0	27.6	16.7
40	46.2	10.7	6.2	12.6	11.0
50	59.1	12.7	9.1	14.4	11.2
60	51.1	12.9	-8.9	17.2	14.8

Overall RMS 18.3 Mean S.D. 13.5

I.A.D. Only Splay Angle 30 degrees (Fig. 8.10)

ANGLE	MEAN	ABS.E	E	RMS	S.D.
-60	-43.0	19.7	17.0	24.3	17.4
-50	-43.2	12.1	6.8	16.9	15.4
-40	-45.7	13.7	-5.7	16.7	15.7
-30	-41.1	17.1	-11.1	19.2	15.6
-20	-21.7	9.1	-1.7	10.8	10.7
-10	-9.7	6.4	0.3	8.1	8.1
0	-1.0	5.1	-1.0	6.3	6.3
10	4.0	6.7	-6.0	9.8	7.7
20	16.4	6.9	-3.6	8.9	8.1
30	34.1	15.0	4.1	19.1	18.6
40	39.9	12.8	-0.1	15.2	15.2
50	39.2	14.9	-10.7	16.2	12.1
60	47.1	16.0	-12.9	18.9	13.8

Overall RMS 15.5 Mean S.D. 12.7

I.A.D. Only Splay Angle 50 degrees (Fig. 8.11)

ANGLE	MEAN	ABS.E	E	RMS	S.D.
-60	-53.2	11.3	6.8	14.5	12.8
-50	-50.1	12.9	-0.1	15.0	15.0
-40	-48.7	13.2	-8.7	15.7	13.0
-30	-44.6	18.3	-14.6	22.0	16.5
-20	-29.0	13.6	-9.0	16.9	14.3
-10	-17.6	9.4	-7.6	12.1	9.4
0	-3.9	10.0	-3.9	13.6	13.0
10	0.4	13.9	-9.6	17.2	14.3
20	30.6	16.2	10.6	22.1	19.4
30	32.4	14.8	2.4	19.5	19.3
40	46.9	17.4	6.9	19.6	18.4
50	43.8	17.4	-6.2	19.6	18.6
60	51.1	12.6	-8.9	17.0	14.5

Overall RMS 17.5 Mean S.D. 15.3

APPENDIX 3

An additional auditory localisation experiment

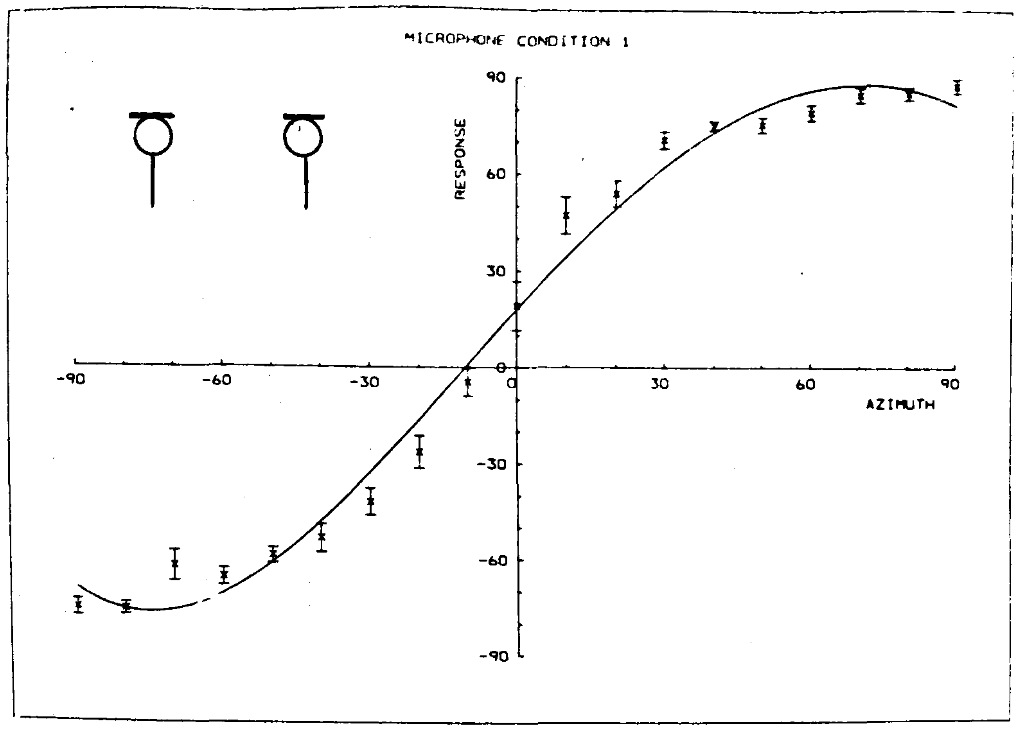
The localisation experiments carried out by Rudlin[16] are fundamental to the design of the binaural information display for the mobility aid. Furthermore, they are of relevance to work being carried out by Heyes[60] on a stereophonic hearing aid for the hard of hearing blind.

In view of the discrepancy that has arisen between the results obtained by Rudlin using cardioid microphones and those obtained with an experimental version of the mobility aid, Heyes has recently carried out a repeat of Rudlin's experiments using very similar experimental conditions. The sounds were presented in the sector 90 degrees to either side of the straight ahead position and the subjects indicated the direction of the perceived sound by the pointing technique. The results obtained are shown in Figs. (a), (b) and (c). Each graph indicates the orientation of the microphones and thus the interaural information presented to the subjects. The azimuth/response curves have been obtained by fitting the best third order polynomial to each subject's data in turn and then calculating the means of the coefficients. A comparison of the curves for the three conditions, namely: interaural time information alone, interaural intensity information alone, and a combination of interaural time and intensity information, shows no marked difference between the responses and so prevents the formulation of a definite conclusion. This prompted Heyes to repeat the conditions for interaural intensity differences alone and a combination of interaural time and intensity differences with different microphone orientations. These are shown in Figs. (d) and (e) together with the resulting curves. A comparison of these two curves clearly shows that the response

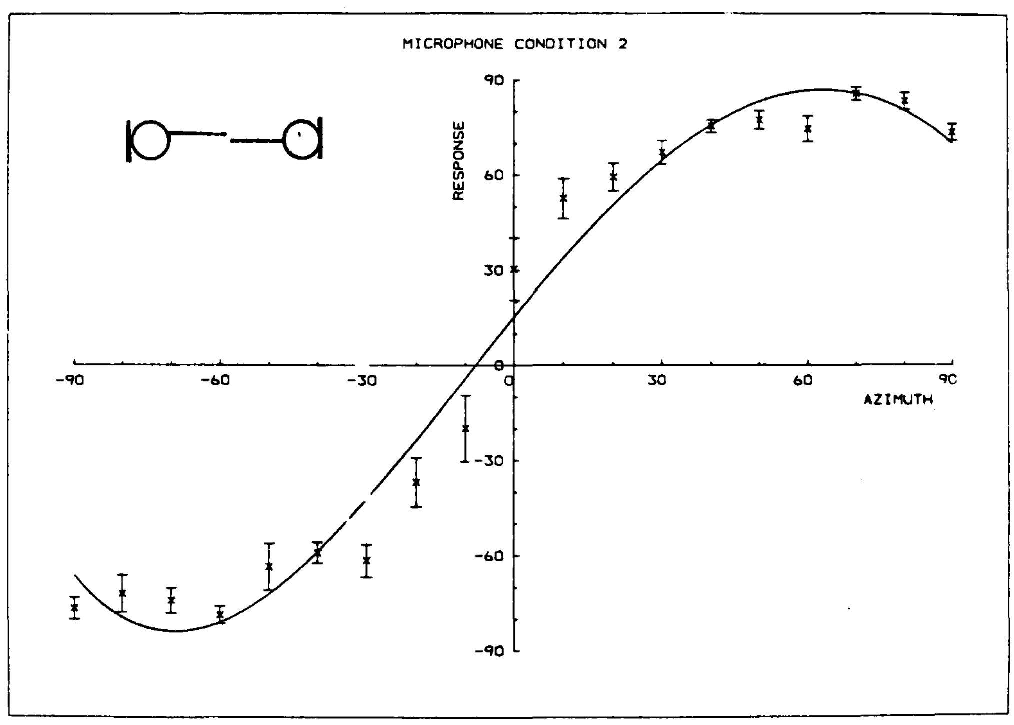
Appendix 3

Figs. A.3 (a)-(f)

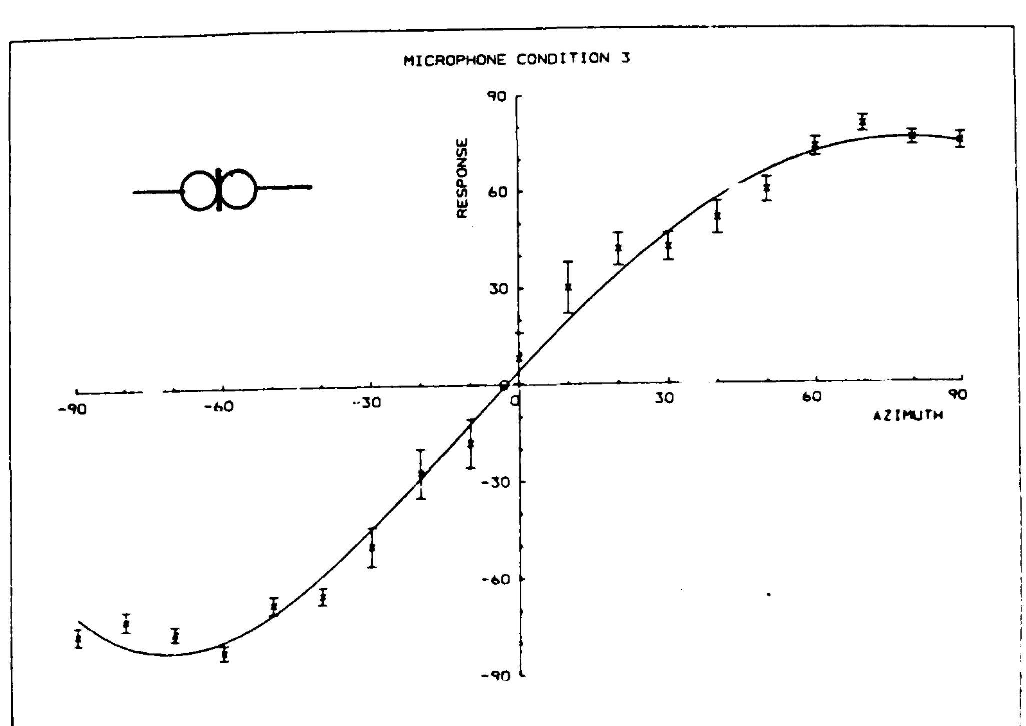
Figs. A.3(a)-(c)



(a) Interaural time information alone

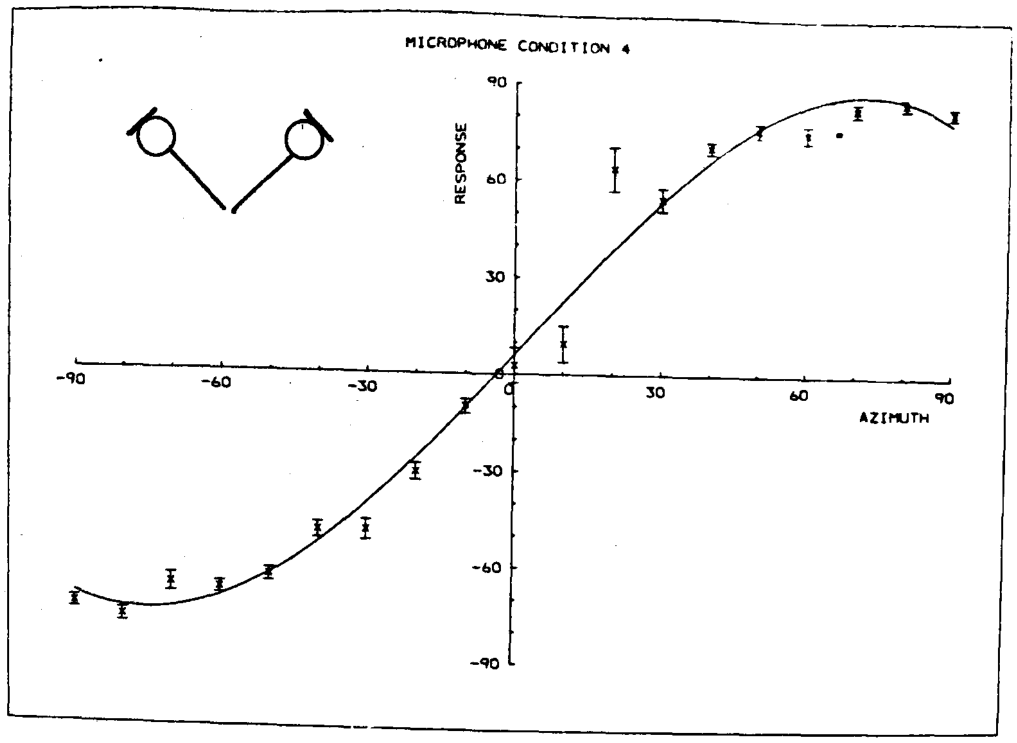


(b) Interaural time and intensity information (180°)

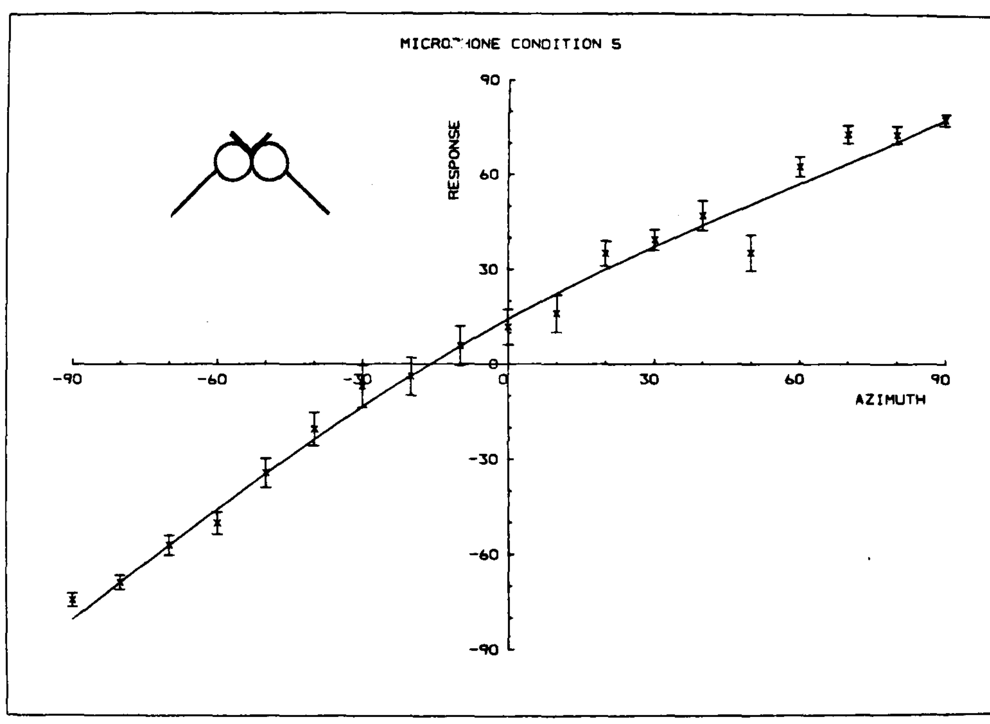


(c) Interaural intensity information alone (180°)

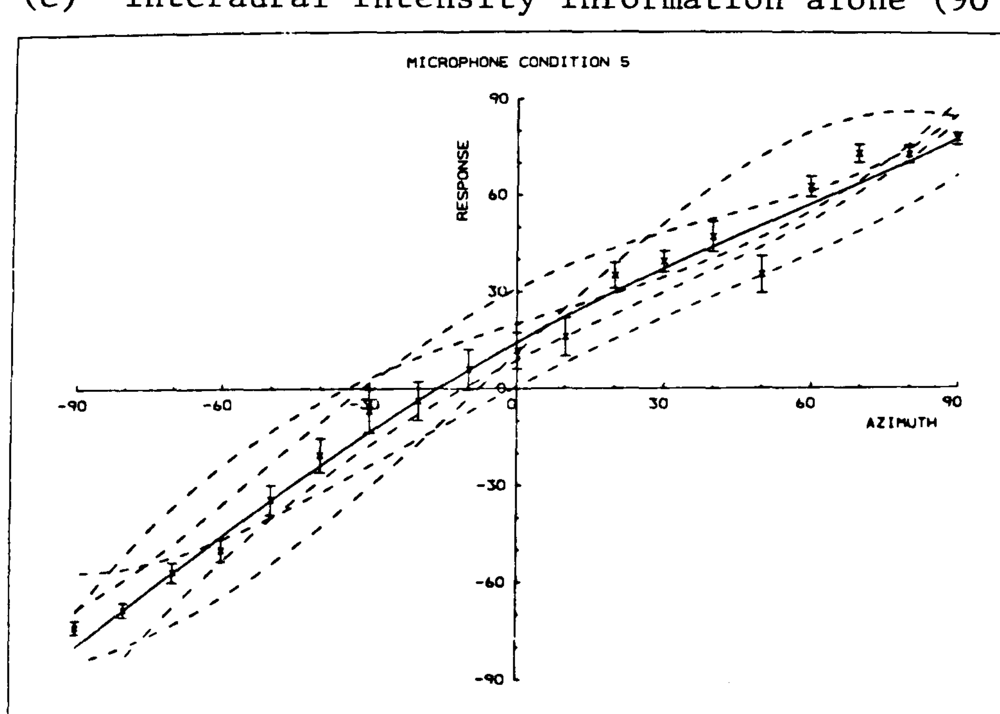
Figs. A.3(d)–(f)



(d) Interaural time and intensity information (90°)



(e) Interaural intensity information alone (90°)



(f) Interaural intensity information alone (90°)
showing individual subject polynomials

for interaural intensity information alone is more accurate than that for a combination of interaural time and intensity information. Fig. (f) shows the individual subject polynomials for the condition shown in Fig. (e) as well as the mean polynomial.

Summarising Heyes' work, it appears that his first three experiments, which were carried out under very similar conditions to those used by Rudlin, have given results which do not permit the same conclusions to be reached as those reached by Rudlin. Furthermore, the additional experiments performed clearly show that for a pair of cardioid microphones crossed at 90 degrees, more accurate localisation cues are obtained with interaural intensity differences alone rather than a combination of interaural time and intensity differences.

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