UNIVERSITY OF SOUTHAMPTON

INSTITUTE OF SOUND AND VIBRATION RESEARCH

THE SCIENTIFIC BASIS FOR THE FITTING OF HEARING AIDS

by

Michael Wald

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GLOSSARY

AVELS	Audio and visual everyday listening situation test
AVQ	Audio visual presentation in quiet
AVN	Audio visual presentation in noise
AQ	Audio only presentation in quiet
AN	Audio only presentation in noise
FADAST	Four alternative auditory disability and speech reading test
SPIN	Speech perception in noise test
QCL	Quietest comfortable level
MCL	Most comfortable level
LCL	Loudest comfortable level
LDL	Loudness discomfort level
SBCL	Speech band comfortable level
MHA	Master hearing aid
KEMAR	Knowles Electronics manikin for acoustic research

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UNIVERSITY OF SOUTHAMPTON

ABSTRACT

FACULTY OF ENGINEERING AND APPLIED SCIENCE
INSTITUTE OF SOUND AND VIBRATION RESEARCH

Doctor of Philosophy

THE SCIENTIFIC BASIS FOR THE FITTING OF HEARING AIDS
by Michael Wald

There has been slow progress in the development of a scientific understanding of the fitting of hearing aids. This has been partly due to the lack of a systematic multi-disciplinary approach to the problems involved. This thesis attempts to overcome these deficiencies by drawing together present day knowledge and technology to develop feasible clinical and research techniques, equipment and test procedures.

Microcomputer controlled digital filters and attenuators were incorporated into a specially designed Master Hearing Aid (MHA) which enabled any desired frequency response characteristic to be delivered to the ear of a patient. Probe tube microphone ear canal measurements provided precise individual real ear audiometric and hearing aid response data. Computer interfacing of an analogue hearing aid test system allowed all acoustic measurements to be integrated on to a computer data base which facilitated hearing aid selection procedures. An Audio and Visual Everyday Listening Situation (AVELS) videotape was developed which allowed both objective and subjective measures of sentences spoken in various signal-to-noise ratio and audio and visual contexts to be used to evaluate hearing aids. An adaptive pair comparison procedure using the MHA and employing audio recordings of continuous passages of speech, was developed to determine just noticeable differences between hearing aid characteristics, and also find the subjectively judged optimum response.

The MHA, AVELS videotape and pair comparison procedures were evaluated on both normally hearing and hearing-impaired subjects, and were shown to provide a practical and flexible method of producing and evaluating optimum hearing aid responses. The pair comparison procedure was a sensitive test of noticeable differences between hearing aids, whilst the AVELS videotape gave a more realistic indication of the importance of these differences in everyday listening situations. Hearing-impaired subjects were found to be less sensitive to changes in frequency response and distortion levels than normally hearing subjects. The results also suggested that the precision obtained using the real ear measurements may be unnecessary for many hearing-impaired subjects. Results of an experiment designed to compare hearing aid fitting procedures pointed to a flaw in widely accepted present fitting procedures that has been overlooked by previous studies.

The widespread use of the tests and procedures developed in this research study would enable a feasible, scientifically based, hearing aid evaluation procedure to be established in both hearing aid clinics and research laboratories, thus better facilitating the continuous step-by-step development of a scientific basis for the fitting of hearing aids.

CHAPTER 1

INTRODUCTION

1.1 General Background

There has in the past been much research concerned with the alleviation of the communication problems of the hearing impaired by means of amplifying speech and environmental sounds. This research has by necessity encompassed a wide range of topics and disciplines, including the diagnosis and specification of the auditory and psychoacoustic deficiencies caused by the hearing impairment, the signal processing and amplification regimes applicable to these problems, and the methods of evaluating the benefits or otherwise of the use of these regimes.

Considering the time and effort that has been spent on a scientific understanding of the problems of fitting hearing aids, it would appear that there has been a very slow acquisition of quantifiably useful knowledge. Braida *et al* (1979) in summing up their extensive review of past research in this field, stated that in their opinion one of the reasons for this slow progress had been the lack of multidisciplinary research staff involved in the research. The author, following his own extensive reviews of the research literature, would agree with their opinion. In many cases, audiometric, acoustic and electroacoustic variables have frequently been inadequately controlled and specified, which together with some lack of understanding of the true role of laboratory experiments has often led to conclusions drawn from these tests being inappropriately applied to real life situations.

Realising the problems involved, an attempt has been made in this thesis to systematically draw together present knowledge and technology to develop feasible clinical and research techniques for hearing aid fitting and evaluation and to use these techniques to examine the scientific basis for the fitting of hearing aids. The author, with his wide experience and background knowledge in most of the disciplines previously indicated as necessary for this approach, was felt to be suited to this task.

The techniques developed in this research study allow for both the accurate specification of acoustic and electroacoustic parameters and the evaluation of benefit gained from hearing aid use. They are therefore also of potential value in the wider fields of speech perception and speech processing research.

1.2 Review of Research Literature

The main research topics pertinent to this thesis are:

1. Speech testing methods and materials
2. Subjective evaluation procedures
3. Hearing aid fitting approaches
4. Acoustic and electroacoustic measurement techniques

The research literature in these topics is reviewed in Chapters 2, 3, 4 and 5 respectively.

In order to draw together existing knowledge for this thesis it was necessary to review many hundreds of references in a wide range of topics. It would not be either feasible or perhaps even necessary to present critical reviews of all this material, so wherever possible the reader will be referred to existing recent but relevant reviews of previous research from which only the salient points and conclusions will be distilled and discussed.

1.3 Aims of Research

The main aims of this thesis are as follows:

(i) The development of a clinically feasible protocol for the scientific fitting of hearing aids.

(ii) The development of hardware and software systems capable of carrying out the required experimental protocols.

(iii) The development of hearing aid evaluation procedures.

(iv) The determination of the relationship between audiometric profiles, hearing aid frequency responses and preferred gain settings.

(v) The study of the relationship between audiometric profiles and optimum hearing aid frequency response.

(vi) The examination of the importance of deviations from the optimum response and its relationship to audiometric measurements.

(vii) The examination and development of techniques for the measurement of real ear hearing aid performance.

The reasons for choosing these aims are discussed in detail in Chapter 6.

1.4 Design, Development and Testing of Systems and Procedures

To fulfil the aims of this research it was necessary to design and develop extensive novel hardware and software systems. These are described fully in Chapters 7 and 8 respectively. It was also necessary to develop new test material and protocols for the hearing aid evaluation procedures and this is described in Chapters 9 and 10. The design and development of these systems and procedures was a major task that took over two years to complete.

The developed tests and protocols were refined and 'calibrated' using normally hearing subjects. This baseline study is described in Chapter 11 and involved eighty hours of subject testing.

Chapter 12 presents the details, results and conclusions of the main study using hearing-impaired subjects. This study required seventy hours of subject testing and some fifty hours of adaptively setting up the individual experimental conditions.

The main findings and new contributions to knowledge provided by this thesis are summarized and discussed in Chapter 13.

CHAPTER 2

SPEECH TESTS

2.1 Introduction

Tests of speech recognition have in the past been used for many different purposes. They have been shown to provide diagnostic information concerning the pathology of hearing impairment (Boothroyd, 1968; Hood and Poole, 1971; Coles, 1972; Priede and Coles, 1976). They have also, for a long time, been used to evaluate the intelligibility of speech communication systems, initially over telephone links (Fletcher and Steinberg, 1929) and thereafter for the objective evaluation of hearing aids and hearing impairment (Fry and Kerridge, 1939).

Since then a large variety of speech tests have been developed, including the use of nonsense syllables (Levitt and Resnick, 1978), phonetically balanced word lists (Peterson and Lehiste, 1962), high frequency word lists (Pascoe, 1975), sentences (Silverman and Hirs h, 1955) and synthetic sentences (Speaks and Jerger, 1965).

These tests have usually been developed with a particular aim in mind. For example, closed-set response tests such as the Rhyme Test (Fairbanks, 1958), the Modified Rhyme Test (House *et al*, 1965) and the more recent FAAF test (Foster and Haggard, 1979) allow information to be obtained concerning consonant confusions. Discussion of the various speech tests can be found in many articles, reviews and books such as Egan (1948), Beranek (1949), Watson (1967), Lyregaard *et al* (1976), Markides (1977), Noble (1978), Levitt and Resnick (1978) and McCormick (1980) and need not be repeated here.

One of the aims of this research was to develop a procedure for evaluating hearing aids in a realistic way. Any speech test used for this purpose should therefore be capable of predicting real-life performance. Due to the current lack of straightforward speech tests that have been shown to predict real-life performance, it was decided to develop an audio visual sentence material test having a range of realistic signal-to-noise ratio conditions. A summary of the evidence favouring this approach will now be given but for an extensive review of research into audio visual and sentence tests the reader is referred to McCormick, 1980.

2.2 Word and Sentence Tests

Giolas and Epstein (1963) found that the scores obtained on word lists did not accurately predict how well an individual understood conventional speech. Niemeyer (1965) found little relationship between scores for word lists and on sentence discrimination tasks and reported sentences as a more reliable indicator of hearing handicap. Various sentence lists have been developed (Fletcher and Steinberg (1929), Fry and Kerridge (1939), Silverman and Hirs h (1955), Harris *et al* (1961), Fry (1961), Speaks and Jerger (1965), Watson (1967), Bench and Bamford (1979), Kalikow (1977a), Markides (1979), Ewertsen (1973), Utley (1946), Taaffe and Lowell (1957), McCormick (1980), Hasselrot (1974). Some of the sentence lists were devised for audio presentation, some for visual presentation and others for audiovisual presentation but the differences in test design approach necessary for these three modes of presentation are not well understood.

For the compilation of equally intelligible sentence lists in the past it has been necessary to adopt an empirical trial and error approach, with the task becoming extremely difficult if a balancing of tests in all of auditory, visual and auditory visual modes is simultaneously required. McCormick (1980) attempted to overcome this problem by presenting the single list of his Visual Auditory and Audio Visual (VAAV) test repeatedly, attempting to correct for learning and practice effects.

2.3 Visual and Audio Visual Tests

An understanding of what constitutes skill in lipreading and how this can be validly quantified is still very sketchy (Oyer and Hardick, 1976) despite much research into the lipreading of syllables (Erber, 1972; Shoop and Binnie, 1979), words (Berger, 1972; Binnie, 1974) and sentences (Taafe and Wong, 1957; Lowell, 1974). Systematic research into the interaction between auditory and visual skills in audio visual reception of speech has been scarce although the results of the work of such researchers as Erber (1971, 1974), McGurk and MacDonald (1976) and McCormick (1980) indicate the importance of this interaction.

Very little research has been undertaken in using sentence materials in audio visual speech discrimination. Ewertsen (1973), in Denmark, attempted to use questions requiring one word responses to evaluate real-life speech perception; however, he appeared to have little success, due partly to the problem of balancing his lists for equal difficulty. Plant *et al* (1982), following tests of an English translation of this test, felt that it would be of use as a rehabilitative procedure. However, little evidence was presented to support this view. Binnie (1974) using CID sentence materials found that the use of unrelated sentences was easier to apply and score than using everyday discourse materials.

2.4 Situational and Context Clues

Although it has been shown that situational or contextual clues improve comprehension (Garsteki and O'Neil, 1980; Garrett *et al*, 1980; Pelson and Prather, 1974; Smith and Kitchen, 1972), attempts to include these factors in reliable tests have been largely unsuccessful, principally due to the lack of knowledge and apparent complexities of the problem.

2.5 The Speech Perception in Noise Test

Duffy and Giolas (1974) investigated the relationship between word predictability and sentence intelligibility and concluded that "careful selection of key words based on their 'predictability status' is a method of controlling or influencing the intelligibility of sentences". This idea was taken up by one recently developed test (Kalikow *et al*, 1977a, b), the Speech Perception in Noise (SPIN) test, which has attempted to control for a large number of factors including word predictability, familiarity, context, rhythm and length structure, semantic, acoustic and phonetic factors.

Sentence lists were constructed using phonetically balanced, reasonably familiar key words of relatively homogeneous intelligibility in predictable and non-predictable contexts. The low predictable sentences of a carrier phrase type combined minimum predictability with variable rhythmic structure. The key words and sentence lists were tested empirically to establish that the design criteria had been carried out successfully.

The authors (Kalikow *et al*, 1977b) suggest that "some combination of performance on the PH (high predictability) and PL (low predictability) sentences in the SPIN test can predict some aspects of the ability of a hearing impaired individual to comprehend speech in everyday situations", and that "what is also needed is some measure of the extent to which performance on the SPIN test can be used to predict some measure of actual performance in the real world". They also discuss the possibility that the SPIN test might be used to ascertain the degree of success a patient may obtain in real life when fitted with a hearing aid, using questionnaires or a larger set of testing procedures that sample a wide variety of communicative situations that are presumably more representative of everyday speech reception. The original study compared the SPIN test lists for equivalence for both normally hearing subjects and hearing impaired subjects with a range of hearing impairments.

Although the SPIN test has not yet been used extensively in research or clinical contexts, a few studies have investigated its use. Owen (1981) showed that the difference between PH and PL sentences was significantly related to the subject's hearing loss and the S/N ratios used in the administration of the SPIN sentences. Morgan, Kamm and Velde (1981) looked at the equivalence of the forms of the SPIN test with normally hearing subjects and found that the list pairs of companion forms (which contained the same key words but in sentences of different predictability) and seven of the original 10 list forms were equivalent. They also obtained critical differences for test scores and reliability measures. Results were consistent with the critical differences for 25 and 50 items word lists calculated using a binomial model (Thornton and Raffin, 1978). Their findings therefore suggested that the test was reasonably well constructed in terms of equivalence and reliability.

Martin *et al* (1983) investigated the use of an audio visual version of the SPIN test, with no competing message, in assessing the high and low predictability word discrimination in profoundly and totally deaf patients under different conditions of lip reading alone, and lip reading in combination with a prosthetic device. This device was either a multiple channel cochlear implant, a powerful hearing aid or a simple hand-held vibrotactile device. The image of the speaker's face on the monitor screen was adjusted to be approximately life size. The results showed that the PH sentences were more easily lip read than the PL sentences, and that some subjects obtained no benefit from the use of an auditory or vibrotactile device in addition to lip reading. The test appeared to be fairly difficult, scores ranging between 12-46%. It was felt that "it is important to establish criteria of speech intelligibility by which to compare a multiple channel cochlear implant with the patient's performance with a hearing aid, and to provide the patient with information by which he can make his choice"; the SPIN test sentences used as a lip reading test appeared to go some way in fulfilling this function.

An attempt to examine the linguistic predictability of the BKB sentences developed by Bamford, Kowell and Bench (Bench and Bamford, 1979) has been made by Pearce and Coles (1980). However, this was only a written exercise using university students and no studies have been undertaken of the use of the spoken BKB sentence test either audio visually or with respect to its predictability.

McCormick (1980) developed the VAAV test which presented 10 sentences in visual, auditory, and audio visual mode and scored the sentences phonetically. Great trouble was taken to make the sentences everyday in nature, phonetically balanced and of homogeneous length with a variety of structures. The test proved successful as a means of determining the outcome of a rehabilitative programme when corrections for learning effects were made. In describing the reasons for the development of this VAAV test of everyday sentences, McCormick states that whilst the SPIN test appears to relate more closely to everyday communication experience than any other test, he was not able to use it as the details of the test had not been released when his research was being planned.

2.6 The Use of Live and Recorded Speech Tests

McCormick (1980) found no significant difference between audio visual scores on colour or black and white video recorded or live presentations. Some previous studies (Winkelaar *et al*, 1976; Jeffers and Barley, 1971) had found real-life presentation easier, while a recent study (Ronnberg, 1983) claimed to have found an interaction between the type of hearing loss, medium of presentation and type of material. McCormick carefully matched his live and recorded presentations for image size, quality and consistency of presentation and felt this was an important reason for the similarity in his results between presentation modes.

The suggestion from previous research is therefore that a recorded presentation can be successfully made to simulate results of a live presentation if the live presentations are controlled in certain ways. Obviously if the subject was allowed to see more of the presenter or was allowed to move his position during the test in the live presentation mode, then a difference in results might be expected.

2.7 The Use of Competing Noise in Speech Tests

A large proportion of everyday listening situations involve listening to speech in a background of noise. Pearsons *et al* (1977) report on the various noise levels and signal/noise ratios encountered in different environments, the S/N ratios in real-life situations rarely being worse than 0 dB. Licklider and Miller (1951) claimed that S/N ratios should exceed 6 dB for satisfactory communication. However, this will obviously depend on many variables including the listening ability of the subjects, the material involved and whether visual or contextual clues are available.

Various attempts have been made to characterise typical noise spectra, e.g., Aniansson (1974), Niemeyer (1976), and many studies compromised by using white or pink noise due to its ease of production. Speech shaped noise and occasionally speech shaped noise amplitude modulated to follow the amplitude envelope of the speech have been employed in attempts to make speech tests more difficult or more realistic.

The effectiveness of competing speech has also been shown to depend on the number of competing speakers (Carhart *et al*, 1970, 1975) and multi-talker babble was used as the competing message/background noise for the SPIN test due to its face validity of being a real-life problem.

2.8 The Speech Spectrum and Speech Levels

The spectrum of speech has been measured by many workers in the past, e.g., Dunn and White (1940); Pascoe (1978); Byrne (1977); Niemoeller *et al* (1974), but the most extensive study of the speech spectrum and speech level used in everyday life has been that of Pearsons *et al* (1977). The range of spectra they present explains to some extent differences in previous studies. These could be due to such things as vocal effort, speaker differences, type and length of material spoken, recording set up and equipment (e.g., microphone position), the number of speakers averaged and the analysis method.

There appears to have been some misunderstanding by some past research workers as to the differences and applicability of % bandwidth spectra of speech (e.g., 1/3 octave) and level per cycle spectra of speech. Byrne has attempted to correct the speech spectrum for critical bandwidths, feeling this is a better indication of how the ear would perceive the speech. However, the modifications to a 1/3 octave analysis are minor.

The averaging technique used to measure the speech will also affect the results; many different and often inadequate methods having been used in the past, such as rating the level of frequent peaks on a sound level meter set to r.m.s. which is inevitably prone to lack of precision and observer biases and errors.

Methods that average spectra between individuals blur the individual characterization of a particular speaker (e.g., main formant peaks) and can lead to an average spectrum that is unrealistic in terms of any individual speech spectrum. The variations in speech spectra that have been shown to occur (Pearsons *et al*, 1977) due to vocal effort indicate that the practice of electronic attenuation of speech levels for diagnostic test purposes should not be applied if real-life spectra and levels are required. To simulate a decrease in level due to distance actual distance recordings must be made, while to simulate a change in level due to vocal effort actual recordings of different vocal efforts should be made.

2.9 Methods of Scoring Speech Tests and the Reliability of Speech Tests

Speech tests have generally been scored by scoring individual phonemes, or individual words. The former approach has the advantage of supplying a greater number of scored items in a given test and allowing information to be obtained regarding the actual phonemes substituted or omitted. The latter approach has the advantages of speed and simplicity. In scoring sentence material generally a key word approach has been used (Kalikow, 1977) again for ease of administration. However, McCormick (1980) designed his VAAV test to use a phonemic scoring approach of every phoneme in the test. This required taping the response for later analysis, and also the unwieldly task of phonemically balancing every phoneme in the task, while retaining the everyday nature of the sentences.

In evaluating the results of speech tests, whether for diagnostic purposes, hearing aid evaluations, handicap evaluations or evaluating the outcome of rehabilitative procedures, it is necessary to take into account the variability inherent in speech discrimination measurements. One important source of variability can be due to differences between lists on a speech test. Attempts can be made to minimize these inequivalences (Campbell, 1965; Hood and Poole, 1977). However, differences between lists for individual subjects can still exist. Problems caused by list differences can be largely overcome by using a closed response set format as the same words can be repeated. However, this is difficult with sentence materials and so a compromise must be made whether to use one list and adjust for learning effects or use multiple lists and attempt equivalence between lists.

Statistical fluctuations are another important source of variability in speech tests, exactly the same score being unlikely to be obtained by a subject upon each presentation of a particular test. The practical applicability of mathematically modelling the performance on speech tests has been shown by Lyregaard (1973), Hagerman (1976) and Thornton and Raffin (1978) while Raffin and Thornton (1980) and Raffin and Schafer (1980) have published confidence levels for differences between speech discrimination scores based on this binomial model.

Much of the previous research into the reliability of speech tests for hearing aid comparisons has failed to take this mathematical/theoretical limitation of speech tests into account in their analysis of results. Rules of thumb have been assumed (e.g., 10%) for a 'significant' difference between hearing aids to have been obtained whereas actual critical differences can be calculated.

Dillon (1982) gives a comprehensive review of the variables involved in speech testing. In addition to the inter-list differences and statistical variation already mentioned he points out that discrimination ability may vary in time due to such factors as motivation, fatigue and amount of hearing.

He also points out that intra subject variability due to the factors mentioned above should not be confused (as it often has been) with inter subject variability. A low value of inter subject variability for normal subjects is not necessarily a desirable feature of a test, as it could suggest a test that is insensitive to differences between subjects. Dillon also points out that the use of correlation coefficients to examine test reliability is of little use, as high correlations will be obtained if a sample population with a wide range of speech discrimination abilities is chosen and a low correlation will be observed if similar discrimination abilities are exhibited.

The variability of speech tests containing items of different degrees of difficulty can also be predicted by binomial theory (Hagerman, 1976) once the difficulties and numbers of items are known. However, no attempt has been made in the past to apply binomial theory to the scoring of sentences where every item is scored (McCormick, 1980) as the individual items will no longer be independent and the binomial distribution will no longer be appropriate.

The conclusion to be drawn from previous research into this field is that to increase the reliability of a test the number of independent items must be increased, and therefore the length of the test and the time taken to administer it must also be increased. This obviously has implications for the application of speech tests in a clinical environment where time is extremely limited: e.g., Shore *et al* (1960) claimed that the tests available were not reliable enough to show up differences between hearing aids. This aspect will be further investigated in Chapter 3.

2.10 Continuous Discourse Tracking

A novel and interesting method for evaluating the reception of speech is a discourse tracking procedure which has been described by Filippo and Scott (1978). A person designated the 'talker' reads to a 'receiver' (the person whose receptive communication ability is being measured) a prepared text, segment by segment. The receiver attempts to repeat back what is read to him word-by-word after each segment or whenever he wants to interrupt. If the receiver is correct in his repetition, the talker reads the next segment, if there has been any error the talker must elicit the correct response from the receiver by any verbal means at his disposal before proceeding to the next segment. Non speech-related cues such as gesture are not allowed. The score is measured on the words per minute the receiver is able to repeat correctly.

The philosophy behind this test is that it has the face validity of being a realistic measure of actual communication exchange as takes place for the hearing impaired. Its developer felt it had many uses, such as training speech reception or production with aids to lip reading as well as the evaluation of hearing aids. They also pointed out that results on discourse tracking may differ from results from more conventional speech tests due to the testing of different skills.

There are, however, many problems in the actual implementation of discourse tracking techniques as a clinical test. Learning effects are large, both tester and receiver having to work together for a long period to learn which are the best strategies to use. Results are very dependent upon the type of material used and the interaction between the individual tester and receiver. The actual effectiveness of discourse tracking as a real-life communication simulation has not been validated and one could question the absence of allowable gesture and of context and situation cues. The task itself is fairly difficult to master, as well as possessing the unrealistic necessity to repeat everything that is said and is time consuming and frustrating to carry out. Due to the large learning effects and the great dependence on type of material, it would appear that very lengthy tracking sessions are required to get a reliable estimate of the subject's tracking ability, especially if results are to be used to compare hearing aids or rehabilitative regimes.

The tracking task by its very nature requires live face-to-face communication and cannot be used in a recorded form, therefore being very intensive in its use of talker/experimenter time and effort.

Continuous discourse tracking would therefore appear to be an as yet unvalidated, interesting and useful research tool with little immediate clinical application. Perhaps with further research it may in the future be developed into a standard evaluative procedure.

2.11 Conclusion

No existing speech test has been shown to successfully and sensitively predict the real-life performance of an individual. Although McCormick's VAAV test managed to demonstrate the improvement in performance of a group of subjects following a rehabilitative programme, it was not designed to predict differences between hearing aids for an individual. For this task it would appear that a test involving a wider range of realistic speech material and test conditions is required.

The Audio Visual Everyday Life Simulation (AVELS) test was devised for this purpose, and its development is described in Chapter 9. The AVELS test is based on the SPIN test sentence material as this is the most appropriate validated material at present devised.

CHAPTER 3

SUBJECTIVE EVALUATION OF HEARING AIDS

3.1 Introduction

Previous research into the subjective evaluation of hearing aids can be grouped under three main headings. Firstly, research to establish the reliability of subjective evaluations of speech intelligibility and quality as compared to 'conventional' objective speech testing (i.e., experimenter scored word and sentence tests of intelligibility); secondly, to examine subjective perception of the effects of different electroacoustic factors; and, thirdly, as a measure of user benefit or satisfaction of an aid to evaluate fitting procedures and protocols. The majority of research has been in the first category, comparing subjective evaluation with objective laboratory measures of speech discrimination. The interpretations of the results of these researches have often been influenced by the unfounded intuitive beliefs of those involved, that where subjective judgements differ from objective measurements, it is the former that are in error, whereas the 'validity' of the objective measures goes unquestioned.

3.2 Comparison of Subjective and Objective Measures

3.2.1 Review of research literature

Davis *et al* (1946 and 1947 in the Harvard Report), realizing the problems of measuring the statistical significance and reliability of differences on speech tests (but not applying statistical methods in their study) accepted the importance of subjective judgements of the quality of the aid, even though these were not always compatible with the greatest intelligibility. They also felt that intelligibility was not likely to be much impaired unless the quality was very poor. The majority of subjects preferred the quality of their own aid in the Harvard Study, even if articulation tests showed the Master Hearing Aid was better. Davis *et al* felt this preference might be due to subjects' familiarity with their own aid.

Although Davis *et al* advocated that use of a +6 dB/octave response for everybody was the 'best' fitting rule, they admitted that the selection of aids on the basis of subjective judgements did not give discrimination scores much worse than the 'best' aid in any case. This result appears to have been overlooked by many workers who have used or reported the findings of the Harvard Report. One of the main claims of the report was that apart from special cases the use of speech tests and other audiometric matching approaches were "either too arbitrary (confined to special conditions and requirements), too elaborate (impractical), or inconclusive (statistically unreliable)". There appears to be no finding in their report that would discredit the use of subjective judgements of hearing aids as a valid selection criteria.

Watson and Knudsen (1940) claimed that subjects' judgements of the optimum hearing aid frequency gain response were not reliable as they chose an aid response which accentuated most of the frequencies they already heard best. This claim, however, was not backed up by experimental evidence.

Although Carhart (1946a,b) used a large battery of speech discrimination tests, he also undertook a subjective evaluation which placed most emphasis on the results of a listening hour, where the subject heard a repertoire of music, speech and environmental noises and then rated the aid's performance. This session was regarded as serving both a screening and training function. Carhart claimed that the method was highly discriminative between instruments and therefore enabled the less desirable ones to be rejected. The final choice of aid which was made by the clinician, however, mainly depended on the results of the speech tests and tolerance tests. Zerlin (1962) reported a selection procedure where subjects were required to compare the intelligibility of pairs of recordings of speech which had been recorded through hearing aids. The subjects were able to switch between the pairs of recordings which each lasted 30 seconds. The subjects' preference rankings were compared with scores on speech tests and were found to be reasonably reliable, and sensitive to differences between aids.

Thompson and Lassman (1970) investigated the subjective preferences between flat and high pass amplification for subjects for whom they had previously measured the speech discrimination ability (Thompson and Lassman (1969)). They found that subjects preferred listening through the flat system in spite of having obtained better speech discrimination with the high pass system. However, the speech material was different for each test, and the improvements in speech discrimination using the high pass amplification were very small and probably not significant. None of the subjects were normally hearing aid users and Thompson and Lassman interpreted the results as perhaps suggesting that the subjects were preferring what they were used to rather than what gave them the best discrimination.

Gray and Speaks (1977) asked patients with sensorineural losses to make subjective judgements of the percentage intelligibility of connected discourse. They compared these judgements with various objective measures on CID W-22 word lists presented at a wide range of signal-to-noise ratios and levels and found reasonable correlations for only a few of the measures, these being at high S/N ratios and low sound pressure levels. The subjective judgements, however, proved highly repeatable.

Speaks *et al* (1972) found that normally hearing listeners could track and give reliable estimates of the percentage intelligibility of connected discourse which corresponded closely with objective scores on CID sentence tests. Their findings supported those of Pollack and Decker (1958) who, in addition, concluded that a six category subjective rating was better than a fixed binary yes/no decision about how confident a subject was that they had received a message correctly. Speaks *et al* concluded that a speech test could therefore be designed to predict the actual understanding of everyday speech. However, they do not discuss why this would be more useful than directly using the subjective judgements themselves.

Punch and Howard (1978) used paired comparisons of CID sentences recorded through hearing aids on Kemar and presented in quiet and in multi-talker babble. For subjective judgements of the quiet condition, listeners were required to select the aid based on clarity as it was assumed there would be 100% intelligibility, while for the babble condition they were to select based on 'understandability'. The subjective preferences gave better test-retest reliability than did the objective scoring of the sentence lists, which was poor and did not discriminate among aids. The test-retest reliability was better for the quiet than for the babble condition and the subjective judgements in one condition could not be predicted from the judgements in the other. This finding is perhaps not that surprising since subjects were making their judgements based on different criteria.

Tonning (1978) compared subjects' speech in noise scores with their subjective preferences obtained after talking to the subjects and letting them listen to environmental noises. None of the patients achieved better discrimination scores with aids they had rejected subjectively than with those they had preferred.

Punch and Parker (1981) evaluated the relationship between pairwise judgements of the quality and relative intelligibility of connected discourse and the measured intelligibility on a nonsense syllable test for subjects with sensorineural losses. Stimuli were processed and recorded through hearing aids and the Zwislocki coupler of an acoustic manikin. Relatively high test-retest reliability was obtained for all three measures, and a positive relationship found between relative intelligibility judgements and measured phonemic identification. The correspondence between quality judgements and either of the other two measures was not good and this was taken to indicate the importance of using the correct instructions if the choice of the aid with the best intelligibility was desired. The problem of reducing the potential reliability of subjective judgements by forcing subjects to make a choice by using a binary decision was pointed out by Punch and Parker. However, they favoured a statistical approach to overcome this, rather than use the subject confidence rating as suggested by Pollack and Decker.

Once again the researchers seem almost obsessed by the need to find subjective judgements that match speech scores, and even suggest that the use of nonsense syllables for the subjective preference tasks may improve the agreement. They seem to miss the point that the eventual aim of the exercise is to find the aids which give the best real-life performance.

Studebaker *et al* (1982), using both normally hearing and hearing-impaired subjects with a wide range of audiometric configurations, compared speech intelligibility judgements on continuous discourse processed by pairs of binaural hearing aids on a manikin, with scores on speech tests using words. Competing noise was used in all cases at 0 and +7 dB S/N ratios. Signals were replayed through hearing aid receivers and individual earmoulds, and were equalized to present the signals as if the subject was wearing the original hearing aid (i.e., the double ear canal resonance effect was removed by ensuring that the replay system presented a flat response when measured in a Zwislocki coupler).

Subjects were allowed to switch back and forth between the pairs of hearing aid processed speech and adjust the levels for maximum intelligibility. However, the lack of separate attenuators in the equipment used would suggest that changes in the relative intensity between any pair would be difficult to achieve, as they would have to be adjusted each time the switch was made.

Subjects were also asked to rate how confident they were about their choice on a five point scale ranging from "very clear difference" to "no difference". A tournament approach was used to obtain rankings between aids as all the aids could not be tried against each other due to the large number of conditions involved. Estimates of reliability from repeats of the pair comparison procedure and the speech tests showed that the subjective comparison produced more reliable aid rankings than the objective scoring, especially when the differences between aids in terms of discrimination scores was small. Confidence ratings were related strongly to the size of the performance differences between the aids. Studebaker appears to assume that speech tests are the valid method of selecting aids and examines how well the subjective judgements predict this selection. However, as the subjective judgements were made for connected discourse while the speech tests were word lists, the 'best' aid for one condition need not be the best for the other. This might go some way to explain the finding that "On occasions individuals chose aids that were clearly not best for them or the group".

Montgomery, Schwartz and Punch (1982), using data from bilaterally symmetric sensorineural loss patients obtained in a previous study (Punch and Parker, 1981), looked at the reliability of subjects' own estimation of intelligibility in the context of an elimination tournament, and correlated these judgements with nonsense syllable identification scores. They also examined the effects of 'seeding' and tournament structure on the outcome of the preferred aid. They found that while group performance was 'reliable' and correlated well with nonsense syllable identification, the distribution of tournament winners for individual listeners was "unreliable" and highly dependent on the effects of seeding. They admit that this was due partly to the similarity of aids employed. They also forced the subjects to make a binary yes/no choice even if no preference was held. They, too, drew the conclusion that if subjective ratings do not correlate well with speech scores there is something amiss with the subjective judgements; for example, they may not be judging intelligibility. They do not appear to consider that in comparing subjective judgements of connected discourse with scores on word lists, the latter may be an 'invalid' or 'unreal' approach.

Lawton and Cafarelli (1978) conducted a laboratory study to test the hypothesis that smoothing the frequency response and extending the bandwidth of the NHS BE-11 aid would improve both speech discrimination and sound quality for a broadly defined group of hearing-impaired adults. They used high frequency words in both quiet and noise to measure speech discrimination, and female continuous speech and semantic differential scales to evalulate subjective sound quality. Individual data was not analysed but group data showed that the smoothed wide band aid gave both the best speech discrimination and the best sound quality. The actual ranking of the other aids in the study did appear to some extent to vary between the speech in noise, speech in quiet and sound quality conditions. A follow-up study by Wald and Rice (1981) was carried out to investigate whether the laboratory findings of Lawton and Cafarelli would be substantiated when similar modified hearing aids and moulds were worn by a similar group of patients in their usual everyday situations, and so give an indication of the ability of the special laboratory tests to predict 'real-life' performance. To the author's knowledge this has been one of the few attempts to conduct both rigorous 'laboratory' and 'field' trials of hearing aids and to attempt to validate the laboratory methods as well as scientifically examining the reasons for subject real-life performances. The trial was conducted in the form of a pair comparison tournament, the subjects being allowed to take home and compare each pair of aids for one month in their everyday situations. The results of the field trials failed to substantiate those of the laboratory study; the best aid in the laboratory often being the worst aid in the field study, and *vice versa*. This apparent contradiction could perhaps be understood by considering the validity of the laboratory tests used, and the differences between the subjects in the two studies. A wide range of audiometric configurations of sensorineural loss had been used in both studies, but the mean losses at all frequencies were similar (within 10 dB). Closer examination of the field study results showed that those subjects with a hearing loss configuration more similar to those in the laboratory study tended to prefer the aid shown to be the best in the laboratory study. One implication of this is that in order to validate the real-life applicability of laboratory tests the same subjects should be used for both laboratory and field trials. Unfortunately this had not been possible for the Wald and Rice, and Lawton and Cafarelli studies.

The laboratory tests used by Lawton and Cafarelli were conducted in an anechoic room, using no visual, semantic or context clues, and employing high frequency weighted word lists with a criterion of 2% differences for statistical significance. They were therefore perhaps somewhat unrealistic in environment and content and although statistically significant, the small intelligibility differences involved may not have been noticeable in real-life situations. The results of these studies therefore show the importance of proving the real-life validity of laboratory tests before using them to make predictions about real-life aid performance. The development of such valid laboratory tests is one of the aims of this research study.

Harford and Fox (1978) compared the benefits of high pass extended bandwidth amplification on 9 subjects with sensorineural hearing losses by means of speech tests of monosyllable words in competing sentence noise, and by subjective judgements after a field trial. Subjective ratings of listening difficulty in various situations was rated on a 10 point scale and the ratings compared for the experimental aids and the subject's own aid. The high pass aid gave better speech scores for the majority of patients and was also preferred in the noisy situations encountered during the field trial. However, in the quiet situations the patients preferred their own aid. This suggests that the speech tests used predicted reasonably well the subjective preferences experienced in similar real-life situations. The results also suggest that a wide range of 'real-life' situations must be presented in a laboratory test as the patients' preferences may be different in different situations.

Haggard, Foster and Iredale (1981) followed a sample of postaural hearing aid users over an 18 month period subsequent to issue and found that questionnaire responses suggested in many cases reliable subjective benefits in specific listening situations where no advantage in speech discrimination in aided and unaided free field tests had been observed. They concluded that subjective questionnaire measures were important when evaluating overall effectiveness of aid provision.

3.2.2 Conclusions

The following conclusions can be drawn from previous research into the relationship between subjective and objective hearing aid evaluations:

(i) Subjective judgements of hearing aids can be reliable, especially if subjects' own confidence ratings are taken into account rather than forcing binary yes/no choices. A thorough comparison between the reliability of subjective judgements and objective speech tests has yet to be undertaken and would require careful consideration of many factors including time available, material used, instructions given, methodology employed and the range of hearing aids to be examined.

(ii) Subjective judgements do not necessarily correspond to objective discrimination score measures. The reasons for this lack of correspondence have yet to be thoroughly investigated, but one important reason would appear to be that different test materials have usually been used for the objective and subjective evaluations. In addition, instructions given for subjective evaluations have often been vague and ambiguous, resulting in the possibility that a wide range of different criteria may have been employed by those making the judgements (e.g., 'Quality' or'Preference' or 'Intelligibility').

The test conditions for the subjective and objective evaluations have often been different with apparently arbitrary signal-to-noise levels and noise types being employed.

(iii) Objective speech tests have appeared to give fairly poor predictions of real-life benefit and hearing aid preferences. The search for subjective measures that correlate well with laboratory speech tests would therefore appear to be of mainly academic interest. It would appear to be of more importance to develop measures either 'subjective' or 'objective' that correlate well with real-life performance and preferences.

It would appear that, due to our limited knowledge of the relationship between speech perception and real-life preferences, a subjective approach is, at present, the most promising one to follow.

3.3 Subjective Perception of Electroacoustic Factors

3.3.1 Interaction between stimuli and electroacoustical factors

Witter and Goldstein (1971) used normally hearing listeners to make quality judgements of pairs of male and female hearing aid processed speech. The hearing aids were specified in terms of frequency response and distortion measures. Frequency range and a crude measure of transient response seemed to be the best electroacoustic predictors of preferences. There appeared to be some interaction between stimuli and hearing aids in quality judgements although no definite conclusions could be drawn. Punch (1978) used hearing aid processed male and female speech as a paired comparison task for ten normal and ten high frequency sensorineural loss subjects with little or no experience of amplification. Five hearing aids were compared in this way and the preferences for the two groups of listeners were highly similar and largely independent of stimulus. The relationship between subjective judgements and electroacoustic characteristics proved somewhat equivocal, partly due to the problem of separating out the complex interactions that are perhaps possible between electroacoustic parameters.

The researchers felt that the test-retest correlations were not sufficiently high for all the conditions for adequate clinical reliability. This reliability, however, could not have been helped by the use of the ambiguous instruction "which is of better quality or which is the one you'd prefer to listen to for a long period of time?". Subjects were also forced to make a preference even if none existed.

Gabrielsson (1975a,b; 1979b) has used a multidimensional scaling approach to obtain subjective quality judgements. This approach while of academic interest, has to date provided little of immediate application to the fitting of hearing aids. His results have, however, shown there is an interaction between programme material and sound reproducing system. The implication of this in terms of subjective hearing aid evaluation is that it is important to use programme material that is important and representative for the hearing aid user.

The results of these studies also indicate that hearing-impaired subjects can make reliable judgements of subjective quality on a variety of dimensional scales. Gabrielsson *et al* (1980) also examined the relationship between sound quality judgements made by subjects when wearing their hearing aids at home with their judgements in the clinic and found a fair correspondence between them. The researchers in this study also point out that people's memory of sound quality is known to be poor and that trying to improve the sensitivity of the tests by using fast pairwise comparisons is not possible with real hearing aids and earmoulds. They state that they have no solution to this problem.

The procedures and systems developed during this research study do however solve this problem and are described in detail in Chapters 7, 8 and 10.

3.3.2 Distortion

Gabrielsson *et al* (1976) had earlier investigated the detectability of amplitude distortion of music, speech and sinusoids for normal and sensori-neural hearing-impaired subjects. 'Power series distortion' generated by means of an amplitude distortion synthesiser was used, and the task involved was to identify the recorded pair as same or different. The actual levels of distortion used will not be discussed here as they cannot be directly compared with hearing aid distortion. The hearing-impaired subjects in general had higher thresholds of distortion than the normal hearing subjects, there being big inter-subject variations within each group.

All comparisons were made at the subjects' most comfortable levels, which were similar for all stimuli.

3.3.3 Smoothing the frequency response

Various studies have indicated that the presence of peaks in the hearing aid frequency response may result in a deterioration of sound quality as judged subjectively (Gabrielsson and Sjogren, 1977; Lawton and Cafarelli, 1978; Killion and Tillman, 1982; Killion, 1979). Cooper (1984), however, using subjective ratings of hearing aid processed stimuli, found that there was an interaction of electroacoustic characteristics with stimulus material and that smoothed frequency responses were found to reduce perceived sound quality in some cases. The improvement in sound quality obtained by damping response peaks as demonstrated by the Lawton and Cafarelli anechoic laboratory study was not substantiated by Wald and Rice (1981) in their field study. Their results indicated that while smoothing the frequency respone resulted in an increase in the subjective 'smoothness' and 'mellowness' of the aids for many subjects, it also reduced the clarity and overall quality of the aids. This study drew attention to the necessity to consider the electro-acoustic characteristics of the aid as a whole as well as the audiological profile of the subject because what might be an undesired 'peak' to one person might be a preferred 'mid frequency boost' to another!

Bucklein (1981) examined the detectability of peaks and dips in the frequency response for normally hearing subjects using a variety of recorded stimuli in a binary yes/no pair comparison task. He found that peaks were more noticeable than dips, and that peaks when detected were always unpleasant; this was not so for dips. The audibility of small frequency response irregularities was rendered more difficult with solo instrumentals as test stimuli as opposed to continuous spectra stimuli. The detection only occured when the stimuli spectrum fell into the frequency region of the irregularity. A wide range of frequencies, 'widths' and 'heights' of the peaks and dips were used and no definitive rules for detection were formulated. However, 10 dB narrow peaks were generally detectable whereas 20 dB narrow valleys were not.

The interpretation of these results in terms of noticeable quality changes is equivocal as the relative levels between the stimuli used in this experiment were not varied. It is therefore possible that the change in loudness caused by the addition of a peak contributed to its detection. This hypothesis is supported by the study of Byrne, Christen and Dillon (1981) which showed for normally hearing subjects that the addition of peaks to the frequency response resulted in increases in perceived loudness. These changes corresponded closely with predictions based on Zwicker's loudness model. The studies of Lawton and Cafarelli (1978) and Wald and Rice (1980) found that subjects increased their hearing aid gain when the peaks were smoothed out, thus demonstrating the subjective change in loudness. Martin (1978) also found slight loudness and quality effects for the peaks used in his study on normal and hearing-impaired listeners.

3.3.4 <u>The effect of bandwidth limitation</u>

Franks (1982) examined the subjective effects of bandwidth on normal and hearing-impaired listeners using paired comparisons of hearing aid processed music. The normally hearing subjects indicated a perception of, and a preference for, extended range in both the high and low frequencies whereas the hearing-impaired subjects gave accurate perception and preference judgements for only the low frequencies. The hearing-impaired subjects were, however, required to listen through hearing aids and as binary yes/no decisions were required the results are possibly more an indication of the bandwidth of the hearing aids used than of the abilities of the hearing-impaired subjects. This study, like many others in this field, also failed to control properly the processing, recording and presentation of the stimuli and so did not present the signal to the ear of the listener as if he had been wearing the aid. The problems involved in the recording and reproduction of hearing aid processed signals will be discussed more fully in Chapter 10. Lawson and Chial (1982) obtained speech quality magnitude estimates (SQME) of high and low pass filtered speech and of various amounts of rectified distorted speech. The subjects were presented with a standard passage and then a distorted or filtered passage. The SQME varied systematically as a function of signal degradation by LP and HP filtering and linear rectification. Both the normal and hearing-impaired listeners (moderate high frequency losses) were capable of making reasonably accurate reliable judgements of signal quality. The hearing-impaired subjects appeared to be less sensitive to changes in low pass filtering and linear rectification while having similar sensitivity to high pass filtering.

Killion and Tillman (1982) compared music, speech and speech noise processed by experimental hearing aids with that processed by various sound reproduction systems including both headphones and loudspeakers. By careful recording and equalization techniques the recorded processed music was presented to the ears of the listeners with the same responses as the original sound reproduction systems. Subjects were required to compare how closely the various systems duplicated a high fidelity reference system by means of 'A.B.A.' comparisons, the reference system being heard both before and after the test system. Both 'trained' and 'untrained' normally hearing listeners were used and gave similar results and good test-retest reliability for their ratings.

The hearing aids gave the best fidelity ratings and were also the best electroacoustically in terms of smoothness and flatness of response. One hearing aid with an 8 kHz bandwidth rated as well as another with a 16 kHz bandwidth but a slightly less smooth response. This appeared to support the conclusions of Snow (1931) and Fletcher (1942) that an 8 kHz upper cut-off frequency gave substantially complete fidelity for orchestral music.

Killion and Tillman found that the trained listeners gave less inter-subject variability in their ratings and less programme-system interaction than did the untrained subjects. It therefore appeared that the trained subjects were a more homogeneous group in their ratings, and were apparently able to "listen through" the programme material to rate the 'quality' of the sound reproducing systems. However, it is important to note that subjects were not required to express preferences for the systems but just to compare them.

Muraoka, Iwahara and Yamada (1981) used a pair comparison procedure to find the detectability of high frequency cut-off by presenting pairs of stimuli to be judged as identical or different. Music was used for the source as they had previously shown that detectability did not depend upon the type of source, merely upon the abundance of adequate high frequency components. Levels were set according to listeners' individual preferences and no time limit was imposed in making the judgements. A large number of trained listeners with a wide range of ages were used. Few people managed to reliably detect cut-off frequencies higher than 14 kHz. No attempt was made to relate subjective preferences or quality judgements to these detectability measures.

3.3.5 Conclusions

From the results of previous research it would appear that the subjective effects of changes in the electroacoustic characteristics of hearing aids can depend upon the stimuli used, the test environment employed, the signal level, the presentation system, the criteria used to elicit the subjective judgements and the training and audiological profile of the subject.

Although there seems to be some consensus that smooth, wideband, low distortion hearing aids are generally preferred, there has actually been little conclusive quantifiable experimental evidence.

We can conclude therefore, that due to the limitations in present knowledge, if we required to determine in the laboratory or clinic the subjective importance that variations in the electroacoustic parameters of hearing aids have in everyday life, it would seem prudent to use aids, presentation methods, stimuli, environments, levels and criteria characteristic of real-life situations.

3.4 Implications for Research Study

Previous attempts to evaluate hearing aids subjectively have paid inadequate attention to the construction of reliable test procedures (conclusion 3.2.2(i)), appropriate judgement criteria (conclusion 3.2.2(ii)) and realistic test conditions and material (conclusion 3.2.2(iii)). The problems of presenting fast pairwise comparisons through real hearing aids and earmoulds have been found insurmountable (3.3.1) while the results of all past pair comparison procedures may have also been confounded due to subjects not being allowed the opportunity to equalize loudness levels (3.3.3)

Therefore past subjective evaluation procedures have been unsatisfactory due both to a lack of understanding of the acoustic and psychoacoustic principles involved and also to the lack of available test material and presentation systems. The tests and systems developed for this research study attempt to ovecome all these problems and are described in detail in Chapters 7, 8, 9 and 10.

CHAPTER 4

HEARING AND FITTING PROCEDURES

4.1 Introduction

With the recent advances of knowledge and technology in electronics, electroacoustics and earmould acoustics (Lybarger, 1980; Egolf, 1980; White *et al*, 1980; Killion, 1980, 1981, 1982a,b; Wald *et al*, 1984) virtually any required frequency gain response characteristic can now be produced in a wearable hearing aid. The most pressing problem would therefore appear to be to determine which characteristics are actually required for a particular individual.

Systematic research into the fitting of hearing aids has been undertaken for well over 40 years. In this time there have been many hundreds of research studies resulting in numerous scientific papers and other contributions by those who fit and evaluate hearing aids. It therefore appears surprising that in spite of this vast expenditure of time and effort there is little consensus as to the optimum approach to fitting hearing aids to hearing-impaired individuals. From studying the published literature it would seem that in addition to the complexity of the topic, the main factors that have contributed to this slow progress are inadequate personnel (see section 1.1), poor measurement techniques (Chapter 5), inappropriate tests (Chapters 2 and 3) and a lack of knowledge regarding the functioning of the hearing system (section 4.7). Due to one or more of these factors, many previous research findings become reduced to virtual "anecdotal' evidence so that they cannot be compared in a scientific way. It would therefore lend little to the understanding of the work of this research study, or to the readability of this thesis to include a detailed account and criticism of all previous work in this field. For this, the reader is referred to the many recent reviews of past research (Lybarger, 1978; Levitt, 1978; Studebaker, 1979, 1980, 1982; Harford, 1979; Braida *et al*, 1979; Byrne, 1979; Zelnick, 1982, 1983).

The use of speech tests or subjective judgements for hearing aid fitting has been discussed in Chapters 2, 3, 9 and 10. The main body of research into hearing aid fitting has been concerned with the possibilities of using audiometric measurements to select hearing aids for an individual, for even if speech tests or subjective judgements were favoured as the final method of selecting the optimum aid, some means of swift 'preselection' of aids would still be required.

In this chapter an attempt will be made to draw together the main findings of past research into hearing aid fitting based largely on audiometric measurements.

4.2 Research into Fixed Frequency Gain Characteristics

4.2.1 Research review

Davis *et al* (1947), in the Harvard report, compared the effects of response slope and high and low frequency cut offs using speech tests and subjective judgements. They concluded that a hearing aid characteristic with a slope of 0 to +6 dB per octave, a bandwidth from 300 Hz - 4 kHz, and limitation of the maximum output was the best characteristic for the subjects they studied. Minor variations from the ideal response were found to be relatively unimportant, and for the 'usual' hard-of-hearing patient any detailed fitting was deemed to be wasteful of time and effort. They admit that their findings did not necessarily hold for irregular or sharply sloping losses, or restricted frequency ranges. However, many of the 18 subjects used in the study had large conductive components to their hearing loss and therefore the results of this study may not be wholly applicable to the present-day hearing aid user population which has predominantly sensorineural losses.

It has been suggested that the limited number of hearing aid responses and the type of test material used may also have led to the finding that a 'fixed' aid response was the 'best' for everybody (Resnick, 1977). However, Sheets and Hedgecock (1949) using over 100 subjects with a wide variety of audiogram slopes and a commercial hearing aid, interpreted their results as supporting the Harvard study conclusions.

Radley *et al*, in the Medresco report (1947), conducted an extensive study to determine the optimum frequency gain characteristic and examined the effects of low and high frequency cut offs and slope of response. Based on the results of their tests and theoretical calculations, the authors concluded that the optimum response was one that rose at 12 dB/octave to 750 Hz and was flat or rising at 5 dB/octave above 750 Hz. The bandwidth necessary needed only to extend to 4000 Hz. Those subjects with hearing losses of less than 45 dB did equally well without the 12 dB/octave cut below 750 Hz.

Knight (1967) re-examined the findings of the 1947 Medresco report using insert earphones and hearing-impaired subjects selected according to pathology, and concluded, in agreement with the 1947 report, that marked changes of frequency response from the specified 'optimum' curve had a small effect on articulation efficiency.

The Harvard and Medresco studies' recommended optimum responses appear at first glance to be in close agreement. However, as pointed out by Resnick (1977), unlike the Harvard response the Medresco recommended response included a correction to account for head diffraction effects. If the correction was not used, the Medresco response would have a greater slope than the Harvard response. It is difficult to accurately compare the responses, however, due to the different measurement techniques involved.

4.2.2 Conclusions

The results of the Harvard and Medresco studies appear to indicate that a rising hearing aid response limited to an upper cut-off frequency of 4 kHz provides a reasonably good hearing aid fitting in terms of speech intelligibility for a large proportion of hearing aid users. The results of the studies also indicated that very sensitive tests would be required to show that changes in response result in reliable improvements in intelligibility. For the reasons already indicated it would seem unwarranted to draw more far reaching or specific conclusions than these.

4.3 Selective Amplification

4.3.1 Audiogram mirroring

Attempts to correct the impaired hearing threshold to normal by providing gains which mirrored the hearing loss were found not to be satisfactory very early on in the history of hearing aid fitting (Watson and Knudsen, 1940). The loudness discomfort levels of subjects with sensory hearing losses vary little from those of normally hearing people and so audiogram mirroring often leads to overamplification.

Pascoe (1975) conducted an experiment which used speech tests to compare hearing aid frequency responses specified by both functional gains measured using unaided and aided thresholds and acoustic gains in a coupler. His results showed that responses that mirrored their audiograms gave the best results for his eight subjects. However, although the shape of the frequency gain curve mirrored the audiogram, the overall gain level was well below that required to bring the aided thresholds down to normal levels. The subjects used had relatively mild losses with slopes of approximately 6 dB/octave and therefore had a reasonably large residual hearing area.

The mean slope of the audiogram mirroring functional gain was ≃ +6 dB/octave. This differed from the +6 dB/octave coupler response used in the experiment mainly in terms of the extra high frequency emphasis required to reinstate the 'lost' ear canal resonance (see Section 5.2). The other 3 responses tried in the experiment had the least high frequency emphasis and gave the worst results.

Skinner (1980) examined whether Pascoe's findings also held true for subjects with noise-induced hearing losses having normal thresholds below 1 kHz and losses greater than 40 dB above 2 kHz. She found that this was not the case; the subjects performing best when the level of speech energy above 2 kHz was 0 to 15 dB above that below 1 kHz. When the balance between the high and low frequency energy was greater than 20 dB there was a significant decrease in all the listeners' scores.

Providing amplification which causes the subjects' aided threshold to parallel the normal audibility curves (mirroring the shape of the audiogram) seems therefore to provide close to optimum intelligibility for mild or gently sloping losses but not for severe or steeply sloping losses.

4.3.2 Most comfortable equal loudness contour fitting

Watson and Knudsen (1940) found that a frequency gain response that mirrored the equal loudness curve at the most comfortable listening level for 1 kHz pure tones gave the best articulation scores. For conductive losses it was, however, little better than 'flat' amplification.

Davis *et al* (1947) claimed that it was impractical to carry the Watson and Knudsen procedure out clinically due to the time necessary to obtain reliable loudness matches, even though for the subjects used in their study the method predicted the optimum aid 90% of the time.

Pascoe (1975) stated that for the subjects in his study a response based on most comfortable equal loudness contours would not be much different from the optimum responses found.

Many recent studies have investigated the use of a slightly modified procedure from that used by Watson and Knudsen. This involves presenting a frequency gain response that amplifies the speech spectrum such that it follows the most comfortable equal loudness contour (Byrne and Tonisson, 1976; Shapiro, 1976; Skinner *et al*, 1982; Byrne, 1982; Skinner, 1983).

Byrne and Tonisson used a formula to estimate the most comfortable level (MCL) from audiometric thresholds, claiming in accordance with Davis, that equal loudness measurements were unreliable. Their prediction formula was derived from measurements of preferred listening levels (Byrne and Fifield, 1974) and supported by the data of Brooks (1973) and Boorsma and Courtoy (1974).

Similar approaches based on measurements of audiometric thresholds have been put forward by Berger (1979) and McCandless and Lyregaard (1983). All these prescriptive fitting methods involve some arbitrary adjustment at low frequencies to remove the possible effects of low frequency masking and are dependent on the actual speech spectra used.

Hearing aid fitting based on 'comfort levels' had in fact been patented many years before by Balbi (1935) who estimated the MCL by bisecting the difference between the threshold of hearing and the upper tolerance threshold. Lybarger (1944) also filed a patent application for a similar approach which, however, reduced the gain in the lower frequencies before mirroring the MCL.

Barford (1972) devised an interesting variant on the most comfortable equal loudness contour fitting approach. His method was to present the levels of speech that are exceeded 10% of the time at the same subjective loudness for hearing-impaired subjects as for the normally hearing. This method required the hearing-impaired subjects to have normal hearing at some frequency to enable the loudness balance to be carried out and therefore is of restricted application.

Some researchers have claimed that most comfortable levels must be measured empirically as they cannot be accurately predicted from threshold measurements (Watson and Knudsen, 1940; Shapiro, 1975; Kamm *et al*, 1978). Others (Christen and Byrne, 1980; Berger and Soltisz, 1981), claim that the low reliability of MCL measurements means that estimates based on threshold measures are better. However, Cox and Bisset (1982) examined the reliability of measures of the 'upper limit of the comfortable loudness range' and found it to be similar to that of measures of pure tone thresholds. Walden *et al*(1977) and Shapiro (1978) claim also to have shown that hearing aid users are reliable in their settings of comfortable levels.

The apparent disagreement by researchers over the use and reliability of suprathreshold measurements such as MCL may be due in part to the different procedures used. The stimuli, measurement technique, instructions and methods employed, have all been shown to affect the results (Dirks and Kamm, 1976; Stephens *et al*, 1977; Beattie and Cullibrk, 1980).

Apart from the work of Skinner (1983) there has actually been little research to investigate whether this most comfortable equal loudness contour approach produces the optimum response. The results published by Berger and Byrne only suggest that a 'satisfactory' fitting is obtained.

Lippman *et al* (1981) found that for the five subjects used in their study, presenting the speech spectrum to follow the Most Comfortable Level curve resulted in similar speech intelligibility scores to an audiogram mirroring approach but better subjective quality ratings.

4.3.3 Relationship between frequency response and preferred Listening Levels

There have been many studies that have attempted to relate the subject's 'most comfortable' or preferred gain settings to his threshold hearing loss (Brooks, 1973; Martin, 1973; Byrne and Fifield, 1974; Martin *et al*, 1976; Berger *et al*, 1980). The results of these studies have generally been interpreted as providing validation for the hypothesis that subjects' preferred gain levels are approximately half their hearing loss even though there were very wide individual variations in the results from all these studies. This data has been used to develop frequency specific hearing aid fitting 'prescription' approaches in spite of there having been little systematic study to examine how the preferred gain levels vary with frequency response. The understanding of the relationship between preferred levels and frequency response is important as it may enable the choice of a 'comfortable' response that provides the best intelligibility to be made. In this way, hearing aid fitting approaches based on audiometric measurements may be improved.

Byrne and Christen (1979) examined the most comfortable listening levels for various frequency responses for normally hearing subjects. They claimed that the effect of changing the frequency response was predictable and could be calculated in accordance with Zwicker's method (1960) for estimating the loudness of complex sounds. They also showed that Zwicker's method of loudness calculations satisfactorily predicted the loudness of low and high pass filtered speech as measured by Fastl (1977). Zwicker's procedure was originally devised for calculating the loudness of steady state sounds. However, Zwicker (1977) suggested that it could also be used for temporally variable sounds such as speech. This was based on the finding by Fastl (1977) that running speech and speech-like noise are found to be equally loud when the peaks of speech were equal to the steady state level of the noise.

Byrne *et al* (1981) examined the effects of peaks in hearing aid frequency response curves on the comfortable listening levels of normally hearing subjects. They found that the reduction in most comfortable level occurring when peaks were introduced agreed closely with predictions based on Zwicker's method.

The authors of these studies point out that the findings cannot necessarily be applied to hearing-impaired subjects. Wald and Rice (1981), however, found that damping out aid response peaks resulted in hearing-impaired subjects turning up the aid gain slightly in accordance with the results of Byrne *et al* (1981). Davis *et al* (1947) measured the change in gain relative to a flat response for comfortable listening for the various responses used in their study. The average differences for the hearing-impaired subjects, used in their study were found to be very similar to those for normal subjects and Davis *et al* concluded that "with respect to loudness matching at high intensity levels, the ears of the hard-of-hearing seem to function quite normally".

4.3.4 Conclusion

There appears to be a fair agreement in the research literature that providing amplification which presents bands of speech at the most comfortable level for that band gives a close to optimum hearing aid fitting. There appears to be some controversy over the reliability of comfort measures, which may be due to the variety of approaches adopted.

Attempts to predict preferred listening levels from audiometric threshold measures have generally not taken into account such important factors as the frequency response of the hearing aid and the subject's perception of loudness at suprathreshold for different frequencies and levels. The interactions for the hearing-impaired between frequency response, loudness perception and preferred listening levels have not been adequately studied considering the potential importance for the scientific fitting of hearing aids.

In the light of the agreement over the superiority of the 'comfortable level' approach to hearing aid fitting it is perhaps surprising that there have been very few studies that have actually tested its validity.

There has, for instance, been no examination of how the actual levels hearing-impaired individuals choose to listen to broadband speech relate to their most comfortable level for the individual bands. One would expect that the method of presenting the broadband speech signal such that each band of the speech is at the most comfortable level for that band would result in an overall level that was too loud, due to summation between bands. This theoretical consideration would therefore suggest that the most comfortable level equal loudness contour fitting method should not work, yet this has never before been pointed out by any worker in the field.

The author therefore decided to investigate in this study the detailed relationship between narrow band and broadband comfortable listening levels for various stimuli and frequency responses.

4.4 Theoretical Approaches to Hearing Aid Fitting

For normally hearing listeners the effect of varying the frequency-gain characteristics of speech transmission systems can generally be well predicted by articulation theory (French and Steinberg, 1947; Kryter, 1962). This uses a measure called the articulation index (AI) which is a weighted average over frequency bands of the proportion of the speech signal that is available to convey information to a given listener. The weights reflect the relative contribution of the bands to speech transmission. Adjustments are made to account for masking effects.

Radley *et al* in the Medresco Report (1947) attempted to determine the optimum frequency gain characteristic for hearing-impaired subjects based on articulation theory but ignoring inter-band masking of speech and assuming a limited total power. Fletcher (1952) analyzed the results of the Harvard study (Davis *et al*, 1947) according to a version of articulation theory that claimed to predict the experimental results well. However, both these two theoretical studies involved many arbitrary assumptions including the sensation level at which the speech is listened to.

More recently, studies by Macrae and Brigden (1973) and Aniansson (1974) have shown that articulation theory can, to some extent, predict the speech reception performance of hearing-impaired subjects. Dugal *et al* (1982) using data from the research of Skinner (1976) have attempted with some success to examine the predictions of articulation theory using proficiency factors to account for differences between listeners. They claim that a thorough investigation of the problems was possible for the first time as accurate information was available on the detection and discomfort thresholds of individuals, the speech spectrum used in the speech tests, the background noise, the presentation levels and the relevant parameters of the speech transmission system. In addition, the subjects used for Skinner's study had relatively homogeneous audiometric configurations.

However, before articulation theory could be successfully widely used to predict optimum hearing aid frequency responses, much more knowledge must be gained into the importance and effects of such things as listening conditions, type of speech material and audiological factors other than detection and discomfort thresholds. In addition, successful application of articulation theory requires some method of predicting the actual listening levels the subject will prefer (Byrne, 1982). The present state of knowledge on this has been described already in Section 4.3.3.

The use of articulation theory to predict optimum frequency response characteristics while promising much for the future, has therefore little to offer at present. This is due mainly to our lack of knowledge concerning the perceptual effects of sensorineural impairments on speech reception.

4.5 Selection of hearing aids for persons with predominantly only low frequency residual hearing

Little research appears to have been carried out with adults with predominantly only low frequency residual hearing, to examine whether a different approach to hearing aid fitting is required.

Rice (1965), Leckie and Ling (1968), Morris (1977), Byrne (1978) all suggest that for hearing-impaired children having predominantly low frequency residual hearing it might be beneficial to extend the low frequency response and restrict the high frequency bandwidth. This approach allows the maximum low frequency aid gain to be used before the onset of discomfort or feedback. However, no definite audiometric criteria have been put forward as to when this form of amplification would be of benefit and so an empirical evaluation procedure is required.

4.6 Maximum Output

Although there appears to be general agreement in the literature that the maximum output of the hearing aid should not exceed discomfort levels (Davis *et al*, 1947; Byrne (1978); Berger (1979); McCandless and Lyregaard (1983)) there appears to have been little empirical research to verify how best the required maximum output could be predicted from audiometric discomfort level measures.

Measurements of loudness discomfort, like those of most comfortable levels, have been shown to be dependent on stimuli, methodology and instructions (Stephens and Anderson, 1971; Beattie *et al*, 1979; Beattie *et al*, 1980).

The limitation of output at individual frequencies to below the loudness discomfort level does not necessarily mean that the output for any broadband input signal will also be limited to below the loudness discomfort level. This point does not seem to have been appreciated by many workers in the field. For example, Skinner and Miller (1983), using a master hearing aid that limited the maximum power in each of nine half octave channels to below the listener's uncomfortable listening levels in that band, expressed surprise at their finding that many broadband sounds were still uncomfortably loud for many of their subjects.

Various forms of compression and limiting have in the past been used to limit the maximum output of aids but rarely has it been specified how much of the speech waveform, if any, has been affected by this limiting. It is therefore difficult to scientifically compare the various approaches to limiting the maximum output of hearing aids. The uses of compression and limiting are further discussed in the following section.

4.7 'Speech Processing' Hearing Aids

Sensorineural hearing impairment can result in a degradation of many perceptual abilities in addition to the elevation of absolute thresholds. Those that have been investigated include reduced frequency discrimination (Hoekstra and Ritsma, 1977), reduced frequency selectivity (Florentine *et al*, 1980; Tyler *et al*, 1982; Humes, 1982) and poor temporal discrimination (Tyler and Summerfield, 1980; Zwicker and Schorn, 1982).

It may perhaps be possible in the future to use signal processing of the speech signal to compensate for some of these deficits (Evans, 1978). Fourcin, (1980), for example, suggests that portable speech pattern analysers will make it feasible to provide restricted speech pattern combinations tailored to the needs of the individual to give stimulation by acoustic or direct electrical means. At the present time, however, apart from the use of linear frequency selective amplification to overcome the loss of absolute sensitivity and of frequency transposition (Velmans, 1983) to aid those who have lost high frequency hearing, the main deficit that 'speech processing' hearing aids have attempted to compensate for is the reduced dynamic range of residual hearing in cochlear pathology. This reduction in dynamic range is due to the elevation of absolute thresholds without a corresponding elevation in discomfort levels (Hood and Poole, 1966). The techniques that have generally been used to attempt to compensate for this reduction in dynamic range have been various forms of amplitude compression. These include 'limiting' which protects the ear from painful peak sound levels, 'automatic volume control' (AVC) which keeps the long-term average presentation level near that corresponding to maximum intelligibility and 'syllabic compression' which alters the short-term intensity relations among speech elements to improve intelligibility. There have in the past been many studies using various forms of compression with different compression ratios, compression thresholds, numbers of channels and attack and decay time constants, e.g., Fleming and Rice (1969), Villchur (1973), Drysdale and Gregory (1978), Lawrence, Moore and Glasberg (1983) and King and Martin (1984). For the interested reader, the principles of amplitude compression are described by Barford (1978a) while Braida *et al* (1979, 1982) give a comprehensive account of research in this field.

Despite the large amount of research, any conclusions regarding the benefits of the various forms of amplitude compression have been inconclusive and often inconsistent. Braida *et al* suggest that this may have been partly due to the inadequate technical specification of the equipment used and the lack of testing in controlled experiments in a wide variety of realistic situations.

The majority of amplitude compression systems that have been used in the past have involved compressing or limiting loud sounds while leaving quiet sounds relatively unaffected. This approach is perhaps surprising considering the well-documented findings (Steinberg and Gardner, 1937; Hood and Poole, 1966) that for subjects with a cochlear pathology loudness levels tend to be normal at high intensities.

Wald (1977) designed a novel system in accordance with these findings that allowed a large dynamic range to be 'squeezed' into the narrow residual range of a hearing-impaired subject while restoring the 'intensity relationships' for hearing-impaired subjects to near normal. This was achieved by compressing the dynamic range of quiet sounds while leaving loud sounds relatively unaffected. A similar approach was developed by Barford (1978) and Lippman *et al* (1981).

Wald implemented his system by mixing an amplified but limited or compressed signal with an amplified linear version of the signal. By increasing the gain of the compressed channel the amount of 'speech processed' signal could be varied according to the requirements of the dynamic range of the individual. The simplicity of this approach allows for a practical wearable aid to be made that can easily be adjusted by the user. In contrast, the systems developed by Barford and Lippman required laboratory based computer equipment with extensive complex audiometric tests to set up the system parameters for each individual.

Only informal listening tests were carried out by Wald. Barford and Lippman used multichannel compressors and found little, if any, benefit over a linear system.

4.7.1 Conclusion

No speech processing system has yet been shown to provide a conclusive advantage over linear amplification. Although it might be expected that future advances will take place, it will still be necessary to find methods for determining the optimum 'linear' amplification system with which the speech processing aid must be compared. It will also still be necessary to be able to 'shape' the frequency response of whatever system is developed.

The development of successful speech processing systems appears to have been greatly hampered by the lack of realistic laboratory based evaluation procedures as in the majority of cases field trials of wearable speech processing aids were not feasible and so studies were conducted only in the laboratory.

4.8 Binaural Fittings of Hearing Aids

The advantages of binaural fitting of hearing aids over monaural fittings have been extensively studied and reported, for example by Mackeith and Coles (1971), Markides (1977) and by many researchers, in Libby *et al* (1980) and so will not be reiterated here.

At the present time the accepted method for fitting binaural aids would seem to be to select the aid for each ear independently based on monaural fitting procedures. Although it is possible that special selection techniques based on binaural functioning may in certain cases prove advantageous (for example split-band amplification; Franklin, 1975) there is not enough research evidence at present to warrant such an approach.

4.9 Distortion in Hearing Aids

Although it is generally accepted that distortion in hearing aids should be kept as low as possible, there is little consensus as to the relative importance of different types of distortion (e.g., transient, harmonic, intermodulation) and to the most appropriate method by which they should be measured.

In addition there has been no systematic study as to how the various types and degrees of distortion in hearing aids interact with the various degrees and types of hearing impairment.

This knowledge is important as differences in distortion between hearing aids may confound the investigation of other variables such as frequency response.

4.10 Rehabilitation Procedures

The selection of the most appropriate frequency gain characteristic of a hearing aid is, of course, only one aspect in the rehabilitation of hearing-impaired persons. Other important considerations include such factors as ergonomic and cosmetic aspects of hearing aids, the use of environmental aids, auditory training, hearing tactics training, speech reading practice, and counselling. These factors will, however, not be discussed further in this thesis. For a more thorough description of rehabilitation programmes as a whole, the reader is referred to Markides (1977), Brooks (1979) and Goldstein and Stephens (1981).

CHAPTER 5

HEARING AID MEASUREMENTS

5.1 Introduction

The two main reasons for measuring the electroacoustic performance of hearing aids are to establish whether the aid is performing according to specification, and to determine if the aid's performance is suitable for a particular individual. The former use requires only that the measurement technique be repeatable, while the latter requires the aid's performance be specified in terms of the individual user's actual requirements. This chapter will concern itself with the most commonly used performance measures, the frequency/gain characteristics of the aid.

5.2 Couplers

The 2 cc coupler was introduced by Romanow (1942) as a simple means of making objective repeatable measurements on hearing aids with insert receivers. The coupler was used to measure the 'transmission gain' of a hearing aid by measuring the ratio of the output generated in the coupler, to the level of the free sound field in which the microphone was placed.

To measure the actual gain the aid provided to an individual's ears, Romanow described a monaural loudness balance subjective calibration procedure in which the ratio of the pressure in the sound field without an aid to the pressure with an aid adjusted to give it an equal loudness sensation was measured. Littler (1936) described a similar procedure using aided and unaided threshold measurements to measure the gain provided.

These subjective procedures can provide a functional estimate of the "insertion gain" which can be defined as the ratio of the sound pressure at the eardrum of the individual when the aid is worn to the sound pressure at the eardrum without the aid.

The transmission gain or "acoustic gain" of a hearing aid measured on a 2 cc coupler will be different from the insertion gain of the hearing gain measured on a person for three main reasons:

(i) The coupler is an inadequate simulation of the acoustic impedance of the occluded human ear.

(ii) No allowance is made for head or body diffraction effects.

(iii) No allowance is made for the loss of the natural open ear canal resonance when the ear canal is occluded.

Empirical data has been used to 'correct' for these differences using 'universal' correction factors, but these can only account for differences between coupler measurements and the 'average' human being. It is possible to try and derive a 'correction' for the difference between transmission gain and insertion gain for a particular individual, but differences in microphone position and interactions of couplers and real ears with hearing aid receivers prevent the accurate application of any standard correction factors.

Surprisingly the importance of these differences in response between individual and average ears and couplers has not been systematically investigated in spite of the great interest in their quantification.

Although the limitations of transmission gain measurements have been documented and well understood for a long time (Romanow, 1942) they have often been overlooked by both researchers and dispensers. A consequence of this may have been that the development of the scientific fitting of hearing aids based on audiometric measurements has been inhibited. The differences between hearing aid responses in couplers and in real ears might have caused the relationships between optimum hearing aid responses and audiometric measures to have become clouded.

5.3 Ear Simulators and Manikins

Endeavours to overcome these difficulties have resulted in the development of 'ear simulators' (Zwislocki, 1970, 1971; Bruel and Kjaer Type 4157) which attempt to model the impedance of the average human ear and the transfer impedance to the eardrum, and of Manikins (Burkhard and Sachs, 1975) which simulate the effects of head and body diffraction effects and ear canal acoustics.

Although the use of ear simulators and Manikins for hearing aid measurements appears to give results close to those obtained using human subjects, there will still be individual variations from these 'average' measurements. If precise measures of hearing aid responses on individuals are required it is therefore still necessary to use measurement techniques involving that individual rather than ear simulators or Manikin. These techniques will be discussed in the following sections.

This brief account of the development and use of couplers and ear simulators for hearing aid measurements has only described the most salient points relevant to this research. For a more thorough review of their development the reader is referred to Lower (1980).

5.4 Coupler Calibrated Behavioural Measures

One reason why hearing aid performance criteria cannot be directly related to audiometric thresholds is that the sound delivery systems are different and the measurements quantitatively expressed in different ways.

If audiometric thresholds were obtained using the actual hearing aid receiver and earmould fitted to the subject, then both audiometric thresholds and hearing aid performance could be expressed in terms of the same coupler measurements and hence directly compared. Corrections would then only be needed for individual head diffraction effects and the interaction effects of the couplers and individual real ears with different hearing aid receivers. Penney and Goodwin (1984) have recently reported the use of a similar procedure.

5.5 Aided and Unaided Behavioural Threshold Measures

The difference between aided and unaided behavioural thresholds can be used to give a measure of the insertion gain of an aid on an individual. This "functional gain" method of estimating insertion gain has been used both in the past (Littler, 1936) and more recently by Pascoe (1975) and Skinner (1976). One attraction of this method is its simplicity and the directness of its approach. However, there are various problems that can prevent accurate results being obtained.

The use of pure tones in non-anechoic free field testing results in standing waves being set up in the test room. This can cause large variations in the SPL of the pure tone over small distances and so lead to difficulties with accurate calibration of the system and interpretation of the results. The use of narrow band noise or warble tones to overcome these problems will lead to more repeatable results, but less frequency specific information concerning the hearing aid response (Dillon and Walker, 1982). These problems can of course occur for all free field testing methods whether behavioural or objective.

The relevance of these problems will depend on the hearing aid fitting criteria used. For example, pure tone measurements of aided and unaided thresholds may be unnecessary for hearing aid fitting based on the attempt to present the one-third octave bands of speech at some predetermined level above aided thresholds or aided comfort levels. This only requires behavioural aided measurements using one-third octave band stimuli.

In contrast, hearing aid fitting based upon providing certain pure tone aided thresholds calculated from pure tone unaided thresholds will require both aided and unaided pure tone measurements to be undertaken.

The measurement of accurate behavioural aided thresholds can also be prevented by masking caused by ambient, physiological or hearing aid noise (Macrae and Frazer, 1980). However, with the low noise levels of present day hearing aid microphones (Killion, 1976) hearing aid noise would appear only to be a problem when aided thresholds lower than 20 dB SPL are required. This is much lower than that recommended by hearing aid prediction methods for the majority of hearing-impaired subjects and so would not appear to be a problem.

Another problem that is inherent in all behavioural measures is intra-subject variability. Since the difference between two behavioural measures is required for the determination of insertion gain the total variability will be increased. An additional drawback when the gains of a large number of hearing aids on an individual are required is that the use of behavioural measures will be fatiguing and very time consuming.

5.6 Loudness Balance Measures

The use of subjective aided and unaided loudness balance measures for determination of hearing aid insertion gain has already been described (Section 5.2). This procedure suffers from similar disadvantages to those derived for behavioural threshold measures (Section 5.5). Although being a supra-threshold measure, its accuracy can still be affected by microphone noise. This can reduce the subjective loudness of the test signal due to the phenomenon of partial masking (Lower, 1980). Lower, however, used normally hearing subjects; the majority of hearing-impaired subjects for whom microphone noise would be below threshold would presumably not be affected by this partial masking phenomenon.

The subject's task of balancing loudness would also appear to be subjectively more difficult than that of providing behavioural thresholds, and the method therefore not as universally acceptable.

5.7 Acoustic Reflex Measures of Insertion Gain

Measurements of aided and unaided acoustic reflex thresholds have been used (Tonisson, 1975; Lower, 1980) to estimate the insertion gain of hearing aids. One advantage of this method is that no behavioural responses of the subject are required, although passive cooperation does help reduce artifacts. It is therefore appropriate for babies, very young children, or anyone from whom behavioural responses are difficult to obtain.

Acoustic reflex thresholds appear to be completely independent of the levels of microphone noise (Lower, 1980). The greater drawback of the use of acoustic reflex thresholds however is their restricted applicability, as they are often not present in losses with a conductive element, severe sensory impairment and pathologies that result in non-intact neural reflex pathways.

There is also a certain amount of subjective judgement required in the interpretation of the meter deflection indicating the presence of the reflex.

5.8 Probe Microphone Measures

5.8.1 Existing techniques

Behavioural functional gain measures give an indication of the insertion gain of a hearing aid on an individual but as pointed out in Section 5.5, may be subject to certain 'errors'. A direct measure of the insertion gain of a hearing aid can be made by measuring the actual sound pressure level in the ear canal of a person in the unaided and aided condition. As insertion gain is a relative measure, so long as both aided and unaided measurements are made at exactly the same position, the actual point of measurement is immaterial. It would therefore appear unnecessary to measure at the eardrum as there is no evidence that the actual absolute sound pressure level at the eardrum of an individual yields any additional information useful for the fitting of hearing aids.

The main advantage of objective ear canal sound pressure level measurements is that they can be quick and require no behavioural response.

There have been many studies describing probe microphone measurements in open ears (Wiener and Ross, 1946; Djupesland and Zwislocki, 1972) and in occluded ears using insert earphones (Dalsgaard and Jenson, 1976;

Harford, 1981). Sachs and Burkhard (1971) have shown theoretically that for frequencies above approximately 3 kHz it is necessary to make measurements at a position 3-5 mm beyond the earmould tip to avoid any anomalous sound distribution near the mould. Lawton (1979) has surveyed the sound field within the occluded ear canal documenting the standing wave patterns that occur due to sound reflections at the eardrum and indicating the likely errors occurring from estimating eardrum sound pressure levels from measures at distances from the eardrum. However, where relative measures of aided and unaided sound pressure levels are required any errors caused by changes in probe position will generally cause errors in SPL measurements of considerably less than 5 dB for frequencies up to 6 kHz. Errors will be even less for measures closer to the eardrum or for lower frequencies. Greater errors occur for higher frequencies due to the larger standing wave ratios that occur at these frequencies (Lawton, 1984). The adequate repeatability of ear canal sound pressure measurements has been verified by Harford (1980) and Pederson, Lauridson and Birk Nielson (1982). The two methods that have been used most often for measurements of ear canal sound pressure levels are probe tube microphones and miniature ear canal microphones. The latter method, pioneered by Harford, has been made possible by the introduction of a very small miniature broad frequency flat-response electret microphone (Knowles model EA 1934).

Both methods have been shown to give similar results (Pederson, Lauridson and Birk Nielson, 1982). The advantage of the probe tube method is that it is less invasive, while the ear canal microphone approach has the advantages of not requiring modification of the earmould to allow a probe tube to enter the ear canal, and of not requiring removal and replacement for the measurements of both aided and unaided responses.

Attempts to use a probe tube that is placed between the mould and the auditory canal (Pederson, 1982) resulted in leaks and a consequent reduction in the low frequency responses of the aids tested. The presence of the probe tube (Lower, 1980) or the canal microphone (Harford, 1980) in the ear canal has been shown not to effect ear canal sound pressure levels significantly.

5.8.2 Sound field levels

Various methods have been used to control the sound field for ear canal sound pressure measurements. If a hearing aid was a totally linear device at all frequencies and levels, then the absolute levels of the sound field at any frequency would be unimportant for insertion gain measurements so long as they were the same for both unaided and aided test conditions. However, because hearing aid frequency responses are input level dependent and to ensure that test conditions are easily repeatable, a flat sound field in the absence of the subject is often required. This can be achieved by either tape recording the required signals or 'remembering' the required correction curve using a digital memory (e.g., B & K audio test station 2118).

5.8.3 Aided threshold prediction using ear canal measurements

For hearing aid fitting techniques that do not require insertion gain measures but are based upon aided thresholds, it is possible to use ear canal sound pressure level measurements to predict aided thresholds for any hearing aid from only one behavioural threshold measurement. This can be accomplished by measuring the sound pressure level in the ear canal for the behavioural thresholds at the same position at which the hearing aid performance is measured. The behavioural threshold could be measured using headphone, insert receiver or free field presentation (Fig. 5.1). The free field aided threshold can be found by subtracting the gain provided by the aid at the measurement position from the measured behavioural threshold.

5.8.4 A new non-invasive probe microphone measurement technique

A novel 'non-invasive' probe microphone approach to predicting aided thresholds from objective sound pressure level measurements was developed by the author during this study. Instead of measuring the sound pressure levels at a point in the ear canal for behavioural thresholds and for hearing aid performance, it is measured at the entrance to the earmould tubing. This method therefore requires behavioural thresholds to be measured using an insert receiver and the subject's hearing aid earmould.

The SPL measurement is made by fitting a 'T-piece' between the hearing aid and earmould tubing as shown in Fig. 5.2. This technique is based on the transmission line principle that the pressure distribution along a transmission line is independent of the source impedance. 'T-piece' measurements to predict aided thresholds are therefore independent of receiver type and so are accurate, even if a different receiver is used for the behavioural threshold measurement than for the hearing aid. This is the identical principle that allows relative ear canal sound pressure level measurements to be used instead of actual eardrum position measurements being necessary. If the same earmould is used for all measurements, the 'T-piece' measurement technique can be thought of as treating the earmould and its tubing as an extension of the ear canal. Empirical measurements using Kemar (Appendix A) validate the theory.

The greatest advantage of this technique is that it is totally non-invasive and can be used with all subjects. In contrast, ear canal SPL measurements have up till now not been possible for those with small ear canals (e.g., young children and babies). Probe tube measures are also difficult to use where there is little space in the earmould for the probe bore (vented moulds, acoustic horn). Ear canal measurements using ear canal microphones require caution and care in their use and so present a safety risk.

5.9 Difference between Occluded and Unoccluded Listening

There has in the past been much speculation as to whether identical sound pressure levels at the eardrums are perceived identically for unoccluded and occluded listening. This speculation was due to the observed differences between monaural minimum audible field (MAF) and minimum audible pressure (MAP) as described by Munson and Wiener (1952).

Recent research evidence (Rudmose, 1962; Villchur, 1969; Killion, 1978) has shown that these differences are caused by artifacts in the experiments such as a physiological noise and calibration errors ignoring ear and head acoustics and real ear-coupler differences. Lower (1980) using probe tube ear canal SPL measurements to measure the insertion gain of hearing aids and acoustic reflex thresholds to measure the 'behavioural' or 'functional' gain of the aids showed that the two methods were in excellent agreement. The insertion gain of an aid can therefore be considered as predicting accurately the 'true' functional gain of the aid.

5.10 Conclusions

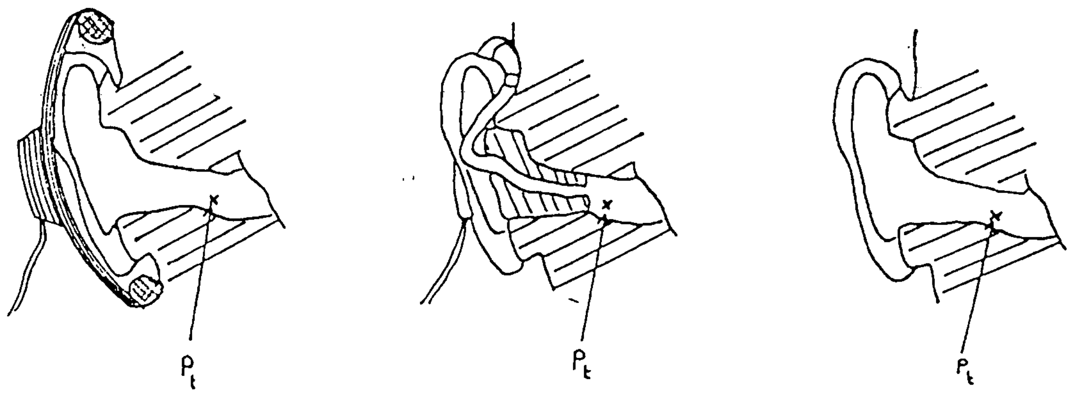
There are a wide range of techniques available for measuring hearing aid performance, each having its advantages and disadvantages. The required accuracy of measures of the performance of hearing aids on individuals has yet to be determined. Repeated behavioural measures are relatively simple but time consuming and fatiguing to carry out and of limited reliability. Hearing aid, ambient and physiological noise must be taken into account for the correct interpretation of the results.

Coupler, ear simulator and Manikin measurements are quick and repeatable but require individual correction factors to be used if complete accuracy is required. The correction factors will be the smallest for Manikin measures as they are closest to the 'average' real person. Probe microphone measures using the actual individual provide quick and accurate measures of hearing aid performance, and using the appropriate method can be carried out on all people. 'T-piece' earmould SPL measurements can be carried out on all ear canals and all earmould types, but require behavioural thresholds to be obtained with the individual's hearing aid earmould. This method, however, cannot provide insertion gain measures.

Ear canal microphone techniques are independent of earmould type and whilst they can provide insertion gain measures are invasive, present a potential safety risk and cannot be used on small ear canals or ear canals containing significant deposits of wax. Ear canal probe tube microphone measures can be considered as a compromise between the advantages and disadvantages of the other two methods. They are less invasive than the ear canal microphone technique but cannot be used for all earmould types or on all subjects.

Figure 5.1

Ear canal measurements for aided threshold prediction

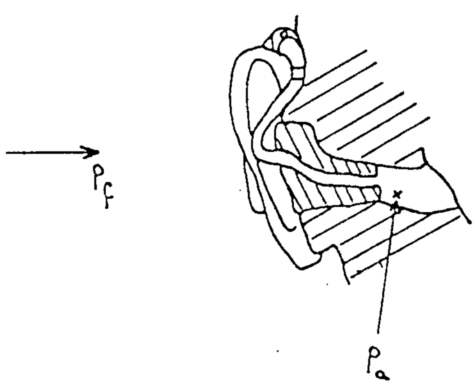


headphone threshold   insert receiver threshold   free field threshold

at probe microphone position in ear canal

behavioural threshold is P_t

P=SPL ,dB



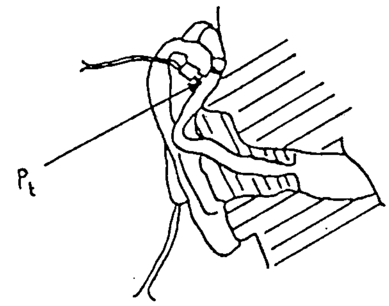
acoustic gain $G=P_a - P_f$

Free field aided threshold for this gain setting = $P_t - G$

Figure 5.2

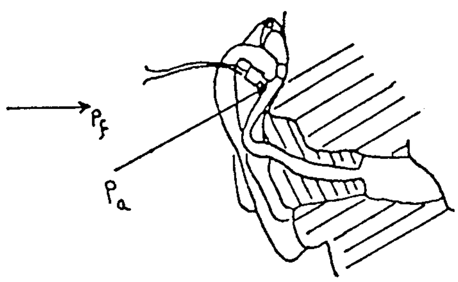
T piece technique

insert receiver threshold



behavioural threshold is P_t

at T piece probe microphone position



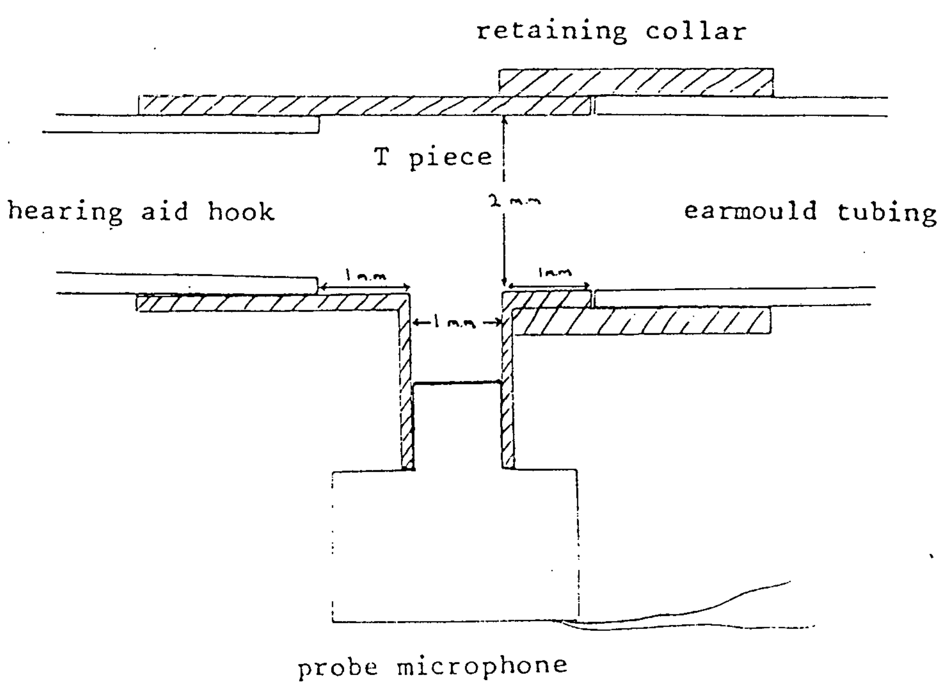
P=SPL ,dB

acoustic gain $G=P_a - P_f$

at T piece probe microphone position

Free field aided threshold for this gain setting = P_t - G

detailed view of T piece



CHAPTER 6

RESEARCH AIMS

6.1 Introduction

From the extensive review of previous research summarised in Chapters 2-5 of this thesis, certain areas of knowledge important to a scientific approach to the fitting of hearing aids were found to be poorly understood and in need of further study. These findings shaped the aims of this whole study, which were summarised in Chapter 1, Section 1.3, and are now described in more detail.

6.2 The Development of a Clinically Feasible Protocol for the Scientific Fitting of Hearing Aids

Much of the research into the fitting of hearing aids has been laboratory based using specialized equipment and tests that are not feasible to use in a normal hearing aid clinic due to limitations of complexity, cost and time. These studies have also generally involved very small numbers of subjects and have rarely included home trials of aids.

If a simplified protocol could be developed for clinic use there would be two important benefits. Firstly, clinical hearing aid fitting would benefit from the most up-to-date knowledge and techniques of hearing aid fitting. Secondly, the results from routine hearing aid fitting in hearing aid clinics could be used to further research into the improvement of fitting protocols. In this way the development of a scientific basis for the fitting of hearing aids could be a continuous step-by-step development based on a pooling of knowledge and ideas between clinics and research laboratories. This would be a great improvement on the current rather disjointed development, where the lack of communication between clinic and research laboratory appears to have characterized much of the previous work in this field.

The tests developed for this research study have therefore been designed with possibilities for use in a normal hearing aid clinic very much in mind. A simplified protocol was established, based on the results reported in Chapters 11 and 12 and subsequently field trialled as outlined in Appendix Q.

6.3 The Development of Hardware and Software Systems Capable of Carrying Out the Required Experimental Protocols

The research aims described in this chapter necessitated the development of a specially constructed computer based hardware system with associated software. This allowed for accurate presentation of required real ear responses to patients and the flexible manipulation of these hearing aid responses and gain settings under either experimenter or subject control. Accurate adaptive presentation of audiometric stimuli and the integration of audiometric, electroacoustic and subjective information onto a data base that facilitated computer hearing aid selection were also required.

No similar system has, to the author's knowledge, ever been developed and therefore the extensive hardware, interfacing and software development undertaken during this study was entirely original. The hardware is described in detail in Chapter 7, while the software development is described in Chapter 8.

6.4 The Development of Hearing Aid Evaluation Procedures

In adopting an approach to hearing aid fitting and evaluation, certain assumptions are often made that are unsupported by any previous extensive empirical research. One pragmatic approach is to assume that the hearing aid user in his real-life everyday situation is capable of choosing a satisfactory or optimum aid himself. An 'expert based' approach assumes that the hearing aid user is incapable of choosing the aid for himself and requires the 'expert' clinician using objective measurements to select the aid. These are of course extremes in approach, most clinicians using some 'intuitive' combination of user subjective judgements and clinician objective measurements.

However, if the aims of the hearing aid fitting approach are not explicitly stated, or an evaluative approach used that does not allow the achievement of any stated aims to be tested, very little progress in developing a scientific approach to the fitting of hearing aids can be made.

In the past, the term 'a scientific approach' has tended to be associated synonymously with objective measurements. Subjective user judgements were felt to be somehow 'non scientific' and therefore not valid. This has resulted in objective tests such as the measurement of speech discrimination ability being extensively employed (Chapter 2). Clinicians and researchers appear to have taken on faith the untested assumption that results from these tests will predict results in the real-life performance of the aids. The relevance of laboratory or clinic tests to real-life situations is difficult to determine objectively, as either their relevance must be accepted on grounds of face validity or 'non scientific' subjective judgements must be used to validate the relevance of the 'scientific' objective measurements. If the latter approach is adopted then the subjective judgements are in fact being accepted as the true, valid results.

We can therefore see that attempts to use a purely objective 'scientific' approach to hearing aid fitting must inevitably be based on assumptions that have not been scientifically tested. This paradox appears to have been overlooked by many workers in the field whose attempts at producing simple but unrealistic objective tests have often disregarded even face validity due to the assumed 'unreliability' of possibly more realistic but more complex tests.

Subjective responses have often been claimed to be proved 'invalid' by their lack of correspondence with objective measures, whereas in fact one could equally well interpret the results as showing that the objective measures are invalid due to their lack of correspondence with subjective responses (Section 3.2).

A great deal of previous research has proved fruitless through a failure to understand the above points. Rather than spending much time and effort on research using and developing objective tests with little or no proven 'real-life' validity, a more fruitful avenue to follow is the development of laboratory and clinic tests which predict real-life performance and subjective preferences. In this way valid aid preselection could be made from those aids that may be of possible benefit. Some sort of preselection is necessary whatever approach to hearing aid fitting and evaluation is used (Section 10.1).

If valid laboratory tests that predict real-life performance can be devised both clinical and research work will benefit by the reduction in necessary field/home trials of aids (Section 4.7).

Other aspects of subjective judgements requiring research are test-retest reliability of judgements, and any accustomization effects that exist. For example, if subjects were not reliable in their subjective judgements of aids, or liked any aid as long as they were used to it, there would be important implications for the fitting of hearing aids.

The knowledge gained concerning the relationship between audiometric factors, objective tests and subjective judgements can also be valuable in fitting hearing aids to those aid users whose subjective judgements cannot be obtained or relied upon (e.g., babies and young children).

Two original hearing aid evaluation procedures were developed for this research study to overcome previous deficiencies. A pair comparison test, described fully in Chapter 10, allowed the rapid comparison of hearing aid responses and levels under subject control and solved many of the problems encountered in the past, as detailed in Section 3.4. The second evaluation procedure was an Audio and Visual Everyday Life Simulation (AVELS) test which presented a realistic laboratory simulation of real-life listening situations and so overcomes some of the problems of previous speech tests, which have not predicted real-life performance well. The AVELS test, described in detail in Chapter 9, was designed to be both subjectively rated and objectively scored for the evaluation of hearing aids. The objective scoring allows analysis of errors to be made for rehabilitative treatment programmes as well as statistical analysis for comparison of group data.

6.5 The Determination of the Relationship between Audiometric Profiles, Hearing Aid Frequency Responses and Preferred Gain Settings

The performance of hearing-impaired subjects using a hearing aid, whether in laboratory/clinic tests or in their everyday life will depend to some extent on the amount of amplification they use. Any attempt to predict whether one hearing aid is better than another either by a theoretical approach or by laboratory tests, must therefore take into account the user's preferred gain setting if the results are to relate well to real-life use and performance (Section 4.3.3). Based on experience, intuition and some reported research findings, 'rule of thumb' approaches have often been used in the past to relate audiometric data, hearing aid frequency responses and preferred gain settings. However, wide variations in this relationship have been found for given individuals. Due to limitations in the construction and reporting of previous research studies it is not possible to determine whether there was an actual unpredictable subjective variation in preferred gain settings, or whether the relationship was confounded by inadequate specifications of audiometric criteria and real ear hearing aid responses and inadequate analysis of the data using too simplistic a model.

However, even if the determination of preferred gain settings proved to be possible only by means of empirical testing, it might still be feasible to predict the preferred gain setting for one frequency response from that measured empirically for another. If even this was not found to be possible, then any theoretical approach without empirically measuring the preferred gain setting for each aid involved, would be doomed to failure.

In addition, any non linearities in the system such as limiting by means of peak clipping or compression, add to the difficulties of theoretically predicting preferred gain settings and little data is presently available concerning this.

The experiments detailed in Chapters 11 and 12 examine the relationships between preferred gain, frequency response and audiometric data for both normally hearing and hearing-impaired subjects using carefully controlled measurement techniques.

6.6 The Study of the Relationship between Audiometric Profiles and Optimum Hearing Aid Frequency Response

If the optimum electroacoustic parameters could be determined purely from audiometric measures, without empirically having to test the possibly vast range of alternative hearing aids, a much quicker and more satisfactory method of hearing aid fitting would be possible (Section 4.4).

Previous research has been equivocal in its findings, due partly to inadequacies in accurate and wide ranging specifications of electroacoustic parameters and audiometric measures and partly to a lack of adequate evaluative techniques to determine the optimum electroacoustic parameters.

The adaptive pair comparison technique described in Chapter 10 was used to enable a wide range of response parameters to be evaluated rapidly by subjects. Optimum frequency responses, as specified by existing state of the art evaluation and prescription techniques were also compared.

All acoustic and audiometric data were specified in terms of ear canal sound pressure levels through the use of probe tube microphone measurements.

6.7 The Examination of the Importance of Deviations from the Optimum Response and its Relationship to Audiometric Measurements

In comparing an almost infinite possible range of electroacoustic parameters it is important to know which parameters are of importance and how great a change in a particular dimension is required to give a corresponding change in performance and subjective judgements. This knowledge would enable the clinician and also the manufacturer to simplify and restrict the range of alternatives offered and tested, resulting in possible savings in both time and money.

Much time has been spent in the past quantifying acoustic and electro-acoustic differences between measurements in real ears, couplers, ear simulators and Manikins (Chapter 5) as well as determining the acoustic effects of such things as earmoulds and earmould plumbing variations, microphone placement and damping (Section 4.1).

However, little is known whether differences in these acoustic effects result in objective performance or subjectively perceived effects (Section 3.3). This information is valuable as if it could be shown that certain deviations from the 'optimum' were of no importance, for example the difference between real ear and ear simulator mesurements, then considerable simplifications in the approach to hearing aid fittings could be made. The effects of these changes in frequency response were examined using the Master Hearing Aid and the pair comparison test.

The necessity of making individual empirical measurements of the importance of deviations from the optimum response will be determined by whether or not these measurements may be predicted from audiometric data.

6.8 The Examination and Development of Techniques for the Measurement of Real Ear Hearing Aid Performance

In order to investigate the relationship between audiometric data and hearing aid responses and also to compare hearing aid responses, the measurements must be done in a comparable way. This has not been the case in the majority of previous research; audiometric data has usually been collected using free-field or headphone presentations, whereas hearing aid data has been obtained from coupler measurements.

There is a wide range of possible techniques for overcoming this problem, including one novel method developed during this research study and described in Section 5.10. Although some of the methods may be more appropriate for clinical use than others, little research has previously been undertaken concerning the necessity, advantages, disadvantages and fesibility of the various approaches. This study examines these problems by comparing the use of probe microphone, headphone, free field, coupler and ear simulator measurements.

CHAPTER 7

DEVELOPMENT OF A COMPUTER CONTROLLED MASTER HEARING AID TEST FACILITY

7.1 Introduction

The needs of this study could not be met using any existing proprietary system and therefore a computer controlled system was developed by the author. Although some standard equipment was used, much of the system described in this chapter was developed using hardware and software specially conceived and built over an eighteen month period.

The main components of the system are:

(i) Programmable Master Hearing Aid

(ii) Hearing aid measurement system

(iii) Microcomputer and peripherals

7.2 Programmable Master Hearing Aid

7.2.1 Requirements

The study required a system that would deliver any required frequency characteristic to the subject's ear with subject control of level and characteristic (Section 6.3). It was considered important to be able to provide responses required by any 'fitting rule' or 'system' and to examine the effect of small variations from these responses.

Subject control of level was required as the performance at the level the subject chooses is what matters in real life, not the performance at a level determined for him. This is especially important when comparing two frequency gain characteristics as the relative levels will affect the subject preference and performance (see Section 3.4). Equating the loudness levels for one hearing-impaired subject will not necessarily provide equal levels for a different subject, and so independent control of levels is required.

When comparing two frequency gain characteristics it is also necessary to allow the subject control of which characteristic is presented rather than switch between them after a set period of time. This is because some people may take longer to adjust levels and make up their minds than others.

7.2.2 Hardware

7.2.2.1 Limitations of existing analogue equipment

Existing analogue equipment could not fulfil the requirements satisfactorily. The use of a one-third octave graphic equaliser would give an approximately accurate frequency response (to the nearest 1/3 octave) but has a poor phase response and cannot be set with speed or precision. Commercial 'Master Hearing Aids' have very limited ranges of responses while even a versatile analogue Master Hearing Aid such as that built for the RNID (Martin and Evans, 1980) suffers from the drawback of unpredictable interactions between controls in addition to not being generally available.

7.2.2.2 Digital filter

The requirements could, however, be met using a novel digital filter designed at the Institute of Hearing Research that had recently become available. This could, within certain constraints (see Appendix B) present virtually any desired frequency response within milliseconds of being programmed by the microcomputer.

Following visits to and consultations with both the designer of the filter (Trinder, 1981) and of the filter creation algorithm (Caine, 1981) the author developed a 'Master Hearing Aid' system based around two IHR digital filters.

The digital filters were programmed via eight data lines and the output strobe line of a microcomputer parallel input/output port (see Appendix C). The required TTL clock pulses were obtained from a Wavetek synthesizer/function generator Model 171 and monitored by a Racal universal counter (9835) (Figure 7.1). The digital filter input and outputs were filtered by analogue Barr and Stroud filters to prevent aliasing.

The clock rate was set at 30 kHz and the anti-aliasing filter at 10 kHz with cut-off rates of 48 dB/octave. The highest frequency passed by the digital filter was programmed to be 8 kHz giving no in-band aliasing for input frequencies less than 23 kHz while those created by input frequencies above 23 kHz would be greater than 58 dB down. Since the wideband signals used were speech which has little energy at these high frequencies, these specifications were considered to be more than adequate.

The digital filter has a 12 bit input analogue to digital converter and a 16 bit output digital to analogue converter. This provides an input dynamic range of greater than 60 dB and an output dynamic range of greater than 80 dB. This is adequate considering the dynamic range (signal/noise ratio) of the taped stimuli (< 60 dB) if the input levels to the digital filter are adjusted to make full use of its available dynamic range.

The filter can be swiftly reprogrammed to adjust its gain and so optimize the output signal to noise ratio. The distortion levels of the filter were also more than adequate (for complete detailed specifications see Appendix D).

7.2.2.3 Digital attenuators

The presentation of any required signal level to a subject with adequate precision and repeatability requires stepped attenuators. If the subject is required to have similar precise control of the required level this becomes unwieldly if not impossible to achieve using conventional analogue attenuators. The actual levels set would be difficult to monitor and there would also be problems with bias over the subjects' setting of the controls. It was therefore decided to use custom built digital attenuators controlled by microcomputer. The digital attenuator allows 0-99 dB attenuation in 1 dB steps and can be operated manually by means of push buttons in addition to remotely via eight data input lines. The attenuating electronics consisted of passive resistor networks switched by relays and so gave a very good frequency response and dynamic range (see Appendix C).

7.2.2.4 Hearing aid transducers

In addition to the computer controlled digital filter and digital attenuator the Master Hearing Aid required input and output transducers. To include all the normal room, head and pinna acoustic effects (Chapter 5) and facilitate direct comparison of results with current behind-the-ear hearing aids, the Master Hearing Aid used the case, microphone, preamplifier with battery supply, and receiver of the NHS BE51 hearing aid. A 680 acoustic ohms resistance (Knowles) was placed at the end of the hook to damp out earmould tubing resonances. The use of the microphone close to the preamplifier with shielded leads and a separate output power amplifier operating from 24 volts allowed high output with a wide frequency response and low noise and cross talk to be achieved.

As shown in Figure 7.1 the output from the microphone preamplifier was fed to a custom built amplifier/anti-aliasing filter to boost the band limited signal to that required by the Neve compressor (± 5 V max). This compressor allowed for a wide range of input compression ratios, thresholds and attack and recovery times to be employed if required. The output of this compressor/limiter was attenuated to give the levels required at the input of the digital filter (± 1 V pp max).

Two digital filters were used in series (with appropriate anti-aliasing filters) to provide for greater flexibility in programming responses. The output of the filters went to a power amplifier, the digital attenuator, the output compressor/limiter and finally via a shielded lead, to the hearing aid receiver mounted in the hearing aid body.

In this way a high fidelity Master Hearing Aid with flexibility of computer control of frequency response and gain, was achieved. Full specifications are provided in Appendix E.

7.2.2.5 Subject Control Box

Subjects were required to control both the level of the signal and which of a pair of signals they wanted to listen to, and also to indicate they had finished making a judgement. The 'response box' was required to be comfortable to hold in the hand and simple to operate and understand. As shown in Figure 7.2 a red and a green button with red and green lights

(L.E.D.s) positioned above them were provided for choice of stimuli, while grey buttons next to yellow lights and marked 'LOUDER' and 'QUIETER' were provided for control of volume. Another button with a yellow light by the side and marked 'STOP' was provided for stopping the stimuli and signalling when a decision had been made. This same response box was used for different tests, the buttons always retaining their 'meaning' but not all were always necessary. For measurement of comfortable levels the red and green channel selection buttons were not required, whereas for measurement of loudness discomfort only the 'Quieter' button was needed. For the pair comparisons all the buttons were used. In this way the requirements of ease of operation and ease of understanding were accomplished and no subjects found any difficulty in very quickly picking up the operation of the response box.

A cable connected the buttons/response box to the computer parallel input port from which the power for the L.E.D. lights was drawn. The buttons were wired so that the requisite data line was grounded when its particular button was depressed. Full details of the response box are given in Appendix C, and the layout of the buttons and lights shown in Figure 7.2.

7.3 Hearing Aid Measurement System

7.3.1 Requirements

It was necessary to be able to measure the performance of hearing aids in couplers or ear simulators and in real ears. Coupler or ear simulator measurements were required to check the functioning of the aids and to relate and compare their characteristics. Real ear response measurements were required to ascertain the actual gain that the hearing aid provided for each individual subject.

7.3.2 Hardware

7.3.2.1 Limitations of existing equipment

Although the measurement of hearing aid performance can be accomplished using existing analogue equipment, the data is recorded on a chart by a pen trace. This limits the accuracy due to problems of pen width and chart alignment and makes for time consuming and unwieldly measurements for analysis. In addition, when probe microphones are used for real ear measurement, corrections must be made for their response and so further measurements and numerical manipulation must take place. This is also the case for the measurement of insertion gain and the measurement of differences between aids. Also, to programme the digital filter to simulate an aid response would entail entering all the information by hand into the computer.

7.3.2.2 Computer control of analogue test system

The possibility of using the computer to do all the hearing aid measurement functions (i.e., signal generation and analysis and recording of responses) was considered, but due to constraints of time and limited available technical assistance it was decided to use computer control and data recording and manipulation of an analogue signal generator and analysis system. With no commercially available system with a computer interface and no company agreeing to modify their equipment as a 'one off', a B & K audio test station (Type 2116) was purchased with the aim of modifying it for interface with the computer.

Computer control was required of all the functions of the B & K audio test station in addition to the recording by the computer of the test parameters and frequency and dB level of the signal measured.

This necessitated the building of an interface box which connected to the computer input/output ports and to the requisite points on the audio test station circuit boards. The system was designed to require a minimum of modification to the standard B & K circuit boards, but due to what appeared to be rather 'close to the limit' design by B & K in some of the logic timing circuits, some modification to the width of timing pulses proved to be necessary to allow the modified test station to function correctly. This would not be necessary for more recent B & K audio test stations.

The frequency output from the 2116 audio test station was available on eight data lines (coded as 256 logarithmically spaced frequencies between 100 Hz and 10 kHz) while the rms dB signal level output was obtained as an analogue signal and transformed to a digital signal via an 8 bit A-D converter, thus giving the dB levels from 26 dB to 154 dB to the nearest 0.5 dB. To give the correct recording of the signal level it was also necessary to monitor the 'frequency change pulse' to know when a shift in frequency was occurring, and the 'range shift pulse' to know when a shift in range was taking place. A facility was also incorporated that allowed the B & K audio test station to return to manual control automatically when not under computer control. Full specifications and circuit diagrams are given in Appendix F.

7.3.2.3 Probe microphone system

For real ear measurements a Knowles Electret wideband subminiature microphone (EA 1934) was used due to its wide flat frequency response and its similar sensitivity to the B & K 1/2" microphone used for the hearing aid and ear simulator measurements.

A probe tube approach was taken rather than the more direct ear canal microphone technique. This was due to two main factors. Firstly, 'Safety and Ethics' approval would have necessitated medical supervision for a canal microphone approach. Secondly, as high power and high gain aids were to be used with no venting of moulds, there was no difficulty with using the probe tube approach. The 'T-piece' method described in Section 5.8.4 was not used as insertion gain measures were required.

The probe microphone was powered by four 1.5 V batteries contained in a power supply/interface box which was worn around the subject's neck.

For real ear hearing aid and Manikin measurements a single driver loudspeaker was built which would deliver the required levels of the pure tone test signal.

7.4 Choice of Computer and Peripherals

The hearing aid measurement and Master Hearing Aid systems were based around a microcomputer. The choice of microcomputer was constrained by various factors. The rapid programming and calculation for the digital filter required machine code or fast high level language programming. The large amounts of data obtained, stored and manipulated in a hearing aid data base required disk storage. The control and interface with a wide variety of equipment required flexible input/output facilities. Reliability, ease of repair and availability of software favoured a micro-computer that was similar to others available within the University.

Within the budgetary constraints of the project it was decided that a Cromemco System 3 (see Appendix D) suited the purpose well.

However, it was also required to plot frequency response graphs on the video screen and to obtain hard copies of these plots. For this purpose a Televideo 920B terminal, the newly developed Midelectron 'Graffix' system, and an Anadax 1500 graphics dot matrix printer were purchased. Full specifications of this equipment are given in Appendix D and a schematic diagram of the computer interfaces is shown in Figure 7.3.

Figure 7.1

Master Hearing Aid

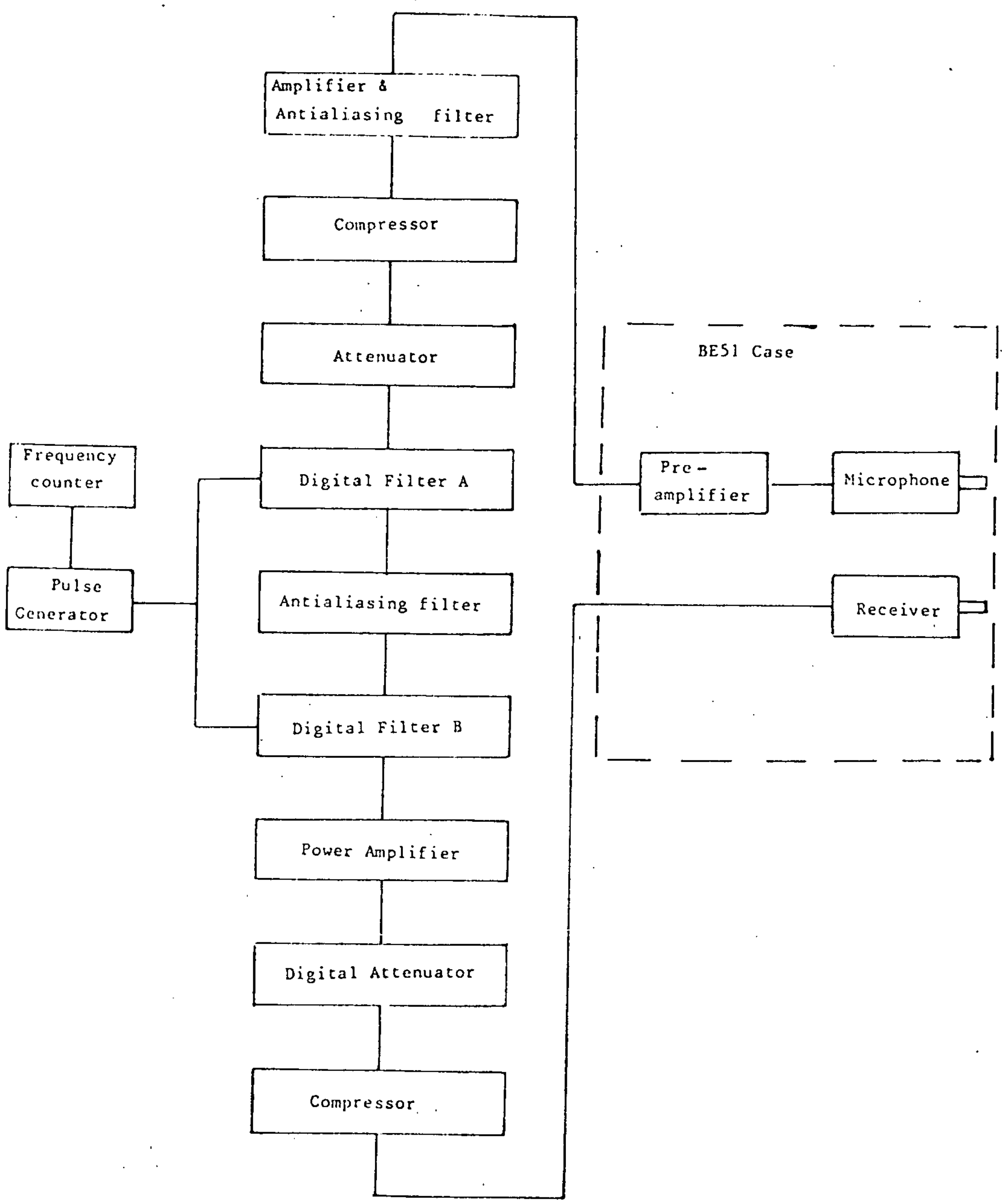


Figure 7.2

Subject Response Box

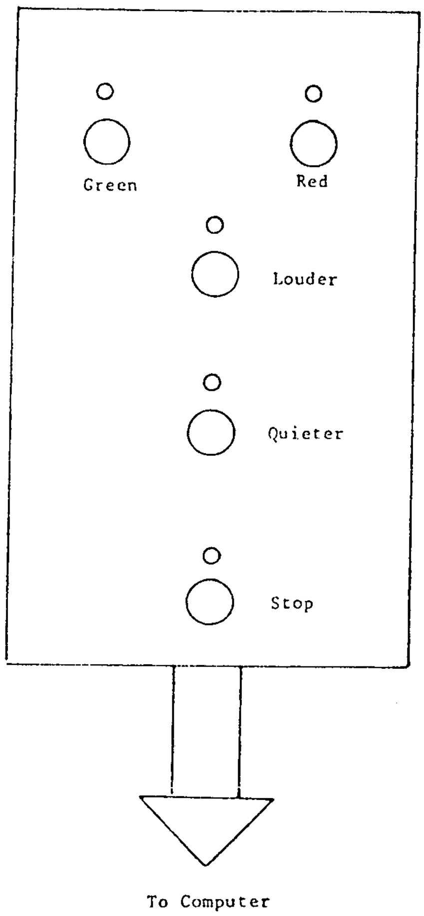
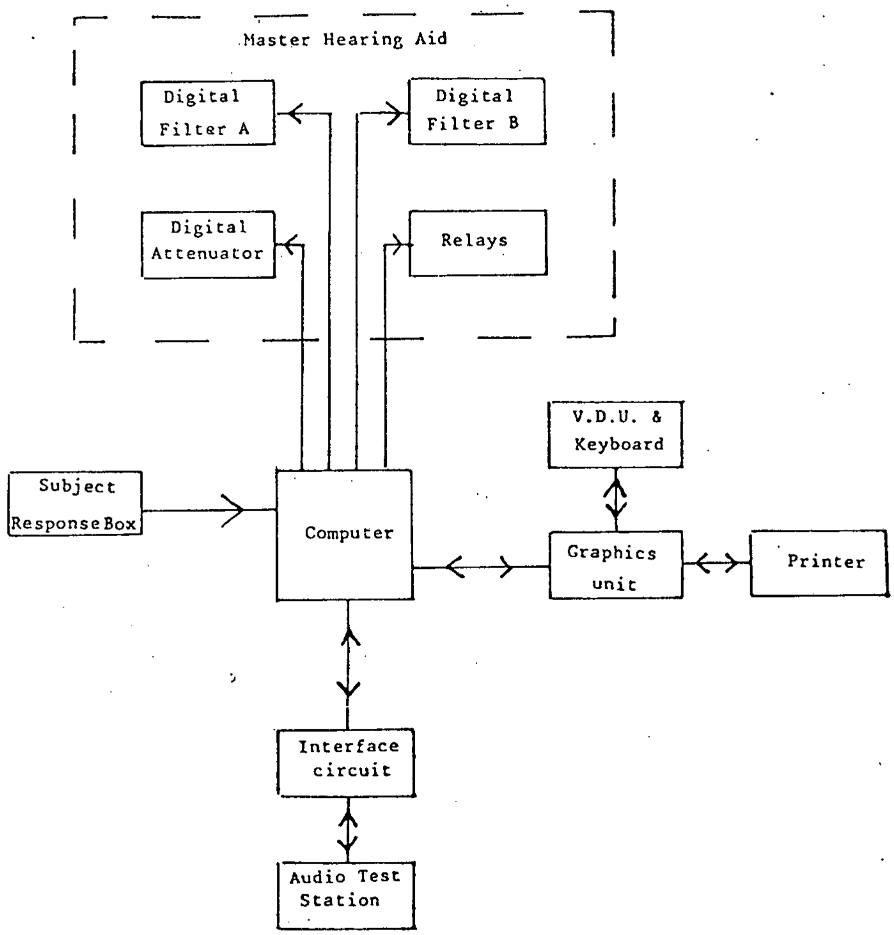


Figure 7.3

Computer and peripherals schematic



CHAPTER 8

COMPUTER SOFTWARE DEVELOPMENT

8.1 Introduction

The decision to use a microcomputer based master hearing aid and hearing aid measurement system necessitated the writing of a large number of wide ranging computer programs, all of which were written over a two year period by the author. Due to severe time constraints and the fact that the author was the sole user of the system, the emphasis placed on the software development was to produce programs that worked, rather than write elegant well-structured programs that were very 'user friendly'.

All programs were written in FORTRAN as it was the most suitable high level language available for the microcomputer system. However, it is not by any means an ideal language with which to write programs involving graphic displays, data base manipulation, user interactive programs and hardware control. The programs are reproduced in Appendix G with documentation in the comment statements. Only the main functions and salient points in the development of the programs will be described here.

8.2 Graphics Software

As described in Section 7.4, a 'Graffix' system was used for the presentation of a memory mapped graphics display that could be dumped to the dot matrix printer. This system is 'Tektroniks compatible' thus allowing 'Industry standard' graphics packages to be run. Unfortunately, the standard packages available were too large to be incorporated as part of the hearing aid measurement programs, as they were designed to plot data from files and not to be used on short plotting and display routines within a large program. It was therefore necessary to write the graphics display routines from scratch. The routines developed allowed axes to be drawn and labelled, data points to be scaled and plotted and written material presented. A subroutine was used to change the X and Y coordinates into the codes required for the Graffix unit to facilitate efficient programming.

8.3 B & K Audio Test Station

8.3.1 Introduction

As described in Section 7.3.2.2, the microcomputer was required to control all the functions of the B & K audio test station and to monitor the frequency and level of the input and output signals.

8.3.2 Calibration

For the calibration procedure of the microphone of the audio test station the computer program only displayed changes in calibrated level on the screen and ended the calibration procedure if no change had occurred within approximately 20 seconds. A prompt was then displayed on the screen asking if the calibration had finished. If the user replied 'No' the calibration routine was repeated. This was found to be a quick, easy to use and read calibration procedure.

8.3.3 Correction curve read in

The B & K audio test station has the facility to store the correction voltages required to produce a flat sound pressure level at its input microphone for its output frequency sweep. It does not use a compressor microphone for this purpose.

The correction curve read in program included a validation facility to check that the curve had in fact been read in correctly. This was found to be useful, as the audio test station took some time for its memory to stabilize after being switched on.

8.3.4 Function and level select

The program asks the user for the required test function, including the required sound pressure level and the facility to measure frequency response and 2nd and 3rd harmonic and intermodulation (difference) distortion.

The program checks that this information has been correctly received by the audio test station by monitoring the X and Y position of the pen that indicates the mode and level on the test chart (Figure 8.1).

8.3.5 Data collection

The actual frequency output of the test station signal was monitored directly on the relevant eight data lines. This was multiplexed with the measured sound pressure level information and selected by sending a chip select pulse. A change of frequency was also monitored by interrogating the computer's input latch which was triggered by the frequency change pulse from the audio test station.

After each change in frequency a stop signal was sent by the computer and when the data acquisition was finished, a 'run' signal was sent. This was a safeguard measure as there was no other way that the audio test station could check that the computer program was ready to move on to the next frequency.

In actual fact the computer program ran much faster than the audio test station and this 'safeguard' procedure was not strictly necessary. It did, however, provide the facility of a keyboard controlled pause in the program, resulting in a corresponding pause of the audio test station run.

The frequency information was necessary for data recording and plotting and also for the correct monitoring of the distortion tests, as these began and ended at different frequencies from the frequency response test.

The B & K audio test station obtains its large dynamic range (over 100 dB) by means of autoranging amplifiers. It was therefore necessary for the computer to monitor whether an autorange was taking place so as not to record incorrect data. Time delay routines were built into the program to allow the levels to settle after each frequency or range change.

As the speed of the A-D converter and the computer program were more than adequate to deal with the speed of the audio test station runs, it was decided to use a data check routine to help eliminate any contamination of the data by spurious noise. Data was therefore only accepted if two consecutive repeated measurements were found to be identical.

Frequency and level information were coded and plotted using the graphics routines.

8.4 Data Base Software

The data base software allowed all the details of the hearing aid response measurements to be stored. In addition to recording the test mode and level and frequency response data, the data base program requested and recorded information concerning the aid settings used and three lines of 'free comments'. A more detailed description of the data base is given in Appendix G.

Hearing aid data could be accessed and displayed according to record number or test condition information.

The FORTRAN random access file structure was based on 128 byte 'records'. One 128 byte record was used for hearing aid settings and test condition information. This was more than adequate and allowed for any possible expansion in the type of aid settings or test conditions available. Two 128 byte records were used for the frequency response data. This comprised of the 256 discrete frequencies and 8 bit level information obtained from the audio test station. The three 'free comment' lines took up another two 128 byte records as each line consisted of 80 characters. Each hearing aid test therefore took up five 128 byte 'records'. A separate data file was set up for each hearing aid and could be accessed using its eleven character 'name'.

Hearing aid records could also be erased or overwritten by means of an erase 'flag' written into their records.

8.5 Data Manipulation Software

To allow hearing aid response data to be manipulated for such purposes as aid comparisons and insertion gain measurements, a facility was included to allow rescaling of responses, inversion of responses and subtraction of responses. The manipulated responses could be displayed on the screen and printed on the dot matrix printer.

8.6 Hearing Aid Selection Software

Programs were written to allow the computer to select from the data base the aid which most closely fitted that required by a selection procedure. The hearing aid data base included much information that was not required for 'selection' procedures (e.g., 256 discrete frequencies, many test conditions and three comment lines), and so to speed up the selection program another 'selection file' was set up to automatically record only the information required by the selection procedure, i.e., the aid gains and maximum outputs at audiometric measurement frequencies.

Another file recorded general information concerning each hearing aid, such as which controls it contained and its microphone position.

The selection programs were written for two formally defined hearing aid selection procedures (Byrne, 1976, 1978; Berger, 1979). The computer selection programs selected the 10 best aids according to each procedure based on the relevant audiometric information for the subject.

Since data was not recorded for all possible aid gains, the program took the maximum aid gain and iteratively reduced it to obtain the best 'fit' for that aid. The best fit was arbitrarily defined as occurring when the minimum sum of the modulii of the differences between the required and actual gains at each specified audiometric frequency was obtained. This method assumes equal importance of each frequency and was used because no information is given in the prediction methods as to how to compare aids that differ from the required response.

Each hearing aid tone control setting could be included in this selection program as could different earmould options. The facility also allowed the use of real ear aid measurements for the selection procedures as well as coupler or simulator measurements.

8.7 Digital Filter Software

The digital filter required two programs for its operation. One to create the filter files which contained the coefficients for the digital filter, the other to load the coefficients into the digital filter.

The filter creation program was supplied with the digital filter by the manufacturer. It is based on a "host windowing transform" (Appendix B). A selection of Hilbert transformers was also supplied which could create filters with various 'transition bandwidths' (cut-off rates) and impulse response lengths.

Empirical experimentation based on theoretical considerations was used to find the optimum combination of data points, transition bandwidth, sampling rates and impulse response lengths that gave smooth frequency responses with adequate low frequency cut-off rates and adequate bandwidth.

The filter creation program calculated filters to fit the requirements supplied in a data file. This data file contained the required 'breakpoints' in terms of frequency and gain. The 'host window' approach to filter creation treats these breakpoints as the cut-off limits of low pass filters, and so the filters it produces appear as a series of steps between the 'breakpoints'. In order to create filters with smooth slopes it is necessary to put in sufficient data points. It would, however, be tedious to have to calculate and type in the required 60 data points for the creation of, say, a +6 dB/octave filter.

A program was therefore written that 'joined up' required breakpoints with 'straight lines' of required slope and interpolated the 60 data points for the filter data file.

Programs were also written that allowed the addition of and subtraction of filter data files to create new filters. The filter load program enabled either of the two digital filters to be loaded with the required filter and the gain rescaled if required.

8.8 Digital Filter Aid Simulation

To enable the digital filter to be programmed automatically to simulate any hearing aid response in the data base, a routine was written that took the required number of frequency/gain data points and coded them into the correct form for the digital filter creation program.

8.9 Pair Comparison Software

The pair comparison test (Chapter 10) required computer control of the digital attenuator and digital filter and channel relays in response to inputs from the subject response box and the terminal keyboard.

Programs were written that loaded the required filter responses to be compared into the digital filter when the subject pressed the corresponding channel button. The attenuator levels were reduced or increased by the computer according to whether the 'louder' or 'quieter' buttons on the subject response box were pressed. The program 'remembered' the attenuator settings for each channel, and so they did not require readjustment when the channels were switched back and forth repeatedly. For the comparison of tape recorded stimuli on two channels the computer switched tape channels via relays as well as digital filter responses. The subject indicated the end of the comparison by pressing the 'stop' button on the response box.

To facilitate speedy and error free loading of filter files, a program was written that allowed data files of all the required pairs of filters and their scale values to be created and numbered. This meant that during the pair comparison tests all that was required of the operator to call up a new pair of filters was to type in a number rather than the actual filter file names and scale values.

8.10 Uncomfortable Listening Level Software

A program was written that 'automatically' found the subject's uncomfortable listening level. The computer increased the level of the test stimulus in 5 dB steps until the patient signalled it was uncomfortably loud by pressing the 'quieter' button on the subject response box. The program then reduced the level by 10 dB, and then continued increasing the level in 5 dB steps. This was repeated for three or more presentations until at least 50% of the ascending presentation levels were signalled as being uncomfortably loud. This level was then taken to be the uncomfortable listening level. The program also controlled the digital filtering of the required test stimuli so that the uncomfortable listening level for various frequency responses could be measured.

8.11 Comfortable Listening Level Software

This program was similar to that for the pair comparison test except that only one 'channel' was operative. The subject could therefore adjust the level of the signal for the required filter response.

8.12 Utility Programs

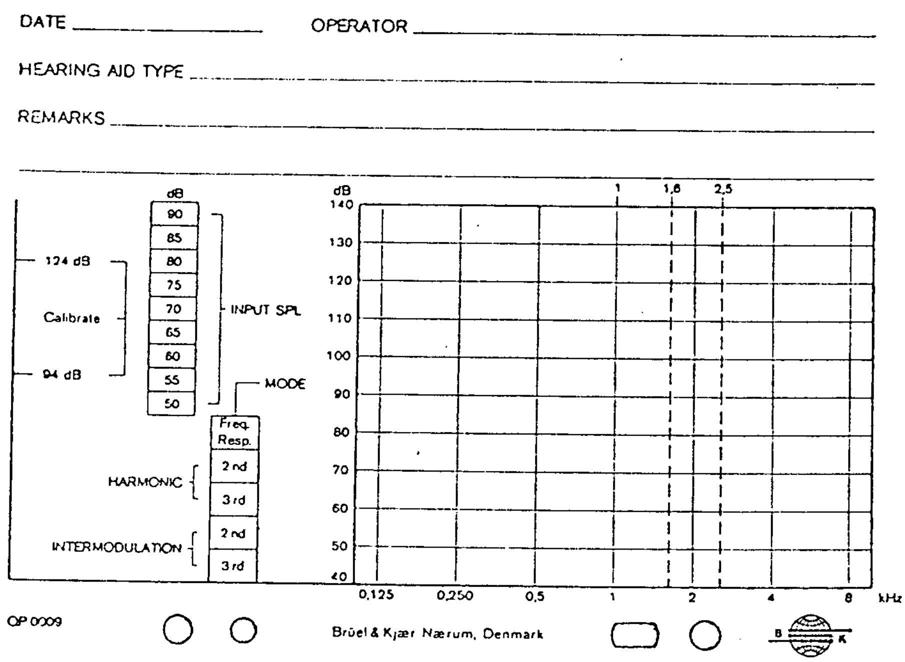
Two utility programs were written to facilitate the writing and implementation of the other programs.

The first one enabled messages to be displayed on the screen without having to write them into format statements as must usually be done in FORTRAN.

The second program allowed the user to input file names without having to pad out any unused characters with spaces as was required by the normal FORTRAN routine. For example, to open a file called TEST.AID, FORTRAN requires the user to input TEST _ _ _ _ . AID. These two programs were written as subroutines and used extensively in the writing of the main programs.

Figure 8.1

Audio Test Station response chart



CHAPTER 9

DEVELOPMENT OF AUDIO AND VISUAL EVERYDAY LIFE SIMULATION (AVELS) TEST

9.1 Introduction

Up until the present time there does not exist any clinical test, objective or subjective that has been shown to satisfactorily predict the real-life performance of hearing impaired subjects using hearing aids. From the review of the research literature in Chapters 2 and 3 it would appear that no attempt at constructing such a test has explicitly been made although there have been attempts to incorporate more 'realistic' features into speech tests.

Any single speech test in a clinical situation cannot be representative of real-life due to the wide variety of individuals, environments, situations and stimuli that are involved. This was realised by Davis (1946) when he wrote "The variety of voices, voice levels, speech sounds and acoustic environment introduces great complexity into the apparently simple criterion of 'Intelligibility of Speech'". Comparisons obtained under any arbitrary set of conditions are strictly valid only for those special conditions, and they become more and more uncertain as they are extrapolated or generalized to other conditions. Davis concluded that speech tests were therefore not a very satisfactory way of deciding which was the best hearing aid for an individual. One approach to predicting the overall real-life performance of hearing aids would be to use a test that included in the correct proportions, all the acoustic and linguistic variables that occur in an individual's real life. The complexity involved makes this impractical, as a different test would be required for each individual.

Any test attempting a real-life simulation and capable of clinical applicability must of necessity therefore be a compromise using only those situations felt to be of most importance. As the optimum aid in one situation need not necessarily be the optimum in another, and unless aids with automatically or user switchable characteristics are available, the correct balance between the situations is necessary for the test to predict the overall optimum single aid.

One possible approach to this problem would be to have a range of tests covering different situations and conditions and to select only those that are most appropriate to an individual's real life communication environment. If objective scoring is used then it would be necessary to weight the results of each test in accordance with the importance this situation has for the individual, whereas if subjective judgements are employed then the weighting can be done 'automatically' by the subject.

An important limitation of restricting tests or evaluations to those situations characteristic of a subject's present lifestyle, is that he may be avoiding certain situations due to difficulties experienced at that time. As the 'ideal' hearing aid may enable him to change his lifestyle by helping in these situations, it is important to establish the performance of the person with the aid under all conditions.

9.2 <u>AVELS Test Requirements</u>

The AVELS test was developed to attempt to provide a clinically feasible method of predicting real-life hearing aid performances and preferences. Test material was required that presented meaningful speech stimuli at a variety of realistic levels and S/N ratios in a range of audio and audio visual situations and with a variety of speakers.

A subjectively judged test was required so that subjects could use whatever criteria they employed in real life to evaluate their performance. The added facility of simultaneously being able to objectively score the test was also required. This provided information as to the possible reasons behind the subjective judgements and indications as to possible refinements and simplifications of the tests. The objective scoring facility also provides information concerning the particular speech perception difficulties experienced by the subjects, for example, phoneme confusions. This could be potentially useful for communication training and rehabilitation programmes.

In addition to speech material, sounds of an everyday nature were included in the AVELS test. The aim was to allow subjects the chance to judge the subjective quality of real-life sounds other than speech, and to determine any potential discomfort that they might cause.

The environmental sounds used in the test and the subtitles presented simultaneously on the screen are shown in Table 9.1. The subtitles informed the subject what the sounds were, as some hearing-impaired subjects might find it difficult to identify the sound alone.

Table 9.1 Environmental sounds, SPL in dB(A)

	rms fast	peak
whispering	58	73
birdsong	56	69
washing machine	87	93
traffic	74	85
train	78	91
aeroplane	80	88
inside a car	65	79
shouting	83	94
door slamming	76	90
telephone ringing	73	84
bicycle bell	66	80
water running	52	67
music	78	91

Note: Ambient noise level in room was ≃ 35 dB(A) L_{eq}.

9.3 Avels Test Speech Material

It was decided to use the SPIN test sentences for the AVELS test for the following main reasons:

(i) Content

The SPIN test sentences were considered to be the best attempt so far to produce an extensive representative set of everyday listening tasks. This view was also expressed by McCormick

(1980). It was not considered feasible nor necessary to develop new speech material for this study.

The fact that the SPIN test was designed for 'American Standard English' was not regarded as a problem as only a few lists in the SPIN test contain 'Americanisms' and these lists were omitted in the AVELS test.

(ii) <u>Scoring procedure</u>

The SPIN test material is designed for last word scoring; this allowed an efficient method of objectively scoring the test while not interfering unduly with the individual's subjective rating of the AVELS test. The use of a sentence test in which every word was scored would have been practically difficult to administrate and would have detracted from the real-life aspect of the test as far as the subjective ratings were concerned, as in real life people are not required to repeat every word that is said to them. The 'compromise' of having only to repeat the last word in each sentence also ensures that the subject is motivated to listen to the material, and so not only assesses the quality but also subjectively judges the intelligibility as well. By contrast this motivation occurs naturally in real-life situations where a person listens because he wants to understand what is being said.

The SPIN test was designed to be scored by the key word using whole word scoring. Due to the phonetic balance it would be possible to score it phonemically although no standardization of the test using phonemic scoring has to the author's knowledge been attempted.

The recording of the subject's responses would facilitate this approach.

9.4 Repeated Measures Design

In using tests to compare hearing aid performance one has the choice of either repeating the same test twice or of using different tests of equal difficulty. The former approach suffers from the drawback of involving learning or memory effects, the subject remembering some of the words or clues from a previous test. The latter approach involves the difficult tasks of developing tests that are equally difficult for all people. The development of equivalent tests involving all the conditions and constraints required for a 'real-life' audio visual test was felt to be too formidable a task to attempt for this study. The AVELS test was therefore designed to be used repeatedly with corrections being made for learning effects. The possibility of using subsets of the test as alternative lists was, however, included in the design.

9.5 Test Conditions Used

Little information is available concerning the range of conditions experienced by normally hearing or hearing-impaired subjects and their relative frequency of occurrence and relative importance. An empirical approach to the choice of test conditions was therefore adopted.

Speech and background noise levels in various environments have been measured by Pearsons *et al* (1977), and these were taken as guidelines for the levels of the AVELS test. The majority of listening situations for most people involve face-to-face contact. There are of course obvious exceptions to his, notably people who spend a lot of time using the telephone or other wired or wireless audio communications systems; but their problems will be complicated by the limitations of the communication system itself and so were not included in the test. The inclusion of a test condition over the telephone line could have been included, but the uncontrollable factors of the particular telephone receiver and any inductive coupling used in real life also led the author to leave this condition out of the AVELS test. However, to give some examples of 'limited quality' speech, radio and television simulations were presented.

To represent the range of situations where visual speech reading clues are available in addition to an audio only condition, two levels of ease of speech reading were included. A 'good speech reading' condition was presented where a clear view of the face and neck was available, and a 'poor speech reading' condition where the head was angled down. Attempts to try and vary the lighting conditions during recording of the test to obtain a full face poor visual condition were not successful, as they upset the video camera's colour balance and so were abandoned.

A male and a female talker were included in the test to give examples of the speech of both sexes.

9.6 Competing noise

9.6.1 Type of noise

Speech has been shown to be one of the most commonly encountered and most difficult competing noises in real life. Competing messages can lead to difficulties in identifying and attending to the test stimuli, especially if they are of greater interest. Whilst there may be real-life effects they are difficult to control and interpret. Gaps in competing messages can lead to test items being heard too easily and while this again is a real-life occurrence it may detract from the efficiency of the test.

To overcome these problems while still retaining the speech quality of the competing signal multi-talker speech babble was used. This type of competing message was also used for the original SPIN test for similar reasons.

9.6.2 Recording procedures

The babble was produced by recording 4 male and 4 female talkers reading aloud. Approximately 30 seconds of 16 speaker babble were obtained by mixing together two separate sections of the 8 talker babble and a continuous recording of this material was produced using a tape loop. In addition 'babble shaped noise' was mixed in to fill any possible remaining gaps in the signal. The noise, shaped to the long term average r.m.s. spectrum of the babble, was below the level of the babble, had no effect on its spectrum and was not subjectively noticeable.

9.7 AVELS Test Details

One hundred sentences comprising lists 2.1 and 2.4 of the SPIN material were used for the AVELS test, providing one hundred different scored key words. The actual sentences and conditions used are shown in Table 9.2. The SPIN material provides the possibility of using two lists with the same key words but in different high and low predictability locations. This approach was not, however, felt to be an advantage for the AVELS tests as subjects might discover this pattern. The use of lists with different key words also allowed a wider range of words to be included.

The 100 sentences were presented by two talkers, 84 by a male talker and 16 by a female talker. Two of the sentences were used without visual clues as a radio simulation and two with visual clues as a television simulation. This was achieved by recording the talker's voice after it had been picked up by a microphone and processed by the amplifier and loudspeaker of a domestic television. This set up is shown in Fig. 9.1.

Eight of the sentences spoken by the male speaker were used to present speech from different directions: behind, in front and from the left and right of the subject. The remaining seventy-two sentences spoken by the male speaker were divided into two groups of thirty-six. This allowed for comparisons between various room conditions (e.g., reverberant and non-reverberant conditions) or provided the possibility of using two 36 sentence lists for a short version of the test. Each of these thirty-six sentence sections had twelve sentences spoken with no visual clues (blank screen), twelve with good visual clues (full face looking straight ahead) and twelve with poor visual clues (face looking down).

Table 9.2 AVELS test sentences and conditions

SPIN Form 2.1

Predictability		
(H)	1.	The watchdog gave a warning growl.
(H)	2.	She made the bed with clean sheets.
(L)	3.	The old man discussed the dive.
(L)	4.	Bob heard Paul called about the strips.
(L)	5.	I should have considered the map.
(H)	6.	The old train was powered by steam.
(H)	7.	He caught the fish in his net.
(L)	8.	Miss Brown shouldn't discuss the sand.
(H)	9.	Close the window to stop the draught.
(H)	10.	My T.V. has a twelve-inch screen.
(L)	11.	They might have considered the hive.
(L)	12.	David has discussed the dent.
(H)	13.	The sandal has a broken strap.
(H)	14.	The boat sailed along the coast.
(H)	15.	Crocodiles live in muddy swamps.
(L)	16.	He can't consider the crib.
(H)	17.	The farmer harvested his crop.
(H)	18.	All the flowers were in bloom.
(L)	19.	I am thinking about the knife.
(L)	20.	David does not discuss the hug.
(H)	21.	She wore a feather in her cap.
(L)	22.	We've been discussing the crates.
(L)	23.	Miss Black knew about the doll.
(H)	24.	The Admiral commands the fleet.
(L)	25.	She couldn't discuss the pine.
(L)	26.	Miss Black thought about the lap.
(H)	27.	The beer drinkers raised their mugs.
(H)	28.	She was hit by a poisoned dart.
(H)	29.	The bread was made from whole wheat.
(L)	30.	Mr. Black knew about the pad.
(L)	31.	You heard Jane called about the van.
(H)	32.	I made the phone call from a booth.
(L)	33.	Tom wants to know about the cake.
(L)	34.	She's spoken about the bomb.
(H)	35.	The cut on his knee formed a scab.
(L)	36.	We hear you called about the lock.
(L)	37.	The old man discussed the yell.
(H)	38.	His boss made him work like a slave.
(H)	39.	The farmer baled the hay.
(L)	40.	They're glad we heard about the track.
(H)	41.	A termite looks like an ant.
(H)	42.	Air mail requires a special stamp.
(H)	43.	Football is a dangerous sport.
(L)	44.	Sue was interested in the bruise.
(L)	45.	Ruth will consider the herd.
(H)	46.	We saw a flock of wild geese.
(L)	47.	The girl talked about the gin.
(L)	48.	Paul can't discuss the wax.
(H)	49.	Drop the coin through the slot.
(L)	50.	I hope Paul asked about the mate.

Table 9.2 cont.

<u>SPIN Form 2.4</u>

Predictability

(L)	1.	I want to speak about the crash.
(H)	2.	Harry slept on the folding cot.
(L)	3.	She's glad Jane asked about the drain.
(H)	4.	The doctor charged a low fee.
(L)	5.	He had considered the robe.
(L)	6.	I haven't discussed the sponge.
(H)	7.	The guilty one should take the blame.
(L)	8.	You cannot have discussed the grease.
(H)	9.	The biscuits were kept in a jar.
(H)	10.	Let's invite the whole gang.
(L)	11.	Mr. White discussed the cruise.
(H)	12.	The sport shirt has short sleeves.
(L)	13.	They knew about the fur.
(L)	14.	We've spoken about the truck.
(H)	15.	The cushion was filled with foam.
(H)	16.	How long can you hold your breath?
(L)	17.	She wants to talk about the crew.
(H)	18.	The cow was milked in the barn.
(H)	19.	That accident gave me a scare.
(H)	20.	The kitten climbed out on a limb.
(L)	21.	You're glad she called about the bowl.
(L)	22.	The man could not discuss the mouse.
(H)	23.	He tossed the drowning man a rope.
(L)	24.	You hope they asked about the vest.
(L)	25.	You want to talk about the ditch.
(H)	26.	Stir your coffee with a spoon.
(L)	27.	We hear she called about the drum.
(H)	28.	Bob stood with his hands on his hips.
(H)	29.	The teacher sat on a sharp tack.
(L)	30.	She might have discussed the ape.
(H)	31.	The storm broke the sailboat's mast.
(H)	32.	At breakfast he drank some juice.
(H)	33.	He hit me with a clenched fist.
(L)	34.	Peter knows about the raft.
(L)	35.	The old man considered the kick.
(L)	36.	We have not thought about the hint.
(H)	37.	The team was trained by their coach.
(L)	38.	Bill hopes Paul heard about the mist.
(H)	39.	The king wore a golden crown.
(H)	40.	The sand was heaped in a pile.
(L)	41.	The boy can't talk about the thorns.
(L)	42.	Miss Brown will speak about the grin.
(H)	43.	The duck swam with the white swan.
(H)	44.	Let's decide by tossing a coin.
(L)	45.	She has a problem with the goal.
(L)	46.	Jane didn't think about the brook.
(L)	47.	He hears she asked about the deck.
(H)	48.	He got drunk in the local bar.
(H)	49.	The girl swept the floor with a broom.
(L)	50.	The class will consider the blast.

Table 9.2 cont.

	AVELS TEST First 36 test items						SPIN TEST	
Key word	Speech level	Noise level	Speaker	Condi- tion	Speech source	No. of phonemes	Form no.	Sentence no.
crop	1	0	M	GV	F	4	2.1	17
bloom	1	0	M	GV	F	4	"	18
knife	1	1	M	GV	F	3	"	19
hug	1	1	M	GV	F	3	"	20
cap	1	2	M	GV	F	3	"	21
crates	1	2	M	GV	F	5	"	22
wheat	2	0	M	GV	F	3	"	29
pad	2	0	M	GV	F	3	"	30
van	2	2	M	GV	F	3	"	31
booth	2	2	M	GV	F	3	"	32
cake	2	3	M	GV	F	3	"	33
bomb	2	3	M	GV	F	3	"	34
ant	1	0	M	PV	F	3	"	41
stamp	1	0	M	PV	F	5	"	42
sport	1	1	M	PV	F	4	"	43
bruise	1	1	M	PV	F	4	"	44
herd	1	2	M	PV	F	3	"	45
geese	1	2	M	PV	F	3	"	46
drain	2	0	M	PV	F	4	2.4	3
fee	2	0	M	PV	F	2	"	4
robe	2	2	M	PV	F	3	"	5
sponge	2	2	M	PV	F	5	"	6
blame	2	3	M	PV	F	4	"	7
grease	2	3	M	PV	F	4	"	8
bowl	1	0	M	AO	F	3	"	21
mouse	1	0	M	AO	F	3	"	22
rope	1	1	M	AO	F	3	"	23
vest	1	1	M	AO	F	4	"	24
ditch	1	2	M	AO	F	3	"	25
spoon	1	2	M	AO	F	4	"	26
drum	2	0	M	AO	F	4	"	27
hips	2	0	M	AO	F	4	"	28
tack	2	2	M	AO	F	3	"	29
ape	2	2	M	AO	F	2	"	30
mast	2	3	M	AO	F	4	"	31
juice	2	3	M	AO	F	3	"	32

Table 9.2 cont.

	AVELS TEST Second 36 test items						SPIN TEST	
Key word	Speech level	Noise level	Speaker	Condi- tion	Speech source	No. of phonemes	Form no.	Sentence no.
doll	1	0	M	GV	F	3	2.1	23
fleet	1	0	M	GV	F	4	"	24
pine	1	1	M	GV	F	3	"	25
lap	1	1	M	GV	F	3	"	26
mugs	1	2	M	GV	F	4	"	27
dart	1	2	M	GV	F	3	"	28
scab	2	0	M	GV	F	4	"	35
lock	2	0	M	GV	F	3	"	36
yell	2	2	M	GV	F	3	"	37
slave	2	2	M	GV	F	4	"	38
hay	2	3	M	GV	F	2	"	39
track	2	3	M	GV	F	4	"	40
gin	1	0	M	PV	F	3	"	47
wax	1	0	M	PV	F	3	"	48
slot	1	1	M	PV	F	4	"	49
mate	1	1	M	PV	F	3	"	50
crash	1	2	M	PV	F	4	2.4	1
cot	1	2	M	PV	F	3	"	2
jar	2	0	M	PV	F	2	"	9
gang	2	0	M	PV	F	3	"	10
cruise	2	2	M	PV	F	4	"	11
sleeves	2	2	M	PV	F	5	"	12
fur	2	3	M	PV	F	2	"	13
truck	2	3	M	PV	F	4	"	14
foam	1	0	M	AO	F	3	"	15
breath	1	0	M	AO	F	4	"	16
crew	1	1	M	AO	F	3	"	17
barn	1	1	M	AO	F	3	"	18
scare	1	2	M	AO	F	3	"	19
limb	1	2	M	AO	F	3	"	20
fist	2	0	M	AO	F	4	"	33
raft	2	0	M	AO	F	4	"	34
kick	2	2	M	AO	F	3	"	35
hint	2	2	M	AO	F	4	"	36
coach	2	3	M	AO	F	3	"	37
mist	2	3	M	AO	F	4	"	38

Table 9.2 cont.

	AVELS TEST Final 28 test items						SPIN TEST	
Key word	Speech level	Noise level	Speaker	Condition	Speech source	No. of phonemes	Form no.	Sentence no.
broom	1	2	M	TV sim.	F	4	2.4	49
blast	1	2	M	TV sim.	F	5	"	50
deck	1	2	M	Radio sim.	F	3	"	47
bar	1	2	M	Radio sim.	F	2	"	48
crown	1	2	M	AO	F	4	"	39
pile	1	2	M	AO	F	3	"	40
thorns	1	2	M	AO	B	4	"	41
grin	1	2	M	AO	B	4	"	42
swan	1	2	M	AO	L	4	"	43
coin	1	2	M	AO	L	3	"	44
goal	1	2	M	AO	R	3	"	45
brook	1	2	M	AO	R	4	"	46
growl	1	0	F	GV	F	4	2.1	1
sheets	1	0	F	GV	F	4	"	2
dive	1	1	F	GV	F	3	"	3
strips	1	1	F	GV	F	6	"	4
map	2	0	F	GV	F	3	"	5
steam	2	0	F	GV	F	4	"	6
net	2	2	F	GV	F	3	"	7
sand	2	2	F	GV	F	4	"	8
draft	1	0	F	AO	F	5	"	9
screen	1	0	F	AO	F	5	"	10
hive	1	1	F	AO	F	3	"	11
dent	1	1	F	AO	F	4	"	12
strap	2	0	F	AO	F	5	"	13
coast	2	0	F	AO	F	4	"	14
swamps	2	2	F	AO	F	6	"	15
crib	2	2	F	AO	F	4	"	16

Key:

Speech level	Noise level	Speaker	Condition	Speech source
1 ≃ 65 dB(A)	0 = Quiet	M = male	GV = good visual	F = front
2 ≃ 55 dB(A)	1 = 65 dB(A)	F = female	PV = poor visual	B = back
	2 = 55 dB(A)		AO = audio only	L = left
	3 = 45 dB(A)			R = right

Each of these groups of twelve sentences was made up of pairs of sentences presented at two vocal efforts, 'raised' and 'casual', and with three background noise levels, 'quiet' and two babble conditions 10 dB apart in level.

The sixteen sentences spoken by the female talker were divided into two groups of eight sentences and one group presented with good visual clues (full face) and the other without visual clues (blank screen). Each of these groups of eight sentences was made up of pairs of sentences presented at two vocal efforts 'raised' and 'casual' and with two background noise levels 'quiet' and a babble condition at a level equivalent to the loudest babble condition for the male speech.

9.8 Characteristics of Male and Female Talkers

The author was the male speaker for the recording the AVELS test and his wife was the female speaker. Both have had extensive experience of communicating with deaf people and have been informed on numerous occasions that they are fairly easy to lipread. The author's closely trimmed beard was therefore considered not to be a problem. Both speakers use southern English and have no accents or peculiarities of speech.

The 'good visual' conditions of the AVELS test were therefore considered to be close to an optimum real-life situation whereas the 'poor visual' conditions were more representative of the average situation.

9.9 Speech Spectra, Levels and Characteristics

The spectra of the speech of the male and female speakers and the multi-talker babble are shown in Figures 9.2 to 9.5. These spectra are similar to published speech spectra (Pearsons *et al*, 1977) and so appear to be fairly representative.

The speech was recorded for different realistic vocal efforts to obtain speech at different levels rather than using unrealistic electronic attenuation. The vocal efforts involved affected the speech characteristics and spectrum shape to some extent as shown by Pearsons *et al.*

9.10 Recording Procedure

The AVELS test was recorded in an acoustically treated room of approximate dimension 12' × 10' × 8'. The video recording was made using a Sony DXC1200 colour camera (with a Canon zoom lens) linked to a Sony VO2850 colour video 'U'matic recorder. This system offered the highest possible quality of recording in the absence of professional broadcast equipment. The arrangement used for the recording is shown in Fig. 9.6.

The lighting consisted of three Malham 1 kW Halogen flood lamps directed at white paper reflectors. This was arranged so that facial or laryngeal movements were not occluded by shadow and normal lingual movements were observable while the talker's face still kept a normal appearance. This was accomplished after much trial and error, as a compromise was required between the extremes of lighting the face so that no shadows were produced but the face looked 'washed out' and providing a good 'depth' to the face but obscuring the tongue and teeth and throat with shadows. The lighting arrangement was also required not to distract the talker when looking at the camera. The zoom lens was adjusted so that the final display of the talker on the 20 inch Sony colour video monitor was exactly lifesize.

The fidelity of the video recording was optimized using a BBC colour camera Gray scale test chart Number 57 in conjunction with a BBC Flesh Tone Reference Test Chart Number 61. Other adjustments were made precisely according to manufacturers' instructions. A large matt pale blue background was used, that produced a pleasant shadow-free backdrop.

To pick up, measure and deliver the audio signal to the video recorder a B & K 1" microphone and measuring amplifier were used. A 1 kHz calibration tone and pink noise were also recorded on to the tape to facilitate the calibration of editing, analysis and presentation. The microphone was positioned some 15" away from the speaker's mouth such that it was out of the field of view of the camera, and did not cast any visible shadows.

The author operated the video recorder by means of a remote control unit. A television monitor placed directly behind the camera allowed the correct position of the talker's screen image to be maintained. Attempts at reading the sentence material from 'cue cards' tended to give a stilted unnatural delivery and a somewhat staring, unnatural visual effect. These effects are often apparent even in the delivery of experienced professional broadcasters. To overcome this problem for the recording of the AVELS test the speakers memorized each sentence immediately before speaking it. This successfully produced a natural visual and auditory delivery of the sentence material.

As inevitably some mistakes were made by the speakers, no attempt was made to record a continuous fully correct run-through of the test. Many repeat takes were made of various sections of the test to ensure that the optimum recordings and presentations had been made, as it was important that the final test only included material that had been delivered in a natural and realistic manner.

The appropriate optimum recordings of the various sentences were at a later date edited together by the author to make the continuous complete version of the test. The environmental sounds were recorded live through a 1" B & K microphone, using a Nagra IVSJ tape recorder and were calibrated to provide realistic sounds at the correct real-life level.

9.11 Editing

The author produced the edit master of the AVELS test on a 'U'matic editing suite. Care was taken to obtain equal intervals of approximately five seconds between the sentences to enable the subject's responses to be given. The video signal was faded to grey during these five second periods. The calibration tones recorded on the master tapes ensured that the correct levels were recorded onto the edit master.

After editing the final version of the speech material of the test the competing multi-talker babble was edited on to the second audio channel. The babble was always faded out during the 5 second inter-sentence intervals. This allowed for the optimum monitoring by the experimenter of the subject's verbal responses during the test. If the babble continued during the subject response periods the test also becomes one of the experimenter's listening ability as well as the subject. Many speech in noise tests (e.g., FADAST test, see Appendix H) suffer from this problem.

All audio recordings were made to give the optimum signal to tape noise ratio while still maintaining the fidelity of the signal. The relative levels of the sentence stimuli were not altered during the editing process. The competing babble levels for each particular condition were obtained using a stepped attenuator to maintain accuracy.

The subtitles for the environmental noises were recorded on a professional subtitling machine and were transferred on to the edit master tape along with the environmental noises. Copies of the edit master were then produced to be used in the research study.

The final version of the videotape was shown to the professional staff at the editing/production studio and was regarded as satisfactory. It was felt to be of as high a quality as could have been achieved without the availability of broadcast standard equipment.

9.12 Presentation Equipment, Environment and Methods

The AVELS test was presented through the equipment shown in Figure 9.7 and in the room shown in Figure 9.8. The television monitor, video playback machine, amplifiers and loudspeakers used were of the highest quality. Their specifications are given in Appendix D.

To obtain as realistic a presentation as possible, the speech loudspeaker was placed as close as possible to the 'mouth position' of the image on the monitor screen. A thick iron shield was required to prevent the stray magnetic field emanating from the loudspeaker affecting the television picture.

For the majority of test conditions the speech signal was presented from this loudspeaker, the babble being presented through the three identical loudspeakers placed behind and to each side of the subject. For the test condition requiring the speech to come from these directions a switching circuit was built that enabled the speech to be routed through any one loudspeaker while routing the babble through the other three. Stepped attenuators were used to present the speech and babble at their required levels. The calibration tones and noises recorded at the start of the test tape allowed these levels to be checked quickly to ensure consistency throughout the experiments.

The subject sat facing the video monitor, 1.8 metres from the speech loudspeaker. This was shown to be in the reverberant field (Appendix I). The subject's ears were slightly above the horizontal axis of the loudspeaker and slightly below the ear level at the video face image. This was felt to be a good compromise, as it permitted an adequately flat free field response of the test system at the subject's ear while maintaining a realistic position for face-to-face communication.

The author's original idea of presenting the stimuli in various acoustic environments had to be abandoned as it was neither possible to easily modify the rooms to give changes in reverberation time nor to use a variety of rooms. The possibilities of using artificial reverberation or re-recording the audio track in different rooms were considered but rejected on the grounds of them being unvalidated as providing realistic simulations.

9.13 Test and Subject Response Monitoring

The author, sitting in the control room, monitored the test presentation and the subject's responses using a television monitor, electret microphone, amplifier and headphones. The subject responses were also tape recorded for later analysis in case of possible ambiguities. The subjects were observed visually by a one-way mirror to ensure that no problems arose and that they maintained their correct seated position throughout the test. These arrangements allowed the author to ensure that any problems with the test presentation or subject responses would be noticed immediately.

Figure 9.1

T.V & Radio simulation audio recording method

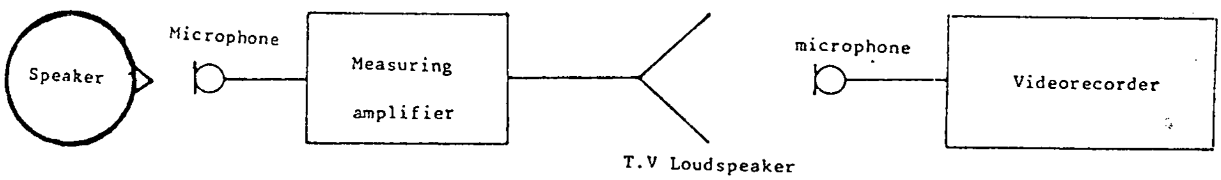


Figure 9.2

Typical spectra of AVELS test male speech

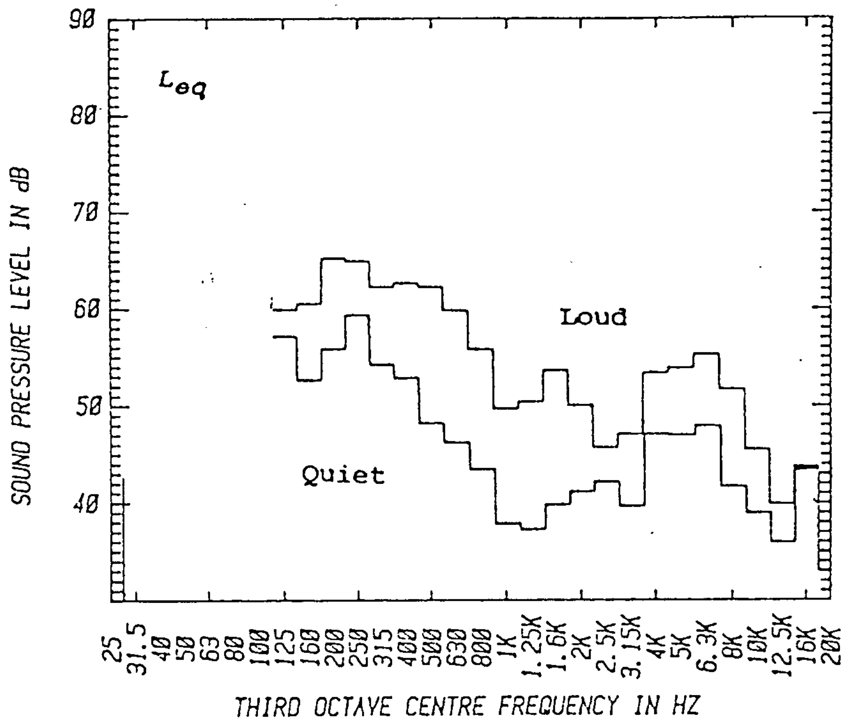


Figure 9.3

Typical spectra of AVELS test female speech

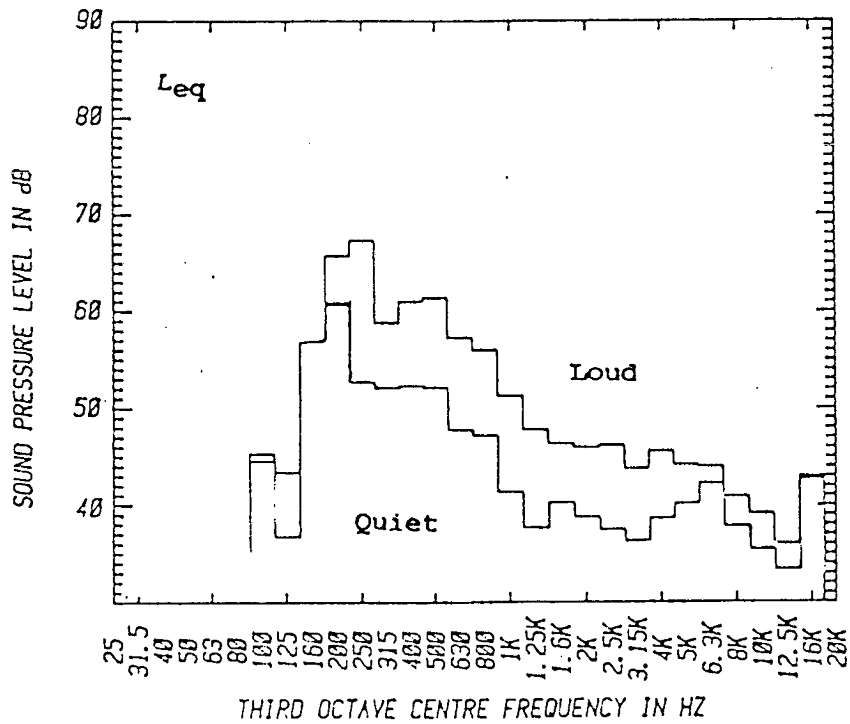


Figure 9.4

AVELS test television simulation speech spectrum

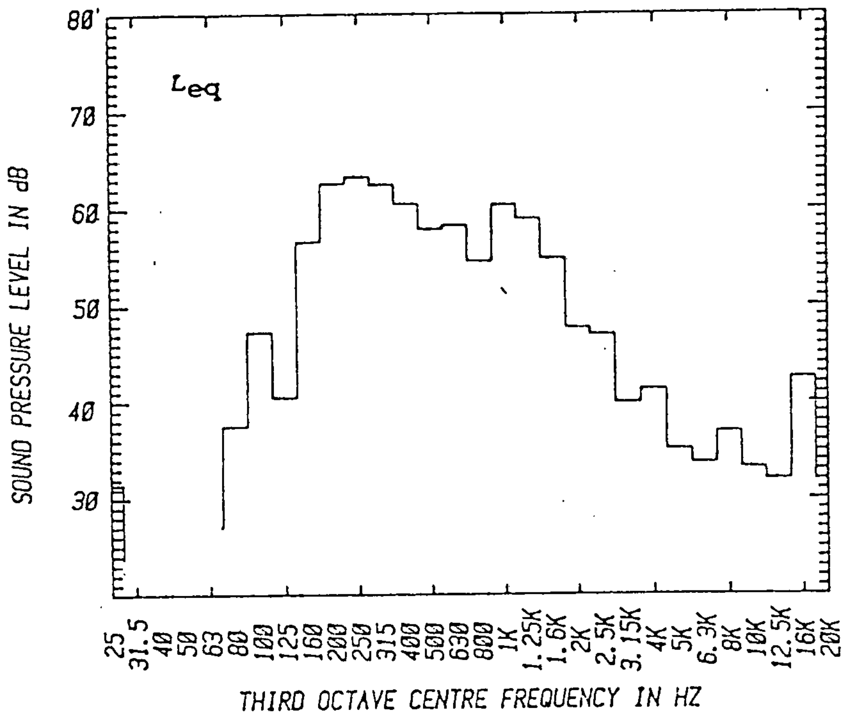


Figure 9.5

Speech babble spectra

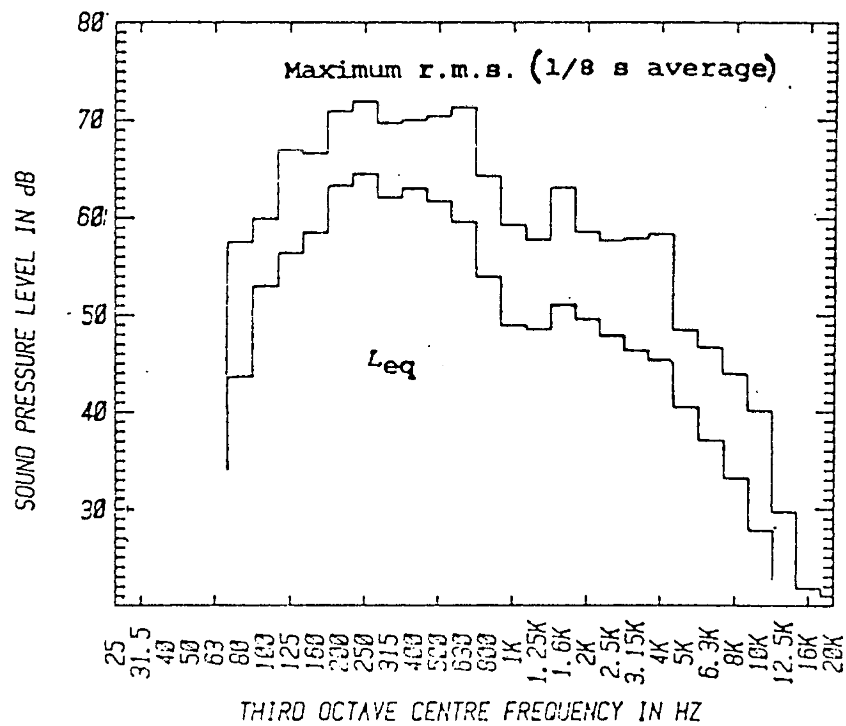


Figure 9.6

AVELS recording arrangement

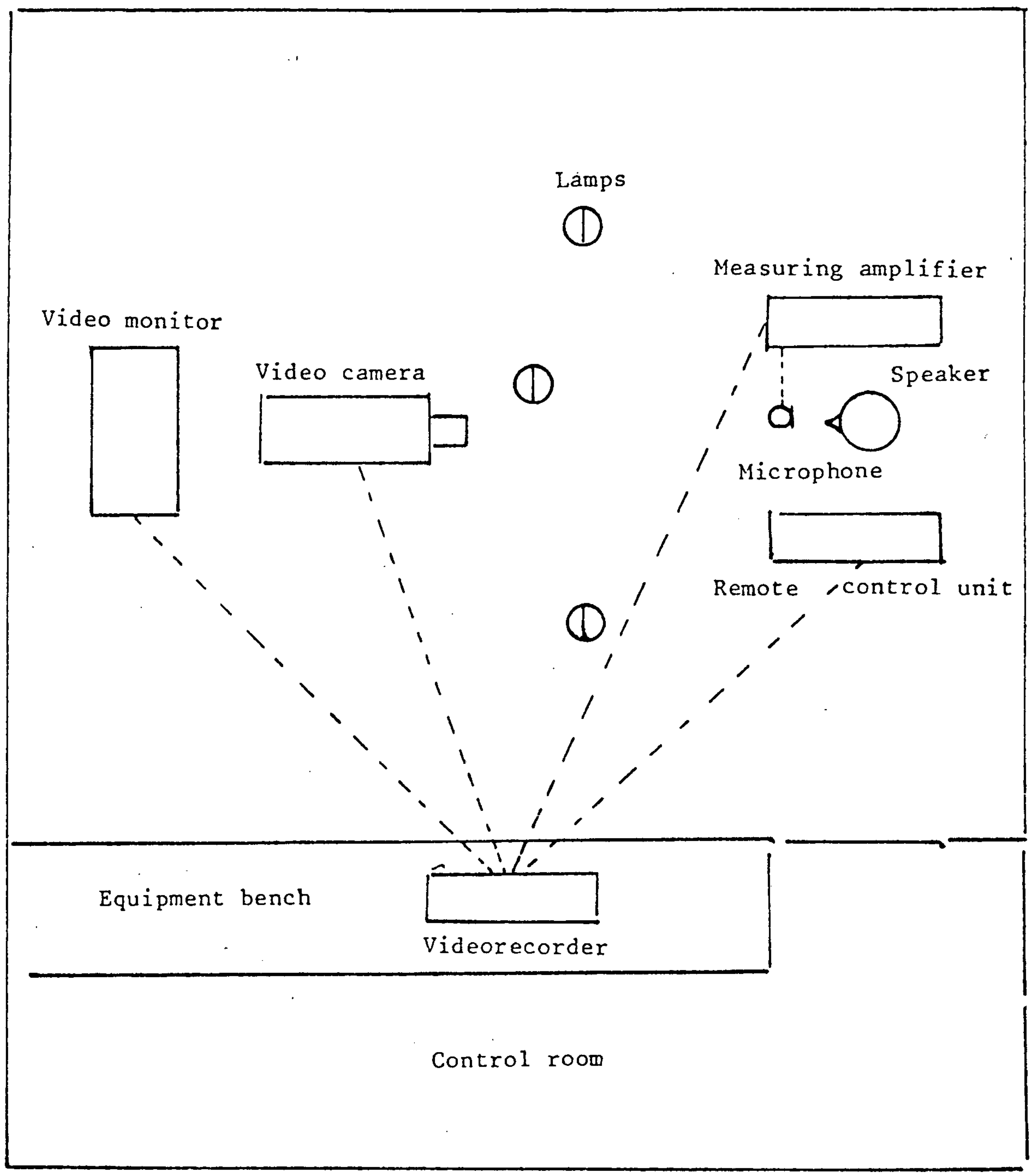


Figure 9.7

Audio & Visual presentation system

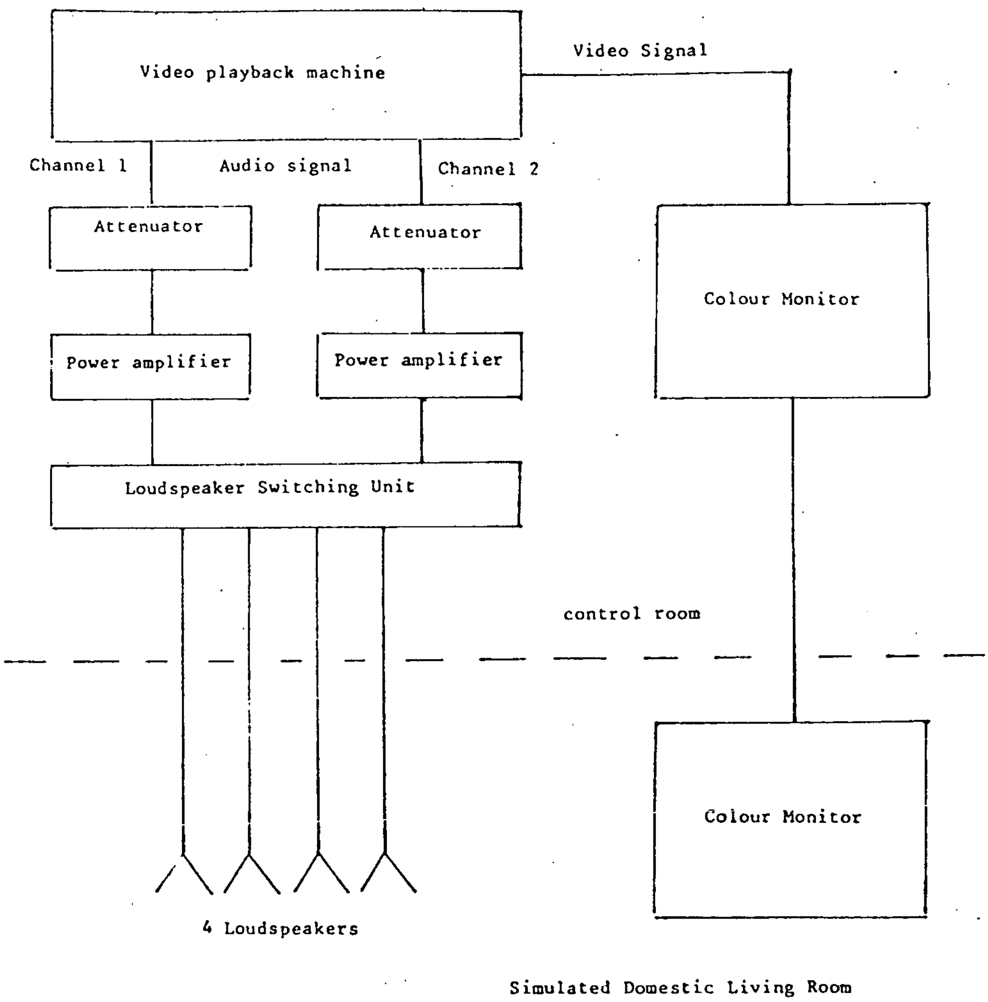
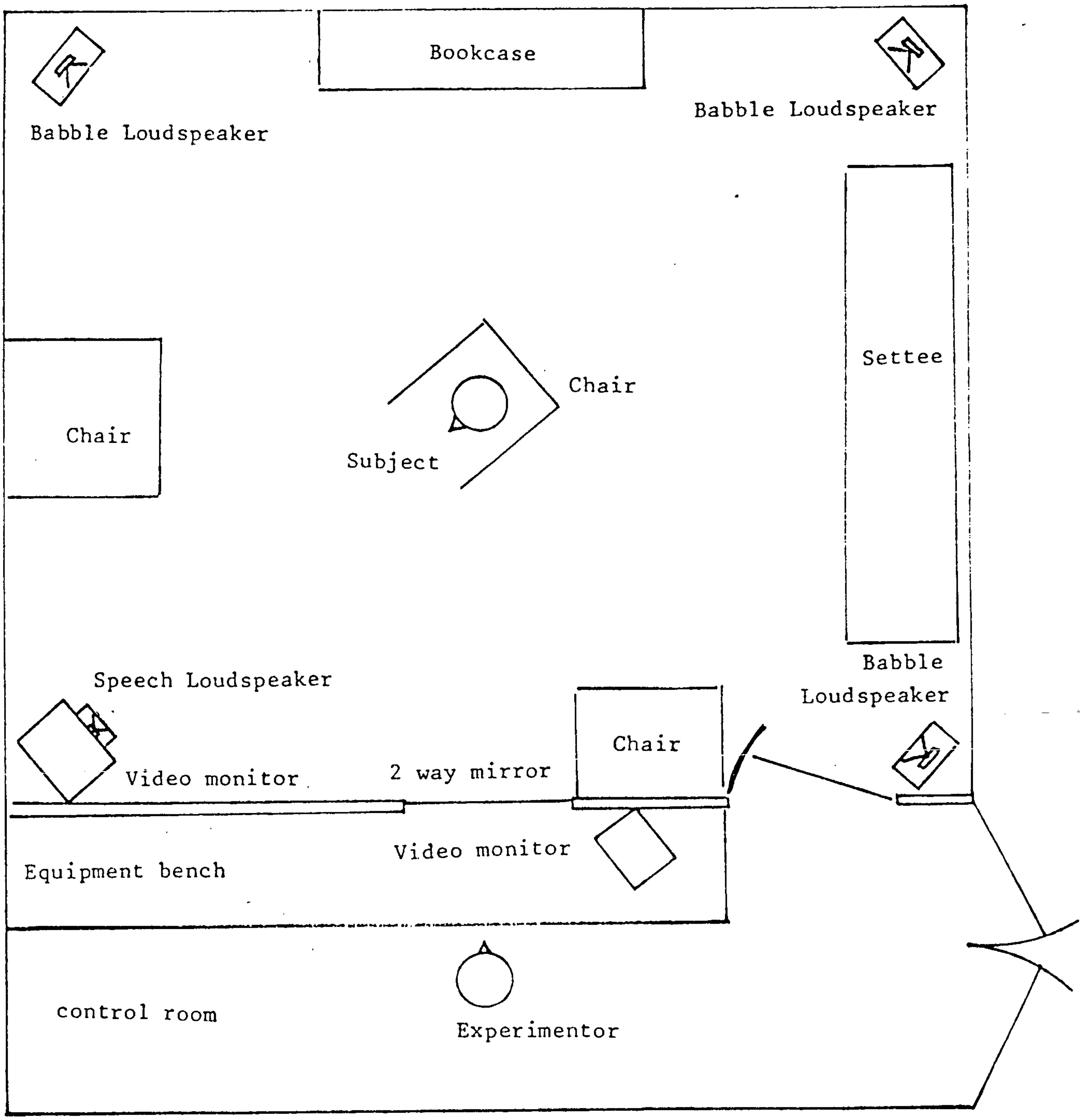


Figure 9.8

Simulated domestic living room presentation arrangement



CHAPTER 10

PAIR COMPARISON TEST

10.1 Design Philosophy

There is an almost infinite range of hearing aid electroacoustic parameters that it would be theoretically possible to produce. It would obviously not be feasible to try out all these 'aids' either in the clinic or at home, and so some pre-selection of suitable aids must take place whether an 'objective testing' or 'subjective choice' approach to hearing aid fitting is used. One way of preselecting aids would be to only try out aids with parameters that the user would perceive as different from each other.

If the 'just noticeable differences' (JND) for individuals could be established for specified electroacoustic parameters, then only aids with differences greater than those subjects' 'JND' need to be tested or compared. For example, if no difference could be perceived between two aids differing only in high frequency cut-off, one extending to 2 kHz and the other to 6 kHz, then there would appear to be little point in comparing aids differing only in frequencies above 2 kHz.

By assessing in this way the potential of subjects to benefit from 'fine tuning' of hearing aid parameters, better use could be made of available clinic time and resources. Those patients shown to have little potential for benefit from more extensive hearing aid fitting procedures could have their needs better served by spending that time on other rehabilitative procedures.

10.2 Design Criteria

It is possible that differences not noticed in one situation may be noticed in another, and so it is important to use a test for noticeable differences that is very sensitive to small changes. A very sensitive subjective test of noticeable differences appeared to be a paired comparison method where the subject is at liberty to compare the two 'aids' by switching instantaneously between them using the Master Hearing Aid described in Chapter 7, adjusting their respective levels as required while listening to the appropriate stimulus materials. The subject is allowed to take as long as required to make his decision. Bearing in mind the 'real-life' applicability aims of the study, it was decided that continuous speech would be used as the stimulus material as it is the most significant stimuli that the majority of people in real life are required to understand or listen to. Informal listening tests and previous research findings suggested that for the restricted bandwidth to be considered the use of other source material such as music gave little if any benefit in sensitivity over speech as a stimuli and gave JND's that were very dependent upon the particular music used.

In addition to finding JND's, pair comparisons can be used to make subjective preference judgements between hearing aids. If these judgements are related to real-life judgements, then this pair comparison procedure would provide a very efficient tool for deciding which was the 'best aid' for a particular individual; the 'best aid' here meaning that which the individual would choose if he could try out all the aids in his real-life situation.

However, because this pair comparison approach would appear to provide a very sensitive test, it would not be unreasonable to suppose that differences and preferences found between aids may not necessarily be substantiated in real-life situations. In everyday life it is not normally possible to compare aids instantaneously, for time must be taken to physically swap the aids over by which time the 'source' stimuli may have altered and memory effects may have become involved.

It would of course be theoretically possible to overcome this problem by having wearable master aids that could have the required characteristics compared by a simple user accessible switch. One might question the need for such measures on the grounds that any differences between aids that could not be noticed with a short gap between comparisons may indeed by 'unimportant'. This would, however, be a value judgement unless substantiated by empirical or theoretical evidence.

Another plausible reason for expecting real-life comparison results to be less sensitive than laboratory tests in quiet is that the effect of unpredictable background noises and room environments may mask differences. Real-life preferences may also differ from laboratory pair comparison preferences due to the wider range of stimuli and environments involved.

10.3 Subjective Ratings

Subjects were therefore asked to rate their pair comparison subjective responses by stating how sure they were that differences really existed and how great their preference was. These confidence and preference ratings gave an indication of the importance of any differences or preferences and could conceivably be used as a means of 'calibrating' the laboratory tests in terms of real-life responses.

For additional information, the subjects were also asked to indicate the reasons for their preferences.

The confidence and preference ratings and the reasons for expressed preferences also have the potential of indicating to the clinician which parameters may be most fruitfully further evaluated in finding the optimum hearing aid for a particular individual. For example, if changes in the low frequency cut-off elicit very confident differences and the strongest effect on preferences, then this might appear to be the most important variable to examine further.

Based on previous research findings and on the use of confidence and preference ratings it was decided not to use repeated measures to examine or improve the statistical reliability of individuals' judgements. This allowed a wider range of aids to be compared within the available time.

However, to given an overall indication of whether the subject was reliable, understood the task in hand and was adjusting the relative levels correctly, identical frequency responses were also presented at random during the test.

10.4 Test Material

Continuous speech recordings were made of the author reading from popular paperback books on the subjects of soil, black holes (astronomy) and the seashore. These passages were chosen to be interesting while not being so captivating as to override the task in hand, which was to make subjective difference and preference judgements. It was necessary to use material that held the subject's attention, as in real life the important listening situations will generally be ones where the subject is motivated to listen. The language content, due to the popular nature of the books, was at a homogeneous and appropriate level of difficulty.

The comments from subjects involved in the study indicated that they enjoyed and were interested in listening to this material and suggest that the aims in the choice of material had been satisfactorily fulfilled.

10.5 Recording Equipment, Environment and Methods

10.5.1 Continuous discourse master tape

The continuous passages were recorded on a Nagra IVSJ using a 1" B & K freefield microphone in an anechoic chamber. The author read the passages aloud after acquainting himself with their content and practising until fluency was achieved. Any sections with errors that still occurred were repeated correctly and then edited out during preparation of the master tape.

Approximately ten minutes of continuous discourse were obtained in this way and copied three times on to the master tape to produce a thirty minute continuous discourse tape consisting of three ten-minute repeats of the passages. Measurements of the speech showed that the levels and spectra remained relatively constant over the whole of the tape (Figure 10.1 and Tables 10.1 and 10.2).

10.5.2 Hearing aid processed speech tape

In addition to using the continuous discourse tape as stimulus material for the pair comparison test, recordings of hearing aid processed speech were also prepared. One use of this was to compare speech processed by a

Table 10.1 Statistical analysis of typical 2-minute speech section.

		L_{in}	250	500	800	1 k	1.6 k	2 k	3.15 k	4 k	6.3 k	8 k
						1/3 octave centre frequency (Hz)						
L_1	(dB)	77	67	68	63	64	60	57	53	59	57	54
L_5	"	75	64	65	58	59	56	52	50	55	54	50
L_{10}	"	73	61	63	56	56	53	50	48	53	51	47
L_{50}	"	68	53	54	43	45	43	41	40	43	38	36
L_{90}	"	54	37	41	31	31	32	33	32	33	28	27
L_{eq}	"	69	57	59	52	52	51	47	44	49	46	43
dB(A)L_{eq}		65										

Table 10.2 Analysis of 5 × 2-minute continuous speech samples.

(dB)	L_{in}	250	500	800	1 k	1.6 k	2 k	3.15 k	4 k	6.3 k	8 k
					1/3 octave centre frequency (Hz)						
Mean	69	57	59	51	50	50	48	45	48	46	43
SD	0	1	0	1	2	1	1	2	1	1	1

Table 10.3 Hearing aid processed speech tape.

Pair	Green Channel			Red Channel		Speech dB
1	Kemar open ear (left)			Kemar open ear (right)		70
				Aid	Gain	
2	Kemar open ear (left)			BE51L	Max-30 dB	70
3	Kemar open ear (left)			BE51H	Max-30 dB	70
	Aid	Gain	Peak Clip	Aid	Gain	
4	BE51L	Max-30 dB	0	BE51L	Max-30 dB	70
5	"	Max-20 dB	0	"	"	70
6	"	Max-15 dB	0	"	"	70
7	"	Max-10 dB	0	"	"	70
8	"	Max-5 dB	0	"	"	70
9	"	Maximum	0	"	"	70
10	"	Maximum	5	"	"	70
11	"	Maximum	10	"	"	70
12	"	Maximum	10	"	Max-35 dB	75
13	"	Maximum	10	"	Max-40 dB	80

real wearable hearing aid operating within its linear range, with that processed by a simulation of this hearing aid using the digital filter, and Master Hearing Aid described in Chapter 7.

The results of this comparison would indicate whether the digital filter was accurate enough to simulate real hearing aid frequency responses. Another use for the hearing aid processed speech tapes would be to examine the detectability of distortion produced by hearing aids. It would have been possible to use the Master Hearing Aid to simulate this distortion but as the precise characteristics of this type of distortion are difficult to specify, it was decided to use a real hearing aid so that the results could be directly applicable to clinical&real-life situations.

It is perhaps pertinent to mention here some of the inherent limitations in the use of master hearing aids. Although a master hearing aid can attempt to simulate various parameters of real hearing aids, there is no way that it could hope to mirror them all. For example, each hearing aid has its own particular relationship between input and output level and distortion. One method that has been widely used for comparing hearing aids that appears to overcome some of these problems is to compare recordings of hearing aid processed material. However, this still has the drawback that generally the material is recorded at a certain level through the aid and may be replayed at another, thus excluding realistic level-dependent effects.

For this study the hearing aid processed speech was recorded through hearing aids and ear simulators mounted in their normal positions on KEMAR in an anechoic room. The recordings were made on a Nagra IVSJ using half-inch B & K pressure microphones. The source material used was the continuous discourse tape described in Section 10.4.

To make the recordings for the comparison of the Master Hearing Aid with the real hearing aid, one channel was recorded through Kemar's open left ear while the other was recorded through the hearing aid operating linearly in Kemar's right ear. To check that any differences that might be noticed were not due to differences between the two recording channels, recordings were also made through both of Kemar's open ears for comparison.

For the recordings of the distorted processed speech, two hearing aids were used. One was operating well within its linear range, the other had increasing amounts of distortion applied. This was accomplished firstly by increasing the aid gain to maximum in 5 dB steps, then by introducing peak clipping via the aid's present control in 5 dB steps and finally by increasing the input signal in 5 dB steps.

The aid gains and output levels were monitored on an oscilloscope and on the built-in sound level meter of the Nagra IVSJ. A 1 kHz calibration tone was used to check the gain, output and clipping levels.

The system used is shown in Figure 10.2 and the various comparisons recorded are detailed in Table 10.3. The statistical level distributions and distortion measures of the undistorted and distorted speech are given in Appendix J..

Copies of the master tape were made to give one minute sections of the above comparisons separated by coloured leader tape for ease of identification and location.

10.6 Presentation Equipment, Environment and Method

The continuous passages were always replayed through the same Nagra IV SJ on which they were recorded. The Master Hearing Aid equipment used for the pair comparison test has already been described in Chapter 7. The room dimensions, loudspeaker and subject position were described in Chapter 9 and shown in Figure 9.7, the sound reproduction system used being that used for the replay of the speech channel of the AVELS test.

For the pair comparisons using hearing aid processed speech, computer operated relays were used to direct the appropriate tape channel through the system. These relays were disabled for pair comparisons using the unprocessed speech, as shown in Figure 10.3.

For all comparisons involving hearing aid processed speech equalization was carried out using the digital filter to ensure that the signal heard by the subject was identical to that he would have heard had he been wearing the actual hearing aid.

10.7 <u>Instructions to Subjects</u>

The instructions for the pair comparisons are shown in Appendix K. Before proceeding with the tests it was established that the subjects understood the instructions and were competent at following them out. The importance of adjusting the relative levels of each channel was stressed and the tests only commenced when the subjects had demonstrated an ability to do this. The subjects were encouraged to readjust the levels if required at any time during the comparison, the purpose of this being to ensure that a difference or preference was not expressed solely on the basis of one channel being 'louder' than another. All previo us studies of pair comparisons have failed to exclude this possibility.

Figure 10.1

Typical spectra of continuous discourse
(128 second Average)

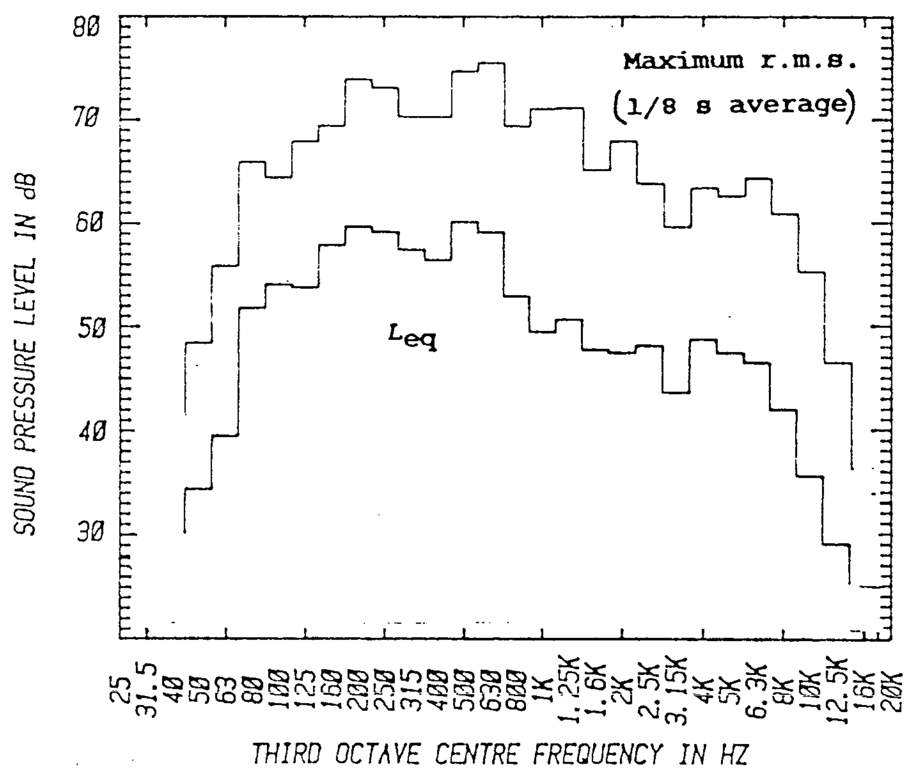


Figure 10.2

Hearing aid processed speech recording arrangement

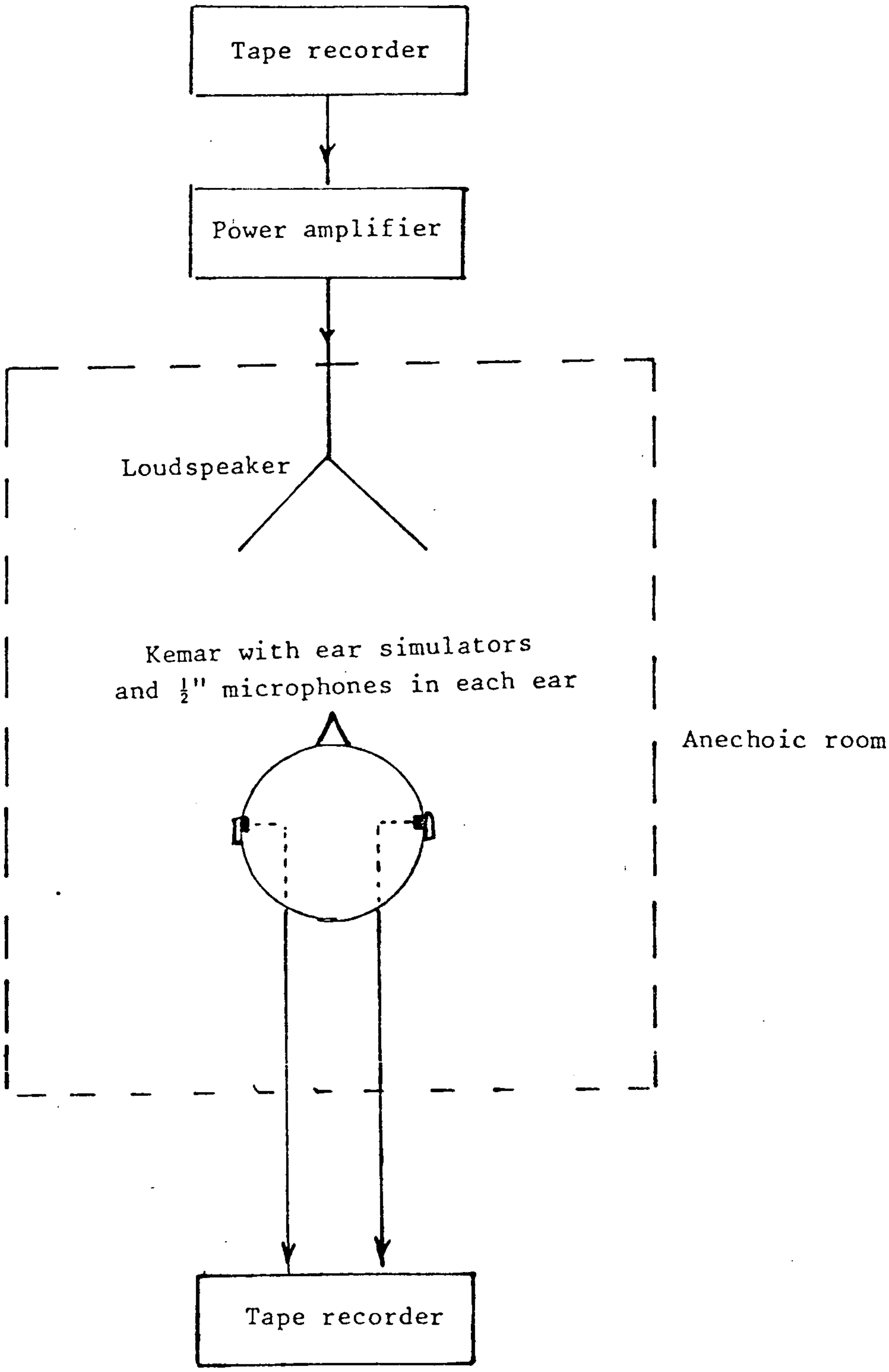
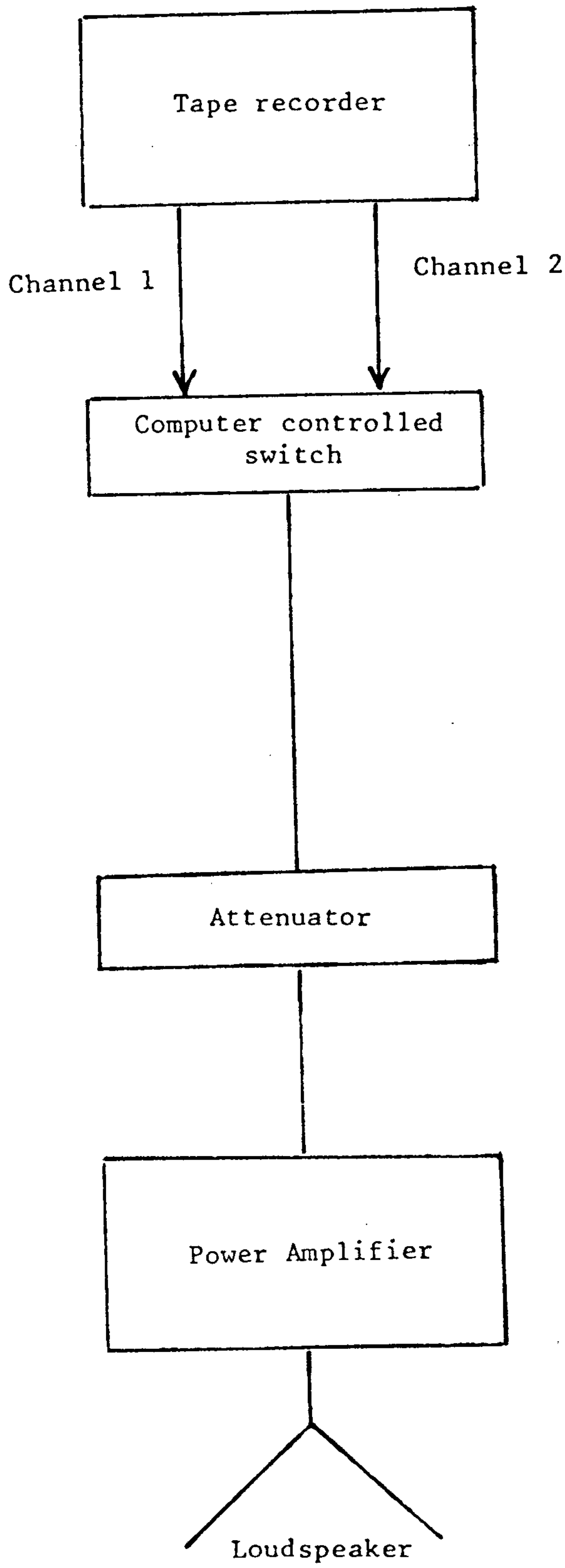


Figure 10.3

Hearing aid processed speech presentation system



CHAPTER 11

BASELINE STUDY: CALIBRATION OF TESTS ON NORMALLY HEARING SUBJECTS

11.1 Introduction

The experimental protocols and audio visual (AVELS) and pair comparison tests developed in this thesis required validation and calibration on normally hearing subjects in order to provide baseline measures for the main study on hearing-impaired subjects. The four main aims of the study were to:

(i) determine the learning effects of the Audio-Visual Everyday Life Simulation (AVELS) test

(ii) determine the sensitivity of the pair comparison procedure

(iii) determine the optimum stimuli to use for comfort level measures

(iv) validate the feasibility of the experimental protocol.

11.2 AVELS Test: Calibration Study

11.2.1 Introduction

The principal purpose of this experiment was to examine the subject learning and test-retest reliability effects for repeated applications of the AVELS test. Any learning effects or test-retest variability should ideally be as low as possible so that real differences in scores (or subjective judgements) for different hearing aid conditions would not be confounded. If an estimate of the learning effects can be determined then allowance can be made for them in the test scores.

Additional objectives were to:

(i) obtain baseline scores for the AVELS test on normally hearing subjects

(ii) compare whole word and phoneme scoring of the AVELS test

(iii) compare results on the AVELS test with results on the FADAST test (Appendix H) in order to see if different abilities are being tested

nefield & Foster 1983

(iv) examine the equivalence of subsets of the AVELS test.

11.2.2 Subjects

Eighteen normally hearing subjects with hearing thresholds better than 20 dB HL (BS 2497) at audiometric frequencies between 250 Hz and 8 kHz and who had no previous history of ear lesions or tinnitus and were otologically normal took part.

Estimates of subjects' critical ratios (which indicate the bandwidth of the subjects' auditory filter) were made at 2 kHz from measurements of masked and unmasked thresholds. The importance of critical ratio measures and the procedure used are described in detail in Appendix L. Subjects' ages, mean audiometric thresholds (500 Hz, 1 kHz, 2 kHz, 4 kHz) and critical ratio estimates are shown in Table 11.1. The majority of subjects had some previous experience of audiological tests although none had ever participated in audiovisual speech tests.

Table 11.1 Subject data

Subject No.	Age	BEST EAR Critical ratio, dB	Average threshold, dB HTL
1	23	20.0	1.25
2	24	22.5	1.25
3	35	20.0	2.50
4	31	22.5	3.75
5	49	22.5	3.75
6	23	22.5	5.00
7	31	20.0	3.75
8	24	20.0	6.25
9	24	17.5	5.00
10	35	20.0	13.75
11	36	27.5	6.25
12	19	22.5	8.75
13	30	22.5	0.00
14	23	22.5	1.25
15	30	20.0	13.75
16	29	22.5	8.75
17	39	22.5	11.25
18	20	22.5	-2.50

11.2.3 Experimental design and methodology

Two separate learning effects could be envisaged for repeated measures of the AVELS test. The first is a 'practice effect' due to familiarity with the test procedure and the test items. This would tend to increase scores obtained using repeated measures of the test under identical conditions. The second type of 'learning effect' is an "order dependent memory effect" which might be expected to occur when the test is repeated under more difficult conditions; for example, a 'poorer' hearing aid. In this case, test items correctly identified under the first 'easier' test condition might be remembered and therefore correctly identified under the second more difficult test condition. This would not have been the case had the more difficult condition been tested first.

In order to estimate the importance of these practice and order effects an experimental design was adopted that divided the subjects into three groups and tested repeated measures of the AVELS test under identical consecutive 'difficult' conditions and under both 'easy' followed by 'difficult' conditions and 'difficult' followed by 'easy' conditions. The experimental design is shown in Table 11.2. It can be seen that each group of six subjects received two identical difficult conditions and one easy condition.

Table 11.2 Experimental design

	AVELS TEST CONDITIONS		
Group	1st presentation	2nd presentation	3rd presentation
1	Difficult	Easy	Difficult
2	Easy	Difficult	Difficult
3	Difficult	Difficult	Easy

The 'easy' condition presented the test at a 'speech to babble' ratio of 0 dB in the worst case, whereas the 'difficult' condition presented the test at a 'speech to babble' ratio of −10 dB in the worst case. The actual levels used are shown in Table 11.3. The 'easy' condition used levels and signal-to-noise ratios that were representative of real life as measured by Pearsons *et al* (1977). The 'difficult' condition was used to depress the scores to those that would be more representative of hearing-impaired subjects.

Table 11.3 AVELS test presentation levels

	Loud speech	Loud babble
Easy condition	65 dB(A)	65 dB(A)
Difficult condition	55 dB(A)	65 dB(A)

The AVELS test was presented and scored as described in Chapter 9. All subjects were also tested on the FADAST test immediately before the AVELS test began. The FADAST test which is described more fully in Appendix H presents words audiovisually in a background of noise. There are four possible choices for each word spoken and these are displayed on the television monitor under the speaker's face during the test. A comparison of the results of the AVELS and FADAST tests would indicate if the tests were measuring different skills and therefore had different uses.

Each AVELS test presentation lasted fifteen minutes; the FADAST test also took fifteen minutes. The experimental session consiting of audiometric testing, a single presentation of the FADAST test and three presentations of the AVELS test, took a total time of approximately 1 hour 30 minutes for each subject.

11.2.4 Results

11.2.4.1 Subject group similarity

The results of the AVELS and FADAST tests are shown in Table 11.4. The AVELS scores are for the percentage of words correctly recognized (whole word scoring). No significant differences were found between the three groups for their scores on the FADAST test, AVELS easy condition or AVELS difficult condition for the first or third presentations (Table 11.5). This suggests that the groups were similar in their audio and audiovisual speech reception abilities.

A difference was found between the scores of group 2, 2nd presentation and group 3, 2nd presentation. However, for group 2 this was the initial presentation of the difficult condition, and followed the easy condition, while for group 3 it was the repeat presentation of the difficult condition and came before the easy condition. The difference in scores was therefore due to learning effects.

Table 11.4 Test scores, %

Subject No.	Group	FADAST test	AVELS test presentation 1st	2nd	3rd
1	1	93	67	90	75
2	1	94	76	92	77
3	1	91	70	90	76
4	1	96	69	90	74
5	1	88	70	88	75
6	1	87	64	91	74
7	2	92	91	77	75
8	2	91	93	80	83
9	2	89	89	78	84
10	2	89	84	75	74
11	2	82	82	68	67
12	2	93	92	74	81
13	3	90	64	66	86
14	3	87	67	68	86
15	3	95	69	69	89
16	3	91	64	67	89
17	3	96	69	69	91
18	3	94	65	67	87
Mean	1	91.50	69.33	90.17	75.00
SD	1	3.51	3.98	1.33	1.26
Mean	2	89.30	88.50	75.33	77.33
SD	2	3.90	4.50	4.18	6.53
Mean	3	92.17	66.33	67.66	88.00
SD	3	3.43	2.34	1.21	2.00

Table 11.5 Differences between subject groups' mean scores

Test	t_{10} values Groups 1 & 2	Groups 1 & 3	Groups 2 & 3
FADAST	0.94	0.31	1.24
AVELS easy	0.83	2.02	0.23
AVELS difficult - 3rd presentation	0.78	-	-
AVELS difficult - 1st presentation	-	1.45	-
AVELS difficult - 2nd presentation	-	-	3.94**

significance levels: $p < 0.05$* $p < 0.01$** $p < 0.001$***

Note: These significance levels will be used throughout the study unless otherwise indicated.

11.2.4.2 Learning effects

If the changes in the scores for each of the 12 subjects of groups 2 and 3 who repeated the difficult conditions consecutively are analysed (Table 11.6) a slight significant mean practice effect of 1.67% is found. If the practice effects for the subjects of group 2 and group 3 are analysed separately only that for the latter group is found significant with a mean value of 1.33% although a non-significant mean practice effect of 2.0% occurred for group 2.

For the six subjects in group 1 who repeated the difficult condition following the presentation of the easy condition, a mean improvement in scores of 5.67% over that obtained upon initial presentation of the difficult condition was measured. This improvement can be taken to be due to the combination of any practice effects and any 'memory' or 'order' effect.

Another possible approach to estimating the memory/order learning effects is to find the difference between the mean scores of all the difficult conditions presented before and after the easy condition. The justification for this approach is the similarity in scores between subject groups and the small practice effects found. As shown in

Table 11.6, a slightly greater mean learning effect of 8.11% is found using this analysis.

As no significant differences were found between the scores of the easy test condition for the three groups even though they were presented with this condition in a different order (Table 11.5), it would appear that no significant learning effect occurs when an easy condition follows a difficult condition.

Table 11.6 Mean % learning effects for complete AVELS test (whole word scoring)

Change in scores for repeat presentation of difficult condition

Group	No. of subjects	Mean	SD	t	95% confidence limits
2 & 3	12	1.67	2.77	2.08*	± 1.44
1	6	5.67	2.94	4.72**	± 2.42
2	6	2.00	3.90	1.26	± 2.22
3	6	1.33	1.21	2.69*	± 1.00

Group	Degrees of freedom	Mean	t	95% confidence limits
1,2,3	16	8.11	6.37***	± 2.22

11.2.4.3 Equivalent test forms

An analysis of the differences between the scores (whole word scoring) for the test forms consisting of the first 36 items and the second 36 items of the AVELS test is shown in Table 11.7 with the raw data given in Table 11.8. There is no significant difference between scores on the test forms for the easy condition or for the difficult condition presented first. There is, however, a significant difference for the difficult condition presented in second or third position and this difference also results in a difference between test forms when analysed across all test conditions. The reason for this difference may be understood by examining the learning effect for the two test forms (Table 11.9). Similar

Table 11.7 Difference between 1st and 2nd 36 item % scores (whole word scoring)

Condition	No. of tests	Mean	SD	t	95% confidence limits
All	54	2.36	5.86	4.39*	± 1.60
Easy	18	2.00	5.44	1.55	± 2.57
Difficult 1st presentation	12	0.47	6.25	0.26	± 3.62
Difficult 2nd & 3rd presentation	24	3.58	5.94	2.96**	± 2.43

learning effects to those for the complete test (Table 11.6) are found. However, it can be seen that there is a significantly greater effect for the first test form than the second test form.

One might hypothesize that this is due to the fact that the test forms were always presented in the same order and the 1st test form will be more easily remembered. However, the possibility that there is a real difference in the learning effect for the two forms cannot be ruled out.

In any case, if the two test forms were used to compare two hearing aids, they would be only presented once and, therefore, they would be equivalent as shown by the analysis in Table 11.7 for the initial presentation. The scores on the first test form correlate highly ($r = 0.933$) with those for the complete test (Table 11.10), thus showing that the scores on the shortened test form are representative of those on the whole test.

11.2.4.4 Phoneme scoring

Analysis of the difference between the word scores and phoneme scores for the AVELS test shows that phoneme scores were nearly 9% higher for the difficult conditions and 6% higher for the easy condition (Table 11.11). The raw phoneme scores are given in Table 11.12. The correlation coefficient for the whole word and phoneme scores is 0.967 indicating a close correspondence.

Table 11.8 Scores on test forms (no. of words correct)

Subject No.	1st presentation 1st form	1st presentation 2nd form	2nd presentation 1st form	2nd presentation 2nd form	3rd presentation 1st form	3rd presentation 2nd form
1	24	27	30	34	25	29
2	29	27	35	34	28	28
3	27	25	34	33	29	28
4	28	25	34	32	31	26
5	27	27	32	32	29	28
6	24	23	32	33	28	26
Mean	26.5	22.3	27.8	33.0	28.3	27.5
7	33	32	29	29	30	26
8	34	33	31	27	33	29
9	32	28	29	28	32	32
10	31	29	28	26	29	26
11	29	30	25	25	25	25
12	32	34	29	25	29	30
Mean	26.8	31.0	28.5	26.7	29.7	28.0
13	25	23	26	24	32	30
14	25	26	26	26	33	29
15	25	27	26	27	33	31
16	25	23	26	23	33	33
17	26	26	27	25	33	33
18	23	27	25	26	32	31
Mean	21.5	25.3	26.0	21.8	32.7	31.2

Total number of words on each form = 36

Table 11.9 Mean learning effects (%) for 1st and 2nd 36 items (whole word scoring)

1st 36 items

Group	No. of subjects	Mean	SD	t
2 & 3	12	3.25	2.31	4.84**
1	6	5.08	4.78	2.61

2nd 36 items

Group	No. of subjects	Mean	SD	t
2 & 3	12	1.61	6.08	0.92
1	6	5.08	2.72	4.58**

Table 11.10 Relationship between scores on 1st 36 items and whole test

Conditions	No. of tests	Slope	Intercept	Correlation Coefficient
All	54	0.98	11.28	0.933***

Table 11.11 Relationship between word and phoneme scores (%)

(i) Difference between phoneme and word scores

Condition	No. of tests	Mean	SD	t
Difficult	36	8.59	2.51	20.55***
Easy	18	6.13	2.03	4.24***

(ii) Correlation between phoneme and word scores

Condition	No. of tests	Mean	SD	t
All	54	0.83	20.92	0.967***

Table 11.12 Phoneme scoring (%)

Subject no.	1st presentation 1st form	2nd form	Total	2nd presentation 1st form	2nd form	Total	3rd presentation 1st form	2nd form	Total
1	74.2	78.5	71.5	92.7	97.5	9.61	79.0	86.8	80.8
2	87.9	81.0	81.7	99.2	95.9	97.2	85.5	86.8	84.8
3	81.5	84.3	82.8	97.6	96.7	95.5	87.1	90.9	85.9
4	86.3	81.0	79.7	97.6	91.7	94.9	91.9	84.3	84.8
5	86.3	80.2	79.7	95.2	95.0	94.9	89.5	84.3	84.8
6	75.8	71.1	71.8	95.2	96.7	95.8	87.1	82.6	83.7
Mean	82.00	79.35	77.87	96.25	95.58	95.73	86.68	85.96	84.13
SD	5.85	4.56	4.96	2.33	2.08	0.86	4.38	2.92	1.78
7	96.8	95.9	96.6	83.9	82.6	80.3	88.7	81.0	82.3
8	96.8	96.7	96.9	91.9	86.0	88.2	96.0	88.4	89.9
9	95.2	95.9	96.3	87.9	83.5	84.2	95.2	86.0	92.4
10	94.4	93.4	94.1	82.3	81.8	81.7	87.1	81.0	82.0
11	91.1	92.6	91.8	75.0	74.4	74.1	75.8	76.9	73.8
12	93.5	98.3	93.5	87.1	81.8	82.5	87.9	86.8	85.9
Mean	94.63	95.47	94.87	84.68	81.68	81.83	88.45	83.35	84.38
SD	2.17	2.12	2.05	5.80	3.89	4.66	7.28	4.41	6.62
13	83.1	72.7	74.9	83.1	77.7	76.9	92.7	92.6	93.2
14	79.0	82.6	78.9	84.7	81.0	81.7	96.8	90.9	93.5
15	76.6	84.3	78.3	79.0	79.3	75.8	96.8	91.7	94.6
16	79.8	78.5	75.8	79.8	74.4	74.9	95.2	95.9	94.1
17	76.6	77.7	74.6	83.9	78.5	78.6	96.8	95.9	96.6
18	79.0	81.8	77.2	85.5	81.0	78.3	95.2	95.0	94.6
Mean	79.02	79.60	76.62	82.67	78.65	77.70	95.58	93.67	94.43
SD	2.41	4.20	1.79	2.67	2.47	2.42	9.78	2.21	1.20

Learning effects calculated from phoneme scores for the complete AVELS test (Table 11.13) and for the 1st and 2nd 36 items (Table 11.14) are similar to those measured using whole word scoring (Tables 11.6 and 11.9 respectively). Differences between the 1st and 2nd 36 items scored phonemically (Table 11.15) are also similar to those measured by whole word scoring (Table 11.7).

It would therefore appear that the results using phoneme scoring follow closely those using word scoring, although the absolute scores are raised slightly.

11.2.4.5 AVELS test sections

The test items which make up the 'easy' AVELS test can be divided into four types:

(i)	AVQ:	Audio visual presentation in 'quiet'	(> 0 dB S/N ratio)
(ii)	AVN:	Audio visual presentation in 'noise'	(0 dB S/N ratio)
(iii)	AQ:	Audio only presentation in quiet	(> 0 dB S/N ratio)
(iv)	AN:	Audio only presentation in noise	(0 dB S/N ratio)

The means, standard deviation and range of score for each of these 'sections' are given in Table 11.16, which shows how the addition of noise reduces the score for both the audio visual and the audio only tests. The fact that subjects scored higher on the audio only in noise (AN) section of the AVELS test than they did on the audio visual in noise (AVN) section, would seem to indicate a difference in difficulty of the test items involved. The absolute scores on the different sections of the AVELS test cannot therefore be directly compared to indicate the relative difficulty a subject has under these conditions. The scores on normally hearing subjects can, however, be used as a baseline with which to compare those of hearing-impaired subjects and thus indicate their possible relative handicap under different conditions.

Table 11.13: Learning effects (%) for complete AVELS test (phoneme scoring)

Group	No. of subjects	Mean (%)	SD	t	95% confidence limits
2 & 3	12	1.81	2.72	2.30*	± 1.41
1	6	6.25	3.55	4.31**	± 3.73

Table 11.14: Learning effects (%) for 1st and 2nd 36 items (phoneme scoring)

<u>1st 36 items</u>

Group	No. of subjects	Mean	SD	t
2 & 3	12	3.70	2.78	4.60**
1	6	4.70	4.43	2.60*
<u>2nd 36 items</u>				
2 & 3	12	0.27	3.29	0.28
1	6	6.61	3.00	5.39**

Table 11.15: Difference between 1st and 2nd 36 items, % scores (phoneme scoring)

Condition	No. of tests	Mean (1st-2nd)	SD	t	95% confidence limits
All	54	1.72	4.25	2.97**	± 1.16
Easy	18	0.58	3.02	0.82	± 1.50
Difficult (1st presentation)	12	1.03	5.56	0.44	± 3.53
Difficult (2nd and 3rd presentations)	24	2.92	4.41	3.24**	± 1.86

Table 11.16: Comparison of % scores on AVELS test sections for 'easy' condition

Test section	AVELS			
	AVQ	AVN	AQ	AN
Mean %	98.25	69.44	97.92	78.47
Standard deviation %	2.44	8.81	3.71	14.09
Maximum score, %	100.00	87.50	100.00	100.00
Minimum score, %	93.75	56.25	87.5	50.0

11.2.4.6 Comparison of AVELS and FADAST test scores

The correlations between scores on the AVELS easy condition and the FADAST test are given in Table 11.17. It can be seen that there is a small but significant correlation between the total scores on the AVELS test and on the FADAST test. The FADAST test presents all its test words audiovisually in a background of noise. It might be expected therefore that there would be a closer correlation with the audiovisual in noise (AVN) section of the AVELS test. This was not the case as is shown in Table 11.15. A significant correlation however was obtained with the audio only in noise (AN) section.

The AVELS test therefore appears to be testing somewehat different skills than is tested by the FADAST tests. The abilities to extract information from the linguistic and prosodic features of speech may be two such skills as the AVELS test consists of sentences with syntactic, semantic and contextual clues, whereas the FADAST test involves only isolated words.

Table 11.17 Comparison of scores on AVELS and FADAST tests

Range of scores (%)

Test	AVELS 'easy'	FADAST
Mean	89.29	91.00
SD	2.44	3.63
Maximum score	93	96
Minimum score	82	82

Correlation between AVELS test and FADAST

AVELS test	Slope	Intercept	Correlation
Whole test	0.50	43.80	0.614**
AVN section	0.86	-8.56	0.350
AN section	2.38	-139.89	0.618**

Table 11.18: Relationship between audiometric measurement and test scores

Audiometric measurement	Test	Correlation coefficient
Average threshold, Hz) (500, 1k, 2k, 4k)	FADAST	0.071
" " " "	AVELS ' easy'	-0.009
Critical ratio	FADAST	0.366
" "	AVELS 'easy'	0.422

11.2.4.7 Relationship between audiometric measures and speech test scores

From the results of the analysis shown in Table 11.18, there appears to be little correlation between audiometric threshold measurements and results on the FADAST or AVELS test.

Measures of critical ratios show a larger although still non-significant correlation with speech scores. The possible importance of critical ratio measures as a predictor of speech scores is indicated by subject 11 who, although having normal pure tone thresholds, has a much larger critical ratio than the other 17 subjects (Table 11.1) and also obtained test scores on the FADAST and AVELS test well below all other subjects (Table 11.4). Subject 11 stated that he found conversation in noisy situations such as parties particularly difficult (more so than his wife) and so often avoided these situations. This indicated that both the AVELS and FADAST test might successfully give some prediction of real life performance.

11.2.5 Conclusions

The following conclusions can be drawn for the subjects and conditions used in this experiment.

(i) Learning effects and test-retest reliability.

When repeated measures of the complete AVELS test are presented under identical conditions only small mean practice effects occur of approximately 1.5%. The 95% confidence limits cover differences between 0% and 3%. When the AVELS test is presented with a 'difficult' condition following an 'easy' condition, a total mean learning effect of between 5.5% and 8% may be expected. The 95% confidence limits covered values between 3% and 10%. These values may overestimate the learning effects for only two presentations of the AVELS test, as they were calculated from the results of an experiment that presented the AVELS test three times.

When the AVELS test is presented with an easy condition following a difficult condition there would appear to be no significant learning effects.

(ii) Equivalent test forms

The test forms consisting of the 1st and 2nd 36 items were equivalent for initial presentations, and scores on these forms correlated highly with those of the complete AVELS tests. Scores on the 1st 36 items were higher following repeated presentations of the AVELS test, suggesting greater learning effects. This could possibly be due to the presentation order of the test forms which remained constant throughout the test.

It would therefore appear feasible to use the two test forms rather than repeated measures of the complete test when comparing two hearing aids, as each test form need only be used once.

(iii) Phoneme scoring

Although slightly higher absolute scores for phoneme scoring than word scoring were obtained there was a high correlation between the two methods of scoring. In addition, similar learning effects and standard deviations occurred. Apart from providing some information on the type of phonemic error, there would seem to therefore be little advantage in the use of phonemic scoring to offset its greater complexity.

(iv) AVELS test sections

The absolute scores on the separate section of the AVELS test cannot be directly compared to indicate how subjects cope under these different conditions, due to the demonstrated lack of equivalence between these sections of the test. This outcome was not unexpected, due to the difficulties involved in attempting to balance the test for all conditions (described in Chapters 2 and 9) and does not detract from the usefulness of the test sections. The scores for normally hearing subjects may be used as baseline scores with which to compare those of hearing-impaired subjects and thus indicate their relative performance under the various conditions.

(v) Comparison of AVELS and FADAST test scores

Although some relationship between AVELS and FADAST scores was found, the two tests appear to be testing somewhat different abilities. This indicates that the inclusion of linguistic and prosodic features in an audiovisual test does provide extra information regarding subjects' speech reception abilities. If there was a very high correlation between the two tests this conclusion could not have been drawn, even though the AVELS test has the face validity of employing more realistic test conditions.

(vi) Relationship between audiometric measures and speech test scores

No significant correlation was found between AVELS or FADAST test scores and audiometric measures of threshold or critical ratios. This may have been due in part to the small range of test scores and audiometric measures exhibited by the subjects. However, critical ratio measures hold some promise as a possible predictor of low scores, as the subject who had the largest critical ratio (but normal thresholds) scored the lowest on both the AVELS and FADAST tests.

11.2.6 Test retest reliability

The test-retest reliability for the AVELS test for the subjects and conditions used can be estimated from the standard deviations obtained for the repeat measures (Table 11.6). These will obviously only be rough approximations due to the relatively small number of tests and subjects involved.

For the consecutive repeats of the complete AVELS test difficult conditions, 95% of the subjects had differences of between -3.8 and 7.1% (± 1.96 standard deviations from the mean). For the difficult/ easy/ difficult conditions, 95% of the differences were between -0.1 and 11.4. Similar estimations can be made for the differences between the 1st and 2nd test forms (Table 11.7). For the initial difficult presentations, 95% of the differences were between -11.8 and +12.7. These critical differences can be expected to vary with absolute scores.

The values of these critical differences have implications for the significance of differences in scores obtained with the AVELS test for hearing-impaired subjects using two different hearing aids. If the complete AVELS test is repeated using two different hearing aids, then scores with the second aid will have to be more than 7.1% higher or 3.8% lower to indicate a 95% significance difference between the aids. If the second aid scores significantly lower, then the actual differences between the aids can be underestimated by up to 11.4% due to memory/learning effects.

If the two test forms are used to compare the aids, then differences of 12% to 13% are required for 95% significance. Critical differences are therefore lower for the complete test, due mainly to the greater number of independent items involved.

11.2.7 Implications for clinical and laboratory use of the AVELS test

The AVELS test was designed to be used as a subjective test for comparing hearing aids. The small learning effects found in this study suggest that they should not seriously confound any subjective preferences, as subjects will not find the test much easier for the repeat presentation. The insignificant differences found between test form scores indicate that the separate test forms can also be presented to obtain subjective preferences.

The scoring of the AVELS test allows for an indication of subjects' communication difficulties, comparisons of group data for evaluation of hearing aid fitting protocols and comparisons of individual hearing aid fittings. The learning effects found in this study can be used to correct for the results of group data.

The critical differences found give indications as to the differences in scores between hearing aids that is required to obtain significance. As was made clear in Chapters 2, 3, 9 and 10, speech tests are not efficient at significantly discriminating small differences between hearing aids due to the statistics involved, and there is no reason why the AVELS test should be any more efficient. Repeated measures of the complete AVELS test will be more sensitive to scored differences between hearing aids than the use of separate test forms due to the greater number of items involved. However, this greater sensitivity need not also apply to subjective judgements.

11.3 Pair Comparison Test Subjective Calibration

11.3.1 Aims

The aims of this study were to:

(i) Determine the sensitivity of the pair comparison procedures for normally hearing subjects;

(ii) determine the 'optimum' frequency response of the master hearing aid system for normally hearing subjects as a baseline measure with which to compare hearing-impaired subjects' preferences;

(iii) verify that the Master Hearing Aid could simulate real hearing aids satisfactorily;

(iv) examine the effects of frequency response changes on comfortable listening levels.

11.3.2 Subjects

Ten normally hearing subjects with hearing thresholds better than 20 dB HL (BS 2497) at audiometric test frequencies between 125 Hz and 8 kHz and who had no history of ear lesions or tinnitus and were otologically normal took part in this study.

Estimates of subject critical ratios were made at 2 kHz from measurements of masked and unmasked thresholds. Subjects' ages, mean audiometric thresholds (500, 1 kHz, 2 kHz, 4 kHz) and critical ratio estimates are shown in Table 11.19. The majority of the subjects had some previous experience of audiological tests although none had ever previously participated in subjective pair comparison tests.

Table 11.19: Subject data

Subject No.	Age	BEST EAR Critical ratio dB	Average threshold dB HTL
1	32	25	0
2	26	22.5	-3.75
3	38	22.5	1.25
4	31	20	3.75
5	23	20	1.25
6	23	22.5	-5.00
7	24	20	6.25
8	24	17.5	5.00
9	24	22.5	-2.50
10	30	22.5	0

11.3.3 Experimental design and methodology

The pair comparison test material and test protocol were as described in Chapter 10. The actual pairs of master hearing aid frequency responses used are listed in Table 11.20 and illustrated in Figure 11.1. The frequency responses and their 'names' are generally self-explanatory but will be described in more detail here along with the reasons for choosing them. There is an almost infinite range of responses that could have been tried and so some method of selecting those to be used had to be adopted. Pairs 1 to 24 were selected on the basis of informal listening trials by the author and colleagues as covering a wide range of slight deviations from the flat low pass 8 kHz condition that included variations in low and high cut-off frequency (pairs 1 to 4), low and high frequency slopes (pairs 5 to 20) and mid-frequency cuts and boosts (pairs 21 to 24). These were felt to provide an adequate range to determine the optimum response for the normally hearing subjects and provide adequately small changes to examine the sensitivity of the procedure. In addition to these responses it was possible to load the digital filter/master hearing aid with any of many hundreds of other frequency responses if required, although, as anticipated, this did not prove necessary. The creation and checking of these large numbers of filter files took many hundreds of hours.

Table 11.20 Pair comparison digital filter responses

Pair	Green	Red
1	HP 150 Hz	LP 8 kHz
2	HP 200 Hz	LP 8 kHz
3	LP 8 kHz	LP 7 kHz
4	LP 6 kHz	LP 8 kHz
5	LP 8 kHz	+ 1.5 dB/oct
6	+ 3 dB/oct	LP 8 kHz
7	-1.5 dB/oct	LP 8 kHz
8	- 3 dB/oct	LP 8 kHz
9	LP 8 kHz	-3 dB/oct below 500 Hz
10	LP 8 kHz	-6 dB/oct below 500 Hz
11	+ 3 dB/oct below 500 Hz	LP 8 kHz
12	+ 6 dB/oct below 500 Hz	LP 8 kHz
13	- 3 dB/oct above 4 kHz	LP 8 kHz
14	- 6 dB/oct above 4 kHz	LP 8 kHz
15	- 3 dB/oct above 2 kHz	LP 8 kHz
16	LP 8 kHz	- 6 dB/oct above 2 kHz
17	LP 8 kHz	+ 3 dB/oct above 4 kHz
18	LP 8 kHz	+ 6 dB/oct above 4 kHz
19	+ 3 dB/oct above 2 kHz	LP 8 kHz
20	+ 6 dB/oct above 2 kHz	LP 8 kHz
21	5 dB mid cut	LP 8 kHz
22	LP 8 kHz	10 dB mid cut
23	5 dB mid boost	LP 8 kHz
24	LP 8 kHz	10 dB mid boost
25	5 dB peaks	LP 8 kHz
26	LP 8 kHz	10 dB peaks
27	15 dB peaks	LP 8 kHz
28	LP 8 kHz	Inverse of Kemar open ear response
29	LP 8 kHz	Zwislocki - 2 cc
30	Kemar - 2 cc	LP 8 kHz
31	Kemar - Zwislocki	LP 8 kHz
32	Master aid/Kemar	Master aid/Insertion gain
33	Master aid/Zwislocki	Master aid/2 cc
34	Master aid/2 cc	Master aid/Insertion gain
35	Master aid/Kemar	Maste aid/Zwislocki
36	LP 4 kHz	LP 6 kHz
37	LP 3 kHz	LP 2 kHz
38	LP 4 kHz	LP 3 kHz

Pairs 25, 26 and 27 were included to examine the effects of peaks in the frequency response. Pair 28 was presented to determine the effect of the 'loss' of the natural ear canal resonance that occurs when an earmould is fitted. Pairs 29 to 35 examine whether the differences between coupler, ear simulator, and manikin responses can be noticed; pairs 28 to 31 using a flat frequency response for comparison and pairs 32 to 35 using a 'typical' hearing aid response for comparison. Pairs 36, 37, 38 present a variety of high frequency cut-offs at 2, 3, 4 and 6 kHz for comparison with each other. The allocation of the frequency respones to the red or green channel was made at random.

Each pair was presented only once, the subjects taking as long as they required to make their decisions and choices. A 'dummy' condition was presented at random to test the reliability of the subject' choices. This consisted of comparing two identical low pass 8 kHz responses. Not all pairs were required to be used for all people, as when a difference was noticed for a pair comparison and a preference expressed for the LP 8 kHz response no further testing of that parameter was carried out. For example, if for pair 1 a preference was expressed for the LP 8 kHz response, then pair 2 was not presented and the subject proceeded to pair 3. If, however, no difference was found between the responses of pair 1, or no preference was expressed, or a preference was expressed for the HP 150 Hz condition, then the subject proceeded to pair 2. This protocol allowed the subject's noticeable differences and optimum responses to be determined with the minimum number of presentations. It required the not unreasonable assumption that if a subject noticed and disliked the change in a certain parameter (e.g., increase in high frequency cut-off) he would also notice and dislike an even greater change in that parameter.

To test whether the master hearing aid could simulate a real hearing aid satisfactorily and determine the noticeable difference for peak clipped speech, the hearing aid processed speech tape was used as described in Chapter 10. The hearing aid simulations were derived using the hardware and software described in Chapters 7 and 8 and their responses are shown in Figure 11.2. All presentations of hearing aid processed speech were equalised to allow the subject to listen as if he was actually wearing the hearing aid. This was accomplished by 'subtracting' Kemar's open ear free field to eardrum response as shown in Figure 11.3. The pairs compared are shown in Table 11.21.

Table 11.21 Pair comparison pairs: Hearing aid processed speech

Pair	Green	Red
1	Kemar open ear (left)	Kemar open ear (right)
2	BE51L simulation	BE51L
3	BE51H simulation	BE51H
4	BE51L	BE51L
5	BE51L slight distortion	BE51L
6	BE51L more distortion	BE51L
7	BE51L more distortion	BE51L
8	BE51L more distortion	BE51L
9	BE51L more distortion	BE51L
10	BE51L more distortion	BE51L

For this experiment using normally hearing subjects it was not feasible to use the microphone and insert earphone of the Master Hearing Aid. This was because the individual earmoulds could not provide sufficient attenuation to prevent all free field sounds from being heard directly, in addition to through the Master Hearing Aid. While this would be a realistic listening situation for hearing-impaired subjects, for the purposes of this study on normally hearing subjects it would introduce another variable which would confound the results. To overcome this problem the taped stimuli were fed electrically into the master hearing aid and presented through the loudspeaker in free field. This arrangement is shown in Figure 11.4. The experiment can therefore be considered as testing the pair comparison procedure under the condition of normally hearing subjects wearing ideal 'transparent' hearing aids with unity insertion gain. This would appear to be a useful baseline with which to compare the results of hearing-impaired subjects.

The pair comparison test took approximately 1 hour 30 minutes for each subject.

11.3.4 Results

11.3.4.1 Frequency response noticeable differences and preferences

Subjects were asked for confidence ratings for different judgements and also preference ratings as described in Chapter 10 and indicated in the instructions shown in Appendix K. The results are shown in Table 11.22 in the form of the number of subjects who gave each rating. The totals for the preferences for each response have also been included. The asterisks denote when the response was the flat low pass 8 kHz condition.

(i) Optimum response

The majority of subjects did not prefer any deviation from the 'flat' low pass 8 kHz condition. For most comparisons the majority of subjects preferrred the flat low pass 8 kHz condition, while in a few comparisons no differences were noticed and no preferences expressed. For the normally hearing subjects the optimum frequency response was therefore the flat low pass 8 kHz condition.

(ii) Sensitivity of pair comparison test

Most subjects noticed a difference in nearly all the comparisons. Only for four pairs (9, 11, 13 and 17) did one half or more of the subjects not notice any difference. These conditions were the 3 dB/octave cut or boost below 500 Hz or above 4 kHz. The only other conditions where more than one subject did not notice any difference from the LP 8 kHz responses were pairs 1, 3, 14, 21, 23, 25. These were the HP 150 Hz, LP 7 kHz, -6 dB/octave above 4 kHz, 5 dB midcut, 5 dB mid boost and 5 dB peaks, conditions.

Two subjects found no difference between the hearing aid responses on the Zwislocki ear simulator and the 2 cc coupler (pair 33) and three subjects found no difference between the hearing aid responses on Kemar and the Zwislocki ear simulator (pair 35). These coupler, ear simulator and Manikin differences were noticed by everybody for the LP 8 kHz condition (pairs 29 and 31).

For the majority of comparisons, when differences were noticed, subjects were moderately, very or completely sure of these differences. In only nine comparisons (2.8%) were subjects 'not at all sure' if there

Table 11.22 Pair comparison ratings

	Difference						Red preference				Green preference				No pref.
Pair	4	3	2	1	0	N	3	2	1	Total	3	2	2	Total	
1	2	1	1	3	0	3	0	0	3	3*	0	0	1	1	3
2	2	3	2	0	0	0	1	1	4	6*	0	1	0	1	0
3	4	2	1	1	1	1	0	0	0	0	0	2	5	7*	2
4	3	0	0	0	0	0	0	0	2	2*	0	0	1	1	0
5	6	1	1	2	0	0	0	0	2	2	0	1	7	8*	0
6	2	0	0	0	0	0	1	1	0	2*	0	0	0	0	0
7	7	0	2	1	0	0	2	1	3	6*	0	0	2	2	2
8	4	0	0	0	0	0	0	1	3	4*	0	0	0	0	0
9	3	1	0	1	0	5	0	0	3	3	0	1	1	2*	0
10	6	1	0	1	0	0	0	0	1	1	1	2	4	7*	0
11	1	2	1	0	0	6	0	1	3	4*	0	0	0	0	0
12	1	2	2	1	0	0	0	2	4	6*	0	0	0	0	0
13	1	0	0	1	0	8	0	0	2	2*	0	0	0	0	0
14	1	1	1	1	1	3	0	0	3	3*	0	0	1	1	1
15	3	0	4	1	1	1	0	1	3	4*	0	0	3	3	2
16	4	0	1	0	1	0	0	0	2	2	1	0	2	3*	1
17	2	0	0	1	0	7	0	0	0	0	0	0	2	2*	1
18	1	1	1	3	1	1	0	0	3	3	0	0	4	4*	0
19	4	1	2	1	1	1	0	1	2	3*	0	3	1	4	2
20	4	2	1	0	0	0	1	5	1	7*	0	0	0	0	0
21	2	1	1	2	1	3	1	0	2	3*	0	0	3	3	1
22	4	1	1	1	0	0	0	0	0	0	1	2	3	6*	1
23	2	1	2	1	0	4	0	1	3	4*	0	1	0	1	1
24	4	1	0	1	0	0	0	0	1	1	2	2	1	5*	0
25	2	2	1	1	1	3	0	1	2	3*	0	1	1	2	2
26	6	0	1	0	0	0	0	0	0	0	3	4	0	7*	0
27	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0
28	7	2	1	0	0	0	0	0	0	0	4	3	3	10*	0
29	8	2	0	0	0	0	0	0	0	0	7	3	0	10*	0
30	8	2	0	0	0	0	4	5	1	10*	0	0	0	0	0
31	3	1	4	1	0	1	0	0	6	6*	0	0	1	1	2
32	6	2	2	0	0	0	0	4	4	8	0	0	1	1	1
33	4	3	0	1	0	2	0	1	3	4	0	0	3	3	1
34	5	2	1	1	1	0	0	1	5	6	0	0	2	2	2
35	2	1	0	4	0	3	0	0	5	5	0	0	0	0	2
36	6	3	1	0	0	0	2	4	3	9	0	0	0	0	1
37	7	1	2	0	0	0	0	0	0	0	0	5	5	10	0
38	7	2	1	0	0	0	0	0	0	0	2	5	3	10	0
TOT.	144	45	38	31	9	52	12	31	79	172	21	36	60	117	28

Total no. of comparisons = 319. LP 8 pref. in 144 comparisons.
LP 8 not pref. in 32 comparisons

* denotes LP 8 kHz response

When a difference was noticed subjects generally expressed a preference; in only 28 comparisons (9.5%) was no preference expressed. For the majority only slight preferences were expressed. This, however, is perhaps not surprising considering the small changes in frequency response that were being compared.

11.3.4.2 Hearing aid simulation and distortion

(i) Hearing aid simulation

The results for the hearing aid processed speech comparisons are shown in Table 11.23. The results for pair 1 show that there was a noticeable difference between the responses of Kemar's left and right open ears for only two subjects.

Table 11.23 Hearing aid processed speech ratings

	Difference						Red Preferred				Green Preferred				No Pref
Pair	4	3	2	1	0	N	3	2	1	T	3	2	1	T	
1	0	1	0	1	0	8	0	0	0	0	0	0	1	1	1
2	1	0	1	3	0	5	0	0	2	2	0	0	2	2	1
3	1	0	1	3	3	2	0	0	3	3	0	0	0	0	5
4	1	0	0	1	0	8	0	0	2	2	0	0	0	0	0
5	1	1	3	1	2	2	0	1	5	6	0	0	0	0	2
6	4	1	3	2	0	0	0	4	6	10	0	0	0	0	0
7	5	1	4	0	0	0	3	3	4	10	0	0	0	0	0
8	5	0	2	0	0	0	0	4	3	7	0	0	0	0	0
9	5	0	1	0	0	0	2	4	0	6	0	0	0	0	0
10	3	1	0	0	0	0	4	0	0	4	0	0	0	0	0

The results for pairs 2 and 3, the hearing aid simulation comparisons, indicate that not all of the subjects found the simulated and real hearing aid processed speech identical. However 8 subjects found little or no difference between them. Considering that only 8 subjects found the responses of Kemar's right and left ears (through which the recordings were made) identical, and that only the 3 dB/octave cut above 4 kHz condition (pair 13) in Table 11.22 was found to give a less noticeable difference, the simulations would appear to be good representations of the real hearing aids. In addition, the simulated response resulted in a reduction of tape noise due to its filtering action. Attempts were made to 'mask' this by low level high frequency noise but were not completely successful. It is therefore conceivable that some subjects were using differences in tape noise to notice differences between the simulated and real hearing aid processed speech.

(ii) Noticeable distortion

Pair 4 compared speech processed through two hearing aids operating well below saturation. They appeared to be nearly identical, differences between them being no greater than those for the open ear condition.

Pairs 5 to 10 present increasing amounts of distortion through the green channel. The actual distortion levels are shown in Appendix J. Very slight amounts of distortion were noticed by all subjects, preferences and confidence ratings getting greater as the levels of distortion were increased.

11.3.4.3 Reliability

The 'dummy' pair of identical LP 8 kHz responses was presented three times at random to each subject during the pair comparison test. All ten subjects on all three occasions found no difference between the responses. The pair comparison test was therefore extremely reliable in not 'detecting' differences that did not exist.

11.3.5 Conclusions

(i) The optimum frequency response for the normally hearing subjects was the flat response with the widest bandwidth (LP 8 kHz). The conclusion can therefore be drawn that the speech presentation system (tape recorder, speech tape, loudspeaker, amplifiers and room) needed no equalization to provide a baseline from which to measure optimum hearing aid responses for the hearing-impaired. Although the speech presentation system had been measured electroacoustically to be flat from 100 Hz to 8 kHz it was necessary to establish whether this would also prove to be the subjective optimum.

(ii) The pair comparison test proved to be a very sensitive procedure for determining subjective differences and preferences between hearing aid responses. Very small differences in frequency response were noticed and preferences expressed.

(iii) Differences in frequency response corresponding to different hearing aid measurement methods (transmission gain/insertion gain and 2 cc coupler/Zwislocki ear simulator/Kemar manikin) were noticed by almost all the subjects for almost all the comparisons.

(iv) 10 dB peaks in the frequency response were noticed and found moderately or much worse by all subjects, whereas 5 dB peaks were not noticed by all subjects and no clear preferences were expressed.

(v) The master hearing aid simulation of real hearing aids proved satisfactory.

(vi) Slight amounts of peak clipping distortion were noticed by all subjects.

(vii) The pair comparison procedure proved reliable in that subjects did not 'detect' differences that did not exist.

(viii) The pair comparison procedure was found simple to carry out by all subjects.

11.3.6 Effects of frequency response changes on comfortable listening levels

11.3.6.1 Introduction

Subjects were instructed, for the pair comparison experiment, to adjust the levels of each channel to their comfortable listening level. These levels could be determined from the digital attenuator settings displayed on the computer VDU and any changes in comfort settings due to changes in the frequency response could be found. In this way, any variations in the absolute comfort levels during the experiment would not affect the relative comfort levels for each pair.

11.3.6.2 Results

The results for low and high frequency cut-offs, mid frequency boost and cuts, peaks and low and high frequency slopes are shown in Table 11.24. The change in comfort level shown is relative to the LP 8 kHz response. For example, rolling off the low frequencies at 150 Hz

Table 11.24 Comfort level differences

Pair	A	B	N	Mean (A-B)	SD	t	95% confidence limits	L_{eq} (A-B)
1	HP 150 Hz	LP 8 kHz	10	+1.60	1.08	4.71*	± 0.62	-0.9
2	HP 200 Hz	LP 8 kHz	7	+1.86	2.04	2.41*	± 1.50	-1.8
3	LP 7 kHz	LP 8 kHz	10	+0.50	1.08	1.46		-0.0
4	LP 6 kHz	LP 8 kHz	3	+1.00	2.00	0.80		-0.0
36	LP 4 kHz	LP 6 kHz	10	-0.60	2.01	0.94		-0.1
37	LP 3 kHz	LP 4 kHz	10	+0.40	0.84	1.50		-0.1
38	LP 2 kHz	LP 3 kHz	10	+0.30	0.68	1.41		-0.0
9	-3 dB/oct. below 500 Hz	LP 8 kHz	8	-1.50	1.07	3.9***	± 0.72	+0.7
10	-6 dB "	LP 8 kHz	8	-1.75	1.04	4.78***	± 0.69	+3.2
11	+3 dB "	LP 8 kHz	10	+0.50	0.85	1.86*	± 0.49	-0.7
12	+6 dB "	LP 8 kHz	6	+2.00	1.67	2.93*	± 1.37	-1.6
15	-3 dB "	LP 8 kHz	10	+0.30	0.68	1.41		-0.2
16	-6 dB "	LP 8 kHz	5	0.00	0.00			-0.3
19	+3 dB "	LP 8 kHz	10	-0.30	0.48	1.96	± 0.28	+0.2
20	+6 dB "	LP 8 kHz	8	-0.50	0.76	1.87		+0.8
13	-3 dB "	LP 8 kHz	9	+0.44	1.42	0.94		-0.0
14	-6 dB "	LP 8 kHz	8	-0.25	0.71	1.00		-0.0
17	+3 dB "	LP 8 kHz	8	-1.62	1.30	3.5**	± 0.87	+0.0
18	+6 dB "	LP 8 kHz	8	0.00	0.53	0.00		+0.0
21	5 dB mid cut	LP 8 kHz	10	+0.60	1.35	1.41		-0.2
22	10 dB mid cut	LP 8 kHz	7	+0.71	1.25	1.50		-0.4
23	5 dB mid boost	LP 8 kHz	10	-1.50	1.51	3.14**	± 0.88	+0.5
24	10 dB mid boost	LP 8 kHz	6	-4.50	3.89	2.84*	± 2.48	+1.0
25	5 dB peaks	LP 8 kHz	10	-1.80	1.81	3.14**	± 1.05	+0.2
26	10 dB peaks	LP 87 kHz	6	-2.66	1.75	3.73**	± 1.66	+1.1

*significance * p < 5% ** p < 1% *** p < 0.1%

(pair 1) required a mean extra output of 1.6 dB to achieve an equivalent comfort setting, whereas adding 10 dB peaks (pair 26) resulted in a mean reduced output of 2.7 dB. The actual measured change in L_{eq} level for the responses relative to the LP 8 kHz response is also given.

The results show that changes in low frequency cut-offs or slopes result in changes in most comfortable listening levels, whereas changes in high frequency cut-offs or slopes have very little effect. For example, the comparison between comfort levels for the widely differing LP8 and LP2 conditions (interpolated from the other LP comparisons as they were not directly compared) shows a change in level much lower than the effect of rolling off the low frequency at only 150 Hz.

The insertion of peaks or mid frequency boost into the frequency response results in changes in comfort levels whereas attenuating the mid frequencies or inserting "dips" have little effect.

In order to compare the comfort settings for the 1.5 dB/octave and 3 dB/octave slopes, Table 11.25 shows the frequencies at which the levels were equal. This appeared to be at approximately 500 Hz.

Table 11.25 Frequencies at which comfort levels equate

Pair	A	B	N	SD (dB)	Equiv. freq.	95% confidence limits	L_{eq} equiv. freq.
5	LP 8 kHz	+1.5 dB/oct	10	1.73	480 Hz	300–760 Hz	480 Hz
7	LP 8 kHz	-1.5 dB/oct	10	2.39	440 Hz	230–830 Hz	440 Hz
8	LP 8 kHz	-3 dB/oct	5	1.30	480 Hz	360–640 Hz	380 Hz

A comparison of the subjective comfort settings and the measured L_{eq} values for the pairs of frequency responses shown in Tables 11.24 and 11.25 indicate that adjusting levels to give equal L_{eq} values gives nearly equivalent comfort levels. For example, for pair 2, subjects turned up the level of the HP 200 Hz response by a mean of 1.86 dB which compensated for a reduction in L_{eq} level of 1.80 dB. Some responses, for example, 5 dB peaks (pair 25), did not give such a close correspondence between L_{eq} values and comfort settings.

11.3.6.3 Conclusion

For the normally hearing subjects in this experiment, changes in low frequency response or mid frequency boosts or peaks in the response affect the most comfortable listening level whereas changes in high frequency response or mid frequency cuts or dips have little effect. For sloping responses of 1.5 and 3 dB/octave comfort settings appeared to equate the levels at approximately 500 Hz. L_{eq} measures of speech levels for the various responses appeared to be a reasonable predictor of comfort level settings, as approximately equal L_{eq} levels of speech gave equal comfort levels.

11.4 Comfortable Listening Levels: The Effects of Stimuli and Instructions

11.4.1 Introduction

The importance and problems of measuring subjects' comfortable listening levels were described in Chapter 4. To decide on the stimuli and instructions for determination of comfortable listening levels for the hearing aid fitting protocol, this experiment examined the:

(i) effect of instructions on comfortable listening level settings

(ii) effect of stimuli on comfortable listening level settings

(iii) reliability of comfortable listening level settings

(iv) relationship between narrow band and broad band comfortable listening levels.

11.4.2 Subjects

The same ten subjects who took part in the pair comparison experiment (Table 11.19) took part in this experiment.

11.4.3 Experimental design and methodology

11.4.3.1 Stimuli

Three types of broadband stimuli (continuous speech, continuous babble and pink noise) and four types of narrowband stimuli (1/3 octave speech, 1/3 octave babble, 1/3 octave noise and pure tones) were used. These stimuli present a wide range of conditions, from most to least real-life-like stimuli. The speech stimuli is the most 'realistic' whereas babble has less variations of level with time. The noise and pure tones are more 'artificial' stimuli but can be more easily calibrated and standardised and have the least variation over time. The speech stimuli were taped and were identical to that used for the pair comparison tests (Chapter 10) while the taped babble stimuli were similar to that recorded for the AVELS test (Chapter 9).

11.4.3.2 Instructions

The levels at which subjects can comfortably listen to sounds have been shown to cover a range. This experiment was designed to examine this range by eliciting listening levels at the QUIETEST COMFORTABLE LEVEL (QCL), the MOST COMFORTABLE LEVEL (MCL), and the LOUDEST COMFORTABLE LEVEL (LCL). The instructions are shown in Appendix K. Attempts to also measure levels at which subjects found the stimuli uncomfortable had to be abandoned due to problems with distortion of the signals at very high output levels.

11.4.3.3 Reliability

Estimates of the reliability of these comfort levels were obtained by repeating the broadband and 2 kHz narrowband conditions during the experiment.

11.4.3.4 Protocol

The pure tone measurement tests were presented first in free field in the anechoic room. All other conditions were presented in the same simulated living room used for the AVELS and pair comparison tests. All measurements of QCL were obtained before measurements of MCL which were obtained before measurements of LCL. This was so that listening to louder sounds did not affect settings for quieter sounds and also so that subjects did not require frequent reinstruction due to a change in task. For each instruction (QCL, MCL and LCL) the test conditions were randomly presented. The complete range of conditions is shown in Table 11.26. The experiment took approximately 1 hour 30 minutes for each subject to complete.

Table 11.26 Stimuli, tests and conditions

Stimuli	Test	Condition (Hz)								
Pure tone	QCL						2 k			
Pure tone	LCL	250	500	750	1 k	1.5 k	2 k	3 k	4 k	6 k
		1/3 octave bands at centre frequency								
Speech	QCL broadband						2 k			
Speech	MCL broadband						2 k			
Speech	LCL broadband	250	500	750	1 k	1.5 k	2 k	3 k	4 k	6 k
Babble	QCL broadband						2 k			
Babble	MCL broadband						2 k			
Babble	LCL broadband						2 k			
Noise	QCL broadband						2 k			
Noise	MCL broadband						2 k			
Noise	LCL broadband						2 k			

11.4.3.5 Measurement of levels

All levels of stimuli quoted are L_{eq} sound pressure levels. The levels of the taped stimuli are those averaged over the whole tape. The actual variations in levels over the tape are given in Table 10.2.

11.4.3.6 Equipment and presentation system

The pulsed pure tone stimuli were presented in free field in an anehoic room (Appendix M). The binaural free field thresholds of the subjects were also measured using the same system and are shown in Table 11.27, for comparison with comfortable level measures.

The pure tone comfort measures were obtained by ascending in 5 dB steps and descending in 10 dB steps until consistent results were obtained (at least 2 out of 3 similar responses and greater than 50%).

Table 11.27 Free field binaural thresholds (dB SPL)

Frequency (Hz)	Mean	SD
250	16.5	3.34
500	8.0	4.08
750	6.5	3.37
1000	6.5	4.24
1500	7.5	5.99
2000	1.0	4.22
3000	-4.0	4.22
4000	1.0	6.14
6000	12.0	3.94
8000	27.0	5.16

The speech, babble and noise stimuli were presented using the same system as for the pair comparison experiment; the speech and babble being tape recorded, whereas the pink noise was produced by a pink noise generator. The third octave stimuli were obtained using the digital filter. Subjects were allowed as long as required to make the comfort level adjustments. Levels were adjusted by the subject in the same manner as for the pair comparison procedure using the software described in Chapter 8.

11.4.4 Results

11.4.4.1 Absolute levels

The mean and standard deviation sound pressure levels for the various comfort settings are shown in Tables 11.28 and 11.29. It can be seen that all the stimuli were found comfortable to listen to over a dynamic range of some 20 to 30 dB, the MCL coming approximately midway between the QCL and the LCL.

Subjects sometimes expressed difficulty in selecting the QCL for the non-speech broad band stimuli and all the narrow band stimuli, but all managed to accomplish the task. This difficulty may have contributed to the slightly higher standard deviations obtained for the QCL tests than the MCL or LCL tests. Broad band stimuli appear to give lower standard deviations than narrow band stimuli, and speech stimuli give lower

Table 11.28 Comfort levels (dB SPL)

2 kHz

Stimuli	QCL Mean	QCL SD	MCL Mean	MCL SD	LCL Mean	LCL SD
1/3 oct. Speech	55.75	11.40	63.10	5.46	71.60	5.62
1/3 oct. Babble	47.25	10.92	57.05	7.17	66.40	8.03
1/3 oct. Pink noise	45.00	13.68	60.00	9.64	69.90	10.17
Pure tone	43.50	14.42			70.50	11.11

Broad band

Stimuli	QCL Mean	QCL SD	MCL Mean	MCL SD	LCL Mean	LCL SD
Speech	50.80	7.17	64.65	2.84	79.90	3.75
Babble	53.10	7.08	64.55	4.00	79.85	5.17
Pink noise	47.45	9.04	60.75	7.43	74.25	10.96

Table 11.29 LCL 1/3 octave speech and pure tone comfort levels (dB SPL)

	LCL			
	1/3 octave speech		pure tone	
Frequency (Hz)	Mean	SD	Mean	SD
250	80.80	5.39	71.50	10.34
500	76.60	3.66	69.50	11.30
750	75.70	4.57	69.00	11.00
1000	72.90	5.76	72.50	11.56
1500	72.90	7.37	74.90	11.71
2000	71.60	5.62	70.50	11.11
3000	68.40	5.50	58.90	13.05
4000	64.50	7.20	61.50	12.03
6000	68.80	8.00	69.50	12.48

standard deviations, than non-speech stimuli. Subjects sometimes voluntarily expressed the opinion that comfort levels were easier to set for the broad band speech stimuli.

11.4.4.2 Difference between stimuli

The differences in the comfort levels for the stimuli are shown in Tables 11.30 and 11.31. The only significant difference between the speech and pure tone comfort measures was for the LCL instructions at 250 Hz, 500 Hz and 3 kHz.

For the speech and noise measures significant differences were found for the 2 kHz QCL and broad band MCL conditions. For the speech and babble measures significant differences were found for all tests at 2 kHz.

11.4.4.3 Reliability

The test re-test reliabilities for the various conditions are shown in Table 11.32. All stimuli gave good test retest reliability, the only condition giving a significant change in comfort setting on retest being the 1/3 octave noise at 2 kHz for the QCL instruction.

11.4.4.4 Relationship between narrow band and broad band comfort settings

The mean 1/3 octave narrow band and broad band comfort settings for the speech stimuli are shown in Figure 11.5. The levels for the broad band speech stimuli were obtained by measuring the 1/3 octave speech spectrum for the broadband comfort level.

The LCL narrow band levels appear to follow the contour of the speech spectrum but at a 10-15 dB higher level. The MCL 2 kHz narrow band level appears to be at some 19 dB above the broad band 2 kHz level, whereas the QCL 2 kHz narrow band comfort level is approximately 25 dB above the broad band 2 kHz level.

The subjects' mean binaural free field thresholds are shown for comparison with the comfort levels.

Table 11.30 Comfort level differences between stimuli

SPEECH - BABBLE

	Test	Mean (dB)	SD	t	95% confidence limits
QCL	2 kHz 1/3 oct.	6.50	5.36	3.83**	± 3.83
MCL	2 kHz 1/3 oct.	6.05	3.33	5.75***	± 2.38
LCL	2 kHz 1/3 oct.	5.20	4.29	3.83**	± 3.07
QCL	Broadband	-2.30	3.94	1.84	
MCL	Broadband	0.10	2.89	0.11	
LCL	Broadband	0.05	2.98	0.05	
SPEECH - NOISE					
QCL	2 kHz 1/3 oct.	8.20	8.13	3.19*	± 5.82
MCL	2 kHz 1/3 oct.	4.10	6.61	1.96	
LCL	2 kHz 1/3 oct.	1.70	5.64	0.95	
QCL	Broadband	3.35	6.88	1.54	
MCL	Broadband	3.90	5.36	2.30*	± 3.83
LCL	Broadband	5.65	9.08	1.95	
1/3 octave SPEECH - PURE TONE					
QCL	2 kHz	10.65	16.18	2.08	
LCL	2 kHz	1.10	9.67	0.36	

Table 11.31 LCL 1/3 octave speed - pure tone comfort level differences

Frequency (Hz)	Mean (dB)	SD	t	95% confidence limits
250	9.30	10.40	2.83*	± 7.44
500	7.10	9.26	2.43*	± 6.62
750	6.70	9.51	2.23	
1000	0.30	9.45	0.10	
1500	-1.90	10.75	0.56	
2000	1.10	9.67	0.36	
3000	9.50	12.09	2.49*	± 8.64
4000	3.00	11.09	0.86	
6000	0.70	10.23	0.22	

Table 11.32 Reliability of comfort level settings

		Mean (dB)	SD	t
1/3 oct. speech at 2 kHz	QCL	0.9	4.86	0.59
1/3 oct. babble at 2 kHz	QCL	1.5	4.35	1.08
1/3 oct. noise at 2 kHz	QCL	2.1	2.64	2.51*
Broadband speech	QCL	-1.50	2.30	2.04
Broadband babble	QCL	0.80	4.13	0.60
Broadband noise	QCL	-0.90	3.10	0.92
1/3 oct. speech at 2 kHz	MCL	-1.50	2.54	1.86
1/3 oct. babble at 2 kHz	MCL	0.70	2.36	0.94
1/3 oct. noise at 2 kHz	MCL	2.80	4.05	2.19
Broadband speech	MCL	0.50	1.78	0.89
Broadband babble	MCL	1.30	1.83	2.25
Broadband noise	MCL	-0.90	3.55	0.85
1/3 oct. speech at 2 kHz	LCL	0.80	4.29	0.59
1/3 oct. babble at 2 kHz	LCL	-0.40	3.27	0.39
1/3 oct. noise at 2 kHz	LCL	1.00	3.27	0.97
Broadband speech	LCL	0.40	2.07	0.61
Broadband babble	LCL	0.10	2.28	0.14
Broadband noise	LCL	0.90	2.80	1.01

11.4.5 Summary of results

(i) All stimuli (pure tones, speech, babble and noise) gave good test retest reliability.

(ii) For the majority of test conditions no differences between test stimuli were found, although for a few conditions small but significant differences were measured: In these conditions the speech stimulus gave slightly higher comfort levels.

(iii) Subjects stated that the broad band speech stimulus was the easiest to set comfort levels for, while the QCL was the hardest instruction to carry out.

(iv) Subjects found all stimuli comfortable to listen to over a dynamic range of 20 to 30 dB, the MCL coming approximately midway between the QCL and the LCL.

(v) For the speech stimuli, the LCL narrow band levels followed the contour of the speech spectrum but were 10-15 dB above the LCL broad band speech 1/3 octave levels. The MCL 2 kHz narrow band level was 19 dB above the MCL broad band 2 kHz 1/3 octave level and the QCL 2 kHz narrow band level was 25 dB above the QCL broad band 2 kHz 1/3 octave level.

11.4.6 Conclusions and discussion

11.4.6.1 Choice of stimuli

Speech stimuli would appear to be the best stimuli for obtaining both narrow and broad band comfort level settings for the following reasons:

(i) The speech stimulus was found the easiest to set comfort settings for.

(ii) The speech stimuli gave as good test-retest reliability as the other stimuli.

(iii) No systematic differences were found between the speech stimuli comfort levels and those of the other stimuli.

(iv) Speech is the most realistic real-life stimulus.

(v) The relationship between narrow band and broad band comfort levels can be studied most directly by using narrow and broad band versions of the same stimuli.

The pure tone results were very similar to the narrow band speech stimuli results and so would appear to be an adequate substitute. The pure tone measures were obtained using 5 dB steps in an ascending manner as would be often used in clinical procedures. There are therefore two methodological differences in the way in which the pure tone and speech band comfort levels were determined. Firstly, the pure tone levels were measured in 5 dB steps whereas the speech band levels were measured in 1 dB steps. The pure tone measures could therefore overestimate the LCL by up to 5 dB. Secondly, the speech levels were under direct subject control whereas the pure tone levels were under experimenter control and so might result in a conservative judgement of LCL by the subject. These two factors may conceivably have cancelled each other out in this experiment. Alternatively, they may have masked real differences between pure tone and speech band comfort levels. Any such differences might, however, be expected to be fairly small.

11.4.6.2 Relationship between narrow band and broad band comfort levels

The finding that the Loudest Comfortable Levels for 1/3 octave speech stimuli were some 10–15 dB higher than the 1/3 octave band levels for the loudest comfortable level for broad band speech supports the hypothesis put forward in Section 4.3.3. This suggested that the widely accepted approach to hearing aid fitting of amplifying the broad band speech signal such that each band was at the most comfortable level for that band could result in an overall level that was above the most comfortable level for the broad band speech signal. For the normally hearing subjects used in this study, this approach would result in speech levels 10–15 dB too loud. Attempts to present band levels at MCL for normally hearing subjects might, however, still result in the broad band levels being close to LCL due to their large range of comfortable listening levels, and so still not cause discomfort. This may well not be the case with hearing-impaired subjects, who would have to compensate by reducing the gain setting of their hearing aid.

The absolute speech level also has only a slight effect on the subjective relative loudness between high and low speech frequencies for normally hearing subjects due to their equal loudness contours being almost parallel. For hearing-impaired subjects this need not be the case and so the absolute speech level might greatly affect the subjective loudness balance between low and high frequencies. A frequency response that is found to give the optimum balance at one gain setting might therefore not be appropriate at another gain setting. Additionally, for the normally hearing subjects the decrease in the difference between narrow and broad band 1/3 octave comfort levels for the LCL over the QCL conditions suggests a reduction in loudness summation as the broad band stimuli level is increased. This would result in a difference between broad band and narrow band loudness growth functions and have possible implications for hearing aid fitting using compression amplification or equal loudness contour fitting as described in Chapter 4.

11.5 Summary of Baseline Study Results and Conclusions

11.5.1 AVELS test

The AVELS test was shown to have small practice effects and equivalent test forms. Phoneme scoring gave similar results to whole word scoring and the AVELS test sections provided baseline scores for comparing how hearing-impaired subjects perform under the various audio visual and noise conditions. Some relationships between AVELS and FADAST test scores were found although they appear to be testing somewhat different abilities; the AVELS test involving the contribution of linguistic and prosodic factors, while the FADAST test does not. There was some indication that critical ratio measures bore a closer relationship to test scores than audiometric threshold measures for normally hearing subjects. The AVELS tests proved to be simple to administer and gave little difficulty for the subjects in understanding the task required.

The learning effects found for presentations of a difficult condition following an easy condition indicate that artefactual improvements of 3% to 10% could occur in the score for the difficult condition.

The small learning effects found suggested that repeat measures of the complete AVELS tests would not confound subjective judgements of hearing aids. The use of separate test forms would also seem to be satisfactory for subjective judgements.

Estimates of test retest reliability indicate that repeated measures of the complete AVELS test would be more sensitive to differences between hearing aids (small critical differences) than would the use of separate test forms if objective scoring was employed. This is due to the larger number of items involved. The use of phoneme scoring added little to the sensitivity of the test.

11.5.2 Pair comparison test

The pair comparison test proved to be very sensitive in reliably determining subjective differences and preferences between hearing aid responses. The task involved was quickly understood and found easy to carry out. The speech presentation system was found to be subjectively optimum without recourse to equalisation. The Master Hearing Aid simulation of hearing aids proved satisfactory. Comfortable listening levels were dependent on frequency response in a fairly predictable manner.

11.5.3 Comfortable listening levels: the effects of stimuli and instructions

Speech, pink noise, babble and pure tones were found to give reliable, reasonably similar comfort level settings for all instructions. However, speech appeared to be the first choice stimulus as it was found the easiest to set comfort settings for and has the most face validity as a realistic real-life stimulus. Subjects found stimuli comfortable to listen to over a range of 20 to 30 dB, the most comfortable level coming approximately midway between the quietest and loudest comfortable levels.

There appeared to be a reduction in the loudness summation for broad band stimuli as the levels increased. The LCL levels for the 1/3 octave speech stimuli were some 10–15 dB higher than the 1/3 octave band levels for the LCL broad band speech, although they followed a similar contour. This could have important implications for hearing aid fitting as it might not be possible for subjects to tolerate speech presented such that its third octave bands are at MCL as has been widely advocated (Section 4.3.3).

Figure 11.1

Pair Comparison Digital Filter Responses

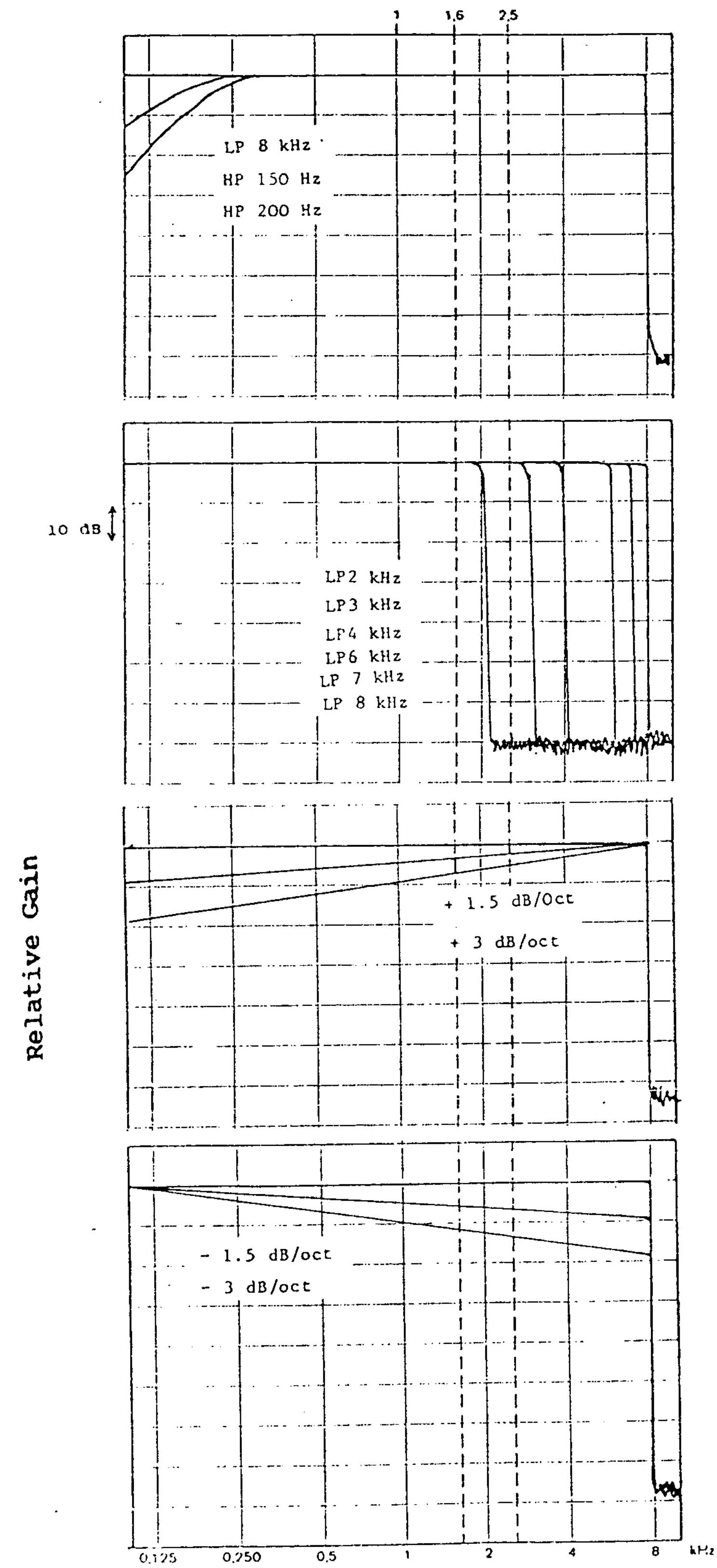


FIGURE 11.1 cont.

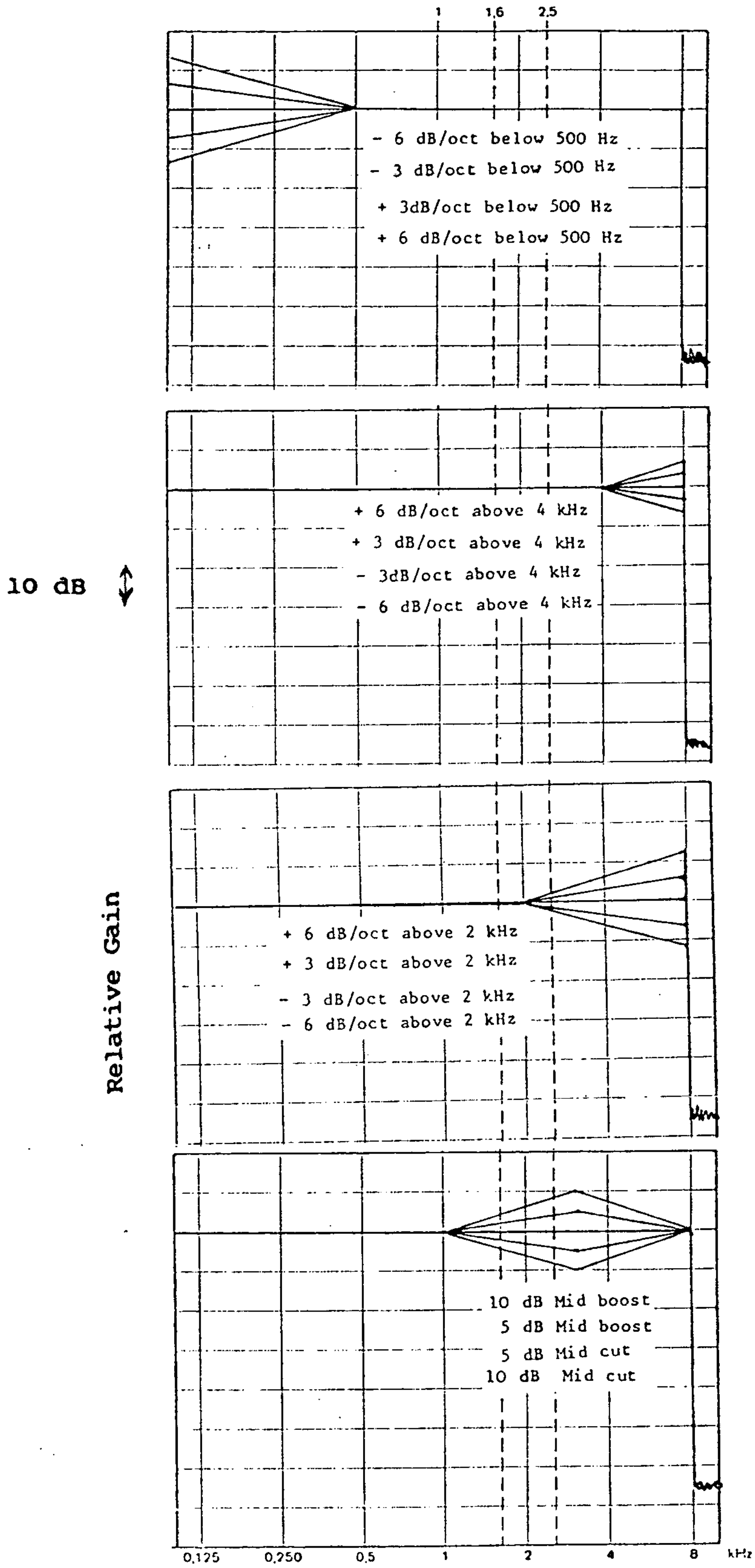


FIGURE 11.1 cont

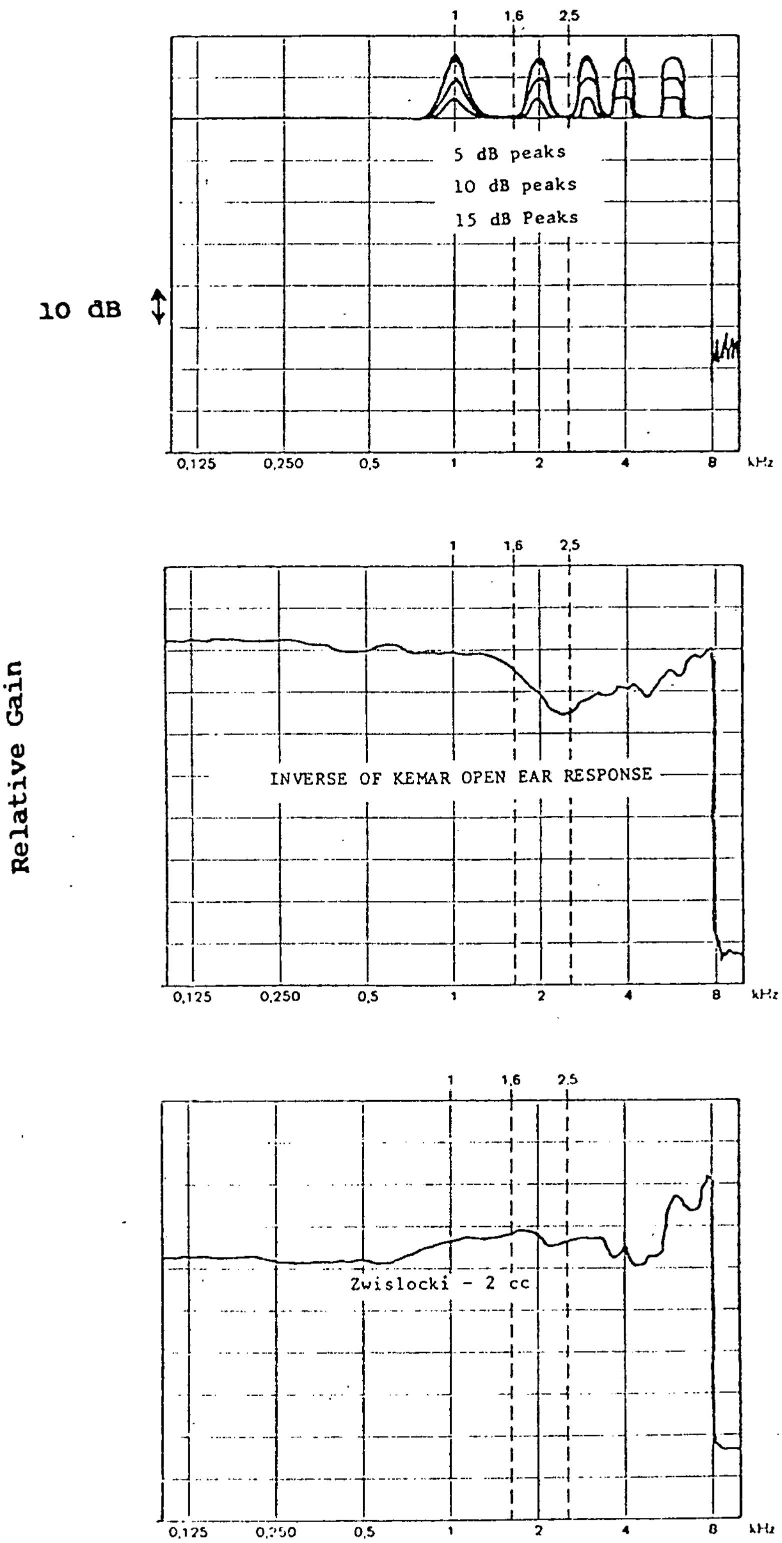


FIGURE 11.1 cont.

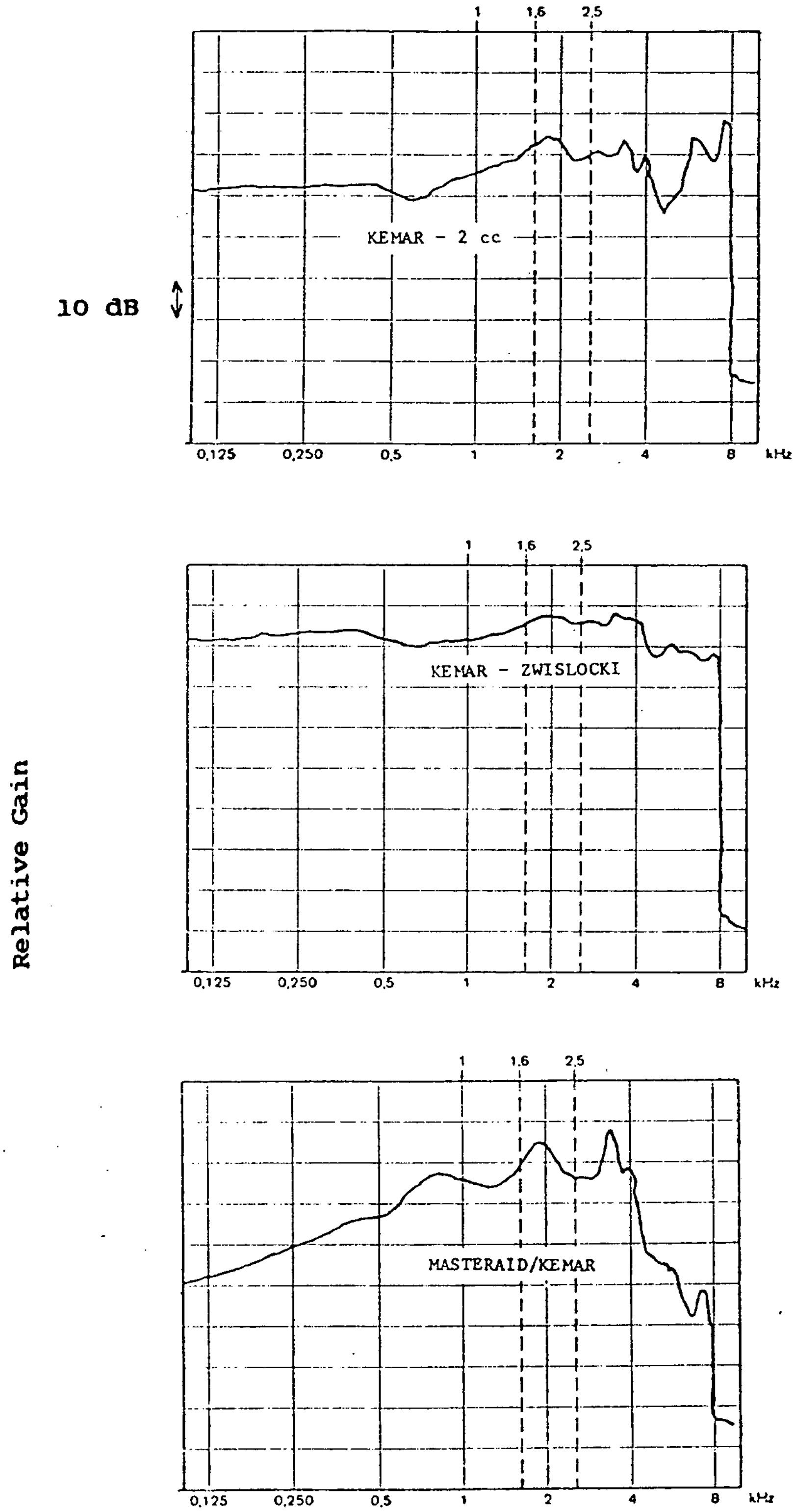


FIGURE 11.1 cont.

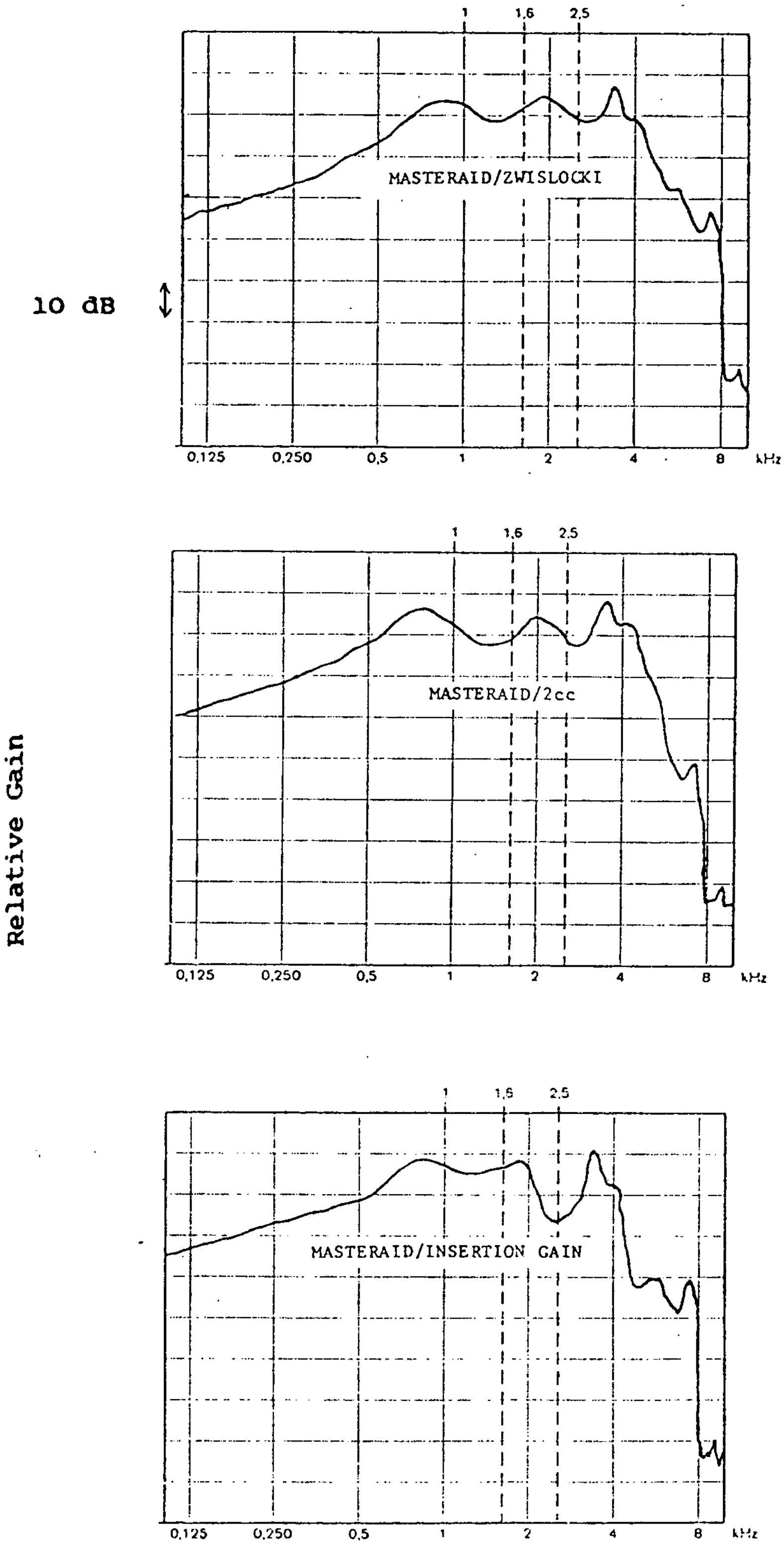
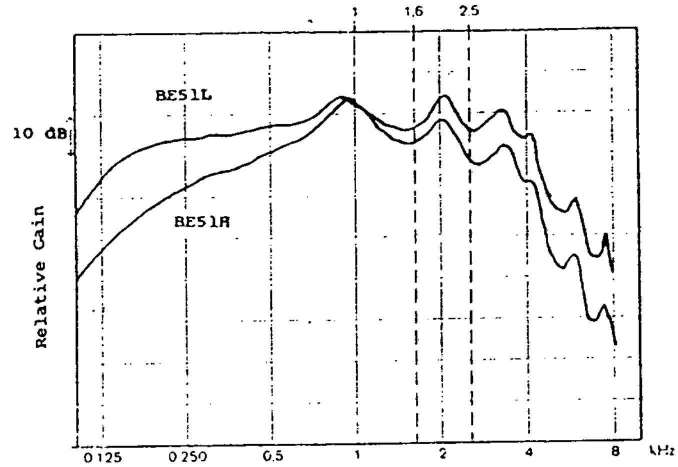


Figure 11.2

Hearing aid simulations



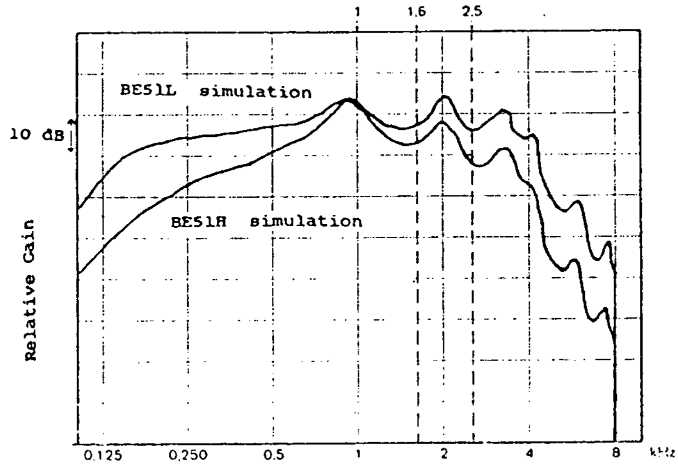


Figure 11.3

Inverse of Kemar open ear response

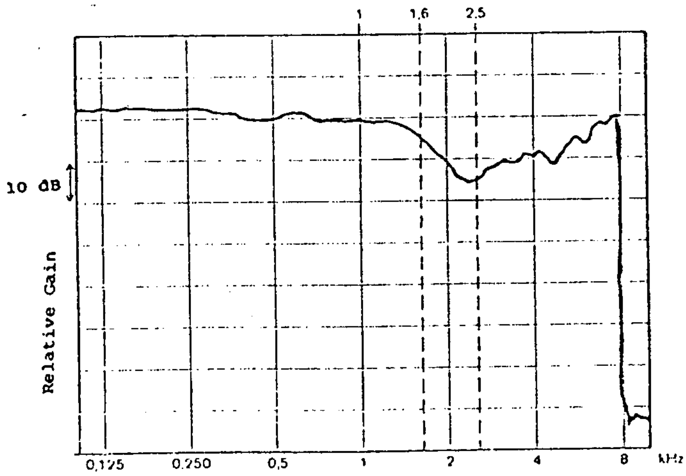


Figure 11.4

Pair Comparison presentation system (Baseline Study)

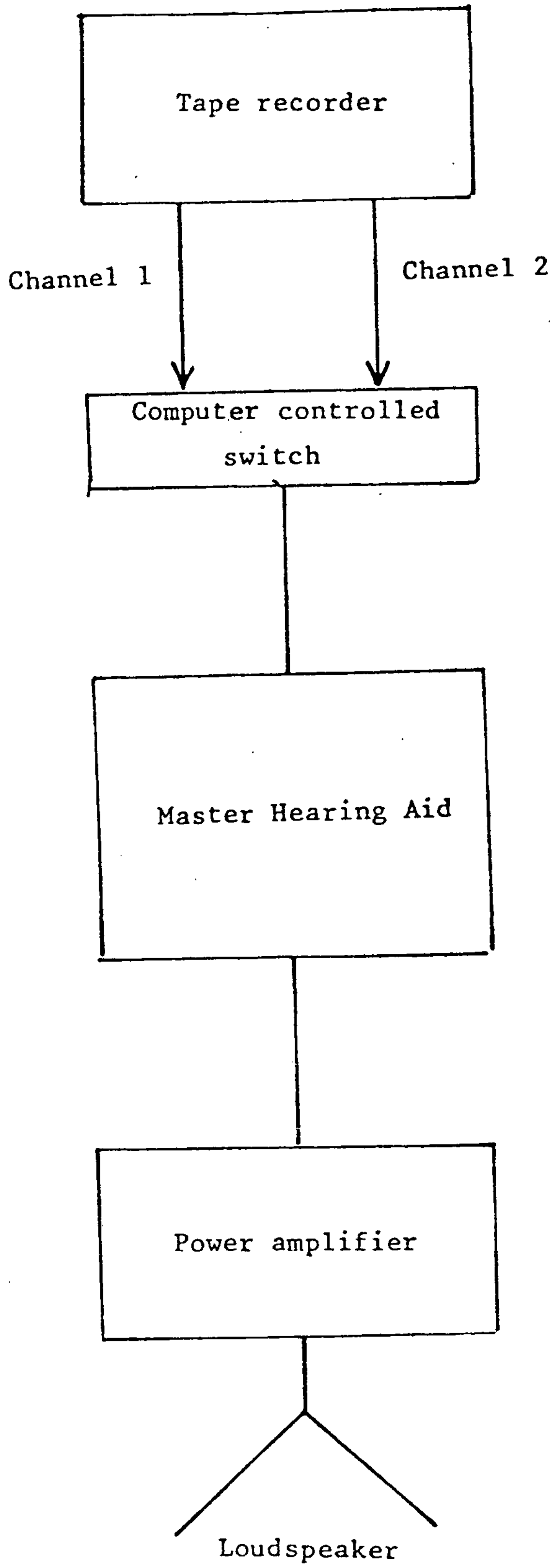
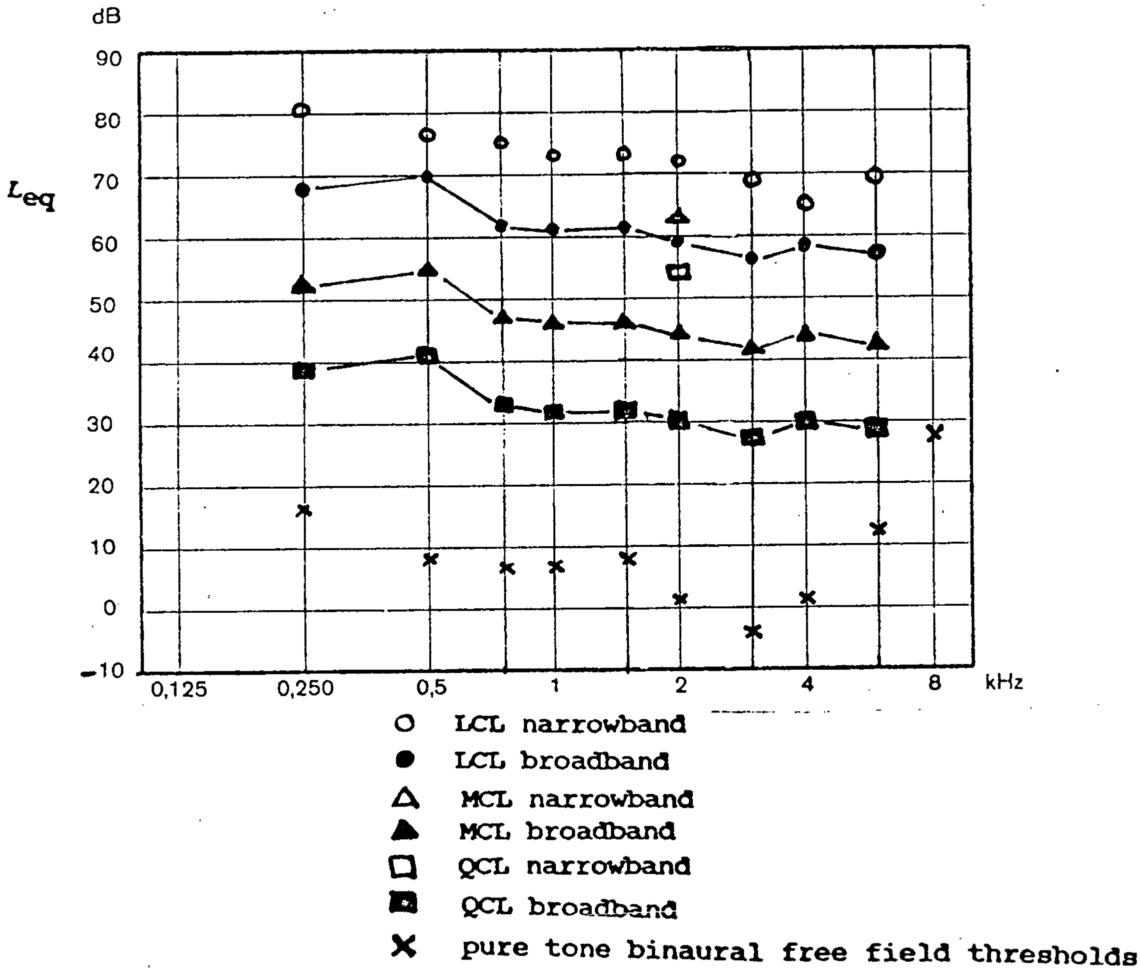


Figure 11.5

Mean narrow and broadband 1/3 octave band comfort levels



CHAPTER 12

MAIN STUDY

12.1 Introduction

The study described in Chapter 11 provided baseline measures of the AVELS and pair-comparison tests for normally hearing subjects. It also provided information concerning the relationship between frequency response, stimuli bandwidth and comfort settings. The main study was designed to obtain similar information for severely hearing-impaired subjects. The frequency response differences used for the pair comparison procedure were selected from those that were noticed by all the normally hearing subjects. For comfort level measurements, running speech was used as this had been shown to be the most satisfactory stimulus (section 11.5.3). Subjects listened through individual earmoulds fitted to the master hearing aid and measurements of ear canal sound pressure levels were made for all subjects.

In addition to the pair comparisons used in the normally hearing study, a comparison of hearing aid fitting procedures was carried out using the AVELS and pair comparison tests.

The correlation between AVELS test scores and real-life intelligibility judgements was also determined to indicate how well the AVELS tests predicted real-life performance.

12.2 Subjects

Twelve moderate to severely hearing-impaired subjects of predominantly cochlear pathology and selected from the files of a local hospital hearing aid clinic volunteered to participate in the experiment. Their pathology and audiometric thresholds were verified using air and bone conduction audiometry, tympanometry and tests of acoustic reflex and tone decay. Critical ratios at 2 kHz were measured using the method described in Appendix L, but due to the magnitude of hearing loss satisfactory estimates were not possible for seven of the subjects. All subjects were experienced users of National Health Service BE30 or BE50 series aids and had measurable hearing at all frequencies between 250 Hz and 8 kHz.

This group of subjects was chosen as possibly having the potential to benefit from changes in frequency response while also relying to some extent on visual cues for receptive communication. Subjects' audiometric profiles obtained under headphones are shown in Tables 12.1 and 12.2, whereas further details of subjects obtained by interview questionnaire are given in Appendix L.

Table 12.1 Subject headphones A.C. thresholds for test ear, dB HTL

	250	500	750	1k	1.5k	2k	3k	4k	6k	8k
1	55	65	75	75	75	70	65	75	80	80
2	55	65	60	60	40	50	45	50	80	70
3	35	45	50	55	85	85	85	80	95	70
4	70	75	80	85	85	85	80	85	80	80
5	50	65	65	65	80	90	85	95	105	80
6	40	65	85	95	90	90	85	80	80	80
7	50	55	55	55	40	50	55	65	65	65
8	45	60	60	55	50	75	80	100	95	80
9	65	70	85	95	95	80	75	80	75	75
10	95	90	85	90	80	75	80	80	85	100
11	40	55	60	65	65	65	80	85	110	90
12	55	65	65	55	50	50	45	40	55	50
Mean	54.6	64.6	68.8	70.8	69.6	72.1	71.7	76.3	83.8	78.3
SD	16.3	11.2	12.6	16.4	19.8	15.3	15.3	17.2	15.7	14.2

Table 12.2 Subject audiometric data

	Test Ear			Non Test Ear	
Subject No.	AC average 500 Hz-4 kHz dB HTL	Air-bone gap	2 kHz CR	AC average 500 Hz-4 kHz dB HTL	Air-bone gap
1	71.25	0	30	No response (115)	-
2	56.25	30	30	62.50	35
3	66.25	27.5	-	66.25	26.25
4	82.50	0	-	83.75	0
5	78.75	35	-	42.50	12.5
6	82.50	0	-	87.5	0
7	56.25	15	22.5	75.0	15
8	72.50	0	-	40.0	0
9	81.25	25	-	96.25	35
10	83.75	0	-	18.75	0
11	66.25	0	36.5	51.25	0
12	52.5	0	27.5	50.0	0
Mean	70.83	11.04		65.73	10.31
SD	11.34	14.36		27.17	14.28

12.3 Experimental Procedures

12.3.1 Earmoulds

Temporary earmoulds were made for all subjects using silicone impression material. This method provided earmoulds with a perfect seal as verified using the manometer and pump of the impedance meter. As the subjects would wear the earmould only for a few hours in the laboratory it was not necessary to manufacture more permanent earmoulds. Temporary ear tips or foam plugs were not used, as repeatable probe tube and hearing aid measurements required that the earmould could be positioned by a subject to the same depth every time it was inserted. The silicone earmoulds were core bored to take the 2 mm bore sound tube and the 1 mm bore probe tube. The sound tube length was that necessary to allow the earmould coupled to the hearing aid to be worn comfortably by the subject. The sound tube ended flush with the earmould tip while the probe continued 5 mm past the earmould tip.

12.3.2 Probe microphone measures

The miniature microphone used for all probe tube measurements was a Knowles electret EA 1934. Either 30 mm or 40 mm of 1 mm internal bore vinyl tubing was used as the probe tube, depending on the individual ear canal geometry. Calibration curves for the probe tube microphone system are given in Appendix O. The sensitivity of the probe microphone was similar to that of the B & K half inch microphone and so allowed for simple coupling to the input of the audio test station. The probe microphone was powered by a battery contained in a small box hung around the subject's neck.

Probe microphone measures were made of subjects' pure tone thresholds and insertion gains using the master hearing aid and individual temporary earmoulds. The pure tone threshold measurements were made using the master hearing aid receiver electrically driven from the external output of the audiometer, as shown in Figure 12.1. Insertion gains were determined by measuring the individuals' *in situ* acoustic gains at the probe tube position with and without the master hearing aid, as shown in Figure 12.2. Similar probe tube positions for these measurements were ensured by noting the position of the probe microphone relative to landmarks of the individual's ear, and taping the microphone in position. All probe tube gain measurements were obtained in an anechoic room with the subject sat facing the coaxial loudspeaker which was driven from the audio test station and delivered a constant sound pressure level at the subject's position. The master hearing aid digital filter was set to the LP 8 kHz (flat from 100 Hz to 8 kHz) condition for these measurements.

The probe tube microphone measurement experiment took approximately one hour for each subject.

12.3.3 Measurements of comfort levels

Subjects' LCLs were determined under headphones using the same procedure as in the previous study. Subjects' MCLs and LCLs were determined using one third octave bands of speech in a similar way to that described in the study in Section 11.4, except that the hearing-impaired subjects listened through the headworn master hearing aid and their individual earmoulds. The output level from the loudspeaker remained constant and the subject controlled attenuator determined the hearing aid gain.

In addition to measurements of MCL and LCL, measurements of loudness discomfort levels (LDL) were made using the programme described in Chapter 8 and their instructions shown in Appendix K.

The range of comfort measures determined in this experiment is shown in Table 12.3. For the few subjects for whom LCLs at all frequencies could not be determined due to the output limits of the system, MCL measures at all frequencies were obtained.

The measurement of comfort level for each subject took approximately one hour.

Table 12.3 Stimuli and conditions for comfort or discomfort tests

Stimuli	Test	Condition (band centre frequency, Hz)								
Pure Tone	LCL		500		1000		2000		4000	
Speech	MCL Broadband		500				2000			
	LCL Broadband	250	500	750	1000	1500	2000	3000	4000	6000
	LDL Broadband		500				2000			

12.3.4 Pair comparison test

12.3.4.1 Introduction

The pair comparison procedure was similar to that used in the normally hearing study except that the hearing-impaired subjects listened through the headworn master hearing aid, as described in Chapter 7. The pair comparison test required approximately one hour of testing for each subject.

12.3.4.2 Selection of frequency responses

The normally hearing subjects used in the study in Chapter 11 were required to give their judgements on frequency responses that varied slightly from the hypothesized optimum (i.e., LP 8 kHz). The initial choice of a close to optimum response allows for an efficient adaptive protocol to home in on the actual optimum. For hearing-impaired subjects a good first estimate of the optimum response would appear to be a hearing aid response which placed each third octave band of the speech signal at the subject's most or loudest comfortable level for that band. The philosophy and widespread support for this approach were discussed in Chapter 4.

Since the MCL and LCL for one third octave bands of speech for each subject had been determined using the master hearing aid, these attenuator levels could be used to create the individual's digital filter response that accomplished this required 'speech band comfortable level' (SBCL) hearing aid fitting. The pair comparisons were presented relative to this 'SBCL' response.

So that a complete new set of digital filter responses did not have to be created for each person, the SBCL response was programmed into digital filter B, whereas the pairs of responses for comparison were programmed into filter A. The filters were in series as shown in Figure 12.3.

The responses used for the pair comparison are listed in Table 12.4 and shown in Figure 12.4. Pair 1 is a dummy response presented at random, as described in the pilot study, to determine the reliability of subjects' responses. Pairs 2, 3 and 4 determined the optimum high pass cut-off frequency. Pairs 5, 6, 7, 8, 9 and 10 determined the optimum low pass frequency, while pairs 11, 12, 13, 14, 15 and 16 determined the

Table 12.4 Pair comparison digital filter responses

Pair	Green	Red
1	LP 8 kHz	LP 8 kHz
2	HP 400 Hz	LP 8 kHz
3	LP 8 kHz	HP 250 Hz
4	HP 400 Hz	HP 250 Hz
5	LP 8 kHz	LP 2 kHz
6	LP 8 kHz	LP 4 kHz
7	LP 2 kHz	LP 6 kHz
8	LP 6 kHz	LP 8 kHz
9	LP 6 kHz	LP 4 kHz
10	LP 2 kHz	LP 4 kHz
11	+ 6 dB/octave	LP 8 kHz
12	+ 3 dB/octave	LP 8 kHz
13	+ 6 dB/octave	+ 3 dB/octave
14	LP 8 kHz	-6 dB/octave
15	- 3 dB/octave	LP 8 kHz
16	- 3 dB/octave	- 6 dB/octave
17	10 dB peaks	LP 8 kHz
18	10 dB mid-boost	LP 8 kHz
19	20 dB mid-boost	LP 8 kHz
20	LP 8 kHz	10 dB mid cut
21	20 dB mid cut	LP 8 kHz
22	LP 8 kHz	Inverse of Kemar open ear response
23	LP 8 kHz	Zwislocki - 2 cc diff.
24	Kemar - individual insertion gain diff.	LP 8 kHz

optimum slope. Any interaction between low and high pass cut-off frequency and slope was checked by repeating the optimum slope comparison with the optimum low and high pass cut-off frequencies.

Pair 17 determined the effect of peaks in the frequency response. Pairs 18, 19, 20, 21 examined the effect of mid frequency boost and cut. Pair 22 determined the effect of the loss of natural ear canal resonance, while pair 23 examined whether the difference between coupler and ear simulator measurements was of importance. Pair 24 was presented to determine whether subjects found any difference or preference between a hearing aid specified in terms of its insertion gain measured on Kemar and one specified by its insertion gain on that subject. The results of this comparison would indicate whether individual insertion gain measurements might be of value in hearing aid fitting. The Kemar-individual insertion gain difference response was obtained by programming the digital filter with the difference between the master aid's insertion gain measured on Kemar and the master aid's insertion gain measured on the individual subjects by the probe microphone method described in 12.3.2. This response was therefore different for each subject.

In addition, the same hearing aid processed speech tape used in the study on normally hearing subjects was used to examine the effects of hearing aid distortion for the hearing-impaired subjects.

In order to adaptively find the optimum response in the most efficient manner, pairs with the greatest difference were compared first. If no difference was noticed, no further pairs were compared for that parameter. The narrower bandwidths or flattest sloped responses were arbitrarily taken as the optimum response when no difference was noticed. When no preference was expressed, the widest bandwidth or the flattest response were assumed to be the optimum. This approach therefore assumed the response closest to the SBCL was the optimum unless proved otherwise. If extra bandwidth provided no noticeable difference, the narrower bandwidth response allowed greater output with less chance of feedback. The protocol followed is indicated in Figure 12.5.

12.3.4.3 Comparison of hearing aid prescription procedures

The most widely used hearing aid prescription procedures would appear to be (Section 4.3.2):

(i) some form of Speech Band Comfortable Level approach

(ii) the Berger (1979) method

(iii) the Byrne and Tonnison (1976, 1978) method.

Both Berger's and Byrne and Tonnison's procedures specify required aided thresholds based on individual's unaided hearing thresholds. Their procedures are outlined in Appendix P, and were described briefly in Chapter 4.

Other possible approaches to hearing aid fitting include providing a fixed response for everyone or the other extreme of allowing each person to choose their preferred response.

The outcome of the pair comparison test provided the individuals' subjectively chosen 'preferred' response and showed whether modifications of the SBCL approach were required.

To compare hearing aid selection procedures the subjectively chosen response, 'Berger' prescribed response, 'Byrne and Tonnison' prescribed response and a fixed response were programmed into digital filter A and a pair comparison test conducted, as indicated in Table 12.5. Digital filter B was not used for this part of the experiment. The fixed response was the LP 8 kHz condition which simulated for the master hearing aid a National Health Service BE50 series aid. The required digital filter responses to obtain each individual's Berger, and Byrne and Tonnison prescribed Master Hearing Aid responses were calculated from the probe tube measures of each individual's thresholds and acoustic gain, described in 12.3.2.

Table 12.5 Hearing aid selection procedure, pair comparison

	Green	Red
1	Chosen	Berger
2	Byrne	Chosen
3	Chosen	Fixed
4	Byrne	Berger
5	Berger	Fixed
6	Fixed	Byrne

The subjectively chosen response was derived by adding the SBCL response to the optimum pair comparison response determined as in Section 12.3.4.2. These individual calculations and programming of the digital filter took approximately two hours per subject.

12.3.5 Audiovisual tests

12.3.5.1 FADAST test

The FADAST test was presented to subjects wearing their own hearing aid(s). The presentation levels, equipment and scoring procedure were identical to those used for the study on normally hearing subjects.

12.3.5.2 AVELS test

The best and worst aid responses found using the pair comparison test of the four hearing aid selection procedures were used for the AVELS test. Subjects received three presentations of the complete AVELS test. The presentation protocol was similar to that used for the normal study. One aid response was used for the first presentation, the other for the second presentation and the first aid response used again for the third presentation. Learning effects for the test could thus be determined. Subjects were not told that the third presentation used an identical aid response to that used for the first presentation. All presentation levels were those used for the 'easy' condition in the study on normally hearing subjects.

Subjects were asked for their subjective preferences after the first two presentations of the test. For half the subjects the best aid on the pair comparison test was presented first, for the other half the worst aid was presented first. Subjects were asked the same questions as in the pair comparison test (Appendix K.1 (1),(2),(3)&(4))

Environmental sounds

Subjects listened to the environmental sounds section of the AVELS videotape using their own hearing aids and were asked to give their judgements on how true to life they were.

The audiovisual testing session lasted approximately 1 hour 15 minutes for each subject.

12.4 Results

12.4.1 Acoustic measurements

12.4.1.1 Relationship between probe microphone and headphone threshold measurements

Probe microphone measurements of threshold were made as described in 12.3.2. Headphone thresholds could be expressed in terms of dB SPL measured in the 6 cc calibration coupler (BS 2497). It was of interest to see by how much these widely differing methods of expressing hearing thresholds differed, to give some indication of the errors that might be involved in hearing aid fitting procedures based on headphone measurements. The analysis is shown in Table 12.6. It can be seen that although some significant differences exist, they are not generally large differences, except for 6 kHz.

Table 12.6 Differences between probe microphone and headphone measurement of threshold

Frequency (Hz)	250	500	1 k	1.5 k	2 k	3 k	4 k	6 k	8 k
Mean probe-headphone difference (dB)	-2.92	1.25	0.67	4.67	4.92	1.42	-2.25	-13.7	2.54
SD	4.44	3.19	3.94	5.10	3.73	4.14	5.07	8.76	9.13
t	2.28*	1.36	0.59	3.17**	3.44**	1.18	1.54	5.18***	0.93

12.4.1.2 Relationship between individual occluded ear probe microphone measures and ear simulator measures

Probe microphone measures were taken of the sound pressure level developed in individuals' ear canals for sounds presented by the electrically driven master hearing aid receiver through the individual's ear mould. Sound pressure levels developed in the Zwislocki ear simulator were also measured. The mean differences between the individual probe microphone measures and the standard Zwislocki coupler measures are shown in Table 12.7. The Zwislocki coupler measurements were made with a fixed sound tube length of 45 mm. Subjects' earmoulds had sound tube lengths ranging between 42 mm and 50 mm with a mean of 46 mm. These variations of sound tube length produce significant changes in the Zwislocki coupler measurements

Table 12.7 Differences between individual occluded ear canal probe microphone and Zwislocki ear simulator sound pressure levels

Freq. (Hz)	250	500	750	1 k	1.5 k	2 k	3 k	4 k	6 k	8 k
Mean diff. (dB) (Probe-Zwis.)	-2.67	-1.96	-1.71	-2.83	-3.38	-4.04	-4.46	-5.92	-6.92	1.08
SD	3.54	3.61	3.07	3.75	2.06	2.68	3.70	4.12	6.99	8.18
t	3.54**	1.88	1.93	2.62*	5.68***	5.23***	4.18**	4.97***	3.12**	0.46

only above 4 kHz and of the order of 2 to 3 dB and so cannot explain all the differences found between real ear probe microphone measures and Zwislocki coupler measures.

Differences due to the probe position for the ear canal measurements not being at the eardrum would also only affect the higher frequency measurements and would lead to an underestimation of eardrum SPL. Differences between ear canal probe microphone measures and Zwislocki ear simulator measurements at high frequencies might therefore to some extent be accounted for by variations in tubing length and in measurement position, whereas differences in low frequency measurements require a different explanation.

Since all subjects' earmoulds were shown to give a good acoustic seal, the only apparent explanation seems to be that the Zwislocki ear simulator is slightly underestimating the equivalent occluded ear canal volume of the subjects used in this experiment and so measured a slightly higher sound pressure level than the probe microphone method. For example, an underestimation of equivalent volume by 25% would lead to an increase in sound pressure level of approximately 2 dB. All the earmoulds used in this study had substantial meatal portions and would represent typical well-fitting earmoulds for higher power hearing aids. If earmoulds with smaller meatal portions had been used, the Zwislocki ear simulator would have underestimated subjects' occluded ear canals by an even greater degree. This underestimation of real ear canal occluded volumes by the Zwislocki ear simulator is supported by the recent work of Lawton (1984), who measured ear canal volumes of cadaver ears.

12.4.1.3 Relationship between free field and headphone measures of threshold

Estimates of subjects' free field unoccluded monaural thresholds were made by adding their individual free field to ear canal probe position transfer function to their probe position occluded thresholds. These sound pressure levels were converted to hearing level by subtracting the binaural minimum audible field (BS 3383). A further correction of 3 dB was then made to account for monaural rather than binaural listening.

The mean differences between these interpolated monaural thresholds and subject headphone thresholds are shown in Table 12.8. Free field and headphone thresholds appear in good agreement except at 250 Hz and 500 Hz. The largest spread in results occurs at the highest frequencies.

Table 12.8 Difference between free field and headphone thresholds

Freq. (Hz)	250	500	750	1 k	1.5 k	2 k	3 k	4 k	6 k	8 k
Mean diff. (FF-head-phone)	7.25	5.08	-0.67	-0.75	0.00	-0.67	-3.33	-4.50	-5.49	6.18
SD	3.86	4.38	4.37	4.69	5.27	4.96	5.30	7.45	11.33	10.92
t	6.51***	4.02**	0.53	0.55	0.00	0.47	2.18	2.09	1.68	1.88

12.4.1.4 Comparison between open ear acoustic gain of subjects and Kemar

The mean difference between the subjects' free field to microphone open ear acoustic gains and Kemar's free field to 'eardrum' microphone acoustic gain are shown in Table 12.9. Small but significant differences were found. Inter-subject variations were small, being greatest at the highest frequencies. This might be expected due to variations in individual head and pinna diffraction effects.

12.4.1.5 Differences between subjects' and Kemar's occluded free field acoustic gain

The difference between the occluded free field acoustic gain of the master hearing aid measured by probe microphone in the ear canals of the subjects and by 'ear drum' microphone for Kemar are shown in Table 12.10. Some significant differences were found somewhat similar to those found for the Zwislocki ear simulator measurements shown in Table 12.7. This suggests that it may be the ear simulator and measurement techniques rather than the manikin that are largely responsible for the differences.

12.4.1.6 Subject-Kemar insertion gain difference

Differences between master hearing aid insertion gains measured on subjects and on Kemar are shown in Table 12.11. Only a few significant differences were found, the largest being at the highest frequencies. The greatest inter-subject variation was also at the highest frequencies.

12.4.2 Pair comparison test

12.4.2.1 Sensitivity and reliability

(1) *Sensitivity*

The results of the pair comparison tests are shown in Table 12.12. Generally, the hearing-impaired subjects were less sensitive to differences than the normally hearing subjects used in the preliminary study.

Virtually all subjects noticed and disliked the low frequency roll-off at 400 Hz (Pair 2), whereas approximately equal numbers preferred, disliked or did not notice the roll-off at 250 Hz (Pair 3). For the low pass conditions tested, nobody noticed the loss of frequencies above 6 kHz (Pair 8), only two people noticed the loss of frequencies above 4 kHz (Pair 6) and only two people did not notice the loss of frequencies above 2 kHz (Pair 5). Three of the subjects actually preferred the LP 2 kHz condition, whereas seven disliked it.

Table 12.9 Differences between subjects' and Kemar's open ear acoustic gain

Freq. (Hz)	250	500	750	1 k	1.5	2 k	3 k	4 k	6 k	8 k
Mean diff. (dB) (Sub-Kemar)	1.50	-2.92	-1.25	-3.67	1.50	-1.42	2.33	-0.25	-8.50	-5.42
SD	0.80	1.31	1.76	3.08	2.71	3.03	4.46	4.35	6.23	7.39
t	6.51***	7.70***	24.5*	4.12**	1.91	1.62	1.81	0.20	4.73***	2.54*

Table 12.10 Difference between subjects' and Kemar's occluded free field acoustic gain

Freq. (Hz)	250	500	750	1 k	1.5	2 k	3 k	4 k	6 k
Mean diff. (dB) Sub-Kemar	-0.83	-2.83	-3.08	-1.58	-2.00	-2.83	-5.33	-7.92	-4.42
SD	3.24	3.69	3.99	3.80	4.18	3.21	3.58	4.58	7.05
t	0.89	2.66*	2.67*	1.44	1.66	3.06*	5.17***	5.99***	2.17

Table 12.11 Differences between subjects' and Kemar's insertion gain

Freq. (Hz)	250	500	750	1 k	1.5	2 k	3 k	4 k	6 k
Mean diff. (dB) (Sub-Kemar)	-2.33	1.08	-1.83	2.08	-3.50	-1.42	-7.67	-7.67	4.08
SD	3.77	4.12	3.74	4.81	4.60	5.38	6.27	5.10	8.90
t	2.14	0.91	1.65	1.56	2.64*	1.01	4.24**	5.20***	1.58

Table 12.12 Pair comparison ratings.

	Difference						Red preferred				Green preferred				No pref.
Pair	#4	3	2	1	0	No	3	2	1	Total	3	2	1	Total	
2	5	4	1	1	0	1	4	3	3	10*	1	0	0	1	0
3	1	2	0	3	0	4	0	0	3	3	1	1	1	3*	0
4	0	0	0	0	0	1	0	0	0	0	0	0	0	0	0
5	6	3	1	0	0	2	2	1	0	3	4	2	1	7*	0
6	1	0	1	0	0	6	1	0	0	1	0	0	1	1*	0
7	1	0	0	0	0	0	1	0	0	1	0	0	0	0	0
8	0	0	0	0	0	1	0	0	0	0*	0	0	0	0	0
9	1	0	0	0	0	1	0	0	0	0	1	0	0	1	0
10	1	0	0	0	0	0	0	0	0	0	1	0	0	1	0
11	4	2	0	1	1	1	5	0	1	6*	0	0	2	2	0
12	2	3	2	2	0	0	2	2	3	7*	0	0	1	1	1
13	0	0	0	1	0	1	0	0	0	0	0	0	1	1	0
14	1	3	2	0	0	1	1	0	0	1	2	1	1	4*	1
15	2	4	1	0	1	0	0	1	3	4*	1	3	0	4	0
16	2	0	1	1	0	0	0	0	1	1	1	1	1	3	0
17	1	3	1	1	0	6	2	2	2	6*	0	0	0	0	0
18	1	3	2	2	0	4	1	1	5	7*	0	1	0	1	0
19	0	1	2	0	0	0	1	1	1	3*	0	0	0	0	0
20	0	1	4	1	0	4	0	0	1	1	0	3	1	4*	1
21	0	1	1	0	0	1	0	0	1	1*	1	0	0	1	0
22	3	4	2	0	0	3	2	0	0	2	1	3	2	6*	1
23	1	1	2	0	0	8	0	0	0	0	0	3	1	4*	0
24	3	1	2	1	0	5	2	0	3	5*	0	1	1	2	0
Totals	36	36	25	14	2	50	24	11	27	62	14	19	14	47	4

Total no. of comparisons = 163

LP 8 kHz preferred in 78 comparisons
LP 8 kHz not preferred in 23 comparisons

* denotes LP 8 kHz response

\# Numerical Categories are those indicated on the Pair Comparison instructions shown in Appendix K.1

Table 12.13 Hearing aid processed speech ratings

	Difference						Red preferred				Green preferred				No pref
Pair	4	3	2	1	0	No	3	2	1	Total	3	2	1	Total	
5	0	0	0	2	0	10	0	0	2	2	0	0	0	0	0
6	0	0	1	1	0	8	0	1	1	2	0	0	0	0	0
7	0	0	1	0	0	7	0	0	1	1	0	0	0	0	0
8	1	0	0	0	0	6	1	0	0	1	0	0	0	0	0
9	2	0	1	0	0	3	3	0	0	3	0	0	0	0	0
10	1	0	1	0	0	1	0	0	2	2	0	0	0	0	0
11	1	0	0	0	0	0	1	0	0	1	0	0	0	0	0

Virtually all subjects noticed the effect of changes in response slope (pairs 11 to 16), the majority preferring either the 'flat' LP 8 kHz or -3 dB/octave condition.

Nobody preferred the 10 dB peaks (pair 17), half of the subjects preferring the LP 8 kHz response, the other half not noticing any difference.

The mid-frequency boost and cut responses (Pairs 17 to 20) were noticed and disliked by the majority of subjects although four subjects failed to detect the 10 dB conditions (Pair 17 and Pair 20). The loss of natural ear canal resonance (Pair 22) was also noticed and disliked by three-quarters of the subjects.

Two-thirds of the subjects failed to detect a difference between the Zwislocki ear simulator and the 2 cc coupler responses (Pair 23) whereas over half noticed the difference between their own and Kemar's insertion gain (Pair 24). The majority of these disliked the frequency response as specified for Kemar when compared.

Much larger amounts of distortion were needed by the hearing-impaired subjects than the normally hearing subjects before differences could be noticed (Table 12.13).

(ii) *Reliability*

As with the normally hearing subjects in the pilot study, the hearing-impaired subjects in this experiment proved reliable in their judgements in that nobody expressed differences or preferences for the dummy test conditions where no difference between responses existed.

12.4.2.2 Optimum preferred response

The majority of subjects either preferred the broadest bandwidth responses or did not notice any difference between these and narrower bandwidths. However, one subject preferred the LP 2 kHz condition, another the LP 4 kHz condition and another the LP 6 kHz condition. One subject preferred the HP 400 Hz condition and four subjects preferred the HP 250 Hz condition.

Regarding slope of response, six subjects preferred the 'flat' LP 8 kHz condition, four subjects preferred the -3 dB/octave condition, and one subject preferred each of the +3 dB/octave, +6 dB/octave, -6 dB/octave responses respectively. Therefore over one half of the subjects preferred a response differing from the flat condition, which was the 'SBCL' response. These preferences are shown in Table 12.14.

No interactions were found between slope and cut-off frequency (i.e., subjects who preferred a certain slope for the widest bandwidth condition also preferred that slope for their preferred narrower bandwidth condition).

Table 12.14 Subject preferences relative to SBCL response

Subject No.	Slope (dB/octave)	Cut-off frequency (Hz) Low	High
1	-3	250	4000
2	-3	100	6000
3	+3	400	2000 *
4	+6	400 *	4000 *
5	0	250 *	2000 *
6	-3	100	4000 *
7	-6	250 *	2000
8	0	250	4000 *
9	-3	250	4000 *
10	0	250	4000 *
11	0	100	4000 *
12	0	250 *	6000 *

* denotes no difference noticed between this and the wideband condition (low frequency cut-off = 100 Hz or high frequency cut-off = 8 kHz)

12.4.2.3 Optimum hearing aid fitting procedure

The results of the pair comparison tesets for the four fitting procedures described in Section 12.3.4.3 and Table 12.5 are given in Table 12.15. Table 12.16 shows the number of times each procedure was chosen or not chosen as the preferred aid response in the pair comparison tests. The strength of preference expressed and the number of times no preference or differences were found are also shown.

Table 12.15 Pair comparison ratings (hearing aid selection procedure)

	Differences						Red preferred				Green preferred				No preference
Pair	4	3	2	1	0	No	3	2	1	Total	3	2	1	Total	
1	4	3	2	3	0	1	1	3	3	7	2	0	2	4	0
2	4	3	2	2	0	1	0	1	2	3	4	2	1	7	1
3	2	2	4	1	1	2	3	0	1	4	0	3	2	5	1
4	2	2	2	2	0	4	2	0	0	2	1	1	3	5	1
5	3	1	3	2	1	2	2	1	1	4	2	0	3	5	1
6	4	1	4	1	0	2	1	3	2	6	1	1	1	3	1
Totals	19	11	17	11	2	12	9	8	9	26	10	7	12	29	5

Table 12.16 Preferred aid response

	Preferred				Not preferred					
Response	3	2	1	Total	3	2	1	Total	No Difference	No Preference
Chosen	2	4	6	12	8	5	5	18	4	2
Berger	5	3	6	14	5	2	6	13	7	2
Byrne	6	6	6	18	3	2	3	8	7	3
Fixed	6	2	3	11	3	6	7	16	6	3

From these tables it can be seen that differences between fitting procedures were found and preferences were expressed. The aid response selected by the Byrne and Tonnison procedure was preferred in more comparisons than any other response and not preferred in less. The aid responses of the fixed (LP 8 kHz) conditions and the 'chosen' condition were preferred in the fewest number of comparisons and not preferred in the greatest number. However, when the fixed response was preferred it was found much better in the majority of cases whereas when it was not preferred it was only found slightly or moderately worse. In contrast when the chosen response was preferred it was usually found only slightly or moderately better and when it was not preferred it was often found much worse. The aid response chosen by the Berger prediction method was preferred in approximately the same number of comparisons as it was not preferred.

The rankings of the aid responses based on all the comparisons would therefore appear to be

1. Byrne
2. Berger
3. Fixed
4. Chosen

If the number of times an aid response was found to be the best or equal best of the four responses for individual subjects is tabulated a similar ranking occurs.

However, if individual subjects' comparison results are examined to find how many times any aid response was found better than all of the other three or worse than all of the other three then a slightly different picture occurs (Table 12.17). The 'chosen' aid response was found to be exclusively the best aid by 3 people. A similar number of people found the Byrne response exclusively the best aid. Only one person exclusively

Table 12.17 Best response

Aid Response	Exclusive best	Best or equal best
Chosen	3	3
Berger	2	4
Byrne	3	6
Fixed	1	4

found the fixed aid response better than all other three. It would therefore appear that the chosen response although being preferred in the fewer number of comparisons was found to be better than all three other responses by three people.

Although no hearing aid selection procedure was successful in predicting the preferred aid for all subjects, the Byrne and Tonnison procedure appeared more successful than the other procedures.

12.4.2.4 Relationship between responses and audiometric profiles

The four frequency gain responses used for the hearing aid selection pair comparison test for each subject at their chosen comfort gain setting are shown in Figure 12.6. Also shown are subjects' pure tone thresholds and 1/3 octave band speech comfort levels. All measurements are shown relative to the 1/3 octave speech band levels delivered to the individual through the fixed frequency response. To interpret the frequency responses in terms of absolute frequency gain characteristics on Kemar one must add the frequency response of the fixed response shown in Figure 12.7. To interpret the data in terms of absolute speech band levels in the ear canal of individual subjects one must additionally add the speech spectrum shown in Figure 12.8 and correct for Kemar minus individual frequency gain differences. An example of this procedure is shown in Figure 12.9. All the data have not been presented in this way as it lends little to the interpretation of the results and makes deviations from the fixed response more difficult to observe. In addition, the frequency gain responses shown in Figure 12.6 are direct representations of the digital filter responses used in the master hearing aid, as the fixed response corresponded to the LP 8 kHz setting of the filter.

The frequency gain responses of the sloping conditions of the pair comparison test for individual subjects at their chosen gain setting are shown in Figure 12.10. Also shown are the subjects' pure tone thresholds and 1/3 octave speech band comfort levels. All measurements are shown relative to the individual's SBCL response. The actual SBCL responses are shown in Figure 12.11 relative to the LP 8 kHz responses. To interpret the frequency responses of Figure 12.10 in terms of absolute frequency gain characteristics one must add the individuals' SBCL response and gain setting shown in Figure 12.11 and the LP 8 kHz response of the master aid shown in Figure 12.7. All the data have not been presented in this way as it lends little to the interpretation of the results and makes deviations from the SBCL response difficult to observe. In addition the frequency gain responses shown in Figure 12.10 are direct representations of the digital filter A responses used in the pair comparison tests (Section 12.3.4.2) whereas the SBCL responses shown in Figure 12.11 correspond to the settings of digital filter B.

12.4.2.5 Effect of frequency response changes on comfort level settings

The changes in comfort settings relative to the LP 8 kHz condition for the low pass, high pass, peaks, mid-frequency boost and mid-frequency cut responses are shown in Table 12.18. The changes in L_{eq} level of the responses are also shown. It might be expected that since these responses were all presented relative to the SBCL response, loudness relationships would be similar to those found for normally hearing subjects (Section 11.3.6, Table 11.24). This does not seem to be the case. High frequencies appear to have a greater effect and low frequencies a lesser effect than for normally hearing subjects.

The frequencies at which comfort levels were equal for the sloping conditions are shown in Table 11.19. There is a wider spread of results than for the normally hearing subjects. Also, responses seem to equate levels at higher frequencies, but the correspondence with L_{eq} measures is also not so good. Individual results can be seen in the plots of

Figures 12.10. The effects of variations in comfort settings, while not identical to normally hearing subjects, still appear to be predictable within 5 to 10 dB.

12.4.2.6 Differences between pure tone and speech band comfort levels

Measurements of LCL were made for both pure tone and 1/3 octave speech band levels. Although pure tone measurements were made using headphones, and 1/3 octave speech measurements using the master hearing aid, the two could be directly compared as pure tone threshold measurements were made with both headphone and master hearing aid. Table 12.20 shows the mean differences between the speech band and pure tone LCL relative to threshold. Measures of LCL were not possible for a few subjects at a few frequencies due to output limitations of the equipment.

No significant differences were found.

Table 12.18 Comfort level differences

Pair	A	B	N	Mean A-B (dB)	SD	t	95 confidence limits	L_{eq} A-B
2	HP 400 Hz	LP 8 kHz	12	1.83	2.76	2.30*	± 1.43	-3.5
3	HP 250 Hz	LP 8 kHz	10	0.80	1.69	1.50		-2.2
5	LP 2 kHz	LP 8 kHz	12	4.17	4.00	3.61***	± 2.07	-0.2
6	LP 4 kHz	LP 8 kHz	8	1.75	1.83	2.70*	± 1.23	-0.1
17	10 dB peaks	LP 8 kHz	12	-3.75	2.77	4.69***	± 1.44	+1.1
18	10 dB mid boost	LP 8 kHz	10	-3.70	2.16	5.42***	± 1.54	+1.0
20	10 dB mid cut	LP 8 kHz	7	3.29	2.43	3.58**	± 1.78	-0.4

Table 12.19 Frequencies at which comfort levels equate

Pair	A	B dB/octave	N	SD dB	Equiv freq Hz	95% conf. limits Hz	Leq equiv. freq.
11	LP 8 kHz	+ 6	9	3.11	770	590–1020	1200
12	"	+ 3	9	3.75	1140	590–2210	740
14	LP 8 kHz	– 6	7	3.27	900	630–1270	340
15	"	– 3	8	3.52	620	310–1220	380

Table 12.20 Differences in LCL between pure tone and 1/3 octave speech band measures

Frequency (Hz)	N	Mean 1/3 octave – pure tone	SD	t
500	9	-3.7	10.68	1.04
1000	9	-0.78	16.66	0.14
2000	8	3.75	7.65	1.39
4000	7	-2.43	10.23	0.63

12.4.2.7 Relationship between ULL, LCL and MCL

The differences between ULL and LCL and the differences between LCL and MCL for the boadband and narrow band speech stimuli are shown in Table 12.21. The ULL is some 9 to 10 dB above LCL for both broad band and narrow band stimuli. The LCL-MCL difference is slightly higher for broad band stimuli than for the narrow band stimuli as was found in the normal study. These differences indicate the small range over which the hearing-impaired subjects can comfortably listen and therefore points to the importance of the correct gain setting for satisfactory aid use. One could also note here the possible contribution some form of compression amplification might bring to ensuring that the perceived speech level varied little whatever the actual speech input level.

Individual subjects ULL, LCL, and MCLs are shown in Figure 12.6.

Table 12.21 Relationship between ULL, LCL and MCL.

Difference	Stimuli	n	Mean diff. (dB)	SD	t	95% confidence limits
ULL - LCL	Broadband	11	9.36	6.10	6.10***	± 3.33
ULL - LCL	500 Hz	10	9.80	5.57	5.56***	± 3.23
ULL - LCL	2 kHz	10	9.20	5.49	5.30***	± 3.18
LCL - MCL	Broadband	12	6.75	5.17	4.52***	± 2.68
LCL - MCL	500 Hz	10	3.70	4.40	2.27*	± 2.55
LCL - MCL	2 kHz	10	4.00	3.65	3.46**	± 2.12

12.4.2.8 Reliability of comfort settings

The standard deviations of ten or more comfort settings for each individual for the LP 8 kHz response during the pair comparison test ranged between 0.55 dB and 3.20 dB for the 12 subjects. This suggests a good test retest reliability as all subjects had 95% confidence limits of less than ± 2.5 dB about their mean comfort settings.

The comfort settings for the pair comparison tests were, on average, some 4 dB higher than the MCL settings for the MCL determination experiment. This suggests that subjects were listening at levels approaching LCL for the more demanding task of the pair comparison test.

12.4.2.9 Relationship between narrow band and broad band comfort levels

The mean difference between the 1/3 octave band levels of the broad band and narrow band comfort settings are shown in Table 12.22. The broadband comfort levels are for the fixed LP 8 kHz master aid response. The differences tend to decrease as the stimulus level increases from MCL to UCL suggesting that loudness summation decreases with stimulus level.

Table 12.22 Difference between broad band and narrow band comfort levels

Test	Frequency band, Hz	n	Mean diff.	SD	t	95% confidence limits
MCL	2000	10	20.2	8.01	7.97***	±5.64
LCL	2000	10	17.0	7.10	7.57***	±5.08
UCL	2000	10	15.0	5.37	6.47***	±3.84
MCL	500	10	14.0	5.97	7.41***	±4.27
LCL	500	10	10.7	5.90	5.73***	±4.22
UCL	500	10	10.0	5.37	5.88***	±3.84

These results are very similar to those for the normally hearing subjects in their study and suggest that loudness summation effects are similar for the two groups at these suprathreshold levels. For the broad band stimuli presented through the SBCL response the mean difference between its 1/3 octave band levels at MCL and the narrow band comfort levels was 18.25 dB, very similar to that for the fixed LP 8 kHz response. The individual subjects' broad band and narrow band comfort levels are shown in Figures 12.6 and 12.10. The difference of 14 to 20 dB between 1/3 octave band levels for broad band speech at MCL and narrow band speech at MCL has important implications for hearing aid fitting procedures.

The widely accepted method of trying to present the speech spectrum such that each 1/3 octave band is at the MCL for that band cannot successfully accomplish its aims. Subjects will turn down the gain of the aid by some 20 dB until the broad band speech signal is comfortable to listen to. This effect is clearly apparent for all subjects in Figures 12.6 and 12.10.

The loudness relationship between the low and high frequencies may vary with level depending on the subjects' equal loudness contours and loudness growth relationships. Therefore the effects of listening at up to 20 dB below the 'prescribed' level will vary depending on audiometric configurations.

These effects explain why many subjects with apparently usable hearing up to 8 kHz failed to notice differences in the higher frequencies in the pair comparison tests. As can be seen from Figures 12.6 and 12.10, very few subjects were receiving the higher frequency elements of speech above their threshold of hearing.

Also for subjects with a small but apparently usable range of hearing of some 20-30 dB, the chosen comfort setting for broad band speech can result in very little speech being above threshold.

12.4.2.10 Relationship between predicted and actual user gain

The hearing aid prescription procedures of Berger and Byrne specify the required aided thresholds the subjects should achieve. This is calculated from the predicted gain the subject will choose to use based on a modification of the (1/2 × hearing loss) principle. Table 12.23 presents the differences between the predicted and actual gains used by the subjects for the aid responses specified by the Byrne and Berger prediction methods. It can be seen that Byrne's method predicts the chosen gain setting well on average, whereas Berger's method specifies on average 12.67 dB higher gain than is actually used. It is important to realise that these gain settings are very dependent on the input speech level. If louder speech was used, lower gain settings would be chosen and likewise higher settings for quieter speech. However, the speech level used for this experiment (65 dB(A)) is representative of real-life levels and similar to that used by Byrne and Berger in the formulation of their prescribed settings.

Table 12.23 Difference between predicted and actual user gain

Prediction method	Mean difference actual-predicted	SD	t
Berger	-12.67	8.34	5.26***
Byrne	-1.50	9.93	0.52

It is also important to realise that although Byrne's prediction method predicted the chosen gain on average, there was a large variation between subjects. For example, one subject used 16 dB less gain than predicted and another 14 dB more gain than predicted. From the frequency response plots of Figure 12.6 it can be seen that there is no simple universal relationship between chosen gain settings, frequency response thresholds and narrow band comfort levels. However, some significant correlation coefficients for linear regression can be found. For example, Table 12.24 shows the relationships between insertion gain measures and hearing loss at different frequencies for the 'fixed' LP 8 kHz response. The strongest relationship is for the lowest frquencies of 250 and 500 Hz.

12.4.3 AVELS test results

12.4.3.1 Introduction

The aid responses used for the AVELS test are shown in Table 12.25. For two subjects (Nos. 1 and 8) the best and worst responses from the hearing aid selection procedures were not used because no differences had been found. Instead of this, two responses that had shown differences for these subjects in the pair comparison test were used. Half the subjects

Table 12.24 Relationship between insertion gain and hearing loss

Frequency insertion-gain, Hz	Frequency hearing loss, Hz	Correlation	Slope	Intercept
250	250	0.820**	0.37	-2.62
500	500	0.776**	0.56	-9.02
750	750	0.446	0.31	12.05
1000	1000	0.390	0.20	20.34
2000	2000	0.317	0.23	21.06
4000	4000	0.150	0.08	15.40
Av. 500-4000	Av. 500-4000	0.251	0.22	17.88
Av. 250-2000	Av. 250-2000	0.480	0.43	4.76

Table 12.25 Aid responses used for AVELS test

Subject no.	1st presentation	2nd presentation
1	+6 dB/octave	LP 8 kHz *
2	Berger	LP 8 kHz *
3	LP 8 kHz	Byrne *
4	Chosen	LP 8 kHz *
5	Chosen *	Berger
6	Berger *	LP 8 kHz
7	Byrne *	Chosen
8	HP 400 Hz	LP 8 kHz *
9	Chosen *	LP 8 kHz
10	Berger *	LP 8 kHz
11	LP 8 kHz	Byrne *
12	Berger *	Chosen

* denotes preferred aid in pair comparison test

had their 'preferred response' presented first and the other half second. The results of the FADAST and AVELS tests are shown in Table 12.26. Subject 4 did not finish the final presentation due to becoming distressed at the low score. Subject 10 did not finish the final presentation due to the lack of time available and also listened binaurally with her good ear unoccluded to obtain reasonable scores. Whole word scoring will be used throughout for analysis of results for reasons of simplicity and because the pilot study showed up no advantages of phoneme scoring.

Table 12.26 Audiovisual test scores

Subject no.	FADAST test, %	AVELS test presentation, % 1st	2nd	3rd
1	76	56	52	57
2	79	58	66	72
3	95	62	67	69
4	73	15	21	
5	86	68	72	75
6	89	32	31	32
7	84	63	56	67
8	92	53	51	59
9	75	54	60	66
10	91	78	81	
11	73	39	37	42
12	82	70	61	70

12.4.3.2 Learning effects

An analysis of the average learning effects for this experiment is given in Table 12.27. An overall mean improvement of 5.4% was found between the first and third presentations. No significant improvement was found between the first and second presentations. The mean improvement of 3.10% between the best scores in either of the first or second presentation and the scores in the third presentation indicates a practice effect of this order over the three presentations. However, no separate analysis of practice and memory/order effects is possible due to the lack of significant differences between the 'best' and 'worst' PC aids. For the normal study the 'best' and 'worst' conditions were satisfactorily controlled. The mean overall learning effect is similar to that found for that study for the 'difficult, easy, difficult' order of presentation, even though no great differences between aids were found.

Table 12.27 AVELS test learning effects (%)

Difference	n	mean	SD	t	95% confidence limits
2nd presentation - 1st presentation	12	0.59	5.53	0.37	
3rd presentation - 1st presentation	10	5.40	4.81	3.54***	±2.79
3rd presentation - best of 1st or 2nd	10	3.10	2.38	4.12***	±1.38
best PC - worst PC	12	1.25	5.41	0.80	

One might therefore speculate that had greater differences between aids existed a slightly higher learning effect may have occurred. As was pointed out in the conclusions to the normal study, the repetition of the AVELS test three times gives a slightly unrealistic overestimation of the learning effects for only two presentations of the test. For the comparison of two hearing aids only two presentations need be used. Taking this into consideration, the normal study estimation of a 4% maximum mean learning effect for two presentations of the test would appear to also apply to the hearing-impaired subjects in this experiment.

12.4.3.3 Test form equivalence

No significant differences between scores on the first and second 36 items of the AVELS test were found (Table 12.28). Learning effects for the two test forms were similar to those for the test as a whole (Table 12.29). As found in the normal study, a slightly greater learning effect was found for the first 36 items. The scores on the 1st test form correlated highly with those for the complete test (Table 12.30), thus indicating that scores on a shortened test could be representative of those on the whole test.

Table 12.28 Differences between 1st and 2nd 36 items, % scores

Presentation	n	mean	SD	t
1st	12	-3.00	6.53	1.59
2nd	12	0.46	3.53	0.46
3rd	10	0.00	7.86	0.00
All	34	0.89	6.14	0.85

Table 12.29 AVELS % leraning effects for 1st and 2nd 36 items

1st 36 items

Difference	n	mean	S.D.	t	95% confidence limits
2nd presentation - 1st presentation	12	1.46	3.79	1.33	
3rd presentation - 1st presentation	10	4.60	5.04	2.88*	± 2.92
3rd presentation - best of 1st and 2nd	10	2.70	3.43	2.49*	± 1.99
Best P.C. - Worst P.C.	12	1.45	3.79	1.33	

2nd 36 items

Difference	n	mean	S.D.	t	95% confidence limits
2nd presentation - 1st presentation	12	-0.33	4.30	0.43	
3rd presentation - 1st presentation	10	2.69	3.97	2.15*	± 2.30
3rd presentation - best of 1st and 2nd	10	1.59	2.90	1.73	
Best P.C. - Worst P.C.	12	1.86	3.87	1.65	

Table 12.30 Relationship between scores on 1st 36 items and whole test

Conditions	n	correlation coefficient
All presentations	32	0.965***

Table 12.31 Comparison of % scores on AVELS test sections

Section	AVQ	AVN	AQ	AN
Mean	75.52	43.23	57.81	15.63
SD	15.93	18.55	33.93	16.96
Maximum score	93.75	68.75	100.00	50.00
Minimum score	43.75	12.50	6.25	0.00

The two test forms would therefore appear to give equivalent scores for both normally hearing and hearing-impaired subjects over a wide range of conditions and scores.

12.4.3.4 AVELS test sections

The range of scores on the four 'sections' of the AVELS test (AVQ, AVN, AQ and AN) are shown in Table 12.31. A comparison with the results on normally hearing subjects shows the dramatic effect that the loss of visual cues has for the hearing-impaired subjects, especially in the noisy condition. The scores for all sections of the test are reduced but especially the audio only condition. A much wider range of scores was obtained for the hearing-impaired subjects, indicating both the diversity of their receptive communication abilities and the feasibility of the AVELS test to measure them.

12.4.3.5 Comparison of AVELS and FADAST test scores

The AVELS test scores covered a wider range than the FADAST scores, suggesting that it is a more sensitive measure of subjects' abilities (Table 12.32). Correlations between the FADAST and AVELS test were not significant, indicating that somewhat different abilities were possibly being tested. A comparison with results of the normal study shows that for the AVELS test no hearing-impaired subject scored higher than any normally hearing subject. For the FADAST test, however, there was considerable

Table 12.32 Comparison of % scores on AVELS and FADAST tests

Test	AVELS	FADAST
	Range of Scores	
Mean	59.25	82.91
SD	18.36	7.77
Maximum	81	92
Minimum	21	73

Correlation between tests

AVELS test	FADAST	correlation coefficient
Whole test	FADAST	0.404
AVN section	FADAST	0.220
AN section	FADAST	0.433

overlap. The AVELS test therefore appears to distinguish more clearly the communication limitations of the hearing-impaired.

12.4.3.6 Real-life intelligibility ratings

The results of the questionnaire on real-life receptive communication skills (Appendix N) are given in Table 12.33. No significant correlations were found for a linear regression of the overall 'understanding' rating and the complete audiovisual test scores. Attempts to average the individual ratings (QV, NV, QA and NA) only slightly increased the non-significant correlation coefficient (Table 12.34).

The information regarding the frequency of occurrence of the differing audio and audiovisual conditions in the subjects' real lives was included to determine whether weightings of the intelligibility ratings for the individual conditions could predict the overall ratings. No simple weighting procedure was, however, found to give good predictions, indicating that any existing relationship would appear to be fairly complex.

Significant correlations were, however, found for a linear regression of the scores on the separate test conditions and the real-life ratings of the subjects' receptive communication abilities in these types of conditions (Table 12.35).

Table 12.33 Subjects' questionnaire results

Sub-ject no.	Occur				Avoid				Understand				
	QV	NV	QA	NA	QV	NV	QA	NA	QV	NV	QA	NA	All
1	5	4	2	2	0	3	1	5	7	2	5	0	5
2	5	5	5	5	5	5	5	5	8	4	4	1	2
3	3	5	1	1	3	3	5	4	7	7	3	5	6
4	4	2	5	2	0	0	5	5	5	4	1	0	4
5	3	3	3	3	0	3	0	2	9	7	8	5	6
6	5	5	2	0	0	3	5	5	9	5	1	0	4
7	4	3	1	1	0	3	2	4	7	4	5	2	6
8	3	2	1	1	0	5	5	5	7	5	3	0	7
9	4	0	1	5	0	2	5	5	7	6	3	1	5
10	5	3	4	1	5	2	4	1	9	6	6	1	7
11	5	3	5	0	0	3	3	5	9	5	5	0	5
12	4	2	3	5	4	5	3	5	5	2	4	0	7

Table 12.34 Relationship between overall test scores and intelligibility ratings

Test	Rating	Correlation
AVELS	Overall	0.413
AVELS	Averaged	0.555
FADAST	Overall	0.502
FADAST	NV	0.471

Table 12.35 Linear regression of scored and rated corresponding conditions

Test section	correlation coefficient	slope	intercept
Audio only (AQ and AN)	0.759**	11.59	8.90
Audiovisual (AVQ and AVN)	0.614*	8.14	18.56
All (AQ, AN, AVQ and AVN)	0.720**	7.17	16.32

The individuals' scores and ratings are shown in Figure 12.12. If these scores are compared with the range of scores obtained in the baseline study (Section 11..2.4.5), it is interesting to note that very few of the hearing-impaired subjects scored as well as the worst normally hearing subject in any test condition.

The reasonable correlations between speech scores and real-life ratings are even more impressive when it is considered that the speech tests were conducted monaurally with the non-aided ear plugged and using experimental aid responses. In real life, five subjects wore binaural aids, and three subjects (3, 6, 11) generally wore their aid in the other ear to that used for the experiment. It was not feasible to present the AVELS test a fourth time using the aids used in real life to overcome these differences.

12.4.3.7 Relationship between test scores and audiometric thresholds

The AVELS test scores were significantly correlated with subjects' thresholds (Table 12.36). Better correlations were obtained with bone conduction thresholds than air conduction thresholds. No significant correlations were found for the FADAST test scores.

Table 12.36 Relationship between audiometric thresholds and test scores

Audiometric threshold measurement	Test	Correlation	Slope	Intercept
Air conduction average (500, 1, 2, 4 k)	AVELS	0.591*	-0.60	98.48
"	FADAST	0.249	-0.13	91.31
Bone conduction average (500, 1, 2, 4 k)	AVELS	0.728*	-0.59	92.79
"	FADAST	0.326	-0.11	89.28

12.4.3.8 Subjective judgements on AVELS tests

Only three subjects claimed to notice a difference between the aid responses used for the AVELS test. Two subjects (nos. 3 and 6) preferred the aid that was worst in the pair comparison test, while only one subject (no. 10) preferred the aid that was best in the pair comparison. All three subjects preferred the aid that scored highest in the AVELS test.

All twelve subjects felt that the AVELS test presented very realistic listening situations. Nine of the subjects thought that the situations were of similar difficulty to those experienced in their everyday life. The remaining three subjects thought that the AVELS test situations were more difficult than their real life experiences and each gave a different reason for this opinion. One stated that in real life the subject matter was usually known, whereas in the AVELS test the sentences were spoken out of context. Another claimed that in real life he often asked people to repeat what they had said. The third explained that in real life she avoided any situation with background noise.

12.4.3.9 Environmental sounds

All subjects expressed the opinion that the environmental sounds were very realistic.

12.5 Summary of Conclusions of Main Study

12.5.1 Measurement methods

(i) Below 4 kHz, headphone, freefield and probe tube measures of threshold gave comparable results. Mean differences were generally less than 5 dB. These mean differences and individual variations increased for higher frequencies, presumably due to head and pinna diffraction effects.

(ii) Mean sound pressure levels developed in real ears were within 5 dB of Zwislocki ear simulator measures up to 4 kHz. Slightly larger differences and individual variations occurred at higher frequencies. Differences found would appear to be partly due to individual variations and probe positioning. Differences would also appear to be due to a discrepancy between the occluded ear canal volumes of the subjects used in this study and that represented by the Zwislocki ear simulator. A rough estimate would suggest that the simulator may have up to 25% less volume than real occluded ears and thus gave sound levels up to 2 dB higher than in real ears. Lawton's (1984) recent measurements on cadaver ears would support this contention.

(iii) Insertion gain measurements for individual subjects were in good agreement with measurements on Kemar, the greatest discrepancies and inter-subject variations being at the highest frequencies.

12.5.2 Pair comparison test

(i) Sensitivity

The hearing-impaired subjects were less sensitive to differences between pairs of responses than the normally hearing subjects. Response changes that resulted in low frequency effects (such as low frequency cut-off and slope of response variation) tended to be noticed by most people, whereas higher frequency effects such as high frequency cut-offs, peak clipping and coupler differences, were not so easily noticed. Very few subjects noticed the effect of band width above 4 kHz. Peaks in the response were noticed by only half the subjects, who all disliked them. Mid-frequency cuts and boosts were noticed by the majority of subjects, as was the difference between their own and Kemar's insertion gain and the loss of 'natural ear canal resonance'.

(ii) Reliability

The hearing-impaired subjects made reliable judgements.

(iii) Chosen response

Over one half of the subjects chose a response differing slightly from the SBCL response. The most frequent modifications being a low frequency cut-off of 250 Hz and/or a reduction in slope of -3 dB/octave. Frequencies above 4 kHz were rarely detected by subjects.

(iv) Hearing aid selection procedures

No hearing aid selection procedure was successful in predicting the preferred aid for all subjects. However, the Byrne and Tonnison procedure was the most successful and the chosen aid (modified SBCL) the least successful.

12.5.3 Comfort levels

(i) The effect of frequency response changes on comfort settings, while still being roughly predictable, appears to differ from that for normally hearing subjects.

(ii) No significant differences between pure tone and speech band LCL were found.

(iii) Hearing-impaired subjects had approximately half the dynamic range of the normally hearing subjects between MCL and LCL. Approximately 10 dB of 'headroom' above LCL existed before discomfort (ULL) occurred.

(iv) Subjects exhibited good reliability in their comfort settings.

(v) Subjects tended to listen at levels approaching LCL rather than at MCL for the more demanding task of the pair comparison test.

(vi) Differences between broad band and narrow band comfort levels were similar to those found for normally hearing subjects. Loudness summation effects decreased slightly as stimulus intensity increased. The difference of 14 dB to 20 dB between band levels for broad band speech at MCL and narrow band speech at MCL, means that hearing aid fitting procedures that try to present the speech spectrum such that each band is at the MCL level for that band cannot be successful. Subjects will turn the gain of the aid down by some 20 dB until the broad band speech signal is comfortable to listen to. This means that some subjects with an appaentIy usable range of hearing between threshold and discomfort (e.g., 30 dB) may in fact receive little benefit from amplification.

For other subjects, the relative balance between high and low frequencies may be altered, if equal loudness contours are not parallel. These effects are shown clearly in Figures 12.6 and 12.10 and explain why many subjects with residual hearing up to 8 kHz failed to notice the difference between responses differing in the higher frequencies in the pair comparison tests. Very few subjects received any speech signal above threshold for frequencies at 4 kHz or above.

(vii) No simple relationship was found that accurately related chosen gain settings, frequency responses, and audiometric profiles. However, some significant correlations at low frequencies were found.

Byrne and Tonnison's hearing aid prescription procedure predicted chosen gain settings reasonably well on average, whereas Berger's procedure specified gains on average approximately 13 dB higher than were actually used. There were, however, large individual variations in gain settings for both procedures.

12.5.4 AVELS test

(i) Similar learning effects were found for hearing-impaired subjects as were found for the nomally hearing subjects.

(ii) No significant differences between shortened test forms was found. The shortened test forms gave similar results to the complete test.

(iii) The AVELS test appeared to be more sensitive than the FADAST test and distinguished more clearly between hearing-impaired and normally hearing subjects. The AVELS test scores do not correlate well with the FADAST test scores, indicating that a different ability is being tested, probably involving prosodic and linguistic features.

(iv) The hearing-impaired subjects were affected to a greater extent by the loss of visual cues and noise than the normally hearing subjects, as shown by the scores on the various sections of the AVELS test.

(v) A close relationship was found between the scores on the sections of the AVELS test and the subjects intelligibility ratings for the corresponding real-life situations.

(vi) No simple relationship was found between the subjects' intelligibility ratings for the individual real-life situations and their overall intelligibility ratings.

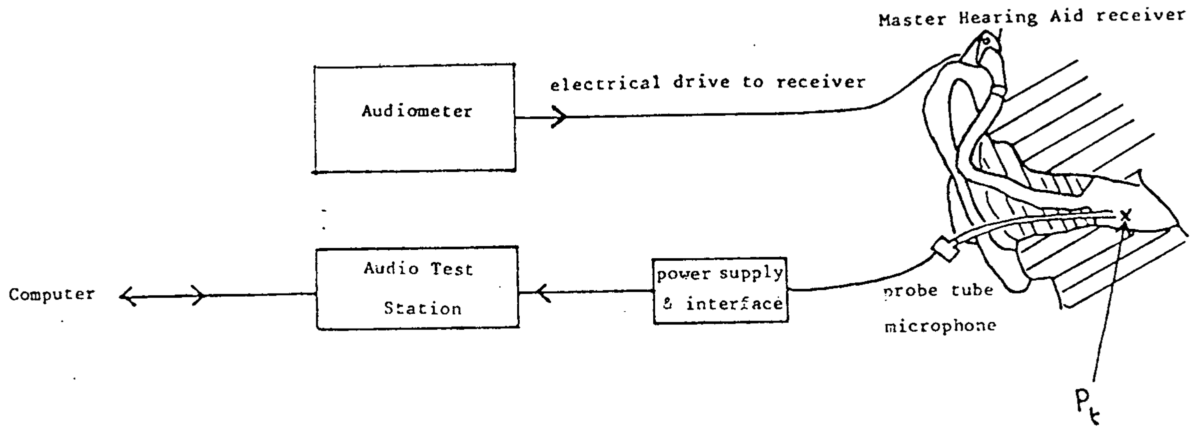
(vii) A significant relationship was found between AVELS test scores and audiometric threshold measurements.

(viii) Whereas preferences were expressed by all subjects in the pair comparison tests for the pairs of responses used in the AVELS test, only three of the subjects also expressed a preference after the AVELS test. These preferences were always for the higher scoring aid. Two of the subjects preferred the aid that was not preferred by them in the pair comparison test.

This would indicate that the subjective AVELS test is less sensitive than the pair comparison test and can also lead to different preferences.

Figure 12.1

Pure tone threshold measurement technique

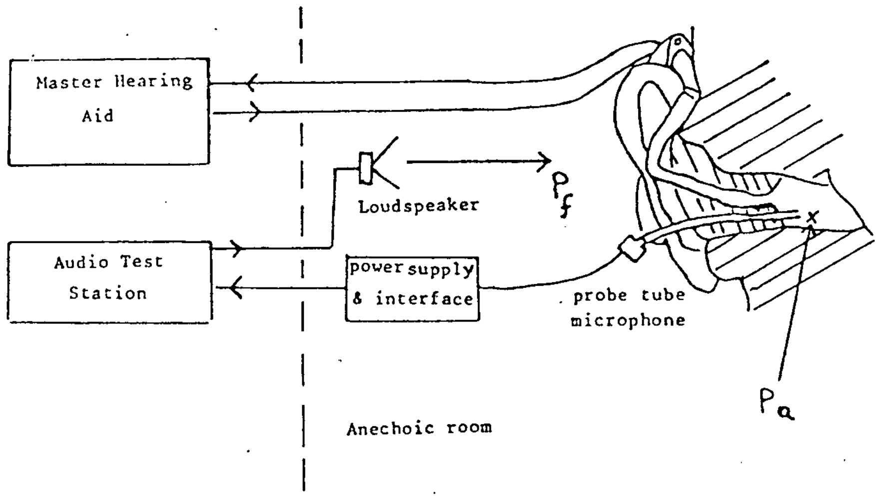


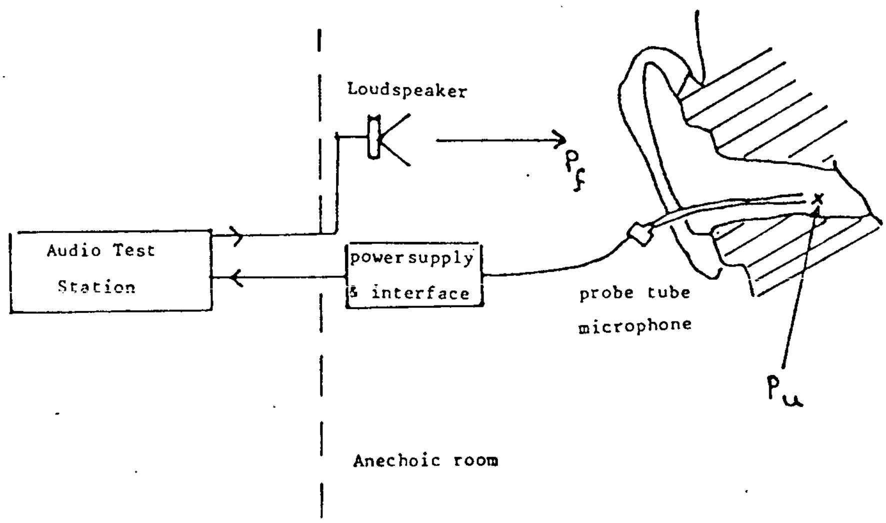
behavioural threshold is P_t

at probe microphone position in ear canal

Figure 12.2

Insertion gain measurement technique





Insertion gain = $P_a - P_u$

P represents SPL,dB

Figure 12.3

Digital Filter A & B

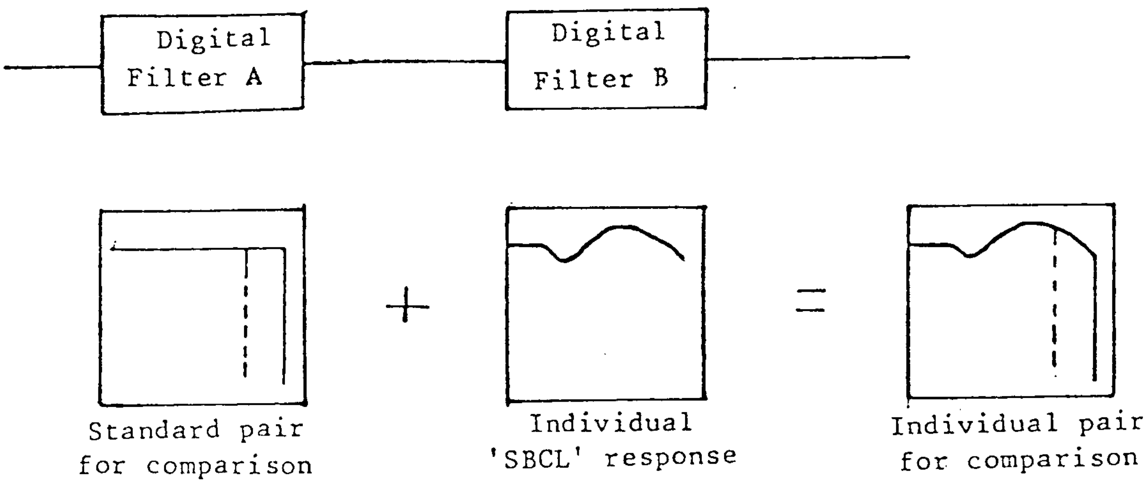


Figure 12.4

Pair Comparison Digital Filter Responses

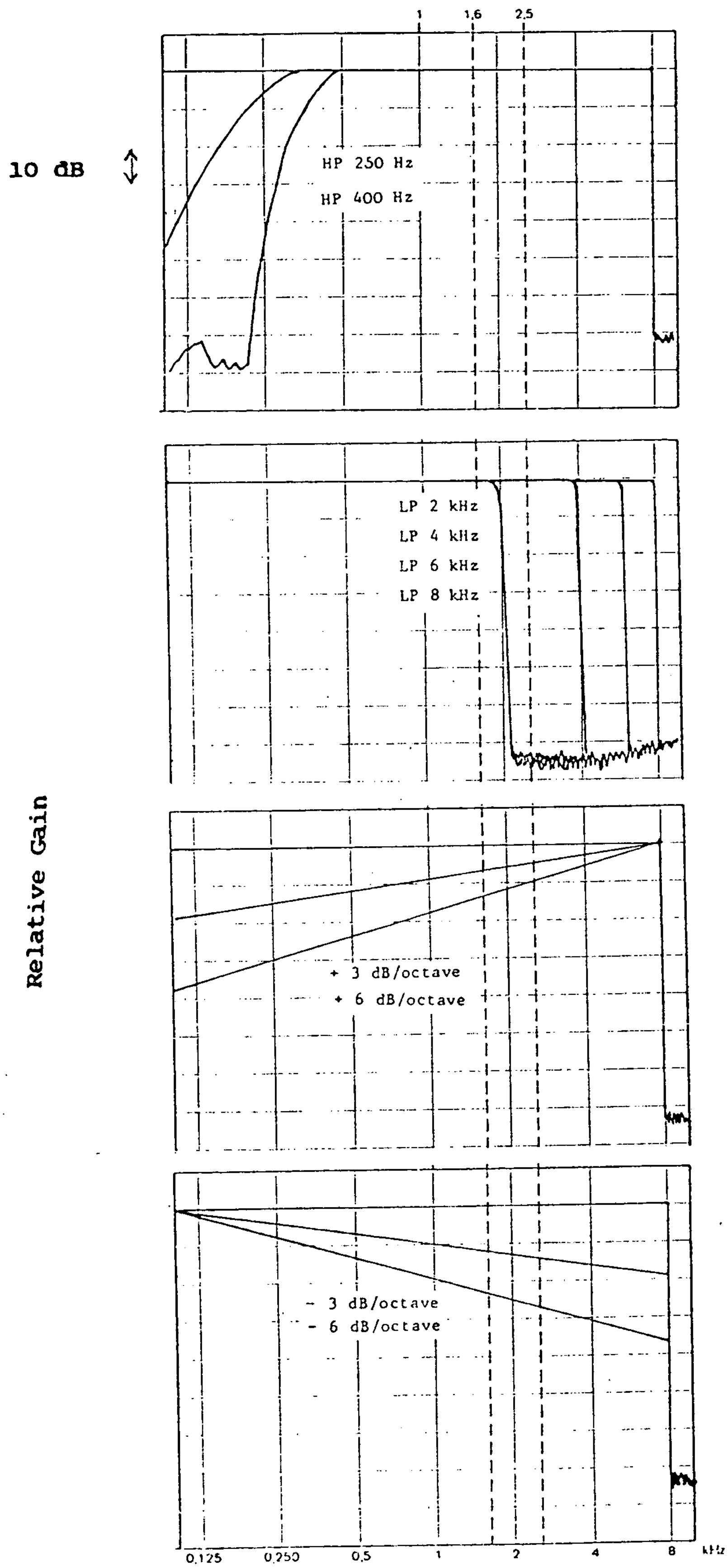


FIGURE 12.4 cont.

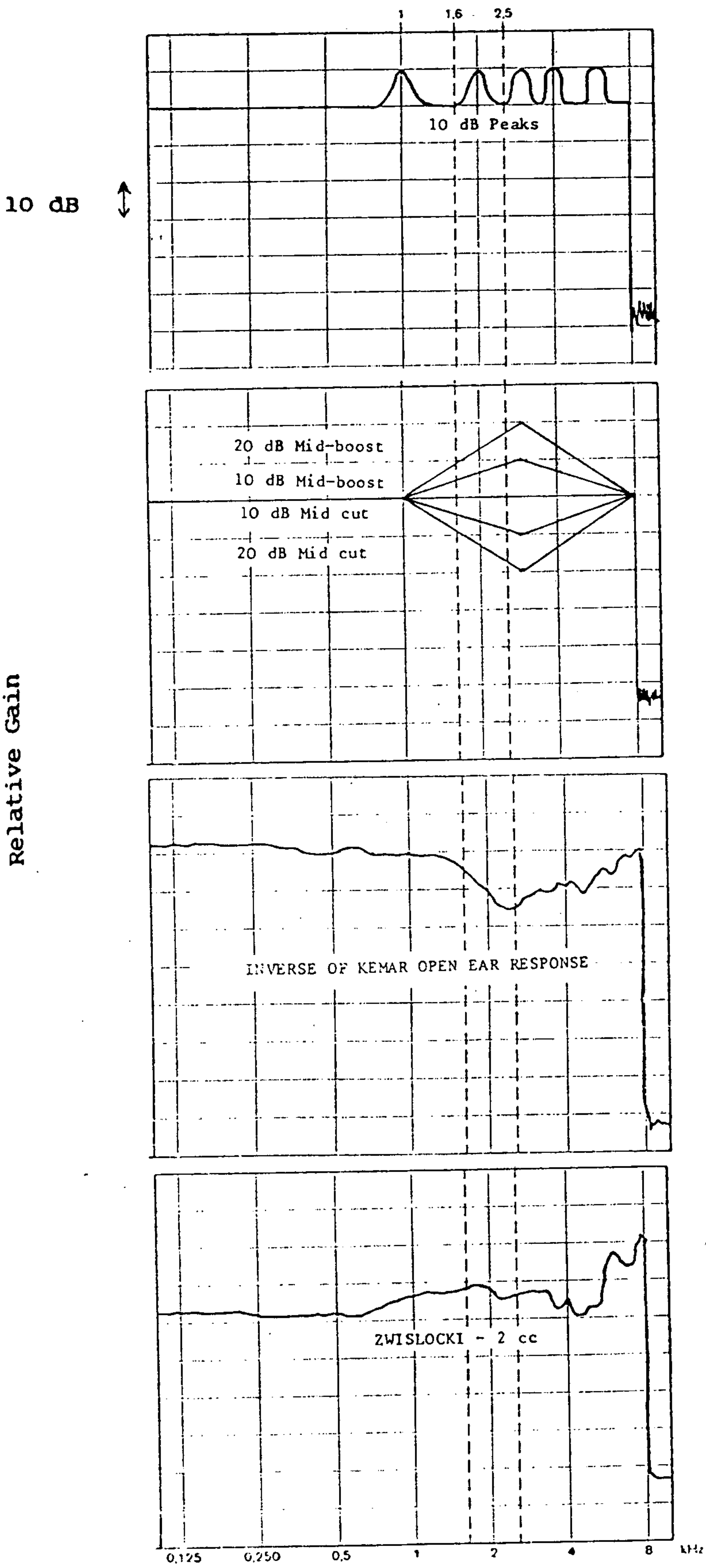


FIGURE 12.4 cont.

Kemar insertion gain minus subject's insertion gain

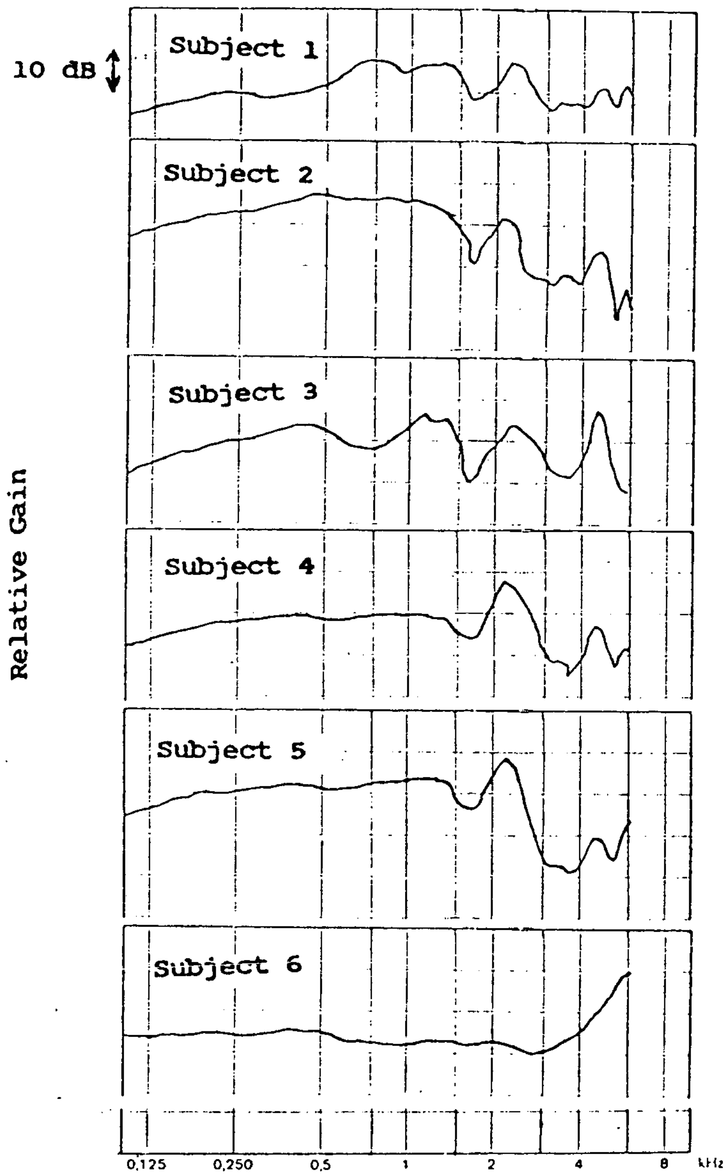


FIGURE 12.4 cont.

Kemar insertion gain minus subject's insertion gain

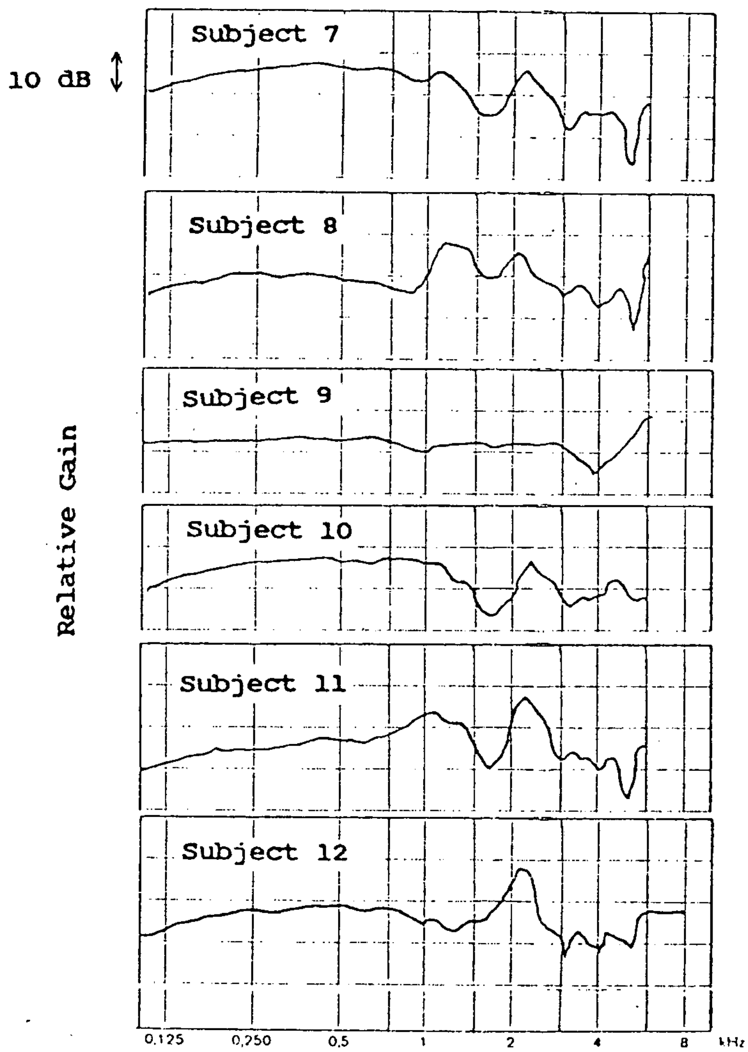


Figure 12.5

Pair Comparison Protocol

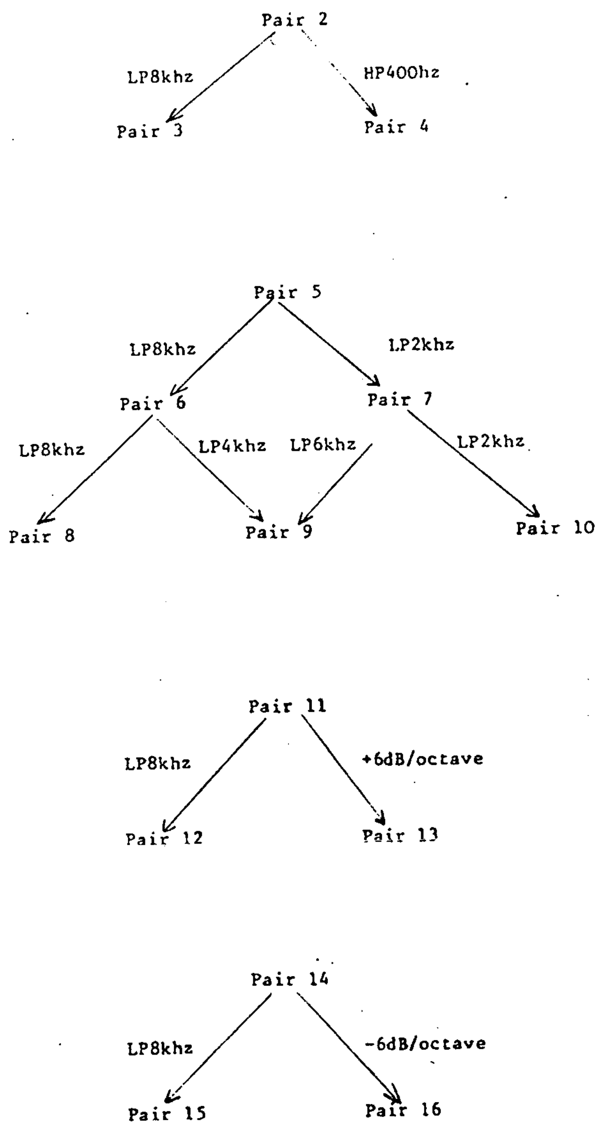


Figure 12.6

Individual hearing aid selection responses

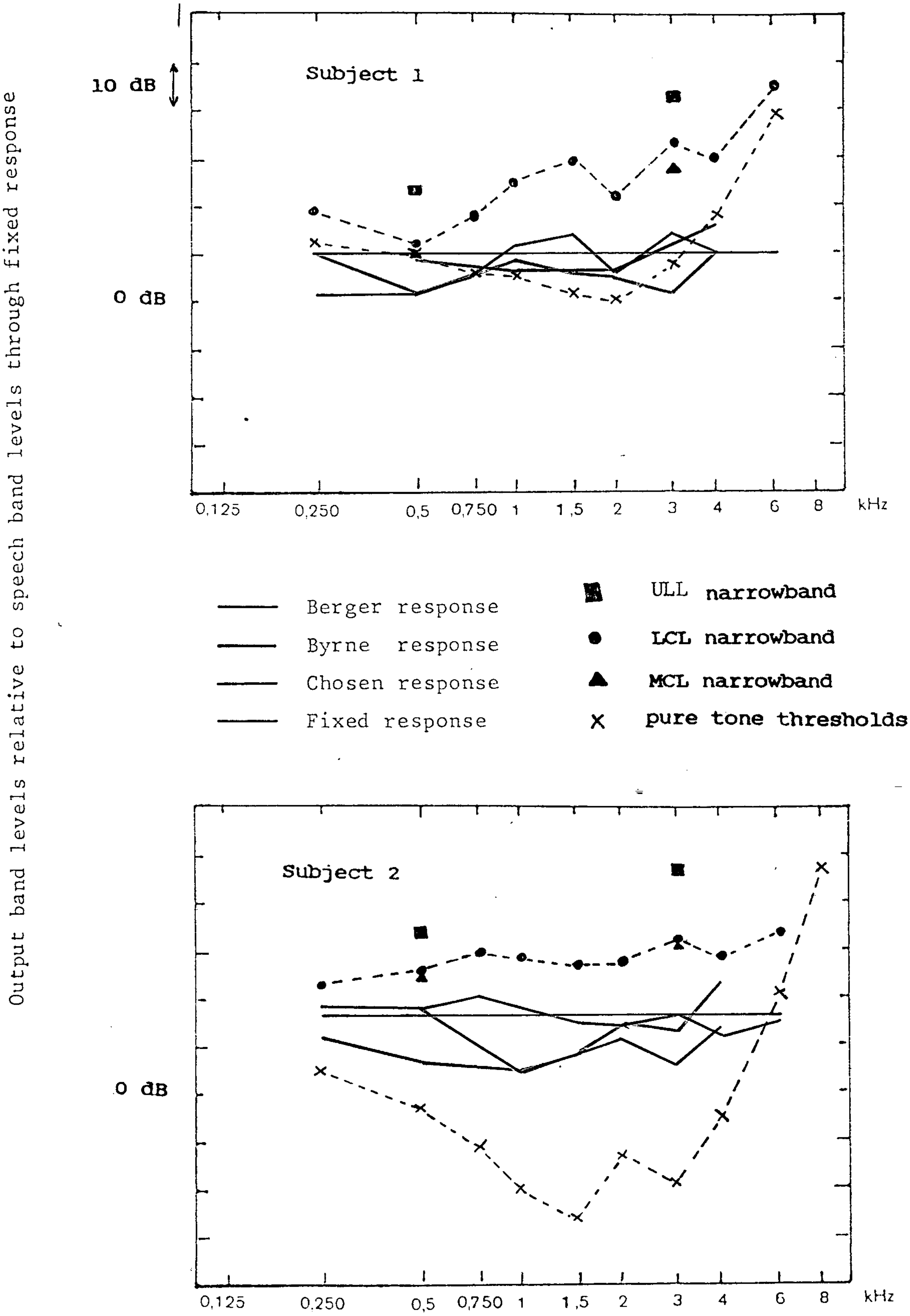


Figure 12.6 cont.

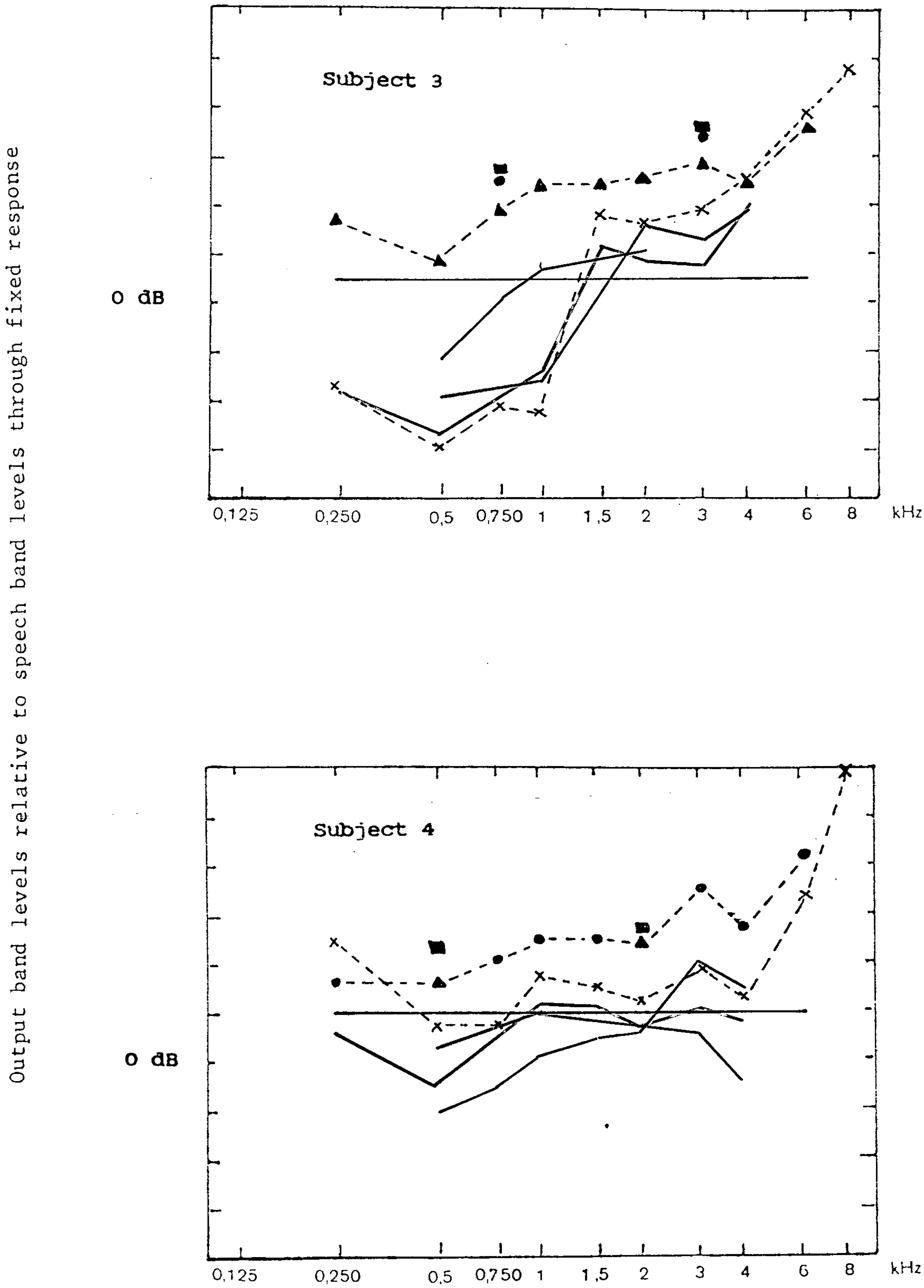


Figure 12.6 cont.

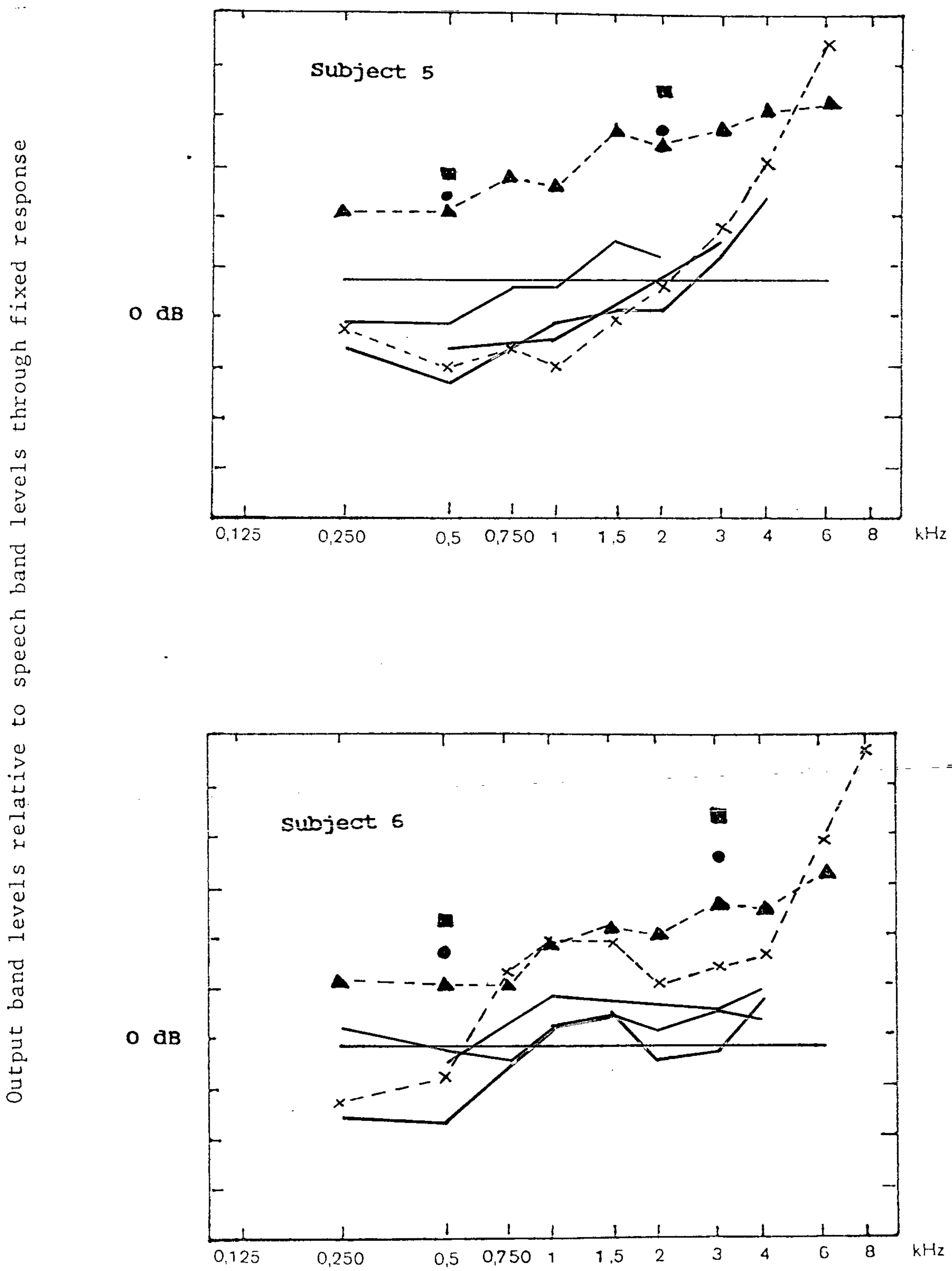
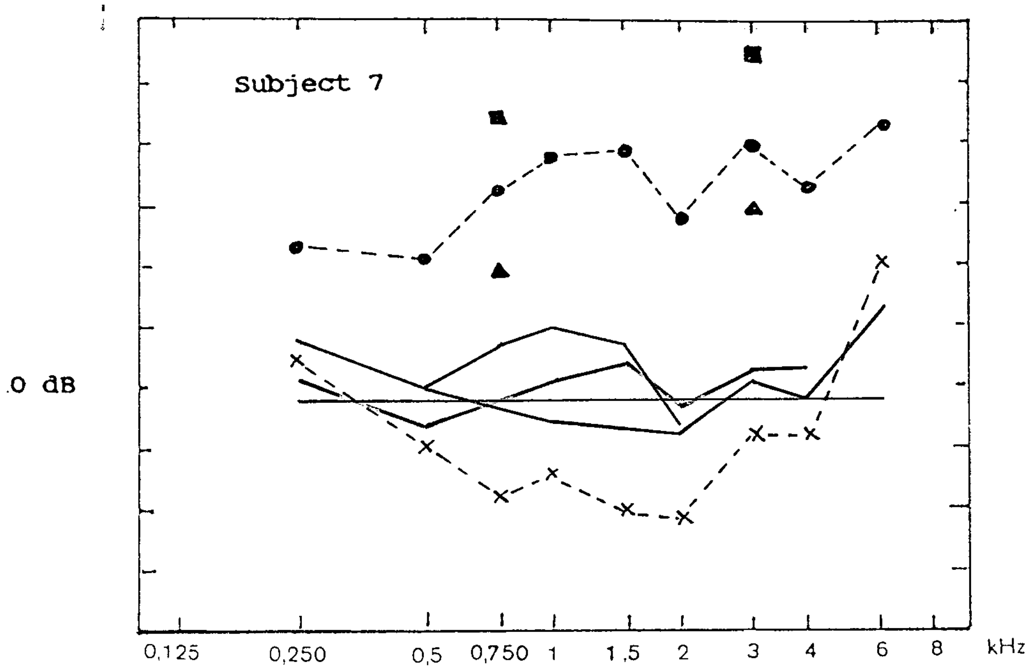


Figure 12.6 cont.

Output band levels relative to speech band levels through fixed response



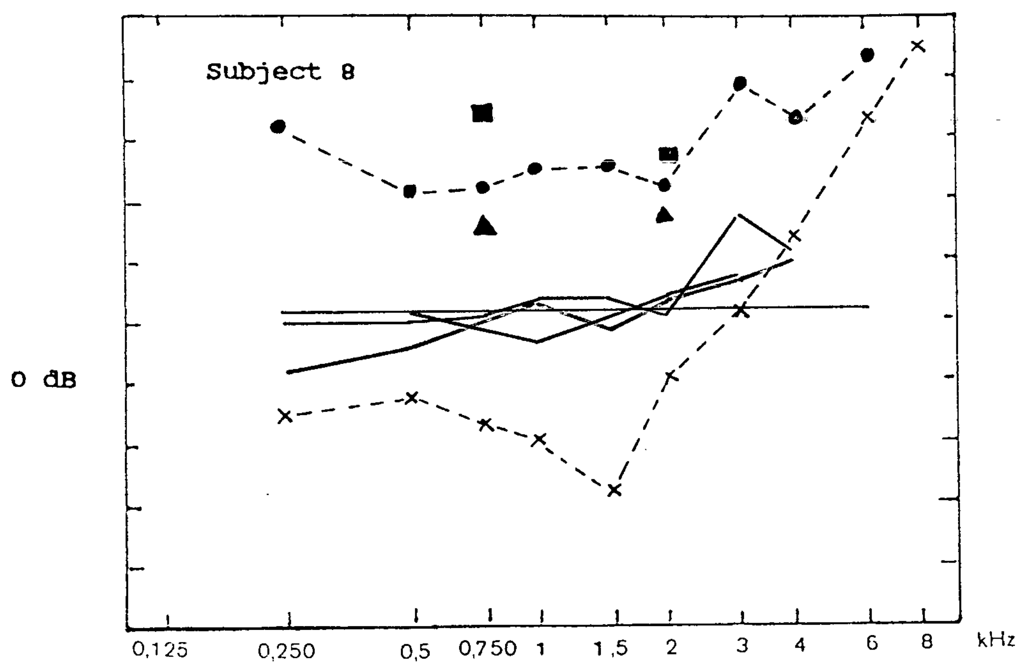
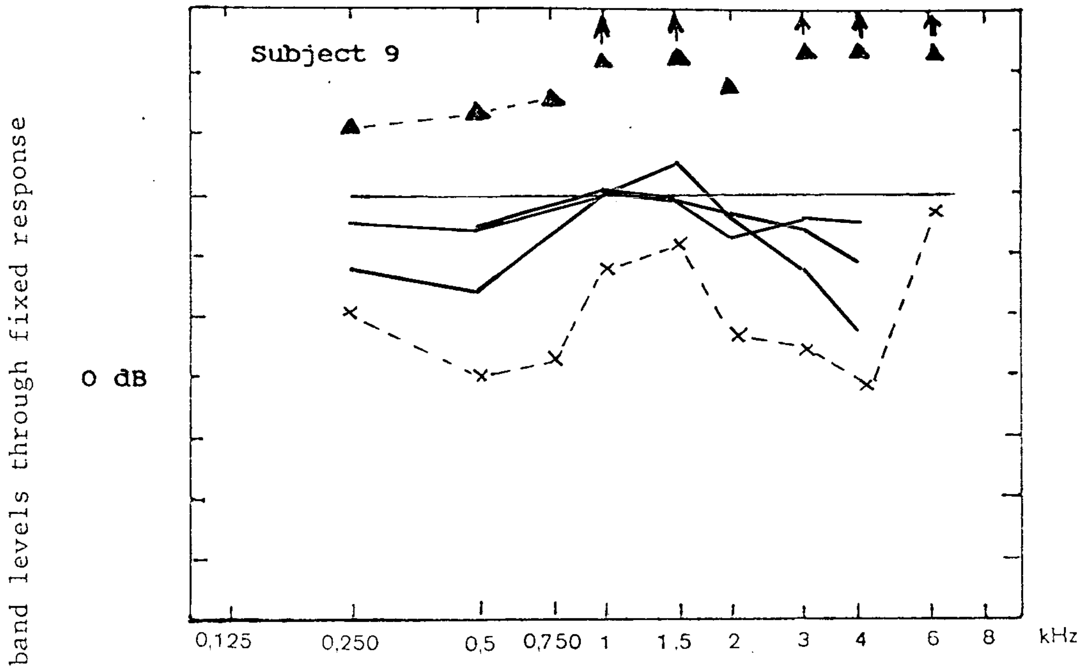


Figure 12.6 cont.



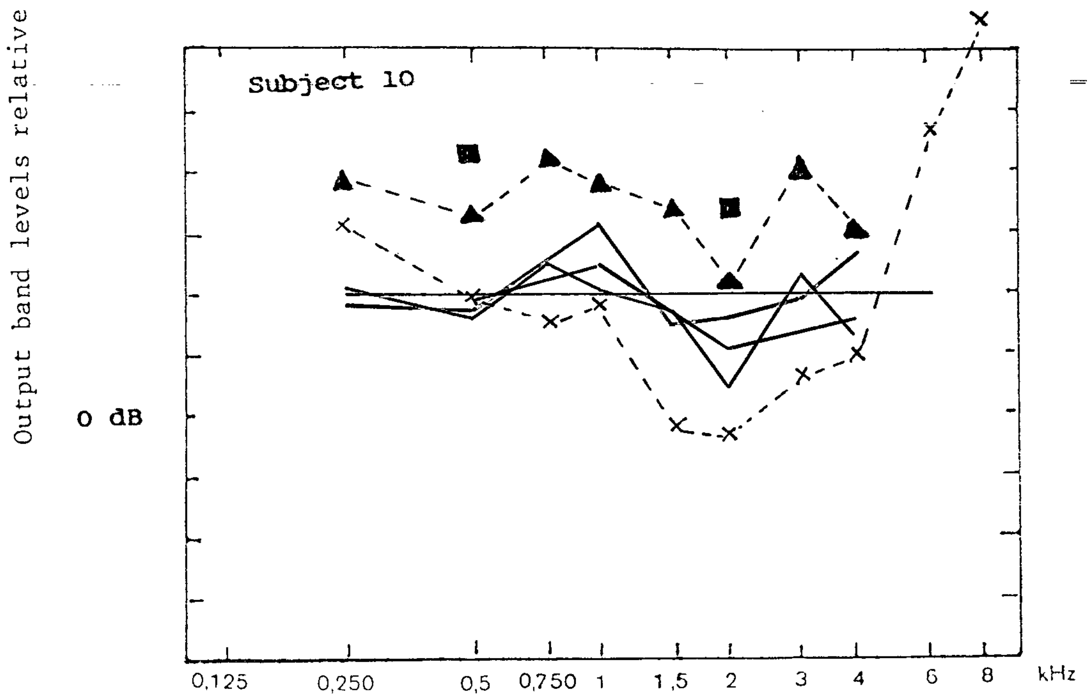


Figure 12.6 cont.

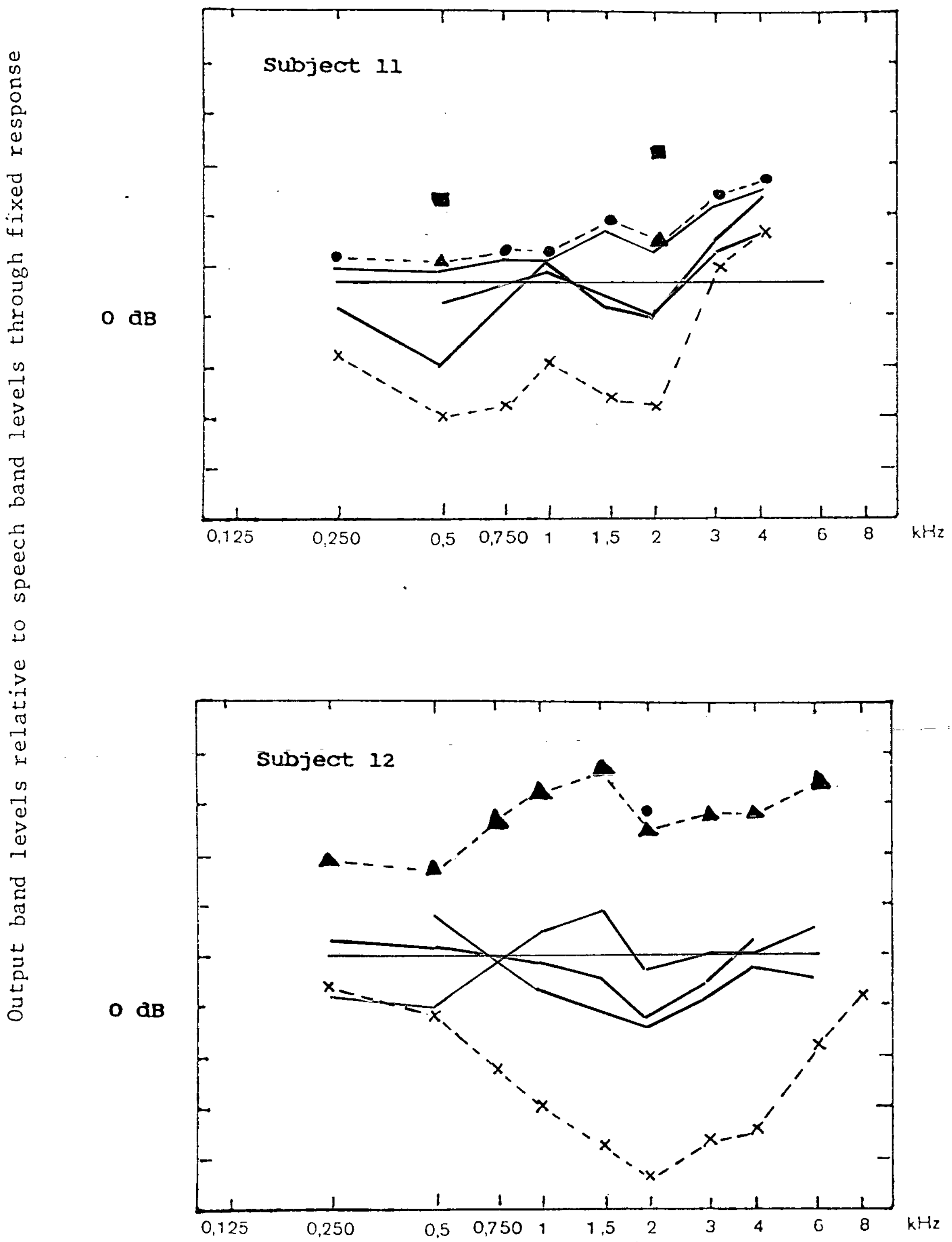


Figure 12.7

Fixed response

Output SPL (Kemar)

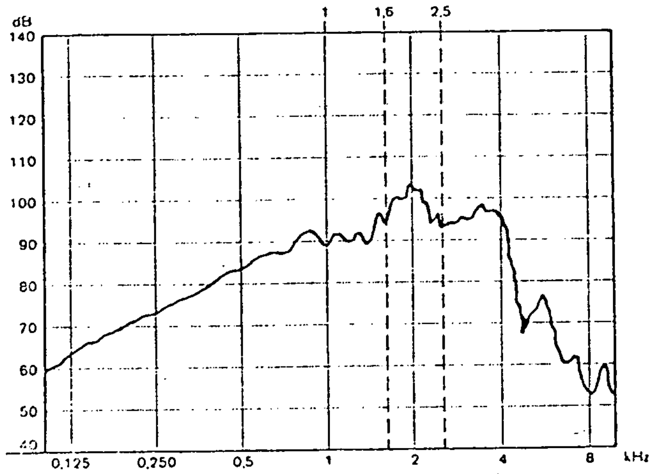


Figure 12.8

Speech spectrum

L_{eq}

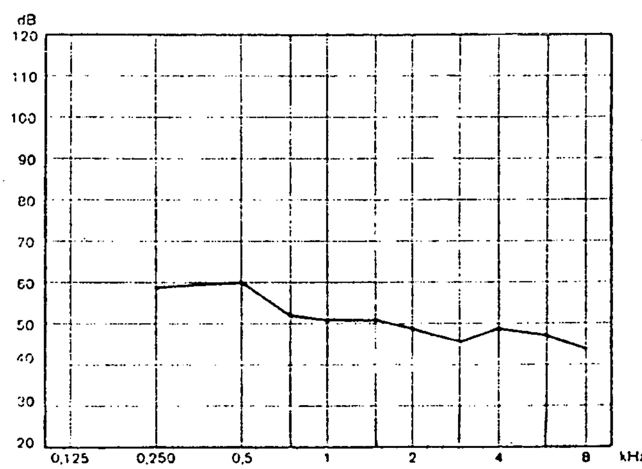
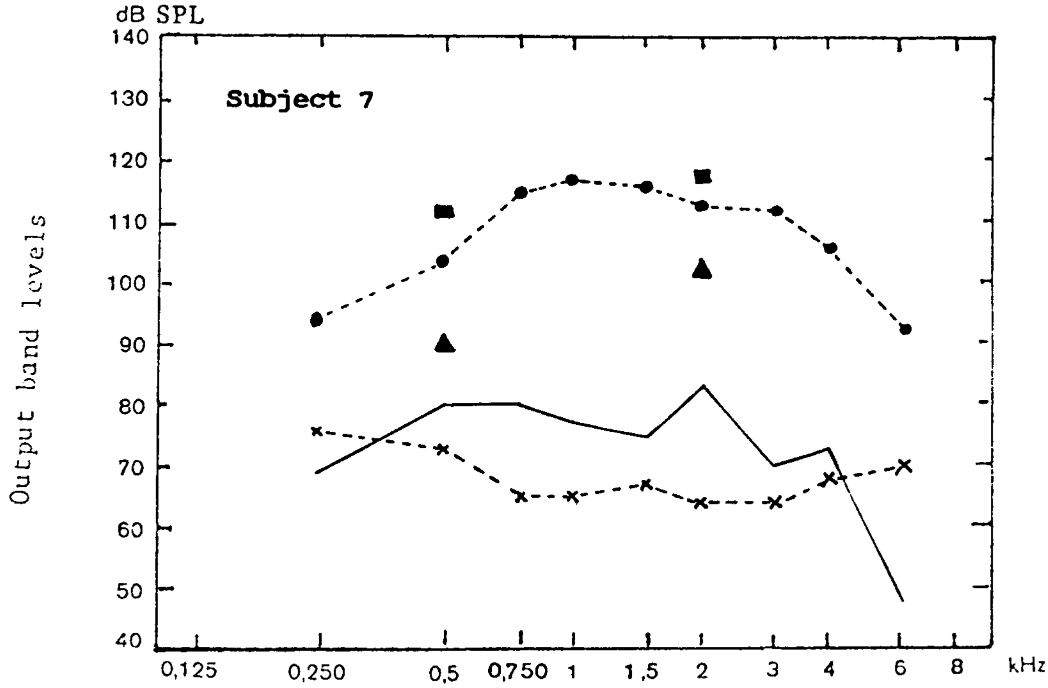


Figure 12.9

Example of individual response corrected to dB SPL

at probe microphone position in ear canal



ULL narrowband

LCL narrowband

MCL narrowband

pure tone thresholds

Fixed response

Figure 12.10

Comfort settings for sloping response conditions

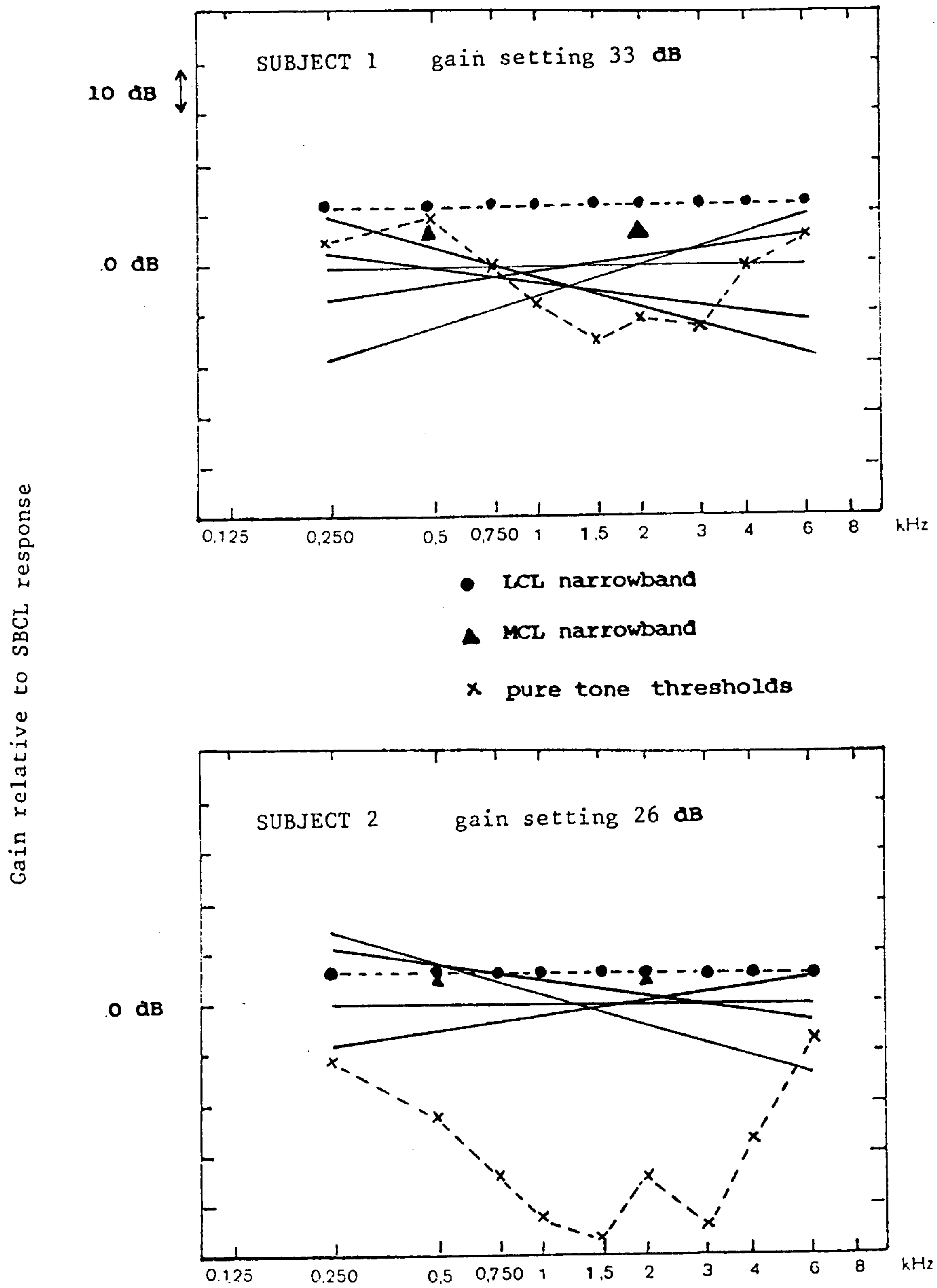
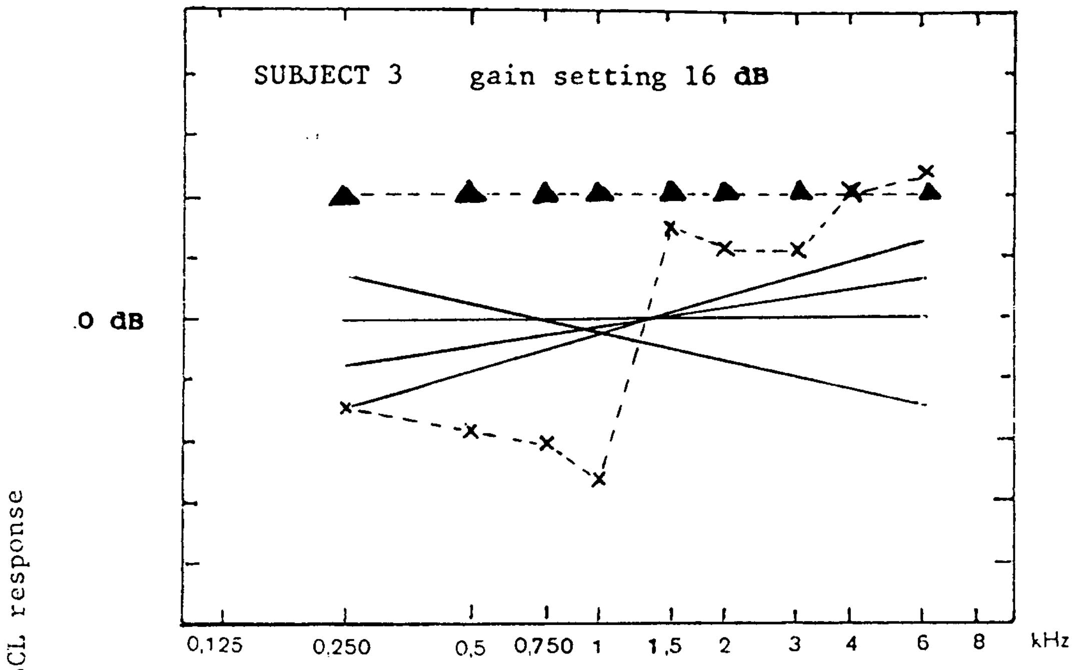


Figure 12.10 cont.



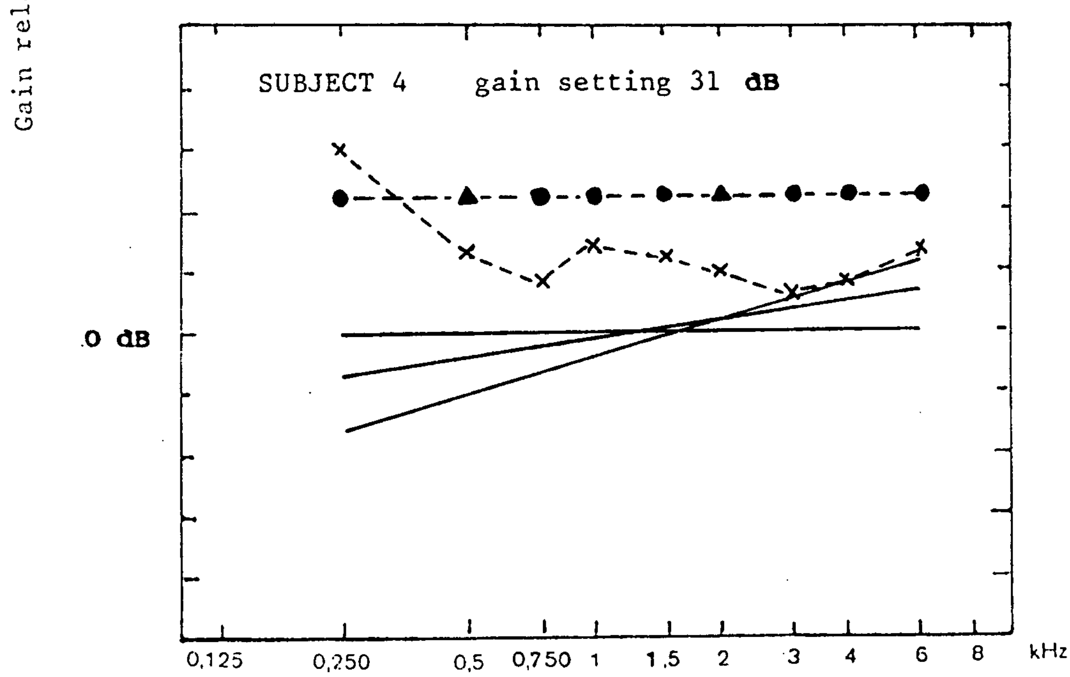
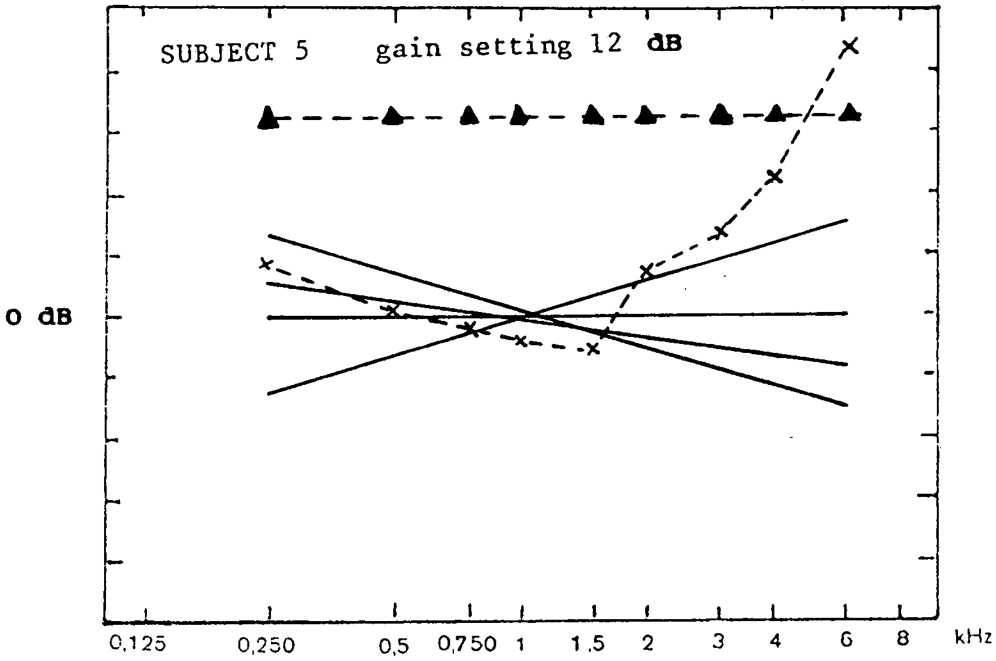
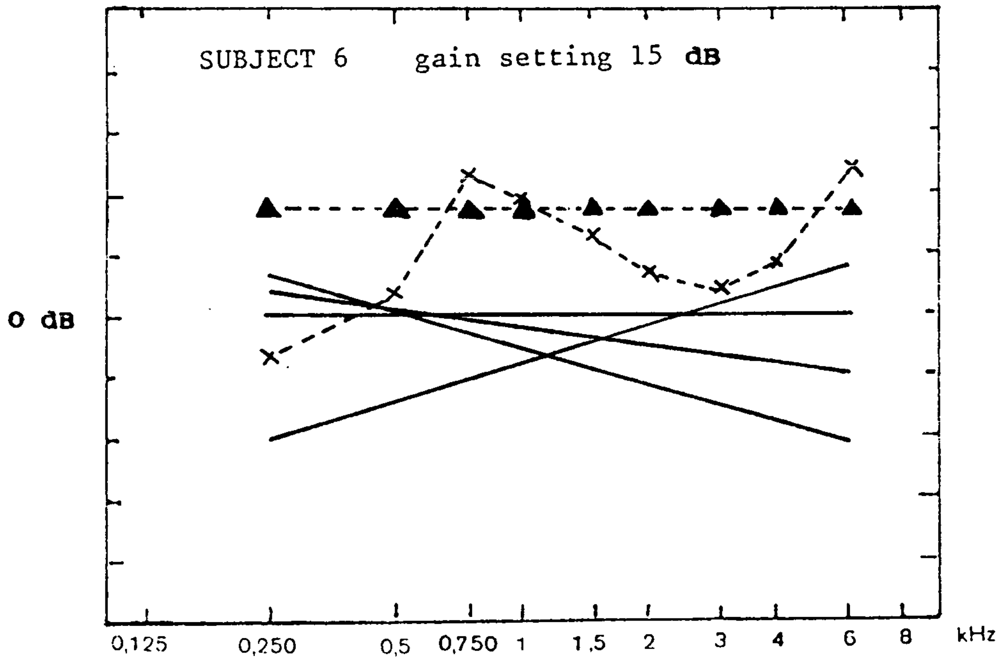


Figure 12.10 cont.

Gain relative to SBCL response





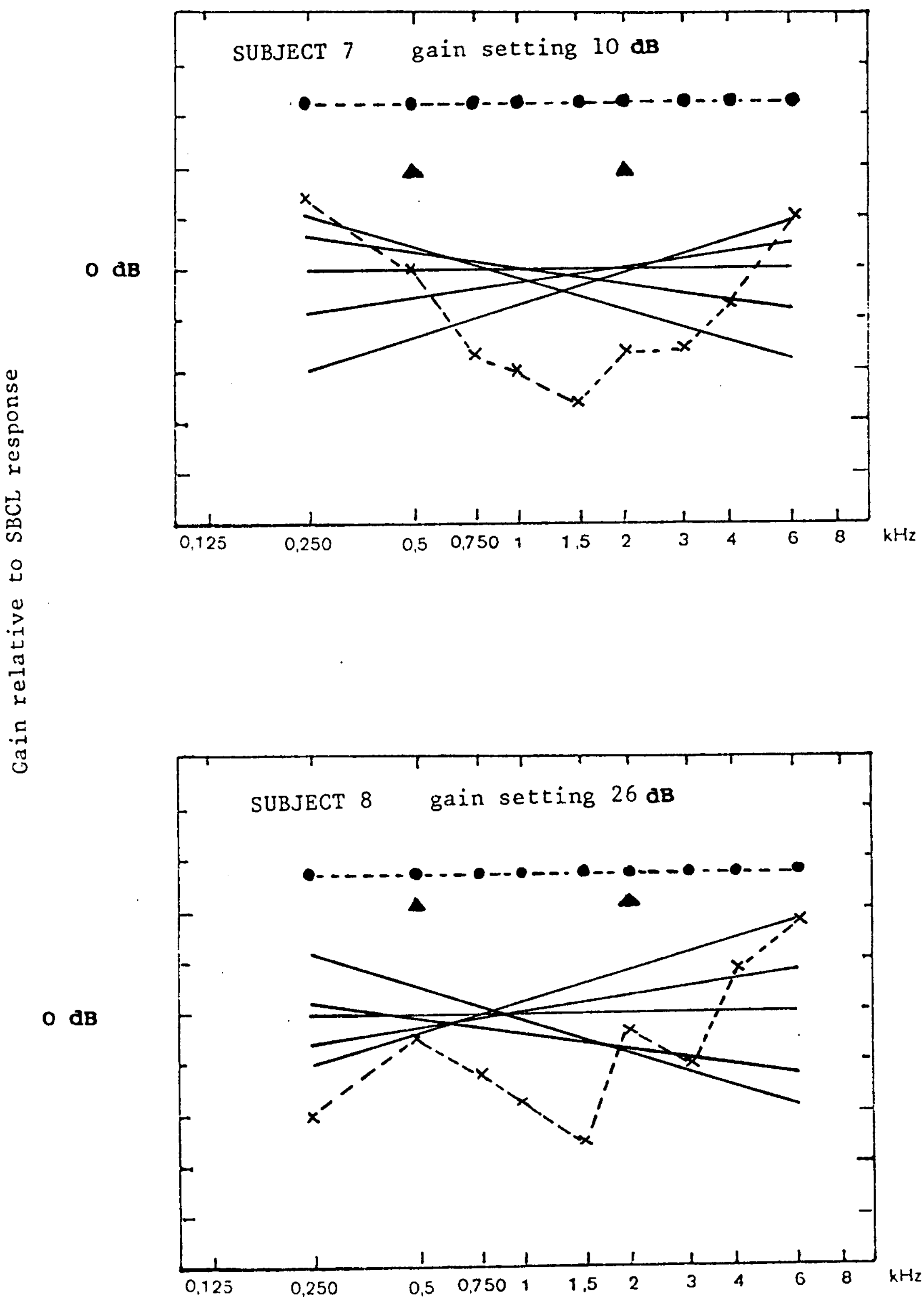

Gain relative to SBCL response
SUBJECT 7 gain setting 10 dB
0 dB
0,125
0,250
0,5
0,750
1
1,5
2
3
4
6
8
kHz
SUBJECT 8 gain setting 26 dB
0 dB
0,125
0,250
0,5
0,750
1
1,5
2
3
4
6
8
kHz

Figure 12.10 cont.

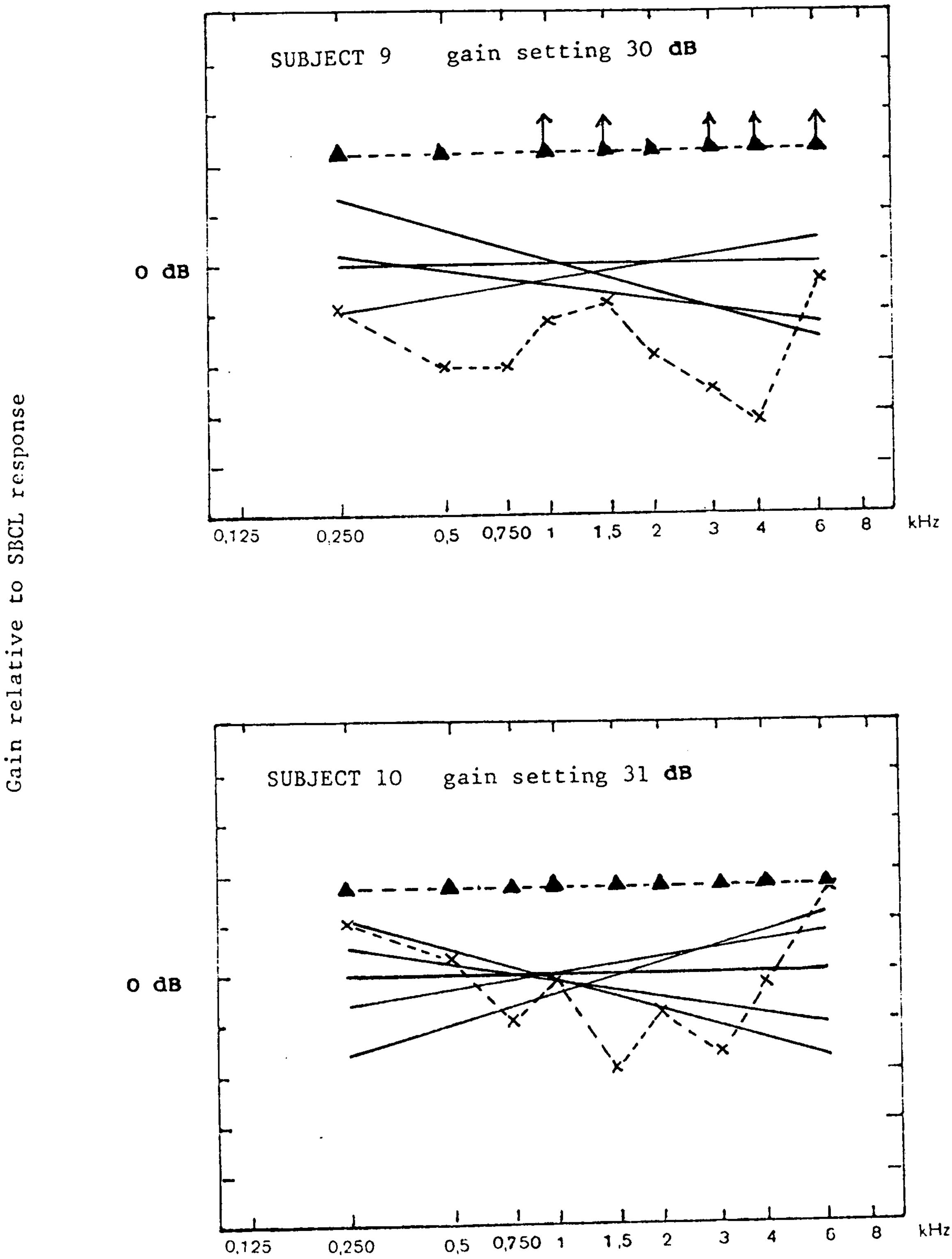
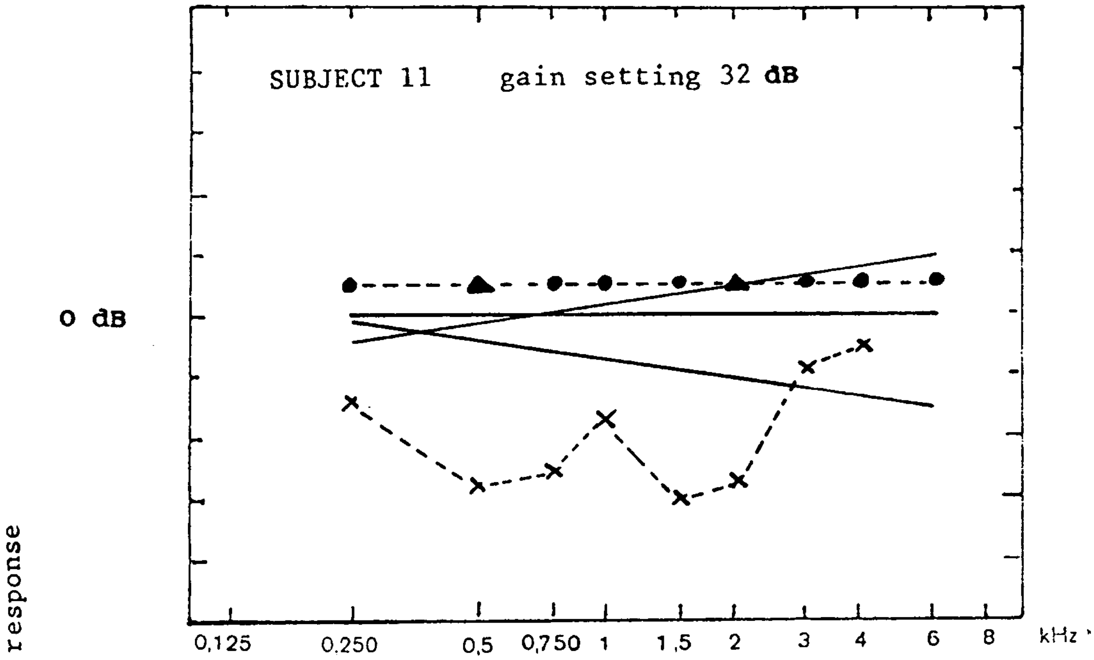


Figure 12.10 cont.



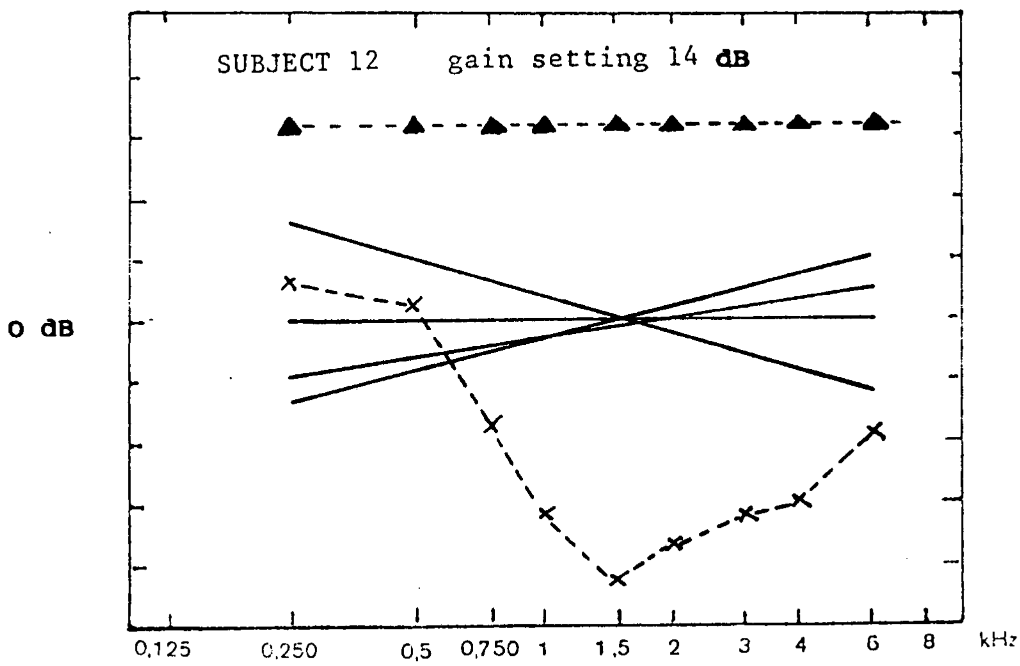


Figure 12.11

Individual 'SBCL' response

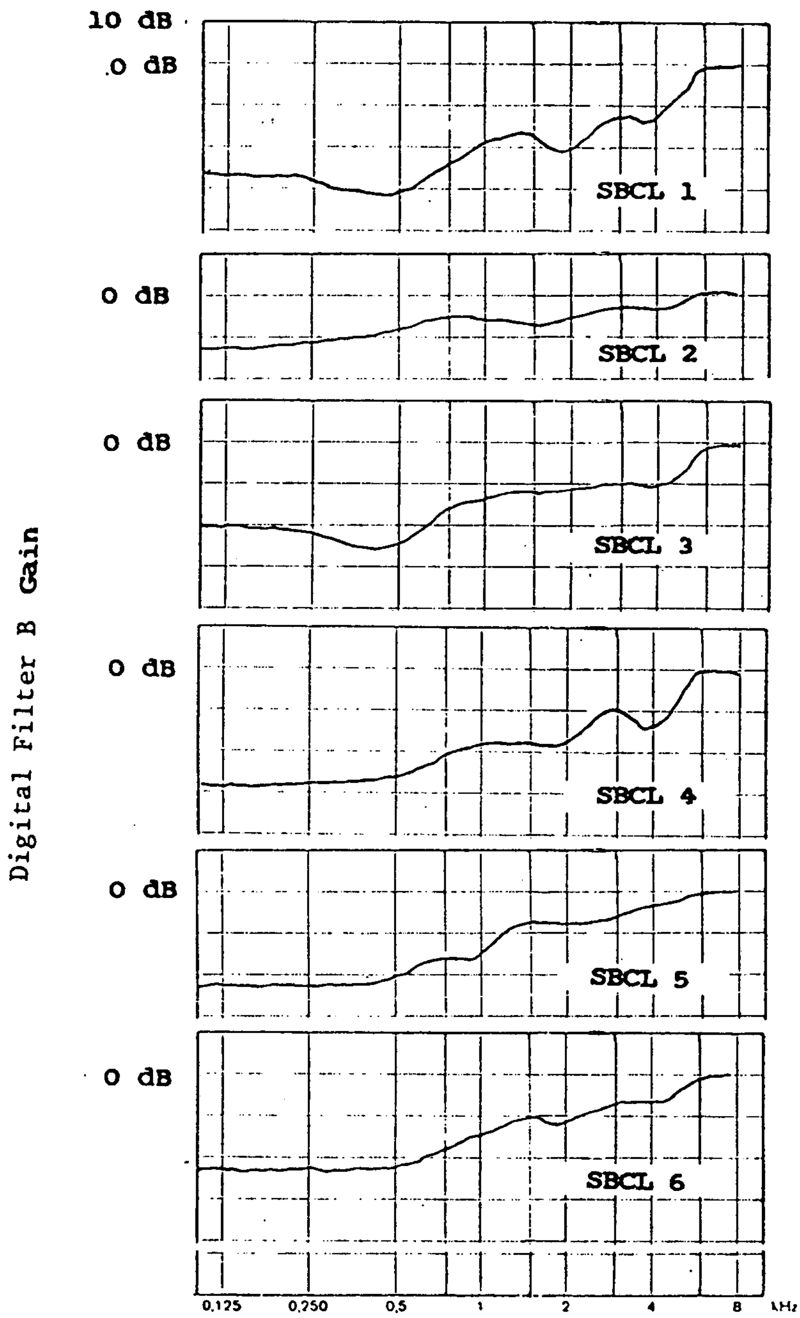


Figure 12.11 cont.

Individual 'SBCL' response

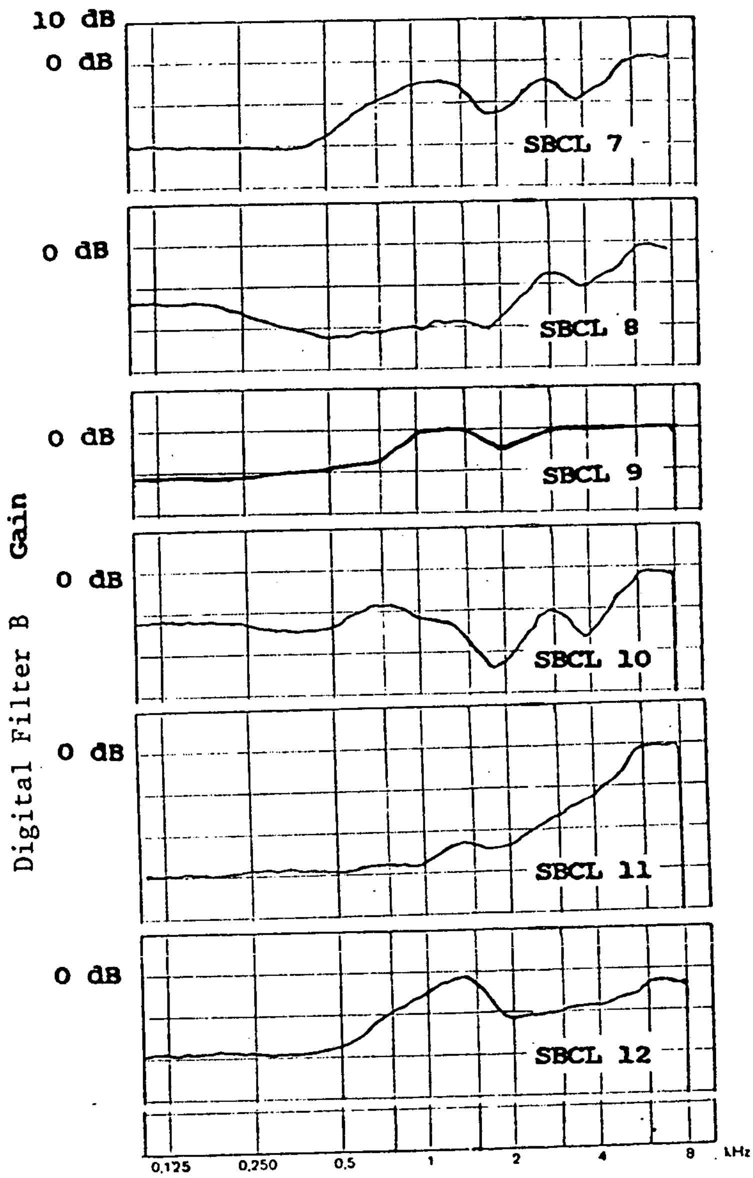
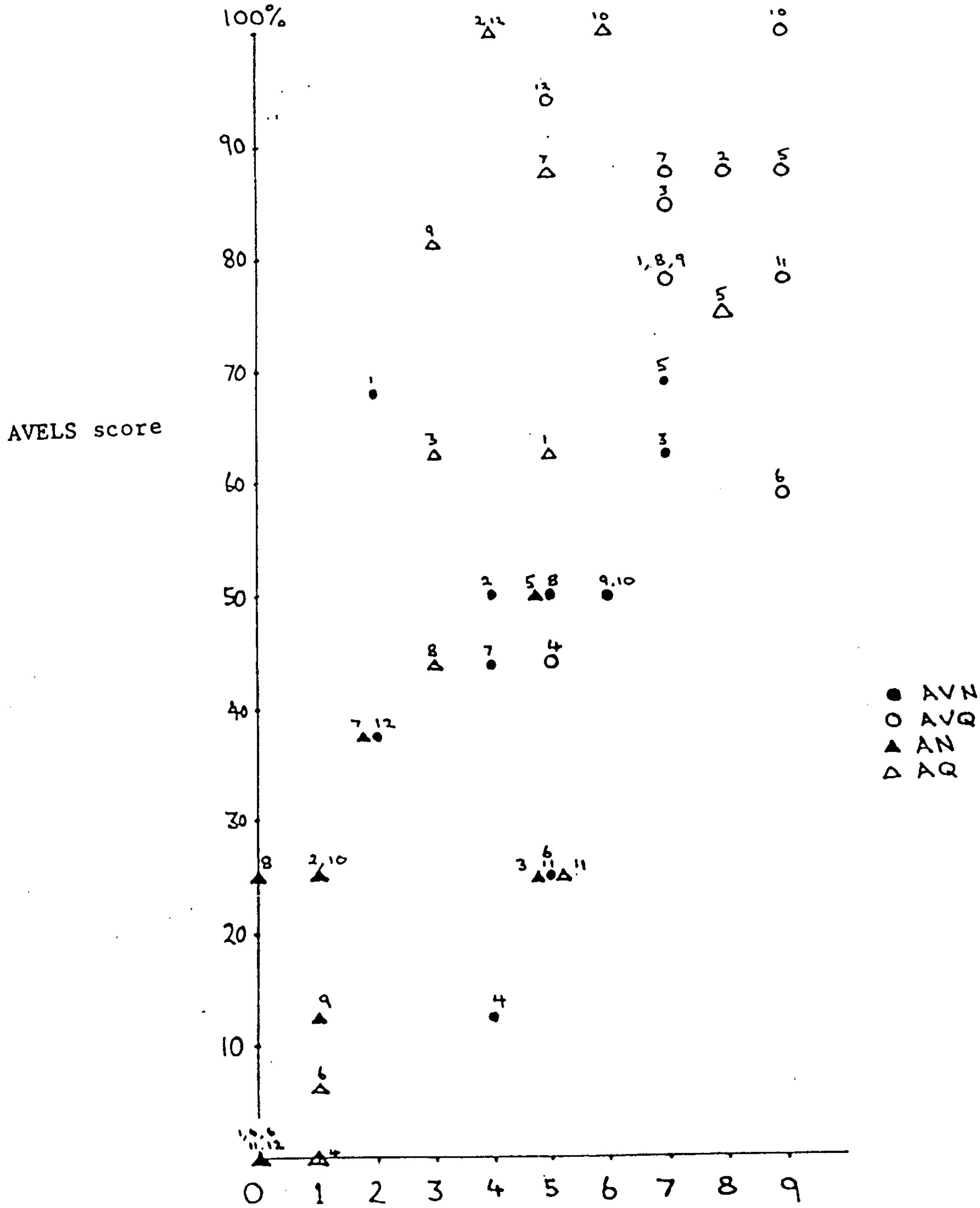


Figure 12. 12

Intelligibility scores and ratings



CHAPTER 13

CONCLUSION: FULFILMENT OF RESEARCH STUDY AIMS

13.1 Introduction

The main aims of this study were described in Chapter 6. The results and conclusions from Chapters 11 and 12 will now be drawn together to show how these aims have been filfilled.

13.2 The Development of a Clinically Feasible Protocol for the Scientific Fitting of Hearing Aids

The tests and evaluation procedures developed in this research study may be simplified to establish an evaluation procedure that is feasible within the constraints of skills, cost and time that exist in hearing aid clinics.

The pair comparison procedure can be accomplished using a somewhat restricted range of aid responses by means of analogue replay equipment and recorded hearing aid processed speech. A simple analogue channel switching and level control interface could take the place of the computer controlled system developed for the Master Hearing Aid used in this study. Alternatively, a digitally controlled system could be developed based around a reasonably cheap microcomputer to allow similar flexibility to the present system. This microcomputer would also allow hearing aid responses to be stored on a database and computer assisted hearing aid fitting to be used.

Ear canal acoustic measurements were shown to be probably unnecessary in obtaining audiometric and hearing aid measurements to the accuracy required by high power aid users. Headphone, freefield, coupler, ear simulator or manikin measurements proved to give results that were generally not noticeably different when corrections for the differences between insertion and acoustic gains were taken into account (12.5.1). However, the simple procedure of measuring audiometric data through a hearing aid receiver and the subject's earmould and calibrating the equipment on the same ear simulator used for the hearing aid measurements would allow greater accuracy in measurements than using headphone presentations (5.4). Where subjects demonstrate sensitivity to very small response changes it is possible to use one of the probe microphone techniques described in Section 5.8 to obtain greater accuracy in measurement, while also allowing rapid testing of a range of hearing aids. The T-piece probe measurement technique (5.8.4) developed during this research study is the simplest and least invasive of these procedures and can also be used with all earmould types and all subjects.

Whereas the pair comparison test gives an indication of the subject's potential to benefit from response changes the AVELS test gives a more realistic indication of how the subject will find a particular aid in real life (12.5.4). The shortened test forms of the AVELS test can be used where time is at a premium.

A simplified clinically feasible protocol is at present undergoing laboratory and field tests with a large number of subjects and is described in Appendix Q.

13.3 The Development of Hardware and Software Systems Capable of Carrying Out the Required Experimental Protocols

The Master Hearing Aid system described in detail in Chapters 8 and 9 allowed the evaluation procedures (13.2) to be successfully and reliably carried out. This system was simple to operate and its great flexibility in measuring, storing, presenting and manipulating hearing aid responses also makes it an important research tool with greater capabilities than any other system at present in use.

13.4 The Development of Hearing Aid Evaluation Procedures

The pair comparison and AVELS tests proved to be valuable and complementary procedures. The pair comparison test was shown to be a clinically feasible, simple, sensitive and reliable procedure for indicating subjects' potential to benefit from changes in frequency response (12.5.2). The AVELS test was shown to be a less sensitive but probably more realistic test of hearing aid performance (12.5.2). The AVELS test can be administered using either repeated measures with corrections for learning effects, or as two equivalent shortened test forms of 36 items each. Word or phoneme scoring can be used.

13.5 The Determination of the Relationship between Audiometric Profiles, Hearing Aid Frequency Responses and Preferred Gain Settings

The successful use of appropriate instructions to obtain reliable indications of subjects' usable comfortable dynamic ranges was demonstrated (12.4.2). Speech stimuli were found to be realistic and reliable for comfort level determination, while pure tones were shown to give similar results to narrowband speech. The relationship between hearing aid frequency responses and comfortable listening levels was shown to be complex and only broadly understood for hearing-impaired subjects. It was not accurately predictable from audiometric thresholds (12.5.3).

13.6 The Study of the Relationship between Audiometric Profiles and Optimum Hearing Aid Frequency Response

No hearing aid selection procedure was found to successfully predict all subjects' preferred aids, although Byrne and Tonnison's prescription method was the most successful (12.5.2(iv)). The SBCL (Speech Band Comfortable Level) approach to hearing aid fitting was found not to be successful. This was because the attempt to present the speech signal such that individual bands of speech are at the MCL for each band, results in presentation levels up to 20 dB too loud, and leads to subjects reducing the output level by some 20 dB. The response therefore proved not to be a good starting point for the adaptive pair comparison procedure to find the chosen aid response (12.5.2(iii), 12.5.3(vi)).

13.7 The Examination of the Importance of Deviations from the Optimum Response and its Relationship to Audiometric Measurements

Response changes in low frequencies were more noticeable than changes in high frequencies (12.5.2). Cutting off frequencies below 400 Hz was disliked by almost all the hearing-impaired subjects whereas approximately equal numbers preferred, disliked or failed to notice a frequency roll off below 250 Hz. Restricting the high frequencies from 8 kHz to 4 kHz had little noticeable effect while limiting them to 2 kHz was disliked by the majority of subjects.

The 10 dB peaks at typical behind-the-ear aid-frequencies were always disliked if detected, and this therefore indicates the potential value of smoothing the hearing aid response. 5 dB peaks were neither generally detected nor always disliked.

The loss of natural ear canal resonance was noticed and disliked by the majority of subjects and therefore stresses the necessity of considering the true insertion gains when deciding on the required acoustic 'transmission' gain for a subject.

For the majority of severely hearing-impaired subjects in this study the difference between 2 cc and Zwislocki coupler responses was not noticed, most probably due to the difference being mainly at higher frequencies above 2 kHz. Differences between hearing aid responses as measured on Kemar and in subjects' own ears were, however, noticed by over half the subjects on the pair comparison, although the differences were mainly slight.

Hearing-impaired subjects were less sensitive to distortion caused by 'peak clipping' than normally hearing subjects, most probably because peak clipping produced harmonic distortion at high frequencies.

The implications of these findings for the majority of users of higher power aids would seem to be that a bandwidth of 250 Hz-4 kHz would be satisfactory, while peaks in the frequency response should preferably be kept below 10 dB. The use of a 2 cc coupler rather than the Zwislocki ear simulator for aid response specification will not noticeably affect the hearing aid response for the majority of subjects, whereas the failure to specify aid responses in terms of insertion gains rather than acoustic gains may be noticed. Distortion caused by the restriction of maximum output through peak clipping will be less noticeable for the severely hearing-impaired than for normally hearing subjects although no actual general quantitative guidelines can be given due to the dependency on the individual, stimuli and actual components of distortion caused in the particular hearing aid.

These conclusions are of course based on the results of this research study and require substantiating on a larger number of subjects using a field trial in addition to laboratory tests to establish their widespread applicability. This substantiation field/laboratory study is at present in progress and is described in Appendix Q.

13.8 The Examination and Development of Techniques for the Measurement of Real Ear Hearing Aid Performance

Probe microphone measurement techniques were developed for accurate specification of audiometric profiles and hearing aid responses. However, headphone and freefield thresholds and ear simulator and manikin hearing aid measurements were shown to be reasonably similar to real ear probe microphone measurements, giving measurement errors relative to probe microphone measurements that would generally not have been subjectively noticed in real life for the hearing-impaired subjects used in this study (12.5.1).

CHAPTER 14

SUGGESTIONS FOR FURTHER WORK

The evaluation procedures developed in this research should be extensively tested on a larger number of subjects to confirm the findings of this study for the severely hearing-impaired and to establish the applicability of the procedures and results to subjects with a wider range of impairments. The development of the clinically feasible protocol described in this thesis and summarized in Section 13.2 has made this possible to carry out with a minimum expenditure of time, skills and cost.

Further extensive studies are required to establish the relationship between frequency response, audiometric profile and comfortable gain setting because fitting methods involving the shaping of the speech spectrum so that it follows an equal loudness contour at the chosen comfort setting must at present use a trial-and-error approach in the absence of any valid theoretical framework.

The possibility of using limiting or compression to allow a higher comfortable level to be tolerated also requires investigation; this could be implemented using the Master Hearing Aid system and evaluation procedures developed in this study.

The widespread use of scientific hearing aid evaluation protocols such as the one developed in this thesis, would allow a large amount of relevant information concerning the optimum and satisfactory fitting of hearing aids to be efficiently obtained in both the hearing aid clinics and research laboratories. This information could be used to improve the fitting of present-day hearing aids and to suggest possible improvements in future hearing aid design.

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APPENDIX A

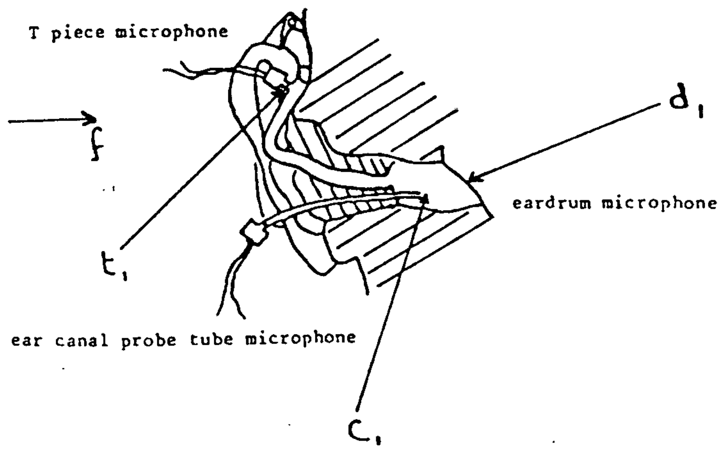
T-PIECE TECHNIQUE VALIDATION EXPERIMENT

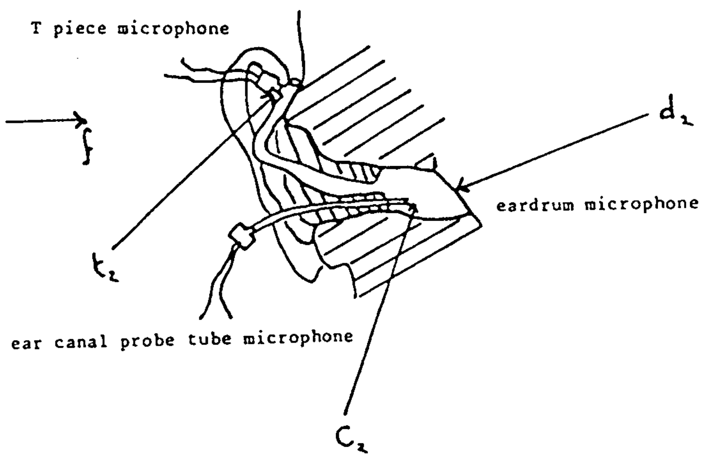
An experiment was undertaken to validate the T-piece technique described in Chapter 5. Measurements were made on Kemar in an anechoic room using a T-piece microphone, an ear canal probe tube microphone and an 'eardrum' microphone, for two widely different conditions, as shown in Figure A.1. The first condition involved the use of a hearing aid coupled to an earmould, whereas the second condition did not use the hearing aid; the earmould tubing being left open and unconnected. For both conditions, a swept sinewave of constant level in the absence of Kemar, was delivered from a single driver loudspeaker and the sound pressure level at the three microphone positions was measured and recorded using the audio test station and computer.

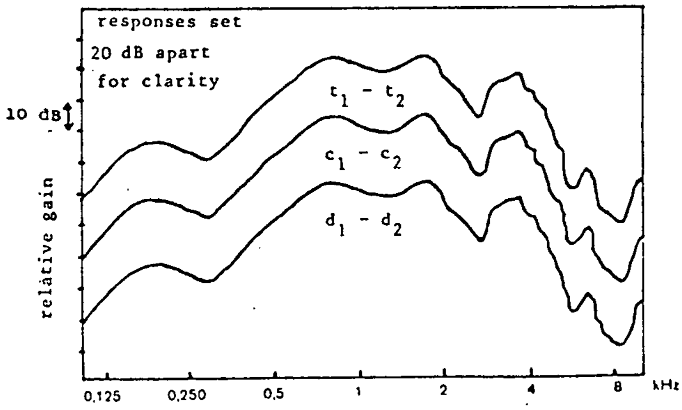
The differences between the measured sound pressure levels for the two conditions at each of the three microphone positions is shown in Figure A.1. As predicted by transmission line theory (section 5.8.4), relative measures between the two conditions are identical for the T-piece microphone, ear canal microphone and eardrum microphone.

Figure A.1

T Piece technique validation







APPENDIX B

IHR UNIVERSAL DIGITAL FILTER - HOST WINDOWING TECHNIQUE

The IHR Universal Filter is a direct convolution Finite Impulse Response (F.I.R.) filter, capable of real-time operation at sample rates up to 60 kHz. The impulse response is programmable up to 512 points for symmetric inpulse responses.

The technique of Host Windowing (Abed and Caine, 1978) allows efficient and close-to-optimal creation of the required filters. This technique can be explained in terms of a windowed ideal low pass filter of length N which may be factorized into a host filter $h_o(nT)$ and a trigonometric window $w_o(nT)$

$$h(nT) = h_o(nT)w_o(nT), \qquad n = 0, 1, \ldots, N - 1.$$

Once a host filter has been optimized, whole families of filters can be produced. Band pass filters can be designed by first subtracting the trigonometric windows of two low pass filters. By extending this principle, multiband filters can be designed efficiently. The transition width (i.e., cut-off rate) and ripple of the host filter will determine the transition width and ripple of the target filter.

The choice of host filter involves a compromise between sampling rate, impulse response length and transition width. The sampling rate is constrained to be at least twice the maximum input frequency to prevent aliasing. For minimum ripple in the filter the maximum possible impulse response length for a given transition width is required, and is constrained by the chosen sample rate and the calculation speed of the digital filter.

The chosen host filter used throughout this study had a sampling rate of 30 kHz, an impulse response length of 398 points and a transition width of 180 Hz. This provided the required flexibility for filter response creation.

APPENDIX C

COMPUTER INTERFACES

C.1 TU-ART Digital Interface

The Cromemco TU-ART (Twin Universal Asynchronous Receiver and Transmitter) provides two channels of duplex serial data exchange, two channels of parallel data exchange and ten interval timers.

Digital filter A was connected to the output of parallel port, device A.

The subject response box was connected to the input of parallel port, device B.

The digital attenuator was connected to the output of parallel port, device B.

The base address of device A was 160. The base address of device B was 180.

The pin configurations of the port connectors are shown in Figure C.1.

C.2 8PIO

The Cromemco 8PIO is an S-100 bus compatible, eight parallel input/output board. Each input/output port pair are latched and share 8 bi-directional lines. I/O port 6 is wired to software monitor and control the other seven 8PIO ports.

The 8PIO also features a pair of relays rated at 2 amps at 28 Volts D.C. and activated by data bits 0 and 1 of port 1.

The B & K interface was connected to ports 2, 3, 4 and 5 on connectors J2 and J3.

Digital filter B was connected to port 7.

The base address of the 8PIO board was set to 128.

The pin configurations of the port connectors are shown in Figure C.1.

C.3 Digital Attenuator

The digital attenuator gave 0 to 99 dB attenuation by means of a passive resistor network switched by electromagnetic relays. Remote operation via the microcomputer TU-ART parallel port was via the 8 data lines using binary coded decimal:

TU-ART data bit	Attenuation value (dB)
0	1
1	2
2	4
3	8
4	10
5	20
6	40
7	80

C.4 Subject Response Box

The pin connections of the subject response box to the input of parallel port device B were:

	TU-ART pin	Response box button
Data bit 0	7	Quieter
1	19	Louder
2	6	Stop
3	18	Red select
4	5	Green select
Ground	14	
5 V	3	L.E.D.

Figure C.1

TU-ART Digital Interface — Cromemco

J2 Parallel A

Pin	Name	Signal Direction	Voltage Level	Function
1	Invert	Input	TTL	Used for normal/reverse address switching. See discussion under "Switch Selectable Options."
2	Input Strobe A	Output	TTL	When active indicates that data present on input bits Ø-7 is being sampled.
3	Vcc	Output	+5V	
4	Bit 6	Input	TTL	
5	Bit 4	Input	TTL	
6	Bit 2	Input	TTL	
7	Bit Ø	Input	TTL	
8	Disable	Input	TTL	Turns the output drivers for the parallel output bits OFF.
9	Output Strobe A	Output	TTL	Indicates that data is present on parallel output bits Ø-7.
10	Bit 6	Output	TTL	
11	Bit 4	Output	TTL	
12	Bit 2	Output	TTL	
13	Bit Ø	Output	TTL	
14	Signal Ground	Output	ØV	
15	SENS A	Input	TTL	Interrupt request, input to IC 4 55Ø1 (A)
16	Bit 7	Input	TTL	
17	Bit 5	Input	TTL	
18	Bit 3	Input	TTL	
19	Bit 1	Input	TTL	
20	NMI	Input	TTL	Non maskable interrupt. This pin is tied directly to pin 12 of the S-1ØØ bus. Consult the Z-8Ø manual for use. Only usable with the Cromemco ZPU card.
21	Wait	Input	TTL	This pin is tied directly to pin 72 (PRDY) of the S-1ØØ bus. Forces the CPU to wait when held low.
22	Bit 7	Output	TTL	
23	Bit 5	Output	TTL	
24	Bit 3	Output	TTL	
25	Bit 1	Output	TTL	

8PIO — Cromemco

EIA Connector J1 Thru J4 PIN-OUTS

Pin	Connector J1	Connector J2	Connector J3	Connector J4
1	OUTPUT STROBE 6	OUTPUT STROBE 4	OUTPUT STROBE 2	OUTPUT STROBE Ø
2	OUTPUT ENABLE 6	OUTPUT ENABLE 4	OUTPUT ENABLE 2	OUTPUT ENABLE Ø
3	LATCH INPUT 6	LATCH INPUT 4	LATCH INPUT 2	LATCH INPUT Ø
4	BIT DØ PORT 6	BIT DØ PORT 4	BIT DØ PORT 2	BIT DØ PORT Ø
5	BIT D1 PORT 6	BIT D1 PORT 4	BIT D1 PORT 2	BIT D1 PORT Ø
6	BIT D2 PORT 6	BIT D2 PORT 4	BIT D2 PORT 2	BIT D2 PORT Ø
7	BIT D3 PORT 6	BIT D3 PORT 4	BIT D3 PORT 2	BIT D3 PORT Ø
8	BIT D4 PORT 6	BIT D4 PORT 4	BIT D4 PORT 2	BIT D4 PORT Ø
9	BIT D5 PORT 6	BIT D5 PORT 4	BIT D5 PORT 2	BIT D5 PORT Ø
1Ø	BIT D6 PORT 6	BIT D6 PORT 4	BIT D6 PORT 2	BIT D6 PORT Ø
11	BIT D7 PORT 6	BIT D7 PORT 4	BIT D7 PORT 2	BIT D7 PORT Ø
12	OUTPUT STROBE 7	OUTPUT STROBE 5	OUTPUT STROBE 3	OUTPUT STROBE 1
13	OUTPUT ENABLE 7	OUTPUT ENABLE 5	OUTPUT ENABLE 3	OUTPUT ENABLE 1
14	LATCH INPUT 7	LATCH INPUT 5	LATCH INPUT 3	LATCH INPUT 1
15	BIT DØ PORT 7	BIT DØ PORT 5	BIT DØ PORT 3	BIT DØ PORT 1
16	BIT D1 PORT 7	BIT D1 PORT 5	BIT D1 PORT 3	BIT D1 PORT 1
17	BIT D2 PORT 7	BIT D2 PORT 5	BIT D2 PORT 3	BIT D2 PORT 1
18	BIT D3 PORT 7	BIT D3 PORT 5	BIT D3 PORT 3	BIT D3 PORT 1
19	BIT D4 PORT 7	BIT D4 PORT 5	BIT D4 PORT 3	BIT D4 PORT 1
2Ø	BIT D5 PORT 7	BIT D5 PORT 5	BIT D5 PORT 3	BIT D5 PORT 1
21	BIT D6 PORT 7	BIT D6 PORT 5	BIT D6 PORT 3	BIT D6 PORT 1
22	BIT D7 PORT 7	BIT D7 PORT 5	BIT D7 PORT 3	BIT D7 PORT 1
23	+OPTO D3 PORT 4	+OPTO D4 PORT 4	RELAY DØ PORT 1	RELAY D1 PORT 1
24	-OPTO D3 PORT 4	-OPTO D4 PORT 4	RELAY DØ PORT 1	RELAY D1 PORT 1
25	GROUND RETURN	GROUND RETURN	GROUND RETURN	GROUND RETURN

APPENDIX D

EQUIPMENT SPECIFICATION

D.1 IHR Universal Digital Filter Specifications

Digital

25-way D connector

Connects to 8 data bits and output strobe.

Pin out corresponds to that of Cromemco TU-ART parallel port.

Integral 12-bit analogue to digital converter.

Integral 16-bit digital to analogue converter.

Maximum sampling rate 60 kHz

Maximum impulse response length 516 points symmetric
256 points arbitrary.

Analogue input

Maximum undistorted input ± 1.0 V pp

20 Mohm input impedance

TTL sampling clock

Analogue output

Maximum ± 1.0 V pp

1 ohm output impedance

Slew rate: 30,000,000 V/s

Spurious outputs

Input quantising noise: -65 dB

Output quantising noise: -80 dB

Harmonic distortion (1 kHz in): -60 dB at 2 kHz
-75 dB at 3 kHz

D.2 Neve Limiter Compressor 2264

Input impedance	10,000 ohms
Noise	< −75 dBm
Frequency response	± 0.25 dB, 20 Hz to 20 kHz
Distortion at 1 kHz	0.05% (no compression/limiting)
Limit threshold variable	+4 dBm to +15 dBm; in 0.5 dB steps
Limit ratio	> 100:1
Limit attack time	4 ms
Limit recovery time	switchable, 50 ms to 5 s
Compression ratio	adjustable from 1.5:1 to 6:1
Compression threshold	variable from −20 dBm to +10 dBm, in 2 dB steps
Compressor recovery time	variable 50 ms to 1.5 s
Gain	0–20 dB, switchable

D.3 TPA 25D Power Amplifier (H/H electronic)

Maximum power output	45 volts rms into 7.5 ohm
Frequency response	± 0.1 dB, 20 Hz to 20 kHz
Total harmonic distortion	<0.1%, 20 Hz to 20 kHz
Signal-to-noise ratio	100 dB
Damping factor	100
Slewing rate	> 10 V/μs

D.4 KEF Model 101 Loudspeaker

Frequency range	90 Hz–30 kHz, ±2 dB at 2m on reference axis
Directional characteristics:	Within 2 dB of response on reference axis up to 20,000 Hz for ±5° vertically up to 20,000 Hz for ±20° horizontally
Maximum output	100 dB SPL on program peaks
Characteristic sensitivity level:	81 dB at 1 m on reference axis for pink noise input of 1 W (anechoic conditions)
Distortion	2nd harmonic < 1% from 120 Hz–20 kHz 3rd harmonic < 1% from 70 Hz–20 kHz measured at 1 m on reference axis at mean SPL of 90 dB, anechoic conditions
Nominal impedance	8 ohm
Dimensions	348 × 180 × 197 mm

D.5 <u>Sony CVM 2000 PSB</u> (20" colour receiver monitor)

D.6 <u>Sony VO-2850P U-matic Videocassette recorder</u>

Video horizontal resolution	240 lines (colour)
Video signal-to-noise ratio	> 43 dB
Audio frequency response	50 Hz-15 kHz
Audio signal-to-noise ratio	> 45 dB
Audio wow and flutter	0.2% rms

D.7 <u>Cromemco System 3 Microcomputer</u>

Cromemco 64 KZ 64 k-byte S-100 bus compatible read-write memory board.

16 FDC floppy disc controller

2 × 8" Perscii2 disc drives (2 × 1.2 megabytes)

Televideo 920b terminal

Anadax graphics printer model DP9501

D.8 <u>GRAFFIX</u> (Midlectron)

Self-contained Tektronix 4010 compatible graphics unit with 512 × 256 dot resolution.

D.9 <u>Kamplex Clinical Impedance Audiometer AZ7</u>

D.10 <u>Amplaid 207 Clinical Audiometer</u>

Pulsed tones at 2.5 Hz, 50% duty cycle, 40 ms rise and decay time.

D.11 <u>Barr & Stroud Variable Filter EF3</u>

Hatfield Instruments switched attenuator.

D.12 <u>Nagra IV-SJ Reel-to-reel Tape Recorder.</u>

D.13 <u>Racal Universal Counter 9835</u>

D.14 <u>Wavetek Synthesizer/Function Generator Model 171</u>

APPENDIX E

MASTER HEARING AID SPECIFICATIONS

The digital filter was programmed for the LP8 kHz response and the measurements made using the B & K audio test station and a Zwislocki coupler.

Maximum gain > 60 dB (limited by acoustic feedback)	
Distortion at maximum gain and 80 dB SPL input	
Swept 2nd harmonic	< 1% at all frequencies
Swept 3rd harmonic	"
Swept 2nd intermodulation (difference)	< 1% at all frequencies
Swept 3rd intermodulation (difference)	"

Signal-to-noise ratio	> 60 dB
Equivalent input noise	≃ 25 dB

Maximum undistorted sine wave output (monitored on oscilloscope)

Frequency	Output, dB SPL
125	122
250	122
500	126
1 k	140
2 k	142
4 k	140
6 k	122
8 k	116

APPENDIX F

B & K AUDIO TEST STATION INTERFACE

The circuit diagrams of the interface between the B & K audio test station and the Cromemco System 3 microcomputer are shown in Figure F.1. The electronic components were all contained on an S-100 bus circuit board which could be slotted in to the microcomputer S-100 bus. However, to allow ease of access for development, the board was housed in a separate case and was connected to the microcomputer 8PIO board by two 25-way ribbon cables and associated connectors. Output plugs C2 and C3 on the interface board were connected to 8PIO connectors J2 and J3, respectively. The interface board was connected to the audio test station by a 40-way ribbon cable (connector C1) to communicate digital information. A single screened lead (C4) was additionally used to transfer the analogue signal level from test point C on the audio test station generator and analyser board ZE0301 to the input of the A/D converter on the interface board.

The only modifications necessary to the circuit of the audio test station were the insertion of 10 kΩ resistors in series with the switches on the front panel circuits and the breaking of the tracks connecting the mode switches. These modifications, detailed in Figure F.2, ensured that computer control did not interfere with the normal manual operation of the test station.

Pins 1-17 and 26, 27 of connector C1 were connected directly to the appropriate points on the test station front panel circuits as shown in Figure F.2. The other pins of connector C1 were connected directly to the appropriate points on the test station generator and analyser board ZE031 and the logic board ZD0223, as shown in the audio test station service manual. These connections included the digitally coded frequency information (pins 18-25), the range shift signal (pin 39), the frequency change pulse (pin 35) and the power supply connections (pins 28, 36, 37, 38).

The 8-bit frequency and level information were multiplexed on to the same eight data lines (C2 pins 4-11) using the chip enable signal (C3 pin 17). Connector C3 pin 18 was connected to the initiate conversion pin of the A/D converter while pins 15 and 3 on connector C2 monitored the status of the conversion.

Analogue switches were used to automatically isolate the interface board from the audio test station when the computer was not in use.

Figure F.1

Audio Test Station interface

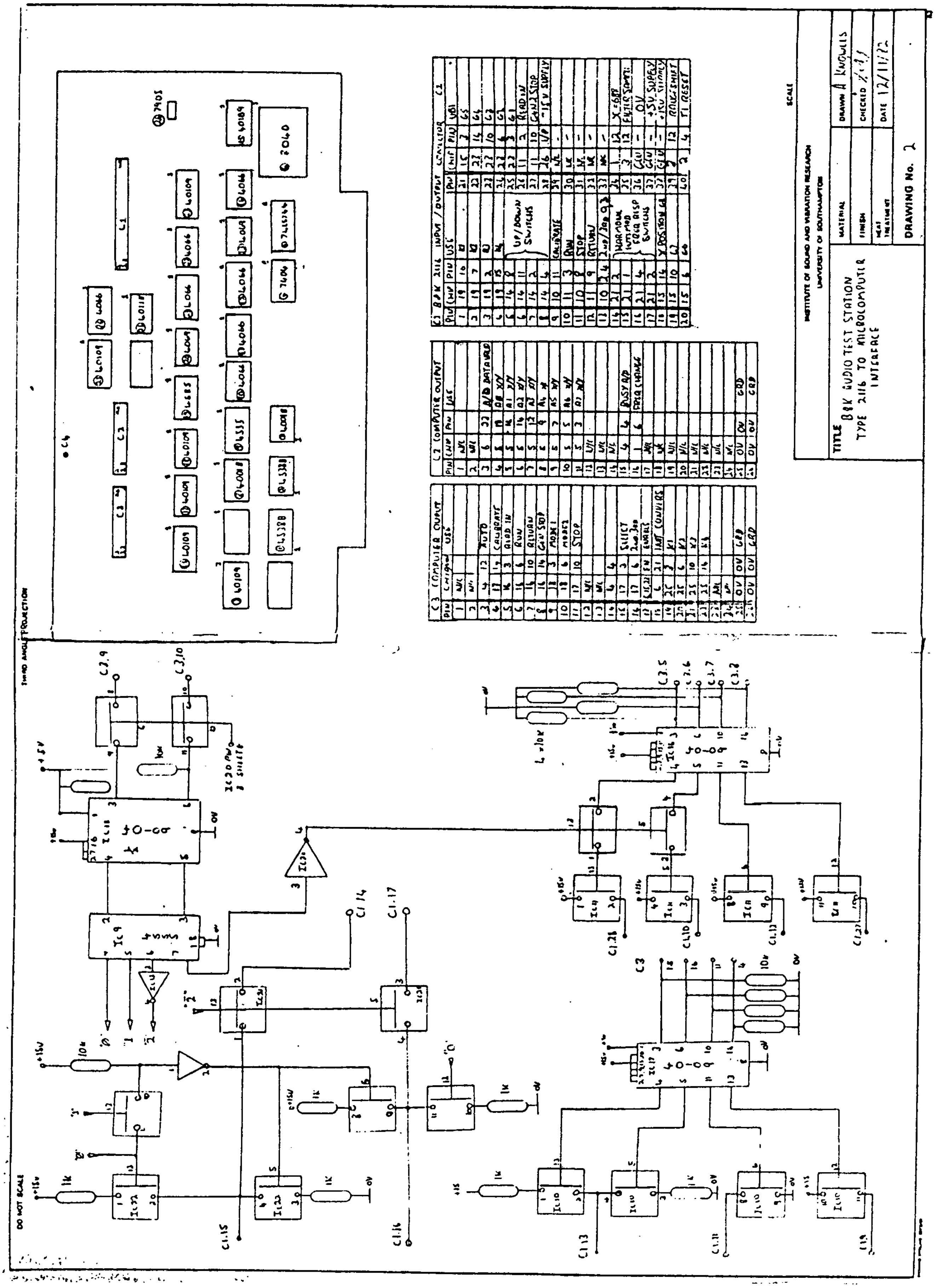


Figure F.1 cont.

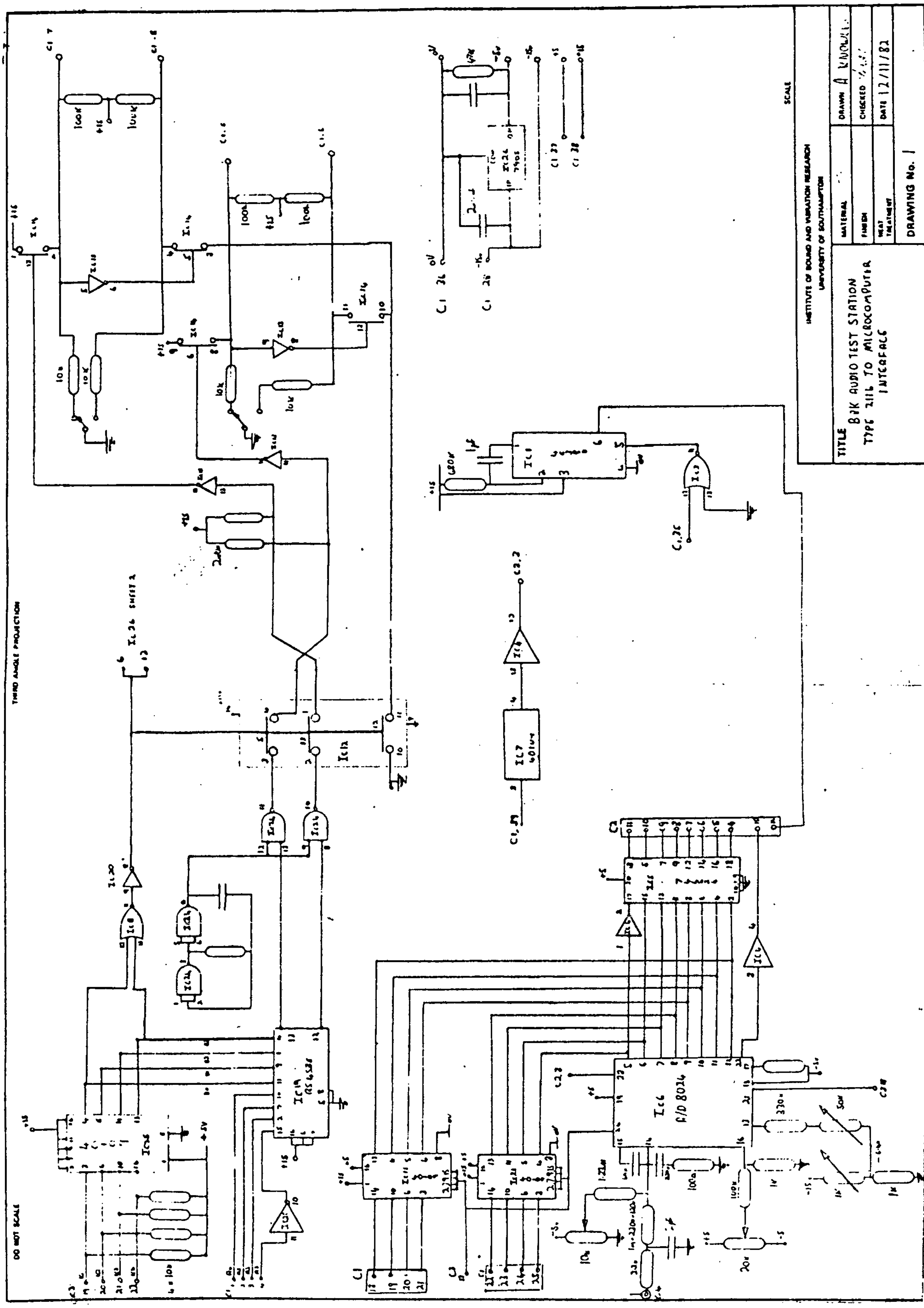


Figure F.2

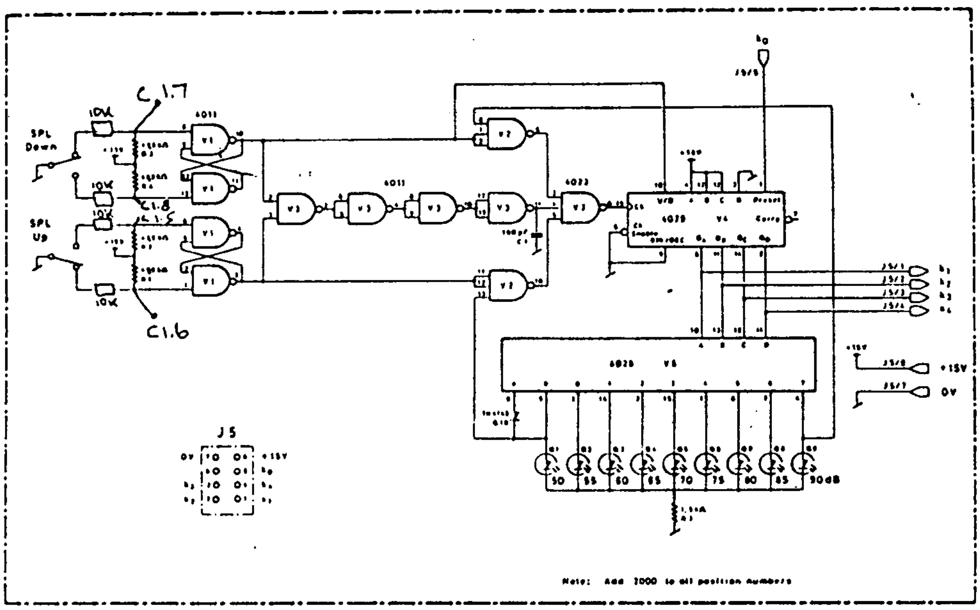
Audio Test Station

Front panel circuit modification

BRÜEL & KJÆR
Nærum - Denmark

Front Panel Circuit

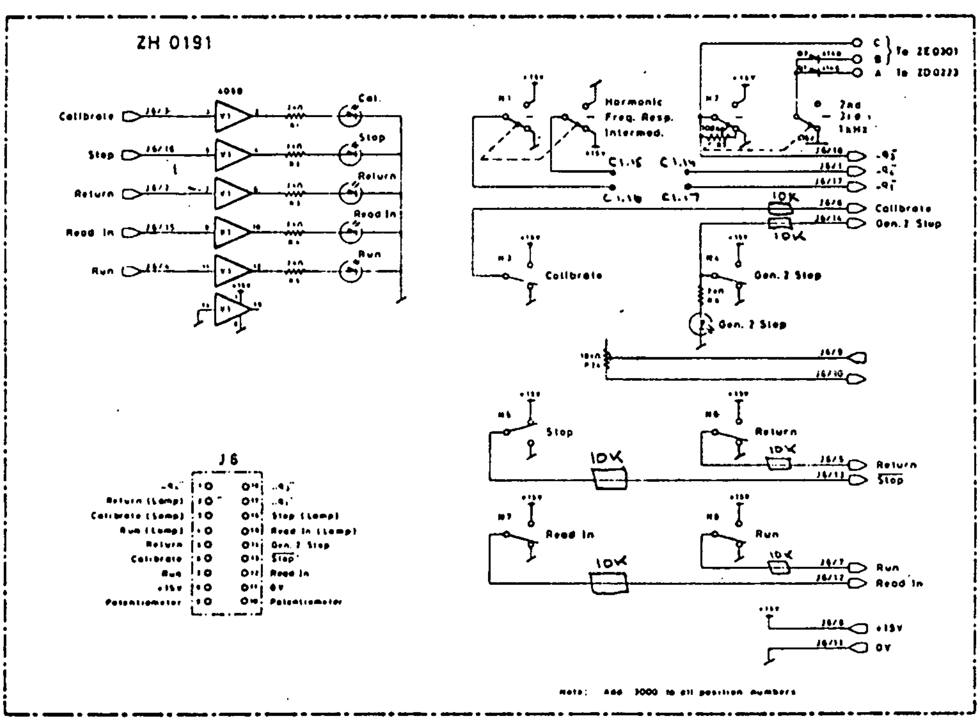
ZH 0190



BRÜEL & KJÆR
Nærum - Denmark

Front Panel Circuit

ZH 0191



APPENDIX G

COMPUTER PROGRAMS

G.1 The computer programs described in Chapter 8 are listed here with explanations in the comment statements.

PROGRAM CALCOR

Calibration and correction curve read in for B & K audio test station (8.3.2, 8.3.3).

PROGRAM BKRSDF

SUBROUTINE BKTSIF

SUBROUTINE ENCOD

Function and level selection (8.3.4): Data base software (8.4): Data base manipulation (8.5): Data collection (8.3.5): Graphics software (8.2): Digital filter aid simulation (8.8).

PROGRAM FLUPDT

Creates selection file UPDATFIL.DAT (8.6).

PROGRAM AUDATA

Creates audiometric data files (8.6).

PROGRAM MSADFT

SUBROUTINE BYPRSB

SUBROUTINE BGPRSB

SUBROUTINE BSTFTS

SUBROUTINE WRTBYR

SUBROUTINE WRTBER

Selects and writes out the 10 best aids for both Byrne and Berger's prediction methods (8.6).

PROGRAM DTAINT

Interpolates intermediate points between breakpoints for digital filter (8.7).

PROGRAM LCLINT

Calculates the normalized filter response required to present speech at the loudest comfortable level in each band (SBCL) from the attenuator settings in the comfortable level determination experiments (12.3.4.2).

PROGRAM PREFER

Calculates the normalized filter response required to present the 'chosen' response, by combining the SBCL response with the preferred slope and low and high pass cut-offs (12.3.4.3).

PROGRAM LOADAB

Loads digital filters with required responses and allows gain to be rescaled if required (8.7).

PROGRAM PAIRCM
SUBROUTINE BTIN

Controls digital filter and digital attenuator and responds to subject response box for pair comparison experiment (8.9).

PROGRAM PRCMSW

Similar to PROGRAM PAIRCM but also switches channels on tape recorder to allow hearing aid processed speech to be used for pair comparison (8.9).

PROGRAM PRCMFN

Creates file of filter names and scale values for pair comparison (8.9).

PROGRAM ULL3X
SUBROUTINE SUBULL

Uncomfortable listening level determination program (8.10).

PROGRAM MCL3X
SUBROUTINE SUBMCL

Comfortable listening level determination program (8.11).

PROGRAM C&ULFN

Creates files of filter names and scale values for programs ULL3X and MCL3X.

SUBROUTINE MKOPEN

File name formatting utility (8.12).

SUBROUTINE MSSG

Write to VDU screen utility (8.12).

G.2 Hearing Aid Data Base Structure

Hearing aid files

Each hearing aid response record consisted of 5 × 128 byte sections (Fortran 'records').

1st 128-byte record

1st byte	–	erase/update flag
2nd,3rd"	–	record no.
4th byte	–	test mode
5th "	–	date
6th "	–	test level
7th "	–	measurement method used
8th "	–	gain setting
9th "	–	bass cut setting
10th "	–	hi tone setting
11th "	–	peak clipping setting
12th "	–	compression setting
13th "	–	ear mould configuration

2nd and 3rd 128-byte records – comments

4th and 5th 128-byte records – 256 hearing aid response levels

Test name file

A list of the hearing aids stored in the data base and technical information on these aids is contained in the file TESTNAME.DAT.

Each record in this file contains the following information:

1st-12th bytes	Hearing aid file name
13th byte	Aid type
14th "	Compression present/absent/type
15th "	Peak clipping present/absent
16th "	Tone controls
17th "	Maximum gain control present/absent
18th "	Directional/non-directional microphone
19th "	Microphone position

Selection file

The gain and output information for each aid response that is necessary for the implementation of the various selection procedures is stored in the file UPDATFIL.DAT. Each record contains the following information:

1st-12th bytes	Hearing aid file name
13th-14th "	Record no. of gain data
15th-16th "	Record no. of maximum output data
17th-27th "	Maximum gains at 11 audiometric frequencies
28th-38th "	Maximum outputs at 11 audiometric frequencies

```
      PROGRAM CALCOR
C
C
C     M WALD MAR 1983
C
C
      BYTE FC
      BYTE PRNTC,XINITL,XTEST,XLEV,XPLOT
      INTEGER CALC,ERROR
      REAL*4 DATAIN(2,256)
      BYTE RETURN,RUNO,STOPO,INIT,ENABLE,RESET
      BYTE DATA(256),XPOS,CAL,Z
      BYTE SPACE,BUF(20),DSK
      BYTE GAL,CA(1),VE(1),VEC,BL(1),BLK,WH(1),WHT
      BYTE TVAL,ERA,YESNO(20),PNT,ES(1),ESC
      BYTE TVA(1),ER(1),PN(1),IO1(4),IO2(4),IO3(4)
      DIMENSION X(256),XA(256),Y(256),IXA(256)
      DATA TVAL/24/,ERA/25/,PNT/28/,VEC/29/,ESC/27/,BLK/127/
     1,IVDU/1/ ,CAL/31/,WHT/97/,SPACE/32/,IN2/7/
C
C     ENCODE GRAPHICS
C     ---------------
C
      ENCODE(TVA,6010)TVAL
      ENCODE(ER,6010)ERA
      ENCODE(PN,6010)PNT
      ENCODE(GA,6010)CAL
      ENCODE(VE,6010)VEC
      ENCODE(BL,6010)BLK
      ENCODE(WH,6010)WHT
      ENCODE(ES,6010)ESC
C
C     SET UP X FREQ ARRAY
C     -------------------
C
      CALL MSSG(IVDU,'CALCULATION OF FREQUENCY SPACING IN PROGRESS\')
      RINC=10**(2.0/255.0)
      DO 2 I=1,256
      X(I)=0.1*RINC**(I-1)
      IXA(I)=350.0+300*ALOG10(X(I)/0.1)
2     CONTINUE
C
C
C     SET OUTPUT PORTS 3 AND 4 ON 8PIO TO OUTPUT
C     -------------------------------------------
C
      CALL OUT(134,12)
C
C     ASK FOR CALIBRATE?
C     -----------------
C
5     CALL MSSG(IVDU,'CALIBRATE ? Y/N >\')
      READ(IVDU,1000)YESNO
      IF (YESNO(1).NE.89) GOTO 14
C
C     SEND CALIBRATE
C     --------------
C
      CALL OUT(130,1)
C
C     INITIATE CONVERSION
C     -------------------
C
      CALL OUT(131,8)
      Z=0
C
C     SET LOOP TIME
C     -------------
C
9     DO 13 I=1,10000
10    DO 11 J=1,100
11    CONTINUE
C
C     READ IN dB
C     ----------
C
      CAL=INP(132)
C
C     DON'T PRINT IF dB UNCHANGED
C     ---------------------------
C
      IF(CAL.EQ.Z)GOTO 12
      DB=CAL
C
C     CONVERT FROM 2'S COMPLEMENT
C     ---------------------------
C
      DB=140-(DB+103)/2
      WRITE(IVDU,2000)DB
12    Z=CAL
13    CONTINUE
      CALL MSSG(IVDU,'FINISHED CALIBRATION YET ? Y/N >\')
      READ(IVDU,1000)YESNO
      IF(YESNO(1).EQ.78)GOTO 9
14    CALL OUT(130,8)
      CALL MSSG(IVDU,'READ IN CORRECTION CURVE ? Y/N >\')
      READ(IVDU,1000)YESNO
      IF (YESNO(1).EQ.89)GOTO 15
       CALL MSSG(IVDU,'CHECK 90 Db LEVEL ? Y/N >\')
       READ(IVDU,1000)YESNO
       IF(YESNO(1).EQ.89)GOTO 48
       GOTO 5
C
C     SEND ENABLE X
C     -------------
C
15    CALL OUT(131,252)
C
C     SEND READ IN
C     ------------
C
      CALL OUT(130,6)
C
```

```
C       LOOK FOR X=0 START OF MEASURING RANGE
C       ------------------------------------
C
        Z=99
        DO 18 I=1,200
17      XPOS=INP(132)
        IF((XPOS.EQ.-128).AND.(I.GT.10))GOTO 19
        IF(XPOS.EQ.Z)GOTO 17
        IXPOS=XPOS
        IXPOS=IXPOS+128
18      Z=XPOS
19      CONTINUE
        CALL MSSG(IVDU,'CORRECTION CURVE BEING READ IN\')
        DO 40 I=1,1000
        CALL OUT(131,252)
20      XPOS=INP(132)
C
C       LOOK FOR END OF READ IN
C       ----------------------
C
        IF((XPOS.EQ.-128).AND.(I.GT.10))GOTO 45
        IXPOS=XPOS
        IXPOS=IXPOS+128
21      IF(IXPOS.LT.I-1)GOTO 20
        IF(IXPOS.EQ.I-1)GOTO 22
22      CONTINUE
C
C       INITIATE CONVERSION
C       ------------------
C
        CALL OUT(131,248)
        Z=0
C
C       READ IN dB
C       ----------
C
23      CAL=INP(132)
C
C       DON'T PRINT IF dB CHANGED
C       -------------------------
C
        IF(CAL.EQ.Z)GOTO 25
        Z=CAL
        DO 24 J=1,200
24      CONTINUE
        GOTO 23
25      DB=CAL
C
C       CONVERT FROM 2'S COMPLEMENT
C       -------------------------
C
        DB=140-(DB+103)/2
40      CONTINUE
45      CONTINUE
        CALL OUT(130,8)
C
C       INITIALIZE Y ARRAY
C       -----------------
C
48      DO 50 I=1,256
50      Y(I)=0.0
C
C       SET NORMAL X START POS
C       ---------------------
C
        IXBGIN=0
C
          XTEST=1
51      IMODE=64
        IORDER=0
C
C        LEVEL
C       ------
C
          XLEV=90
60      XLEVL=((XLEV-55)/5)*16-128
65      CONTINUE
C
C         INITIALISE COMMANDS
C         ------------------
C
        RUNO=4+IMODE
        STOPO=IMODE-128
        RETURN=8
        INIT=IORDER+XLEVL+8
C
C         CLEAR GRAFFIX SCREEN
C         -------------------
C
        WRITE(IVDU,6010)PN,ER
C
C       PLOT AXES
C       ---------
C
C
C       HORIZONTAL
C       ----------
C
        DO 74 IY=250,682,36
         IX1=350
         IX2=950
         CALL ENCOD(IX1,IY,IO1)
         CALL ENCOD(IX2,IY,IO2)
         WRITE(IVDU,2010)VE,IO1,IO2
74       CONTINUE
C
C       VERTICAL
C       --------
C
         DO 76  I=1,7
         X1=0.0625*2**I
C
C       LOG SPACING
```

```
C          -----------
C
            IX=350.0+300*ALOG10(X1/0.1)
            IY1=250
            IY2=682
            CALL ENCOD(IX,IY1,IO1)
            CALL ENCOD(IX,IY2,IO2)
            WRITE(IVDU,2010)VE,IO1,IO2
            IF(I.GT.3)GOTO 75
            IX=IX-50
            CALL ENCOD(IX,230,IO3)
C
C          LABELS
C          ------
C
            WRITE(IVDU,3010)PN,ES,BL,IO3,ES,WH,GA,X1
            GOTO 76
75           IX=IX-7
            CALL ENCOD(IX,230,IO3)
            IX1=X1
            WRITE(IVDU,3500)PN,ES,BL,IO3,ES,WH,GA,IX1
76           CONTINUE
C
C          LEFT AND RIGHT 'UPRIGHTS'
C          -------------------------
C
            CALL ENCOD(350,250,IO1)
            CALL ENCOD(350,682,IO2)
            WRITE(IVDU,2010)VE,IO1,IO2
            CALL ENCOD(950,250,IO1)
            CALL ENCOD(950,682,IO2)
            WRITE(IVDU,2010)VE,IO1,IO2
C
C          Y LABELS
C          --------
C
            DO 77 I=1,13
           IY=200+36*I
           IY1=20+10*I
           CALL ENCOD(290,IY,IO1)
77          WRITE(IVDU,10000)PN,ES,BL,IO1,ES,WH,GA,IY1
C
C          'Db'
C          ----
           CALL ENCOD(290,695,IO1)
           WRITE(IVDU,11010)PN,ES,BL,IO1,ES,WH,GA
C
C          'KHz'
C          -----
C
           CALL ENCOD(960,230,IO1)
           WRITE(IVDU,12000)PN,ES,BL,IO1,ES,WH,GA
           WRITE(1,6010)TVA
C
C            FREE RUN A/D
C            ------------
C
```

```
           ENABLE=IORDER+XLEVL+12
C
C           SEND ENABLE X
C           -------------
C
            CALL OUT(131,ENABLE)
C
             CALL MSSG(IVDU,'CHECK CORRECTION CURVE READ IN OK\')
           CALL MSSG(IVDU,'TO START PRESS ANY  KEY EXCEPT T\')
           CALL MSSG(IVDU,'PRESS T TO TERMINATE PROGRAMME----\')
C          -------------
           PAUSE
C          -------------
C           SEND RUN
C           --------
C
            CALL OUT(130,RUNO)
C
C           LOOK FOR X=0 START OF MEASURING RANGE
C           -------------------------------------
C
C
C          LOOK FOR FREQ CHANGE BY MASKING OUT LATCH INP(133)
C          TO ENSURE ACTUALLY STARTED
C          --------------------------------------------------
C
100        IF(.NOT.(INP(134).OR.-33))GOTO 110
           GOTO 100
110        CONTINUE
            Z=99
            DO 145 I=1,200
120         XPOS=INP(132)
            IF((XPOS.EQ.-128).AND.(I.GT.10))GOTO 150
            IF(XPOS.EQ.Z)GOTO 120
            IXPOS=XPOS
            IXPOS=IXPOS+128
            IF((IXPOS.NE.90).AND.(IXPOS.NE.120))GOTO 145
C
C           INITIATE CONVERSION
C           -------------------
C
            CALL OUT(131,INIT)
C
C           READ IN dB
C           ----------
C
            CAL=INP(132)
            DB=CAL
C
C           CONVERT FROM 2'S COMPLEMENT
C           ---------------------------
C
C
C          CHECK LEVEL
C          -----------
C
            IF(IXPOS.EQ.120)GOTO 122
```

```
      IDB=5*IFIX((7-DB)/12)+50
      WRITE(IVDU,5500)IDB
      IF(IDB.EQ.XLEV)GOTO 121
      CALL MSSG(IVDU,'LEVEL ERROR\')
      CALL OUT(130,RETURN)
      GOTO 5
121   GOTO 140
C
C     CHECK MODE
C     ----------
C
122   ITS=(CAL+7)/17
      GOTO(125,127,129,131,133),ITS
125   CALL MSSG(IVDU,'FREQ RESP\')
      GOTO 135
127   CALL MSSG(IVDU,'2nd HARMONIC\')
      GOTO 135
129   CALL MSSG(IVDU,'3rd HARMONIC\')
      GOTO 135
131   CALL MSSG(IVDU,'2nd INTERMOD\')
      GOTO 135
133   CALL MSSG(IVDU,'3rd INTERMOD\')
C
C      SEND STOP IN TIME AS HAS NO EFFECT IN PARAMETER RANGE
C      -----------------------------------------------------
C
       CALL OUT(130,STOPO)
C
135   IF(ITS.EQ.XTEST)GOTO 140
      CALL MSSG(IVDU,'TEST MODE ERROR\')
      CALL OUT(130,RETURN)
      GOTO 5
140   CALL OUT(131,ENABLE)
145   Z=XPOS
150   CONTINUE
      CALL OUT(130,STOPO)
C
C     IF INTMOD GET XPOS CORRECT
C     --------------------------
C
      IF(IMODE.NE.32)GOTO 167
165   XPOS=INP(132)
      IF(XPOS.NE.(IXBGIN-128))GOTO 165
C
C      ADD 1 AS LOOP INDEX STARTS AT 1 NOT 0
C      -------------------------------------
C
167   IXBGIN=IXBGIN+1
C
C      REMOVE FREE RUN A/D FROM ENABLE
C      -------------------------------
C
      ENABLE=IORDER+XLEVL+4
C
C      SET ERROR COUNT TO 0
C      --------------------
C
      ERROR=0
C
168   DO 400 I=IXBGIN,256
C
C      IF FIRST POSITION OMIT FREQ CHANGE CHECK
C      ----------------------------------------
C
      IF(I.EQ.IXBGIN)GOTO 175
C
      CALL OUT(131,ENABLE)
      RESET=INP(133)
      CALL OUT(130,RUNO)
C
C     LOOK FOR FREQ CHANGE BY MASKING OUT LATCH INP(133)
C     --------------------------------------------------
C
170   IF(.NOT.(INP(134).OR.-33))GOTO 175
      GOTO 170
C
C
C      STOP
C      ----
C
175   CALL OUT(130,STOPO)
C
C      PLOT PREVIOUS POINT
C      -------------------
C
      IY=250.0+3.6*(DB-30.0)
      N=I-1
      CALL ENCOD(IXA(N),IY,IO1)
      WRITE(IVDU,2010)PN,IO1,GA,TVA
C
C      INPUT NEW X POS
C      ---------------
C
      XPOS=INP(132)
176   IXPOS=XPOS
      IXPOS=IXPOS+128
C
C
C     CHECK FOR POSITION ERROR
C     ------------------------
C
      IF((XPOS.EQ.-128).AND.(I.GT.10))GOTO 180
      IF(IXPOS.EQ.I-1)GOTO 180
      CALL MSSG(IVDU,'X POS ERROR\')
C
C      WRITE FREQ
C      ----------
C
180   WRITE(IVDU,14000)X(I)
C
C     CHECK FOR RANGE SHIFT IN PROGRESS BY MASKING LATCH INP(130)
C     -----------------------------------------------------------
C
      RESET=INP(130)
```

```
185   IF(.NOT.(INP(134).OR.-5))GOTO 187
        GOTO 190
187     CONTINUE
      RESET=INP(130)
      GOTO 185
C
C      INITIATE CONVERSION
C      -------------------
C
190    CALL OUT(131,INIT)
       Z=-128
C
C       WASTE TIME TO ALLOW LMS DETECTOR IN B&K TO SETTLE
C       ----------------------------------------------
C
191     DO 192 N=1,500
192     CONTINUE
C
C     CHECK IF BUSY BY MASKING LATCH INP(131)
C     ---------------------------------------
C
194     RESET=INP(131)
195   IF(.NOT.(INP(134).OR.-9))GOTO 200
      GOTO 195
C
C      READ IN dB
C      ----------
C
200    CAL=INP(132)
C
C      DON'T PRINT IF dB CHANGED
C      -------------------------
C
       IF(CAL.EQ.Z)GOTO 250
       Z=CAL
C
C       WASTE TIME TO ALLOW B&K RMS DETECTOR TO CHANGE
C       ---------------------------------------------
C
240     DO 245 N=1,500
245     CONTINUE
       GOTO 194
250    DB=CAL
C
C      CONVERT FROM 2'S COMPLEMENT
C      ---------------------------
C
       DB=140-(DB+103)/2
       Y(I)=DB
        WRITE(IVDU,2000)DB
C
C      LOOK FOR END OF RUN
C      -------------------
C
       IF((XPOS.EQ.-128).AND.(I.GT.10))GOTO 450
C
C       CHECK IF 90dB
C       -------------
C
         IF((Y(I).GT.91.5).OR.(Y(I).LT.88.5))ERROR=ERROR+1
400     CONTINUE
450     CONTINUE
        CALL OUT(130,RETURN)
         WRITE(IVDU,2000)(Y(I),I=1,256)
         IF(ERROR.NE.0)GOTO 460
         CALL MSSG(IVDU,'CORRECTION CURVE OK\')
         GOTO 5
460      WRITE(IVDU,13000)ERROR
         GOTO 5
1000    FORMAT(20A1)
1020   FORMAT(1H ,10X,F6.3,7X,F6.1)
2000    FORMAT(1X,F5.1,'dB')
2010    FORMAT(9A1)
3000    FORMAT(1X,I5)
3010    FORMAT(10A1,F5.3)
3500    FORMAT(10A1,I1)
4000   FORMAT(F12.0)
5000   FORMAT(1X,2F12.4)
5500   FORMAT(I2,'Db')
6000   FORMAT(1X,F12.4)
6010    FORMAT(2A1)
7000   FORMAT(A1)
8000   FORMAT(1X,F6.3,',',F6.1)
9000   FORMAT(2F12.0)
10000  FORMAT(10A1,I3)
10010  FORMAT(I2)
11000  FORMAT(I1)
11010  FORMAT(10A1,'dB')
12000  FORMAT(10A1,'kHz')
13000    FORMAT(1X,I3,'ERROR(S)')
14000    FORMAT(1X,F6.3,'KHz')
        END
        SUBROUTINE ENCOD(IX,IY,IO)
C
C        THIS SUBROUTINE ENCODES X AND Y CORDINATES INTO GRAFFIX CODES
C        -----------------------------------------------------------
C
        BYTE IO(4)
        IC=IX/32+32
        ID=IX-32*(IC-34)
        IA=IY/32+32
        IB=IY-32*(IA-35)
        ENCODE(IO,5000)IA,IB,IC,ID
        RETURN
5000    FORMAT(4A1)
        END
```

```
      PROGRAM BKRSDF
C
C     M WALD JUNE 1983
C     ----------------
C
C
C     THIS PROGRAM FINDS DIFFERENCE BETWEEN TWO RESPONSES
C     ---------------------------------------------------
C
      IMPLICIT BYTE(X)
       BYTE PRNTG
       BYTE READAL
       BYTE SEEREC
       BYTE REWR
       BYTE SPACE
       BYTE DAT(20)
       BYTE DATE(20)
       BYTE DATARA(256),DATARB(256),DATADF(256)
       BYTE DATAI(256)
       BYTE DATAO(256)
       BYTE YESNO(3)
       BYTE BUF(20)
       BYTE COM(240),TYPE,AVC,PC,TONE,MAXCC,MICDIR,MICPOS
       BYTE COMM(240),FILE(12),ZFILE(11)
       BYTE ENDTS,DISC,NEWFIL
       BYTE TVA(1),TVAL,CA(1),CAL,PNT,PN(1)
       DATA SPACE/32/,  IVDU/1/, IN/6/, DISC/0/,   XERAS/2/, XNEW/0/
     1 ,ZFILE/11*32/,TVAL/24/,CAL/31/,PNT/28/
       ENCODE(TVA,10000)TVAL
       ENCODE(CA,10000)CAL
       ENCODE(PN,10000)PNT
C
C      SET INITL FLAG
C      ------------
C
       XINITL=0
       GOTO 34
5     IF(XDIF.EQ.0)GOTO 17
      IF(XDIFB.EQ.0)GOTO 28
C
C     COMPUTE DIFERENCE BETWEEN RESPONSES AND PLOT
C     --------------------------------------------
C
        IF(XINV.EQ.1)GOTO 9
      CALL MSSG(IVDU,'DO YOU WANT  A-B (1) OR  B-A (2)     1/2 > \')
      READ(IVDU,10010)XAB
      IF(XAB.EQ.2)GOTO 7
      DO 6 I=1,256
6     DATADF(I)=DATARA(I)-DATARB(I)
      GOTO 11
7     DO 8 I=1,256
8     DATADF(I)=DATARB(I)-DATARA(I)
      GOTO 11
C       INVERSE
C       -------
C
9       DO 10 I=1,256
10      DATADF(I)=0-DATARB(I)
C
C     PLOT DIFF
C     ---------
C
11    XPLOT=1
      CALL BKTSIF(PRNTG,XINITL,DATADF,XTEST,XLEV,XPLOT)
        WRITE(IVDU,10000)TVA
        CALL MSSG(IVDU,'RESCALE ? Y/N > \')
        READ(IVDU,10000)YESNO
        IF(YESNO(1).NE.89)GOTO 13
        CALL MSSG(IVDU,'RESCALE VALUE ? > \')
        READ(IVDU,10750)IRSCAL
        DO 12 I=1,256
12      DATADF(I)=DATADF(I)-2*(IRSCAL)
        GOTO 11
13      CALL MSSG(IVDU,'DO YOU WANT INVERSE RESPONSE OF DIFF?\')
        CALL MSSG(IVDU,'Y/N >\')
        READ(IVDU,10000)YESNO
        IF(YESNO(1).NE.89)GOTO 34
C       INVERSE
C       -------
C
        DO 14 I=1,256
14      DATADF(I)=0-DATADF(I)
        GOTO 11
C
C       TRANSFER READ OR WRITTEN DATA FOR RESCALING
C       -------------------------------------------
C
17      IF(XREAD)GOTO 19
        DO 18 I=1,256
18      DATADF(I)=DATAI(I)
        GOTO 24
19      DO 20 I=1,256
20      DATADF(I)=DATAO(I)
C
C       CHECK IF RESCALE OF NORMAL PLOT REQUIRED
C       ----------------------------------------
C
24      CALL MSSG(IVDU,'RESCALE FOR FILTER ?\')
        READ(IVDU,10000)YESNO
        IF(YESNO(1).NE.89)GOTO 34
        CALL MSSG(IVDU,'RESCALE VALUE ? > \')
        READ(IVDU,10750)IRSCAL
        DO 25 I=1,256
25      DATADF(I)=DATADF(I)-2*(IRSCAL)
      XPLOT=1
      CALL BKTSIF(PRNTG,XINITL,DATADF,XTEST,XLEV,XPLOT)
        WRITE(IVDU,10000)TVA
        GOTO 24
C
C       RESET FLAG FOR RECORD B
C       -----------------------
C
28      XDIFB=1
```

```
      GOTO 35
C
C     CHECK IF DIFF BETWEEN RESPONSES REQUIRED
C     --------------------------------------
C
34    CALL MSSG(IVDU,'DO YOU WANT DIFFERENCE BETWEEN RESPONSES?\')
      CALL MSSG(IVDU,'Y/N >\')
      READ(IVDU,10000)YESNO
C
C     SET DIFF FLAG
C     ------------
C
      XDIF=0
      IF(YESNO(1).EQ.89)XDIF=1
      IF(XDIF.EQ.0)GOTO 36
      XDIFB=0
        CALL MSSG(IVDU,'DO YOU WANT INVERSE RESPONSE ?\')
        CALL MSSG(IVDU,'Y/N >\')
        READ(IVDU,10000)YESNO
        XINV=0
        IF(YESNO(1).EQ.89)XINV=1
        IF(XINV.EQ.1)XDIFB=1
35    IF(XDIFB.EQ.0)CALL MSSG(IVDU,'RECORD A\')
      IF(XDIFB.EQ.1)CALL MSSG(IVDU,'RECORD B\')
C
C       CHECK IF ANOTHER RECORD REQUIRED
C       ----------------------------
C
36      IF(XINITL.EQ.0)GOTO 37
       CALL MSSG(IVDU,'ANOTHER RECORD FOR THIS AID ? >\')
       READ(IVDU,10000)YESNO
      XANREC=0
       IF (YESNO(1).EQ.89) XANREC=1
       ENDFILE IN
      IF(XANREC.EQ.1)GOTO 80
C
C     ASK IF WANT TO SEE ALL AID NAMES
C     -------------------------------
C
37     CALL MSSG(IVDU,'DO YOU WANT TO SEE ALL AIDNAMES IN FILE ?\')
      CALL MSSG(IVDU,'Y/N >\')
      XREADA=0
      READ(IVDU,10000)YESNO
      IF(YESNO(1).NE.89)GOTO 38
      XREADA=1
      GOTO 40
C
C      ASK FOR HAID FILENAME
C      -------------------
C
38     CALL MSSG(IVDU,'HEARING AID ? >\')
       READ(IVDU,10000)BUF
       IF (BUF(1).EQ.SPACE)STOP
C
C     OPEN FILE OF AIDNAMES
C     --------------------
C
40    CALL OPEN(IN,'TESTNAMEDAT',0)
C
C     SET NEW FILE AND SPACE FLAGS
C     --------------------------
C
      NEWFIL=0
      K=0
C
C     GO TO START OF FILE AND FIND FIRST SPACE IN RECORD
C     --------------------------------------------------
C
      DO 55 I=1,1000
      READ(IN,REC=1,ERR=65,END=58)FILE,TYPE,AVC,PC,TONE,MAXGC,MICDIR
     1,MICPOS
      IF(XREADA.NE.1)GOTO 42
      WRITE(IVDU,12000)FILE,TYPE,AVC,PC,TONE,MAXGC,MICDIR,MICPOS
      GOTO 55
42    IF(FILE(1).NE.SPACE)GOTO 45
      IF(K.EQ.0)K=I
      GOTO 55
C
C     CHECK IF FILENAME ALREADY EXISTS
C     ------------------------------
C
45    DO 50 J=1,12
      IF(BUF(J).NE.FILE(J))GOTO 55
50    CONTINUE
      GOTO 70
55    CONTINUE
58    IF(XREADA.NE.1)GOTO 60
      ENDFILE IN
C
C     RESET READ ALL FLAG
C     ------------------
C
      XREADA=0
      GOTO 18
60    NEWFIL=1
      CALL MSSG(IVDU,'NEW FILE\')
      CALL MSSG(IVDU,'--------\')
      CALL MSSG(IVDU,' \')
C
C     CHECK IF REQUIRE AID NAME IN AID NAME FILE
C     ----------------------------------------
C
      CALL MSSG(IVDU,'ENTER AID NAME IN "TESTNAME.DAT" FILE ? Y/N>\')
      READ(IVDU,10000)YESNO
      IF(YESNO(1).NE.89)GOTO 70
C
      CALL MSSG(IVDU,'WHAT TYPE OF AID IS IT? \')
      CALL MSSG(IVDU,'POST AURAL > 1\')
      CALL MSSG(IVDU,'BODY       > 2\')
      CALL MSSG(IVDU,'IN THE EAR > 3\')
      CALL MSSG(IVDU,'           > \')
      READ(IVDU,10750)TYPE
      CALL MSSG(IVDU,'HAS AID GOT AVC? \')
      CALL MSSG(IVDU,'NO              > 0\')
```

```
      CALL MSSC(IVDU,'INPUT       > 1\')
      CALL MSSC(IVDU,'OUTPUT      > 2\')
      CALL MSSC(IVDU,'    > \')
      READ(IVDU,10750)AVC
      CALL MSSC(IVDU,'HAS AID GOT PEAK CLIPPING? \')
      CALL MSSC(IVDU,'NO  > 0\')
      CALL MSSC(IVDU,'YES > 1\')
      CALL MSSC(IVDU,'    > \')
      READ(IVDU,10750)PC
      CALL MSSC(IVDU,'HAS AID GOT TONE CONTROL? \')
      CALL MSSC(IVDU,'NO                  > 0\')
      CALL MSSC(IVDU,'BASS CUT ONLY       > 1\')
      CALL MSSC(IVDU,'HIGH CUT ONLY       > 2\')
      CALL MSSC(IVDU,'BASS AND HIGH CUT > 3\')
      CALL MSSC(IVDU,'    > \')
      READ(IVDU,10750)TONE
      CALL MSSC(IVDU,'HAS AID GOT MAX GAIN CONTROL? \')
      CALL MSSC(IVDU,'NO  > 0\')
      CALL MSSC(IVDU,'YES > 1\')
      CALL MSSC(IVDU,'    > \')
      READ(IVDU,10750)MAXGC
      CALL MSSC(IVDU,'HAS AID GOT A DIRECTIONAL MICROPHONE? \')
      CALL MSSC(IVDU,'NO  > 0\')
      CALL MSSC(IVDU,'YES > 1\')
      CALL MSSC(IVDU,'    > \')
      READ(IVDU,10750)MICDIR
      CALL MSSC(IVDU,'HAS AID GOT A FF OR DF MICROPHONE? \')
      CALL MSSC(IVDU,'FORWARD  FACING      > 1\')
      CALL MSSC(IVDU,'DOWNWARD FACING      > 2\')
      CALL MSSC(IVDU,'    > \')
      READ(IVDU,10750)MICPOS
C
C     WRITE IN NEW FILENAME
C     --------------------
C
      DO 62 J=1,12
62    FILE(J)=BUF(J)
      IF(K.EQ.0)K=I
      WRITE(IN,REC=K,ERR=65,END=70)FILE,TYPE,AVC,PC,TONE,MAXGC
     1,MICDIR,MICPOS
      GOTO 70
65    CALL MSSC(IVDU,'READ/WRITE ERROR\')
      ENDFILE IN
      GOTO 17
70    ENDFILE IN
      CALL MKOPEN(IN,BUF,DISC)
C
C      ASK IF READ OR WRITE?
C      -------------------
C
80     CALL MSSC(IVDU,'READ OR WRITE ? (R/W) >\')
       READ(IVDU,10000)REWR
C
C      SET AND TEST READ FLAGS
C      ---------------------
C
       READADL=0
       XCHECK=1
       XREAD=0
       IREC=0
       PRNTG=0
       IF (REWR.EQ.82)XREAD=1
       IF(XREAD.NE.1)GOTO 85
C
C     CHECK IF DATA REQUIRED TO  BE SEEN ?
C     ------------------------------------
C
      CALL MSSC(IVDU,'DO YOU WISH TO SEE DATA PLOT AS WELL ? Y/N>\')
      IPLOT=0
      READ(IVDU,10000)YESNO
      IF(YESNO(1).EQ.89)IPLOT=1
C
C     IF REQUIRE DIFF THEN CANNOT READ ALL REC
C     ----------------------------------------
C
      IF(XDIF.EQ.1)GOTO 85
       CALL MSSC(IVDU,'DO YOU WANT TO SEE ALL RECORDS ? >\')
       READ(IVDU,10000)READAL
       IF(READAL.EQ.89)GOTO 970
85     CALL MSSC(IVDU,'DO YOU KNOW RECORD No ? (Y/N) >\')
       READ(IVDU,10000)YESNO
       IF(YESNO(1).NE.89)GOTO 90
       CALL MSSC(IVDU,'WHAT IS RECORD No ? >\')
       READ(IVDU,10750)IREC
       IF(XREAD.NE.1)GOTO 90
       GOTO 970
C      ASK FOR TEST
C      -----------
C
90     CALL MSSC(IVDU,'TEST REQUIRED ? \')
       CALL MSSC(IVDU,'FREQUENCY RESPONSE-                  > 1\')
       CALL MSSC(IVDU,'2nd HARMONIC DISTORTION-             > 2\')
       CALL MSSC(IVDU,'3rd HARMONIC DISTORTION-             > 3\')
       CALL MSSC(IVDU,'2nd INTERMODULATION DISTORTION-      > 4\')
       CALL MSSC(IVDU,'3rd INTERMODULATION DISTORTION-      > 5\')
       CALL MSSC(IVDU,'   >\')
       READ(IVDU,10010)XTEST
C
C
C      ASK FOR INPUT LEVEL
C      -----------------
C
200    CALL MSSC(IVDU,'INPUT LEVEL ? >  \')
       READ(IVDU,10010)XLEV
C
C      ASK FOR COUPLER
C      --------------
C
300    CALL MSSC(IVDU,'COUPLER/KEMAR ? \')
       CALL MSSC(IVDU,'ZWISLOCKI COUPLER-      >  1\')
       CALL MSSC(IVDU,'2cc COUPLER-            >  2\')
       CALL MSSC(IVDU,'B&K EAR SIMULATOR-      >  3\')
       CALL MSSC(IVDU,'KEMAR 0  deg incident   >  4\')
       CALL MSSC(IVDU,'KEMAR 90 deg incident   >  5\')
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```
      CALL MSSC(IVDU,'KEMAR 180deg incident  >  6\')
      CALL MSSC(IVDU,'KEMAR 270deg incident  >  7\')
      CALL MSSC(IVDU,'   >\')
      READ(IVDU,10010)XCOUP
C
C     ASK FOR CAIN
C     -----------
C
400   CALL MSSC(IVDU,'CAIN ? \')
      CALL MSSC(IVDU,'MAX-      > 1\')
      CALL MSSC(IVDU,'MIN-      > 2\')
      CALL MSSC(IVDU,'USER-     > 3\')
      CALL MSSC(IVDU,'   >\')
      READ(IVDU,10010)XCAIN
C
C     ASK FOR BASS CUT
C     ---------------
C
     XBASC=0
     IF(TONE.EQ.0)COTO 700
     IF(TONE.EQ.2)COTO 600
500   CALL MSSC(IVDU,'BASS CUT ? \')
      CALL MSSC(IVDU,'NONE-     > 0\')
      CALL MSSC(IVDU,'MAX -     > 1\')
      CALL MSSC(IVDU,'USER-     > 2\')
      CALL MSSC(IVDU,'   >\')
      READ(IVDU,10010)XBASC
C
C     ASK FOR HIGH CUT
C     ---------------
C
     XHIC=0
     IF(TONE.EQ.1)COTO 700
600   CALL MSSC(IVDU,'HIGH CUT ? \')
      CALL MSSC(IVDU,'NONE-     > 0\')
      CALL MSSC(IVDU,'MAX-      > 1\')
      CALL MSSC(IVDU,'USER-     > 2\')
      CALL MSSC(IVDU,'   >\')
      READ(IVDU,10010)XHIC
C
C     ASK FOR PEAK CLIP
C     ----------------
C
700   XPCL=0
      IF(PC.EQ.0)COTO 800
750   CALL MSSC(IVDU,'PEAK CLIPPING ? \')
      CALL MSSC(IVDU,'NONE-     > 0\')
      CALL MSSC(IVDU,'MAX-      > 1\')
      CALL MSSC(IVDU,'USER-     > 2\')
      CALL MSSC(IVDU,'   >\')
      READ(IVDU,10010)XPCL
C
C     ASK FOR COMPRESSION
C     ------------------
C
800   XCOM=0
      IF(AVC.EQ.0)COTO 900
850   CALL MSSC(IVDU,'COMRESSION ? \')
      CALL MSSC(IVDU,'NONE-     > 0\')
      CALL MSSC(IVDU,'MAX-      > 1\')
      CALL MSSC(IVDU,'USER-     > 2\')
      CALL MSSC(IVDU,'   >\')
      READ(IVDU,10010)XCOM
C
C     ASK FOR EARMOULD PLUMBINC
C     ------------------------
C
900   CALL MSSC(IVDU,'EARMOULD/PLUMBINC ? \')
      CALL MSSC(IVDU,'STANDARD TUBINC-      > 1\')
      CALL MSSC(IVDU,'ALTERNATE PLUMBINC-   > 2,3,4,5,6,7,8,9\')
      CALL MSSC(IVDU,'   >\')
      READ(IVDU,10010)XEARM
C
C     ASK FOR DATE
C     -----------
C
950   CALL MSSC(IVDU,'DATE ?  (DAY/MONTH/YEAR)  >\')
      READ(IVDU,10000)DATE
C
C     ASK FOR ANY COMMENTS
C     -------------------
C
      CALL MSSC(IVDU,'ANY COMMENTS ? (up to 3 lines) > \')
     WRITE(IVDU,10040)
      READ(IVDU,10020)COMM
C
C     READ BACK INFO. TO CHECK IF CORRECT
C     -----------------------------------
C
      IRE=0
      XTES=XTEST
      DO 955 I=1,20
955   DAT(I)=DATE(I)
      XLE=XLEV
      XCOU=XCOUP
      XCAI=XCAIN
      XBAS=XBASC
      XHI=XHIC
      XPC=XPCL
      XCO=XCOM
      XEAR=XEARM
      DO 956 I=1,240
956   COM(I)=COMM(I)
      COTO 2000
960   CALL MSSC(IVDU,'IS ENTRY CORRECT ?>\')
      READ (IVDU,10000) YESNO
      IF (YESNO(1).NE.89) COTO 90
C
C    SKIP DATA COLLECT IF READ
C    ------------------------
C
     IF(XREAD.EQ.1)COTO 970
C
C     ASK FOR DATA
```

```
C       ------------
C
        XPLOT=0
        CALL BKTSIF(PRNTG,XINITL,DATAI,XTEST,XLEV,XPLOT)
C
C      TRANSFER DATA TO RECORD A OR B
C      ------------------------------
C
       IF(XDIF.EQ.0)GOTO 964
       IF(XDIFB.EQ.1)GOTO 962
       DO 961 M=1,256
961    DATARA(M)=DATAI(M)
       GOTO 964
962    DO 963 M=1,256
963    DATARB(M)=DATAI(M)
964    CONTINUE
C
        WRITE(IVDU,10000)TVA
        IF(PRNTG.NE.89)GOTO 968
        WRITE(IVDU,10000)PN,CA
        GOTO 2050
965     PRNTG=0
        WRITE(IVDU,10000)TVA
C
C        CHECK IF STILL WANT TO WRITE RECORD ?
C        -----------------------------------
C
968    CALL MSSG(IVDU,'IS RECORD OK TO WRITE ? Y/N >\')
       READ(IVDU,10000)YESNO
       IF(YESNO(1).NE.89)GOTO 5
C
C       READ FILE TO CHECK IF NEW RECORD
C       --------------------------------
C
C
C       SET FLAG
C
970     XFLAG=0
        XCHECK=0
        SEEREC=0
        ICAP=0
C
C       LOOK THROUGH RECORDS
C       --------------------
C
        DO 1010 I=1,100
        IF(IREC.EQ.I)GOTO 980
        IF(IREC.NE.0)GOTO 1000
980     J=(5*I-4)+1
        WRITE(IVDU,10500)J
        READ(IN,REC=J,ERR=1050,END=3000)XUPDT,IRE,XTES,DAT,XLE,XCOU,XCAI,
      1XBAS,XHI,XPC,XCO,XEAR
C
C      LOOK FOR SPACE IN FILE (FOR ERASED RECORD XUPDT=2)
C      --------------------------------------------------
C
        IF(XUPDT.EQ.XERAS)ICAP=I
        J=J+1
        READ(IN,REC=J,ERR=1050,END=3000)COM
        J=J+2
        READ(IN,REC=J,ERR=1050,END=3000)DATAO
        IF(IREC.NE.0)GOTO 2000
        IF(READAL.EQ.89)GOTO 2000
        IF (XFLAG) GOTO 1000
        IF ((XTES.EQ.XTEST).AND.(XLE.EQ.XLEV).AND.(XCOU.EQ.XCOUP).AND.
      1(XCAI.EQ.XCAIN).AND.(XBAS.EQ.XBASC).AND.(XHI.EQ.XHIC).AND.
      2(XPC.EQ.XPCL).AND.(XCO.EQ.XCOM).AND.(XEAR.EQ.XEARM)) GOTO 1200
1000    CONTINUE
1010    CONTINUE
        GOTO 3000
1050    CALL MSSG(IVDU,'READ ERROR\')
        GOTO 5
C
C       READ OUT RECORD IF 'READ' SET
C       -----------------------------
C
1200    IF(XREAD)GOTO 2000
        CALL MSSG(IVDU,'RECORD ALREADY EXISTS\')
        CALL MSSG(IVDU,'DATE                    REC No\')
        WRITE(IVDU,10300)DAT,IRE
        CALL MSSG(IVDU,'DO YOU WISH TO SEE RECORD ? >\')
        READ(IVDU,10000)SEEREC
        IF (SEEREC.EQ.89) GOTO 2000
C
C       READ MORE ?
C       -----------
C
1300    CALL MSSG(IVDU,'DO YOU WISH TO SEE IF MORE RECORDS
      1ALREADY EXIST ?\')
        READ(IVDU,10000)SEEREC
        IF (SEEREC.EQ.89) GOTO 1000
C
C       WRITE ?
C       -------
C
1500    SEEREC=0
        CALL MSSG(IVDU,'DO YOU WISH TO WRITE IN A NEW RECORD ? >\')
        READ(IVDU,10000)YESNO
        IF (YESNO(1).EQ.89)XFLAG=1
        IF (XFLAG) GOTO 1000
        GOTO 5
2000    CONTINUE
C
C       READ OUT TEST
C       -------------
C
2050    GOTO (2100,2110,2120,2130,2140),XTES
2100    CALL MSSG(IVDU,'FREQUENCY RESPONSE\')
        GOTO 2200
2110    CALL MSSG(IVDU,'2nd HARMONIC DIST\')
        GOTO 2200
2120     CALL MSSG(IVDU,'3rd HARMONIC DIST\')
        GOTO 2200
2130     CALL MSSG(IVDU,'2nd INTERMOD DIST\')
```

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      GOTO 2200
2140   CALL MSSG(IVDU,'3rd INTERMOD DIST\')
C
C
C     READ OUT INPUT LEVEL
C     ------------------
C
2200  WRITE(IVDU,10030)XLE
C
C     READ OUT COUPLER
C     ---------------
C
       GOTO (2310,2320,2330,2340,2350,2360,2370),XCOU
2310   CALL MSSG(IVDU,'ZWISLOCKI COUPLER\')
       GOTO 2400
2320   CALL MSSG(IVDU,'2cc COUPLER\')
       GOTO 2400
2330   CALL MSSG(IVDU,'B&K EAR SIMULATOR\')
       GOTO 2400
2340   CALL MSSG(IVDU,'KEMAR 0  deg incident \')
       GOTO 2400
2350   CALL MSSG(IVDU,'KEMAR 90 deg incident\')
       GOTO 2400
2360   CALL MSSG(IVDU,'KEMAR 180deg incident \')
       GOTO 2400
2370   CALL MSSG(IVDU,'KEMAR 270deg incident \')
C
C     READ OUT GAIN
C     ------------
C
2400   GOTO (2410,2420,2430),XGAI
2410   CALL MSSG(IVDU,'MAX-GAIN   \')
       GOTO 2500
2420   CALL MSSG(IVDU,'MIN-GAIN   \')
       GOTO 2500
2430   CALL MSSG(IVDU,'USER-GAIN  \')
C
C     READ OUT BASS CUT
C     ---------------
C
2500  IF(TONE.EQ.0)GOTO 2680
      IF(TONE.EQ.2)GOTO 2580
       IB=(XBAS+1)
       GOTO (2510,2520,2530),IB
2510   CALL MSSG(IVDU,'NO- BASS CUT\')
       GOTO 2600
2520   CALL MSSG(IVDU,'MAX-BASS CUT\')
       GOTO 2600
2530   CALL MSSG(IVDU,'USER-BASS CUT\')
C
C     READ OUT HIGH CUT
C     ---------------
C
2580   WRITE(IVDU,10040)
2600  IF(TONE.EQ.1)GOTO 2690
       IH=(XHI+1)
       GOTO (2610,2620,2630),IH
2610   CALL MSSG(IVDU,'NO-HIGH CUT\')
       GOTO 2700
2620   CALL MSSG(IVDU,'MAX-HIGH CUT\')
       GOTO 2700
2630   CALL MSSG(IVDU,'USER-HIGH CUT\')
C
C     READ OUT PEAK CLIP
C     -----------------
C
2680  WRITE(IVDU,10040)
2690  WRITE(IVDU,10040)
2700  IF(PC.EQ.0)GOTO 2780
       IP=(XPC+1)
       GOTO (2710,2720,2730),IP
2710   CALL MSSG(IVDU,'NO-PEAK CLIPPING\')
       GOTO 2800
2720   CALL MSSG(IVDU,'MAX-PEAK CLIPPING\')
       GOTO 2800
2730   CALL MSSG(IVDU,'USER-PEAK CLIPPING\')
C
C     READ OUT COMPRESSION
C     ------------------
C
2780   WRITE(IVDU,10040)
2800  IF(AVC.EQ.0)GOTO 2890
       IC=(XCO+1)
       GOTO (2810,2820,2830),IC
2810   CALL MSSG(IVDU,'NO-COMPRESSION\')
      GOTO 2900
2820   CALL MSSG(IVDU,'MAX-COMPRESSION\')
      GOTO 2900
2830   CALL MSSG(IVDU,'USER-COMPRESSION\')
C
C     READ OUT EARMOULD PLUMBING
C     -----------------------
C
2890   WRITE(IVDU,10040)
2900   IF (XEAR-1)2920,2910,2920
2910   CALL MSSG(IVDU,'STANDARD TUBING\')
      GOTO 2950
2920   CALL MSSG(IVDU,'ALTERNATE PLUMBING\')
      WRITE(IVDU,10400)XEAR
C
C     READ OUT DATE
C     -----------
C
2950  WRITE(IVDU,10700)DAT
C
C     READ OUT RECORD No
C     ----------------
C
      WRITE(IVDU,10500)IRE
C
C     READ OUT COMMENTS
C     ----------------
C
      CALL MSSG(IVDU,'COMMENTS>\')
```

```
      WRITE(IVDU,10025)COM
      IF((PRNTG.EQ.89).AND.(XREAD.NE.1))GOTO 965
      IF(PRNTG.EQ.89)GOTO 2960
C
C     CHECK IF JUST CHECK OR LOOKING FOR MORE
C     ------------------------------------
C
      IF(XCHECK)GOTO 960
C
C     READ OUT DATA IF REQUIRED
C     -------------------------
C
C
C    TRANSFER DATA TO RECORD A OR B
C     ------------------------------
C
      IF(XDIF.EQ.0)GOTO 2958
      IF(XDIFB.EQ.1)GOTO 2956
     DO 2955 M=1,256
2955 DATARA(M)=DATAO(M)
     GOTO 2958
2956 DO 2957 M=1,256
2957 DATARB(M)=DATAO(M)
2958 CONTINUE
     IF(IPLOT.EQ.0)GOTO 2970
      XPLOT=1
      CALL BKTSIF(PRNTG,XINITL,DATAO,XTES,XLE,XPLOT)
      IF(PRNTG.NE.89)GOTO 2970
      GOTO 2050
2960  PRNTG=0
      WRITE(IVDU,10000)TVA
2970  IF(SEEREC.EQ.89) GOTO 1300
C
C     CHECK IF WANT TO OVER-WRITE A PARTICULAR RECORD
C     --------------------------------------------
C
      IF((XREAD.NE.1).AND.(IREC.NE.0))GOTO 2975
      IF(IREC.NE.0)GOTO 5
      GOTO 1000
2975  IEND=1
      CALL MSSG(IVDU,'IS THIS THE RECORD YOU WANT TO OVER WRITE ?\')
      READ(IVDU,10000)YESNO
      IF(YESNO(1).NE.89)GOTO 5
      CALL MSSG(IVDU,'ARE YOU POSITIVE ? TYPE YES IN FULL >\')
      READ(IVDU,10000)YESNO
      IF((YESNO(1).EQ.89).AND.(YESNO(2).EQ.69).AND.(YESNO(3).EQ.83))
     1GOTO 3010
      GOTO 5
C
C     RECORD DOES NOT EXIST
C     ---------------
C
3000  IF(IREC.EQ.0)GOTO 3010
      CALL MSSG(IVDU,'NO EXISTING RECORD !\')
      GOTO 5
C
C     CHECK IF WRITE REQUIRED
C     ---------------------
C
3010  IF(XREAD) GOTO 5
      IF(SEEREC.EQ.89) GOTO 1500
C
C     WRITE IN RECORD (IF NEWFIL SET NEWFIL FLAG TO 0)
C     ------------------------------------------------
C
     IF(NEWFIL.EQ.1)WRITE(IN,REC=1,ERR=1050,END=1300)XNEW
      IF(ICAP.NE.0)I=ICAP
      IREC=I
      J=(5*I-4)+1
      WRITE(IVDU,10500)J
      WRITE(IN,REC=J,ERR=1050,END=1300)XNEW,IREC,XTEST,DATE,
    1XLEV,XCOUP,XGAIN,XBASC,XHIC,XPCL,XCOM,XEARM
      J=J+1
      WRITE(IN,REC=J,ERR=1050,END=1300)COMM
      J=J+2
      WRITE(IN,REC=J,ERR=1050,END=1300)DATAI
4000  GOTO 5
10000  FORMAT(20A1)
10010  FORMAT(I2)
10020  FORMAT(80A1)
10025  FORMAT(1X,80A1)
10030  FORMAT('INPUT LEVEL >',I2,'dB')
10040  FORMAT(1X)
10300  FORMAT(1X,20A1,I3)
10400  FORMAT(1X,I2)
10500  FORMAT(1X,'RECORD No > ',I3)
10600  FORMAT(1X,80I1)
10650  FORMAT(80I1)
10700  FORMAT(1X,'DATE > ',20A1)
10750  FORMAT(I3)
10800  FORMAT(1X,I4)
12000  FORMAT(1X,12A1,'  TYPE-',I2,'  AVC-',I2,'  PC-',I2,'  TONE-',I2
    1,'  MAXCC-',I2,'  MICDIR-',I2,'  MICPOS-',I2)
      END
```

```
      SUBROUTINE BKTSIF(PRNTG,XINITL,DATA,XTEST,XLEV,XPLOT)
C
C
C      M WALD DEC 1982
C
C
C       THIS SUBROUTINE PLOTS DATA FROM B&K AUDIO TEST STATION
C       ------------------------------------------------------
C
       BYTE PRNTG,XINITL,XTEST,XLEV,XPLOT
       INTEGER CALC
       REAL*4 DATAIN(2,256)
      BYTE RETURN,RUNO,STOPO,INIT,ENABLE,RESET
       BYTE DATA(256),XPOS,CAL,Z
      BYTE SPACE,BUF(20),DSK
       BYTE GAL,GA(1),VE(1),VEC,BL(1),BLK,WH(1),WHT
       BYTE TVAL,ERA,YESNO(20),PNT,ES(1),ESC
       BYTE TVA(1),ER(1),PN(1),IO1(4),IO2(4),IO3(4)
      DIMENSION X(256),XA(256),Y(256),IXA(256)
      DATA TVAL/24/,ERA/25/,PNT/28/,VEC/29/,ESC/27/,BLK/127/
     1,IVDU/1/ ,GAL/31/,WHT/97/,SPACE/32/,IN2/7/
C
C     ENCODE GRAPHICS
C     ---------------
C
       ENCODE(TVA,6010)TVAL
      ENCODE(ER,6010)ERA
      ENCODE(PN,6010)PNT
      ENCODE(GA,6010)GAL
      ENCODE(VE,6010)VEC
      ENCODE(BL,6010)BLK
      ENCODE(WH,6010)WHT
      ENCODE(ES,6010)ESC
C
C       ONLY IF FIRST TIME CALCULATE FREQUENCIES
C       ----------------------------------------
C
       IF(XINITL.NE.0)GOTO 3
C
C      SET OUTPUT PORTS 3 AND 4 ON 8PIO TO OUTPUT
C      ------------------------------------------
C
       CALL OUT(134,12)
C
C      SET UP X FREQ ARRAY
C      -------------------
C
        CALL MSSC(IVDU,'CALCULATION OF FREQUENCY SPACING IN PROGRESS\')
       RINC=10**(2.0/255.0)
       DO 2 I=1,256
       X(I)=0.1*RINC**(I-1)
       IXA(I)=350.0+300*ALOG10(X(I)/0.1)
2      CONTINUE
C
C     ASK IF ERASE PREVIOUS PLOT
C     --------------------------
C
3      CALL MSSC(IVDU,'ERASE PREVIOUS PLOT Y/N >\')
      READ(IVDU,1000)YESNO
      IF(YESNO(1).NE.89)GOTO 8
      WRITE(IVDU,6010)PN,ER
C
C     PLOT AXES
C     ---------
C
C
C     HORIZONTAL
C     ----------
C
      DO 4 IY=250,682,36
       IX1=350
       IX2=950
       CALL ENCOD(IX1,IY,IO1)
       CALL ENCOD(IX2,IY,IO2)
       WRITE(IVDU,2010)VE,IO1,IO2
4     CONTINUE
C
C     VERTICAL
C     --------
C
       DO 6  I=1,7
       X1=0.0625*2**I
C
C     LOG SPACING
C     -----------
C
       IX=350.0+300*ALOG10(X1/0.1)
       IY1=250
       IY2=682
       CALL ENCOD(IX,IY1,IO1)
       CALL ENCOD(IX,IY2,IO2)
       WRITE(IVDU,2010)VE,IO1,IO2
       IF(I.GT.3)GOTO 5
       IX=IX-50
       CALL ENCOD(IX,230,IO3)
C
C     LABELS
C     ------
C
       WRITE(IVDU,3010)PN,ES,BL,IO3,ES,WH,GA,X1
       GOTO 6
5      IX=IX-7
       CALL ENCOD(IX,230,IO3)
       IX1=X1
       WRITE(IVDU,3500)PN,ES,BL,IO3,ES,WH,GA,IX1
6      CONTINUE
C
C     LEFT AND RIGHT 'UPRIGHTS'
C     -------------------------
C
       CALL ENCOD(350,250,IO1)
       CALL ENCOD(350,682,IO2)
       WRITE(IVDU,2010)VE,IO1,IO2
```

```
      CALL ENCOD(950,250,IO1)
      CALL ENCOD(950,682,IO2)
      WRITE(IVDU,2010)VE,IO1,IO2
C
C     Y LABELS
C     --------
C
      DO 7 I=1,13
      IY=200+36*I
      IY1=20+10*I
      CALL ENCOD(290,IY,IO1)
7     WRITE(IVDU,10000)PN,ES,BL,IO1,ES,WH,GA,IY1
C
C     'Db'
C     ----
C
      CALL ENCOD(290,695,IO1)
      WRITE(IVDU,11010)PN,ES,BL,IO1,ES,WH,GA
C
C     'kHz'
C     -----
C
      CALL ENCOD(960,230,IO1)
      WRITE(IVDU,12000)PN,ES,BL,IO1,ES,WH,GA
      WRITE(1,6010)TVA
C
C     CHECK IF JUST PLOT REQUIRED ?
C     -----------------------------
C
8     IF(XPLOT.EQ.1)GOTO 452
C
C
C     ASK FOR CALIBRATE?
C     ------------------
C
15    CALL MSSG(IVDU,'CALIBRATE ? Y/N >\')
      READ(IVDU,1000)YESNO
      IF (YESNO(1).NE.89) GOTO 24
C
C     SEND CALIBRATE
C     --------------
C
      CALL OUT(130,1)
C
C     INITIATE CONVERSION
C     -------------------
C
      CALL OUT(131,8)
      Z=0
C
C     SET LOOP TIME
C     -------------
C
19    DO 23 I=1,10000
20    DO 21 J=1,100
21    CONTINUE
C
C     READ IN dB
C     ----------
C
      CAL=INP(132)
C
C     DON'T PRINT IF dB UNCHANGED
C     ---------------------------
C
      IF(CAL.EQ.Z)GOTO 22
      DB=CAL
C
C     CONVERT FROM 2'S COMPLEMENT
C     ---------------------------
C
      DB=140-(DB+103)/2
      WRITE(IVDU,2000)DB
22    Z=CAL
23    CONTINUE
      CALL MSSG(IVDU,'FINISHED CALIBRATION YET ? Y/N >\')
      READ(IVDU,1000)YESNO
      IF(YESNO(1).EQ.78)GOTO 19
24    CALL OUT(130,8)
C
C     INITIALIZE Y ARRAY
C     ------------------
C
48    DO 50 I=1,256
50    Y(I)=0.0
C
C     SET NORMAL X START POS
C     ----------------------
C
      IXBGIN=0
C
C     TEST REQUIRED
C     -------------
C
      GOTO(51,52,53,54,55),XTEST
51    IMODE=64
      IORDER=0
      GOTO 57
52    IMODE=0
      IORDER=1
      GOTO 57
53    IMODE=0
      IORDER=2
      GOTO 57
54    IMODE=32
      IORDER=1
      IXBGIN=49
      GOTO 57
55    IMODE=32
      IORDER=2
      IXBGIN=29
57    CONTINUE
C
C     LEVEL
```

```
C      ------
C
59     IF(XLEV.NE.50)GOTO 60
       XLEVL=16
       GOTO 65
60     XLEVL=((XLEV-55)/5)*16-128
65     CONTINUE
C
C        COMMANDS
C        --------
C
       RUNO=4+IMODE
       STOPO=IMODE-128
       RETURN=8
       INIT=IORDER+XLEVL+8
C
C        FREE RUN A/D
C        ------------
C
       ENABLE=IORDER+XLEVL+12
C
C        SEND ENABLE X
C        -------------
C
        CALL OUT(131,ENABLE)
       CALL MSSC(IVDU,'TO START PRESS ANY  KEY EXCEPT T\')
       CALL MSSC(IVDU,'PRESS T TO TERMINATE PROGRAMME----\')
C      -------------
       PAUSE
C      -------------
C       SEND RUN
C       --------
C
        CALL OUT(130,RUNO)
C
C       LOOK FOR X=0 START OF MEASURING RANGE
C       -------------------------------------
C
C
C      LOOK FOR FREQ CHANGE BY MASKING OUT LATCH INP(133)
C      ---------------------------------------------
C
100    IF(.NOT.(INP(134).OR.-33))GOTO 110
       GOTO 100
110    CONTINUE
        Z=99
        DO 145 I=1,200
120     XPOS=INP(132)
        IF((XPOS.EQ.-128).AND.(I.GT.100))GOTO 150
        IF(XPOS.EQ.Z)GOTO 120
        IXPOS=XPOS
        IXPOS=IXPOS+128
        IF((IXPOS.NE.90).AND.(IXPOS.NE.120))GOTO 145
C
C       INITIATE CONVERSION
C       -------------------
C
        CALL OUT(131,INIT)
C
C       READ IN dB
C       ----------
C
        CAL=INP(132)
        DB=CAL
C
C
C      CHECK LEVEL
C      -----------
C
        IF(IXPOS.EQ.120)GOTO 122
        IDB=5*IFIX((7-DB)/12)+50
        WRITE(IVDU,5500)IDB
       IF(IDB.EQ.XLEV)GOTO 121
       CALL MSSC(IVDU,'LEVEL ERROR\')
       CALL OUT(130,RETURN)
       GOTO 3
121     GOTO 140
C
C      CHECK MODE
C      ----------
C
122     ITS=(CAL+7)/17
        GOTO(125,127,129,131,133),ITS
125     CALL MSSC(IVDU,'FREQ RESP\')
        GOTO 135
127     CALL MSSC(IVDU,'2nd HARMONIC\')
        GOTO 135
129     CALL MSSC(IVDU,'3rd HARMONIC\')
        GOTO 135
131     CALL MSSC(IVDU,'2nd INTERMOD\')
        GOTO 135
133     CALL MSSC(IVDU,'3rd INTERMOD\')
C
C       SEND STOP IN GOOD TIME AS HAS NO EFFECT ON PARAMETER RANGE
C       ---------------------------------------------
C
       CALL OUT(130,STOPO)
C
135    IF(ITS.EQ.XTEST)GOTO 140
       CALL MSSC(IVDU,'TEST MODE ERROR\')
       CALL OUT(130,RETURN)
       GOTO 3
140     CALL OUT(131,ENABLE)
145     Z=XPOS
150     CONTINUE
         CALL OUT(130,STOPO)
        CALL MSSC(IVDU,'X=0\')
C
C      IF INTMOD GET XPOS CORRECT
C      ------------------------
C
        IF(IMODE.NE.32)GOTO 167
        CALL OUT(130,STOPO)
165    XPOS=INP(132)
```

```
      IF(XPOS.NE.(IXBGIN-128))GOTO 165
C
C       ADD 1 AS LOOP INDEX STARTS AT 1 NOT 0
C       -----------------------------------
C
167   IXBGIN=IXBGIN+1
C
C       REMOVE FREE RUN A/D FROM ENABLE
C       -------------------------------
C
        ENABLE=IORDER+XLEVL+4
C
168    DO 400 I=IXBGIN,256
C
C       IF FIRST POSITION OMIT FREQ CHANGE CHECK
C       ----------------------------------------
C
        IF(I.EQ.IXBGIN)GOTO 178
C
       CALL OUT(131,ENABLE)
        RESE=INP(133)
       CALL OUT(130,RUNO)
C
C     LOOK FOR FREQ CHANGE BY MASKING OUT LATCH INP(133)
C     --------------------------------------------------
C
170   IF(.NOT.(INP(134).OR.-33))GOTO 175
      GOTO 170
C
C       STOP
C       ----
C
175     CALL OUT(130,STOPO)
C
C       PLOT PREVIOUS POINT
C       -------------------
C
       IY=250.0+3.6*(DB-30.0)
        N=I-1
       CALL ENCOD(IXA(N),IY,IO1)
       WRITE(IVDU,2010)PN,IO1,GA,TVA
C
C       INPUT NEW X POS
C       ---------------
C
178    XPOS=INP(132)
       IXPOS=XPOS
       IXPOS=IXPOS+128
C
C
C      CHECK FOR POSITION ERROR
C      ------------------------
C
       IF((XPOS.EQ.-128).AND.(I.GT.10))GOTO 180
       IF(IXPOS.EQ.I-1)GOTO 180
       CALL MSSG(IVDU,'X POS ERROR\')
C
```

```
C       WRITE FREQ
C       ----------
C
180    WRITE(IVDU,14000)X(I)
C
C     CHECK FOR RANGE SHIFT IN PROGRESS BY MASKING LATCH INP(130)
C     -----------------------------------------------------------
C
        RESET=INP(130)
185   IF(.NOT.(INP(134).OR.-5))GOTO 187
        GOTO 190
187   RESET=INP(130)
      GOTO 185
C
C      INITIATE CONVERSION
C      -------------------
C
190    CALL OUT(131,INIT)
       Z=-128
C
C       WASTE TIME TO ALLOW LMS DETECTOR IN B&K TO SETTLE
C       -------------------------------------------------
C
191     DO 192 N=1,500
192     CONTINUE
C
C     CHECK IF BUSY BY MASKING LATCH INP(131)
C     ---------------------------------------
C
194   RESET=INP(131)
195   IF(.NOT.(INP(134).OR.-9))GOTO 200
      GOTO 195
C
C      READ IN dB
C      ----------
C
200    CAL=INP(132)
C
C      DON'T ACCEPT VALUE IF dB CHANGED
C      --------------------------------
C
       IF(CAL.EQ.Z)GOTO 250
       Z=CAL
C
C       WASTE TIME TO ALLOW LMS DETECTOR IN B&K TO SETTLE
C       -------------------------------------------------
C
240     DO 245 N=1,500
245     CONTINUE
C
       GOTO 194
250    DB=CAL
       DATA(I)=CAL
C
C      CONVERT FROM 2'S COMPLEMENT
C      ---------------------------
C
```

```
      DB=140-(DB+103)/2
      Y(I)=DB
       WRITE(IVDU,2000)DB
C
C     LOOK FOR END OF RUN
C     ------------------
C
      IF((XPOS.EQ.-128).AND.(I.GT.10))GOTO 450
400   CONTINUE
450   CALL MSSG(IVDU,'X=0\')
      CALL OUT(130,RETURN)
     GOTO 456
C
C     CONVERT FROM 2'S COMPLEMENT INTO Db
C     -----------------------------------
C
452   DO 455 I=1,256
      DB=DATA(I)
      DB=140-(DB+103)/2
      Y(I)=DB
      IY=250.0+3.6*(DB-30.0)
      CALL ENCOD(IXA(I),IY,IO1)
      WRITE(IVDU,2010)PN,IO1,GA,TVA
455   CONTINUE
456   CALL MSSG(IVDU,'WRITE IN DATA FILE ? Y/N >\')
      READ(IVDU,1000)YESNO
      IF(YESNO(1).NE.89)GOTO 700
457   DO 460 I=1,256
      Y(I)=Y(I)-130.0
460   CONTINUE
       WRITE(IVDU,5000)(X(I),Y(I),I=1,256)
      DO 465 J=1,60
      RJ=J-1
      FREQ=0.1+7.9*RJ/59.0
      I=1.0+ALOG10(10*FREQ)/2.0*255.0
      Y(J)=Y(I)
      XA(J)=X(I)
465   CONTINUE
      Y(61)=-200.0
      XA(61)=8.5
C
C     WRITE IN DATA FILE
C     ------------------
C
      CALL MSSG(IVDU,'DATA FILENAME > \')
      READ(IVDU,1000)BUF
      IF(BUF(1).EQ.SPACE)GOTO 700
467   CALL MSSG(IVDU,'DISK? A=1,B=2 >\')
      READ(IVDU,11000)DSK
      IF((DSK.LT.1).OR.(DSK.GT.2))GOTO 467
      CALL MKOPEN(IN2,BUF,DSK)
      WRITE(IN2,8000,END=470)(XA(I),Y(I),I=1,61)
470   CONTINUE
      REWIND IN2
      READ(IN2,9000,END=480)((DATAIN(J,NBAND),J=1,2),NBAND=1,61)
480   ENDFILE IN2
      CALL MSSG(IVDU,' \')
      CALL MSSG(IVDU,'       BANDTOP(KHZ)      GAIN(DB)\')
      CALL MSSG(IVDU,' \')
      DO 490 NBAND=1,61
      WRITE(IVDU,1020)(DATAIN(J,NBAND),J=1,2)
490   CONTINUE
700   XINITL=1
800   CALL MSSG(IVDU,'GRAPHICS PRINT OF DETAILS REQUIRED ? Y/N \')
      READ(IVDU,7000)PRNTG
      IF(PRNTG.NE.89)GOTO 900
      CALL ENCOD(5,450,IO1)
      WRITE (IVDU,1000)PN,ES,BL,IO1,ES,WH,GA
900   RETURN
1000   FORMAT(20A1)
1020  FORMAT(1H ,10X,F6.3,7X,F6.1)
2000   FORMAT(1X,F5.1,'dB')
2010   FORMAT(9A1)
3000   FORMAT(1X,I5)
3010   FORMAT(10A1,F5.3)
3500   FORMAT(10A1,I1)
4000  FORMAT(F12.0)
5000  FORMAT(1X,2F12.4)
5500  FORMAT(I2,'Db')
6000  FORMAT(1X,F12.4)
6010   FORMAT(2A1)
7000  FORMAT(A1)
8000  FORMAT(1X,F6.3,',',F6.1)
9000  FORMAT(2F12.0)
10000 FORMAT(10A1,I3)
10010 FORMAT(I2)
11000 FORMAT(I1)
11010 FORMAT(10A1,'dB')
12000 FORMAT(10A1,'kHz')
14000 FORMAT(1X,F6.3,'KHz')
       END
       SUBROUTINE ENCOD(IX,IY,IO)
C
C       THIS SUBROUTINE ENCODES X AND Y CORDINATES INTO GRAFFIX CODES
C       -------------------------------------------------------------
C
       BYTE IO(4)
       IC=IX/32+32
       ID=IX-32*(IC-34)
       IA=IY/32+32
       IB=IY-32*(IA-35)
       ENCODE(IO,5000)IA,IB,IC,ID
       RETURN
5000   FORMAT(4A1)
       END
```

```
      PROGRAM FLUPDT
C
C     MIKE WALD OCT 1982
C     -----------------
C
      IMPLICIT BYTE(X)
      INTEGER FREQP(11)
      BYTE TSTEND,XLE(100),XBAS(100),XHI(100),XPC(100),XCO(100)
      BYTE SPACE
      BYTE FILE(12),DATE(20),DATA(256)
      DIMENSION IMAXO(11),XMAXC(11),XMAXO(11),IR(2),IRE(100)
      DATA IVDU/1/,     IN/6/,  INA/7/,  INB/8/ ,SPACE/32/
     1,FREQP/13,52,90,113,129,151,167,189,205,228,244/
C
C     OPEN FILE OF AID NAMES
C     ---------------------
C
      CALL OPEN(INA,'TESTNAMEDAT',0)
C
C     OPEN UPDATE FILE
C     ----------------
C
      CALL OPEN(INB,'UPDATFILDAT',0)
C
C     READ THROUGH AID NAME FILE
C     -------------------------
C
      DO 1000 I=1,10000
      READ(INA,REC=I,ERR=1100,END=1200)FILE
      WRITE(IVDU,10000)FILE
C
C     OPEN AID FILE
C     -------------
C
      CALL MKOPEN(IN,FILE,0)
C
C     SET FOUND FLAGS
C     ---------------
C
      N=0
      XLEVCF=0
      XLEVMF=0
C
      READ(IN,REC=1,ERR=1100,END=900)XFLAG
      IF(XFLAG.EQ.1)GOTO 900
      DO 30 J=1,10000
      K=(5*J-4)+1
      READ(IN,REC=K,ERR=1100,END=35)XFLAG,IREC,XTES,DATE,XLEV,XCOUP
     1,XGAIN,XBASC,XHIC,XPCL,XCOM,XEARM
C
C     CHECK IF NOT ERASED OR UPDATED AND ZWIS AND FREQ RES AND 60 OR
C     90 Db INPUT AND MAX GAIN
C     --------------------------------------------------------------
C
      IF((XFLAG.NE.0).OR.(XTES.NE.1).OR.(XCOUP.NE.1)
     1.OR.(XGAIN.NE.1).OR.((XLEV.NE.60).AND.(XLEV.NE.90)))GOTO 30
 20   N=N+1
      IF(XLEV.EQ.60)XLE(N)=0
      IF(XLEV.EQ.90)XLE(N)=1
      XBAS(N)=XBASC
      XHI(N)=XHIC
      XPC(N)=XPCL
      XCO(N)=XCOM
      IRE(N)=IREC
      GOTO 30
 30   CONTINUE
C
C     CHECK FOR BOTH 60 AND 90 dB RECORDS
C     ----------------------------------
C
 35   NA=N-1
      DO 700 J=1,NA
      XBASC=XBAS(J)
      XHIC=XHI(J)
      XPCL=XPC(J)
      XCOM=XCO(J)
      XLEV=XLE(J)
      IREC=IRE(J)
      JA=J+1
      DO 600 K=JA,N
      IF((XHIC.NE.XHI(K)).OR.(XPCL.NE.XPC(K)).OR.(XCOM.NE.XCO(K))
     1.OR.(XLEV.EQ.XLE(K)))GOTO 600
C
C     IR(1) CONTAINS REC No OF 60 dB AND IR(2) OF 90 dB
C     ------------------------------------------------
C
      IF(XLEV.EQ.1)GOTO 45
      IR(1)=IREC
      IR(2)=IRE(K)
      GOTO 50
 45   IR(1)=IRE(K)
      IR(2)=IREC
 50   CONTINUE
```

```
C
C     FIND MAX GAIN
C     -------------
C

      WRITE(IVDU,12000)IR(1)
      CALL MSSG(IVDU,'FREQ      > 125   250   500   750  1000  1500  200
      IRR=5*IR(1)
      READ(IN,REC=IRR,ERR=1100,END=900)DATA
      DO 60 L=1,11
      M=FREQP(L)
      IMAXO(L)=DATA(M)
      IMAXO(L)=IMAXO(L)+128
C
C     ROUND UP TO NEAREST 1DB
C     -----------------------
C
      IMAXO(L)=140-(IMAXO(L)-26)/2
60    XMAXG(L)=IMAXO(L)-60
      WRITE(IVDU,13000)XMAXG
C
C     FIND MAX OUTPUT
C     ---------------
C

      WRITE(IVDU,12000)IR(2)
      IRR=5*IR(2)
      READ(IN,REC=IRR,ERR=1100,END=900)DATA
      DO 65 L=1,11
      M=FREQP(L)
      IMAXO(L)=DATA(M)
      IMAXO(L)=IMAXO(L)+128
C
C     ROUND UP TO NEAREST 1DB
C     -----------------------
C
      IMAXO(L)=140-(IMAXO(L)-26)/2
      XMAXO(L)=IMAXO(L)-50
65    CONTINUE
      WRITE(IVDU,14000)IMAXO
C     GO TO END OF FILE OR IN A SPACE AND WRITE IN RECORD
C     ---------------------------------------------------
C
90    DO 100 L=1,10000
      READ(INB,REC=L,ERR=1100,END=200)TSTEND
      IF(TSTEND.EQ.SPACE)GOTO 200
100   CONTINUE
200   WRITE(INB,REC=L)FILE,IR(1),IR(2),XMAXG,XMAXO
C
C     UPDATE FLAG
C     -----------
C
      XUPFLG=1
      WRITE(IN,REC=1,ERR=1100,END=900)XUPFLG
600   CONTINUE
700   CONTINUE
900   ENDFILE IN
1000  CONTINUE
1100  CALL MSSG(IVDU,'READ ERROR\')
1200  ENDFILE IN
      ENDFILE INA
      ENDFILE INB
10000 FORMAT(1X,12A1)
11000 FORMAT(1X,'FLAG >',I5)
12000 FORMAT(1X,'REC No >',I5)
13000 FORMAT(1X,'MAX GAIN  >',11(I4,2X))
14000 FORMAT(1X,'MAX OUTPUT>',11(I4,2X))
      END
```

```
      PROGRAM AUDATA
C
C     MIKE WALD OCT 1982
C     -----------------
C
C
C     THIS PROGRAM ENTERS AUDIOMETRIC DATA INTO FILE
C     ----------------------------------------------
C
      INTEGER FREQ(11)
      BYTE YESNO,SPACE,ENDTES
      BYTE UNATHR(11),MONMAF(11),LDL(11),BUF(20)
      DATA MONMAF /24,14,9,7,7,6,4,0,-1,8,18/,IVDU/1/
      DATA FREQ/125,250,500,750,1000,1500,2000,3000,4000,6000,8000/
      DATA SPACE/32/     ,IN/6/
C
C     ASK SUBJECTS FILENAME
C     --------------------
C
5     CALL MSSG(IVDU,'SUBJECT FILENAME ? >\')
      READ(IVDU,2000)BUF
      IF(BUF(1).EQ.SPACE)STOP
C
C     CHECK IF FILE EXISTS
C     --------------------
C
      CALL SBFLNM(BUF,M)
      IF(M.EQ.1)GOTO 10
      CALL MSSG(IVDU,'FILE ALREADY EXISTS\')
      GOTO 5
C
C     READ IN THRESHOLDS
C     ------------------
C
10    DO 100 I=1,11
20    WRITE(IVDU,1000)FREQ(I)
30    READ(IVDU,1100,ERR=40)IUNATH
C
C     CHECK IF VALID
C     --------------
C
      IF((IUNATH.GE.-10).AND.(IUNATH.LE.120).AND.((FLOAT
     1(IUNATH)/5).EQ.(IUNATH/5)))GOTO 90
      CALL MSSG(IVDU,'THRESHOLDS MUST BE MULTIPLE OF 5 AND <= 120
     1 AND >= -10\')
40    WRITE(IVDU,1000)FREQ(I)
      GOTO 30
90    UNATHR(I)=IUNATH
100   CONTINUE
      CALL MSSG(IVDU,'ARE THESE CORRECT ? \')
      WRITE(IVDU,1400)
      WRITE(IVDU,1150)(UNATHR(I),I=1,11)
      CALL MSSG(IVDU,'(Y/N) >\')
      READ(IVDU,1200)YESNO
      IF(YESNO.NE.89)GOTO 10
C
C     READ IN LDL
C     -----------
C
200   DO 300 I=1,11
      WRITE(IVDU,1050)FREQ(I)
230   READ(IVDU,1100,ERR=240)ILDL
C
C     CHECK IF VALID
C     --------------
C
      IF((ILDL.GE.-10).AND.(ILDL.LE.120).AND.((FLOAT
     1(ILDL)/5).EQ.(ILDL/5)))GOTO 290
      CALL MSSG(IVDU,'LDL MUST BE MULTIPLE OF 5 AND <= 120
     1 AND >= -10\')
240   WRITE(IVDU,1000)FREQ(I)
      GOTO 230
290   LDL(I)=ILDL
300   CONTINUE
      CALL MSSG(IVDU,'ARE THESE CORRECT ? \')
      WRITE(IVDU,1400)
      WRITE(IVDU,1170)(LDL(I),I=1,11)
      CALL MSSG(IVDU,'(Y/N) >\')
      READ(IVDU,1200)YESNO
      IF(YESNO.NE.89)GOTO 200
      CALL OPEN(IN,BUF,0)
      WRITE(IN,REC=1)UNATHR,LDL
      ENDFILE IN
      CALL MSSG(IVDU,'ANOTHER SUBJECTS AUDIOMETRIC DATA ? > (Y/N)\')
      READ(IVDU,3000)YESNO
      IF(YESNO.EQ.89)GOTO 5
1000  FORMAT('INPUT THRESHOLD (dB HTL) AT ',I4,'Hz >')
1050  FORMAT('INPUT LDL (dB HTL) AT  ',I4,'HZ >')
1100  FORMAT(I3)
1150  FORMAT(1X,'THRESHOLDS dB HL>   ',11I5)
1170  FORMAT(1X,'LDL dB HL         >   ',11I5)
1200  FORMAT(A1)
1400  FORMAT(1X,'FREQUENCY         > .125k .25k  .5k  .75k  1k 1.5k
     1  2k    3k    4k    6k    8k ')
2000  FORMAT(20A1)
3000  FORMAT(A1)
      END
```

```
      PROGRAM MSADFT
C
C     MIKE WALD OCT 1982
C     -----------------
C
C
C     THIS PROGRAM FINDS REQUIRED AID
C     --------------------------------
C
      INTEGER SSPLA(11),SSPLMX(11),SSPLMN(11),BESTA(10,3),BESTB(10,3)
      INTEGER BMXODA(10,11),BMXODB(10,11),BMXGDA(10,11),BMXGDB(10,11)
      BYTE YESNO,MAXG(11),MAXO(11),FILE(12),BFILEA(10,12),BFILEB(10,12)
      BYTE UNATH(11),AIDTHA(11),AIDTHB(11),REARGA(11),REARGB(11),
     1MAXCG(11),LD(11),FREA(7),BUF(20),SPACE,NUMA,FLAG,FREB(6),NUMB
      BYTE INSG(11)
      DATA SPACE /32/   ,IN/6/   ,NUMA/7/   ,NUMB/6/
      DATA IVDU/1/,FREA/2,3,5,6,7,8,9/,FREB/3,5,7,8,9,10/
      DATA INSG/0,0,2,4,4,4,12,12,12,8,-6/
C
C     OBTAIN AUDIOMETRIC DATA ON SUBJECT
C     ----------------------------------
C
10    CALL MSSG(IVDU,'SUBJECTS FILENAME >\')
      READ(IVDU,2000)BUF
      IF(BUF(1).EQ.SPACE)STOP
      CALL OPEN(IN,BUF,0)
      READ(IN,REC=1,ERR=20,END=30)UNATH,LD
      GOTO 40
20    CALL MSSG(IVDU,'READ ERROR\')
      GOTO 35
30    CALL MSSG(IVDU,'FILE EMPTY\')
35    ENDFILE IN
      GOTO 10
40    ENDFILE IN
C
C     WRITE OUT DATA
C     --------------
C
      WRITE(IVDU,2400)
      WRITE(IVDU,2150)UNATH
      WRITE(IVDU,2400)
      WRITE(IVDU,2170)LD
      CALL BYPRSB(UNATH,LD,SSPLA,REARGA,AIDTHA)
      CALL BGPRSB(UNATH,LD,SSPLMX,SSPLMN,REARGB,AIDTHB,MAXCG)
C     GO THROUGH HEARING AID DATA AND COMPARE
C     ---------------------------------------
C
      CALL OPEN(IN,'UPDATFILDAT',0)
      FLAG=1
      DO 200 I=1,10000
      READ(IN,REC=I,ERR=400,END=210)FILE,IREC,IREC2,MAXG,MAXO
      DO 50 J=1,11
50    MAXG(J)=MAXG(J)+INSG(J)
      IF(FILE(1).EQ.SPACE)GOTO 200
      IF(IREC.EQ.0)GOTO 200
      CALL BSTFTS(REARGA,SSPLA,BESTA,FILE,IREC,MAXG,MAXO,BMXGDA,
     1BMXODA,FREA,NUMA,FLAG,BFILEA)
      CALL BSTFTS(REARGB,SSPLMX,BESTB,FILE,IREC,MAXG,MAXO,BMXGDB,
     1BMXODB,FREB,NUMB,FLAG,BFILEB)
      FLAG=0
200   CONTINUE
210   CALL WRTBYR(AIDTHA,REARGA,SSPLA,BESTA,BFILEA,BMXGDA,BMXODA)
      CALL WRTBER(AIDTHB,REARGB,SSPLMX,BESTB,BFILEB,BMXGDB,BMXODB,
     1SSPLMN,MAXCG)
310   ENDFILE IN
      CALL MSSG(IVDU,'ANOTHER FITTING ?  (Y/N) >\')
      READ(IVDU,1200)YESNO
      IF(YESNO.EQ.89)GOTO 10
      GOTO 500
400   CALL MSSG(IVDU,'READ ERROR ON UPDATE FILE\')
      ENDFILE IN
500   STOP
1200  FORMAT(A1)
2000  FORMAT(20A1)
2400  FORMAT(1X,'FREQUENCY        > .125K .25K  .5K  .75K  1K 1.5K
     1  2K    3K    4K    6K    8K ')
2150  FORMAT(1X,'THRESHOLDS dB HL>   ',11I5)
2170  FORMAT(1X,'LDL dB HL         >   ',11I5)
      END

      SUBROUTINE BYPRSB(UNATH,LD,SSPL,REARGA,AIDTHR)
C
C     THIS SUBROUTINE PREDICTS REQUIRED RESPONSE
C     ------------------------------------------
C
      INTEGER SSPL(7)
      BYTE TABLE(7,22),UNATH(11),AIDTHR(7),MONMAF(7),REARGA(7),
     1LD(11),FRE(7),UNATHR(7),LDL(7)
      DATA MONMAF /14,9,7,6,4,0,-1/,FRE/2,3,5,6,7,8,9/
      DATA TABLE /
C HTL          FREQ(kHz)
C dB(ISO).25 .5  1 1.5 2  3  4
C------------------------------
     0-     31,24, 3, 2, 1, 2, 2,
     5-     34,27, 6, 5, 4, 5, 4,
    10-     36,29, 8, 7, 6, 7, 6,
    15-     39,32,11,10, 9,10, 9,
    20-     42,35,14,13,12,13,12,
    25-     44,37,16,15,14,15,14,
    30-     47,40,19,18,17,18,17,
    35-     50,43,22,21,20,21,20,
    40-     53,46,25,24,23,24,23,
    45-     55,48,27,26,25,26,25,
    50-     58,51,30,29,28,29,28,
    55-     61,54,33,33,31,32,31,
    60-     63,56,35,34,33,34,33,
    65-     66,59,38,37,36,37,36,
    70-     69,62,41,40,39,40,39,
    75-     71,64,43,42,41,42,41,
    80-     74,67,46,45,44,45,44,
    85-     77,70,49,48,47,48,47,
    90-     80,73,52,51,50,51,50,
    95-     82,75,54,53,52,53,52,
   100-     85,78,57,56,55,56,56,
   105-     88,81,60,59,58,59,58/
```

```
C      WORK OUT PREDICTION
C      -------------------
C
       DO 100 I=1,7
       J=FRE(I)
       UNATHR(I)=UNATH(J)
       LDL(I)=LD(J)
       J=UNATHR(I)/5+1
       AIDTHR(I)=TABLE(I,J)
       REARGA(I)=UNATHR(I)+MONMAF(I)-AIDTHR(I)
       SSPL(I)=LDL(I)
       SSPL(I)=SSPL(I)+MONMAF(I)
100    CONTINUE
       RETURN
       END


      SUBROUTINE BCPRSB(UNATH,LD,SSPLMX,SSPLMN,REARGA,AIDTHR,MAXCG)
C
C     MIKE WALD OCT 1982
C     ------------------
C
C
C     THIS PROGRAM PREDICTS REQUIRED AID RESPONSE
C     -------------------------------------------
C
      INTEGER SSPLMX(6),SSPLMN(6),LDL(6)
      BYTE TABLE(6,23),UNATH(11),AIDTHR(6),MONMAF(6),REARGA(6),
     1LD(11),FRE(6),UNATHR(6),AUDZRO(6),MINSPL(6),MAXCG(6)
      DATA MONMAF /9,7,4,0,-1,8/,IVDU/1/,FRE/3,5,7,8,9,10/
      DATA AUDZRO /11,7,9,9,9,9/,MINSPL /75,75,72,71,70,68/
      DATA TABLE /
C HTL          FREQ(KHz)
C dB(ISO) .5  1  2  3  4  6
C----------------------------
     0-      0, 0, 0, 0, 0, 0,
     5-      0, 0, 0, 0, 0, 0,
    10-      0, 0, 0, 0, 0, 0,
    15-      0, 0, 0, 0, 0, 0,
    20-     20, 7, 7, 8, 9,10,
    25-     20, 9, 8,10,12,12,
    30-     21,11,10,12,14,15,
    35-     22,13,12,14,17,17,
    40-     23,15,13,16,19,20,
    45-     24,17,15,19,21,22,
    50-     25,19,17,21,24,25,
    55-     28,21,18,23,26,27,
    60-     30,22,20,25,28,30,
    65-     33,24,22,27,31,32,
    70-     35,26,23,29,33,35,
    75-     38,28,25,31,36, 0,
    80-     40,30,27,33,38, 0,
    85-     43,32,28,35,40, 0,
    90-     45,34,30,37, 0, 0,
    95-     48,36,32,39, 0, 0,
   100-     50,38,33,41, 0, 0,
   105-     53,39,35, 0, 0, 0,
   110-     55,41,37, 0, 0, 0/
C
C      OBTAIN AUDIOMETRIC DATA
C      -----------------------
C
       DO 100 I=1,6
       K=FRE(I)
       UNATHR(I)=UNATH(K)
       LDL(I)=LD(K)
       J=UNATHR(I)/5+1
C
C      CONVERT TO FF THRESH
C      --------------------
C
       AIDTHR(I)=TABLE(I,J)+MONMAF(I)
C

C      REAL FUNCTIONAL GAIN
C      --------------------
C
       REARGA(I)=UNATHR(I)-TABLE(I,J)

C
C      MAX COUPLER GAIN
C      ----------------
C
       MAXCG(I)=REARGA(I)+10
       IF(I.EQ.3)MAXCG(I)=MAXCG(I)+2
       IF(I.EQ.4)MAXCG(I)=MAXCG(I)+3
C
C      MAX SSPL
C      --------
C
       SSPLMX(I)=LDL(I)+AUDZRO(I)
C
C      MIN SSPL
C      --------
C
       SSPLMN(I)=MAXCG(I)-10+MINSPL(I)
100    CONTINUE
       RETURN
       END
```

```
      SUBROUTINE BSTFTS(REARGA,SSPL,BEST,FILE,IREC,MAXG,MAXO,BMAXGD,
     1BMAXOD,FRE,NUM,FLAG,BFILE)
C
C     MIKE WALD OCT 1982
C     -----------------
C
C
C     THIS PROGRAM FINDS AND SORTS BEST AIDS
C     ------------------------------------
C
      INTEGER SSPL(11),BEST(10,3),OPT,DIFF(100),IMAXO(11),H
      INTEGER BMAXOD(10,11),BMAXGD(10,11),IMAXOD(11),IMAXGD(11)
      BYTE MAXG(11),MAXO(11),FILE(12),BFILE(10,12)
      BYTE REARGA(11),FRE(11),NUM,FLAG
C     SET BEST  ARRAY-DIFF TO 999 AND REC,FILE,BELOW MAXG TO 0
C     -------------------------------------------------------
C
      IF(FLAG.NE.1)GOTO 105
      DO 105 I=1,10
      BEST(I,2)=999
      BEST(I,3)=0
      BEST(I,1)=0
      DO 103 J=1,12
103   BFILE(I,J)=0
      DO 105 J=1,11
      BMAXGD(I,J)=0
      BMAXOD(I,J)=0
105   CONTINUE
C
C     REJECT IF SSPL TO BIG
C     --------------------
C
      DO 110 J=1,11
      IMAXO(J)=MAXO(J)
110   IMAXO(J)=IMAXO(J)+50
      DO 115 J=1,NUM
      K=FRE(J)
      IMAXOD(J)=SSPL(J)-IMAXO(K)
      IF(IMAXO(K).GT.SSPL(J))GOTO 200
115   CONTINUE
C
C     WORK OUT DIFF BETWEEN REQUIRED AND ACTUAL FREQ/GAIN
C     --------------------------------------------
C
      DO 140 L=1,100
      DIFF(L)=0
120   DO 130 J=1,NUM
      K=FRE(J)
C
C     SEE IF DECREASING GAIN FROM MAX IMPROVES FIT
C     ------------------------------------------
C
130   DIFF(L)=DIFF(L)+IABS(REARGA(J)-MAXG(K)+L-1)
      IF(L.EQ.1)GOTO 140
      IF(DIFF(L).GT.DIFF(L-1))GOTO 143
140   CONTINUE
143   DO 145 J=1,NUM
      K=FRE(J)
145   IMAXGD(J)=MAXG(K)-REARGA(J)-L+2
C
C     SORT 10 BEST FITS
C     ----------------
C
150   OPT=DIFF(L-1)
      DO 160 J=1,10
      IF(BEST(J,2).GT.DIFF(L-1))GOTO 170
160   CONTINUE
      GOTO 200
170   IF(J.EQ.10)GOTO 190
      M=10-J
      DO 180 K=1,M
      N=11-K
      DO 175 H=1,12
175   BFILE(N,H)=BFILE(N-1,H)
      DO 179 H=1,NUM
      BMAXGD(N,H)=BMAXGD(N-1,H)
179   BMAXOD(N,H)=BMAXOD(N-1,H)
      BEST(N,1)=BEST(N-1,1)
      BEST(N,2)=BEST(N-1,2)
      BEST(N,3)=BEST(N-1,3)
180   CONTINUE
190   BEST(J,1)=IREC
      BEST(J,2)=OPT
      BEST(J,3)=L-2
      DO 195 K=1,12
195   BFILE(J,K)=FILE(K)
      DO 197 K=1,NUM
      BMAXGD(J,K)=IMAXGD(K)
197   BMAXOD(J,K)=IMAXOD(K)
200   CONTINUE
      RETURN
      END
```

```
      SUBROUTINE WRTBYR(AIDTHR,REARGA,SSPL,BEST,BFILE,BMAXCD,BMAXOD)
C
C     MIKE WALD OCT 1982
C     -----------------
C
C
C     THIS PROGRAM WRITES REQUIRED AND BEST AID RESPONSES
C     -----------------------------------------------
C
      INTEGER SSPL(11),BEST(10,3),BMAXOD(10,11),BMAXCD(10,11)
      BYTE AIDTHR(11),REARGA(11),FILE(12),BFILE(10,12)
      DATA IVDU/1/
C
C     WRITE OUT PREDICTION
C     --------------------
C
      CALL MSSG(IVDU,'BYRNE PREDICTIONS\')
      CALL MSSG(IVDU,'-----------------\')
      CALL MSSG(IVDU,' \')
      WRITE(IVDU,1400)
      WRITE(IVDU,1500)(AIDTHR(I),I=1,7)
      WRITE(IVDU,1600)(REARGA(I),I=1,7)
      WRITE(IVDU,1700)(SSPL(I),I=1,7)
C
C     WRITE OUT FITTINGS
C     -----------------
C
      DO 250 I=1,10
      IF(BEST(I,2).NE.999)GOTO 250
      IF(I.NE.1)GOTO 270
      CALL MSSG(IVDU,'NO AID SATISFACTORY\')
      GOTO 310
250   CONTINUE
270   K=I-1
      WRITE(IVDU,2300)
      DO 300 I=1,K
      WRITE(IVDU,2200)(BFILE(I,J),J=1,12),BEST(I,1),BEST(I,2),BEST(I,3),
     1(BMAXCD(I,J),J=1,7),(BMAXOD(I,J),J=1,7)
300   CONTINUE
310   CONTINUE
1400  FORMAT(1X,'FREQUENCY        >  250Hz 500Hz  1kHz 1.5kHz  2kHz
     1  3kHz  4kHz')
1500  FORMAT(1X,'AIDED THRESH SPL>',7I6)
1600  FORMAT(1X,'REAL EAR GAIN dB>',7I6)
1700  FORMAT(1X,'SSPL dB SPL     >',7I6)
2200  FORMAT(1X,12A1,I4,I5,I5,6X,7I3,5X,7I3)
2300  FORMAT(1X,'FILENAME   RECNo DIFF-BELOW-MXCD.25 .5 1 1.5  2
     1. 3  4 MXOD.25 .5 1 1.5  2  3  4')
      RETURN
      END
```

```
      SUBROUTINE WRTBER(AIDTHR,REARGA,SSPLMX,BEST,BFILE,BMAXCD,BMAXOD,
     1SSPLMN,MAXG)
C
C     MIKE WALD OCT 1982
C     -----------------
C
C
C     THIS PROGRAM WRITES REQUIRED AND BEST AID RESPONSES
C     -----------------------------------------------
C
      INTEGER SSPLMX(11),BEST(10,3),BMAXOD(10,11),BMAXCD(10,11),
     1SSPLMN(11)
      BYTE AIDTHR(11),REARGA(11),FILE(12),BFILE(10,12),MAXG(11)
      DATA IVDU/1/
C
C     WRITE OUT PREDICTION
C     --------------------
C
      CALL MSSG(IVDU,'BERGER PREDICTIONS\')
      CALL MSSG(IVDU,'------------------\')
      CALL MSSG(IVDU,' \')
      WRITE(IVDU,1400)
      WRITE(IVDU,1500)(AIDTHR(I),I=1,6)
      WRITE(IVDU,1600)(REARGA(I),I=1,6)
      WRITE(IVDU,1900)(MAXG(I),I=1,6)
      WRITE(IVDU,1700)(SSPLMX(I),I=1,6)
      WRITE(IVDU,1800)(SSPLMN(I),I=1,6)
C
C     WRITE OUT FITTINGS
C     -----------------
C
      DO 250 I=1,10
      IF(BEST(I,2).NE.999)GOTO 250
      IF(I.NE.1)GOTO 270
      CALL MSSG(IVDU,'NO AID SATISFACTORY\')
      GOTO 310
250   CONTINUE
270   K=I-1
      WRITE(IVDU,2300)
      DO 300 I=1,K
      WRITE(IVDU,2200)(BFILE(I,J),J=1,12),BEST(I,1),BEST(I,2),BEST(I,3),
     1(BMAXCD(I,J),J=1,6),(BMAXOD(I,J),J=1,6)
300   CONTINUE
310   CONTINUE
1400  FORMAT(1X,'FREQUENCY        >  500Hz  1kHz  2kHz
     1  3kHz  4kHz  6kHz')
1500  FORMAT(1X,'AIDED THRESH SPL>',6I6)
1600  FORMAT(1X,'REAL EAR GAIN dB>',6I6)
1700  FORMAT(1X,'SSPLMAX dB SPL  >',6I6)
1800  FORMAT(1X,'SSPLMIN dB SPL  >',6I6)
1900  FORMAT(1X,'MAX GAIN      dB>',6I6)
2200  FORMAT(1X,12A1,I4,I5,I5,7X,6I3,5X,6I3)
2300  FORMAT(1X,'FILENAME   RECNo DIFF-BELOW-MAXCD .5  1  2
     1  3  4  6 MXOD .5  1  2  3  4  6')
      RETURN
      END
```

```
      PROGRAM DTAINT
C
C     MIKE WALD MARCH 1983
C
C
      REAL LF,LFG
      BYTE FILE(12),SPACE,YESNO(20),Y
      DIMENSION FREQ(70),X(256),BRKPNT(70,2),GAIN(70)
      DATA IVDU/1/,IN/6/,INA/7/,SPACE/32/,Y/89/
C
C     SET UP 256 LOG SPACED FREQ POINTS
C     -------------------------------
C
      RINC=10**(2.0/255.0)
      DO 10 I=1,256
      X(I)=0.1*RINC**(I-1)
10    CONTINUE
        OCTAVE=1/ALOG10(2.0)
C
C     SET UP 'NEAREST' 60 LIN SPACED POINTS
C     ------------------------------------
C
      DO 20 J=1,60
      RJ=J-1
      FRE=0.1+7.9*RJ/59.0
      I=1.0+ALOG10(10*FRE)/2.0*255.0
      FREQ(J)=X(I)
20    CONTINUE
C
C     ASK FOR DATAFILE NAME
C     ---------------------
C
      CALL MSSG(IVDU,'DATAFILE NAME ? >\')
      READ(IVDU,1030)FILE
      IF(FILE(1).EQ.SPACE)STOP
      CALL MKOPEN(IN,FILE,0)
C
C       SET FLAG
C       --------
C
21      IFLAG=0
C
C     ASK FOR BREAKPOINTS
C     -------------------
C
      CALL MSSG(IVDU,'ENTER BREAKPOINTS IN FORMAT FREQ(KhZ),GAIN(Db)\')
      CALL MSSG(IVDU,'FREQ MUST BE IN INCREASING ORDER AND BETWEEN
     10.1Khz AND 8Khz\')
      CALL MSSG(IVDU,'TO END LIST ENTER 0,0\')
      CALL MSSG(IVDU,' \')
      DO 30 I=1,60
22    READ(IVDU,1000)BRKPNT(I,1),BRKPNT(I,2)
C
C       CHECK IF END OF LIST
C       --------------------
C
      IF(BRKPNT(I,1).EQ.0)GOTO 32
C
C     CHECK IF FIRST ENTRY
C     --------------------
C
      IF((I.EQ.1).AND.(BRKPNT(I,1).LT.0))GOTO 24
      IF(I.EQ.1)GOTO 27
C
C     CHECK IF SLOPE RELATIVE TO PREVIOUS OR NEXT POINT AND CALCULATE
C     ---------------------------------------------------------------
C
        IF(IFLAG.EQ.0)GOTO 23
        BRKPNT(I-1,1)=ABS(BRKPNT(I-1,1))
        BRKPNT(I-1,2)=ALOG10(BRKPNT(I-1,1)/BRKPNT(I,1))
     1*SLOPE*OCTAVE+BRKPNT(I,2)
        WRITE(IVDU,1070)BRKPNT(I-1,1),BRKPNT(I-1,2)
        CALL MSSG(IVDU,' \')
        IFLAG=0
        GOTO 30
C
C     CHECK IF VALID POINT
C     --------------------
C
23    IF((BRKPNT(I,1).GT.BRKPNT(I-1,1)).AND.(BRKPNT(I,1).GE.0.1).AND.
     1(BRKPNT(I,1).LE.8.0))GOTO 30
C
        IF(BRKPNT(I,1).GT.0)GOTO 28
24      CALL MSSG(IVDU,'SLOPE ? Db/OCTAVE >\')
        READ(IVDU,1050)SLOPE
      IF(I.NE.1)GOTO 25
      IFLAG=1
      GOTO 30
25      CALL MSSG(IVDU,'RELATIVE TO PREVIOUS(0) OR NEXT(1)
     1  POINT ? (0/1) >\')
        READ(IVDU,1060)IREL
        IF(IREL.EQ.1)IFLAG=1
C
C     CHECK IF SLOPE RELATIVE TO PREVIOUS OR NEXT POINT AND CALCULATE
C     ---------------------------------------------------------------
C
        IF(IFLAG.EQ.1)GOTO 30
        BRKPNT(I,1)=ABS(BRKPNT(I,1))
        BRKPNT(I,2)=SLOPE*OCTAVE*ALOG10(BRKPNT(I,1)/
     1BRKPNT(I-1,1))+BRKPNT(I-1,2)
        WRITE(IVDU,1080)BRKPNT(I,1),BRKPNT(I,2)
        CALL MSSG(IVDU,' \')
      GOTO 30
27    IF((BRKPNT(I,1).GE.0.1).AND.
     1(BRKPNT(I,1).LE.8.0))GOTO 30
28    CALL MSSG(IVDU,'FREQ MUST BE IN INCREASING ORDER AND BETWEEN
     10.1Khz AND 8Khz\')
      CALL MSSG(IVDU,' \')
      GOTO 22
30    CONTINUE
C
C     WRITE OUT AND CHECK BREAKPOINTS
C     -------------------------------
```

```
C
32    CALL MSSG(IVDU,' \')
      CALL MSSG(IVDU,'       BANDTOP(Khz)      GAIN(Db)\')
      CALL MSSG(IVDU,' \')
      DO 34 J=1,I
      WRITE(IVDU,1020)(BRKPNT(J,K),K=1,2)
34    CONTINUE
      CALL MSSG(IVDU,'IS DATA CORRECT Y/N ? >\')
      READ(IVDU,1030)YESNO
      IF(YESNO(1).NE.Y)GOTO 21
40    K=I-1
C
C     INTERPOLATE DATA VALUES
C     ----------------------
C
      J=1
      DO 200 I=1,K
C
C     CHECK THAT NO BREAKPOINTS SAME FREQ
C     -----------------------------------
C
      IF(BRKPNT(I,1).EQ.BRKPNT(I+1,1))GOTO 200
C
      LF=ALOG10(BRKPNT(I,1))
      LFG=BRKPNT(I,2)
      HF=ALOG10(BRKPNT(I+1,1))
      HFG=BRKPNT(I+1,2)
      SLOPE=(HFG-LFG)/(HF-LF)
      RINT=(LFG*HF-HFG*LF)/(HF-LF)
85    GAIN(J)=SLOPE*ALOG10(FREQ(J))+RINT
      IF(J.EQ.60)GOTO 210
      J=J+1
      IF(I.EQ.K)GOTO 90
      IF(J.EQ.60)GOTO 90
      IF(FREQ(J).GE. BRKPNT(I+1,1))GOTO 200
90    GOTO 85
200   CONTINUE
210   GAIN(61)=-200
      FREQ(61)=10.0
C
C     WRITE OUT BREAKPOINTS
C     --------------------
C
      CALL MSSG(IVDU,' \')
      CALL MSSG(IVDU,'       BANDTOP(Khz)      GAIN(Db)\')
      CALL MSSG(IVDU,' \')
      DO 220 J=1,61
      WRITE(IVDU,1020)FREQ(J),GAIN(J)
220   CONTINUE
C
C     WRITE INTO DATA FILE
C     ------------------
C
      WRITE(IN,1040)(FREQ(J),GAIN(J),J=1,61)
      ENDFILE IN
C
C     ANOTHER CALCULATION ?
C     --------------------
C
      CALL MSSG(IVDU,'ANOTHER CALCULATION ? (Y/N) > \')
      READ(IVDU,1030)YESNO
      IF(YESNO(1).EQ.Y)GOTO 20
C
1000  FORMAT(2F12.0)
1020  FORMAT(1H ,10X,F6.3,7X,F6.1)
1030  FORMAT(20A1)
1040  FORMAT(1H,F6.3,',',F6.1)
1050    FORMAT(F12.0)
1060    FORMAT(I1)
1070    FORMAT('PREVIOUS VALUES ARE >',2F12.3)
1080    FORMAT('CORRECT VALUES ARE >',2F12.3)
      END
```

```
      PROGRAM LCLINT
C
C     MIKE WALD MARCH 1983
C
C
      REAL LF,LFG
      BYTE FILE(12),SPACE,YESNO(20),Y
      DIMENSION FREQ(70),X(256),BRKPNT(70,2),GAIN(70),SPEECH(11,2)
      DATA IVDU/1/,IN/6/,INA/7/,SPACE/32/,Y/89/
        DATA SPEECH/0.1,0.25,0.50,0.75,1.0,1.5,2.0,3.0,4.0,6.0,8.0,
     1              -6.0,-6.0,-4.0,-2.0,-1.,0.0,0.0,0.0,0.0,0.0,0.0/
C
C     SET UP 256 LOG SPACED FREQ POINTS
C     -------------------------------
C
      RINC=10**(2.0/255.0)
      DO 10 I=1,256
      X(I)=0.1*RINC**(I-1)
10    CONTINUE
C
C     SET UP 'NEAREST' 60 LIN SPACED POINTS
C     -------------------------------------
C
      DO 20 J=1,60
      RJ=J-1
      FRE=0.1+7.9*RJ/59.0
      I=1.0+ALOG10(10*FRE)/2.0*255.0
      FREQ(J)=X(I)
20    CONTINUE
C
C     ASK FOR UCL DATAFILE NAME
C     -----------------------
C
      CALL MSSG(IVDU,'UCL DATAFILE NAME ? >\')
      READ(IVDU,1030)FILE
      IF(FILE(1).EQ.SPACE)STOP
      CALL MKOPEN(IN,FILE,0)
C
C     ASK FOR ATTENUATOR VALUES
C     -----------------------
C
21      CALL MSSG(IVDU,'ENTER ATTENUATOR VALUE FOR 250 Hz>\')
        READ(IVDU,1000)BRKPNT(2,2)
        CALL MSSG(IVDU,'ENTER ATTENUATOR VALUE FOR 500 Hz>\')
        READ(IVDU,1000)BRKPNT(3,2)
        CALL MSSG(IVDU,'ENTER ATTENUATOR VALUE FOR 750 Hz>\')
        READ(IVDU,1000)BRKPNT(4,2)
        CALL MSSG(IVDU,'ENTER ATTENUATOR VALUE FOR 1  KHz>\')
        READ(IVDU,1000)BRKPNT(5,2)
        CALL MSSG(IVDU,'ENTER ATTENUATOR VALUE FOR 1.5KHz>\')
        READ(IVDU,1000)BRKPNT(6,2)
        CALL MSSG(IVDU,'ENTER ATTENUATOR VALUE FOR 2  KHz>\')
        READ(IVDU,1000)BRKPNT(7,2)
        CALL MSSG(IVDU,'ENTER ATTENUATOR VALUE FOR 3  KHz>\')
        READ(IVDU,1000)BRKPNT(8,2)
        CALL MSSG(IVDU,'ENTER ATTENUATOR VALUE FOR 4  KHz>\')
        READ(IVDU,1000)BRKPNT(9,2)
        CALL MSSG(IVDU,'ENTER ATTENUATOR VALUE FOR 6  KHz>\')
        READ(IVDU,1000)BRKPNT(10,2)
C
C       SET ATT VALUE OF 100Hz=ATT VALUE OF 250 Hz
C       -----------------------------------------
C
        BRKPNT(1,2)=BRKPNT(2,2)
C
C       SET ATT VALUE OF 8Khz=ATT VALUE OF 6Khz
C       -------------------------------------
C
        BRKPNT(11,2)=BRKPNT(10,2)
C
C
C     WRITE OUT BREAKPOINTS
C     -------------------
C
32    CALL MSSG(IVDU,' \')
      CALL MSSG(IVDU,'          BANDTOP(Khz)       ATT(Db)\')
      CALL MSSG(IVDU,' \')
      DO 34 J=1,11
      WRITE(IVDU,1020)SPEECH(J,1),BRKPNT(J,2)
34    CONTINUE
      CALL MSSG(IVDU,'IS DATA CORRECT Y/N ? >\')
      READ(IVDU,1030)YESNO
      IF(YESNO(1).NE.Y)GOTO 21
C
C       TAKE ATT VALUE FROM SPEECH VALUE
C       ------------------------------
C
        DO 70 I=1,11
        BRKPNT(I,2)=SPEECH(I,2)-BRKPNT(I,2)
C
C       PUT IN FREQ
C       ----------
C
        BRKPNT(I,1)=SPEECH(I,1)
C
70      CONTINUE
C
C       NORMALISE ATT VALUES SO THAT LARGEST IS ZERO
C       -------------------------------------------
C
C       FIRST FIND LARGEST VALUE
C       -----------------------
C
        DO 80 J=1,11
        L=J+1
        DO 75 K=L,11
        IF(BRKPNT(J,2).LT.BRKPNT(K,2))GOTO 80
75      CONTINUE
C
C       THIS IS LARGEST VALUE
C       --------------------
C
        RLRGST=BRKPNT(J,2)
        GOTO 82
```

```
80      CONTINUE
C
C       TAKE LARGEST VALUE AWAY FROM EACH TO NORMALISE TO 0
C       -------------------------------------------------
C
82      DO 83 K=1,11
        BRKPNT(K,2)=BRKPNT(K,2)-RLRGST
83      CONTINUE
C
C
C     WRITE OUT BREAKPOINTS
C     --------------------
C
      CALL MSSG(IVDU,' \')
      CALL MSSG(IVDU,'        BANDTOP(Khz)        ATT(Db)\')
      CALL MSSG(IVDU,' \')
      DO 84 J=1,11
      WRITE(IVDU,1020)SPEECH(J,1),BRKPNT(J,2)
84    CONTINUE
C
C
C     INTERPOLATE DATA VALUES
C     ----------------------
C
      J=1
      DO 200 I=1,10
      LF=ALOG10(BRKPNT(I,1))
      LFG=BRKPNT(I,2)
      HF=ALOG10(BRKPNT(I+1,1))
      HFG=BRKPNT(I+1,2)
      SLOPE=(HFG-LFG)/(HF-LF)
      RINT=(LFG*HF-HFG*LF)/(HF-LF)
85    GAIN(J)=SLOPE*ALOG10(FREQ(J))+RINT
        IF(J.EQ.60)GOTO 210
        J=J+1
      IF(I.EQ.10)GOTO 90
        IF(J.EQ.60)GOTO 90
      IF(FREQ(J).GE. BRKPNT(I+1,1))GOTO 200
90    GOTO 85
200   CONTINUE
210   GAIN(61)=-200
      FREQ(61)=10.0
C
C     WRITE OUT BREAKPOINTS
C     --------------------
C
      CALL MSSG(IVDU,' \')
      CALL MSSG(IVDU,'        BANDTOP(Khz)        GAIN(Db)\')
      CALL MSSG(IVDU,' \')
      DO 220 J=1,61
      WRITE(IVDU,1020)FREQ(J),GAIN(J)
220   CONTINUE
C
C     WRITE INTO DATA FILE
C     --------------------
C
      WRITE(IN,1040)(FREQ(J),GAIN(J),J=1,61)

      ENDFILE IN
C
C     ANOTHER CALCULATION ?
C     --------------------
C
      CALL MSSG(IVDU,'ANOTHER CALCULATION ? (Y/N) > \')
      READ(IVDU,1030)YESNO
      IF(YESNO(1).EQ.Y)GOTO 20
C
1000  FORMAT(F12.0)
1020  FORMAT(1H ,10X,F6.3,7X,F6.1)
1030  FORMAT(20A1)
1040  FORMAT(1H,F6.3,',',F6.1)
      END
```

```
      PROGRAM PREFER
C
C     MIKE WALD NOVEMBER 1983
C
C
      BYTE FILE(12),SPACE,YESNO(20),Y
      DIMENSION FREQA(70),FREQB(70),FREQP(70),
     1 GAINA(70),GAINB(70),GAIN(70)
      DATA IVDU/1/,IN/6/,SPACE/32/,Y/89/
C
C     ASK FOR UCL DATAFILE NAME
C     ------------------------
C
20    CALL MSSG(IVDU,'UCL DATAFILE NAME ? >\')
      READ(IVDU,1030)FILE
      IF(FILE(1).EQ.SPACE)STOP
C
C     READ IN FREQ AND GAIN VALUES
C     ----------------------------
C
      CALL MKOPEN(IN,FILE,0)
      READ(IN,1010,END=50)(FREQA(I),GAINA(I),I=1,70)
50    ENDFILE IN
C
C     ASK FOR PREFERED SLOPE FILE NAME
C     --------------------------------
C
      CALL MSSG(IVDU,'PREF SLOPE FILE NAME (FLAT=RETURN)? >\')
      READ(IVDU,1030)FILE
C
C     CHECK IF FLAT
C     -------------
C
      IF(FILE(1).NE.SPACE)GOTO 100
      DO 60 I=1,61
      GAIN(I)=GAINA(I)
60    CONTINUE
      GOTO 220
C
C     READ IN FREQ AND GAIN VALUES
C     ----------------------------
C
100   CALL MKOPEN(IN,FILE,0)
      READ(IN,1010,END=150)(FREQB(I),GAINB(I),I=1,70)
150   ENDFILE IN
C
C     ADD GAINS TOGETHER
C     ------------------
C
      DO 200 I=1,61
      GAIN(I)=GAINA(I)+GAINB(I)
200   CONTINUE
      WRITE(IVDU,2000)(GAINA(I),GAINB(I),GAIN(I),I=1,61)
C
C     FIND PREFERED LOW FREQ CUTOFF
C     -----------------------------
C
220   CALL MSSG(IVDU,'WHAT IS PREFERRED LOW FREQ CUT OFF ?\')
      CALL MSSG(IVDU,'100 HZ   > 1\')
      CALL MSSG(IVDU,'250 HZ   > 2\')
      CALL MSSG(IVDU,'400 HZ   > 3\')
      CALL MSSG(IVDU,' >\')
      READ(IVDU,1050)ILOW
C
C     FIND PREFERED HIGH FREQ CUTOFF
C     ------------------------------
C
      CALL MSSG(IVDU,'WHAT IS PREFERRED HIGH FREQ CUT OFF ?\')
      CALL MSSG(IVDU,'2  KHZ   > 2\')
      CALL MSSG(IVDU,'4  KHZ   > 4\')
      CALL MSSG(IVDU,'6  KHZ   > 6\')
      CALL MSSG(IVDU,'8  KHZ   > 8\')
      CALL MSSG(IVDU,' >\')
      READ(IVDU,1050)IHIGH
      IF(IHIGH.EQ.2)IHIGH=16
      IF(IHIGH.EQ.4)IHIGH=31
      IF(IHIGH.EQ.6)IHIGH=46
      IF(IHIGH.EQ.8)IHIGH=61
```

```
C
C       NORMALISE MAX GAIN TO 0
C       ----------------------
C
C       FIRST FIND LARGEST VALUE
C       ------------------------
C
        JLOW=ILOW+1
        JHIGH=IHIGH-2
        KHIGH=IHIGH-1
        DO 280 J=JLOW,JHIGH
        L=J+1
        DO 275 K=L,KHIGH
        IF(GAIN(J).LT.GAIN(K))GOTO 280
275     CONTINUE
C
C       THIS IS LARGEST VALUE
C       ---------------------
C
        RLRGST=GAIN(J)
        GOTO 282
280     CONTINUE
        RLRGST=GAIN(KHIGH)
C
C       TAKE LARGEST VALUE AWAY FROM EACH TO NORMALISE TO 0
C       ----------------------------------------------------
C
282     DO 300 K=ILOW,IHIGH
        GAIN(K)=GAIN(K)-RLRGST
300     CONTINUE
C
C       SET MAX & MIN FREQ GAIN TO -200
C       -------------------------------
C
        GAIN(IHIGH)=-200.0
        GAIN(ILOW)=-200.0
C
C       ASK FOR PREFER FILE NAME
C       ------------------------
C
        CALL MSSC(IVDU,'PREFERED OUTPUT FILE NAME ? >\')
        READ(IVDU,1030)FILE
        IF(FILE(1).EQ.SPACE)STOP
        CALL MKOPEN(IN,FILE,0)
C
C       WRITE INTO DATA FILE
C       --------------------
C
        WRITE(IN,1040)(FREQA(I),GAIN(I),I=ILOW,IHIGH)
        ENDFILE IN
C
C       ANOTHER CALCULATION ?
C       ---------------------
C
        CALL MSSC(IVDU,'ANOTHER CALCULATION ? (Y/N) > \')
        READ(IVDU,1030)YESNO
        IF(YESNO(1).EQ.Y)GOTO 20
C
1010    FORMAT(2F12.0)
1030    FORMAT(20A1)
1040    FORMAT(1H,F6.3,',',F6.1)
1050    FORMAT(I1)
2000    FORMAT(1H ,F6.1,'      ',F6.1,'      ',F6.1)
        END
```

```
      PROGRAM LOADAB
C
C
C     M WALD     AUG 1983
C
C
      BYTE SPACE,VAL,PORT,A,B,AB
      BYTE BUF(12),SCALE
      INTEGER ATT
      INTEGER*2 ICODEA(1408),ICODEB(1408)
      EQUIVALENCE (IVAL,VAL),(IPORT,PORT),(ISCALE,SCALE)
      DATA SPACE/32/, IVDU/1/, IN/6/, IEND/999/,A/65/,B/66/
C
C
C      SET 8PIO PORT 7 TO OUT
C      ---------------------
C
       CALL OUT(134,-115)
C
C
C
1     CALL MSSG(IVDU,'LOAD FILTER A(TUART) OR B(8PIO)PROGRAM\')
      CALL MSSG(IVDU,'--------------------------------------\')
C
C      ASK FOR FILTER FILE NAME
C      ------------------------
C
10     CALL MSSG(IVDU,'FILTER FILE NAME ? >\')
       READ(IVDU,1000)BUF
       IF(BUF(1).EQ.SPACE)STOP
C
C      ASK FOR FILTER A OR B
C      ---------------------
C
12     CALL MSSG(IVDU,'LOAD FILTER A OR B ? (A/B) >\')
       READ(IVDU,1000)AB
       IF(AB.EQ.A)PORT=164
       IF(AB.EQ.B)PORT=135
       IF(AB.NE.A.AND.AB.NE.B)GOTO 12
       CALL MKOPEN(IN,BUF,0)
C
C
C     READ IN FILE
C
C
      READ(IN,ERR=30,END=30)ICODEA
25    ENDFILE IN
      GOTO 100
30    CALL MSSG(IVDU,'READ ERROR\')
      ENDFILE IN
       GOTO 10
100   DO 160 J=1,1408
              IVAL=ICODEA(J)
              IF(IVAL.EQ.IEND)GOTO 170
              CALL OUT(PORT,VAL)
160   CONTINUE
C
C       DISPLAY NORMAL SCALE VALUE
C       --------------------------
C
170     J=J-1
171     ISCALE=ICODEA(J)-15
        IF(ISCALE.GE.1.AND.ISCALE.LE.16)GOTO 175
                CALL MSSG(IVDU,'BAD FILE FORMAT\')
                GOTO 10
175     CALL MSSG(IVDU,' \')
        CALL MSSG(IVDU,'NORMAL SCALE VALUE:\')
        WRITE(IVDU,1020)ISCALE
C
C       ASK FOR SCALE FACTOR
C       --------------------
C
180     CALL MSSG(IVDU,'SCALE VALUE ?(1-16)>\')
        READ(IVDU,1100)ISCALE
        IF(ISCALE.LT.1.OR.ISCALE.GT.16)GOTO 10
C
C       OUTPUT NEW SCALE
C       ----------------
C
        ISCALE=ISCALE+15
        CALL OUT(PORT,SCALE)
        GOTO 180
C
C
1000  FORMAT(20A1)
1020   FORMAT(1H ,I2)
1100  FORMAT(I2)
C
C
      END
```

```
      PROGRAM PAIRCM
C
C
C     M WALD     JUNE 1983
C
C
      BYTE SPACE,VAL
       BYTE FILEA(13,40),FILEB(13,40),FILE(13,80)
      BYTE BUF(12),SCALEA,SCALEB
      INTEGER ATT
      INTEGER*2 ICODEA(1408),ICODEB(1408)
      EQUIVALENCE (IVAL,VAL)
      DATA SPACE/32/, IVDU/1/, IN/6/, IEND/999/
C
C
1     CALL MSSG(IVDU,'PAIR COMPARISON PROGRAM\')
      CALL MSSG(IVDU,'-----------------------\')
C
C      ASK FOR NAME OF FILE CONTAINING PAIRS FOR COMPARISON
C      ----------------------------------------------------
C
       CALL MSSG(IVDU,'FILENAME OF FILE OF PAIRS FOR COMPARISON ? >\')
       READ(IVDU,1000)BUF
       IF(BUF(1).EQ.SPACE)STOP
       CALL MKOPEN(IN,BUF,0)
       READ(IN,END=2)((FILE(J,I),J=1,13),I=1,80)
       ENDFILE IN
       IPAIR=40
       GOTO 3
2      ENDFILE IN
C
       IPRINT=0
C
C      SET No OF PAIRS FLAG
C      --------------------
C
       IPAIR=(I-1)/2
C
C      TRANFER INTO FILEA AND FILEB
C      ----------------------------
C
3      DO 6 I=1,IPAIR
       IA=2*I-1
       IB=2*I
       DO 4 J=1,13
4      FILEA(J,I)=FILE(J,IA)
       DO 5 J=1,13
5      FILEB(J,I)=FILE(J,IB)
6      CONTINUE
C
C     SET FILE FLAG
C     -------------
C
7      K=0
C
C      RESET ATT TO 99
C      ---------------
C
       CALL OUT(180,153)
C
C
C     FIND NEXT PAIR
C     --------------
C
8      CALL MSSG(IVDU,'NEXT PAIR ?\')
       CALL MSSG(IVDU,'ENTER 0 IF NEW SET OF PAIRS REQUIRED\')
C
C      ONLY PRINT LIST 1ST TIME
C      ------------------------
C
       IF(IPRINT.EQ.1)GOTO 10
       DO 9 I=1,IPAIR
9      WRITE(IVDU,2000)I,(FILEA(J,I),J=1,12),FILEA(13,I),
     1 (FILEB(J,I),J=1,12),FILEB(13,I)
10     CALL MSSG(IVDU,' > \')
       READ(IVDU,1100)NXTPR
       IF((NXTPR.GT.IPAIR).OR.(NXTPR.LT.0))GOTO 8
       IF(NXTPR.EQ.0)GOTO 1
C
       IPRINT=1
C
14     DO 15 I=1,12
15     BUF(I)=FILEA(I,NXTPR)
       SCALEA=FILEA(13,NXTPR)+15
       GOTO 20
16     DO 17 I=1,12
17     BUF(I)=FILEB(I,NXTPR)
       SCALEB=FILEB(13,NXTPR)+15
20     CALL MKOPEN(IN,BUF,0)
C
C
C     READ IN FILE
C
C
      IF (K.EQ.1) GOTO 22
      READ(IN,ERR=30,END=30)ICODEA
      GOTO 25
22    READ(IN,ERR=35,END=35)ICODEB
25    ENDFILE IN
      GOTO 40
30    CALL MSSG(IVDU,'READ ERROR\')
      ENDFILE IN
      GOTO 14
35    CALL MSSG(IVDU,'READ ERROR\')
      ENDFILE IN
      GOTO 16
40    K=K+1
      IF(K.EQ.1)GOTO 16
50    CALL MSSG(IVDU,'ATTENUATOR SETTING G? >\')
      READ(IVDU,1100,ERR=50)ATT
      IF ((ATT.LT.0).OR.(ATT.GT.99))GOTO 50
      M=ATT
60    CALL MSSG(IVDU,'ATTENUATOR SETTING R? >\')
```

```
      READ(IVDU,1100,ERR=50)ATT
      IF ((ATT.LT.0).OR.(ATT.GT.99))GOTO 60
      L=ATT
      N=0
       GOTO 100
95    CALL BTIN(K,ATT,N)
      IF (N.EQ.1)GOTO300
      IF (K.EQ.1)GOTO 200
      L=ATT
100   DO 160 J=1,1408
              IVAL=ICODEA(J)
              IF(IVAL.EQ.IEND)GOTO 170
              CALL OUT(164,VAL)
160   CONTINUE
170    IF(SCALEA.EQ.15)GOTO 180
       CALL OUT(164,SCALEA)
180           ATT=M
      GOTO 95
200   M=ATT
      DO 260 J=1,1408
              IVAL=ICODEB(J)
            , IF(IVAL.EQ.IEND)GOTO 270
              CALL OUT(164,VAL)
260   CONTINUE
270    IF(SCALEB.EQ.15)GOTO 280
       CALL OUT(164,SCALEB)
280   ATT=L
      GOTO 95
300   CALL MSSG(IVDU,'CONTINUE ? >\')
      READ(IVDU,1000)BUF
      IF (BUF(1).EQ.27)GOTO 7
      N=0
      GOTO 95
C
C
1000  FORMAT(20A1)
1100  FORMAT(I2)
       FORMAT(1X,'PAIR',I2)
2000   FORMAT(1X,'PAIR ',I2,'  G:',12A1,' SCALE-',I2,'  R:',12A1,
     1  ' SCALE-',I2)
C
C
      END
```

```
      SUBROUTINE BTIN(K,ATT,N)
C
C
C     READS VALUE OF RESPONSE BUTTONS
C
C
      IMPLICIT INTEGER(A,B)
      GOTO 115
96    DO 100 I=1,100
      DO 100 J=1,100
100   BUT=INP(180)+256
      IF((BUT.EQ.239).AND.(K.EQ.1))GOTO 150
      IF((BUT.EQ.247).AND.(K.EQ.2))GOTO 160
101   IF((BUT.EQ.235).OR.(BUT.EQ.243))  GOTO 106
102   IF((BUT.EQ.246).OR.(BUT.EQ.238))  GOTO 107
104   IF((BUT.EQ.245).OR.(BUT.EQ.237))  GOTO 108
105   GOTO 100
106   CONTINUE
      N=1
      RETURN
107   ATT=ATT-1
      IF (ATT.LT.0)GOTO 110
      GOTO 115
108   ATT=ATT+1
      IF( ATT.GT.99)GOTO 112
      GOTO 115
110   ATT=0
      GOTO 100
112   ATT=99
      GOTO 100
115   A=ATT/10
120   B=6*A+ATT
130   CALL OUT(180,B)
      IF (K.EQ.1)GOTO 138
135   WRITE(1,1000)ATT
      GOTO 140
138   WRITE(1,1100)ATT
140   GOTO 96
150   K=2
      CALL OUT(180,153)
      RETURN
160   K=1
      CALL OUT(180,153)
      RETURN
1000  FORMAT(I3,4HDb-G)
1100  FORMAT(I3,4HDb-R)
      END
```

```
      PROGRAM PRCMSW
C
C
C     M WALD     JUNE 1983
C
C
      BYTE SPACE,VAL
       BYTE FILEA(13,40),FILEB(13,40),FILE(13,80)
      BYTE BUF(12),SCALEA,SCALEB
      INTEGER ATT
      INTEGER*2 ICODEA(1408),ICODEB(1408)
      EQUIVALENCE (IVAL,VAL)
      DATA SPACE/32/, IVDU/1/, IN/6/, IEND/999/
C
C      SET 8PIO PORT 1 TO OUT
C      ---------------------
C
       CALL OUT(134,2)
C
C
1     CALL MSSG(IVDU,'PAIR COMPARISON 2 CHANNELS\')
      CALL MSSG(IVDU,'--------------------------\')
C
C      ASK FOR NAME OF FILE CONTAINING PAIRS FOR COMPARISON
C      ----------------------------------------------------
C
       CALL MSSG(IVDU,'FILENAME OF FILE OF PAIRS FOR COMPARISON ? >\')
       READ(IVDU,1000)BUF
       IF(BUF(1).EQ.SPACE)STOP
       CALL MKOPEN(IN,BUF,0)
       READ(IN,END=2)((FILE(J,I),J=1,13),I=1,80)
       ENDFILE IN
       IPAIR=40
       GOTO 3
2      ENDFILE IN
C
C      SET No OF PAIRS FLAG
C      --------------------
C
       IPAIR=(I-1)/2
C
C      TRANFER INTO FILEA AND FILEB
C      ----------------------------
C
C
3      DO 6 I=1,IPAIR
       IA=2*I-1
       IB=2*I
       DO 4 J=1,13
4      FILEA(J,I)=FILE(J,IA)
       DO 5 J=1,13
5      FILEB(J,I)=FILE(J,IB)
6      CONTINUE
C
C     SET FILE FLAG
C     -------------
C
7     K=0
C
C      SET ATT TO 99
C      -------------
C
       CALL OUT(180,153)
C
C
C     FIND NEXT PAIR
C     --------------
C
8      CALL MSSG(IVDU,'NEXT PAIR ?\')
       CALL MSSG(IVDU,'ENTER 0 IF NEW SET OF PAIRS REQUIRED\')
       DO 9 I=1,IPAIR
9      WRITE(IVDU,2000)I,(FILEA(J,I),J=1,12),FILEA(13,I),
     1 (FILEB(J,I),J=1,12),FILEB(13,I)
       CALL MSSG(IVDU,' > \')
       READ(IVDU,1100)NXTPR
       IF((NXTPR.GT.IPAIR).OR.(NXTPR.LT.0))GOTO 8
       IF(NXTPR.EQ.0)GOTO 1
14     DO 15 I=1,12
15     BUF(I)=FILEA(I,NXTPR)
       SCALEA=FILEA(13,NXTPR)+15
       GOTO 20
16     DO 17 I=1,12
17     BUF(I)=FILEB(I,NXTPR)
       SCALEB=FILEB(13,NXTPR)+15
20     CALL MKOPEN(IN,BUF,0)
C
C
C     READ IN FILE
C
C
      IF (K.EQ.1) GOTO 22
      READ(IN,ERR=30,END=30)ICODEA
      GOTO 25
22    READ(IN,ERR=35,END=35)ICODEB
25    ENDFILE IN
      GOTO 40
30    CALL MSSG(IVDU,'READ ERROR\')
      ENDFILE IN
      GOTO 14
35    CALL MSSG(IVDU,'READ ERROR\')
      ENDFILE IN
      GOTO 16
40    K=K+1
      IF(K.EQ.1)GOTO 16
50    CALL MSSG(IVDU,'ATTENUATOR SETTING G? >\')
      READ(IVDU,1100,ERR=50)ATT
      IF ((ATT.LT.0).OR.(ATT.GT.99))GOTO 50
      M=ATT
60    CALL MSSG(IVDU,'ATTENUATOR SETTING R? >\')
      READ(IVDU,1100,ERR=50)ATT
      IF ((ATT.LT.0).OR.(ATT.GT.99))GOTO 60
      L=ATT
      N=0
       GOTO 100
95    CALL BTIN(K,ATT,N)
```

```
      IF (N.EQ.1)GOTO300
      IF (K.EQ.1)GOTO 200
      L=ATT
100   DO 160 J=1,1408
              IVAL=ICODEA(J)
              IF(IVAL.EQ.IEND)GOTO 170
              CALL OUT(164,VAL)
160   CONTINUE
170    IF(SCALEA.EQ.15)GOTO 175
       CALL OUT(164,SCALEA)
C
C      SWITCH CHANNELS
C      ---------------
C
175    CALL OUT(129,0)
180           ATT=M
      GOTO 95
200   M=ATT
      DO 260 J=1,1408
              IVAL=ICODEB(J)
              IF(IVAL.EQ.IEND)GOTO 270
              CALL OUT(164,VAL)
260   CONTINUE
270    IF(SCALEB.EQ.15)GOTO 275
       CALL OUT(164,SCALEB)
C
C      SWITCH CHANNELS
C      ---------------
C
275    CALL OUT(129,1)
280   ATT=L
      GOTO 95
300   CALL MSSC(IVDU,'CONTINUE ? >\')
      READ(IVDU,1000)BUF
      IF (BUF(1).EQ.27)GOTO 7
      N=0
      GOTO 95
C
C
1000  FORMAT(20A1)
1100  FORMAT(I2)
1500   FORMAT(1X,'PAIR',I2)
2000   FORMAT(1X,'PAIR ',I2,'  G:',12A1,' SCALE-',I2,'  R:',12A1,
     1 ' SCALE-',I2)
C
C
      END
```

```
      PROGRAM PRCMFN
C
C
C     M WALD      JUNE 1983
C
C
      BYTE SPACE
       BYTE FILE(13,80)
      BYTE BUF(12)
      DATA SPACE/32/, IVDU/1/, IN/6/, IEND/999/
C
C
1     CALL MSSG(IVDU,'PAIR COMPARISON FILENAME FILE CREATE PROGRAM\')
      CALL MSSG(IVDU,'--------------------------------------------\')
C
C      SET No OF PAIRS FLAG
C      -------------------
C
       IPAIR=40
       DO 5  I=1,40
C
C      ODD NO FILES=A EVEN NO=B
C      ------------------------
C
       IA=2*I-1
       IB=2*I
       CALL MSSG(IVDU,'ANOTHER PAIR ? Y/N >\')
       READ(IVDU,1000)BUF
       IF(BUF(1).NE.89)GOTO 6
       WRITE(IVDU,1500)I
C
C     ASK FOR FILENAMES OF PAIRS FOR COMPARISON
C     -----------------------------------------
C
      CALL MSSG(IVDU,' \')
      CALL MSSG(IVDU,'FILENAME A ? >\')
      READ(IVDU,1000)(FILE(J,IA),J=1,12)
       IF(FILE(1,IA).EQ.SPACE)STOP
       CALL MSSG(IVDU,'SCALEA ? >\')
       READ(IVDU,1100)FILE(13,IA)
      CALL MSSG(IVDU,' \')
      CALL MSSG(IVDU,'FILENAME B ? >\')
      READ(IVDU,1000)(FILE(J,IB),J=1,12)
      IF(FILE(1,IB).EQ.SPACE)STOP
       CALL MSSG(IVDU,'SCALEB ? >\')
       READ(IVDU,1100)FILE(13,IB)
5      CONTINUE
       GOTO 8
6     IPAIR=I-1
       IF(IPAIR.EQ.0)GOTO 1
8      DO 9 I=1,IPAIR
       IA=2*I-1
       IB=2*I
9      WRITE(IVDU,2000)I,(FILE(J,IA),J=1,12),FILE(13,IA)
     1 ,(FILE(J,IB),J=1,12),FILE(13,IB)
       CALL MSSG(IVDU,'ARE ENTRIES CORRECT ? Y/N > \')
       READ(IVDU,1000)BUF
       IF(BUF(1).EQ.89)GOTO 50
       CALL MSSG(IVDU,'WHICH PAIR DO YOU WISH TO CORRECT ?>\')
       READ(IVDU,1100)ICORR
       IA=2*ICORR-1
       IB=2*ICORR
10    CALL MSSG(IVDU,' \')
      CALL MSSG(IVDU,'FILENAME A ? >\')
      READ(IVDU,1000)(FILE(J,IA),J=1,12)
       IF(FILE(1,IA).EQ.SPACE)GOTO 10
       CALL MSSG(IVDU,'SCALEA ? >\')
       READ(IVDU,1100)FILE(13,IA)
20    CALL MSSG(IVDU,' \')
      CALL MSSG(IVDU,'FILENAME B ? >\')
      READ(IVDU,1000)(FILE(J,IB),J=1,12)
      IF(FILE(1,IB).EQ.SPACE)GOTO 20
       CALL MSSG(IVDU,'SCALEB ? >\')
       READ(IVDU,1100)FILE(13,IB)
       GOTO 8
50     CALL MSSG(IVDU,'WHAT IS FILENAME OF FILE OF NAMES ? >\')
       READ(IVDU,1000)BUF
       CALL MKOPEN(IN,BUF,0)
       WRITE(IN,END=100)((FILE(J,I),J=1,13),I=1,IB)
100    ENDFILE IN
       GOTO 1
1000  FORMAT(20A1)
1100  FORMAT(I2)
1500   FORMAT(1X,'PAIR',I2)
2000   FORMAT(1X,'PAIR ',I2,'  A:',12A1,' SCALEA-',I2,'  B:',12A1,
     1 ' SCALEB-',I2,)
C
C
      END
```

```
      PROGRAM ULL3X
C
C
C     M WALD     AUG 1983
C
C
      BYTE SPACE,VAL
       BYTE FILEA(13,40),FILEB(13,40),FILE(13,80)
      BYTE BUF(12),SCALEA,SCALEB
      INTEGER ATT
      INTEGER*2 ICODEA(1408),ICODEB(1408)
      EQUIVALENCE (IVAL,VAL)
      DATA SPACE/32/, IVDU/1/, IN/6/, IEND/999/
C
C
1     CALL MSSG(IVDU,'ULL DETERMINATION PROGRAM\')
      CALL MSSG(IVDU,'-------------------------\')
C
C      ASK FOR NAME OF FILE CONTAINING NAMES FOR ULL
C      ---------------------------------------------
C
       CALL MSSG(IVDU,'FILENAME OF FILE OF NAMES FOR ULL ? >\')
       READ(IVDU,1000)BUF
       IF(BUF(1).EQ.SPACE)STOP
       CALL MKOPEN(IN,BUF,0)
       READ(IN,END=2)((FILE(J,I),J=1,13),I=1,80)
       ENDFILE IN
       IPAIR=40
       GOTO 3
2      ENDFILE IN
C
C      SET No OF PAIRS FLAG
C      -------------------
C
       IPAIR=(I-1)/2
C
C      TRANFER INTO FILEA AND FILEB
C      ---------------------------
C
3      DO 6 I=1,IPAIR
       IA=2*I-1
       IB=2*I
       DO 4 J=1,13
4      FILEA(J,I)=FILE(J,IA)
       IF(FILEA(1,I).EQ.SPACE)GOTO 7
6      CONTINUE
       GOTO 8
7      IPAIR=I-1
C
C
C     FIND NEXT NAME
C     --------------
C
8      CALL MSSG(IVDU,'NEXT NAME ?\')
       CALL MSSG(IVDU,'ENTER 0 IF NEW SET OF NAMES REQUIRED\')
       DO 9 I=1,IPAIR
9       WRITE(IVDU,2000)I,(FILEA(J,I),J=1,12),FILEA(13,I)
        CALL MSSG(IVDU,' > \')
        READ(IVDU,1100)NXTPR
        IF((NXTPR.GT.IPAIR).OR.(NXTPR.LT.0))GOTO 8
        IF(NXTPR.EQ.0)GOTO 1
14      DO 15 I=1,12
15      BUF(I)=FILEA(I,NXTPR)
        SCALEA=FILEA(13,NXTPR)+15
        GOTO 20
20      CALL MKOPEN(IN,BUF,0)
C
C
C      READ IN FILE
C
C
       READ(IN,ERR=30,END=30)ICODEA
25     ENDFILE IN
       GOTO 50
30     CALL MSSG(IVDU,'READ ERROR\')
       ENDFILE IN
        GOTO 8
50     CALL MSSG(IVDU,'ATTENUATOR SETTING G? >\')
       READ(IVDU,1100,ERR=50)ATT
       IF ((ATT.LT.0).OR.(ATT.GT.99))GOTO 50
        GOTO 100
95     CALL SUBULL(ATT)
        GOTO 8
100    DO 160 J=1,1408
               IVAL=ICODEA(J)
               IF(IVAL.EQ.IEND)GOTO 170
               CALL OUT(164,VAL)
160    CONTINUE
170     IF(SCALEA.EQ.15)GOTO 180
        CALL OUT(164,SCALEA)
180    CONTINUE
       GOTO 95
300    CALL MSSG(IVDU,'CONTINUE ? >\')
       READ(IVDU,1000)BUF
       IF (BUF(1).EQ.27)GOTO 7
       N=0
       GOTO 95
C
C
1000   FORMAT(20A1)
1100   FORMAT(I2)
1500    FORMAT(1X,'NAME',I2)
2000    FORMAT(1X,'NAME ',I2,'   G:',12A1,' SCALE-',I2)
C
C
       END
```

```
      SUBROUTINE SUBULL(ATT)
C
C     M WALD AUG 1983
C     ---------------
C
C     FINDS UNCOMFORTABLE LISTENING LEVEL
C     -----------------------------------
C
      IMPLICIT INTEGER(A,B)
      BYTE SPACE,YESNO,Y
      INTEGER ULL5(100,2),ULL1(100,2),THRESH,LEV
      DATA IVDU/1/, IN/6/, SPACE/32/, Y/89/
C
C     CHECK IF ANOTHER RUN REQUIRED
C     -----------------------------
C
      IFLG=0
10    WRITE(IVDU,15)
15    FORMAT(' PRESS ANY KEY TO START ULL DETERMINATION :PRESS T TO
     1 TERMINATE RUN')
      WRITE(IVDU,1400)
      READ(IVDU,1500)YESNO
      IF(YESNO.EQ.84)RETURN
C
C     SET MATRIX TO INITIAL VALUES
C     ----------------------------
C
      DO 20 I=1,100
      ULL5(I,2)=0
      ULL5(I,1)=0
      ULL1(I,1)=0
      ULL1(I,2)=0
20    CONTINUE
      I5THR=-99
      I1THR=-99
      IASC=0
C
C     SET INITIAL ATTENUATOR LEVEL
C     ----------------------------
C
      IF(IFLG.EQ.1)ATT=ATT+15
      IFLG=1
      GOTO 45
C
C     CHECK IF BUTTON PRESSED
C     -----------------------
C
30    DO 40 I=1,2000
      IF(IASC.EQ.0)GOTO 35
      BUT=INP(180)+256
      IF((BUT.EQ.245).OR.(BUT.EQ.237).OR.(BUT.EQ.253)) GOTO 50
35    DO 40 J=1,100
40    CONTINUE
      BUT=INP(180)+256
      IF((BUT.EQ.245).OR.(BUT.EQ.237).OR.(BUT.EQ.253)) GOTO 50
      ATT=ATT-5

      IASC=1
      IF(ATT.LT.0)ATT=0
      LEV=ATT+1
      ULL5(LEV,1)=ULL5(LEV,1)+1
45    A=ATT/10
      B=6*A+ATT
      CALL OUT(180,B)
      WRITE(1,1000)ATT
      GOTO 30
C
C     CHECK IF ASCENDING THRESHOLD
C     ----------------------------
C
50    IF(IASC.EQ.0)GOTO 550
C
C     CHECK IF 5db THRESHOLD REACHED
C     ------------------------------
C
      LEV=ATT+1
      INUM=(ULL5(LEV,1)+1)/2
510   ULL5(LEV,2)=ULL5(LEV,2)+1
      IF((ULL5(LEV,2).GE.2).AND.(ULL5(LEV,2).GE.INUM)) I5THR=ATT
      IF(I5THR.NE.-99)GOTO 980
550   CONTINUE
      ATT=ATT+10
      IASC=0
      IF(ATT.GT.99)ATT=99
      GOTO 45
600   CONTINUE
      INUMB=0
      ATT=ATT+5
      IASC=0
      GOTO 645
```

```
C
C       CHECK IF BUTTON PRESSED
C       ----------------------
C
630     DO 640 I=1,2000
        IF(IASC.EQ.0)GOTO 635
        BUT=INP(180)+256
        IF((BUT.EQ.245).OR.(BUT.EQ.237).OR.(BUT.EQ.253)) GOTO 650
635     DO 640 J=1,100
640     CONTINUE
        BUT=INP(180)+256
        IF((BUT.EQ.245).OR.(BUT.EQ.237).OR.(BUT.EQ.253)) GOTO 650
        ATT=ATT-1
        IASC=1
        IF(ATT.LT.0)ATT=0
        LEV=ATT+1
        ULL1(LEV,1)=ULL1(LEV,1)+1
645     IA=ATT/10
        IB=6*A+ATT
        CALL OUT(180,IB)
        WRITE(1,1000)ATT
        GOTO 630
C
C       CHECK IF ASCENDING THRESHOLD
C       ---------------------------
C
650     IF(IASC.EQ.0)GOTO 950
C
C       CHECK IF 1db THRESHOLD REACHED
C       ------------------------------
C
        LEV=ATT+1
        INUM=(ULL1(LEV,1)+1)/2
910     ULL1(LEV,2)=ULL1(LEV,2)+1
        IF((ULL1(LEV,2).GE.2).AND.(ULL1(LEV,2).GE.INUM)) I1THR=ATT
        IF(I1THR.NE.-99)GOTO 980
950     CONTINUE
        ATT=ATT+2
        IASC=0
        IF(ATT.GT.99)ATT=99
        GOTO 645
980     CALL OUT(180,153)
        WRITE(IVDU,1400)
        WRITE(IVDU,1200)I5THR
        WRITE(IVDU,1400)
        WRITE(IVDU,985)
985     FORMAT('BREAKDOWN OF LEVELS REQUIRED ? Y/N >')
        READ(IVDU,1500)YESNO
        IF(YESNO.NE.Y)GOTO 999
        WRITE(IVDU,1250)
        DO 990 I=1,100
        IF(ULL5(I,1).EQ.0)GOTO 990
        K=I-1
        WRITE(IVDU,1300)K,(ULL5(I,J),J=1,2)
990     CONTINUE
991     WRITE(IVDU,1400)
        GOTO 999
        WRITE(IVDU,1350)
        DO 995 I=1,100
        IF(ULL1(I,1).EQ.0)GOTO 995
        K=I-1
        WRITE(IVDU,1300)K,(ULL1(I,J),J=1,2)
995     CONTINUE
999     CONTINUE
        GOTO 10
1000    FORMAT(I3,'Db')
1200    FORMAT('      5Db ULL=',I2,'   1Db ULL=',I2)
1250    FORMAT('  5DB ULLS')
1300    FORMAT(I3,'Db -',I2,'PRES ',I2,'COR')
1350    FORMAT('  1DB ULLS')
1400    FORMAT(1X)
1500    FORMAT(A1)
        END
```

```
      PROGRAM MCL3X
C
C
C     M WALD     AUG 1983
C
C
      BYTE SPACE,VAL
       BYTE FILEA(13,40),FILEB(13,40),FILE(13,80)
      BYTE BUF(12),SCALEA,SCALEB
      INTEGER ATT
      INTEGER*2 ICODEA(1408),ICODEB(1408)
      EQUIVALENCE (IVAL,VAL)
      DATA SPACE/32/, IVDU/1/, IN/6/, IEND/999/
C
C
C      SET 8PIO PORT 7 TO OUT
C      ------*--------------
C
       CALL OUT(134,-115)
C
C
C
1      CALL MSSC(IVDU,'MCL DETERMINATION PROGRAM\')
       CALL MSSC(IVDU,'-------------------------\')
C
C       ASK FOR NAME OF FILE CONTAINING NAMES FOR MCL
C       ---------------------------------------------
C
        CALL MSSC(IVDU,'FILENAME OF FILE OF NAMES FOR MCL ? >\')
        READ(IVDU,1000)BUF
        IF(BUF(1).EQ.SPACE)STOP
        CALL MKOPEN(IN,BUF,0)
        READ(IN,END=2)((FILE(J,I),J=1,13),I=1,80)
        ENDFILE IN
        IPAIR=40
        GOTO 3
2       ENDFILE IN
C
C
C     SET No OF PAIRS FLAG
C     --------------------
      IPAIR=(I-1)/2
C
C     TRANFER INTO FILEA AND FILEB
C     ----------------------------
C
3     DO 6 I=1,IPAIR
      IA=2*I-1
      IB=2*I
      DO 4 J=1,13
4     FILEA(J,I)=FILE(J,IA)
      IF(FILEA(1,I).EQ.SPACE)GOTO 7
6     CONTINUE
      GOTO 8
7     IPAIR=I-1
C
C
C     FIND NEXT NAME
C     --------------
C
8      CALL MSSC(IVDU,'NEXT NAME ?\')
       CALL MSSC(IVDU,'ENTER 0 IF NEW SET OF NAMES REQUIRED\')
       DO 9 I=1,IPAIR
9      WRITE(IVDU,2000)I,(FILEA(J,I),J=1,12),FILEA(13,I)
       CALL MSSC(IVDU,' > \')
       READ(IVDU,1100)NXTPR
       IF((NXTPR.GT.IPAIR).OR.(NXTPR.LT.0))GOTO 8
       IF(NXTPR.EQ.0)GOTO 1
14     DO 15 I=1,12
15     BUF(I)=FILEA(I,NXTPR)
       SCALEA=FILEA(13,NXTPR)+15
       GOTO 20
20     CALL MKOPEN(IN,BUF,0)
```

```
C
C
C       READ IN FILE
C
C
        READ(IN,ERR=30,END=30)ICODEA
25      ENDFILE IN
        GOTO 50
30      CALL MSSG(IVDU,'READ ERROR\')
        ENDFILE IN
         GOTO 8
50      CALL MSSG(IVDU,'ATTENUATOR SETTING G? >\')
        READ(IVDU,1100,ERR=50)ATT
        IF ((ATT.LT.0).OR.(ATT.GT.99))GOTO 50
         GOTO 100
95      CALL SUBMCL(ATT)
         GOTO 300
100     DO 160 J=1,1408
                IVAL=ICODEA(J)
                IF(IVAL.EQ.IEND)GOTO 170
                CALL OUT(135,VAL)
160     CONTINUE
170      IF(SCALEA.EQ.15)GOTO 180
         CALL OUT(135,SCALEA)
180     CONTINUE
        GOTO 95
300     CALL MSSG(IVDU,'CONTINUE ? >\')
        READ(IVDU,1000)BUF
        IF (BUF(1).EQ.27)GOTO 8
        N=0
        GOTO 95
C
C
1000    FORMAT(20A1)
1100    FORMAT(I2)
1500     FORMAT(1X,'NAME',I2)
2000     FORMAT(1X,'NAME ',I2,'  G:',12A1,' SCALE-',I2)
C
C
        END
```

```
        SUBROUTINE SUBMCL(ATT)
C
C
C         M WALD AUG 1983
C
C       READS VALUE OF RESPONSE BUTTONS FOR MCL
C
C
        IMPLICIT INTEGER(A,B)
        GOTO 115
96      DO 100 I=1,100
        DO 100 J=1,100
100     BUT=INP(180)+256
101     IF((BUT.EQ.235).OR.(BUT.EQ.243).OR.(BUT.EQ.251)) GOTO 106
102     IF((BUT.EQ.246).OR.(BUT.EQ.238).OR.(BUT.EQ.254)) GOTO 107
104     IF((BUT.EQ.245).OR.(BUT.EQ.237).OR.(BUT.EQ.253)) GOTO 108
105     GOTO 100
106     CALL OUT(180,153)
        RETURN
107     ATT=ATT-1
        IF (ATT.LT.0)GOTO 110
        GOTO 115
108     ATT=ATT+1
        IF( ATT.GT.99)GOTO 112
        GOTO 115
110     ATT=0
        GOTO 100
112     ATT=99
        GOTO 100
115     A=ATT/10
120     B=6*A+ATT
130     CALL OUT(180,B)
135     WRITE(1,1000)ATT
140     GOTO 96
1000    FORMAT(I3,4HDb-G)
        END
```

```
      PROGRAM C&ULFN
C
C
C     M WALD     JUNE 1983
C
C
      BYTE SPACE
       BYTE FILE(13,80)
      BYTE BUF(12)
      DATA SPACE/32/, IVDU/1/, IN/6/, IEND/999/
C
C
1     CALL MSSC(IVDU,'ULL FILENAME FILE CREATE PROGRAM\')
      CALL MSSC(IVDU,'-------------------------------------------\')
C
C      SET No OF NAMES FLAG
C      --------------------
C
       IPAIR=40
       DO 5  I=1,40
C
C      ODD NO FILES=A EVEN NO=B
C      ------------------------
C
       IA=2*I-1
       IB=2*I
       CALL MSSC(IVDU,'ANOTHER NAME ? Y/N >\')
       READ(IVDU,1000)BUF
       IF(BUF(1).NE.89)COTO 6
       WRITE(IVDU,1500)I
C
C     ASK FOR FILENAMES
C     -----------------
C
      CALL MSSC(IVDU,' \')
      CALL MSSC(IVDU,'FILENAME A ? >\')
      READ(IVDU,1000)(FILE(J,IA),J=1,12)
       IF(FILE(1,IA).EQ.SPACE)STOP
       CALL MSSC(IVDU,'SCALEA ? >\')
       READ(IVDU,1100)FILE(13,IA)
5      CONTINUE
       COTO 8
6     IPAIR=I-1
       IF(IPAIR.EQ.0)COTO 1
8      DO 9 I=1,IPAIR
       IA=2*I-1
       IB=2*I
9 .    WRITE(IVDU,2000)I,(FILE(J,IA),J=1,12),FILE(13,IA)
       CALL MSSC(IVDU,'ARE ENTRIES CORRECT ? Y/N > \')
       READ(IVDU,1000)BUF
       IF(BUF(1).EQ.89)COTO 50
       CALL MSSC(IVDU,'WHICH NAME DO YOU WISH TO CORRECT ?>\')
       READ(IVDU,1100)ICORR
       IA=2*ICORR-1
       IB=2*ICORR
10    CALL MSSC(IVDU,' \')
      CALL MSSC(IVDU,'FILENAME A ? >\')
      READ(IVDU,1000)(FILE(J,IA),J=1,12)
       IF(FILE(1,IA).EQ.SPACE)COTO 10
       CALL MSSC(IVDU,'SCALEA ? >\')
       READ(IVDU,1100)FILE(13,IA)
       COTO 8
50     CALL MSSC(IVDU,'WHAT IS FILENAME OF FILE OF NAMES ? >\')
       READ(IVDU,1000)BUF
       CALL MKOPEN(IN,BUF,0)
       WRITE(IN,END=100)((FILE(J,I),J=1,13),I=1,IB)
100    ENDFILE IN
       COTO 1
1000  FORMAT(20A1)
1100  FORMAT(I2)
1500   FORMAT(1X,'NAME',I2)
2000   FORMAT(1X,'NAME ',I2,'  A:',12A1,' SCALEA-',I2,)
C
C
      END
```

```
      SUBROUTINE MKOPEN(LUN,FILE,DSK)
C
C     MIKE WALD OCT 1982
C     -----------------
C
C
C     THIS SUBROUTINE FORMATS FILENAME AND OPENS FILE
C     -----------------------------------------------
C
      INTEGER LUN
      BYTE DSK,FILE(12),ZFILE(11)
      DATA ZFILE/11*32/    ,IVDU/1/
C
C
      DO 10 I=1,11
10    ZFILE(I)=32
      IFLAG=0
      IEND=0
      DO 100 I=1,12
      IF((FILE(I).EQ.46).AND.(IFLAG.EQ.1))GOTO 500
      IF(FILE(I).EQ.46)IFLAG=I
      IF(IFLAG.EQ.1)GOTO 500
      IF(FILE(I).EQ.32)IEND=I
      IF((IFLAG.GT.9).OR.((IEND-IFLAG).GT.4))GOTO 500
100   IF(IEND.NE.0)GOTO 120
      IEND=13
120   IF(IFLAG.EQ.0)GOTO 500
      I=IFLAG-1
      DO 130 J=1,I
130   ZFILE(J)=FILE(J)
      K=IEND-1
      I=IFLAG+1
      DO 150 J=I,K
      L=J+9-I
150   ZFILE(L)=FILE(J)
      WRITE(IVDU,1000)LUN,ZFILE,DSK
      CALL OPEN(LUN,ZFILE,DSK)
      RETURN
500   CALL MSSG(IVDU,'ILLEGAL FILENAME\')
      RETURN
1000  FORMAT(1X,'LUN',I1,'FILENAME',11A1,'DSK',I1)
      END
```

```
      SUBROUTINE MSSG(LUN,TEXT)
C
C
C     M WALD     JUNE  1982
C
C
C     SENDS A MESSAGE TO SPECIFIED LUN
C
C
      BYTE TEXT(80)
      BYTE EOS
      DATA EOS/'\'/
C
C
      DO 10 N=1,80
              IF(TEXT(N).EQ.EOS)GOTO 20
10    CONTINUE
C
20    NC=N-1
      IF(NC.EQ.0)RETURN
      WRITE(LUN,1000)(TEXT(N),N=1,NC)
      RETURN
C
1000  FORMAT(1H ,80A1)
C
      END
```

APPENDIX H

FADAST TEST (Summerfield & Foster 1983)

The FADAST, a Four -alternative Auditory Disability and Speech reading Test, is a derivative of the rhyme test. On each of the 100 trials a single word is presented audio visually for identification. Words are selected from 25 4-item vocabularies constrained by combinations of two vowels and two consonants. The test has been recorded in monochrome video by an English male speaker and takes 20 minutes to administer.

The speech was presented at a level of 58 dB(A) from the front loudspeaker, while the noise was presented at 66 dB(A) from the three corner speakers.

The vocabulary and correct responses are shown in Figures H.1 and H.2, while the test instructions are given in Figure H.3.

Figure H.1

STRUCTURE OF VOCABULARIES

		CONSONANT	
VOWEL		pin	tin
		pen	ten

1	swap	slop	swoop	sloop
2	bull	wool	bell	well
3	seize	she's	Sue's	shoes
4	hat	cat	hot	cot
5	rum	run	ram	ran
6	sat	fat	seat	feet
7	but	gut	bet	get
8	heel	seal	hail	sail
9	pin	tin	pen	ten
10	sank	thank	sink	think
11	beet	meat	boat	moat
12	rung	rug	ring	rig
13	mill	pill	mail	pale
14	pat	bat	peat	beat
15	luck	lug	lock	log
16	do	two	doe	toe
17	neigh	Tay	knee	tea
18	dot	not	debt	net
19	suck	sung	sack	sang
20	Joy's	noise	Joe's	nose
21	right	light	rout	lout
22	slap	snap	slip	snip
23	tease	cheese	two's	choose
24	yacht	got	yet	get
25	sigh	tie	soy	toy

Figure H.2

FADAST RESPONSES

Circle respose made, except for practice. Underlining indicates correct response.

(Practice : 01 $\underline{2}$ 02 $\underline{3}$ 03 $\underline{2}$ 04 $\underline{4}$ 05 $\underline{3}$ 06 $\underline{2}$ 07 $\underline{2}$ 08 $\underline{2}$ 09 $\underline{3}$ 10 $\underline{4}$)

01-01	0:$\underline{1}$:2:3:4	02-01	0:1:2:$\underline{3}$:4	03-01	0:1:$\underline{2}$:3:4	04-01	0:1:$\underline{2}$:3:4
01-02	0:1:2:3:$\underline{4}$	02-02	0:1:2:3:$\underline{4}$	03-02	0:$\underline{1}$:2:3:4	04-02	0:1:2:$\underline{3}$:4
01-03	0:1:2:$\underline{3}$:4	02-03	0:1:$\underline{2}$:3:4	03-03	0:1:2:3:$\underline{4}$	04-03	0:1:2:$\underline{3}$:4
01-04	0:$\underline{1}$:2:3:4	02-04	0:1:$\underline{2}$:3:4	03-04	0:1:$\underline{2}$:3:4	04-04	0:1:$\underline{2}$:3:4
01-05	0:1:$\underline{2}$:3:4	02-05	0:1:2:$\underline{3}$:4	03-05	0:1:2:$\underline{3}$:4	04-05	0:1:$\underline{2}$:3:4
01-06	0:1:2:$\underline{3}$:4	02-06	0:1:$\underline{2}$:3:4	03-06	0:$\underline{1}$:2:3:4	04-06	0:1:2:$\underline{3}$:4
01-07	0:1:2:3:$\underline{4}$	02-07	0:1:2:3:$\underline{4}$	03-07	0:1:2:3:$\underline{4}$	04-07	0:1:2:3:$\underline{4}$
01-08	0:1:2:$\underline{3}$:4	02-08	0:$\underline{1}$:2:3:4	03-08	0:1:$\underline{2}$:3:4	04-08	0:1:2:3:$\underline{4}$
01-09	0:1:2:3:$\underline{4}$	02-09	0:$\underline{1}$:2:3:4	03-09	0:1:2:$\underline{3}$:4	04-09	0:1:2:$\underline{3}$:4
01-10	0:$\underline{1}$:2:3:4	02-10	0:$\underline{1}$:2:3:4	03-10	0:1:$\underline{2}$:3:4	04-10	0:$\underline{1}$:2:3:4
01-11	0:1:2:3:$\underline{4}$	02-11	0:1:2:3:$\underline{4}$	03-11	0:1:2:3:$\underline{4}$	04-11	0:1:2:3:$\underline{4}$
01-12	0:$\underline{1}$:2:3:4	02-12	0:$\underline{1}$:2:3:4	03-12	0:$\underline{1}$:2:3:4	04-12	0:1:2:$\underline{3}$:4
01-13	0:1:$\underline{2}$:3:4	02-13	0:1:$\underline{2}$:3:4	03-13	0:1:2:3:$\underline{4}$	04-13	0:1:2:$\underline{3}$:4
01-14	0:1:2:$\underline{3}$:4	02-14	0:1:2:3:$\underline{4}$	03-14	0:1:$\underline{2}$:3:4	04-14	0:1:2:3:$\underline{4}$
01-15	0:1:$\underline{2}$:3:4	02-15	0:1:2:3:$\underline{4}$	03-15	0:1:2:$\underline{3}$:4	04-15	0:$\underline{1}$:2:3:4
01-16	0:1:$\underline{2}$:3:4	02-16	0:1:2:$\underline{3}$:4	03-16	0:$\underline{1}$:2:3:4	04-16	0:1:2:$\underline{3}$:4
01-17	0:1:2:3:$\underline{4}$	02-17	0:$\underline{1}$:2:3:4	03-17	0:1:2:3:$\underline{4}$	04-17	0:1:$\underline{2}$:3:4
01-18	0:1:$\underline{2}$:3:4	02-18	0:1:2:$\underline{3}$:4	03-18	0:1:2:3:$\underline{4}$	04-18	0:1:2:3:$\underline{4}$
01-19	0:$\underline{1}$:2:3:4	02-19	0:1:2:3:$\underline{4}$	03-19	0:$\underline{1}$:2:3:4	04-19	0:$\underline{1}$:2:3:4
01-20	0:1:$\underline{2}$:3:4	02-20	0:1:$\underline{2}$:3:4	03-20	0:1:2:$\underline{3}$:4	04-20	0:1:$\underline{2}$:3:4
01-21	0:1:2:$\underline{3}$:4	02-21	0:1:2:$\underline{3}$:4	03-21	0:$\underline{1}$:2:3:4	04-21	0:$\underline{1}$:2:3:4
01-22	0:1:2:$\underline{3}$:4	02-22	0:1:$\underline{2}$:3:4	03-22	0:1:2:$\underline{3}$:4	04-22	0:$\underline{1}$:2:3:4
01-23	0:1:2:3:$\underline{4}$	02-23	0:1:2:$\underline{3}$:4	03-23	0:1:2:$\underline{3}$:4	04-23	0:1:$\underline{2}$:3:4
01-24	0:$\underline{1}$:2:3:4	02-24	0:1:$\underline{2}$:3:4	03-24	0:1:2:3:$\underline{4}$	04-24	0:1:2:3:$\underline{4}$
01-25	0:$\underline{1}$:2:3:4	02-25	0:$\underline{1}$:2:3:4	03-25	0:1:$\underline{2}$:3:4	04-25	0:$\underline{1}$:2:3:4
Score	/25		/25		/25		/25
Total							/100

Figure H.3

Instructions to subjects

This test has been designed to find out how well you can hear words when you can see the person who is talking as well as listening to him. If you normally wear spectacles or contact lenses for watching the televison, please wear them during the test. Similarly, if you normally wear a hearing aid, please wear it during the test with the volume set to a comfortable level; if necessary, adjust the volume during the practice trials with which the test starts.

I'm going to ask you to sit in front of the television and concentrate on the face of the man who will appear on the screen.
You will hear a noise in the background. Ignore the noise.
Four words will appear in a subtitle at the bottom of the screen. You should read them to yourself.
After a few seconds a number will change on the left-hand-side of the screen and a man's voice will call out the new number. Pay careful attention to the man on the screen. He will now speak one of the four words shown in the subtitle.
Your job is to decide which of the four words he has spoken. I want you to tell me the word and the number beneath it at the bottom of the screen so that I can record them on the score sheet.
A new set of four words will then appear, and we shall continue to repeat this procedure.

Don't worry if these instructions seem complicated or if at first you find the test a bit difficult. We shall start by doing 10 different sets of words just for practice. To help you, the four words in the first subtitle will be:

PIN	TIN	PEN	TEN
1	2	3	4

and the man on the screen will say the second word in the list, "TIN", so that you should call out "TIN - two".
On the second trial, the four words will be:

SUCK	SUNG	SACK	SANG
1	2	3	4

and the man of the screen will say the third word in the list, "SACK", so that you should call out "SACK - three".

On some trials you will find that it is easy to work out which word has been spoken. On others it will be more difficult. Don't worry. If you are unsure which word has been spoken, do not spend a long time thinking about it, just guess.

After the practice trials, we shall run through the test trials. The whole test will last for about 15 minutes.

If you have any questions please ask.

APPENDIX I

PRESENTATION ROOM ACOUSTICS

I.1 The reverberation times for the presentation room were:

Octave centre frequency Hz	RT s
31.5	0.94
63	0.89
125	0.65
250	0.54
500	0.38
1000	0.34
2000	0.35
4000	0.35
8000	0.36
Overall	0.58

I.2 The loudspeaker responses at the subject's head position are shown in Figure I.1, while the variation in response with distance is shown in Figure I.2.

Figure I.1

Simulated domestic living room Loudspeaker Response

response at subject's head position

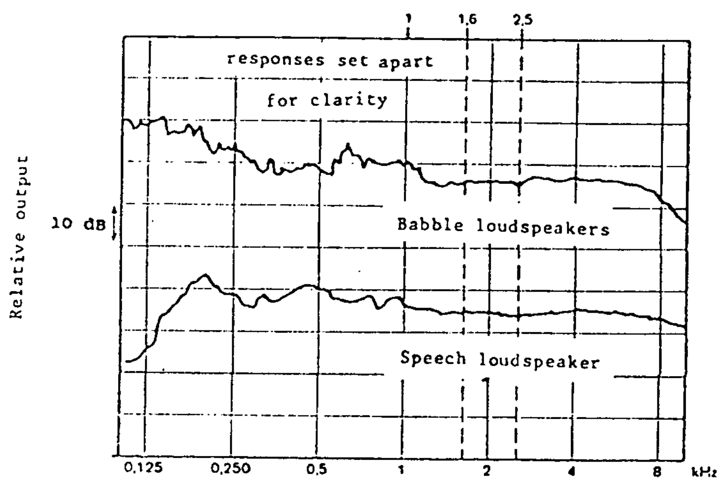
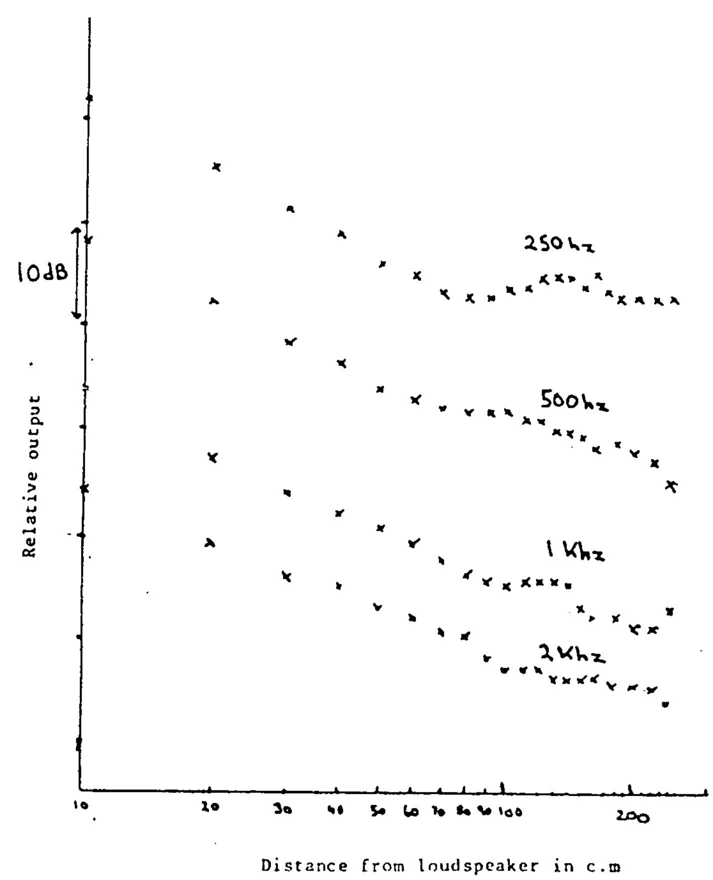


Figure I.2

Change in level with distance



APPENDIX J

HEARING AID PROCESSED SPEECH ANALYSIS

Two indications of the amount of clipping used for the hearing aid processed speech tape are shown in Table J.1. The differences '$L_1 - L_{99}$' indicate the dynamic range of the speech between the levels exceeded for 99% and for 1% of the time. It can be seen that this dynamic range is reduced as overload occurs.

The total harmonic distortion at 500 Hz gives an indication of the distortion of the speech signal. These measures were made using a 500 Hz pure tone of the same intensity as the speech used for the recordings. The distortion of the speech signal was greatest for frequencies around 500 Hz as speech has the greatest level and the aid has the lowest saturation level for this frequency band.

It should be noted that the figures shown in Table J.1 are only two of the possible ways of quantifying the distortion levels of the speech. Since speech is a complex time-varying signal they can only give an indication of the actual 'distortions' involved.

However, the actual speech levels and gain settings used for the recording of the hearing aid processed speech tape were realistic and therefore the results of the subjective pair comparison judgements are indicative of how noticeable these distortions are in real life.

Table J.1 Clipping and distortion measures

Pair	$L_{99} - L_1$ dB		Total harmonic distortion (500 Hz) %	
	Green	Red	Green	Red
4	39	39	3	3
5	39	39	4	3
6	38	39	5	3
7	38	39	7	3
8	35	39	10	3
9	31	39	22	3
10	27	39	25	3
11	27	39	28	3
12	24	39	31	3
13	21	39	50	3

APPENDIX K

TEST INSTRUCTIONS

The following instructions were given to subjects for the tests used in this study. The instructions were given in both written and verbal form. In all cases it was established that the subjects understood the requirements of the tests before proceeding further.

K.1 Pair Comparison Instructions

Adjust the level of the sound so that it is at the most comfortable level for you to listen to.

Do this for each of the channels: green and red.

When both the green and red channels are at your most comfortable listening levels, I want you to compare the channels by switching beween them. Do this for as long as you feel is necessary in order to answer the following questions.

(1) Do you think there is a difference between the green and red channels?

Yes/No

If you think there is a difference answer the following questions.

(2) How sure are you that there really is a difference?

4	3	2	1	0
completely sure	very	moderately	slightly	not at all sure

(3) Which channel did you prefer? Red/green/no preference.

If you had a preference, answer the following questions.

(4) How much better did you find your preferred channel?

3	2	1
much better	moderately better	slightly better

(5) Why did you prefer this channel?

(i) easier to understand the speech

(ii) preferred the quality of the sound

(iii) other reason

K.2 Threshold in Noise Instructions

I am interested in finding out the faintest tones you can hear in a background of noise.

Try to ignore the noise which you will hear all the time.

As soon as you hear the tone, press the button.

Keep the button pressed for as long as you hear the tone, only release the button when you can no longer hear the tone.

No matter how faint the tone, press the button when you hear it and only release the button when you no longer hear it.

K.3 Threshold Instructions

I am interested in finding out the faintest sounds you can hear.

As soon as you hear the sound, press the button.

Keep the button pressed for as long as you hear the sound. Only release the button when you can no longer hear the sound.

No matter how faint the sound, press the button when you hear it and only release the button when you no longer hear it.

K.4 QCL Instructions (Pure Tone)

I am interested in finding the quietest sounds you can comfortably listen to.

Press the button when the sound is quieter than you could comfortably listen to for a long time. Release the button when the sound is comfortable to listen to.

K.5 LCL Instructions (Pure Tone)

I am interested in finding the loudest sounds you can comfortably listen to.

Press the button when the sound is louder than you could comfortably listen to for a long time. Release the button when the sound is comfortable to listen to.

K.6 MCL Instructions

Adjsut the level of the sound so that it is at the most comfortable level for you to listen to for a long time.

K.7 QCL Instructions

I am interested in finding the quietest sounds you can comfortably listen to.

Adjust the level of the sound until it is the quietest you could comfortably listen to for a long time.

K.8 <u>LCL Instructions</u>

I am interested in finding out the loudest sounds you can comfortably listen to.

Adjust the level of the sound until it is the loudest you could comfortably listen to for a long time.

K.9 <u>ULL Instructions</u>

I would like to find out at what levels you find sounds uncomfortably loud.

Press the button when the sound becomes uncomfortably loud and release the button when the sound is no longer uncomfortably loud.

When you press the button the sounds will not get any louder.

K.10 <u>AVELS Instructions</u>

You are going to hear some sentences spoken and will sometimes see the speaker on the TV screen. Sometimes you will also hear noises in the background.

I would like you to repeat aloud the last word in each sentence immediately after the sentence has been spoken.

Even if you are unsure, repeat what you think the word was.

APPENDIX L

CRITICAL RATIO DETERMINATION

The critical ratio (CR) is defined as the ratio between the SPL of the masked threshold of a pure tone and the power spectrum density of the masking noise. The critical bandwidth can be estimated from the measure of critical ratio and gives an indication of the resolving power of the ear's 'auditory filters'. More direct measurements of the auditory filter shape are possible, but require far more difficult and time-consuming tests. The larger the critical ratio, the wider the 'auditory filter' and the poorer the resolving power of the ear.

If a white noise of power spectrum density N dB/Hz just masks a pure tone of level T dB, then the critical ratio in dB is given by $T - N$ and the critical bandwidth estimate will be $10^{T-N/10}$ (i.e., a critical ratio of 20 dB indicates a critical bandwidth of 100 Hz).

Figure L.1 shows the power spectrum density in dB/20 Hz for the speech noise measured in the B & K 4153 artificial ear for a 90 dB dial setting. The power spectrum was similar for the left and right headphone receivers. The power spectrum/Hz at 2 kHz is 45 dB/Hz. The spectrum of the speech noise approximates a white noise spectrum fairly well over the frequencies of interest and so no correction was made. Since relative measures between subjects are the main interest of this study, this approximation is satisfactory.

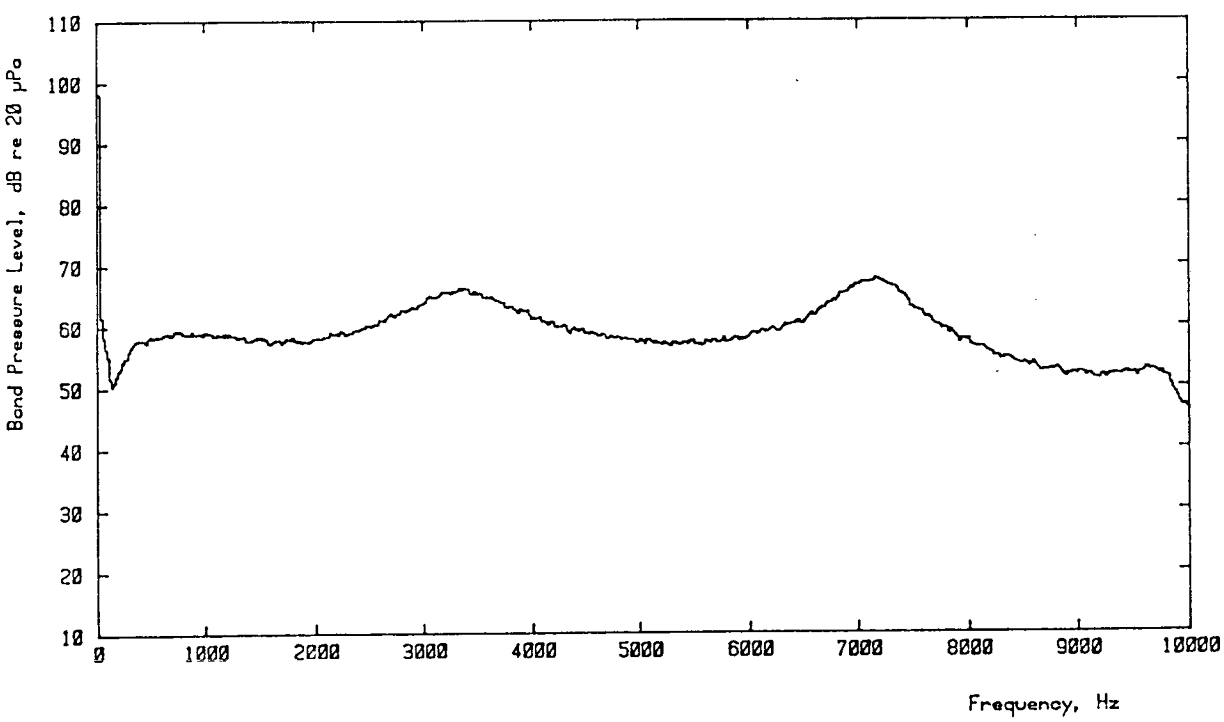
The pure tone dial reading of 90 dB HTL corresponded to an SPL of 100 dB.

The level of masking noise used for each subject was the loudest level they could comfortably listen to (LCL), and therefore varied from subject to subject. This level was chosen so as to measure critical ratios at similar loudness levels for subjects and therefore estimate the resolving power of their auditory filters at realistic listening levels. Critical ratios were measured at 2 kHz as this is an important speech frequency and within the range of residual hearing of the majority of the hearing-impaired population.

Figure L.1

Speech noise level /20 hz

Right ear   90 dB dial level



APPENDIX M

FREE FIELD PURE TONE TEST SYSTEM

The pulsed pure tones were produced by the Amplaid 207 clinical audiometer. All free field audiometry was performed in the anechoic room using an H/H power amplifier and an 8" diameter single driver loudspeaker for amplification and delivery of the test signals.

Subjects sat 1 metre from the loudspeaker with their heads centred on axis. The sound pressure level varied by less than ±1 dB within a sphere of 6" about this central position.

The noise level of the equipment was sufficiently low to allow free field binaural hearing threshold measurements better than -10 dB HTL to be made.

APPENDIX N

QUESTIONNAIRES AND SUBJECT INFORMATION (SELF ASSESSMENT)

Subject no.	1	2	3	4	5	6	7	8	9	10	11	12
Age	64	54	50	39	61	42	62	52	52	36	60	69
Sex (M/F)	M	M	F	F	M	F	M	M	F	F	F	F
Married (Y/N)	Y	Y	Y	Y	Y	Y	Y	Y	Y	Y	Y	N
Living alone (Y/N)	N	N	N	N	N	N	N	N	N	N	N	Y
Out to work (Y/N)	Y	Y	Y	N	Y	Y	Y	Y	N	N	Y	N
Years aware of hearing difficulty	25	38	39	20	4	16	20	45	35	6	22	60
No. of years aided	10	15	2	14	4	9	20	29	35	3	6	6
Good lipreader (Y/N)	N	N	Y	N	N	N	Y	Y	Y	N	N	?
Lipreading helps (Y/N)	Y	N	Y	Y	Y	Y	Y	Y	Y	Y	Y	Y
Come to terms with hearing loss (Y/N)	Y	N	Y	Y	Y	Y	Y	Y	Y	Y	Y	Y
Cope with hearing loss (0-3) (not al all - very well)	3	0	3	3	3	2	3	3	3	2	3	3
Emotional response (hearing measurement scale section IV)	9	25	15	16	15	19	10	23	18	24	23	26
No. of h/day aid worn (A = All day)	A	A	A	A	A	A	A	A	A	4	A	A
Sound quality of own aid (0-8) (bad - good)	7	4	7	7	7	4	7	8	7	5	9	8
Understand speech aided (0-9) (not at all - perfectly)	5	2	6	4	6	4	6	7	5	7	5	7
Understand speech unaided (0-9)	0	0	1	0	2	1	1	5	0	5	1	0
Satisfaction with aid (0-9)	9	3	6	8	8	5	6	9	9	6	8	8

Figure N.1

INITIAL QUESTIONNAIRE - HEARING AID USE

Subject No.

Name:

Aid

Aided Ear

Left	Right	Either	Both

1 (a) On average for how long each day do you feel you can usefully wear your aid?

(b) What was the main reason for not wearing it longer?

(show card A)

(b) Reason

1. Never
2. Only occasionally
3. 1 - 2 hours a day
4. 2 - 4 hours a day
5. 4 - 8 hours a day
6. More than 8 hours a day

(a) ☐ (b) ☐

2.(a) How long did you take to get used to your aid?

(b) What was your initial reaction to your aid?

(show card B)

(1) Immediately
(2) A few hours
(3) A few days
(4) A few weeks
(5) Still not used to it

(a) ☐ Initial Reaction (b) ☐

V 3 (Ask only after initial videotape session)

The videotape contained a range of listening tasks some more difficult than others. Taking everything into account, how did you feel the listening tasks compared with those you experience in your daily life?

(a) More difficult/less difficult/of the same difficulty?

(If more or less difficult ask (b) and (c))

(b) How much more/less difficult did you find it?

Very much (3)/moderately (2)/slightly (1)

(c) Why do you think you found it .... ?

(a) ☐

(b) ☐

(c) ☐

V 4 (a) Did you feel that the environmental sounds sounded realistic ?

(If no ask (B))

(b) Why ?

(a) ☐

(b) ☐

6.

I am interested in how useful you find your aid in various situations.

I will now describe some situations.

Ask (a) through to (f) for each situation.

(a) Has this situation occurred? (read out situation) Yes (1)/ No (0)

Situation		(a) Occur Y/N	(b) Avoid	(c) Important	(d) Difficulty	(e) Use Y/N	(g) MAIN REASON
(1) Talking with a person who has a particularly quiet voice	1						
(2) Talking with 1 person in a quiet place when you cannot see their face	2						
(3) Talking with 1 person in a quiet place (eg. at home)	3						
(4) Talking with 1 person in a noisy place (eg. on a bus, in a shop)	4						
(5) Talking with 2 or more persons in a quiet place	5						
(6) Talking with 2 or more persons in a noisy place (eg. at a party)	6						
(7) At a cinema or theatre	7						
(8) Listening to music	8						
(9) Listening to environmental sounds (eg. birds, rain, cars, footsteps, bells)	9						
(10) Listening to telephone	10						
(11) Listening to T.V. (on microphone [ ] , on loop [ ])	11						
(12) Listening to speech in a hall (eg. church or public meeting)	12						
(13) Having to decide from which direction the sound is coming	13						
Any Other							

(b) If NO ask if the situation has been avoided due to hearing problems

YES (1)/NO (0)

(c) How important is it for you to hear in this situation?

Very (3) Moderately (2) Slightly (1)

(d) How much difficulty do you have in this situation?

Great (3) Moderate (2) Slight (1) None (0)

(e) Have you found you can use the aid in this situation?

Yes (1)/ No (0)

If NO ask (g)

(f) How helpful do you find the aid in this situation?

Very (3) Moderately (2) Slightly (1) Not (0)

(g) What was the main reason for you not using the aid?

5. I would now like to talk about problems you have had with your aid

(Ask (a) thru (e)) for each problem.

(a) Have you had any problems with .... ?

		Problem
		Own Aid
(1) Not enough volume	1	
(2) Adjusting and handling the aid (ergonomic)	2	
(3) Discomfort from the earmould	3	
(4) Inserting the earmould	4	
(5) Condensation affecting the sound	5	
(6) Having to clean the earmoulds	6	
(7) Wind noise	7	
(8) Background noise	8	
(9) Whistling (feedback)	9	
(10) Needing to often adjust the volume control setting	10	
(11) Discomfort from sounds being too loud	11	
(12) The appearance of the aid or earmould	12	
Any other		

(If yes to question (a) ask (b))

(b) Do you regard it as a serious (3), moderate (2), or slight (1) problem?

7. In what situations do you find the aid

(a) most useful ☐

(b) least useful ☐

8(a) Could you please rate the quality of the sound for your aid on each of the following scales.

(show card C)

											Own aid
Clear	4	3	2	1	0	1	2	3	4	Hazy	
Harsh	4	3	2	1	0	1	2	3	4	Mellow	
Pleasant	4	3	2	1	0	1	2	3	4	Unpleasant	
Rough	4	3	2	1	0	1	2	3	4	Smooth	
Good	4	3	2	1	0	1	2	3	4	Bad	

For instance on the first scale if you feel the sound is extremely clear rate it 'clear'

If you feel it is extremely Hazy, rate it 'Hazy 4'

If you have no feelings about whether it is hazy or clear rate it '0'

(b) How would you describe the sound quality of your aid? ☐

9. Could you think about how well you understand with and without hearing aid?

9. Understand perfectly

8.

7.

6. without aid ☐

5.

4. with aid ☐

3.

2.

1.

0 Not understand at all

---

10 Considering everything, how satisfied are you with the aid?

9. Extremely satisfied

8.

7.

6. Own aid ☐

5.

4.

3.

2.

1.

0 Not at all satisfied

---

11 What improvements would increase your satisfaction?

Anything else?

NAME ....

COULD YOU KINDLY ANSWER THE FOLLOWING QUESTIONS BY PUTTING A RING AROUND THE APPROPRIATE NUMBER

HOW OFTEN DO THESE SITUATIONS OCCUR ?

(Explanation of rating scale)

(Frequently--5 4 3 2 1 0--Never)
(If a situation occurs frequently ring No. 5)
(If it never occurs ring No. 0)
(If it does occur but not frequently,ring an appropriate No.between 1 and 4)

Situations where somebody is talking to you and ......

...it is quiet and you can see the talker's face	5 4 3 2 1 0
...it is noisy and you can see the talker's face	5 4 3 2 1 0
...it is quiet and you can NOT see the talker's face	5 4 3 2 1 0
...it is noisy and you can NOT see the talker's face	5 4 3 2 1 0

HOW OFTEN DO YOU AVOID THESE SITUATIONS BECAUSE YOU FIND THEM DIFFICULT ?

(Explanation of rating scale)

(Frequently--5 4 3 2 1 0--Never)
(If a situation is avoided frequently ring No. 5)
(If it is never avoided ring No. 0)
(If it is avoided but not frequently,ring an appropriate No.between 1 and 4)

Situations where somebody is talking to you and......

...it is quiet and you can see the talker's face	5 4 3 2 1 0
...it is noisy and you can see the talker's face	5 4 3 2 1 0
...it is quiet and you can NOT see the talker's face	5 4 3 2 1 0
...it is noisy and you can NOT see the talker's face	5 4 3 2 1 0

HOW WELL DO YOU UNDERSTAND SOMEBODY TALKING.. ?

(Explanation of rating scale)

(Perfectly-9 8 7 6 5 4 3 2 1 0-Not at all)
(If you understand perfectly in this situation ring No. 9)
(If you do not understand at all in this situation ring No. 0)
(If you do understand but not perfectly,ring an appropriate No.between 1 and 8)

In situations where ......

...it is quiet and you can see the talker's face	9 8 7 6 5 4 3 2 1 0
...it is noisy and you can see the talker's face	9 8 7 6 5 4 3 2 1 0
...it is quiet and you can NOT see the talker's face	9 8 7 6 5 4 3 2 1 0
...it is noisy and you can NOT see the talker's face	9 8 7 6 5 4 3 2 1 0

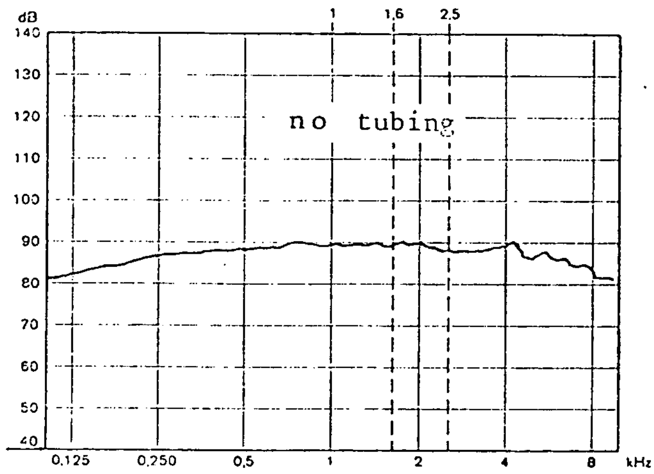
APPENDIX O

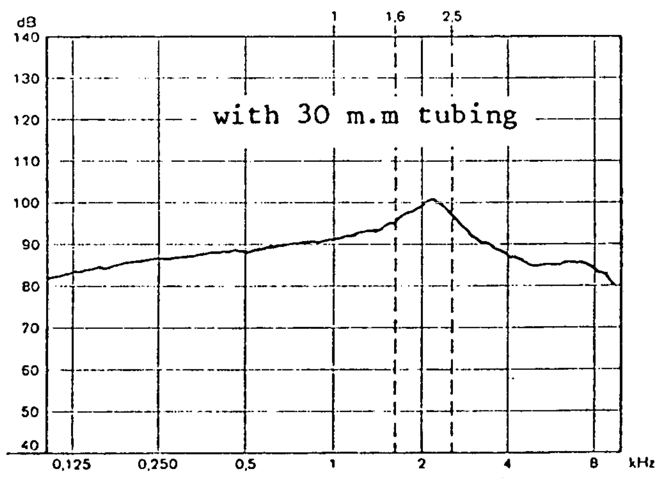
PROBE TUBE MICROPHONE CALIBRATION

The calibration curves shown in Figure O.1 were obtained in an anechoic room using the B & K audio test station 2116. The audio test station output a 90 dB swept sine wave at the probe tube microphone position. It can be seen that the probe tube microphone is of similar sensitivity to the B & K half inch microphone that was used to calibrate the test station.

Figure 0.1

probe microphone calibration Response





APPENDIX P

HEARING AID SELECTION PROCEDURES

P.1 Byrne Selection Table

Unaided threshold HTL	Required aided threshold, dB SPL 250	500	1 k	1.5 k	2 k	3 k	4 k Hz
0	31	24	3	2	1	2	2
5	34	27	6	5	4	5	4
10	36	29	8	7	6	7	6
15	39	32	11	10	9	10	9
20	42	35	14	13	12	13	12
25	44	37	16	15	14	15	14
30	47	40	19	18	17	18	17
35	50	43	22	21	20	21	20
40	53	46	25	24	23	24	23
45	55	48	27	26	25	26	25
50	58	51	30	29	28	29	28
55	61	54	33	33	31	32	31
60	63	56	35	34	33	34	33
65	66	59	38	37	36	37	36
70	69	62	41	40	39	40	39
75	71	64	43	42	41	42	41
80	74	67	46	45	44	45	44
85	77	70	49	48	47	48	47
90	80	73	52	51	50	51	50
95	82	75	54	53	52	53	52
100	85	78	57	56	55	56	56
105	88	81	60	59	58	59	58

P.2 Berger Selection Table

Unaided threshold HTL	Required aided threshold, HTL 500	1 k	2 k	3 k	4 k	6 k Hz
20	20	7	7	8	9	10
25	20	9	8	10	12	12
30	21	11	10	12	14	15
35	22	13	12	14	17	17
40	23	15	13	16	19	20
45	24	17	15	19	21	22
50	25	19	17	21	24	25
55	28	21	18	23	26	27
60	30	22	20	25	28	30
65	33	24	22	27	31	32
70	35	26	23	29	33	35
75	38	28	25	31	36	
80	40	30	27	33	38	
85	43	32	28	35	40	
90	45	34	30	37		
95	48	36	32	39	Not possible	
100	50	38	33	41	to aid	
105	53	39	35			
110	55	41	37			

Berger suggests reducing the required aided threshold for conductive/ mixed losses by one-fifth of the air bone gap. Thresholds below the solid line are claimed not to be aidable. The required aided thresholds in dB HTL are converted to dB SPL by adding the monaural minimum audible field, which is estimated from the binaural minimum audible field by the addition of 3 dB.

APPENDIX Q

FIELD TRIALS OF SIMPLIFIED TEST PROTOCOL

Q.1 Introduction

The pair comparison procedure and AVELS test require extensive field trials to verify the relationship between results on these tests and performance in real life situations. In addition, the sensitivities to variations in response found for the hearing-impaired subjects in the main research study (13.7) requires validating on a larger number of patients.

With these aims in mind, an extensive field and laboratory study was designed and is at present being completed. The main design features of this study will be described in this chapter. The results will be available after publication of this study and will be reported elsewhere.

Q.2 Subjects

50 hearing-impaired users of monaural NHS BE30 and BE50 series higher power post-aural aids volunteered to take part in the study. They were all experienced hearing aid users and had obtained their aids from the same hospital hearing aid clinic.

Q.3 Experimental Design

Two hearing aids were given to each subject to compare using the pair comparison procedure, the AVELS test and two weeks of home trial. The aim of this design was to establish how the results of these three different methods of aid evaluation compared and to see how well the laboratory tests predicted real-life performance. The aids used were samples of the BE30 and 50 series aids so that results could be directly applied to NHS higher power aids. Each subject was asked to compare two samples of the same aid type, one set to the lowest tone setting, the other to the highest. The aid responses used are shown in Figure Q.1. The subject's existing earmould was used if it provided a satisfactory fit and seal. Since each subject's earmould was used for all their comparisons, the relative change in response would remain similar whatever the actual earmould acoustics that existed.

The pair comparison test was carried out using digital filter Master Hearing Aid simulations of the actual aid responses.

Wherever possible the subject was allotted the same type of aid as he or she already possessed. This was so that the experiment could test the often stated but as yet untested hypothesis that subjects prefer the aid response that they are used to (3.2). The subjects were, however, informed that these were all experimental aids which just used the same cases as the standard NHS aids. This was done in order to remove any bias in their preferences.

The AVELS test was administered using the shortened test forms.

The two hearing aids were then given to the subject to try out at home for a two week trial. The performance and relative preferences were then established by interview using the questionnaire shown in Appendix R. The details and philosophy of the field trial and questionnaire design will not be given here as they follow the procedure developed by Wald in the 1981 Wald and Rice study which is reproduced in full in Appendix R.

In addition to comparing the two simulated aid responses, the pair comparison test was used to establish the effect of changes in the hearing aid response. These included changes in low frequency cut off, high frequency cut off, low frequency slope, high frequency slope and interactions of these parameters. The effect of distortion and coupler/manikin differences was also tested. The actual pairs of responses used are shown in Table Q.1 and Figure Q.1.

Q.4 Experimental Protocol

Following standard clinical audiometric evaluation, the questionnaire shown in Appendix N was administered by interview. This obtained details about the subject, including the history of their hearing problem and remediation and the present status of their communication ability.

The subjects' hearing aids and earmoulds were checked for satisfactory functioning and the subjects each allotted their two experimental aids.

The shortened forms of the AVELS test were administered followed by the environmental sounds. Each subject was then required to give his preferences using the questionnaire shown in Appendix R.

Subjects were then required to undertake the pair comparison test using the master hearing aid and the responses shown in Table Q.1. The test protocol and instructions were similar to those used in the main study (Chapter 12)

The two hearing aids were then taken home by the subject for a few weeks trial which was followed by an evaluation through interview and questionnaire.

Table Q.1 Pairs of digital filter responses for comparison.

Pair	Green	Red
1	LP 8 kHz	LP 8 kHz
2	BE31H	BE31N
3	BE32N	BE32H
4	BE33H	BE33N
5	BE51L	BE51H
6	BE52L	BE52H
7	LP 8 kHz	LP 3 kHz
8	LP 3 kHz	LP 4 kHz
9	LP 8 kHz	LP 4 kHz
10	HP 250 Hz	LP 8 kHz
11	HP 250 Hz	HP 400 Hz
12	HP 400 Hz	LP 8 kHz
13	+6 dB/octave	−12 dB/octave
14	−6 dB/octave	+6 dB/octave
15	LP 8 kHz	−12 dB/octave
16	LP 8 kHz	+6 dB/octave
17	−6 dB/octave	LP 8 kHz
18	−12 dB/octave	−6 dB/octave
19	−12 dB/octave above 1 kHz	+6 dB/octave above 1 kHz
20	−6 dB/octave above 1 kHz	+6 dB/octave above 1 kHz
21	LP 8 kHz	−12 dB/oct. above 1 kHz
22	LP 8 kHz	+6 dB/octave above 1 kHz
23	−6 dB/octave above 1 kHz	LP 8 kHz
24	−6 dB/octave above 1 kHz	−12 dB/oct. above 1 kHz
25	+6 dB/octave below 1 kHz	−12 dB/oct. below 1 kHz
26	−6 dB/octave below 1 kHz	+6 dB/octave below 1 kHz
27	−12 dB/octave below 1 kHz	LP 8 kHz
28	LP 8 kHz	+6 dB/octave below 1 kHz
29	−6 dB/octave below 1 kHz	LP 8 kHz
30	−6 dB/octave beloiw 1 kHz	−12 dB/oct. below 1 kHz
31	15 dB peaks	LP 8 kHz
32	10 dB peaks	LP 8 kHz
33	−12 dB/octave	−12 dB/oct. HP 250 Hz
34	+6 dB/octave LP 3 kHz	+6 dB/octave
35	Inverse of Kemar open ear response	LP 8 kHz
36	Kemar - 2cc	LP 8 kHz

FIGURE Q.1

PAIR COMPARISON FILTER RESPONSES

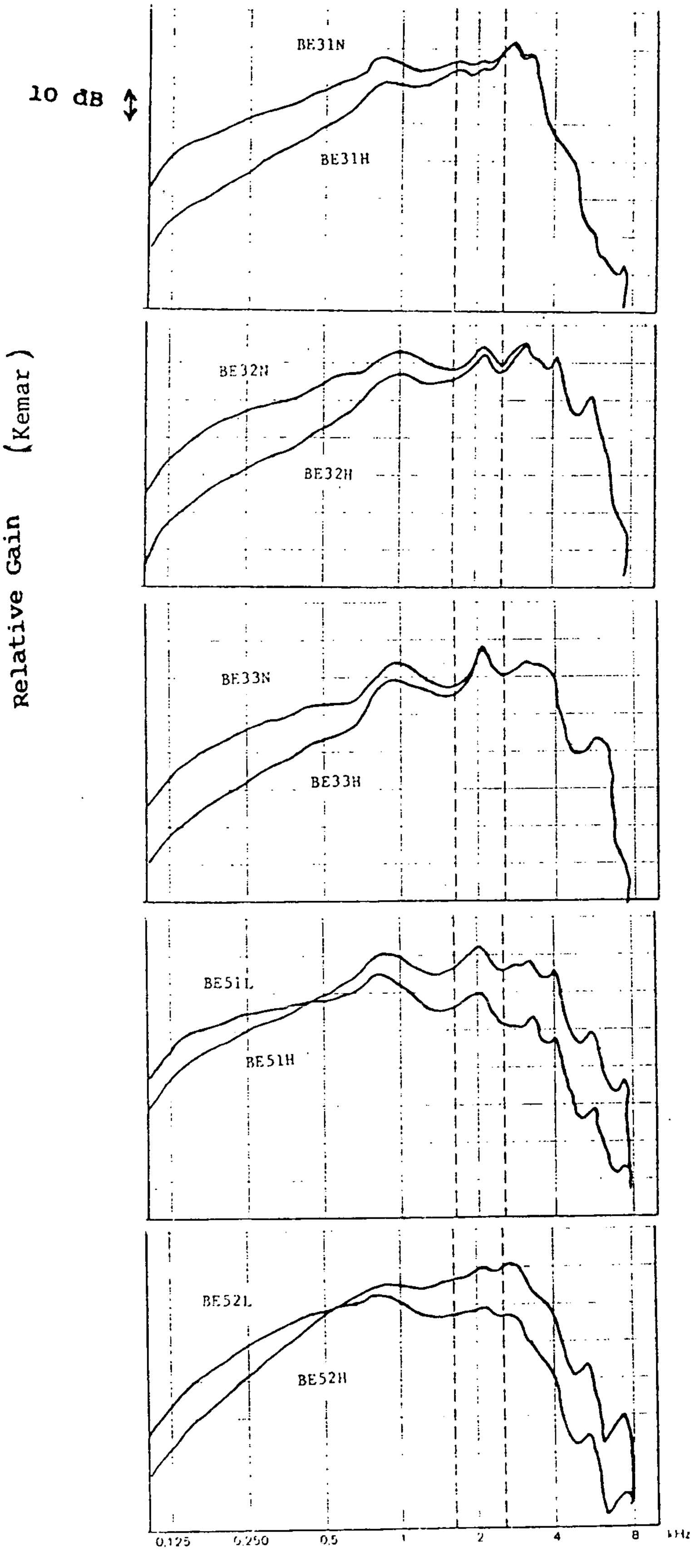


FIGURE Q.1 cont.

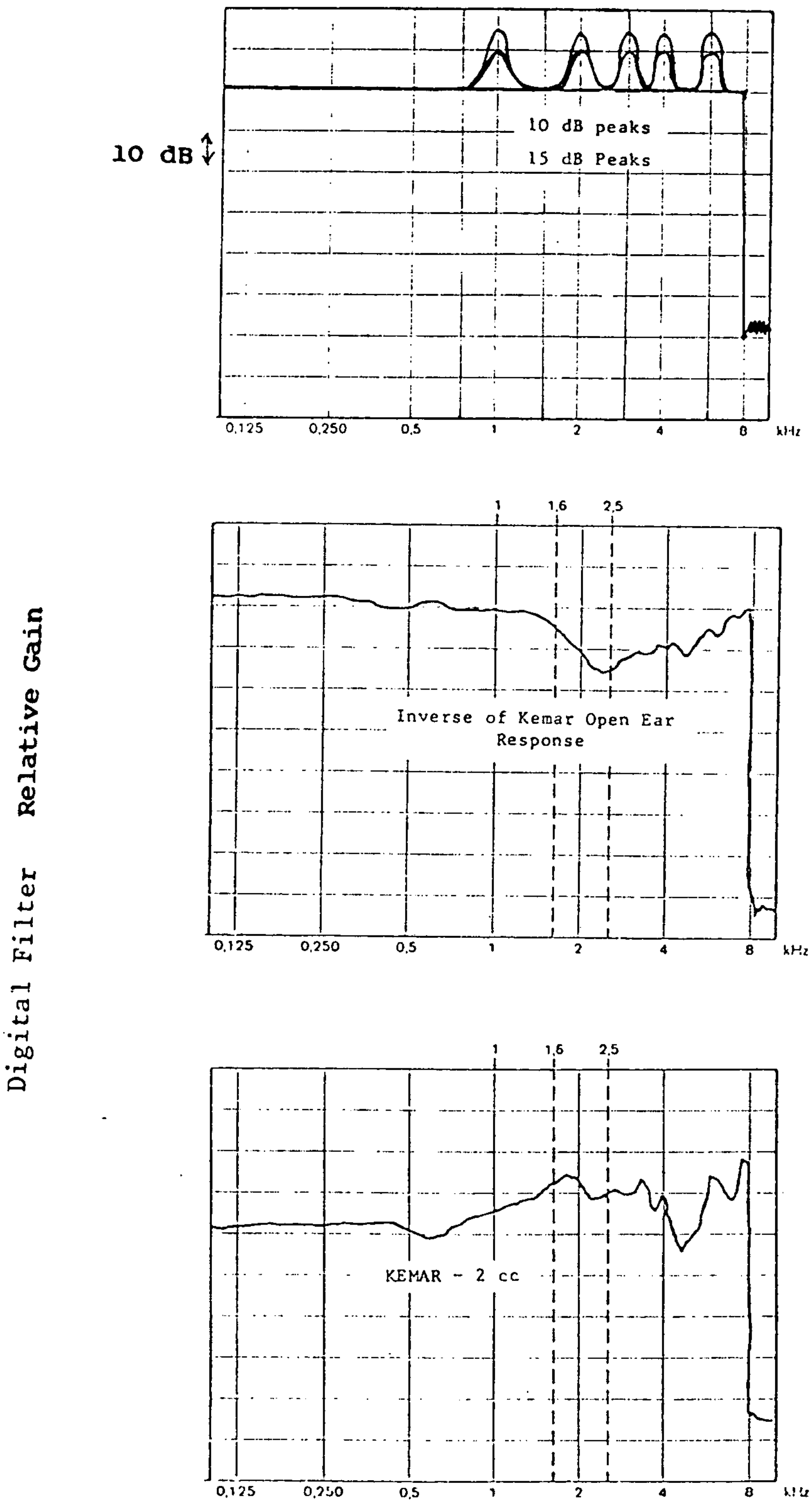


FIGURE Q.1 cont.

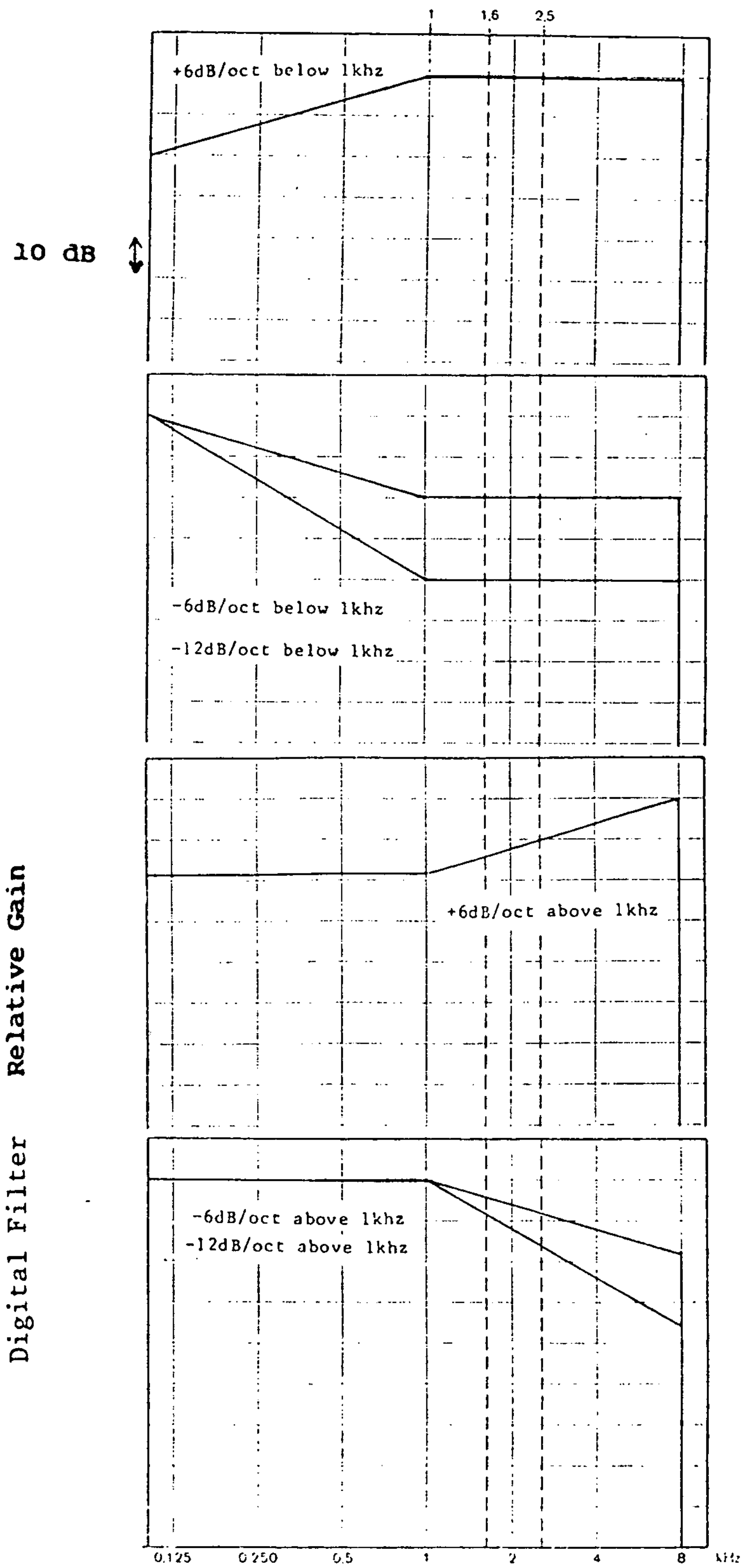
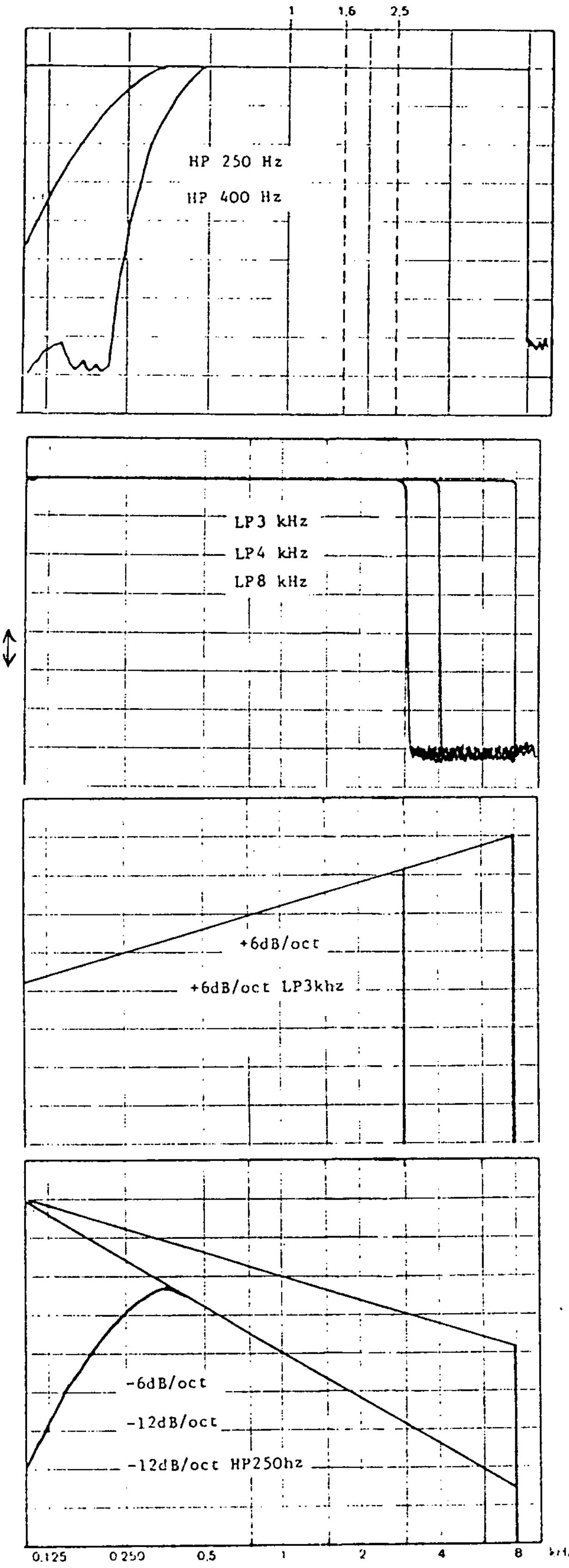


FIGURE Q.1 cont.

Digital Filter Relative Gain

10 dB



Appendix R

Reprint of M Wald and CG Rice contract report 81/7

A STUDY OF BENEFITS FROM
ELECTRO-ACOUSTICAL MODIFICATIONS
TO THE NHS BE HEARING AIDS

DHSS Agreement No. 26/79

M. Wald and C.G. Rice

Contract Report 81/7

CONTENTS

CONTENTS contd..

ABSTRACT

A previous laboratory study at I.S.V.R. found that smoothing and extending the frequency response of the N.H.S. BE-11 hearing aid improved both speech discrimination and perceived sound quality. The field study described here was conducted to determine whether these improvements would be substantiated in every day use.

The results suggested that in order to achieve the subjective benefits of these modifications special account needed to be taken of the audiometric configuration of the subjects. Firm recommendations regarding the implementation of the modifications cannot at present be made, but their further development as regards manufacture, acoustic performance and fitting would appear to be worthwhile.

1.0 INTRODUCTION

A recent laboratory study by Lawton and Cafarelli (reference 1) tested the hypothesis (based on the results of published research (references 2, 3, 4, 5, 6, 7, 8) ,that smoothing the frequency response and extending the bandwidth of the N.H.S. BE-11 aid would improve both speech discrimination and sound quality for a broadly defined group of hearing impaired adults. The study concluded that in the laboratory situation the hypothesis was accepted.

This present field study was designed to investigate whether the laboratory findings of Lawton and Cafarelli were substantiated when modified hearing aids and moulds were worn by a similar group of patients in their usual everyday situations. The results should also therefore, give an indication of the ability of the special laboratory tests to predict 'real life' performance. Many published research findings appear not to have involved both rigorous 'laboratory' and 'field' testing. Either only a laboratory study has been conducted or the field tests have been conducted in an informal way with only anecdotal reporting of results, and so the true value of their results cannot be properly assessed.

2.0 EXPERIMENTAL DESIGN

2.1 General purpose of experiment

The general purpose of the experiment was to test for subjective and objective differences between hearing aid systems of slightly differing response characteristics. This was to be achieved by allowing hearing impaired subjects to wear various behind the ear aid and earmould systems in their everyday situations and give judgements on system sound quality, speech intelligibility and their preferences by means of interview and questionnaire.

As with the laboratory study the differing frequency responses of the experimental hearing aid sytems were achieved by modifying the National Health Service BE-11 aid and its associated earmould and sound tube system. The high frequency response of the basic BE-11 was improved by replacing the transducers by wide band units. The sound tube was modified by changing from a single diameter to a multiple diameter, damped tube which smoothed the frequency response.

2.2 Hearing aid and earmould modifications

The four hearing aid/earmould combinations were similar to those used in the laboratory study, ie :-

Standard BE-11 + Standard Earmould	Abbreviated BE
Modified transducer BE-11 + Standard Earmould	Abbreviated BETr
Standard BE-11 + Modified Earmould	Abbreviated BETb
Modified transducer BE-11 + Modified Earmould	Abbreviated BETrTb

The frequency response of each aid system is given in Figure 1, and the 1 kHz gain, HAIC gain and HAIC frequency range given in Table 1. These responses are similar to those used in the laboratory study, the main difference being that the modified transducer aids in the field study used a subminiature microphone rather than the impractically mounted larger one used in the laboratory study. The resonance damping due to the modified mould filters and the extended bandwidth due to the transducers can be clearly seen in Figure 1. The transducers in each of the trial aids are listed in Table 2, while examples of the standard and modified aids are shown in Figure 2.

The standard ear moulds were made with 1.9 mm internal diameter tubing and the modified earmoulds (specified 6R10 in Reference 9) were made as a step-wise approximation to an acoustic horn with internal dimensions as shown in Figure 3. They had similar sound tubes to those used in the laboratory experiment except that the modified mould had its 4 mm section formed by the mould internal bore rather than by tubing. (Figure 4).

2.3 Pilot Study

Before proceeding with the main study it was necessary to develop the techniques and procedures required to produce acceptable wearable modified ear moulds. Twenty one of the twenty eight people who took part in the original laboratory study readily agreed to try out these prototype skeleton earmoulds supplied commercial under sub-contract.

Certain problems were revealed during this pilot trial.

2.3.1 Fabrication problems

(i) Each fitting presented a unique mould-tubing-anatomical fit problem and so on average took three times as long to fabricate as a conventional ear mould.

(ii) The deeper meatal penetration, and larger bore occasionally necessitated building up the impression and consequently required individual adjustment of the earmould when fitting.

(iii) A few subjects required 'pressure release' venting on account of the tightness of fit of the mould. In these cases a lateral vent parallel to the 4 mm bore was drilled. This required considerable care not to damage the mould.

(iv) The construction of the acoustical tubing required two glued joints, which could cause mechanical breakdown to occur either due to the reduced flexibility of the tubing or to inadequate adhesion.

(v) Each mould required individual testing to ensure it conformed to specification.

2.3.2 User problems

(i) Condensation became trapped between the damping elements, especially when the moulds were worn all day.

(ii) Initial comfort and insertion problems sometimes occurred due to tightness of fit of the moulds.

(iii) The cosmetic effect of the tubing 'sticking out' was mentioned by a few people.

2.3.3 Implications

It was accepted that some of these problems may be overcome by using purpose made 'preformed' tubing, while the condensation problem might be relieved by relocation of the damping elements or the use of only one element of different resistance value. For the main study however it was decided to continue with present fabrication methods and in order to overcome the problem of condensation all subjects were supplied with an earmould dehumidifier and cleaner (both HAL-HEN products) with instruction as to their regular daily use.

It should be noted that the subjects in this pilot study almost unanimously preferred the modified earmoulds to their own N.H.S. earmoulds. Whether this was partly due to the better fit of the moulds was not investigated for this pilot group.

The pilot study also offered an opportunity to develop and finalise the questionnaire for the main study.

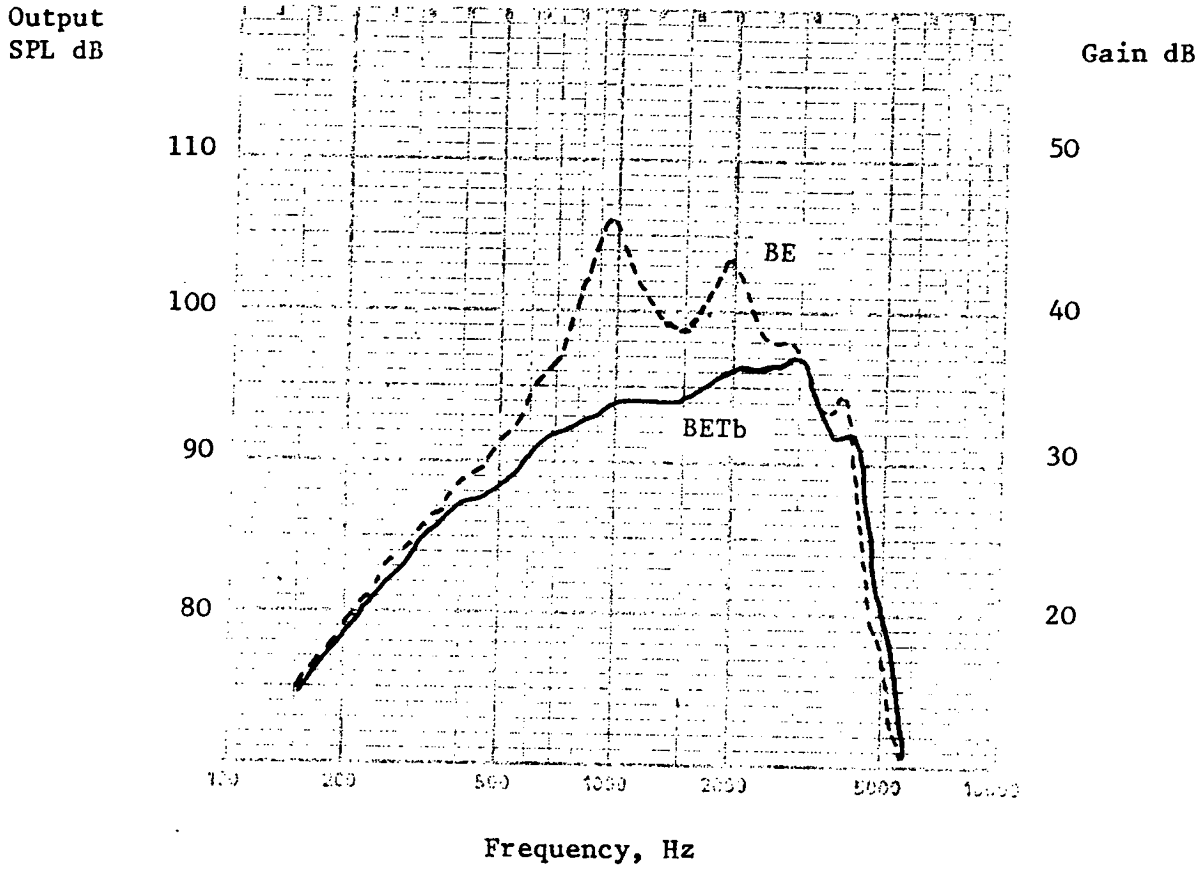
TABLE 1

Typical performance of aid/earmould combinations

Measured on Zwislocki coupler	BE	BETb	BETr	BETrTb
1 kHz gain dB	49 (50.05)	39 (39.5)	47 (49.0)	36 (39.5)
HAIC gain dB	45.0 (47.3)	38.7 (39.5)	42.0 (45.0)	35.0 (39.3)
HAIC Freq. Range,Hz	300-4.3k (270-4.3k)	200-5.5k (170-5.7k)	350-6.2k (200-6.4k)	220-7k (100->8k) estimated as outside controlled range of test box.

Measured on 2cc coupler	BE	BETb	BETr	BETrTb
1 kHz gain dB	46 (47.0)	34 (33.5)	44 (45.0)	31 (33.0)
HAIC gain dB	40.0 (40.5)	32.7(32.7)	38.0(39.0)	30.0 (31.7)
HAIC Freq. Range,Hz	300-4.2k (220-4.1k)	170-5.4k (130-5.0k)	300-5.0k (160-5.2k)	170-6.5k (110-6.5k)

Figures in brackers refer to previous laboratory study hearing aid responses (reference 1)



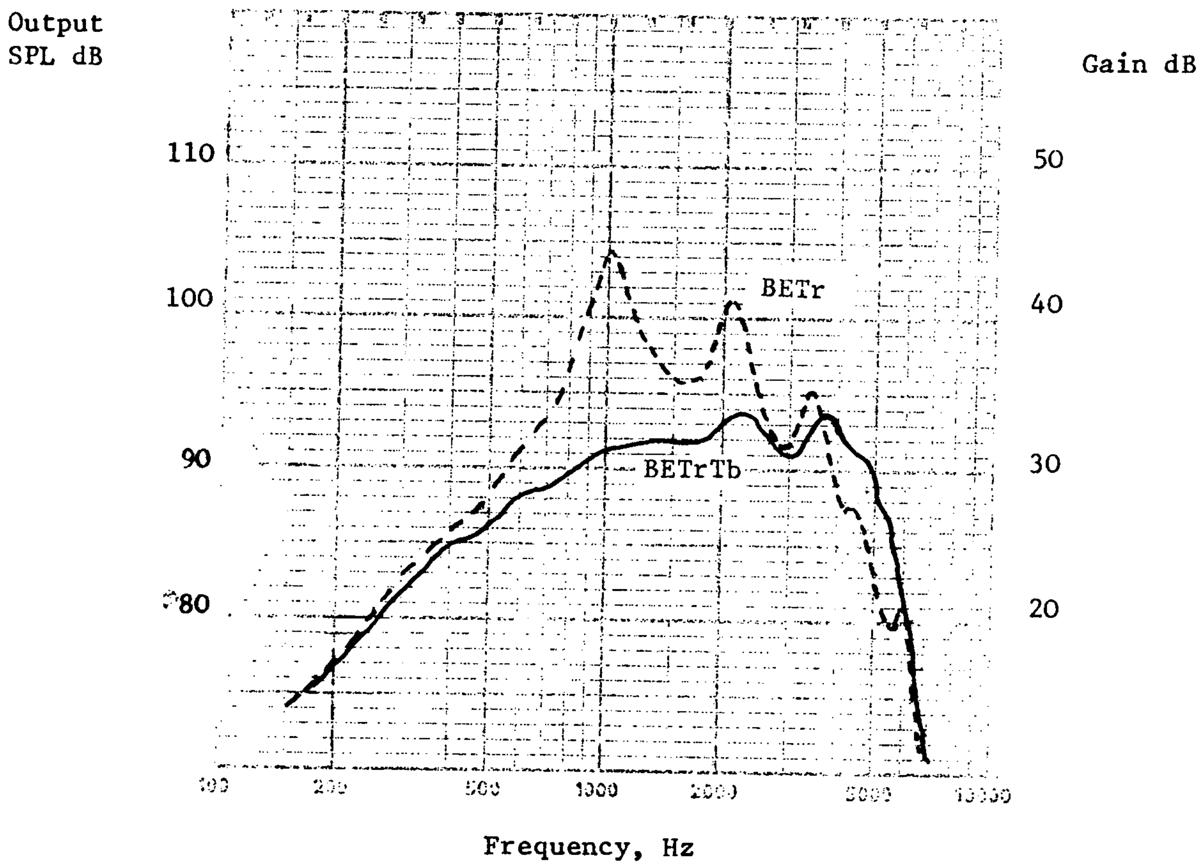
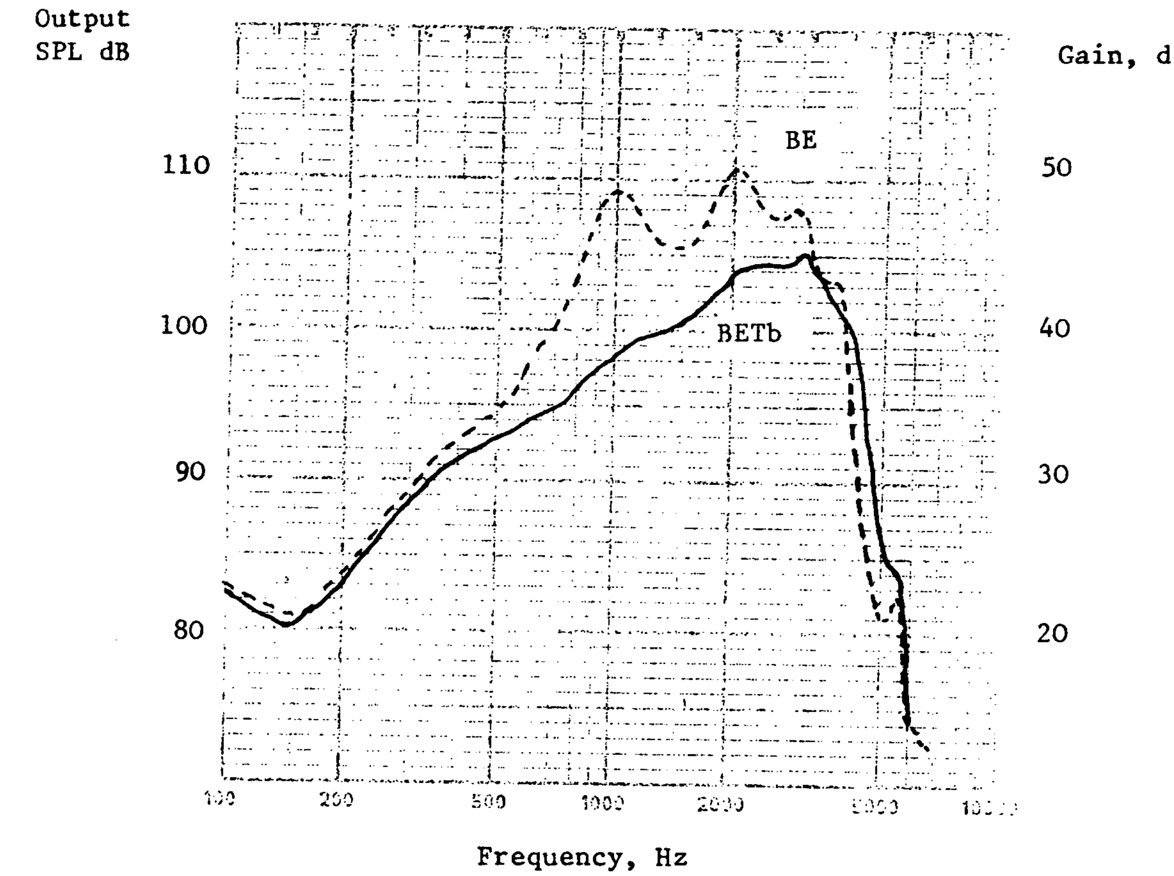
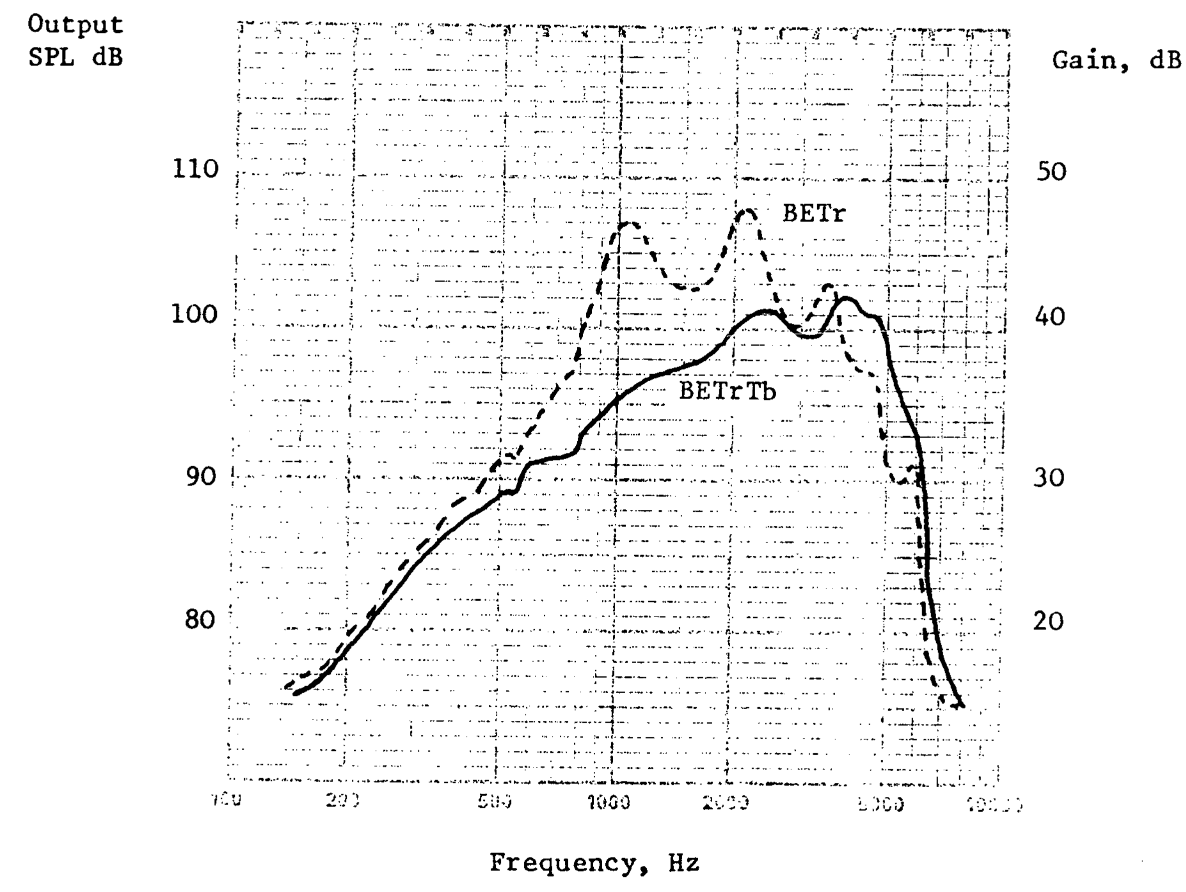


Figure 1a Comprehensive frequency response of each experimental aid measured using 2 cc acoustic coupler (60 dB input, maximum gain setting)





<u>Figure 1b</u> Comprehensive frequency response of each experimental aid, measured using Zwislocki artificial ear (60 dB input, maximum gain setting).

TABLE 2

Knowles Electronics Ltd. transducers used in the two models of experimental hearing aid

	microphone	receiver
Standard aids	BT 1751(electret)	BK 2606 (magnetic)
Modified transducer aids	EA 1934(electret)	BK2839 (magnetic)

FIGURE 2

Example of modified transducer aid and standard aid .

FIGURE 3 Modified sound tube.

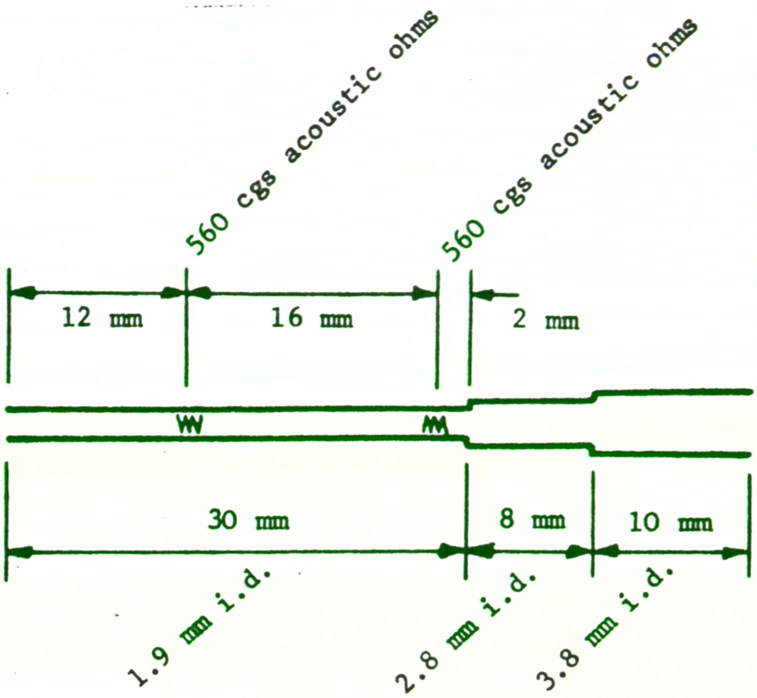


FIGURE 4 Examples of Modified and Normal Moulds.

2.4 Questionnaire Design

The original laboratory study provided information concerning speech discrimination in noise and quiet, subjective sound quality judgements and overall preference.

The aim of the questionnaire was to obtain similar information in the form of both numerical scale and qualitative comparison ratings while also identifying everyday situational factors that were of importance .

The final questionnaire (see Appendix 1) following pilot studies included:

(i) A semantic differential scale for sound quality judgements (similar to that used in the laboratory study, based on work by B.C. Grover and M.C. Martin of R.N.I.D.)

(ii) Absolute numerical rating scales for (a) speech intelligibility
(b) aid satisfaction

(iii) Qualitative comparisons between pairs of aids for
- (a) Overall preference
- (b) Sound quality
- (c) Speech intelligibility
- (d) Problems experienced
- (e) Situational factors
- (f) Reasons for preference
- (g) Hours of use
- (h) Time required for adaptation
- (i) Other factors

2.5 Main study design

Since each subject was required to try each aid for a month (to allow for any learning effects or situational variations) it was decided to run the experiment by asking the subjects to compare the aids in pairs in order to remove any problems that might have occurred if the subjects had been required to compare aids tried a few months apart. The thirty-six subjects were randomly assigned to 6 groups with 6 people in each group.

Each subject had three trial periods of a month's duration, the assessments being made at the end of each trial month through interview and recorded on the questionnaire.

The subjects preferred aids, one chosen from the two aids compared in the first month's trial, the other chosen from the remaining two aids in the second trial period were themselves compared during the final trial month.

This design is shown below :

	GROUP					
MONTH	I	II	III	IV	V	VI
FIRST	BE & BETr	BETb & BETrTb	BE & BETb	BETr & BETrTb	BE & BETrTb	BETr & BETb
SECOND	BETb & BETrTb	BE & BETr	BETr & BETrTb	BE & BETb	BETr & BETb	BE & BETrTb
THIRD	aids preferred in first and second months					

3.0 EXPERIMENTAL METHOD

3.1 Electroacoustic measurements

All aids and moulds were tested before and after each trial period to ensure they were performing to specification. All aids and moulds were within $\pm$ 2 dB of their specified frequency response, and variations in the maximum gain of the systems were less than 5 dB. A few subjects required lateral venting of their moulds to relieve the feeling of 'pressure', but this had no significant effect on the frequency response as the vents were $\approx$ 20 mm long x 0.5 mm diameter.

The moulds were checked using an electrically driven receiver for ease of standardisation of results. They were mounted using a specially made adapter tightly sealed to a Zwislocki coupler.

3.2 Subjects

Thirty-six volunteer patients acted as subjects in this experiment. Potential subjects were drawn from the files of the hearing aid departments of two local hospitals as sensorineural hearing impaired users of the N.H.S. BE-10 series hearing aids.

Each subject was given an otoscopic examination, and underwent standard clinical pure tone air and bone conduction audiometry to verify the diagnosed sensorineural hearing loss. Twenty three males and thirteen females were finally selected to act as subjects. Their mean age was 54.5 years (median 52 years range 20 - 65).

The test ear for each subject was chosen as the ear normally aided. Two subjects had binaural fittings but agreed to revert to the monaural requirements for the duration of the trial. Apart from one subject the hearing loss in the chosen aided ear did not exceed 70 dB HL when averaged over the middle frequencies (500 Hz - 2 kHz). The test ears, 22 right and 14 left, which were fitted with the trial hearing aids had the mean, median and range of hearing loss shown in Figure 5. The majority of the ears had pure sensorineural hearing loss, although there was a nominal conductive element (less than 10 dB averaged over 500 Hz to 4 kHz) in a few subjects.

10 people were fitted in their worst ear
13 people were fitted in their best ear
13 people had symmetrical losses

The aids used by the subjects before the trial were as follows:

BE 11 - 23 subjects

BE 12 - 10 subjects

BE 13 - 2 subjects

BE 14 - 1 subject

Three subjects had their own aids set to the 'Bass cut' position.

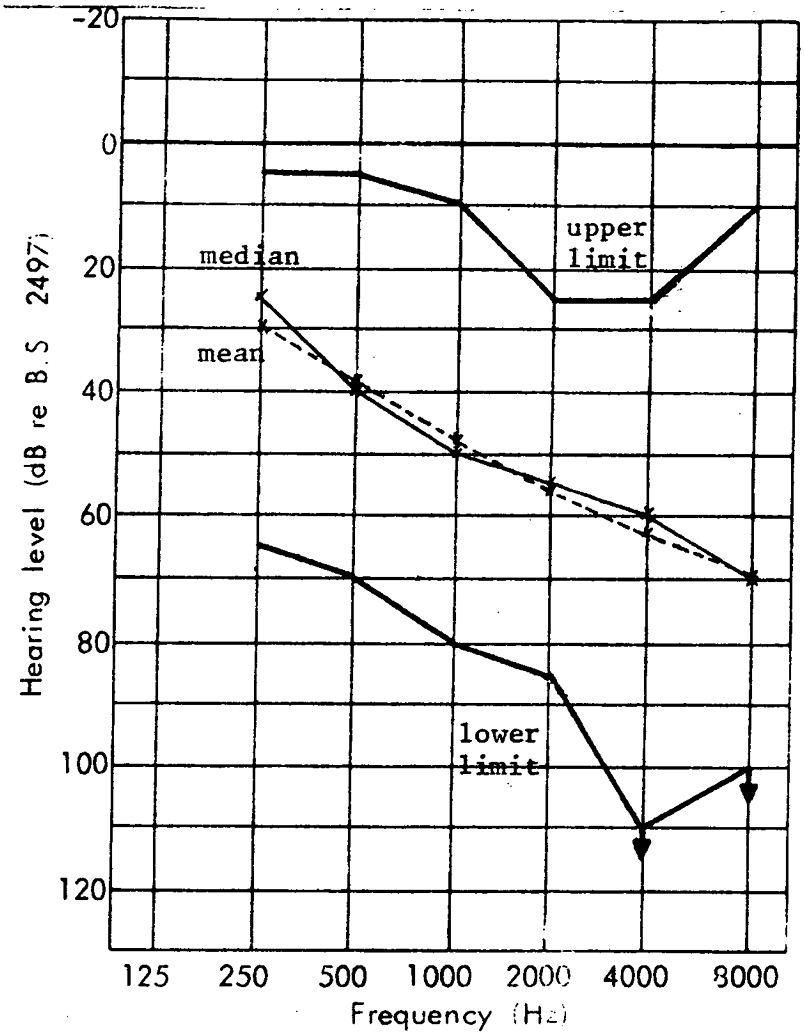
3.3 Experimental procedures

Those people found suitable to participate were advised of the details of the experiment and their written consent obtained. Appointments were then made for impressions to be taken and earmoulds made. The finished moulds were then checked and if necessary modifications were carried out to ensure they conformed to design specification and to ensure similarity of fit between standard and modified moulds. Subjects then returned for the fitting of their earmoulds when any alterations required to ensure a satisfactory fit were made, following which the response characteristics of the moulds were checked once more. They were then given their first pair of aid/earmould combinations, batteries, earmould dehumidifiers and cleaner and instructions as to trial procedure, purpose of the trial and cleaning and condensation clearing instructions (see Appendix 2). An appointment was also made for the following month's assessment; before leaving it was established that the trial procedure and instructions were fully understood and that the subject was competent at insertion and removal of the earmoulds.

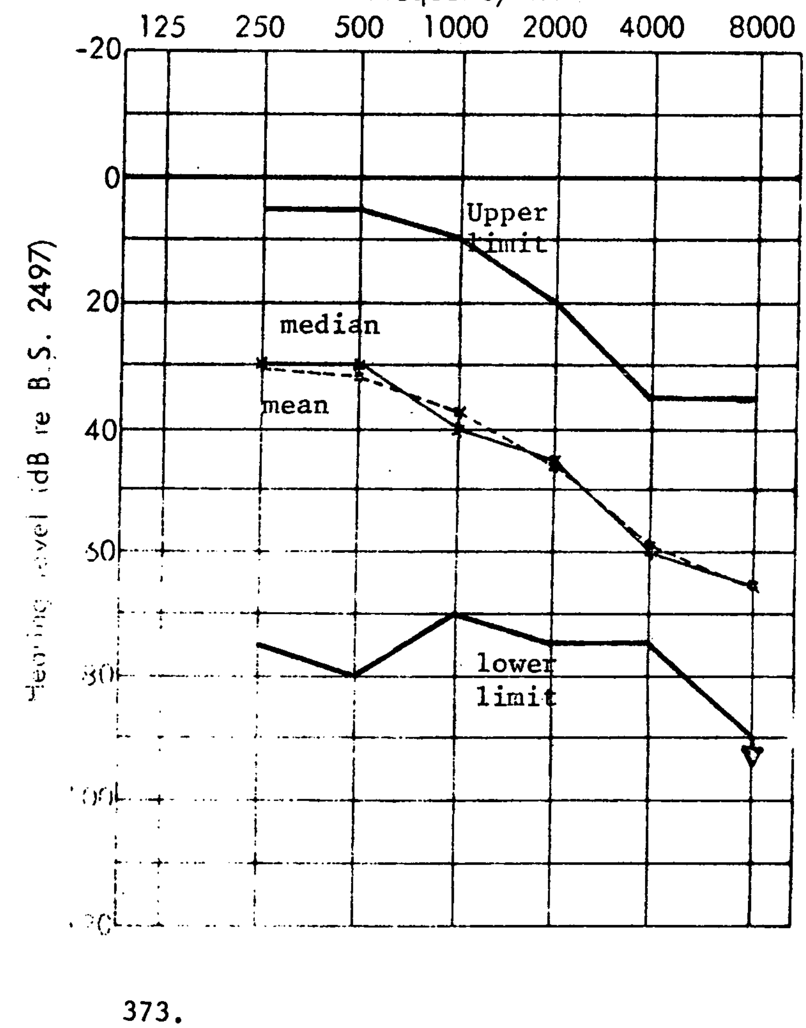
To remove a possible source of bias the modified moulds were referred to as 'banded tube', the standard mould as 'clear tube', while the modified aids as 'red aid' and the standard aid as 'black aid'. The aids were identified by a small inconspicuous coloured spot marked on the case, (Fig. 2)

Figure 5 Mean (dashed line), median (solid line) and limits of subject's hearing in ear chosen for experimental aids.

Field Study



Previous Laboratory Study



4.0 RESULTS and CONCLUSIONS

4.1 General observation

(i) No difference was generally experienced between the various aid systems in respect of condensation problems, comfort, mould insertion, cleaning, hours of use, wind noise, feedback or the appearance of the aid or earmould.

(ii) Situational factors appeared to have little relevance to choices as the preferred system was generally preferred in all situations.

4.2 Speech Intelligibility Judgements

The questionnaire included two main questions regarding speech intelligibility. Question 4(i) requested a qualitative comparison between pairs of trial aid systems, while question 9 asked for a numerical scale rating for each aid system. The results from the first two trial months are shown in Figure 6.

Numerical Ratings

The mean numerical ratings suggest that BE and BETb (the narrow bandwidth) systems were rated as providing better speech intelligibility than the wide band systems BETr and BETrTb, the earmould modifications appearing to have little influence on ratings.

Qualitative comparisons

The qualitative comparisons show that BE was rated as better in more, and worse in less comparisons than any other trial system.

It should be noted that in almost one half of all the trial comparisons no preferences were expressed between systems while moderate or strong preferences were expressed in one third of the comparison. However, the individual subjects' preferences (not noted in figure 6) show that only ten subjects expressed no preference in both their trials while over half the subjects expressed moderate or strong preferences in at least one trial.

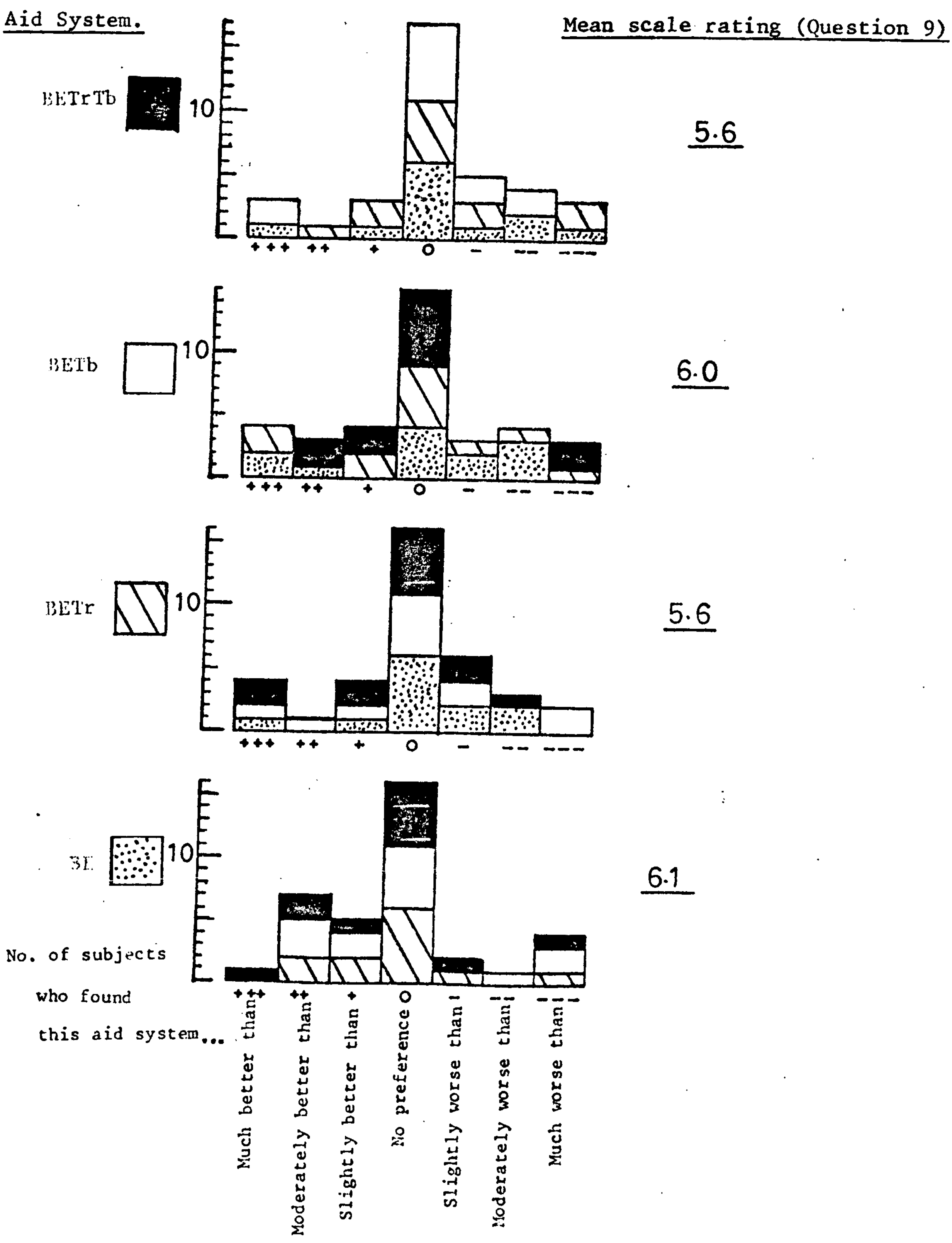
CONCLUSION

The aid systems were ranked in the following way for speech understanding

Higher Intelligibility	BE BETb
Lower Intelligibility	BETr BETrTb

Figure 6. Speech Intelligibility Ratings

Qualitative comparison (Question 4(i))



4.3 Sound Quality Judgements

The questionnaire included two main questions regarding sound quality. Question 4(ii) requested a qualitative comparison between pairs of trial aid systems while question 8 asked for numerical scale ratings for each system. The results from the first two trial months are shown in Figures 7 and 8 and table 3.

Numerical ratings

The mean semantic differential ratings (figure 8 and table 3) show that BETb and BETrTb, the systems with the modified moulds, were rated as smoother and mellower but not as clear as the other two systems. The system with the modified aid and mould BETrTb was rated as the least pleasant and least clear but also as the most smooth and most mellow. It was apparent from the subjects responses during the interview that smoothness and mellowness were not regarded by everybody as desirable end points on the semantic differential scales.

Qualitative comparisons

The qualitative comparisons show that BE was rated as better in more, and worse in less comparisons than any other trial aid system.

It should be noted that in slightly less than one third of all the trial comparisons no preferences between systems were expressed, while moderate or strong preferences were expressed in one third of the trials. However, the individual subject preferences (not noted in Figure 7) show that only three subjects expressed no preferences in both their trials, while over half the subjects expressed moderate or strong preference in at least one trial.

CONCLUSIONS

From the results of the qualitative comparisons and the semantic differential ratings it appears that clarity was the most important feature contributing to sound quality preferences.

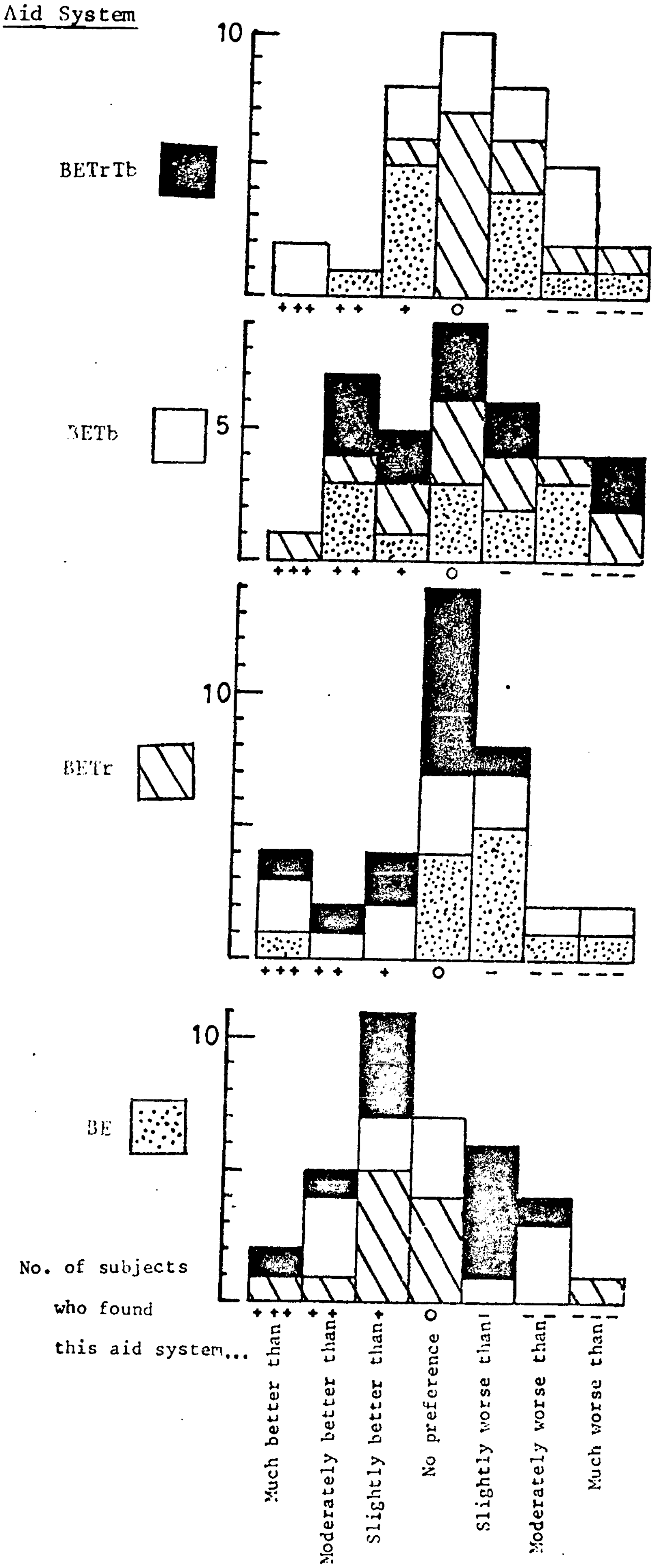
The aid systems were ranked in the following way for sound quality

Best sound quality	BE
	BETb, BETr
Worst sound quality	BETrTb

Figure 7

SOUND QUALITY

Qualitative Comparison (Question 4 ii)



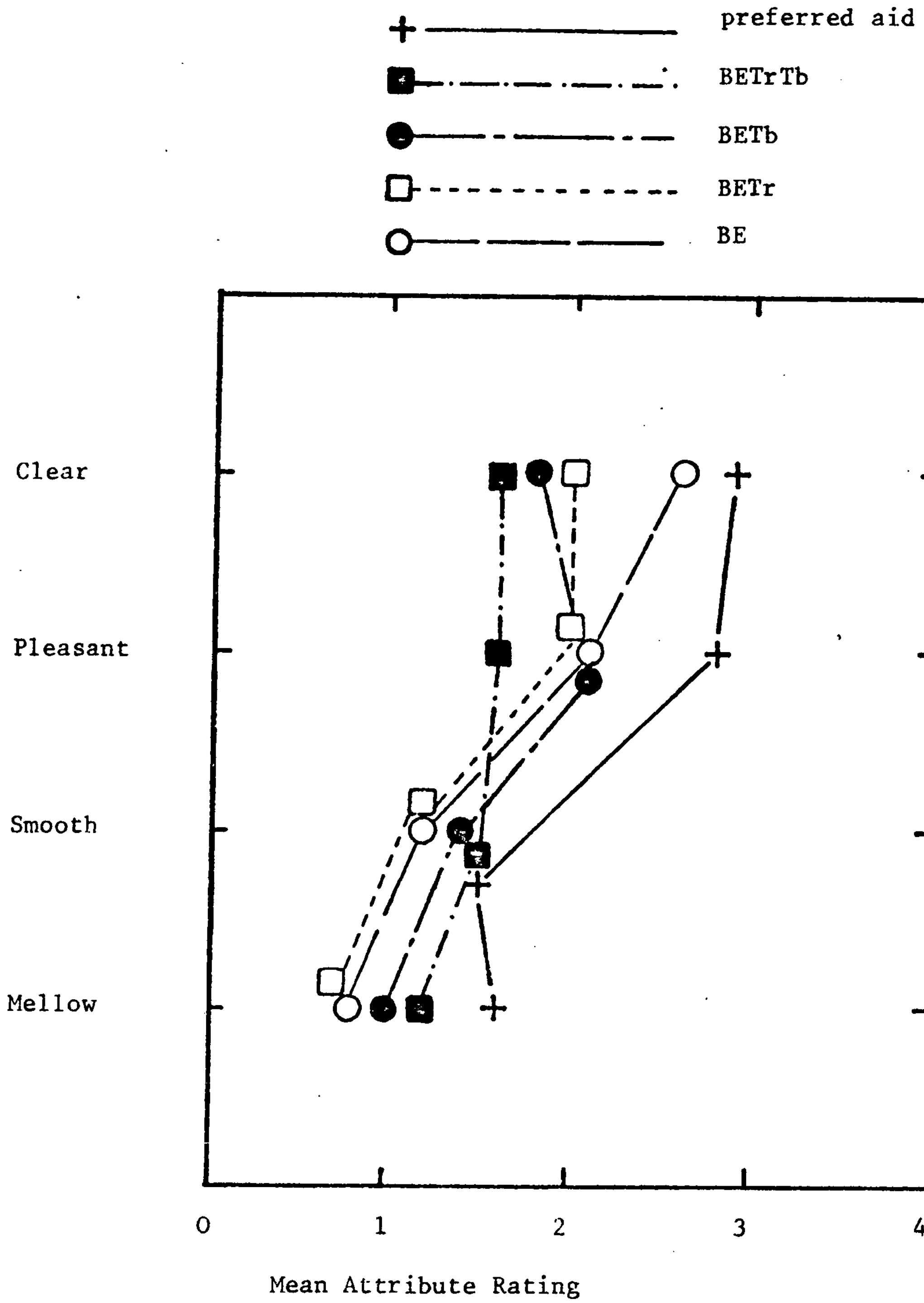


Figure 8 Mean attribute ratings of each experimental aid, and for the aid preferred by each subject.

TABLE 3 Mean ratings of each aid on semantic differential scales

(Laboratory study ratings in brackets, reference 1)

	BE	BETr	BETb	BETrTb	Mean over aids	Preferred aid
CLEAR	2.6(2.2)	2.0(2.4)	1.8(2.7)	1.6(2.7)	2.0 (2.5)	2.9 (3.6)
PLEASANT	2.1(2.4)	2.0(2.1)	2.1(2.6)	1.6(2.9)	2.0 (2.5)	2.8 (3.6)
SMOOTH	1.2(1.3)	1.2(1.6)	1.4(2.0)	1.5(2.3)	1.3 (1.8)	1.5 (2.8)
MELLOW	0.8(1.2)	0.7(1.7)	1.0(1.8)	1.2(1.8)	0.9 (1.6)	1.6 (2.0)
MEAN OVER ATTRIBUTES	1.7(1.8)	1.5(2.0)	1.6(2.3)	1.5(2.4)	-	2.2 (3.0)

4.4 Pair comparison preferences and satisfaction ratings

The questionnaire included two main questions regarding overall preferences. Question 3 requested a qualitative comparison between pairs of aid systems while question 10 asked for a numerical rating for the subject's overall satisfaction with each aid system. The results from the first two trial months are shown in Figure 9.

Qualitative comparisons

The qualitative comparisonsshow that the systems with the narrow band aids BE and BETb were preferred in more trials and found worse in less than the systems with the wide band aids.

In less than one sixth of the trial comparisons no preferences between systems were expressed while moderate or strong preferences were expressed in over one third of the trials. The individual subject preferences (not indicated in figure 9) show that only three subjects expressed no preferences in both their trials, while over half the subjects expressed moderate or strong preferences for at least one aid system.

Numerical rating

The numerical satisfaction ratings also indicated the subjects' slightly greater satisfaction with BE and BETb, the systems using the narrow band aids.

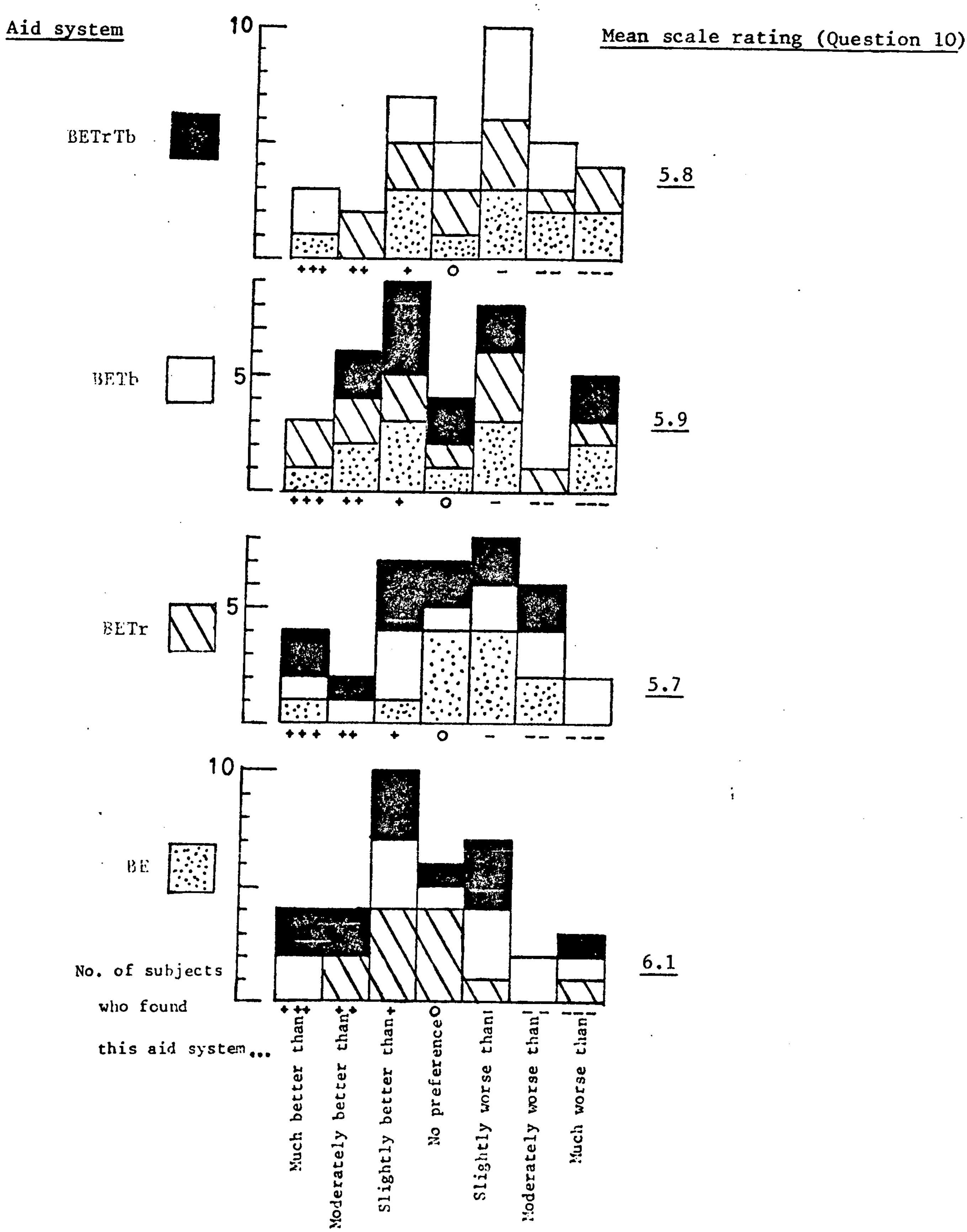
CONCLUSIONS

The aid systems were ranked in the following way for overall preference and satisfaction.

Preferred more often and giving greater satisfaction	BE, BETb
Preferred less often and giving lower satisfaction	BETr, BETrTb

Figure 9 Comparison preferences and satisfaction ratings

Qualitative comparison (Question 3)



4.5 Final choices of aid systems

The subject's final choices of their preferred aid systems are shown in Figure 10.

Not included in the above figure are the results for six subjects. Three people found one or more of the aid systems too quiet, and two subjects' preferences were due solely to mould discomfort. One subject expressed no preference between any of the aid systems.

CONCLUSION

The rank ordering of aid systems for subjects final preferences is as follows

BE	preferred by 13 people
BETb	preferred by 10 people
BETr	preferred by 9 people
BETrTb	preferred by 7 people

4.6 Subjects' own aid comparisons

Two thirds of the subjects preferred their final choice trial aid system to their own N.H.S. aid and mould. Only four subjects definitely preferred their own N.H.S. aid and mould to all of the trial systems.

As suggested by the pilot study the fit of the subjects' earmoulds appeared to be an important factor governing their preference; however this was not investigated in any controlled way.

4.7 Implication for future aid and mould provision

The final choices of aid system show that about half the subjects found one or other of the modifications an improvement on the standard BE system. This suggests that a large number of NHS BE series users would benefit from the availability of a wider range of hearing aids. Possible guidelines for predicting subject aid preferences are indicated from the field study results and are discussed in the next chapter.

The results also suggest that completely replacing the BE system by any single one of the modified systems would leave more people displeased than pleased by their change of aid.

It should of course be noted that although these are the results for the people who participated in this study, the applicability of these findings to the hearing aid user population as a whole is open to further discussion.

4.8 User gain settings of aid systems

At the end of each trial period each subject's user aid gain was measured as set when listening to the author's conversational voice at a distance of approximately 1 metre. These gain settings are shown in Figures 11 and 12 plotted against average hearing loss (500 Hz - 2 kHz).

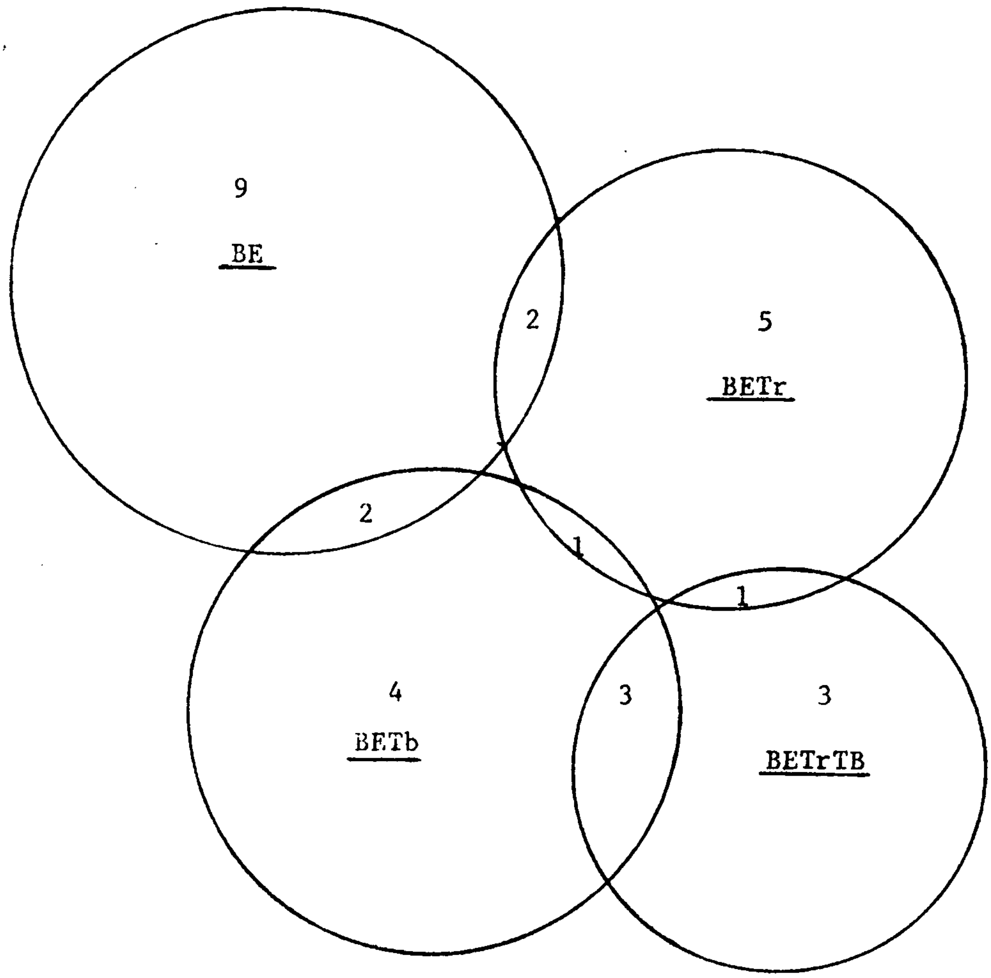
The mean 500 Hz and 1 kHz gain and the correlation coefficients, slopes and intercepts of best fit straight lines are as follows.

	BE	BETr	BETb	BETrTb
Mean 500 Hz gain (dB)	18.3	17.1	21.6	18.0
Mean 1 kHz gain (dB)	32.3	32.1	26.6	24.0
Correlation coefficient	0.43	0.53	0.50	0.50
Slope of fitted line	0.34	0.41	0.36	0.34
Intercept for 1 kHz gain	15.9	13.3	9.4	7.9

This suggests that the systems with the modified moulds required the aids to be turned up slightly in gain compared to the systems with the standard moulds to compensate for the loss in sound power due to resonance damping.

Figure 10 Final preferences

(a) Venn diagram showing subject final preference



(b) The same data shown in table form (laboratory study preferences in brackets, reference 1)

	chose standard mould	chose either mould	chose modified mould	total
Chose standard aid, BE	9 (1)	2	4 (6)	15
Chose either aid	2		3	
Chose modified aid, BETr	5 (3)	1	3 (18)	9
TOTAL	16		10	

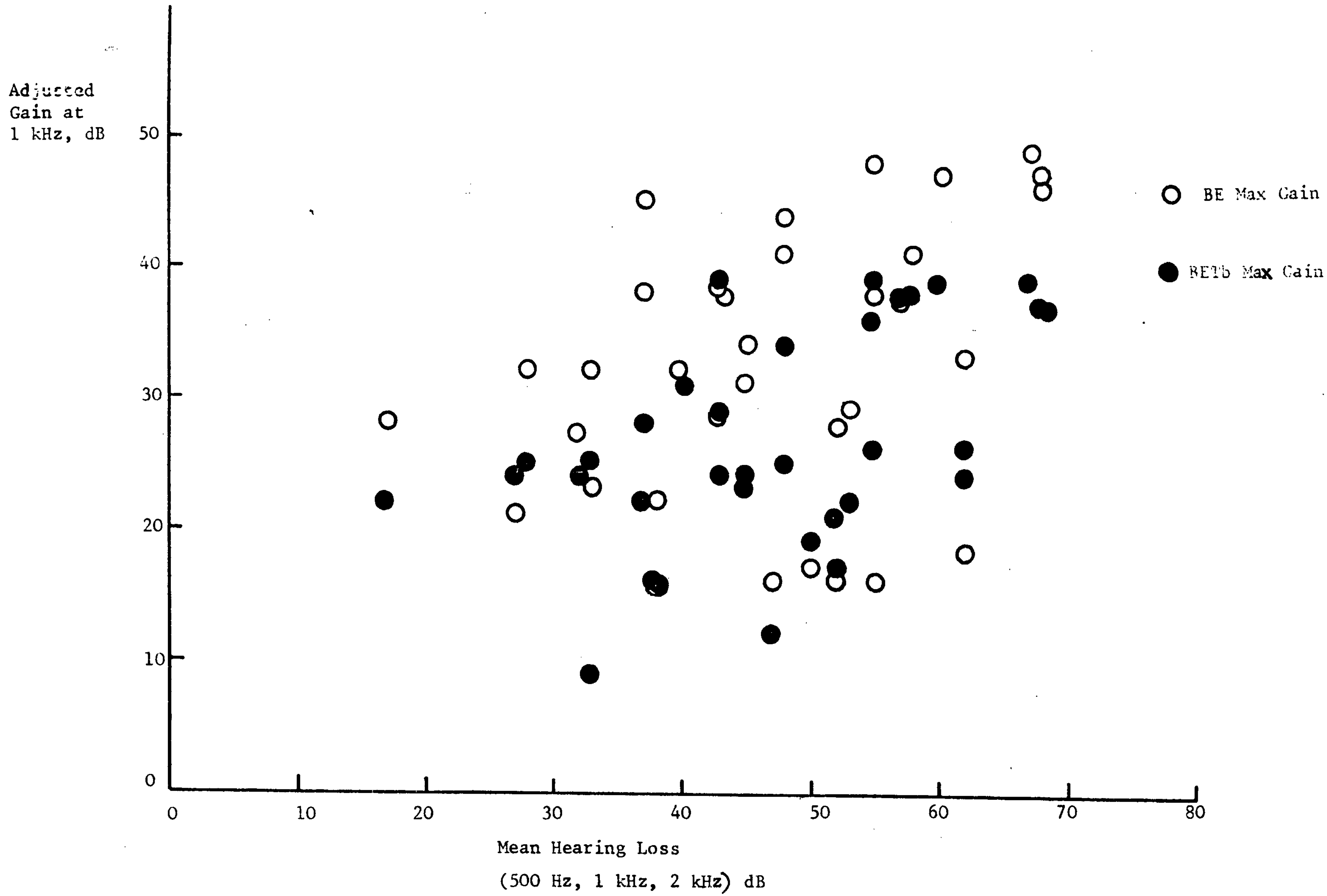


Figure 11 Relation between mean Hearing Level and 1 KHz gain for BE (open symbols) and BETb (closed symbols)

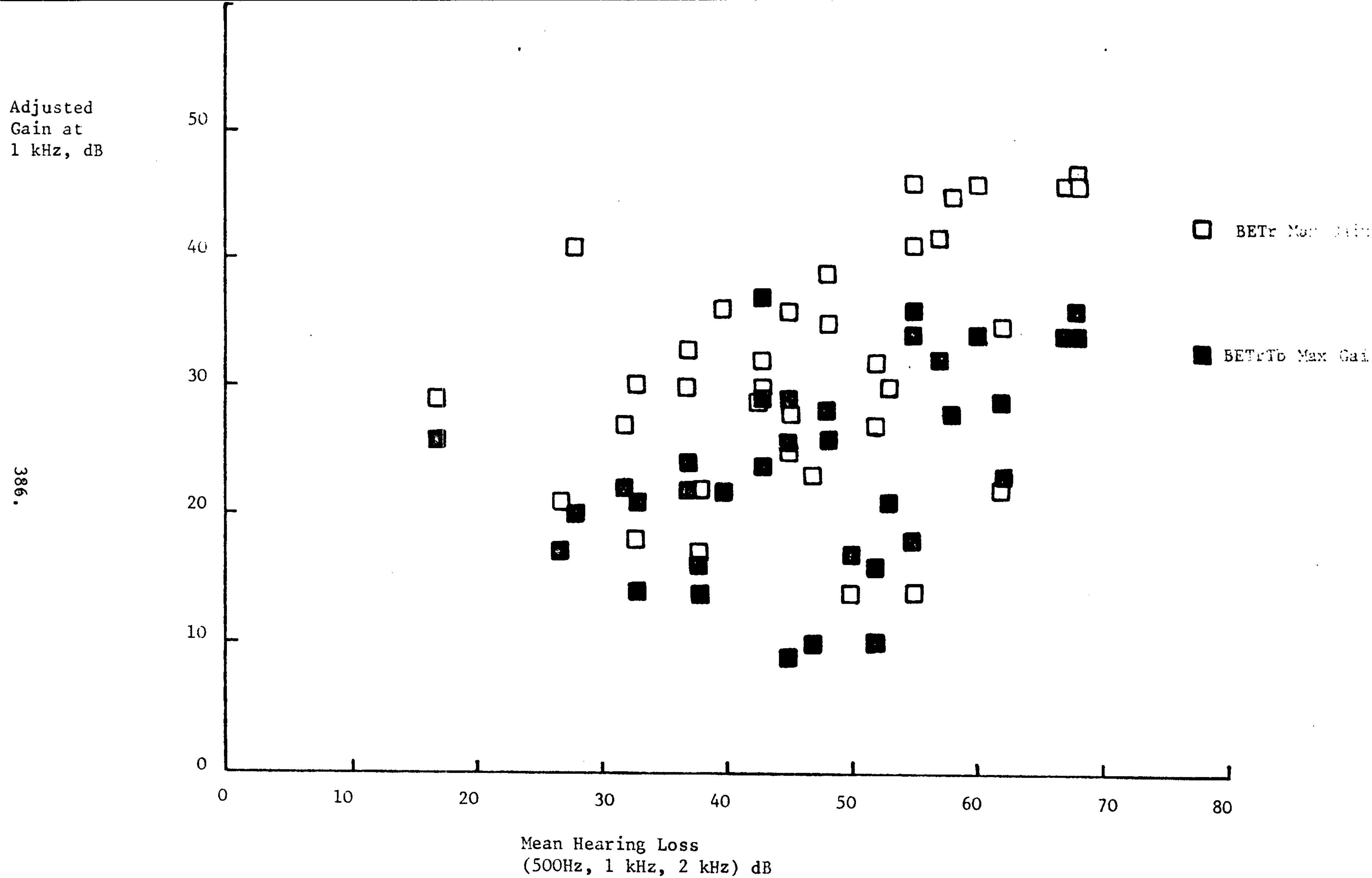


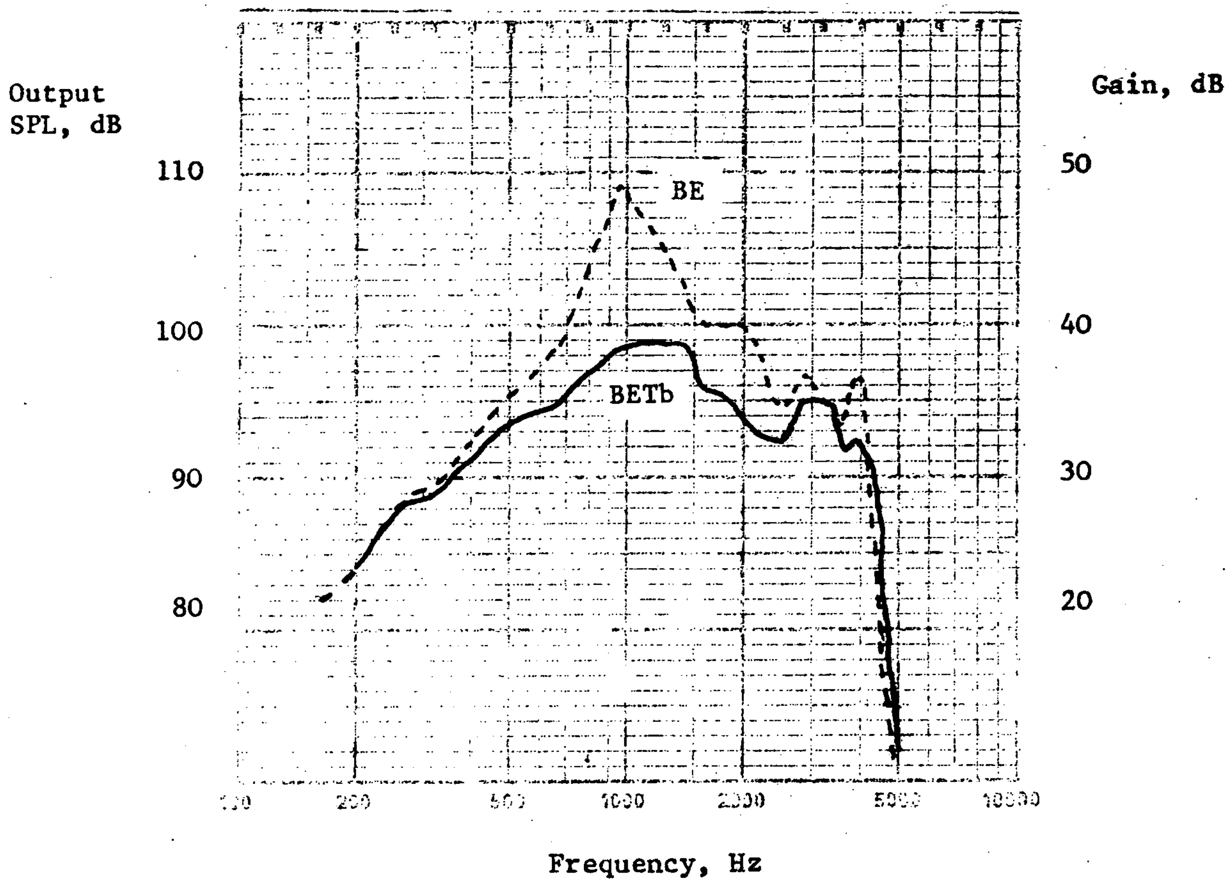
Figure 12 Relation between Mean Hearing Level and 1 kHz gain for BETr (open symbols) and BETrTb (closed symbols)

4.9 Preference for Bass Cut

Following the subjects' final assessment at the end of their third month, their chosen aid was adjusted to the 'BASS CUT' position and they were asked to comment whether they found this better, worse, or no different while listening to the author talking in a normal conversational voice at a distance of approximately one metre. Due to the limitations of time it was not possible to test the effect of Bass cut in a more rigorous manner or on all the aid systems. There were no investigations on the effect of bass cut in the original laboratory study and so no comparisons of results would have been possible. Twelve of the thirty-six subjects expressed a preference for 'bass cut' and these preferences were spread evenly among the aid systems.

4.10 Insertion gain measurements

The 'insertion gains' of the aid systems were measured on Kemar in an anechoic chamber for sounds from an incidence of 0^{o} (directly ahead). The frequency responses are shown in figure 13 and demonstrate how all aid systems are deficient in gain above 1 kHz. The reinsertion of the lost ear canal resonance (apparent from the insertion gain plots compared to the coupler response plots) could be accomplished through an alternative earmould plumbing configuration (reference 9). However as pointed out in this reference, with a microphone location other than in the concha with a skeleton mould, this compensation would only be correct for the designated sound source direction. The importance of this effect requires investigation as does the reinsertion of the ear canal resonance for the more severely hearing impaired.



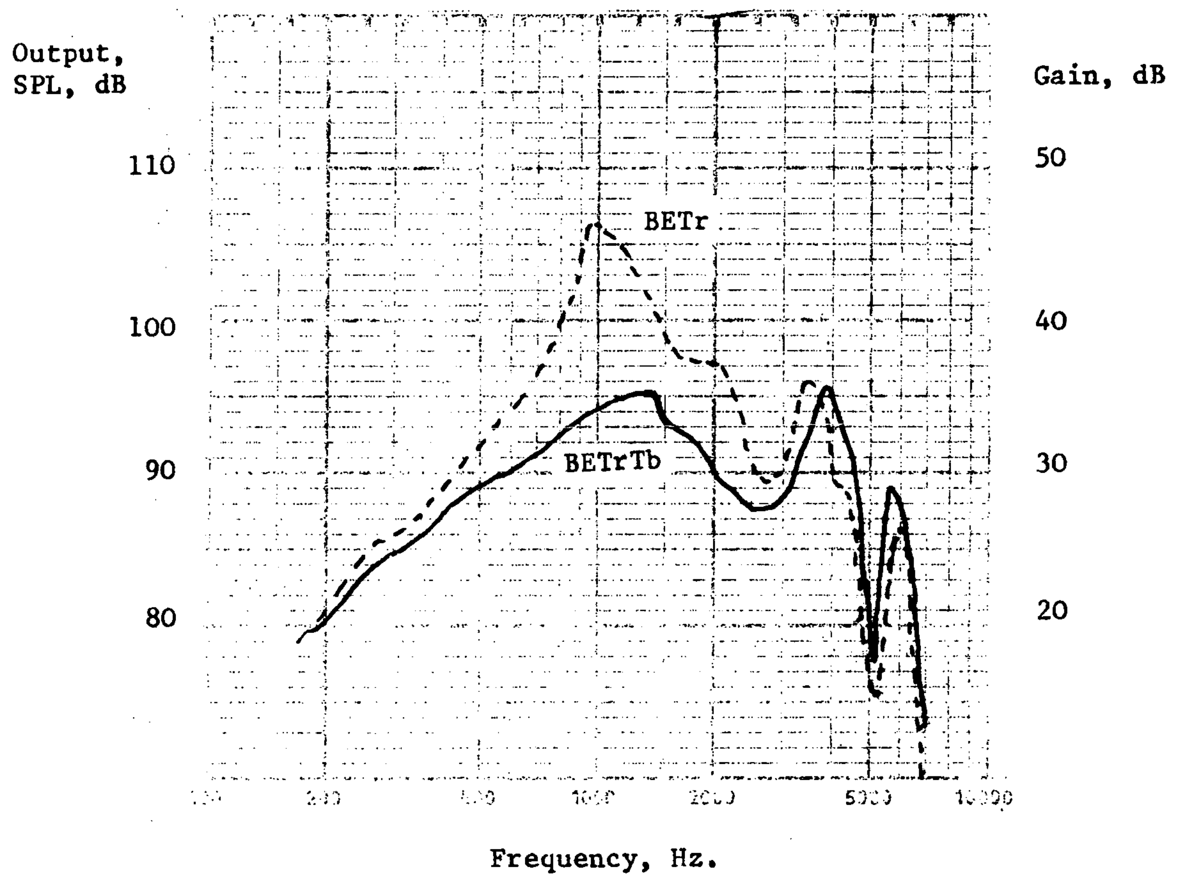


Figure 13 Comprehensive insertion gain and output of each experimental aid measured using Kemar and Zwislocki artificial ear (60 dB input, maximum gain setting) for pure tones incident from the front.

5.0 POSSIBLE FACTORS RELATING TO SYSTEM PREFERENCES

5.1 General factors

The preferences of subjects seem to bear no apparent relation to their occupation, lifestyle, noise environment, age, sex, previous aid type, hours of aid use, problems experienced, or hearing loss in unaided ear.

5.2 Audiometric factors

By examining the subjects' audiograms and aid preferences the following relationships are suggested by the 'trends' of preference but are in no way definitive.

5.2.1 Mould preferences

Aid systems with modified moulds tended to be preferred by those people with a lesser average hearing loss and a less steep slope in their pure tone audiogram between 250 Hz and 1 kHz. Aid systems with the standard moulds, BE and BETr tended to be preferred by those subjects with a greater average hearing loss and a steeper slope in their pure tone audiograms between 250 Hz and 1 kHz (Figure 14).

Support for these findings may be drawn from two published research studies.

The very rigorously designed study by Pascoe (reference 6) found that optimum speech discrimination scores were obtained with aid 'real ear responses' that effectively 'mirrored the audiogram' and so required a steeper slope of response for those subjects with steeper losses.

The study by Jerger and Thelin (reference 8) concluded that the SSI (synthetic sentence identification) scores of those subjects with 'steeper' hearing losses appeared to rank aids in a different way from those with less steep losses, the latter group's SSI scores ranking aids in a similar way to those of normally hearing subjects.

The results of these studies suggest that the improvement in sound quality experienced by normally hearing people when resonance peaks in the response are damped out (reference 10) may also occur for those with flatter and lesser hearing losses but not necessarily for those with steeper and greater losses.

One possible explanation for this may perhaps be that for subjects with greater or steeper losses the resonance power around 1 kHz may decrease the masking effect of the frequencies below 1 kHz (reference 11), while for those with flatter audiograms and a lesser hearing loss the resonance peaks may only detract from the quality and intelligibility of the speech.

If the main reason for the preference of some subjects for the standard mould systems in this study was the resonance peak energy providing release from low frequency masking, then a similar effect might also be obtained by damping out the resonant peaks with the modified mould while also cutting the bass response electronically or through venting; this would also improve the high frequency relative to the low (see figure 15). This effect was not tested in the present study and would require higher power aids to compensate for the decrease in available sound power due to both the damping and the reduced bass response.

5.2.2 Aid preferences

Wideband aids tended to be preferred by people with a less steep slope in their P.T.A. from 250 Hz - 4 kHz. It appears therefore that these subjects had enough high frequency hearing to benefit from the extra bandwidth when the aid gain was adjusted to their usual listening level, governed mainly by the low frequency power output. The preference of the other subjects for narrow band aid systems may perhaps be due to the irregularities of the wide band aid response between 2 kHz and 4 kHz. Also, for those with poor high frequency hearing the increase in bandwidth may increase the distortion while providing little extra benefit.

5.2.3 Bass Cut preferences

Bass cut tended to be preferred by those people with a steeper slope in their P.T.A. from 250 Hz - 2 kHz. The bass cut has its main effect in this region and so may provide release from low frequency masking while allowing the user aid gain and consequently the high frequency output to be increased.

5.3 Distortion

Aid systems with the modified moulds required on average slightly higher gain than those with the standard moulds This could possibly result in more distortion for those with greater hearing losses who have their aids set at higher gain levels. Those subjects who preferred the standard mould systems set their gain levels on average a few dB higher than those who preferred the modified moulds. However, distortion measurements and oscilloscope traces showed little noticeable distortion for pure tone inputs of up to 75 dB at maximum aid gain settings. The effects of transients or of peaks in the speech waveform were not investigated and so the possibility of distortion effects cannot be totally ruled out.

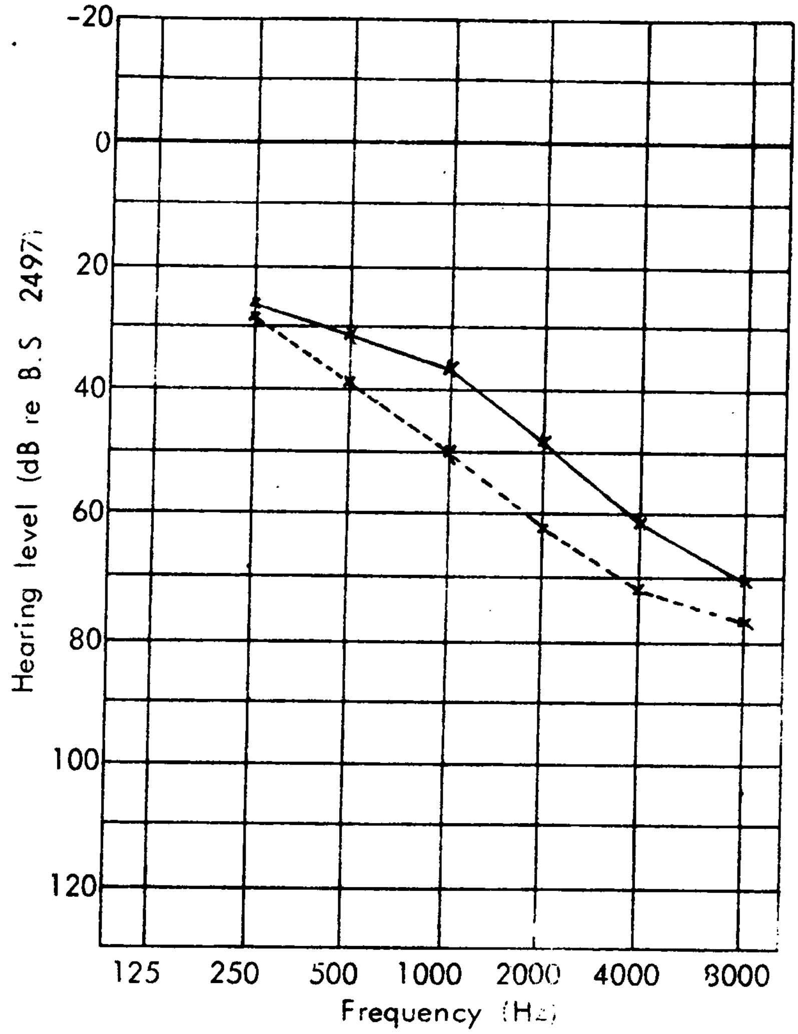
Figure 14

Mean audiometric data for two 'groups' of the subjects in the field study.

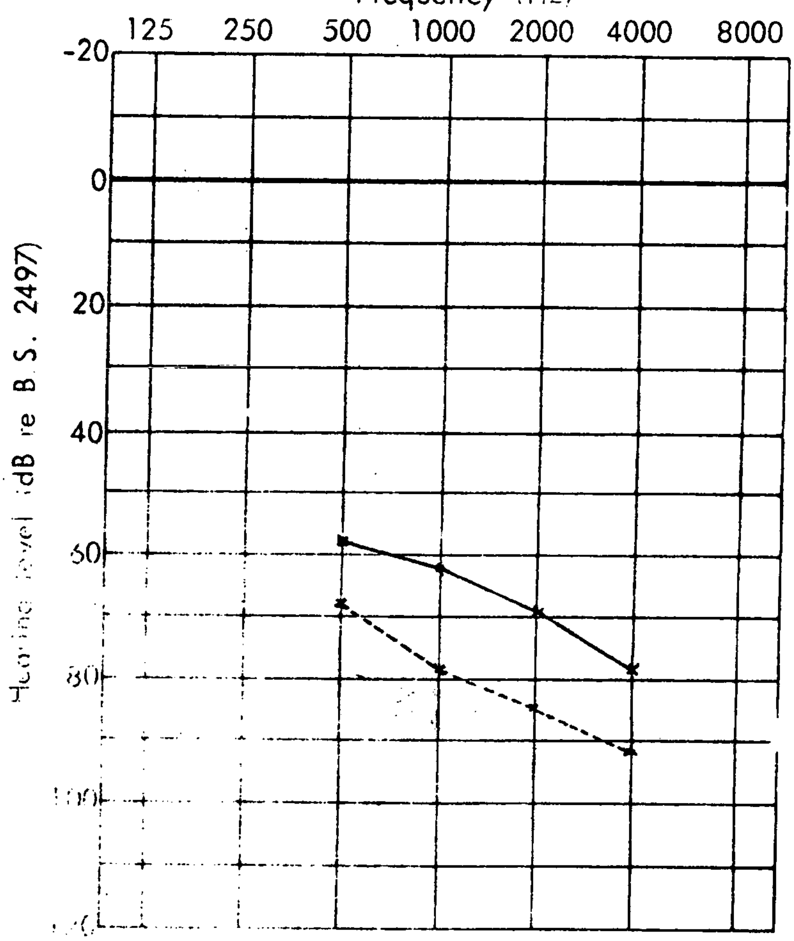
people who preferred systems with modified moulds (solid lines)

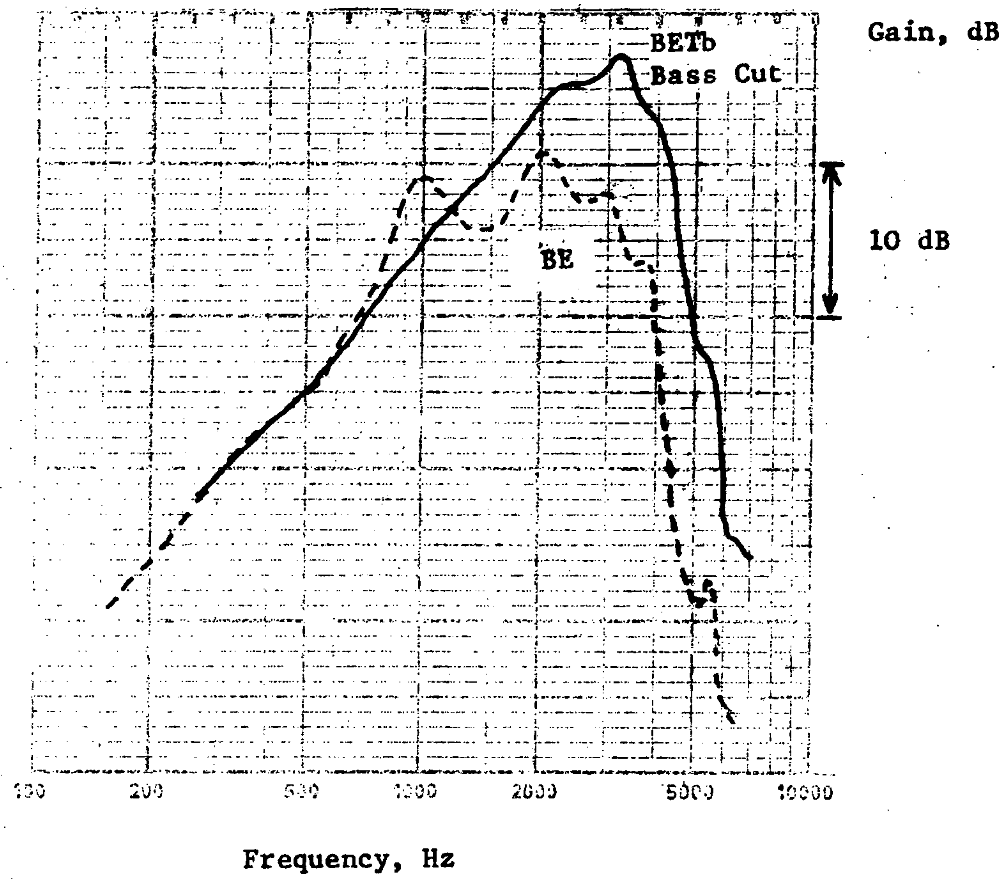
people who preferred systems with normal moulds (dashed lines)

Pure tone thresholds



Most comfortable levels.





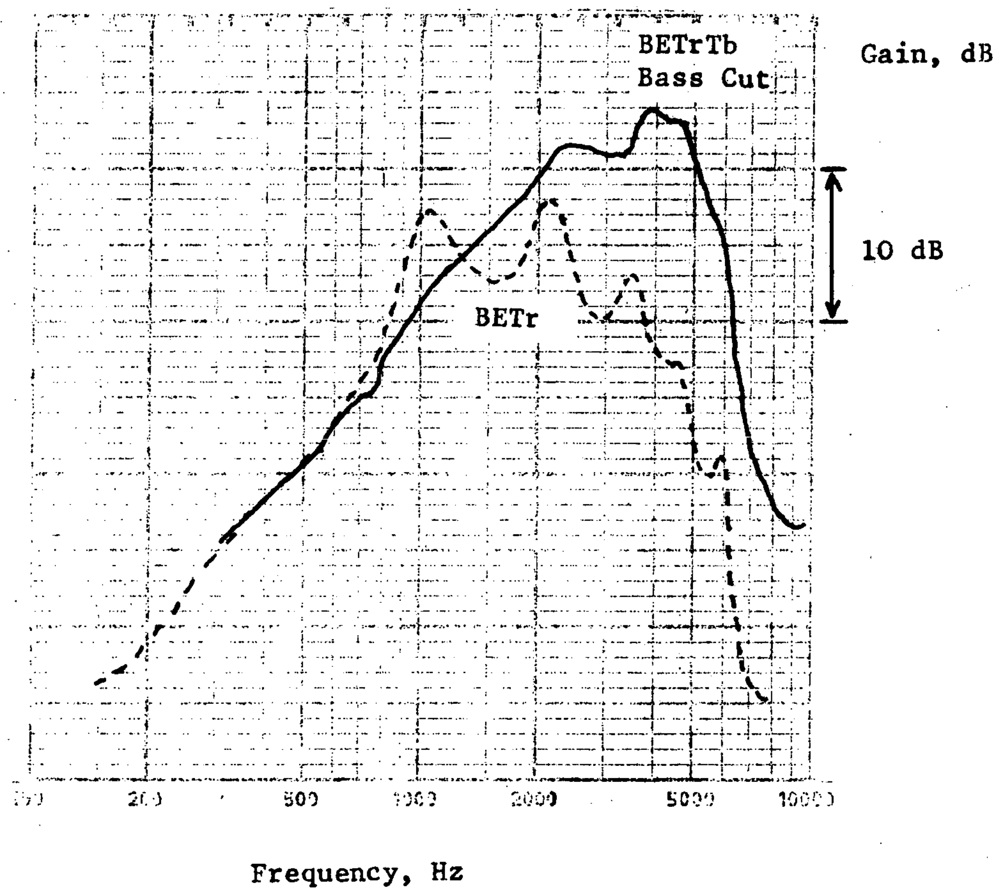


Figure 15. Relative responses of modified mould systems with aids set to bass cut response and normal mould systems with aids set to normal response.

6.0 COMPARISON OF FIELD STUDY AND LABORATORY STUDY RESULTS

6.1 Summary of field study results

6.1.1 Speech Intelligibility

The aid systems were ranked in the following way for benefit to the aid user

Higher intelligibility	BE
	BETb
Lower intelligibility	BETr
	BETrTb

6.1.2 Sound quality

Subjective judgements of sound quality showed that the aid system with the modified transducers and modified mould BETrTb was rated the least pleasant and least clear but also the most smooth and most mellow. The system with the standard aid and mould, BE appeared to provide the most preferred sound quality.

6.1.3 Overall preferences

The aid systems were ranked in the following order by overall preferences.

BE
BETb
BETr
BETrTb

6.2 Summary of laboratory study results (from reference 1)

6.2.1 Speech intelligibility

The aid systems were ranked in the following way for benefit to the aid user

Higher intelligibility	BETrTb
	BETr
	BETb
Lower intelligibility	BE

The results indicated that extra high frequency bandwidth is important for improved aided speech intelligibility.

6.2.2 Sound quality

Subjective judgements of sound quality showed that the aid systems with modified moulds were judged as clearer, smoother, more mellow and more pleasant to listen to than the standard BE 11 response.

6.2.3 Overall preferences

The aid systems were rated by overall preferences in the following order.

BETrTb
BETb
BETr
BE

6.3 Differences between results of the laboratory and field studies

The results from the previous laboratory study appear to differ from those of the present field study. The laboratory study found that BETrTb, the aid system with the extended and smoothed response provided the best intelligibility and sound quality and was preferred by the largest number of subjects, while BE, the system with the standard BE 11 and normal mould provided the worst intelligibility and sound quality and was the least preferred system. The field study found that BE provided the best intelligibility and sound quality and was preferred the most often while BETrTB was judged as providing the worst intelligibility and sound quality and was the least preferred aid system.

This apparent contradiction may perhaps be understood by considering four possibilities

(a) Validity of laboratory studies
The laboratory study may have used tests that were poor at predicting real-life preferences.

(b) Subject group differences
The groups of subjects used in the two studies may have differed significantly in some respect.

(c) Aid system differences
The aid systems used in the two studies may have differed significantly in some respect.

(d) Test ear differences.

Considering each of the above possibilities in turn:-

6.3.1 Validity of laboratory study

(i) The speech intelligibility tests were looking for 2% differences in high frequency consonant weighted word lists.

This might have favoured the wide band aid systems. In real life where grammatical sentence clues and visual speech reading clues are available this small difference may not necessarily give a large or even noticeable difference in aid performance.

(ii) The subjective sound quality tests were conducted in an anechoic room, which could highlight any 'resonance' or 'ringing' effect due to the earmould as there would be no competing room resonances. This might have favoured the systems with the modified moulds.

The combined effects of (i) and (ii) above might have been to strongly favour BETrTb, the system with both the modified wide band transducer and modified damped earmoulds, in the overall preference selection.

6.3.2. Subject group differences

The mean P.T.A. of the subjects used in the two studies differed slightly, that of the field study group having a slightly steeper slope and greater average loss (see Figure 5). However, if the field study group is divided into those who preferred the modified moulds and those who preferred the standard moulds, it is interesting to note that the mean P.T.A. of the former group is very similar to that of the laboratory group (see Figures 14 and 5). This suggests that the differing results between laboratory and field studies may in part be due to the subjects in the latter study having on average a slightly greater hearing loss and steeper slope of loss between 250 Hz and 1 kHz.

6.3.3 Aid differences

The aids used in the two studies had slightly different frequency responses due to the different transducers employed. In particular the wideband aid used in the field study had a slightly reduced bass response and a slightly reduced response for frequencies above 3 kHz when compared to that used in the laboratory study.

However the relative differences between the four aid system responses were similar for the two studies and so the above mentioned transducer differences appear to offer no satisfactory explanation for the contradictory results of the laboratory and field studies.

6.3.4 Test ear differences

In the laboratory study the subject's non aided ear was plugged with an efficient earplug whereas in the field study the subject's non aided ear was not plugged.

7.0 DISCUSSION

It would appear that the differences between the results of the laboratory study and the field study may be due partly to the perhaps unrealistic laboratory tests and partly to the slight but in retrospect perhaps important differences in mean hearing loss between the groups of subjects who took part.

The lack of any conclusive relationship between audiogram shape and aid system preference in this field study is not surprising considering their subtle rather than gross variations. In addition to the fact that the experiment was not designed to discover this relationship the following points are important to note.

7.1 Suprathreshold measures

Since people listen at suprathreshold levels the subjects' most comfortable listening levels and loudness discomfort levels are important in affecting aid response preferences. Simple rules such as MCL = $\frac{1}{2}$ (ULL + H. threshold) while perhaps sometimes giving reasonable results over a group may involve large individual deviations (reference 12). The MCL and ULL pure tones were measured during the initial audiometric screening and as shown in figure 14 the MCL's appear to show a relationship to aid system preference in a similar way to that shown by the pure tone audiograms.

The relationship between comfortable listening levels, aid responses and user gain settings is of great importance in understanding aid preferences, however at present little is known about this relationship (references 13, 18).

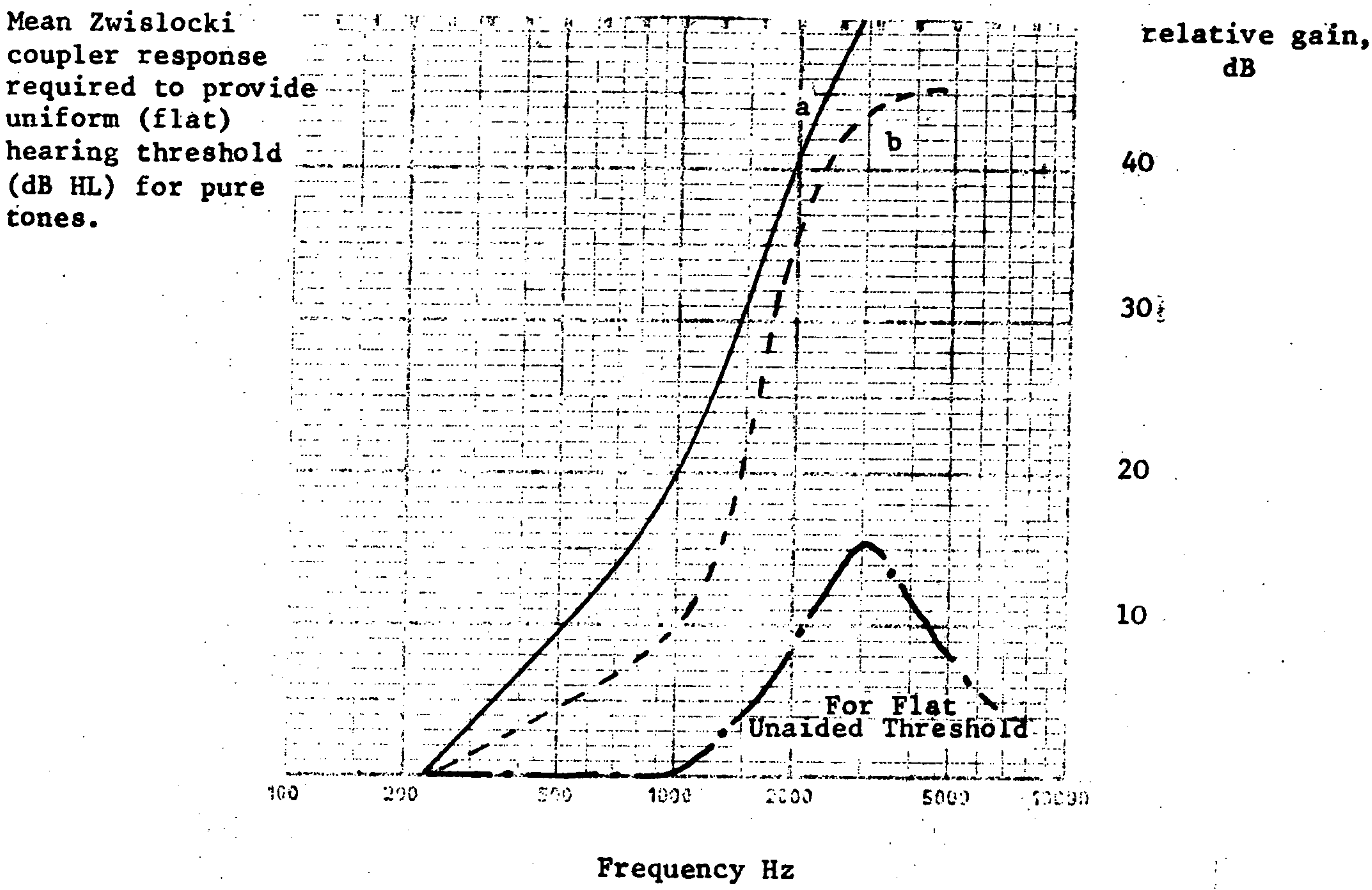
Theoretical coupler responses that would be required to provide uniform hearing thresholds (audiogram mirroring), and uniform comfortable listening levels for a simplified flat (dB HL) speech spectrum are shown in figure 16 for the two 'subject group mean audiograms' in figure 14. One can see how a lower gain is required when MCL's rather than pure tone thresholds are taken into consideration and also how deficient all the aid system responses were in their high frequency gain.

7.2 Eardrum S.P.L.

Measurements of actual or relative eardrum S.P.L's for hearing threshold and aided user gain were not made either directly by probe microphone techniques or indirectly through aided and unaided free field threshold and supra threshold measurements. The variations and their importance between individual subjects eardrum S.P.L's and the standard coupler/Kemar responses are therefore unknown.

7.3 Frequency resolution ability

The frequency resolution ability of a hearing impaired person may affect the extent to which release from masking is required and therefore aid response preferences. The importance of this factor is at present not well understood and no attempt was made to measure it in this study.



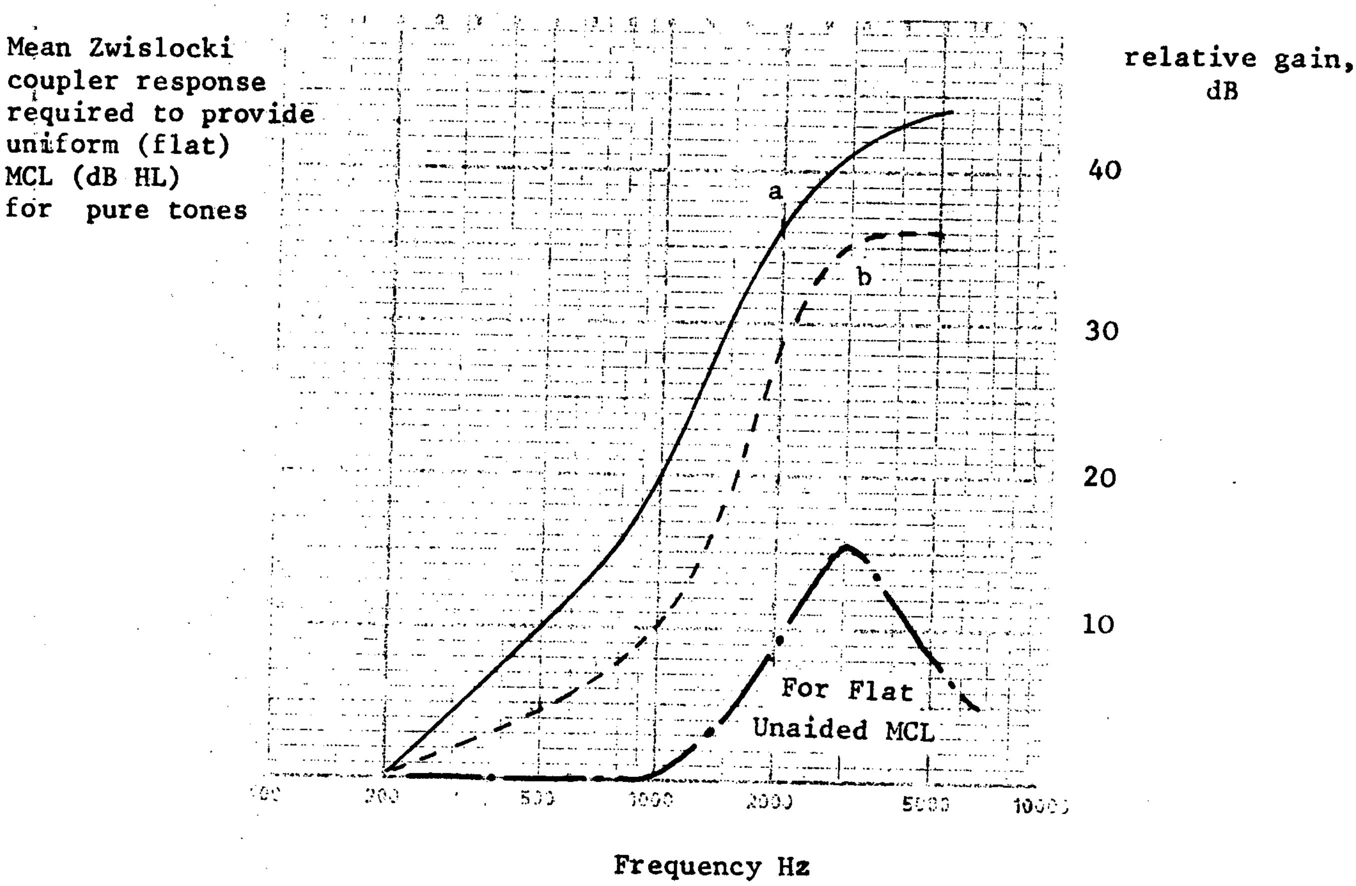


Figure 16 Mean required coupler responses for (a) people who preferred systems with normal moulds (solid lines) and (b) people who preferred systems with modified moulds (dashed lines) calculated from audiometric data (Fig. 14) + insertion loss (ie coupler gain, Fig. 1, minus insertion gain, Fig. 13)

8.0 RECOMMENDATIONS

The field study results suggest that a large number of people would benefit from:

(a) Better fitting earmoulds.

(b) The availability of higher power ear level aids.

(c) A wider choice of ear level aid and earmould systems.

Tentative relationships were found which suggest fitting guidelines for a wider choice of systems that may be of use to the N.H.S. However, as discussed in the section 7, further research (both laboratory and field studies) is required to gain more specific knowledge of the main variables affecting hearing aid performance and preferences. Only then can scientific rules or guidelines for the selection and fitting of hearing aids more consistent with subject preference and objective performance and more generally applicable than those in past or present use be developed, (ref. 13, 14, 15, 16).

The recommendation of the laboratory study that users of the NHS BE 11 with a wide range of audiometric configurations switching to the BETrTb system would be pleased with the new quality of amplified sound and would achieve greater speech reception, was not substantiated by the results of the field study. For the group of subjects involved the results indicate that such a change would leave more people displeased than pleased. This would also be the case for a switch to the BETr system, and changing to the BETb system would leave as many people pleased as would be displeased, although fewer people would then have their first choice aid. Aid preferences appeared to be related to audiometric configuration and the applicability of the above findings to the hearing aid population as a whole, is open to further discussion.

Firm recommendations regarding the immediate implementation by the DHSS of the aid and earmould modifications investigated in this study might be premature. However the results suggest that further immediate experimentation as to the optimum earmould tubing system, for both ease of manufacture and acoustic performance, would be worthwhile, as would a wider dissemination of the findings of this report.

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Subject No.

Name:

Aid X :

1. On average for how long each day do you feel you can usefully wear each aid.
(show card A)

(1) Never

(2) Only occasionally

(3) 1 - 2 hours a day

(4) 2 - 4 hours a day

(5) 4 - 8 hours a day

(6) More than 8 hours a day

Aid X : ☐

Aid Y : ☐

3. You have tried the aids for a few weeks

(a) All things considered which did you prefer?

(If preference expressed ask (b) about preferred aid)

(b) How much better did you find Aid ______?

(read out categories)

4. (a) Are these important to you

(i) Understanding what people say

(ii) The quality of the sound you hear

(Ask (b) then (c) for (i) then (ii)

(b) As to .... which aid did you find better ?

(If preference experienced ask (c))

(c) How much better did you find Aid .... ?

(read out categories)

Aided Ear	Left	Right	Either	Both

Aid Y :

2. How long did you take to get used to each aid?

(show card B)

(1) Immediately

(2) A few hours

(3) A few days

(4) A few weeks

(5) Still not used to it

Aid X : ☐

Aid Y : ☐

(a) Prefer			(b) better		
X	Y	No Preference	much better	Moderately better	slightly better

	(a) important		(b) prefer			(c) better		
	Yes	No	X	Y	No preference	much better	moderately better	slightly better
(i)								
(ii)								

5. I would now like to talk about problems you have had with either or both of the aids.

(Ask (a) thro' (e) for each problem.

(a) Have you had any problems with ....?

(3) Discomfort from the earmould

(4) Inserting the earmould

(5) Condensation affecting the sound

(6) Having to clean the earmoulds

(7) Wind noise

(8) Background noise

(9) Whistling

(10) Needing to often adjust the volume control setting

(11) Discomfort from sounds being too loud

(12) The appearance of the aid or earmould

Any other

(If yes to question (a) ask (b))

(b) With which aid/aids did you have this problem?

If no preference expressed ask (d)

(c) Do you regard it as a serious, moderate, or slight problem.

(d) Which aid was worse in this respect?

	(a) Prob.	(b), (c) Aid X			(b), (c) Aid Y			(d) Worse		
	No	Ser.	Mod.	Slight	Ser.	Mod.	Slight	X	Y	No pref.
3										
4										
5										
6										
7										
8										
9										
10										
11										
12										

6. You said that all things considered you preferred Aid ..... and found it ..... better. I am interested whether you found it ..... better in all situations or whether there were any particular situations where this was not so.

I will now describe some situations.

Ask (a) through to (f) for each situation.

(a) Has this situation occurred? (read out situation)

(13) Talking with 1 person in a quiet place (eg at home)

(14) Talking with 1 person in a noisy place (eg. on a bus, in a shop)

(15) Talking with 2/3 persons in a quiet place

(16) Talking with 2/3 persons in a noisy place

(17) At a cinema or theatre

(18) Listening to music

(19) Listening to environmental sounds (eg. birds, rain, cars, footsteps)

(20) Listening to telephone

(21) Listening to T.V. (on microphone ☐ , on loop ☐)

(22) Listening to speech in a hall (eg church or public meeting)

(23) Any other

(If yes to (a) ask (b)

(b) Is it important for you to be able to hear in this situation?

(c) Have you found you can use the aid in this situation?

(If yes to (c) ask (d)

(d) Which aid do you prefer in this situation?

(If preference expressed ask (e)

(e) How much better did you find Aid ....?

(read out categories)

(f) What was the main reason for your preference?

	(a) Occurred		(b) Import		(c) Use			(d) Pref			(e) Better			Main (f) Reason
	Yes	No	Yes	No	X	Y	No	X	Y	No pref.	much better	mod better	slightly better	
(13)														
(14)														
(15)														
(16)														
(17)														
(18)														
(19)														
(20)														
(21)														
(22)														
(23)														

7. Just to recap,

You said that all things considered you preferred Aid ....

Yes ☐ No ☐

What was the main reason you preferred it?

---

8. Could you please rate the quality of the sound for each aid on each of the following scales.

(show card C)

406.

											Aid X		Aid Y	
Clear	4	3	2	1	0	1	2	3	4	Hazy				
Harsh	4	3	2	1	0,	1	2	3	4	Mellow				
Pleasant	4	3	2	1	0	1	2	3	4	Unpleasant				
Rough	4	3	2	1	0	1	2	3	4	Smooth				

For instance on the first scale if you feel the sound is extremely clear rate it 'clear 4'

If you feel it is extremely Hazy, rate it 'Hazy 4'

If you have no feelings about whether it is hazy or clear rate it '0'

9. Could you think about how well you understand with each hearing aid?

(show card d)

9. Understand perfectly
8.
7
6.
5. Aid X : ☐
4. Aid Y : ☐
3.
2.
1.
0 Not understand at all

---

10 Considering everything, How satisfied are you with each aid?

(show card e)

9. Extremely satisfied
8.
7.
6. Aid X : ☐
5. Aid Y : ☐
4.
3.
2.
1.
0 Not at all satisfied.

---

11 What improvements would increase your satisfaction?

Anything else?

Appendix 2

MONTH'S TRIAL PROCEDURE

Name: Start: day

Finish: day

MONTH'S TRIAL HEARING AID + EARMOULD TUBE COMBINATIONS

1st Combination:

2nd Combination:

WEEKS 1 & 2 ie day to day

Wear the combination on alternate days.

ie. On the first day wear the 1st combination

On the second day wear the 2nd combination

On the third day wear the 1st combination again

On the fourth day wear the 2nd combination again

and so on for two weeks.

WEEKS 3 & 4 ie. day to day

Carry both combinations around with you and compare them in the various situations you encounter.

If you have any problems please ring MIKE WALD on extension 753 or leave a message with Mrs. Jenny Andrews on extension 2288.

(Southampton University telephone number (0703) 559122)

IMPORTANT NOTES ON THE PURPOSE OF THE TRIAL

BOTH the EARMOULD TUBING and the HEARING AID can affect the sound you hear.

We are therefore interested in the effect of the COMBINATION of the hearing aid and the earmould tubing.

It is essential for the purposes of this trial that each month you try out ONLY the combinations of hearing aid and earmould tubing allotted for that month. Over the three month trial period you will be given all the possible combinations to try out.

The combinations you are trying each month will be shown diagrammatically on the inside of your hearing aid box as well as in that month's trial procedure instructions.

Even if you decide early on in the month's trial that you prefer one aid/earmould tube combination PLEASE PERSEVERE with both combinations as you may perhaps find your first impressions altered as you get used to the aids and earmoulds and try them in various situations.

INSTRUCTIONS FOR CLEANING EARMOULDS

AND CLEARING CONDENSATION

CLEANING

The Earmoulds can be cleaned in the normal way. However it may be more difficult to clear the banded tube if it gets filled with fluid. Therefore it is probably easier to clean the earmould with a piece of dry tissue.

CONDENSATION

The earmould tubing may become partially or completely blocked with condensation. This may depend on such things as the length of time you wear the earmould, the weather conditions and the situations you encounter.

To ensure that the earmould tubing is always kept clear of condensation throughout the trial we are supplying you with:

1. An earmould cleaner
2. A Dri-Aid kit

Please use these in the following way.

LAST THING AT NIGHT

(a) Remove the Earmould/tubing from the Aid

(b) Blow the tubing clear with the earmould cleaner

(c) Place the earmoulds (not the aids) in the Dri-Aid pouch

(d) Roll down the top and snap shut.

IN THE MORNING

(a) Remove the earmoulds from the pouch

(b) Reseal the pouch

(c) Blow the tubing clear with the earmould cleaner

If the earmould tubing becomes blocked by condensation at any time, it can be blown clear with the earmould cleaner.

INSTRUCTIONS ON REACTIVATING SILICA GEL IN METAL CONTAINER

If the chemical in the window turns pink, heat the metal container in an oven at approximately 350°F until a blue colour is restored.

Allow to cool before re-using

Repeat as required