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PERCEPTION OF FILTERED SPEECH BY HEARING-IMPAIRED LISTENERS AND BY NORMALLY-HEARING LISTENERS WITH SIMULATED HEARING LOSS

City University of New York PH.D. 1982

PERCEPTION OF FILTERED SPEECH
BY HEARING-IMPAIRED LISTENERS
AND BY NORMALLY-HEARING LISTENERS
WITH SIMULATED HEARING LOSS

by

PAUL MILNER

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Abstract

PERCEPTION OF FILTERED SPEECH
BY HEARING-IMPAIRED LISTENERS
AND BY NORMALLY-HEARING LISTENERS
WITH SIMULATED HEARING LOSS

by

Paul Milner

Advisors: Prof. Louis D. Braida (MIT)
Prof. Harry Levitt (CUNY)
Prof. Irving Hochberg (CUNY)

Six subjects participated in a study to measure the perception of filtered nonsense syllables and test the ability of Articulation Theory to predict intelligibility performance. Two of the listeners had normal hearing, three had bilateral sensorineural hearing loss and one had a sensorineural loss in one ear and normal hearing in the other. Ten conditions of low-pass, high-pass and band-pass filtered speech were presented at a minimum of five different presentation levels ranging from near threshold to near discomfort level. All listeners heard the speech samples with no added background noise. In addition, the

normally-hearing subjects listened to the speech materials in a background of shaped white noise designed to simulate the hearing loss of selected hearing-impaired subjects. All listening was done in a sound-proof room with headphones.

Performance-intensity functions were determined for all the conditions and Articulation Index calculations were performed to determine the accuracy of predicted intelligibility performance versus the observed performance of the listeners.

The results of this study have shown that for all subjects, intelligibility of the materials used remained relatively high whether high-pass filtered at 700 Hz or low-pass filtered at 2800 Hz, and heard with sufficient intensity. The performance-intensity functions of the normally-hearing listeners with simulated high frequency hearing loss were similar to listeners with relatively flat sensorineural hearing loss.

Rollover in performance occurred at high presentation levels that were below reported discomfort thresholds. Articulation Theory calculations generally predicted lower intelligibility scores than were observed for most conditions and most subjects. Although spread out, A.I. values for the normally-hearing listeners when tested in quiet fell closer to a curve of expected performance than for these listeners when tested in noise and for the hearing-impaired listeners.

Errors in the predictions were generally systematic, in that observed performance for the narrowest band conditions showed the greatest difference from the predicted scores. Calculated Articulation Index values did not exhibit the expected monotonicity assumed by Articulation Theory. Methods of computing the Articulation Index appear to require revision to account for the differences between the observed data and performance predicted by Articulation Theory.

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Without question, sincere thanks and appreciation go to my long-suffering subjects who, probably, are still hearing filtered CV's in their sleep. They are Frank Gauntt, Joan Gulovsen, Tina Thompson, Jean Trelawny and finally, my parents, George and Rose Milner who, with love, warmth and kindness prepared me for this but never dreamed they would have a direct part in it.

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Paul Milner

December 1981

TABLE OF CONTENTS

ABSTRACT..3

ACKNOWLEDGEMENTS..................................6

TABLE OF CONTENTS.................................7

LIST OF TABLES...................................10

LIST OF FIGURES..................................11

CHAPTER I. INTRODUCTION..........................16

CHAPTER II. PREVIOUS STUDIES ON FILTERED SPEECH PERCEPTION..........................22

Studies of Normally-Hearing Listeners

Studies of Hearing-Impaired Listeners

CHAPTER III. ARTICULATION THEORY.................43

Previous Studies Incorporating Articulation Theory Predictions

Articulation Index Calculation Procedure

CHAPTER IV. SUBJECTS.............................71

Simulation of Hearing Loss in Normals

CHAPTER V. EXPERIMENTS...........................86

Speech Signals

Filtering of Speech Samples

Experimental Setup and Calibration

Earphone Calibration

Experimental Procedures

Presentation Conditions

CHAPTER VI. RESULTS 104

Introduction

Filtered Speech Perception - Normally-Hearing Listeners

Filtered Speech Perception - Hearing-Impaired Listeners

Normally-Hearing Listeners with Simulated Loss - Comparison to Tests in Quiet

Normally-Hearing Listeners with Simulated Loss - Comparison to Hearing-Impaired Listeners

Articulation Theory Calculations

CHAPTER VII. DISCUSSION OF RESULTS.......................176

Introduction

Filtered Speech Perception

Influence of Spectral Components Below 700 Hz

Influence of Spectral Components Above 2800 Hz

Comparison of Present Data to Previous Studies

Comparison of Normal Listeners to French and Steinberg

Other Studies of Filtered Speech Perception

Implication of Saturation and Rollover

Predictions of Rollover by Articulation Theory

CHAPTER VII. (continued)

Additivity of Articulation Index

Monotonicity of Articulation Index Functions

Summation of Percentage Correct Scores

Articulation Theory Predictions - General Considerations

CHAPTER VIII. SUMMARY AND CONCLUSIONS....................243

APPENDIX A. RMS AND PEAK LEVELS FOR TALKERS............247

APPENDIX B. MASKING NOISE FOR SIMULATED LOSS...........248

APPENDIX C. FREE FIELD CALIBRATION DATA................249

APPENDIX D. ANALYSIS OF VARIANCE.......................250

APPENDIX E. SUBJECT DATA..............................251

REFERENCES..278

LIST OF TABLES

Table 1. One-Third Octave Band Weights for Kryter (1962a) and ANSI Standard S3.5 - 1969....................47

Table 2. Percentage Correct Scores, Level for Maximum Performance and Presence of Rollover at High Intensity: Normally-Hearing Listeners in Quiet............107

Table 3. Percentage Correct Scores, Level for Maximum Performance and Presence of Rollover at High Intensity: Subject F.G. - Hearing-Impaired................115

Table 4. Percentage Correct Scores, Level for Maximum Performance and Presence of Rollover at High Intensity: Subject G.M. - Hearing-Impaired................121

Table 5. Percentage Correct Scores, Level for Maximum Performance and Presence of Rollover at High Intensity: Subject J.G. - Right Ear (Impaired)............126

Table 6. Percentage Correct Scores, Level for Maximum Performance and Presence of Rollover at High Intensity: Subject T.T. - Right Ear (Impaired)...........130

Table 7. Percentage Correct Scores, Level for Maximum Performance and Presence of Rollover at High Intensity: Normally-Hearing Listeners with Simulated Loss.133

Table 8. Component Bands and Summation Bands for Tests of Additivity of A.I. Values.........221

Table 9. Probability Analysis of Additivity Normally-Hearing Listeners in Quiet and in Noise..........................234

Table 10. Probability Analysis of Additivity Hearing-Impaired Listeners.....................235

LIST OF FIGURES

Figure 1. Relation between Articulation Index and Various Measures of Speech Intelligibility.....49

Figure 2. Audiogram - Subject F.G. 74

Figure 3. Audiogram - Subject G.M. 76

Figure 4. Audiogram - Subject T.T. 78

Figure 5. Audiogram - Subject J.G. 79

Figure 6. Audiogram - Subject R.M. 81

Figure 7. Audiogram - Subject J.T. 82

Figure 8. List of CV Syllables and Subject Instructions..87

Figure 9. FFT Frequency Response - Low-Pass Filters................................90

Figure 10. FFT Frequency Response - High-Pass Filters................................91

Figure 11. FFT Frequency Response - Band-Pass Filters................................92

Figure 12. Performance - Intensity Functions: Subject J.G. - Left Ear [Normally-Hearing]....108

Figure 13. Performance - Intensity Functions: Subject J.T. - Right Ear [Normally-Hearing]...119

Figure 14. Performance - Intensity Functions: Subject R.M. - Right Ear [Normally-Hearing]...110

Figure 15. Performance - Intensity Functions: Subject F.G. - Left Ear [Impaired]............116

Figure 16. Performance - Intensity Functions: Subject F.G. - Right Ear [Impaired]...........117

Figure 17. Performance - Intensity Functions: Subject G.M. - Left Ear [Impaired]............122

Figure 18. Performance - Intensity Functions: Subject G.M. - Right Ear [Impaired]...........123

Figure 19. Performance - Intensity Functions:
Subject J.G. - Right Ear [Impaired]............128

Figure 20. Performance - Intensity Functions:
Subject T.T. - Right Ear [Impaired]............131

Figure 21. Performance - Intensity Functions:
Subject J.G. - Left Ear
[Simulated Hearing Loss]........................134

Figure 22. Performance - Intensity Functions:
Subject J.T. - Right Ear
[Simulated Hearing Loss]........................135

Figure 23. Performance - Intensity Functions:
Subject R.M. - Right Ear
[Simulated Hearing Loss]........................136

Figure 24. Performance - Intensity Functions:
Subject J.G. - Left Ear Tested in Quiet
Compared with Left Ear Simulated Loss
Low-Pass Filtered Speech........................139

Figure 25. Performance - Intensity Functions:
Subject J.G. - Left Ear Tested in Quiet
Compared with Left Ear Simulated Loss
High-Pass Filtered Speech.......................140

Figure 26. Performance - Intensity Functions:
Subject J.G. - Left Ear Tested in Quiet
Compared with Left Ear Simulated Loss
Unfiltered and Band-Pass Filtered Speech......141

Figure 27. Performance - Intensity Functions:
Subject J.T. - Right Ear Tested in Quiet
Compared with Right Ear Simulated Loss
Low-Pass Filtered Speech........................142

Figure 28. Performance - Intensity Functions
Subject J.T. - Right Ear Tested in Quiet
Compared with Right Ear Simulated Loss
High-Pass Filtered Speech.......................143

Figure 29. Performance - Intensity Functions:
Subject J.T. - Right Ear Tested in Quiet
Compared with Right Ear Simulated Loss
Unfiltered and Band-Pass Filtered Speech......144

Figure 30. Performance - Intensity Functions:
Subject R.M. - Right Ear Tested in Quiet
Compared with Right Ear Simulated Loss
Low-Pass Filtered Speech........................145

Figure 31. Performance - Intensity Functions:
Subject R.M. - Right Ear Tested in Quiet
Compared with Right Ear Simulated Loss
High-Pass Filtered Speech......................146

Figure 32. Performance - Intensity Functions:
Subject R.M. - Right Ear Tested in Quiet
Compared with Right Ear Simulated Loss
Unfiltered and Band-Pass Filtered Speech......147

Figure 33. Performance - Intensity Functions:
Subject J.G. - Left Ear with Simulated Loss
Compared to Right Ear Actual Loss
Low-Pass Filtered Speech.......................149

Figure 34. Performance - Intensity Functions:
Subject J.G. - Left Ear with Simulated Loss
Compared to Right Ear Actual Loss
High-Pass Filtered Speech......................150

Figure 35. Performance - Intensity Functions:
Subject J.G. - Left Ear with Simulated Loss
Compared to Right Ear Actual Loss
Unfiltered and Band-Pass Filtered Speech......151

Figure 36. Performance - Intensity Functions:
Subject J.T. - Right Ear with Simulated Loss
Compared to J.G. - Right Ear Actual Loss
Low-Pass Filtered Speech.......................153

Figure 37. Performance - Intensity Functions:
Subject J.T. - Right Ear with Simulated Loss
Compared to J.G. - Right Ear Actual Loss
High-Pass Filtered Speech......................154

Figure 38. Performance - Intensity Functions:
Subject J.T. - Right Ear with Simulated Loss
Compared to J.G. - Right Ear Actual Loss
Unfiltered and Band-Pass Filtered Speech......155

Figure 39. Performance - Intensity Functions:
Subject R.M.- Right Ear with Simulated Loss
Compared to G.M. - Left Ear Actual Loss
Low-Pass Filtered Speech.......................156

Figure 40. Performance - Intensity Functions:
Subject R.M. - Right Ear with Simulated Loss
Compared to G.M. - Left Ear Actual Loss
High-Pass Filtered Speech......................157

Figure 41. Performance - Intensity Functions:
Subject R.M. - Right Ear with Simulated Loss
Compared to G.M, - Left Ear Actual Loss
Unfiltered and Band-Pass Filtered Speech......158

Figure 42. Articulation Index Functions:
Subject J.G. - Left Ear [Normally-Hearing]....161

Figure 43. Articulation Index Functions:
Subject J.T. - Right Ear [Normally-Hearing]...162

Figure 44. Articulation Index Functions:
Subject R.M. - Right Ear [Normally-Hearing]...163

Figure 45. Articulation Index Functions:
Subject F.G. - Right Ear [Impaired]...........164

Figure 46. Articulation Index Functions:
Subject F.G. - Left Ear [Impaired]............165

Figure 47. Articulation Index Functions:
Subject G.M. - Right Ear [Impaired]...........166

Figure 48. Articulation Index Functions:
Subject G.M. - Left Ear [Impaired]............167

Figure 49. Articulation Index Functions:
Subject J.G. - Right Ear [Impaired]...........168

Figure 50. Articulation Index Functions:
Subject T.T. - Right Ear [Impaired]...........169

Figure 51. Articulation Index Functions:
Subject J.G. - Left Ear
[Simulated Hearing Loss]......................170

Figure 52. Articulation Index Functions:
Subject J.T. - Right Ear
Simulated Hearing Loss].......................171

Figure 53. Articulation Index Functions:
Subject R.M. - Right Ear
[Simulated Hearing Loss]......................172

Figure 54. Performance - Intensity Functions:
Combined Results, Subjects J.G. and J.T.,
[Normally-Hearing]............................187

Figure 55. Performance - Intensity Functions:
Combined Results, Subjects J.G. and J.T.,
Compared to French and Steinberg Data,
Low-Pass Filtered Speech......................188

Figure 56. Performance - Intensity Functions:
Combined Results, Subjects J.G. and J.T., Compared to French and Steinberg Data, High-Pass Filtered Speech......................189

Figure 57. Performance - Intensity Functions:
Observed and Predicted Functions
Subject J.G. Left Ear, Normally-Hearing
Talker B, (Male)..............................216

Figure 58. Performance - Intensity Functions:
Observed and Predicted Functions
Subject J.G. Right Ear, Impaired
Talker B, (Male)..............................217

Figure 59. Performance - Intensity Functions:
Observed and Predicted Functions
Subject F.G. Left Ear, Impaired
Talker D, (Female)............................218

Figure 60. Additivity of A.I. Values:
Subject J.T. - Normally-Hearing, Tested in Quiet
Summation of LP 700, BP 700-1400, BP 1400-2800
and HP 2800 Hz Bands..........................225

Figure 61. Performance - Intensity Functions:
Subject J.T. - Normally-Hearing, Tested in Quiet
Summation of LP 700, BP 700-1400, BP 1400-2800
and HP 2800 Hz Bands..........................226

Figure 62. Additivity of A.I. Values:
Subject J.T. - Normally-Hearing, Tested in Noise
Summation of LP 700, BP 700-1400, BP 1400-2800
and HP 2800 Hz Bands..........................227

Figure 63. Performance - Intensity Functions:
Subject J.T. - Normally-Hearing, Tested in Noise
Summation of LP 700, BP 700-1400, BP 1400-2800
and HP 2800 Hz Bands..........................228

Figure 64. Additivity of A.I. Values:
Subject F.G. - Hearing-Impaired, Left Ear
Summation of LP 700, and HP 700 Hz Bands......229

Figure 65. Performance - Intensity Functions:
Subject F.G. - Hearing-Impaired, Left Ear
Summation of LP 700, and HP 700 Hz Bands......230

CHAPTER I

INTRODUCTION

A listener with a sensorineural hearing loss frequently exhibits a reduction in the ability to understand speech signals that in themselves are unaltered. Hearing aids are designed to help the individual with a hearing impairment overcome the loss in sensitivity by selectively amplifying sound as a function of frequency and improve his or her speech intelligibility. The primary goal of fitting a hearing aid is, therefore, to match speech input signals to the residual auditory function of the person with the hearing loss. However, one of the major difficulties in fitting a hearing aid to a specific hearing loss is selecting the appropriate electroacoustic characteristics of an aid that will satisfactorily accomplish that purpose. For hearing aids with linear amplification, the most important specification of performance is the functional frequency-gain characteristic of the hearing aid. Although considerable research in selecting frequency-gain characteristics has been carried out (see Braida et al., 1979), no completely satisfactory method of individual fitting of hearing aids has emerged. There are several reasons why this remains a problem.

Fitting hearing aids is frequently complicated by the fact that standard electroacoustic measurements of hearing aid performance may not accurately reflect the actual gain of the hearing aid when fitted on an individual. Hearing aid fitting usually takes place in an environment that is very different from the normal everyday circumstances in which a hearing aid is worn. Such problems in hearing aid fitting occur as well with listeners having only conductive hearing impairments. These persons, however, do not normally suffer from the distortions in the perceived speech signal experienced by listeners with sensorineural or mixed hearing loss. Today, however, a better understanding of the problems of fitting hearings aids is taking place as seen, for example, in Levitt (1978), and Studebaker and Hochberg (1980).

Another significant reason for the difficulty in fitting hearing aids is the inadequate understanding of speech perception and auditory processing in individuals with hearing impairment, especially in those with sensorineural hearing loss. For these listeners, reduced sensitivity is often accompanied by limited dynamic range and other distortions within the impaired auditory system. Selecting the optimum frequency-gain characteristic, therefore, becomes a complex problem, one not solved by merely "mirroring the audiogram" to overcome the frequency dependent loss in auditory sensitivity. A third reason for the difficulty in fitting hearing aids, and one that applies to all types of hearing impairments, is related to differences in characteristics of various hearing aids and the familiarity that any given hearing aid user has with those characteristics, i.e., the degree of training that any listener may undergo in the process of hearing aid fitting and use.

It has been suggested (e.g., Radley et al.,1947; Fletcher, 1951; Dugal et al., 1980) that the problem of selecting frequency-gain characteristics for hearing aids is somewhat analogous to the design of speech communications systems such as telephone circuits. Typically, such systems can be designed according to the predictions of Articulation Theory.(1)
(Collard, 1930; French and Steinberg, 1947; Beranek, 1947; Kryter, 1962a). Articulation Theory implies that intelligibility of speech sounds may be predicted from measurements of intensity and spectral distribution of the speech signal and the intensity and spectral distribution of any noise masking the speech. According to Articulation

---

(1) Articulation Theory here refers to concepts based on measuring the "articulation" capabilites of voice communications systems, a term first used by telephone researchers and described fully by Egan (1948). It does not apply to aspects of articulation in speech production.

Theory, a single number (in the range between 0 and 1.0) called the Articulation Index, predicts speech intelligibility for a variety of listening situations and speech materials. For listeners with normal hearing, listening to speech at normal intensity levels, intelligibility as a function of intensity level and signal-to-noise ratio usually may be predicted as well by Articulation Theory as by real-voice intelligibility measurements (Kryter, 1962b). Computation of the Articulation Index is achieved by summing the relative contibutions to the audibility of speech of a number of contiguous spectral bands over the range from 200 to 6000 Hz. The bandwidth, and, therefore, the number of bands depends on the specific method of calculation used to compute the Articulation Index (see Kryter, 1962a). The effects of altering frequency-gain characteristic also can be predicted reasonably well by Articulation Theory. A number of investigators (e. g., Radley et al., 1947; Fletcher, 1952; Wilber, 1964; Aniansson, 1974; Dugal, Braida and Durlach, 1980) applied Articulation Theory to listeners with hearing loss. In addition, Radley et al., (1947) and Dugal, Braida and Durlach (1980) studied the extent to which Articulation Theory can be used to select an optimum frequency-gain characteristic for a hearing aid.

In the form of Articulation Theory developed by Dugal, Braida and Durlach (1980) to analyze the results of a study by Skinner (1976), hearing loss is modelled as an external masking noise in the shape of the hearing loss of the the impaired ear. One way to examine the validity of this model is to examine the results of normally-hearing listeners with a simulated sensorineural loss created by adding an external masking noise to the speech signal, since Articulation Theory should predict the effect of listening in noise for these listeners as it should for listening in quiet. Therefore, in addition to evaluating listeners with sensorineural hearing loss, a third aspect of this thesis will be to evaluate normally-hearing listeners with shaped white noise added to the speech signal to create, as it were, an artificial hearing loss, similar in shape to the audiometric configurations of the hearing-impaired subjects.

Many investigators have sought to determine the importance of spectral contributions to speech understanding in normally-hearing and hearing-impaired listeners by testing them with speech signals spectrally limited by filtering or noise. The results of the early studies of this nature on normally-hearing listeners led to the formulation of Articulation Theory. While Articulation Theory has been shown to predict performance for normally-hearing and conductively-impaired listeners, the results of similar studies with listeners having sensorineural loss are less clear. This may be due, in part, to masking effects not well understood for sensorineural impairments, distortions and other non-linearities in the impaired ear, and effects that may be related to reduced tolerance levels. In many studies, only one class of hearing-impaired subjects was tested for a limited set of conditions. For example, all the subjects may have had one type of hearing loss. Measurements also have not covered a wide range of conditions and levels, nor has there been systematic and controlled training and testing of both normally-hearing and impaired listeners during the course of the study.

This thesis was designed to provide a response to some of the limitations cited above. The specific aims of this study are first, the thorough training and testing of both normally-hearing and hearing-impaired subjects with sensorineural hearing loss under precisely controlled test conditions using band-limited speech. Secondly, the significance that the spectra of the speech materials contribute will be measured by filtering the speech in various combinations of high-pass, low-pass and band-pass conditions for both the normally-hearing and the hearing-impaired listeners. The final goal of this study is to test the ability of Articulation Theory to predict speech intelligibility for these listeners and examine some of the fundamental assumptions of Articulation Theory.

CHAPTER II

PREVIOUS STUDIES ON FILTERED SPEECH PERCEPTION

Research on the perception of filtered speech has taken place since the early years of the development of voice communications systems. Many studies have been done on the perception of speech by both normally-hearing and hearing-impaired listeners. Braida et al., (1979) reviewed a large number of these, but their primary focus was on papers related to amplification and signal processing schemes for hearing aids. The present review will examine only papers that have a direct bearing on the research of this thesis. In addition, several studies have examined their results using Articulation Theory. These will be discussed in a later chapter.

Perception of filtered speech is a function of several important variables. These include the presentation level of the speech signal, the bandwidth of the filters and their cutoff frequencies, the slopes of the filter skirts, the test materials employed and the characteristics of the talkers' voices. In the papers to be discussed, however, frequently one or more of these important variables is either considered lightly or ignored altogether.

STUDIES OF NORMALLY-HEARING LISTENERS

One of the most comprehensive papers of the study of filtered speech is the report of French and Steinberg (1947), summarizing many years of research studies at Bell Telephone Laboratories. Two aspects of their report are important. The first is their insight into the nature of speech perception as signal bandwidth is reduced by low-pass or high-pass filtering. The second is the refinement of procedures to estimate the performance of listeners mathematically from measurements of the available speech spectrum, its intensity distribution and any noise simultaneously masking the speech. This latter process has been termed Articulation Theory and will be examined later in this report.

French and Steinberg described experiments in which the high-pass and low-pass cutoff frequencies of filters placed in a speech transmission circuit were systematically varied. The test materials they used were nonsense consonant-vowel-consonant (CVC) monosyllables. When the unfiltered speech power level was held fixed, performance decreased monotonically as the available bandwidth was reduced. They also observed that for any one filtered condition, performance scores increased as a function of signal intensity, reaching a maximum at a gain of 0 to 10 dB above the orthotelephonic condition. Performance usually diminished at higher intensities.

The orthotelephonic condition (Inglis, 1938) refers to the situation of normal face-to-face conversation at a distance of one meter between a normally-hearing listener and a normal talker in a quiet room. This implies a specific overall level and frequency-gain characteristic. A communications system has an orthotelephonic gain of unity (0 dB) if the levels of this condition are maintained from transmitter to receiver, independent of signal processing in the transmission channel. An orthotelephonic gain of 0 dB is assumed to refer to a free field sound pressure level of 65 dB SPL.

French and Steinberg analyzed their data with the goal of arriving at a theory that would predict the performance of telephone systems without the need for articulation or intelligibility tests. They first divided the speech spectrum into two regions, with a crossover frequency somewhere between 1500 Hz and 1900 Hz, each of which contributed equally to overall percentage intellgibility. They empirically determined 20 frequency bands over the range from 250 Hz to 7000 Hz that contributed equally and independently to percentage intelligibility. Within each band they determined the important parameters that contributed on an acoustic basis to the intelligibility of speech. Some of these parameters included the long-term spectral level of speech in the band, the threshold of audibility of speech, and the long-term spectrum levels and effective masking levels of any noise masking the speech signals. Each of the 20 bands contributed a value of 0.05 to the total "Articulation Index" whose maximum value was unity. These findings formed the basis of Articulation Theory, which will be discussed in a later chapter.

Pollack (1948) examined the effects of high-pass and low-pass filtering in normally-hearing listeners with speech partially masked by broadband noise at 82 dB SPL. Seven low-pass conditions with cutoff frequencies ranging from 425 Hz to 3950 Hz and six high-pass conditions with cutoff frequencies ranging from 350 Hz to 2375 Hz were tested. No specification of filter slopes was given other than that they were "sharp cut-off filters." He used the Harvard PB-50 monosyllabic words (Egan, 1948) for his test materials. Several important results of this study are somewhat different from the data reported by French and Steinberg. They observed for both high-pass and low-pass conditions a uniform monotonic increase in performance as available bandwidth increased along with a consistent increase in the slope of the intelligibility function. Pollack also observed this but with several important differences. First, for both high-pass and low-pass conditions, performance increased to a maximum level more rapidly, i.e., over a narrower range of intensities than for the unfiltered conditions. Secondly, for the high-pass conditions, intelligibility scores remained high as information up to 580 Hz was removed. In fact, with the high-pass filter cutoff frequency at 350 Hz, maximum scores were higher than for the unfiltered condition. Pollack also addressed the significance of band interactions. He observed, for example, that even though scores for a high-pass condition with the cutoff frequency at 2375 Hz were extremely low, when this information was added to the low-pass condition with the same cutoff frequency, greater improvements occurred in percentage correct performance than one would have expected assuming the independence of spectral contributions as described by French and Steinberg.

Miller and Nicely (1955) studied perceptual confusions of nonsense syllables in low-pass and high-pass conditions with masking noise added. Their filter cutoff rate was 24 dB per octave. Under progressively more severe high-pass conditions, they observed that error patterns were scattered and unpredictable. Under progressively more severe low-pass conditions, however, the results were more structured and showed patterns of error similar to those from uniform-noise masking at low signal-to-noise ratios. (See, for example, Miller, 1947.)

They observed that:

> ". . . high frequency components of speech are relatively weak and therefore more susceptible to masking by the uniform spectrum of noise."

Perhaps the most important outcome of their study, however, was the analysis using consonant confusion matrices of the perceptual errors made by the listeners under the various conditions of filtering. This technique is extremely powerful in developing a more detailed understanding of speech perception. The data of the present study were collected in a manner that would also permit consonant confusion analysis.

Castle (1963) studied normal listeners under several band-pass conditions using Harvard PB-50 words and a single male talker. Band center frequencies varied from 1080 Hz to 1920 Hz and band widths varied from 240 Hz to 1800 Hz.. In addition, the cutoff slopes of the band-pass filters were varied from 30 to 120 dB per octave by cascading one, two, three or four band-pass filters set to the same cutoff frequencies. He found, as expected, that as the bandwidth increased, percentage intelligibility increased.

Castle also found that the filter skirts also had a major effect on intelligibility scores. For the narrowest band conditions, with band limits of 960-1200 Hz, 1400-1680 Hz and 1800-2400 Hz, intelligibility scores were near 100%

when only one filter, with skirts of 30 dB/octave was used. When four filters were combined, producing skirts of 120 dB/octave, scores dropped to between 25% and 35%. For the wider band conditions, the slope of the filter skirts became less critical. Intelligibility scores became nearly equal for the widest band conditions, regardless of filter slope. His results clearly point out the dependence of intelligibility performance on the slope of the filter skirts. Analog filters also possess other characteristics that were not addressed by Castle or any other of the investigators studying filtered speech perception. These include in-band distortion, flatness of the passband, and frequency-dependent phase effects. In the present study, these issues are not dealt with directly. However, the use of digital filtering eliminated these specific variables. At the same time, several new ones were introduced that express some of the problems of using this alternative approach to filtering of speech. These are related to differences in the manner of specifying and generating digital filters and the processing of signals by digital filters. These issues will be dealt with later in this report.

Palva (1965) has also shown that the filter characteristics are extremely important. He examined the perception of normally-hearing listeners for band-pass filtered speech for which the high-pass and low-pass cutoff frequencies were the same. When the slope of the filters was 30 dB per octave, and the filters set to high-pass and low-pass cutoff frequencies between 1200 Hz and 2000 Hz, maximum performance averaged greater than 80 percentage correct identification. When the filter slopes were increased to 60 dB per octave, with high-pass and low-pass cutoff frequencies between 1560 Hz and 2040 Hz, performance dropped to a maximum of about 50 percent. Performance improved, as expected, as bandwidth increased for a fixed center frequency. In addition to the band-pass conditions above, Palva studied low-pass and high-pass filtering conditions as well. He observed that performance was critically dependent on vowel discrimination since performance maxima occurred when filter pass bands coincided with either first or second vowel formant regions. These results are not surprising in view of the high vowel content of the Finnish disyllabic words he used for his tests.

In a recent study by Wang, Reed and Bilger (1978), normally-hearing listeners were tested with several conditions of filtered speech to compare their performance with the results of an earlier study (Bilger and Wang, 1976) in which subjects with sensorineural hearing loss were evaluated with unfiltered speech. Their goal was to determine whether normally-hearing subjects listening to high-pass and low-pass filtering conditions would approximate the performance of the sensorineural listeners.

Speech signals used for their experiments consisted of four 48-syllable sets: two consonant-vowel (CV) sets and two vowel-consonant (VC) sets. The filter conditions were six high-pass conditions with low cutoff frequencies of 355, 710, 1400, 2000, 2800 and 4000 Hz; and five low-pass conditions with high cutoff frequencies of 2800, 1400, 1000, 710 and 500 Hz. A wide-band condition of 80-5600 Hz. was also included. Thus for the low-pass conditions, the lower limiting frequency was 80 Hz, and for the high-pass conditions, the upper limiting frequency was 5600 Hz. Filter slopes were 48 dB per octave. A reference presentation level was established at 95 dB SPL re a 1000 Hz tone for the wide-band speech and all filtered speech conditions were presented at a constant spectrum level. Thus the overall intensity level of the signals was dependent on the available bandwidth of the filtered speech.

The results of their study were initially presented as curves of percentage correct syllable identification vs. filter cutoff frequency. As expected, performance diminished as available spectrum was reduced by either high-pass or low-pass filtering. In the high-pass conditions, however, performance did not diminish until the low cutoff frequency was above 700 Hz. For the low-pass filter conditions, performance did not diminish until the high cutoff frequency was reduced to 2800 Hz.

The primary purpose of their study was to analyze patterns of consonant confusions obtained under the different filter conditions. They further analyzed their data using a procedure called sequential information analysis (SINFA). Discussion of these analyses are not relevant to the topic of this thesis; however, since similar speech materials were used for this research (CV's), a similar anaylsis using the results of this thesis is possible.

Wang, Reed and Bilger concluded that,

> "Patterns of consonant confusions generated by subjects with sensorineural hearing loss are like those generated by normal-hearing subjects in response to the appropriate instrumental distortion of speech. . . . The pattern on consonant confusions for subjects with sloping, high-frequency sensorineural hearing loss is like that for normal-hearing subjects listening to low-pass filtered speech, . . . the pattern of confusions for subjects with flat sensorineural hearing loss is like that for normal-hearing subjects listening to high-pass filtered speech."

In many of the studies cited above, the researchers found a "crossover" frequency, dividing the spectrum into regions of equal contribution to articulation or intelligibility. French and Steinberg found that point to be between 1700 and 1900 Hz. These were average values for both male and female talkers at high presentation levels. Pollack (1948) found that the crossover frequency increased from 800 Hz to 1620 Hz as presentation intensity increased. Miller (1951) reported values of 1660 Hz for male voices and 2140 Hz for female voices. In the Miller and Nicely study, however, the crossover frequency was 1550 Hz for female voices. They observed for their highly restricted test vocabulary a great dependence of the crossover frequency on ". . . different linguistic features: 450 cps for nasality, 500 cps for voicing, 750 cps for affrication, 1900 cps for place of articulation and 2200 cps for duration. What crossover point we get depends on how we load the test vocabulary with these different features."

Webster (1964) reported on a conference that attempted to determine the significant frequencies in speech perception. He specifically addressed the issue of variability of the crossover frequency. In particular, he observed that:

> ". . . noise-masked speech has a crossover or importance frequency as much as an octave lower than the crossover frequencies of filtered speech in quiet. . . . In simple terms, as listening conditions get worse (more noise or presumably, greater hearing losses), the importance frequency gets lower."

Other factors affecting the crossover frequency include talker differences and test materials. For example, whereas French and Steinberg's talkers were both male and female, Miller and Nicely used only female talkers. For Palva's male talker, the crossover frequency was a low 1100 Hz. Other than talker differences, significant differences occur in the vocabulary structures of the test materials (Hirsh, Reynolds and Joseph, 1954). French and Steinberg used consonant-vowel-consonant monosyllables, Miller and Nicely used 16 consonants followed only by the vowel /a/, and Palva's Finnish words were trochaic disyllables. For the CV and VC syllables of Wang, Reed and Bilger's study, the crossover frequency was about 2000 Hz. The variability of this crossover frequency as a function of the filter conditions, talker differences, test materials and noise masking levels implies that it may be of limited use in itself as an indicator of spectral contribution to speech intelligibility. It simply points to the spectral regions at which half the contribution to speech intelligibility lie above and half of which lie below for a specific set of measurement conditions.

STUDIES OF HEARING-IMPAIRED LISTENERS

In most of the studies discussed so far, all subjects had normal hearing and the conditions of measurement were systematically controlled. This, frequently, has not been the case in studies with hearing-impaired listeners. Another problem in nearly all studies with impaired listeners, concerns significant subject variability. One can usually base conclusions on data obtained on a relatively small number of subjects with normal hearing and have reasonable confidence that the results will hold for most of the normally-hearing population. Hearing loss, on the other hand is a highly individualized phenomenon. One cannot always assume with the same assurances for normally-hearing listeners that test results obtained for any one impaired listener will obtain for another, even though audiometric data appear similar.

Studies with hearing-impaired listeners using filtered speech usually have been conducted to determine effective frequency-gain characteristics for hearing aid fittings. The primary question concerned the degree of high frequency emphasis to include in the frequency-gain characteristic since hearing loss is usually most pronounced at the higher frequencies.

LaBenz (1953,1956) reported experiments in which both normally-hearing and hearing-impaired listeners heard low-pass and band-pass filtering conditions. He used filters whose slopes were 30 to 33 dB per octave. Test materials were phonetically balanced (PB) monosyllabic word lists. Low-pass cutoff frequencies were 500, 1000, 1500 and 2000 Hz. Band-pass conditions were 200-500 Hz, 250-750 Hz, 500-1000 Hz, 750-1500 Hz, 1500-2000 Hz and 2000-3000 Hz. For both kinds of filtering, listeners with sensorineural hearing loss had greatly reduced performance scores relative to normally-hearing and conductively-impaired listeners. However, for the low-pass conditions, performance approached that of normally-hearing listeners. The presentation levels he chose for each condition were not fixed in sound pressure level but varied in sensation level from subject to subject and from normals to impaired subjects. Thus, observed differences in performance for his subjects may be partly due to differences in sensation levels of the speech materials.

Castle (1964) studied hearing-impaired listeners using a variety of band-pass conditions. Filter slopes were 30 dB per octave, and the Harvard PB-50 words served as his test materials. Castle defined three different classes of filtering other than wideband. The first he called "matching bandwidth" and used one or more filters which selectively amplified the most sensitive frequencies in the subject's audiogram. The second he called "mirror bandwidth" condition used filters which selectively amplified regions of least sensitivity. Finally, he provided a number of conditions he called "selective narrow bandwidth" which used a variety of one or more disjoint bandpass filter configurations. In describing his data, he did not indicate whether 1, 2, 3 or 4 filters were used in combinations to vary the slopes of the filter skirts as he did in his study with normals. Some observations can be made, however.

Six of Castle's subjects showed improved performance for some of the filtered speech conditions but only two of these subjects showed improvements greater than test-retest variations. He concluded that "among individuals with sensorineural type hearing losses, the appropriateness of a given type of selective amplification cannot be readily ascertained from examining the audiogram." Unfortunately, since the conditions heard by the hearing-impaired listeners differed from those heard by the normally-hearing listeners, the results of his studies cannot be compared directly. Braida et al. (1979) further pointed out that Castle's data present an unclear picture since test-retest variations are comparable to many of the differences in scores observed for different filter combinations.

Ambrose (1972) described a study on the effects of bandwidth and harmonic distortion in both normally-hearing and hearing-impaired listeners. Three band-pass conditions were tested: 300-4800 Hz, 600-1200 Hz and 1200-2400 Hz. Filter slopes were 90 dB/octave. Distortion conditions were produced by disabling the negative feedback in a standard audio line amplifier. Distortion levels were determined by measuring total harmonic distortion of a 400 Hz and 1000 Hz tone. In addition to no measurable distortion (0%), levels of 5%, 10% and 20% total harmonic distortion were tested. In all cases but two, mean intelligibility scores decreased as the measured degree of harmonic distortion increased. The exceptions were a slight increase in performance by the sensorineural impaired listeners when distortion increased from 5% to 10% for the widest band condition (300-4800 Hz) and the 1200-2400 Hz band condition. Ambrose, however, did not indicate that these differences were significant. The differences, most likely, are not significant in view of the low scores reported for these conditions. One problem, however, is the fact that since instantaneous speech levels vary greatly, the actual amount of distortion present at any given time will also vary. As in LaBenz's study, performance for sensorineural listeners was poorer than for normally-hearing and conductively impaired listeners, independent of the degree of harmonic distortion, for the wide-band (300-4800) and high-band (1200-2400) cases. For the low band (600-1200) condition, performance was roughly the same for all listeners. The scores for the sensorineural listeners for the wide band condition were, however, significantly greater than for the two narrow band conditions. He concluded that listeners with sensorineural hearing loss cannot make use of high frequency cues and recommended amplification of frequency regions of greater sensitivity. That is, he recommended "matching" the audiogram, rather than "mirroring" the audiogram by selectively amplifying regions of reduced sensitivity. This conclusion does not clearly follow from his results, especially in view of the improved performance in the sensorineural listeners for the wide band condition. Ambrose did not test a condition combining the two narrow bands which would have demonstrated the influence of the high band on performance scores.

In a later study, Ambrose and Neal (1973) measured three band-pass conditions on normal and impaired listeners. The filters were 200-1500 (low), 500-2000 (middle) and 1000-3000 (high) Hz. Filter slopes in this study were only 24 dB/octave vs. 90 db/octave in the previous study by Ambrose. These shallow filter slopes alone can account for the high scores achieved by the normally-hearing listeners for all the filter conditions. The hearing-impaired listeners' scores, however, ranged from 2% to 90%. Test materials for this study were the CID W-22 monosyllabic words (Hirsh et al., 1952), a set known to be easier than the Harvard PB word lists. In general, the impaired group scored higher for the high band condition than for the low and middle bands. Presentation levels were not consistent in SPL across all conditions but were adjusted to 40 dB above the SRT (spondee reception threshold) unless this level was uncomfortable for the subject. The generally improved performance of the sensorineural listeners for the high band condition relative to the lower bands contradicts the implication of Ambrose's earlier study in which he claimed that listeners with high frequency sensorineural loss cannot make use of high frequency cues. However, the importance of the added information above and below the filter cutoff frequencies cannot be discounted since the filter skirts were relatively shallow.

Several studies have examined combinations of disjoint bands both in the same ear and in different ears, presenting, for example a high band to one ear and a low band to the other. Franklin (1969, 1975) studied both normally-hearing subjects and subjects with moderate to severe sensorineural hearing loss. Rosenthal, Lang and Levitt (1975) studied only normally-hearing subjects. These studies showed that perception of high frequency bands may be enhanced by the inclusion of low frequency bands at relatively low energy levels in the same or different ears. These effects may account for the good performance of the sensorineural listeners to the high band condition in the Ambrose and Neal study. Since the filter slopes were only 24 dB/octave, some low frequency energy was present. As noted earlier, Pollack (1948) observed that scores for his listeners were better for the high pass condition at high levels when information below 350 Hz was filtered out. He concluded that information below this frequency may be audible at low levels and may enhance the audibility of speech by "adding to the overall level, while at high levels they may mask the weak high frequency sounds."

Of the studies examined, those that used hearing-impaired listeners as subjects show some reasonably consistent conclusions regarding their performance. Subjects with high frequency sensorineural hearing losses listening to unfiltered speech respond in a manner roughly similar to normally hearing subjects listening to low-pass filtered speech. This was especially true for hearing-impaired subjects whose low frequency thresholds were near normal. Normally-hearing subjects listening to high-pass filtered speech respond more like subjects with flat sensorineural hearing losses, e.g., Wang, Reed and Bilger (1978).

In the papers reviewed, we have seen a varied array of experimental conditions designed to provide insight into the importance of spectral contibutions to speech understanding in both normally-hearing and hearing-impaired listeners. At the beginning of this chapter, the important variables in the perception of filtered speech were described. In spite of the many differences among the studies, it is clear that limiting the available spectral and amplitude information reduces intelligibility. In the studies cited, however, it is difficult to make direct comparisons between results obtained for normal listeners and impaired listeners. Even when the same investigator studied both normal and impaired listeners, differences existed in presentation levels, test conditions, levels of subject training, talkers and test materials.

As pointed out by Braida et al.,

> " . . . studies of normal listeners . . . have often differed from the studies of impaired listeners in that they have employed more precise stimulus control, better trained speakers, better trained listeners, and more systematic variation of both filtering condition and the intensity level at which the material is presented."

There has yet to be a controlled study which systematically tests both normally-hearing and hearing-impaired listeners under identical training and testing conditions over a wide range of filter conditions and presentation levels, which uses a consistent set of speech materials for all test conditions, which tests conditions of simulated hearing loss in normals under the same conditions, and finally, which uses the results of the study to test Articulation Theory to predict performance of the subjects.

CHAPTER III

ARTICULATION THEORY

One of the goals of this research project is to examine the accuracy with which Articulation Theory predicts the performance of subjects with sensorineural hearing loss and of normally-hearing subjects with and without simulated hearing loss. If the predictions of performance prove accurate, the theory can then be used to predict the effects of manipulating the frequency-gain characteristic for a linear amplification system to be used as a hearing aid by a listener with a hearing loss. The optimum characteristic could be specified as that which maximizes the value of the Articulation Index, given the hearing loss of the listener.

Articulation Theory states that in any frequency band, speech contributes to intelligibility only if speech energy peaks in that band are audible, above the level of any masking noise present in that band, but below discomfort thresholds. An important assumption of the theory is that the contribution of each band is independent of that of any other band. The result of calculations based on Articulation Theory is a number ranging from 0 to 1.0 called the Articulation Index (A.I.). This value represents the overall fractional part of the total reproduced usable dynamic range of speech available to a listener relative to the total audible speech range. This proportion in each band may be reduced by increasing amounts of external masking noise, by attenuation due to external filtering or other spectral shaping, and by reducing the audibility of the speech information either by insufficient intensity or by hearing loss.

Several different methods of calculating the A.I. have been described (Collard, 1930; French and Steinberg, 1947; Beranek, 1947; and Kryter, 1962a). Although Collard's procedure was the earliest reported, French and Steinberg's method is most commonly used as the basis for later improvements in A.I. calculations. They divided the frequency spectrum from 250 Hz to 7000 Hz into 20 bands, each of which was empirically determined to contribute equally to the total Articulation Index. The bands were not uniform in either bandwidth or percentage bandwidth. In each band, they determined a factor 'W', which they defined as being

> "equal to the fraction of 1/8th second time intervals in which the speech intensity in the particular band is of sufficient intensity to be heard. Stated differently, it is the fraction of these intervals in which the speech intensity in a band exceeds the intensity which corresponds to an effective sensation level of 0 dB."

In order to calculate 'W', one needed these factors: the level of the speech expressed as the long term average intensity per cycle (i.e., spectrum level), the difference in dB between the intensity "in a critical band exceeded by 1 percent of 1/8th second intervals of received speech" and the long term average intensity in the same band, the effective masking of an external noise, and the long term spectrum level of the masking noise. French and Steinberg found from the data of Dunn and White (1940) of cumulative level distributions of speech that speech became inaudible when its long term average intensity per cycle was reduced to about 30 dB below the single frequency threshold. The threshold level of speech in any frequency region, they pointed out, was "determined by the most intense sounds in that region." Speech levels greater than 30 dB above the threshold level contributed little further to intelligibility. They also determined that the difference between the level in a critical band exceeded by 1 percent of the 1/8th second intervals and the long term average intensity was about 12 dB. The value W consisted of the sum of these factors divided by the 30 dB effective dynamic range of speech. Finally, the Articulation Index was determined by summing the W's for each band and dividing by 20 (or multiplying by 0.05). The assumption of band independence was modified to account for the spread of masking of external noise as well as that of the speech signal itself.

Beranek (1947) described a simpler version of the A.I. calculation procedure. He also observed that, in general, the total useful dynamic range of speech is about 30 dB in any band. He altered the 20 bands to span frequencies from 200 Hz to 6100 Hz, stating that the range described by French and Steinberg was based on levels for combined male and female talkers. Beranek clearly stated that his range was for male voices only. Beranek assumed that the peak levels of speech fell 12 dB above the R.M.S. long term average level and minimum speech levels fell 18 dB below in each band. The A.I. was then calculated by determining the fractional part of the 30 dB dynamic range available for each of the 20 bands, multiplying by 0.05 and summing the individual contributions for the total A.I. He provided simplified graphical methods of computing the A.I. by indicating the effective threshold of speech in each of the 20 bands, the maximum discomfort levels and the long term R.M.S. levels of speech with peaks 12 dB above and minima 18 dB below the R.M.S. level.

Kryter (1962a) further refined this procedure and incorporated methods based on weighted signal-to-noise ratios in the standard one-third octave or one-octave wide bands found in commercially available sound measuring equipment. The weights were designed to incorporate the differences between the 20 bands of equal contribution and the standardized filter bandwidths. These methods were eventually adopted as an American National Standard (ANSI S3.5-1969). Two sightly different sets of weights are available, one as published by Kryter (1962a), and the other as published in the ANSI standard. Table 1 shows both sets of weights for the one-third octave method. The newer, ANSI weights give slightly greater emphasis to the regions above 1000 Hz. One-octave weights are obtained simply by summing the weights of the constituent one-third octave bands.

TABLE 1

WEIGHTS FOR ONE-THIRD OCTAVE BAND METHOD OF COMPUTING THE ARTICULATION INDEX

BAND CENTER FREQUENCY (HZ)	KRYTER (1962a)	ANSI (1969)
200	0.0003	0.0004
250	0.0007	0.0010
315	0.0010	0.0010
400	0.0016	0.0014
500	0.0017	0.0014
630	0.0017	0.0020
800	0.0027	0.0020
1000	0.0030	0.0024
1250	0.0033	0.0030
1600	0.0037	0.0037
2000	0.0036	0.0038
2500	0.0030	0.0034
3150	0.0027	0.0034
4000	0.0026	0.0024
5000	0.0017	0.0020

The relationship between A. I. and performance of normal listeners on various kinds of speech materials is shown in Figure 1 (Kryter, 1962a). Although the ANSI standard states that the methods are considered valid for only male talkers, the procedures are often used without modifications for both male and female talkers unless the specific spectra of the actual talkers being utilized are incorporated into the computations.

In addition to French and Steinberg (1947), Pollack (1948), discussed earlier, and Hirsh, Reynolds and Joseph (1954) performed Articulation Index calculations to compare the A.I. predicted scores with scores determined from normally-hearing listeners. They found, generally, that their results were consistent with the theoretical predictions. Hirsh, Reynolds and Joseph studied filtered speech, however, the main thrust of their work was a comparison of the intelligibility of different speech materials, such as syllable, words and sentences. They were not concerned specifically with the issues under investigation in this thesis. Levitt and Rabiner (1967) used A.I. methods to predict the gain in intelligibility for binaural masking experiments in which low-frequency portions of the speech spectrum were phase-shifted interaurally and presented with high-level Gaussian noise masking. They found that predictions were "fairly consistent with data obtained in an earlier experiment."

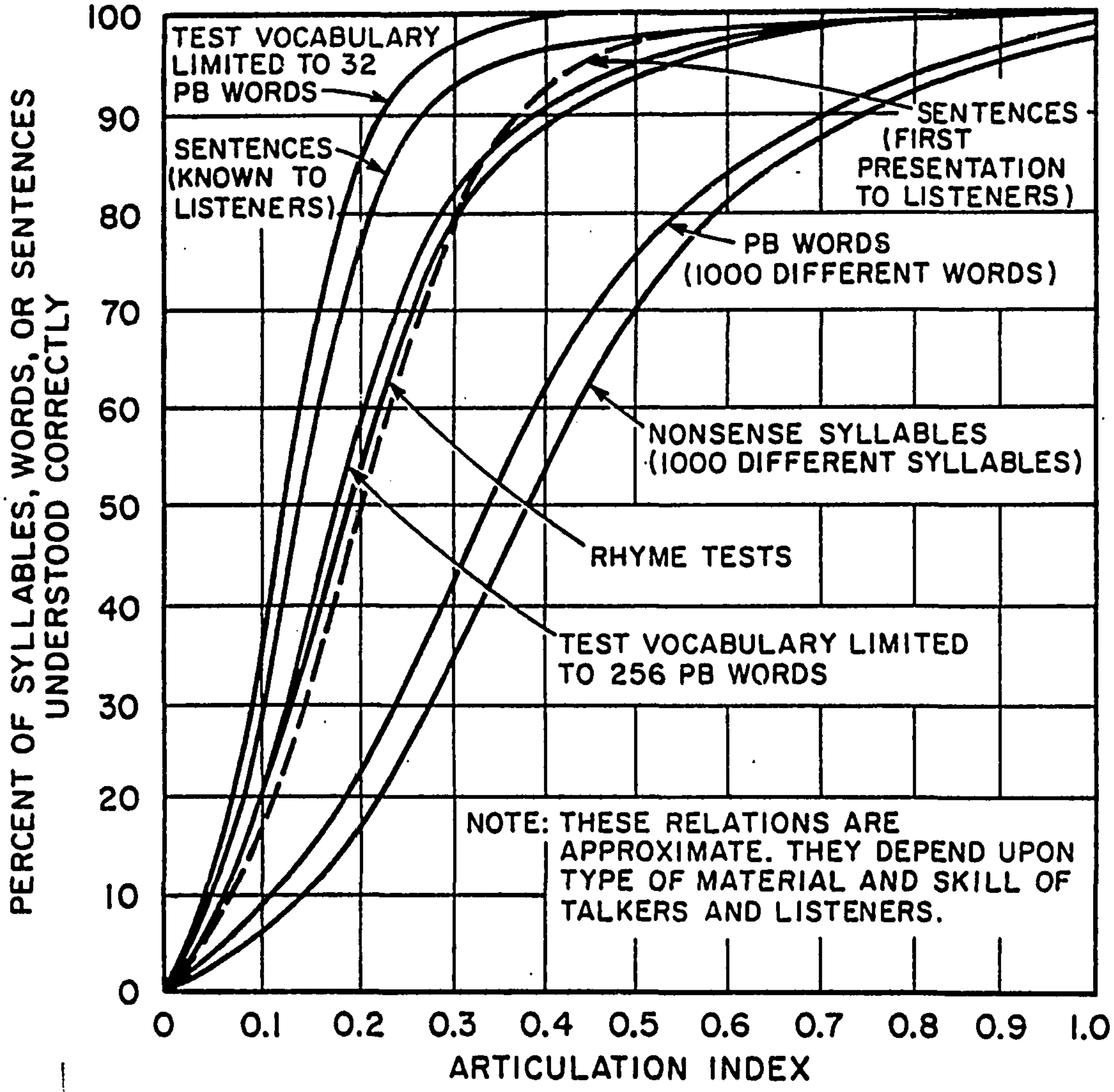


FIGURE 1

Relation between Articulation Index

and Various Measures of Speech Intelligibility

(from ANSI S3.5-1969)

In this study, the Articulation Index is calculated using a modification of the procedures suggested by both French and Steinberg, Kryter and the ANSI method. It is described in detail by Dugal, Braida and Durlach (1980). However, prior to discussing the specific aspects of this method, I will examine the results of several earlier studies that employed Articulation Theory calculations to analyze data obtained with hearing-impaired listeners.

<u>PREVIOUS STUDIES INCORPORATING ARTICULATION THEORY PREDICTIONS</u>

One of the earliest reported attempts to compare Articulation Theory calculations to results obtained with hearing-impaired listeners is that by Ackroyd (1948). Using Collard's approach to Articulation Theory, Ackroyd described an analytical procedure using a simple application of calculus of variations to arrive at predictions of the optimum frequency-gain characteristic of a hearing aid for maximum intelligibility of speech given a specific hearing loss. Although no comparative data from Radley et al., are described, he concluded the the results were "in the main confirmed by [experimental] articulation tests."

Fletcher (1952) analyzed the data of the so-called Harvard Study (Davis et al., 1947) using a version of Articulation Theory developed by Fletcher and Galt (1950) which, in part, is derived from French and Steinberg. He considered a conductive hearing loss as a frequency dependent attenuation, and a sensorineural loss as fictitious internal masking noise. He did not consider discomfort levels since the Harvard Study employed limiting in its Master Hearing Aid to avoid uncomfortably loud levels. Fletcher predicted performance scores that he felt were in good agreement with experimentally observed data. To achieve this agreement, however, he used "proficiency factors", numbers less than 1.0, to reduce the Articulation Index to account for the reduced speech discrimination abilities of listeners with sensorineural hearing loss.

The first reported study to use the now common Kryter method in a study of hearing-impaired listeners was by Wilber (1964). She used the one-octave band method to predict the speech discrimination scores of a large group of normally-hearing and hearing-impaired listeners. Speech discrimination is used here in the audiological context and is essentially synonymous with articulation testing or intelligibility testing. One crucial difference, however, exists between these terms. That difference is subject training which is not usually part of routine clinical speech discrimination procedures, but is usually assumed in articulation or intelligibility testing. Wilber's study was an attempt to provide a direct correlation between clinical measures of speech understanding and predicted performance based on A.I. calculations. Therefore, standard clinical procedures were used with her subjects and no training took place with the test materials prior to initial presentation.

Wilber used 59 subjects, twenty normally-hearing listeners, twenty unilaterally impaired listeners with conductive loss and nineteen listeners with unilateral sensorineural impairments diagnosed as Meniere's disease. The normally-hearing subjects were students ranging in age from 18 to 25 years. The conductively-impaired group ranged in age from 18 to 59 years. Within this group, bone conduction thresholds were within 20 dB of normal in the impaired ear with a difference between air conduction and bone conduction threshold (air-bone gap) of at least 10 dB. Their average hearing loss was approximately 45 dB. The sensorineurally-impaired subjects ranged in age from 24 to 74 years. Mean hearing loss for this group was approximately 40 dB; however, greater variability was present for both the normal ears and impaired ears for this group compared to the normally-hearing and conductively-impaired subjects. The shape of these subjects' audiograms were generally flat rather than sloping.

All subjects were given three different audiometric measurements to determine thresholds, one using standard clinical procedures with the usual 5 dB accuracy, one using

Bekesy swept-frequency audiometry and finally using one-octave-wide bands of white noise. In addition, speech reception thresholds (SRT) were obtained from responses to the CID spondee lists (Hirsh et al., 1952). Speech perception testing was performed with all subjects listening through TDH-39 earphones. Speech materials were drawn from different versions of a single 50-word phonetically-balanced CVC-type list developed at Northwestern University. A single male talker spoke the words and each test was presented only once to each ear without training at each of three presentation levels. The three presentation levels were +4, +8 and +12 dB relative to the measured SRT levels for each subject. For the impaired subjects, all tests were performed first with the pathological ear and then in the normally-hearing ear. This was done in an attempt to provide some control over learning factors since the same set of 50 words was used for all tests, although in different order. For the normally-hearing group, the first-tested ear was randomly assigned. No filtering or masking was used on the speech materials with the exception of contralateral masking in the normally-hearing ear of the impaired subjects if threshold differences between ears were greater than 40 dB.

Data points were averaged over all subjects within a group according to type of loss, i.e. normal, conductive and sensorineural. The average results indicated that scores improved as sensation level increased. The dependence of percentage correct scores on sensation level for the normally-hearing subjects, the normally-hearing ears of the impaired subjects and the conductively impaired ears were essentially similar and well within the deviation normally expected for untrained listeners. Scores for the sensorineural ears were generally 10 to 15 percentage points lower at the same sensation levels as compared to the others listeners; however, the slope of the function was similar. No saturation effects such as a flattening or rollover of the articulation functions was observed in the mean data presented. This, of course, does not preclude the occurrence of these effects in individual subjects. This was not reported by Wilber, however.

Articulation Index calculations were performed for all conditions tested using the one octave band method described by Kryter (1962a). They were performed for each of the three different threshold measures obtained for each subject. Since thresholds obtained with the three measures were different, the calculated A.I. values were different in each case. However, the A.I. values as a function of sensation level were similar across subject groups with the notable exception of the sensorineurally impaired ears. For these cases, all calculated A.I. values were substantially higher at each sensation level than the corresponding case for both the normally-hearing and conductively impaired ears. Wilber attributed this divergence to a large relative discrepancy between SRT and other threshold measures for the sensorineural group. She felt that since speech discrimination scores for the sensorineural listeners were poorer, SRT scores would be elevated above the commonly used three frequency (500-1000-2000 Hz) average of pure tone thresholds. Since the A.I. calculations were based on sensation levels relative to the SRT rather than the specific SPL presentation level of speech, the calculated A.I. values were probably too large for these cases. In converting sensation level to SPL in the A.I. calculation, the SPL values will be much higher relative to the pure tone thresholds and therefore result in higher A.I. values. The result of this error is higher predicted scores for the sensorineurally impaired listeners than were actually the case. These subjects did much more poorly than the A.I. would predict. On the other hand, observed scores for the normally-hearing and conductively impaired listeners were much closer to the scores predicted by Articulation Theory.

The accuracy of the predictions depended somewhat on which threshold measure was used in the calculation. Predicted performance was best using either Bekesy audiometry or filtered noise thresholds for the A.I. computations with the normally-hearing and conductively impaired group. For the sensorineural group, however, conventional audiometric thresholds predicted performance scores closer than the other methods. Since thresholds for this method were higher than the other two for this group, the resultant threshold SPL's used in the A.I. calculation had the effect of lowering the A.I. and making the predicted results closer to the observed data.

The results of Wilber's study, in general, do not provide a clear indication of the value of using Articulation Theory to predict the performance in impaired ears, at least as related to clinical measures of hearing function. Incorporating SRT levels in the calculations rather than actual presentation levels in SPL resulted in erroneously high A.I. values for the listeners with sensorineural hearing loss. The presentation levels must be independent of a psychophysical threshold measure such as the SRT that itself is defined around a highly specific vocabulary and set of measurement conditions. Furthermore, some accuracy is lost in the A.I. calculations by using the one-octave band method since variations in speech spectrum and hearing thresholds in impaired listeners may not adequately be accounted for in such a wide band. Other factors related to the training of the listeners may play a significant role in the errors since the assumptions of Articulation Theory in predicting performance are based upon extensive training with the test materials used.

A different approach to examining Articulation Index calculations and measurements of speech perception in hearing-impaired listeners was taken by Macrae and Brigden (1973). They measured the perception of "everyday sentences" (Davis and Silverman, 1970) in 309 impaired listeners in "everyday listening situations". The vast majority of these subjects (80%) had sensorineural impairments, 20% had mixed hearing losses and 2 subjects had purely conductive losses. Age range of the subjects was from 23 to 86 years. Audiometric configurations for these subjects were not indicated; however, a distribution of the three-frequency (500, 1000 and 2000 Hz) average hearing loss in the better ear was plotted. The peak of the distribution (about 58%) occurred in the "10 to 19 dB" range.

The sentence materials were presented to the subjects via loudspeaker in a background of shaped random noise designed to simulate traffic noise or with recorded cafeteria noise. Since the cafeteria noise proved a more effective masker, most tests were performed using that rather than the simulated traffic noise. The masking noises were reproduced stereophonically through loudspeakers.

Macrae and Brigden used the 20-band method of calculating the Articulation Index described by Kryter (1962a). They modified the procedure by using the audibility threshold of the listener as the minimum noise spectrum if that threshold exceed the noise spectrum or the masking spectrum in any band. They determined an Articulation Index function for the sentences, based on scoring of keywords in each sentence, and used this curve to predict intelligibility performance for their impaired listeners. In analyzing their results, they compared predicted performance using the three-frequency average hearing loss and the Articulation Index. For the measurements made in quiet and with the random noise background at speech presentation levels of 60 dB and 80 dB long term RMS level, the A.I. predictions were deemed more accurate as an indicator of performance than the three-frequency average loss. They did not use the A.I. to predict performance at 60 dB nor at 80 dB SPL when the cafeteria noise was added since they observed that their normally-hearing subjects scored much better than would be predicted on the basis of Articulation Index calculations. They did not analyze their data in terms of type or degree of hearing loss.

Another approach to measuring speech perception in "everyday listening situations", was taken by Aniansson (1974) who measured listeners with normal hearing and listeners with high frequency sensorineural hearing loss. Recordings were made in the living room of an apartment with a binaural artificial head which picked up speech played through a loudspeaker. The speech material consisted of three lists of 50 Swedish monosyllabic phonetically-balanced words. A variety of competing "everyday noises" were contrived to diminish the intelligibility of speech. These included recorded traffic noise, radio broadcasts and one, two or three simultaneous voices reading newspaper text. Subjects, both normal and impaired, listened through binaural headphones to eliminate several of the external factors that would influence performance with live-voice or loudspeaker presentations. In a preliminary study Aniansson found no statistically significant differences between direct listening and earphone listening for the speech in traffic noise conditions. He found somewhat better performance with earphone listening for the multiple competing speakers since visual distractions were removed as compared to direct listening. He therefore conducted all his experiments with binaural earphones.

Aniansson's next experiments were measurements on a large set of both normally-hearing and hearing-impaired subjects with bilateral sensorineural hearing loss, measured in quiet and in the different situations of everyday listening. The impaired subjects had varying degrees of high frequency hearing loss, presumably noise-induced since they were all workers in an automobile factory and under 50 years of age. He grouped his impaired subjects into four categories according to the approximate frequency above which the loss was 50 dB or greater. Three cutoff frequencies were used: 2000, 3000 and 4000 Hz. The fourth category was a group which he called unilateral 3000 Hz, composed of subjects whose loss in one ear was somewhat less at 3000 Hz than the 50 dB criterion but exhibited loss greater than 50 dB at 4000 Hz and above.

The results of this set of experiments show, as expected, that the lower the upper limit of hearing, the poorer the performance. This was true in all the listening situations tested. Aniansson observed, however, that the impaired listeners with hearing up to 4000 Hz. did nearly as well as the normal listeners. He also noted that the competing noise and voice situations had a greater effect in reducing performance for those listeners whose upper hearing limit was 3000 Hz or lower as compared to the normals and the listeners with hearing up to 4000 Hz.

Aniansson next attempted to assess the relative importance of various spectral regions for three of his everyday listening situations with an ultimate goal of examining spectral regions contributing to intelligibility, as did French and Steinberg and others. His experiments here consisted of measurements of seven low-pass and high-pass filtered speech conditions in normally-hearing subjects. The high-pass cutoff frequencies were 950, 1500, 2000 and 2600 Hz; the low-pass cutoff frequencies were 1400, 2300 and 3100 Hz. Filter slopes were 48 dB per octave. The three everyday listening situations he used were speech with simulated traffic noise at a signal-to-noise ratio of 0 dB, the same situation with the addition of a voice from a radio and finally with three competing voices added. The traffic noise was simulated by shaping braod-band noise with a spectrum decreasing at 5 dB per octave. It was reproduced through a loudspeaker placed about one meter outside a closed window. As expected, under all conditions, as spectral information was removed by either high-pass or low-pass filtering, intelligibility scores diminished. The highest scores were obtained for the unfiltered speech in the 0 dB traffic noise condition and the lowest were obtained for speech with the three competing speakers. He analyzed his results to look for regions of comparable performance. That is, he wanted to find out which low-pass and which high-pass conditions gave equivalent results for the three listening situations. With this in mind, he found that the low-pass at 2300 Hz and high-pass at 1500 Hz were comparable, and that low-pass at 3100 Hz and high-pass at 950 Hz were comparable. No specific benefit was evident from his results of high-pass filtering under the noisy conditions. The lowest high-pass cutoff frequency was, however, 950 Hz, and is quite high relative to the the 350 Hz high-pass cut-off used by Pollack in which intelligibility improved in noise.

The final aspect of Aniansson's work was an analysis according to Articulation Theory of a portion of the results of the filtering experiments just described and a portion of the results of the experiments with the hearing-impaired subjects. He analyzed those situations using simulated traffic noise without any additional voices - real or radio. He used the one-third octave band method as described by Kryter (1962a) and the ANSI Standard. Plotted points of A.I. versus percentage correct scores showed that predicted performance was slightly below measured for nearly all the filtered speech conditions heard by the normally-hearing listeners. For the low-pass 1400 Hz and low-pass 2600 Hz conditions, however, predicted performance was better than the observed performance. Predicted scores for the hearing-impaired and normally-hearing subjects were nearly equal to the observed scores for the higher signal-to-noise condition, whereas observed scores were better than predicted for the low signal-to-noise condition.

Aniansson concluded that, because of a wide spread of individual data, it is "difficult to predict the individual speech intelligibility exactly with the aid of tone audiometry." However, he felt that A.I. calculations may be useful in "checking the relevance of a particular result" and that "deviation from expected value might indicate retrocochlear lesions or malingering." He did not provide any clinical validation of these concepts.

It is important to contrast Aniansson's study with

Wilber's since she found that A.I. did not predict performance scores for listeners with sensorineural hearing loss. Aside from individual differences in subjects, etilogy of hearing loss (noise-induced vs. Meniere's disease) and character of loss (high frequency vs. flat), the most significant difference is that Aniansson's calculations were based on actual SPL measures of the speech and noise in the listening room and not adjusted by additional factors such as the SRT as Wilber did. In addition, sound pressures under earphones are not the same as sound pressures in a free field or semi-reverberant field such as Aniansson's living room (Villchur, 1972, Lippmann, 1981). Even though his tests were performed under earphones, he obtained measures of his signals in the actual listening room and related them to the earphone levels. Since the Articulation Theory is based on measures obtained in a sound field, approriate corrections must be applied to earphone data to allow direct comparison to the classical studies of French and Steinberg, etc. This issue will be discussed more thoroughly in Chapter V.

Most recently, Dugal (1978) and Dugal, Braida and Durlach (1980) examined how Articulation Theory can be used to predict the performance of listeners with sensorineural hearing loss. In addition, the authors tried to predict the optimum frequency-gain characteristic of a hearing aid for these impaired listeners. Using data from a study by

Skinner (1976) on listeners with noise-induced sensorineural hearing loss, the authors found that predictions with Articulation Theory were in "relatively good agreement" with Skinner's data. They attempted to determine the optimum frequency-gain characteristic that yielded the largest Articulation Index for a listener corresponding roughly to "an average of Skinner's impaired listeners." Using a "proficiency factor" to adjust the A.I. values to account for differences in subject training and abilities, a good fit was found for the predicted optimum frequency-gain characteristic with Skinner's observed optimum characteristic.

This study, however, is limited to data reported for subjects with one etiology of hearing loss. The speech material used by Skinner, and on which predicted performance is based, is a highly specialized set of 50 monosyllabic words (Pascoe, 1975) whose Articulation Index function is uncertain since it was never measured independently. Since this set of materials is small, the Articulation Index function is likely to be quite steep, going from poor performance to high performance over a narrow range of A.I. values. The resolving power of the Articulation Index to differentiate among various conditions is somewhat limited in this case since relatively low values of A.I. may be associated with very high expected intelligibility. Perhaps the most important result of the work of Dugal and Dugal

Braida and Durlach that is of direct relevance to this thesis is the specific form of Articulation Theory developed by these authors. It is this form that will be used to analyze the results of the experiments in this research.

ARTICULATION INDEX CALCULATION PROCEDURE

In this thesis, the Articulation Index is calculated using a modification of the procedures suggested by both French and Steinberg and Kryter and the ANSI Standard method.

The basic form can be expressed as:

$$A.I. = P * \sum_{i=1}^{15} BI(i) * BE(i)$$

where P = proficiency factor

BI(i) = Importance of Band i

BE(i) = Efficiency of Band i

i = Band index

The A.I. is determined using the fifteen standard one-third octave bands with center frequencies from 200 Hz to 5000 Hz. The band importance for each band is simply the weight described by Kryter (Table 1) that reflect the relationship to French and Steinberg's 20 bands of equal contribution to the Articulation Index of the 15 bands. The weights reflect both the importance per cycle and the increasing bandwidths of the one-third octave bands. In this thesis the weights described in the ANSI Standard will be used, as listed earlier in Table 1, rather than Kryter's weights, since most current A.I. calculations will be likely to use these. Differences in the results using either weights are negligible. Band Efficiency is the proportion of the speech signal in the given band that is above the threshold of audibility or masked threshold and below the discomfort level of a listener.

The proficiency factor, P, as originally described by Fletcher (1952), may reflect the amount of training of the listeners on a particular set of utterances and the precision with which the talkers recite the test materials. This constant should be independent of listening conditions. Dugal, Braida and Durlach used the proficiency factor as a fitting parameter. If one assumes that the speakers are trained and enunciate clearly, and that the listeners' are highly trained, then this factor should equal unity. It has been shown (Milner, 1973) that using trained listeners over a long period results in relatively stable performance. For the calculations of predicted performance in the present study, a proficiency factor of 1.0 was used for all listeners and for all conditions evaluated.

Since the talkers used in this study were not professional speakers or especially trained in voice techniques, proficiency factors of slightly less than 1.0 were incorporated into the Articulation Index calculations to account for their occasional poor enunciation. The following proficiency factors were used to adjust the calculated A.I. for the talkers' characteristics:

Talker D (female): 0.90
Talker P (male) : 0.90
Talker B (male) : 0.96
Talker M (female): 1.00

These values were determined empirically to provide a good fit to the data for early analytical results and were retained and, of course, used in all calculations for all conditions.

Measurements of the R.M.S. spectrum of the four voices used in this study were incorporated into the Articulation Index calculations rather than using the average speech spectrum described by Kryter. The values of the of R.M.S. levels and peak levels for the four talkers are described in Appendix A. Articulation Index calculations were done separately for each talker but in the analysis reported, the A.I. values were averaged across the four talkers to provide a more generalized result. As is seen from this data, the speech peaks were not assumed to be 12 dB above the R.M.S. level; however, maximum dynamic range in the band was held to 30 dB.

In using Articulation Theory, hearing loss will be modelled in the manner described by Dugal, Braida and Durlach. In this model, the subject's threshold elevation is treated as if caused by an internal masking noise. External masking noise, if present, is then added with appropriate corrections for the upward and downward spread of masking. The methods of Kryter (1962a) and French and Steinberg for treating the spread of masking are incorporated into the procedure. Also included is a method, based on data obtained by Bilger and Hirsh (1956), in which the downward spread of masking of a band of noise is roughly the same at all frequencies below that band, is proportional to the sensation level of the noise, and is less severe for high frequency noise than for low frequency noise. As described by Dugal, Braida and Durlach (1980), "The masked threshold in a given band is determined by computing the total equivalent power of the internal noise required to account for the quiet threshold, the effective external noise, and the equivalent noise representing the masking caused by speech in other bands. The threshold is then taken to be the total equivalent noise power."

No corrections for reverberation were incorporated since the speech materials were recorded in an anechoic environment and all listening was done with headphones. The external masking noise used to simulate hearing loss was incorporated in the calculations for the normally-hearing listeners with simulated loss, otherwise, no external noise was included in the calculations. Appendix B lists the one-third octave band levels of the noise signals used to simulate hearing loss in each of the three subjects who were tested in this manner. An additional correction (Lippmann, 1981) to the calculation was incorporated to account for the difference between earphone sound pressures and free field sound pressures at each of the one-third octave band center frequencies. This will be discussed in detail in Chapter V, relating to calibration procedures.

CHAPTER IV

SUBJECTS

Six subjects participated in this study. Three subjects had bilateral sensorineural hearing loss, one had a unilateral sensorineural loss and two subjects had normal hearing. The hearing-impaired subjects were specifically selected to have different kinds of sensorineural hearing impairments since a more general test of Articulation Theory predictions would be provided. As noted earlier, many studies have focussed on only one type or degree of hearing loss. It was felt that, in view of the large amount of data to be collected from each subject, significant initial differences between a few subjects would provide more insight into the results than a small amount of data collected on a large number of subjects. In particular, because the subjects were thoroughly trained on each of the many conditions evaluated, a consistency in the data set would result even though a small number of subjects would be used. Further, since such a small number of subjects were tested, no attempt was made to average the data over the listeners.

Prior to beginning the experimental study, all subjects, normally-hearing and hearing-impaired, underwent complete otological and audiological evaluations at either the Massachusetts Eye and Ear Infirmary or the M. I. T. Medical Department. None showed any significant otopathology other than those related to their hearing impairments. All thresholds were determined using standard clinical audiometers, and standard clinical procedures with expected variability of plus or minus 5 dB in the accuracy of thresholds. All the audiometers used were calibrated prior to performing these tests and met the requirements of ANSI Standard S3.6-1969 Specifications for Audiometers. The clinical tests performed were pure-tone air and bone conduction thresholds, speech reception thresholds (SRT), speech discrimination tests using the CID W-22 lists (Hirsh et al., 1952) and other tests as necessary to characterize the hearing loss. These included reflex and reflex decay measurements, alternate binaural loudness balance (ABLB) tests, tone decay and measures of most comfortable loudness (MCL) for speech of discomfort thresholds (UCL) for both tones and speech. UCL levels were determined by asking the subject to indicate when the particular signal became intolerably loud. In addition to these initial evaluations, repeated pure tone threshold measurements were made on each subject using a clinical audiometer in the laboratory, with the same TDH-49 earphones used for the experiments. This was done to monitor any possible change in the subject's hearing during the course of the study.

Subject F.G. was a 48 year old male with bilateral symmetric sloping high frequency loss above 1000 Hz. His audiogram is presented is Figure 2. Speech discrimination scores were good in both ears with maximum scores of 96% in both left and right ears. Discomfort thresholds in both ears were similar and found to be near those of normally-hearing listeners. F.G.'s hearing loss was noise-induced due to many years of rifle and pistol target shooting without the benefit of hearing protectors. The onset of this loss was documented in an audiogram of 1954 showing diminished hearing above 2000 Hz. This subject still participated regularly in sport shooting during the study; however, he did so wearing both ear plugs and ear muffs simultaneously. His hearing thresholds were monitored during the course of the research and no changes were observed. Both ears of F.G. were tested since there were differences in thresholds at the higher frequencies.

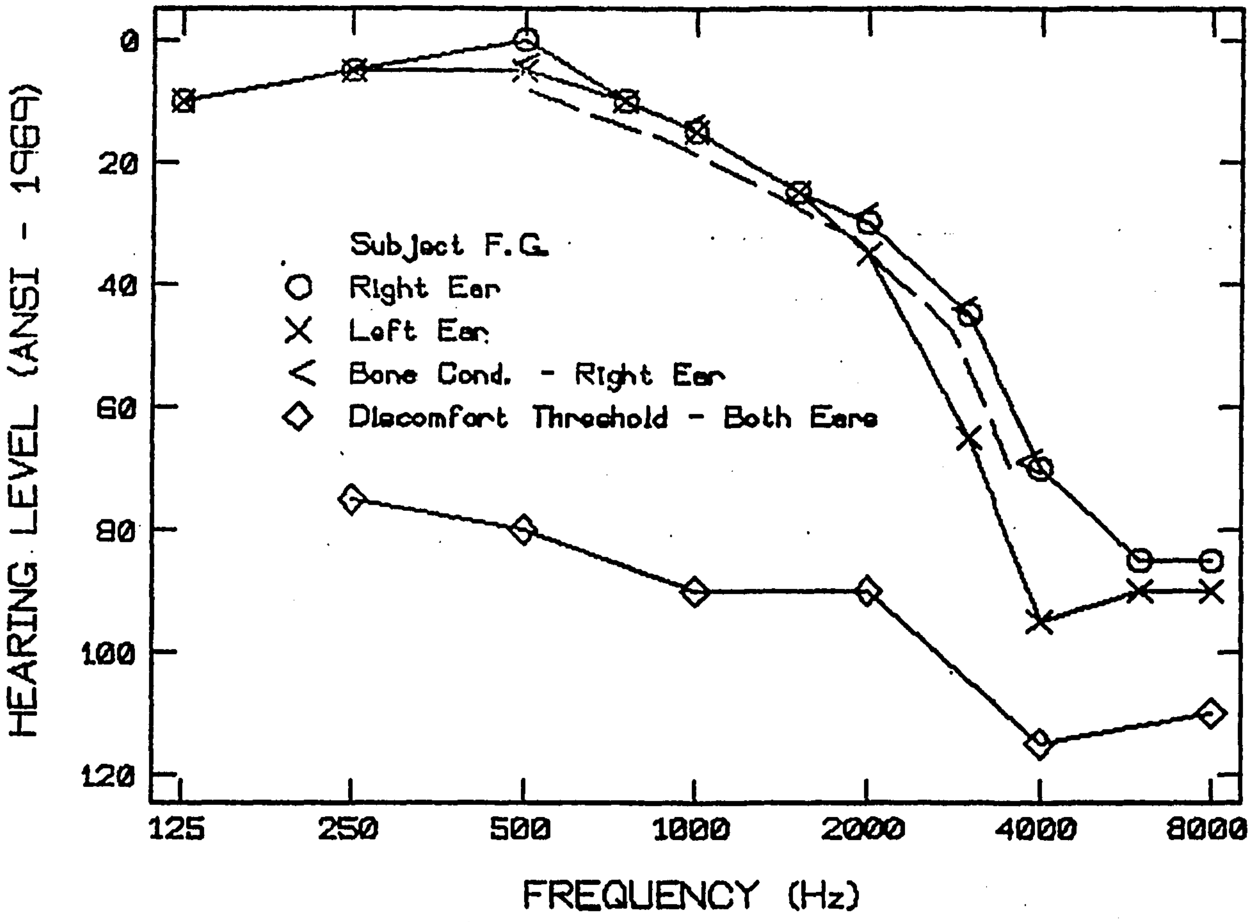


FIGURE 2

Audiogram - Subject F.G.

Subject G.M. was a 70 year old male with asymmetrical bilateral sensorineural hearing loss (Figure 3). His left ear had a moderately sloping loss, whereas his right ear had a somewhat flatter sensorineural loss. Low frequency hearing in the left ear was better than in the right ear; but high frequency hearing in the right ear was better than in the left. Speech discrimination scores were somewhat reduced in both ears with scores of 80% in his right ear and 88% in his left ear. Etiology for the hearing impairment was unclear, but there was probably a component of presbycusis in the loss. Onset of this loss had been gradual over the last ten years. There was no significant history of noise exposure nor any other otological abnormalities, although there was some familial history of hearing loss. Since notable differences in performance for this listener's two ears were observed during this study, both ears were tested independently.

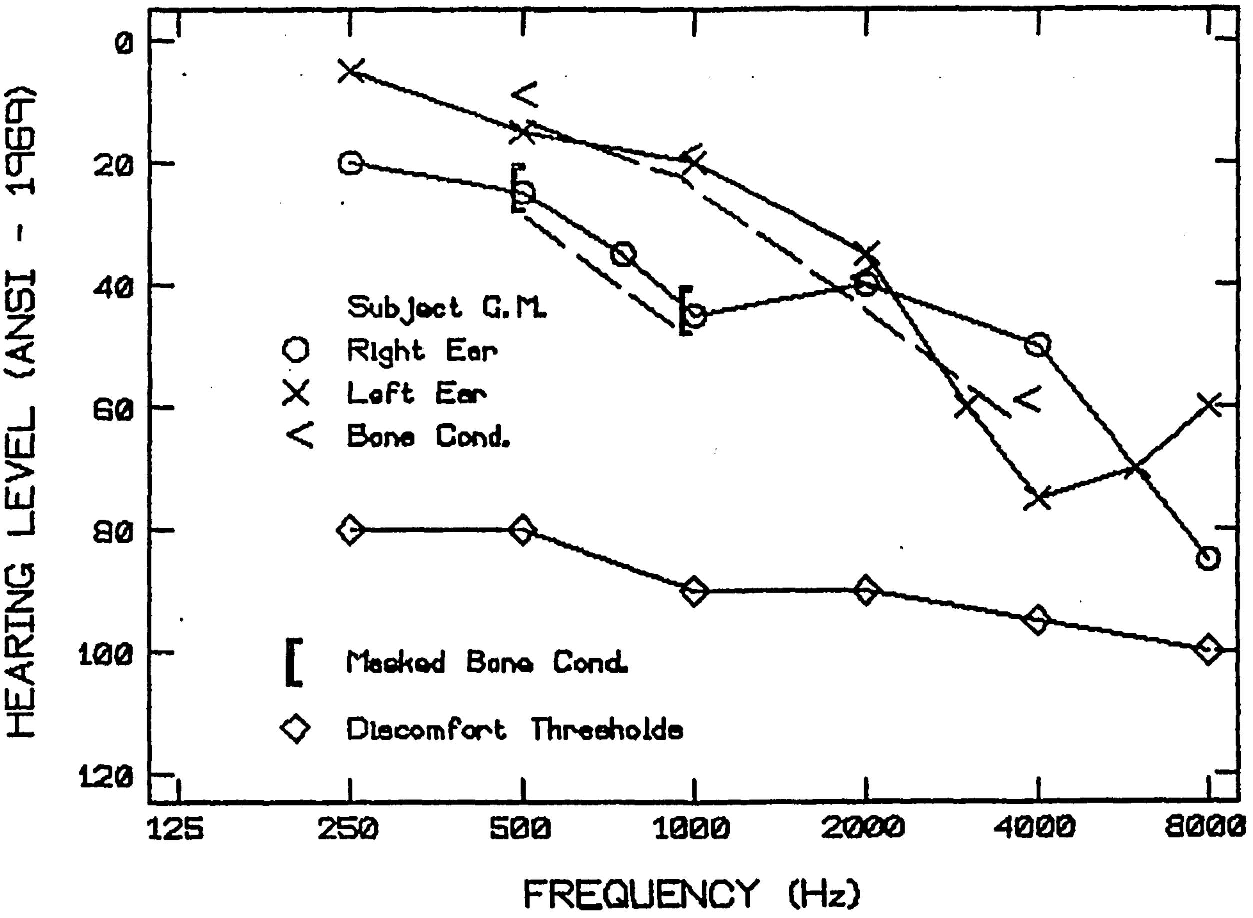


FIGURE 3

Audiogram - Subject G.M.

The third subject with bilateral sensorineural hearing impairment was T.T., a 27 year old female with severe congenital hearing loss (Figure 4). She was the only subject who regularly wore a hearing aid. Speech discrimination scores with CID W-22 words (Hirsh et al., 1952) were good (88%) in her right ear and poor (28%) in her left ear. She wore an aid only in the right ear and only the right ear was tested during this study. Discomfort levels in her right ear were found to be quite high whereas, in her left ear, thresholds for discomfort were much lower.

Subject J.G. was a 43 year old female with normal hearing in her left ear and a mild to moderate gradual sloping sensorineural hearing loss in her right ear (Figure 5). The onset of the hearing loss was sudden with no clearly defined etiology or other symptoms such as vertigo or tinnitus frequently associated with sudden hearing loss. Speech discrimination scores were 100% in her left (normal) ear and 88% in her right ear. Other audiological tests suggest a cochlear pathology. Results on Alternate Binaural Loudness Balance (ABLB) tests and reflex threshold measurements indicate recruitment and reduced dynamic range, although measured discomfort thresholds were somewhat higher than would have been expected. This listener was tested both as a hearing-impaired listener and as a normally-hearing listener.

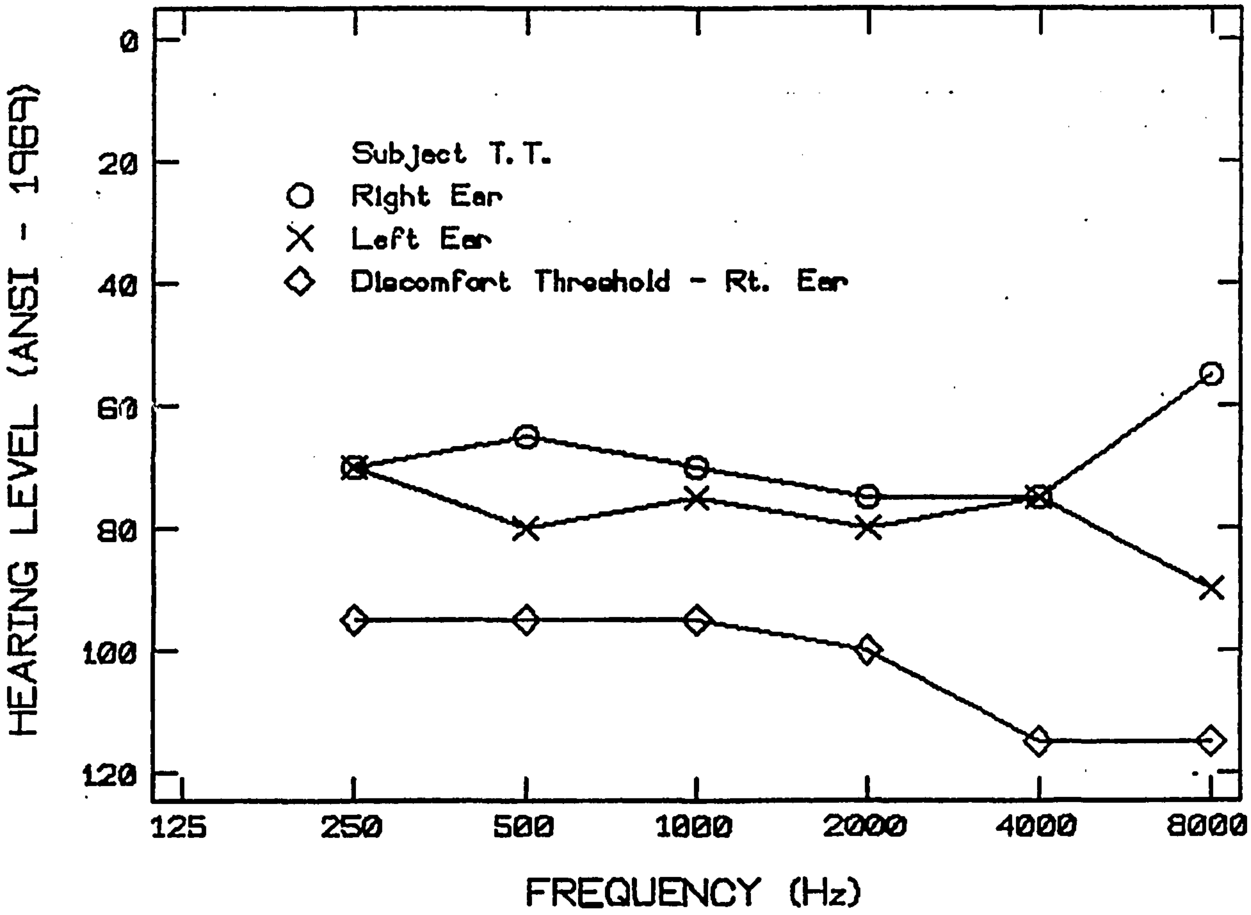


FIGURE 4

Audiogram - Subject T.T.

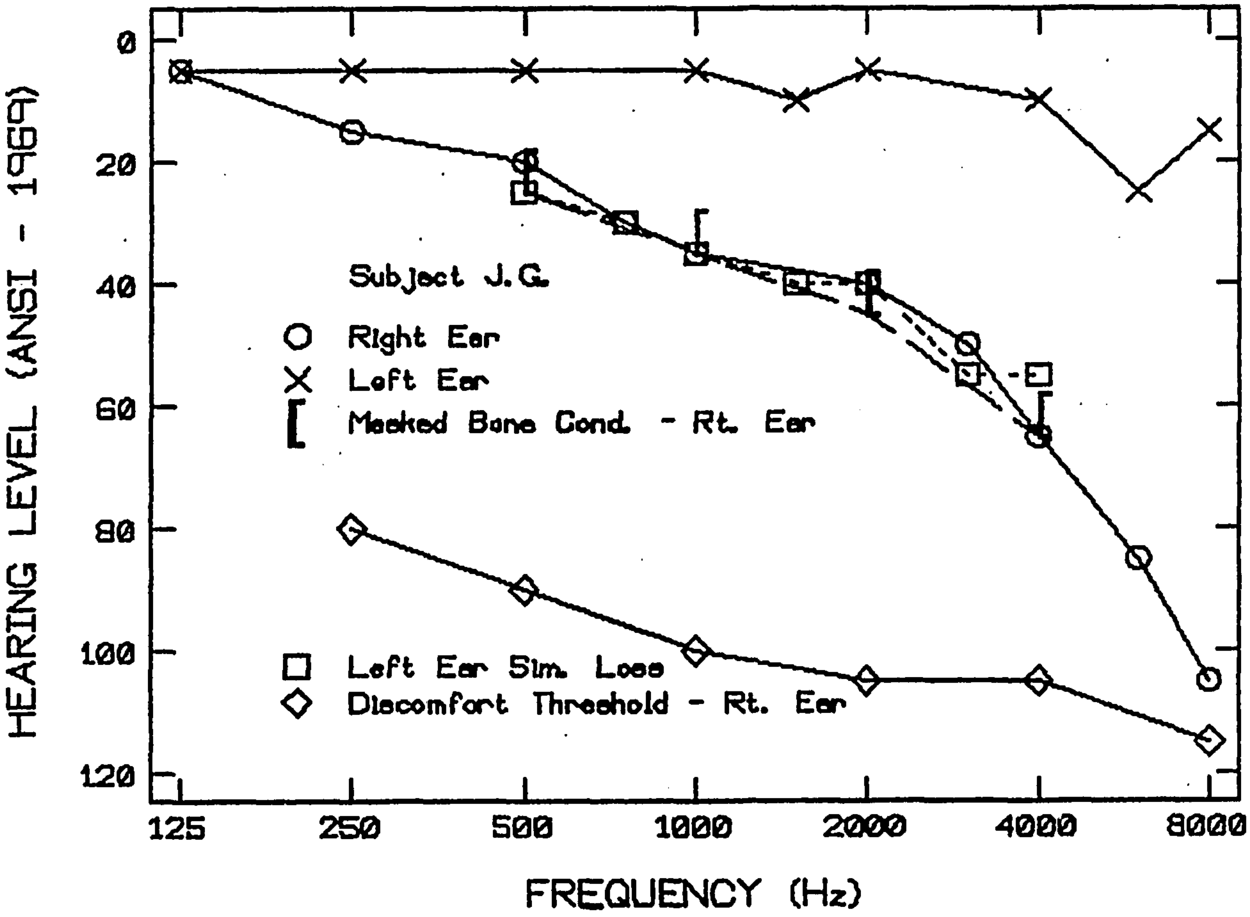


FIGURE 5

Audiogram - Subject J.G.

The two remaining subjects, R.M. and J.T. had hearing thresholds within normal limits in both ears (Figures 6 and 7) and had no history of any otological problems. Subject R.M. was a 67 year female and was tested as an age control to Subject G.M. Subject J.T. was a 26 year old female and was tested as a normally-hearing young adult control subject.

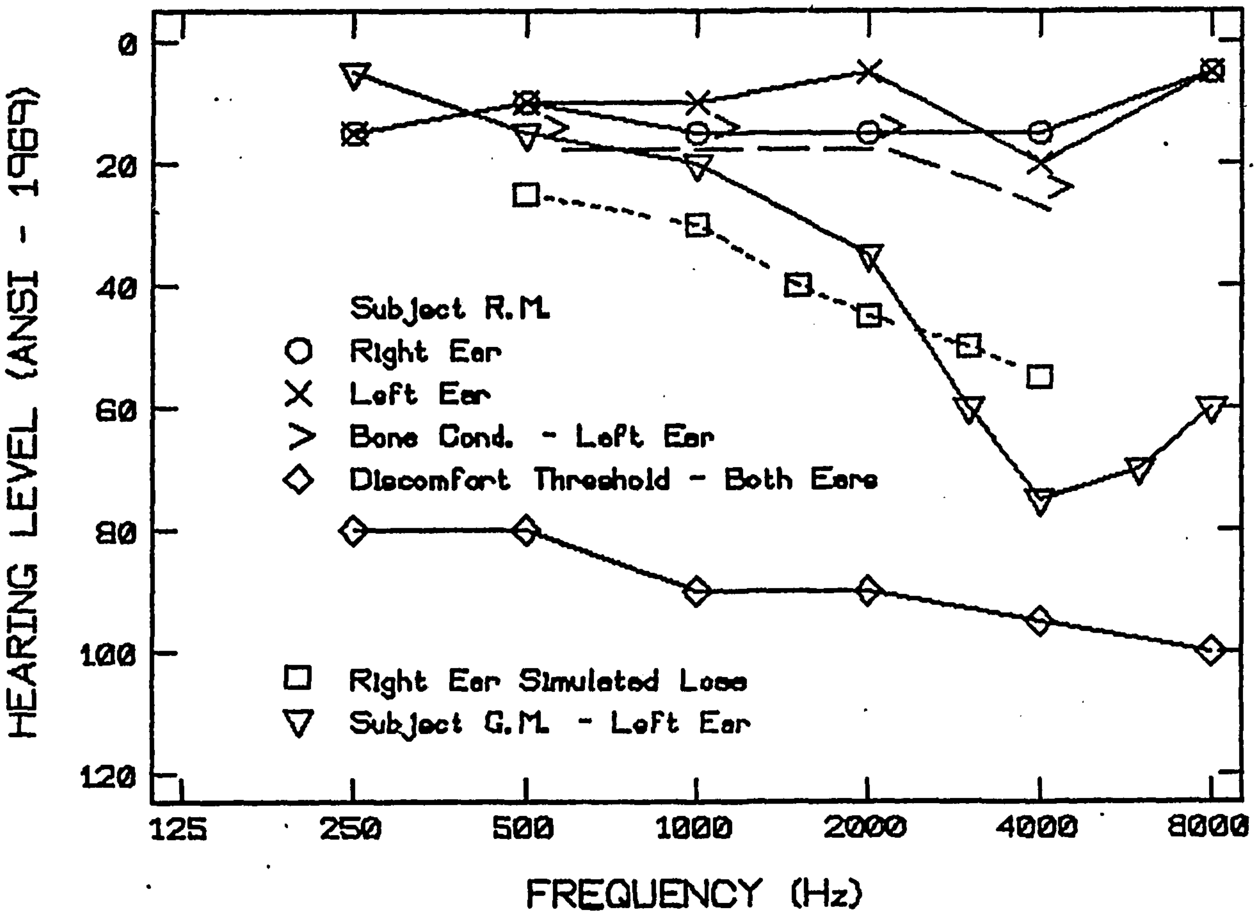


FIGURE 6

Audiogram - Subject R.M.

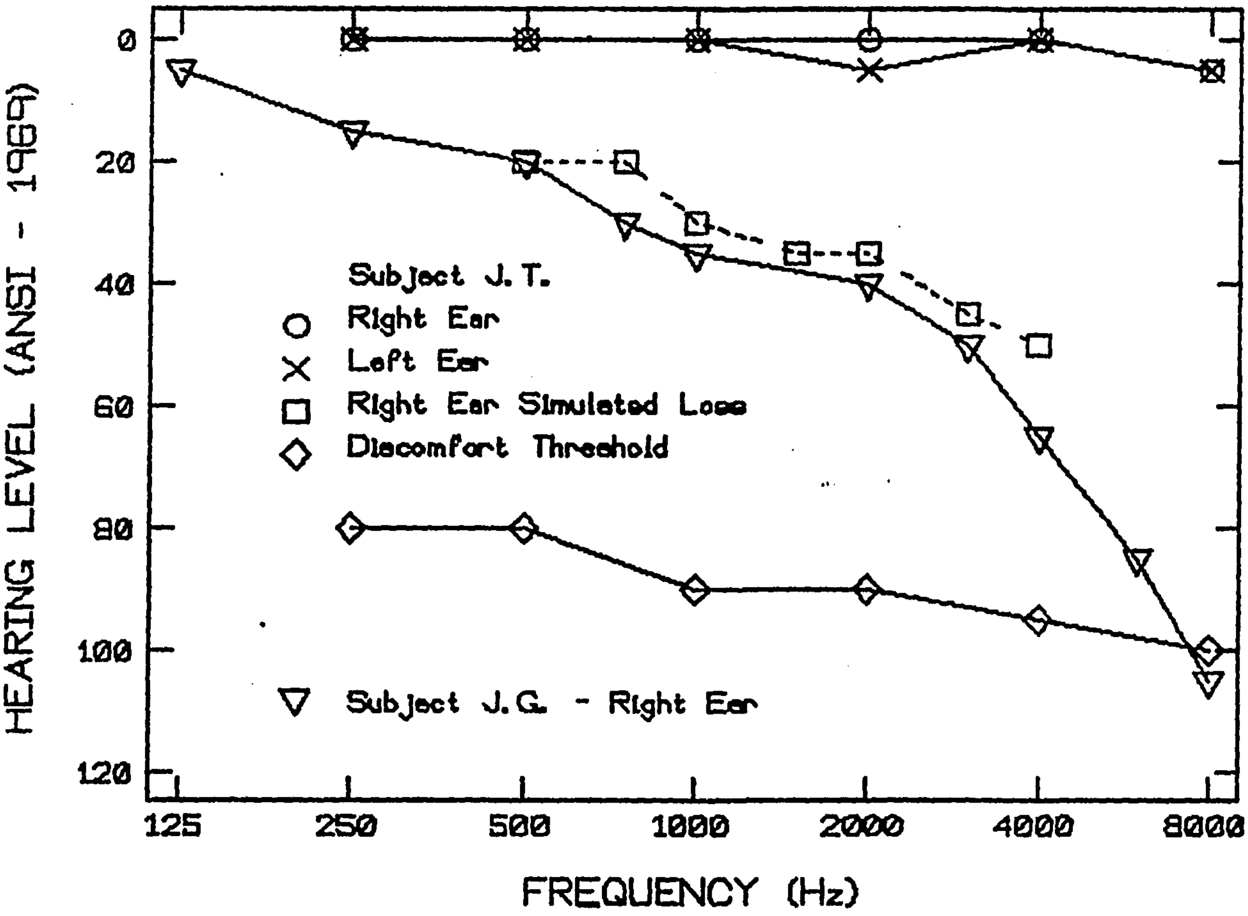


FIGURE 7

Audiogram - Subject J.T.

SIMULATION OF HEARING LOSS IN NORMALS

This study measured the performance of the listeners with normal hearing in quiet and under masking conditions simulating hearing loss. The simulated loss was created by introducing shaped wide band noise masking in the test ear at an intensity that was independent of speech presentation level. The masking signal consisted of white noise shaped by a General Radio 1/3-octave band spectrum shaper. The intensity and shape of the masker were selected to simulate the audiogram of the hearing losses in the subjects with impaired hearing. Appendix B lists the one third octave band noise levels used to create the simulated loss for each subject.

Shown in the audiograms for listeners J.G., R.M. and J.T. (Figures 5 - 7) are the masked thresholds and the hearing loss of the subjects whose impairments were simulated. For Subject J.G., with a unilateral hearing loss, the masker was chosen to create a threshold shift in her normal ear as close as possible the to the hearing loss in her impaired ear. In Subject R.M., the masker was chosen initially to simulate the hearing loss of G.M.'s right ear. However, due to much poorer performance in G.M.'s right ear as compared to his left, performance comparisons will be made with the moderate high frequency sloping loss of G.M.'s left ear. even though the simulated loss is somewhat greater than the actual below 3000 Hz. Subject J.T. had a simulated loss more closely matched to Subject J.G. Since Subject T.T.'s loss is so severe, no attempt was made to simulate it since the masking noise would have had to be very intense.

Threshold shifts for the simulated loss were measured using a sine wave generator in the laboratory's digital computer as an audiometric tone source. Due to the upper limit of the computer's digital-to-analog converter, masked thresholds only up to 4000 Hz could be obtained. Masking existed beyond that frequency, however, with the noise spectrum following the shape of the desired threshold shift. Calibration of the speech audiometer for these measurements was obtained by determining the sine wave voltage required to produce the necessary sound pressure level at each test frequency as measured with the earphone placed on a standard artificial ear. Masked and unmasked thresholds could then easily be obtained. To verify that the computer-derived thresholds were accurate, unmasked audiograms were obtained using both the computer-generated tones and speech audiometer combination and another clinical pure-tone audiometer using the same set of earphones. Differences of 5 dB or less were observed between thresholds obtained with the two methods. Masked thresholds were then obtained using the computer generated tones and the intensity and shape of the masking noise adjusted until the desired threshold shift was achieved at each test frequency. Since the speech audiometer was used to determine masked thresholds, accuracy of these values was plus or minus 2 dB, the resolution of the audiometer attenuator.

During the course of this study, pure-tone audiometric thresholds were monitored to observe any possible temporary or permanent threshold shifts especially after exposure to stimuli near discomfort levels. No changes were observed in any listener although one impaired subject, F.G., occasionally reported tinnitus following testing at a high level. If that occurred, no further testing took place in that ear on that day.

CHAPTER V

EXPERIMENTS

SPEECH SIGNALS

For this study, the primary source of speech signals for all experiments were original natural voice recordings of 72 consonant-vowel (CV) syllables recorded in an anechoic environment by each of four different talkers - two male and two female. These recordings were already available since they had been created for earlier studies in the laboratory. The 72 CV's consisted of all combinations of 24 initial consonants and the three vowels /a/, /i/ and /u/ (as in bah, bee, boo). The full set is shown in Figure 8, along with the instructions to the subject. (The voiced consonant "th" was orthographically represented as XH to permit direct entry into the computer during testing.) Each utterance was spoken three times by each talker, thereby producing a total of 864 utterances for the entire set of recorded CV's.

INSTRUCTIONS FOR CONSONANT IDENTIFICATION EXPERIMENT

1. TYPE THE NAME OF THE SOUND YOU HEAR AND PRESS "RETURN."
2. IF YOU MAKE AN ERROR, PRESS "DELETE" AS MANY TIMES AS NECESSARY TO ERASE THE INCORRECT LETTERS.
3. YOU MAY TAKE A BREAK FROM TESTING AT ANY TIME AND LATER RESUME THE EXPERIMENT BY TYPING "RETURN."
4. THE FOLLOWING LIST OF SYLLABLES ARE THE ONLY ONES RECOGNIZED BY THE COMPUTER. IF YOU TYPE ANYTHING ELSE AND PRESS "RETURN" THE COMPUTER WILL TYPE "?" AND GIVE YOU ANOTHER CHANCE.

BA (AS IN BOX)	BI (AS IN BEE)	BU (AS IN BOOT)
CHA (AS IN CHOP)	CHI (AS IN CHEAP)	CHU (AS IN CHOOSE)
DA	DI	DU
FA	FI	FU
GA (AS IN GAP)	GI	GU
HA	HI	HU
JA (AS IN JACK)	JI	JU
KA	KI	KU
LA	LI	LU
MA	MI	MU
NA	NI	NU
PA	PI	PU
RA	RI	RU
SA	SI	SU
SHA	SHI	SHU
TA	TI	TU
THA (AS IN THIN)	THI	THU
VA	VI	VU
WA (AS IN WALL)	WI	WU
WHA (AS IN WHAT)	WHI	WHU
XHA (AS IN THAT)	XHI	XHU
YA	YI	YU
ZA	ZI	ZU
ZHA (AS IN AZURE)	ZHI	ZHU

FIGURE 8

List of CV Syllables and

Subject Instructions

The recorded syllables were low-pass filtered at 4500 Hz and converted to 12 bit digital samples at a sampling rate of 10,000 Hz. All the digitized speech waveforms were normalized to equal RMS (root-mean-square) levels relative to the vowel intensities and stored as waveform files on a large disk memory accessed by the laboratory's Digital Equipment Corp. PDP-11/55 computer system.

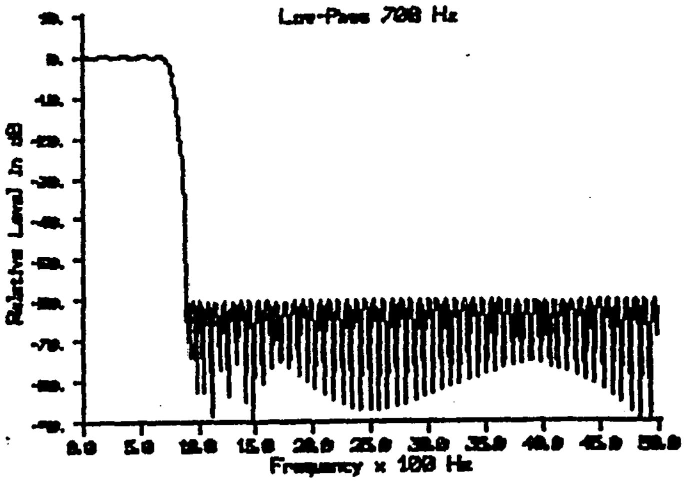
FILTERING OF SPEECH SAMPLES

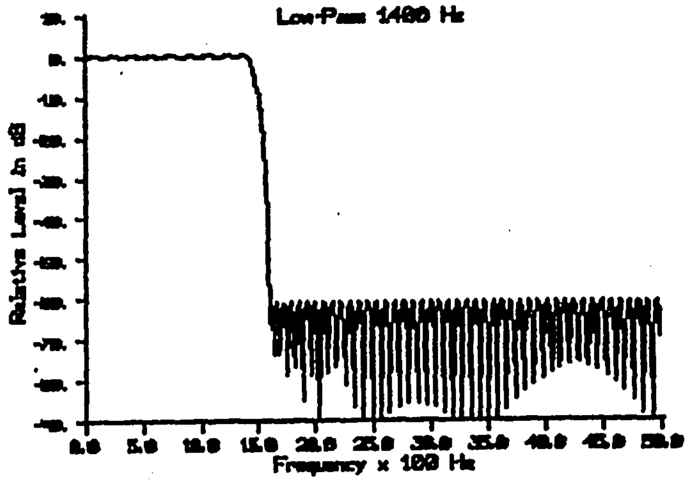
The basic set of 864 CV waveforms comprised the "unfiltered" condition. However, the total bandwidth was restricted to 4500 Hz resulting from the use of anti-aliasing filters required by the computer's analog-to-digital and digital-to-analog conversion. Nine additional conditions of filtering were studied:

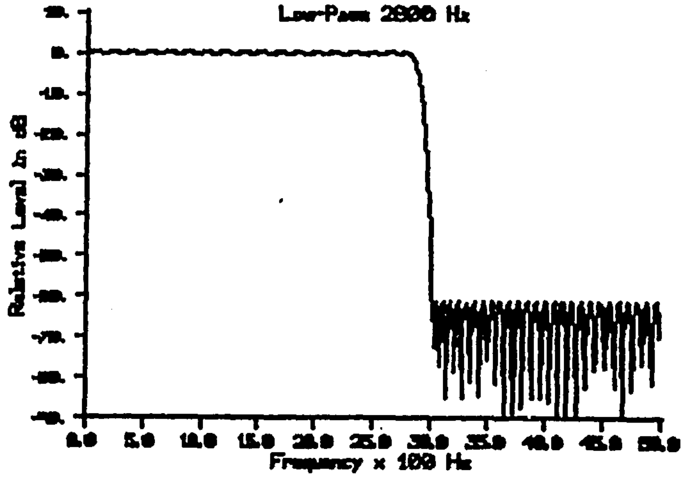
1. Low-pass filtered at 700 Hz
2. Low-pass filtered at 1400 Hz
3. Low-pass filtered at 2800 Hz
4. High-pass filtered at 700 Hz
5. High-pass filtered at 1400 Hz
6. High-pass filtered at 2800 Hz
7. Band-pass filtered from 700 to 1400 Hz
8. Band-pass filtered from 1400 to 2800 Hz
9. Band-pass filtered from 700 to 2800 Hz

Filtering was accomplished by using digital filtering techniques on the stored CV waveforms. For each of the nine conditions listed above, all 864 waveforms were processed using a convolution process with digital finite impulse response filters designed with a computer program described by McClellan, Parks and Rabiner (1973). Each filtered condition, therefore, produced a new set of 864 waveforms.

Digital filtering techniques were used for several reasons. One is the simplicity of computer signal processing. In addition, digital filters have extremely sharp slopes and no phase distortion. Figures 9, 10 and 11 show the characteristics of the nine filters obtained using a Fast Fourier Transform technique. All filters have a uniform transition region of 200 Hz and a stop band attenuation of at least 60 dB. This finite attenuation meant that unlike analog filters which attenuate progressively beyond the cutoff frequency, there is information present in the stop-band region, albeit, in this case attenuated by 60 dB. At high presentation levels, this information might become audible, especially for relatively intense low frequency vowel energy.



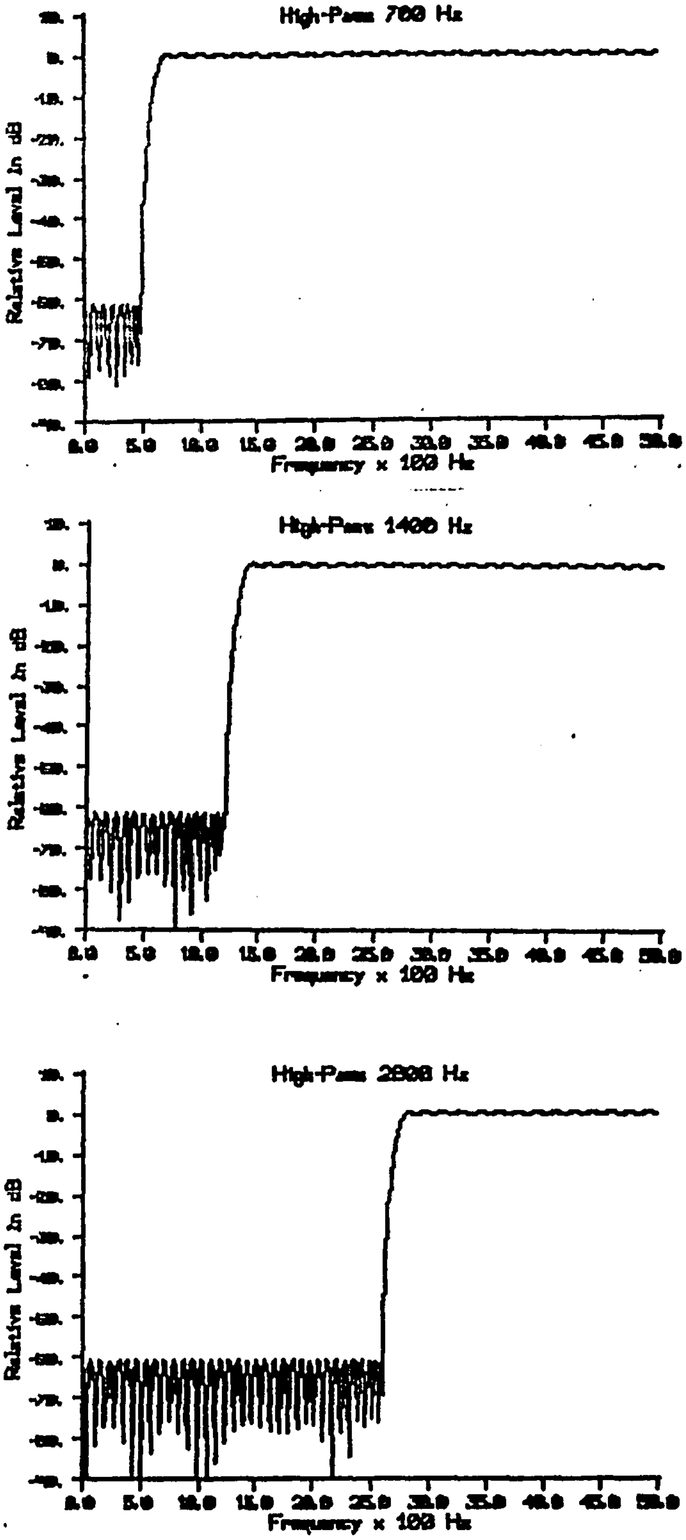




<u>FIGURE 9</u>

FFT Frequency Response

Low-Pass Filters



<u>FIGURE 10</u>

FFT Frequency Response

High-Pass Filters

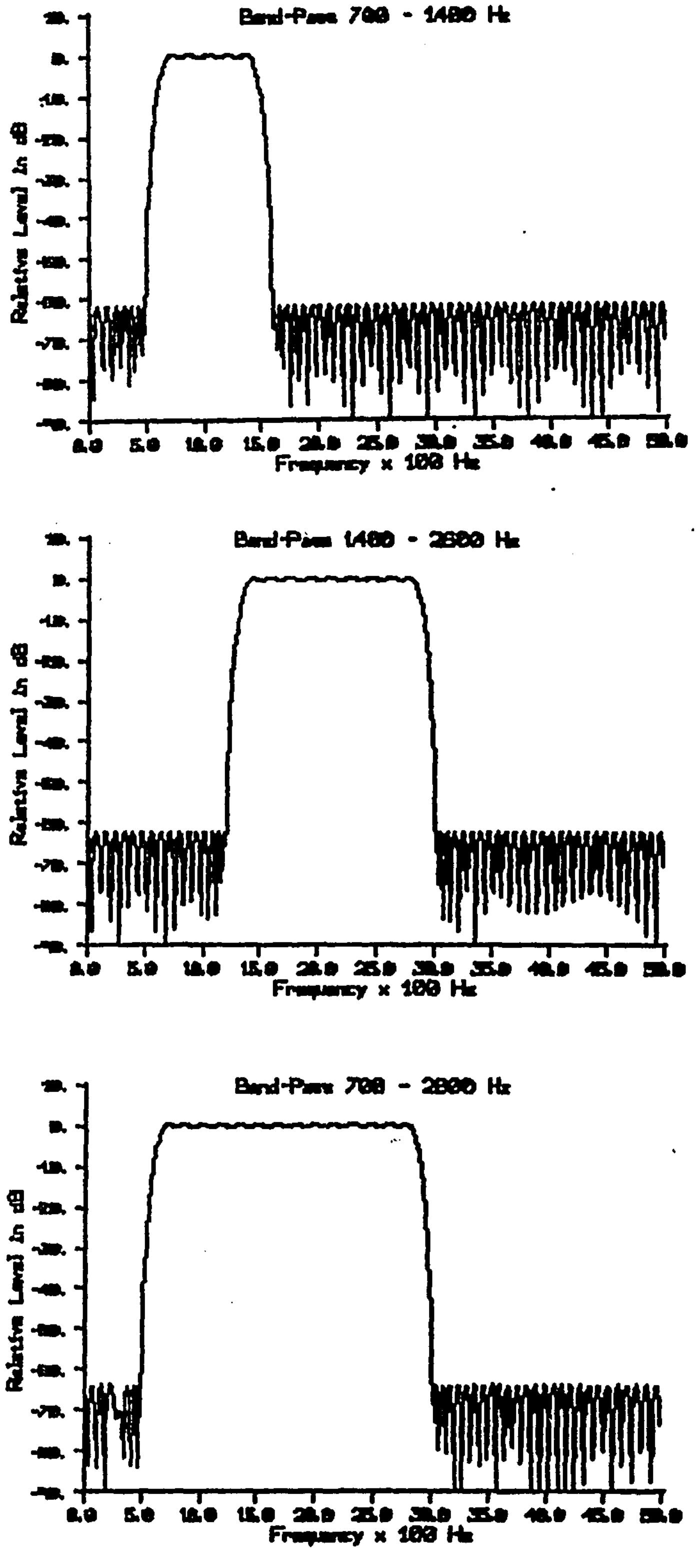


FIGURE 11

FFT Frequency Response

Band-Pass Filters

For every condition, each subject heard exactly the same set of utterances. Since the CV's were stored on a computer disk, the experimenter had direct and immediate random access to each utterance stored on that disk. This capability permitted the use of a powerful training technique that will be described later.

EXPERIMENTAL SETUP AND CALIBRATION

Using computer signal processing for all speech samples permitted an extremely simple experimental setup. The output of the computer digital-to-analog converter was connected directly to the input of a Grason-Stadler Model 162 speech audiometer. A matched pair of Telephonics TDH-49 earphones with "Zwislocki" type circumaural cushions were used for listening by all subjects. Calibration of the audiometer consisted simply of verifying that the output was free of significant noise and distortion and that the "gain" was accurately 20 dB, corresponding to the standard hearing threshold level for spondees. Calibration measurements were made in accordance with the ANSI Standard S3.6-1969 Specifications for Audiometers. Repeated calibration checks during the course of this study showed no changes.

Calibration of the speech levels was determined by playing the unfiltered speech condition and adjusting the input calibration level of the audiometer so that "frequent peaks" of the unfiltered CV's read 0 VU, implying that the gain of the audiometer is exactly 20 dB relative to the level indicated on the main attenuator. A precise sinewave voltage at 1000 Hz from the computer's sinewave generator was then determined that would cause the VU meter to read 0 VU. That sinewave then became the reference signal for regular calibration adjustments to the audiometer.

EARPHONE CALIBRATION

Articulation Theory is based on measurements of speech and noise signals in free acoustic field conditions. Thus, to perform A.I. calculations and permit direct comparison of the data of this study to other studies involving A.I. calculations, it was necessary to determine the equivalent free field sound pressure levels for the signals presented to the subjects under earphones. Lippmann (1981) measured the equivalent free-field sound pressure on a special set of earphones (Telephonics Model 556, Villchur, 1970), designed to reduce the variability of the standard audiometric TDH-39/MX41-AR (or TDH-49) earphone/cushion combination. For the present study, Zwislocki-type circumaural cushions were used rather than the supra-aural MX41-AR. However, it was still necessary to determine the equivalent free field calibration of these earphones.

In order to obtain the free field calibration of the

TDH-49 headphones used in this study, pure-tone auditory thresholds were obtained with these headphones and the Telephonics 556 set. Bekesy continuous swept frequency thresholds were determined for both the right and left earphones of each set using a Grason-Stadler Type E-800 Bekesy audiometer. Since relative rather that absolute threshold differences were of interest, calibration of the audiometer was unnecessary. Four clinical audiologists with normal hearing were used as listeners to reduce the amount of training and familiarization required to perform the measurements. Appendix C shows the table of differences in thresholds between the two sets of earphones, the free field correction from Lippmann and the total correction applied to the A.I. calculation procedures at each of the one third octave band center frequencies. The overall level difference between free-field SPL and coupler SPL was determined by comparing the unfiltered speech level as read on the speech audiometer VU meter to the level when the same signals were passed through a one-third octave band equalizer which had been adjusted to duplicate the characteristic of the free-field corrections. The overall level difference was found to be 6 dB and, therefore, a correction of -6 dB (1) was applied to the presentation levels in all calculations of A.I. and shown in the P.I. function plots.

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(1) An additional correction of -6 dB was required for all the filtered speech samples since the computer program which filters the waveforms scaled all the input waveforms by 0.5 to avoid overloading data buffers during processing.

Presentation levels will be reported in dB free field Sound Pressure Level (SPL), although during the experiments, presentation levels were read directly in dB Hearing Level (re 20 dB SPL) on the audiometer dial. All filtered speech conditions were presented at hearing levels relative to the unfiltered condition. Since no other compensations in overall system gain were made for the filtered CV's, presentation levels for the different conditions and subjects reflect actual differences in the levels of the filtered speech samples and absolute differences in the auditory sensitivites of the subjects for each condition. All filtered CV utterances were analyzed for absolute levels of each syllable relative to the unfiltered CV. No change in the relative spectral levels occurred as a result of the digital filtering process.

EXPERIMENTAL PROCEDURES

Each subject was tested individually while seated in front of a computer video terminal in a sound proof room. Two types of experimental procedures were used - a training procedure and an absolute identification procedure. For both types of experiments, the computer selected an utterance at random, from a predetermined list of utterances and played out the waveform to the listener. The listener then typed into the computer one of 72 possible responses representing the syllable heard. If the response was not one of the 72 permissible choices, the computer asked again for a response to be typed in. The response was saved in a data file for future analysis. A complete list of all utterances presented during an experiment was contained in the file along with the subject's respones and the cumulative score in percentage correct. An important feature of this data file is the ability to analyze error responses on the data similar to the analyses of Miller and Nicely (1955) and Wang, Reed and Bilger (1978) During the course of an experiment, the experimenter could monitor the subject's progress on another computer terminal observing either the entire response file or just the most recent trial along with a graphic display of the subject's cumulative performance since the beginning of the experiment.

The training experiment was designed to provide rapid and effective learning of the discriminations required by the subject through the use of immediate feedback when errors were committed (Miller et al., 1975). If the subject typed in a correct response, a new sample was played. The computer maintained an internal count of the tokens correctly identified and the cumulative number of correct responses. These tokens were then given a lower probablilty of selection than the incorrectly identified items so that the subject would have more practice on the difficult items. As a motivation to the subject, the "correct count" was displayed on the subject's terminal.

If the subject typed in an incorrect response, the subject heard a "beep" tone informing him that an error had occurred. The computer then displayed on the terminal both the response CV typed in by the subject and the signal CV. Simultaneously, the computer alternately played to the subject the signal CV and the response CV spoken by the same talker while the terminal's cursor indicated the utterance being played. The subject set his or her own pace in the experiment since the computer would not present a new signal or initiate a feedback sequence until the RETURN key was typed on the terminal. Since this was a training experiment, no fixed number of trials occurred. The computer continued to play samples until expressly stopped by the experimenter. This usually took place after 40 to 50 minutes of listening or after about 500 trials were completed, whichever came sooner. This was done to avoid subject fatigue. A training experiment took place with each subject at the beginning of every day of testing and when a new condition was tested on any given day.

In the identification experiments, the CV's were played to the subject at random with equal probability for the unplayed signals from the entire set of utterances. No feedback was given and a new trial was initiated as soon as the response to a given trial was entered. For these experiments, each token was played once to the subject and the experiment was concluded when the list was exhausted. The subject had the option in both the identification and training experiment to simply press the RETURN key without entering a response. These skipped trials were not counted in the cumulative score but the tokens were put back into the utterance list and presented later in the experiment. The subjects were not encouraged to use this option except under occasional circumstances when a computer processing error occurred or a distraction caused the subject to miss the utterance.

For the training experiments the list of CV utterances played to the subject consisted of two of the three utterances of all 72 CV's spoken by all four talkers. Thus, the set of utterances consisted of 576 possible tokens.

This was done to prevent the subject from learning any idiosyncratic cues for any one utterance since any item may be repeated during feedback. Not all tokens were necessarily played, however, since a finite time limit of between 40 and 50 minutes was placed on the duration of a training experiment to avoid subject fatigue. The token list for the identification experiments consisted of the remaining utterance, spoken by each talker but not used in training, for a total of 288 items. An identification experiment would generally take between 15 and 20 minutes to complete. A listener's test day typically started with one training experiment followed by five or six identification experiments, usually for one filtering condition at several different presentation levels. The order of presentations of the three different utterances of the CV's was randomized so the a subject did not hear the same one in succession on any day.

Each subject heard all ten filtered conditions at a minimum of five different presentation levels. As a rule, at least two runs of 288 trials were performed at each presentation intensity. Some exceptions occurred because of the large volume of data collected, especially for Subject J.G. who was tested both as a hearing impaired subject and as a normally-hearing listener. During the course of this study, however, each subject listened to about 40,000 CV syllables per condition, or over 4000 per filter condition.

Since all subjects but one were used for more than one ear-condition, these subjects heard between 80,000 and 120,000 CV's.

PRESENTATION CONDITIONS

The presentation levels for training were determined for each condition, filtered and unfiltered, by playing the CV's to the subject and asking him or her to report when most comfortable loudness (MCL) was achieved. In some cases, and in particular for the two narrow band-pass conditions, MCL was difficult to determine since level variation from syllable to syllable was great and depended on the particular spectral shape of the utterance. For example, for the Band-Pass 700-1400 Hz condition, the second formant of the vowel /a/ fell within this band, whereas it did not for /i/ and /u/. Thus, the CV's with /a/ were, on the average, the same level as the unfiltered CV's. However, the /i/and /u/ syllables were 20 to 40 dB below the average unfiltered CV level. MCL was, therefore, not used for any precise considerations in performance but rather used as a guideline in determining the range of presentation levels to the subject.

Presentation levels usually extended over a 40 to 50 dB range, from about 20 to 30 dB below the estimated MCL to about 15 to 20 dB above. Level increments were either 6 dB and 10 dB, as necessary to define useful performance-intensity functions for the test condition. Training and identification experiments were first run on the unfiltered speech and then on the wider band filtered conditions to increase the learning rate for the task as well as familiarity with the materials. A practical limit was placed on the test order, however, by the limited computer storage space for the digitized speech waveforms. Thus certain tests could not be run until all subjects had completed a specific number of conditions.

Initially, training experiments were conducted at the MCL determined for the specific filter condition. Subsequent training experiments, occuring after a series of identification experiments for the same filter condition, were run at a level at or near the performance maximum for the condition. After training, identification experiments were run on that condition at decreasing levels in 10 or 20 dB steps until a lower audibility limit or practical performance limit was reached. Intensity levels were then increased in 6 or 10 dB increments. The size of the increment depended on the observed slope of the performance-intensity function. This procedure was designed to avoid auditory fatigue that might affect subsequent experiments on any given day. Presentation levels were increased until maximum tolerable level was reached or performance began to roll over. If rollover occurred well below discomfort threshold, then intensity was increased further in 6 dB steps until the loudness discomfort limit was reached. This was not done with the normally-hearing subjects since a flattening of performance was felt to be an adequate stopping point without reaching the limits imposed by safety precautions. One should recall that if a subject reported any aftereffects such as tinnitus testing was stopped in that ear for that day.

Contralateral masking was used in the non-test ear whenever any potential for cross hearing existed at higher presentation levels. The masking signal was the "speech noise" masker built into the Grason-Stadler speech audiometer. Masking levels were determined individually for each subject based on an assumption of 40 dB interaural attenuation. Whenever presentation levels were greater than 40 dB above the threshold of the non-test ear, masking was introduced at a maximum level of 20 dB below the hearing level of the speech signals being played. If the overall level of the filtered speech signals were sufficiently low, as in the high-pass 1400 Hz and high-pass 2800 Hz conditions, masking was reduced or eliminated to avoid the danger of overmasking.

CHAPTER VI

RESULTS

INTRODUCTION

Data for this thesis consist of percentage correct syllable identification scores for each of the ten conditions of filtered speech at each tested intensity. These data are plotted as performance-intensity functions for each filter condition and each of twelve cases as defined in Chapter IV. In addition, consonant and vowel confusion data were collected for the CV syllables for all subjects and conditions to permit confusion analyses at a future date.

In the report, data will be presented in two basic forms:

(1) Performance vs. Intensity (P.I.) functions

Mean percentage correct scores vs. presentation level in dB Free Field SPL are plotted for all ten filter conditions (including wideband or "unfiltered" speech). For these data, intelligibility scores were averaged across the four talkers and represent both consonant and vowel errors. That is, if either the consonant or vowel component of the CV utterance was incorrectly identified, then the entire utterance was scored as incorrect.

(2) <u>Articulation Index (A.I.) functions</u>

Observed percentage correct scores for each of the ten filtered speech conditions are plotted against the calculated Articulation Index. Superimposed on these plots is a curve of percentage correct vs. Articulation Index derived by interpolating a curve between the 1000 PB words curve and the 1000 nonsense syllable curve in Figure 1. This curve is not a function based on the observed data but a reference curve for comparative purposes. This will be discussed later in this report.

Appendix E contains the individual results for each subject and ear condition used for plotting the graphs in this chapter. Included in the tables are the mean percentage correct syllable intelligibility score for each filter condition, at each presentation level, averaged for the four talkers. Also listed is the calculated Articulation Index averaged for the four talkers, the total number of CV tokens heard by the subject for each data point, and the standard deviation of the intelligibility scores across the four talkers at each level.

Overall reliability of the individual scores was quite high. Direct estimates of test-retest reliability were performed on subjects for whom test points were repeated, and a split-half method of analysis was used for those conditions which were run only once. Generally, intelligibility differences of greater than 3 percentage points were significant. This criterion was used for determining whether saturation or "rollover" in intelligibility score was apparent in any subject's performance at high presentation levels. Therefore, in the tables that follow, if rollover is indicated, intelligibility score was more than 3 percentage points below the maximum score at a presentation level above that found to show maximum performance. If no rollover is indicated, then the presentation level shown is the level at which the maximum score occurred. A third indication, designated by "-", means that scores at presentation levels higher than that at which the maximum intelligibility occurred were lower than the maximum, but by less than three percentage points.

FILTERED SPEECH PERCEPTION - NORMALLY-HEARING LISTENERS

Performance vs. Intensity (P.I.) functions for the three normally-hearing listeners tested in quiet are shown in Figures 12, 13 and 14. Table 2, following, shows for the normally-hearing listeners, the maximum scores obtained for each condition tested, the presentation level in dB sound pressure level specified relative to the unfiltered speech level at which the maximum score was achieved and whether rollover occurred.

TABLE 2

PERCENTAGE CORRECT SCORES, LEVEL FOR MAXIMUM PERFORMANCE AND PRESENCE OF ROLLOVER AT HIGH INTENSITY

NORMALLY-HEARING SUBJECTS IN QUIET

FILTER	PERCENTAGE CORRECT			LEVEL (SPL)			ROLLOVER		
	JG	JT	RM	JG	JT	RM	JG	JT	RM
UNFILT.	93.9	94.8	90.4	62	56	56	Y	Y	-
LP 700	37.2	37.4	33.1	86	76	76	N	Y	Y
LP 1400	62.6	57.9	57.5	76	76	76	-	N	N
LP 2800	87.2	88.9	85.2	56	76	76	N	N	N
HP 700	91.0	92.4	89.5	56	66	76	N	-	N
HP 1400	76.5	77.3	73.3	66	66	76	-	N	N
HP 2800	38.2	49.6	30.2	66	96	76	N	N	Y
BP .7-1.4K	55.2	54.9	49.8	66	86	66	Y	N	Y
BP1.4-2.8K	72.9	71.3	67.0	66	76	66	-	-	-
BP .7-2.8K	82.7	83.0	81.0	66	86	76	Y	N	-

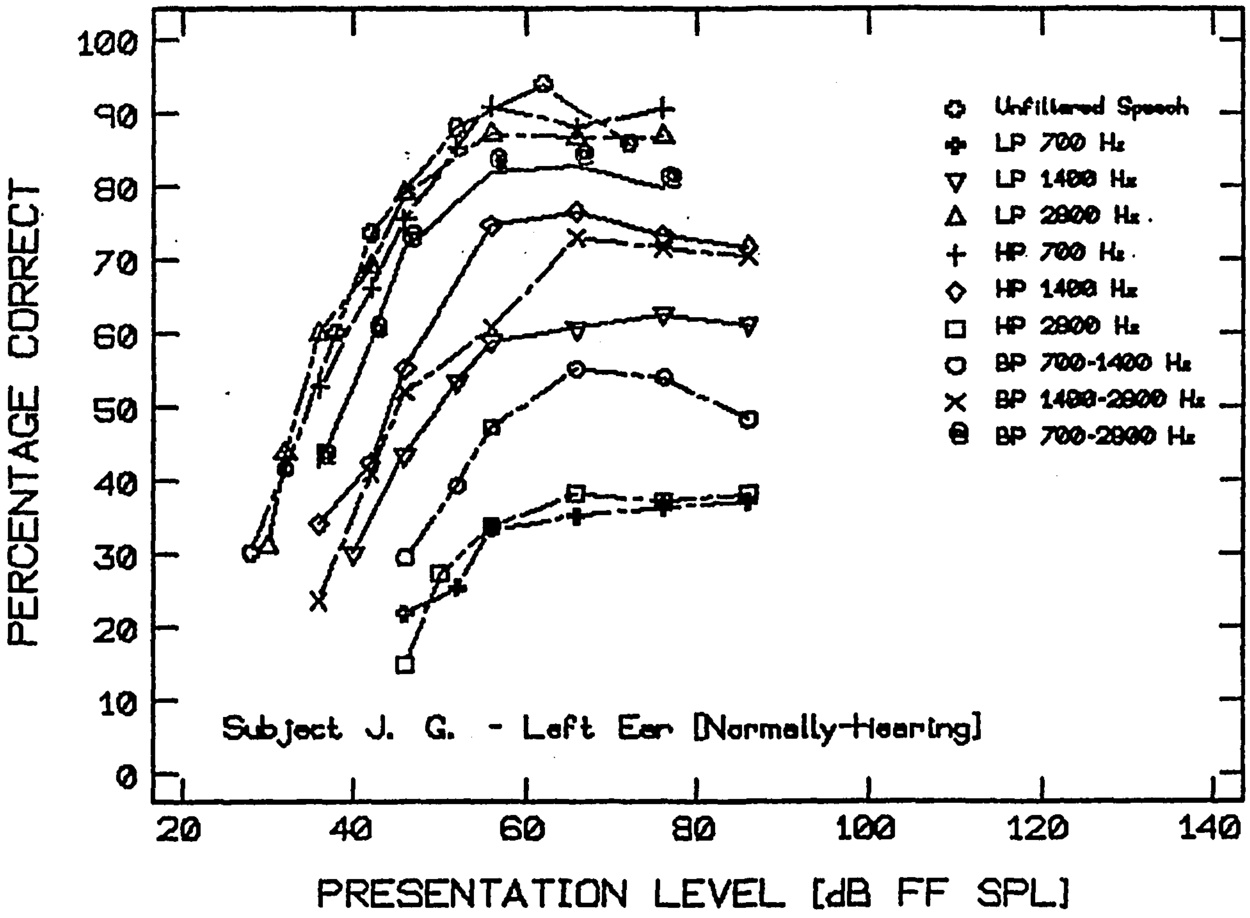


FIGURE 12

Performance - Intensity Functions:

Subject J.G. - Left Ear [Normally-Hearing]

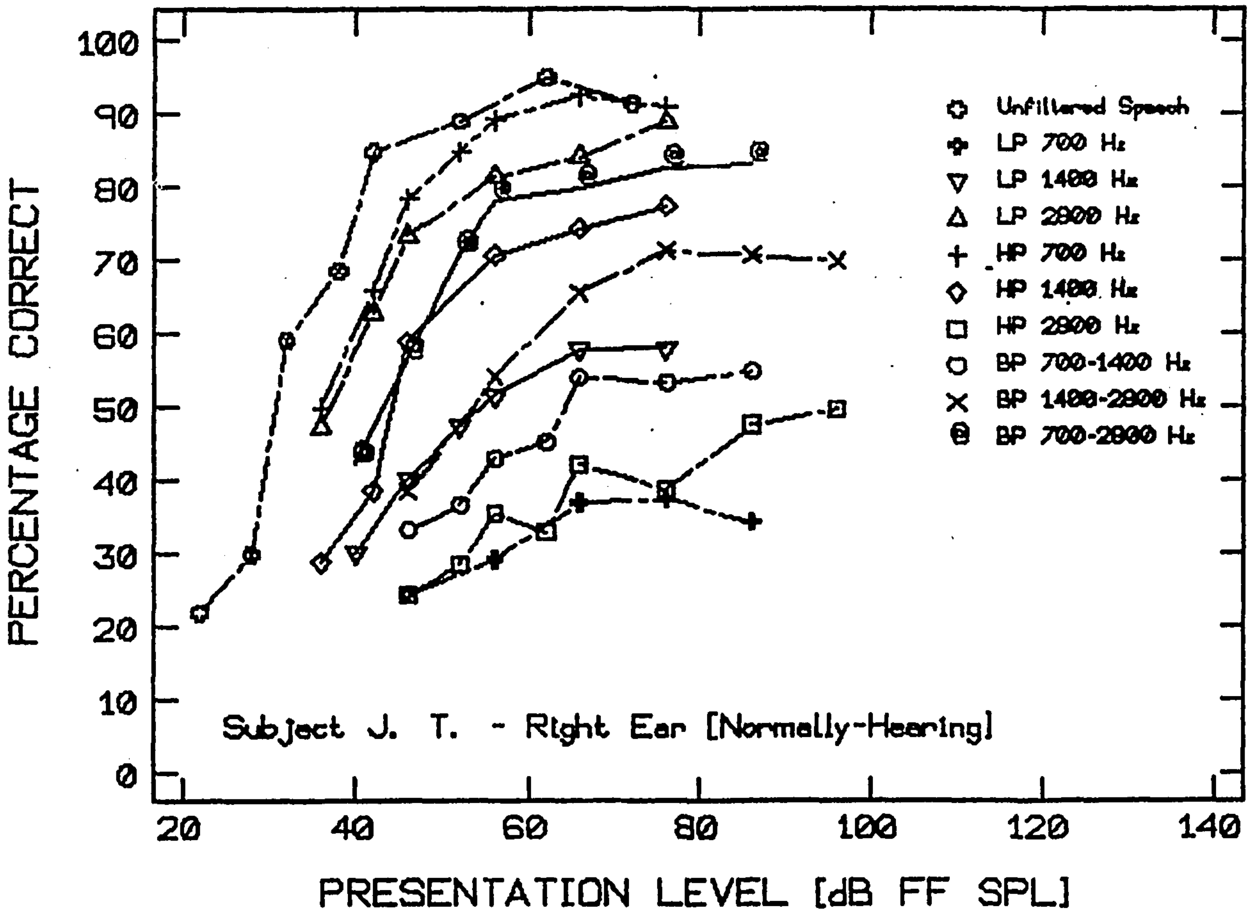


FIGURE 13

Performance - Intensity Functions:

Subject J.T. - Right Ear [Normally-Hearing]

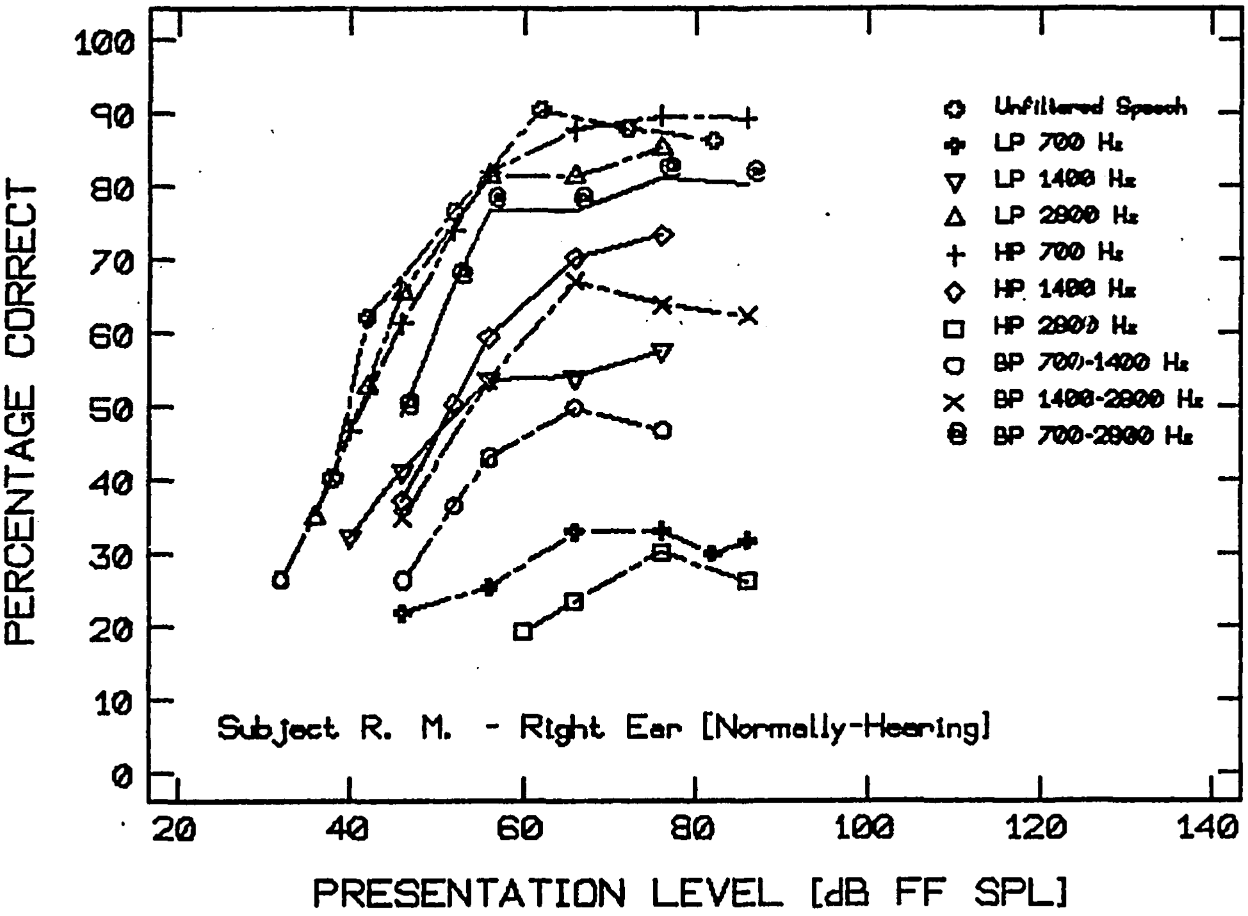


FIGURE 14

Performance - Intensity Functions:

Subject R.M. - Right Ear [Normally-Hearing]

The performance-intensity functions for the normally-hearing listeners exhibited characteristic performance improvements as presentation level increased and as the amount of available spectral information increased. For the unfiltered speech, maximum performance was achieved at a presentation level of 62 dB SPL (FF) for all three subjects. One hundred percent performance was not achieved by any listener. Among the reasons for this are the fact that the "unfiltered speech" has an upper limit of 4500 Hz, the use of untrained talkers, the general difficulty of the speech materials (e.g., Hirsh, Reynolds and Joseph, 1954), and the absence of a carrier phrase (Egan, 1948). French and Steinberg's maxima occur at between 0 and +10 dB "orthotelephonic response." 0 dB orthotelephonic response corresponds to an RMS sound pressure level of 65 dB SPL (Kryter, 1962a). Peak performance is achieved at essentially the same presentation levels as for French and Steinberg. A discussion of these issues and a comparison of the data of the present study with French and Steinberg's data and other investigators, as well, will be made in the next chapter.

Of particular interest are the curves for the high-pass 700 Hz condition. Performance for this condition, although poorer than the unfiltered speech at low presentation levels, eventually equals performance for that condition at high levels. For unfiltered speech at lower presentation levels, the information below 700 Hz obviously contributes to intelligibility. However, at higher levels, this information may not be essential. In addition, at high levels, there may be a greater spread of masking of more intense low frequency vowel energy to the high frequency components. However, in the high-pass 700 Hz condition at high presentation levels, the energy below 700 Hz may not be completely absent from the signals used in this study due to the characteristics of the digital filters used. The filter cutoff rates were such that all energy below 500 Hz was present but attenuated by 60 dB. Between 700 Hz and 500 Hz the slope of the attenuation was linear, equivalent to a rate of about 105 dB per octave. At the maximum tested level of 76 dB SPL, this low frequency energy may be about 16 dB SPL, and therefore may be just barely audible. Pollack (1948) and Rosenthal, Lang and Levitt (1975) observed that contributions of low frequencies at low energy levels may be significant. Pollack also observed that extreme high frequency information is important. This is confirmed by the improved performance for the conditions when information above 2800 Hz is added to the low-pass and band-pass conditions with upper cutoff frequencies of 2800 Hz.

In comparing the results for R.M. in Figure 14 to the results of J.G. and J.T. in Figures 12 and 13, one observes for R.M. somewhat steeper slopes of the performance-intensity functions and lower scores at the same presentation levels. The maximum scores were also slightly lower for all conditions, and in particular for HP 2800 Hz. This reduced performance was likely due to the slight elevation in threshold shown by R.M. as well as possible age effects that are not well understood.

As seen in Table 2, rollover in performance occurred for listeners J.G. and J.T. for the unfiltered speech, and for listeners J.T. and R.M. for the LP 700 Hz filtered condition. In addition, rollover occurred for other filtered conditions but not consistently the same ones for each subject. It is not clear from this result whether rollover was a significant factor for the normally-hearing subjects, including R.M., the older adult control at the levels tested. A more important observation is that a performance plateau was reached by each subject and was generally maintained over a 30 to 40 dB range in presentation level. For all filter conditions but one, maximum performance was achieved between 60 and 65 dB SPL. The exception was the HP 2800 Hz filter condition which, for subject J.T., showed increased intelligibility as presentation level increased. This increase in intelligibility may also have been due to the finite attenuation characteristics of the digital filters used. The filters were designed to have the stop band at 200 Hz above or below the cutoff frequency, with a stop band attenuation of 60 dB. For the HP 2800 filter, this meant that components below 2600 Hz were present but attenuated 60 dB relative to the pass band information. The filter skirts for this filter were equivalent to 800 dB per octave over this 200 Hz band. At high presentation levels, some of the more intense low frequency vowel energy was probably audible, increasing intelligibility scores at those levels. Since subject J.T. was an audiologist and experienced in administering and listening to speech tests, she may have been more aware of these low intensity cues than the other subjects.

PERCEPTION OF FILTERED SPEECH - HEARING-IMPAIRED LISTENERS

SUBJECT F.G.

For Subject F.G., with bilateral noise-induced high frequency hearing loss, testing was performed on both ears since his hearing loss was somewhat asymmetrical. His left ear exhibited greater hearing loss above 2000 Hz than his right. Performance-intensity functions are shown in Figures 15 and 16. Table 3 shows the maximum scores, the level at which they occurred and whether rollover occurred.

TABLE 3

PERCENTAGE CORRECT SCORES, LEVEL FOR MAXIMUM PERFORMANCE AND PRESENCE OF ROLLOVER AT HIGH INTENSITY

SUBJECT F.G. - HEARING-IMPAIRED

FILTER	PERCENTAGE CORRECT		LEVEL (SPL)		ROLLOVER	
	RIGHT	LEFT	RIGHT	LEFT	RIGHT	LEFT
UNFILT.	79.7	71.1	94	74	Y	Y
LP 700	37.6	35.6	78	78	Y	-
LP 1400	52.6	52.4	68	68	Y	Y
LP 2800	79.4	76.9	78	88	N	Y
HP 700	83.3	82.8	98	98	Y	-
HP 1400	62.9	57.8	88	88	Y	Y
HP 2800	35.0	34.2	104	108	Y	Y
BP .7-1.4K	51.6	48.8	78	78	Y	-
BP1.4-2.8K	61.7	59.6	88	88	-	-
BP .7-2.8K	79.6	79.5	98	88	Y	Y

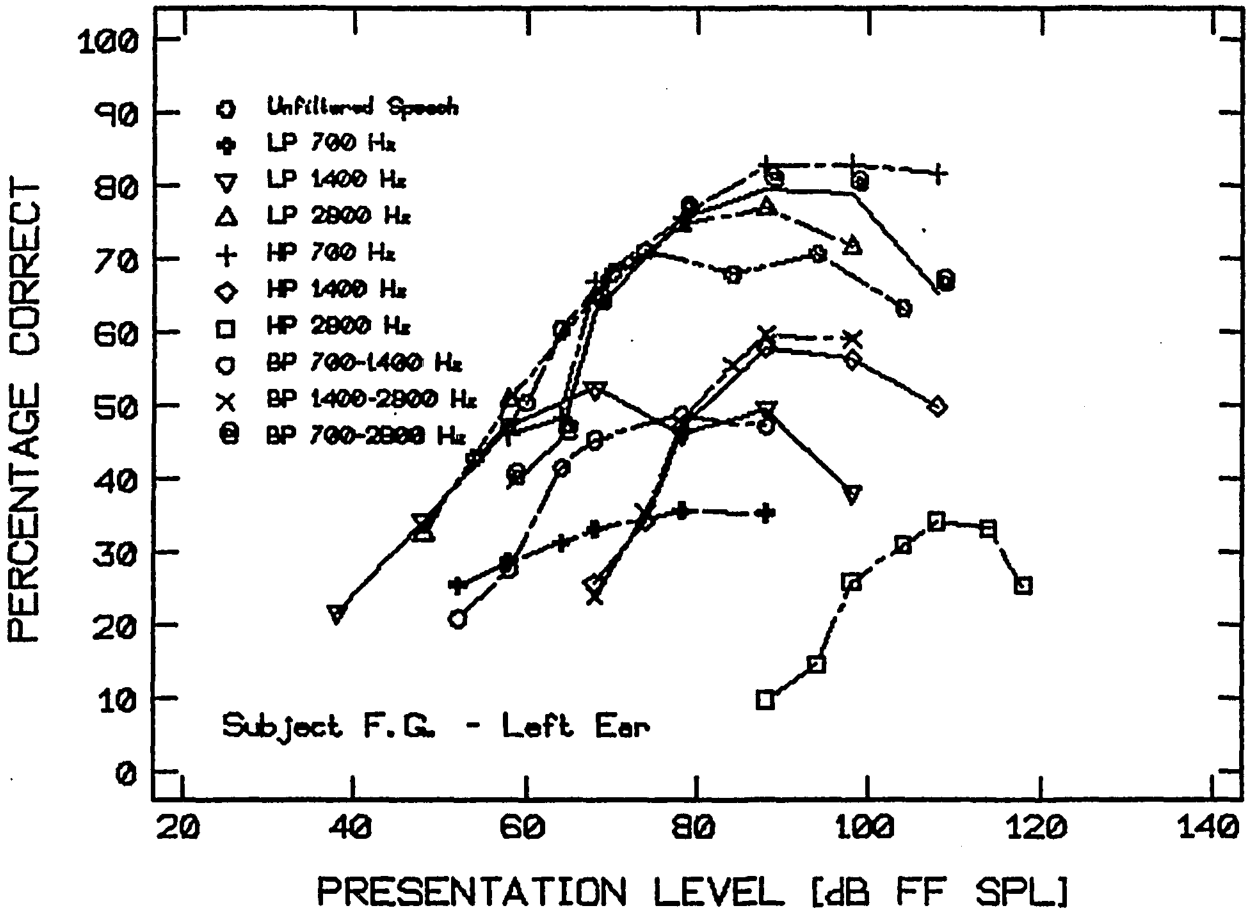


FIGURE 15

Performance - Intensity Functions:

Subject F.G. - Left Ear [Impaired]

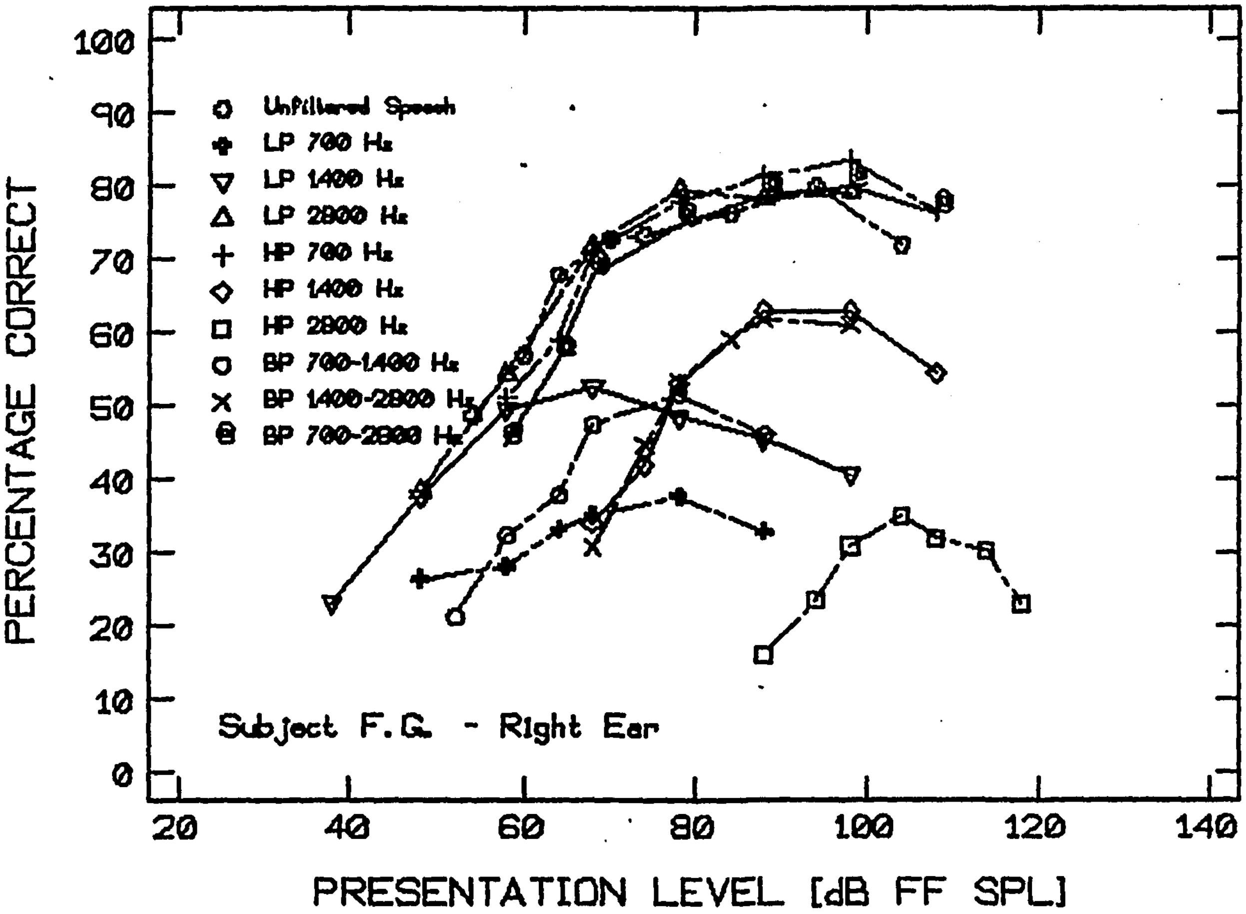


FIGURE 16

Performance - Intensity Functions:

Subject F.G. - Right Ear [Impaired]

Overall maximum performance occurred for the HP 700 Hz condition in both ears, with little difference between their scores. However, maximum performance for the unfiltered condition in the left (poorer) ear was nearly 10 percentage points lower and was reached at 20 dB lower intensity than for the right. Both ears exhibited rollover for this condition, but the left ear much sooner. On the other hand, no significant rollover occurred in the left ear for the HP 700 Hz condition. This was the only case for the high-pass conditions for which no rollover occurred. When low frequency information was removed below 700 Hz, F.G. was better able to make use of his residual high frequency hearing. Differences in maximum performance between left and right ears also occured for the HP 1400 Hz condition. Maximum performance scores between right and left ears for the HP 2800 Hz condition were nearly equal and at essentially the same presentation levels. His scores for the HP 2800 Hz condition were also near that for the normally-hearing listeners. As noted, due to the digital filters, some low frequency energy may become audible at high presentation levels. His maximum performance occurred at 104 dB SPL in both ears. Thus, vowel energy in the low frequencies is at 44 dB SPL, no doubt audible to this listener.

Since F.G.'s hearing below 1000 Hz was near normal in both ears, it is not surprising that performance for the LP

700 Hz condition was comparable to that for the normals. For LP 2800 Hz speech, scores in the right ear were slightly higher and at a lower intensity than the left. The right ear showed no rollover whereas the left did. For LP 700 Hz and LP 1400 Hz maximum performance scores were nearly identical and occurred at the same test level. However, rollover for these conditions occurred only in the right ear for LP 700 Hz, but in both ears for LP 1400 Hz. It may be important that for the LP 1400 Hz condition, rollover occurred at a relatively low presentation level. Maximum performance for this condition was achieved at 70 dB SPL, a level lower than that for any other filter condition. It is likely that at higher levels, even the moderately intense low frequency energy available for this condition masked what little high frequency information may be available to this subject. Similar to performance in F.G.'s left ear, as level increased, performance diminished to nearly the same as for the LP 700 Hz condition.

Scores increased for the three band-pass conditions from the BP 700-1400 Hz to BP 1400-2800 Hz and BP 700-2800 Hz. Rollover occurred in both ears for the BP 700-2800 Hz case, in the right ear for BP 700-1400 Hz, but in neither ear for BP 1400-2800. No obvious pattern appeared in the conditions that exhibited rollover, although more conditions exhibited rollover in the right or better ear. This is contrary to what one would intuitively expect; that is, that saturation or rollover would occur in the ear with the more severe hearing loss.

It is possible that rollover would eventually occur as level increased even in those conditions for which it was not observed during this study. Extra precautions were taken, however, in F.G.'s case to keep presentation levels below the threshold of discomfort. The reason for this was that he occasionally reported tinnitus after testing at relativly high intensity; however, he did not indicate that the levels were uncomfortable. If tinnitus did occur, no further testing in that ear took place that day.

SUBJECT G.M.

For subject G.M., ear difference was much more significant than for F.G. Figures 17 and 18 show the performance-intensity functions for this subject, and Table 4 shows the maximum performance scores, level and rollover conditions.

TABLE 4

PERCENTAGE CORRECT SCORES, LEVEL FOR MAXIMUM PERFORMANCE AND PRESENCE OF ROLLOVER AT HIGH INTENSITY

SUBJECT G.M. - HEARING-IMPAIRED

FILTER	PERCENTAGE CORRECT		LEVEL (SPL)		ROLLOVER	
	RIGHT	LEFT	RIGHT	LEFT	RIGHT	LEFT
UNFILT.	58.5	78.3	84	74	Y	Y
LP 700	28.0	29.0	78	78	-	Y
LP 1400	31.1	47.0	88	68	Y	Y
LP 2800	55.1	71.2	88	78	-	Y
HP 700	57.9	75.2	88	88	Y	Y
HP 1400	37.6	53.8	94	88	-	Y
HP 2800	24.0	29.7	114	98	N	Y
BP .7-1.4K	26.0	39.9	94	68	Y	Y
BP1.4-2.8K	35.2	58.3	108	84	Y	Y
BP .7-2.8K	55.6	69.6	104	78	N	Y

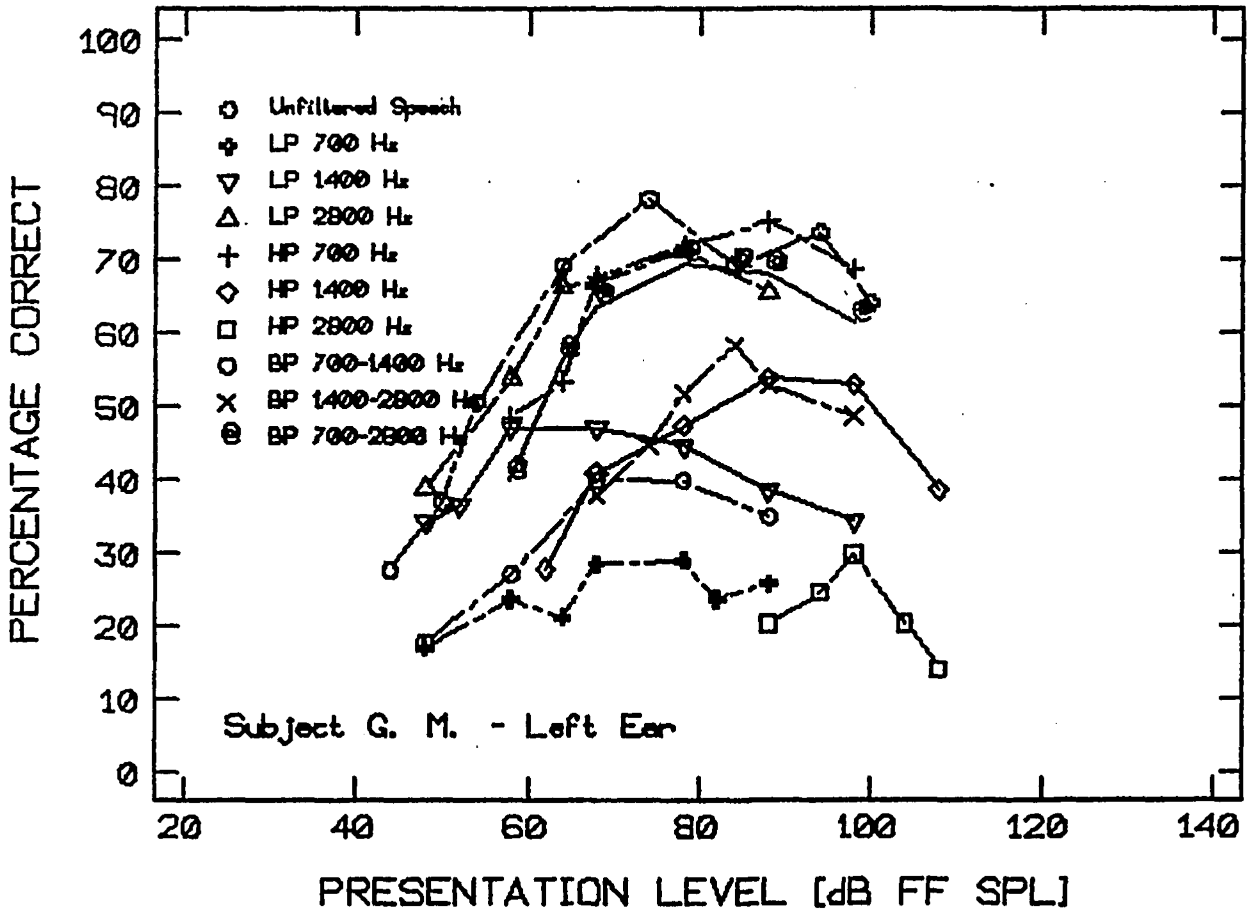


FIGURE 17

Performance - Intensity Functions:

Subject G.M. - Left Ear [Impaired]

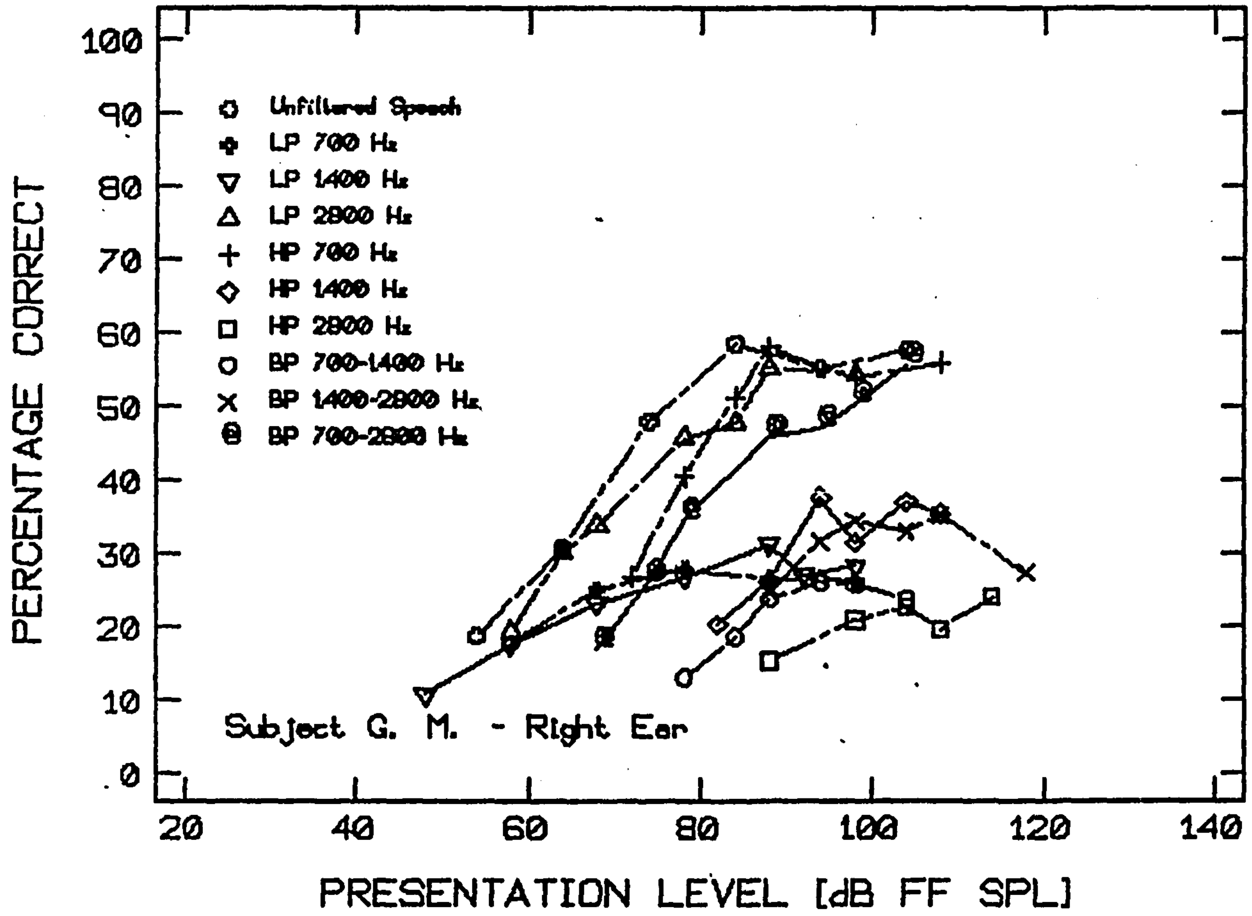


FIGURE 18

Performance - Intensity Functions:

Subject G.M. - Right Ear [Impaired]

In all cases, maximum scores in the left ear were higher and occurred at lower intensities than in the right. Rollover took place for every tested condition in the left ear, and for five of the ten conditions in the right. However, in the right ear, rollover did not occur in any predictable manner.

In G.M.'s left ear, significantly higher performance was obtained for the four widest band conditions: unfiltered speech, LP 2800 Hz, HP 700 Hz and BP 700-2800 Hz. Similar to the results for F.G., the scores for the HP 1400 Hz and BP 700-1400 Hz conditions were lower and nearly equal. Intelligibility for the LP 700 Hz and HP 2800 Hz conditions were the poorest. Unlike for F.G., however, overall maximum performance in both ears was not achieved for the HP 700 Hz condition, but for the unfiltered speech. G.M.'s maximum scores were also lower overall than F.G.'s. Possibly due to factors associated with age and etiology, G.M. may have been less able to use residual high frequency hearing as F.G. seemed to be able to do. In G.M.'s left ear, rollover for the LP 1400 Hz condition occurred at a relatively low presentation level, as it did for F.G.

Scores in G.M.'s right ear were, as noted, lower overall than in the left ear even at high presentation levels. Highest scores again were achieved for the four widest band conditions. At the highest level tested, these scores were nearly equal. However, the slopes of the P.I. curves appear significantly different. This effect was likely due to poorer low frequency thresholds in the right ear, as compared to the left. The somewhat better high frequency thresholds in the right ear, however, allow his performance to improve as intensity increases but not to the levels reached by the left ear. This may have been the result of greater presbycusic component to the hearing loss in this ear, as well as some unmeasured retrocochlear involvement, as evidenced by slightly poorer audiometric speech discrimination performance. Although performance for the BP 700-1400 Hz and BP 1400-2800 Hz bands are nearly equal and relatively poor, performance for the resultant band, BP 700-2800 Hz, is substantially better.

SUBJECT J.G.

Subject J.G. was tested in her impaired right ear. (J.G. was also tested as a normal listener in her normally-hearing left ear.) Table 5 indicates the maximum performance scores, the level at which the maximum occurred, and whether rollover occurred for the indicated condition.

TABLE 5

PERCENTAGE CORRECT SCORES, LEVEL FOR MAXIMUM PERFORMANCE AND PRESENCE OF ROLLOVER AT HIGH INTENSITY

SUBJECT J.G. - RIGHT EAR (IMPAIRED)

FILTER	PERCENTAGE CORRECT	LEVEL (SPL)	ROLLOVER
UNFILT.	69.5	104	N
LP 700	30.8	98	-
LP 1400	43.8	88	N
LP 2800	70.8	98	N
HP 700	72.3	98	N
HP 1400	51.9	98	-
HP 2800	23.2	108	Y
BP .7-1.4K	40.2	88	N
BP1.4-2.8K	55.2	108	-
BP .7-2.8K	66.2	98	Y

Performance-intensity functions for J.G. are shown in Figure 19. In comparing the relative scores for the ten conditions, higher performance scores were obtained for the wider band conditions, as was true for G.M. and F.G. Similarly, performance for the high-pass 700 Hz condition was greater than for the unfiltered condition. Poorest performance occurred for the HP 2800 Hz condition. The hearing loss in J.G.'s right ear was similar to G.M.'s left ear in shape and degree of hearing loss. Age and etiology differences between the subjects would normally preclude direct comparisons. However, because of similarities in the threshold curves, some comparisons are of interest. Her maximum scores were generally lower than G.M.'s scores and occur at higher levels than for G.M. Rollover was observed for two conditions, HP 2800 Hz and BP 700-2800 Hz. In particular, the rollover was especially great for BP 700-2800 Hz. For several other conditions, the drop in score was very slight. It is likely, however, that at increased intensity, a reduction in score would have occurred.

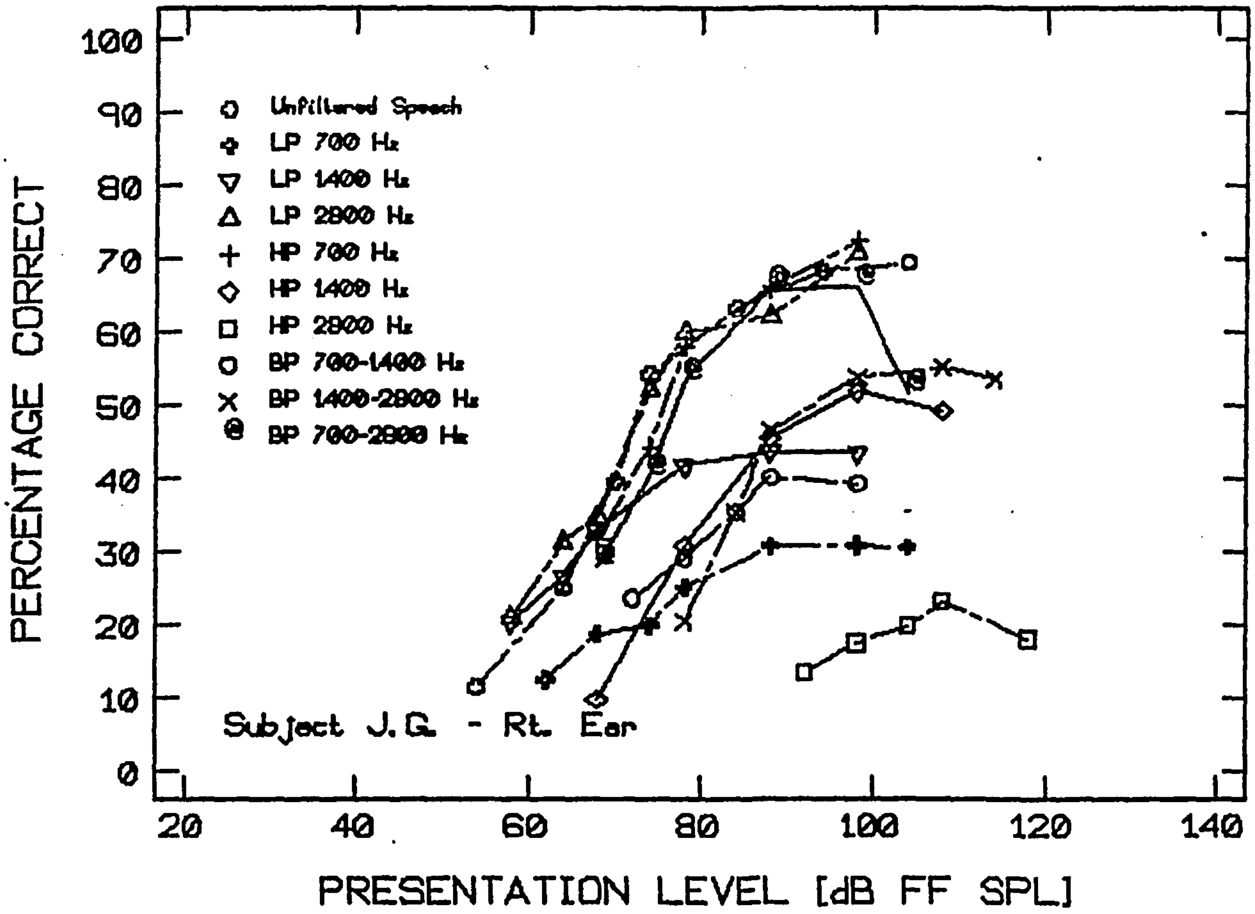


FIGURE 19

Performance - Intensity Functions:

Subject J.G. - Right Ear [Impaired]

SUBJECT T.T.

Subject T.T.'s results are shown in Table 6 and Figure 20. As for the other impaired listeners, the widest band conditions exhibited higher intelligibility scores relative to the other filtered conditions. However, unlike the results of the other impaired listeners, the results for the high-pass 700 Hz condition showed no improvement over the unfiltered condition at any presentation level. For T.T., the high-pass 2800 Hz band was barely audible and therefore contributed little useful information. The importance of the 1400-2800 Hz band appeared significant when this band was added to those below it in frequency.

TABLE 6

PERCENTAGE CORRECT SCORES, LEVEL FOR MAXIMUM PERFORMANCE AND PRESENCE OF ROLLOVER AT HIGH INTENSITY

SUBJECT T.T. - RIGHT EAR

FILTER	PERCENTAGE CORRECT	LEVEL (SPL)	ROLLOVER
UNFILT.	67.0	120	N
LP 700	29.2	114	-
LP 1400	39.4	124	N
LP 2800	63.4	114	Y
HP 700	65.0	118	-
HP 1400	35.3	124	Y
HP 2800	11.5	108	Y
BP .7-1.4K	38.7	124	N
BP1.4-2.8K	31.4	124	-
BP .7-2.8K	57.3	128	N

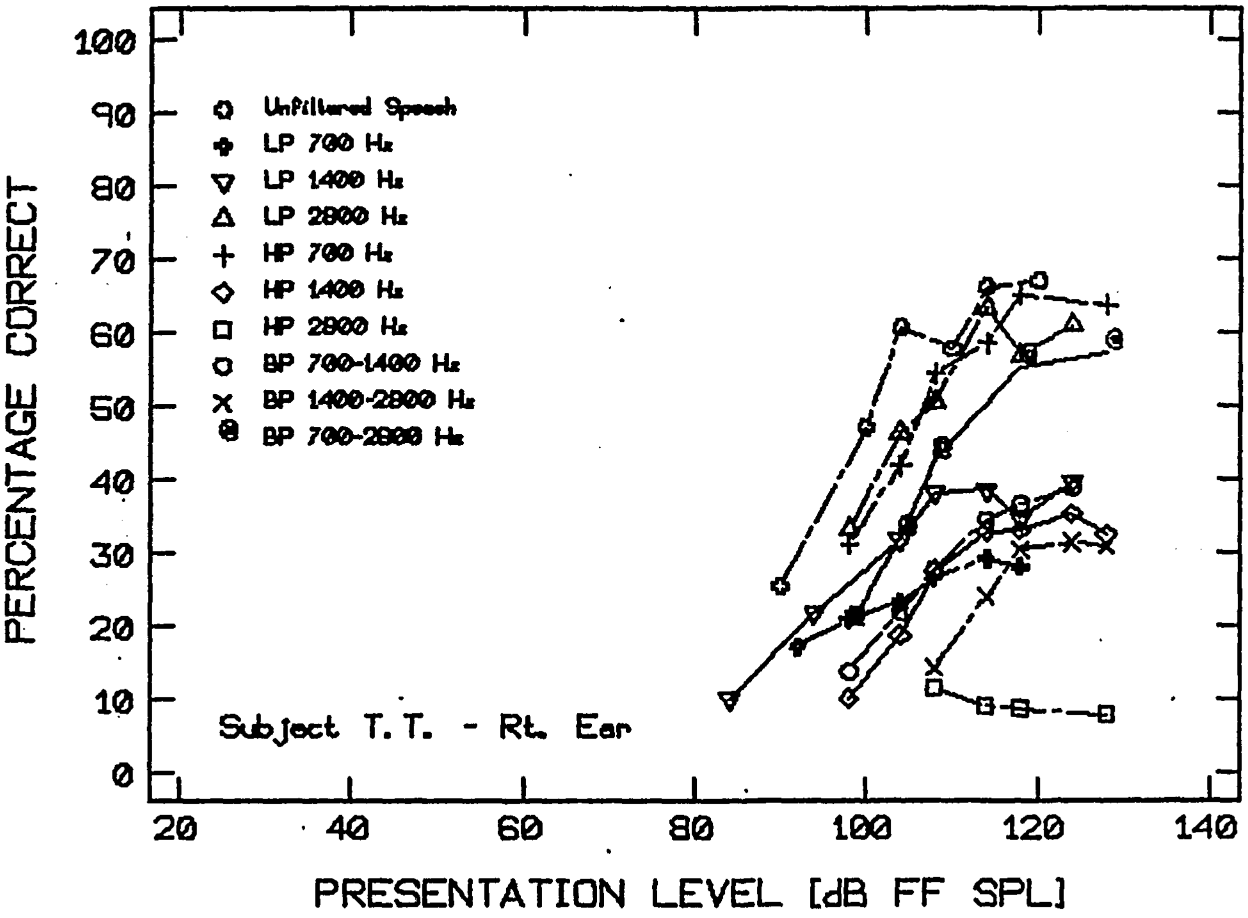


FIGURE 20

Performance - Intensity Functions:

Subject T.T. - Right Ear [Impaired]

Based on the 3% criterion mentioned earlier, rollover of performance occurred for three conditions. It is particularly interesting to observe the high presentation levels at which maximum performance was achieved. She never reported that these levels were uncomfortable even though for some conditions the speech audiometer used to control presentation levels was set to the maximum possible output. T.T. regularly wore a hearing aid in the ear that was tested for this study. She may, therefore, have had a greater tolerance to loud and possibly distorted speech than the other impaired listeners who were inexperienced with hearing aids. As a hearing aid wearer she was accustomed to hearing "filtered" speech. This may have accounted somewhat for the relatively high scores for the wider band filtered conditions.

NORMALLY-HEARING LISTENERS WITH SIMULATED LOSS - COMPARISON TO TESTS IN QUIET

The results for the three normally-hearing subjects tested with simultaneous masking noise to simulate hearing loss are presented in Table 7, which shows the maximum scores obtained, the presentation levels and whether rollover occurred for that condition. Performance-intensity functions for the three listeners are shown in Figures 21, 22 and 23.

TABLE 7

PERCENTAGE CORRECT SCORES, LEVEL FOR MAXIMUM PERFORMANCE AND PRESENCE OF ROLLOVER AT HIGH INTENSITY NORMALLY-HEARING LISTENERS WITH SIMULATED LOSS

FILTER	PERCENTAGE CORRECT			LEVEL (SPL)			ROLLOVER		
	JG	JT	RM	JG	JT	RM	JG	JT	RM
UNFILT.	88.2	92.0	86.5	92	92	92	N	N	Y
LP 700	40.6	40.1	34.8	76	86	86	N	Y	Y
LP 1400	57.8	58.3	53.9	86	76	86	N	Y	Y
LP 2800	85.8	87.2	81.9	96	86	96	N	N	N
HP 700	85.9	90.8	82.3	96	96	96	N	N	N
HP 1400	75.6	80.7	68.4	102	96	106	Y	N	N
HP 2800	46.2	55.9	32.3	112	106	106	N	Y	-
BP .7-1.4K	52.5	52.1	46.5	86	86	86	Y	Y	Y
BP1.4-2.8K	71.5	70.8	62.2	96	96	96	Y	Y	Y
BP .7-2.8K	79.3	84.6	76.9	96	96	96	N	N	N

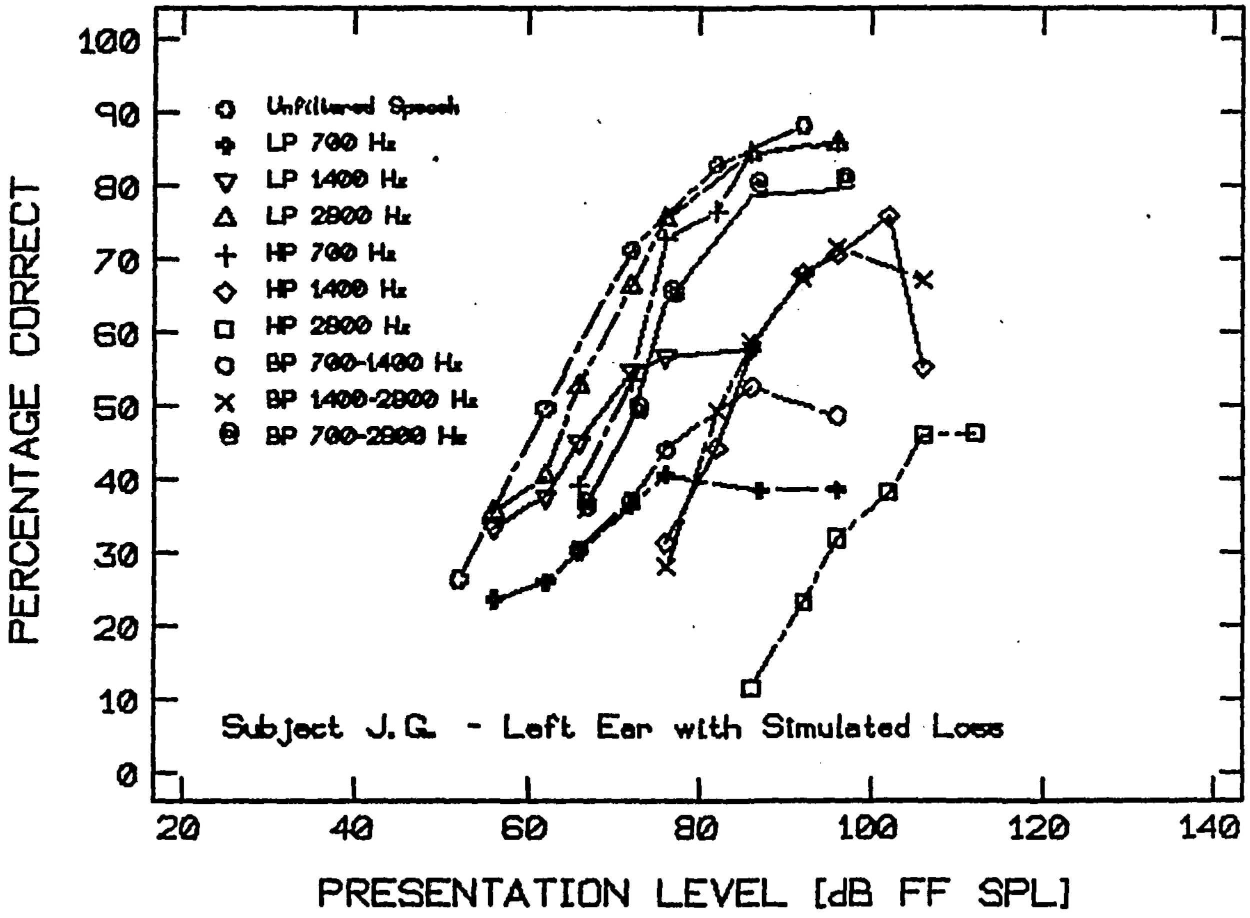


FIGURE 21

Performance - Intensity Functions:

Subject J.G. - Left Ear [Simulated Hearing Loss]

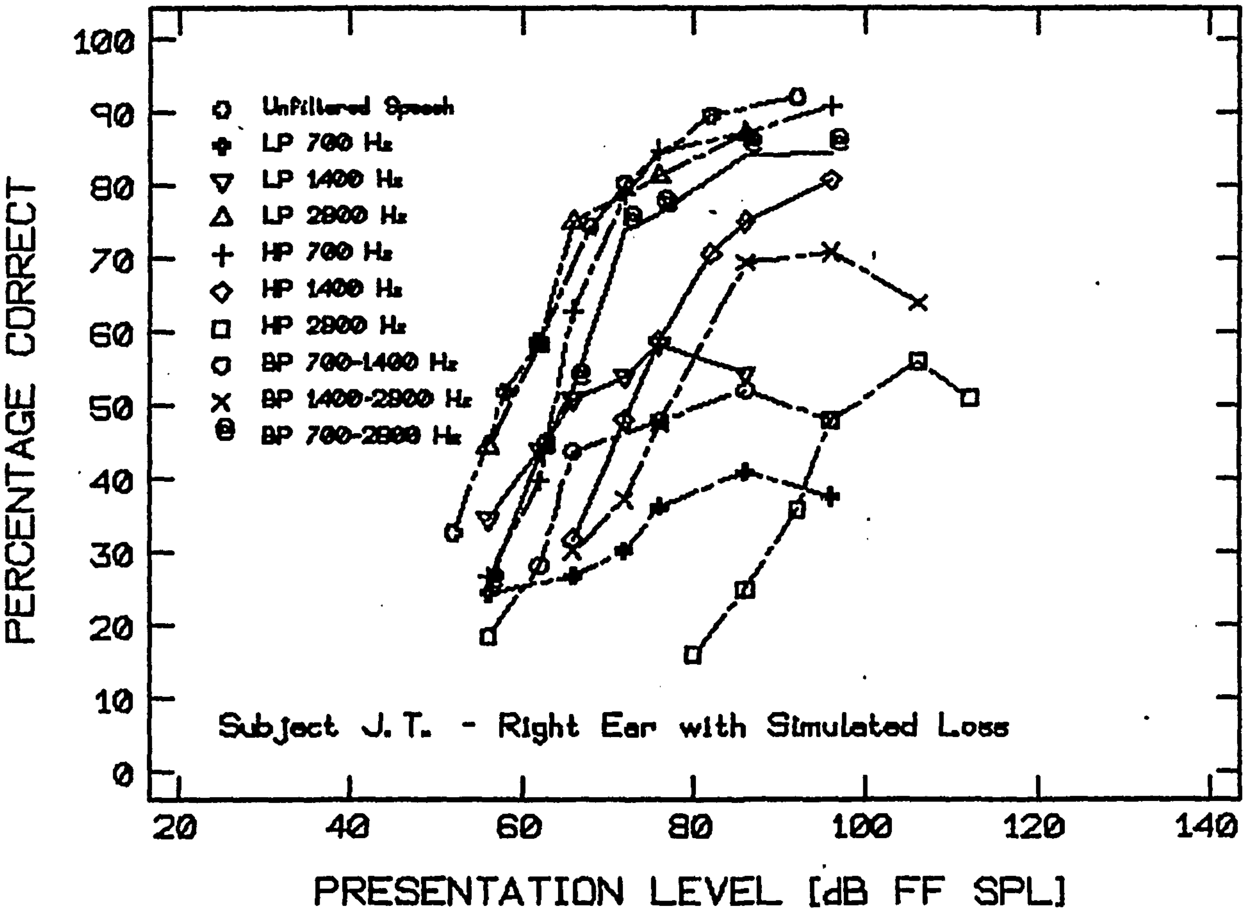
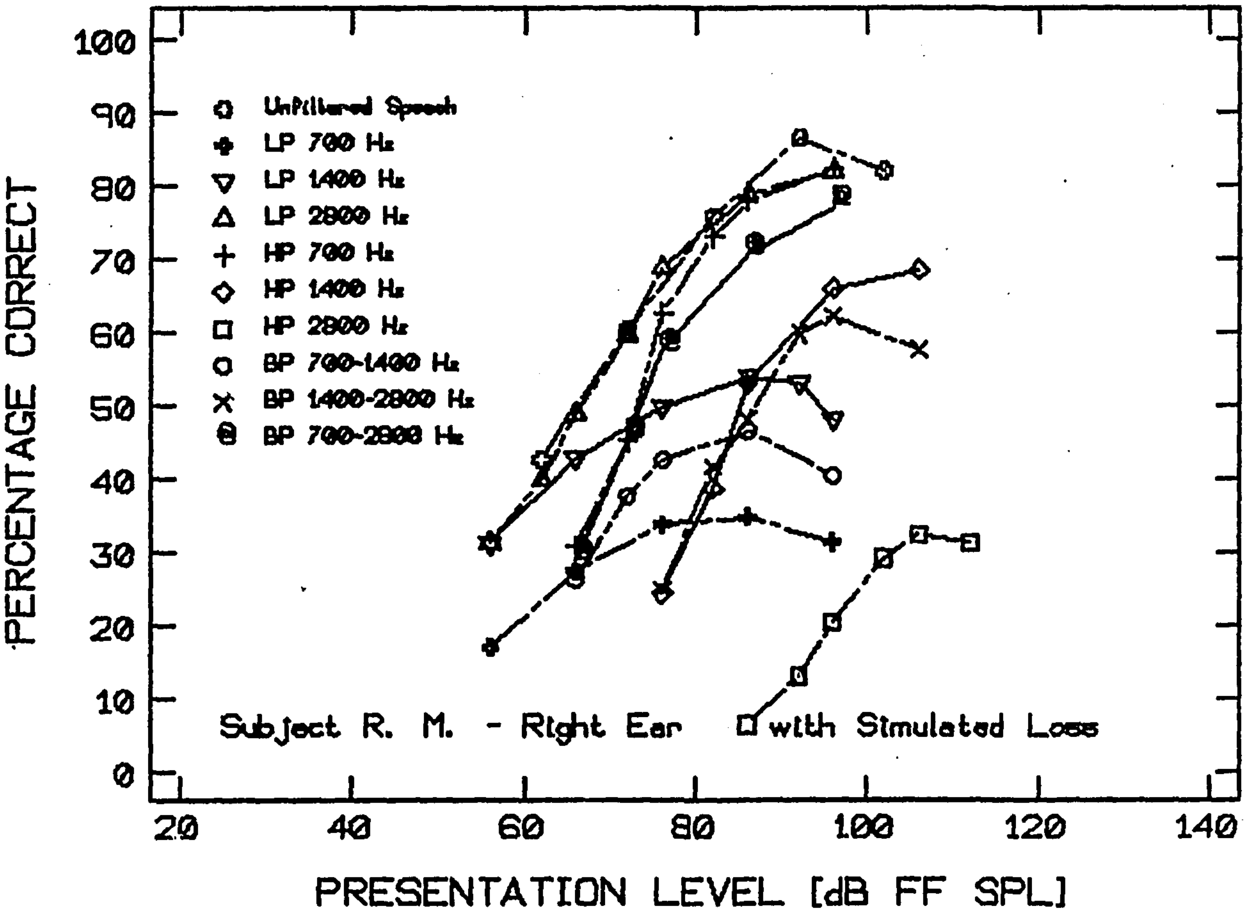


FIGURE 22

Performance - Intensity Functions:

Subject J.T. - Right Ear [Simulated Hearing Loss]



<u>FIGURE 23</u>

Performance - Intensity Functions:

Subject R.M. - Right Ear [Simulated Hearing Loss]

As might be expected, for all subjects tested with ipsilateral masking, P.I. functions shifted as a function of intensity as a result of the threshold shift caused by the masking. Maximum scores achieved by each subject were generally lower than those of the unmasked conditions. Exceptions to this occurred for the LP 700 Hz and HP 2800 Hz conditions. Slopes of the functions were generally steeper, however, than for the conditions tested in quiet, especially for the high-pass and band-pass conditions. As for the hearing-impaired subjects, P.I. functions for the wider band conditions were clustered and represented the highest levels of performance. In none of the cases did the HP 700 Hz condition at high levels exceed the performance of the unfiltered speech as it did for most of the impaired subjects. Figures 24 through 32 replot the performance-intensity functions for listeners J.G., J.T. and R.M. when tested in quiet (Figures 12, 13 and 14) and the P.I. functions of Figures 21, 22 and 23 to permit direct comparison of the results of the normally-hearing listeners when tested in quiet and with masking noise.

One important observation in examining the results of the testing with simulated loss is that rollover occurred for essentially the same conditions for which it occurred in the unmasked cases, the narrow band-pass filters and low-pass 700 Hz filtered speech. Even as level increased to overcome masking, with only one exception, rollover did not occur in the wider band conditions as it did for the impaired subjects. As level increased above the masking noise, performance scores achieved nearly the same maximum levels as for the unmasked conditions.

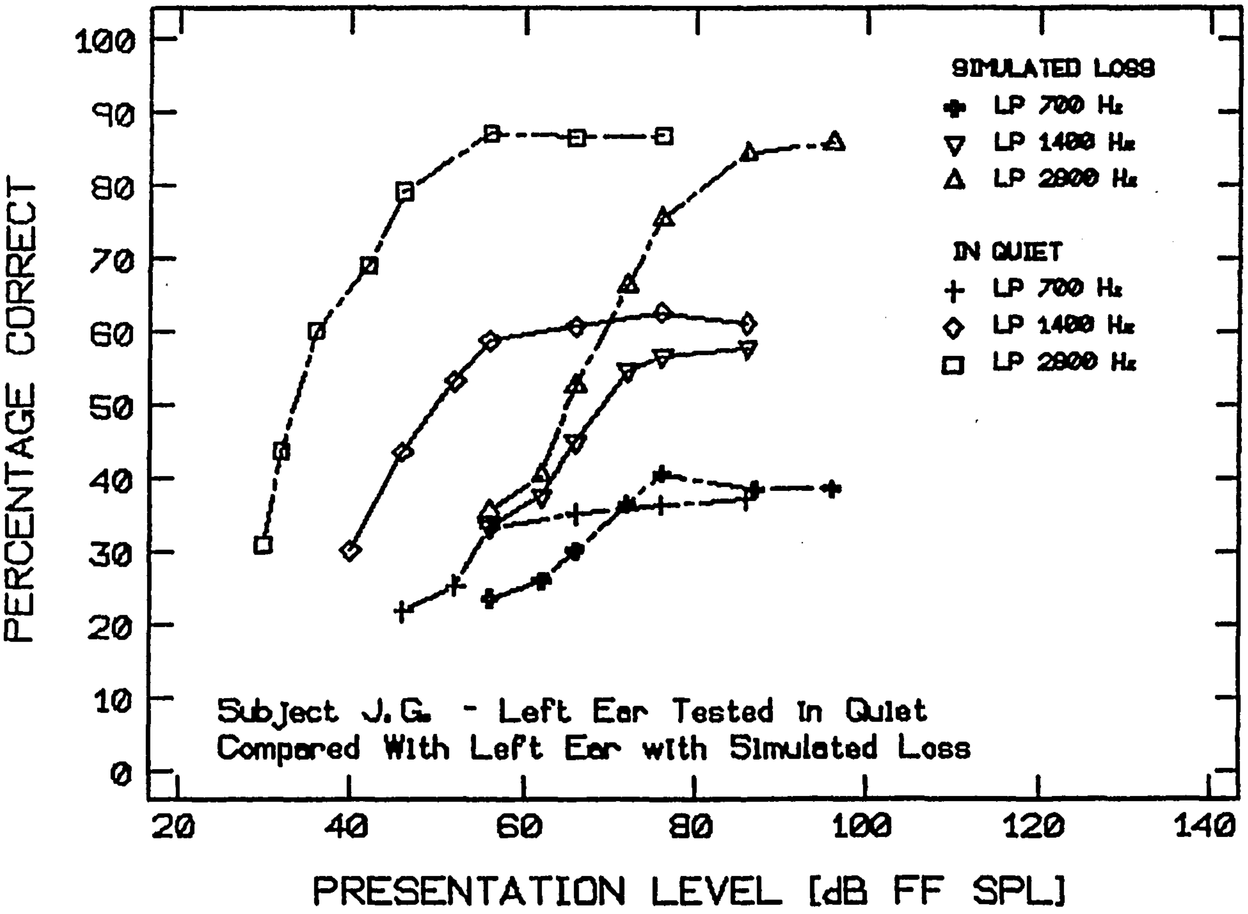


FIGURE 24

Performance - Intensity Functions:

Subject J.G. - Left Ear Tested in Quiet

Compared with Left Ear Simulated Loss

Low-Pass Filtered Speech

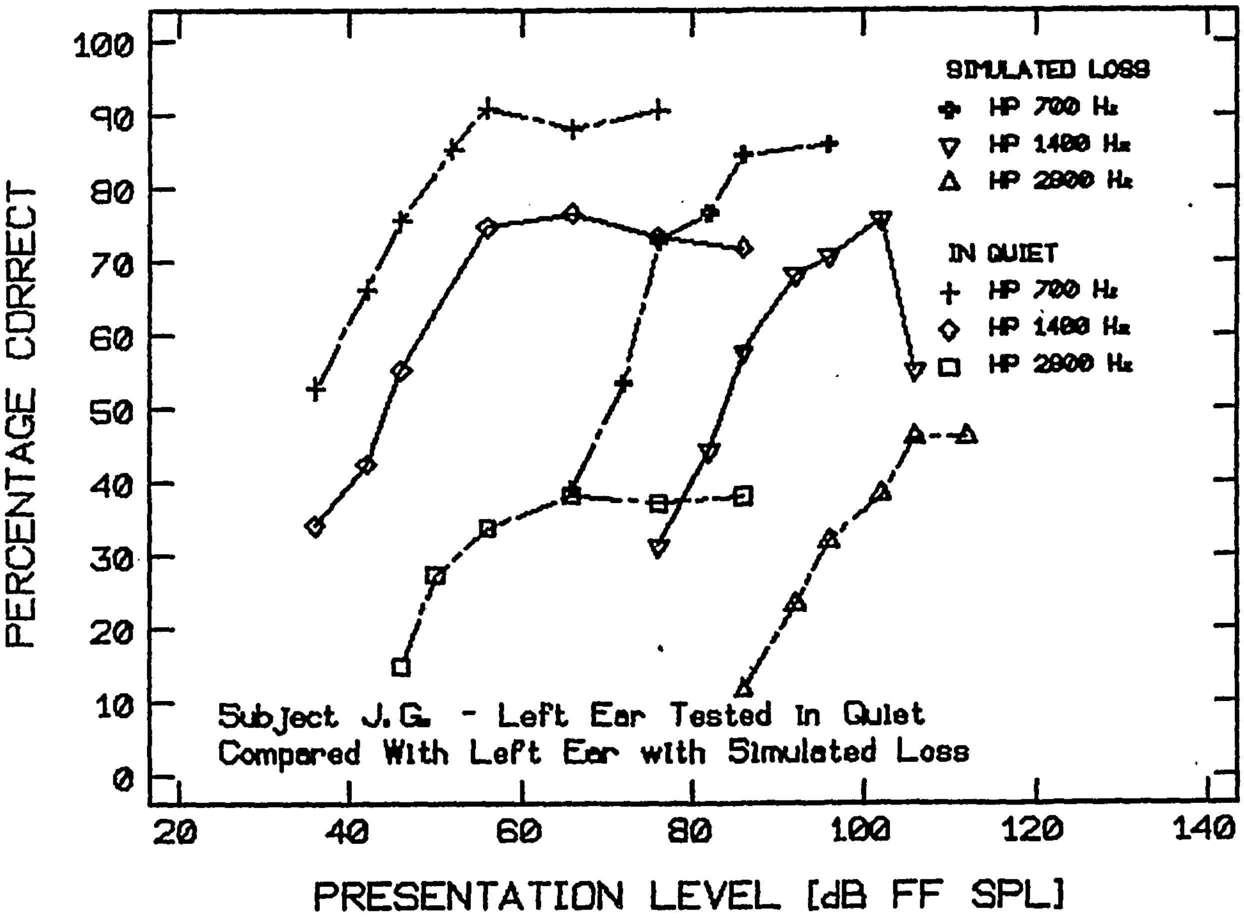


FIGURE 25

Performance - Intensity Functions:

Subject J.G. - Left Ear Tested in Quiet

Compared with Left Ear Simulated Loss

High-Pass Filtered Speech

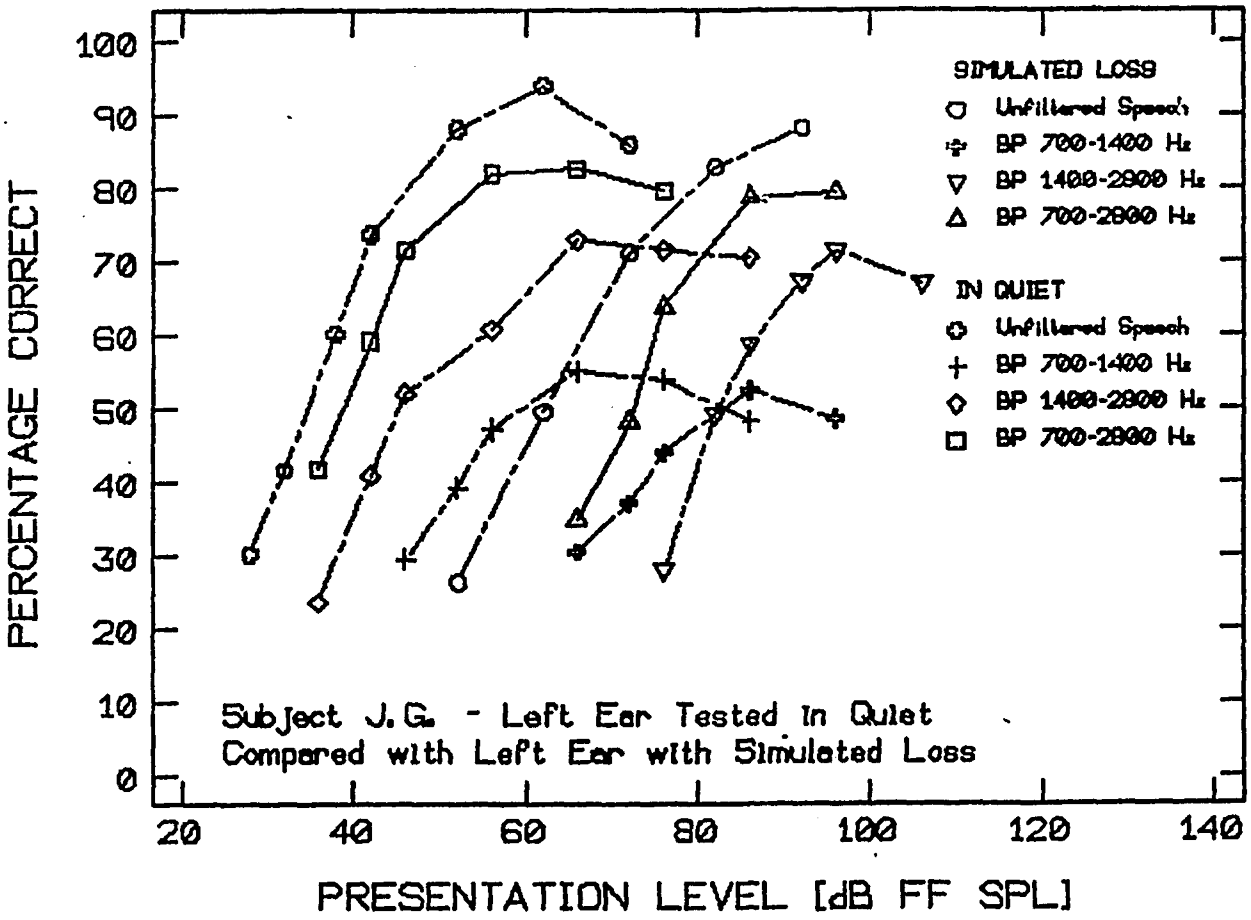


FIGURE 26

Performance - Intensity Functions:

Subject J.G. - Left Ear Tested in Quiet

Compared with Left Ear Simulated Loss

Unfiltered and Band-Pass Filtered Speech

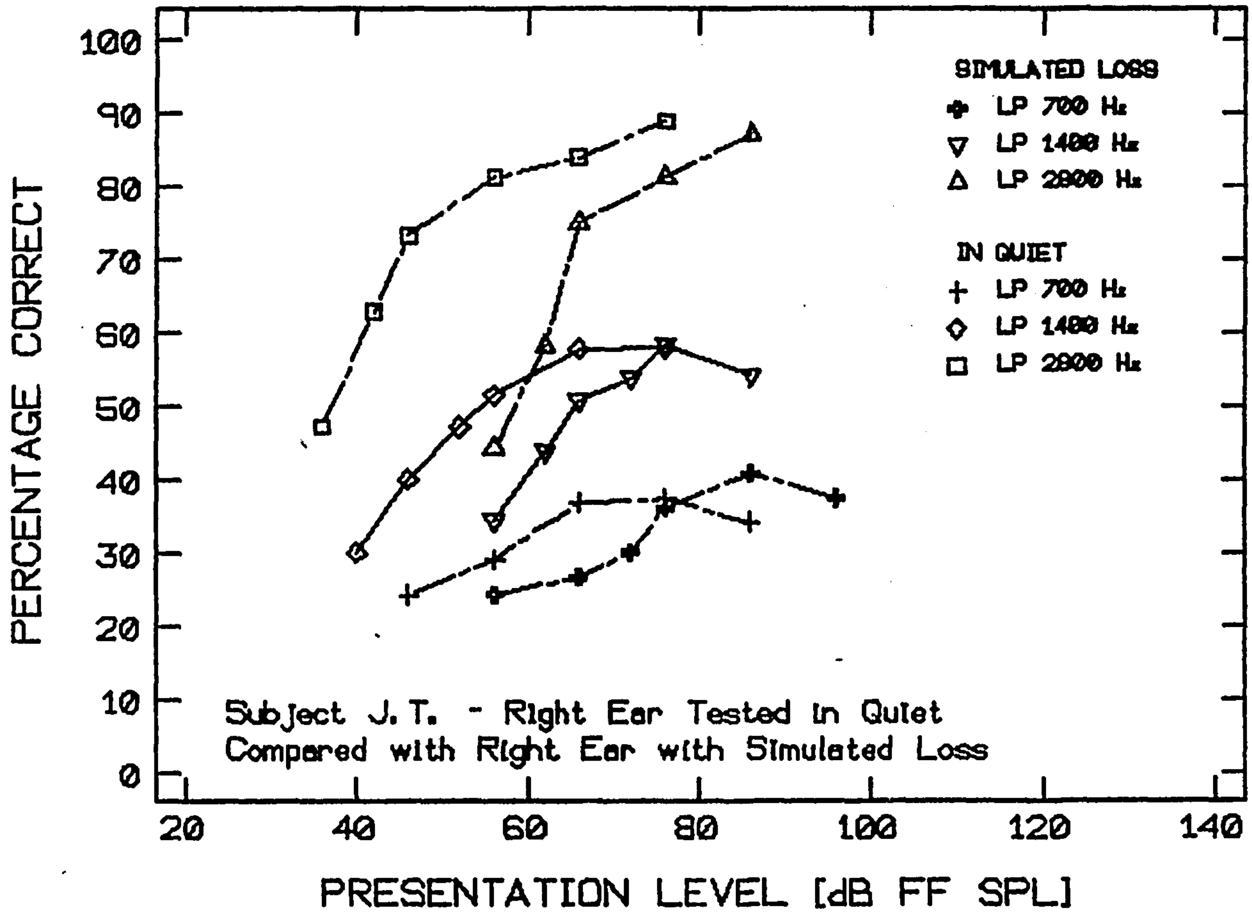


FIGURE 27

Performance - Intensity Functions:

Subject J.T. - Right Ear Tested in Quiet

Compared with Right Ear Simulated Loss

Low-Pass Filtered Speech

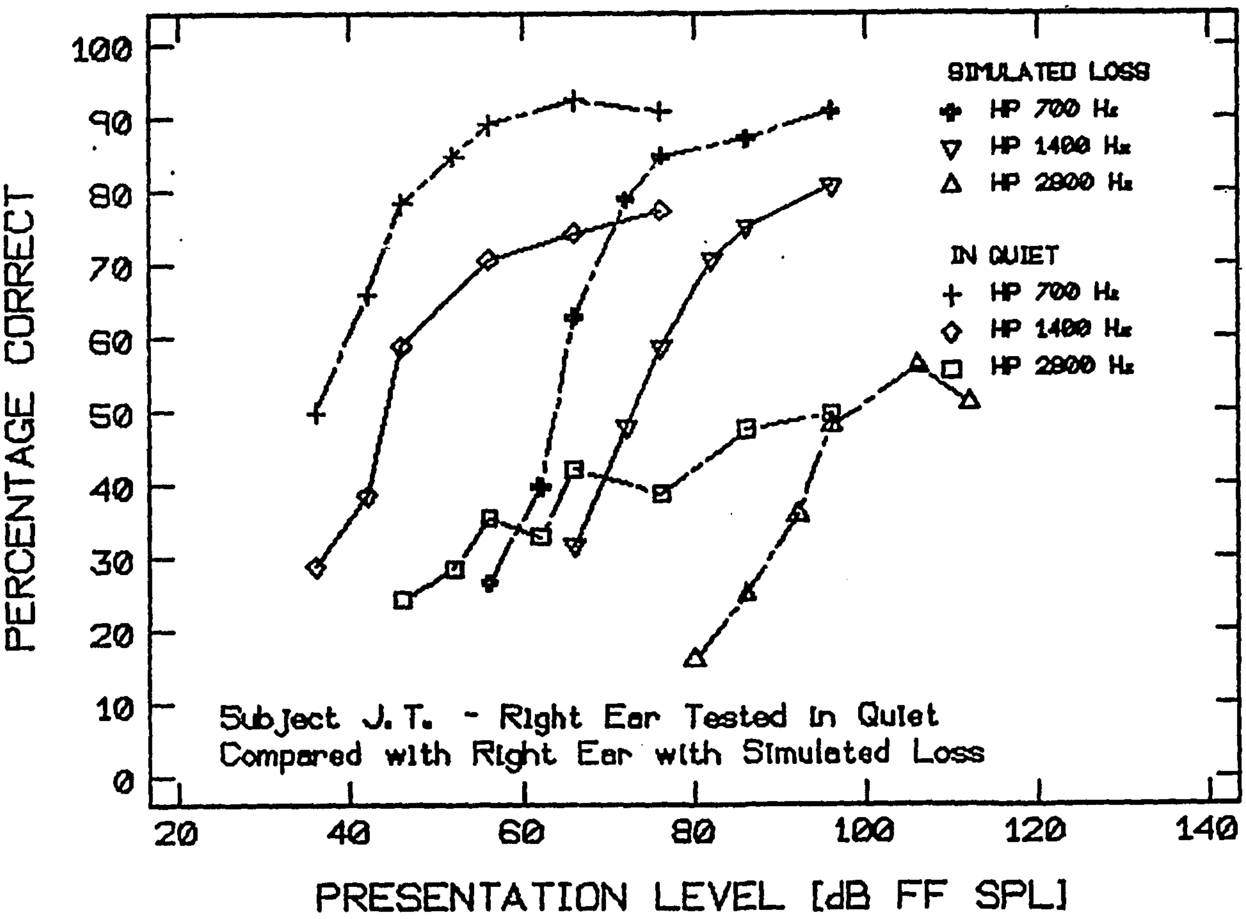


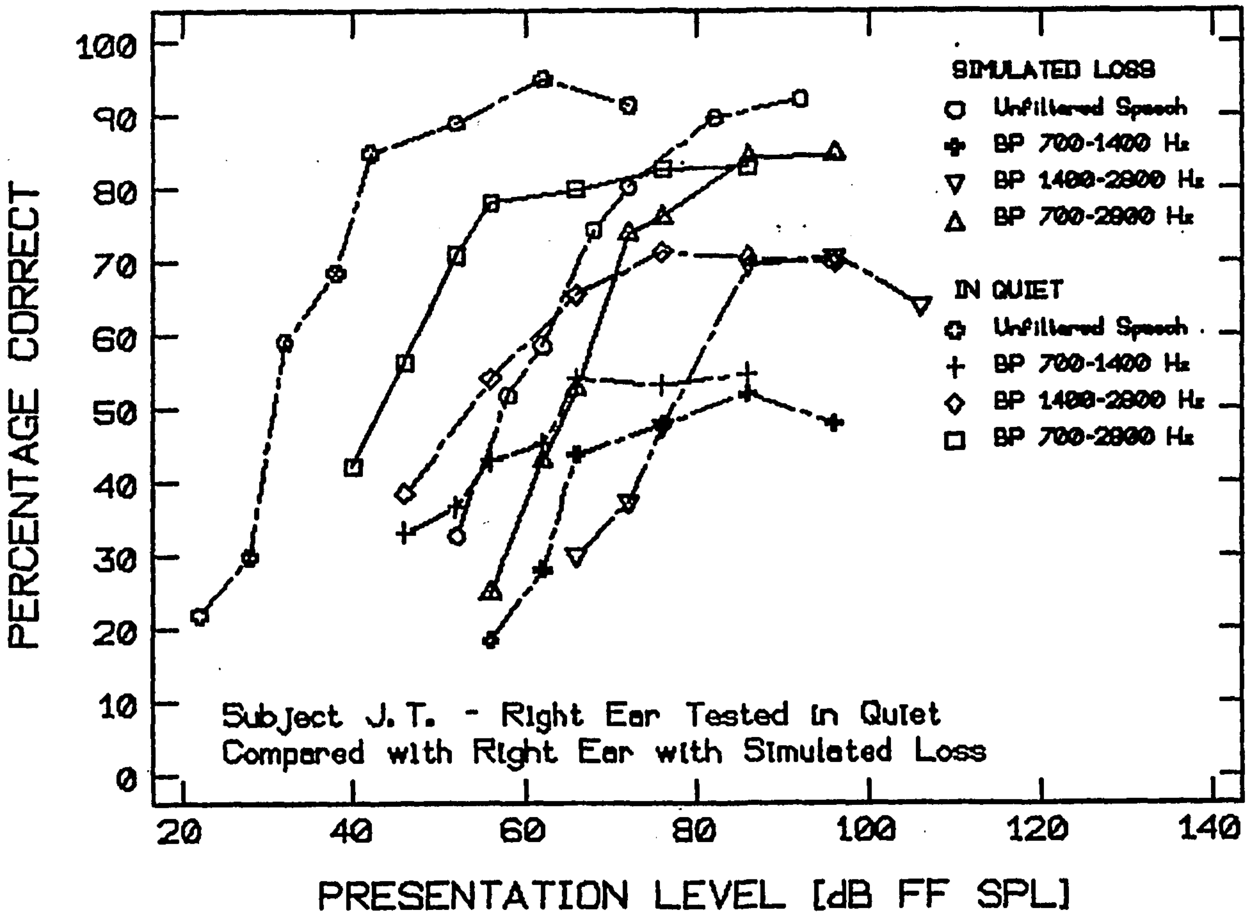
FIGURE 28

Performance - Intensity Functions:

Subject J.T. - Right Ear Tested in Quiet

Compared with Right Ear Simulated Loss

High-Pass Filtered Speech



<u>FIGURE 29</u>

Performance - Intensity Functions:

Subject J.T. - Right Ear Tested in Quiet

Compared with Right Ear Simulated Loss

Unfiltered and Band-Pass Filtered Speech

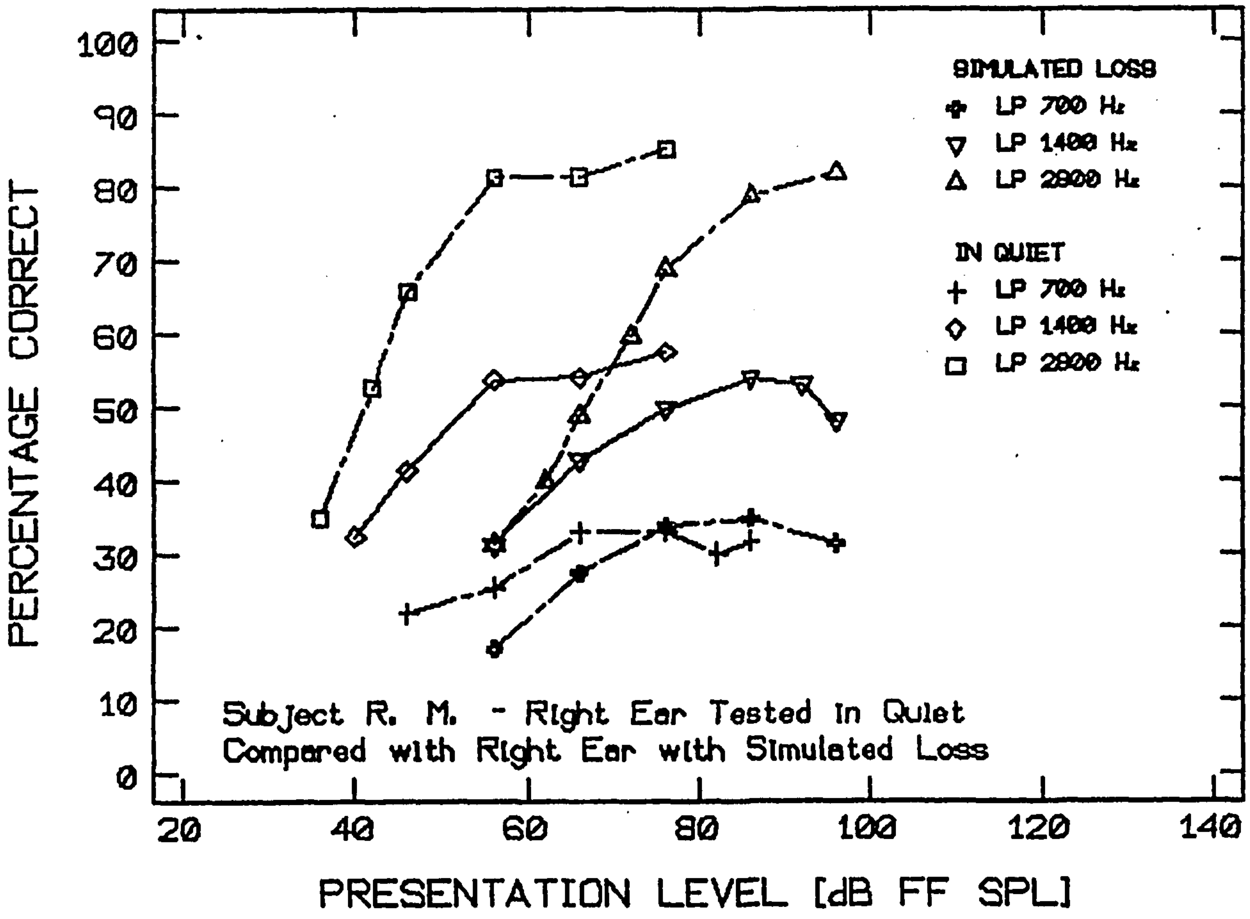


FIGURE 30

Performance - Intensity Functions:

Subject R.M. - Right Ear Tested in Quiet

Compared with Right Ear Simulated Loss

Low-Pass Filtered Speech

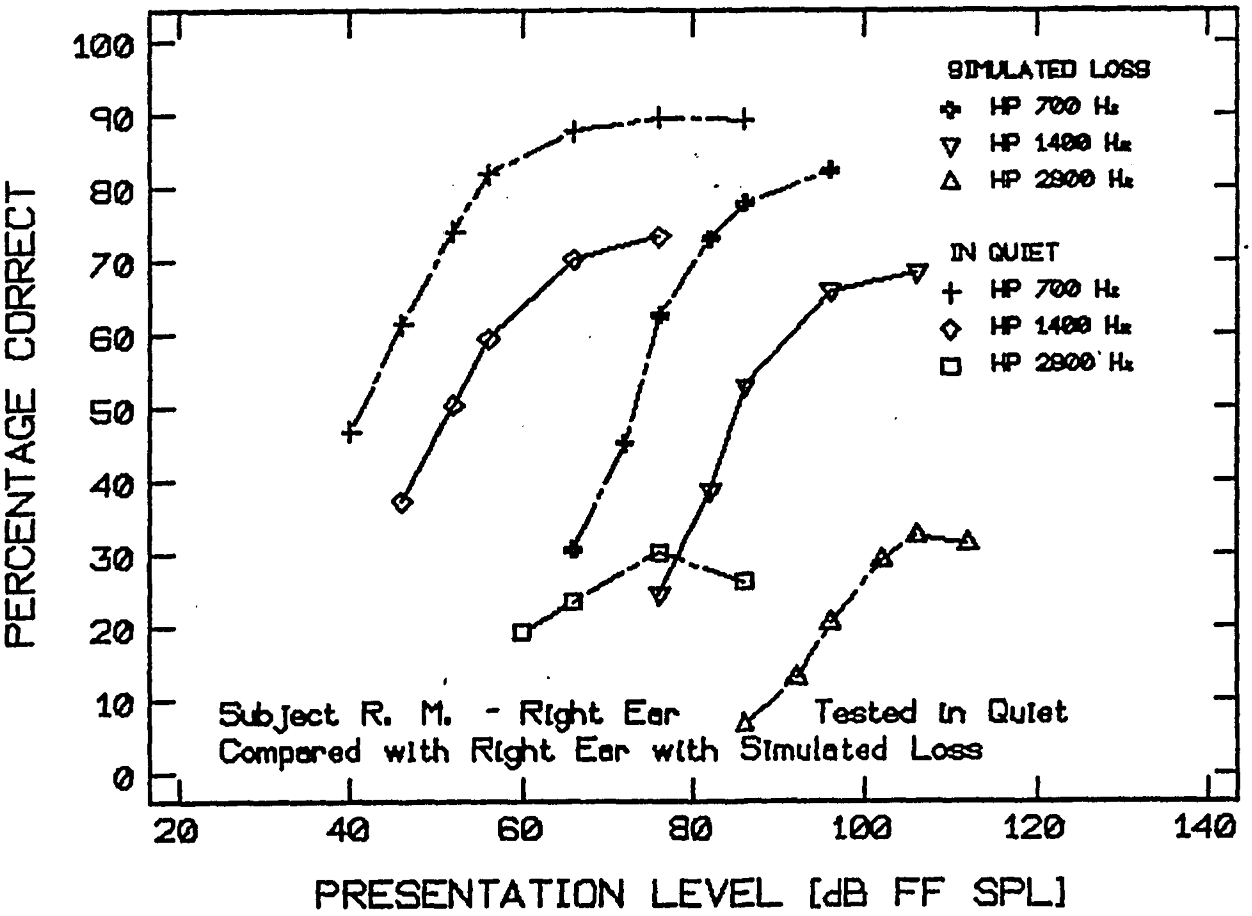


FIGURE 31

Performance - Intensity Functions:

Subject R.M. - Right Ear Tested in Quiet

Compared with Right Ear Simulated Loss

High-Pass Filtered Speech

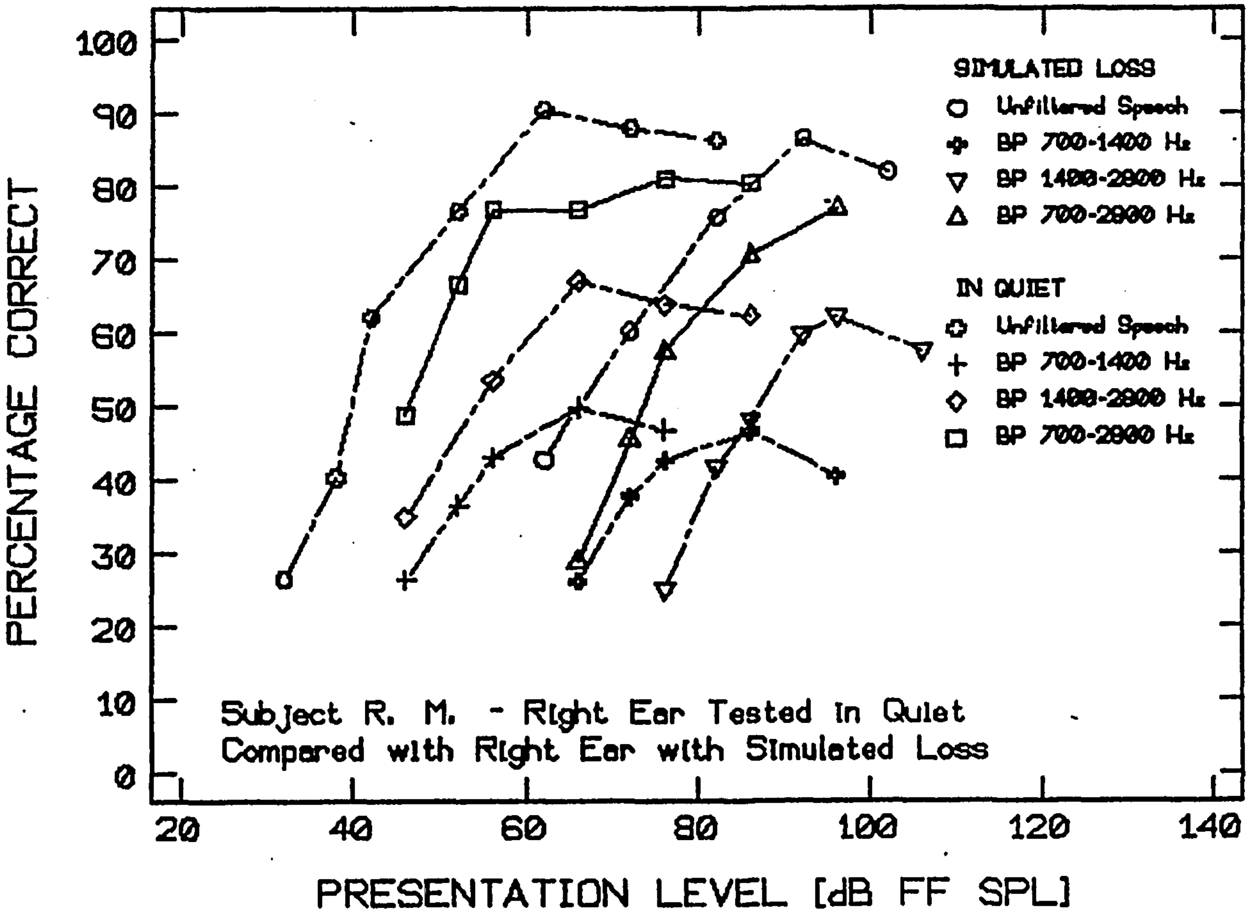


FIGURE 32

Performance - Intensity Functions:

Subject R.M. - Right Ear Tested in Quiet

Compared with Right Ear Simulated Loss

Unfiltered and Band-Pass Filtered Speech

NORMALLY-HEARING LISTENERS WITH SIMULATED LOSS - COMPARISON TO HEARING-IMPAIRED LISTENERS

In general, the performanace-intensity functions of the simulated hearing loss conditions appear to be more like those of the flatter loss subjects, J.G., T.T. and G.M.'s right ear rather than the sloping high frequency losses of F.G. or G.M.'s left ear. Comparison of the results for J.G. with performance in her impaired right ear is seen in Figures 33, 34 and 35. Higher scores occurred for all conditions with the simulated loss, even though the masked pure-tone thresholds and the hearing loss were closely matched. Rollover did not occur for the wider band conditions but did occur for HP 1400 Hz, BP 700-1400 Hz and BP 1400-2800 Hz. For the HP 1400 Hz filter, rollover was severe. However, the significance of this is not well understood.

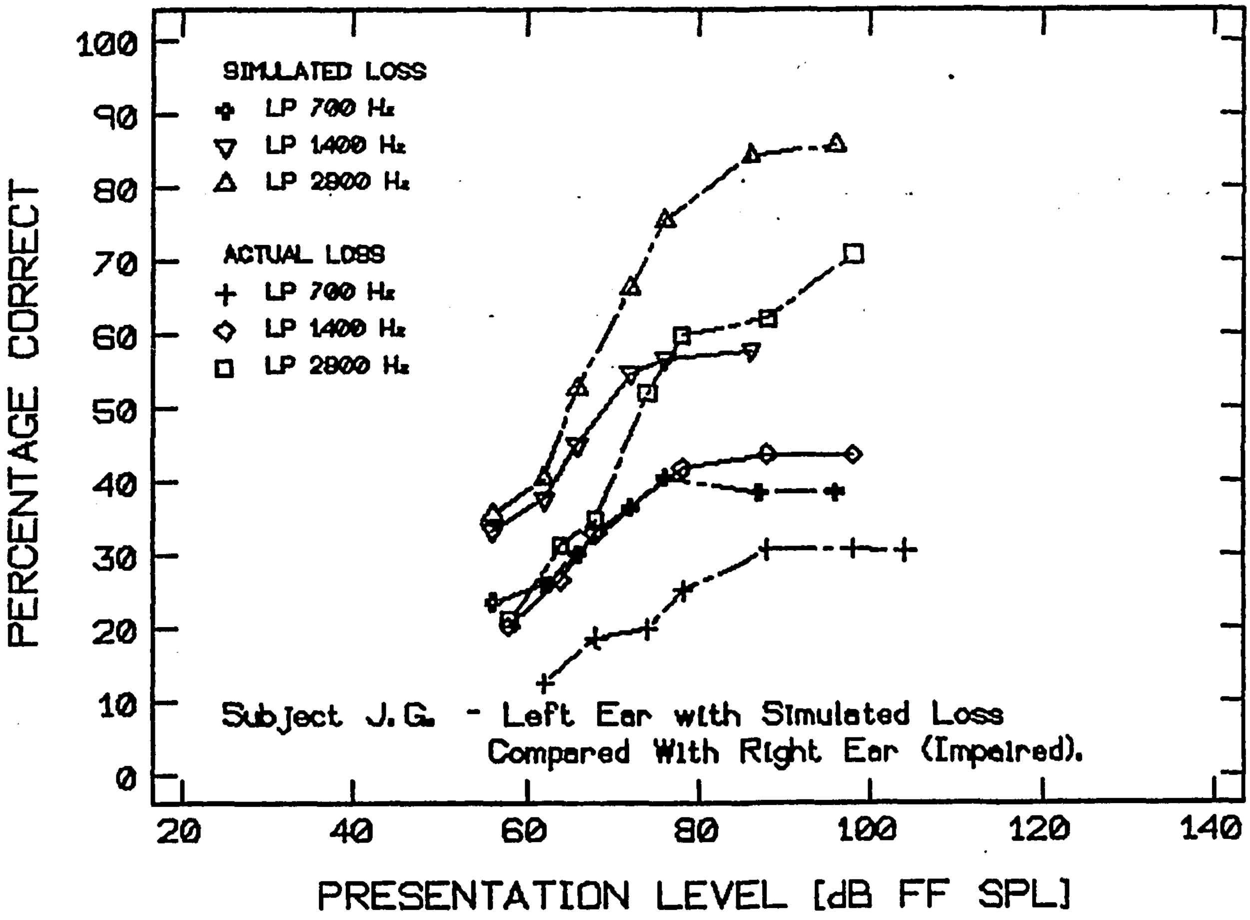


FIGURE 33

Performance - Intensity Functions:

Subject J.G. - Left Ear with Simulated Loss

Compared to Right Ear Actual Loss

Low-Pass Filtered Speech

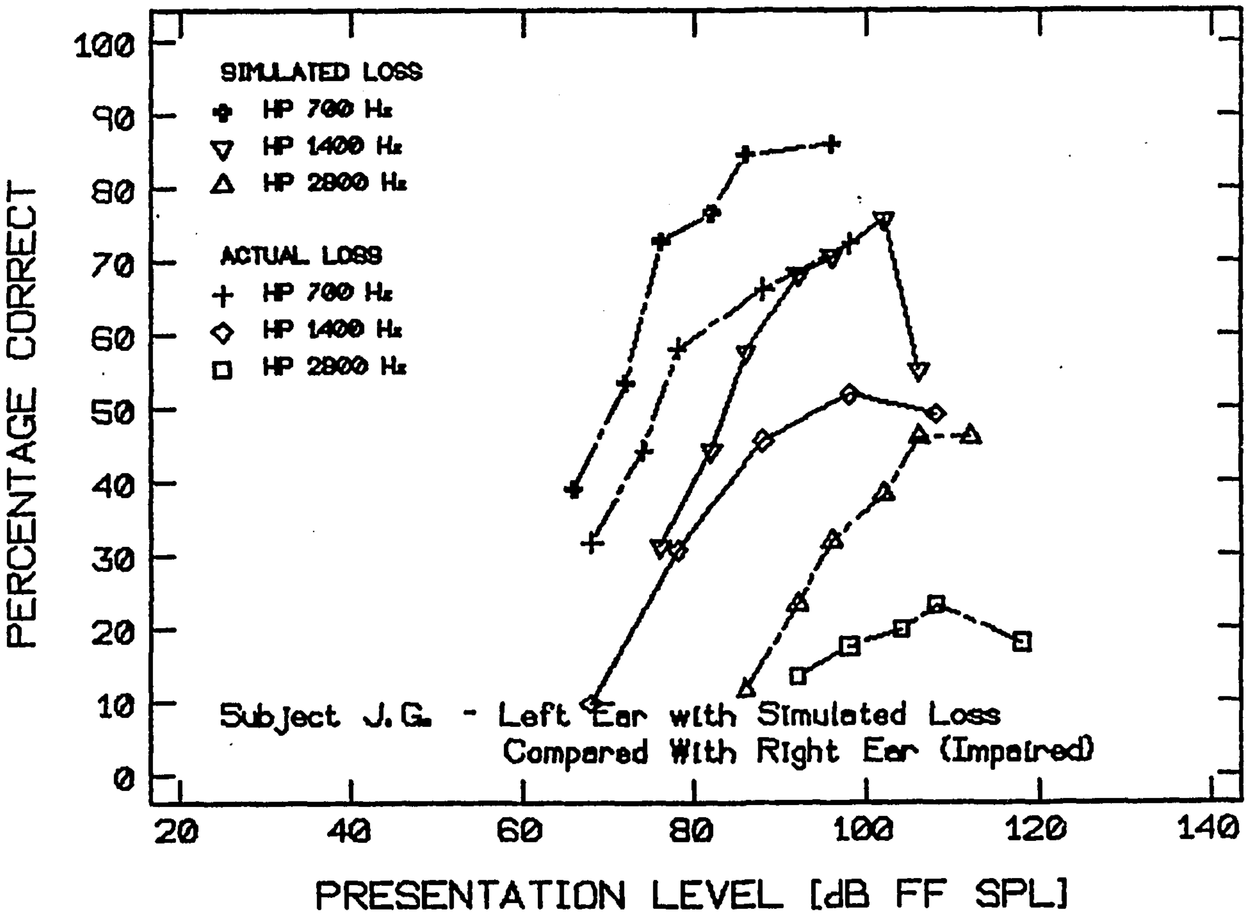


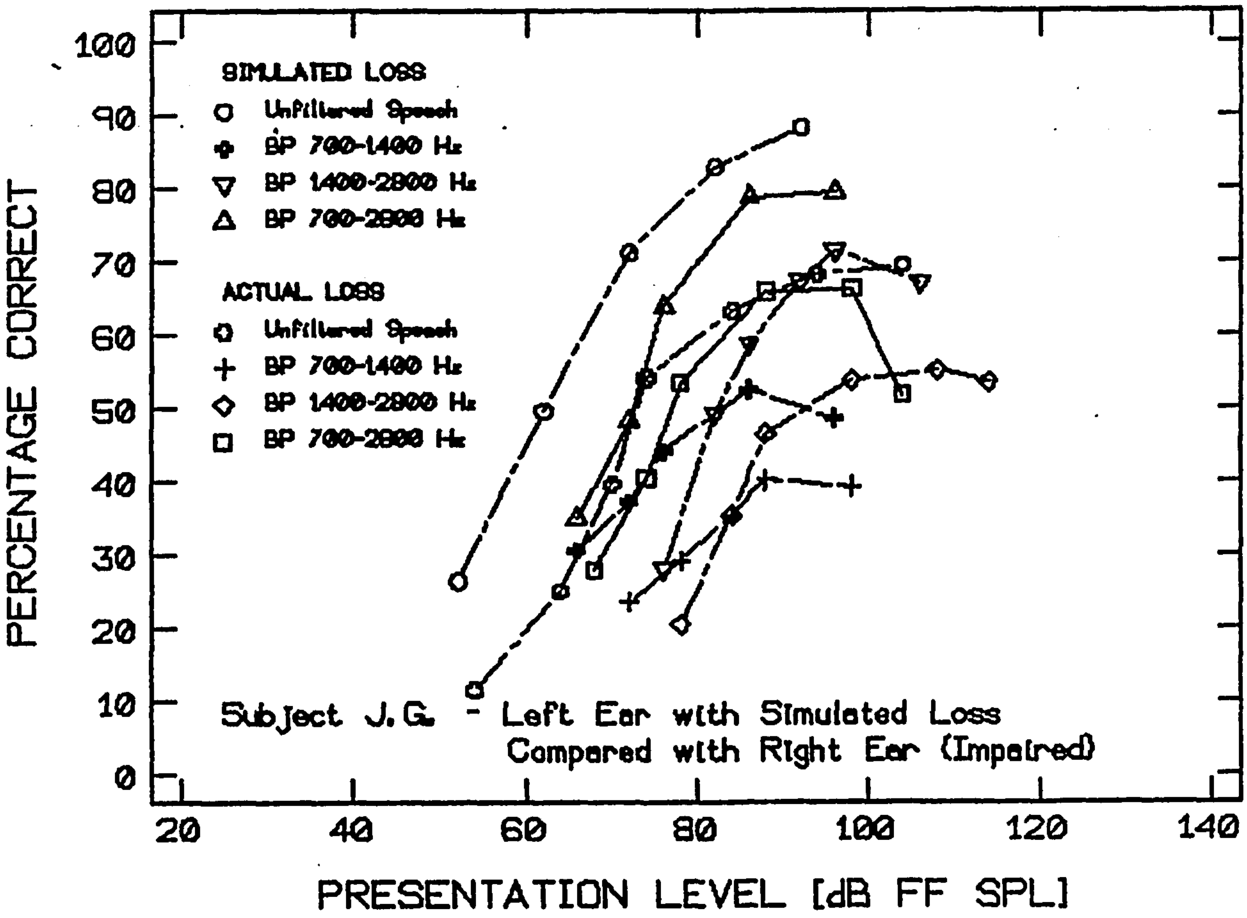
FIGURE 34

Performance - Intensity Functions:

Subject J.G. - Left Ear with Simulated Loss

Compared to Right Ear Actual Loss

High-Pass Filtered Speech



<u>FIGURE 35</u>

Performance - Intensity Functions:

Subject J.G. - Left Ear with Simulated Loss

Compared to Right Ear Actual Loss

Unfiltered and Band-Pass Filtered Speech

Figures 36, 37 and 38 compare the results of J.T. with simulated hearing loss to J.G.'s right ear, which was similar in configuration of simulated vs. actual hearing loss. In every case, J.T.'s intelligibility scores were higher than J.G.'s. These differences are partly accounted for by fact that J.T.'s simulated loss was not as great as the actual loss. For J.T,. rollover occurred as for J.G. in the LP 700 Hz, BP 700-1400 Hz, and BP 1400-2800 Hz bands. Rollover also took place for the HP 2800 Hz condition. It did not, however, take place for the HP 1400 Hz band.

Figures 39, 40 and 41 compare R.M.'s simulated loss performance with G.M.'s left ear, which her simulated loss was selected to match. In every case, R.M.'s performance at low presentation levels was poorer than G.M.'s. As presentation level increased, however, R.M.'s scores increased at a much greater rate and eventually exceeded G.M.'s scores. For R.M., rollover occurred in the unfiltered speech, LP 700 Hz, LP 1400 Hz, BP 700-1400 Hz and BP 1400-2800 Hz and HP 2800 Hz bands.

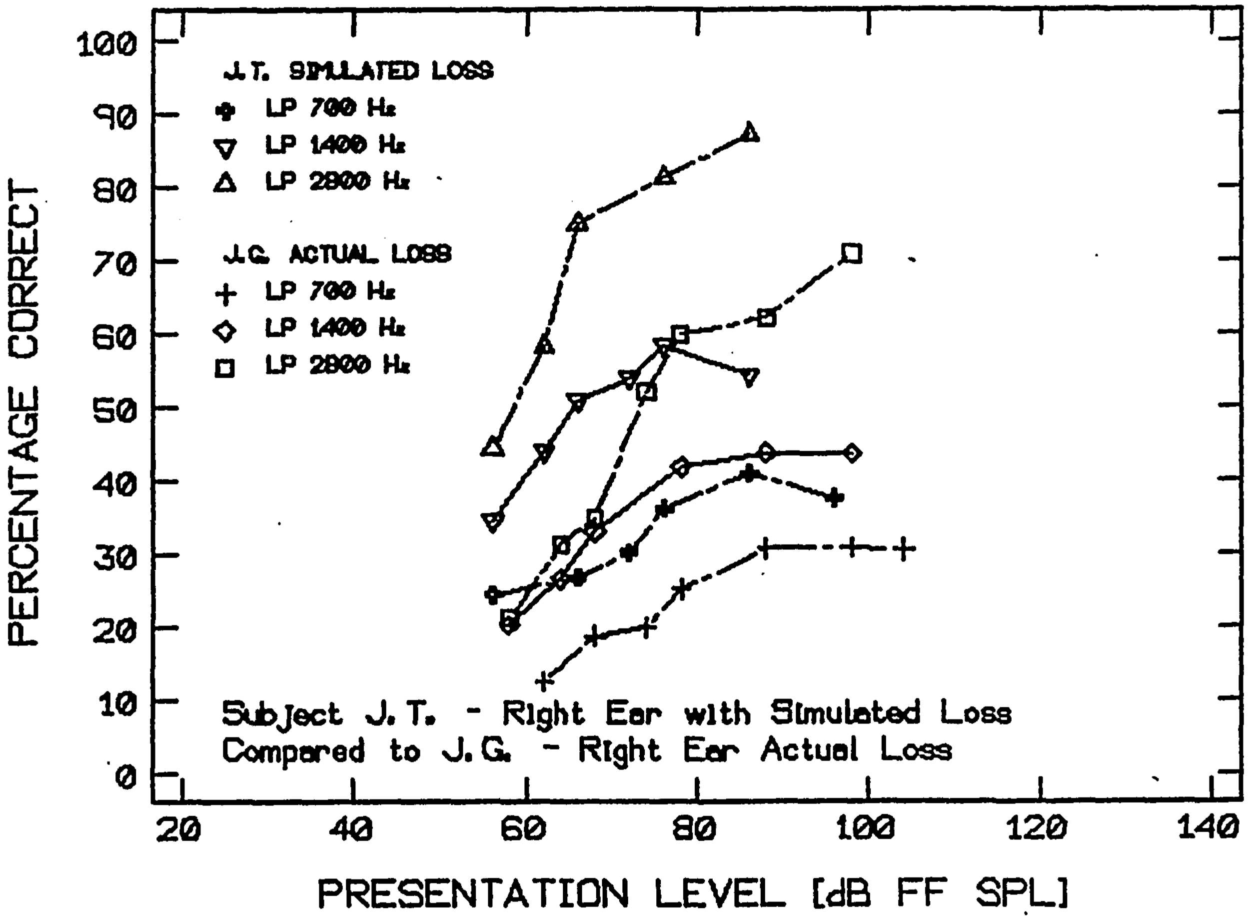


FIGURE 36

Performance - Intensity Functions:

Subject J.T. - Right Ear with Simulated Loss

Compared to J.G. - Right Ear Actual Loss

Low-Pass Filtered Speech

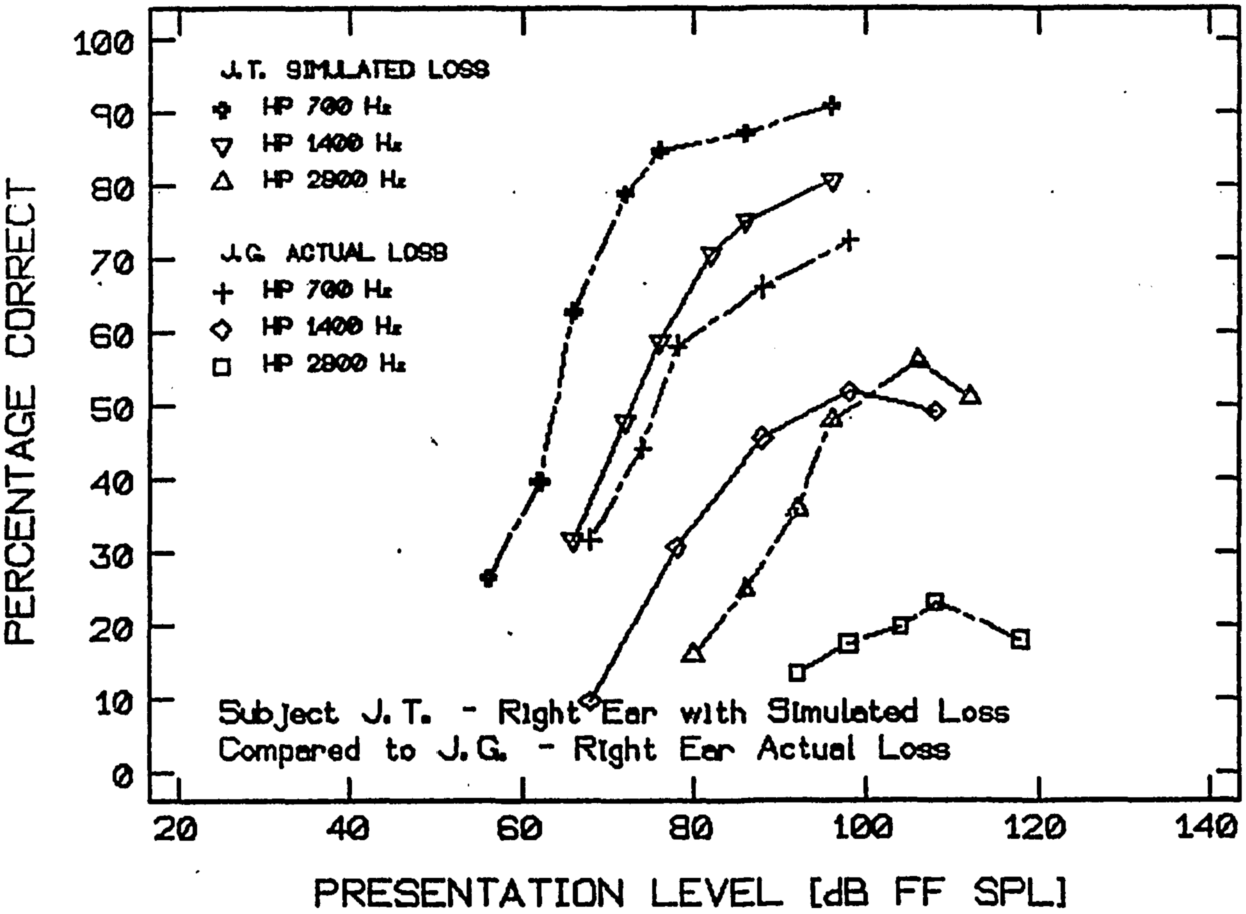


FIGURE 37

Performance - Intensity Functions:

Subject J.T. - Right Ear with Simulated Loss

Compared to J.G. - Right Ear Actual Loss

High-Pass Filtered Speech

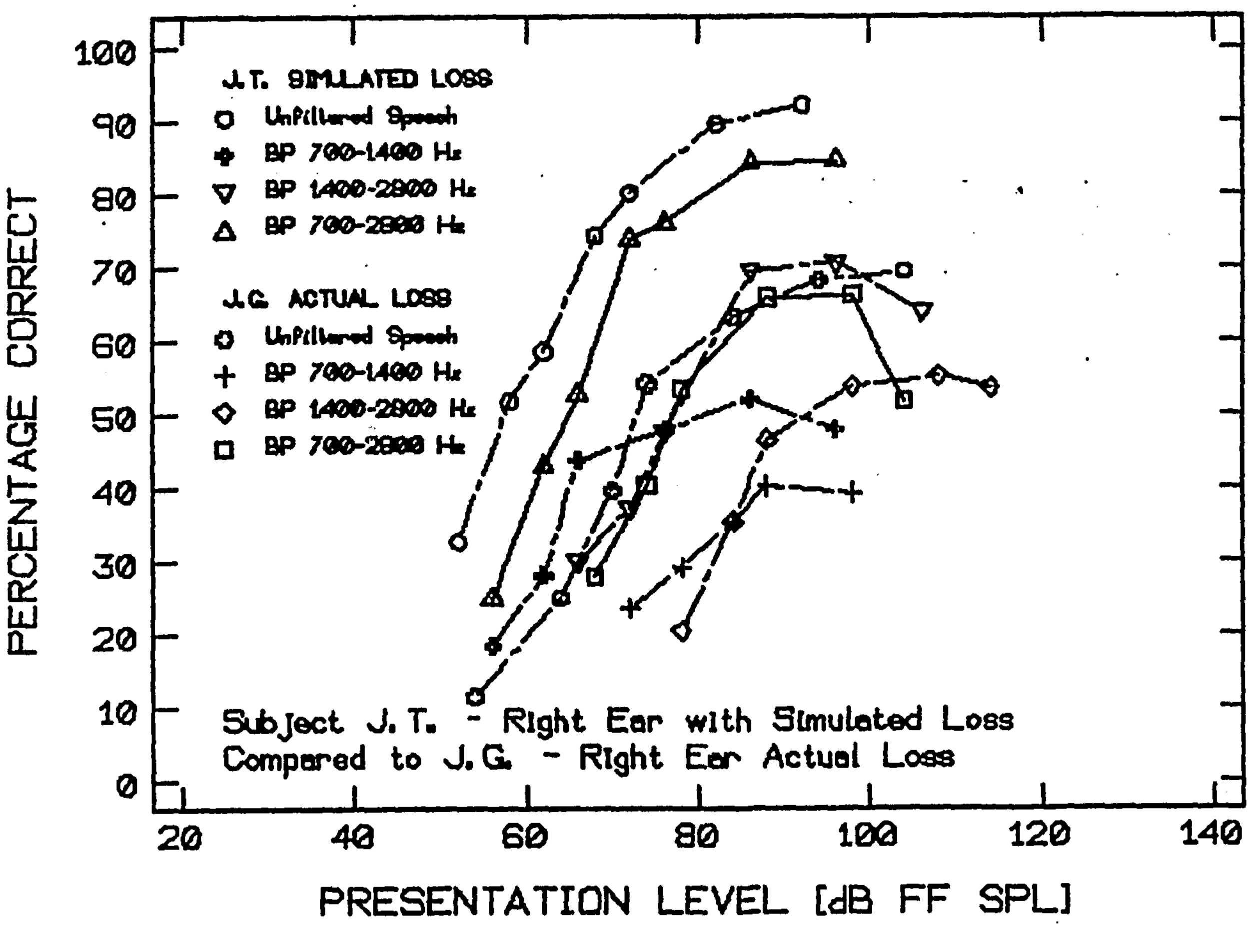


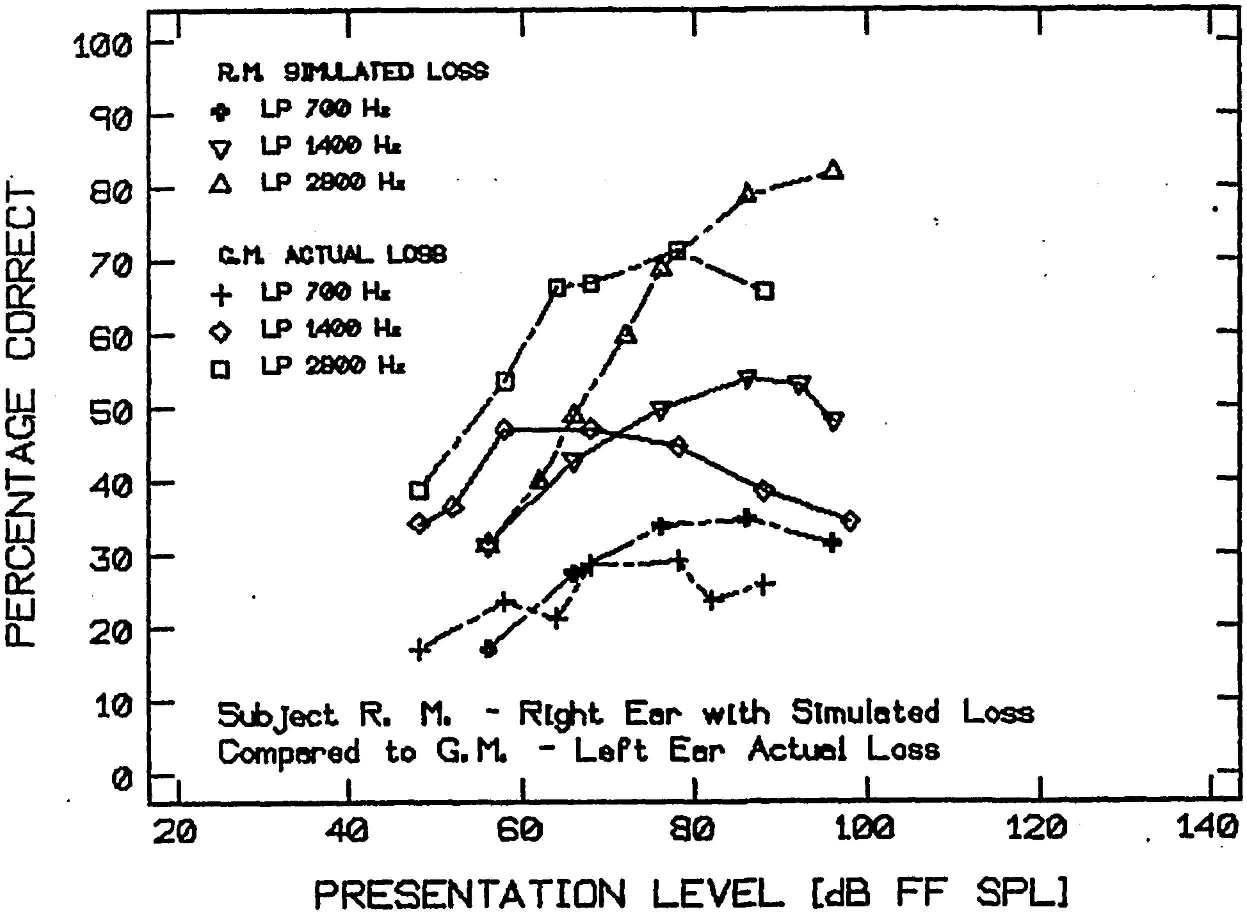
FIGURE 38

Performance - Intensity Functions:

Subject J.T. - Right Ear with Simulated Loss

Compared to J.G. - Right Ear Actual Loss

Unfiltered and Band-Pass Filtered Speech



<u>FIGURE 39</u>

Performance - Intensity Functions:

Subject R.M. - Right Ear with Simulated Loss

Compared to G.M. - Left Ear Actual Loss

Low-Pass Filtered Speech

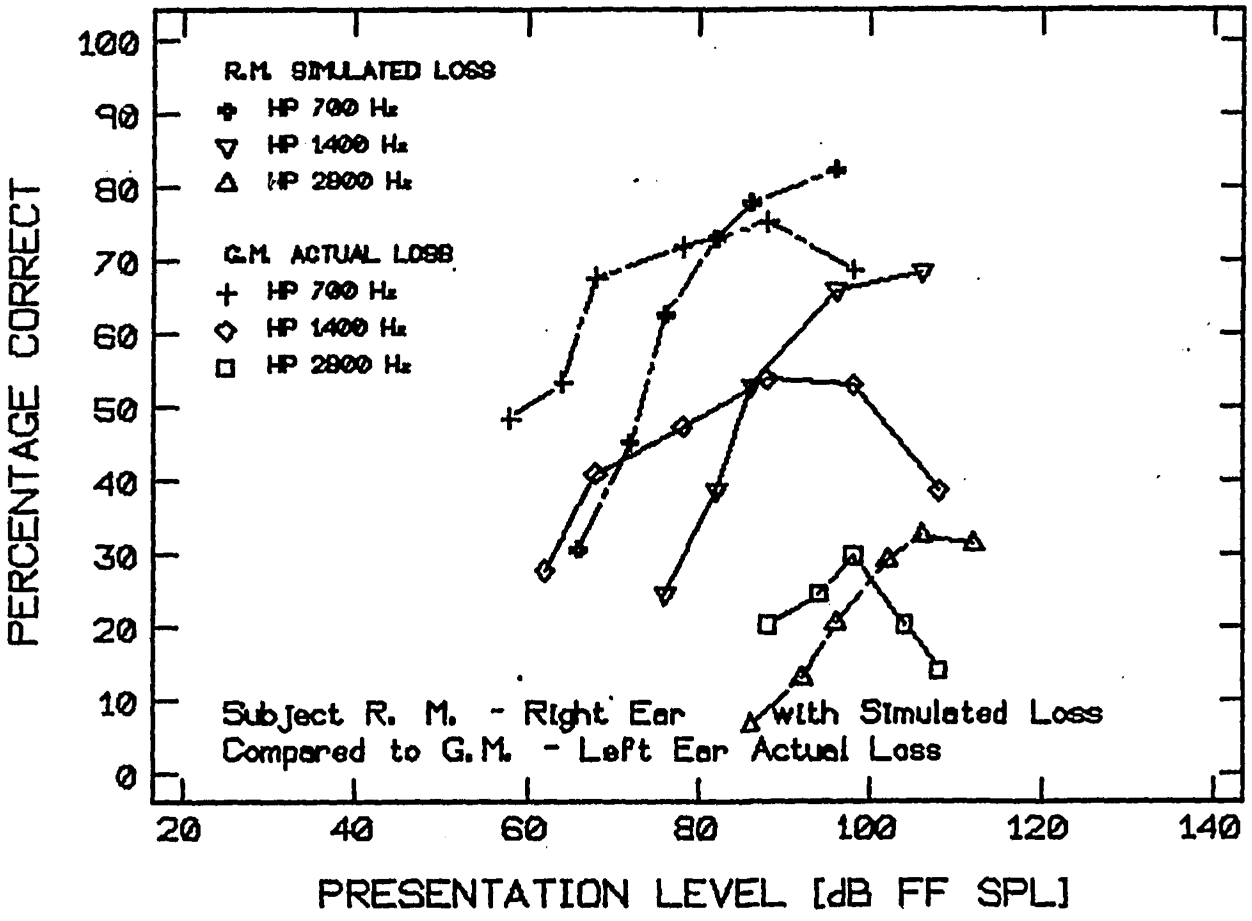


FIGURE 40

Performance - Intensity Functions:

Subject R.M. - Right Ear with Simulated Loss

Compared to G.M. - Left Ear Actual Loss

High-Pass Filtered Speech

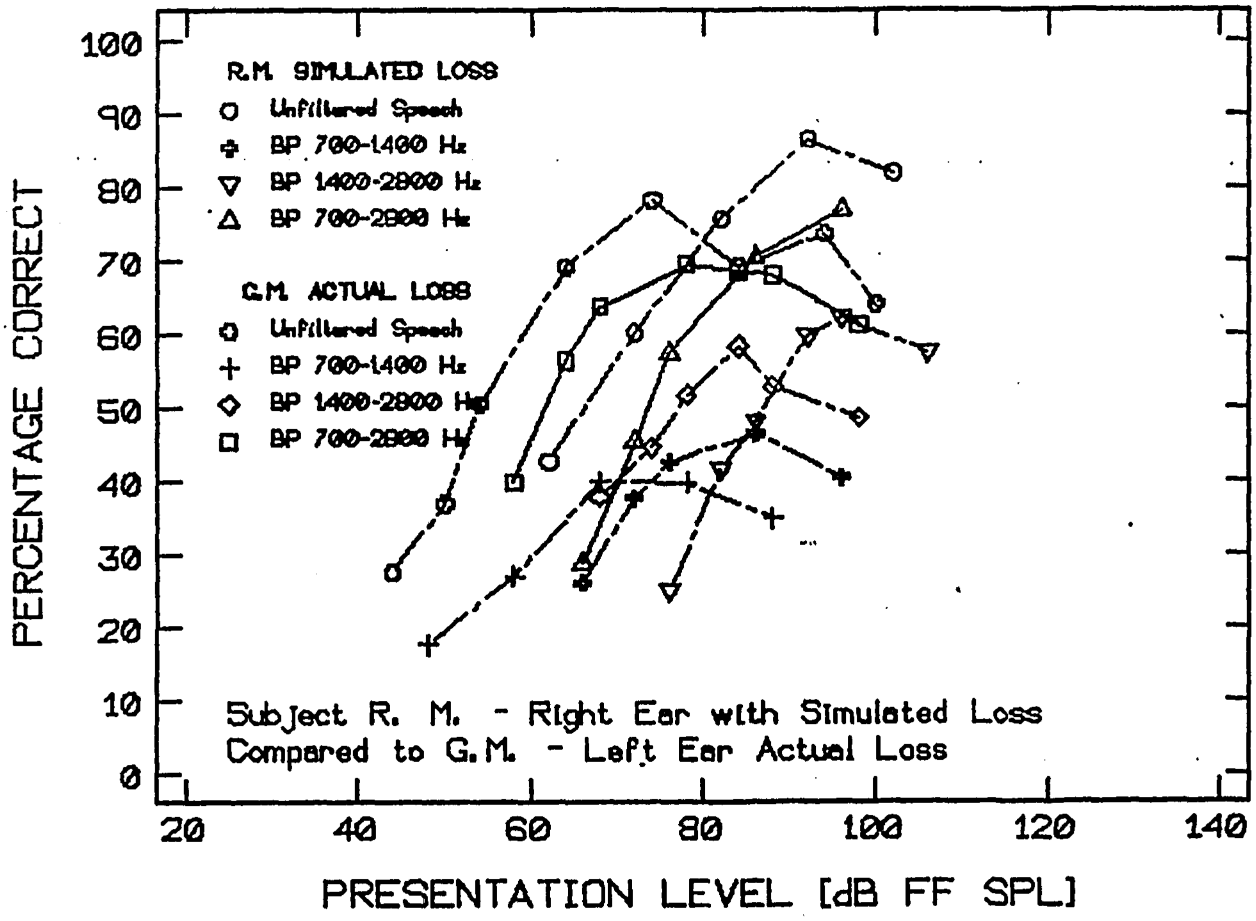


FIGURE 41

Performance - Intensity Functions:

Subject R.M. - Right Ear with Simulated Loss

Compared to G.M. - Left Ear Actual Loss

Unfiltered and Band-Pass Filtered Speech

ARTICULATION THEORY CALCULATIONS

The results of calculations of Articulation Index are shown in the following figures:

Normally-Hearing Subjects

Figure 42: Subject J.G. Left Ear
Figure 43: Subject J.T Right Ear
Figure 44: Subject R.M. Right Ear

Hearing-Impaired Subjects

Figure 45: Subject F.G. Right Ear
Figure 46: Subject F.G. Left Ear
Figure 47: Subject G.M. Right Ear
Figure 48: Subject G.M. Left Ear
Figure 49: Subject J.G Right Ear
Figure 50: Subject T.T. Right Ear

Normally-Hearing Listeners with Simulated Loss

Figure 51: Subject J.G. Left Ear
Figure 52: Subject J.T. Right Ear
Figure 53: Subject R.M. Right Ear

The graphs show observed percentage correct syllable identification plotted against the calculated Articulation Index. In performing the calculations of Articulation Index, a proficiency factor of unity was used for the listeners and proficiency factors of one or slightly less than one were used for the talkers, as described in Chapter III. The calculated values of A.I. presented here and in Appendix E are the arithmetic average of the individual values for each talker. Also plotted on each graph is a reference curve of percentage correct vs. Articulation Index derived from the curves of Figure 1 by interpolating a curve between the PB 1000 word performance curve and the nonsense syllable curve. The curve was obtained from a seventh order polynomial regression using a least-squares method of determination. Although the reference curve shown may not be appropriate for the speech materials used in this study, it is presented in these graphs to provide a reference from which comparative assessments of the data may be made. Points that lie to the left or above the curve shown imply that the performance of the subject for that condition is better than would be predicted by Articulation Theory using the curve plotted. Points lying to the right or below the curve, therefore, are points that are lower than would be predicted by the Theory using this curve.

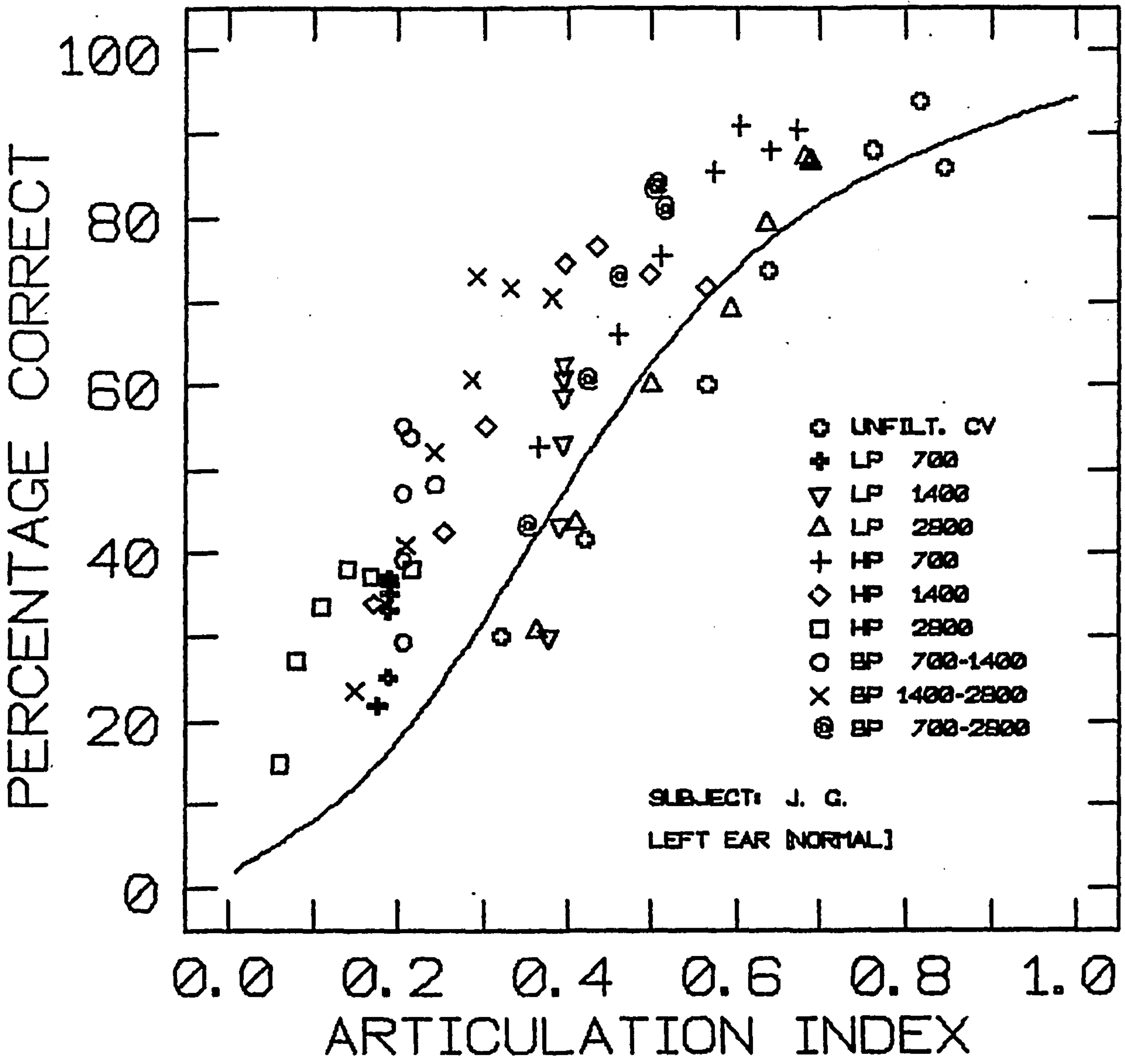
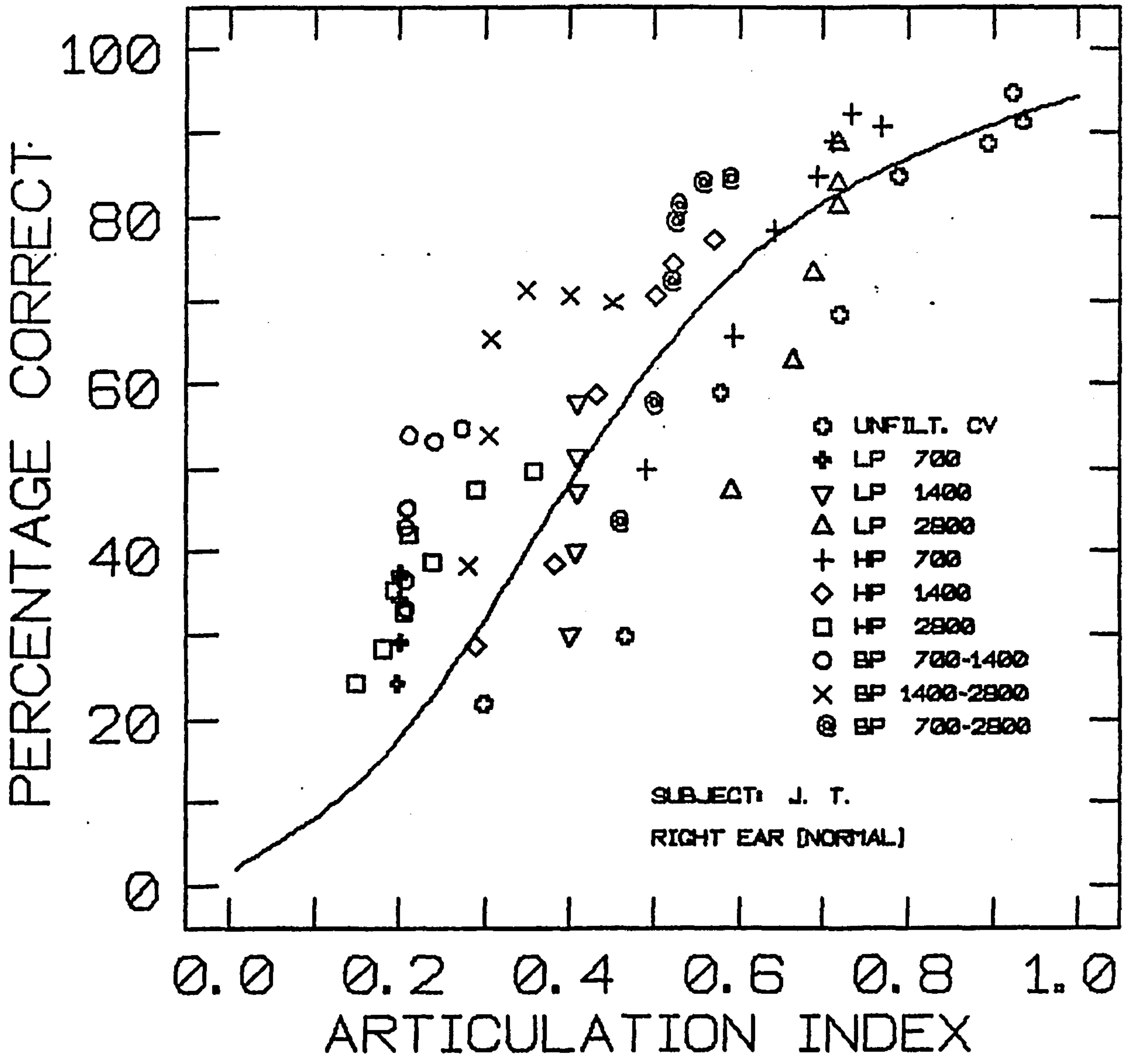


FIGURE 42

Articulation Index Functions

Subject J.G. - Left Ear [Normally-Hearing]



<u>FIGURE 43</u>

Articulation Index Functions

Subject J.T. - Right Ear [Normally-Hearing]

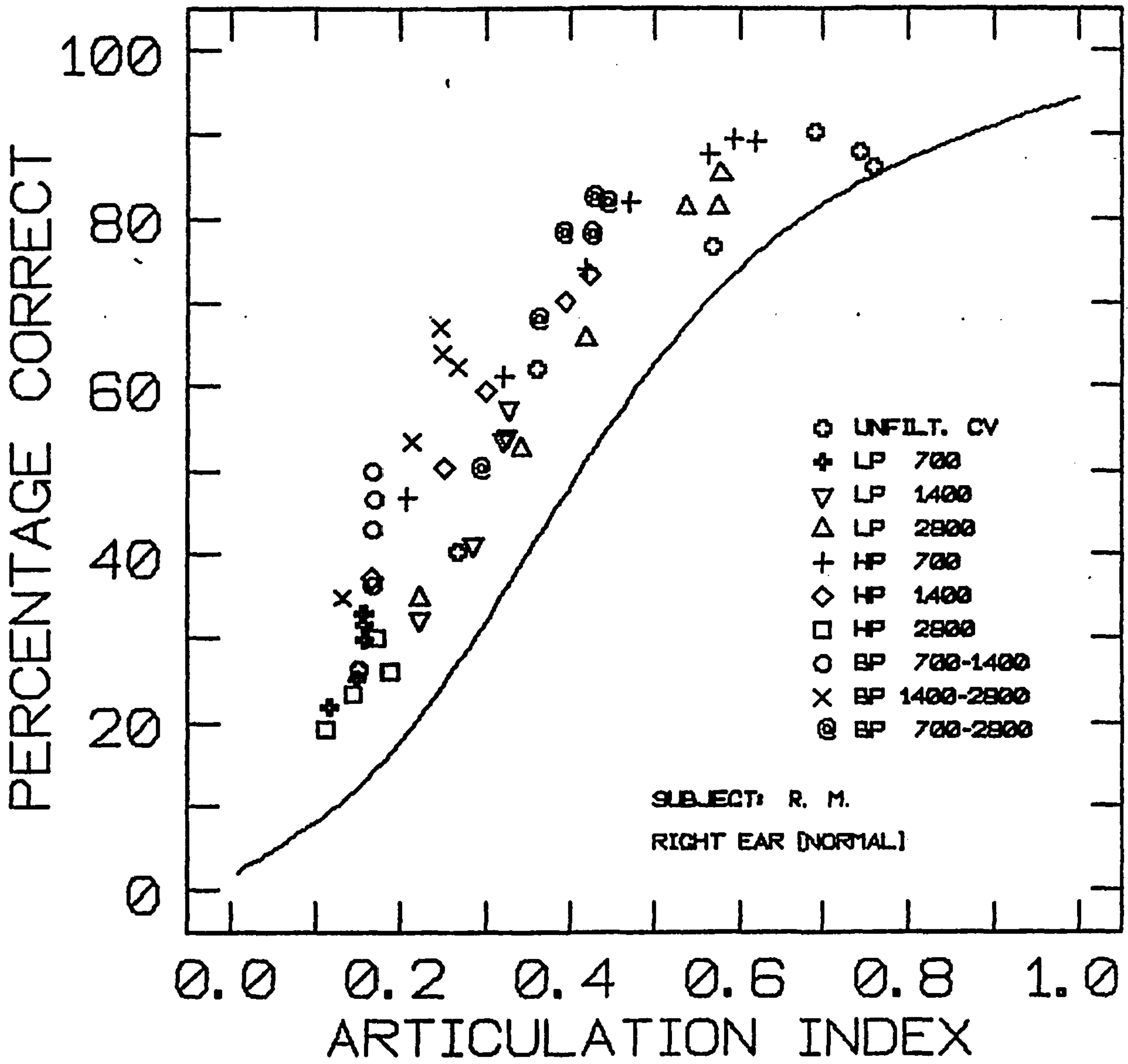


FIGURE 44

Articulation Index Functions

Subject R.M. - Right Ear [Normally-Hearing]

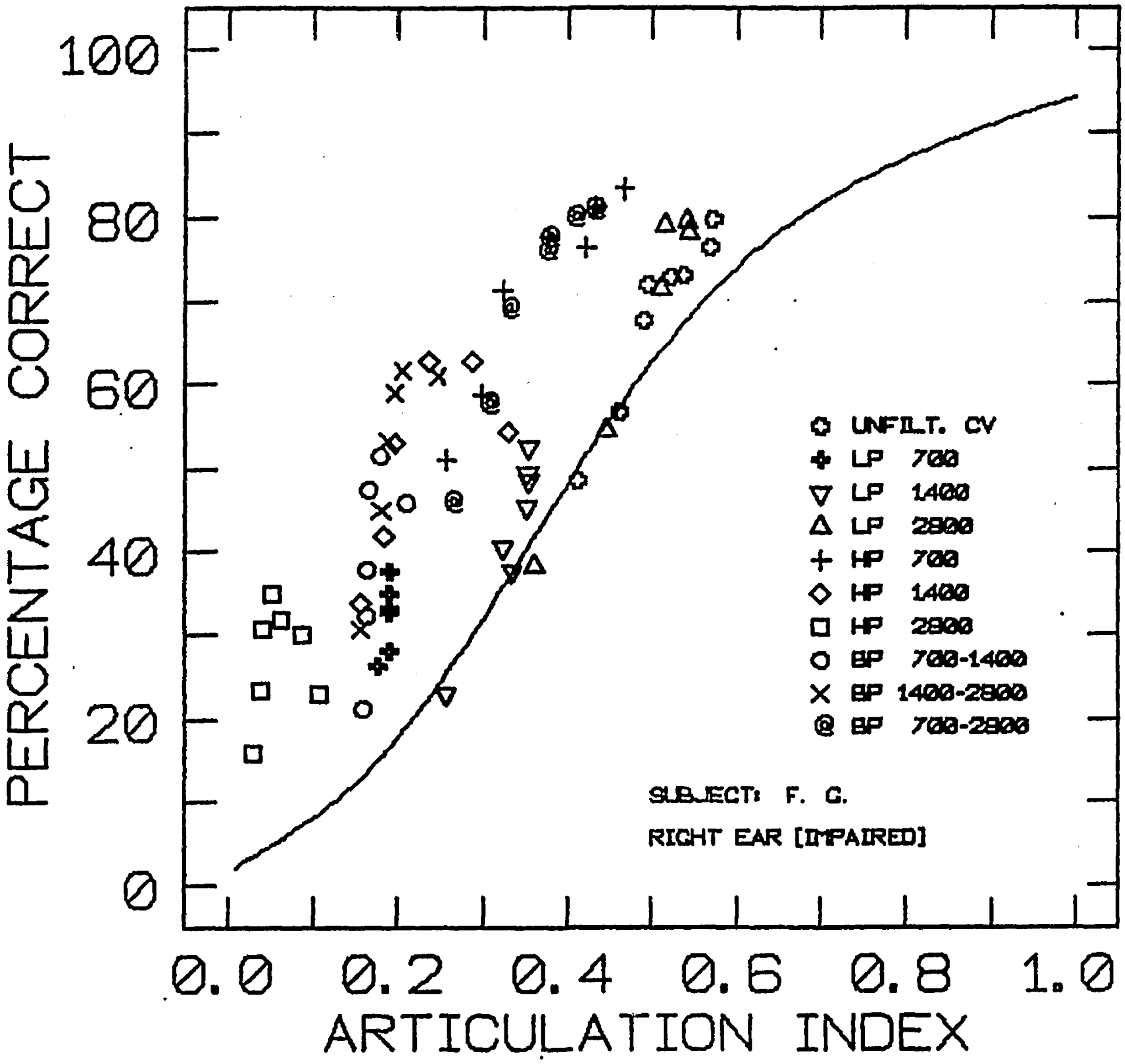


FIGURE 45

Articulation Index Functions

Subject F.G. - Right Ear [Hearing-Impaired]

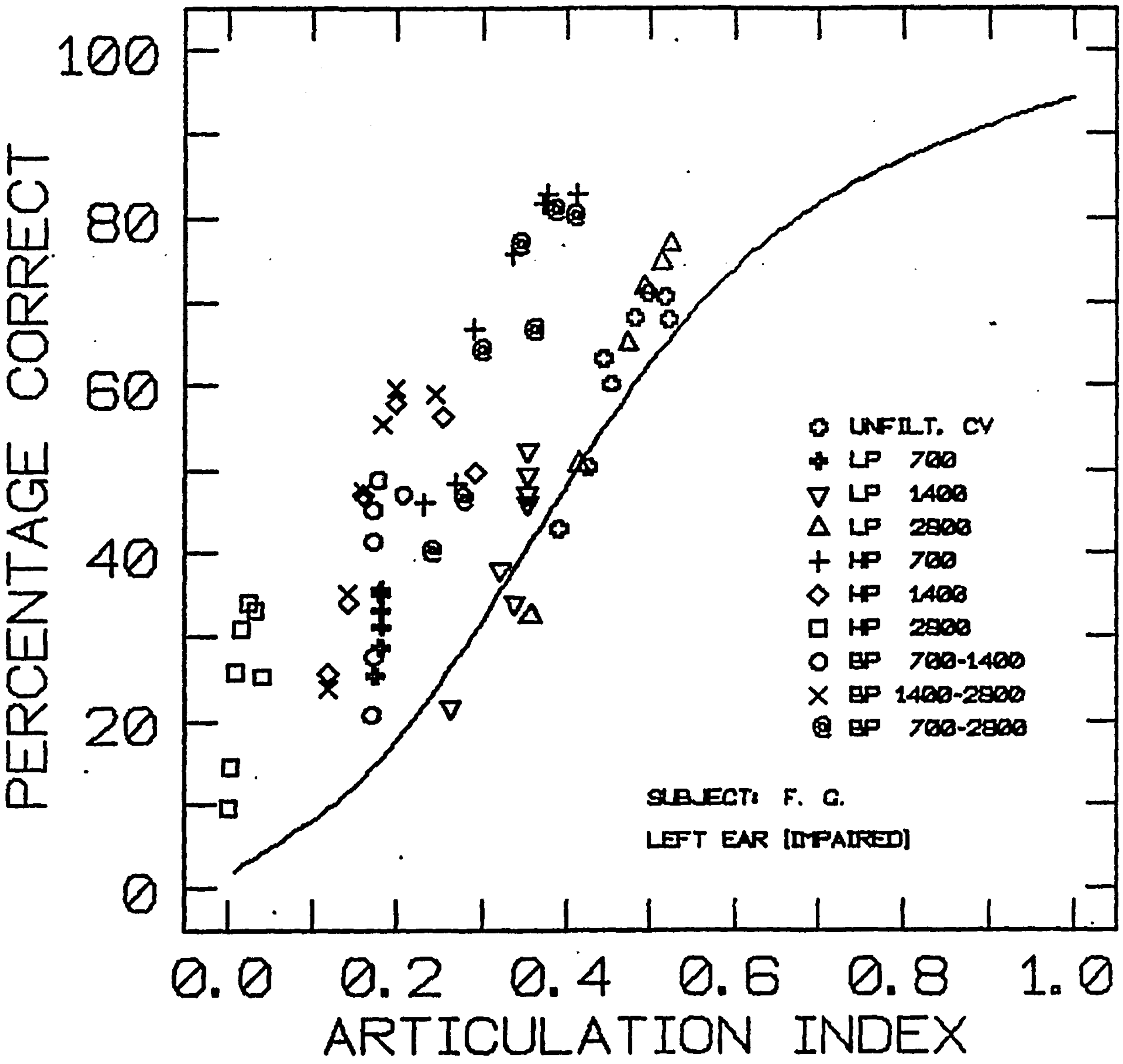


FIGURE 46

Articulation Index Functions

Subject F.G. - Left Ear [Hearing-Impaired]

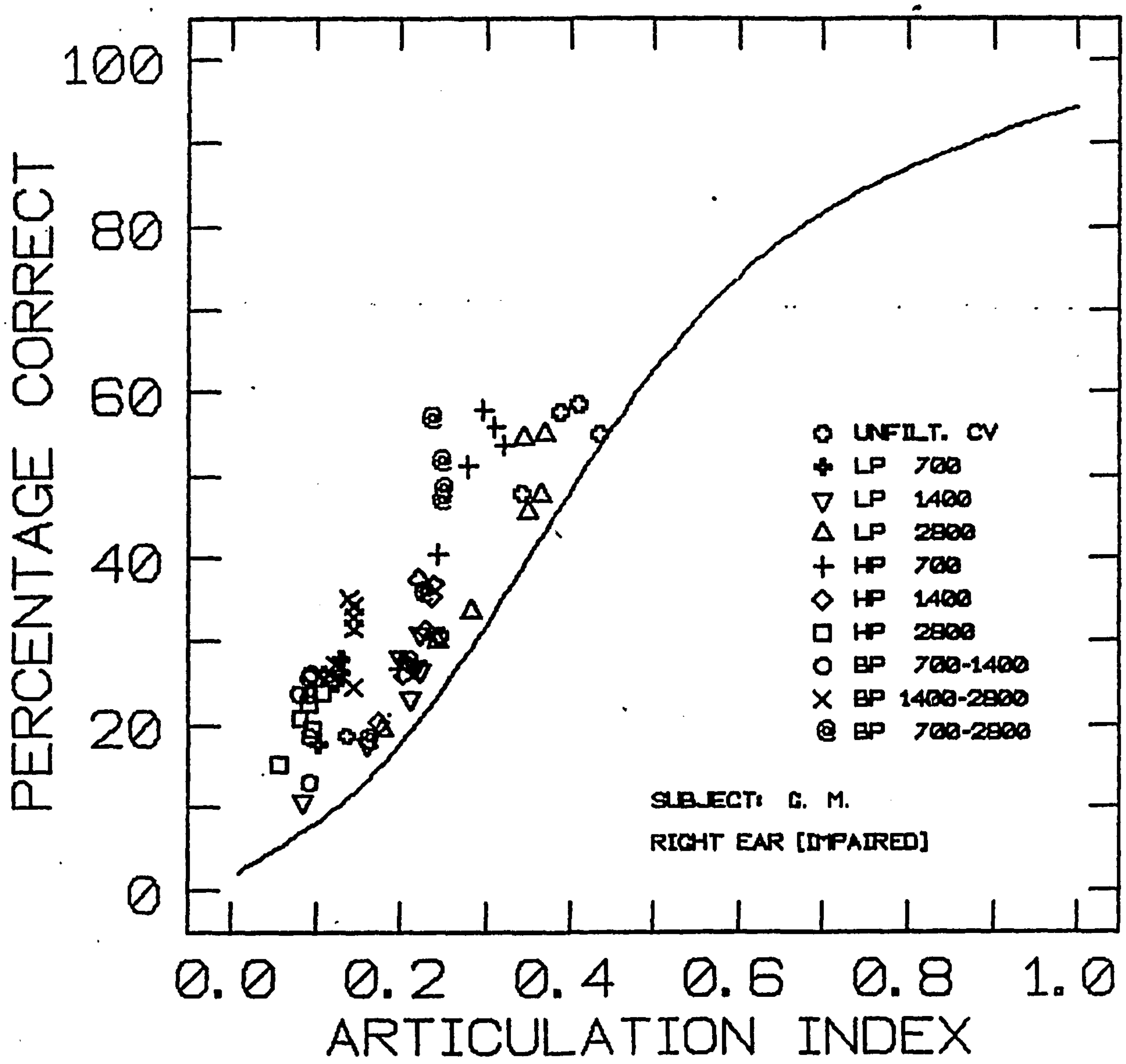


FIGURE 47

Articulation Index Functions

Subject G.M. - Right Ear [Hearing-Impaired]

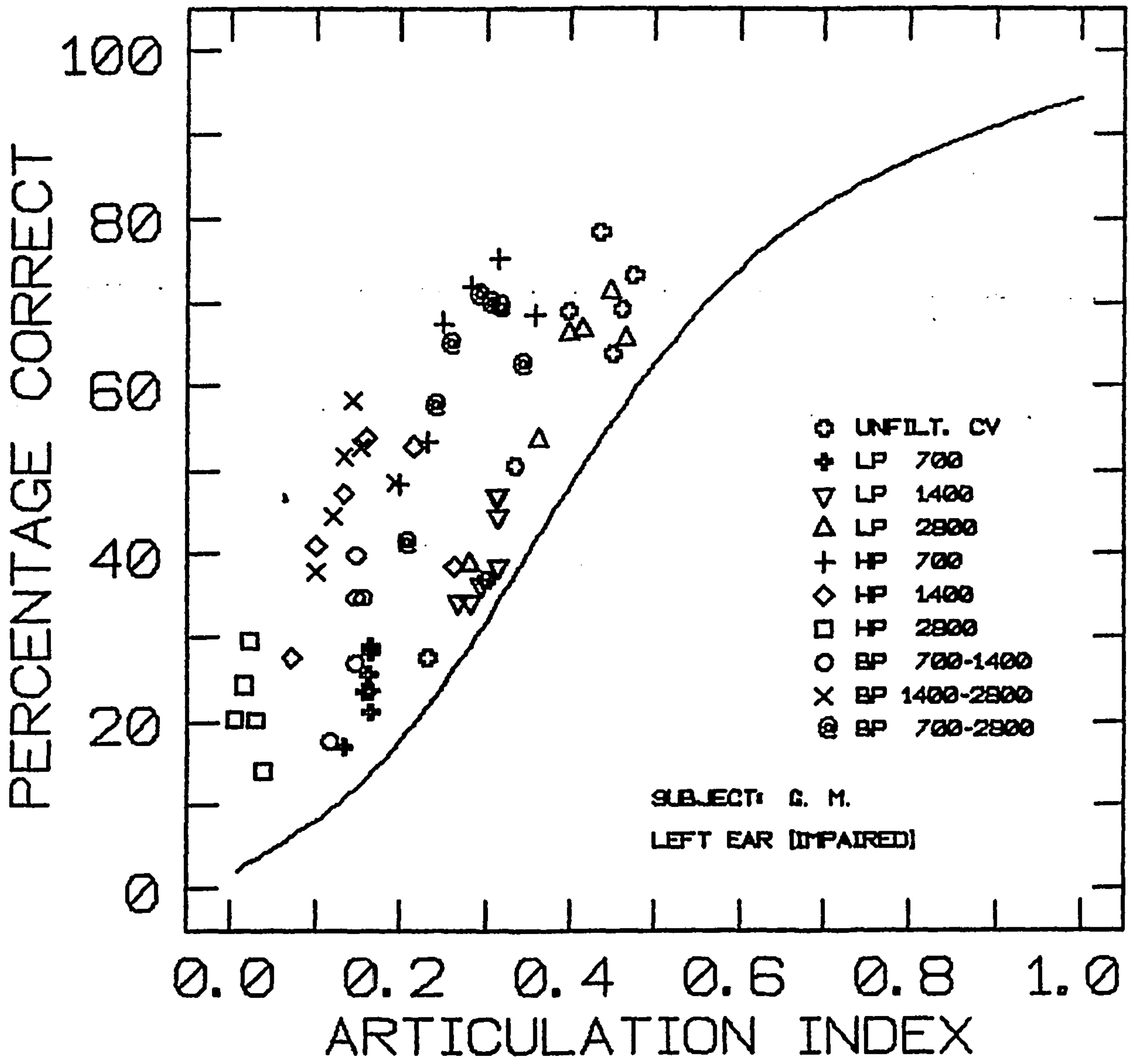


FIGURE 48

Articulation Index Functions

Subject G.M. - Left Ear [Hearing-Impaired]

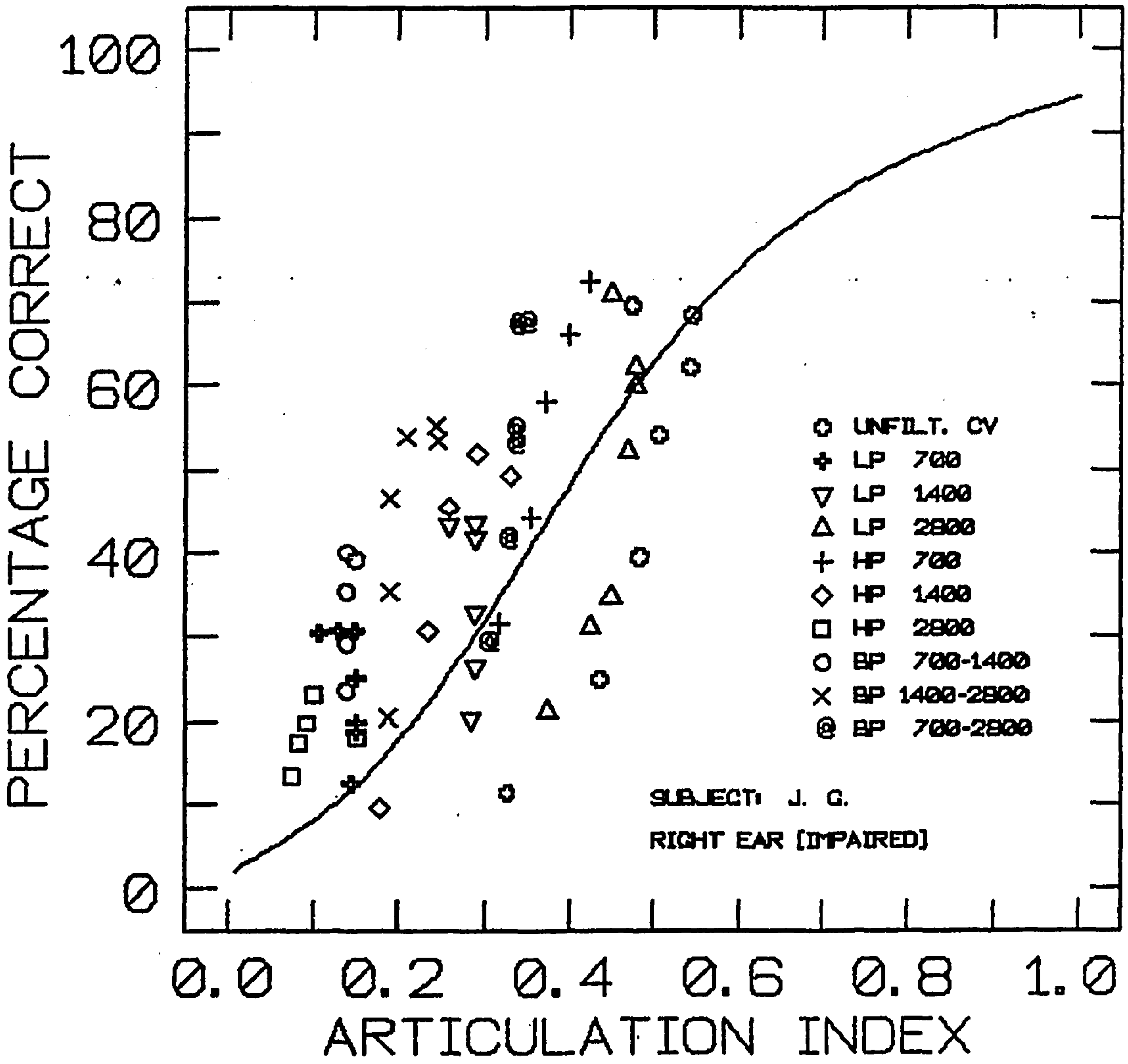


FIGURE 49

Articulation Index Functions

Subject J.G. - Right Ear [Hearing-Impaired]

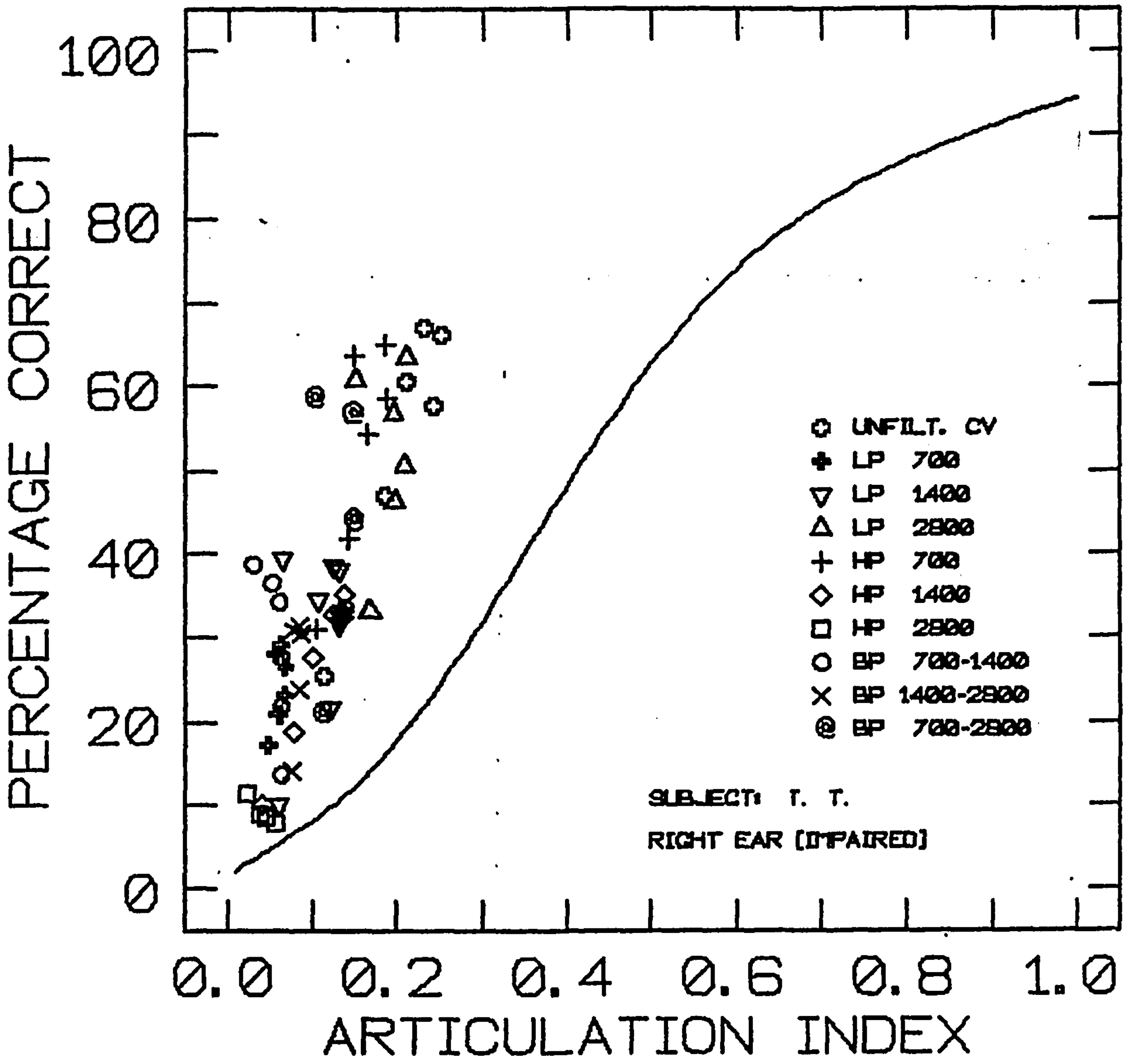


FIGURE 50

Articulation Index Functions

Subject T.T. - Right Ear [Hearing-Impaired]

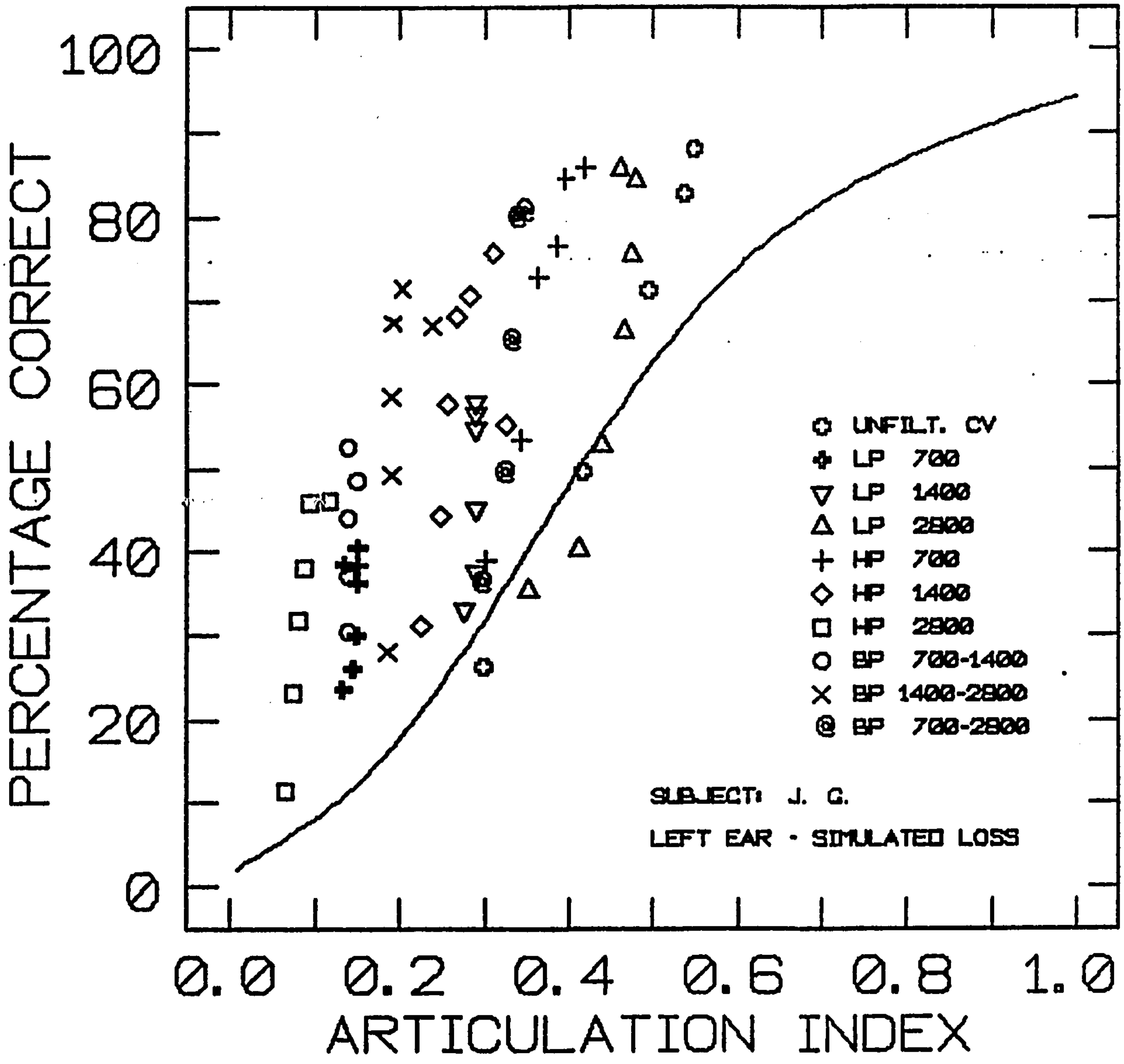


FIGURE 51

Articulation Index Functions

Subject J.G. - Left Ear [Simulated Hearing Loss]

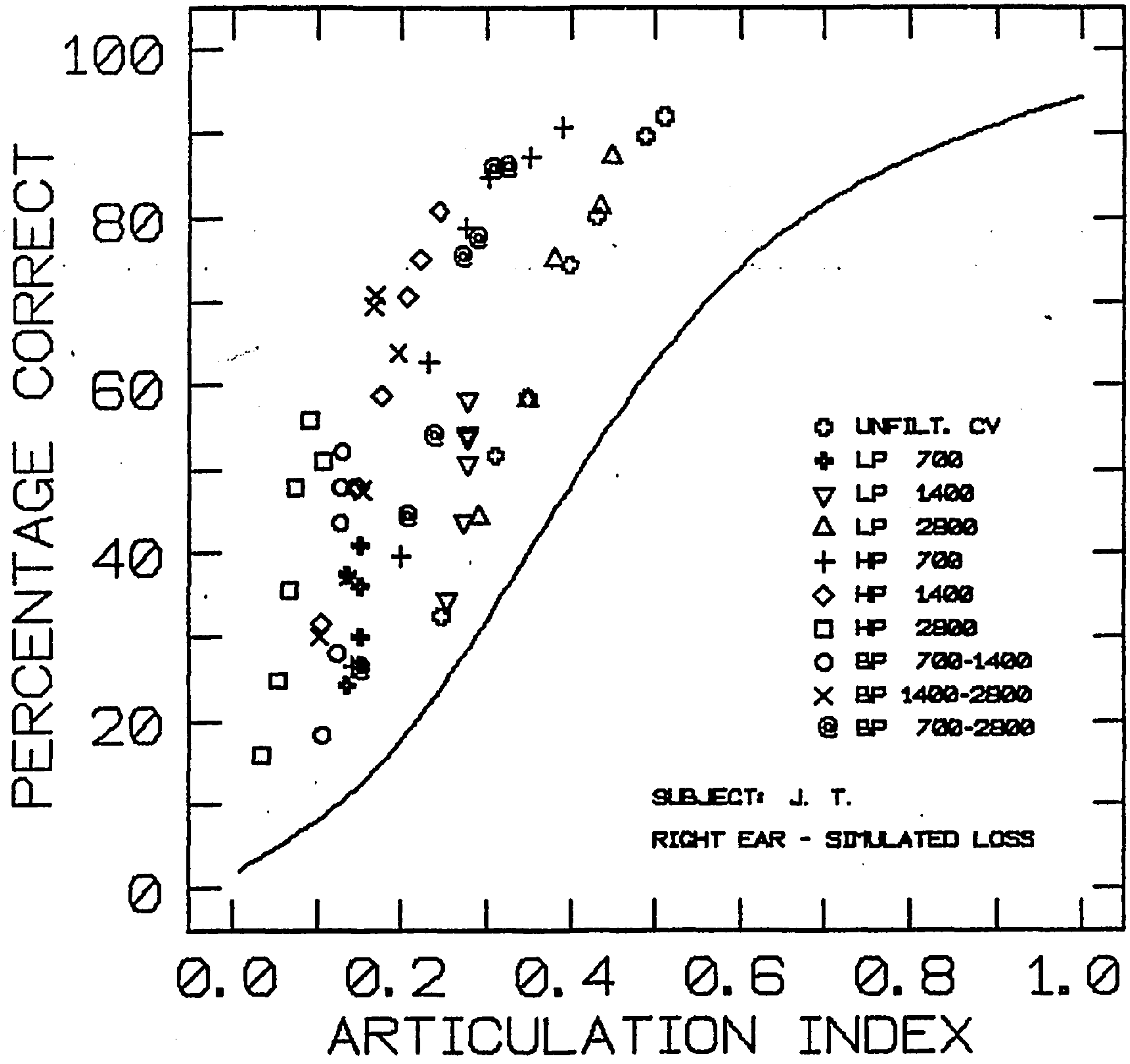


FIGURE 52

Articulation Index Functions

Subject J.T. - Right Ear [Simulated Hearing Loss]

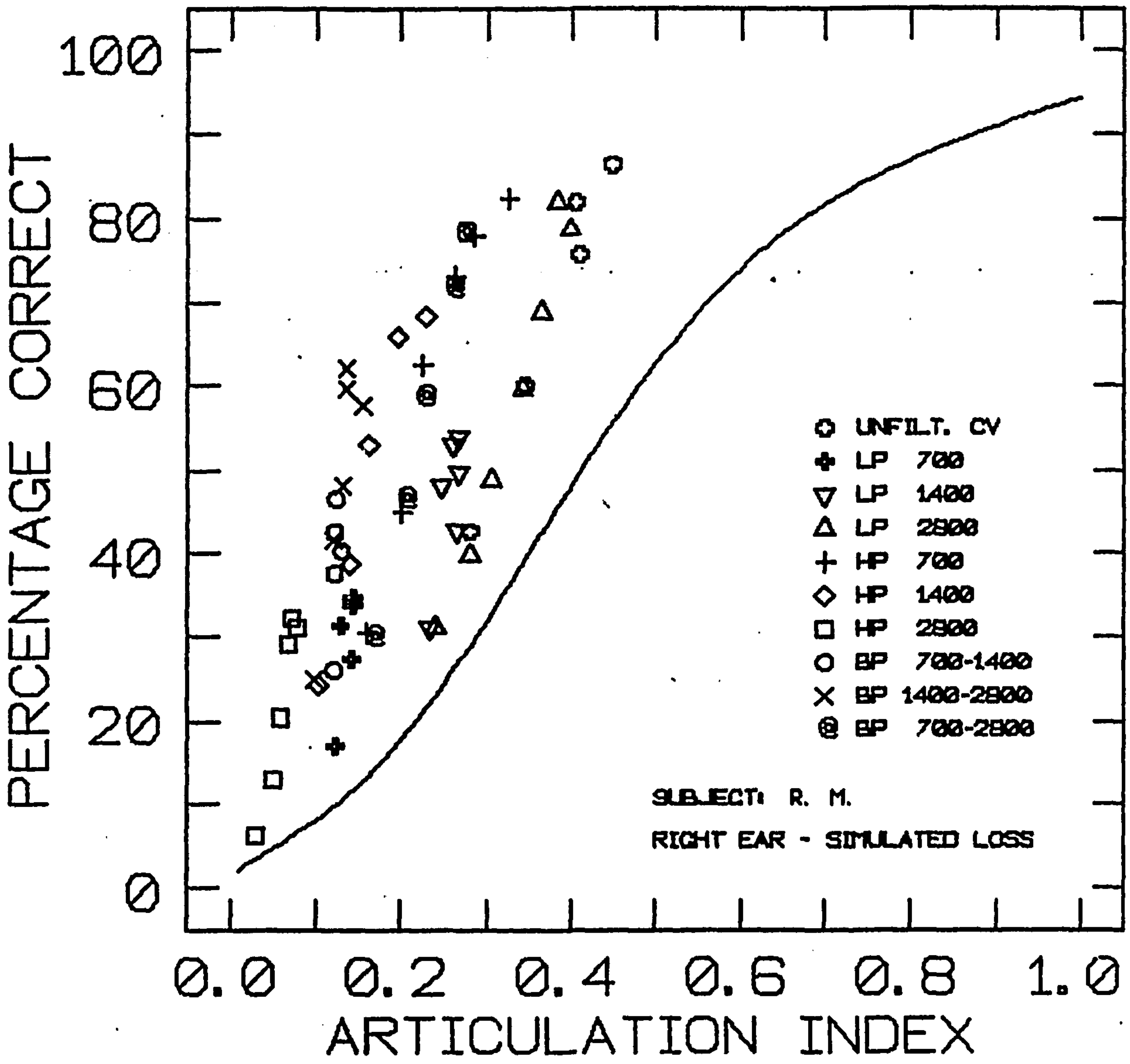


FIGURE 53

Articulation Index Functions

Subject R.M. - Right Ear [Simulated Hearing Loss]

As seen in all the graphs, most of the data points generally do not fall close to the reference curve shown. However, the distribution of calculated A.I. values is systematically grouped by bandwidth, with the A.I. values for the widest band conditions placed toward the right of the cluster of points and the values for the narrowest band conditions placed toward the left. The A.I. values for the narrowest bands including LP 700 Hz, BP 700-1400 Hz, BP 1400-2800 Hz and HP 2800 Hz were, as they should have been, the lowest values for each subject. A.I. values for the two-band combinations, LP 1400 Hz, HP 1400 Hz and BP 700-2800 Hz, were generally somewhat closer to the reference curve. For the three-band combinations LP 2800 Hz, HP 700 Hz and also the unfiltered speech (all four bands), calculated A.I. values fell closest to the calibration curve, and for J.G. and J.T., fell below the curve.

For the normally-hearing subjects listening in quiet, calculated A.I. values fell over most of the range from near zero to near unity. Significantly, however, for some conditions, such as LP 700 Hz and LP 1400 Hz for Subjects J.G. and J.T., the percentage correct score increased even though calculated A.I. values changed little while for other conditions A.I. increased but scores changed very little. These phenomena will be discussed in the next chapter. For both J.G. and J.T., the A.I. values for unfiltered speech fell below the reference curve whereas for

R.M. the values fell above the curve. For R.M., however, the points show the least scatter of the three normally-hearing listeners tested in quiet and all are above the curve. For J.G. the points show more relative scatter, with several values falling below the curve and for J.T., the points exhibit the greatest degree of scatter with still more points falling below the curve. Using the reference curve to predict performance for these listeners will generally predict poorer than observed performance for most conditions, particularly the bands of narrowest and medium bandwidths. For the widest band conditions, predicting performance using this curve will generally result in better performance than observed, especially for J.G. and J.T.

In contrast to the normally-hearing listeners tested in quiet, the results for the hearing-impaired listeners generally indicate that using the reference curve to predict performance will result in poorer intelligibility scores than those observed for these subjects for nearly all conditions tested. The exceptions consist of several points for F.G. which lie on or close to the curve and points for J.G. which lie on or below the curve. In all cases, these values are for unfiltered speech, LP 1400 Hz and LP 2800 Hz. For T.T. all points fell significantly above the curve and show less scatter than for the other listeners. Since her hearing loss was the most severe of the subjects tested, the A.I. values calculated would be expected to be the lowest.

However, relative to the other listeners, her performance was generally much better than would be expected for the low values of A.I. calculated.

The distribution of percentage correct vs. A.I. points for the three normally-hearing listeners tested with noise to simulate hearing loss were similar to the impaired listeners in that most points fell above the reference curve. However, the results were not at all like the results of the hearing losses they were intended to simulate. Although the range of calculated A.I. values was similar, with maximum calculated A.I. values only between 0.5 and 0.6, intelligibility scores were significantly higher than for similar conditions for the impaired listeners, indicating that Articulation Theory as used in this study would significantly underestimate performance for the normally-hearing listeners when tested with masking noise. A detailed analysis and discussion of the implications of these results will be given in the next chapter.

CHAPTER VII

DISCUSSION OF RESULTS

INTRODUCTION

This chapter examines several aspects of the data presented in the previous chapter. First, some observations about the performance-intensity functions are discussed. Then, these results are compared to the results of several earlier studies in which the investigators also measured the perception of filtered speech in normally-hearing and hearing-impaired listeners. Other issues such as the existence and importance of rollover and discomfort limits are examined. Also discussed are several issues relating to the predictions of intelligibility using Articulation Theory including an analysis of the additivity and monotonicity of the A.I. values as tests of the calculations performed in this report and the assumptions of the independence of the band A.I.'s. Finally, recommendations in changing the A.I. calculation procedures in light of these issues are suggested.

FILTERED SPEECH PERCEPTION

The normally-hearing listeners and the hearing-impaired listeners exhibited similar characteristics in performance-intensity functions in that their scores generally increased as presentation level increased and as more spectral information became available. However, as seen in the data in the previous chapter, once maximum performance for a condition was reached, many instances occurred where performance decreased as presentation level was increased further. This was true for both normally-hearing and hearing-impaired subjects. This "rollover" effect will be discussed later in this chapter. At present, the discussion will focus on two aspects of filtered speech perception observed in the data resulting from eliminating the lowest band, LP 700 Hz, or the highest band, HP 2800 Hz, or both at the same time.

INFLUENCE OF SPECTRAL COMPONENTS BELOW 700 HZ

In normal speech signals, most of the energy is primarily in low frequencies. However, this region, below 700 Hz in this study, may have negative influences on intelligibility that are most apparent at high presentation levels, particularly for listeners with sensorineural high frequency hearing loss. A comparison of the results for conditions of filtered speech with and without this energy for both normally-hearing and hearing-impaired listeners demonstrates these effects.

For the normally-hearing subjects (Figures 12, 13 and 14), intelligibility of the HP 700 Hz condition at low presentation levels was lower than that of the unfiltered speech. At high presentation levels, however, intelligibility scores for the HP 700 Hz condition were equal or nearly equal to the unfiltered speech. The same was true when these listeners were tested with masking noise to simulate hearing loss. It is likely, however, that because of the characteristics of the digital filters used in this study, part of the improvement in scores for the HP 700 Hz condition at high levels resulted from some low frequency energy being audible at very high presentation levels. The influence of the energy below 700 Hz may be seen in comparing intelligibility scores for the LP 1400 Hz and BP 700-1400 Hz conditions. For the normally-hearing listeners in quiet and in noise, intelligibility scores for the LP 1400 Hz condition were higher than for BP 700-1400 Hz at low presentation levels. However, at higher presentation levels, performance for BP 700-1400 Hz was nearly equal to LP 1400 Hz indicating that energy at normal intensity levels below 700 Hz was not essential for relatively good speech intelligibility of the CV syllables for this condition. It is also possible that as presentation levels increase for LP 1400 Hz, the upward spread of masking of low freqeuncy energy will limit any improvement in intelligibility.

For the impaired subjects, the effect of removing energy below 700 Hz was dependent on the configuration of the hearing loss. For listener F.G. (Figures 15 and 16), who had nearly normal hearing at low frequencies, the presence of this energy in unfiltered speech caused his performance to diminish when presented at normal and higher levels as compared to the same presentation levels when this energy was removed by filtering (or, at least reduced to very low intensities because of the filter characteristics). At high presentation levels, scores for both unfiltered speech and LP 2800 Hz filtered speech were lower than for HP 700 Hz and BP 700-2800 Hz filtered speech. The effect was more noticeable in the left ear which had a greater high frequency hearing loss than the right ear.

Intelligibility scores nearly equal to the wideband speech were the result at high presentation levels for the HP 700 Hz and BP 700-2800 Hz conditions for measurements in J.G.'s impaired right ear (Figure 19) and G.M.'s left ear (FIgure 17), in which the loss was more steeply sloping in the high frequencies than his right ear. In addition, for these listeners, when the LP 700 Hz band was added to BP 700-1400 Hz to produce LP 1400 Hz, intelligibility for the LP 1400 Hz filtered speech was higher than BP 700-1400 Hz at low intensities. As presentation level increased, differences between scores for LP 1400 Hz and BP 700-1400 Hz diminished and at the highest tested levels were nearly equal for each subject. For listener T.T. (Figure 20)and the right ear of G.M. (Figure 18), which have flatter hearing losses, the results were more like those of the normal listeners tested in quiet and in noise. In these listeners, removing low frequency energy did not result in the degree of improved performance relative to those conditions which contain this information as it did for the normally-hearing listners or listeners with sloping losses. However, scores for LP 1400 vs. BP 700-1400 were much closer at high presentation levels, as they were for the sloping loss subjects just cited.

A possible cause of the effects observed for the LP 700 Hz band is the upward spread of masking (French and Steinberg, 1947; Bilger and Hirsh, 1956; Kryter, 1962a) of relatively intense low frequency vowel energy at high presentation levels. This may tend to reduce intelligibility for listeners with sloping high frequency hearing loss. As seen in the results of F.G. in Figures 15 and 16, significant improvements in performance resulted when information below 700 Hz was removed by low-pass filtering, especially in his left ear which suffered from a somewhat greater high frequency hearing loss than his right ear. The fact that some low frequency energy may be present in high-pass filtered conditions such as HP 700 Hz and BP

700-2800 Hz also should not be forgotten, however. Since its intensity is greatly reduced, the spread of masking would be reduced or eliminated. Other researchers have shown that low frequency energy at low intensities may contribute to intelligibility (e.g., Franklin, 1969 and 1975; Rosenthal et al., 1975). The results of this study are consistent with the findings of these investigators.

INFLUENCE OF SPECTRAL COMPONENTS ABOVE 2800 HZ

As the low frequency region contains most of the energy in a normal speech signal, so, it is generally regarded that the higher frequency region contains most of the intelligibility information. Where that region specifically lies is partially addressed by examining the results obtained for conditions including the highest frequency region tested in this study, from 2800 Hz to 4500 Hz. Since 4500 Hz was the absolute upper limit of the speech signals used, the contributions of energy beyond this frequency cannot be determined from these data.

For the normally-hearing listeners, Figures 12, 13 and 14, improved intelligibility resulted for the unfiltered speech and HP 700 Hz conditions as compared to the LP 2800 Hz and BP 700-2800 Hz conditions. When these listeners were tested in noise simulating the sloping hearing losses (Figures 21, 22 and 23), the inclusion of the HP 2800 Hz band also improved intelligibility, but not to the degree observed for the tests in quiet. The observed results for HP 1400 Hz filtered speech, which, in effect, consists of adding HP 2800 Hz to BP 1400-2800 Hz, were also higher for the wider band condition for the tests both in quiet and in noise. For normal listener J.G., however, the improvement in quiet was not as great as for the other two normal listeners; and in noise, there was no observable difference in her scores for these two conditions. Scores for J.G. and J.T. for the HP 2800 Hz condition when tested in noise at the highest presentation levels, however, were higher than when tested in quiet. This may have been due to the energy below the 2800 Hz cutoff becoming audible at the high presentation levels as a result of the finite attenuation characteristics of the digital filters.

For the hearing-impaired listeners, the importance of the HP 2800 Hz band was dependent on the subjects' hearing loss configuration. For the subjects with high frequency sloping losses, F.G. (Figures 15 and 16), J.G. (Figure 19) and the left ear of Subject G.M. (Figure 17), no differences were apparent in the intelligibility scores between HP 700 Hz and BP 700-2800 Hz. For the subjects whose losses were flatter, T.T. (Figure 20) and G.M.'s right ear (Figure 18), adding the HP 2800 Hz band to BP 700-2800 Hz resulted in higher relative intellgibility scores at low and moderate presentation levels. At the highest tested levels for these two subjects, however, there were virtually no differences between scores for any of the four conditions: unfiltered, LP 2800 Hz, HP 700 Hz and BP 700-2800 Hz. When HP 1400 Hz and BP 1400-2800 Hz are compared, the addition of the information above 2800 Hz made little or no difference in intelligibility score for any of the subjects with the exception of T.T. Even though she described the HP 2800 Hz condition as difficult to hear by itself, she displayed higher intelligibility scores for the HP 1400 Hz condition at lower presentation levels than for the BP 1400-2800 Hz condition particularly at the lower presentation levels. At the highest levels tested, however, the differences were smaller, although the scores for the HP 1400 Hz were still higher. None of the subjects had high scores for the HP 2800 Hz condition alone, although F.G. did better than the others. However, because of his better low frequency hearing, he may have heard some low level information below the cutoff frequency when presented at high intensities.

Intelligibility scores for three of the four narrowest band conditions LP 700 Hz, BP 700-1400 Hz and HP 2800 Hz were the lowest scoring conditions for all subjects under all test conditions. Scores for the fourth condition, BP 1400-2800 Hz, were higher than these except for Subject T.T., who did better on the BP 700-1400 Hz condition.

These results show that if the presentation level is sufficiently high, the upper and lower extremes of the available speech bandwidth were not essential for good speech intelligibility of the test materials used in this study for any of the listeners tested. However, removing information below 700 Hz had a lesser effect on reducing intelligibility at higher presentation levels than removing information above 2800 Hz. It also seems apparent from these results that the band from 1400-2800 Hz contributes the major portion of the intelligibility of the CV syllables used in the present research. The results of the experiments with normally-hearing listeners tested with simulated hearing loss show that under these two filter conditions they performed similarly to the listeners with flat sensorineural hearing loss than to listeners whose hearing loss was predominantly in the higher frequencies.

COMPARISON OF PRESENT DATA TO PREVIOUS STUDIES

In Chapter II, several papers were examined in which the investigators measured the intelligibility of filtered speech in both normally-hearing and hearing-impaired listeners in a variety of listening conditions. While none of these studies duplicate the conditions of the present study, especially in regard to the digital processing of speech, it is instructive to compare some of their results to those of this study to find any general trends that might lead to better understanding of the perception of band restricted speech in both normally-hearing and hearing-impaired listeners. The earliest detailed investigations were those performed at Bell Telephone Laboratories in the late 1920's, and reported by Fletcher (1929). French and Steinberg (1947) summarized this and later efforts to understand the factors associated with speech intelligibility. Their report will be examined first.

COMPARISON OF NORMAL LISTENERS TO FRENCH & STEINBERG

Since the data reported by French and Steinberg (1947) are often considered to be representative of the performance of normally-hearing subjects listening to filtered speech, it was of interest to make direct comparisons of the results of this study with their results. For this comparison, the results for J.G. and J.T. were combined since they were normally-hearing young listeners. The Performance-Intensity (P.I.) curves for their combined results are shown in Figure 54. The data for these functions are in Appendix E. Figures 55 and 56 reproduce the curves of French and Steinberg for the filtered conditions closest to those tested in this study along with the present data replotted from Figure 54 to allow easy comparisons of the data. Figure 55 compares low pass conditions; Figure 56 compares high pass conditions. French and Steinberg originally plotted their data relative to the orthotelephonic condition. This was converted to sound pressure level by assuming that 0 dB orthotelephonic gain equals 65 dB SPL.

In Figure 55, four of French and Steinberg's low-pass filter conditions are plotted along with three of the present low-pass conditions and the "unfiltered" speech condition. The quotations marks here refer to the fact the the unfiltered condition is actually low-pass filtered at 4500 Hz. Thus, the unfiltered speech may be compared to French and Steinberg's low-pass 4500 Hz condition.

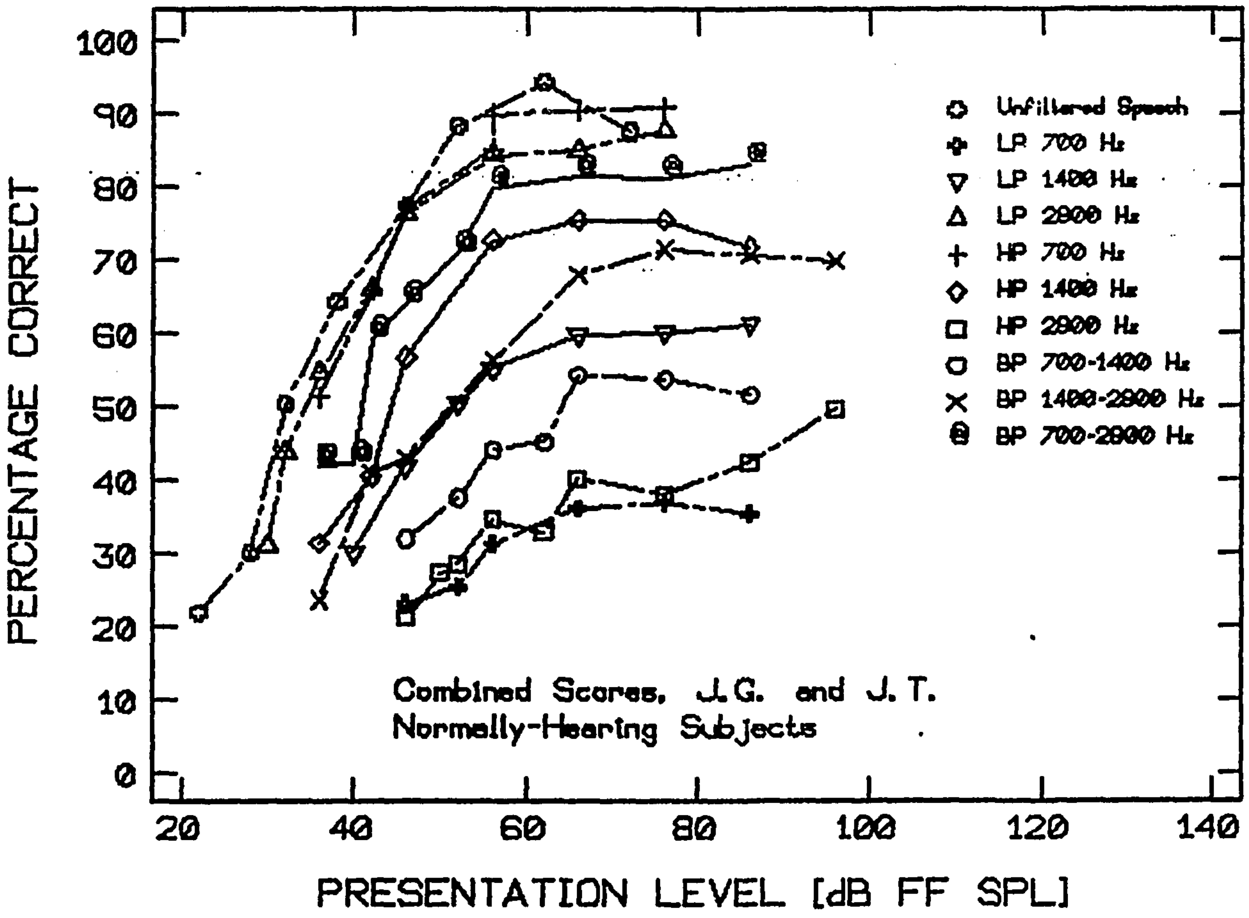
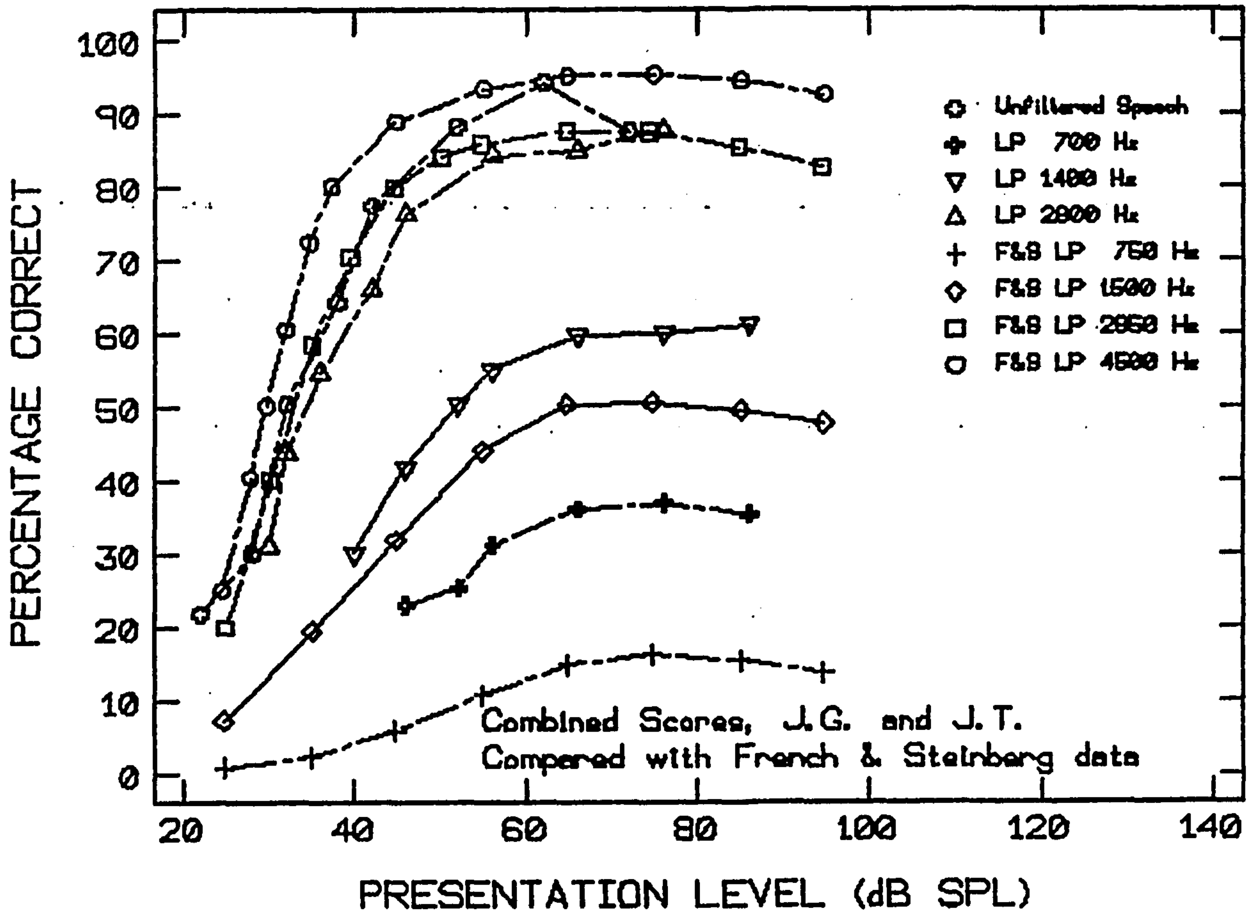


FIGURE 54

Performance - Intensity Functions

Combined Results, J.G. and J.T. [Normally-Hearing]



<u>FIGURE</u> <u>55</u>

Performance - Intensity Functions

Combined Results, J.G. and J.T.

Compared to French and Steinberg Data

Low-Pass Filtered Speech

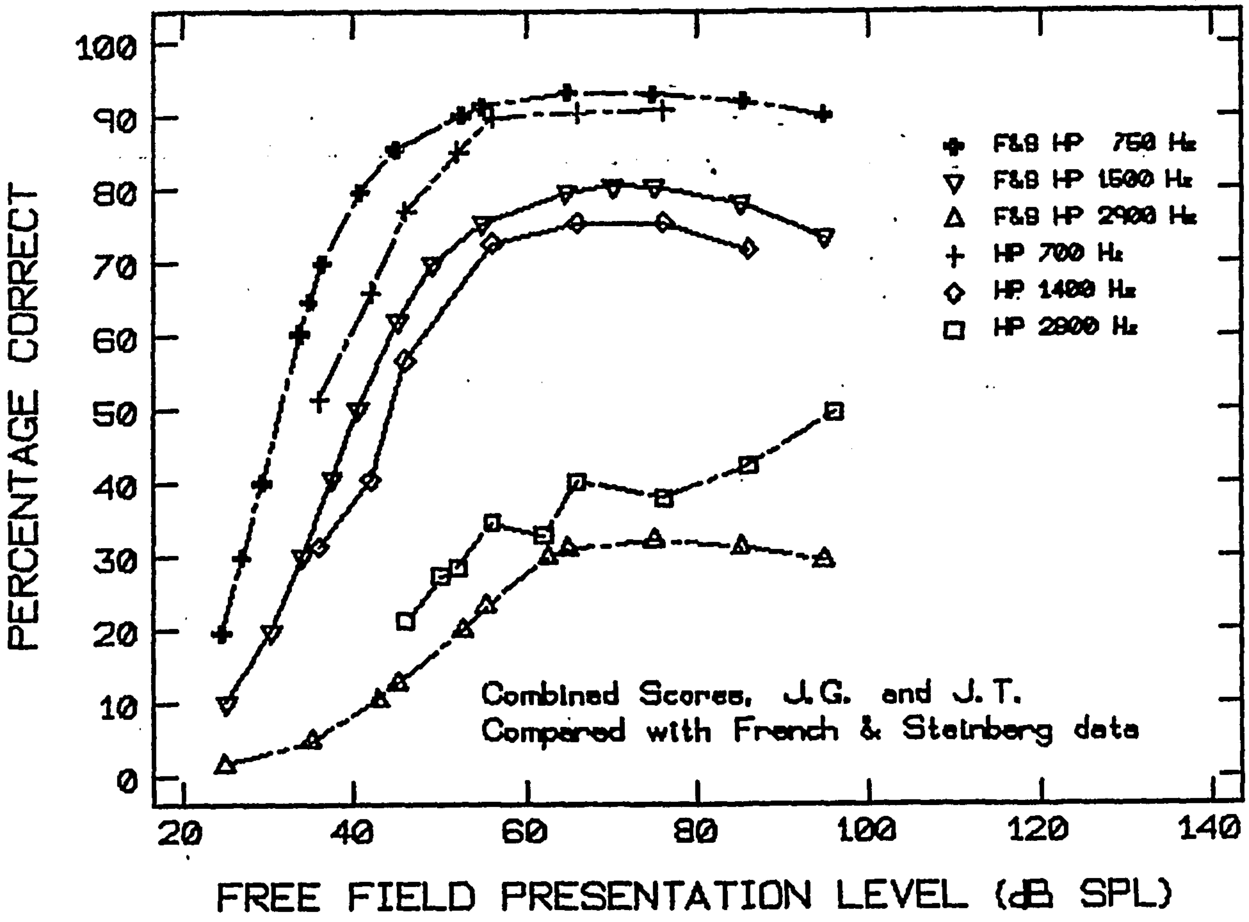


FIGURE 56

Performance - Intensity Functions

Combined Results, J.G. and J.T.

Compared to French and Steinberg Data

High-Pass Filtered Speech

In comparing the present unfiltered condition to French and Steinberg's LP 4500 Hz, one notes that maximum performance was nearly equal and that it occurred at approximately the same level. However, as level decreased, scores for the present study dropped somewhat more rapidly than for French and Steinberg. The differences between the present wideband data and those of French and Steinberg are primarily one of a translation of the curves as a function of intensity since the slopes of the curves are similar. Although the presentation levels used in this experiment were calibrated relative to free field sound pressure levels, there, nevertheless, appears to be a systematic difference between the specification of presentation levels used in this study and the levels indicated by French and Steinberg.

Returning to Figure 55, one also observes differences in performance for the LP 2800 Hz and French and Steinberg's LP 2850 Hz conditions similar to the wider band conditions. French and Steinberg's reported scores are generally higher than those obtained in this study. Again, there appears to be a difference in the specification of presentation levels since the slopes of the curves are similar. However, for the low-pass 1400/1500 Hz and low-pass 700/750 Hz conditions, scores achieved by the listeners in this study were significantly higher than French and Steinberg, even though the cutoff frequencies were slightly lower and the filter skirts significantly steeper.

High-pass filter conditions are compared in Figure 56. French and Steinberg's scores are higher for the HP 700/750 Hz and HP 1400/1500 Hz conditions. As for the low pass cases, there appears to be a systematic level difference since the slopes of the P.I. functions are similar. For HP 2800 Hz, however, scores in the present study were higher and increased above 80 dB SPL. This is most likely due to low frequency energy becoming audible at very high presentation levels. As noted earlier, digital filters do not attenuate the signal progressively, but pass energy in the "stop band" at a finite attenuation level determined by the filter specifications. These filters had a stop band attenuation of approximately 60 dB, Therefore, at 100 dB SPL presentation level, some vowel energy may have been present at 40 dB SPL.

Differences between the results of this study and French and Steinberg may be attributable to several factors. These include differences in procedures, speech levels speech materials, subject and talker differences, filter characteristics and listening conditions. Major procedural differences occurred in the manner of presentation of the speech materials and the training and response manner of the subjects. For the French and Steinberg report, the subjects heard the speech signals read by natural voices over a simulated telephone link. They responded verbally or wrote the responses from a virtually unlimited response set. In the present study, the subjects' responses were restricted to the 72 possible choices of the utterance. This limited response set may account for the relatively higher performance scores achieved by the listeners in this study for the more difficult filtered speech conditions such as LP 700 Hz and HP 2800 Hz.

In French and Steinberg's report, speech levels are specified in dB referred to the "orthotelephonic condition." The "0 dB" orthotelephonic condition has been assumed to be 65 dB sound pressure level. However, this assumption is based on averaged data over many talkers. It is possible that due to individual differences in the talkers reported by French and Steinberg, the speech level at 0 dB orthotelephonic gain was not 65 dB. These individual differences could account for some of the shift in the performance-intensity functions for similar conditions between their data and those of the present report.

Since high quality audio signal storage media such as records or tape did not exist at the time of the research reported by French and Steinberg, all material was uttered on the spot. In the present study, all filtered utterances were derived from the same set of speech samples and stored in digital form on a computer. Although precautions were taken to avoid having the listeners remember any particular utterance or set of utterances, the fact that the same utterances were used repeatedly no doubt accounted for some of the differences between the results. In addition, since the subjects in this study were trained extensively on each condition, as shown by Milner (1973), performance levels would be expected to be higher than when subjects were given less rigorous training, unless testing continued over a long period of time with the same subjects.

Differences in the test materials probably accounted somewhat for the higher scores in the LP 700 Hz and LP 1400 Hz conditions of this study as compared to French and Steinberg. The set of CV's employed for this study used only three vowels environments /a/, /i/ and /u/. The listeners also had to choose one consonant out of 24 to successfully identify the syllable correctly. French and Steinberg's subjects listened, however, to CVC's in an open response format, and therefore, had much lower a priori probabilities of getting equal or higher scores for difficult conditions. In addition, scores for final consonants are usually lower than for initial consonants under difficult listening conditions, further decreasing the intelligibility score.

Differing filter characteristics were partly responsible for differences between the present study and

French and Steinberg in the results for the high pass filter conditions. The digital filters used in this study had extremely sharp slopes. Since digital filters attenuate linearly as a function of frequency rather than logarithmically, the slope characteristics are different as well. The equivalent attenuation rates ranged from about 100 dB/octave to about 800 dB/octave. French and Steinberg used standard analog filters whose slopes were unspecified, but not likely as steep as the 100 dB/octave of the shallowest digital filter. In addition to the digital filters, a 4500 Hz anti-aliasing filter was used that had a cutoff slope of approximately 120 dB/octave. As Castle (1963) and Palva (1965) observed, the slopes of the filter skirts are a critical variable in the perception of filtered speech. At the higher presentation levels, the finite attenuation of the digital filters also contributed to increased intellgibility, particularly for the high pass conditions. Another characteristic of the digital filters is the absence of phase distortion. This may also have had a slight influence on the results of this study, tending to improve scores relative to similar analog filters where phase distortion may be significant.

Talker variation may have also accounted for performance differences between the studies. The talkers used in this study were not "trained" talkers in the sense of having professionally trained voices, as have been used in many studies of speech perception. In fact, the precision of these talkers in reciting some of the CV syllables used here was, on occasion, quite poor. Evidence for this is the fact that no normally-hearing listener achieved 100 percent intelligibility for the 4500 Hz (wideband) speech condition. For the data reported by French and Steinberg, professionally trained talkers were not used, either. However, they did use their listening crews as both talkers and listeners (or as Fletcher (1929) referred to them, "callers and observers"). That is, for a series of articulation tests, each member of the crew served as both listener and talker, giving these subjects an advantage in familiarity of voices and speaking mannerisms that was not available to the subjects used here. Thus for their listeners, French and Steinberg's talkers were probably more intelligible for comparable conditions of speech degradation than for the talkers and listeners used in this study.

For the speech materials used in this study, Each of the four talkers spoke each CV syllable three times. Obvious differences in the same CV spoken by the different talkers were observed. To balance out this effect, each training run included two utterances and subsequent identification runs used the third utterance not heard during training. Later identification runs used one of the three utterances but presented in a random order to avoid memorization. As seen in Appendix E, the standard deviations across talkers for many of the points were quite high. This meant that for a particular level and filter condition, intelligibility scores were highly talker dependent. No specific pattern of individual talker variability was observed, however, nor was any pattern clear by grouping male vs. female. For any given filter condition, the talker having the highest percentage correct score at one presentation level was not necessarily the same talker or the same sex of talker at other presentation levels for that condition.

Another source of variation resulted from the specification and control of speech levels. In this study, the unfiltered CV's were normalized to equal vowel RMS levels. It is not likely that any level control was used in the articulation testing reported by French and Steinberg other than verbal instructions to the talkers to attempt to maintain a uniform speaking level. These differences likely resulted in reduced speaker variability in the present study. It is possible, therefore, that part of the apparent disagreement in presentation levels and steepness of the performance-intensity functions between these two studies results from the way in which speech presentation levels were determined.

OTHER STUDIES OF FILTERED SPEECH PERCEPTION

One difficulty in making comparisons of this research to other studies is the fact that most other studies of filtered speech perception consisted of experiments at one test level. Very few investigators tested subjects over a wide range of intensities, and none evaluated both normally-hearing and hearing-impaired listeners under identical training and test conditions. Another problem is that with one exception, speech materials used by the investigators differed considerably from those of the present study. Of those that tested a range of intensities, primarily French and Steinberg, just discussed, and Pollack (1948) have relevance to this research. Pollack's study can be compared to the results obtained here with the normally-hearing listeners tested with masking noise to simulate hearing loss. Using monosyllabic phoetically-balanced words Pollack tested normally-hearing listeners with several high-pass and low-pass filtered speech conditions in the presence of a constant intensity white noise signal. The noise level was about 81 dB SPL, much more intense than the maskers used in the present study which averaged about 65 to 70 dB SPL. At the highest levels tested, however, maximum intelligibility scores were comparable for comparable conditions. Pollack found that maximum performance for masked high-pass filtered speech with a cutoff of 350 Hz was higher than that for the unfiltered condition. Since the lowest high-pass cutoff frequency in this study was 700 Hz, no direct comparison may be made regarding this effect in the normally-hearing listeners. However, for all the hearing-impaired listeners except T.T., the maximum intelligibility for the HP 700 Hz condition was greater than that of the unfiltered speech. However, it is important to note that at Pollack's next highest high-pass cutoff, 580 Hz, the effect disappeared, consistent with the performance of the normal listeners tested with the noise-simulated hearing loss.

Pollack observed differences in the "gain functions" of the high-pass and low-pass filter conditions, noting that the slopes of the high-pass functions were much steeper than the low-pass functions. A similar effect was seen in this study, where one may observe that the minimum performance scores for the high-pass conditions occurred at significantly higher presentation levels than for the the low-pass conditions and rose sharply to maximum performance levels comparable to the levels in the unmasked cases.

Another observation in comparing the present study to Pollack is that none of his smoothed "gain functions" exhibited rollover or reduced performance. Only one data point indicated by Pollack appeared to show a reduction in performance at the highest tested level. This is surprising considering that his presentation levels seem to be quite intense. His data are presented in dB orthotelephonic gain. Pollack indicated that his overall speech level was 68 dB SPL, based on VU meter indications at 0 dB orthotelephonic gain. Thus Pollack's maximum presentation level is about 108 dB SPL. For the subjects in the present study, rollover in the performance-intensity functions occurred at levels below 100 dB for some conditions.

LaBenz (1956) used monosyllabic phonetically-balanced words to test 100 adult subjects consisting of a nearly uniform mix of normally-hearing, conductive, mixed and sensorineural ("perceptive") hearing losses. All testing was done at 30 dB relative to spondee threshold (SRT). Relative to the normals, conductive and mixed loss subjects, the subjects with sensorineural loss scored lower as cutoff frequencies increased in low-pass conditions. For band-pass conditions consisting of varying widths with cutoff frequencies ranging from 250 Hz to 3000 Hz, highest scores for all listeners were achieved for the bands of 1500-2000 Hz and 2000-3000 Hz. However, performance of the sensorineurally-impaired subjects was substantially poorer than for the others.

Compared to the results of the present study, LaBenz's sensorineurally-impaired subjects performed much more poorly than the impaired subjects used here, except for T.T. and the right ear of G.M. Direct comparisons are difficult since LaBenz's data was averaged over his 22 subjects with sensorineural loss. Further, since his presentation levels were specified relative to the subject's SRT, not in absolute sound pressure level, and since these thresholds were not specified, it is not possible to make direct comparisons of performance versus level.

LaBenz claimed that the subjects with sensorineural hearing loss performed similarly to normally-hearing listeners hearing low-pass filtered speech. The results of the present study agree with this finding as far as comparisons between the studies are possible. Generally, the subjects with high frequency sloping losses tested for the present research had reduced intelligibility scores for the wideband condition compared to the normal listeners. Their maximum scores fell in a range where the results for a low-pass condition might have fallen had such a condition been tested. LaBenz did not test high-pass conditions so one cannot see if performance for high-pass conditions with low cutoff frequencies improved relative to wideband speech for his subjects with sensorineural loss.

Castle (1963) studied normally-hearing listeners using monosyllabic words read by a single male talker. Some comparisons with specific conditions studied in the present research may be made. In this research, Castle studied band-pass filters of varying width and filter skirt slopes.

Only three of Castle's many conditions match closely to those studied here. They are band-pass 720-1440 Hz, 1440-2400 Hz and 720-2400 Hz. The scores achieved by Castle's subjects for these conditions were dependent on the filter skirt configuration. Only one such configuration, the steepest skirts, was used for the 720-2400 Hz condition and the subjects' scores averaged about 99%. For the other two conditions, scores for 720-1440 Hz ranged from 60% to 70%, and for 1440-2400 Hz ranged from 70% to 90%. The maximum scores for the comparable conditions in this study were somewhat lower than Castle's, but the relative ordering of performance was the same: BP 700-1400 Hz was the poorest (av. score 54%), next highest was BP 1400-2800 Hz (av. score 71%) and highest was BP 700-2800 Hz (av. score 81%).

Castle (1964) also tested hearing-impaired listeners with band-pass filtered speech. Unfortunately, the conditions between the normal and impaired groups of subjects were not the same and thus cannot be compared directly. Comparison of the data of the hearing-impaired subjects in the present study to Castle's data obtained with hearing-impaired subjects is difficult since a uniform set of experiments was not performed on all subjects. Different subjects listened to different conditions of filtering, and no subject listened to all. His subjects also had more severe hearing losses than the subjects used here except for T.T. Castle's "Subject VIII" had a hearing loss similar to

T.T., and for his "maximum bandwidth (60-20160 Hz)" condition compared to the unfiltered condition of this study, these two listeners performed similarly. Castle's designation of 60-20160 Hz bandwidth is obviously unrealistic, since his subjects listened using earphones which typically have a frequency range of about 80-6000 Hz.

Palva (1965) studied the performance of normally-hearing subjects listening to Finnish two-syllable words through various low-pass, high-pass and band-pass filters. No masking was used so comparisons may only be made with the present listeners tested in quiet. For Palva's high-pass conditions, subjects generally did more poorly than those in the present study at comparable filter cutoff frequencies whereas for low-pass conditions they did better. For example, in the present research, maximum performance for HP 700 Hz was nearly equal to the wideband condition. Palva's subjects' performance diminished when the high-pass cutoff was above 500 Hz. For LP 2800 Hz, intelligibility scores for the subjects in this study were lower than wideband, whereas for Palva's low-pass 2000 Hz, performance was essentially the same as his wideband condition. These differences were due, most likely, to the greater vowel content of the Finnish words Palva used. This material should be affected less by low-pass filtering than by high-pass filtering. The band-pass filter conditions he studied increased in bandwidth steps from extremely narrow to 1-1/2 octave wide filters. Band center frequencies varied from about 300 to 5000 Hz. The one octave case is directly comparable to the band-pass filters BP 700-1400 and BP 1400-2800 Hz, studied for this research. Palva's results were the reverse of the outcome of this study, with higher scores for the lower octave band. Palva's overall maximum score occurred for the bands 900-1800 Hz and 1080-2160 Hz. Again, this result was most likely due to the nature of the speech materials.

Another study of normally-hearing and hearing-impaired subjects listening to filtered speech was that of Ambrose (1972). Using CID monosyllabic words, (Hirsh et al, 1952) he examined three conditions of band-pass filtered speech and varied the amount of distortion present in the speech signal. The three band pass conditions were 300-4800 Hz, 600-1200 Hz and 1200-2400 Hz. The zero percent distortion condition is the only one that was comparable to any condition of this study. Scores for the 600-1200 Hz condition were only around 10-15% for both the normally-hearing and sensorineurally impaired listeners. For the 1200-2400 Hz condition, scores averaged around 45% for the normals and only 20% for the sensorineurally impaired subjects. The 300-4800 Hz case is somewhat like the unfiltered and the HP 700 Hz conditions of this report. For this condition, Ambrose's normally-hearing subjects scored very high, 95%; his sensorineural impaired listeners, scored only 60%, somewhat lower than the impaired subjects' scores for the HP 700 Hz condition in this study. He gave no detailed descriptions of his impaired listeners, so no direct comparisons can be made.

In a later study, Ambrose and Neal (1973) tested subjects with sensorineural impairments using three band-pass conditions. Comparison with the data of this study would not be meaningful since the skirts of filters they used were extremely shallow - 24 dB/octave. Scores for these subjects ranged from 2% to 90%. Their results however, exhibited characteristics similar to the results of this study; maximum scores for the higher band conditions were higher than for the lower band conditions.

The study by Aniansson (1975) also differed in many ways from the present research. His voices were recorded while reading a small set of Swedish monosyllabic words in a home living room for binaural earphone presentation. He also recorded "everyday noises" consisting of recorded traffic noise and competing speech from a radio and real voices along with the speech materials to create a variety of "everyday listening situations." His subjects all had, presumably, noise-induced hearing loss, of varying magnitudes. In Aniansson's study, filtered speech tests were performed only on the normally-hearing listeners. The impaired listeners heard only unfiltered speech under the various conditions of everyday listening. Aniansson tested his subjects at only one level, 60 dB(C-weighted), which corresponds approximately to the same level in this study since he measured his sound levels in the recording room at the time the materials were recorded.

Only one subject in the present study, F.G., had a hearing loss that was specifically diagnosed as noise-induced. F.G.'s hearing in the right ear was closest to Aniansson's "L 4000" group of subjects, whose hearing loss was 50 dB or more above 4000 Hz. His left ear was similar to the "L 3000" group, defined similarly. In quiet, Aniansson's L 4000 group averaged 90% and the L 3000 subjects averaged 88%. For the unfiltered speech in this study, F.G.'s maximum scores were 80% in his right ear and 71% in his left ear, significantly below Aniansson's subjects' scores. Furthermore, F.G.'s right ear score was the highest of any of the impaired subjects for the unfiltered speech.

Aniansson also tested a group of normally-hearing listeners under several conditions of filtered speech. All tests, however, were conducted with one of three kinds of background noise. None was tested in quiet. It is virtually impossible to compare these situations with those of this study since the noise spectra used are very different. Also, binaural recording situations provide a perceptual advantage when listening to speech masked by external noise, especially when the noise is recorded binaurally at the same time as the speech. For example, in a 0 dB signal-to-noise situation with traffic noise as the masker, the average intelligibility score for Aniansson's subjects was 94%. In a monaural situation, with white noise masking speech with a signal-to-noise ratio of 0 dB, a score of 50% to 60% would be expected.

The study by Wang, Reed and Bilger (1978) has some relevance to the present research since they used speech materials similar to those used here. Nonsense syllables consisting of both consonant-vowel (CV) pairs and vowel-consonant (VC) pairs were presented under earphones to eight normally-hearing listeners. The talker of these materials was unspecified and only one token per syllable was presented, as contrasted to four for this study. A single presentation level was used that was described as "95 dB SPL re a 1000 Hz tone" for the wide band condition. A constant spectrum level was used for all filter conditions. This appears to correspond to a presentation level of between 85 and 90 dB FF SPL for the data of the present study. Not all subjects were tested at that level for the present research since it was felt to be somewhat intense for the normally-hearing subjects used here and the listeners usually reached maximum performance below that level. Since the major findings of Wang, Reed and Bilger were presented in terms of Sequential Information Analysis of perceptual confusions, no direct comparison to these data can be made at present. The data of this study, however, may be similarly analyzed and this should be the direction of future efforts regarding this data.

Wang, Reed and Bilger did present averaged curves of percentage correct as a function of filter cutoff frequency. A comparison of these data shows similarites in the performance of subjects for both low-pass and high-pass conditions. For their subjects, identification score in low-pass conditions did not decrease significantly until the cutoff frequency was below 2800 Hz, as was true for the results of this study as well. However, scores for low-pass 700 Hz were much higher for Wang, Reed and Bilger. As for the results of this study, scores in high-pass conditions for Wang, Reed and Bilger's subjects did not decrease until the cutoff frequency exceeded 700 Hz. However, scores for high-pass 2800 Hz filtered speech were higher for Wang, Reed and Bilger than for the present study. Thus, as far as one may compare, the data of the studies appear similar except at the extreme ranges of high-pass and low-pass filtering. (No band-pass conditions were tested Wang, Reed and Bilger). These differences are possibly due to the different filter characteristics. Their filters were analog filters with slopes of 48 dB/octave. The authors made no mention of waveform amplitude normalization as was done on the speech signals for this study.

In comparing the results of these studies to the present investigation, another factor must be considered as well as those discussed above, including speech materials, talkers, presentation levels and filter characteristics. Most of the studies cited used monosyllabic words for the speech signals. Excluding the foreign language materials of Palva and Aniansson, the English words were probably spoken with a carrier phrase preceeding the test word. Although this fact was not clearly stated by the investigators, it is customary practice to use a phrase such as "you will say . . . " or "you will write . . . " proir to each key word. It is well known (Egan, 1948) that the presence of a carrier phrase will affect intelligibility, particularly of initial consonants. With a carrier phrase the listener is prepared for each test word and has the advantage of additional cues to phoneme identification through coarticulation effects.

Wang, Reed and Bilger and the present author presented nonsense syllables without a carrier phrase. The primary effect of this would be to increase initial consonant errors for CV syllables relative to similar materials which used a carrier phrase to introduce the syllable. Therefore, differences in performance between these studies and other testing similar conditions are likely to occur due to differences in the manner of speech sample presentation.

IMPLICATIONS OF SATURATION AND ROLLOVER

One important characteristic of the performance-intensity functions, especially for the hearing-impaired subjects, was the occurrence of saturation or rollover in intelligibility performance as presentation level increased above the point where the subject achieved maximum score. As discussed earlier, rollover for these data was defined as a decrease in performance of 3% or greater at a presentation level higher than that at which the maximum intelligibility occurred. For the normally-hearing listeners tested in quiet, and with masking noise to simulate hearing loss, such rollover occurred sporadically for the three listeners with no apparent pattern to the effect. However, as seen in Tables 2 - 7, and in the performance-intensity functions in the previous chapter, there is evidence for rollover in many conditions.

Several hypotheses exist for the presence of rollover. One is the upward spread of masking that, according to Articulation Theory, will cause intelligibility to diminish as masking noise in low frequency regions increases in intensity above certain threshold levels. Speech contains more low frequency energy relative to high and, as noted by French and Steinberg (1947), Kryter (1962a) and others, at high intensities the low frequencies of speech will contribute to the upward spread of masking. In this study, spread of masking effects may have caused rollover at levels below discomfort thresholds for the low-pass conditions. For the hearing-impaired listeners with high frequency sensorineural loss, low level consonant energy is crucial to intelligibility. The spread of masking effects will be likely to have greater influence on intelligibility since the the masking extends into regions where listeners have impaired hearing mechanisms.

Another reason for rollover is the possible increase in internal distortion, especially in impaired auditory systems, caused by high intensity speech near discomfort thresholds. Since speech consists of a distribution of sound levels, brief peak speech levels may have been associated with such internal distortion. but not for a long enough time period that a subject experienced discomfort. It is important to realize the difficulty in accurately specifying the speech levels. Since root-mean-square speech levels were referred to a sinewave level in calibration, it is probable that for some of the speech signals used in this study, peak values exceeded the levels assumed for the Articulation Index calculations (see Appendix A) and extended above discomfort thresholds.

In examining the observed intelligibility performance of this study, rollover occurred more often for the impaired subjects than for the normally-hearing subjects. However, as with the normally-hearing subjects, there was no discernable pattern to its occurrence. For example, for subject F.G., with better high frequency hearing in his right ear than in his left, rollover occurred in his right ear for eight out of ten test conditions and in his left ear for six out of ten conditions. Those conditions for which it occurred in both ears (See Table 3) were unfiltered speech, LP 1400 Hz, HP 1400 Hz, HP 2800 Hz, and BP 700-2800 Hz. Rollover did not occur for the BP 1400-2800 Hz condition in either ear. For the other impaired subjects, the conditions in which rollover occurred do not lend themselves to simple explanation. Only for the tests conducted in the left ear of Subject G.M. did rollover occur for every condition.

The results obtained for subject F.G. demonstrate the apparent effects of the spread of masking and increased distortion. For this listener, rollover occurred at a lower level for unfiltered speech than for HP 700 Hz when the low frequency energy is removed by filtering. His performance at high levels for HP 700 Hz improved relative to unfiltered speech indicating that he was able to make use of his residual high frequency hearing in the absence of intense low frequency energy.

PREDICTIONS OF ROLLOVER BY ARTICULATION THEORY

The presence of rollover in the observed data raises the question of whether rollover would occur in the performance-intensity functions predicted by Articulation Theory. Additional analyses of the data were performed to determine when the Theory would predict rollover as the presentation levels reached or exceeded discomfort thresholds. Three examples of this analysis are shown in Figures 57, 58 and 59. These figures plot individual performance-intensity functions for each of the ten conditions of filtered speech heard for the condition analyzed. Each set of ten plots show the results for a single talker and a single subject. Figure 57 shows the results for Subject J.G.'s normally-hearing left ear and Figure 58 for her impaired right ear, both for Talker B (male); Figure 59 shows the results for the left ear of Subject F.G., Talker D (female). Indicated in each individual plot are the observed scores for each condition (open circles) and a line showing the predicted performance based on Articulation Theory calculations extending above and below the levels actually tested using the reference curve shown in Figures 42 - 53 to predict the intelligibility score. The reader must be cautioned that these predictions are based on the reference Articulation Index function which is a rough estimate of a function based on the observed data. Also shown in the curves at the levels for which observed data points exist is the 2-sigma expected range for the predicted score assuming a binomial distribution. Therefore the predictions shown in these figures are, in reality, a test of the methods used to calculate the A.I. values determined in this study.

The manner of the fit of the observed data to the predicted performance curve enables one to observe the accuracy of predictions using the previously described reference curve more easily than in the Articulation Index plots of Figures 42 - 53. In Figure 57, the best fit of observed to predicted data for Subject J.G.'s normally-hearing ear occurred only for unfiltered speech and LP 2800 Hz. The results for the same subject's impaired ear, Figure 58, showed relatively poor fits for all conditions. For Subject F.G., shown in Figure 59, reasonably good fits of ovserved and predicted performance are found for the wideband and low-pass conditions, but very poor fits elsewhere. These observations are similar to those made earlier, that for most listeners, observed points for the low-pass and wideband conditions fell closer to the reference curve than the high-pass and band-pass conditions.

As seen in all three figures, predicted intelligibility score showed rollover as level continued to increase for all conditions except HP 2800 Hz. The reasons for this are the influence of the discomfort levels specified for each listener. As peak speech levels in any band reach the discomfort levels, that band contributes no further to the A.I. value. As additional peak levels rise above discomfort thresholds, the calculated A.I., and therefore, the predicted percentage correct will diminish. Also, the spread of masking increases at high intensities and thus diminishes the calculated A.I. value. For the HP 2800 Hz condition, the presentation levels were so high that the calculated A.I. values reflected the increasing influence of the finite 60 dB attenuation of low frequency energy by the digital filters.

It is important to note that the specification of the discomfort levels for an individual subject will influence the point at which rollover occurs in the predicted performnace-intensity function. Dugal, Braida and Durlach (1978) specified discomfort levels which averaged 90 dB SPL. These levels are lower than those used here. Had lower estimates of discomfort thresholds been used for the subjects in this study, rollover would have been predicted to occur at lower intensities, possibly near the points at which it was observed in the listeners. This point suggests the need to carefully specify the discomfort thresholds for subjects used in studies in which A.I. calculations will be performed.

In the three figures, 57, 58 and 59 one observes several conditions for which rollover actually occurred at presentation levels lower than those at which predicted rollover occurred. This occurred, for example, for J.G. for unfiltered speech, LP 700 Hz, HP 1400 Hz, HP 2800 Hz and BP 700-1400 Hz when tested in her normally-hearing ear in quiet. In her impaired ear, it only occurred for the HP 1400 Hz and BP 700-2800 Hz conditions. For F.G., in his left ear, this occurred also for HP 1400 Hz and HP 2800 Hz. In these examples, however, only one condition showed predicted performance to be close to observed performance. This was for J.G. listening to unfiltered speech when tested in quiet. As seen in these three figures and earlier in Chapter VI, Articulation Theory as applied to this study did not accurately or uniformly predict performance for all conditions and all subjects. One primary reason is, as noted, the choice of the reference curve of percentage correct vs. Articulation Index. However, other fundamental factors relating to calculation of the Articulation Index must also be considered. These include the assumptions of band independence as evidenced by the additivity of the band A.I. values and the monotonicity of observed and predicted A.I. functions and performance-intensity curves. It is also clear that without further understanding of the causes of early rollover, one may not successfully apply Articulation Theory to select an appropriate frequency-gain characteristic for a hearing aid for hearing-impaired listeners.

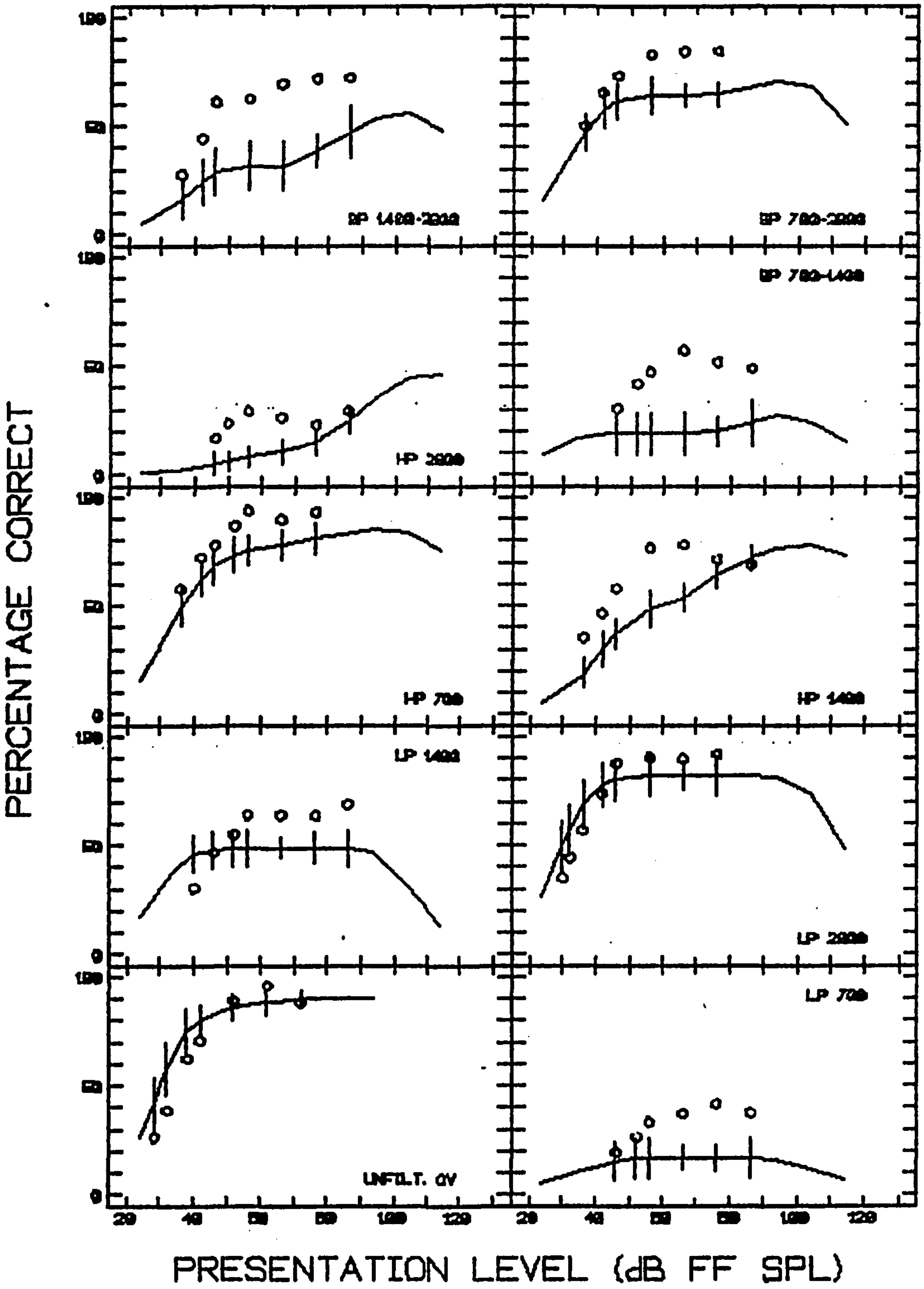


FIGURE 57

Performance - Intensity Functions:

Observed and Predicted Functions

Subject J.G. Left Ear, Normally-Hearing

Talker B, (Male)

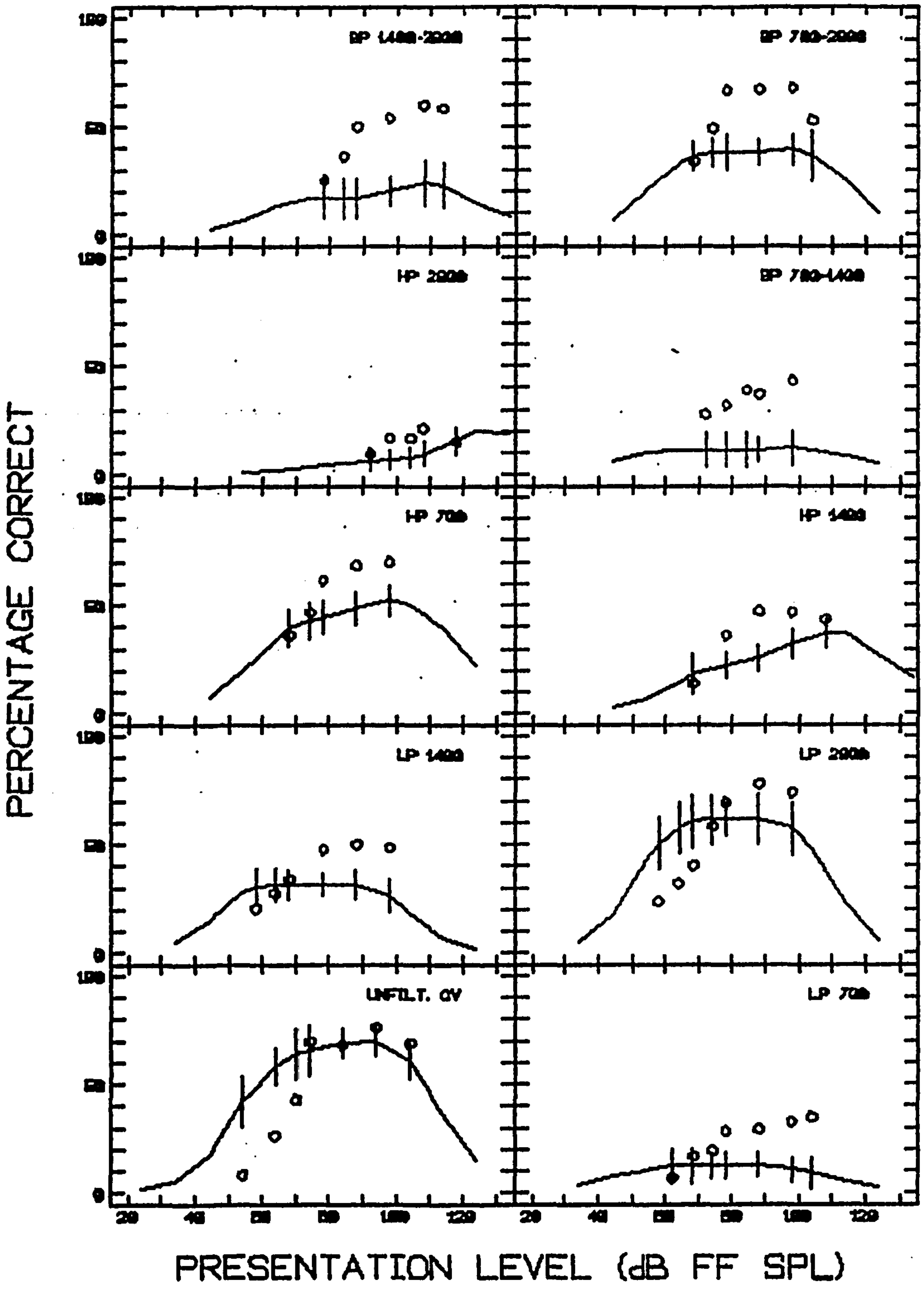


FIGURE 58

Performance - Intensity Functions:

Observed and Predicted Functions

Subject J.G. Right Ear, Impaired

Talker B, (Male)

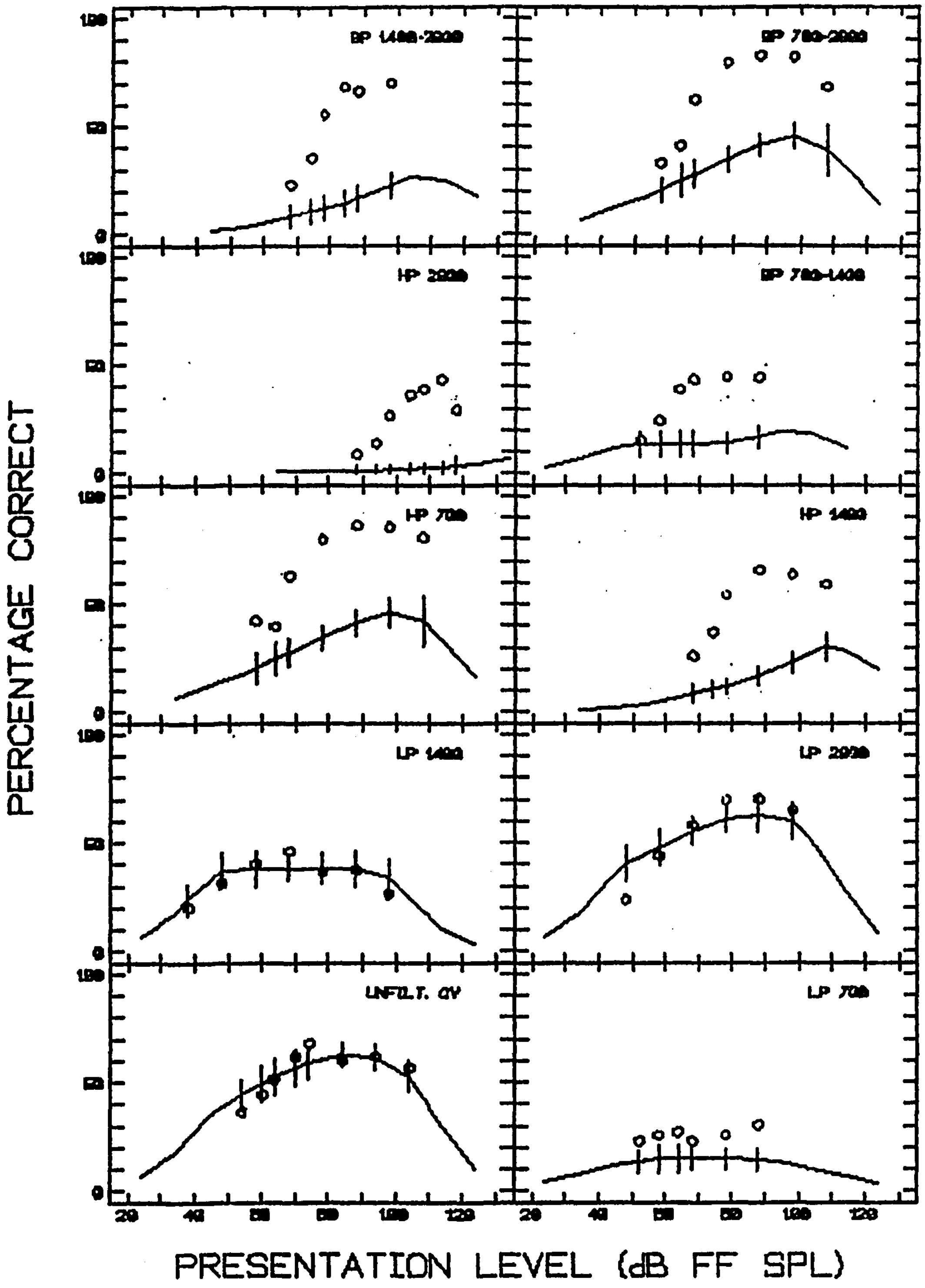


FIGURE 59

Performance - Intensity Functions:

Observed and Predicted Functions

Subject F.G. Left Ear, Impaired

Talker D, (Female)

ADDITIVITY OF ARTICULATION INDEX

An important assumption of Articulation Theory is the independence of the contributions of each of the 15 one-third octave bands (or 20 equally intelligible bands of French and Steinberg, if their method is used). One way to verify this assumption is to examine the additivity of the Articulation Index values calculated and percentage correct scores predicted for the narrow bands of speech that, when combined, constitute the wider band conditions presented to the listeners. If additivity obtains, the sum of the A.I. values for each component band will equal the A.I. value for the broader band which includes all the component bands. Furthermore, the percentage correct score for the broader band should be correctly predicted from the sum of the A.I. values of the component bands. These assumptions should hold true if speech peaks are below discomfort thresholds and the spread of masking does not contribute to altering the A.I. value for any condition.

Another important assumption is that the Articulation Index functions are monotonic, that for any A.I. value a single predicted percentage correct score exists and increases as the A.I. value increases. If no external noise is present, predicted performance intensity functions should monotonically increase as presentation levels increase until they reach the limits imposed by either loudness discomfort thresholds or the maximum A.I. value for a given bandwidth condition. If the band limit is reached, the curves will become flat; if discomfort levels are reached, or if external noise is sufficiently intense that the spread of masking influences the A.I. value, the P.I. curves will roll over as A.I. values diminish for further increases in presentation level. In calculating A.I. values speech peak levels in any band do not increase beyond discomfort levels, but remain there as either overall speech level increases or effective masking noise increases.

In the discussion to follow, the terms summation or sum band will be used to describe this broader band which is composed of the component filter bands. It must be understood that the listeners in this study never actually heard wide-band conditions that were created from a direct combination of the narrower bands as would occur by summing filter outputs. The wide band conditions were created simply by filtering within the specified limits. For the ten filter conditions tested in this study, tests of additivity and monotonicity were performed on sixteen different combinations of component and summation bands. The tests were performed to determine whether the calculations of this study followed the basic assumptions. These sixteen combinations are show in Table 8.

TABLE 8

COMPONENT BANDS AND SUMMATION BANDS
FOR TESTS OF ADDITIVITY OF A.I. VALUES

COMPONENT BANDS	SUMMATION BAND
1. LP 700 + HP 700	UNFILTERED
2. LP 1400 + HP 1400	UNFILTERED
3. LP 2800 + HP 2800	UNFILTERED
4. LP 700 + BP 700-2800 + HP 2800	UNFILTERED
5. LP 1400 + BP 1400-2800 + HP 2800	UNFILTERED
6. LP 700 + BP 700-1400 + LP 1400	UNFILTERED
7. LP 700 + BP 700-1400+BP 1400-2800 + HP 2800	UNFILTERED
8. LP 700 + BP 700-2800	LP 2800
9. LP 1400 + BP 1400-280	LP 2800
10. LP 700 + BP 700-1400 + BP 1400-2800	LP 2800
11. BP 700-1400 + HP 1400	HP 700
12. BP 700-2800 + HP 2800	HP 700
13. BP 700-1400 + BP 1400-2800 + HP 2800	HP 700
14. LP 700 + BP 700-1400	LP 1400
15. BP 1400-2800 + HP 2800	HP 1400
16. BP 700-1400 + BP 1400-2800	BP 700-2800

For each of the sixteen cases, the calculated Articulation Index values of the component bands were summed and compared to the A.I. of the summation band. Performance-intensity functions were plotted for each of the sixteen additivity conditions, showing in each case, the P.I. function for each component band and the resultant summation band. Since the result of this analysis consists of 384 (16 cases x 12 conditions x 2 plots) graphs, only a small number of representative plots will be presented here.

Before discussing these specific examples, however, some general observations may be made regarding these calculations. In the A.I. computations, all of the sums of the A.I. values of the component bands were equal to or greater than the A.I. value of the summation bands. At presentation levels below 80 dB SPL, the sum of the A.I.'s was equal to the A.I. of the summation band. At higher presentation levels, however, the sum of the component band A.I.'s exceeded the A.I. of the summation band for several of the subjects. These included nearly all sixteen cases for subject F.G. in both ears, J.G. in her impaired ear and in her normal ear tested in quiet and J.T. in her normal ear tested in quiet.

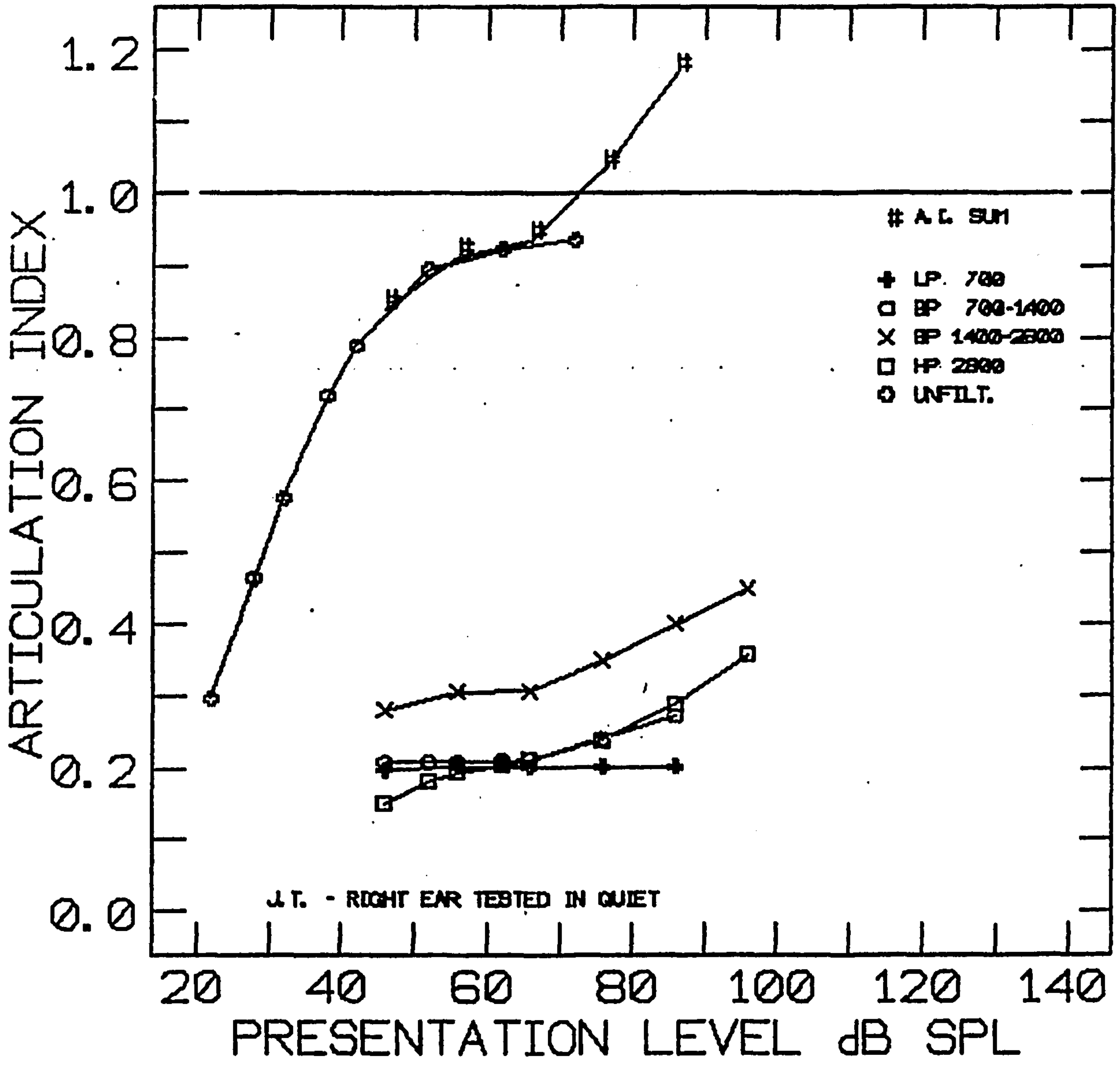
The sum of the A.I.'s exceeded the A.I. of the sum band in only a few cases for G.M.'s left ear, for J.G.'s normal left ear tested in noise and R.M.'s right ear tested in noise. It did not occur at all for G.M.'s right ear, T.T.'s right ear, R.M.'s right ear tested in quiet and J.T.'s right ear tested in noise. In addition, the sum of the component band A.I. values exceeded unity for four specific cases. This was, of course, improper, since, by definition, the maximum A.I. value is unity. The sums greater than the A.I. of the summation band and the sums greater than unity occurred as a result of the filter characteristics as incorporated into the calculation procedure. As noted earlier, the digital filters had a finite stop band attenuation of 60 dB. When the A.I. values were computed for each component band at high presentation levels, the stop band regions were incorporated into the total band A.I. unless reduced or eliminated by threshold shifts or external masking. When adjacent band A.I.'s are added, the stop band contribution of the higher (or lower) component band is added to the pass band contribution of the lower (or higher) component band. This results in a higher A.I. value than that for the summation band. The reader should understand that these circumstances exist because of the methods of computation used, and not by any fundamental fault of Articulation Theory.

Examining specific examples of this additivity analysis, Figure 60 exhibits one result for Subject J.T.'s right ear tested in quiet. Shown is an analysis for the the case of the four component bands summing to equal the overall unfiltered speech condition. The sum of the component band A.I.'s was computed only for those presentation levels where data points exist that are common to all component conditions. The P.I. functions corresponding to this case are shown in Figure 61. All performance levels were substantially below that of the overall wide band condition. Notably, in comparing these two figures, one can observe that the component band with the highest A.I. values, BP 1400-2800 Hz, also demonstrated the highest component band intelligibility.

Masking noise improved the additivity of the calculated

Articulation Index values since the low level information of the filter stop bands is effectively removed from the calculations. Figures 63 and 64 show the results for Subject J.T. with simulated hearing loss for the same additivity case just described. In this example, the A.I. sum is essentially equal to the A.I. of the sum band.

An example for a hearing-impaired listener is seen in Figures 64 and 65. In this case, the analysis was for Subject F.G., with the case of LP 700 combining with HP 700 to equal the unfiltered speech. As in the previous examples, the sum of the component band A.I. values was equal to or greater than the A.I. of the sum band. Additivity for these cases, therefore, appears to obtain.



<u>FIGURE 60</u>

Additivity of A.I. Values

Subject J.T. - Normally-Hearing, Tested in Quiet

Summation of LP 700, BP 700-1400,

BP 1400-2800 and HP 2800 Hz Bands

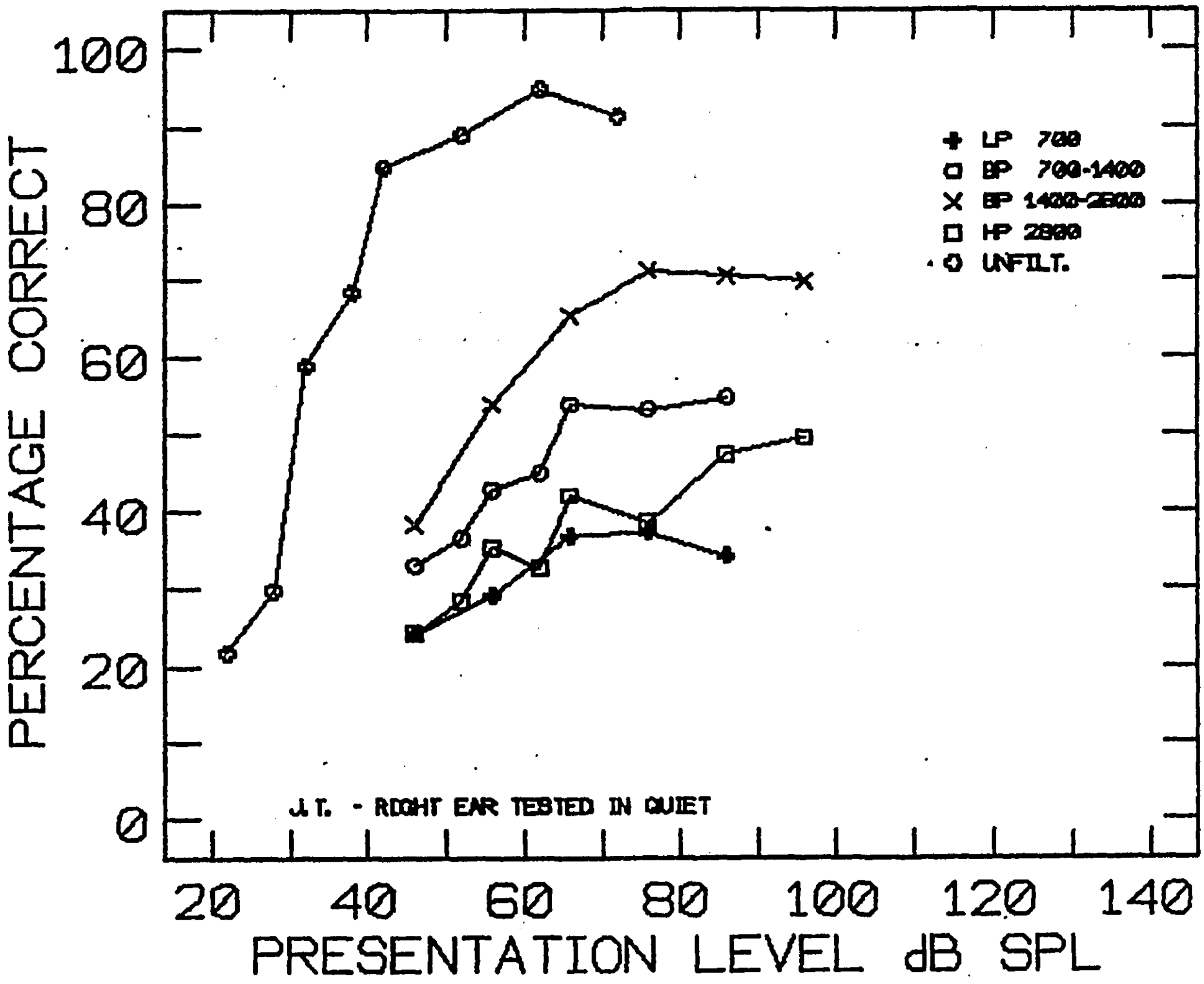


FIGURE 61

Performance - Intensity Functions

Subject J.T. - Normally-Hearing, Tested in Quiet

Summation of LP 700, BP 700-1400,

BP 1400-2800 and HP 2800 Hz Bands

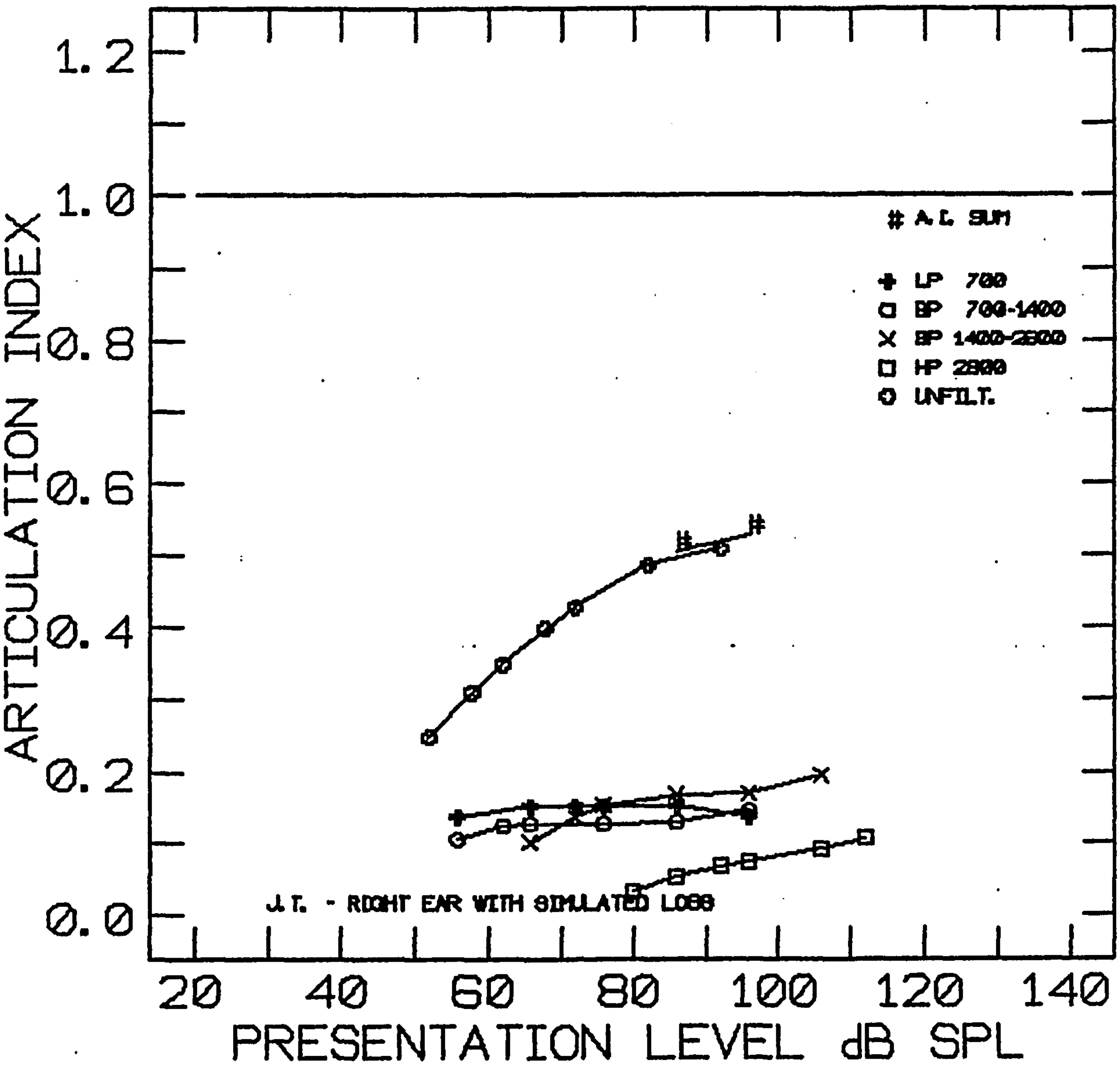


FIGURE 62

Additivity of A.I. Values

Subject J.T. - Normally-Hearing, Tested in Noise

Summation of LP 700, BP 700-1400,

BP 1400-2800 and HP 2800 Hz Bands

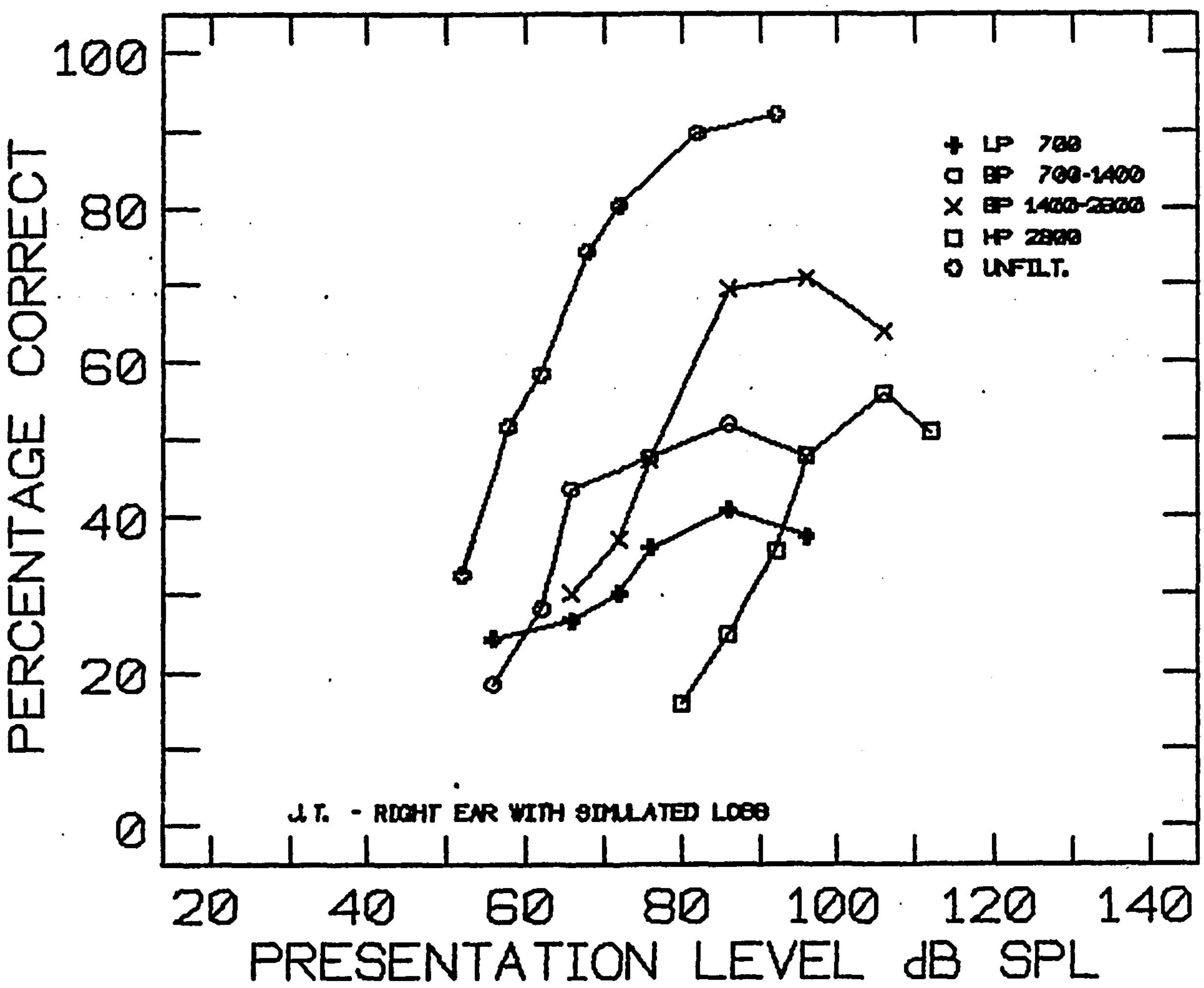


FIGURE 63

Performance - Intensity Functions

Subject J.T. - Normally-Hearing, Tested in Noise

Summation of LP 700, BP 700-1400,

BP 1400-2800 and HP 2800 Hz Bands

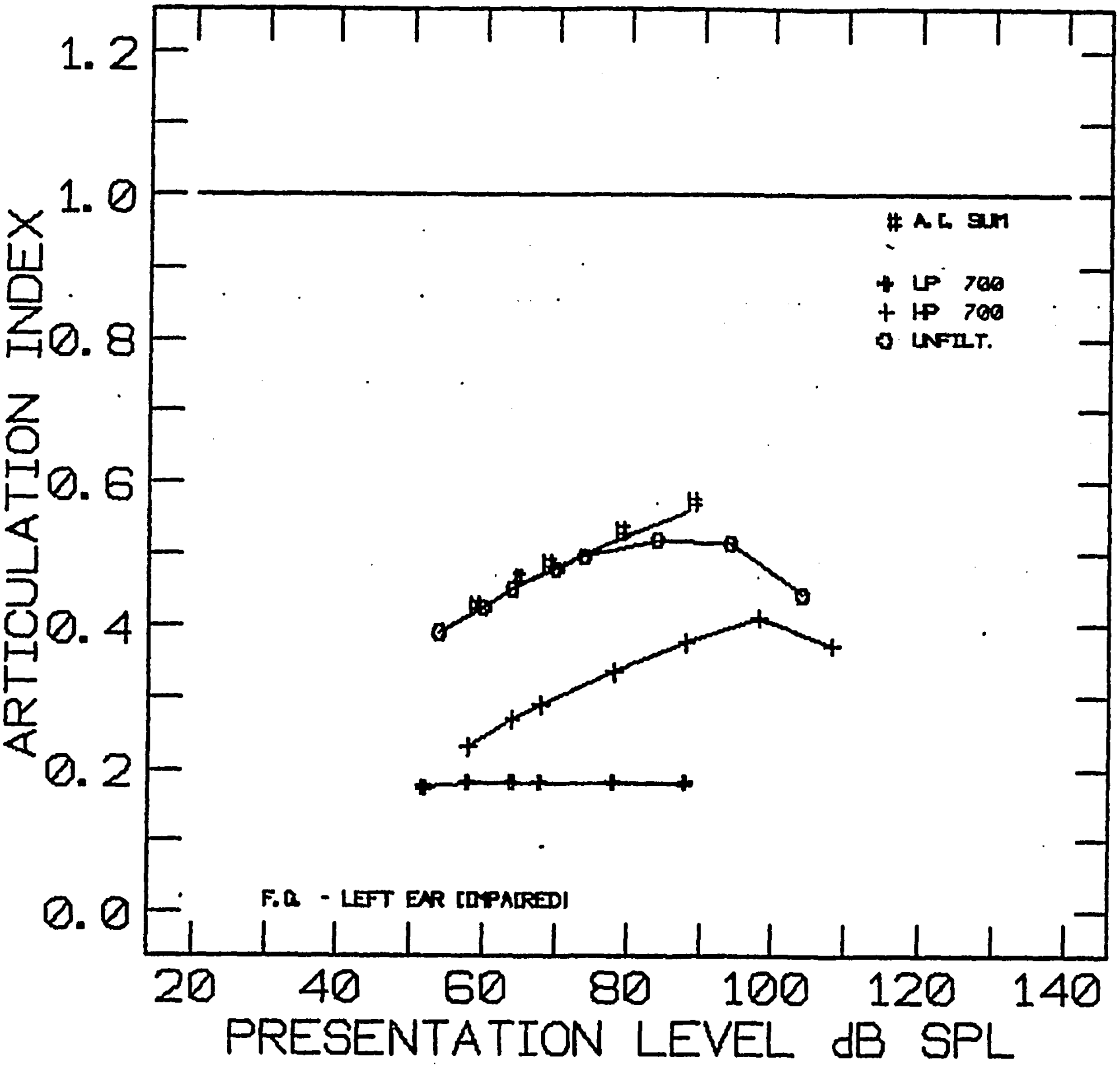


FIGURE 64

Additivity of A.I. Values

Subject F.G. - Hearing-Impaired, Left Ear

Summation of LP 700 and HP 700 Hz Bands

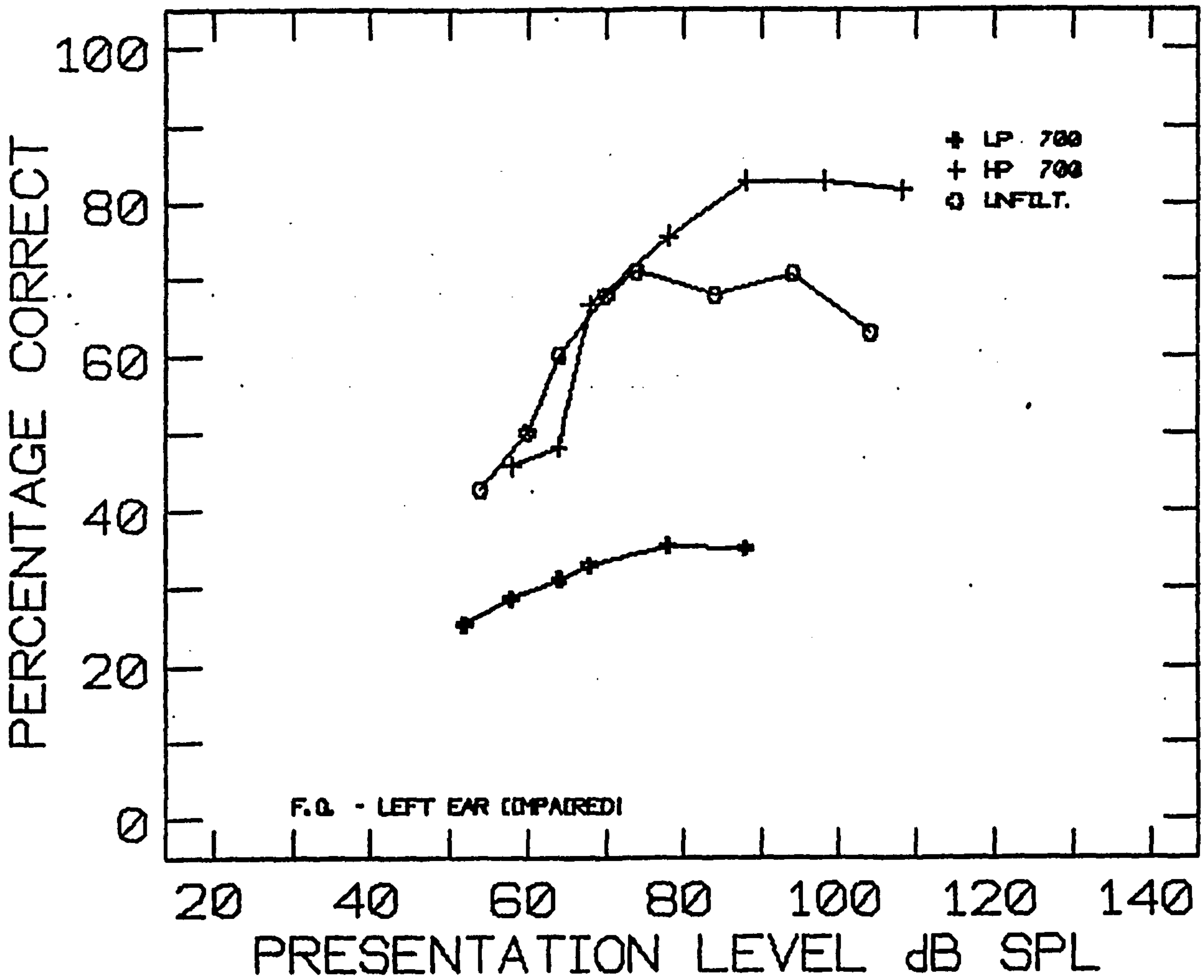


FIGURE 65

Performance - Intensity Functions

Subject F.G. - Hearing-Impaired, Left Ear

Summation of LP 700 and HP 700 Hz Bands

MONOTONICITY OF ARTICULATION INDEX FUNCTIONS

Another important assumption of Articulation Theory is that of the monotonicity of the Articulation Index functions and the percentage correct scores predicted by those functions. This implies that as calculated A.I. values increase, whether by adding additional spectrum information in adjacent bands, or by increasing the intensity of the speech signal, or by reducing the influences of external noise and masking, the percentage correct score predicted by the A.I. will also increase. Further, a given A.I. value may be achieved in different ways. However, for any specific curve of A.I. vs. percentage correct, the same value of A.I. will produce the same value of percentage correct. The curves presented in Figure 1 and Figures 42 - 53 of this report are typical examples of A.I. functions for different sets of speech materials. It is clear, however, from an examination of the observed data in the figures just cited that the assumptions of monotonicity do not hold. For all subjects, there are a number of conditions for which the observed percentage correct diminished even though the calculated A.I. values for these points were higher than points with higher percentage correct values. There were, in addition, several conditions for each subject for which the calculated A.I. values remained constant but had several values of observed percentage correct. Such conditions were LP 700 Hz, LP 1400

Hz and BP 1400-2800 Hz, where the calculated A.I. values were at the maximum possible band A.I. over a range of tested presentation levels. According to the assumptions of monotonicity, if the A.I. remains constant, the percentage correct score should remain constant. As seen in the observed data, however, this was not the case.

SUMMATION OF PERCENTAGE CORRECT SCORES

The relationships of the additivity of the observed percentage correct values was explored using a probability model. By definition, the percentage correct scores are probabilities simply multiplied by 100. If the events are statistically independent, the probability of a joint event is defined by the sum of the individual probabilites less the product of the probabilites. If the assumptions of additivity in Articulation Theory obtains, the relationship between the probabilites may then be expressed by stating that the probability of the sum of two or more independent bands is equal to the sum of the probabilities of the bands minus the product of the probabilities of these bands. For example, for two bands,

$$P(b1 + b2) = p(b1) + p(b2) - p(b1)p(b2)$$

Tables 9 and 10, following, provide the results of such computations for the twelve subject ear-conditions for ten two-band combinations. All probabilities have been converted back to percentage correct. Included in each table are the observed maximum scores for each band (obs), the observed maximum score for the sum band, and the score predicted by the above relationship (pre). An example of the calculations for the first case shown in Table 9 is given below.

The example applies to the case of Subject J.G., tested in quiet in her normally-hearing left ear. The summation tested is that of low-pass 700 Hz and high-pass 700 Hz, which combine to the wideband condition. From Table 2, the observed scores of 37% and 91% are obtained for the LP 700 Hz and HP 700 Hz conditions, respectively. Dividing each score by 100 gives the observed probabilities for each condition. Applying the relationship described, one obtains the following:

$$0.94 = .37 + .91 - (.37)(.91)$$

The value of 0.94 obtained is the predicted probability for the summation band, in this case the unfiltered or wideband condition, based on these observed scores. Multiplying 0.94 by 100 gives a predicted percentage correct score of 94% which, as seen in Table 2 and Table 9, corresponds to the observed score of 93.9% (or 94%) for the unfiltered condition.

TABLE 9

PROBABILITY ANALYSIS OF ADDITIVITY

NORMALLY-HEARING LISTENERS IN QUIET AND IN NOISE

LISTENER/CONDITION

FILTER	J.G. QUIET		J.G. NOISE		J.T. QUIET		J.T. NOISE		R.M. QUIET		R.M. NOISE	
	obs	pre	obs	pre	obs	pre	obs	pre	obs	pre	obs	pre
LP 700	37		37		33		41		40		35	
HP 700	91		92		90		86		91		82	
UNFILT.	94	94	95	95	94	93	88	92	92	95	86	88
LP 1400	63		58		58		58		58		54	
HP 1400	76		77		73		76		81		68	
UNFILT.	94	91	95	90	94	89	88	90	92	92	86	85
LP 2800	87		89		85		86		87		82	
HP 2800	38		49		30		46		56		32	
UNFILT.	94	92	95	94	94	90	88	92	92	94	86	88
LP 700	37		37		33		41		40		35	
BP 728	83		83		81		79		85		77	
LP 2800	87	89	89	89	85	87	86	88	87	91	82	85
LP 1400	63		58		58		58		58		54	
BP 1428	73		71		67		71		71		62	
LP 2800	87	90	89	88	85	86	86	88	87	88	82	83
BP 728	83		83		81		79		85		77	
HP 2800	38		49		30		46		56		32	
HP 700	91	89	92	91	90	87	86	89	91	93	82	84
BP 714	55		55		50		52		52		46	
HP 1400	76		77		73		76		81		68	
HP 700	91	94	92	90	90	87	86	88	91	91	82	83
LP 700	37		37		33		41		40		35	
BP 714	55		55		50		52		52		46	
LP 1400	63	72	58	72	58	67	58	72	58	71	54	65
BP 1428	73		71		67		71		71		62	
HP 2800	38		49		30		46		56		32	
HP 1400	76	83	77	85	73	77	76	84	81	87	68	74
BP 714	55		55		50		52		52		46	
BP 1428	73		71		67		71		71		62	
BP 728	83	88	83	87	81	84	79	86	85	86	77	79

TABLE 10

PROBABILITY ANALYSIS OF ADDITIVITY

HEARING IMPAIRED LISTENERS

LISTENER/CONDITION

	F.G.		F.G.		G.M.		G.M.		J.G.		T.T.	
FILTER	RIGHT		LEFT		RIGHT		LEFT		RIGHT		RIGHT	
	obs	pre	obs	pre	obs	pre	obs	pre	obs	pre	obs	pre
LP 700	38		36		28		29		31		30	
HP 700	83		83		58		75		72		65	
UNFILT.	80	89	71	89	58	70	78	82	70	81	67	76
LP 1400	53		52		31		47		44		39	
HP 1400	63		58		38		54		52		35	
UNFILT.	80	83	71	80	58	57	78	76	70	73	67	60
LP 2800	79		77		55		71		71		63	
HP 2800	35		35		24		30		23		11	
UNFILT.	80	86	71	85	58	66	78	80	70	78	67	67
LP 700	38		36		28		29		31		30	
BP 728	80		80		56		70		66		57	
LP 2800	79	88	77	87	55	68	71	79	71	77	63	70
LP 1400	53		52		31		47		44		39	
BP 1428	62		60		35		58		55		31	
LP 2800	79	82	77	81	55	55	71	78	71	75	63	58
BP 728	80		80		56		70		66		57	
HP 2800	35		35		24		30		23		11	
HP 700	83	87	83	87	58	67	75	79	72	74	65	62
BP 714	52		49		26		40		40		39	
HP 1400	63		58		38		54		52		35	
HP 700	83	82	83	79	58	54	75	72	72	71	65	60
LP 700	38		36		28		29		31		30	
BP 714	52		49		26		40		40		39	
LP 1400	53	70	52	67	31	47	47	57	44	59	39	57
BP 1428	62		60		35		58		55		31	
HP 2800	35		35		24		30		23		11	
HP 1400	63	75	58	74	38	51	54	71	52	65	35	39
BP 714	52		49		26		40		40		39	
BP 1428	62		60		35		58		55		31	
BP 728	80	82	80	80	56	52	70	75	66	73	57	58

In Table 9, one may see that for the normally-hearing listeners tested both in quiet and noise, the predicted values of percentage correct fall generally within 5 percentage points of the observed score for all but two summation conditions: LP 700 + BP 700-1400 = LP 1400 and BP 1428 + HP 2800 = HP 1400. For the first of these, the differences between observed and predicted is substantial, with the predicted score at least nine or more percentage points greater than the observed score.

For the impaired listeners shown in Table 10, significant differences between observed and predicted scores occurred for all summation cases that included the LP 700 Hz band. As for the normal listeners, the predicted score was higher than the observed score. Note also, that as for the normally-hearing subjects, predicted scores were higher than observed for BP 1428 + HP 2800 = HP 1400. It is clear from this analysis that the LP 700 Hz band has a negative influence on the performance of hearing impaired listeners, an observation made earlier when discussing the characteristics of the performance-intensity functions. It seems likely that on the basis of this analysis, further research into the effects of this low frequency energy may provide greater insight to the effects of the spread of masking in listeners with sensorineural hearing loss.

ARTICULATION THEORY PREDICTIONS - GENERAL CONSIDERATIONS

The overall results of the calculations of Articulation Index have shown that for the data obtained in this study, Articulation Theory did not accurately predict intelligibility performance of the listeners for all conditions evaluated. This was especially true for the hearing-impaired subjects and the normally-hearing subjects listening to speech masked with broadband shaped noise to simulated hearing loss. Although the selection of the reference curve used to predict intelligibility was not necessarily appropriate, some data points fell along this curve, indicating that a reference curve determined for some of the data points would not have differed greatly from the curve shown. However, such a curve must be obtained for each subject. This curve, once obtained, should apply to all conditions evaluated for any given subject, not just a few conditions. Nor should a different curve be required for a subject when tested in quiet and tested in a noise environment. These problems imply that a reexamination of the fundamental assumptions and parameters of Articulation Theory is necessary to determine the elements that might be modified to permit greater accuracy in predicting the performance of the listeners in this study.

In Articulation Theory, the primary variables are the

Band Efficiency and the Band Importance. The Band Efficiency is determined primarily from physical measurements that describe the communications channel. However, band efficiency is modified by the effects of the spread of masking that may further reduce the band efficiency if the noise in adjacent bands is sufficiently intense. Kryter's procedure (and, of course, the ANSI Standard procedure) incorporate corrections for both upward and downward spread of masking. Included in the procedures used in this research are additional corrections for the self-masking of speech as discussed by French and Steinberg. In the calculations performed for this research, hearing loss was modelled as an external noise. It is possible that the combination of the self masking of speech and spread of masking of this noise has great influence on the band efficiency of the higher frequency bands. Furthermore, for the hearing-impaired listeners, the effect of the hearing impairment is clearly greater than that of the noise alone.

The Band Importance weights were derived by Kryter (1962a) and others from empirically determined functions for normally-hearing listeners. However, since the normally-hearing listeners in this study exhibited the same problem with low freqeuncy energy as the listeners with hearing loss, a redetermination of the Band Importance weights may be necessary to more accurately reflect the significance of low frequency energy relative to high frequency for these speech materials. The normally-hearing listeners obviously have very good high frequency hearing; however, the impaired subjects may have a residual high frequency capability not accounted for in the calculations when intense low frequency information is removed.

The suggestions of Dugal, Braida and Durlach (1980) to expand the experimental studies to test Articulation Theory under a wider variety of conditions were met to a large degree by this study. One clear problem is the generally better performance of the listeners relative to the performance predicted by the theory. For hearing-impaired subjects with high frequency hearing loss the factors of the in-band and inter-band masking of speech itself must be better understood. The theory as presently stated does not entirely account for the improved performance of these subjects at high levels as low frequency spectral information is removed. Spread of masking of the low frequency speech energy clearly is one cause of the improved performance. However, according to the theory, except for the effects of spread of masking, removing any portion of the available speech spectrum will result in a reduced performance since the sum of the Band Importance weighting factors decreases. Any masking or hearing loss that removes any portion of the audible speech area must also by theory reduce performance.

Dugal, Braida and Durlach (1980) found articulation functions that predicted performance for hearing-impaired listeners with some accuracy, although they could not do so for normally-hearing listeners since articulation index values fell in a high range (A.I. > 0.85). This meant that predicted scores would be near 100%. They observed that a better fit of the results of their analysis was obtained if a proficiency factor of less than 1.0 was used for their normally-hearing listeners as well as their impaired listeners. Incorporating a proficiency factor to fit data to the results may be reasonable for untrained subjects. However, since all subjects in this study were highly trained, they were assumed to have a proficiency factor of 1.0. The only variable upon which the proficiency factor could operate here is the listener since all subjects heard the same speech materials spoken by the same talkers, and received essentially the same amount of training. For a subject listening under different test conditions, but assuming the same amount of training for each, the proficiency factor should remain the same. Thus, for the normally-hearing subjects tested in noise listening to the same talker, the proficiency factor must be the same as when the same subjects are tested in quiet. In addition, and perhaps more fundamental, is the issue of monotonicity under such conditions. As noted, earlier, for a given subject, talker, listening condition and speech materials, an single A.I. value should correspond to a single value of percentage correct. This was clearly not the case for the normally-hearing listeners when listening in quiet and in noise.

As discussed in Chapter III, investigators such as Pollack (1948) and Aniansson (1974) found reasonably close agreement between observed and predicted performance using Articulation Theory. Kryter (1962b) performed calculations using the results of several studies of speech perception and also found good agreement between observed and predicted scores. Wilber (1964) found that observed and predicted scores were close for her subjects with normal hearing and conductive impairments. In contrast, for her subjects with sensorineural hearing loss, predictions of "speech discrimination" were poor. For her listeners, however, the predicted scores were substantially higher than the observed scores. This is the opposite of the results of the A.I. calculations of the present report. Here, observed scores were generally much higher than the predicted scores, especially for the listeners with sensorineural hearing loss and the normally-hearing listeners with simulated hearing loss.

The calculations that were performed to test the basic assumption of additivity in the Articulation Theory procedures used in this study showed that the assumptions of additivity were reasonable. However, the assumption of monotonicity that resulted in curves such as those shown in Figures 1 and 42 - 53 did not hold for the data of the present study. The violation of this basic assumption must be thoroughly understood along with other modifications to the theory.

From the assumptions of independence of French and Steinberg and Beranek's twenty bands of equal contribution to the A.I., Kryter derived the Band Importance weights used in the standard methods of calculating A.I. and used in this report. Clearly, in the results of this study, the bands, as described by French and Steinberg and Beranek, did not appear to maintain equal contribution as intensity and available spectral information was changed. This was particularly true for the listeners with sensorineural hearing loss. Therefore, the Band Importance weights described by Kryter should be altered in some way to reflect the variation in intelligbility as spectral information and presentation level is varied and as the available residual hearing of an impaired subject is varied.

CHAPTER VIII

SUMMARY AND CONCLUSIONS

SUMMARY

Six subjects participated in a study to measure the perception of filtered speech and test the ability of Articulation Theory to predict intelligibility performance of these listeners. Two of the listeners had normal hearing, three had bilateral sensorineural hearing loss and one had a sensorineural loss in one ear and normal hearing in the other. Recorded consonant-vowel nonsense syllables were digitally filtered to produced nine conditions of low-pass, high-pass and band-pass filtered speech. All subjects listened to all conditions including unfiltered speech at a minimum of five different presentation levels ranging from near threshold to near discomfort level. All listeners heard the speech samples with no added background noise. In addition, the normally-hearing subjects listened to the speech materials in a background of shaped white noise designed to produce in these listeners masked auditory threshold shifts comparable to selected hearing-impaired subjects. All listening was done in a sound-proof room with headphones.

The basic data of this thesis consist of sets of performance-intensity functions for all the conditions heard by each subject. Using the model of Articulation Theory developed by Dugal, Braida and Durlach (1980), Articulation Index calculations were performed to determine the accuracy of predicted intelligibility performance versus the performance obtained by the listeners.

CONCLUSIONS

The results of this study have shown that:

(1) For the normally-hearing subjects listening in quiet and noise and the hearing-impaired subjects listening in quiet who participated in this study, speech intelligibility of the materials used remained relatively high whether high-pass filtered at 700 Hz or low-pass filtered at 2800 Hz, and heard with sufficient intensity.

(2) For all the listeners in this study, rollover in performance occurred at high presentation levels that were below reported discomfort thresholds. Identification of when that might occur was not specifically clear from the results of this study.

(3) For listeners with sloping sensorineural hearing loss, removing the high frequency energy from 2800 Hz up to the

4500 Hz cutoff of the material presented here, had little effect on reducing intelligibility. A more significant effect resulted from removing the energy below 700 Hz. At sufficiently high presentation levels, intellgibility performance usually equalled or exceeded that achieved for the unfiltered condition incorporating the spectral information below 700 Hz.

(4) The performance-intensity functions of the normally-hearing listeners with simulated high frequency hearing loss were similar to listeners with relatively flat sensorineural hearing loss. Eliminating low frequency energy for these subjects did not have as dramatic effect on their intelligibility performance as it did on the impaired subjects with mainly high frequency hearing loss. That is, the simulation of sloping loss in the normally-hearing listeners was more like a flat loss rather than a sloping loss.

(5) Estimated predictions of intelligibility using an Articulation curve derived from 1000-monosyllabic words were generally lower than the observed scores for most conditions and most subjects. Although somewhat spread out, A.I. values for the normally-hearing listeners when tested in quiet fell closer to the curve of expected performance than for these listeners when tested in noise and for the hearing-impaired listeners. Errors in the predictions were

generally systematic, in that observed performance scores for the narrowest band conditions showed the greatest difference from the predicted scores.

(6) The basic assumption of additivity of the Articulation Index and percentage correct scores predicted by it was reaffirmed by the data of this study. However, the assumption of monotonicity was found to be violated by the data observed here. For many conditions, different percentage correct scores were observed for the same value of Articulation Index. In addition, as Articulation Index increased, observed percentage correct score dropped.

(7) Methods of calculating the Articulation Index appear to require revision to account for the greater spread of masking of low frequency speech energy to high frequency regions and to account for the improved performance in both normally-hearing listeners and listeners with high frequency sensorineural hearing loss when low frequency energy is removed. Band importance weights must be revised to accurately reflect the relative contributions to intelligibility of individual frequency bands. Band efficiency must be determined to reflect the influence of adjacent bands on overall intelligibility, particularly low frequency bands. Also required is a thorough understanding of the lack of monotonicity of the data of this study.

APPENDIX A

RMS AND PEAK LEVELS FOR TALKERS

The following table indicates the 1/3-octave band pressure levels relative to 0 dB SPL for the four talkers used in this study. These values were used in the calculations of Articulation Index.

	TALKER D(F)		TALKER P(M)		TALKER B(M)		TALKER M(F)	
BAND	RMS	PEAK	RMS	PEAK	RMS	PEAK	RMS	PEAK
200	-14	-4	-14	-5	-12	-3	-15	-5
250	-10	2	-8	2	-6	3	-9	2
315	-7	5	-5	5	-5	5	-6	5
400	-5	5	-7	5	-8	3	-8	4
500	-14	-1	-16	-4	-17	-4	-7	6
630	-17	-4	-11	0	-14	-1	-15	-2
800	-15	-1	-9	3	-11	2	-12	2
1000	-13	0	-10	4	-11	3	-12	2
1250	-11	2	-13	0	-13	0	-18	-4
1600	-22	-8	-21	-6	-25	-12	-33	-20
2000	-32	-19	-23	-10	-23	-9	-36	-22
2500	-30	-17	-24	-11	-23	-10	-36	-22
3150	-30	-18	-25	-11	-31	-17	-30	-16
4000	-34	-21	-37	-23	-37	-23	-30	-16
5000	-49	-37	-44	-31	-46	-33	-42	-28

(F) Female Talker
(M) Male Talker

APPENDIX B

MASKING NOISE FOR SIMULATED HEARING LOSS

This appendix lists the one-third octave band sound pressure levels of the masking noise used to simulate hearing loss in each of the normally-hearing subjects. Values are coupler sound pressure levels using an NBS-9A earphone coupler, TDH-49 earphones and MX-41/AR cushions.

BAND CENTER FREQ.	BAND PRESSURE LEVEL (dB re 20 uPa)		
	SUBJECT: J.G.	J.T.	R.M.
200	24	24	24
250	26	26	26
315	28	30	28
400	32	30	31
500	34	33	34
630	36	35	37
800	38	36	40
1000	43	38	43
1250	47.5	39.5	48
1600	51	43	53
2000	54	45	57
2500	57	48	59.5
3150	60	51.5	64
4000	61	53	67.5
5000	61	53	66
Overall Level (C-weighted)	70	68	74

APPENDIX C

FREE FIELD CALIBRATION DATA

This appendix lists the differences in auditory threshold sound pressure levels at the one-third octave band center frequencies between the TDH-49/Zwislocki cushion earphones used in these experiments and the Telephonics Model 556; the free field correction determined by Lippmann (1981); and the total correction applied to the Articulation Index calculations. A level correction of -6 dB was applied to the presentation levels of speech signals to convert from overall sound pressure in the earphone coupler to the equivalent free field sound pressure.

BAND CENTER FREQ.	dB THRESHOLD DIFFERENCE TDH-49 vs. 556	dB FREE FIELD CORRECTION	OVERALL CORRECTION
200	-1.2	- 9.0	-10.2
250	+6.0	- 7.0	- 1.0
315	+1.8	- 4.0	- 2.2
400	+1.2	- 3.5	- 1.3
500	+1.4	- 6.5	- 5.1
630	+3.3	- 6.5	- 3.2
800	+3.4	- 3.5	- 0.1
1000	+1.8	- 6.0	- 4.2
1250	+2.2	- 5.0	- 2.8
1600	+1.3	- 4.0	- 2.7
2000	+5.9	- 2.5	+ 3.4
2500	+1.2	- 6.0	- 4.8
3150	-0.7	-10.0	-10.3
4000	-0.2	- 8.0	- 7.8
5000	-0.6	- 4.5	- 5.1

APPENDIX D

ANALYSIS OF VARIANCE

Analysis of Variance of data for presentation levels at which lowest and highest scores were achieved for each subject ear condition (12), each filter (10), and each talker (4). All data points transformed using arc-sine transformation.

ANALYSIS OF VARIANCE

ARC-SINE TRANSFORMATION

GRAND MEAN = 1.444

SUM OF SQUARES	MEAN SQUARE	DEGREES OF FREEDOM	F-RATIO	VARIANCE COMPONENT
65.137	7.24	9	65.8*	FILTER
1.622	0.54	3	54.0*	TALKER
2.955	0.11	27	11.0*	TALKERxFILTER
123.009	123.01	1	159.5*	LEVEL
13.688	1.52	9	19.0*	LEVELxFILTER
2.312	0.77	3	77.0*	LEVELxTALKER
2.252	0.08	27	8.0*	LEVELxTALKERxFILTER
28.133	2.56	11		SUBJECT
6.748	0.07	99		SUBJECTxFILTER
1.334	0.04	33		SUBJECTxTALKER
2.826	0.01	297		SUBJECTxTALKERxFILTER
4.880	0.44	11		SUBJECTxLEVEL
6.540	0.07	99		SUBJECTxLEVELxFILTER
0.389	0.01	33		SUBJECTxLEVELxTALKER
2.304	0.01	297		SUBJECTxLEVEL xTALKERxFILTER
264.129	0.28	959		TOTAL

*SIGNIFICANT P < 0.001

APPENDIX E

SUBJECT DATA

The following tables show the data used to plot the Performance - Intensity Functions and Articulation Index Functions. The calculated A.I. values and percentage correct scores were averaged over the four talkers.

For each filter condition, the sound pressure level is in dB free field SPL (re 20 uPa). (See Appendix C regarding the free field levels.)

The subjects' proficiency factor for the A.I. values in these tables is 1.00. The talker's proficiency factor is decribed in Chapter III.

In the "FILTER" column, the Band-Pass conditions are abbreviated as follows:

Band-Pass 700-1400 Hz: BP 714
Band-Pass 1400-2800 Hz: BP 1428
Band-Pass 700-2800 Hz: BP 728

In the "NO. OF TOKENS" column, presentation levels at which training occurred may be identified by numbers which are <u>not</u> integer multiples of 288 (i.e. 576, 864, etc).

The standard deviation is calculated across the four talkers to indicate the variability for each condition at each presentation level due to the talkers.

SUBJECT J. G. - LEFT EAR [NORMAL]

FILTER	LEVEL dB SPL	CALC. A.I.	PERCENTAGE CORRECT	NO. OF TOKENS	STD. DEV.
UNFILT.	28	0.3219	30.21	288	11.64
UNFILT.	32	0.4215	41.67	288	9.29
UNFILT.	38	0.5647	60.07	288	5.82
UNFILT.	42	0.6375	73.61	576	4.21
UNFILT.	52	0.7608	88.02	576	3.18
UNFILT.	62	0.8167	93.92	576	3.37
UNFILT.	72	0.8458	85.96	1406	2.36
LP 700	46	0.1757	21.87	288	3.28
LP 700	52	0.1872	25.35	288	2.09
LP 700	56	0.1892	33.33	288	4.54
LP 700	66	0.1897	35.16	791	5.75
LP 700	76	0.1897	36.41	756	4.91
LP 700	86	0.1897	37.15	288	4.59
LP 1400	40	0.3763	30.21	576	3.47
LP 1400	46	0.3906	43.58	576	7.04
LP 1400	52	0.3938	53.30	576	1.43
LP 1400	56	0.3953	58.85	576	5.30
LP 1400	66	0.3958	60.77	2064	2.15
LP 1400	76	0.3958	62.62	795	1.96
LP 1400	86	0.3958	61.11	576	5.86
LP 2800	30	0.3630	30.90	288	6.15
LP 2800	32	0.4099	43.75	288	8.82
LP 2800	36	0.4989	60.07	288	8.21
LP 2800	42	0.5932	69.10	288	3.99
LP 2800	46	0.6339	79.17	288	6.31
LP 2800	56	0.6823	87.16	288	5.36
LP 2800	66	0.6876	86.57	545	4.00
LP 2800	76	0.6876	86.81	288	3.68
HP 700	36	0.3668	52.78	576	7.28
HP 700	42	0.4590	66.14	576	6.79
HP 700	46	0.5105	75.52	576	4.55
HP 700	52	0.5731	85.42	576	4.28
HP 700	56	0.6030	90.98	647	4.19
HP 700	66	0.6388	88.03	701	3.32
HP 700	76	0.6721	90.63	576	3.52

SUBJECT J. G. - LEFT EAR [NORMAL]

FILTER	LEVEL dB SPL	CALC. A.I.	PERCENTAGE CORRECT	NO. OF TOKENS	STD. DEV.
HP 1400	36	0.1733	34.03	576	6.44
HP 1400	42	0.2537	42.53	576	6.77
HP 1400	46	0.3044	55.21	864	7.92
HP 1400	56	0.3969	74.65	576	5.21
HP 1400	66	0.4338	76.53	1090	6.01
HP 1400	76	0.4970	73.24	1085	5.19
HP 1400	86	0.5642	71.68	1066	6.35
HP 2800	46	0.0610	14.93	288	6.35
HP 2800	50	0.0816	27.26	576	8.90
HP 2800	56	0.1098	33.69	576	10.46
HP 2800	66	0.1409	38.20	576	11.01
HP 2800	76	0.1689	37.16	623	12.70
HP 2800	86	0.2166	38.09	875	8.28
BP 714	46	0.2061	29.51	288	5.71
BP 714	52	0.2061	39.24	288	2.37
BP 714	56	0.2061	47.22	288	4.09
BP 714	66	0.2061	55.21	288	4.73
BP 714	76	0.2149	53.99	605	1.89
BP 714	86	0.2438	48.26	288	1.75
BP 1428	36	0.1495	23.61	288	12.57
BP 1428	42	0.2094	40.97	288	6.16
BP 1428	46	0.2433	52.09	288	9.45
BP 1428	56	0.2871	60.76	288	8.05
BP 1428	66	0.2929	72.92	288	6.66
BP 1428	76	0.3316	71.62	769	6.25
BP 1428	86	0.3823	70.49	288	6.55
BP 728	36	0.3429	41.84	576	7.94
BP 728	42	0.4146	59.20	576	7.72
BP 728	46	0.4494	71.53	576	3.88
BP 728	56	0.4932	81.94	576	4.20
BP 728	66	0.4979	82.68	1383	2.31
BP 728	76	0.5067	79.59	1234	4.17

SUBJECT J. T. - NORMALLY-HEARING

FILTER	LEVEL dB SPL	CALC. A.I.	PERCENTAGE CORRECT	NO. OF TOKENS	STD. DEV.
UNFILT.	22	0.2984	21.88	288	7.64
UNFILT.	28	0.4651	29.86	288	8.53
UNFILT.	32	0.5780	59.03	288	11.43
UNFILT.	38	0.7194	68.40	288	7.89
UNFILT.	42	0.7885	84.72	288	4.68
UNFILT.	52	0.8943	88.89	288	3.76
UNFILT.	62	0.9227	94.79	288	2.86
UNFILT.	72	0.9361	91.34	606	2.36
LP 700	46	0.1979	24.31	288	3.31
LP 700	56	0.2020	29.17	288	2.27
LP 700	66	0.2020	36.95	880	2.82
LP 700	76	0.2020	37.39	898	3.77
LP 700	86	0.2020	34.29	504	1.51
LP 1400	40	0.4013	30.04	576	3.82
LP 1400	46	0.4073	40.10	576	3.94
LP 1400	52	0.4096	47.22	576	4.50
LP 1400	56	0.4106	51.57	576	3.69
LP 1400	66	0.4106	57.84	1025	2.10
LP 1400	76	0.4106	57.92	981	2.63
LP 2800	36	0.5907	47.22	288	10.58
LP 2800	42	0.6640	62.85	288	2.87
LP 2800	46	0.6887	73.26	288	4.73
LP 2800	56	0.7173	81.25	288	2.66
LP 2800	66	0.7173	83.97	867	2.98
LP 2800	76	0.7173	88.89	288	4.95
HP 700	36	0.4911	49.83	576	5.82
HP 700	42	0.5919	65.80	576	3.32
HP 700	46	0.6408	78.30	576	5.42
HP 700	52	0.6933	84.72	576	3.93
HP 700	56	0.7094	89.08	869	4.00
HP 700	66	0.7320	92.39	916	3.00
HP 700	76	0.7684	90.80	576	4.07

SUBJECT J. T. - NORMALLY-HEARING

FILTER	LEVEL dB SPL	CALC. A.I.	PERCENTAGE CORRECT	NO. OF TOKENS	STD. DEV.
HP 1400	36	0.2904	28.82	576	10.15
HP 1400	42	0.3834	38.55	576	12.02
HP 1400	46	0.4322	58.86	576	10.73
HP 1400	56	0.5008	70.66	576	8.81
HP 1400	66	0.5209	74.26	1158	7.52
HP 1400	76	0.5710	77.26	1307	6.15
HP 2800	46	0.1508	24.31	576	7.46
HP 2800	52	0.1823	28.47	576	5.26
HP 2800	56	0.1941	35.42	576	10.50
HP 2800	62	0.2054	32.81	576	5.45
HP 2800	66	0.2127	42.19	576	6.30
HP 2800	76	0.2389	38.72	621	8.29
HP 2800	86	0.2910	47.47	743	12.01
HP 2800	96	0.3586	49.65	288	15.02
BP 714	46	0.2086	33.16	576	4.40
BP 714	52	0.2086	36.63	576	1.54
BP 714	56	0.2086	42.94	864	1.90
BP 714	62	0.2098	45.14	288	3.30
BP 714	66	0.2125	54.01	689	10.04
BP 714	76	0.2417	53.27	767	6.74
BP 714	86	0.2735	54.86	288	7.57
BP 1428	46	0.2813	38.37	576	6.55
BP 1428	56	0.3067	54.00	576	7.11
BP 1428	66	0.3081	65.45	576	6.45
BP 1428	76	0.3510	71.25	1069	7.64
BP 1428	86	0.4018	70.59	1099	9.60
BP 1428	96	0.4512	69.79	576	6.91
BP 728	40	0.4504	42.19	576	4.86
BP 728	46	0.4899	56.25	576	2.27
BP 728	52	0.5109	70.83	576	3.72
BP 728	56	0.5153	77.95	693	2.02
BP 728	66	0.5193	79.80	1159	4.85
BP 728	76	0.5484	82.55	1300	4.37
BP 728	86	0.5802	82.99	576	5.96

SUBJECTS J. G. & J. T. COMBINED [NORMALLY-HEARING]

FILTER	LEVEL dB SPL	CALC. A.I.	PERCENTAGE CORRECT	NO. OF TOKENS	STD. DEV.
UNFILT.	22	0.2607	21.88	288	7.64
UNFILT.	28	0.4165	30.04	576	8.70
UNFILT.	32	0.5310	50.35	576	10.06
UNFILT.	38	0.6796	64.24	576	5.93
UNFILT.	42	0.7509	77.32	864	3.91
UNFILT.	52	0.8679	88.31	864	2.68
UNFILT.	62	0.9086	94.21	864	2.98
UNFILT.	72	0.9256	87.53	2012	1.13
LP 700	46	0.1901	23.09	576	2.50
LP 700	52	0.1966	25.35	288	2.09
LP 700	56	0.1976	31.25	576	2.54
LP 700	66	0.1976	36.10	1671	4.13
LP 700	76	0.1976	36.93	1654	4.16
LP 700	86	0.1976	35.33	792	1.76
LP 1400	40	0.3930	30.12	1152	3.35
LP 1400	46	0.4017	41.84	1152	5.26
LP 1400	52	0.4039	50.26	1152	2.35
LP 1400	56	0.4049	55.21	1152	3.17
LP 1400	66	0.4049	59.78	3089	0.84
LP 1400	76	0.4049	60.01	1776	1.99
LP 1400	86	0.4049	61.11	576	5.86
LP 2800	30	0.4407	30.90	288	6.15
LP 2800	32	0.4883	43.75	288	8.82
LP 2800	36	0.5713	53.65	576	8.92
LP 2800	42	0.6521	65.97	576	2.48
LP 2800	46	0.6791	76.22	576	4.99
LP 2800	56	0.7098	84.21	576	2.30
LP 2800	66	0.7098	84.96	1412	3.15
LP 2800	76	0.7098	87.85	576	3.78
HP 700	36	0.4622	51.31	1152	6.17
HP 700	42	0.5598	65.97	1152	4.63
HP 700	46	0.6101	76.91	1152	4.43
HP 700	52	0.6713	85.07	1152	3.32
HP 700	56	0.6939	89.87	1516	3.98
HP 700	66	0.7193	90.44	1617	2.59
HP 700	76	0.7507	90.71	1152	3.56

SUBJECTS J. G. & J. T. COMBINED [NORMALLY-HEARING]

FILTER	LEVEL dB SPL	CALC. A.I.	PERCENTAGE CORRECT	NO. OF TOKENS	STD. DEV.
HP 1400	36	0.2628	31.42	1152	8.05
HP 1400	42	0.3525	40.54	1152	9.18
HP 1400	46	0.4028	56.67	1440	8.79
HP 1400	56	0.4866	72.66	1152	6.82
HP 1400	66	0.5125	75.40	2248	6.53
HP 1400	76	0.5676	75.42	2392	5.61
HP 1400	86	0.6192	71.68	1066	6.35
HP 2800	46	0.1253	21.19	864	6.93
HP 2800	50	0.1514	27.26	576	8.90
HP 2800	52	0.1645	28.47	576	5.26
HP 2800	56	0.1817	34.55	1152	10.35
HP 2800	62	0.1988	32.81	576	5.45
HP 2800	66	0.2062	40.19	1152	8.54
HP 2800	76	0.2348	38.00	1244	10.49
HP 2800	86	0.2849	42.37	1618	9.83
HP 2800	96	0.3521	49.65	288	15.02
BP 714	46	0.2073	31.94	864	3.95
BP 714	52	0.2073	37.50	864	1.36
BP 714	56	0.2073	44.01	1152	1.48
BP 714	62	0.2073	45.14	288	3.30
BP 714	66	0.2082	54.32	977	8.35
BP 714	76	0.2274	53.65	1372	3.04
BP 714	86	0.2587	51.56	576	3.98
BP 1428	36	0.1982	23.61	288	12.57
BP 1428	42	0.2537	40.97	288	6.16
BP 1428	46	0.2774	42.94	864	6.82
BP 1428	56	0.3049	56.25	864	7.15
BP 1428	66	0.3063	67.94	864	6.14
BP 1428	76	0.3492	71.43	1838	6.95
BP 1428	86	0.4000	70.58	1387	8.95
BP 1428	96	0.4493	69.79	576	6.91
BP 728	36	0.3976	41.84	576	7.94
BP 728	40	0.4434	42.19	576	4.86
BP 728	42	0.4610	59.20	576	7.72
BP 728	46	0.4848	63.89	1152	2.51
BP 728	52	0.5068	70.83	576	3.72
BP 728	56	0.5122	79.76	1269	2.25
BP 728	66	0.5131	81.36	2542	3.37
BP 728	76	0.5323	81.10	2534	4.15
BP 728	86	0.5636	82.99	576	5.96

SUBJECT R. M. - NORMALLY-HEARING

FILTER	LEVEL dB SPL	CALC. A.I.	PERCENTAGE CORRECT	NO. OF TOKENS	STD. DEV.
UNFILT.	32	0.1531	26.39	576	4.68
UNFILT.	38	0.2674	40.28	576	3.93
UNFILT.	42	0.3612	62.15	576	3.51
UNFILT.	52	0.5674	76.56	576	2.36
UNFILT.	62	0.6896	90.39	864	3.43
UNFILT.	72	0.7436	87.83	828	3.65
UNFILT.	82	0.7599	86.21	514	8.12
LP 700	46	0.1170	21.87	288	3.08
LP 700	56	0.1499	25.34	288	6.15
LP 700	66	0.1575	32.99	288	3.65
LP 700	76	0.1590	33.09	536	3.30
LP 700	82	0.1590	30.00	310	4.75
LP 700	86	0.1590	31.77	619	3.21
LP 1400	40	0.2233	32.29	576	4.73
LP 1400	46	0.2868	41.32	576	0.89
LP 1400	56	0.3213	53.68	1080	2.26
LP 1400	66	0.3263	54.10	1182	3.96
LP 1400	76	0.3277	57.49	642	4.77
LP 2800	36	0.2237	34.90	576	5.33
LP 2800	42	0.3432	52.60	576	4.26
LP 2800	46	0.4201	65.62	576	8.15
LP 2800	56	0.5363	81.25	576	5.04
LP 2800	66	0.5744	81.35	798	3.58
LP 2800	76	0.5779	85.20	804	3.91
HP 700	40	0.2084	46.70	576	5.27
HP 700	46	0.3206	61.29	576	5.42
HP 700	52	0.4193	73.96	576	6.43
HP 700	56	0.4711	81.89	762	6.54
HP 700	66	0.5633	87.75	1297	4.65
HP 700	76	0.5930	89.54	1107	4.40
HP 700	86	0.6190	89.24	576	4.76

SUBJECT R. M. - NORMALLY-HEARING

FILTER	LEVEL dB SPL	CALC. A.I.	PERCENTAGE CORRECT	NO. OF TOKENS	STD. DEV.
HP 1400	46	0.1674	37.15	288	10.90
HP 1400	52	0.2517	50.35	576	10.02
HP 1400	56	0.3023	59.38	576	11.17
HP 1400	66	0.3945	70.15	1041	10.34
HP 1400	76	0.4235	73.30	1486	9.30
HP 2800	60	0.1123	19.27	576	5.05
HP 2800	66	0.1465	23.44	576	4.79
HP 2800	76	0.1733	30.23	939	9.35
HP 2800	86	0.1876	26.19	897	8.84
BP 714	46	0.1533	26.39	288	2.27
BP 714	52	0.1676	36.46	288	2.37
BP 714	56	0.1688	43.06	288	5.67
BP 714	66	0.1688	49.84	651	7.26
BP 714	76	0.1696	46.49	607	7.41
BP 1428	46	0.1333	34.90	576	12.10
BP 1428	56	0.2150	53.47	576	9.02
BP 1428	66	0.2481	67.01	576	7.11
BP 1428	76	0.2503	63.86	761	6.23
BP 1428	86	0.2676	62.28	751	8.17
BP 728	46	0.2865	48.78	576	7.31
BP 728	52	0.3547	66.49	576	3.42
BP 728	56	0.3837	76.74	576	1.75
BP 728	66	0.4168	76.69	1100	5.01
BP 728	76	0.4198	81.03	1277	4.39
BP 728	86	0.4346	80.44	864	6.67

SUBJECT F. G. - RIGHT EAR [IMPAIRED]

FILTER	LEVEL dB SPL	CALC. A.I.	PERCENTAGE CORRECT	NO. OF TOKENS	STD. DEV.
UNFILT.	54	0.4128	48.61	1152	10.94
UNFILT.	60	0.4608	56.77	576	16.66
UNFILT.	64	0.4896	67.78	598	14.77
UNFILT.	70	0.5209	72.74	576	10.51
UNFILT.	74	0.5372	73.10	771	7.56
UNFILT.	84	0.5672	76.28	1382	7.16
UNFILT.	94	0.5714	79.71	1063	4.29
UNFILT.	104	0.4939	71.88	864	8.74
LP 700	48	0.1768	26.39	576	4.68
LP 700	58	0.1895	28.12	576	3.99
LP 700	64	0.1899	33.16	576	6.67
LP 700	68	0.1899	35.03	983	8.77
LP 700	78	0.1899	37.65	916	9.53
LP 700	88	0.1894	32.81	576	7.95
LP 1400	38	0.2571	23.09	576	2.62
LP 1400	48	0.3337	37.67	576	9.34
LP 1400	58	0.3534	49.46	928	9.22
LP 1400	68	0.3538	52.64	1434	2.83
LP 1400	78	0.3538	48.55	714	8.91
LP 1400	88	0.3533	45.48	576	11.84
LP 1400	98	0.3232	40.63	576	9.99
LP 2800	48	0.3620	38.37	576	7.95
LP 2800	58	0.4456	54.51	576	11.95
LP 2800	68	0.5108	71.54	1046	9.23
LP 2800	78	0.5412	79.45	1135	5.95
LP 2800	88	0.5439	78.12	576	9.69
LP 2800	98	0.5138	78.99	576	8.08
HP 700	58	0.2561	51.04	576	9.26
HP 700	64	0.2999	58.85	576	12.66
HP 700	68	0.3225	71.18	864	9.65
HP 700	78	0.3776	77.73	1264	3.74
HP 700	88	0.4319	81.28	1092	3.47
HP 700	98	0.4651	83.35	1009	2.86
HP 700	108	0.4212	76.39	288	3.59

SUBJECT F. G. - RIGHT EAR [IMPAIRED]

FILTER	LEVEL dB SPL	CALC. A.I.	PERCENTAGE CORRECT	NO. OF TOKENS	STD. DEV.
HP 1400	68	0.1570	33.91	864	8.89
HP 1400	74	0.1834	41.90	864	9.74
HP 1400	78	0.1979	53.01	864	8.47
HP 1400	88	0.2362	62.85	1257	9.49
HP 1400	98	0.2880	62.81	1942	12.41
HP 1400	108	0.3307	54.34	576	13.29
HP 2800	88	0.0308	15.98	576	5.10
HP 2800	94	0.0394	23.44	576	6.24
HP 2800	98	0.0409	30.81	869	4.98
HP 2800	104	0.0533	34.98	883	4.92
HP 2800	108	0.0634	31.85	1268	6.16
HP 2800	114	0.0888	30.21	576	6.33
HP 2800	118	0.1074	22.92	288	7.39
BP 714	52	0.1586	21.18	576	3.61
BP 714	58	0.1640	32.29	576	5.60
BP 714	64	0.1641	37.85	576	5.36
BP 714	68	0.1655	47.40	576	5.85
BP 714	78	0.1797	51.55	956	4.18
BP 714	88	0.2105	45.96	1006	4.45
BP 1428	68	0.1570	30.73	576	10.34
BP 1428	74	0.1803	44.97	576	9.13
BP 1428	78	0.1874	53.30	576	10.57
BP 1428	84	0.1962	59.02	696	9.54
BP 1428	88	0.2054	61.74	1080	11.31
BP 1428	98	0.2471	60.94	576	12.61
BP 728	58	0.2561	44.56	864	12.19
BP 728	64	0.2999	56.43	576	10.84
BP 728	68	0.3225	67.71	864	9.83
BP 728	78	0.3671	74.65	1120	3.86
BP 728	88	0.4012	78.60	1500	5.16
BP 728	98	0.4242	79.57	1083	2.98
BP 728	108	0.3703	76.04	288	5.12

SUBJECT F. G. - LEFT EAR [IMPAIRED]

FILTER	LEVEL dB SPL	CALC. A.I.	PERCENTAGE CORRECT	NO. OF TOKENS	STD. DEV.
UNFILT.	54	0.3913	42.94	864	8.77
UNFILT.	60	0.4267	50.35	576	11.54
UNFILT.	64	0.4520	60.24	576	13.97
UNFILT.	70	0.4805	68.06	576	11.82
UNFILT.	74	0.4976	71.11	765	10.04
UNFILT.	84	0.5207	67.99	1342	12.11
UNFILT.	94	0.5161	70.67	1070	11.81
UNFILT.	104	0.4442	63.19	864	10.23
LP 700	52	0.1749	25.47	864	1.73
LP 700	58	0.1819	28.82	576	6.35
LP 700	64	0.1823	31.25	576	7.63
LP 700	68	0.1823	33.15	927	10.32
LP 700	78	0.1823	35.63	883	12.07
LP 700	88	0.1817	35.30	864	4.54
LP 1400	38	0.2632	21.70	576	5.15
LP 1400	48	0.3394	34.20	576	4.69
LP 1400	58	0.3539	47.23	576	11.99
LP 1400	68	0.3542	52.37	1554	5.72
LP 1400	78	0.3542	46.15	969	8.12
LP 1400	88	0.3537	49.48	576	10.68
LP 1400	98	0.3224	38.19	576	10.81
LP 2800	48	0.3580	32.47	576	7.45
LP 2800	58	0.4139	50.70	576	13.70
LP 2800	68	0.4726	64.95	1031	13.08
LP 2800	78	0.5123	74.69	1143	9.57
LP 2800	88	0.5232	76.91	576	11.43
LP 2800	98	0.4934	71.70	576	11.38
HP 700	58	0.2320	46.01	576	9.44
HP 700	64	0.2698	48.44	576	12.32
HP 700	68	0.2904	66.78	864	7.45
HP 700	78	0.3362	75.60	1235	5.46
HP 700	88	0.3773	82.76	1136	5.32
HP 700	98	0.4118	82.83	1027	4.99
HP 700	108	0.3735	81.60	288	3.08

SUBJECT F. G. - LEFT EAR [IMPAIRED]

FILTER	LEVEL dB SPL	CALC. A.I.	PERCENTAGE CORRECT	NO. OF TOKENS	STD. DEV.
HP 1400	68	0.1184	25.70	864	9.07
HP 1400	74	0.1434	34.21	1152	10.38
HP 1400	78	0.1621	47.09	1385	8.04
HP 1400	88	0.2000	57.82	1442	6.29
HP 1400	98	0.2552	56.27	1453	7.63
HP 1400	108	0.2930	49.77	864	12.51
HP 2800	88	0.0004	9.72	576	2.89
HP 2800	94	0.0040	14.59	576	0.99
HP 2800	98	0.0098	25.93	864	3.98
HP 2800	104	0.0177	30.94	887	4.08
HP 2800	108	0.0253	34.13	981	4.33
HP 2800	114	0.0335	33.25	627	9.06
HP 2800	118	0.0423	25.35	288	7.38
BP 714	52	0.1694	20.83	576	4.57
BP 714	58	0.1719	27.60	576	6.35
BP 714	64	0.1719	41.50	576	5.21
BP 714	68	0.1719	45.32	576	6.74
BP 714	78	0.1781	48.75	922	4.32
BP 714	88	0.2073	47.12	998	6.03
BP 1428	68	0.1184	23.96	576	8.26
BP 1428	74	0.1434	35.25	576	7.18
BP 1428	78	0.1621	47.40	576	8.14
BP 1428	84	0.1829	55.56	576	11.04
BP 1428	88	0.1995	59.60	728	5.08
BP 1428	98	0.2454	59.02	797	12.31
BP 728	58	0.2320	38.89	864	9.52
BP 728	64	0.2698	45.14	576	13.66
BP 728	68	0.2904	62.73	864	11.76
BP 728	78	0.3362	75.46	1105	6.75
BP 728	88	0.3769	79.50	1517	3.79
BP 728	98	0.4020	78.85	1083	4.69
BP 728	108	0.3535	65.28	288	8.71

SUBJECT G. M. - RIGHT EAR [IMPAIRED]

FILTER	LEVEL dB SPL	CALC. A.I.	PERCENTAGE CORRECT	NO. OF TOKENS	STD. DEV.
UNFILT.	54	0.1370	18.71	731	3.88
UNFILT.	64	0.2446	30.73	576	4.51
UNFILT.	74	0.3442	47.92	576	5.87
UNFILT.	84	0.4097	58.51	576	7.74
UNFILT.	94	0.4339	55.08	1414	6.56
UNFILT.	104	0.3880	57.68	1113	3.26
LP 700	58	0.1030	17.71	576	2.30
LP 700	68	0.1235	25.00	576	4.28
LP 700	78	0.1303	27.95	576	3.73
LP 700	88	0.1300	26.46	691	1.82
LP 700	98	0.1110	26.22	640	3.94
LP 1400	48	0.0864	10.76	288	1.75
LP 1400	58	0.1615	17.59	864	1.31
LP 1400	68	0.2133	23.15	864	4.68
LP 1400	78	0.2238	26.63	1286	7.89
LP 1400	88	0.2234	31.12	1370	3.27
LP 1400	92	0.2180	27.08	576	3.00
LP 1400	98	0.1995	28.23	1391	2.96
LP 2800	58	0.1785	19.44	576	5.04
LP 2800	64	0.2444	30.21	288	6.65
LP 2800	68	0.2835	33.68	864	9.28
LP 2800	78	0.3501	45.66	576	6.84
LP 2800	84	0.3654	47.73	711	6.87
LP 2800	88	0.3698	55.12	747	6.70
LP 2800	98	0.3463	54.50	988	7.12
HP 700	72	0.1963	26.74	576	6.35
HP 700	78	0.2435	40.51	864	5.02
HP 700	84	0.2793	51.16	864	4.38
HP 700	88	0.2972	57.89	1764	4.11
HP 700	98	0.3221	53.78	1675	6.46
HP 700	108	0.3113	55.90	576	11.87

SUBJECT G. M. - RIGHT EAR [IMPAIRED]

FILTER	LEVEL dB SPL	CALC. A.I.	PERCENTAGE CORRECT	NO. OF TOKENS	STD. DEV.
HP 1400	82	0.1749	20.26	864	6.23
HP 1400	88	0.2038	26.04	1637	9.06
HP 1400	94	0.2204	37.60	1006	10.07
HP 1400	98	0.2293	31.43	2799	11.08
HP 1400	104	0.2386	36.92	864	10.07
HP 1400	108	0.2377	35.40	625	9.52
HP 2800	88	0.0574	15.28	288	2.27
HP 2800	98	0.0825	20.81	541	5.13
HP 2800	104	0.0924	22.57	288	5.24
HP 2800	108	0.0976	19.52	948	7.71
HP 2800	114	0.1084	23.99	613	7.70
BP 714	78	0.0934	13.02	576	2.08
BP 714	84	0.0934	18.58	576	4.10
BP 714	88	0.0934	23.67	653	3.51
BP 714	94	0.0947	26.04	576	5.68
BP 714	98	0.0927	25.68	631	3.92
BP 714	104	0.0821	23.79	576	8.70
BP 1428	88	0.1464	24.65	576	8.65
BP 1428	94	0.1468	31.60	576	12.80
BP 1428	98	0.1468	34.37	695	8.18
BP 1428	104	0.1462	32.99	576	10.17
BP 1428	108	0.1401	35.22	650	11.05
BP 1428	118	0.1232	27.26	576	15.09
BP 728	68	0.1539	16.90	864	4.17
BP 728	74	0.2005	26.05	576	8.64
BP 728	78	0.2198	34.56	1024	7.14
BP 728	88	0.2399	45.77	1426	4.10
BP 728	94	0.2415	47.10	313	11.41
BP 728	98	0.2396	50.32	1316	7.83
BP 728	104	0.2283	55.56	576	7.08

SUBJECT G. M. - LEFT EAR [IMPAIRED]

FILTER	LEVEL dB SPL	CALC. A.I.	PERCENTAGE CORRECT	NO. OF TOKENS	STD. DEV.
UNFILT.	44	0.2332	27.60	576	7.40
UNFILT.	50	0.3019	36.98	576	7.20
UNFILT.	54	0.3356	50.52	576	10.32
UNFILT.	64	0.3983	69.10	576	10.54
UNFILT.	74	0.4358	78.30	576	4.99
UNFILT.	84	0.4618	69.17	1773	7.07
UNFILT.	94	0.4757	73.17	692	8.75
UNFILT.	100	0.4505	63.98	606	9.79
LP 700	48	0.1343	17.02	288	1.74
LP 700	58	0.1592	23.61	576	2.60
LP 700	64	0.1650	21.18	288	4.15
LP 700	68	0.1650	28.47	576	3.16
LP 700	78	0.1650	29.00	900	4.25
LP 700	82	0.1650	23.61	288	5.89
LP 700	88	0.1646	25.77	647	4.21
LP 1400	48	0.2679	34.26	864	3.83
LP 1400	52	0.2943	36.46	288	4.99
LP 1400	58	0.3118	46.99	864	2.46
LP 1400	68	0.3142	47.02	1405	7.49
LP 1400	78	0.3142	44.67	1714	5.72
LP 1400	88	0.3137	38.55	576	6.72
LP 1400	98	0.2826	34.37	288	4.73
LP 2800	48	0.2804	38.72	576	6.86
LP 2800	58	0.3634	53.65	576	12.29
LP 2800	64	0.3983	66.32	288	15.93
LP 2800	68	0.4157	66.87	984	11.18
LP 2800	78	0.4479	71.24	809	11.33
LP 2800	88	0.4655	65.62	576	14.12
HP 700	58	0.1988	48.38	864	6.85
HP 700	64	0.2333	53.39	1152	4.64
HP 700	68	0.2507	67.60	1051	4.03
HP 700	78	0.2829	71.90	1752	4.16
HP 700	88	0.3148	75.19	1069	4.53
HP 700	98	0.3580	68.63	864	7.53

SUBJECT G. M. - LEFT EAR [IMPAIRED]

FILTER	LEVEL dB SPL	CALC. A.I.	PERCENTAGE CORRECT	NO. OF TOKENS	STD. DEV.
HP 1400	62	0.0733	27.78	576	7.08
HP 1400	68	0.1015	40.98	864	5.22
HP 1400	78	0.1337	47.28	1035	5.13
HP 1400	88	0.1622	53.84	1083	2.24
HP 1400	98	0.2169	52.89	864	6.57
HP 1400	108	0.2626	38.54	288	8.59
HP 2800	88	0.0068	20.31	576	3.94
HP 2800	94	0.0172	24.48	576	6.55
HP 2800	98	0.0233	29.69	576	9.17
HP 2800	104	0.0304	20.25	949	6.76
HP 2800	108	0.0388	14.10	887	5.46
BP 714	48	0.1176	17.74	213	6.02
BP 714	58	0.1472	27.08	576	5.98
BP 714	64	0.1492	34.90	576	1.74
BP 714	68	0.1492	39.94	576	1.65
BP 714	78	0.1492	39.84	747	5.39
BP 714	88	0.1562	34.91	693	0.92
BP 1428	68	0.1015	37.85	576	7.03
BP 1428	74	0.1216	44.62	576	7.80
BP 1428	78	0.1337	51.77	820	5.51
BP 1428	84	0.1458	58.33	576	8.51
BP 1428	88	0.1554	52.88	773	5.68
BP 1428	98	0.1936	48.61	288	12.00
BP 728	58	0.1988	39.93	864	6.98
BP 728	64	0.2333	56.42	576	8.57
BP 728	68	0.2507	63.72	576	4.52
BP 728	78	0.2829	69.55	1625	4.86
BP 728	84	0.2966	68.53	308	6.36
BP 728	88	0.3080	68.04	1109	2.68
BP 728	98	0.3348	61.28	576	9.50

SUBJECT J. G. - RIGHT EAR [IMPAIRED]

FILTER	LEVEL dB SPL	CALC. A.I.	PERCENTAGE CORRECT	NO. OF TOKENS	STD. DEV.
UNFILT.	54	0.3270	11.46	288	2.87
UNFILT.	64	0.4365	25.00	576	5.04
UNFILT.	70	0.4833	39.58	288	10.55
UNFILT.	74	0.5056	54.17	288	17.67
UNFILT.	84	0.5429	62.23	706	6.76
UNFILT.	94	0.5461	68.27	806	9.54
UNFILT.	104	0.4756	69.45	576	3.35
LP 700	62	0.1464	12.50	288	4.40
LP 700	68	0.1513	18.75	288	5.13
LP 700	74	0.1513	19.97	576	5.05
LP 700	78	0.1513	25.17	576	3.60
LP 700	88	0.1509	30.76	623	4.31
LP 700	98	0.1304	30.80	634	3.64
LP 700	104	0.1079	30.56	288	5.89
LP 1400	58	0.2849	20.31	576	3.74
LP 1400	64	0.2901	26.57	576	6.98
LP 1400	68	0.2901	32.99	864	5.27
LP 1400	78	0.2901	41.84	1544	6.02
LP 1400	88	0.2897	43.75	864	5.51
LP 1400	98	0.2599	43.58	576	3.65
LP 2800	58	0.3747	21.18	288	3.08
LP 2800	64	0.4263	31.25	288	5.13
LP 2800	68	0.4491	34.72	288	7.86
LP 2800	74	0.4710	52.08	288	10.73
LP 2800	78	0.4783	59.92	699	12.69
LP 2800	88	0.4797	62.15	288	11.36
LP 2800	98	0.4499	70.83	288	1.97
HP 700	68	0.3177	31.77	576	5.39
HP 700	74	0.3543	44.27	576	5.93
HP 700	78	0.3726	58.15	702	5.45
HP 700	88	0.3994	66.10	792	5.24
HP 700	98	0.4238	72.34	864	3.28

SUBJECT J. G. - RIGHT EAR [IMPAIRED]

FILTER	LEVEL dB SPL	CALC. A.I.	PERCENTAGE CORRECT	NO. OF TOKENS	STD. DEV.
HP 1400	68	0.1788	9.72	288	8.33
HP 1400	78	0.2337	30.78	864	3.86
HP 1400	88	0.2592	45.56	974	6.36
HP 1400	98	0.2928	51.85	864	4.88
HP 1400	108	0.3308	49.19	864	8.23
HP 2800	92	0.0753	13.37	576	2.86
HP 2800	98	0.0827	17.54	576	3.82
HP 2800	104	0.0921	19.80	576	5.79
HP 2800	108	0.1015	23.15	569	4.55
HP 2800	118	0.1516	18.06	576	3.76
BP 714	72	0.1389	23.61	288	3.59
BP 714	78	0.1389	29.17	288	3.40
BP 714	84	0.1390	35.42	288	6.05
BP 714	88	0.1402	40.20	686	3.44
BP 714	98	0.1511	39.24	288	5.93
BP 1428	78	0.1882	20.49	288	9.97
BP 1428	84	0.1900	35.42	288	6.47
BP 1428	88	0.1900	46.53	288	6.26
BP 1428	98	0.2101	53.82	604	6.84
BP 1428	108	0.2439	55.21	288	7.21
BP 1428	114	0.2463	53.47	288	9.98
BP 728	68	0.2979	28.01	864	8.82
BP 728	74	0.3198	40.39	864	9.27
BP 728	78	0.3271	53.47	576	10.38
BP 728	88	0.3302	65.82	1217	4.05
BP 728	98	0.3411	66.21	864	2.80
BP 728	104	0.3284	51.74	288	5.24

SUBJECT T. T. - RIGHT EAR [IMPAIRED]

FILTER	LEVEL dB SPL	CALC. A.I.	PERCENTAGE CORRECT	NO. OF TOKENS	STD. DEV.
UNFILT.	90	0.1148	25.46	864	6.00
UNFILT.	100	0.1866	47.05	576	5.51
UNFILT.	104	0.2120	60.59	576	8.17
UNFILT.	110	0.2434	57.78	2049	3.91
UNFILT.	114	0.2517	66.14	1900	4.10
UNFILT.	120	0.2331	67.01	576	1.33
LP 700	92	0.0474	17.19	576	1.73
LP 700	98	0.0603	20.93	481	4.14
LP 700	104	0.0678	23.26	576	1.66
LP 700	108	0.0690	26.64	640	3.33
LP 700	114	0.0641	29.18	975	2.27
LP 700	118	0.0577	28.12	576	4.83
LP 1400	84	0.0621	10.07	576	2.57
LP 1400	94	0.1216	21.70	576	3.02
LP 1400	104	0.1321	31.60	576	6.28
LP 1400	108	0.1329	38.12	794	4.16
LP 1400	114	0.1256	38.59	1095	2.77
LP 1400	118	0.1084	34.58	814	5.42
LP 1400	124	0.0665	39.41	576	4.26
LP 2800	98	0.1690	33.16	576	8.74
LP 2800	104	0.1980	46.30	864	6.18
LP 2800	108	0.2098	50.53	1305	6.49
LP 2800	114	0.2122	63.37	576	5.85
LP 2800	118	0.1963	56.90	1035	4.63
LP 2800	124	0.1531	60.88	864	7.33
HP 700	98	0.1057	31.13	864	8.00
HP 700	104	0.1438	41.90	864	6.53
HP 700	108	0.1649	54.35	1034	4.39
HP 700	114	0.1878	58.53	712	3.79
HP 700	118	0.1867	65.02	1378	1.86
HP 700	128	0.1514	63.66	864	6.21

SUBJECT T. T. - RIGHT EAR [IMPAIRED]

FILTER	LEVEL dB SPL	CALC. A.I.	PERCENTAGE CORRECT	NO. OF TOKENS	STD. DEV.
HP 1400	98	0.0424	10.07	576	5.24
HP 1400	104	0.0798	18.75	939	9.82
HP 1400	108	0.1010	27.60	539	10.44
HP 1400	114	0.1262	32.82	515	10.46
HP 1400	118	0.1343	33.03	776	9.24
HP 1400	124	0.1401	35.25	576	9.60
HP 1400	128	0.1354	32.41	864	11.70
HP 2800	108	0.0241	11.45	288	5.60
HP 2800	114	0.0396	9.03	288	2.66
HP 2800	118	0.0463	8.68	576	3.03
HP 2800	128	0.0563	7.83	907	2.49
BP 714	98	0.0633	13.72	576	3.47
BP 714	104	0.0640	21.88	576	4.48
BP 714	108	0.0640	27.74	982	1.90
BP 714	114	0.0616	34.40	999	3.76
BP 714	118	0.0525	36.63	576	3.99
BP 714	124	0.0310	38.72	576	4.93
BP 1428	108	0.0769	14.24	576	8.79
BP 1428	114	0.0866	23.96	576	10.29
BP 1428	118	0.0880	30.50	1024	7.83
BP 1428	124	0.0865	31.39	999	7.31
BP 1428	128	0.0791	30.90	576	9.46
BP 728	98	0.1029	19.68	864	8.48
BP 728	104	0.1298	31.94	576	2.60
BP 728	108	0.1409	42.70	989	6.73
BP 728	118	0.1404	55.36	1538	3.67
BP 728	128	0.0951	57.29	864	7.02

SUBJECT J. G. - LEFT EAR SIMULATED LOSS

FILTER	LEVEL dB SPL	CALC. A.I.	PERCENTAGE CORRECT	NO. OF TOKENS	STD. DEV.
UNFILT.	52	0.2998	26.39	288	4.82
UNFILT.	62	0.4173	49.65	288	7.72
UNFILT.	72	0.4949	71.18	576	4.59
UNFILT.	82	0.5373	82.88	751	3.90
UNFILT.	92	0.5489	88.19	288	5.84
LP 700	56	0.1336	23.61	288	3.59
LP 700	62	0.1464	26.04	288	3.48
LP 700	66	0.1507	30.21	288	6.04
LP 700	72	0.1513	36.46	288	6.25
LP 700	76	0.1513	40.62	288	4.30
LP 700	86	0.1513	38.48	649	3.72
LP 700	96	0.1357	38.54	288	4.30
LP 1400	56	0.2779	33.33	576	4.61
LP 1400	62	0.2898	37.67	576	4.34
LP 1400	66	0.2901	45.20	1022	4.82
LP 1400	72	0.2901	54.86	576	3.36
LP 1400	76	0.2901	56.67	970	3.79
LP 1400	86	0.2901	57.81	576	4.66
LP 2800	56	0.3527	35.42	288	5.13
LP 2800	62	0.4118	40.28	288	4.68
LP 2800	66	0.4383	52.78	288	12.53
LP 2800	72	0.4655	66.32	288	4.15
LP 2800	76	0.4755	75.50	651	5.40
LP 2800	86	0.4801	84.37	288	5.83
LP 2800	96	0.4603	85.76	288	7.38
HP 700	66	0.3022	39.06	576	7.25
HP 700	72	0.3437	53.48	576	9.39
HP 700	76	0.3644	72.74	576	5.51
HP 700	82	0.3861	76.54	953	7.28
HP 700	86	0.3958	84.60	1071	2.71
HP 700	96	0.4200	85.94	576	2.49

SUBJECT J. G. - LEFT EAR SIMULATED LOSS

FILTER	LEVEL dB SPL	CALC. A.I.	PERCENTAGE CORRECT	NO. OF TOKENS	STD. DEV.
HP 1400	76	0.2255	31.25	576	5.75
HP 1400	82	0.2472	44.27	576	6.44
HP 1400	86	0.2562	57.68	668	6.23
HP 1400	92	0.2674	68.05	576	5.35
HP 1400	96	0.2835	70.58	735	3.39
HP 1400	102	0.3114	75.69	576	6.24
HP 1400	106	0.3255	55.21	288	8.89
HP 2800	86	0.0661	11.46	288	6.45
HP 2800	92	0.0753	23.26	288	2.63
HP 2800	96	0.0804	31.94	288	7.44
HP 2800	102	0.0890	38.19	288	11.37
HP 2800	106	0.0956	45.98	630	7.48
HP 2800	112	0.1179	46.18	288	10.29
BP 714	66	0.1389	30.56	288	4.09
BP 714	72	0.1389	37.15	288	3.48
BP 714	76	0.1389	44.10	288	7.30
BP 714	86	0.1396	52.54	706	2.63
BP 714	96	0.1496	48.61	288	3.40
BP 1428	76	0.1854	28.12	288	7.97
BP 1428	82	0.1900	49.31	288	9.18
BP 1428	86	0.1900	58.64	639	8.54
BP 1428	92	0.1921	67.36	288	4.88
BP 1428	96	0.2031	71.53	288	5.13
BP 1428	106	0.2397	67.01	288	11.86
BP 728	66	0.2871	34.95	864	5.30
BP 728	72	0.3143	48.09	576	9.89
BP 728	76	0.3243	63.84	1283	5.11
BP 728	86	0.3296	78.67	1457	4.07
BP 728	96	0.3396	79.34	576	6.09

SUBJECT J. T. - RIGHT EAR SIMULATED LOSS

FILTER	LEVEL dB SPL	CALC. A.I.	PERCENTAGE CORRECT	NO. OF TOKENS	STD. DEV.
UNFILT.	52	0.2484	32.64	288	8.97
UNFILT.	58	0.3115	51.74	288	4.15
UNFILT.	62	0.3498	58.57	594	6.30
UNFILT.	68	0.3997	74.31	288	4.61
UNFILT.	72	0.4294	80.21	864	5.98
UNFILT.	82	0.4875	89.64	850	1.69
UNFILT.	92	0.5107	92.01	288	3.08
LP 700	56	0.1367	24.31	288	3.31
LP 700	66	0.1522	26.74	288	4.45
LP 700	72	0.1525	30.21	288	3.28
LP 700	76	0.1525	36.14	638	6.48
LP 700	86	0.1525	40.97	288	5.61
LP 700	96	0.1369	37.50	288	6.80
LP 1400	56	0.2551	34.49	864	3.35
LP 1400	62	0.2755	43.93	576	2.85
LP 1400	66	0.2795	50.92	1001	3.98
LP 1400	72	0.2799	53.99	576	5.59
LP 1400	76	0.2799	58.33	1039	5.64
LP 1400	86	0.2799	54.34	576	6.01
LP 2800	56	0.2919	44.44	288	9.28
LP 2800	62	0.3497	58.33	288	7.61
LP 2800	66	0.3822	75.00	288	4.82
LP 2800	76	0.4337	81.29	811	3.69
LP 2800	86	0.4486	87.15	288	6.84
HP 700	56	0.1430	26.56	576	4.44
HP 700	62	0.1989	39.76	576	8.44
HP 700	66	0.2322	62.85	576	6.53
HP 700	72	0.2769	78.82	576	3.08
HP 700	76	0.3038	84.78	1097	4.33
HP 700	86	0.3532	87.19	1411	3.98
HP 700	96	0.3911	90.80	576	3.73

SUBJECT J. T. - RIGHT EAR SIMULATED LOSS

FILTER	LEVEL dB SPL	CALC. A.I.	PERCENTAGE CORRECT	NO. OF TOKENS	STD. DEV.
HP 1400	66	0.1049	31.71	864	6.25
HP 1400	72	0.1495	47.92	864	7.99
HP 1400	76	0.1764	58.73	997	7.72
HP 1400	82	0.2076	70.60	864	4.61
HP 1400	86	0.2234	75.00	1107	5.57
HP 1400	96	0.2449	80.73	576	6.01
HP 2800	80	0.0340	15.98	288	2.66
HP 2800	86	0.0547	24.91	501	4.92
HP 2800	92	0.0687	35.77	288	8.44
HP 2800	96	0.0752	47.92	288	12.85
HP 2800	106	0.0926	55.90	288	5.24
HP 2800	112	0.1078	51.04	288	12.75
BP 714	56	0.1062	18.40	288	3.47
BP 714	62	0.1245	28.12	288	7.12
BP 714	66	0.1272	43.75	288	4.32
BP 714	76	0.1274	47.84	343	5.36
BP 714	86	0.1299	52.08	288	4.88
BP 714	96	0.1471	47.92	288	4.32
BP 1428	66	0.1027	30.21	288	8.21
BP 1428	72	0.1369	37.16	288	7.56
BP 1428	76	0.1537	47.57	288	9.91
BP 1428	86	0.1687	69.44	288	10.08
BP 1428	96	0.1696	70.80	689	11.05
BP 1428	106	0.1963	63.89	288	15.13
BP 728	56	0.1430	24.83	576	2.68
BP 728	62	0.1988	43.06	576	6.83
BP 728	66	0.2299	52.60	1314	5.30
BP 728	72	0.2643	73.96	576	3.24
BP 728	76	0.2811	76.18	810	6.40
BP 728	86	0.2985	84.26	864	5.38
BP 728	96	0.3158	84.55	576	5.90

SUBJECT R. M. - RIGHT EAR SIMULATED LOSS

FILTER	LEVEL dB SPL	CALC. A.I.	PERCENTAGE CORRECT	NO. OF TOKENS	STD. DEV.
UNFILT.	62	0.2823	42.71	288	3.82
UNFILT.	72	0.3460	60.24	576	8.85
UNFILT.	82	0.4092	75.73	1382	5.73
UNFILT.	92	0.4473	86.46	288	3.66
UNFILT.	102	0.4056	81.94	288	5.89
LP 700	56	0.1239	17.01	576	3.08
LP 700	66	0.1433	27.43	576	2.01
LP 700	76	0.1449	33.86	576	4.58
LP 700	86	0.1449	34.77	693	2.96
LP 700	96	0.1296	31.39	672	3.11
LP 1400	56	0.2350	31.25	576	4.68
LP 1400	66	0.2656	42.78	986	5.61
LP 1400	76	0.2674	49.76	1220	5.68
LP 1400	86	0.2674	53.90	1285	4.49
LP 1400	92	0.2610	53.12	288	5.71
LP 1400	96	0.2479	48.15	864	6.60
LP 2800	56	0.2412	31.25	576	3.67
LP 2800	62	0.2823	39.94	576	4.03
LP 2800	66	0.3063	48.80	730	8.67
LP 2800	72	0.3443	59.72	576	4.50
LP 2800	76	0.3656	68.76	806	12.27
LP 2800	86	0.3992	78.71	799	3.50
LP 2800	96	0.3845	81.94	576	4.84
HP 700	66	0.1617	30.68	864	4.83
HP 700	72	0.2012	45.14	864	5.02
HP 700	76	0.2268	62.58	1139	6.62
HP 700	82	0.2644	73.01	1039	5.92
HP 700	86	0.2862	77.88	1105	4.65
HP 700	96	0.3260	82.29	576	5.71

SUBJECT R. M. - RIGHT EAR SIMULATED LOSS

FILTER	LEVEL dB SPL	CALC. A.I.	PERCENTAGE CORRECT	NO. OF TOKENS	STD. DEV.
HP 1400	76	0.1042	24.42	864	8.97
HP 1400	82	0.1418	38.72	576	8.61
HP 1400	86	0.1630	53.01	1115	5.84
HP 1400	96	0.1961	66.00	1503	6.71
HP 1400	106	0.2301	68.40	576	8.21
HP 2800	86	0.0311	6.43	576	2.68
HP 2800	92	0.0501	13.02	576	3.18
HP 2800	96	0.0587	20.49	658	5.50
HP 2800	102	0.0675	29.17	288	6.31
HP 2800	106	0.0727	32.35	373	5.64
HP 2800	112	0.0795	31.25	288	8.68
BP 714	66	0.1211	26.22	576	1.91
BP 714	72	0.1225	37.68	576	6.09
BP 714	76	0.1225	42.59	640	6.67
BP 714	86	0.1233	46.54	713	3.92
BP 714	96	0.1308	40.45	576	4.96
BP 1428	76	0.0983	25.00	288	9.82
BP 1428	82	0.1220	41.67	288	8.25
BP 1428	86	0.1318	48.09	419	8.60
BP 1428	92	0.1363	59.72	288	9.00
BP 1428	96	0.1374	62.15	288	8.81
BP 1428	106	0.1574	57.64	288	14.23
BP 728	66	0.1617	28.82	576	6.43
BP 728	72	0.1995	45.32	576	5.11
BP 728	76	0.2208	57.36	953	6.97
BP 728	86	0.2551	70.46	686	5.08
BP 728	96	0.2673	76.91	576	10.63

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