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HOFFNUNG, STEVEN

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City University of New York PH.D. 1981

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THE EFFECTS OF OUTPUT LIMITING ON LISTENER PERFORMANCE FOR HEARING-IMPAIRED SUBJECTS

by

STEVEN HOFFNUNG

A dissertation submitted to the Graduate Faculty in Speech and Hearing Sciences in partial fulfillment of the requirements for the degree of Doctor of Philosophy, The City University of New York.

1981

This manuscript has been read
and accepted for the Graduate
Faculty in Speech and Hearing
Sciences in satisfaction of the
dissertation requirement for the
degree of Doctor of Philosophy

9/14/81
Date

Gerald A. Studebaker
Chairman of Examining Committee

9/15/81
Date

Irving Hochberg
Executive Officer

Dr. Harry Levitt

Dr. Irving Hochberg

Dr. Arthur Boothroyd

Examining Committee

The City University of New York

ACKNOWLEDGEMENTS

I wish to express sincerest appreciation to the following for their assistance and support in the completion of this study.

To Professor Gerald A. Studebaker for his devotion to this project from its inception, despite relocating to Memphis, Tennessee. His standard of excellence together with his ability to break down the most complex problem into its simplest components will serve as an inspiration to me in my professional career.

To Professor Harry Levitt for assuming a major role as an advisor in this project. His incisive comments, constructive criticism and availability at a moment's notice were invaluable.

To Professor Irving Hochberg, who as a member of the dissertation committee and Executive Officer of the Speech and Hearing Sciences program, provided me with the counseling and financial support for completing my doctoral studies.

To Dr. Arthur Boothroyd for serving as my outside examiner. His thorough evaluation of a preliminary version of this dissertation was very helpful in the preparation of the final copy.

To Art and Marcia Podwall and the staff of the Syosset Speech and Hearing Center, who provided patience, confidence, moral support and time, without which this study could not have been completed.

To Dr. Richard E.C. White for his extremely helpful technical assistance.

To Mr. Cecil Redmond and the staff and students of the Communication Sciences Laboratory for their continuous assistance and encouragement.

To Dr. Jane Madell and the New York League for the Hard of Hearing for providing the subjects for this study.

To the subjects for their dedication to the study.

To Mike Byers for his invaluable assistance in providing services necessary for the production of this dissertation.

Finally, to my wife Leah and to my children, Avi, Hindy and Tzippy, whose unselfish love, devotion, continuous assistance and personal sacrifices defy description, but will never be forgotten.

This study was supported in part by NIH Grant NS 13514.

TABLE OF CONTENTS

Page

Page

LIST OF TABLES

Table		Page

Table Page

LIST OF ILLUSTRATIONS

Figure		Page

Figure | Page

Figure | Page

CHAPTER I

INTRODUCTION

A longstanding problem involving hearing aid research is the need for reconciliation of the suprathreshold auditory input signal represented by conversational speech with the auditory characteristics of sensorineural hearing impairment. While conversational level speech is characterized by a broad range of intensities and frequencies, sensorineural hearing impairment is characterized by frequency dependent reductions in the dynamic range of hearing with concomitant disturbances in the perceptual correlates of amplitude and frequency.

One attempt at reconciling the acoustic properties of conversational speech with the characteristics of sensorineural hearing impairment has been the utilization of varying degrees of frequency selective amplification in compensating for the frequency dependent reduction in auditory sensitivity. This approach has long been intuitively acknowledged by professionals involved with hearing aid fittings. More recently, the validity of this approach has been demonstrated empirically (Barfod, 1972; Pascoe, 1975; Skinner, 1980; Lippman, 1981). While very early attempts at frequency selective amplification focused on correcting the listener's reductions in threshold sensitivity, current applications emphasize the long term average speech spectrum in evaluating the amount of frequency selective gain required.

The dilemma presented by a broad range of input signals and a reduced dynamic range of hearing has given rise to various methods for processing the amplitude of speech so that it will conform to the residual auditory area (frequency by dynamic range) of the sensorineural impaired individual. These methods include peak clipping and a variety of compression strategies as a means of reducing the broad dynamic range of input signal levels. Both peak clipping and compression limiting are methods which are primarily intended to prevent the amplified output of the input signal from exceeding the tolerance limits of the individual. Other forms of compression which operate over the entire amplification input-output function, represent an approach to signal processing which attempts to overcome loudness distortion introduced by cochlear hearing loss.

It has been accepted clinical practice to recommend compression limiting for cochlear impaired individuals whose dynamic range is severely reduced (Berger, 1977). However, the optimum method of output limiting for individuals with broader dynamic ranges has not been carefully studied since the Harvard hearing aid study (Davis, et al., 1947). Davis and his co-workers concluded that compression limiting is a preferred alternative to output limiting employing peak clipping. Resnick (1977), in a critical examination of the Harvard study, reported that implications and conclusions for current hearing aid applications based on this study are limited by a number of factors. One significant limitation was attributed to differences in the type of hearing loss now being considered as a candidate for hearing aid amplification.

A review of hearing aid fitting procedures (Ross, 1978; Schmitz, 1980) indicates that measurements of comfort and discomfort levels used to determine user gain and output limitation requirements are a part of most clinical and research approaches to hearing aid selection. Recent research (Kamm, et al., 1978; Ventry & Johnson, 1978) indicates that the intensity range between comfortable listening levels and discomfort levels is much narrower than previously thought. This is true even for individuals with moderate sensorineural hearing impairment. To protect individuals from discomfort, therefore, requires hearing aid limiting levels at values not far above preferred listening levels. Reducing the limiting level to these lower values increases the potential for amplitude distortion and interactions with listening conditions and frequency response.

The present study investigated the effects of various peak clipped output limiting levels on the performance of hearing impaired listeners. Performance measures included the measurement of listener-adjusted preferred listening levels and the measurement of speech intelligibility at their preferred listening and other adjacent levels. Effects were examined as a function of presentation level, frequency response and background noise.

CHAPTER II

REVIEW OF RELATED LITERATURE

The following sections review procedural considerations relevant to hearing aid fitting and selection. Emphasis is on the applications of Most Comfortable Loudness (MCL) and Loudness Discomfort Level (LDL) measurements and the potential for interaction between the electroacoustic characteristics, gain and saturation sound pressure level. The implications of these potential interactions will be examined with respect to amplitude distortion and its effects on the quality and intelligibility of speech, providing the rationale for this investigation.

Applications of MCL Measurements to Hearing Aid Selection

The concept of most comfortable loudness level (MCL) measurement has been applied in the hearing aid selection and evaluation process for over forty years. Applications of comfort level measurements to hearing aids can be divided into three basic approaches. The first approach uses comfort level measurements as an *a priori* means of prescribing frequency-gain characteristics. This is true of equal loudness approaches to hearing aid selection.

The earliest application of the equal loudness approach to the prescription of hearing aid frequency-gain characteristics was proposed by Watson and Knudsen (1940). Using a formula whose central component was based on the characteristics of the individual's most comfortable equal loudness curve, these researchers were the first to consider the suprathreshold listening characteristics of the individual in the determination of appropriate hearing aid amplifier curve requirements. This method represented a significant departure from approaches employing uniform amplification or amplifier curves determined by either mirroring the auditory threshold configuration or by bisecting the auditory area of the individual. Davis, et al. (1947), in their report of the well-known Harvard hearing aid study, described this approach to hearing aid fitting as impractical and imprecise.

More recently, fundamental aspects of Watson and Knudsen's method have been reintroduced and systematically refined. Beginning with Shapiro (1975), current proponents of equal loudness-based approaches to prescriptive hearing aid fittings have: 1) carefully considered the acoustical characteristics of the speech input signal (Shapiro, 1975; Byrne & Tonnison, 1976; Berger, 1977; Pascoe, 1978), 2) derived objective methods by which comfort-based equal loudness estimates could be applied to the individual (Byrne & Tonnison, 1976; Berger, 1977), 3) considered the individual's comfortable listening range rather than specific levels (Pascoe, 1978), and

4) included procedures helpful in resolving acoustical discrepancies between various coupler and real-ear based measurements necessary in accurately implementing the prescriptive procedure (Byrne & Tonnison, 1976; Pascoe, 1978). An important and as yet unanswered question relevant to a consideration of prescriptive approaches to hearing aid fittings involves understanding the relationship between specifications based on discrete frequency measurements and the manner in which a hearing-impaired listener will adjust the composite hearing aid system. Evidence for a possible loudness-bandwidth effect on MCL in normal hearing listeners has been the source of considerable controversy in the research literature (Dirks & Kamm, 1976; Dirks & Kamm, 1977; Ventry, 1977a).

The second approach involving comfort level measurements and hearing aids employs MCL as a means of adjusting the level of a pre-selected frequency gain characteristic. This method, first introduced as part of Carhart's (1946a, b) empirical approach to hearing aid selection, has been widely used in both research (McConnell, et al., 1960; Villchur, 1973; Pascoe, 1975; Levitt, et al., 1978; Studebaker, et al., 1980; Lippman, et al., 1981) and clinical practice (Ross, 1978; Schmitz, 1980) in the evaluation of selected hearing aids or hearing aid characteristics. The aided listener's gain for a conversational level speech input is adjusted according to a comfort criterion and the relative merits of a set of hearing aid characteristics are evaluated in terms of the individual's performance on measures of speech intelligibility.

The third MCL method used in hearing aid selection is only indirectly related to the hearing aid fitting process. In this case it is used as a means of pre-selecting desirable hearing aid characteristics. This method, introduced by Markle and Zaner (1966), involves obtaining the individual's unaided MCL for speech under earphones, from which a level representative of conversational level speech is subtracted. This remainder provides a means of estimating appropriate hearing aid gain specifications required by the individual.

Despite its widespread application in hearing aid fitting, the MCL approach has long been criticized for its test-retest variability. This issue was debated in reports issued by Davis, et al. (1947) and Carhart (1946b). Davis, et al. claimed unsuitable variability for MCL based procedures, while Carhart reported adequate clinical reliability. Results of investigations which have reported intrasubject variability for MCL measurements in hearing-impaired subjects using speech or noise-band test materials reveal test-retest differences of from 4-9 dB in 90% of the subjects (Reid, et al., 1977; Walden, et al., 1977; Shapiro, 1978; Ventry & Johnson, 1978). Interstudy differences seemed to vary depending on the instructional set, degree of hearing loss, aided versus unaided listening, and degree of listener experience. The significance of these differences will depend on how much of a difference 4-9 dB makes in the listener's speech intelligibility performance.

In the Harvard hearing aid study (Davis, et al., 1947) various hearing aid frequency responses were evaluated on a group of hearing-impaired listeners with respect to speech intelligibility performance at PB-max, rather than at comfort level gain settings. PB-max was determined by systematically varying presentation level intensity re: the output limiting level of the hearing aid. These researchers reported that a re-analysis of their original results (ibid, p. 72) suggested that had performance at equal comfort levels been used as the criterion measure for evaluating the various experimental conditions, relative differences in performance might have changed.

In another departure from comfort level based gain settings, Shore, et al. (1960) evaluated the reliability of hearing aid selection procedures by clinically evaluating a variety of hearing aids at gain settings intended to correct for the individual's elevated threshold for speech. Studebaker (1980) has pointed out that the unorthodox manner for establishing gain in the Shore, et al. study could have led to significant overamplification affecting the interpretation of their test results.

Total acceptance of a comfort level approach in hearing aid selection does not yet exist as indicated by the contrasting approaches of two hearing aid studies recently reported in the literature. Skinner (1980), reporting on an investigation of high frequency emphasis in a group of sensorineural impaired listeners, compared the relative benefits of her varying frequency response conditions by measuring speech intelligibility performance at five pre-established intensity levels. Lippman, et al. (1981), however, in their study comparing multi-channel amplitude compression and linear amplification in sensorineural hearing loss, employed the subject's most comfortable level, established by means of a bracketing procedure, as the level at which speech intelligibility tests were presented and relative performance compared.

With the exception of very young children, it would appear that most hearing-impaired individuals will independently control the variable volume or gain control setting of their hearing aid (Pascoe, 1978). Thus, it is the listening level ultimately chosen by the listener which will determine the amount of audible signal and the potential benefits of amplification (Byrne & Christen, 1979). The adoption of a loudness based comfort level concept as a criterion measure for establishing hearing aid gain requirement is ostensibly due to the fact that comfort is a concept whose content or face validity (Ventry, 1977b) seems most compatible with an individual's adjustment to a hearing aid and the performance level at which he/she is most likely to function. Because of the procedurally dependent, variable nature of MCL measurement, its validity in determining the absolute gain requirements for an individual's hearing aid using any particular set of procedures is questionable (Byrne & Christen, 1979). Until aspects relevant to the reliability and validity of MCL are adequately identified, its application in hearing aid research and clinical practice will remain in question.

MCL and Speech Intelligibility Performance

One question which emerges from a review of various approaches to hearing aid selection concerns the unknown relationship between intelligibility performance and comfortable listening levels. That is, how closely do comfort based listening levels approximate the point of maximum intelligibility on a listener's performance intensity function?

In audiological testing considerable controversy has been generated over the issue of presentation levels in intelligibility testing, both for diagnostic and rehabilitative purposes (Carhart, 1965; Yantis, et al., 1966; Clemis & Carver, 1967). Researchers have evaluated both normal and hearing impaired populations at a variety of presentation levels, including MCL, in an effort to derive a time conserving method for estimating the PB-max or the maximum potential intelligibility score for the individual with a given test. Results of these studies indicate that while in normal hearing a single level for estimating PB-max on standard speech tests can be used, similar approaches are not possible in cases of cochlear hearing impairment (Carhart, 1965). Unlike performance intensity functions of normal hearing, characterized by steep slopes, wide dynamic ranges and broad plateaus corresponding to maximum intelligibility, the performance intensity functions in cochlear hearing impairment are less well defined. In addition to a reduction in maximum intelligibility, their functions are often less steep, rising to a narrow range of levels while occasionally dropping significantly as levels corresponding to maximum performance are exceeded (Carhart, 1965; Dirks, et al., 1977). Studies reporting on the relationship between PB-max and single level MCL's in cochlear hearing impaired individuals indicate mean MCL's significantly lower than mean PB-max levels as well as most individual MCL's underestimating their respective PB-max levels (Yantis, et al., 1966; Clemis & Carver, 1967; Posner & Ventry, 1977). Despite evidence of MCL procedures underestimating maximum intelligibility performance, hearing aid selection procedures continue to employ MCL's as a yardstick by which gain is specified and relative hearing aid performance evaluated.

Traditionally, clinical and laboratory procedures used in establishing an individual's comfortable listening level have limited the individual's selection to only one level (e.g., Kopra & Blosser, 1968; Ventry, et al., 1971). This practice has been followed despite research reported by Pollack (1952a), which indicated that comfortable loudness is more accurately described as a range of levels. More recently, Pollack's findings have been reaffirmed (Berger & Lowry, 1971; Woods, et al., 1973; Gabriellson, et al., 1974; Dirks & Kamm, 1976; Ventry & Johnson, 1978). Comfortable loudness ranges exceeding 20 dB have been reported for normal hearing listeners by varying either the instructional criteria (Pollack, 1952a; Gabriellson, et al., 1974), the approach mode of stimulus presentation (Woods, et al., 1973; Ventry & Johnson, 1978) or the strategy employed with an adaptive procedure (Dirks & Kamm, 1976). More restricted comfortable loudness ranges have been reported when listening in the presence of varying degrees of background noise (Pollack, 1952a; Beattie & Culibrk, 1980) and as a function of degree of cochlear hearing impairment (Dirks & Kamm, 1976; Ventry & Johnson, 1978; Kamm, et al., 1978). In both of these cases the lower limit of the MCL range was reported to have been elevated to a greater extent than the upper limit.

The relationship between the comfortable listening range and speech intelligibility performance is in need of clarification (Dirks & Morgan, 1979); however, limited available research data from Clemis and Carver (1967) and Posner and Ventry (1977) imply that improvements in intelligibility can be realized as listening levels exceed the lower limits of comfortable listening ranges. More recent findings, reported by Ventry and Johnson (1978) exploring the comfortable listening range, suggest that the upper limit of the MCL range is more likely to correspond to the individual's maximum intelligibility performance (PB-max).

From the standpoint of face validity, it would appear that intelligibility, rather than comfort, is of greater significance to the listener when adjusting his/her listening level. Comfort is more likely to assume a secondary role where intelligibility is maximized by the listener while remaining compatible with comfortable loudness. Results of experimental investigations which have compared speech MCL's for connected discourse in normal hearing listeners with instructional sets emphasizing loudness or intelligibility have been equivocal (Ventry, et al., 1971; Hochberg, 1975). Intelligibility based MCL instructions were found to result in higher MCL's only as a secondary effect, interacting with other experimental variables. A recent study conducted by Posner and Ventry (1977) with cochlear hearing impaired subjects, continued to demonstrate no significant difference between subject-adjusted MCL's when instructed for comfortable loudness or intelligibility. From comments provided by their subjects, these authors concluded that contextual clues from the connected discourse material used for obtaining MCL's may have affected the subjects' ability to make careful intelligibility judgments. They suggested that different results might have been obtained had more demanding intelligibility materials, such as monosyllabic words, been used.

An overall conclusion from the preceding discussion is the need to re-evaluate the MCL concept in terms of its relationship to speech intelligibility. Recent MCL research has implied that the concepts of comfort and intelligibility are more compatible than once thought. An instruction set which fulfills the requirements of comfort and intelligibility is necessary. This notion has led to the development of a new intelligibility-based, subjectively-determined, loudness criterion referred to as the "preferred listening level (PLL)" (Studebaker, et al., 1980). Studebaker has outlined the instructional criteria relevant to this concept, emphasizing the objective of adjusting the listening level so as to maximize the intelligibility of the signal, while not exceeding the loudness discomfort level. Recent hearing aid research with normal and hearing impaired listeners (Studebaker, et al., 1980) supports the reliability of this concept. Given its inherent face validity and apparent reliability, this concept, when applied to the determination of gain requirements, may prove more appropriate for applications of MCL measurements to hearing aid selection.

Applications of LDL[1] Measurement to Hearing Aid Selection

LDL measurement is used to define an upper limiting level for amplified output, beyond which amplified sound would be considered uncomfortable by the listener (Ross, 1978; Schmitz, 1980). Prior to the research reported by Hood and Poole (1966), who investigated the application of LDL measurement as an aid to the differential diagnosis of cochlear versus retrocochlear hearing impairment, little formal attention had been devoted by researchers to factors affecting LDL measurement since an earlier study reported by Silverman (1947). Since the report of Hood and Poole, a number of investigations involving LDL have been reported. Findings reported in these investigations have demonstrated that traditionally applied instructional criteria for measuring LDL based on the Silverman study may have resulted in unrealistically high levels. Using newly defined response criteria which emphasize loudness annoyance, rather than physical discomfort (Hood & Poole, 1966; McCandless & Miller, 1972; Morgan, et al., 1974; Berger, 1976; Beattie, et al., 1979), LDL's are now being reported at significantly lower levels than those first reported by Silverman. LDL's obtained with varying degrees of cochlear hearing impairment have indicated that their

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[1] The measurement of uncomfortable loudness has been variously referred to as "threhsold of discomfort" (TD), the "uncomfortable loudness level" (UCL), and the "loudness discomfort level" (LDL). For purposes of uniformity in this review, this measurement will be referred to as the "LDL".

LDL's are essentially equivalent to those measured in normal hearing individuals up to moderate losses of hearing, while increasing relatively little for even profound hearing losses (Kamm, et al., 1978; Shapiro, 1979; Edgerton, et al., 1980).

Much of the current research being reported on LDL is concerned with its application in hearing aid selection (McCandless & Miller, 1972; McCandless, 1973). Exceeding or underestimating an individual's LDL when fitting a hearing aid can result in annoying overamplification or in unnecessary reduction of the individual's undistorted dynamic range of hearing, either of which can affect the individual's acceptance of amplification (McCandless, 1972; Beattie, et al., 1980). Unaided inputs of pure tones, narrow band noise, or speech stimuli are presented to the listener at increasingly higher intensity levels for determining appropriate electroacoustic saturation level requirements (ANSI 1976), consistent with discomfort levels (Gengel, et al., 1971; Erber, 1973; Shapiro, 1975; Berger, 1977; Beattie, et al., 1980; Hawkins, 1980a).

Reviews of LDL research (Beattie, et al., 1979; Dirks & Morgan, 1979; Hawkins, 1980a; Beattie, et al., 1979, 1980) have identified a considerable amount of inter- and intrastudy variability with respect to LDL measurement. This variability has been attributed to procedural factors, such as instructions, psychophysical method, type of stimulus, pre-exposure and practice effects and calibration factors. The degree of LDL variability suggests that consideration be given to the development of an appropriate method of measuring LDL specifically for its hearing aid application. Beattie, et al. (1980) have proposed an LDL instructional set for this purpose.

In contrast with selection of frequency-gain characteristics for which only some researchers promote a prescriptive, frequency specific approach (e.g., Shapiro, 1975; Byrne & Tonnison, 1976; Berger, 1977; Pascoe, 1978), more widespread endorsement of the use of frequency-specific measurements for selecting hearing aid saturation requirements can be found in the literature (Erber, 1973; Shapiro, 1975; Berger, 1976; Pascoe, 1978; Hawkins, 1980b), with only a few exceptions where speech stimuli are used (e.g., Beattie, et al., 1980). This introduces some of the same problems which affect equal loudness approaches to hearing aid characteristic selection, namely, those concerning acoustical calibration factors and stimulus factors.

With respect to calibration problems, it becomes necessary to accurately relate unaided LDL measurements to real ear hearing aid output levels under conditions of individual hearing aid use. Methods for overcoming some of the acoustical calibration problems relevant to hearing aid applications of LDL have been reported by Pascoe (1978) and Hawkins (1980a). Pascoe described a free-field method by which correction factors can be applied to standard coupler calibration specifications of a hearing aid in order to obtain the target real ear level. In the method reported by Hawkins which bears resemblance to a procedure reported earlier by Erber (1973), unaided LDL's are obtained with a hearing aid type receiver referenced to the same calibrating coupler standard used for expressing hearing aid specifications (ANSI 1976).

With respect to stimulus factors, no information has yet been reported as to whether SSPL 90 curves (ANSI 1976), specified by frequency specific LDL measurements, are compatible with actual hearing aid input stimuli, such as speech or other environmental sounds characterized by different spectral, bandwidth, and temporal characteristics. While research studies indicate only small differences between unaided LDL's for frequency specific stimuli and speech stimuli (Dirks & Kamm, 1976; Hawkins, 1980b) in accordance with summation effects at high intensities (Zwicker, et al., 1957), this effect is in need of further study with hearing impaired individuals exposed to realistic input signals whose characteristics have been altered by the frequency-gain characteristics of a hearing aid.

The above limitations notwithstanding, conclusions based on LDL research have resulted in at least one researcher (McCandless, 1973) suggesting that hearing aid saturation sound pressure levels of from about 100 dB SPL to about 120 dB SPL are appropriate for all but severe-to-profound hearing losses. Based on a review of clinical recommendations reported in the literature and the results of their own investigation of the implication of LDL instructional sets for SSPL selection in hearing aids, Beattie, et al. (1980) discussed individual SSPL specification in terms of an upper and lower limit. They recommend that the upper limit of SSPL requirements be based on LDL measures as long as it does not present a danger of excessive exposure to loud sound. Their recommended lower limit is determined by an assessment of each individual's PB-max to avoid the potential detrimental effects of harmonic distortion secondary to peak clipping on speech intelligibility. These authors stressed that research is needed to ascertain the actual use SSPL settings of hearing impaired subjects in real-life environments before clinically proposed methods of LDL-based SSPL recommendations can be validated. Regardless of what signal is used in the unaided determination of SSPL requirements, Ross (1978) recommended obtaining a measurement of aided LDL for speech as a final determination of whether the output of a hearing aid is appropriate.

The LDL-MCL Relationship: Implications for the Selection of Basic Hearing Aid Parameters

Conventional amplification consists basically of three parameters: frequency shaping, gain and output limiting in the form of peak clipping. The combination of these parameters producing the highest intelligibility scores at comfortable listening levels for conversational level speech appears to remain an open question. While relevant issues were extensively studied in both the Harvard (Davis, et al., 1947) and MedResCo (Radley, et al., 1947) studies, critical reviews of these studies have emerged questioning their rigid conclusions and recommendations (Resnick, 1977; Braida, et al., 1977). In recent years new techniques for processing speech for the hearing impaired have been introduced. These include frequency-selective multi-band compression systems (Villchur, 1973), aimed at restoring some of the characteristics of normal hearing. With the development of new techniques, the need increases for specifying and selecting optimum conventional amplification systems with which proposed improvements can be compared (Studebaker, et al., 1977).

Recent laboratory investigations of the relative benefits of frequency selective amplification have studied the effects of variations in frequency-gain characteristics on speech intelligibility performance independently of output limiting (Thompson & Lassman, 1969; Ambrose & Neal, 1973; Pascoe, 1975; Skinner, 1980; Lippman, et al., 1981). Output limiting was either not a consideration or was considered independently of frequency-gain variations. While hearing aid specifications formerly based on a combination of more liberally defined LDL criteria and more conservatively defined MCL levels resulted in wider available dynamic ranges of amplification less likely to impose any effect on speech evaluated at an individual's MCL, the implications of recent LDL and MCL research significantly increase the likelihood of interaction between frequency-gain characteristics and limiting level. Redefined discomfort level criteria suggest lowering hearing aid saturation requirements. Investigations of MCL have suggested that the upper limits of its range may more closely correspond to levels achieving the desirable goal of maximum intelligibility performance. In one of the only studies investigating both MCL and LDL on a group of subjects with cochlear hearing impairment, Kamm, et al. (1978) reported median LDL's for a group of mild-to-moderately hearing impaired individuals as lying within 8 dB of the median upper limits of their comfortable loudness range for spondaic words.

Current LDL-based hearing aid output recommendations indicate that saturation sound pressure levels should be selected which correspond to the rms level of the LDL measuring stimulus (Ross, 1978; Schmitz, 1980). This practice is in accordance with research findings of LDL studies (Dirks & Kamm, 1976) and early peak clipping and hearing aid studies (Licklider, 1944; Davis, et al., 1947), which indicate that discomfort level appears closely linked to the power level of the measuring stimulus. Consideration of the amplitude characteristics of a speech signal and its relationship to LDL-based SSPL specifications in a hearing aid underscores the potential for interaction between gain and saturation requirements. The relationship between rms levels and peak levels of a speech stimulus, known as its peak factor, has been described as varying from 12-15 dB (Kryter, et al., 1947; Davis, et al., 1947; Wathen-Dunn & Lipke, 1958). Thus, selecting the saturation output of a hearing aid based on a level corresponding to the rms level of an LDL measurement would result in peaks of a speech signal being clipped, whose rms level is 12-15 dB below the actual saturation level. According to Davis, et al. (1947), high frequency emphasis of +9 dB/octave further increases the peak factor of speech. This will further reduce the rms input level at which speech peaks begin to be clipped. Conservatively defining SSPL in this manner will allow as much as 24 dB of peak clipping (Wathen-Dunn & Lipke, 1958) before the power level of the output speech signal approaches LDL rms values. Thus, the combination of reduced SSPL and increased gain requirements for hearing aids will increase the possibility of the peaks of a conversational level speech input exceeding the limits of the amplification system with concomitant distortion.

The possibility of such interactions was acknowledged as early as the Harvard study (Davis, et al., 1947, p. 76) and alluded to more recently by Byrne and Tonnison (1976) who stated:

> "It is conceivable that in some circumstances altering the aid to achieve the most desirable gain at all frequencies may have an adverse effect on other performance characteristics to the detriment of the overall result."
>
> (p. 57)

Effects of Peak Clipping on Listener Performance

Peak clipping is defined as a form of amplitude distortion in which the peaks of the input voltage wave on both sides of the time axis are clipped off, while the center portion of the wave remains intact. This process results in the generation of harmonic and intermodulation distortion products in the output signal which do not appear in the original input signal. Early investigations into the effects of peak clipping on speech intelligibility with normal hearing subjects were performed in order to evaluate both its detrimental effects and potential applications in military radio communication (Licklider, 1944, 1946). These studies revealed that, when presented at comfortable listening levels, as much as 20 dB of peak clipping had little detrimental effect on speech intelligibility. Even infinitely clipped speech (i.e., 40 dB clipping) remained intelligible, with its degree of intelligibility found to improve with listener practice. Adverse effects of peak clipping on the perceived quality of the speech signal, however, were detected at levels as low as 6-12 dB of clipping (Gross & Licklider, 1946).

Further research revealed a number of interactive effects between symmetrical peak clipping and other variables. When high frequency emphasis of the speech signal was introduced prior to peak clipping, an essentially infinite amount of clipping was tolerated without significantly affecting intelligibility (Licklider & Pollack, 1948; Pollack, 1952b). With this form of pre-processing, quality was restored by introducing low frequency emphasis following clipping (Licklider & Pollack, 1948).

When the primary speech signal was presented against a background noise different effects were identified depending on the type of noise and the point at which the noise entered the system. Detrimental effects on speech intelligibility were attributed to peak clipping when noise entered the system prior to clipping (Licklider, 1946; Kryter, et al., 1947). Under these circumstances it was reported that at low signal-to-noise ratios intermodulation between speech and intense low frequency noise was responsible for degraded intelligibility (Kryter, et al., 1947). It should be noted, however, that in general signal-to-noise ratios were loosely defined and the characteristics of the noises studied were often unique to the particular military communication situation under investigation.

In the case of noise entering the system following the clipper, peak clipping was found to improve speech intelligibility at low signal-to-noise ratios under certain conditions of limited peak amplitude handling capacity (Licklider, et al., 1948, Pollack & Pickett, 1959). Clipping put a maximum amount of physical power into the speech wave, offering a unique advantage in certain communication situations. One such situation cited by Licklider (1946) was that of hearing aid listening in which amplitude handling capacity is limited by the individual's discomfort level.

The above findings indicate that certain advantages can be realized when peak clipping speech. These findings were the impetus for a series of investigations conducted by Thomas and his colleagues. They sought to refine infinite peak clipping as a form of signal processing by systematically identifying the effects of several factors on the intelligibility of infinitely clipped speech. In the first of this series of studies, Thomas (1968) concluded that the intelligibility of peak clipped speech depended on the retention of second formant information, while benefitting from the elimination of first formant information. Thomas and Niederjohn (1970) investigated the ideal asymptotic high pass filter slope and cutoff frequency for optimizing the intelligibility of infinitely clipped speech. Results with normal hearing listeners indicated a net gain in intelligibility for high pass filtered, infinitely clipped speech, as compared with unclipped speech, over a wide range of signal-to-noise ratios. All background noise was introduced after the clipping stage. This result represented an improvement over earlier findings reported by Licklider, et al. (1948b) in that even when clipped and unclipped speech were presented at equal average power levels, the filtered clipped speech remained more intelligible.

Having demonstrated the positive effects of high pass filtered, infinitely clipped speech in normal hearing subjects, Thomas and Sparks (1971) evaluated this form of signal processing as a means of enhancing speech intelligibility for hearing impaired listeners. In a quiet background, filtered/clipped speech presented at the same average SPL as unmodified speech was found to result in higher intelligibility scores at all presentation levels in 13 of 17 cases. Thomas and Sparks concluded that filtered/clipped speech can be an effective means of increasing speech discrimination of subjects with varying types and degrees of hearing impairment.

One problem characteristic of aided situations not considered in earlier investigations of Thomas and his colleagues with normal and hearing impaired subjects was the condition in which noise enters the system prior to clipping. This condition had been previously identified by Licklider (1946) as detrimentally interactive with peak clipping, particularly for conditions of high frequency emphasis. Thomas and Ravindran (1974) performed a normative study of the effects of band limited white noise added prior to clipping on speech intelligibility when filtered and infinitely peak clipped. They concluded that despite the fact that the cutoff frequency and asymptotic high pass slope characteristics were not idealized for the test conditions, filtering and peak clipping the speech was an advantageous form of pre-processing at signal-to-noise ratios as low as 0 dB S/N. While the margin of superiority was small when compared with results in a quiet background, filtering and clipping never resulted in reduced performance.

While meticulous methods of calibrating test stimuli were described in all of the studies reported by Thomas and his colleagues, a number of limitations appear evident. A statistical treatment of the data was not presented, standardized speech intelligibility tests were not used, the acoustic output characteristics of the entire system were not described, and conditions of hearing aid use were not adequately simulated. Despite these limitations it appears that these studies underscore the earlier findings of Licklider with respect to the robustness of clipped speech insofar as speech intelligibility is concerned.

More recently, Carhart and Young (1976) reported on a series of studies investigating peak clipping as a form of input signal processing in normal and hearing impaired subjects with respect to its effects on speech intelligibility. These authors argued against the filtering recommendations of the Thomas studies on theoretical grounds, based on the fact that it eliminated too much important first formant information. Instead, these authors filtered the speech in such a way so as to flatten or "whiten" the average speech spectrum. Other features of this experiment which differed from those of the Thomas studies included study of varying levels of clipping, the addition of speech babble noise prior to clipping for both normal and hearing impaired subjects and the use of a real hearing aid receiver in one portion of the experiment, in an effort to more closely simulate conditions of real hearing aid use. Results were reported which indicated that under quiet conditions, whitened only or whitened and peak clipped speech was at least as intelligible as unmodified speech at various presentation levels for normal hearing listeners in a quiet background under headphones. Intelligibility was adversely affected by whitening and 30 dB of clipping in the presence of multitalker babble which entered the system prior to clipping, with the processed speech transduced through an insert hearing aid earphone. This adverse effect was reported, with some qualifications, to be even more pronounced in their group of hearing impaired listeners.

One factor which appears not to have been carefully considered in the Carhart and Young study was the calibration of the peak clipping characteristics of the test system for the speech stimulus. Secondly, when studying the interactive effects of a competing message with whitening and peak clipping, they used hearing impaired subjects with presbycusic hearing loss and an insert earphone with an irregular frequency response. These factors introduced the potential for central processing problems and frequency distortions which may have adversely interacted with their signal processing technique. Thirdly, presentation levels were fixed and could not be individually adjusted by the listener. Finally, the filtering and clipping conditions chosen for this study were unique and can not be readily compared with results based on the idealized parameters carefully developed in the Thomas studies.

The general conclusions of research related to peak clipping effects on speech intelligibility indicate that under quiet listening conditions with normal hearing individuals, peak clipping will not adversely affect speech intelligibility and may, at times, enhance it. A number of open questions still remain with respect to conditions in which noise enters the system prior to clipping and its interactive effects with hearing impairment.

Another group of studies whose conclusions are indirectly related to a consideration of the effects of peak clipping on speech intelligibility involve investigations that have sought to correlate electroacoustic measurements of commercial hearing aid performance with measures of speech intelligibility in both normal and hearing impaired subjects. These studies differ in approach from those discussed previously in that controlled amounts of clipping were not imposed on the speech signal.

Rather, the distortion characteristics through which the speech signal was transduced were measured.

Methodology typically employed in these studies involved obtaining a broad range of electroacoustic measurements for a group of hearing aids and then recording speech intelligibility materials through the hearing aids for subsequent presentation to the listener under earphones. Aid conditions were ranked according to intelligibility performance measures and these rankings were compared with rankings based on electroacoustic measurement performance. Highest rankings were determined by the best speech intelligibility performance among the hearing aids and the best electroacoustic performance within each measurement category (e.g., harmonic distortion, intermodulation distortion). Results comparing hearing aid rankings based on harmonic distortion and intermodulation distortion measurements with intelligibility performance based rankings were inconsistent across studies. While some investigations reported a positive correlation between distortion and intelligibility rankings (Harris, et al., 1961; Jerger, et al., 1966; Jerger, 1967; Kasten & Lotterman, 1967; Jirsa & Hodgson, 1970; Olsen, 1971), others reported either no correlation (Olsen & Wilbur, 1967; Bode & Kasten, 1971) or, in some studies, even a negative relationship (Olsen & Carhart, 1967; Jerger & Thelin, 1968).

Analysis of the differences among these studies reveals that in those studies which demonstrated positive relationships, a combination of unrealistically poor hearing aid and speech conditions were often used. In addition, speech presentation levels were fixed by the experimenter with no consideration of the altered amplitude characteristics of the processed signal. Finally, interaid conditions often varied in a number of ways, giving rise to possible detrimental effects secondary to complex interactions among the various electroacoustic parameters of the hearing aid systems. Curran (1976), in his review of harmonic distortion measurement methods, concluded that although the quality of the signal might be affected by harmonic distortion, performance appears relatively unaffected until extreme levels are reached.

A more basic question than the effects of controlled amounts of amplitude distortion on speech intelligibility relates to the effects of varying peak clipped output limiting levels on a listener's performance with an otherwise linear amplification system. That is, given an ability to adjust one's own listening level, how will changes in the output limiting level of a system whose output is controlled by peak clipping affect the listener's adjustment and subsequent performance? Under conditions where preferred gain levels cause the speech signal to exceed the output limits of the system being used, the distortion products generated introduce the possibility of interactive effects with respect to gain adjustment and subsequent intelligibility performance.

The relationship between gain and maximum output within an otherwise linear amplification system was investigated in a pilot study reported by Martin (1976). A group of sensorineural impaired listeners were instructed to adjust their listening levels for recorded speech to a comfortable level without regard to signal clarity as the maximum output was varied by the examiner. Martin's data indicated that when sufficient output was present, subjects tended to reduce their gain as the output limiting level was reduced. However, with increased reduction of the output limiting,a critical value was reached where individuals attempted to adjust the gain control beyond the output limits of the system in an apparent effort to achieve the relatively high signal levels judged comfortable in the unlimited, high maximum output conditions. These results suggest that when factors relating to intelligibility are de-emphasized, a listener will attempt to increase his/her listening level up to and beyond the limits of the system in order to maintain a comfortable loudness perception.

Interpretations based on Martin's findings are necessarily limited by the paucity of data and the lack of any statistical analysis. A methodological limitation acknowledged by Martin was the use of only a flat frequency response system. Additional limitations of the Martin study include the absence of a background noise condition. Furthermore, the implications of output limiting on speech intelligibility were not considered. Speech intelligibility was overlooked both as an instructional criterion for level adjustments, as well as with respect to the effects of output limiting induced level adjustments on speech intelligibility.

Research Questions

The present study was designed to investigate questions relevant to a better understanding of the potential interactions between hearing aid gain and saturations requirements and their effects on speech intelligibility. Research questions included:

a) What is the relationship between degree of peak clipped output limiting and a hearing impaired listener's preferred listening levels?

b) What is the relationship between degree of peak clipped output limiting and speech intelligibility performance at the preferred listening level?

c) What is the relationship between degree of peak clipped output limiting and speech intelligibility at other listening levels?

d) How will a listener's performance under varying degrees of peak clipped output limiting be influenced by variations in the frequency response of the system?

e) How will a listener's performance under varying degrees of peak clipped output limiting be influenced by the presence of background noise?

CHAPTER III

METHODOLOGY

Description of the Subjects

Subjects conforming to specific selection criteria were recruited for participation in this experiment. Selection criteria included: a) bilateral sensorineural hearing loss (i.e., air-bone gaps not exceeding 10 dB at any test frequency) of a confirmed cochlear nature[2], b) a moderate-to-moderately severe (i.e., 41-70 dB HTL) degree of hearing impairment in the test ear based on the three-frequency (i.e., 500, 1000 and 2000 Hz) pure tone average, c) measurable speech discrimination ability in quiet for the test ear for standard monosyllabic speech discrimination tests (e.g., CID, W-22), d) American English as a native language, e) an education received in a program for normal hearing persons (i.e., did not attend special school or program for the hearing impaired, f) previous monaural hearing aid experience, and g) 20 to 60 years of age.

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[2] Confirmation of the cochlear nature of the hearing loss was determined on the basis of an otological diagnosis and/or an audiological impression. Audiological impressions were based on the results of a test battery sensitive to differentiating cochlear dysfunction from other types of auditory dysfunction (e.g., acoustic impedance test battery, tone decay testing, SISI test, speech intelligibility performance-intensity functions).

Audiometric criteria (nature and degree of impairment, speech discrimination ability) were chosen as representative of: a) good hearing aid candidacy, b) compatibility with the range of saturation levels chosen for study as based on percentile ranks of loudness discomfort levels at .5 and 2 KHz, reported for individuals with equivalent degrees of hearing impairment (Kamm, et al., 1978, p. 675), and c) having the potential for providing discrimination test scores sensitive to differences among the various aided conditions. Non-audiometric criteria were established as a means of reducing the likelihood of: a) discrimination problems related to language deficits (native language and education), b) adaptation to hearing aid processed speech (hearing aid experience), and c) central auditory processing problems (age).

Ten subjects (6 males, 4 females), fulfilling these criteria, participated in this study. Subjects ranged in age from 27 to 59 years of age with a mean age of 42.7. The ear in which each individual wore his/her hearing aid was used as the test ear (3 right, 7 left). Seven of ten subjects reported first noticing their hearing difficulties in childhood (secondary to illness (3), unknown etiology (2), family history-genetic (1), congenital anoxia (1)). Two subjects reported noticing progressive hearing difficulty beginning in adulthood (positive family histories), and one subject developed a hearing loss as a young adult secondary to illness (kidney disorder).

Audiometric and otologic information was provided by a local speech and hearing center, which served as a referral source for all subjects. Pure tone air and bone conduction thresholds were verified for all subjects under standard audiometric pure tone threshold testing conditions (double-walled test booth) using standard clinical procedures (Carhart & Jerger, 1959) and instrumentation (Grason Stadler audiometer, model 1701), during an orientation session conducted prior to the onset of the experiment. Results obtained were in excellent agreement with pure tone data provided by the clinical referral source.

The three-frequency pure tone average of the subjects' test ears ranged from 45-65 dB HTL (re ANSI 1969), while the average of the non-test ear ranged from 35-67 dB HTL. Audiometric configurations of individual subject's test ears varied from relatively flat to progressively downward sloping configurations. One subject had a trough shaped hearing loss with a greater degree of hearing loss in the mid-frequency range than at either the low or high frequencies. At octave intervals within the audiometric frequency range of 250 to 4000 Hz, hearing sensitivity of the test ear was either better than or within 10 dB of the bone-conduction threshold of the non-test ear for 8 of the 10 subjects. Two subjects had air-conduction thresholds for the test ear 15-25 dB poorer than bone-conduction sensitivity levels in the non-test ear at some test frequencies. With the exception of some individual test frequencies for two subjects, bone-conduction thresholds were always within 10 dB of air-conduction thresholds for the test ear. Tables 1 and 2 and Figures 1 and 2 summarize the pure tone characteristics of the test and non-test ears used in the study. Table 3 provides a summary of clinical speech test results for the test and non-test ears consisting of speech reception thresholds and speech discrimination scores as reported by the clinical referral source. These data depict overall symmetry with respect to the auditory characteristics of the test and non-test ears. More detailed information with respect to these measures for individual subjects is available in Appendix A.

Test Area

All testing was performed in a two-room double-walled test suite (Industrial Acoustics Corporation). Test equipment, with the exception of the subject's attenuator console (i.e., gain control) and earphone headset, was located in one room. The subject sat in the other room wearing the earphone headset, with the attenuator console placed on a table before him/her.

Ambient noise in the subject listening area was measured with a sound level meter (B&K, Type 2203) equipped with a field microphone (B&K, Type 4145) and octave band filter set (B&K, Type 1613). Results of these measurements, when corrected for the attenuation characteristics of the earphone arrangement and the internal noise levels

Table 1. Mean pure tone hearing threshold levels (dB HTL) of the test ear, for ten subjects. Standard deviations and lowest and highest thresholds are also included.

	Frequency, Hz				
	250	500	1000	2000	4000
Mean threshold, dB HTL	32.0	43.5	58.0	60.5	64.0
Standard deviation, dB	13.5	8.4	5.1	8.8	16.4
Lowest - highest thresholds, dB HTL	10-55	30-60	50-65	50-80	30-95

Table 2. Mean pure tone hearing threshold levels (dB HTL) of the non-test ear, for ten subjects. Standard deviations and lowest and highest thresholds are also included.

	Frequency, Hz				
	250	500	1000	2000	4000
Mean threshold, dB HTL	30.5	41.0	57.0	60.5	67.0
Standard deviation, dB	12.5	12.0	12.9	11.5	15.4
Lowest - highest thresholds, dB HTL	15-55	20-55	35-80	45-80	45-95

ANSI 1969

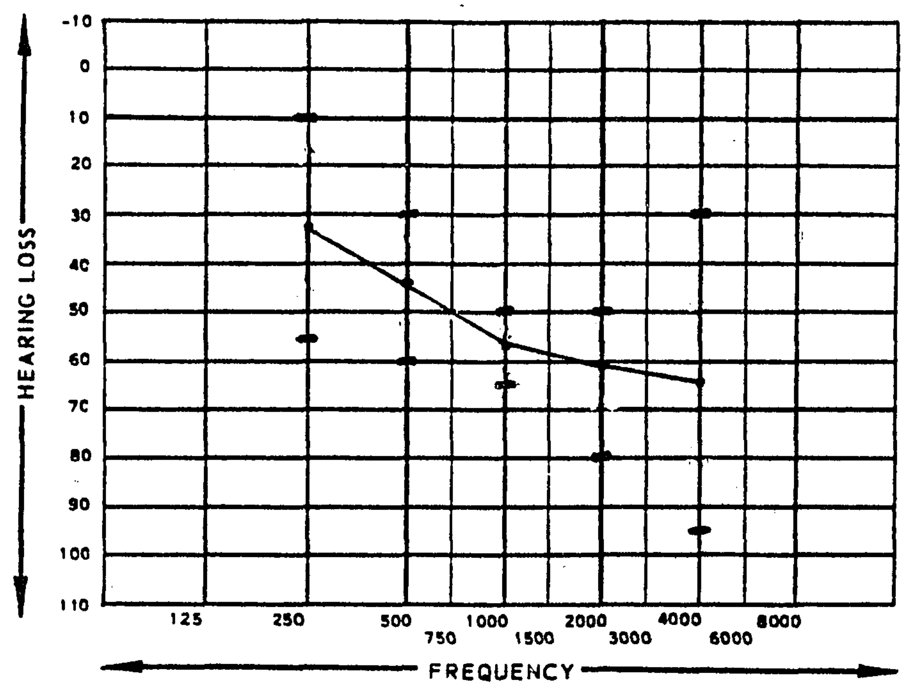

Figure 1. Mean pure tone thresholds of the test ear, for ten subjects. Bars indicate threshold ranges.

ANSI 1969

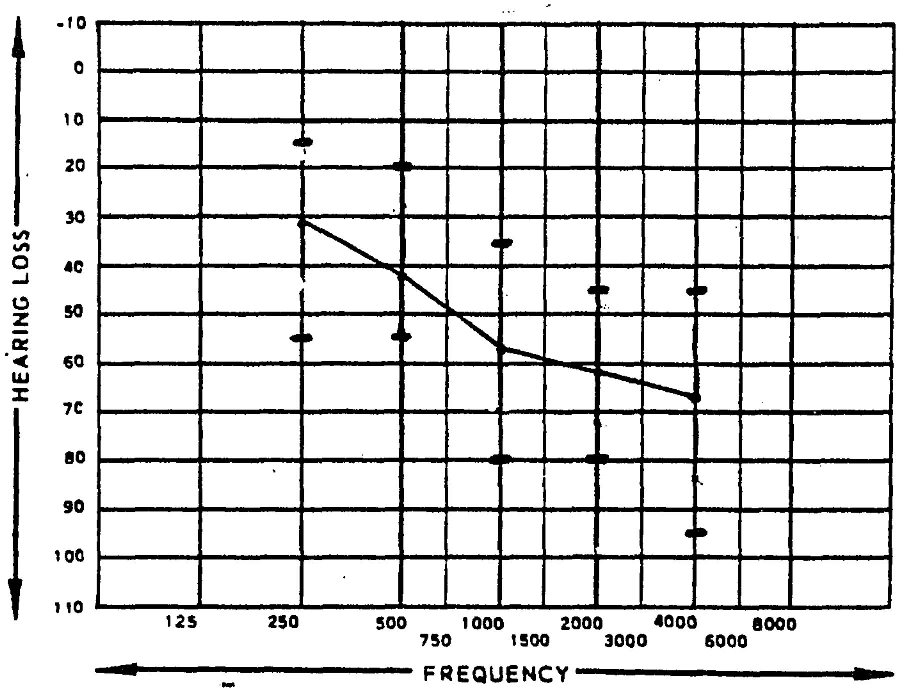

Figure 2. **Mean pure tone thresholds of the non-test ear, for ten subjects. Bars indicate** threshold ranges.

Table 3. Mean speech reception thresholds (SRT, dB HTL) and speech discrimination scores (SDS, %) of the test and non-test ears, for ten subjects. Standard deviations and lowest and highest values are also included.

	Test Ear	Non-test Ear
Mean SRT, dB HTL	45.5	45.0
Standard deviation, dB	5.2	7.7
Lowest - highest SRT's, dB HTL	40-55	35-55
Mean SDS, %	75.5	69.9
Standard deviation, %	19.7	23.2
Lowest - highest SDS, %	36-96	24-96

reported by the manufacturer for the sound level meter used, revealed octave band noise levels not exceeding current standards for permissible ambient noise during audiometric testing (ANSI 1977), over the frequency range of 125-8000 Hz. These levels were well below the audiometric thresholds of all subjects.

Test Equipment

Description

An experimental amplification system was designed which provided 44 dB of variable gain (1 dB steps) prior to output limiting in the form of symmetrical, "hard" (i.e., abrupt overload characteristics) peak clipping. This system used one channel of a two channel wearable master hearing aid (Bolt, Beranek & Newman, WMHA model 206) as its central component. The WMHA was of modular design with separate modules for controlling the method of output limiting (hard or soft peak clipping), saturation sound pressure level (SSPL), frequency response slope, and overall bandwidth. Ancillary equipment, including amplifiers and attenuators, were interfaced with the WMHA in order to achieve the gain and output requirements of the experimental design. Using the WMHA as an electronic component within the system, required that its microphone and earphone be replaced with specially constructed electrical analogs,[3] presenting to the WMHA the same impedance

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[3] designed by Dr. Richard E.C. White.

characteristics as the intended acoustic transducers. The microphone analog consisted of a high impedance input provided principally by the one megohm series resistor in the F.E.T. source follower circuit shown in Figure 3. The receiver analog consisted of a 1:1 transformer with a 600 ohm resistor in parallel with the input.

Peaks of the output signal were limited at one of three saturation sound pressure levels (120, 110 or 100 dB SPL). Electroacoustic components within the WMHA located prior to the limiter stage, included a choice of two frequency response slopes (0 or +6 dB/octave) between a fixed bandwidth of 500 to 6000 Hz. Below 500 Hz the frequency response fell off at approximately 21 dB/octave.

Test signals were introduced electrically into the amplification system from a tape recorder (Revox B-77). Output signals from the system were presented monaurally to the listener via a TDH-49 earphone and a supra-aural muff (MX 41/AR), mounted on a headband.

In addition to the playback system, a DC monitoring system[4] designed for remote monitoring of listening levels as DC voltages, was connected to the subject's attenuator console. This system included a voltage regulated power supply (Heathkit) and a DC voltmeter (Tektronix, model 502). A block diagram of the test system and DC monitoring system is shown in Figure 4.

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[4] designed by Dr. Richard E.C. White, based on a proposal by Professor G.A. Studebaker.

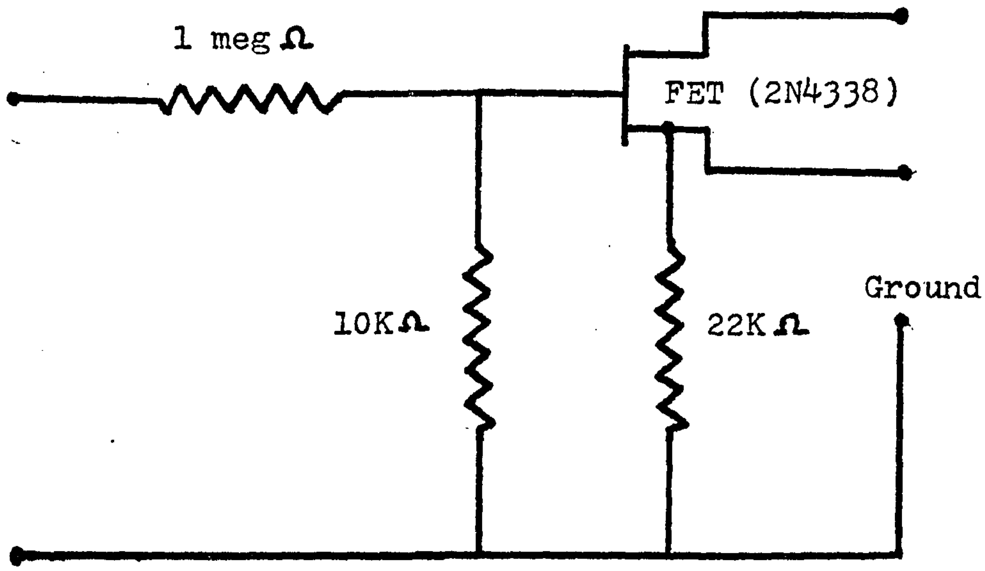

Figure 3. F.E.T. source follower circuit used as the microphone analog with the WMHA.

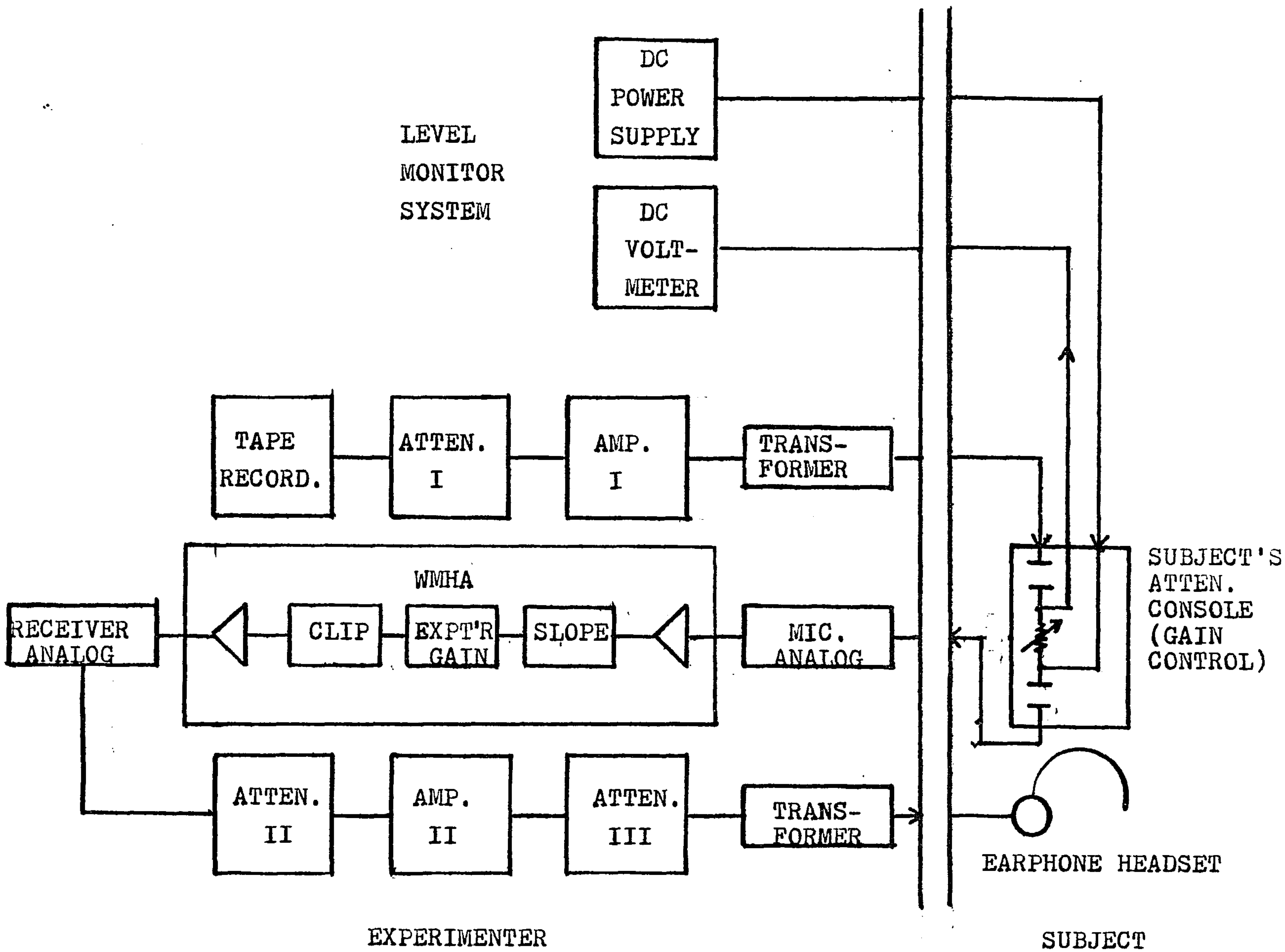

Figure 4. Block diagram of the experimental test system.

Rationale for Electroacoustic Variables

Saturation Levels

Saturation levels of 120, 110 and 100 dB SPL were chosen as representative of the range of loudness discomfort levels currently being reported for hearing impaired subjects representative of the experimental test population (McCandless & Miller, 1972; Kamm, et al., 1978; Shapiro, 1978; Edgerton, et al., 1980). These levels ranged from saturation values expected to impose virtually no amplitude distortion (120 dB SPL) on speech peaks at subject adjusted listening levels, to progressively lower saturation values (110 and 100 dB SPL), which would increase the potential for amplitude distortion and concomitant interaction with other experimental variables.

As a means of validating the choice of limiting levels chosen for study on the actual test group, loudness discomfort level (LDL) measurements were obtained in the present study. LDL's for each subject's test ear were measured at 500, 1000, 2000 and 4000 Hz. These levels were measured in each of two sessions (test and retest). An adjustment method, employing a Bekesy tracking procedure with instructions and signal parameters, equivalent to those used by Morgan, et al. (1974), was used. Signal parameters were achieved by delivering the continuous output of a Grason Stadler clinical audiometer (model 1701) operating in the fixed frequency Bekesy mode through a modular programming system (Grason Stadler, series 1200). The subject, seated in a double-walled, IAC sound-treated test booth, operated a recording attenuator with a fixed attenuation rate of 2.5 dB/second. The order of test frequencies was randomized for each subject and session.

The records of LDL responses obtained with the Bekesy tracking procedure were analyzed for each subject. Troughs in the sawtooth-shaped record were interpreted as corresponding to the point where LDL was reached, causing the subject to attenuate the signal level. The first four troughs for each frequency were eliminated and the mean of the next six troughs was recorded as the LDL for that frequency. This procedure was followed in order to analyze the most stable portion of each subject's response record.

Table 4 summarizes the LDL data obtained for the entire group of subjects for each test session. More extensive analyses were precluded due to either missing or unmeasurable data. These results indicate an essentially equivalent and broad range of LDL's at each test frequency for both test sessions. LDL's ranged from just over 100 dB SPL to levels exceeding 123 dB SPL (output limits), with a median level of approximately 113 dB SPL across all test frequencies for both test sessions. These findings were considered supportive of the choice of limiting levels used in this investigation.

A complete listing of LDL's obtained for each subject in each session is provided in Appendix B. Inspection

Table 4. Highest, median, and lowest pure tone loudness discomfort levels (dB SPL) of the experimental subjects' test ear for four frequencies, test (T) and retest (R).

	Frequency, Hz							
	500		1000		2000		4000	
	T	R	T	R	T	R	T	R
H	>122	>122	>123	>123	>122	>122	>122	>122
M	116.5	112	116.5	111	113	112	113.5	113
L	105	102	102	102	103	103	102	102

KEY:

T: test
R: retest
H: highest
M: median
L: lowest

of individual data indicated a high degree of variability for certain subjects between LDL's obtained for the same test frequency in different sessions. Some of this variability appeared due to difficulties with the Bekesy tracking procedure, with subjects appearing to lose their concentration as the task proceeded. One subject (#8) stated that he suffered a headache from the high levels he tolerated in the first session and was determined not to allow the signal levels to approach those tolerated in the first session again.

LDL data obtained in the present study were compared with LDL's reported by Kamm, et al. (1978) who measured LDL as a function of hearing loss for pure tones (.5 and 2 KHz), using an adaptive procedure. The hearing loss limits at each test frequency (see Table 1) for the subjects used in the present study overlap three of the hearing loss categories defined by Kamm, et al. They reported LDL's ranging in level from approximately 100 dB SPL corresponding to the 25th percentile value in their 31-50 dB HL category to LDL's in excess of 124 dB SPL for listeners in their 51-70 dB HL and 70 dB HL categories. These data appeared essentially consistent with the range of LDL's measured in the present study, lending further support to the choice of output limiting levels selected for use in the present investigation.

Frequency Response Characteristics

Recent studies have verified the efficacy of selective amplification by demonstrating improved intelligibility performance with varying degrees of high frequency emphasis for various populations of sensorineural hearing impairment (Pascoe, 1975; Levitt, et al., 1978; Skinner, 1980; Lippman, et al., 1981). As reviewed in Chapter 2, results of previous research have revealed improved intelligibility performance with high frequency emphasis (e.g., + 6 dB/octave) of speech prior to peak clipping (Licklider, 1946; Thomas & Niederjohn, 1970). A consideration favoring high frequency emphasis of +6 dB to +9 dB/octave in conventional hearing aid amplification systems, with output limiting in the form of clipping, is based on the characteristics of the long term spectrum of conversational speech (Dunn & White, 1940; Byrne, 1977). High frequency emphasis flattens the negatively sloping speech spectrum, allowing more speech energy to be linearly amplified prior to clipping. Because of these issues, frequency response effects were examined using slopes of 0 and +6 dB/octave over the effective bandwidth of the playback system.

Bandwidth

Frequency bandwidth characteristics of the playback system remained constant for all conditions with a low frequency cutoff of approximately .5 KHz and a high frequency cutoff of approximately 6 KHz. The frequency response fell off sharply below and above the respective bandwidth limits.

A low frequency cutoff of 500 Hz was selected based on research findings which have repeatedly indicated (Davis, et al., 1947; Levitt, et al., 1978) that for hearing impaired listeners with residual hearing for mid and high frequencies, amplification below 500 Hz does not significantly contribute to improved performance, while sometimes proving detrimental, due to effects of upward spread of masking (Martin & Pickett, 1970; Danaher & Pickett, 1975).

Results of recent hearing aid research suggest that extending effective amplification to frequencies beyond the 4 KHz upper limit, typical of most commercial wearable hearing aids, can improve speech intelligibility performance in hearing impaired individuals with high-frequency residual hearing (Pascoe, 1975; Skinner & Karstaedt, 1979). The high frequency cutoff of 6 KHz used in this study was based on this finding, with system imposed limitations precluding amplification beyond 6 KHz.

Calibration of the Experimental Test System

Standard Calibration Procedures

A series of calibration procedures were performed at regular intervals during the experiment, in addition to routine acoustical output calibrations performed on a daily basis and electrical output calibrations performed in every subject session. The frequency response of the entire system was calibrated acoustically in a 6 cc coupler (B&K, model 4184) at the earphone output. A constant voltage sweep-frequency input signal from a sine random generator (B&K, model 1034) was delivered through the amplifier of the playback tape recorder (Revox, model B077), whose reproduce-alignment characteristics had been previously verified as flat (± 1 dB) over a frequency range of 250-15,000 Hz, using an NAB reproduce alignment tape (Ampex). The earphone output in the 6 cc coupler was delivered through a sound level meter (B&K, model 2203) to a graphic level recorder (B&K, model 2305), mechanically synchronized with the sine random generator. This measurement was made within the linear amplitude operating range of the system for both slope conditions using the highest saturation level setting (i.e., 120 dB SPL).

Figure 5 shows the frequency response of the system for both slope conditions using identical input signal levels and test settings. These responses verified the slope, rolloff and bandwidth of the system. The 0 dB/octave slope condition appeared flat (± 2 dB) between 600 and 6000 Hz with a bandwidth (3 dB down points) of approximately 550-6000 Hz. Rolloff characteristics of approximately 21 and 42 dB/octave were measured below and above the passband of the system, respectively. Replacing the 0 dB/octave WMHA slope module with a +6 dB/octave module indicated a pivot point in the frequency response around

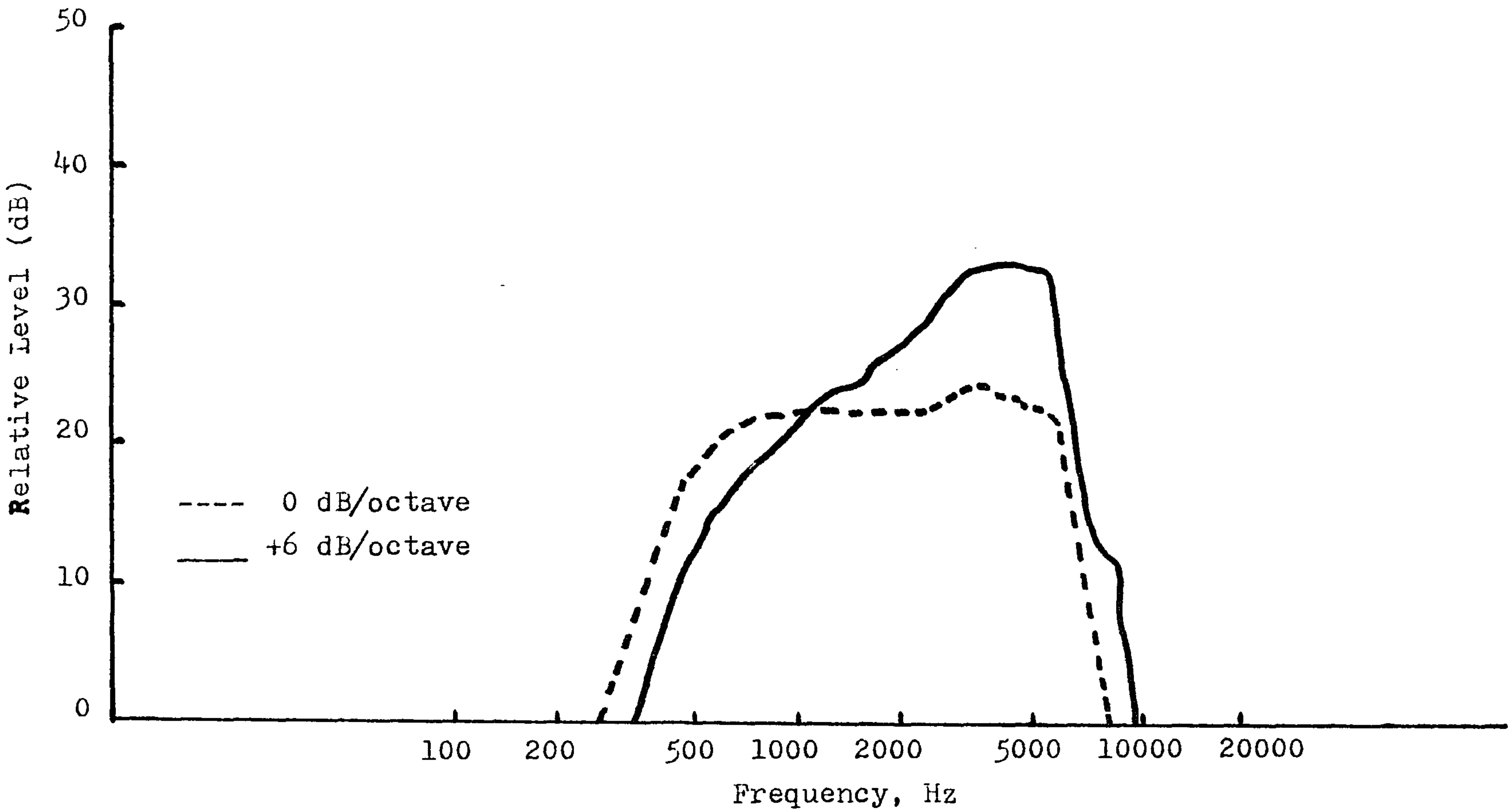

Figure 5. Frequency response curves for the 0 dB/octave and +6 dB/octave slope conditions measured acoustically in a 6-cc coupler at the earphone output.

1100 Hz and a slope of approximately +6 dB/octave between 500 and 6000 Hz. Bandwidth rolloff characteristics appeared essentially unchanged.

Frequency responses were measured in the same 6 cc calibrating coupler as the listener's pure tone threshold levels. An equivalent earphone (TDH-49) was used in each case. Thus, the two frequency responses provided a uniform insertion gain and a +6 dB/octave insertion gain when amplifying over the system's linear operating range.

Electroacoustic saturation characteristics of the test system were derived with the same test equipment. With the WMHA set to the highest saturation level value (120 dB SPL), the level of a 4000 Hz input signal was adjusted at the subject's attenuator to a point where a further decrease in attenuation did not result in any measurable increase in acoustic output. The frequency response of the system was then measured at this same adjusted level for the three WMHA saturation level modules used in this experiment. Figures 6 and 7 show the saturation curves generated for both slope conditions. The upper curve in Figure 7 depicts how, for the input signal level used, the saturation threshold was initially exceeded only in the vicinity of 4000 Hz, with high frequency emphasis for the highest saturation level value. Slight yielding (approximately 2 dB) of the limiter was noted when the saturation threshold was exceeded by 8 dB. With this calibration procedure the saturation threshold was

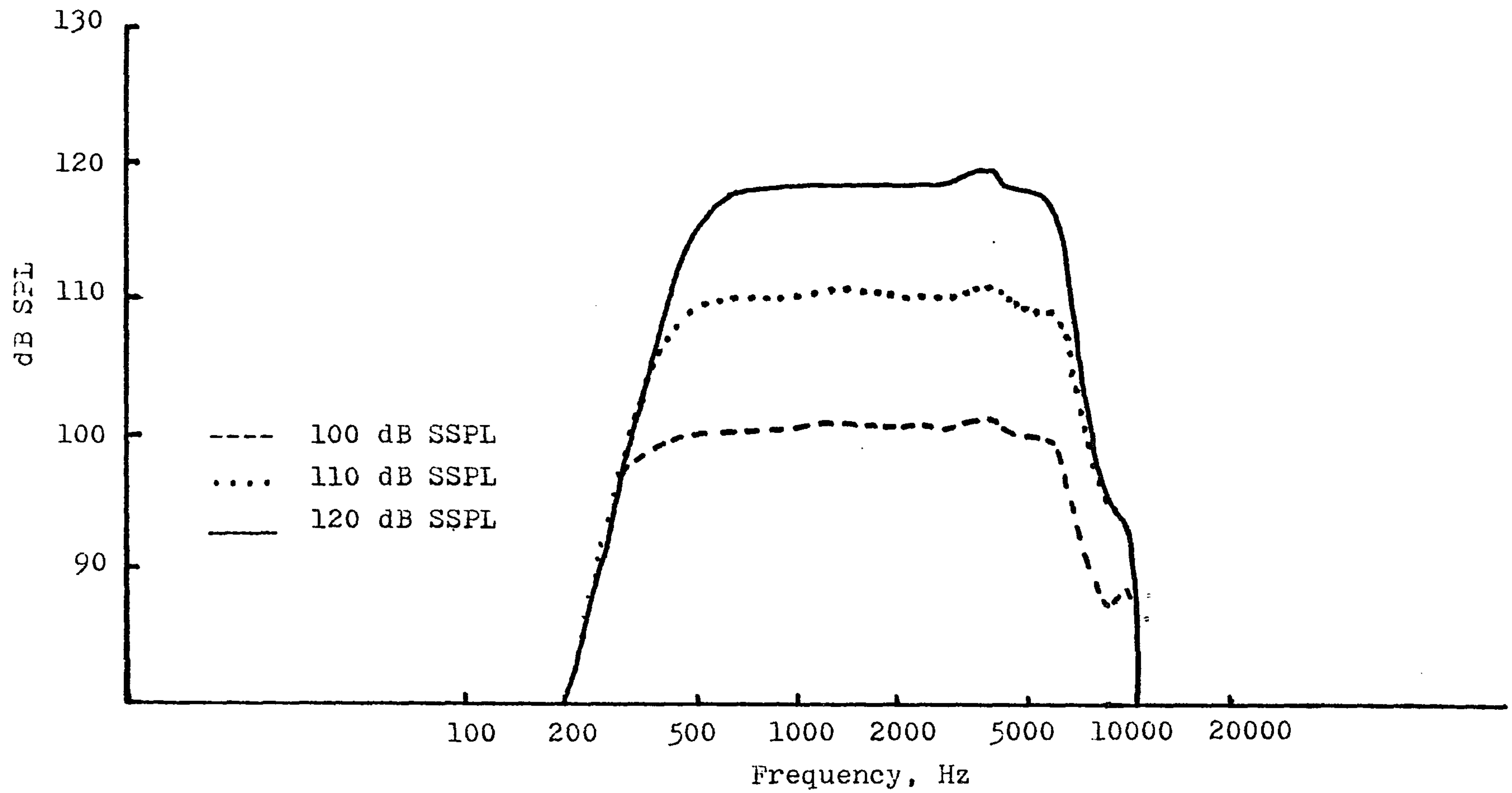

Figure 6. Saturation frequency response curves for the three saturation sound pressure level (SSPL) values used with the 0 dB/octave slope.

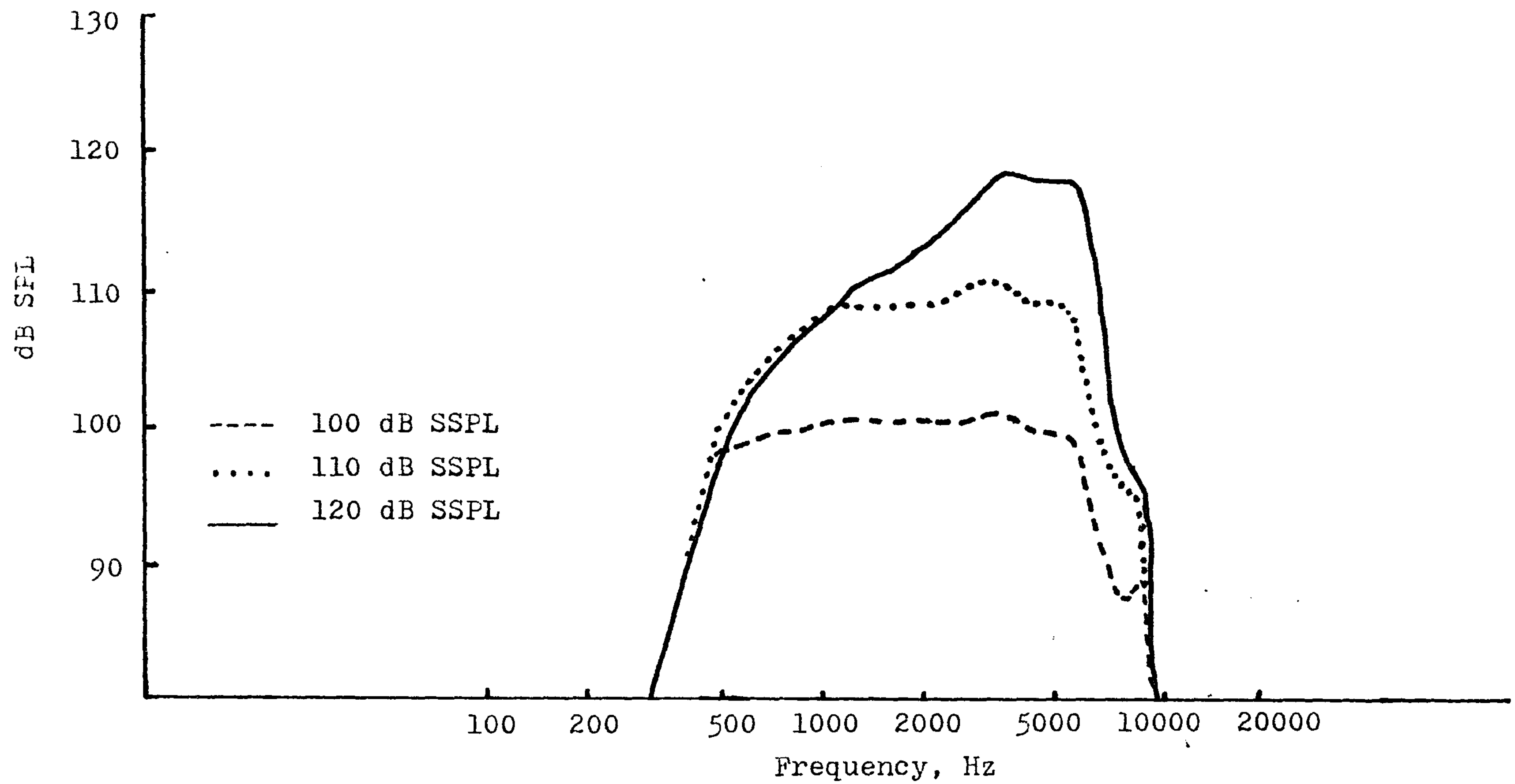

Figure 7. Saturation frequency response curves for the three saturation sound pressure level (SSPL) values used with the +6 dB/octave slope.

increasingly exceeded as saturation level modules were varied from 120 to 100 dB SPL. This resulted in yielding of the clipper only for the two lower saturation level modules (110 and 100 dB SPL). Therefore, Figures 6 and 7 depict less of an interval difference between the upper two saturation levels (120 and 110 dB SPL) than between the lower two saturation levels (110 and 100 dB SPL).

The acoustic output of the test system in a 6 cc coupler, at a level producing approximately 5% total harmonic distortion (THD) with the flat (0 dB/octave) response, was measured for each of the three saturation modules used. A 5% THD level for a pure tone input was established as the "saturation threshold".[5] Harmonic distortion was measured both acoustically at the earphone output and electrically at the input to the earphones using a distortion analyzer (Hewlett Packard, model 332A).

Acoustic distortion measurements were complicated by ambient, low frequency room noises. In order to reduce background noise interference, A-weighting was used only for acoustic distortion analysis. Total harmonic distortion levels for a number of input frequencies, using the flat (0 dB/octave) response, are shown in Table 5. Electrical and acoustical values did not vary from each other by more than 0.6%. These levels closely approximate

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[5] "Saturation threshold" is defined as the point at which clipping becomes just appreciable in the output. 5% THD was a convenient, practical way of defining the boundary for clipping.

Table 5. Total harmonic distortion (THD, %) levels at saturation, for a number of input frequencies, using the flat (0 dB/octave) response. Both electrical and acoustical values are included.

	Input Frequency, Hz				
	500	700	800	900	1600
% THD, electrical	1.1	4.2	4.4	4.5	4.1
% THD, acoustical	1.7	4.8	4.9	5.0	4.3

the 5% THD level designated as the saturation threshold. Output levels of 119.5, 109.5 and 99.5 dB SPL were measured at the saturation threshold for each of the three saturation modules respectively.

The symmetrical peak clipping characteristics of the limiter were verified both acoustically and electrically for the 0 dB/octave slope condition, using an oscilloscope (Tektronix, model 455). Photographs of the oscilloscopic display were taken. Unretouched tracings of these photographs for a 1 KHz pure tone input signal are shown in Figure 8. These photographs reveal essentially equivalent clipping characteristics for both methods of measurement. Slight variations in the shape of the clipping characteristics, ostensibly due to phase shifts, were introduced by the earphone response.

The equivalence of the distortion characteristics of the three WMHA saturation modules used in the experiment was established in the following two ways: a) Saturation thresholds for each module were exceeded by the same amount and equivalent amounts of total harmonic distortion were measured. b) The individual distortion products and their relationship to the fundamental frequency were measured for each module using an octave band analyzer (B&K, model 1613) connected to the sound level meter. This analysis further established the equivalence of the three saturation modules.

Linearity of the subject's attenuator (Attenuator II-44 dB range, 1 dB steps) was checked both acoustically and

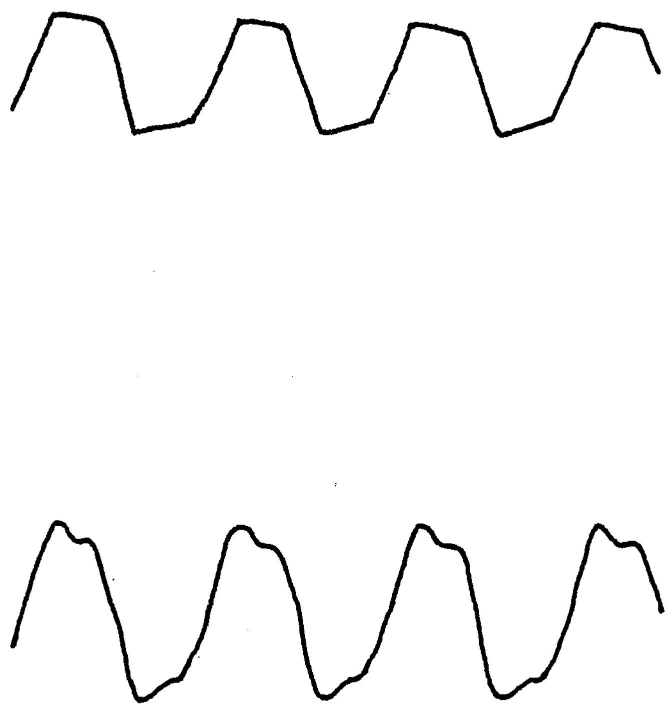

Figure 8. Peak clipping characteristics of the limiter for a 1 KHz pure tone input signal, measured electrically at the input to the earphones (upper tracing) and acoustically in a 6 cc coupler (lower tracing) at the earphone output (tracing of oscilloscope photograph).

electrically with the earphone loaded by a 6 cc coupler, using a voltmeter (Hewlett-Packard, model 400 FL). Individual steps never varied by more than .1 dB from their expected value and a total attenuation range of 43.7 dB was measured between the fully-on and fully-off attenuator positions.

The DC voltage monitoring system, connected to the subject's attenuator, was calibrated using a 35 volt, 20 milliamp input from the regulated low voltage power supply. The subject's attenuator was adjusted in 1 dB steps over its 44 dB range and the corresponding DC voltage levels, as measured on a DC voltmeter, were recorded. These levels were referred to throughout the experiment for determining listener controlled adjustments of the subject's attenuator. Level adjustments indirectly derived in this manner were accurate to within ± 1 dB of the listener's actual adjustment.

Inherent system noise in dB SPL for both slope conditions was measured acoustically at the earphone output in a 6 cc coupler for octave bands between 125 and 8000 Hz. The spectrum level within each octave band was measured and the critical band levels were determined for bands centered around the octave band center frequencies. Two sets of calculations were made, one using Fletcher's (1953) critical bandwidths for masking and the other using the broader critical bandwidths, reported by Zwicker, et al. (1957). For both sets of calculations the levels within a critical band were sufficiently below subjects' thresholds so as not to introduce any potentially interfering background noise levels or masking effects when subjects adjusted their individual listening levels. The overall level of system noise within a 125-8000 Hz band was approximately 46 dB SPL for the 0 dB/octave frequency response slope and approximately 52 dB SPL for the +6 dB/octave frequency response slope. This level was not influenced by the setting of the subject's attenuator.

Special Calibration Procedures

In addition to the standard calibration of the test system, an additional special purpose calibration procedure was developed for establishing the playback system's saturating characteristics with respect to the peak levels of the recorded speech stimuli. This procedure employed a peak equivalent method for equating the transient peaks of the speech test materials with the amplitude of a tone. The trigger level of an oscilloscope was used to produce an output every time a prescribed level was exceeded. The trigger unit had an exceedingly fast time constant (approximately 5 nanoseconds), which was considered suitable for monitoring instantaneous speech peaks. An electronic counter (Grason-Stadler, model 1219) controlled by a timer (Grason-Stadler, model 1216A) and associated logic equipment (Grason-Stadler, 1200 series) was used for counting the speech peaks. The timer was adjusted so that only one trigger per test word could be counted.

Modified versions of the test tapes used in the experiment were required for performing this calibration procedure. These tapes were equivalent recordings of two of the NU#6 test lists used in the study (quiet background) with carrier phrases deleted. These lists enabled determination of the peak characteristics of the test stimuli independently of their carrier phrases. The procedure used in preparing these recordings is provided in this chapter.

This calibration procedure was accomplished in two stages. The first stage involved deriving a calibration test signal for adjusting all components of the playback system located before and after the intended point of limiting. A 1 KHz pure tone calibration input signal, which exceeded the highest peaks of the recorded speech stimuli by 3.5 dB, was used for this purpose. The level of a 1 KHz pure tone input signal from a function generator (Tektronix, model F6501) was adjusted to exceed the level of the recorded calibration signal on the NU#6 test tape by approximately 12 dB. Then the sensitivity of the oscilloscope's trigger unit, placed in parallel with the tape recorder output, was adjusted to be just activated at this level, as measured on the counter. The speech signal was then played back via the tape recorder and the output level of its peaks was compared with the pure tone's peak via the trigger threshold. With the speech signal repeatedly played back via the tape recorder, the level of the tone and the trigger sensitivity were systematically adjusted. In this manner, a calibration signal was obtained which did not exceed the peaks of the recorded speech signal by more than 1 dB. The calibration signal was raised by an additional 3.5 dB, corresponding to the maximum level indicated on the tape recorder's VU meter (i.e., +3 dB VU), providing a convenient reference point and additional headroom between the calibration tone and the peaks of the speech stimulus.

The system's output for the calibration signal was adjusted for a 120 dB SPL saturation threshold (see footnote 5), as measured acoustically through the TDH-49 receiver in a 6 cc coupler. Appropriate adjustments were made, wherever necessary, for ensuring adequate power and amplitude handling capacity for all components outside the limiter, using the highest signal levels anticipated in the study.

The second stage of this calibration procedure involved precisely establishing the saturation and output characteristics for the actual test signal used, as processed by the test system for both slope conditions. Specifically, this involved calibration at the location of the subjects' attenuator such that all listening levels could be specified reliably relative to the saturation threshold for each combination of SSPL and frequency response.

The first step involved adjusting the threshold level of the trigger unit such that a 1 KHz tone at this level produced 5% THD at an output of 120 dB SPL. This identified the onset of clipping (saturation threshold) at 120 dB SSPL. The 1 KHz signal was then replaced with the modified version of an NU#6 test list and the speech level prior to clipping was adjusted at the subject's attenuator to a level where clipping occurred on 10% of the test words (i.e., the electronic counter showed clipping on five words). This adjustment corresponded to a setting of 5 dB of attenuation on the subject's attenuator for the flat frequency response condition. For the +6 dB/octave frequency response, the corresponding setting was found to be 13 dB on the subject's attenuator. This was a result of the high frequency emphasis boosting the energy peaks of the fricatives by as much as 8 dB relative to the flat condition. Attenuator adjusted levels were essentially equivalent for two modified versions of the NU#6 test lists and the criterion of 10% was within 3 dB of no clipping for both frequency response conditions.

The potential peak clipping range of the test system for each combination of SSPL and frequency response can be derived by considering the two extreme conditions. For the 120 dB SSPL, 0 dB/octave condition, 0 dB attenuation is equal to 5 dB of clipping. For the 100 dB SSPL, +6 dB/octave condition, 0 dB attenuation is equal to 33 dB of clipping. Therefore, depending on the attenuator setting and the condition under test, anywhere from 0 to 33 dB peak clipping of the test signal was possible.

Test Materials

Intelligibility Lists

Intelligibility test lists consisted of four randomizations of three word lists (Appendix C) from the Northwestern University Auditory Test #6 (NU#6), developed by Tillman and Carhart (1966). Each list is phonemically balanced and contains fifty monosyllabic words of a consonant-nucleus-consonant (CNC) construction.

Recordings of these NU#6 lists, as prepared by Sommerville (1967), were used in this investigation. These recordings were made using a male talker described as having a "general American dialect." All test words were preceded by the carrier phrase "say the word," and a 4.5 second silent interval separated successive presentations of test words within each list. A 1000 Hz calibration tone corresponding to the peak VU meter reading for the level monitored carrier phrase was recorded with each list. The target test word followed the carrier phrase as a natural utterance.

Research performed by Sommerville with these tapes demonstrated good inter- and intra-list comparability for a group of normal hearing listeners. No significant difference in performance between orders of the same list was measured and a significant difference in performance between lists occurred only at a 4 dB SL presentation level.

Finally, an articulation gain function slope (approximately 5%/dB), typical of such tests, was reported.

Modified Intelligibility Lists

In addition to the test lists described above, three modified lists were prepared. Each list consisted of NU#6 test words with carrier phrases removed by means of a computer implemented editing process. Two of these lists were used for setting the clipping level of the test system as described earlier. These lists were two randomizations of NU#6 word lists (Lists 2A and 3C). Preparation of these lists enabled verification of the equivalency of the system's characteristics for two of the three NU#6 word lists actually used in this experiment.

A third, concatenated word list, was used for preferred listening level adjustments. The concatenated level adjustments list contained 42 NU#6 words (Appendix D), with 14 words randomly selected from each of the three test lists, so as to control for any learning effects taking place during level adjustments which could influence intelligibility performance. In addition to removal of the carrier phrase, interstimulus intervals (i.e., 4.5 seconds) were also eliminated for the purpose of enabling the listener to more efficiently adjust his/her preferred listening level. The list was repeated twice, resulting in an uninterrupted string of monosyllables of approximately 1-3/4 minutes duration.

The rationale underlying the preparation of word lists with the carrier phrase eliminated was that retention of the non-meaningful carrier phrase could have interfered with adjustments of the test system's saturation characteristics, as well as listener adjusted preferred listening levels for the test words. This potential problem was underscored by the fact that the initial /s/ phoneme of the first word of the carrier phrase was found to contain high levels of high frequency energy which, particularly for the high frequency (+6 dB/octave) emphasis condition, almost always exceeded the saturation threshold of the test system before any of the actual test words.

Background Noise

The competing noise used for background noise conditions consisted of a recording of cafeteria noise prepared by Resnick for use in a previous study (Levitt, et al., 1978, pp. 5.36-5.37). Processing of the original tape included extensive equalization and editing. These processes resulted in a two-minute segment, from which all clearly intelligible materials and extreme transients were removed and for which no peaks deviated by more than 1.5 dB from the overall average. Longer segments of this tape were generated by means of a tape loop. This material was selected as a competing signal on the basis that it had more face validity in simulating conditions of actual hearing aid use than noise stimuli which are mathematically more easily defined (e.g., white noise, speech-spectrum noise).

In order to investigate background noise effects within the contexts of this study, two signal-to-noise ratios were used, nominally quiet (i.e., limited by system noise) and +7 dB S/N. An effective signal to system noise ratio of better than +35 dB was achieved under all listening conditions. The +7 dB S/N noise condition was selected on the basis of previous experience with hearing impaired listeners of similar characteristics, for which this condition demonstrated sensitivity in differentiating among a variety of hearing aids (Studebaker, et al., 1980).

Recording of Test Materials - Equipment and Procedure

Intelligibility Test Lists

Copies of each of the recordings of four randomizations of the three NU#6 lists were prepared for both quiet and background noise conditions. All recordings were made using high output, low noise recording tape (Scotch 206). The recording system consisted of two tape recorders (Revox B-77) for playing back the master recordings of the NU#6 test and the cafeteria noise, one amplifier-mixer (Switchcraft, model 308TR) for combining test materials into one channel for the background noise condition and one tape recorder (Ampex 440B) for recording the output signal from the mixer. Signal-to-noise adjustments for the background noise condition were made by adjusting the gain of the mixer's individual channels until the appropriate values were reached for the internal calibration signals on the master tapes, as monitored on an AC voltmeter (Hewlett Packard, model 400 FL), placed in parallel with the input to the recording tape recorder. The 1000 Hz calibration signal on the NU#6 test tape was used to adjust the recording level. This calibration tone, which was set to the peak VU level of the carrier phrase, was conservatively adjusted (-10 VU) to allow sufficient head room (i.e., 16 VU) for the high peak factor characteristic of speech stimuli. A block diagram of the recording system used in recording the intelligibility test lists is shown in Figure 9.

Calibration of the recording system was performed using the same frequency response calibration equipment described earlier for the playback system. The sweep frequency output from the sine random generator was delivered through the individual playback and mixing channels and recorded on the Ampex tape recorder. The recorded sweep frequency signal was subsequently played back into the graphic level recorder which was mechanically synchronized with the sine random generator. Figure 10 shows the frequency response of the recording system. Note that the frequency axis was adjusted for the playback delay. The frequency response was virtually flat ($\pm$ .5 dB) over the frequency range of approximately 100-10,000 Hz. In addition to this overall calibration of the test system, the reproduce characteristics of the playback tape recorders

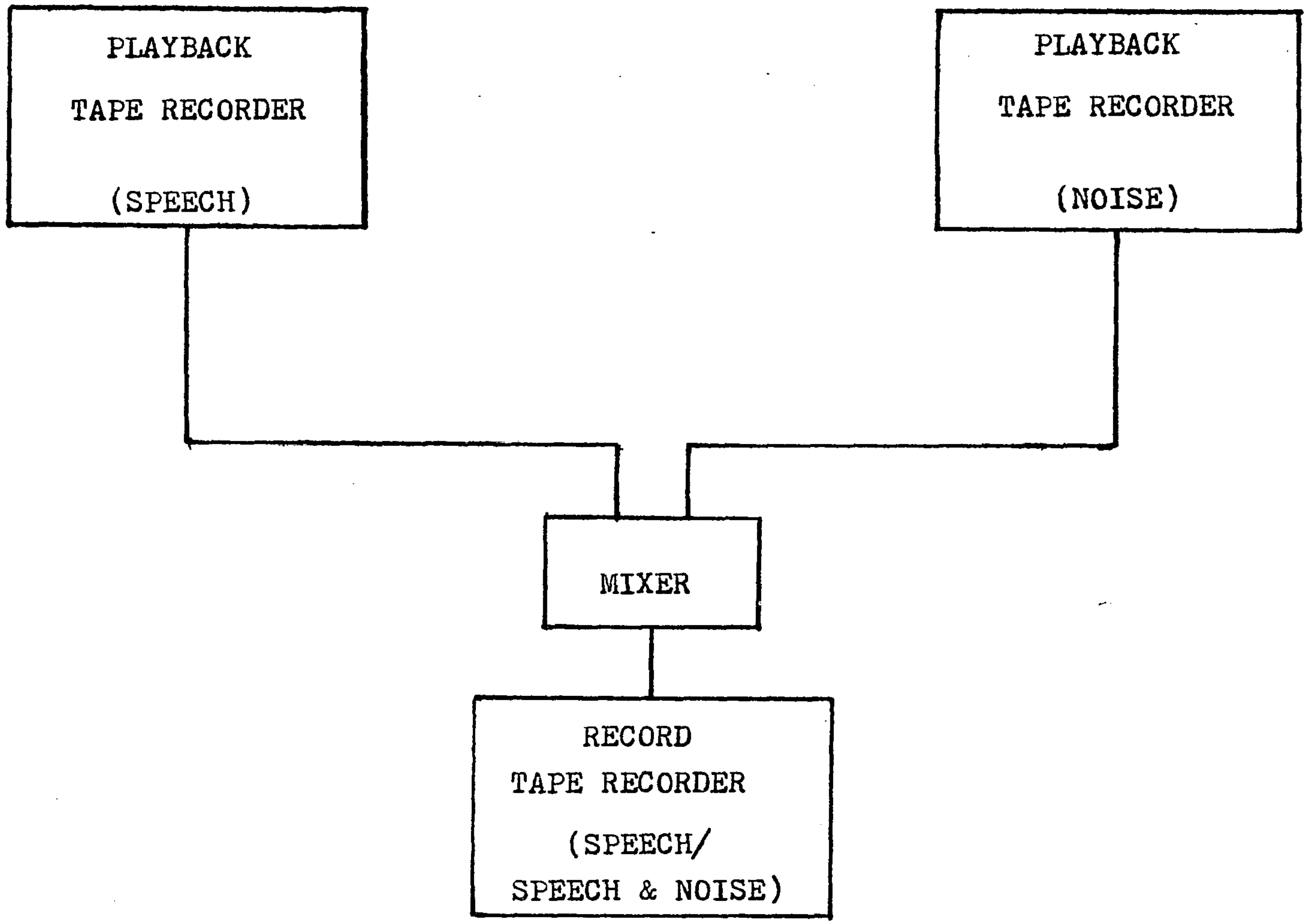

Figure 9. Block diagram of the recording system.

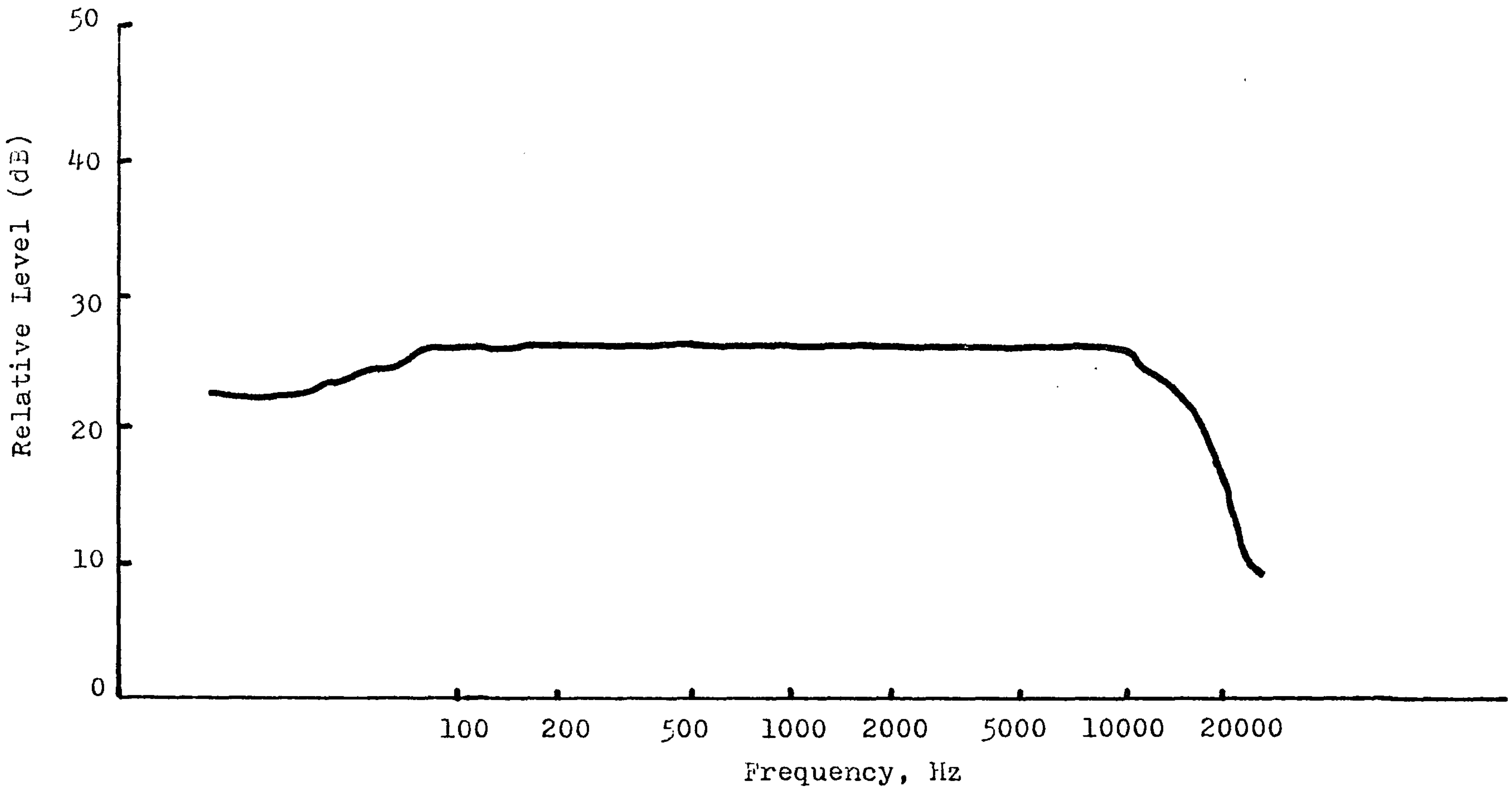

Figure 10. Frequency response of the system used for recording speech intelligibility lists.

were checked and verified as flat ($\pm$ 1 dB) by means of an NAB reproduce-alignment tape (Ampex). The speeds of all tape recorders were also verified.

Modified NU#6 Lists

As noted in a previous section, two modified versions of NU#6 test materials were prepared for calibration and preferred listening level adjustments. Tapes of these versions were dubbed from the standard NU#6 lists while removing their carrier phrases. This was done by means of a specialized editing process employing an interactive computer system.[6] The central component of this system consisted of a DEC PDP 8/e computer (core memory, 16k, 12 bit words) interfaced with a tape recorder (TEAC, model A7030), filter (Allison, model A12B), storage oscilloscope (Tektronix, RM503), keyboard (Digital DEC scope) and control box (custom-built). A block diagram of this system is shown in Figure 11.

Speech was low-pass filtered (approximately 9 KHz) and sampled at a frequency of 20 KHz using a 12 bit quantizer. The disk-stored digitized signal could be played back while being monitored simultaneously through a loudspeaker and visually displayed and stored on the oscilloscope. The control box contained both coarse and fine adjustment knobs for adjusting the start and stop times

[6] This system was developed by Dr. Richard E.C. White. A more complete description of this system can be found in Osberger (1978, pp. 45-51).

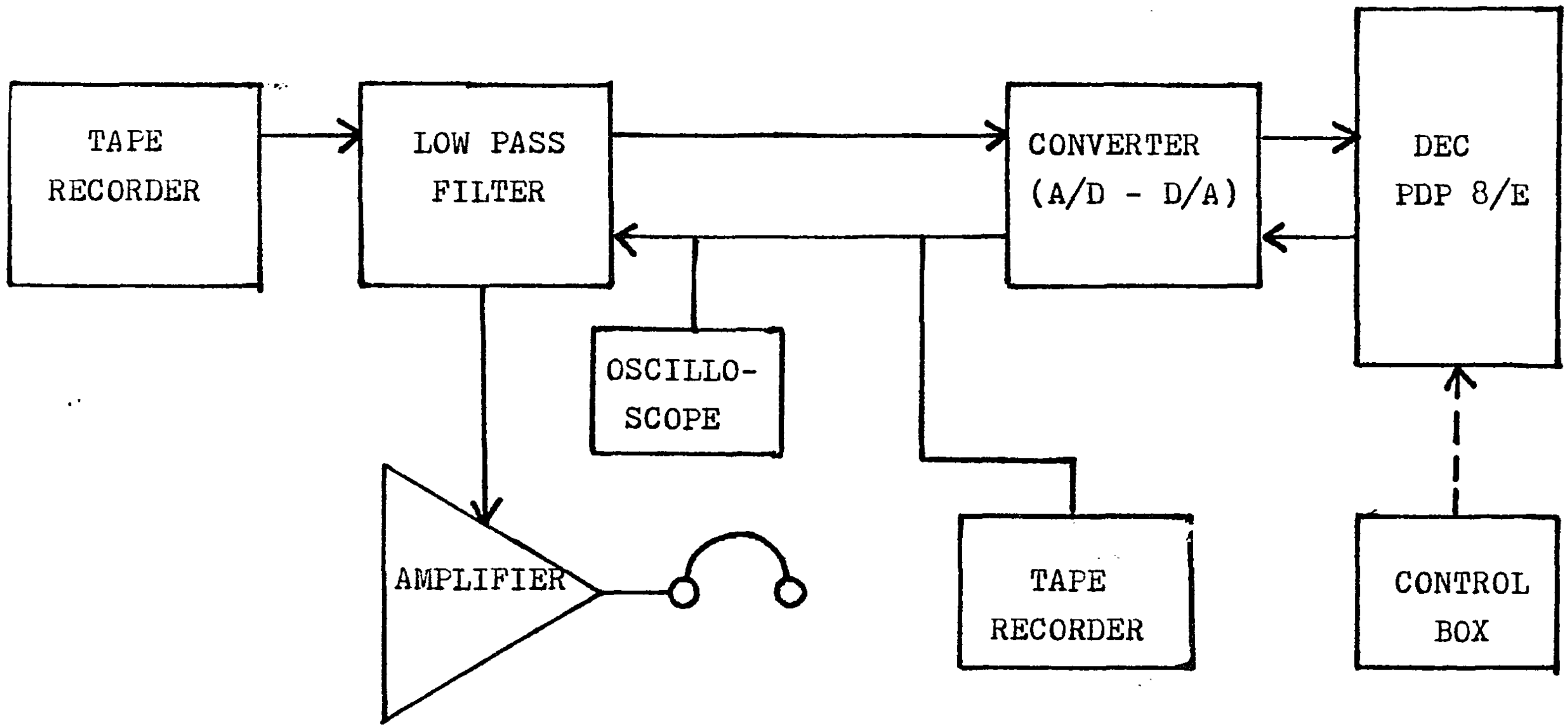

Figure 11. Block diagram of interactive computer system used in the preparation of the modified NU#6 word lists.

of the stored signal, permitting duration adjustments ranging in steps from 0.05 msecs to 3 seconds. The start and stop times of these intervals were displayed on a keyboard monitor.

All NU#6 materials were played into the computer from the TEAC tape recorder with steps taken to insure linear amplitude digitization by adjusting the playback input level to the computer, while monitoring the digitized amplitude level of the signal on the video monitor. Following the editing process, the digitized output signal from the computer was recorded in analog form on an Ampex tape recorder (model 440B). This process fully describes the manner in which the calibration test lists were prepared. The listening level adjustment tape required additional processing. This included removal of the interstimulus interval by manually splicing them out, lengthening the original list of 42 words by adding a duplicate dubbed version to it and finally, preparing a quiet and background noise version of the original tape using the same recording system shown in Figure 9.

Procedures

The experimental test protocol called for obtaining each individual's: a) preferred listening level for each of twelve conditions, b) intelligibility performance score (NU#6) at three presentation levels for each of the twelve test conditions, and c) loudness discomfort level (LDL) at four test frequencies (described earlier). All testing was performed using the ear in which the subject regularly wore his/her hearing aid. With the exception of LDL measurements, all test stimuli (i.e., speech materials) were presented through one TDH-49 earphone and MX41/AR muff, mounted in a conventional spring type headband, with the non-test ear unoccluded. The non-test ear was not occluded in order to eliminate increases in sound pressure delivered to the non-test ear caused by the occlusion effect, while more closely simulating conditions of monaural hearing aid use.

The test protocol was distributed over five sessions conducted at weekly intervals. Each session lasted approximately one and one-half hours. The first session was devoted to orientation and practice with all of the experimental test procedures. This included familiarizing each subject with the entire pool of 150 test words by having the subject read from a randomized version of the list while the examiner read each word aloud. In addition, instructions for all test procedures were read by each subject. These instructions are included in Appendix E. Instructions were also read by each subject and paraphrased aloud by the examiner in subsequent test sessions prior to the administration of each test procedure.

Preferred Listening Levels (PLL's)

Preferred listening levels (PLL's) were obtained twice for each of 12 conditions using a 3x2x2 balanced factorial design. The three factors were: SSPL (120, 110 and 100 dB), background (quiet, +7 dB S/N), and frequency response slope (0 dB/octave, +6 dB/octave). These 12 conditions were presented in a random sequence within each of two experimental test sessions. Subjects adjusted their PLL's for each condition using the control knob on the subject's attenuator while listening to the level adjustment tape. The subject's attenuator setting for each test condition was monitored using the DC voltage monitoring system.

In order to prevent attenuator positional cues from influencing PLL adjustments, three precautionary steps were taken:

a) The attenuator shaft was fitted with a circular knob having no pointer and detents on the control knob were not labeled.

b) Following an individual's PLL adjustment for a test oondition, the examiner reset the position of the control knob to a randomly selected position.

c) The subject's attenuator console actually consisted of two cascaded attenuators, each with a 44 dB attenuation range, providing a total attenuation range of 88 dB. Only one attenuator was controlled by the subject. The other attenuator was controlled by the examiner. The subject was unaware of the setting of the second attenuator.

Intelligibility Testing

Thirty-six intelligibility test conditions were administered to each subject. A 3x2x2x3 balanced factorial design was used. The four factors were: SSPL, frequency response slope, background noise and presentation level. Three presentation levels were used. These three presentation levels were the PLL's obtained in the first test session for each subject. For example, if for a given combination of frequency response and background, the PLL was X dB at an SSPL of 120, Y dB at an SSPL of 110 and Z dB at an SSPL of 100, then levels X, Y and Z were used as presentation levels for intelligibility testing for each combination of SSPL, frequency response and background. Inclusion of these other levels, in addition to the individual's PLL, as an intelligibility test presentation level, provided an opportunity for contrasting the individual's performance at his/her PLL with performance at subjectively less desirable levels, thus evaluating the appropriateness of the subject's choice of PLL. It also provided an opportunity for studying the effects of peak clipping on speech intelligibility, as a function of listening level.

The thirty-six intelligibility conditions were randomly assigned to the four sessions with nine intelligibility conditions run in each session. One NU#6 word list was used for each condition. NU#6 list orders and randomizations were varied across subjects and sessions and evenly distributed within each session.

CHAPTER IV

RESULTS

The main part of the experiment was comprised of two factorial designs. One design was used in establishing preferred listening levels (PLL) twice for twelve experimental conditions per subject (3 SSPL's x 2 backgrounds x 2 frequency response slopes) and the second design was used in the measurement of speech intelligibility for thirty-six experimental conditions per subject (3 SSPL's x 2 backgrounds x 2 frequency response slopes x 3 presentation levels). Data from the second design were also analyzed using various combinations of factors from within the main design.

The primary question with regard to listening level was:

a) What is the relationship between output limiting level and listener adjusted PLL's?

Secondary questions relevant to the study of PLL reliability were:

b) Are PLL's consistent across test sessions?

c) What is the magnitude of PLL test-retest differences?

The primary question with respect to speech intelligibility was:

a) What is the relationship between output limiting and speech intelligibility performance measures at PLL presentation levels?

Secondary research questions relevant to the variables included in the experimental design for intelligibility included:

b) How does a listener's performance at his/her PLL compare with performance at other levels?

c) What is the relationship between output limiting in the form of symmetrical peak clipping and speech intelligibility performance?

Two forms of data relevant to this question were available. The first was provided by analyzing intelligibility performance at a fixed presentation level as output limiting was varied from virtually no limiting (i.e., 120 dB SSPL) to progressively more limiting (i.e., 110 and 100 dB SSPL). Additional analyses relevant to this question were obtained by studying intelligibility performance at various presentation levels for fixed limiting levels. All of the above questions were studied as a function of variations in background noise and frequency response conditions.

Preferred Listening Levels[7]

Factors in the analysis of PLL results included the following: a) three saturation sound pressure levels (SSPL: 120, 110 and 100), b) two frequency response slopes (0, +6 dB/octave), c) two listening backgrounds (quiet, +7 dB S/N), d) time-order (test-retest), and e) subjects ($n=10$). Data were analyzed by means of a factorial analysis of variance.

A summary of the results of this analysis is provided in Table 6. Statistically significant ($p<.05$) main effects are revealed for SSPL, frequency response slope, time-order and subjects. There are significant two-way interactions ($p\leq.001$) between subjects and background and between subjects and frequency response slope. Two- and three-way interactions that approach significance ($.1>p>.05$) are found between SSPL and background, time-order and subjects and background, frequency response slope and time-order.

Effects of Output Limiting

Group Data

Output limiting effects on PLL's were evaluated in terms of the significance of SSPL and its interaction with

[7]Because of difficulties encountered in relating speech peak levels, as measured in this study, to their RMS equivalent and in specifying the absolute level of clipped and/or spectrally shaped speech, all PLL's in this study are expressed in dB attenuation (i.e., relative dB). Appendix F provides information relevant to converting some of these data into absolute listening levels (i.e., dB SPL).

Table 6. Summary table of the analysis of variance for the five factorial PLL design.

SOURCE OF VARIATION	SUMS OF SQUARES	DEGREES OF FREEDOM	MEAN SQUARES	F	SIGNIF.
M	175.30832	2	87.65416	11.171	0.001
B	17.60416	1	17.60416	2.243	0.148
MB	52.85832	2	26.42915	3.368	0.056
F	82.83749	1	82.83749	10.557	0.005
MF	28.67499	2	14.33749	1.827	0.188
BF	1.20417	1	1.20417	0.153	0.701
MBF	0.75833	2	0.37917	0.048	0.953
T	61.00417	1	61.00417	7.774	0.012
MT	1.75833	2	0.87917	0.112	0.894
BT	17.60416	1	17.60416	2.243	0.148
MBT	2.30833	2	1.15417	0.147	0.864
FT	4.53750	1	4.53750	0.578	0.537
MFT	5.42500	2	2.71250	0.346	0.717
BFT	33.00417	1	33.00417	4.206	0.053
MBFT	1.50833	2	0.75417	0.096	0.908
S	1297.42065	9	144.15784	18.372	0.001
MS	123.44167	18	6.85787	0.874	0.611
BS	660.85400	9	73.42822	9.358	0.001
MBS	167.55832	18	9.30879	1.186	0.360
FS	388.95410	9	43.21712	5.508	0.001
MFS	49.90833	18	2.77268	0.353	0.983
BFS	92.92082	9	10.32454	1.316	0.295
MBFS	192.49165	18	10.69398	1.363	0.259
TS	155.28749	9	17.25417	2.199	0.074
MTS	112.82500	18	6.26805	0.799	0.681
BTS	132.68750	9	14.74306	1.879	0.122
MBTS	141.27499	18	7.84861	1.000	0.500
FTS	86.08749	9	9.56528	1.219	0.343
MFTS	201.32500	18	11.18472	1.425	0.229
BFTS	63.62082	9	7.06898	0.901	0.545
MBFTS	141.24165	18	7.84676		
TOTAL	4494.28516	239			

KEY

M: SSPL: 120, 110, 100
B: Background: Quiet, +7 dB S/N
F: Frequency Response: 0 dB/octave, +6 dB/octave
T: Time Order (test-retest)
S: Subjects

other factors. The analysis of variance revealed a significant main effect for SSPL with a two-way interaction between SSPL and background bordering on statistical significance. Means for the main effect and interaction are summarized in Figures 12 and 13, respectively. Figure 12 reveals that subjects reduced their listening level for the lowest limiting level (100 dB SSPL) relative to the other two limiting levels. No change in listening level was indicated between 120 and 110 dB SSPL. In Figure 13 the interaction between SSPL and background reveals that subjects reduced their PLL's more under conditions of background noise than under quiet listening conditions, for the lowest limiting level.

Individual Data

An analysis of individual PLL data proved essentially consistent with the effects observed for group performance. Individual PLL data averaged over the two test sessions are shown in Figures 14 through 17 as a function of SSPL for all combinations of background noise and frequency response. Shaded areas are used to represent the amount each listening level exceeded the saturation threshold for a given SSPL. Raw PLL data for individual test sessions can be found in Appendix G.

These data revealed that, with the highest level of output limiting (i.e., 120 dB SSPL), PLL's did not exceed the saturation threshold for almost all subjects for both

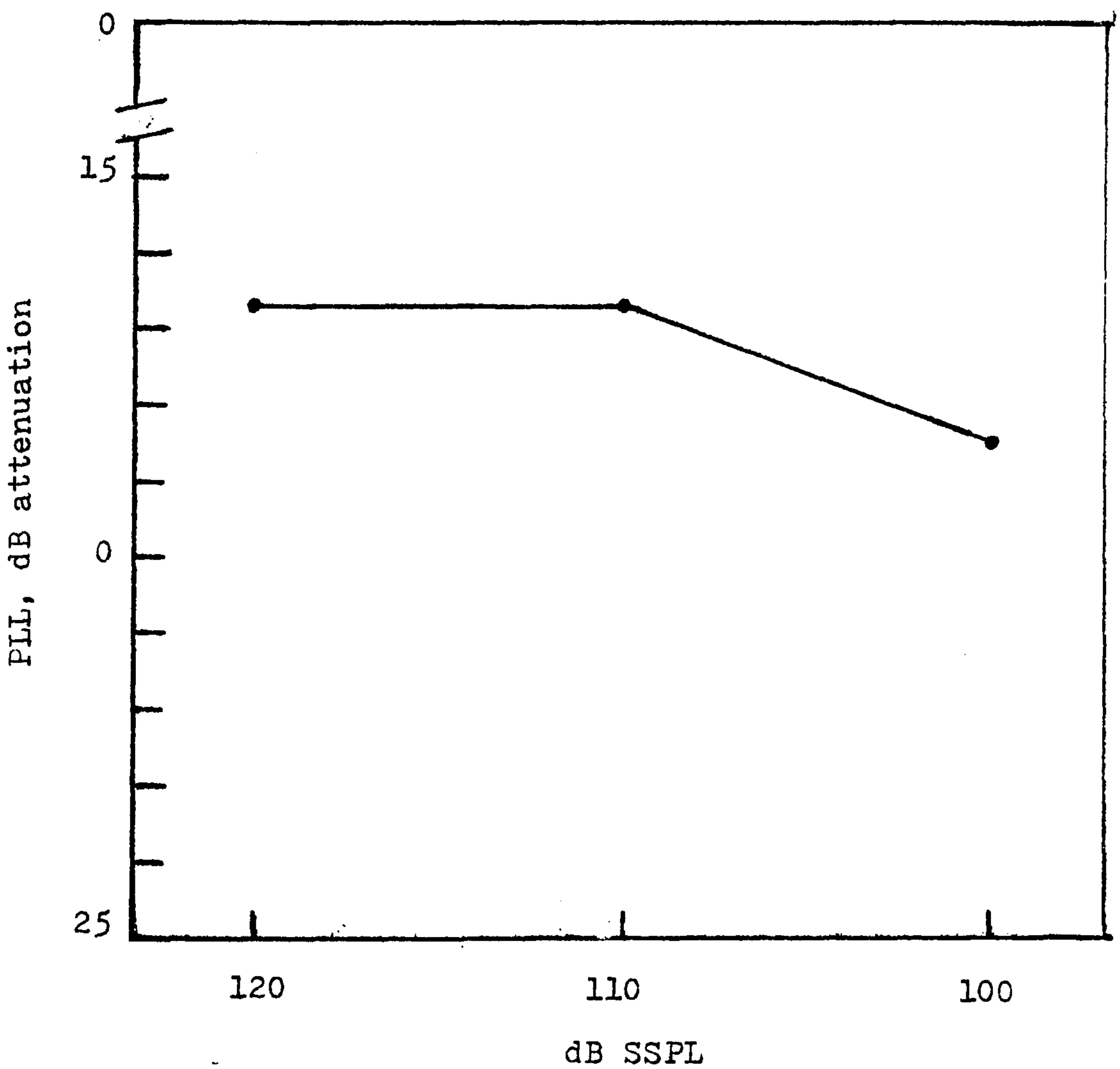

Figure 12. Preferred listening levels (dB attenuation) as a function of output limiting level (dB SSPL).

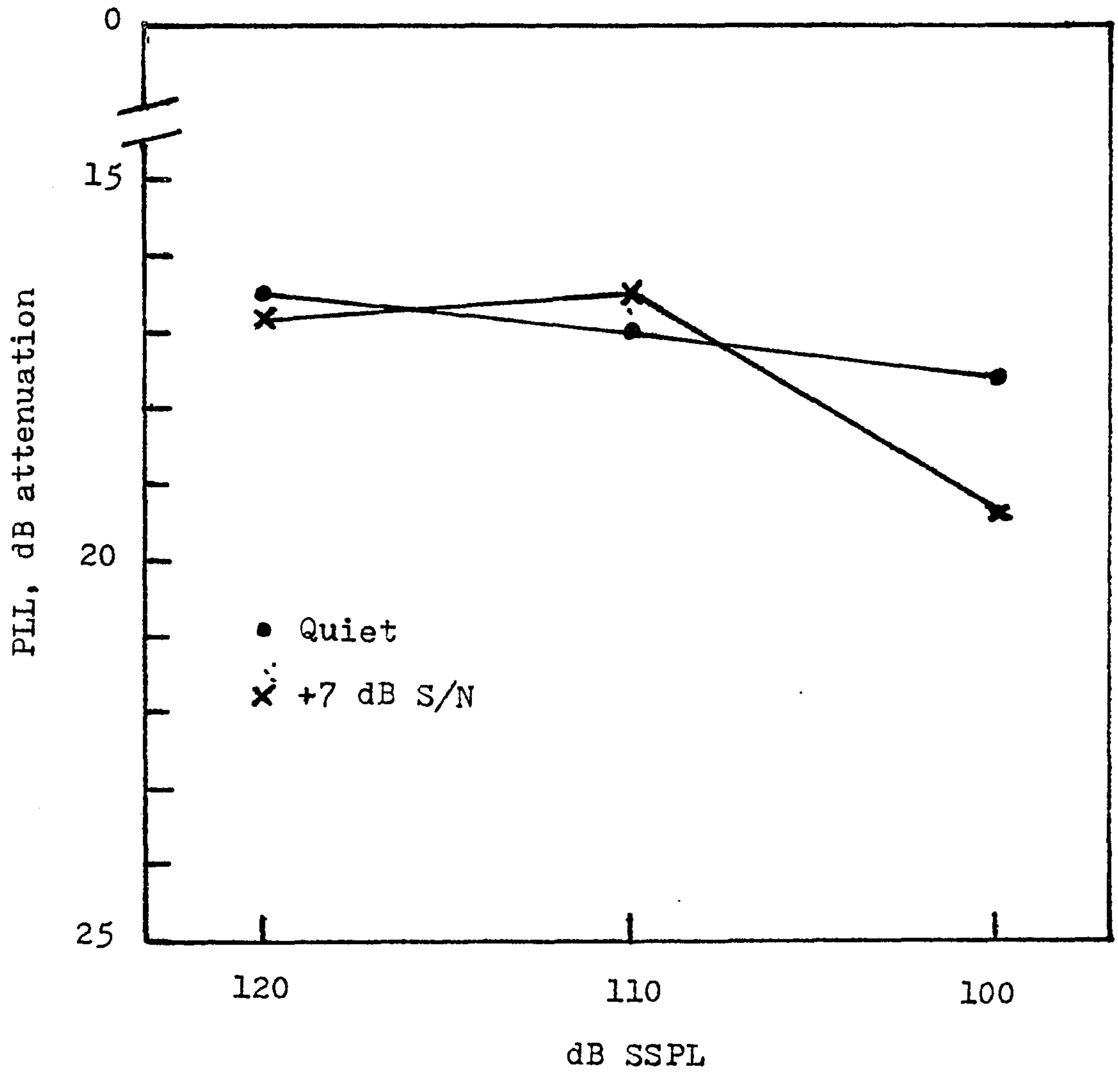

Figure 13. Preferred listening level (dB attenuation) as a function of output limiting level (dB SSPL) and background (quiet, +7 dB S/N).

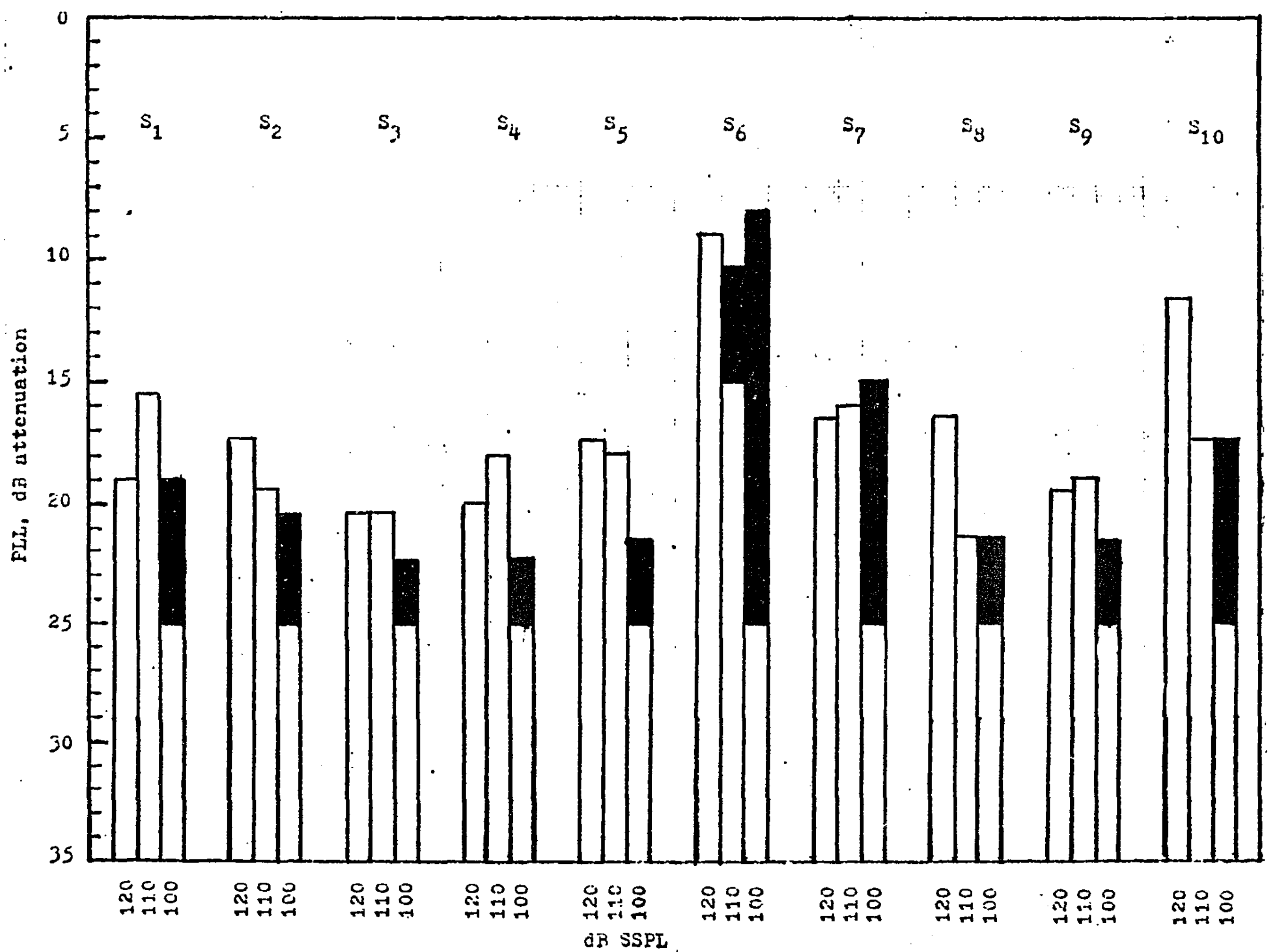

Figure 14. PLL's (dB attenuation) of individual subjects averaged over test sessions (test and retest) for the 0 dB/octave slope in quiet with each output limiting level (dB SSPL). Shaded area depicts the amount each listening level exceeded the saturation threshold for each SSPL value.

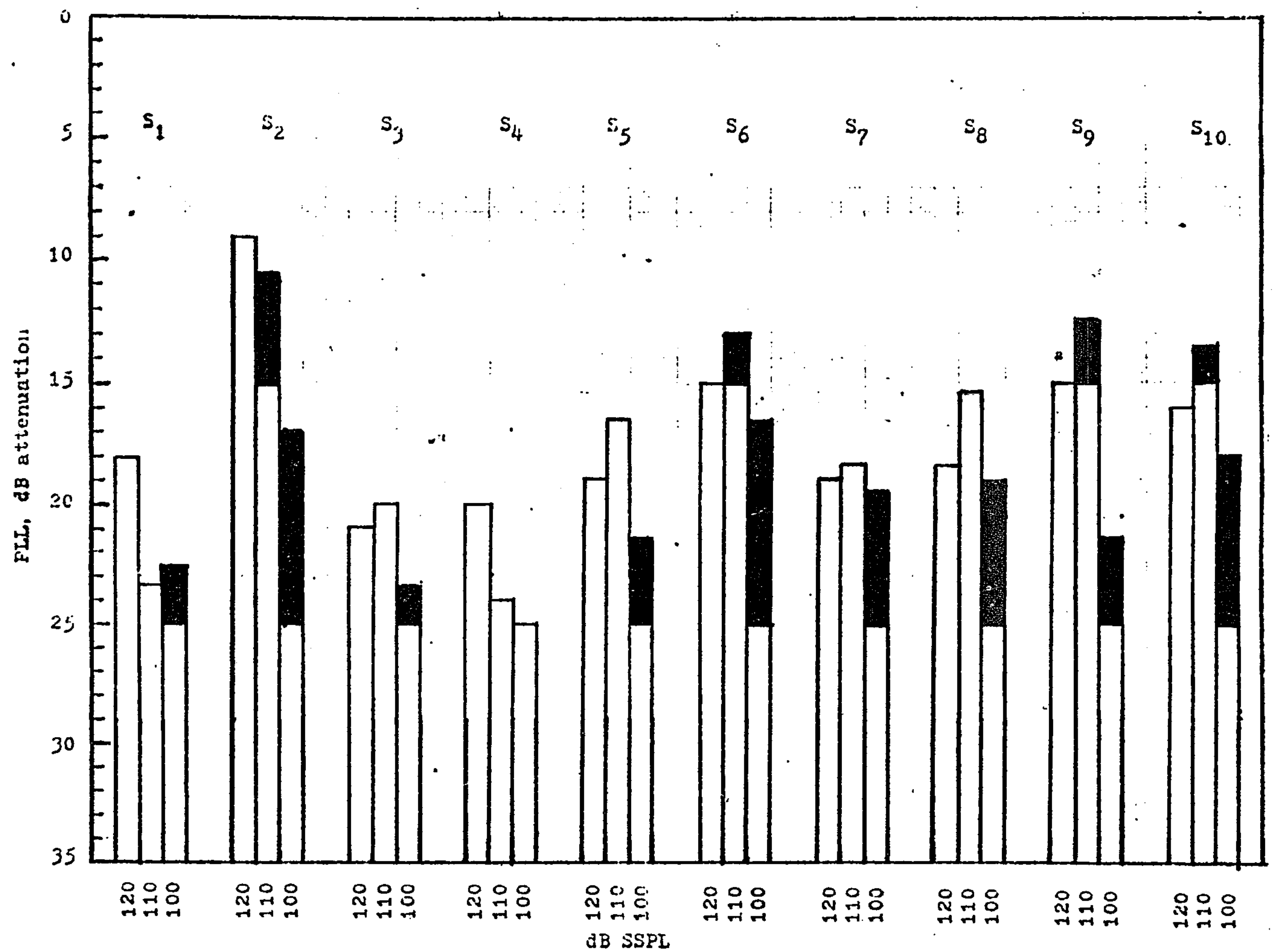

Figure 15. PLL's (dB attenuation) of individual subjects averaged over test sessions (test and retest) for the 0 dB/octave slope in noise (+7 dB S/N) with each output limiting level (dB SSPL). Shaded area depicts the amount each listening level exceeded the saturation threshold for each SSPL value.

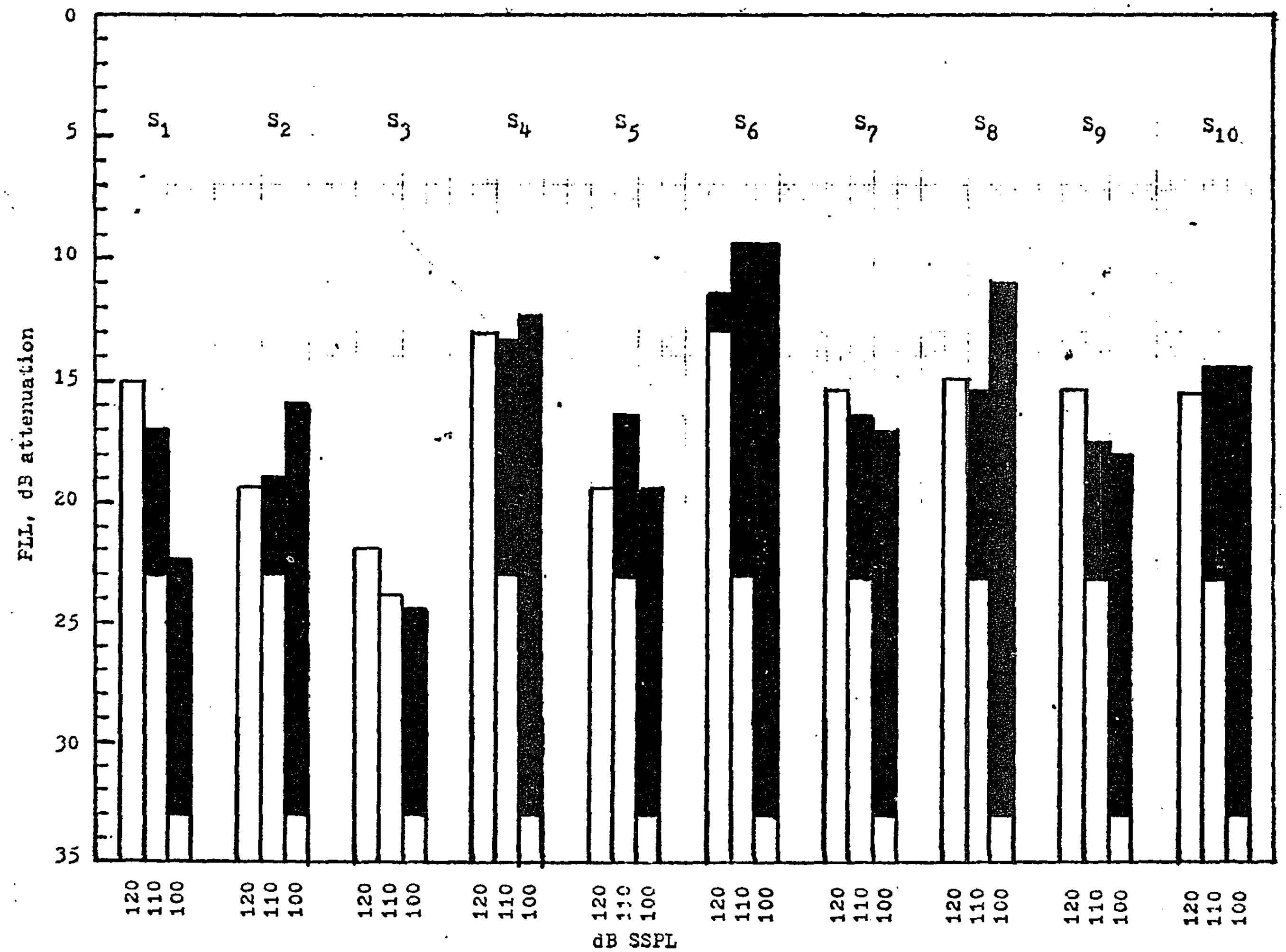

Figure 16. PLL's (dB attenuation) of individual subjects averaged over test sessions (test and retest) for the +6 dB/octave slope in quiet with each output limiting level (dB SSPL). Shaded area depicts the amount each listening level exceeded the saturation threshold for each SSPL value.

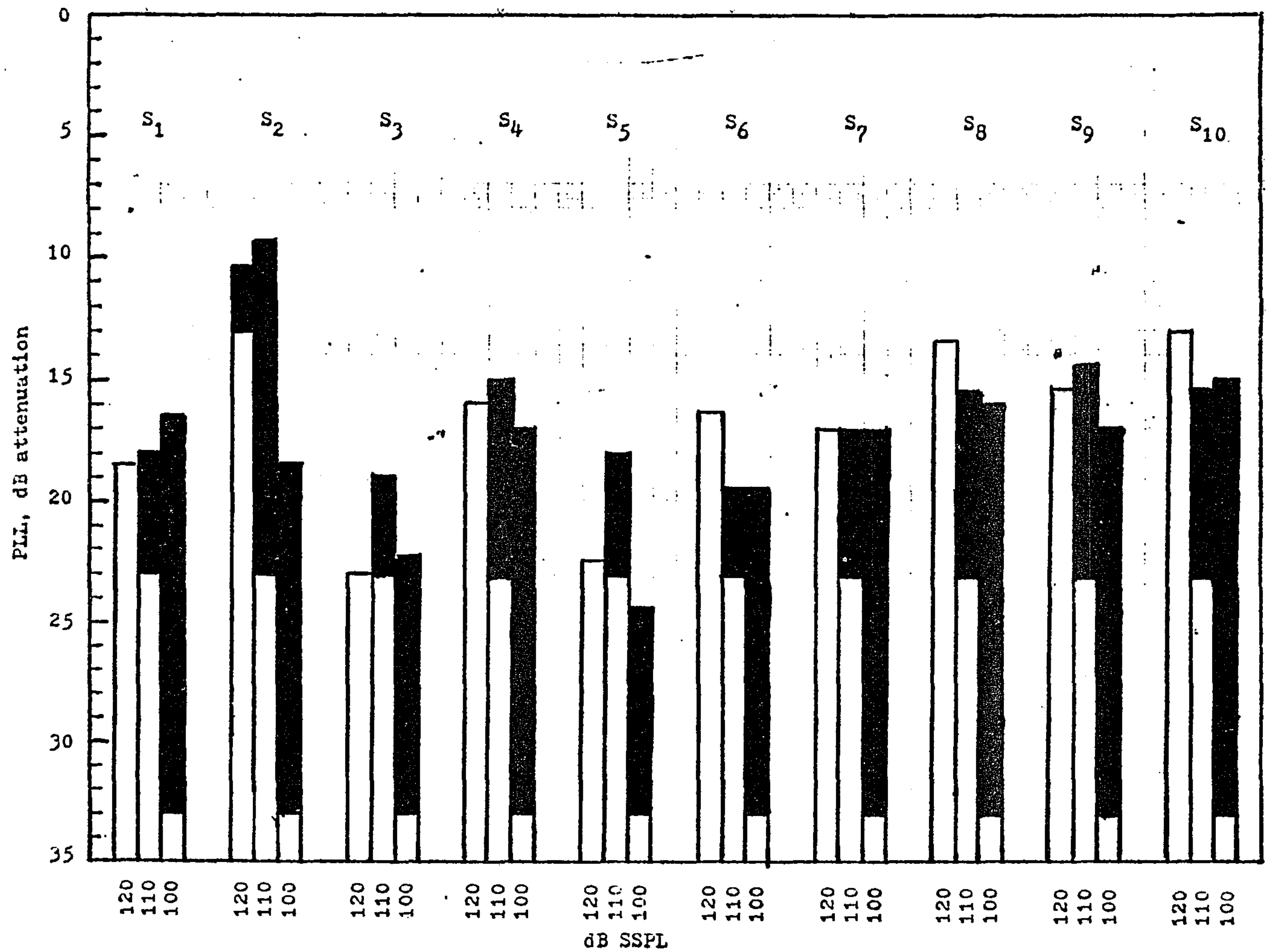

Figure 17. PLL's (dB attenuation) of individual subjects averaged over test sessions (test and retest) for the +6 dB/octave slope in noise (+7 dB S/N) with each output limiting level (dB SSPL). Shaded area depicts the amount each listening level exceeded the saturation threshold for each SSPL value.

the 0 dB/octave and +6 dB/octave slope conditions. In contrast, for the lowest output limiting level (i.e., 100 dB SSPL), PLL's exceeded the saturation threshold for almost all subjects. This was also true for the intermediate limiting level (i.e., 110 dB SSPL) for the +6 dB/octave slope condition. Almost all subjects, whose unlimited PLL (at 120 dB SSPL) equaled or exceeded the saturation threshold of the lowest output limiting level, reduced their listening level for the lowest output limiting level conditions. This effect appeared to be even more marked under conditions characterized by competing noise.

Figures 14 through 17 also reveal the nature of the interactions between background and subjects and between frequency response and subjects. For example, subject #2 consistently had a higher PLL in noise than in quiet. Similarly, subjects #4 and #8 had relatively higher PLL settings for the +6 dB/octave frequency response condition.

<u>Reliability</u>

Data relevant to an analysis of reliability was included in the main experimental design. Table 6 revealed an unexpected, significant main effect for time order. One two-way interaction (time-order x subjects) and one three-way interaction (background x frequency response x time-order) approaching significance were also revealed. Main effect and interaction means for these factors are provided in Table 7. These tables indicate a slightly higher group listening level (1.0 dB) when retesting PLL's. The three-way interaction means reveal that preferred listening levels were higher on retest for the upwardly sloping frequency response condition in quiet, the preferred listening levels for all other conditions remaining essentially the same from test to retest. Finally, the two-way interaction reveals that subjects #3 and #7 listened at much higher levels upon retest than any of the other subjects. The significance of the time-order effect is small in magnitude and appears largely attributable to the PLL results of these two subjects.

The standard deviation of test-retest differences for PLL adjustments was derived from Table 6 by collapsing the sums of squares across all non-significant interactions involving time-order. This analysis resulted in an estimated test-retest standard deviation of 3.14 dB. Similarly, individual data were analyzed with respect to the percentage of total comparisons that test and retest PLL's were within a specified number of dB from each other. These data are summarized in Table 8 and are in good agreement with the estimated standard deviation. From this table it can be seen that the first and second PLL's were within 3 dB of each other for almost 60% of the comparisons and within 6 dB of each other for almost 90% of the comparisons.

Table 7. PLL main effect and interaction means expressed in dB attenuation for factors involving time-order (test-retest).

			Test	Retest
			17.8	16.8
		S#		
TxS		1	18.8	18.7
		2	16.4	14.8
		3	24.1	19.8
		4	17.3	18.5
		5	19.9	19.2
		6	13.5	12.8
		7	18.6	15.5
		8	16.9	16.1
		9	17.7	16.8
		10	14.7	15.7
	Freq. Response	Back-ground		
BxFxT	0	quiet	17.9	17.4
		noise	18.5	17.6
	+6	quiet	17.6	15.1
		noise	17.0	17.0

Table 8. Observed and predicted reliability for PLL judgments, expressed as the percentage of PLL test-retest comparisons that were within a specified number of dB from each other.

dB difference	Observed agreement (%)	Predicted agreement (%, $\sigma_e = 3.14$ dB)
3	57.5	~68
4	66.7	
5	79.2	
6	88.3	~95
7	91.7	
8	97.5	

Other Main Effects

Frequency response slope was identified as a significant main effect in Table 6. An analysis of the mean data revealed that the overall group listened at a slightly higher level (approximately 1.2 dB) for the +6 dB/octave condition re: the 0 dB/octave condition.

Table 6 also revealed a main effect and two-way interactions involving subjects. Because of the heteroteneity of the subject group, these effects were not unexpected.

Speech Intelligibility Performance

The second main factorial design included the following factors: a) three saturation sound pressure levels (SSPL: 120, 110 and 100), b) three presentation levels (X, Y and Z, as explained in the procedure section), c) two frequency response slopes (0 and +6 dB/octave), d) two listening backgrounds (quiet and +7 dB S/N), and e) subjects (n=10). A phoneme scoring procedure (Walker & Boothroyd, 1976) was used in scoring all intelligibility tests (three possible errors per monosyllabic word), making a 150 item list from each 50 word list. This scoring procedure demonstrates greater reliability for phoneme scoring than whole word scoring, while still maintaining a monotonic relationship to whole word scores (Walker & Boothroyd, 1976; Studebaker, et al., 1978). All data for the main design and for the various combination of factors from within the main design were analyzed by means of the same factorial analysis of variance procedure used in the analysis of PLL data. Prior to these analyses, all percentage scores were first transformed into arcsine values,[8] in order to make the error variance independent of the measured score.

Effects of Output Limiting on Intelligibility

Performance at PLL

The effects of output limiting on intelligibility performance at PLL were examined using a factorial design. In this case intelligibility scores at only one level, PLL, were analyzed. A summary table of the analysis of variance is provided in Table 9. No significant main or interactive effects involving SSPL are discernible. Further analysis of the main effects and interaction means suggested a more subtle three-way interaction involving SSPL, background and frequency response, for which the overall F-test does not appear to be sensitive.

Table 10 gives the means in percent intelligibility for the three-way interaction term. From this table it can be seen that no systematic effects of output limiting are obvious for either the 0 or +6 dB/octave slope in quiet or for the 0 dB/octave slope in noise. However, for the +6 dB/octave slope in noise there is a reduction in mean performance as the output limiting level is reduced.

[8] The transformation used was $\theta = 2 \arcsin \sqrt{p}$, (Cohen & Cohen, 1975, p. 256), where p = the proportion correct.

Table 9. Summary table of the analysis of variance for the four factorial intelligibility design, analyzing the effects of output limiting on speech intelligibility at PLL presentation levels. All intelligibility percentage scores were transformed into arcsine values.

SOURCE OF VARIATION	SUMS OF SQUARES	DEGREES OF FREEDOM	MEAN SQUARES	F	SIGNIF.
M	0.02631	2	0.01315	0.431	0.661
B	5.51264	1	5.51264	180.716	0.001
MB	0.02776	2	0.01388	0.455	0.647
F	0.00249	1	0.00249	0.082	0.775
MF	0.04195	2	0.02097	0.688	0.520
BF	0.00185	1	0.00185	0.061	0.803
MBF	0.03923	2	0.01962	0.643	0.542
S	7.40051	9	0.82228	26.956	0.001
MS	0.29633	18	0.01646	0.540	0.900
BS	0.43435	9	0.04826	1.582	0.194
MBS	0.38041	18	0.02113	0.693	0.778
FS	0.10565	9	0.01174	0.385	0.927
MFS	0.48028	18	0.02668	0.875	0.610
BFS	0.11917	9	0.01324	0.434	0.899
MBFS	0.54908	18	0.03050		
TOTAL	15.41802	119			

KEY

M: SSPL: 120, 110, 100

B: Background: Quiet, +7 dB S/N

F: Frequency Response: 0 dB/octave, +6 dB/octave

S: Subjects

Table 10. Mean speech intelligibility performance (%) at PLL, as a function of SSPL, background, and frequency response.

		dB SSPL		
Frequency response	Background	120	110	100
0	quiet	96.1	95.5	95.9
	noise	83.5	82.8	83.9
+6	quiet	95.9	97.1	95.2
	noise	86.0	82.4	81.9

This effect results in changes in the relative performance of the two frequency response slopes in background noise, as output limiting is reduced.

Further analysis of this observation was undertaken on an individual subject basis. This analysis, however, failed to indicate a consistent trend supporting this observation, thus limiting any conclusions which could be derived. Four subjects (#1, #2, #6, #7) demonstrated a significant ($p < .005$) shift from a frequency response superiority favoring the +6 dB/octave slope for the relatively unlimited condition (120 dB SSPL), in favor of the flat response for the maximally limited condition (100 dB SSPL), at presentation levels chosen by the subject for each condition. No significant differences were measured for four subjects (#3, #4, #5, #10) and two subjects (#8, #9) showed a significant shift in the opposite direction.

Performance at PLL vs. Other Presentation Levels

Further analysis was undertaken to determine whether there were any significant differences in intelligibility performance with each output limiting level between PLL presentation levels and the other presentation levels used (Chapter 3, p. 81). A four-factor analysis of variance consisting of the factors, presentation level, background condition, frequency response slope and subjects, was carried out for each of the three SSPL values used. Summary tables of these analyses are shown in Tables 11 through 13.

Table 11. Summary table of the analysis of variance for the four factorial intelligibility design, comparing speech intelligibility performance at PLL vs. other presentation levels with a 120 dB SSPL output limiting level. All intelligibility percentage scores were transformed into arcsine values.

SOURCE OF VARIATION	SUMS OF SQUARES	DEGREES OF FREEDOM	MEAN SQUARES	F	SIGNIF.
L	0.03496	2	0.01748	1.060	0.369
B	5.15654	1	5.15654	312.655	0.001
LB	0.00898	2	0.00449	0.272	0.768
F	0.03192	1	0.03192	1.935	0.178
LF	0.13050	2	0.06525	3.956	0.037
BF	0.00678	1	0.00678	0.411	0.536
LBF	0.00972	2	0.00486	0.295	0.752
S	6.51621	9	0.72402	43.900	0.001
LS	0.14388	18	0.00799	0.485	0.933
BS	0.28552	9	0.03172	1.924	0.114
LBS	0.22989	18	0.01277	0.774	0.704
FS	0.28089	9	0.03121	1.892	0.119
LFS	0.19479	18	0.01082	0.656	0.810
BFS	0.34337	9	0.03815	2.313	0.062
LBFS	0.29687	18	0.01649		
TOTAL	13.67080	119			

<u>KEY</u>

L: Level: X, Y, Z
B: Background: Quiet, +7 dB S/N
F: Frequency Response: 0 dB/octave, +6 dB/octave
S: Subjects

Table 12. Summary table of the analysis of variance for the four factorial intelligibility design, comparing speech intelligibility performance at PLL vs. other presentation levels with a 110 dB SSPL output limiting level. All intelligibility percentage scores were transformed into arcsine values.

SOURCE OF VARIATION	SUMS OF SQUARES	DEGREES OF FREEDOM	MEAN SQUARES	F	SIGNIF.
L	0.02162	2	0.01081	0.617	0.555
B	5.37125	1	5.37125	306.412	0.001
LB	0.05423	2	0.02711	1.547	0.239
F	0.02649	1	0.02649	1.511	0.233
LF	0.02698	2	0.01349	0.769	0.518
BF	0.00472	1	0.00472	0.270	0.615
LBF	0.03179	2	0.01589	0.907	0.576
S	8.19810	9	0.91090	51.964	0.001
LS	0.33340	18	0.01852	1.057	0.454
BS	0.42699	9	0.04744	2.706	0.034
LBS	0.44505	18	0.02473	1.410	0.236
FS	0.17345	9	0.01927	1.099	0.411
LFS	0.20202	18	0.01122	0.640	0.824
BFS	0.37128	9	0.04125	2.353	0.058
LBFS	0.31553	18	0.01753		
TOTAL	16.00288	119			

KEY

L: Level: X, Y, Z
B: Background: Quiet, +7 dB S/N
F: Frequency Response: 0 dB/octave, +6 dB/octave
S: Subjects

Table 13. Summary table of the analysis of variance for the four factorial intelligibility design, comparing speech intelligibility performance at PLL vs. other presentation levels with a 100 dB SSPL output limiting level. All intelligibility percentage scores were transformed into arcsine values.

SOURCE OF VARIATION	SUMS OF SQUARES	DEGREES OF FREEDOM	MEAN SQUARES	F	SIGNIF.
L	0.03199	2	0.01600	0.714	0.507
B	7.77273	1	7.77273	346.811	0.001
LB	0.13106	2	0.06553	2.924	0.078
F	0.01643	1	0.01643	0.733	0.592
LF	0.01144	2	0.00572	0.255	0.780
BF	0.02745	1	0.02745	1.225	0.283
LBF	0.00546	2	0.00273	0.122	0.886
S	6.14680	9	0.68298	30.474	0.001
LS	0.52229	18	0.02902	1.295	0.294
BS	0.90880	9	0.10098	4.506	0.004
LBS	0.45309	18	0.02517	1.123	0.404
FS	0.23013	9	0.02557	1.141	0.386
LFS	0.63965	18	0.03554	1.586	0.168
BFS	0.12215	9	0.01357	0.606	0.777
LBFS	0.40342	18	0.02241		
TOTAL	17.42284	119			

KEY

L: Level: X, Y, Z
B: Background: Quiet, +7 dB S/N
F: Frequency Response: 0 dB/octave, +6 dB/octave
S: Subjects

The primary factor of interest in this analysis is that of level and its interactions with other factors. With the exception of Table 11, statistical significance involving performance differences is only achieved for background and subjects. Such differences across individual subjects and under different background conditions are to be expected. A significant level effect is not observed with any of the output limiting levels.

Table 11 indicates a statistically significant ($p = .037$) interaction between level and frequency response for the lowest output limiting level (i.e., 120 dB SSPL). Mean values for these combined factors revealed that for the +6 dB/octave slope condition, a higher speech intelligibility score was obtained at a presentation level corresponding to that chosen for the 110 dB SSPL conditions (i.e., listening level Y). This presentation level was equal to the highest mean listening level for the group for this combination of conditions.

The only other interaction involving level which approached statistical significance (i.e., $.1 > p > .05$) may be found in Table 13 for the two-way interaction involving level and background ($p = .078$). Mean performance levels for these two factors revealed that, in noise, best performance was obtained at the lowest presentation level, which corresponded to the mean PLL of the group with that output limiting level (i.e., 100 dB SSPL). Thus, with the exception of this trend favoring intelligibility performance at PLL presentation levels with the lowest output limiting level (i.e., 100 dB SSPL), no significant difference in performance between the three presentation levels was observed with any of the output limiting levels used.

Effects of Limiting Levels on Speech Intelligibility

The final analysis of the intelligibility test data centered around issues related to output limiting level in the form of peak clipping and its effects on intelligibility. Although the experimental design did not include experimental conditions with controlled amounts of distortion, the various presentation levels and SSPL values used in the study provided data which could be utilized in studying intelligibility effects related to amplitude distortion.

Two types of analyses were relevant to a study of this issue. The first involved the significance of SSPL as a main effect and its interactions with other factors. This information is contained in the analysis of variance for the main five factor experimental design for intelligibility provided in Table 14.

A significant main effect is seen for SSPL, with a significant two-way interaction between SSPL and background. A two-way interaction between SSPL and subjects approaches significance ($.1 > p > .05$). Three-way interactions

Table 14. Summary table of the analysis of variance for the five factorial intelligibility design, comparing the effects of output limiting levels on speech intelligibility. All intelligibility percentage scores were transformed into arcsine values.

SOURCE OF VARIATION	SUMS OF SQUARES	DEGREES OF FREEDOM	MEAN SQUARES	F	SIGNIF.
L	0.01377	2	0.00689	0.517	0.606
M	0.12941	2	0.06471	4.857	0.013
LM	0.07480	4	0.01870	1.403	0.252
B	18.13693	1	18.13693	1361.277	0.001
LB	0.09737	2	0.04869	3.654	0.035
MB	0.16363	2	0.08181	6.141	0.005
LMB	0.09690	4	0.02423	1.818	0.146
F	0.01516	1	0.01516	1.138	0.293
LF	0.07004	2	0.03502	2.628	0.084
MF	0.05968	2	0.02984	2.240	0.119
LMF	0.09888	4	0.02472	1.855	0.139
BF	0.00771	1	0.00771	0.579	0.542
LBF	0.02071	2	0.01036	0.777	0.529
MBF	0.03125	2	0.01562	1.173	0.321
LMBF	0.02625	4	0.00656	0.493	0.744
S	20.41982	9	2.26887	170.291	0.001
LS	0.27466	18	0.01526	1.145	0.353
MS	0.44130	18	0.02452	1.840	0.059
LMS	0.72492	36	0.02014	1.511	0.110
BS	0.85486	9	0.09498	7.129	0.001
LBS	0.31146	18	0.01730	1.299	0.245
MBS	0.76644	18	0.04258	3.196	0.002
LMBS	0.81656	36	0.02268	1.702	0.057
FS	0.24395	9	0.02711	2.034	0.063
LFS	0.38792	18	0.02155	1.618	0.108
MFS	0.44052	18	0.02447	1.837	0.059
LMFS	0.64853	36	0.01801	1.352	0.185
BFS	0.41381	9	0.04598	3.451	0.004
LBFS	0.53618	18	0.02979	2.236	0.020
MBFS	0.42298	18	0.02350	1.764	0.072
LMBFS	0.47964	36	0.01332		
TOTAL	47.22581	359			

KEY

L: Level: X, Y, Z
M: SSPL: 120, 110, 100
B: Background: Quiet, +7 dB S/N
F: Frequency Response: 0 dB/octave, +6 dB/octave
S: Subjects

for SSPL x frequency response x subjects, and significant four-way interactions, level x SSPL x background x subjects and SSPL x background x frequency response x subjects, approaching significance ($.1 > p > .05$) can also be seen. Means for the main effect (Table 15) suggest a slight decrease ($\leq 1.2\%$) in performance at 100 dB SSPL when compared with 120 or 110 dB SSPL. Two-way interaction means (Table 15) indicate that this effect occurred under conditions of background noise and only for certain subjects, but not in quiet. Further analysis of the three- and four-way interaction terms involved individual subject differences, which were not of direct interest.

Another way in which the effects of limiting levels on intelligibility were studied involved analyzing performance differences at a constant SSPL value, while presentation level was varied. As indicated in the earlier analysis of PLL's, listening levels exceeded the saturation threshold for almost every subject for the 100 dB SSPL value. Thus, varying presentation levels under extreme limiting should have resulted in varying degrees of clipping with concomitant effects on intelligibility performance.

The effects of listening level on intelligibility as a function of all slope and background noise combinations across individual subjects are shown as Figures 18 through 23. These figures reveal that for relatively unlimited conditions (i.e., 120 and 110 dB SSPL), systematic patterns of intelligibility performance were not

Table 15. Speech intelligibility main effect and interaction means (%) for factors involving SSPL from the five-factorial intelligibility design comparing the effects of output limiting levels on speech intelligibility.

		dB SSPL		
		120	110	100
		91.0	91.1	89.9
	Background			
MxB	quiet	96.1	96.2	96.2
	noise	84.2	84.1	80.9
	Subject #			
MxS	1	95.1	95.5	94.7
	2	90.8	93.3	87.7
	3	87.5	85.5	86.0
	4	67.2	65.9	67.8
	5	97.2	97.9	96.1
	6	94.9	96.0	95.6
	7	91.9	94.1	92.5
	8	90.7	87.1	88.2
	9	89.6	87.3	86.9
	10	95.7	95.3	93.7

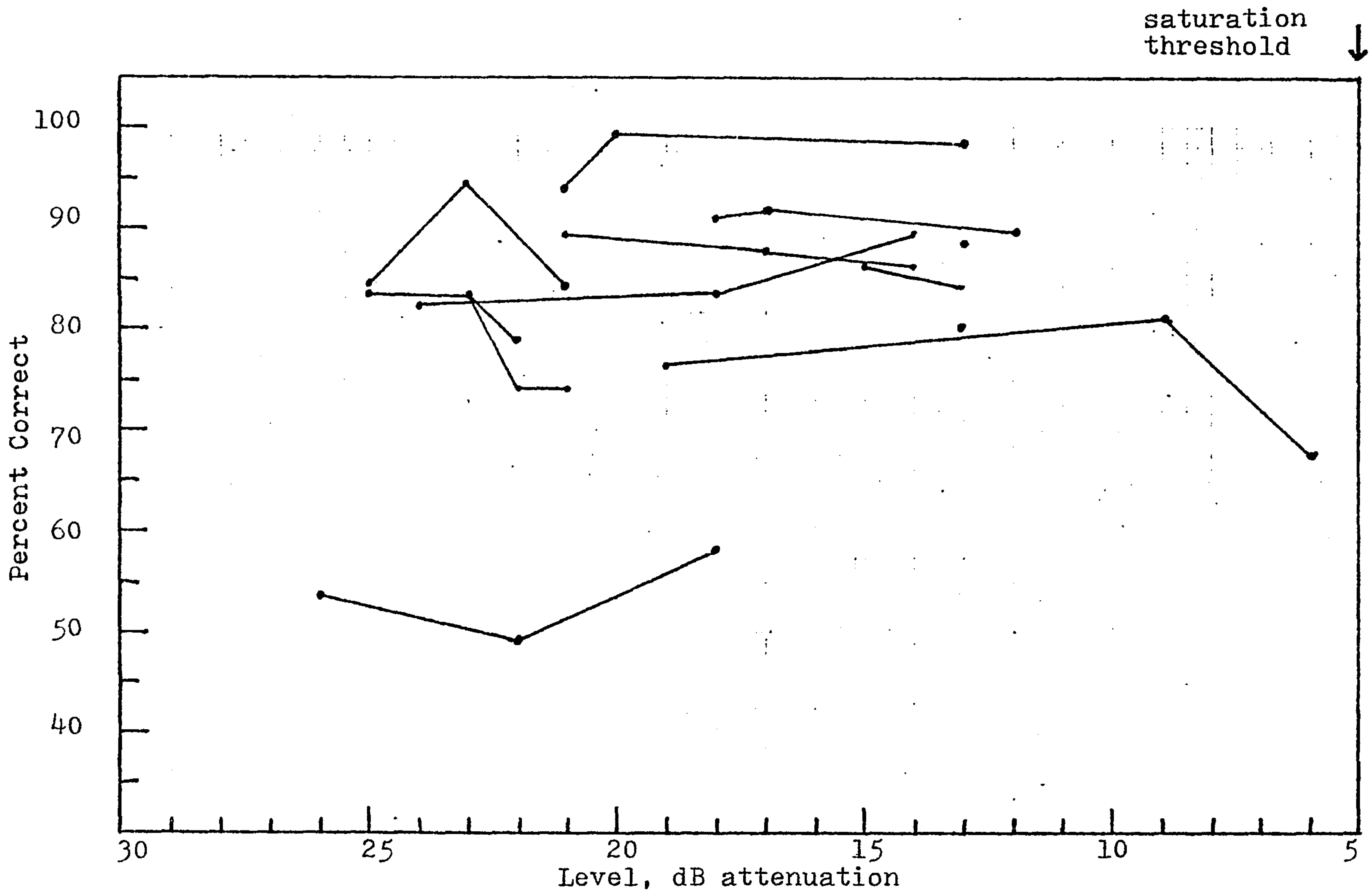

Figure 18. Individual intelligibility performance (%) as a function of presentation level (dB attenuation) for the 0 dB/octave, 120 dB SSPL, +7 dB S/N condition. The saturation threshold is also given.

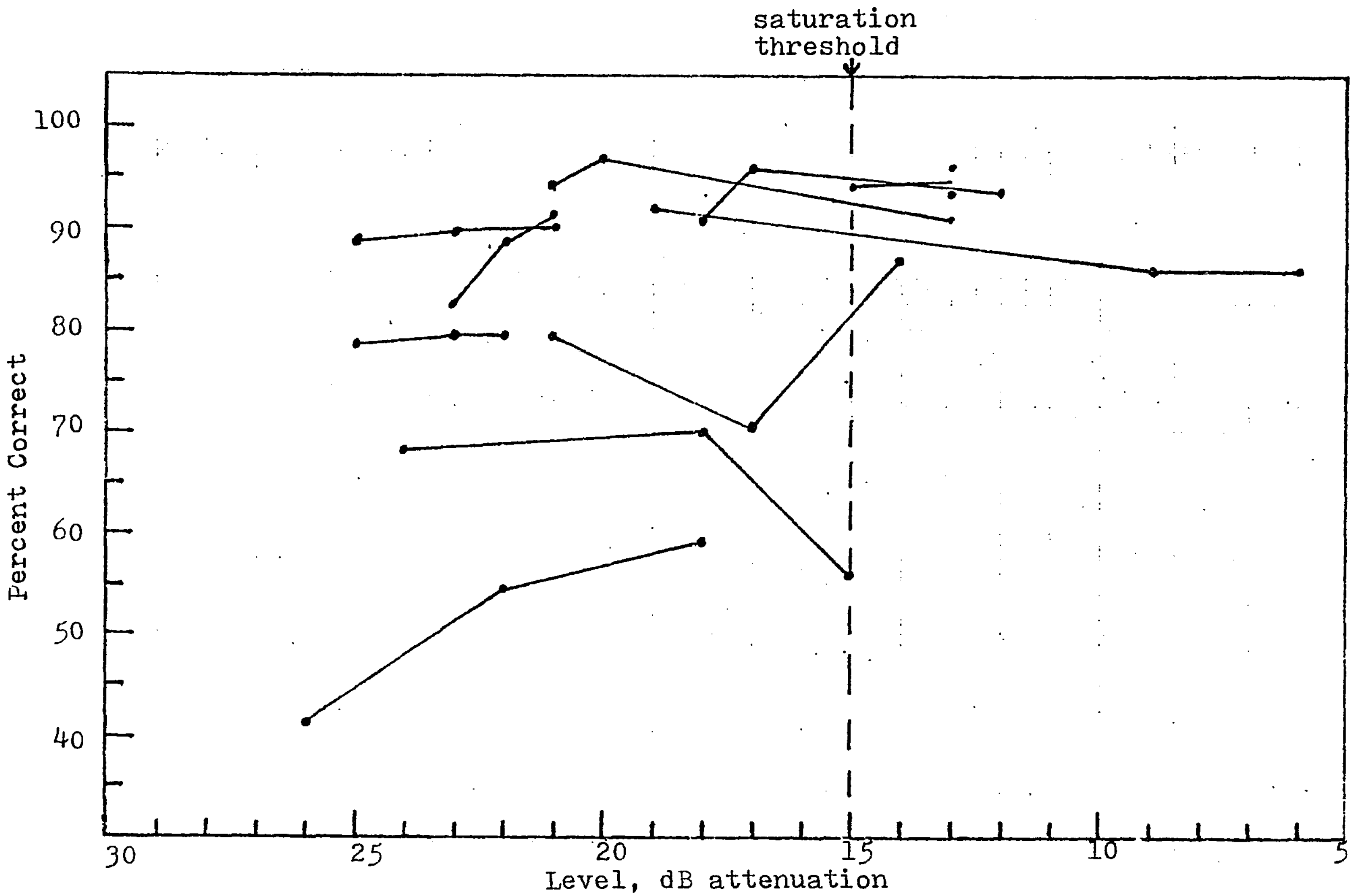

Figure 19. Individual intelligibility performance (%) as a function of presentation level (dB attenuation) for the 0 dB/octave, 110 dB SSPL, +7 dB S/N condition. The saturation threshold is also given.

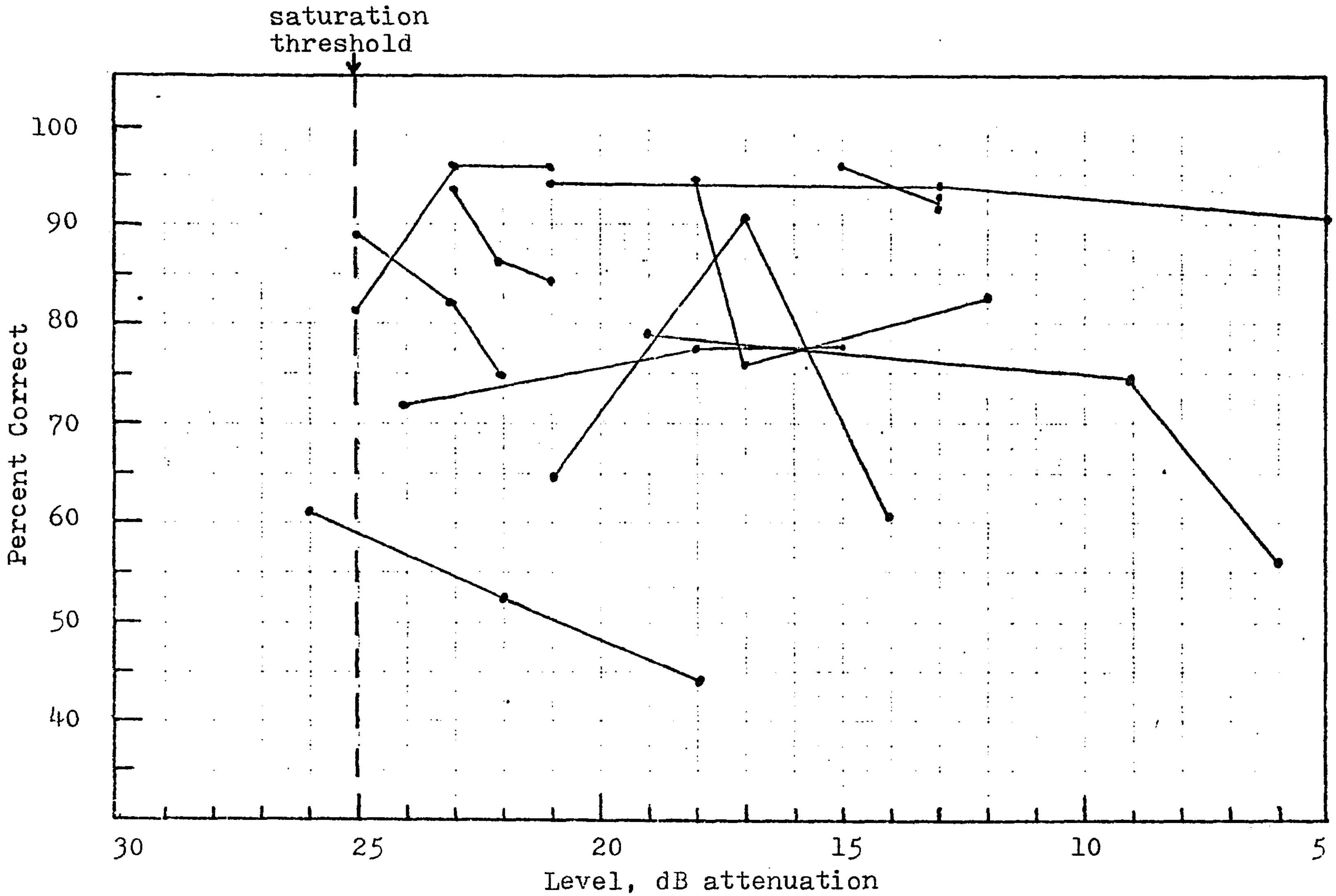

Figure 20. Individual intelligibility performance (%) as a function of presentation level (dB attenuation) for the 0 dB/octave, 100 dB SSPL, +7 dB S/N condition. The saturation threshold is also given.

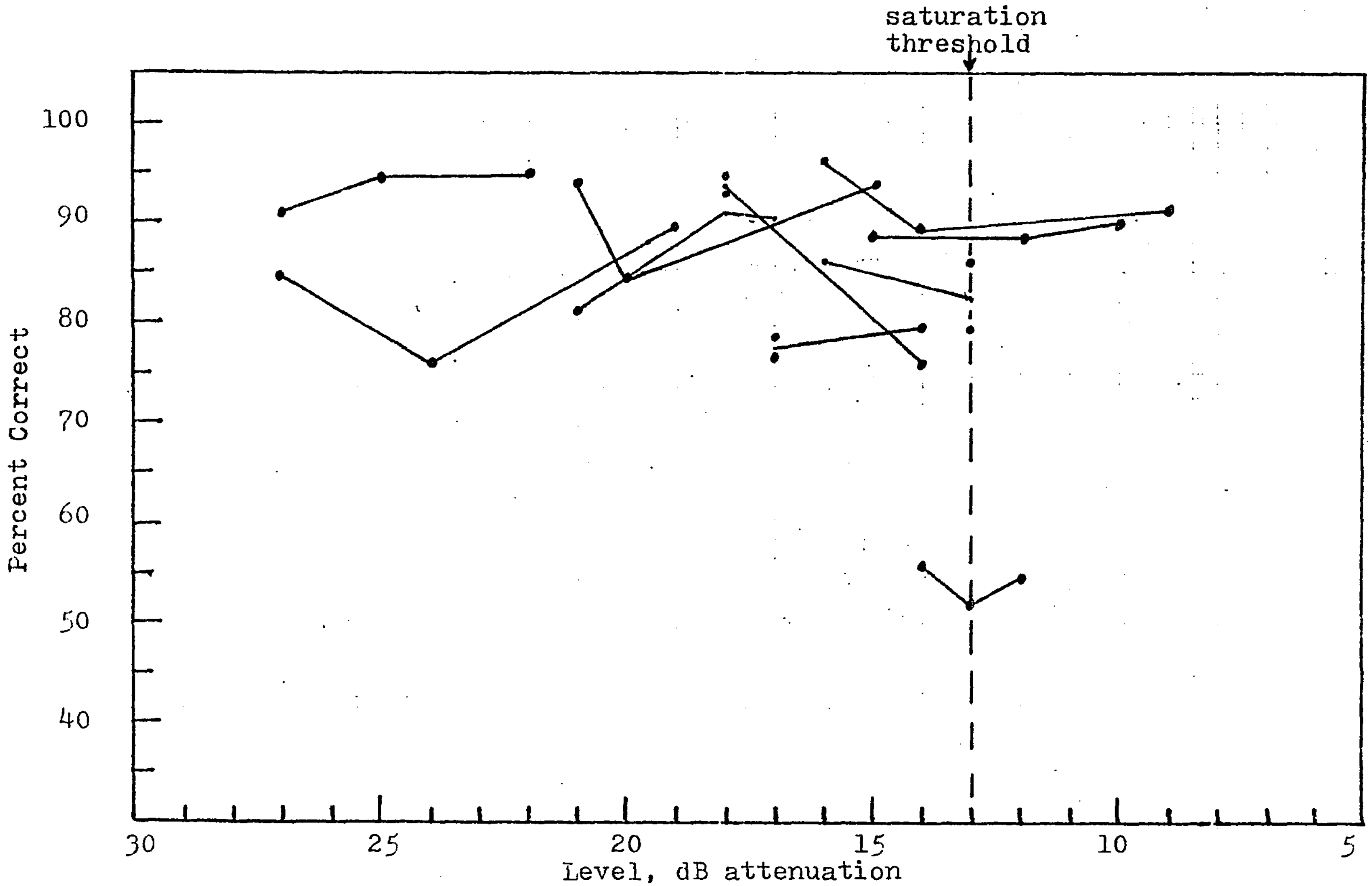

Figure 21. Individual intelligibility performance (%) as a function of presentation level (dB attenuation) for the +6 dB/octave, 120 dB SSPL, +7 dB S/N condition. The saturation threshold is also given.

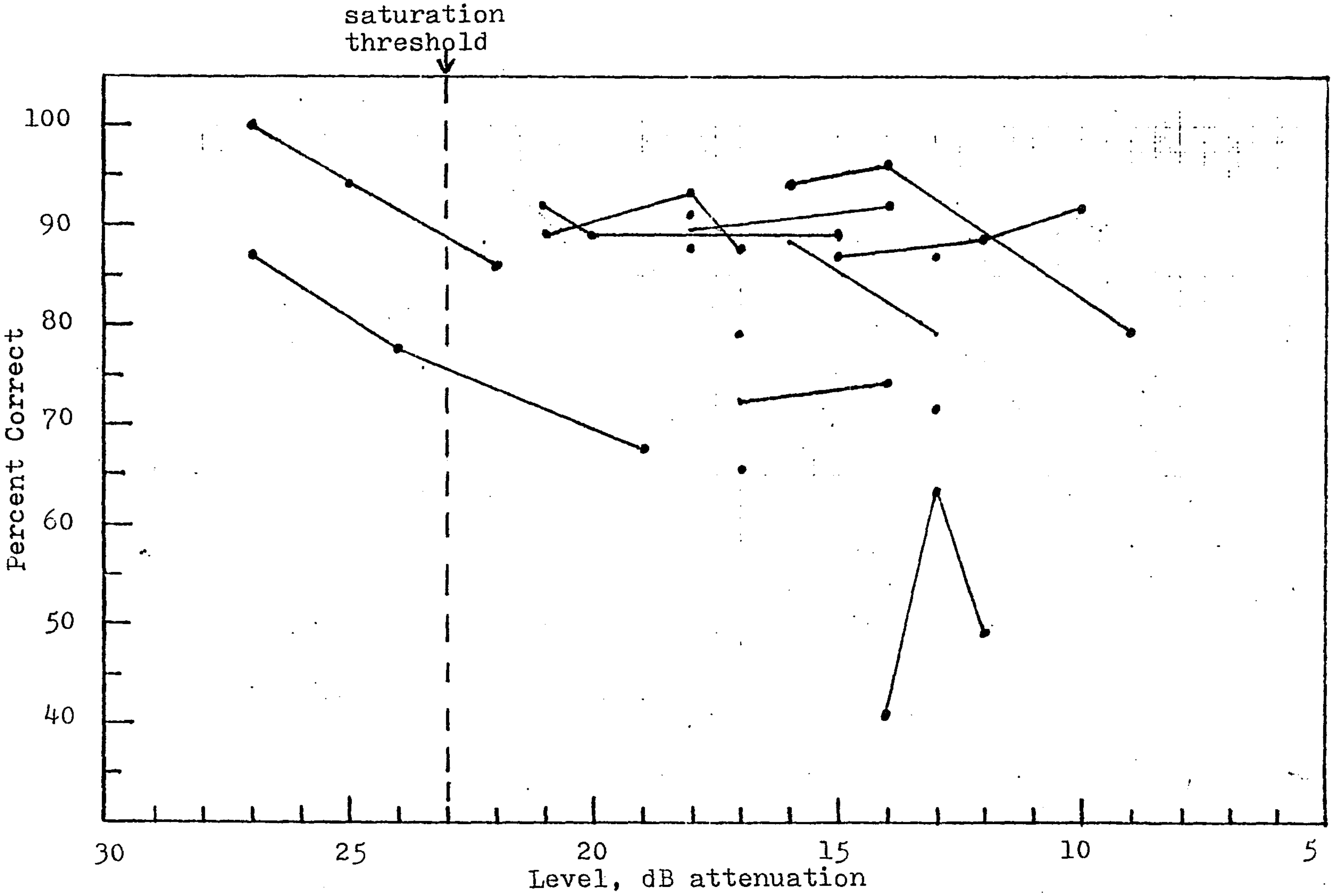

Figure 22. Individual intelligibility performance (%) as a function of presentation level (dB attenuation) for the +6 dB/octave, 110 dB SSPL, +7 dB S/N condition. The saturation threshold is also given.

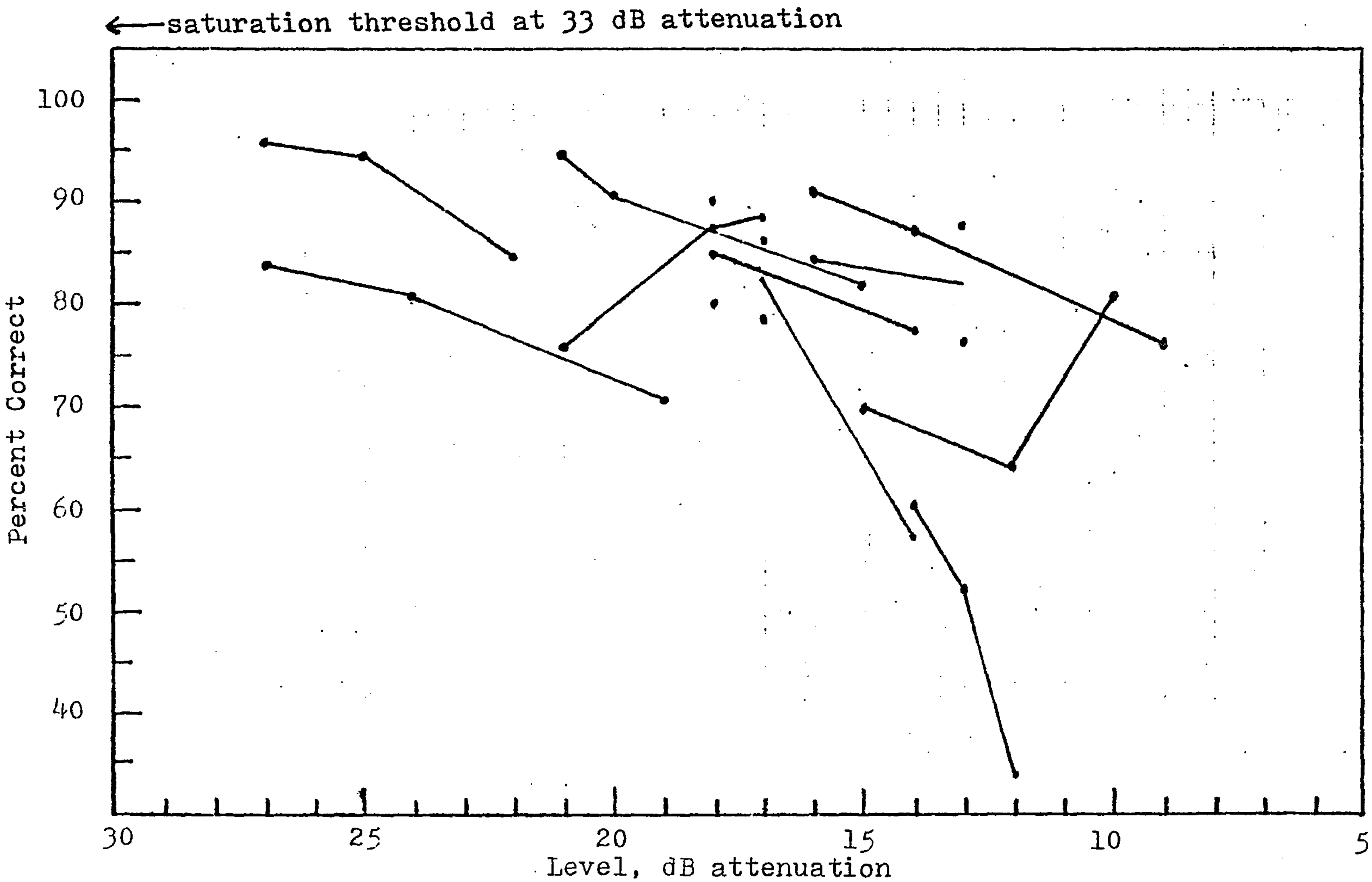

Figure 23. Individual intelligibility performance (%) as a function of presentation level (dB attenuation) for the +6 dB/octave, 100 dB SSPL, +7 dB S/N condition. The saturation threshold is also given.

readily apparent. However, Figures 19 and 22, which represent intelligibility performance under conditions of background noise and severe limiting (100 dB SSPL) for both slope conditions, suggest decreased intelligibility performance at higher listening levels, which were progressively more limited. Furthermore, these figures suggest more of a deleterious effect on intelligibility for the high frequency emphasis condition as presentation level was increased. Similar effects were not revealed for quiet background conditions. Data plots for the quiet background conditions are given in Appendix H.

As a means of further contrasting these differences in performance for the different frequency slopes in noise at different limiting levels, a linear regression line was fitted to the "normalized" arcsine scores (y-axis) and "normalized" listening levels (x-axis) of the individual subjects. Normalization of the data was necessary because of the heterogeneous characteristics of the subject group. Arcsine scores were normalized by assigning a value of 1.0 to the score at the subject's lowest listening level and relative arcsine values to scores obtained at higher listening levels. With respect to listening levels, a value of 10 was assigned to the individual's lowest listening level (i.e., highest attenuation value), and relative dB attenuation values to higher listening levels. The resulting slopes are shown in Figures 24 and 25. These lines represent an average slope or average change in

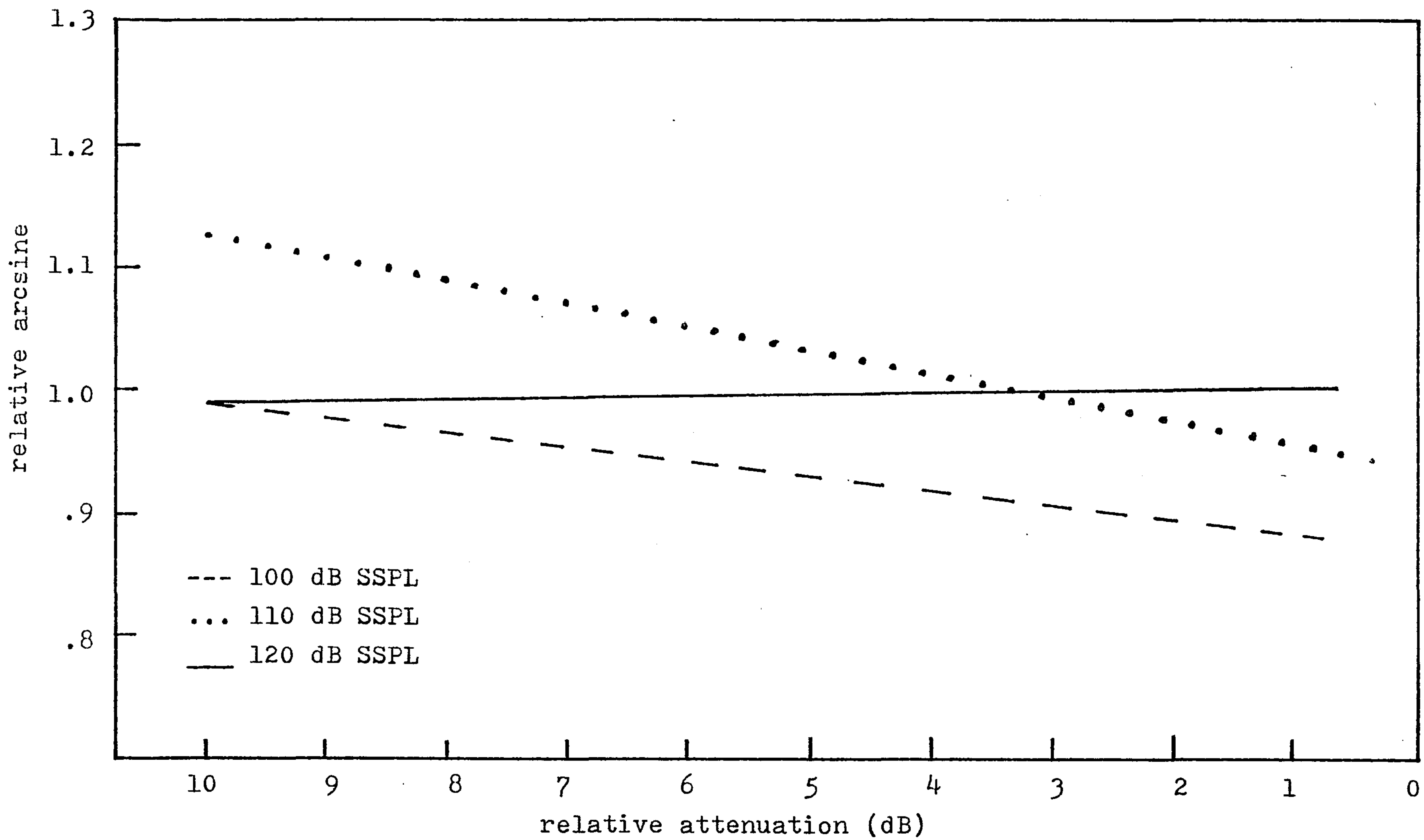

Figure 24. Linear best-fit lines for normalized individual intelligibility scores (relative arcsine) as a function of normalized presentation level (relative attenuation, dB) for the three output limiting levels (dB SSPL) with a 0 dB/octave slope in background noise (+7 dB S/N).

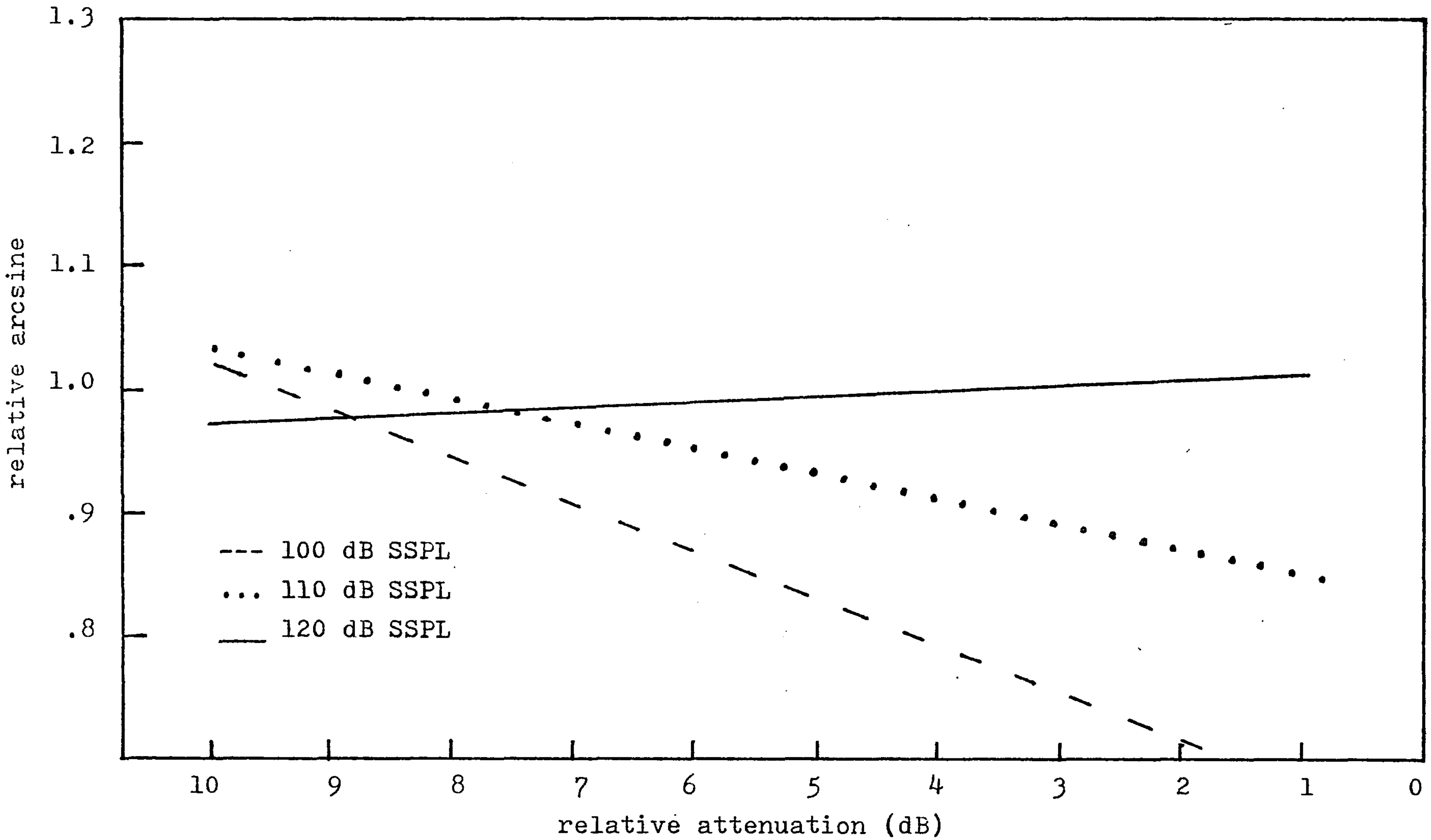

Figure 25. Linear best fit lines for normalized individual intelligibility scores (relative arcsine) as a function of normalized presentation level (relative attenuation, dB) for the three output limiting levels (dB SSPL) with a +6 dB/octave slope in background noise (+7 dB S/N).

performance, under each of the six conditions as listening level is changed. They reveal that, for both frequency response conditions (0 and +6 dB/octave), performance, on average, increased very slightly with decreasing attenuation at 120 dB SSPL. At 110 dB SSPL, average performance decreased with higher listening levels, more for the +6 dB/octave than for the 0 dB/octave slope. At 100 dB SSPL, average performance decreased at about the same rate as at 110 dB SSPL for the 0 dB/octave slope, but at a faster rate for +6 dB/octave.

Table 16 provides information relevant to the calculated values for the linear regression equations for each slope condition. In addition, this table provides information which describes the magnitude of the average individual slope effects of each best-fit curve, expressed in terms of the percentage change which would occur in the 50% performance region for each dB of change in listening level. Percentage changes in terms of intelligibility reduction ranging from .06%/dB for the 0 dB/octave, 120 dB SSPL condition to 1.83%/dB for the +6 dB/octave, 100 dB SSPL condition are revealed.

Table 16. Calculated values for linear best fit functions of Figures 24 and 25. Also given are the percentage changes which would occur in the 50% performance region for each dB of change in listening level.

Freq. Response	dB SSPL	Constant, a	Slope, b (arc-sine/dB)	Corr. Coeff., r	% Change/ dB
	120	1.004	-.0012	.034	.06
0	110	.932	+.0196	.281	.98
	100	.873	+.0117	.193	.59
	120	1.017	-.0045	.064	.23
+6	110	.829	+.0207	.171	1.04
	100	.650	+.0366	.386	1.83

CHAPTER V

DISCUSSION

Introduction

The purpose of this study was to evaluate the effects of output limiting in the form of peak clipping on the performance of hearing impaired listeners. Performance was evaluated in two stages. First, each individual's self-adjusted listening levels were obtained under varying degrees of limiting. Then, intelligibility scores for monosyllabic word lists were obtained under these same conditions at the previously obtained listener adjusted levels and at additional presentation levels. The effects of output limiting on these performance measures were studied as a function of frequency response and background noise.

The experimental hypothesis sought to demonstrate interactive effects between listening levels and output limiting which could affect an individual's intelligibility performance. Previous research reviewed in Chapter 2, indicated a potential for peak clipping to impose qualitative and/or quantitative effects on an individual's perception of speech. In listening circumstances where a listener's unlimited listening levels for speech are high, progressive amounts of limiting in the form of peak clipping could lead to compensatory listening level (gain) adjustments and/or subsequent changes in the listener's intelligibility performance.

In this study a new approach to listener gain adjustments was developed. This approach, called the "Preferred Listening Level" (PLL) instructed the listener to adjust his/her listening level with a primary emphasis on intelligibility, while remaining compatible with listener comfort. The rationale underlying this approach was discussed in Chapter 2. It was hypothesized that the PLL approach would represent an effective compromise between hearing aid level setting approaches based on an individual's maximum speech intelligibility score (PB-max), independent of listener preference, and approaches based on an individual's comfort level, independent of intelligibility performance.

PLL adjustments were obtained using test materials identical to those subsequently used for intelligibility measurements. These materials were contextually more demanding of the listener's attention to intelligibility than standardly used spondaic word or continuous discourse speech materials. With an emphasis on intelligibility, chosen preferred listening levels were expected to more closely approximate the upper limits of the comfort level range. These settings could result in speech peak levels exceeding the lower limits of the range of LDL's now being reported for hearing impaired individuals with moderate hearing impairment.

Preferred Listening Levels

The main question with respect to preferred listening levels concerned the effects of variations in the degree of output limiting on a listener's PLL, as a function of frequency response and background noise. At the same time, factors relevant to an evaluation of the PLL concept itself were examined, as a means of establishing its validity for hearing aid applications. These included an evaluation of its reliability and a comparison of absolute PLL levels in dB SPL with comfort level and gain adjustment data reported in the literature for similar populations of hearing impairment.

Effects of Output Limiting

Before considering the effects of output limiting on PLL, an initial determination was made of each individual's PLL for the highest limiting level (i.e., 120 dB SSPL). This analysis was undertaken in order to determine whether or not the peak levels of the speech adjusted by the listener exceeded the saturation thresholds of the progressively more severe limiting levels chosen for study. It was revealed that, for all subjects, the speech peak levels of PLL's for the least limited condition always exceeded the saturation threshold of the lowest limiting level (100 dB SSPL), and sometimes exceeded the saturation threshold of the intermediate limiting level (110 dB SSPL). PLL's for the highest limiting level, however, virtually never exceeded the saturation threshold. Thus, the potential for interactive effects with peak clipping existed.

Analysis of the relationship between output limiting and listeners' PLL gain[9] adjustments revealed a significant reduction of mean gain adjustments for the lowest output limiting level, when compared with gain adjustments for both the intermediate and highest output limiting levels. Further analysis indicated that this effect was largely attributable to a reduction in mean gain under background noise (+7 dB S/N) conditions.

Individual data averaged across replications revealed substantial individual differences. PLL adjustments ranged from slight increases in gain to gain reductions of as much as 9 dB, as output limiting conditions changed from virtually no limiting to the most severe degree of output limiting (i.e., 100 dB SSPL).

To the extent that SSPL values used in this study were representative of levels now being recommended for use in hearing aids for individuals with similar hearing loss characteristics, the PLL's obtained under essentially unlimited conditions were high enough to be influenced by progressively more output limiting in the form of peak clipping. Reducing the limiting level to its lowest value imposed clipping and associated amplitude distortion of

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[9] In this section, increases and decreases in PLL will be referred to as increases and decreases in gain, respectively, to remain consistent with hearing aid terminology.

speech peaks at the listener's PLL chosen for the unlimited condition. This resulted in relative gain setting reductions, particularly under conditions with background noise. Despite these gain reductions, as the limiting level was progressively reduced, almost all listeners continued to listen at levels where speech peaks exceeded saturation thresholds for that condition. Listeners may not have reduced their gain to a point where clipping of the speech peaks was completely eliminated either because the clipping effects were not audible or because further reductions in gain may have resulted in a loss of audibility. With an intensity range between strong and weak speech sounds of 30 to 50 dB, reducing the gain to eliminate clipping of the speech would have resulted in inaudibility of segments of the overall signal. Thus, it is possible that for conditions of severe limiting, a certain level may be reached where the listener chooses to listen to a somewhat distorted audible signal, rather than to a partially inaudible non-distorted signal.

Listeners appeared to tolerate greater amounts of clipping with high frequency emphasis than with the flat frequency response. This apparently greater tolerance for clipping with high frequency emphasis may be partly explained by acoustic differences revealed by the calibration procedure.

For the 0 dB/octave slope condition, saturation threshold was determined by the low frequency vowel phonemes which were of relatively high intensity and occurred in every word. In contrast, the saturation threshold for the +6 dB/octave condition was initially determined by individual, high frequency and high intensity fricatives (i.e., /s/, /ʃ/), which occurred relatively infrequently. With the vowels made relatively less intense by the high frequency emphasis slope, the saturation threshold was exceeded at a much slower word per dB rate as gain was increased.

Other factors which may explain the differences in listening levels chosen under the 0 dB/octave and +6 dB/octave conditions may include a listener's intent to maintain audibility of a speech signal deprived of much of its power, as well as the fact that with high frequency emphasis many of the distortion products fell beyond the upper bandwidth limits of the system, making them inaudible.

Gain adjustments by the listener may have represented a compromise between audibility, intelligibility, perceived quality and comfort for the individual listener. While the emphasis on intelligibility when instructing listeners implied that reductions in listener gain induced by changes in the degree of limiting should have principally been motivated by intelligibility factors, early research on the effects of amplitude distortion on speech intelligibility (Kryter, et al., 1947; Davis, et al., 1947) indicated that qualitative effects secondary to peak clipping were perceived at degrees of clipping which were not enough to affect intelligibility. The possibility therefore existed that listeners reduced their gain to avoid these qualitative effects imposed by peak clipping. The use of presentation levels in addition to PLL's, as discussed in the section on speech intelligibility performance (p. 138), was helpful in differentiating between these effects.

Comparison with Related Research

Only one study reported in the literature is known to have explored the relationship between output limiting in the form of peak clipping and a listener's gain adjustments. Martin (1976) studied the effects of output limiting on the comfort level adjustments of a group of hearing impaired listeners with moderate-to-severe degrees of sensorineural hearing impairment. His system was described as having a flat frequency response with output limiting in the form of peak clipping varying from approximately 95-130 dB SSPL. Continuous discourse against a quiet background was used as the speech material and listeners were instructed to adjust the gain for comfort and not to pay attention to clarity.

Martin did not provide any statistical analysis, or much detail concerning his data. However, his individual performance functions and conclusion were generally consistent with those measured in this investigation. Highest listener adjusted gain levels were indicated for the highest limiting level, with evidence of trends towards a lowering of individual gain adjustments as the limiting level was decreased. Martin reported that, as the output limiting level was progressively decreased, listeners reduced their gain until a critical level was reached, after which listeners increased their gain in order to maintain a comfortable listening level.

This study differed from Martin's in four important aspects: In this investigation a more homogeneous group of subjects was used. In addition, listening conditions characterized by competing noise were included, listeners were instructed to emphasize intelligibility in their adjustments, and monosyllabic word lists were used as speech materials. These factors may have contributed to the more consistent, individually observed trends and statistically significant group reductions in gain observed in this investigation. The observation made by Martin regarding the existence of a critical point where the listener attempted to increase his/her gain as limiting was increased, is consistent with the suggestion made earlier that, as limiting becomes increasingly severe, listeners will tolerate clipping in order to maintain audibility. With Martin's instructions emphasizing comfortable loudness, listeners were more likely to increase the gain at a certain degree of limiting in order to increase the power of the speech (Wathen-Dunn & Lipke, 1958), which is associated with the perception of the loudness of speech (Davis, et al., 1947).

PLL as a Listening Level Concept

Introduction of the PLL concept in this study was motivated by research indicating that comfort level is a range of levels, the upper limits of which may be more closely associated with a listener's maximum speech intelligibility performance level (Posner & Ventry, 1977; Ventry & Johnson, 1978). Thus, it was expected that use of the PLL concept would more closely approximate an individual's maximum intelligibility performance within his/her comfort range. This level was believed to more validly represent user gain levels under those conditions where speech intelligibility is thought to be more important. Higher absolute listening levels were anticipated using the PLL instructions and speech material than those reported using other instructions and/or other materials (Posner & Ventry, 1977). Valid representation of use gain levels was considered a goal essential to the accurate assessment of the potential effects of output limiting on listener performance.

In an effort to validate the PLL concept as applied in this study, three comparisons were undertaken. First, an estimate of the absolute PLL's measured in this study were compared with studies of comfort level ranges and intelligibility performance functions reported in the literature for similar subject samples. Second, an estimate of gain levels used by subjects in this study was derived and compared with estimates of gain levels for experienced hearing aid users reported in other studies. Finally, the intra-subject reliability of the PLL concept was evaluated and compared with the reliability of similar measurements as reported in the research literature.

In undertaking these comparisons it is clear that experimental procedures and test materials used in other studies often differed from those used in this study. In particular, Dirks, et al. (1977) have pointed out the problems inherent in the calibration of speech levels using a VU meter, as a limiting factor in making inter-study and inter-stimulus comparisons. However, despite these limitations, general conclusions or impressions can be drawn in comparisons with other studies which are helpful in evaluating the PLL concept as representative of the real-life gain adjustment of a hearing aid user.

PLL and the Comfortable Listening Range

In order to provide a reference PLL level in dB SPL with which other speech listening level data based on VU meter calibrations could be compared, mean PLL's in dB attenuation for the unlimited (120 dB SSPL) flat frequency response condition in both a quiet and noise background were converted into dB SPL for the subject group (see Appendix F). Mean PLL's across test sessions for these two conditions were virtually identical. Using a 13 dB ($\pm$ 1 dB) peak factor of speech, defined as the dB difference between instantaneous speech peaks measured on a VU meter and the maximum instantaneous voltage of the highest peaks of speech, a PLL estimate of 101 dB SPL ($\pm$ 1 dB) (σ = 3.5 dB) was derived.

Two investigations (Ventry & Johnson, 1978; Kamm, et al., 1978) have recently been reported which have defined a comfortable listening range for listeners with sensorineural hearing impairment. Both investigations used recorded lists of spondaic words as the speech material. However, different psychophysical methods were used for measuring comfortable listening level. Both descending and ascending approach modes were used in each study. Ventry and Johnson (1978) described a comfortable loudness range for three degrees of sensorineural hearing impairment. The 54 dB HTL mean pure tone average and 45.5 dB mean SRT for the test ear in this investigation places the group of subjects in this study between the categories of moderate (mean PTA = 46.8 dB HTL) and severe (mean PTA = 59.8 dB HTL) defined by Ventry and Johnson. They reported mean comfort level limits of 83 dB SPL and 103 dB SPL for their moderately impaired group and mean comfort level limits of 91 dB SPL and 107 dB SPL for their severely impaired group. Interpolation of Ventry and Johnson's data for the degree of hearing impairment measured in this study would estimate the 101 dB SPL value for PLL measured in this study to be within the upper portion of the range defined by their upper and lower limits of comfortable listening.

Kamm, et al. (1978) studied MCL and LDL functions for varying degrees of cochlear hearing impairment using an adaptive procedure. By varying their adaptive strategies, upper and lower limits of an MCL range were defined. Using a non-linear regression technique, they plotted a best-fit line for these two MCL's in dB SPL as a function of speech reception threshold (p. 676, Figure 2). Extrapolating from their data, an upper limit MCL of approximately 96 dB SPL and a lower limit of approximately 86 dB SPL was found to be associated with a 45 dB SRT. Clearly, the 101 dB SPL estimated as the mean PLL in this study, closely approximates the higher of the two MCL values. These inter-study comparisons suggest that PLL instructions, emphasizing intelligibility, result in listening levels close to the upper limits of the listener's comfort range.

PLL and PB-Max

As reviewed in Chapter 2, no one has investigated the relationship between PB-max and the comfortable loudness range. Ventry and Johnson (1978), based on their own limited findings and a review of research investigations which studied intelligibility performance function in sensorineural hearing impairment, suggested that PB-max may closely correspond with their descending MCL for speech, defining an upper limit of comfortable loudness.

Posner and Ventry (1977) investigated the relationship between speech presentation levels based on comfortable loudness levels for continuous discourse speech materials and monosyllabic speech discrimination in sensorineural hearing impairment, using two sets of instructions. One set of instructions emphasized comfort and another emphasized comprehension. Their experimental group was described as having a mean PTA of 49.2 dB HTL and a mean SRT of 42.6 dB HTL. The mean level at which PB-max was reached for the group was 37.2 dB above the mean SRT. Both instructional sets resulted in listening levels which significantly underestimated the mean level at which PB-max was measured. An admitted shortcoming of their study was the use of continuous discourse material for listening level judgments. In the present study, the 101 dB SPL (81 dB HTL) value for PLL was approximately 36.5 dB above the SRT, approximating the mean sensation level at which PB-max was measured in the Posner and Ventry study. These data support the hypothesis that PLL results in a listening level which maximizes intelligibility performance.

PLL and Gain Adjustments of
Experienced Hearing Aid Users

Results of research investigations reporting aided gain levels of experienced hearing aid users (Brooks, 1973; Byrne & Tonnison, 1976; Berger, et al., 1980) have generally concurred with the original one-half gain rule first recommended by Lybarger (1963). This rule states that aided gain changes at a rate of approximately half the hearing loss, measured in dB HTL. In these investigations the degree of hearing loss has been determined by one or a combination (average) of frequencies used in determining the pure tone average (.5, 1 and 2 KHz).

In addition to being determined by the extent of hearing impairment, aided gain is also determined by the level of the input speech signal. Many recent hearing aid studies have recommended and/or used levels approximating a 70 dB SPL input (Gengel, et al., 1971; Byrne & Tonnison, 1976; Reid, et al., 1977; Walden, et al., 1977) as representative of the average speech level encountered by a hearing impaired subject in everyday listening.

With the system used in the present study, input speech level was not a factor. However, the amount of gain that would be used by the group in listening to conversation can be estimated by subtracting 70 dB SPL from the actual listening level. Assuming such an input level, the reference PLL of 101 dB SPL would be obtained using a gain of 31 dB. This amount of gain is at a ratio of .57 to the group three-frequency pure tone average of the test ear. This ratio is consistent with those reported by other researchers using experienced hearing aid users.

The overall conclusion based on the comparison of the reference PLL level with data reported from other related investigations is consistent with the notion that PLL is representative of a use gain setting, while also approximating the level at which maximum speech intelligibility performance will be achieved. These conclusions are consistent with the findings reported by Yantis, et al.

(1966), who found that the gain adjustment of experienced hearing aid users, as adjusted in their own environment, closely corresponded with the level at which maximum intelligibility performance was measured.

Two important factors which distinguish the present investigation from the Posner and Ventry study, which also used instructions emphasizing intelligibility, include a subject selection criterion of hearing aid experience and the use of monosyllabic words as listening materials. Either one of these factors could have contributed to the higher levels measured in the present study. Future research should evaluate the PLL method used here on listeners without hearing aid experience in order to determine the contribution of experience as a factor. In addition, the relationship of PLL to PB-max should be ascertained.

PLL Reliability

A prerequisite in establishing the usefulness of any clinical measurement is a measure of its variability. If the PLL's measured in this study were not reliable, it would have affected: a) its sensitivity to any effects of output limiting and b) its clinical utility on an individual basis as a means of setting hearing aid gain for differentiating among aided conditions. The analysis of variance revealed an unexpected, significant time-order effect. However, only 1 dB separated the means of the

PLL's measured in the two test sessions. Inspection of the raw data revealed that these differences were largely attributed to reliability problems involving two subjects. The 1 dB test-retest difference was considered small enough so as not to have practical significance.

Intra-subject reliability (Table 8, p. 96) was evaluated in two ways. In the first method, the standard deviation of the time-order effect was calculated based on the total variance of the time-order effect and its interactions. The standard deviation of test-retest error for PLL determined in this manner was 3.14 dB. Thus, based on this method, 68% of test-retest PLL's were within $\pm$3.14 dB of each other and 95% within 6.28 dB.

In the second method, the proportion of individual test-retest measurements differing by a specified number of dB was calculated. These results revealed that 67% of repeated trials were within 4 dB of each other, 88% within 6 dB of each other and 98% of the replications were within 8 dB of each other. An evaluation of these test-retest differences for each frequency response and background noise condition revealed consistency with the overall result. The significance of test-retest differences can only be evaluated by determining the amount of difference that these amounts are likely to make in an individual's intelligibility performance variations.

A comparison of test-retest differences of this investigation and those reported by other investigators for hearing impaired listeners using speech materials, revealed essentially good agreement. Ventry and Johnson (1978) reported test-retest differences within 5 dB for 78% of their subjects for ascending MCL, while 94% of the subjects had test-retest MCL's within 5 dB for descending approaches.

In studies involving the repeatability of hearing aid gain adjustments, Walden, et al. (1977) reported that 90% of their test-retest comparisons of aided gain were within 8.5 dB, while 50% of the test-retest comparisons were within 4 dB. Reid, et al. (1977), in their study of hearing aid gain settings in experienced hearing aid users, found aided gain levels for test and retest to be within 4.4 dB for approximately 88% of their subjects. The present PLL reliability data are comparable to data generated in these other investigations.

The overall conclusions concerning PLL based on related research support its application as a hearing aid gain setting approach. PLL emerges as a reliable measure near the upper limits of an individual's comfort level range. This level closely corresponds to the level necessary for an individual's maximum intelligibility performance. Further research evaluating the PLL approach should: a) include non-experienced hearing aid users, b) measure the relationship between PLL and PB-max on the same subjects, and c) evaluate the implications of test-retest variability on intelligibility performance. In addition, this approach should be evaluated for other listening conditions.

Speech Intelligibility Performance

The second stage of the present study evaluated effects of output limiting on speech intelligibility. For each combination of output limiting, frequency response slope and background noise, speech intelligibility using monosyllabic word lists was evaluated at each individual's previously adjusted PLL and two additional levels (see Chapter 3, p. 81). These additional presentation levels provided data relevant to a general consideration of the effects of peak clipping as a form of amplitude distortion on speech intelligibility. Data provided by this experimental design were also useful in evaluating listeners' strategies when adjusting their PLL's under varying conditions of limiting, and in determining the overall implications of output limiting in the form of peak clipping for hearing aid users as well.

Effects of Output Limiting on Intelligibility Performance at PLL

An analysis of variance comparing intelligibility scores obtained at the group's PLL for the three levels of output limiting revealed no significant differences between intelligibility performance. This result suggested that, for the conditions studied, there was no measurable effect of output limiting on intelligibility when listeners listened at their self-adjusted PLL (gain) levels.

Effects of Limiting Levels on Speech Intelligibility

Results of two separate statistical analyses of intelligibility data suggested deleterious effects associated with the lowest output limiting level, affecting speech intelligibility performance in background noise.

In one analysis the effects of level changes at each SSPL were analyzed. This analysis revealed a trend towards poorer intelligibility performance at increased gain, as the output limiting level was decreased in background noise. A second analysis involved the effects of SSPL on overall intelligibility performance. This analysis revealed a trend toward poorer intelligibility performance in background noise at increased gain settings for the lowest output limiting level.

Linear best-fit functions of the individual normalized data revealed that as the output limiting level was decreased, intelligibility was reduced as gain was increased for the background noise conditions. This effect was greater for the +6 dB/octave slope condition than for the 0 dB/octave slope condition. These analyses indicate that the reduction in gain made by the group under the severe limiting condition was a strategy that helped preserve intelligibility.

Another finding to emerge from these analyses suggested a possible frequency response slope interaction with output limiting. As discussed previously, the analysis of intelligibility performance at PLL suggested that performance with high frequency emphasis in background noise might have been more adversely affected than the flat condition by decreased output limiting levels. Intelligibility performance as a function of listening level and limiting further supported this trend by indicating a steeper roll-off in discrimination performance with increasing level in the high frequency emphasis condition.

Overall findings based on speech intelligibility performance under conditions of varying output limiting level suggest that, as output limiting is decreased, performance is more likely to be reduced with increased presentation levels which exceed PLL and/or the saturation threshold in background noise. This effect was more evident with high frequency emphasis. However, for quiet listening conditions or for unlimited conditions (i.e., high output limiting levels) maximum intelligibility appears to be more effectively preserved over a broader range of presentation levels. It is possible that, with intelligibility performance being very near 100%, these conditions may have been characterized by excessive redundancy which obscured an effect.

Comparison with Previous Research

Unlike research studies which examined the effects of controlled amounts of peak clipping on speech intelligibility, primary emphasis in this study was in controlling the peak clipping output limiting level in an otherwise linear system. This resulted in listening conditions characterized by varying amounts of peak clipping depending on the listening level chosen. This made direct comparisons with earlier studies difficult. However, the resultant effects appeared consistent with the early findings of Licklider and his co-workers.

Licklider, in 1946, summarized the results of a series of investigations on the effects of amplitude distortion on speech intelligibility. He concluded that, in a quiet background, normal listeners demonstrated a high degree of tolerance for peak clipping before intelligibility was significantly affected. This tolerance was enhanced when high frequency emphasis preceded the clipper. In contrast, when noise entered the system at a point ahead of the clipper, an unfavorable interaction resulted, degrading speech intelligibility. This interaction was reportedly more adversely affected by high frequency emphasis ahead of the clipper.

Kryter, et al. (1947) explained the adverse interaction between noise and clipping affecting speech intelligibility as resulting from the low frequency noise acting as a carrier for the speech wave, carrying it beyond the amplitude limits at which the clipper acts. This causes a loss of part of the structural detail of the speech wave. A secondary explanation for the loss of speech intelligibility offered by these authors was based on the intermodulation products generated by the clipping process, which they stated, are essentially unintelligible. These distortion products tend to mask what is left of the speech after clipping. It should be noted, however, that these authors did not differentiate between frequency response slope effects in their explanation of this adverse interaction between clipping and background noise.

Clinical Implications

While the generalizability of findings reported in this investigation are necessarily limited to conditions and procedures used, a number of clinical implications are indicated. Current recommendations for hearing aid characteristics appearing in the literature (see Chapter 2) suggest the use of increased high frequency emphasis, high gain and low saturation sound pressure levels. With these recommendations, the potential exists for adverse interactions of the type revealed in the present study. These could affect speech intelligibility performance under conditions characterized by background noise when peak clipping is used as a form of limiting. While it is possible that the listener can compensate for these effects to a certain extent by reducing his/her gain, the overall effect would be a narrowing of the input operating range over which adequate speech intelligibility can be maintained. Thus, even if intelligibility can be maintained by the listener under some conditions, deleterious effects are likely to occur as a result of rapidly varying speech levels characterizing conversational speech. Therefore, for individuals whose low LDL's require severe output limiting, if the listener's unlimited preferred listening level for speech intelligibility has peaks which exceed the saturation threshold, compression forms of output limiting perhaps should be considered. This recommendation is the same as that previously made in the Harvard hearing aid study (Davis, et al., 1947).

In the present study, intelligibility performance was used as the criterion measure for evaluating adverse effects of output limiting. Significant effects were seen only in background noise. However, the reduced PLL's seen in quiet in this study suggest the possibility of user dissatisfaction with low output limiting levels employing peak clipping, even under quiet listening conditions. These effects may not influence the intelligibility of adequately high level speech signals measurably, but could adversely affect the listener's performance in real-world situations by reducing the intelligibility of low level speech signal inputs.

Suggestions for Further Research

Additional research is needed to clarify some of the issues raised by this investigation. Areas in need of investigation include conditions of high frequency emphasis and different signal-to-noise ratios in order to further clarify the frequency and background effects identified in the present study. In addition, the level or degree of clipping at which speech intelligibility is first affected should be explored by investigating the listener's dynamic range of hearing with fixed intensity level presentations below and above the clipping threshold. In this manner, a more accurate determination could be made of the appropriateness of a conventional peak clipping hearing aid for a given individual with particular gain and saturation requirements.

Speech discrimination tests more sensitive to high frequency effects, such as the nonsense syllable test used by Levitt, et al. (1978), may be more useful in understanding the mechanism by which speech intelligibility performance is affected.

Finally, a conclusion based on results of the present study suggests that the efficacy of compression as an alternative to peak clipping forms of limiting should be investigated. Initial indications from the Harvard hearing aid study suggested that compression limiting was superior to peak clipping for their hearing impaired population. This is in need of experimental verification with the hearing impaired populations now being considered for hearing aid amplification. This is particularly true when considering the more complex signal processing systems now being developed or proposed for individuals with sensorineural hearing loss (Villchur, et al., 1973; Abramovitz, 1979; Lippman, et al., 1980).

Conclusions

The overall conclusions of the present study were that for the hearing-impaired listeners as a group:

1) unlimited speech peaks of PLL's obtained with a high output limiting level (120 dB SSPL) always exceeded the saturation threshold of the lowest output limiting level (100 dB SSPL) ;
2) PLL gain adjustments with the lowest output limiting level were reduced relative to those with higher output limiting levels, principally in background noise (+7 dB S/N);
3) PLL is a reliable measure closely corresponding to the upper limits of the comfort level range associated with PB-max and representative of hearing aid user gain adjustments;
4) reduced output limiting levels appeared to adversely affect speech intelligibility performance under conditions of background noise, especially for high frequency emphasis;
5) PLL reduction was an appropriate strategy for preserving speech intelligibility with low output limiting levels.

However, substantial individual differences were observed which necessarily limit the generalization of the group results to the individual listener.

APPENDICES

APPENDIX A

Individual Subject Audiometric Test Results

Table 17. Pure tone thresholds (dB HTL re: ANSI 1969), for individual subjects, test and non-test ears, as measured in the subject orientation session.

SUBJECT			FREQUENCY (Hz) 250	500	1000	2000	4000	8000
#1	*R	(A)	25	45	55	60	75	70
		(B)		40		45		
	L	(A)	15	20	35	50	65	80
		(B)	20	25	45	45	65	
#2	*R	(A)	55	60	65	70	70	85
		(B)	NR	NR	65			
	L	(A)	35	35	50	50	55	60
		(B)	25	35	50	60	60	
#3	R	(A)	15	35	80	80	80	65
		(B)						
	*L	(A)	10	30	65	55	65	65
		(B)	5	20	NR	60	60	
#4	R	(A)	40	55	65	80	95	NR
		(B)						
	*L	(A)	35	45	60	80	95	NR
		(B)	40	45	65	NR	NR	
#5	R	(A)	15	25	45	45	60	50
		(B)	10	35	40	50	60	
	*L	(A)	20	35	50	50	60	60
#6	R	(A)	45	50	75	55	45	20
		(B)						
	*L	(A)	35	50	60	50	25	5
		(B)	35	40	55	50	30	
#7	R	(A)	35	50	60	65	85	80
		(B)						
	*L	(A)	40	50	60	70	65	65
		(B)	45	60	65	75	70	

Appendix A--continued

SUBJECT			FREQUENCY (Hz) 250	500	1000	2000	4000	8000
#8	R	(A)	30	50	55	55	50	50
		(B)	40	NR	55	60	50	
	*L	(A)	45	50	60	55	50	60
		(B)						
#9	*R	(A)	40	40	50	60	75	95
		(B)	NR	50	55	60	NR	
	L	(A)	55	50	50	65	75	70
		(B)						
#10	R	(A)	25	35	55	55	60	50
		(B)						
	*L	(A)	15	35	55	60	55	55
		(B)	35	50	60	70	65	

<u>KEY</u>

R: Right Ear
L: Left Ear
(A): Air Conduction
(B): Bone Conduction
*: Test Ear
DNT: Did Not Test
NR: No Response

Appendix A--continued

Table 18. Speech reception thresholds (SRT) (dB HTL re: ANSI 1969) and speech discrimination scores (%), for individual subjects, test and non-test ears, as measured by the clinical referral source.

SUBJECT		SRT (dB HTL)	SDS (%)
#1	*R	45	60
	L	35	44
#2	*R	55	96
	L	35	80
#3	R	35	24
	*L	40	60
#4	R	50	48
	*L	50	36
#5	R	40	95
	*L	40	95
#6	R	55	80
	*L	40	88
#7	R	50	84
	*L	50	84
#8	R	45	96
	*L	50	92
#9	*R	40	56
	L	55	60
#10	R	50	88
	*L	45	88

KEY

R: Right Ear
L: Left Ear
*: Test Ear

APPENDIX B

Individual Subject Loudness Discomfort Levels

Table 19. Pure tone loudness discomfort levels (dB SPL) of individual subject's (test ear) for four frequencies, test (T) and retest (R).

	Frequency, Hz							
	500		1000		2000		4000	
S#	T	R	T	R	T	R	T	R
1	115	>122	116	>123	109	120	112	>122
2	118	112	113	111	>122	118	106	113
3	105	109	108	109	106	110	108	112
4	>122	>122	120	>123	>122	>122	>122	>122
5	>122	DNT	>123	DNT	>122	DNT	>122	DNT
6	120	114	118	115	113	107	115	102
7	>122	>122	118	122	113	>122	>122	>122
8	115	102	117	102	>122	107	>122	103
9	109	110	107	109	108	112	112	120
10	105	107	102	109	103	103	102	109

<u>KEY</u>

T: test
R: retest
DNT: did not test

APPENDIX C

Intelligibility Test Lists (NU#6)

List 2A

1. pick	11. dab	21. ton	31. hate	41. lore
2. room	12. numb	22. keg	32. live	42. bite
3. nice	13. juice	23. calm	33. book	43. haze
4. said	14. chief	24. tool	34. voice	44. match
5. fail	15. merge	25. pike	35. gaze	45. learn
6. south	16. wag	26. mill	36. pad	46. shawl
7. white	17. rain	27. hush	37. thought	47. deep
8. keep	18. witch	28. shack	38. bought	48. gin
9. dead	19. soap	29. read	39. turn	49. goal
10. loaf	20. young	30. rot	40. chair	50. far

List 3A

1. base	11. pearl	21. road	31. rat	41. mouse
2. mess	12. search	22. shall	32. void	42. hire
3. cause	13. ditch	23. late	33. wire	43. cab
4. mop	14. talk	24. cheek	34. half	44. hit
5. good	15. ring	25. beg	35. note	45. chat
6. luck	16. germ	26. gun	36. when	46. phone
7. walk	17. life	27. jug	37. name	47. soup
8. youth	18. team	28. sheep	38. thin	48. dodge
9. pain	19. lid	29. five	39. tell	49. seize
10. date	20. pole	30. rush	40. bar	50. cool

List 4A

1. pass	11. sail	21. lease	31. came	41. lose
2. doll	12. yearn	22. long	32. fit	42. near
3. back	13. wife	23. chain	33. make	43. perch
4. red	14. such	24. kill	34. vote	44. shirt
5. wash	15. neat	25. hole	35. judge	45. bath
6. sour	16. peg	26. lean	36. food	46. time
7. bone	17. mob	27. tape	37. ripe	47. hall
8. get	18. gas	28. tire	38. have	48. mood
9. wheat	19. check	29. dip	39. rough	49. dog
10. thumb	20. join	30. rose	40. kick	50. should

Appendix C--continued

List 2B

1. live	11. calm	21. fail	31. juice	41. soap
2. voice	12. book	22. said	32. keg	42. hate
3. ton	13. dab	23. wag	33. gin	43. turn
4. learn	14. loaf	24. haze	34. nice	44. rain
5. match	15. goal	25. white	35. numb	45. shawl
6. chair	16. shack	26. hush	36. chief	46. bought
7. deep	17. far	27. dead	37. gaze	47. thought
8. pike	18. witch	28. pad	38. young	48. bite
9. room	19. rot	29. mill	39. keep	49. lore
10. read	20. pick	30. merge	40. tool	50. south

List 3B

1. sheep	11. rush	21. life	31. cool	41. when
2. cause	12. five	22. pain	32. seize	42. ring
3. rat	13. team	23. base	33. dodge	43. cheek
4. bar	14. pearl	24. mop	34. youth	44. note
5. mouse	15. soup	25. mess	35. hit	45. gun
6. talk	16. half	26. germ	36. late	46. beg
7. hire	17. chat	27. thin	37. jug	47. void
8. search	18. road	28. name	38. wire	48. shall
9. luck	19. pole	29. ditch	39. walk	49. lid
10. cab	20. phone	30. tell	40. date	50. good

List 4B

1. rose	11. join	21. kill	31. perch	41. hole
2. dog	12. check	22. fit	32. chain	42. gas
3. time	13. wheat	23. judge	33. make	43. came
4. such	14. thumb	24. should	34. long	44. vote
5. have	15. near	25. pass	35. wash	45. lean
6. mob	16. lease	26. back	36. food	46. red
7. bone	17. yearn	27. hall	37. mood	47. doll
8. sail	18. kick	28. bath	38. neat	48. shirt
9. rough	19. get	29. tire	39. tape	49. sour
10. dip	20. lose	30. peg	40. ripe	50. wife

Appendix C--continued

List 2C

1. dead	11. nice	21. gin	31. haze	41. chair
2. juice	12. bought	22. pad	32. pick	42. chief
3. merge	13. ton	23. gaze	33. turn	43. keg
4. young	14. shawl	24. live	34. goal	44. soap
5. calm	15. white	25. room	35. voice	45. said
6. bite	16. hate	26. south	36. keep	46. dab
7. rain	17. shack	27. mill	37. thought	47. wag
8. match	18. pike	28. witch	38. far	48. deep
9. book	19. fail	29. tool	39. read	49. learn
10. loaf	20. rot	30. numb	40. hush	50. lore

List 3C

1. mop	11. late	21. pole	31. cool	41. team
2. tell	12. when	22. life	32. five	42. soup
3. germ	13. gun	23. date	33. rush	43. dodge
4. seize	14. name	24. road	34. phone	44. chat
5. good	15. pain	25. talk	35. mouse	45. sheep
6. base	16. ditch	26. cause	36. thin	46. note
7. search	17. rat	27. youth	37. pearl	47. void
8. ring	18. hire	28. bar	38. beg	48. jug
9. half	19. shall	29. lid	39. walk	49. wire
10. mess	20. cheek	30. hit	40. luck	50. cab

List 4C

1. perch	11. make	21. rose	31. yearn	41. hall
2. bath	12. mob	22. kill	32. lean	42. rough
3. back	13. doll	23. came	33. gas	43. peg
4. bone	14. pass	24. food	34. sail	44. get
5. wife	15. sour	25. lose	35. red	45. chain
6. fit	16. dog	26. should	36. wheat	46. join
7. shirt	17. vote	27. check	37. have	47. ripe
8. wash	18. time	28. kick	38. mood	48. judge
9. neat	19. near	29. long	39. dip	49. hole
10. tire	20. lease	30. tape	40. such	50. thumb

Appendix C--continued

List 2D

1. loaf	11. deep	21. gaze	31. chief	41. read
2. chair	12. match	22. live	32. bite	42. merge
3. goal	13. fail	23. keg	33. nice	43. hate
4. shack	14. said	24. calm	34. room	44. learn
5. far	15. wag	25. turn	35. book	45. soap
6. ton	16. pike	26. lore	36. numb	46. young
7. witch	17. haze	27. thought	37. pad	47. voice
8. rot	18. white	28. tool	38. bought	48. mill
9. dab	19. hush	29. rain	39. south	49. gin
10. pick	20. dead	30. keep	40. juice	50. shawl

List 3D

1. germ	11. sheep	21. soup	31. tell	41. late
2. talk	12. mouse	22. pole	32. jug	42. void
3. walk	13. cheek	23. cab	33. pearl	43. rush
4. ditch	14. bar	24. shall	34. base	44. rat
5. mop	15. phone	25. chat	35. team	45. thin
6. hire	16. beg	26. good	36. wire	46. half
7. youth	17. mess	27. note	37. date	47. name
8. pain	18. life	28. hit	38. road	48. lid
9. gun	19. five	29. when	39. cool	49. ring
10. dodge	20. seize	30. luck	40. search	50. cause

List 4D

1. food	11. hole	21. judge	31. lose	41. kick
2. lease	12. wash	22. gas	32. tape	42. mood
3. hall	13. sail	23. bath	33. make	43. lean
4. rose	14. mob	24. yearn	34. tire	44. dog
5. doll	15. ripe	25. time	35. near	45. peg
6. kill	16. bone	26. wife	36. chain	46. sour
7. join	17. such	27. thumb	37. shirt	47. pass
8. dip	18. check	28. have	38. long	48. vote
9. red	19. wheat	29. neat	39. rough	49. perch
10. came	20. should	30. get	40. fit	50. back

APPENDIX D

Word List Used for Level Adjustments

1. white
2. chief
3. ton
4. shack
5. gaze
6. bite
7. goal
8. hire
9. pearl
10. life
11. name
12. hit
13. ring
14. lid
15. shirt
16. pass
17. rose
18. kick
19. red
20. rough
21. hole
22. deep
23. loaf
24. fail
25. pad
26. numb
27. hate
28. lore
29. search
30. name
31. pole
32. bar
33. mouse
34. soup
35. wire
36. join
37. mob
38. judge
39. have
40. near
41. mood
42. perch

APPENDIX E

Subject Instructions

Instructions for Preferred Listening Level

In this exercise you will hear a man reciting a list of words. You will be hearing these words in one ear. These words will sometimes be presented against a background noise which will interfere to some degree with your ability to understand what the reader is saying. You will be able to adjust the overall loudness of the test signal by rotating the dial labeled "loudness control" on the panel in front of you. I want you to adjust the loudness of the speech or the loudness of the speech and noise so that the speech is as intelligible (clear) as possible, but not so high that it is uncomfortably loud. In reaching this loudness level, be sure to adjust the loudness upward until it is clearly too loud and then downward until it is clearly too soft. Repeat this adjustment as often as you require for reaching a final and decisive judgment. Remember, please adjust the speech level so that it is as intelligible (clear) as possible, but not so high that it is uncomfortably loud. Are there any questions?

Appendix E--continued

Instructions for Word Lists

This exercise involves listening to word lists. These word lists consist of one syllable words. Some lists will be spoken in the presence of background noise. Each test word is preceded by the phrase, "Say the word....". Please write each test word that you hear in the numbered spaces on the form in front of you. Each list consists of 50 words and the form has 50 spaces numbered from (1) to (50). Be sure to keep your place on the form as the words are presented. If a word is presented which you do not understand, draw a short line on the form in the appropriate space and move on to the next space. If you understand only part of the word, write any part of the word that you hear in the appropriate space. Please print your responses. Are there any questions?

Appendix E--continued

Bekesy LDL Instructions

This is a test in which you will be hearing sounds in your right or left ear. We want you to decide when the sound is at a level that you think is uncomfortably loud or unpleasantly loud. By "uncomfortably loud" we mean when the sound is so loud that you would choose not to listen to it for any period of time.

Push the button when the loudness of the pulses attains a level to which you would not listen. Keep the button pushed as long as it is at that level. Release the button when the loudness is below the level to which you would not listen. Are there any questions?

APPENDIX F

Conversion of PLL's from dB Attenuation to Absolute Levels

In this study, speech was calibrated with respect to its peak levels. There are a number of difficulties inherent in estimating the level of a speech signal in dB RMS or in VU. The primary difficulty is related to determining the peak factor of speech (see p. 130). Secondary difficulties encountered in this study were associated with the processing of speech with respect to high frequency emphasis (+6 dB/octave) and/or peak clipping.

Davis, et al. (1947) reported a 12 dB peak factor. with respect to the relationship of peak voltage levels to the VU level for their speech signal, when using a flat frequency response. Other authors have reported a peak factor of as much as 14 dB (Wathen-Dunn & Lipke, 1959) for speech. In this study a 13 dB ($\pm$ 1 dB) peak factor was adopted as an estimate of the peak to VU relationship of speech. This figure was used to convert relative listening levels for the 0 dB/octave, unlimited (i.e., 120 dB SSPL) condition to absolute levels corresponding to an equivalent VU based measurement.

Mean dB attenuation levels for PLL for both the

Appendix F--continued

noise and quiet versions of the flat, unlimited condition were 16.75 dB ($\sigma = 3.7$) and 17.05 dB ($\sigma = 3.5$), respectively. Combining across these conditions resulted in a mean level of 16.9 dB attenuation ($\sigma = 3.5$). The saturation threshold for this condition (i.e., the point where 10% of the test words exceeded the 5% THD of a 1 KHz, 120 dB SPL RMS pure tone) was 5 dB on the attenuator. The absolute peaks of speech, however, when measured in this manner, were 3 dB higher. This means that 8 dB of attenuation on the attenuator dial corresponded to a peak speech level of 123 dB SPL. (3 dB is added to account for the peak factor of the pure tone.) Therefore, 17 dB of attenuation is equal to 114 dB SPL peak level. Thus, assuming a 13 dB ($\pm$ 1 dB) peak factor, an estimated VU level of 101 dB SPL ($\pm$ 1 dB) is derived.

APPENDIX G

Individual Subject Preferred Listening Levels

Table 20. PLL's (dB attenuation) of individual subjects for the 0 dB/octave, quiet background condition, with each of three output limiting levels (dB SSPL), test (T) and retest (R).

S#	T			R		
	dB SSPL			dB SSPL		
	120	110	100	120	110	100
1	18	13	19	20	18	19
2	19	20	23	16	19	18
3	24	21	27	17	20	18
4	20	20	19	20	16	22
5	20	10	23	15	26	20
6	11	11	9	7	10	7
7	18	17	14	15	15	16
8	18	24	20	15	18	22
9	17	20	20	22	18	23
10	8	19	16	15	16	19

Appendix G-continued

Table 21. PLL's (dB attenuation) of individual subjects for the 0 dB/octave, +7 dB S/N condition, with each of three output limiting levels (dB SSPL), test (T) and retest (R).

S#	T			R		
	dB SSPL			dB SSPL		
	120	110	100	120	110	100
1	21	25	23	15	22	22
2	6	9	19	12	12	15
3	23	22	25	19	18	22
4	18	26	22	22	22	28
5	21	13	20*	17	20	23
6	13	13	15	17	13	18
7	23	21	22	15	16	17
8	18	15	24	19	16	14
9	17	14	21	13	11	22
10	17	12	18	15	15	18

* S5 initially chose a PLL of "5" for this condition. This level was later changed to "20", the value which was used as presentation level Z for all but one of the 0 dB/octave, +7 dB S/N intelligibility test conditions.

Appendix G-continued

Table 22. PLL's (dB attenuation) of individual subjects for the +6 dB/octave, quiet background condition, with each of three output limiting levels (dB SSPL), test (T) and retest (R).

S#	T			R		
	dB SSPL			dB SSPL		
	120	110	100	120	110	100
1	18	17	22	12	17	23
2	22	22	20	17	16	12
3	25	28	24	19	20	25
4	15	14	14	11	13	11
5	21	18	19	18	15	20
6	11	11	12	12	8	7
7	17	17	18	14	16	16
8	14	18	10	16	13	12
9	16	18	21	15	17	15
10	18	15	14	13	14	15

Appendix G-continued

Table 23. PLL's (dB attenuation) of individual subjects for the +6 dB/octave, +7 dB S/N condition, with each of three output limiting levels (dB SSPL), test (T) and retest (R).

S#	T			R		
	dB SSPL			dB SSPL		
	120	110	100	120	110	100
1	18	18	14	19	18	19
2	10	12	15	11	7	22
3	27	19	24	19	19	21
4	13	12	14	19	18	20
5	25	22	27	20	14	22
6	15	21	20	18	18	19
7	17	18	21	17	16	13
8	13	13	16	14	18	16
9	17	14	17	14	15	17
10	9	14	16	17	17	14

APPENDIX H

Individual Subject's Intelligibility Performance (%) as a Function of Level (dB attenuation) for all Combinations of Output Limiting (dB SSPL) and Frequency Response Slope (0 dB/octave, +6 dB/octave) in a Quiet Background.

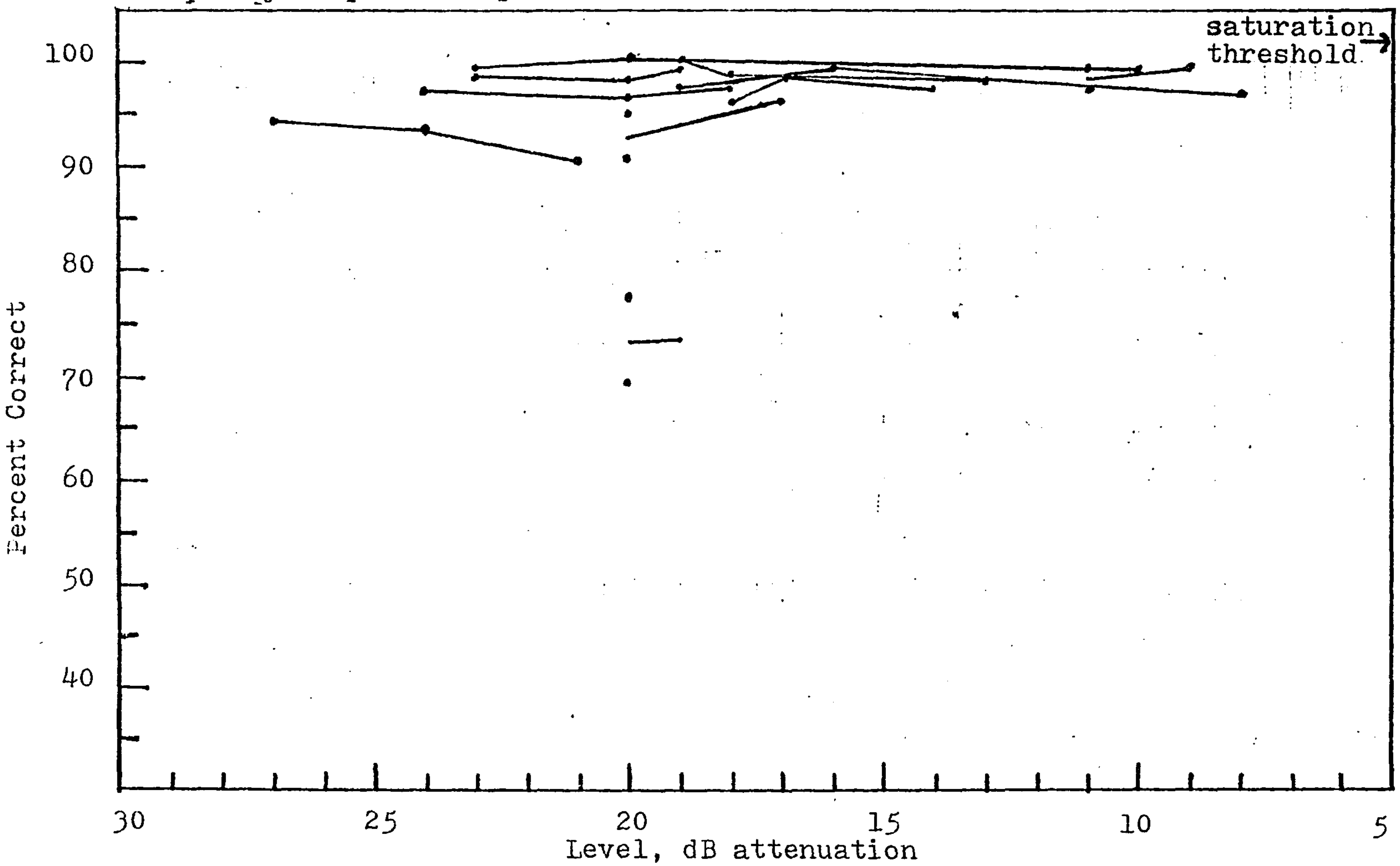

Figure 26. Individual intelligibility performance (%) as a function of presentation level (dB attenuation) for the 0 dB/octave, 120 dB SSPL, quiet condition. The saturation threshold is also given.

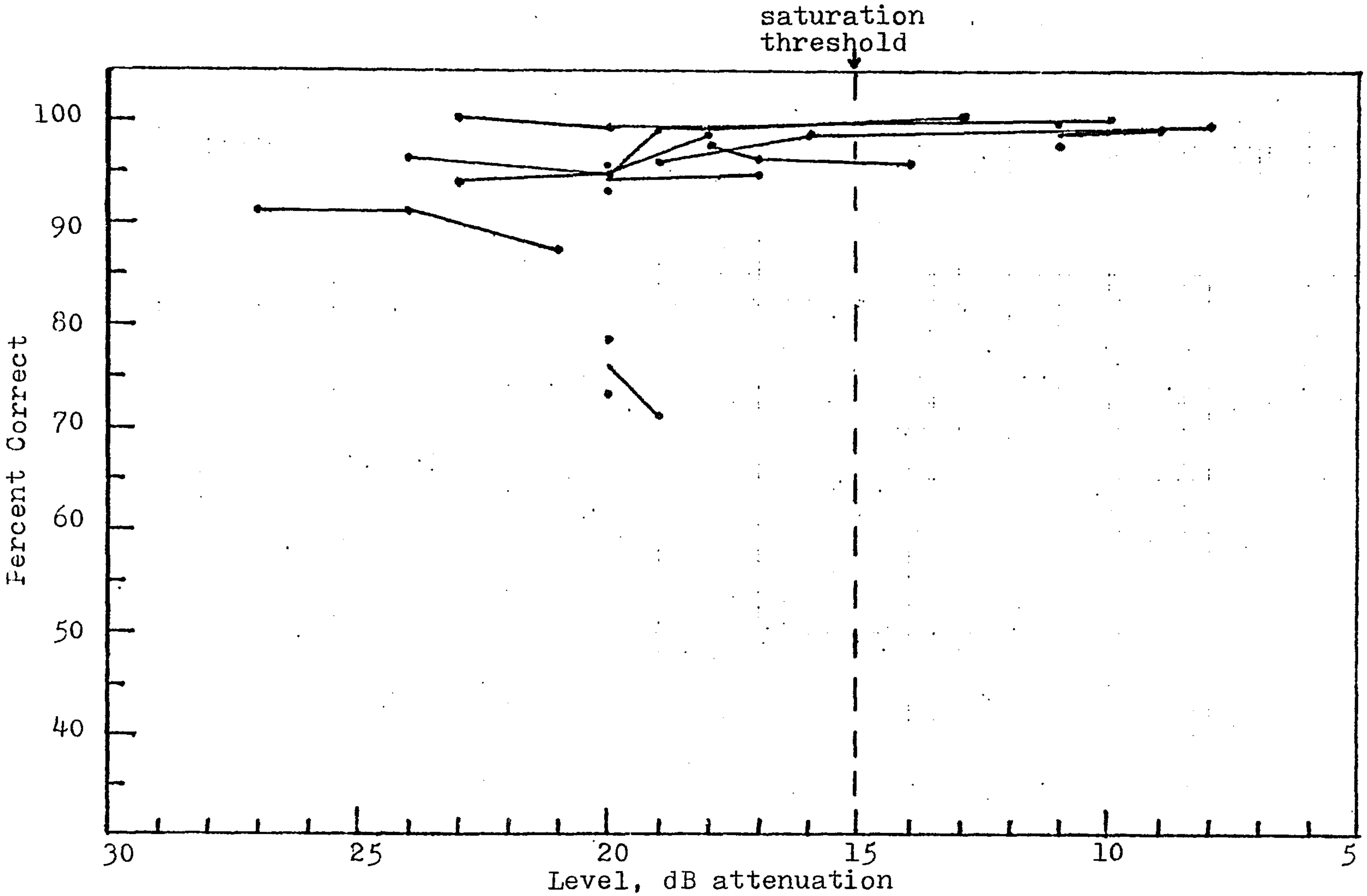

Figure 27. Individual intelligibility performance (%) as a function of presentation level (dB attenuation) for the 0 dB/octave, 110 dB SSPL, quiet condition. The saturation threshold is also given.

Appendix H--continued

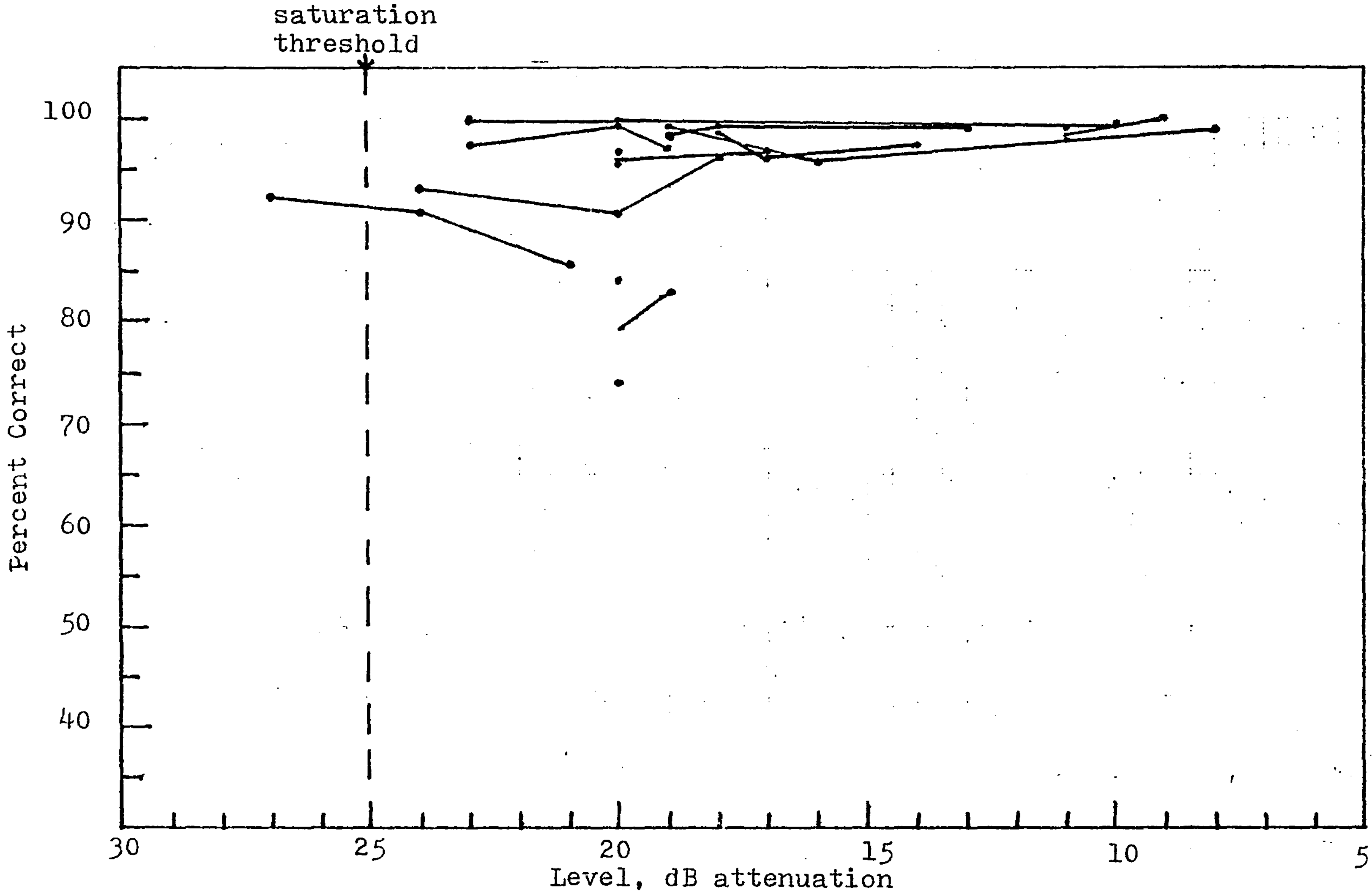

Figure 28. Individual intelligibility performance (%) as a function of presentation level (dB attenuation) for the 0 dB/octave, 100 dB SSPL, quiet condition. The saturation threshold is also given.

Appendix H--continued

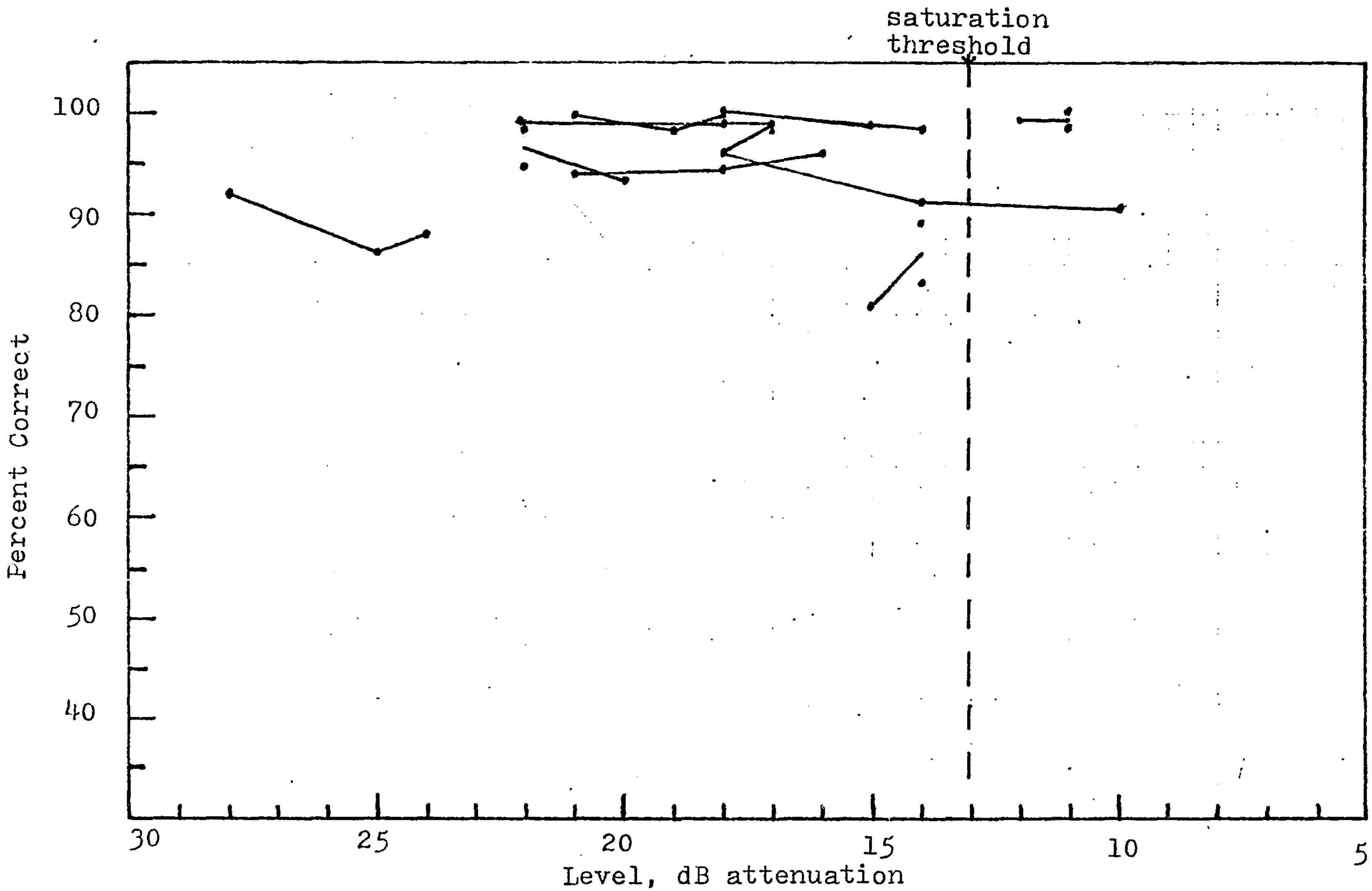

Figure 29. Individual intelligibility performance (%) as a function of presentation level (dB attenuation) for the +6 dB/octave, 120 dB SSPL, quiet condition. The saturation threshold is also given.

Appendix H--continued

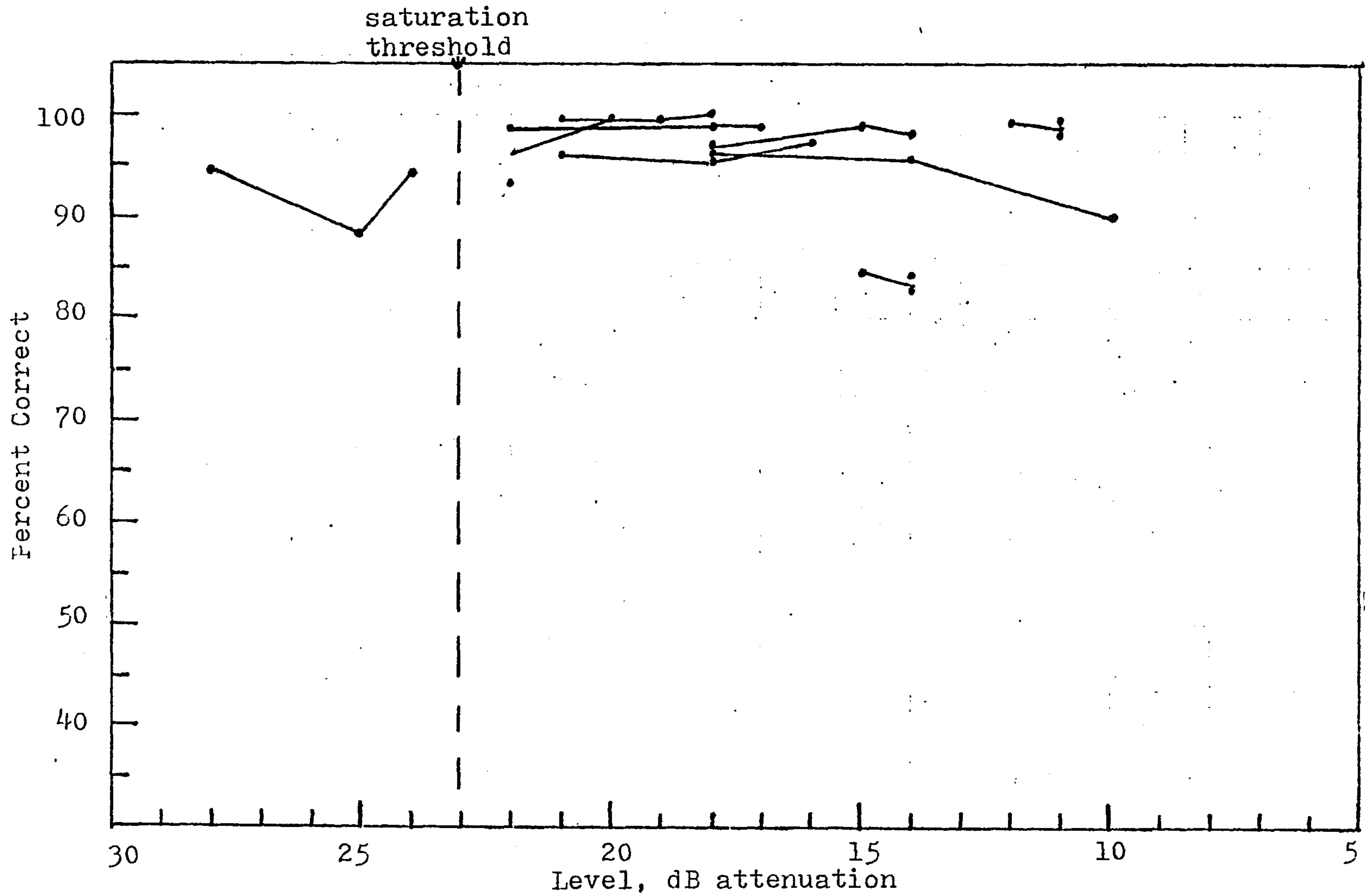

Figure 30. Individual intelligibility performance (%) as a function of presentation level (dB attenuation) for the +6 dB/octave, 110 dB SSPL, quiet condition. The saturation threshold is also given.

Appendix H--continued

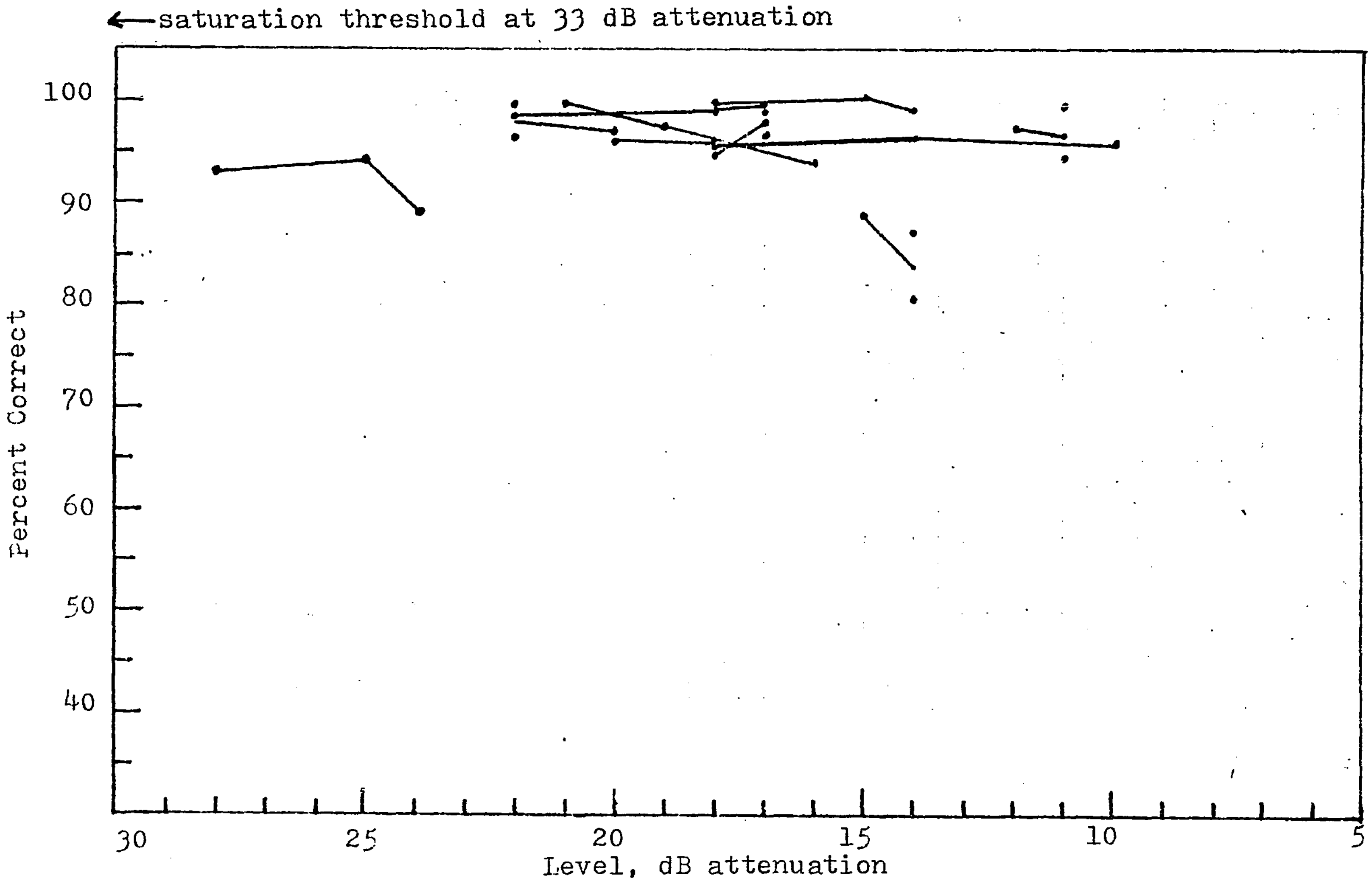

Figure 31. Individual intelligibility performance (%) as a function of presentation level (dB attenuation) for the +6 dB/octave, 100 dB SSPL, quiet condition. The saturation threshold is also given.

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