A COMPARATIVE STUDY
OF
PROPRIETARY AND GENERIC PRESCRIPTIVE PROCEDURES
FOR
NON-LINEAR HEARING AIDS

Register No. A0390001

A Dissertation submitted in part fulflllment of the master's degree (Audiology)
University of Mysore
Mysore

ALL INDIA INSTITUTE OF SPEECH AND HEARING
MANASAGANGOTHRI
MYSORE-570006

MAY- 2005

DEDICATED

TO

ALMIGHTY GOD

&

MY DEAR PARENTS....

CERTIFICATE

This is to certify that the dissertation entitled "A **comparative study of proprietary and generic prescriptive procedures for non-linear hearing aids"** is a bonafide work done in part fulfillment for the degree of **Master of Science (Audiology)** of the student **(Register No. A0390001).**

Prof. M. Jayaram
Director
All India Institute of Speech and Hearing
Manasagangothri
Mysore-570006

Mysore
May, 2005

CERTIFICATE

This is to certify the dissertation entitled **"A comparative study of proprietary and generic prescriptive procedures for non-linear hearing aids"** has been prepared under my supervision and guidance. It is also certified that this dissertation has not been submitted earlier in any other University for the award of any Diploma or Degree.

Manjula.P

Manjula P.
Guide
Lecturer in Audiology
All India Institute of Speech and Hearing
Manasagangothri
Mysore-570006

Mysore
May, 2005

DECLARATION

This dissertation entitled **"A comparative study of proprietary and generic prescriptive procedures for non-linear hearing aids"** is the result of my own study under the guidance of Ms. Manjula P., Lecturer in Audiology; All India Institute of Speech and Hearing, Mysore and has not been submitted earlier in any other university for the award of any Diploma or Degree.

Mysore

May, 2005

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CHAPTER1

INTRODUCTION

INTRODUCTION

Advancement in technology, has led to substantial research in the area of aural rehabilitation and surely, hearing aids are no exception. Various non-linear hearing instruments are now available with complete digital technology. These non-linear hearing aids provide flexible adjustments to meet the desired amplification requirements of hearing impaired individuals. As individuals with sensorineural hearing loss experience an abnormal growth of loudness perception with the increase in input levels, these devices offer an excellent solution for their problem. They provide relatively more amplification for soft sounds and less amplification for loud sounds without manipulation of the volume control switch manually. This is consistent with the two published position statements (American Speech-Language-Hearing Association, 1998; The Pediatric Working Group of the Conference on Amplification for Children With Auditory Deficits, 1996) that demand amplification should provide audibility and comfort for all the input levels without being uncomfortably loud at high levels.

Various prescriptive procedures are used to achieve fitting targets for linear hearing instruments. Generally, linear hearing aids use fitting strategy in which average speech is amplified to a comfortable level, and all other levels of input receive the same amount of gain until the saturation point of amplifier is reached. Because of the loss in the non-linear system (cochlea) the gain characteristics provided by the linear device do not adequately fulfill the requirements of impaired auditory system. This will compromise the audibility of soft input signals and possibly comfort for loud input signals (Jenstad, Pumford, Seewald & Cornelisse, 2000).

Unlike linear hearing aids, non-linear prescription aims at prescribing gain-frequency response for several input levels. As most of non-linear instrument available in market have more than one channel, it is necessary to specify the 1-0 curve for several frequencies at different input levels. 1-0 curve helps in prescribing compression ratios and compression thresholds effectively (Dillon, 2001). Linear fitting procedures, however, provide an indirect measure to verify the performance of non-linear procedures at average input levels by comparing the gain prescribed by non-linear fitting procedures. In contrast, this is not true for low and high input levels and needs further investigation (Dillon, 2001).

Prescriptive procedures for nonlinear hearing aids are based upon different underlying rationales. The idea behind these procedures is either to normalize loudness so that loudness recruitment can be compensated or to maximize speech intelligibility at various input levels (Byrne, 1996). Some of these fitting procedures use threshold and some of others use supra threshold measurements (such as loudness scaling) as input data (Smeds, 2004). Few of these fitting procedures, also known as generic fitting formulae, are applicable to all types or at least a range of aids of roughly the same type. Fitting formulae of this group include LGOB (Allen, Hall, & Jeng, 1990), FIG6 (Killon & Fikret-Pasa, 1993), IHAFF (Cox, 1995), RAB (Ricketts, 1996), ScalAdapt (Kiessling, Schubert, & Archut, 1996), DSL [i/o] (Cornelisse, Seewald, & Jamieson, 1995), NAL-NL1 (Dillon, 1999) and CAMEQ (Moore, Alcantara, Stone, & Glasberg, 1999a; Moore, Glasberg, & Stone, 1999b), and CAMREST (Moore, 2000).

In contrast, some manufacturers of hearing instruments such as Widex, Phonak, GN ReSound and Oticon have devised their own prescriptive procedures, called proprietary fitting formulae for their specific devices to meet fitting requirements. The description of these formulae is usually not available in scientific journals. Moreover, neither they present fitting targets, nor the variation in targets according to degree and configuration of individual hearing loss (Keidser, Brew, & Peck, 2003).

It was a question of thorough research as to which fitting procedure best applies to the need of hearing impaired individuals. Keidser and Grant (2001) compared IHAFF and NAL-NLl in paired comparison test and sentence recognition task. Their findings showed that subjects preferred NAL-NLl in paired comparison test and performed better with NAL-NLl in sentence recognition task in presence of noise than IHAFF.

Similarly Byrne, Dillon, Ching, Katsch, and Keidser (2001) studied gain provided by NAL-NLl, DSL[i/o], FIG6 and IHAFF and found that the gain provided by these procedures varied across frequencies for degree, hearing loss configurations and applied input levels.

Although only a few studies exist for the comparison of the gain prescribed by different generic procedures, there seems to be dearth in literature regarding efficacy of proprietary fitting procedures. Keidser et al. (2003) investigated target insertion gain prescribed by different proprietary and generic fitting procedures and found significant variation among target gain for various hearing loss configurations. When proprietary fitting procedures were compared with other procedures, a variation of 10 to 30 dB in gain was seen at lower and higher frequencies.

Even though, to date, published data explain well, how different fitting procedures provide different target gain for similar hearing loss configuration and how these target gain vary as a function of degree and configuration of hearing loss, there is lack of sufficient information on the efficacy of these procedures on speech understanding ability.

Need for the study

Since hearing aids are now available with the options of proprietary and generic prescriptive procedures, it is necessary to study variation in individual performance, if any, on speech identification task when their devices are programmed with **both** categories of fitting procedures (i.e., proprietary and generic fitting procedures).

Aim of study

The present study attempts **to** investigate whether speech identification scores vary when hearing aid is programmed with generic and proprietary fitting procedures for a given configuration of hearing loss.

CHAPTER2

REVIEW OF LITERATURE

REVIEW OF LITERATURE

Variation in hearing losses in terms of their degree, configuration and type demand that hearing instrument selected for an individual should satisfy the listening need of hearing impaired individuals. The best way to do this is to use a prescriptive procedure that takes care of required target amplification characteristics.

A prescriptive approach to hearing aid fitting is one in which the hearing characteristics of the hearing-impaired individuals are measured, and the necessary amplification characteristics are calculated from them. Of course, this requires a known (or assumed) relationship between the person's listening characteristics and the required amplification characteristics. These required amplification characteristics are often referred to as target gain. Prescriptive formulae can be based on hearing thresholds, supra-threshold loudness judgments and the situations in which the hearing aids need to be worn. However, it is totally impractical to prescribe a hearing aid solely based on prescriptive methods as evaluation of the end results, such as fine tuning of the device according to individual needs is essential in all cases (Dillon, 2001).

Prescriptive selection procedures have come a long way and description of these can be found even before 1950s. According to Knudsen and Jones (1935), the gain needed at each frequency was equal to the threshold loss at the same frequency minus a constant. This is also known as mirroring of the audiogram, as the shape of the gain frequency response equals the inverse of the shape of the hearing loss. The mirroring procedure requires 1 dB increase in additional gain to compensate every 1 dB increase in hearing loss. As differences in loudness growth function exist for individuals with sensorineural hearing loss and individuals with normal hearing, the gain needed to restore normal loudness perception in SN hearing loss is equal to the threshold loss only when the person is listening at threshold. For all higher levels, the amount of gain would be excessive if all gain prescription methods follow mirroring procedure. Mirroring thus leads to excessive gain, especially for those frequencies with the greatest hearing loss (Dillon, 2001).

The next progress in this area was to provide required gain based on the person's most comfortable level (MCL) rather than on their thresholds. Watson and Knudsen (1940) suggested that speech should be amplified sufficiently to make speech energy audible and comfortable. Although their specific formula incorporated MCL, but failed to account the variation of speech energy across frequency.

In 1944, Lybarger proposed half gain rule, which today forms the basis for several current prescriptive procedures. According to the half gain rule, "the amount of gain chosen by hearing impaired individuals was approximately half the amount of threshold loss". Infact, half gain rule and raising speech to MCL, both are functionally the same. However, in cases of mild to moderate sensorineural hearing loss, the threshold of discomfort is little different from that in normal hearing individuals. As MCL is approximately half way between threshold of hearing and discomfort, every 1 dB increase in hearing loss requires MCL to be raised by 0.5 dB. This is why gain is approximately half of the hearing loss. However, half gain rule needs some modification since the loudness for low and high frequency components vary as former being more intense than the later. This can be done simply either by increasing more gain for high frequencies or by decreasing gain for low frequencies or both (Dillon, 2001).

Further, the half gain rule also needs to be modified for severe and profound hearing losses. When hearing thresholds are greater than 60 dB HL, discomfort thresholds are significantly above normal. So the relationship between threshold, MCL, and discomfort does not hold good. In this case, MCL elevates by more than half of the hearing threshold loss, consequently the gain must be more than half of the hearing loss (Dillon, 2001).

Therefore, it is evident that even more than 50 years ago, mainly two auditory attributes, hearing threshold and supra-threshold loudness percept (such as MCL) were used to form foundation for prescription. The link between these is made clear in some procedures where threshold and discomfort levels are measured, but are used to estimate MCL by assuming that MCL bisects the person's dynamic range (Dillon, 2001).

This dichotomous relationship between threshold and loudness perception forms the basis for most current procedures for advanced non-linear hearing aids. So far, prescription procedures for non-linear devices can be broadly classified into two categories.

First, as there is proliferation of hearing instruments some manufacturers have developed their own prescription formulae influenced by parameters of individual hearing aid. These prescriptive procedures are also known as proprietary fitting formulae. Some such methods are based on substantial research (Arlinger, Billermark, Oberg, Lunner, & Hellgren, 1998; Lunner, Hellgren, Arlinger, & Elberling, 1997) but others may not be research based, or rather any such basis may not be published or available for scientific scrutiny. Also, these procedures neither present the fitting targets nor discuss how they vary with hearing loss (Keidser et al., 2003).

A second approach has been used to develop a prescriptive procedure, one that is appropriate to all types or at least a range of aids of roughly the same type. These are also known as generic fitting formulae. Over the years, several such methods are developed. These procedures are mainly dependent on two rationales i.e., either loudness normalization or speech intelligibility maximization.

Loudness normalization means that sounds that appear soft to a normal hearing person should be audible soft, after amplification, to the hearing-impaired person. Similarly, sounds that are comfortable or loud, for the normal hearing person should be comfortable or loud, respectively, after amplification for the hearing aid user. There are basically two aspects of normalization. First, the overall loudness of sounds in normalized. This means for any input level and frequency the sounds would be equally loud for a normal hearing individual and to a hearing impaired person after amplification. Second, the relative loudness of each frequency components of complex sounds will be preserved. According to Byrne et al. (2001), the status of DSL[i/o] is ambiguous in the sense that it is originally described as normalizing loudness but recently described by Seewald, Cornelisse, Black, and Block (1996) as loudness equalization procedure that can be modified to provide normalization. By equalization, it means that all frequency bands of speech will be amplified sufficiently to produce equal loudness of speech.

The aim of speech intelligibility maximization is to maximize speech intelligibility for a specified loudness levels. Such an approach will result in loudness not being normalized in any frequency region, although the overall loudness of broadband sounds may well be normalized (Dillon, 2001).

The majority of fitting procedures such as Loudness Growth in *Vz* Octave Bands (LGOB) (Allen, Hall, & Jeng, 1990), Independent Hearing Aid Fitting Forum (IHAFF) (Cox, 1995), Ricketts and Bentler (RAB) (Ricketts, 1996), ScalAdapt (Kiessling, Schubert, & Archut, 1996), FIG6 (Killon & Fikret-Pasa, 1993) and Desired Sensation Level (input/output) (DSL [i/o]) (Cornelisse, Seewald, & Jamieson, 1995) use loudness normalization rationale. On the other hand, National Acoustic Laboratory Non-Linear, version 1 (NAL-NL1) (Dillon, 1999) is based on speech intelligibility maximization.

The following discussion focuses the origin of different fitting procedures used in the present study, their underlying principles to prescribe gain for a range of input levels and how the gain is dependent for different configurations of hearing loss. The other fitting procedures stated above are briefly described in Appendix-A. Finally, summary of various studies are mentioned which compare the variation in target gain prescribed by those formulae and how individuals perform on speech identification tasks, when fitted with different procedures.

DSL[i/o]

The development of Desired Sensation Level (DSL) started in 1982 with the original published description occurring in 1985 (Seewald, Ross, & Spiro, 1985). This formed the foundation for the DSL method with its goal to make speech sufficiently audible to allow speech perception, without discomfort, for all degrees of hearing loss.

Initially the DSL algorithm focused on calculation of amplified long term average speech spectrum (LTASS) targets for linear gain hearing instruments. A computer-assisted implementation was first released in 1991 (DSL 3.1). The DSL 3.1 algorithm is based on loudness equalization (i.e., setting targets approximately at most comfortable listening level). Because, to date, available research in this regard has indicated that adult listeners generally achieved their best speech intelligibility in the region of MCL (Kamm, Dirks, & Mickey, 1977; Pascoe, 1978; Pascoe, 1988). Work conducted by Skinner and colleagues (Skinner, Pascoe, Miller, & Popelka, 1982; Skinner, Miller, DeFilippo, Dawson, & Popelka, 1986; Skinner, 1993) indicated that scores on measures of speech perception were maximized when the amplified output approximated the preferred listening levels for frequencies between 250 to 6000 Hz. Studies of preferred listening levels in children also indicated that sufficient amplified speech audibility could be achieved near MCL (Macrae, 1986; Erber & Witt, 1977).

DSL[i/o] was first introduced in 1994 (Cornelisse et al., 1995) followed by the most recent version, DSL v4.1 for Windows, in April, 1997 (Seewald , Cornelisse, Ramji, Sinclair, Moodie, & Jamieson, 1997). The DSL[i/o] algorithm was designed for the purposes of fitting hearing instruments with advanced non-linear processing schemes (WDRC and digital devices). While some confusion exists about the exact intent of the algorithm (Byrne et al., 2001), DSL[i/o] is best described as an acoustic mapping algorithm (DSL v4.1 manual p. 21). That is, for a given frequency band, an acoustic input region is mapped onto an acoustic output region. The input dynamic range is generally defined as an extended normal auditory area from the normal hearing threshold of audibility to the hearing impaired individual's upper limit of discomfort. The output dynamic range is defined by the hearing loss (threshold to UCL). Obviously compression is applied given that the input dynamic range is different than the output dynamic range as defined by the hearing loss.

To date, there would appear not to be any comparative studies assessing differences in speech perception ability using the DSL algorithm. There are however, are a number of DSL validation studies that are of interest.

Scollie, Seewald, Moodie, and Dekok (2000) conducted a study to evaluate if the DSL recommended user setting is similar to the preferred listening levels of pediatric hearing instrument users. Results indicated that on average, the DSL recommended volume setting was within 3 dB of preferred listening levels for these children.

Jenstad and colleagues (Jenstad, Pumford, Seewald, & Cornelisse, 2000; Jenstad, Seewald, Cornelisse, & Shantz, 1999) compared the performance of hearing instrument users with linear and WDRC hearing instruments fitted using the DSL method. Performance measures included speech perception and loudness growth. Results of this study indicated consistent speech perception scores for five speech spectra inputs representing essentially extremes in normal listening conditions when using WDRC circuitry fit as per the DSL algorithm. Further, the WDRC circuit fit to DSL[i/o] targets provided more normalized loudness perception.

Independent of the target algorithm, what makes DSL unique is the approach that it takes to the definition of variables that must be considered in the hearing instrument fitting and verification process. Very early in the development of the DSL method the authors recognized that "audibility of the speech signal can be viewed as the most basic prerequisite to auditory linguistic growth and performance" (Seewald & Ross, 1988). The simple fact is that in order to have speech perception you must have speech audibility. Further, it is imperative that the amplified level of speech output in the ear canal be placed at a preferred listening level across the broadest frequency range possible, in a variety of different listening conditions, to achieve maximal performance on speech perception measures. That is, the speech signal must be audible, comfortable and undistorted across a range of listening conditions regardless of the amplification circuitry and prescriptive method used.

Defining audibility of an amplified speech signal is difficult if the audiometric thresholds are referenced in dB HL and the hearing instrument performance is measured in dB SPL. The HL audiogram also infers an average adult ear canal. Consequently, prediction of actual threshold in dB SPL based on a standard HL scale is difficult given the individual variations in ear acoustics (Hawkins, Cooper, & Thompson, 1990). The process may be further complicated by the use of insertion gain as a verification protocol. Targets are not generally presented in the context of the threshold data and consequently the approach does not allow a direct measure of the relative audibility of the amplified speech signal. In particular a match to targets using a static pure tone sweep does not infer a match to targets when speech is the input, specifically when any form of active compression is present.

In order to ensure that the relative audibility of an amplified signal can be determined, DSL refers all assessment data, hearing instrument targets and verification data to real ear SPL. The process emphasizes a very basic scientific principal, that if you want to compare two measures (in this case hearing thresholds and hearing instrument performance), then they have to be measured with the same scale (dB SPL) and referenced to the same point (ear canal). Much of the research that has gone into the development of DSL has focused on the accurate interpretation of threshold data (Scollie, Seewald, Cornelisse, & Jenstad, 1998) and the prediction of the audibility of amplified speech based on accurate threshold measures (Moodie, Seewald, & Sinclair, 1994; Seewald, Moodie, Sinclair, & Scollie, 1999). Consequently, the DSL method converts the HL audiogram into an audiogram in SPLogram (Figure 2.1). Targets are provided based on the degree of hearing loss for various levels of inputs. This measurement format allows a direct comparison between the individual's audiometric characteristics and hearing instrument performance in the ear.

Thus, while there is insufficient data to date to support any particular algorithm as ultimately optimal for speech perception, the strategy applied in DSL is justifiable based on available research. Validation studies would support that, if applied appropriately, the DSL algorithm ensures comfort and acceptable speech perception abilities **over a** wide range of input levels.

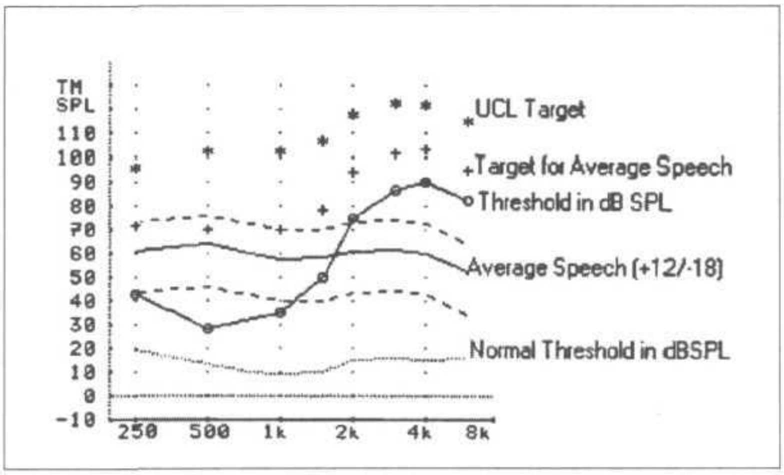

Figure 2.1: SPLogram presentation for a mild to severe hearing loss. All relevant variables have been referenced in dB SPL at the eardrum. (Seewald et al., 1997)

NAL-NL1

The name 'NAL-NL1' stands for National Acoustics Labs, Non-Linear, version 1 and was first described by Dillon in the year **1999. The** underlying assumption behind this procedure like its predecessors, NAL-R and NAL-RP is to maximize speech intelligibility subject to the overall loudness of speech at any level being not more than perceived by a normal hearing person. The main objective of developing NAL-NL1 was to determine the gain for several input levels that would result in maximal effective audibility. It is neither based on loudness normalization nor loudness equalization. However, in this procedure, the loudness of the signal is varied to such an extent where speech intelligibility is maximized (Byrne et al., 2001).

The NAL-NL1 procedure was derived by calculations that combined a loudness model (Moore & Glasberg, 1997) with a modification of the Speech Intelligibility Index (SII) (ANSI, 1993). The only information required by both of models is hearing thresholds and the speech spectrum levels input to the ear after amplification. SII is modified by calculation of an effective audibility correction from the data of Ching and co-workers (Ching, Dillon, Katsch, & Byrne, 1998). Modification of the SII includes effects of hearing loss desensitization (reduction in speech intelligibility as hearing loss increases) and the effects of listening at high sound pressure levels (SPLs). The effect of effective audibility correction is that, for a given amount of speech audibility, the SII contribution is reduced as hearing level becomes greater. The correction is relatively more for the high than for the low frequencies and is larger for high sensation levels than for the low sensation levels. The main implication of effective audibility correction in hearing aid fitting is that it ensures adequate speech understanding at high levels, especially for high frequency hearing loss, and for higher degrees of hearing loss (Byrne etal.,2001).

The NAL-NL1 method is based on a complex equation that specifies insertion gain at each standard 1/3 octave frequency from 125 Hz to 8 kHz at each frequency. At each frequency, the gain depends on threshold at that frequency, the three-frequency average threshold, slope of the audiogram from 500 Hz to 2 kHz and the overall level of a broad-band signal with a long-term spectrum like that of speech. For speech input at any level, gain at each frequency was systematically varied with a high speed computer until the calculated speech intelligibility was maximized, but without the calculated loudness exceeding that loudness calculated for normal hearing people listening to speech at the same level. This process was repeated for many representative audiogram and the optimized gains for each audiogram, for each input level were found. Because this was a very time consuming process, even for a single audiogram at a single input level, an equation was fitted to the complete set of optimized gains. This equation thus summarizes all the optimizations and can be applied to any audiogram. Alternatively, the aid can be prescribed in terms of real ear aided gain (REAG). REAG is deduced from insertion gain by adding the adult average real ear unaided gain (REUG) to the insertion gain target (Dillon, 2001).

To calculate NAL-NL1 gain targets, gain calculations were performed by taking 52 people with different audiometric configurations, for input levels from 30 to 90 dB SPL, in 10 dB increments. For gently sloping, hearing loss audiograms, ranging from mild to severe in degree, the resultant insertion gain was found to be close to that provided by NAL-RP. Consistent with the other NALs, for most types of hearing loss, the mid frequencies of speech were found to be similar in loudness to the lower and higher adjacent speech frequencies. The end aim was thus achieved, i.e., equal loudness of all speech frequency bands, along with maximal speech intelligibility (Byrne et al., 2001).

The prescription can also be expressed as an I/O curve at any frequency, or as a coupler gain-frequency response. Because these are often measured with pure tones, the NAL non-linear software allows for the crest factor 7 band width differences between pure tones and speech signals. The prescription is calculated for pure tones and not for broad band signals, therefore depends on the number of channels with in the hearing aid (Dillon, 2001).

The NAL- nonlinear software programme displays the result as either gain curves at different levels, or I/O curves at different frequencies. For multichannel hearing aids, cross over frequencies, compression thresholds, compression ratios and gains for 50, 65 and 80 dB SPL input levels were also recommended by the NAL software.

Amplification requirements for people with mixed losses are fulfilled by applying the formula to the sensorineural part of the loss (i.e. the bone conduction thresholds) and then adding gain equal to 75% of the conductive part of the loss (i.e., the air bone gap) (Dillon, 2001).

Comparison of gain among procedures

Although various fitting procedures have been used by hearing care professionals for a fairly long time, the relative effectiveness of different procedures (i.e., those that follow loudness normalization or speech intelligibility maximization) remain a controversial issue, in part, because there has been little comparative evaluation of non-linear procedures. Also, it is important to know which rationale works best when listening to a range of input levels that hearing aid users are exposed to in real life situations. Here is the summary of few studies, which attempted to answer some of these questions.

It is obvious that response prescribed by different prescriptions varies as a function hearing loss. Byrne et al. (2001) attempted to compare NAL-NL1 with DSL[i/o], FIG6 and IHAFF for flat, reverse slope, moderately sloping and high-frequency hearing loss for a range of input levels i.e., 50, 65 and 80 dB SPL. Results showed that when compared with other three procedures, NAL-NL1 prescribe less low frequency gain for flat and upward sloping hearing loss, while it prescribes less high-frequency gain for moderately sloping and steeply sloping high frequency loss. The relative differences in gain prescribed by different procedures are not unusual, as these procedures are not based upon single underlying rationale. As it is evident from the previous discussion, NAL-NL1 attempts to give speech the spectral balance required to maximize calculated speech intelligibility. Since low frequency parts of the signal contribute to the loudness of the signal than the intelligibility, NAL-NL1 gives low cut while prescribing gain, which is not the case for other procedures. They attempt to place speech at each frequency at the level needed to give normal loudness for that frequency. For high frequency hearing losses, NAL-NL1 never attempts to give high sensation level encouraged with the fact that ear's ability to extract information decrease in those frequency regions. However, it is not clear as in spite of using same rationale, the other three procedures prescribe different gain for various configurations of hearing loss. This may be in part due to slight differences in their rationale and in the normative data they utilize (Dillon, 2001).

In a similar study, Keidser et al. (2003) attempted to compare target insertion gain prescribed by different proprietary fitting procedures to generic fitting procedures. They compared Digifocus II (Oticon), Claro (Phonak), Danalogic (GN Resound), and Senso Diva (Widex) to NAL-NL1 and DSL[i/o]. Comparison was done across three input levels viz 50, 65 and 80 dB SPL for five audiogram configurations (flat, reverse sloping, gently sloping, steeply sloping with normal low frequency and steeply sloping high-frequency loss with a mild low-frequency loss). Analysis of results was carried out to explore hearing loss configuration dependency on the prescribed insertion gain. This means the change in gain as a function of slope of audiogram. The differences in the targets prescribed by the two generic procedures DSL[i/o] and NAL-NL1 have been reported earlier and these can be described in terms of their assumptions and underlying principles. When proprietary fitting rationales are compared with other procedures, a variation of 10 to 30 dB in gain is seen at lower and higher frequencies. With respect to hearing loss configuration dependency, the loudness normalization and loudness mapping procedures such as DSL[i/o], Claro, Danalogic and Senso Diva all prescribe gain independent of hearing loss configuration. On the other hand, gain prescribed by NAL-NL1 at each frequency depends on degree of loss at several frequencies. Further investigation suggested that Digifocus II is also hearing loss dependent when different acoustic parameters were selected for different audiograms.

Experimental comparisons and evaluations

Stelmachowicz et al. (1998) compared gain prescribed by DSL[i/o], FIG6 and a manufacturer's threshold based algorithm to the adult's user gain for 50 and 80 dB SPL input levels in the Resound, 2-channel WDRC hearing aid. In general, results showed that both the generic procedures do not agree well with the gain actually used by hearing aid wearers. Under all the conditions, both the procedures tended to prescribe more gain than was actually used by the subjects. DSL procedure over-prescribed gain at 500, 2000 and 4000 Hz at both input levels. FIG6 under-prescribed gain for mild and moderate hearing losses, particularly at the 80 dB input level, but over prescribed gain for severe to profound losses. The manufacturer's algorithm provided a closer approximation to the gain actually used by the adults in this study, and the inter-subject variability was found less than that observed for the other two threshold-based procedures. This is not too surprising, as the proprietary formula was a statistical summary of the gains actually used by wearers of precisely this type of hearing aid. The authors recommended that, manufacturer's fitting formula can closely provide loudness based gain values in the subjects who cannot reliably perform loudness judgment. Further, authors asserted that the results of the study cannot be generalized to other types of non-linear devices as only one type of non-linear device was used in this study. The authors failed to clearly mention the experience with prescriptive method with which they were fitted before study and the gain characteristics of the same would have affected the results.

Humes et al. (1999) compared a two-channel WDRC device using DSL[i/o] prescription to a linear hearing aid using NAL-R procedure. The WDRC device gave higher speech intelligibility, particularly at lower input levels and was preferred by 76% of the subjects in the field trial. One possible interpretation is that DSL[i/o] prescribed a more appropriate gain-frequency response for mid level inputs than did NAL-R. The high frequency gain achieved for an input of 70 dB SPL was less than the gain prescribed by both types of aids. Due to differences in the two formulae and the devices used, the mean high-frequency gain achieved for the WDRC instrument was closer to NAL-R than to the DSL[i/o] prescription procedure.

Keidser and Grant (2001) compared NAL-NL1 and IHAFF protocol in a two-channel compression hearing aid. Comparison involved initial laboratory test followed by field trial and final laboratory test. The laboratory test consisted of paired comparison test and sentence recognition task under quiet and noisy situations. Results showed that most of the subjects preferred NAL-NL1 in paired comparison test and performed better with NAL-NL1 in sentence recognition task in the presence of noise than IHAFF. Therefore, authors concluded that, when the two fitting rationale prescribed substantially different responses for a 65 dB SPL input and these differences were achieved in the fitting, then the subjects preferred NAL-NL1.

Smeds and Leijon (2001) compared six threshold-based prescriptive methods for non-linear hearing instruments for a standard audiogram. Out of six procedures two were generic (DSL[i/o] and FIG6) and four device specific methods (ASA, Oticon; Loudness Mapping, Widex; S2000R, Danavox; LGOB Threshold, ReSound). Comparison was done for three simulated listening conditions (soft, average and loud speech). For soft and average speech, comparison was done in quiet environment and for loud speech comparison was done in presence of speech babble. For all the procedures insertion gain measurement, loudness and speech intelligibility index (SII) calculations were carried out in all three listening conditions. Analysis of the results showed that all the fitting methods prescribed different insertion gain-frequency response for the same audiogram in simulated real-life situations. The difference between theoretically prescribed gain and hearing aid implemented gain were found to be large for both DSL[i/o] and FIG6, being about 10 dB in the frequency region 1-2 kHz. The authors stated that it is difficult to draw conclusions based on hearing aid implementations only. The loudness calculations show large differences between the fitting methods. According to authors, based on this study, it is difficult to come to any conclusion as to which method is closest to the loudness desired by normal hearing listener. The differences in SII calculations for amplified speech were small for different fitting strategies. The methods differed substantially only in soft speech condition followed by average speech and negligible differences in loud speech condition. However, the results of the study need to be verified in the presence of noise even for soft and average speech inputs and with hearing aid wearers.

Alcantara, Moore, and Marriage (2001) compared the effectiveness of the CAMEQ, CAMREST and DSL[i/o] fitting procedures in experienced hearing aid users fitted bilaterally with Danalogic 163D digital hearing aids. Immediately after fitting with a specific procedure and one week after fitting, the gains were adjusted when required by the minimum amount necessary to achieve acceptable fittings. The amount of adjustment required provides one measure of the adequacy of the initial fitting. The same process was carried out for all the fitting procedures and fitting order with procedures was counterbalanced across subjects. On the average, the gain adjustments were smallest for the CAMEQ followed by CAMREST and largest for DSL[i/o]. The authors conclude that DSL[i/o] provide more high frequency gain than preferred by adult users. APHAB and speech recognition threshold (SRT) in noise did not differ significantly for all the three procedures. Overall, the CAMEQ and CAMREST procedures give more satisfactory initial fits than DSL[i/o] for experienced adults. The authors have not mentioned whether the individuals included in this study had an experience with CAMEQ or CAMREST procedure. If it is the case, small adjustments for these two procedures were likely to occur.

In a similar study that involved experienced hearing aid users but fitted unilaterally Alcantara, Moore, and Marriage (2004) showed similar results as that of previous one. The authors commented that CAMEQ and CAMREST procedures provide a more initial fitting than DSL[i/o] even for unilaterally, experienced hearing aid wearers. Also, comparison with the previous study based on bilateral fittings suggests that the gain preferences were found to be same for unilateral and bilateral fittings. Further, when adequacy of initial fitting was investigated in experienced versus new users (fitted either unilaterally or bilaterally), Marriage, Moore and Alcantara (2004) showed results in favour of CAMEQ and CAMREST fitting procedures than DSL[i/o] and were consistent with their previous findings.

Smeds (2004) investigated two methods, NormLoudn (generic method) and hearing aid specific method; LessLoudn (Proprietary method) in a Danalogic 163D in field as well laboratory study. After necessary adjustment, the measured gain for the two methods was found to be similar in prescribing gain-frequency shape, but NormLoudn gave more over all gain than LessLoudn. On the subjective preference task most of the subjects favored LessLoudn than NormLoudn in all the test situations. Paired comparison of preference in the laboratory supported the findings in the field. LessLoudn was generally preferred to NormLoudn in all test conditions, except for the soft speech in very soft noise where there was no significant difference for either method. Speech recognition scores were similar for the two fitting strategies. Further, a comparison between the measured gain for NormLoudn and gain prescribed by CAMEQ, NAL-NL1, and DSL[i/o] suggested that all three prescriptive methods overestimate the gain requirement of inexperienced hearing aid users and with mild to moderate hearing loss. Finally, the author concluded that careful adjustment of the over all gain is necessary to avoid overamplification in these individuals. The estimated gain for CAMEQ, NAL-NLl, and DSL[i/o] are likely to occur as they are based on different rationale. Further, no difference in speech recognition scores is suggestive that both NormLoudn and LessLoudn prescribe adequate gains required by hearing impaired individuals.

CHAPTER 3
METHOD

METHOD

Subjects

Two groups of subjects participated in this study.

1- *Hearing-Impaired Group:* Twenty-nine individuals with bilateral (symmetrical or asymmetrical) post-lingual sensorineural (SN) hearing loss in the age range from 18 to 60 (mean age 46) years participated in this study. All the individuals had either moderate flat, moderately-severe flat, gradual sloping or steeply sloping (Carhart, 1945; Lloyd & Kaplan, 1978) hearing loss configuration in the test ear. All participants were naive users of hearing aids. Table 3.1 shows subject's distribution based on hearing loss configurations.

Table 3.1. Subject's distribution based on hearing loss configurations

Hearing loss configuration	No. of Subjects
Moderate flat	**8**
Moderately severe flat	**6**
Gradual sloping	**8**
Steeply sloping	**7**

2- *Normal-Hearing Group:* Twenty-five individuals with normal hearing in the age range from 18 to 50 (mean age 27) years were chosen. All the individuals had audiometric thresholds of 15 dB HL or better with no indication of any middle ear pathology.

Both the groups of individuals were native speakers of Kannada language and had speech identification scores of 75% or more.

Instrumentation

- A calibrated Interacoustics AC40 diagnostic audiometer with sound field speakers.
- FP 40 D hearing aid analyzer.
- PC with Phonak Fitting Guidelines (PFG, version 8.4a) hearing aid fitting software and HiPro interface unit.
- Phonak Perseo 311 dAZ Forte BTE hearing aid with fitting range for moderate to severe hearing loss. The hearing aid has 20 bands with gain of 65 dB and maximum power output of 130 dB.

Test material

Preparation of test material

Test material for speech identification was prepared in Kannada language to evaluate the benefit of different prescriptions. A total of 40 word sets were prepared with each word set containing 3 words. Hence, test material contained a total of 120 meaningful words (List is given in Appendix-B). Words were made with combination of low, mid and high frequency speech sounds. Each word in the word set contained a combination of high and low, low and mid, or high and mid frequency speech sounds. Hence, one set of words included 3 words with all above mentioned combinations.

Test environment

Testing was carried out in sound treated, double-room set-up with ambient noise level with in permissible limits (re: ANSI S3.1-1991, as cited in Wilber, 1994). Since audiometer displays only dB HL values, the HL values were converted in to dB SPL values by adding RETSPL to dB HL. For both signal (speech) and noise (speech shaped noise) RETSPL values were 13 dB (Wilber, 1994).

The output of an audiometer was connected to the speakers. Stimuli (speech material and speech shaped noise) were presented through either of the speakers positioned at 45° or 315° azimuth and 1 meter away from subject depending on the side of test ear. Signal and noise were routed through the same speaker.

Procedure

The study was carried out in two phases in the following manner:

Phase 1

In the first phase, speech material was presented to individuals with normal hearing. The tester presented speech material in presence of speech noise through live voice and subjects had to repeat the words after clinician. Speech material was presented at soft (55 dB SPL) and average (65 dB SPL) (Byrne et al., 2001) levels of input and at 0 and -5 dB SNRs, which consisted a total of four listening conditions. Signal and noise were presented through either of two speakers positioned at 45° or 315°. Few practice items were presented to individuals to familiarize the task. In final speech task, a total of 30 words (i.e., 10 word sets) were presented to each individual. The subjects were instructed to repeat words after clinician. If the subjects correctly repeated word it was considered as correct response, and if not, it was considered as wrong response. Words correctly repeated by subjects were converted in to percentage speech identification (SI) scores. Similarly percentage SI scores were calculated at soft and average input levels and at two SNR conditions. Order of word list was randomized for all individuals across conditions to maintain equal difficulty of the task.

Phase 2

In the second phase, signal-to-noise ratio (SNR) was measured for the words presented in presence of competing speech noise, in aided conditions, involving individuals with SN hearing loss. For the purpose of the study SNR was defined as where individuals correctly repeat two or more than two words in a set of three words presented in competing speech noise. The hearing aid was fitted to each individual unilaterally and to the ear with better speech identification scores. Before fitting to each subject, electroacoustic performance of the hearing aid was measured using FP 40 D hearing aid test system. The hearing aid was attached to a HA-2 coupler and placed in the test chamber. A swept tone function or speech modulated noise was given to verify the hearing aid response. Once the performance of hearing aid was found satisfactory, it was fitted to subjects. The process of hearing aid fitting and SNR measurement was as follows:

A) A Phonak Perseo 31 ldAZ Forte digital behind the ear (BTE) hearing aid coupled with appropriate stock ear mould was used in all test conditions. Hearing aid was programmed using manufacturer's recommendations, with PFG (version 8.4a) software and HiPro interface unit. For each subject, hearing aid was programmed with no changes from predicted fitting with the only input to fitting being individual audiogram. For each individual, the hearing aid was programmed with two generic (NAL-NLl and DSL[i/o]) and one proprietary (Phonak digital/ski slope) prescriptive procedures. In cases of flat and gradual sloping hearing loss configurations, individuals were fitted with 'Phonak digital' and in cases of steep sloping hearing loss; individuals were fitted with 'Phonak ski slope' prescriptive procedures as recommended by the manufacturer.

B) After fitted according to one of the prescriptive procedure, real ear insertion gain (REIG) was measured at 65 dB SPL with digi-speech input signal using FP 40 D hearing aid analyzer. REIG was obtained by subtracting real ear unaided response (REUR) from real ear aided responses (REAR).

C) Once the hearing aid was programmed with one prescriptive method, SNR was measured for words presented in presence of speech noise for each subject. Signal and noise were presented through the same speaker on the side of hearing aid. During testing, the unaided ear was blocked with foam ear plug to avoid any possible participation from that side. Speech materials were presented through live voice by the clinician at soft level (55 dB SPL). SNRs were measured using an adaptive procedure. The noise level was set 15 dB below the signal and varied systematically to measure SNR, where individuals had to respond correctly minimum of two words in a set of three words. If the subject correctly repeated two or more words, the level of noise was increased in 4 dB steps and if not, the level of noise was decreased in 2 dB steps. Further, if the subject again repeated the word correctly, the level of noise was increased by 1 dB, and if not, the previous level was considered as SNR. The same process was carried out for average level (i.e., 65 dB SPL) also. Similarly, SNRs were measured in aided conditions for other two prescriptive methods at both levels. The order of hearing aid fitting with each prescriptive procedure was randomized among subjects.

Analysis

The data obtained for two groups of subjects were tabulated for analysis.

CHAPTER4

RESULTS AND (DISCUSSION

RESULTS

The data obtained from normal and hearing impaired individuals was subjected to statistical analysis using SPSS (version 10) software. Analysis was carried out for both the groups individually. Results for both the groups are outlined below.

1- Normal-Hearing Listeners

A) Speech identification scores measurement

- At 55 and 65 dB SPL.
- At 0 and -5 dB SNR conditions.

2- Hearing-Impaired Listeners

A) Comparison of REIG across prescriptive procedure and for all hearing loss configurations.

B) SNRs in aided condition for all hearing loss configurations.

- For each prescriptive procedure
- At 55 and 65 dB SPL

1- Normal-Hearing Listeners

A) Speech Identification Scores Measurement

Results of mean speech identification (SI) scores for normal hearing individuals (N=25) are shown in Figure 4.1. Error bar shows $\pm$ 1 standard deviation (SD). As evident from the figure, both the left-most bars show mean SI scores for a level of 55 dB SPL and right-most bars show mean SI scores for a level of 65 dB SPL. Grey bar in the figure shows mean SI scores at 0 dB SNR and black bar shows mean SI scores at -5 dB SNR conditions.

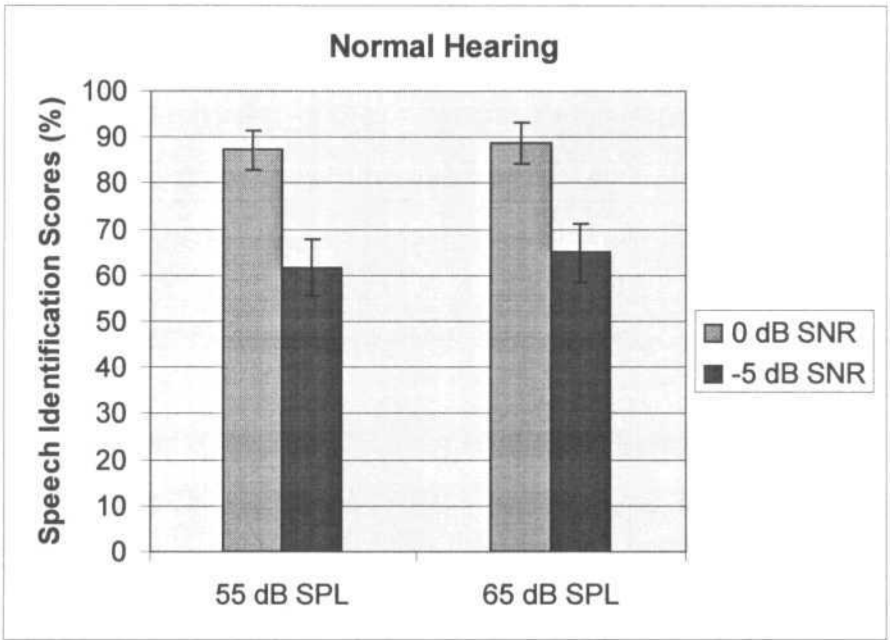

Figure 4.1. Mean SI scores (%) for normal hearing individuals at 55 and 65 dB SPL for 0 and -5 dB SNR conditions.

2- Hearing-Impaired Listeners

A) Comparison of REIG *across prescriptive procedures*

REIG was calculated for moderate flat, moderately severe flat, gradual sloping and steeply sloping SN hearing losses respectively, for an input of 65 dB SPL using digi-speech stimulus. Figure 4.2 to 4.5 show REIG (averaged across individuals) for NAL-NL1, DSL[i/o] and Phonak prescriptive procedures across frequencies from 500 Hz to 6000 Hz.

(i) Moderate Flat hearing loss

Figure 4.2 compares the mean insertion gain prescribed by each fitting procedure, for individuals (N=8) with moderate flat hearing loss. The pattern of insertion gain does not differ across three procedures in mid frequency regions. However, for high and low frequencies DSL[i/o] provided more gain than the other two procedures.

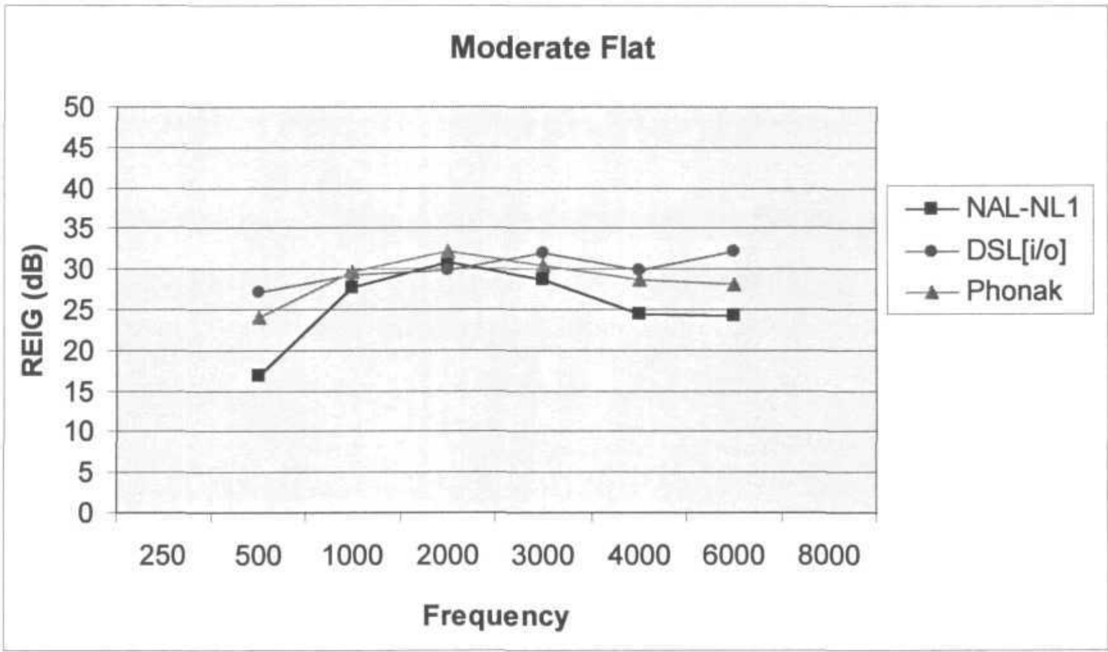

Figure 4.2. Mean insertion gains recommended by three prescriptive procedures for moderate flat hearing loss. Insertion gain is plotted for an input of 65 dB SPL.

(ii) Moderately-Severe Flat Hearing Loss

Mean insertion gain values for individuals (N=6) with moderately-severe flat hearing loss are shown in Figure 4.3 for three fitting methods. As evident from the figure, NAL-NL1 prescribes overall less gain than the other two procedures. DSL[i/o] and Phonak does not show much difference in gains across frequencies. At higher and lower frequencies the difference between NAL-NL1 and DSL[i/o] is found to be more. For example, at 500 Hz NAL-NL1 prescribes 8 dB less gain than DSL[i/o], whereas for 6 kHz NAL-NL1 prescribes approximately 15 dB less gain than DSL[i/o]. The difference between DSL[i/o] and Phonak does not exceed more than 7 dB, especially for low and high frequencies.

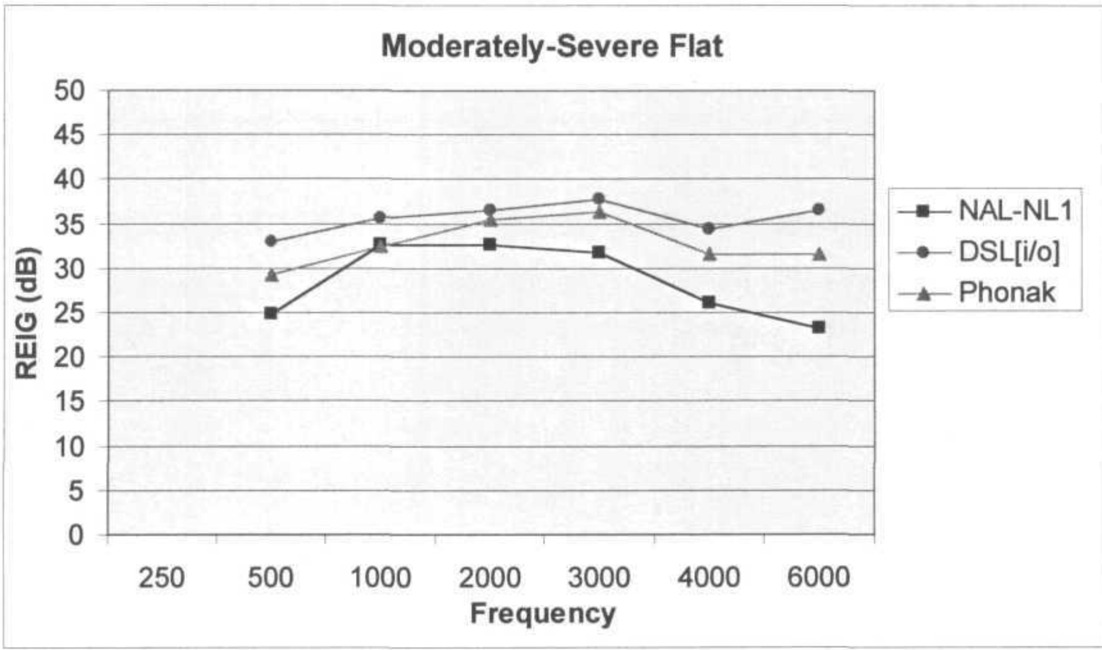

Figure 4.3. Mean insertion gains recommended by three prescriptive procedures for moderately-severe flat hearing loss. Insertion gain is plotted for an input of 65 dB SPL.

(iii) Gradual Sloping Hearing loss

Figure 4.4 shows mean insertion gains prescribed by each fitting procedure for individuals (N=8) with gradual sloping hearing loss. Careful observation of the figure shows that all three procedures prescribe almost equal gain in low and mid frequency regions. However, at higher frequencies NAL-NLl prescribes less gain than the other two procedures. This difference in gain is more marked especially at 6000 Hz.

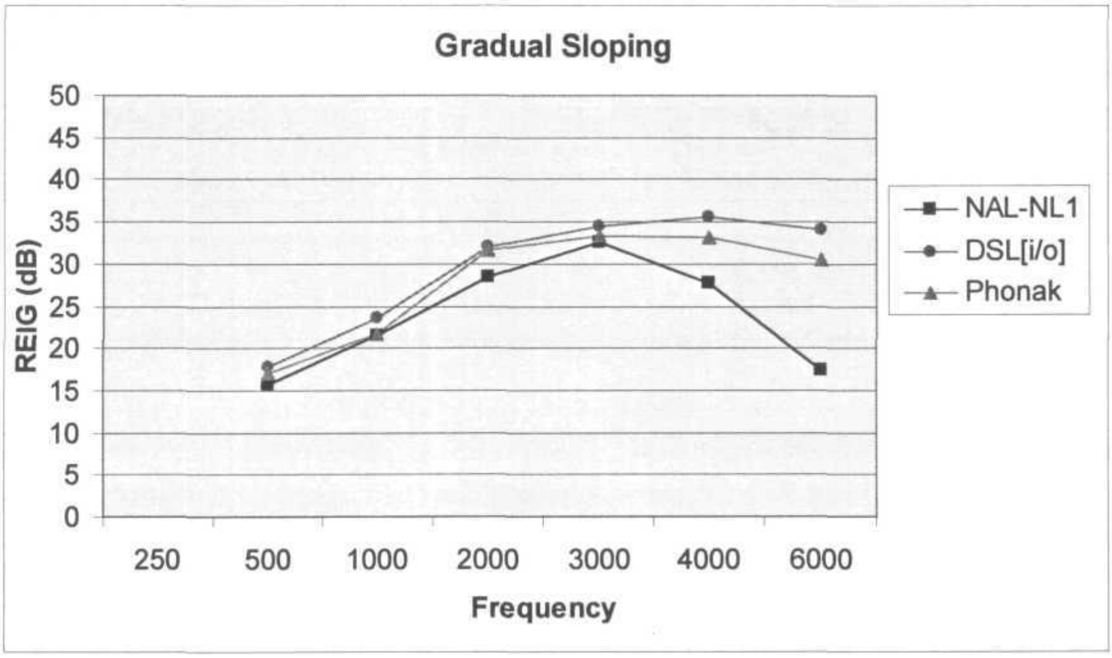

Figure 4.4. *Mean insertion gains recommended by three prescriptive procedures for gradual sloping hearing loss. Insertion gain is plotted for an input of 65 dB SPL.*

(iv) Steeply Sloping Hearing loss

Figure 4.5 depicts mean insertion gains by each fitting procedure for for individuals (N=7) with steeply sloping hearing loss. All the three procedures do not show much variation in gain for low and mid frequencies. The difference in gain becomes large after 3000 Hz for all procedures. NAL-NLl prescribes lesser gain at higher frequencies and gain drops as frequency increases. For example, DSL[i/o] prescribed 46 dB, Phonak 37 dB and NAL-NLl 21 dB gain at 6 kHz.

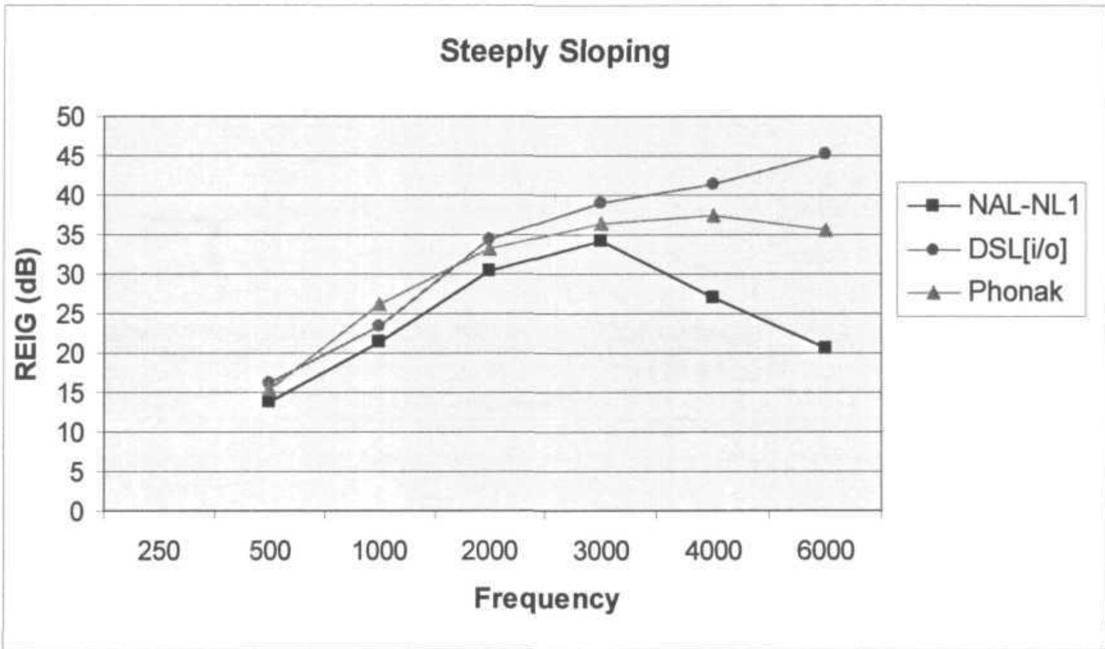

Figure 4.5. Mean insertion gains recommended by three prescriptive procedures for steeply sloping hearing loss. Insertion gain is plotted for an input of 65 dB SPL.

B) Signal-to-noise ratio measurement

As mentioned earlier, for the purpose of the study, signal-to-noise ratio (SNR) was defined as where individual correctly repeat two or more than two words in a set of three words in presence of competing speech noise. The mean SNR (dB SPL) for each hearing loss are shown in Figures 4.6 to 4.9. Error bar shows ± 1 SD. The extreme two left bars show SNR obtained at an input of 55 dB SPL and the remaining two bars show SNR for a level of 65 dB SPL. Note that SNR obtained across three fitting procedures are based on initial fitting and not after adjustments of the gains to achieve acceptable fitting.

A paired sample t-test was administered to find out any significant difference in mean SNR across three procedures:

> For each input level of speech (i.e., 55 and 65 dB SPL).

> For different hearing loss configurations.

The results of t-tests for each configurations of hearing loss are as follows:

***(I) Moderate Flat Hearing** Loss*

The mean SNR and SD (± 1) for moderate flat hearing loss are shown in the Figure 4.6. Analysis of the t- test revealed no significant difference among procedures at each input level (Table 4.1).

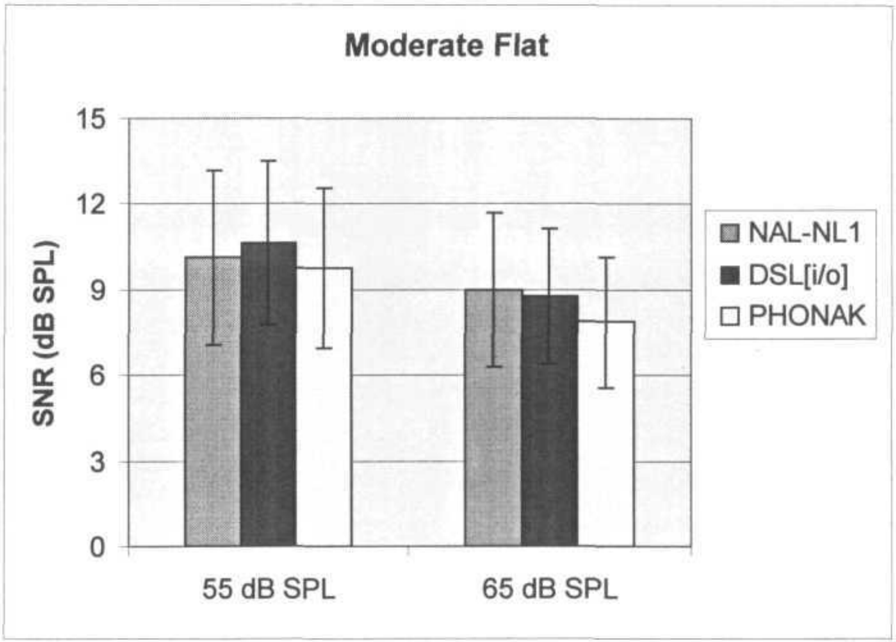

Figure 4.6. Mean SNR (dB SPL) for moderate flat hearing loss at 55 and 65 dB SPL. Error bar shows ± 1 SD.

Table 4.1. Results of t- test at each input level for moderate flat hearing loss.

Level (dB SPL)	Procedures	t-value	Significance
55 dB	NAL-NL1 &DSL[i/ol	-1.08	0.31
	DSL[i/o] & Phonak	0.97	0.36
	NAL-NL1& Phonak	0.44	0.67
65 dB	NAL-NL1 & DSLp/ol	0.32	0.75
	DSLfi/ol & Phonak	1.36	0.21
	NAL-NL1& Phonak	2.04	0.08

(ii) Moderately-Severe Flat Hearing Loss

The mean SNR and SD (± 1) for moderately-severe flat hearing loss are shown in the Figure 4.7. Analysis of the t- test showed no significant difference among procedures at each input level (Table 4.2).

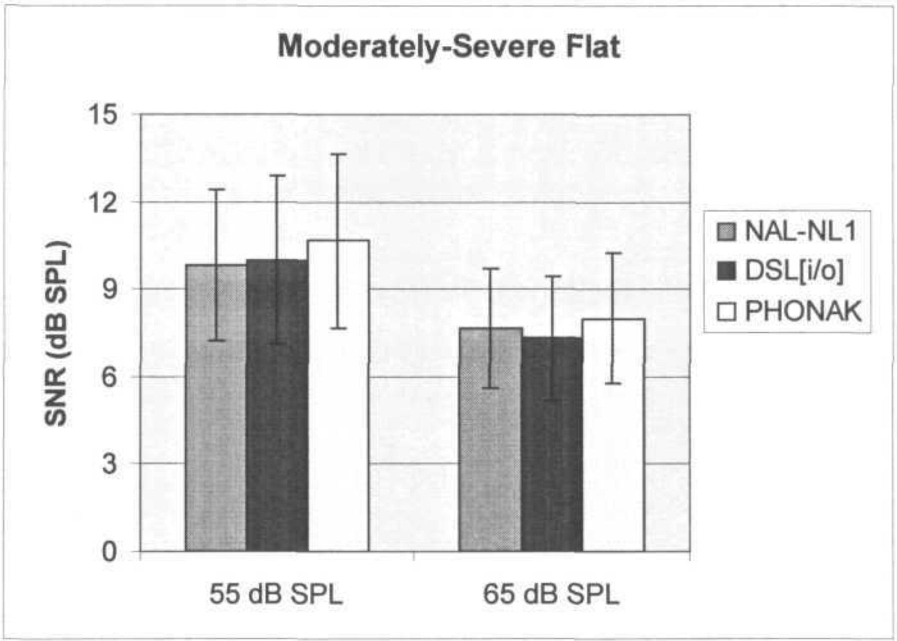

Figure 4.7. Mean SNR (dB SPL) for moderately-severe hearing loss at 55 and 65 dB SPL. Error bar shows ± 1 SD

Table 4.2. Results of t-test at each input level for moderately-sever flat hearing loss.

Level (dB SPL)	Procedures	t-value	Significance
55 dB	NAL-NL1 &DSL[i/o]	-0.41	0.69
	DSL[i/ol & Phonak	-1.08	0.32
	NAL-NLl&Phonak	-1.27	0.25
65 dB	NAL-NL1 & DSL[i/o]	0.50	0.63
	DSL[i/ol & Phonak	-0.75	0.48
	NAL-NLl&Phonak	-0.29	0.78

(Hi) Gradual Sloping Hearing Loss

The mean SNR and SD (±1) for gradual sloping hearing loss are shown in the Figure 4.8. Analysis of the t- test reveals a significant difference (indicated with asterisk) between NAL-NLl-Phonak ($p < 0.05$) and DSL[i/o]-Phonak ($p < 0.05$) for an input of 55 dB SPL. However, no significant difference is found among prescriptive procedures for an input level of 65 dB SPL (Table 4.3).

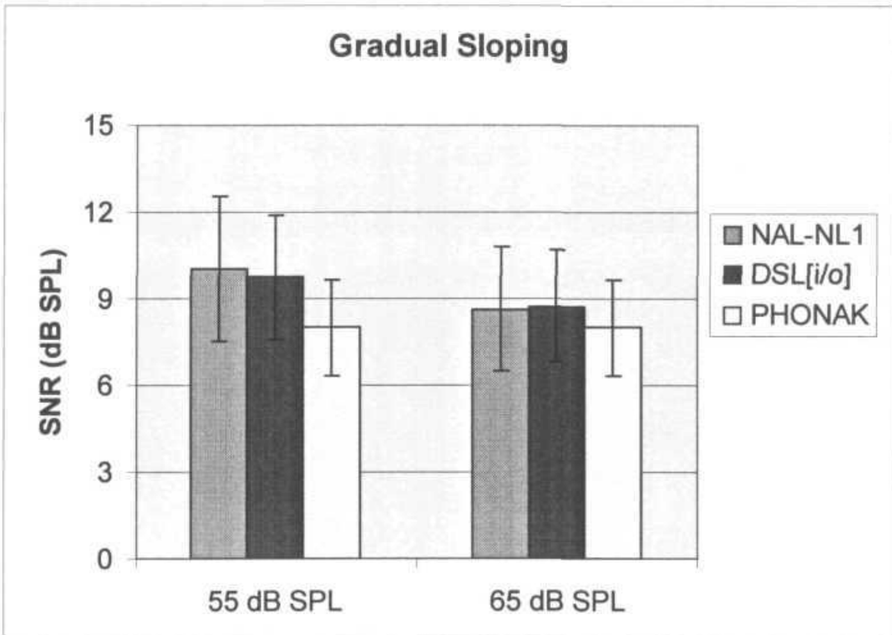

Figure 4.8. Mean SNR (dB SPL) for gradual sloping hearing loss at 55 and 65 dB SPL. Error bar shows ± 1 SD.

Table 4.3. Results of t- test at each input level for gradual sloping hearing loss.

Level (dB SPL)	Procedures	t-value	Significance
55 dB	NAL-NL1 &DSL[i/o]	-0.18	0.85
	DSL[i/o] & Phonak	2.49	0.04*
	NAL-NLl&Phonak	4.33	0.00*
65 dB	NAL-NL1 & DSL[i/o]	1.93	0.09
	DSL[i/ol & Phonak	1.10	0.30
	NAL-NLl&Phonak	1.36	0.21

Note: * significant difference at 0.05 level.

(iv) Steeply Sloping Hearing Loss

The mean SNR and SD (± 1) for steeply sloping hearing loss are shown in Figure 4.9. Analysis of the t- test revealed no significant difference among procedures at each input level (Table 4.4).

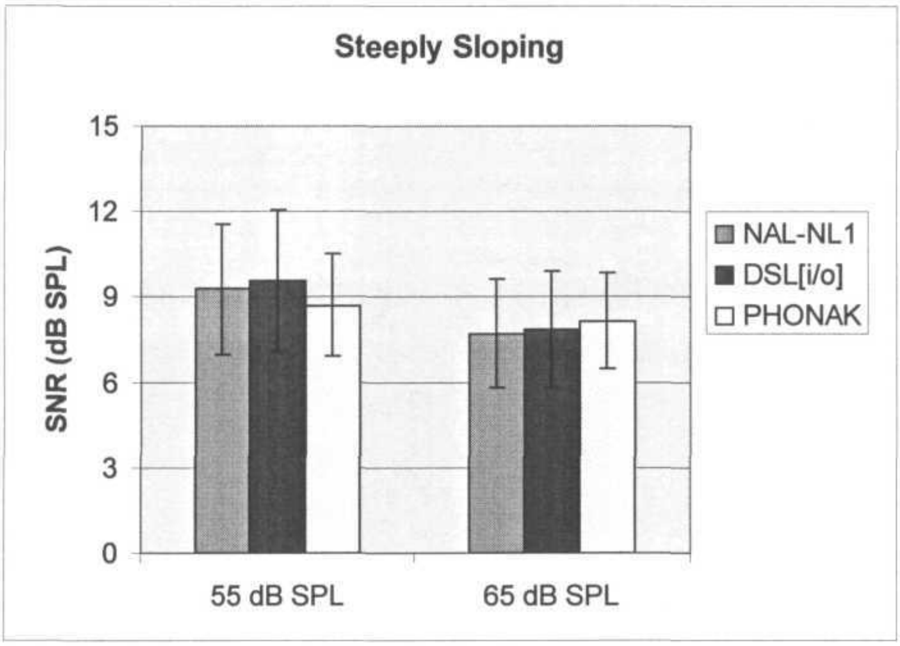

Figure 4.9. Mean SNR (dB SPL) for steeply sloping hearing loss at 55 and 65 dB SPL. Error bar shows ± 1 SD.

Table 4.4. Results of t-test at each input level for steeply sloping hearing loss.

Level (dB SPL)	Procedures	t-value	Significance
55 dB	NAL-NL1 & DSL[i/ol	-0.21	0.83
	DSL[i/o] & Phonak	1.68	0.14
	NAL-NLl&Phonak	0.65	0.53
65 dB	NAL-NL1 &DSL[i/ol	-0.12	0.90
	DSLfi/ol & Phonak	-0.22	0.82
	NAL-NLl&Phonak	-0.70	0.51

DISCUSSION

Speech Identification Scores Measurement

The result of SI scores measured in normal hearing listeners showed that as SNR decreased, the SI scores decreased .The same trend was maintained for both soft (55 dB SPL) and average (65 dB SPL) speech input levels. The results of present study showed agreement with that of Cox, Alaxender, and Gilmore's (1987) in which connected speech test (CST) scores were measured from -2 to -7 signal-to-babble (S/B) ratios. In the present study the SI scores obtained were 87.16% (at 0 dB SNR) and 61.69% (at -5 dB SNR) for 55 dB SPL. For an input of 65 dB SPL, SI scores were found to be 88.75% (at 0 dB SNR) and 64.90% (at -5 dB SNR). According to Cox et al. (1987) the CST scores for 0 dB S/B ratio were 100% and for -5 dB S/B ratio, the CST scores were approximately 63%. The difference in identification scores found in present study and that reported by Cox et al. (1987) could be attributed to the differences in speech sample and competing message used in both the studies. In the present study, only words were used for speech identification whereas, Cox et al. (1987) used passages and subject's had to repeat key words. This implies that, for average levels, when materials such as connected speech are used, they provide additional cues for perception as redundancy for these is higher. At low input levels and low SNRs, probably connected speech and words, both are equally degraded resulting in poor performance.

REIG Measurement

REIG measurement was carried out for NAL-NL1, DSL[i/o] and Phonak (digital/ski slope) prescriptive procedures for an input of 65 dB SPL. For individuals with moderate and moderately-severe flat hearing loss, all the three procedures prescribed nearly same amount of gain in mid frequencies. However, for low frequencies NAL-NL1 prescribed less gain than the other two procedures. The difference in high frequencies were more between DSL[i/o] and NAL-NL1, whereas, differences were found to be less between Phonak (digital/ski slope) and NAL-NL1. Also, the gain differences among procedures were relatively large for moderately severe flat hearing loss than for moderate flat hearing loss.

For sloping hearing loss configurations each procedure prescribed nearly similar gain at or below 3000 Hz. For frequencies above 4 kHz, DSL[i/o] provided maximum gain followed by Phonak (digital/ski slope) and NAL-NL1. The gain was markedly reduced for NAL-NL1 at higher frequencies for both gradual and steeply sloping hearing losses.

The differences in insertion gain for different prescriptive procedures and hearing loss configurations found in present study, can be explained based their rationales. As explained earlier NAL-NL1 is based on speech intelligibility maximization, whereas, DSL[i/o] is based on loudness normalization. NAL-NL1 prescribes relatively less low frequency gain for flat hearing loss configurations as relatively little low-frequency response is required to maximize speech intelligibility index (SII). This is true for both, soft and average levels of input. For sloping, especially steeply sloping hearing losses, NAL-NL1 prescribes less high frequency gain, as amplified signal at those frequencies can not contribute much to enhance speech intelligibility. This is partly due to limited capacity of hearing impaired individuals to use high-frequency information and partly because even maximal contribution of those frequencies to SII is relatively small (Byrne etal,2001).

The rationale behind the DSL[i/o] includes concept of extended dynamic range rather than normal dynamic range. This extended dynamic range is mapped on to the hearing- impaired dynamic range, thus, resulting in more gain frequency response than NAL-NL1 for -

all hearing loss configurations (Byrne et al, 2001).

The insertion gain frequency characteristics found in present study for NAL-NL1 and DSL[i/o], were in consonance with that reported by Byrne et al. (2001) and Keidser et al. (2003). However, for Phonak (digital/ski slope) fitting procedure, the gain frequency characteristics can not be explained on any rationale, since at present, there is no published report which describes the rationale for gain frequency response for Phonak (digital/ski slope) procedure.

SNR Measurement

SNR measurement for words presented in presence of speech noise, in individuals with SN hearing loss showed no significant difference among the three investigated fitting procedures, except for gradual sloping hearing loss at 55 dB SPL. Overall, Mean SNRs for soft input of speech were more than for average levels of speech. This is true because individuals with SN hearing loss need relatively more favourable SNR at softer levels of speech than when speech is presented at an average level.

SNR requirements for a given speech task were found to be highly variable ranging from approximately 5 to 13 dB for both, soft and average input levels. The same

was found for all hearing loss configurations. The variability found in SNRs was due to the fact that individuals with same degree and configurations of hearing loss can have different SNR requirements for a given speech task. This is because loss of outer hair cells (OHCs) causes loss of sensitivity, whereas, inner hair cells loss (IHCs) causes loss of clarity, especially in the presence of noise. Since it is difficult to measure IHC damage in individuals with SN hearing loss, the SNR loss can not be predicted based on individual's audiometric thresholds only. So individuals can have different SNR requirements even if the loss is same and vice-
versa (Killion & Niquette., 2000).

The significant difference among procedures obtained for gradual sloping hearing loss for soft level of speech could be due to the initial fit provided by the Phonak (digital/ski slope) procedure. It appears that the initial fit provided by Phonak (digital/ski slope) procedure more closely matched with preferred listening needs of the individuals with this configuration of hearing loss, which the other two procedures might have failed to do so in the initial fitting. However, the same results were not found for average input of speech for this configuration of hearing loss.

SNR dependency on REIG

All three prescriptive procedures prescribed different insertion gains for similar hearing loss configurations, but SNR measurement showed no significant difference on speech task for all studied hearing loss configurations. This means that even though there were differences in gain frequency response among fitting methods, individuals with SN loss needed nearly same SNR for a given speech task. Only for gradual sloping hearing loss a significant difference was found among procedures at soft input levels of speech. The difference can not be explained in terms of gain frequency characteristics, because for sloping hearing loss configurations DSL[i/o] prescribed maximum and NAL-NL1 prescribed least gain at higher frequencies. Whereas Phonak (digital/ski slope) prescribed more gain than NAL-NL1 but less gain than DSL[i/o] at higher frequencies. Also, the differences in high frequency gain among three procedures were large in steeply sloping hearing losses.

In cases of moderate and moderately-severe flat hearing losses, all the procedures prescribed nearly same amount of gain in mid frequency regions. Small differences in low and high frequencies gains did not result in favourable SNR for one procedure over the other. It seems that all the three procedures were able to restore nearly normal loudness and slight variation in gain did not significantly affect performance on SNR measurement. Thus from the present study, it is difficult to establish a relationship between gain frequency response and SNR requirement of individuals with SN hearing loss.

Further, in this study, high SNRs obtained across procedures for different hearing loss configurations were bound to occur as these were based on initial fit and not based upon preference by subjects or due to fine tuning. This was done mainly to investigate:

- How individuals perform for speech test in presence of noise, when fitted according to original gain frequency response prescribed by each prescriptive procedure.

- Whether theoretical prescriptions and hearing aid implemented versions of prescription differ and in turn affect individual performance in speech tests.

In this study, when original gain frequency responses were compared, the result showed that when individuals with different hearing loss configurations are fitted with original prescriptions (without fine tuning), the SNR requirement does not differ significantly for the same speech task.

It can be inferred from this study that original gain frequency response of investigated prescriptive procedures were preserved in the individual device of this manufacturer. The results of present study contradict with that of Smeds and Leijon (2001) in which the difference between theoretical and hearing aid manufacturer implemented gain frequency response were large. The reason could be attributed to the fact that the devices used in both of the studies were not same and result obtained for one device can not be generalized for all other devices.

It is of importance to mention here that sound quality judgment is also an important aspect of comparison among fitting methods. The present study did not carry out such a measurement as gain frequency responses were not optimized according to individuals needs, but were based upon initial fitting, which wouldn't have given them the accepted sound quality for all the three fitting methods.

Finally, the results of present study indicate that both proprietary and generic prescriptive procedures used in this study provide adequate gain frequency response. When individuals with different hearing loss configurations are fitted either with generic or proprietary prescriptions, nearly similar responses can be expected on speech tests in presence of competing noise.

CHAPTER5

SUMMARY AND CONCLUSIONS

SUMMARY AND CONCLUSIONS

The present study compared two generic (NAL-NL1 and DSL[i/o]) and one proprietary prescriptive procedure (Phonak digital/ski slope) implemented in a commercial digital hearing aid. The three prescriptive procedures were mainly evaluated on the basis of SNR requirement by the SN hearing loss individuals for a given speech task in presence of speech noise. In addition, REIG at an input of 65 dB SPL was also studied to compare gain frequency characteristics of proprietary fitting method with other two generic fitting methods. Two groups of individuals, normal hearing and hearing impaired individuals with different hearing loss configurations participated in this study. To evaluate the benefit of three prescriptive procedures a word list was prepared that included a combination of high, low and mid frequency speech sounds. The study was carried out in two phases.

In the first phase, individuals with normal hearing had undergone speech identification test in presence of speech noise at soft (55 dB SPL) and average (65 dB SPL) input levels. The SI scores were measured at 0 dB and -5 dB SNR conditions.

In the second phase, individuals with SN hearing loss were fitted with the digital BTE hearing aid. The hearing aid was programmed with each of the three prescriptive procedures. REIG was measured with digi-speech signal at an input of 65 dB SPL followed by SNR measurement at soft (55 dB SPL) and average (65 dB SPL) input of speech. The same process is repeated with the hearing aid being programmed with each of three prescriptive procedures. The main findings of the present study were:

- In normal hearing individuals, the SI scores decreased as the SNR became poorer.

- REIG measurement with 65 dB SPL input for three prescriptive procedures showed that-

  NAL-NL1 prescribed less low frequency gain for flat hearing loss configurations and less high frequency gain for sloping, especially for steeply sloping hearing loss configurations. Whereas, DSL[i/o] prescribed maximum gain across frequencies and this trend was seen for all hearing loss configurations. Phonak (digital/ski slope) prescribed gains some what similar to DSL[i/o]. However, overall gains were found to be less than DSL[i/o] across frequencies. The difference in gain frequency characteristics for NAL-NL1 and DSL[i/o] are based on their rationale. Gain frequency characteristics for Phonak (digital/ski slope) can not be explained here on any rationale, as no published report is presently available that describes gain frequency characteristics or rationale used for this proprietary procedure.
- SNR measurement in aided condition for SN hearing loss showed no significant difference among three prescriptive procedures except for gradual sloping hearing loss configuration at soft input. Phonak (digital/ski slope) provided least SNR than the other two generic procedures at 55 dB SPL. This means that when individuals were fitted with this prescriptive procedure, required relatively less favourable signal to noise ratio to perform on a given speech task at soft input level. The difference found for gradual sloping loss at soft input could be due to initial fit provided by this proprietary fitting method.

Finally, it can be inferred from the results of this study that both generic and proprietary prescriptive procedures, when evaluated based on speech task, give similar results in spite of differences in gain frequency response. However, small differences found could be overcome by fine tuning of the device based on individual needs.

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APPENDIX

APPENDIX-A

FIG6

FIG6 gets its name from the Figure 6 of the article in which the underlying data was first outlined by Killon and Fikret-Pasa (1993). According to Killon and Fikret-Pasa (1993) the SN hearing losses can be differentiated from Type I to III based on loudness growth functions. Type I and Type II losses exhibit complete or partial recruitment whereas Type III may or may not show recruitment. Generally, Type III losses are characterized by their intelligibility growth functions, which show that individuals must operate their hearing aids near discomfort levels in order to understand speech in difficult listening situations. For all of these losses, gain requirements should be in such a way that enhances speech intelligibility. In FIG6 procedure, the gain is directly prescribed for each of the input levels of 40, 65 and 95 dB SPL, and is inferred for other levels by interpolation. The FIG6 specifies how much gain is required to normalize loudness, at least for medium and high input levels. Unlike other procedures, it is not based on individual measure of loudness rather it is based on data averaged across a large number of people with similar degrees of threshold loss. This simply means that only hearing thresholds are needed to calculate the required gain.

For low-level (40 dB SPL) input signals, the gain is prescribed on the basis that people with mild or moderate hearing loss should have aided threshold 20 dB above normal hearing threshold. Providing gain more than this would simply make background noise louder and will not increase speech intelligibility significantly. Therefore, except for the first 20 dB of hearing loss, every extra decibel of hearing threshold loss is therefore compensated by an additional decibel of gain. This rule is modified to a half-gain or one-third rule once the unaided threshold exceeds 60 dB HL because providing high gains are likely to cause feedback oscillation.

For comfortable level (65 dB SPL) input signals, the amount of gain prescribed for any degree of hearing loss is equal to the average elevation of MCL for hearing loss, above MCL for normal hearing, using data by Pascoe (1988). With this amount of insertion gain, narrow band sounds perceived as comfortable by a normal-hearing person will also be perceived as comfortable by the hearing-impaired aid wearer.

For high-level (95 dB SPL) input signals, the gain requirements for hearing-impaired individuals are not large. In these cases, little amount of gain is usually sufficient to restore loudness similar to required by normal-hearing listeners. This finding was based upon studies by Lippman et al. (1981) and Lyregaard (1988).

As the gain needed to normalize loudness depends only on hearing threshold and input level, the same formula applies at all frequencies. The FIG6 software performs calculation for required coupler response and compression ratios for high and low input levels at high and low frequencies.

FIG6 formula

For 40 dB SPL input levels:

$IGi = 0$ for $Hi < 20$ dB HL

$IGi = Hi - 20$ for $20 < Hi < 60$ dB HL

$IGi = 0.5\ Hi + 10$ for $Hi > 60$ dB HL

For 65 dB SPL input levels:

$IGi = 0$ for $Hi < 20$ dB HL

$IGi = 0.6\ (Hi - 20)$ for $20 < Hi < 60$ dB HL

$IGi = 0.8\ Hi - 23$ for $Hi > 60$ dB HL

For 95 dB SPL input levels:

$IGi = 0$ for $Hi < 40$ dB HL

$IGi = 0.1\ (Hi - 40)^{1.4}$ for $Hi > 40$ dB HL

IHAFF/Contour

The Independent Hearing Aid Fitting Forum (IHAFF) is the name of a group of individuals who attempted an approach for selection and fitting of non-linear devices (Cox, 1995). Later, the fitting procedure itself was called as IHAFF, which can be applied to any hearing aid with adjustable wide dynamic range compression. Since persons even with same hearing loss often do not give same loudness judgment for a given sound, a standard procedure was needed to measure loudness perception that gives good test retest reliability across professionals and test sessions (Cox, 1995). The IHAFF procedure is based upon loudness scaling method that normalizes loudness at each frequency. The particular loudness scaling method used is called the Contour test. This test determines the levels of pulsed warble tones that correspond to each of seven loudness categories (Table 1).

Table A. 1-Seven categories of loudness used in Contour Test

7- Uncomfortable loud
6- Loud but okay
5- Comfortable, but slightly loud
4- Comfortable
3- Comfortable, but slightly soft
2-Soft
1-Very soft

These seven categories of loudness obtained from normal hearing individuals were compared with three broad regions of loudness perception for warble tones. These three broad regions of loudness; soft, average and loud were used as they represent daily life speech inputs. In IHAFF/Contour protocol, the Visual Input/Output Locator Algorithm (VIOLA) software program presents gain, maximum output and the results of the loudness normalization as three points, on an input-output function at frequencies where loudness scaling is carried out. These three points show the output levels needed to normalize the loudness of 1/3-octave bands of speech, when the complete speech signal is at the levels needed for normal-hearing people to rate its loudness as soft, average, and loud respectively (Cox, 1995).

CAMEQ

The Cambridge procedure for loudness equalization (CAMEQ) was introduced by Moore and colleagues (Moore, Alcantara, Stone, & Glasberg, 1999a). The procedure aims to place as much of the speech spectrum as possible above absolute threshold for a given overall loudness. This is achieved by amplifying speech, so that on average, the loudness is the same for all frequencies within the range 500-5000 Hz; this is the range that is most important for speech intelligibility (Moore et al., 2001). More specifically, the goal is to make the loudness the same in each critical band (Moore & Glasberg, 1997). This goal can also be described as amplifying speech so as to give a 'flat' specific loudness pattern for frequencies from 500 to 5000 Hz (Moore, Glasberg, & Stone, 1999b). The same rationale is behind the NAL-RP procedure for the fitting of linear hearing aids (Byrne & Dillon, 1986). However, with linear hearing aids the goal can only be achieved with for a narrow range of input sound levels. The CAMEQ procedure differs with NAL-RP in a way that this procedure aims to achieve gain frequency responses for a wide range of input sound levels. It also aims to give about the same overall loudness as normal for speech over a wide range of sound levels (Moore et al., 2001).

CAMREST

The Cambridge procedure for loudness restoration (CAMREST) determines the gains needed to restore loudness perception to 'normal' for speech-like stimuli (Moore, 2000). This means not only restoring the overall loudness to normal, but also making the relative loudness of different frequency bands the same as 'normal'. More specifically, the procedure aims to give 'normal' specific loudness patterns (i.e., the same loudness as normal in each critical band) for speech over a wide range of input sound levels (Moore et al., 2001). Restoration of loudness in this way is described by several researchers (Killon & Fikret-Pasa, 1993; Cox, 1995).

Generally speaking CAMEQ tends to prescribe more gain than CAMREST procedure at medium frequencies (i.e., at 1 and 2 kHz) and less gain than CAMREST at low and high frequencies. So, for an individual hearing loss both the procedures produce significantly different frequency-gain characteristics, especially for low to medium (50-65 dB SPL) sound input levels (Alcantara et al., 2004). The CAMEQ and CAMREST procedures are implemented in a computer programme called Camfit. In principle, the two procedures can be applied to any multichannel compression hearing aid and they are not limited to any specific hearing instrument. These can be applied to any device having a number of channels up to 20. It is essential to specify number and bandwidth of the channels when appropriate amplification is given for broadband sounds such as speech by recommending gains measured using sinusoids or narrow bands of noise (Moore et al., 1999b; Moore, 2000). The Camfit programme determines the gains that should be applied in each channel, or at the standard audiometric frequencies, for sinusoids with various input signals. Gains are specified both as insertion gains and real-ear aided gains (Moore etal.,2001).

APPENDIX-B

Word list with a combination of low-mid, low-high and high-mid frequency speech sounds

Low-Mid	Low-High	High-Mid
1) /gu:be/	/nalli/	/tʃa:ku/
2) /me:ke/	/sɛ:bu/	/ko:Li/
3) /bi:ga/	/mola/	/la:ri/
4) /mu:gu/	/bassu/	/d̪a:ra/
5) /rave/	/baLe/	/kivi/
6) /kaNNu/	/d̪ana/	/tʃikka/
7) /ni:ru/	/tʃindi/	/i:ruLLi/
8) /mara/	/ni:vu/	/kuTTu/
9) /kone/	/mi:se/	/tʃakra/
10) /pu:ri/	/t̪inDi/	/ʤinke/
11) /bekku/	/haNa/	/raʤa/
12) /ganTe/	/suma/	/si:re/
13) /ru:pa/	/biLi/	/gaηTe/

14) /nid̪re/	/tand̪e/	/kat̪t̪i:/
15) /kabbu/	/tʃenDu/	/giNi/
16) /magu/	/d̪o:Ni/	/vitʃa:ra/
17) /kappu/	/ʤi:pu/	/se:ru/
18) /bi:ru/	/To:pi /	/ko:ti/
19) /na:ri/	/bila/	/tʃikka/
20) /mu:ru/	/ ba:vi /	/rutʃi/
21) /kemmu/	/ni:li/	/suk^{h}a/
22) /pad̪a/	/baTlu/	/i:ruLLi/
23) /ravi/	/d̪i:pa/	/kelasa/
24) /reppe/	/Dabbi/	/katte/
25) /buguri/	/ hind̪e/	/KuLLi/
26) /kombe/	/ivanu/	/roTTi/
27) /ra:Ni/	/bi:dza/	/ko:su/
28) /ma:rga/	/baTTe/	/iruve/
29) /pennu/	/moLe/	/sari/

30) /gamana/	/tamma/	/guDi/
31) /rama/	/meTlu/	/gedʒdʒe/
32) /be:ru/	/beTTa/	/railu/
33) /maŋga/	/me:dʒu/	/rasa/
34) /guNa/	/ba:Le/	/ka:su/
35) /pa:naka/	/no:vu/	/ke:Lu/
36) /kappe/	/bassu/	/kelavu/
37) /nu:ru/	/ma:tre/	/tʃakli/
38) /gombe/	/noDu/	/kaDDi/
39) /ramja/	/haNNu/	/ka:fi/
40) /nuŋgu/	/beTTa/	/go:De/