EVALUATION OF PERFORMANCE WITH

LINEAR Vs. NON-LINEAR HEARING AIDS

Register No. M 9903

An Independent Project submitted as part fulfillment for the first year M.Sc. (Speech and Hearing) to the University of Mysore

ALL INDIA INSTITUTE OF SPEECH AND HEARING, MYSORE - 570 006

MAY, 2000

Certificate

This is to certify that this Independent Project entitled "EVALUATION OF PERFORMANCE WITH LINEAR Vs. NON-LINEAR HEARING AIDS" is the bonafide work in part fulfillment for the degree of Master of Science (Speech and Hearing) of the student with Register No.M 9903.

Mysore,
May, 2000

Director
All India Institute of
Speech & Hearing
Mysore - 570 006

Certificate

This is to certify that this Independent Project entitled "EVALUATION OF PERFORMANCE WITH LINEAR Vs. NON-LINEAR HEARING AIDS" has been prepared under my supervision and guidance.

Manjula. P

Mysore,
May, 2000

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Declaration

This Independent Project entitled "EVALUATION OF PERFORMANCE WITH LINEAR Vs. NON-LINEAR HEARING AIDS" is the result of my own study under the guidance of Mrs. Manjula, P., Lecturer, Department of Audiology, All India Institute of Speech and Hearing, Mysore, and has not been submitted earlier in any other University for any other Diploma or Degree.

Mysore, Register No. M 9903

May, 2000

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INTRODUCTION

Selection of an appropriate amplification device with the appropriate characteristics has long been a challenge for the hearing health care professionals. The aim of the hearing aid selection is to provide the person with compromised hearing with a means to compensate for the physiologic deficit and avail of auditory sensation as near to normal perception as possible (Dempsey, 1994).

Various audiologic factors determine the choice of hearing aid characteristics of which, dynamic range is of prime importance. Dynamic range is referred to as the range of usable hearing or the range of comfortable loudness for speech (Martin, 1986). The dynamic range is the decibel range between the threshold for tones or speech and the level at which the stimulus (tones or speech) becomes uncomfortably loud. The dynamic range, thus, represents the target area for amplification and the hearing aid fittings must be based on these measures. The closer the amplification envelope can fit consistently into the patients' dynamic range during hearing aid use, the better the fitting.

The dynamic range is usually much reduced in individuals with sensorineural hearing loss. Elevated thresholds will necessarily shift the lower be significantly lowered as a result of recruitment. The narrower the dynamic range, the greater the recruitment and the more complicated is the hearing aid fitting process (Mynders, 1985).

Recruitment is an abnormal growth of loudness for signals at supra-threshold intensities. The phenomenon of recruitment was reported by researchers in 1930s and the earliest definition was by Fowler (1936) who defined loudness recruitment as:

"....the greater change (or the difference) in the increment of loudness in relation to stimulus increase observed in ears with a partial neural loss of hearing as compared with normal ears or ears with only impedance lesions."

Loudness recruitment may be described as the progressive reduction of hearing impairment as the input level of sound increases. It occurs as a result of hair cell damage. The outer hair cells (OHC) probably act as amplifiers to the inner hair cells. For low amplitude acoustic signals, they act as some form of mechanical or electrical amplifier. On the other hand, for high amplitude sounds, they act as a compressor. In a recruiting ear, the OHC damage leads to cessation of the compressor action for loud sounds. This results in what is popularly called the "expanding action" of recruitment.

The expanding action of recruitment has a distorting effect on the loudness relationships among speech sounds. Simulation experiments by Villchur (1974) proved conclusively that recruitment in deaf subjects is a sufficient cause for loss of speech intelligibility, whether or not there were other causes. To compensate for the expanding action, which is a result of lack of compression by the OHCs, we need to have an external compression circuit in the hearing aid i.e., this circuit should amplify the low intensity acoustic signals and compress the high amplitude sounds as required. These modifications in the hearing aid performance can be achieved by means of output limiting (peak clipping or compression) or by frequency response shaping. In our endeavour to redeem the absence of compressor action by damaged OHCs, we must also keep track of the effects of such signal processing methods on speech intelligibility and discrimination.

Villchur (1974) was one of the earliest researchers to suggest the use of amplitude compression to compensate for recruitment and provide better intelligibility. However, there are only a few experimental descriptions of the preocedural aspects to optimize the adjustment of the output limiting function (Dreschler, 1988. Some studies (Dreschler, 1988; Fabry and Olson, 1991; and others) report of no significant differences in speech discrimination for linear vs. full-dynamic-range-compression aids at different compression ratios. The results of the few other studies (Hudgins, et al., 1948; King and Martin, 1984; and others) indicate that the use of output limiting hearing aids reduces distortion and provides better performance over a wide range of speech input levels.

Present day hearing instruments have a choice as to the amount of output limitation to be applied. This usually refers to the compression threshold, or the kneepoint and sometimes to the compression ratio. The degree of output limitation (usually the kneepoints) are marked as 0%, 25%, 50%, 75% and 100% peak clipping/compression. This feature provides more flexibility regarding the amount of output limitation to be applied to the signal. The hearing health care professional is thus able to provide hearing aids which fit into the dynamic range of each hearing impaired individual more accurately, ensuring better patient satisfaction.

To arrive at a perfect balance between satisfying the amplification needs and an adequate degree of speech intelligibility (i.e., as near normal perception as possible), is the aim of the hearing aid selection procedure.

The present study was undertaken with an intention of comparing the effects of various levels of output limiting on the speech perception / discrimination and simultaneously monitor the efficacy of various levels of output limiting for preventing over amplification; thereby reducing tolerance problems.

REVIEW OF LITERATURE

Hearing is a complex multi-step process. A sound wave hitting against the eardrum is converted into mechanical energy by vibration of the ossicles. The mechanical energy is then transferred to the inner ear where it is transformed into bio-electrical energy by the hair cells.

Depending on the intensity of the input sound signal, the outer hair cells perform some form of differential mechanical or electrical amplification. The low intensity acoustic inputs are amplified as much as 10,000 X, but the high intensity sounds are amplified to a much lower extent, viz. 10X. (Ruggero, 1992). Thus, the outer hair cell mechanism provides amplification of low amplitude acoustic input and compression of high amplitude inputs.

The complex speech signals are also subjected to such a differential amplification. The speech signals are composed of low frequency high intensity vowels and high frequency, low intensity consonants. The differential amplification of the low and high intensity sounds by the hair cells, especially the outer hair cells, helps us to perceive clearly both, the high intensity vowels and the low intensity meaning-bearing consonants.

The normal, non-linear amplification process is disrupted when the OHCs sustain damage. This leads to inadequate compression of high amplitude sounds and, in turn, an exaggerated loudness difference among the acoustical elements of speech. This distorted loudness relationship is a characteristic of hair cell loss and has been termed as 'loudness recruitment'. A person with loudness recruitment may be very deaf to weak sounds and progressively less deaf to more intense sounds, until in cases of 'complete' recruitment, at some high level, the listener with impaired hearing has the same loudness response as that of a listener with normal hearing. (Villchur, 1996). This is clear from the diagram shown in Figure 2.1:

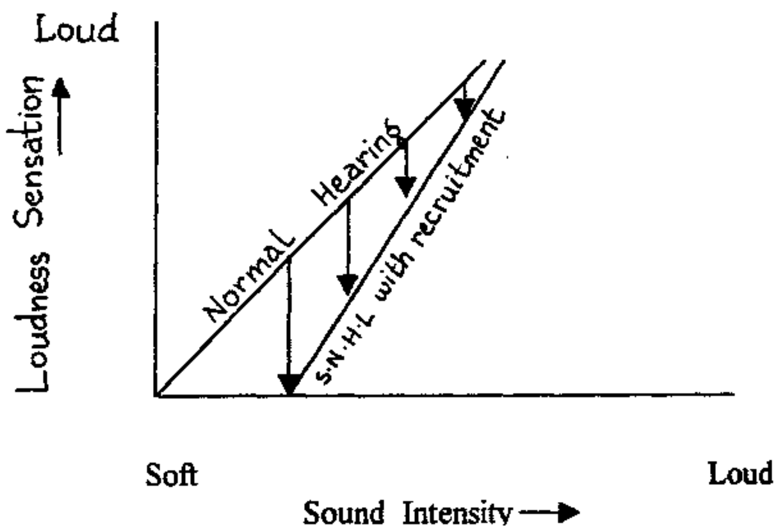


Fig. 2.1: Loudness Responses of individual with normal hearing and person with SNHL (with recruitment), [adapted from Smriga,1993)

Thus, in loudness recruitment, as the input levels increase, the degree of impairment progressively decreases. When the ear of the person exhibiting loudness recruitment is stimulated with loud input sounds, no hearing loss is evident at all. Hence, Libby (1994) describes recruitment as, "not abnormal growth of loudness, but as abnormal loss of sensitivity for quiet sounds".

The phenomenon of recruitment was first reported by Fowler (1936) before the section of Otology of the New York Academy of Medicine. He defined loudness recruitment as-

"Loudness recruitment is the greater change (or the difference) in the increment of loudness in relation to stimulus increase observed in ears with a partial neural loss of hearing as compared with only impedance lesions".

Various theories have been propounded to explain this hair cell phenomenon of recruitment. Most of the initial theories presumed the involvement of nerve fibres as the cause of recruitment. (Steven, 1936; Salvi et al., 1983; Evans, 1976, Kiang et al., 1970). These theories explained loudness recruitment using concepts like steepening of the slope of the function which relates the neural discharge rate to intensity and broadening of tuning curve for high intensity sounds. Though these models seemed logically sound, they did not conform to psychoacoustic and physiologic data. Later research put forth the view that both, the hair cells and the nerve fibers were involved in the sound processing in a recruiting ear (Lorente de No, 1937; Lurie, 1940; Simmons and Dixon, 1966). The more recent theories postulate the involvement of hair cell damage in an attempt to explain the phenomenon of recruitment. (Tonndorf, 1980, 1981; and Killion, 1996). These newer theories explain the physiology of the damaged outer hair cell and are more successful in explaining the expanding action of recruitment. Their explanations concur with the psychoacoustic and physiologic data.

Having described a brief overview of the nature of loudness recruitment, this phenomenon will be further discussed under following heads:

I Effect of loudness recruitment.

II Implications for hearing aid fitting and options available.

(1) Suppression of noise

(2) Output limiting by peak clipping or amplitude compression

III Comparison of peak clipping and compression circuits.

I <u>Effects of loudness recruitment:</u>

The disproportionate loudness growth, which is a hallmark feature of recruitment has two major effects. They are:

(1) Reduced dynamic range.

(2) Distorted speech perception.

(1) *Reduced dynamic range:* Patients who have outer hair cell loss have audiograms that fall between 20 and 75 dB HL (Berlin et al., 1996). Thus, as a result of hearing loss, the lower limit of the dynamic range is

raised. As a result of recruitment, with increase in intensity of sound, loudness perception reached near normal or normal levels. Hence, the dB HL values of the discomfort level of sound will be the same as those of normal hearing individuals (i.e., 90-110 dB HL) as against expected discomfort levels of > 60 dB SL.

This raising of the lower limit and lowering of the upper limit leads to a severely reduced dynamic range. The narrower the dynamic range, the greater is the degree of recruitment.

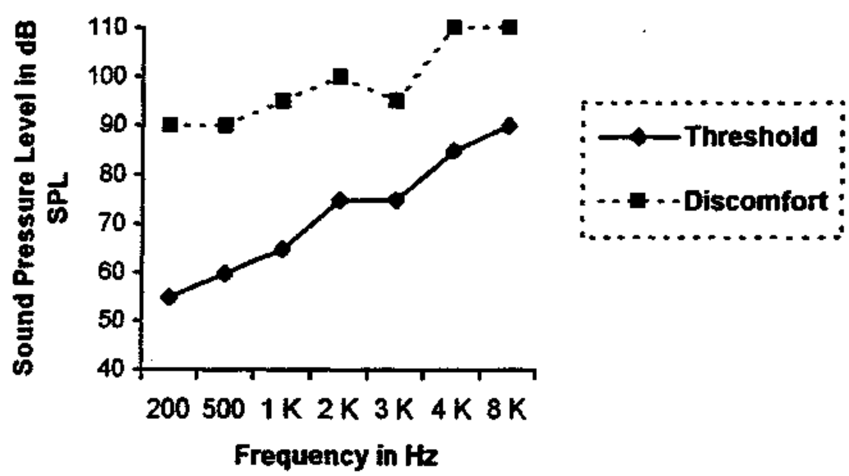


Fig 2.2: Average thresholds and level of discomfort for a person with SNHL and recruitment. [Adapted from Villchur, 1973].

As we can see from the Figure 2.2 above, this reduced dynamic range is not uniform throughout the frequency spectrum. The dynamic range is more narrow in the mid- and high-frequency region, where most of the information bearing elements of speech, i.e., the consonants are located. This could be because of two reasons. Firstly the hearing loss is more at the high frequency regions, and secondly, the loudness growth is steeper at higher frequencies.

The dynamic range of speech requires a hearing range of 45 dB to 50 dB. The range of a normal ear is 100 to 120 dB (i.e., threshold of 0-10 dB HL to a Discomfort level of 100-120 dB HL), but in the impaired ear with sensorineural hearing loss, it is only 15 to 30 dB HL (i.e., threshold of 40-70 dB HL and a discomfort level of < 100 dB HL) [Zelnick (1994)].

(2) *Distorted speech perception :* Fowler (1936) thought of recruitment as an ameliorating factor in hearing impairment, i.e., with decrease in impairment proportional to increasing intensity there would be a lesser degree of handicap due to the hearing loss. But the expanding action of recruitment when applied to the speech spectrum can present a distorted auditory image of the speech. The energy present in the vowel components is significantly greater than the acoustic energy for many of the consonant sounds. Hence, the intense vowel components can be detected with much greater ease than the weak low intensity consonant sounds when the dynamic range is reduced. These distorted loudness differences among the various acoustic speech units comprising the signals leads to a reduced discrimination. Villchur (1974) carried out an experiment to simulate the effect of recruitment on loudness relations in speech. He reported that recruitment was a sufficient cause for loss of

intelligibility in the hearing impaired, whether or not there are other causes.

II <u>Implications of loudness recruitment for hearing aid fitting</u>:

Recruitment is a perplexing problem both for the physician and for the patient. The disproportionate growth in loudness, when compared to normal ear function, presents a serious problem to the proper selection and fitting of hearing aid devices (Schiff and Sandlin,1982). The variations in the presenting picture of problems, with respect to degree of recruitment and its frequency distribution further complicates the issue. As early as in 1937, Steinberg and Gardner understood the implications of recruitment for amplification for hearing impaired persons. They suggested that owing to the expanding action of the hearing loss following hair cell damage, it would be necessary to introduce a corresponding compression in the amplifier. Villchur (1974) also emphasized that compensation for the loudness recruitment is a necessary although possibly insufficient condition for restoring speech intelligibility. This compensation should be in the form of taking the larger dynamic range of speech and fitting it into the smaller dynamic range of the subject. This will require a non-distorting, decreasing gain system with increasing input i.e., an aid which subjects the speech signals to amplitude compression, more at high frequency than at low frequencies.

The linear amplification devices are inefficient in meeting these demands. This is clearly understood from the Figure 2.3:

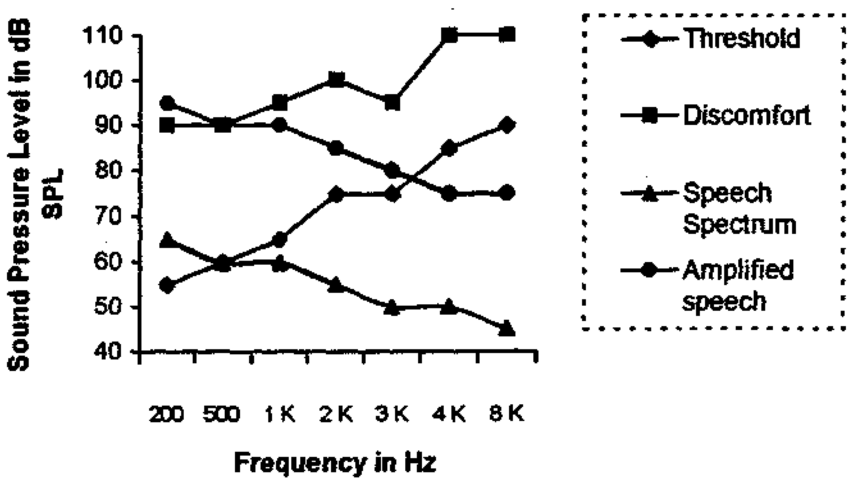


Fig 2.3: Linear amplification of speech spectrum relative to dynamic range of person with recruitment [Adapted from Villchur. 1974).

The linear aid gives uniform amount of amplification across all the frequencies. If we give sufficient gain so as to raise the low frequency sounds (vowels) into the region of audibility, this amount of gain is insufficient to reach even the threshold of audibility at the higher frequencies, where the meaning bearing consonants are situated. If the gain is increased such that the amplified speech in the high frequency low intensity region is within the region of audibility/threshold, a corresponding gain in the low frequency, high intensity region will result in intensities beyond the discomfort levels.

This expansion of the loudness of vowel peaks and loss of perceived consonant vowel relationships in speech should be countered by better compensatory circuitry of the hearing aid (Villchur, 1974). Literature suggests the following methods of compensating for recruitment:

1 Suppression of noise
2 Output limiting by
   (a) Peak clipping
   (b) Amplitude compression.

Sinsle channel

Mum' channel

1) Suppression of noise : Hearing impaired persons with sensorineural involvement typically experience a reduction in speech intelligibility in noise-A reduction in number of usable cochlear hair cell density can predispose the impaired cochlea to processing saturation. This may leave few, if any, remaining hair cells available to process signal information in the presence of noise. This 'busy-line' effect further deteriorates the impaired person's ability to understand speech in noise (Smriga, 1993). Also, there appears to be an efferent neural mechanism in the brain that can assist the cochlea in perceiving signal in the presence of background noise. This natural signal processing advantage is triggered only when the brain is aware of a 'speech-in-noise' input condition. If the impaired cochlear function is unable to transmit this key data to the brain due to the 'busy line' effect, such natural enhancement may not be

triggered in the sensorineurally impaired ear (Smriga, 1993). Furthermore, Smriga, suggested that to maximize the aided performance when fitting patients with sensorineural hearing loss, it is important that the amplifier selected is designed to keep the output level within the user's restricted dynamic range, thus minimizing the busy line effect. He also suggests that the amplifier must maintain the input signal-to-noise ratio available for the normal hearing ear to process, thus triggering the olivocochlear bundle suppression whenever possible.

Tillman, Carhart and Olsen (1970) demonstrated that the hearing aid user required better S-N ratio to understand speech with a hearing aid than without the aid. This finding conforms to the results of Villchur's (1973) study. He compared the speech recognition scores of six hearing impaired subjects under three conditions: linear hearing aid with voice interference at -10 dB; linear hearing aid with interference removed and compression amplification with voice interference at -10 dB. He found that the speech recognition scores for the linear amplification in quiet was higher than the scores in noise. Hence, it seems logical to try to develop an electronic circuit, designed to suppress the noise relative to the speech. However, constructing such a circuit with selective noise suppressing properties is a complicated task. The circuits designed to suppress noise may typically sacrifice rather than restore speech cues.

2) Output limiting:

(a) *Peak Clipping :* Peak clipping can occur when a linear amplifier reaches saturation. Peak clipping occurs when part of the signal wave overshoots the capacity of the amplifier. The part of the wave that exceeds the amplifier limits is cut-off from further signal processing.

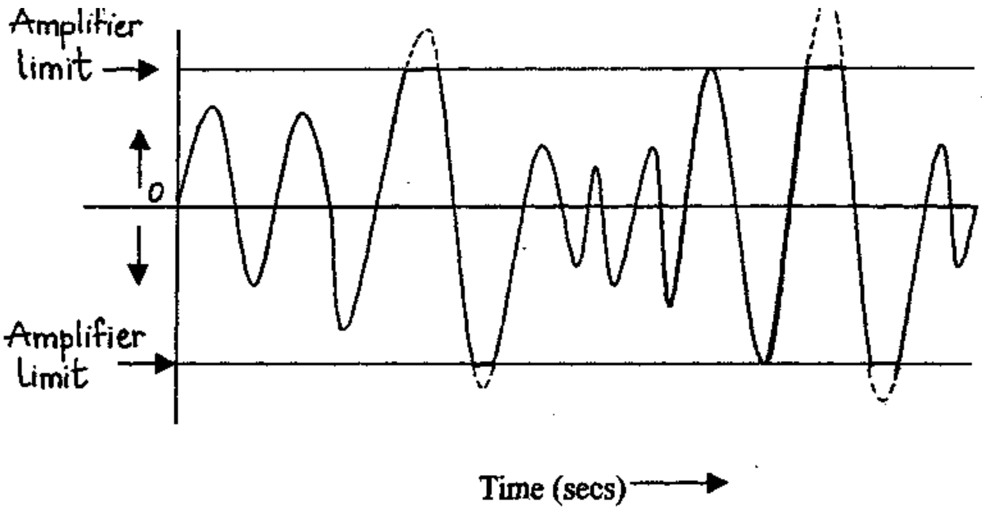


Fig 2. 4: Effect of peak clipping on wave moiphology.

The effect of peak clipping on wave morphology has been shown in Figure 2.4.

Peak clipping may be used as an output limiting strategy by adjusting the amplifier output limit to within the wearer's restricted dynamic range. Thus, the peak clipping may be used to control the output dynamic range of the linear amplifier.

As a result of peak clipping, there is a change in the wave morphology. After peak clipping, the input waveform is flattened or squared. This results in a physical distortion of the signal because its high amplitude elements are now restricted relative to its low amplitude components. This results in a reduction in the sound quality e.g., In a condition where favourable signal to noise ratio exists within the input signal, the first components of this input to be clipped off are the higher amplitude signal components. This disturbs the S-N ratio and leads to distortion. Also, the loudness relations between vowels and consonants are not maintained as a result of peak clipping.

However, unlike the popular misconception that reduced sound quality (as a result of peak clipping) leads to reduced speech intelligibility, studies show otherwise. Olsen (1971) reviewed the early literature, and found suggestions that speed intelligibility is not disrupted appreciably by peak clipping and its resultant harmonic and intermodulation distortion. Staab, (1972) indicated that even excessive peak clipping does not significantly decrease speech discrimination ability. Both these reports indicate that there is no great reduction in speech intelligibility when peak clipping is used as an output limiting device, through speech quality may deteriorate.

These findings may be explained on the basis of the acoustic spectra of speech sounds. Vowel sounds are lower in frequency and higher in intensity and are the first components to be susceptible clipping. The low intensity consonants are relatively unaffected by the effects of peak clipping. Thus, though the sound quality and loudness relationships in speech are distorted, speech intelligibility should not suffer much as the meaning carrying consonants are relatively unaffected.

(b) *Amplitude compression'.* In a peak clipping aid, there is a one-to-one relationship between the dB change in input and the resultant dB change in output until the aid reaches saturation. Once the input plus gain exceeds the maximum output limit of the amplifier peak clipping occurs. This results in harmonic distortion. In addition, some of the information contained in the input signal are not present in the output signal as a result of this peak clipping. Thus, signal fidelity is not maintained. In order to minimize the distortions in signal resulting from peak clipping, a non-linear amplification circuitry may be employed. This is also called the 'compression circuitry', and an automatic monitoring circuit that reduces gain as a function of level of signal that is amplified is called the Automatic Gain Control or AGC.

Such a system acts to decrease gain as a function of the input level to prevent the output signal from reaching saturation when confronted with high inputs. Unlike in peak clipping all of the input content is retained. The non-linear performance structure of the compression amplifier can deliver a more natural loudness growth perception throughout the patient's auditory listening range without under-amplification or over-amplification compromises. This can be graphically represented as shown in Figure 2.5.

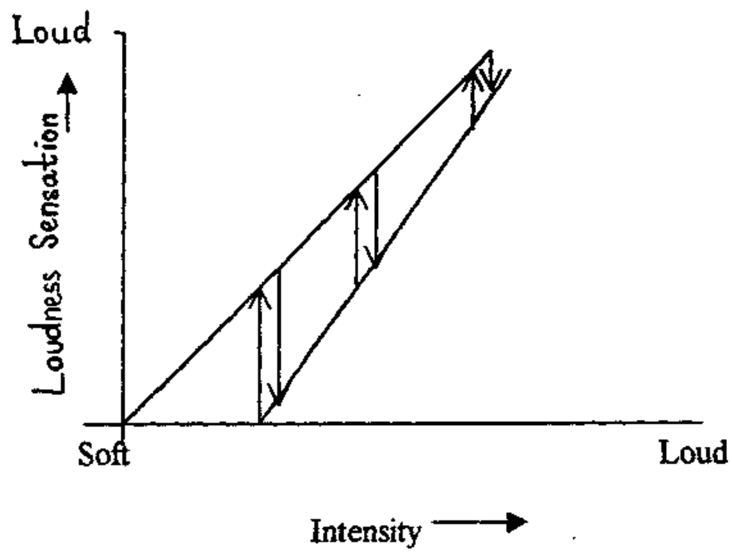


Fig 2.5: Graphical representation of compensatory action of compression amplification for 'expanding action' of recruitment. [Adapted form Smriga. 1993].

Output limitation may be achieved by specifying the compression threshold or the kneepoint. The compression threshold is the point on the input-output function curve, where the gain first deviates from the one-to-one gain relationship. The compression threshold levels are most often employed to achieve compression limiting (a substitute for peak clipping). Depending upon the discomfort levels of the subject, the compression threshold is set such that the aid goes into compression just prior to saturation. The aid may have a high kneepoint if it serves the function solely of compression limiting. In case of dynamic range compression (DRC) a low keneepoint (eg., 65 dB SPL or lower) is employed so that at least a portion of the dynamic range of speech is in compression. A relatively lower kneepoint (eg., 50 dB SPL) is used in a wide dynamic range compression (WDRC) circuit, so that compression occurs over a wider range of inputs. In case of a full dynamic range compression (FDRC) circuit, the entire dynamic range of speech is in compression and in such a case, the kneepoint will be as low as 45 dB SPL (or less).

The kneepoints should be carefully chosen as per the requirements of the subject based on his dynamic range. Fabry and Olsen (1991) carried out a study to judge subject preference for FDRC vs. linear compression Iimiter over a period of one month. They reported no subject preference either objectively or subjectively. Caraway & Carhart (1967) had reached a similar conclusion after they had attempted to improve speech understanding by using amplitude compression. Dreschler (1988) studied the effects of specific compression threshold settings on phoneme identification and found a small but significant increase in identification scores for the lowest setting of compression threshold. More recently, Barker et. al., (1999) have reported that a higher compression threshold was preferred by a majority of his subjects. From these studies, it is apparent that compression limiting does not provide much benefit from point of view of improved speech perception. But, limiting the output power by means of compression amplification is both feasible and desirable, as it not only protects the ear, but at the same time reduces distortion to a minimum, thus maintaining in most cases, the maximum level of performance over a wide range of speech input levels (Hudgins, et al., 1948).

Another variable that is critical to the functioning of the AGC is the compression ratio. The compression ratio is the ratio of the change in dB output to the change in dB input signal. E.g., if for every 5 dB SPL increase in input, there is a 2 1/2 dB SPL increase in output, the aid is said to have a 2.1 compression ratio. Specific ratios of output limiting have varying effects on speech perception and discrimination. Various studies have been carried out to probe into these effects (Kretsinger and Young, 1960; Burchfield, 1970; Dreschler ,1988; Fikret-Pasa, 1993; and Hickson ,et. al., 1995). These studies investigated the effects of different amounts of limiter action vs. peak clipping. Some reported that, compressed speech discrimination scores were much better than those obtained under linear (1:1 ratio) amplification for subject with recruiting ears (Kretsinger and Young, 1960 and Burchfield, 1970). The results of the more recent studies by Dreschler, 1988; Fikret-Pasa, 1993; and Hickson et al., 1995) differ from the earlier studies in that they report no significant differences in performance for linear vs. compression amplification. In fact, in the Fikret-Pasa (1993) study some of the subjects showed superior performance with linear type circuit (compression ratio of 8:1 after 80 dB input) for inputs ranging from 65 to 85 dB SPL.

Hickson et. al., 1995 reported that compression amplification (1.3 and 1.8 ratios) was not found superior to linear amplification in any of the test conditions (quiet condition and two different noise conditions) and was significantly worse than linear amplification in the babble background noise condition.

Most of the studies quoted above concur on the view that extreme'higher output limiting ratios result in deterioration of speech perception scores. On an average, the more the output limiting ratio approximated linear amplification, better discrimination scores were obtained.

The two features of AGC, discussed so for, viz. compression threshold and compression ratio are commonly used to achieve compression limiting, in which compression is used only to limit high output levels. Another type of AGC circuit is the whole range compression in which compression is used at lower levels also in order to compensate for recruitment. Recruitment is a dynamic problem which may present to different extents at different frequencies, i.e., usually, more compression is required at lower frequencies compared to higher frequencies. If an aid is designed to activate at a given input level (say 70 dB) regardless of the frequency content of that input, then is possible for lower frequency background noise to activate compression when the higher frequency speech signals still need full linear amplification to be most intelligible eg., Edgardh (1951) employed a limiter with extreme compression and found that dynamic equalization between vowels and consonant sounds was not obtained unless amplitude control functioned over the entire range. Hence, Caraway and Carhart (1967) advocated that dynamic vowel consonant equalization is required to increase the intelligibility of speech transmission.

To meet this requirement of dynamic vowel-consonant equalization, frequency dependent compression was designed. This involved splitting the signal picked up by the microphone into two or more frequency bands and compression is applied independently in each band. In principle, such multichannel aids can compensate more effectively than single-channel aids for recruitment that varies as a function of frequency (Moore et al., 1998). The level dependent frequency response circuit provides simultaneous adjustment of the frequency response shape as well as gain to get the most appropriate loudness and tonal balance. Studies comparing performance of two channel compression aids over linear amplification aids have equivocal findings. Villchur (1973), reported improved speech intelligibility using frequency selective compression amplification. Lippmann et. al, (1981) found disappointing results for speech intelligibility improvement experiments using multichannel systems (16-channel compression aid) fitted to restore disturbed equal-loudness contours for pure tones. But, in a subsequent study by Lawrence et al, (1983) comparing linear vs. two-channel compression aids, the compression aids proved to be substantially better than the linear aids. The compression aids allowed good speech discrimination over a wide range of sound levels. Fabry and Stypulkowski (1993) studied two-band processors utilizing compression or linear processing in either band. For noisy backgrounds, the linear processing in high band (which contains low intensity speech sounds) was judged to be superior to the compression. Thus, multiband compression aids provide better fits into reduced dynamic range of persons with loudness recruitment. In this context, the study by Souza and Turner (1999) provides much insight. They examined the effects of alternation of temporal information and audibility of speech cues through multichannel compression system. They reported that multichannel compression improved speech recognition under conditions where superior audibility was provided by the two-channel compression system over linear amplification. When audibility of both linearly amplified and multichannel compressed speech was maximized, the multichannel compression had no significant effect on speech recognition score for speech containing both temporal and spectral cues. However, results for signal correlated noise stimuli show that more extreme amounts of multichannel compression can reduce use of temporal information.

The ideal release time is more difficult to define. Various researchers quote the ideal value for release time ranging from 50ms to 500ms. More recently, the average duration of a syllable has been used to arrive at a value of release time. Whole range syllabic compression (WRSC) circuits have a release time that is equal to or shorter than the duration of an average syllable (i.e., 100-50 msec). When release times of 50 msec or less are used, this is sometimes called as phonemic compression. Release time is typically longer than attack time in order to avoid the effect of acoustic fluttering or rapid modulation of intensity between syllables which constantly raise and lower the gain. Compression amplification typically occurs in response to the power produced by vocalic signals, and low intensity- consonants may be reproduced too faintly to maintain speech intelligibility if the release time is too long.

Lynn and Carhart (1963) studied the effects of time constants on speech reception thresholds and intelligibility and demonstrated that speech discrimination decreased as release time increased beyond a critical period. Longer attack times required more attenuation to reach spondee threshold. Johansson (1973) indicated that attack times greater than 20 msec will not assure full protection against discomfort. He also suggested that a release time of 150 msec is satisfactory. Newer circuits incorporate an adaptive compression design that has a variable (or dual) release time that is dependent upon the duration of the input signal (Gitles, et. al., 1987).

One more important factor to be taken into consideration while studying the efficacy of AGC aids are the time-constants viz attack time, release time. The response of a compression system is not instantaneous; some period of time is required for it to operate. This is known as attack time. It is the time lapse from the moment signal amplitude (input) exceeds the limiting level to that instant when gain becomes stabilized at a reduced level. Similarly, when the input level decreases, it takes a moment for the system to cut off and restore the normal gain function. This time lag is referred to as release time. It is defined as the time from the moment input amplitude is decreased to that instant at which gain is again stabilized at the precompressed level.

As long as the attack time is less than the minimum amplitude change perceptible to the human ear (typically less than 10ms) its duration will have no effect on perceived sound quality. If it is longer than 10ms. it is possible for the user to 'hear' a brief period of over shooting before the compression attack is complete. Their period-of distortion will have a negative effect on sound quality (Smriga, 1985).

The compression hearing aids can be classified into input compression and output compression depending on the stage of signal processing at which the compression circuit is activated. The most important difference between these two types of compression is related to what happens when the user adjusts the volume control. For input compression hearing aids, gain and output both change while for output compression, gain changes are independent of maximum hearing aid output. As a result, in noise environments, the user of an input controlled compression hearing aid may lower the hearing aid volume control to minimize gain and output and increase comfort in noise. The output compression limiting devices do not have this

facility and this places undue importance on carefully setting hearing aid **MPO** to below the user's loudness discomfort levels. This variable gives an advantage to input controlled compression aids over output controlled compression aids (Fabry, 1994).

To summarize the discussion on compression hearing aids, automatic gain control can benefit hearing aid users by allowing them to listen to a wider range of sound levels without either strain or discomfort and if time constants are well chosen without adverse effects on speech intelligibility in quiet or in noise (King and Martin, 1984).

The various output limiting circuitry (linear and non-linear) may be successful in changing the nature of the sound waveform so as to curb output intensities from crossing discomfort levels. The exact nature of their effects on speech intelligibility has been a subject of much research. One of the principle problems to be resolved is the kind/type of amplification required vs. subsequent distortions induced/ introduced in the signal as a result of the amplification system, and whether or not the trade-off of signal fidelity for user acceptance is favourable.

III Peak dipping vs. compression circuits:.

All form of compression amplifiers offer advantages over peak clipping:

1. Compression reduces the amount of distortion of signal.
2. Output is limited at a level below the amplitude saturation through gain reduction over the entire signal.
3. Compression amplification permits an expanded dynamic range for hearing aid user since it provides a wider range of input level to the ear, yet still maintains maximum output levels which can be adjusted to tolerance levels of hearing impairment. (Skinner, 1981).

There are indications that compression reduces the relative contribution of frication and enhances the contribution of the second formant in vowels (Dreschler, 1988).

Many studies have been carried out to compare the subject performance on linear peak clipping aids and compression aids, with equivocal results. Davis et al., 1947, Hudgins (1948), Dreschler, (1988), Barker et al, (1999), Jenstad et al, (1999), report that compression hearing aids give much better speech intelligibility scores than peak clipping linear amplifiers. But other researchers (Caraway and Carhart, 1967; Blegvad, 1974; Dreschler et al., 1984 and Biering-Sorensen et al., 1995 and others) found no significant improvement in speech intelligibility for compression aids as compared to peak clipping aids. Blegvad (1974) reasons out that the contrary results of his study could be due to the fact that the degree of compression and gain characteristics were not fitted according to individual requirements of the subjects. The degree of hearing loss and its configuration was not a decisive factor in the subject selection criterion. It must also be noted that most of the studies (Blegvad 1974, and others) used compressor aids with extreme compression settings. This may have distorted the loudness relationships among speech sounds to such an extent that though the loudness of the output sound never reached discomfort levels the output signal was unintelligible to a large extent. Hence in most studies, the subjects preferred the linear amplification with intact loudness relationships and uncomfortable loudness levels to the distorted speech output of the compression aids. Most authors agree that there are a number of variables in any fitting that we cannot consider for a variety of reasons making valid comparisons difficult.

In conclusion though linear signal processor that use peak clipping is not the limiter of choice for most patients, it would be wrong to assume that linear processing is an incorrect choice of amplification. Clinical evidence has shown that listeners often prefer the linear growth in amplification rather than a 2:1 and 3:1 function. This is more often apparent for high frequencies than for low frequencies, and a non-linear circuit often results in a not loud enough complaint despite best efforts to adjust the gain for typical speech levels

(Bentler, 1994). Thus, the linear processor still retains its place in hearing aid fitting.

As can be surmised from the various, studies carried out on subject performance with linear and non-linear amplification devices, the results are varied. Most results seem to reveal that the linear amplification devices do not lead to distortion and lowered speech intelligibility, except perhaps at very high intensities. Most of the subjects preferred to use the hearing aids with linear circuits with high thresholds of output limiting and large ratios. Despite these conflicting results, most new hearing aid technologies continue to incorporate complex compression circuits to modify the input sound stimuli so as to reproduce normal loudness relations in speech sounds for persons with perceptive deafness and recruitment. This is because the compression aids are a more desirable choice of output limiting for all patients in spite of the fact that there is no subjective difference in performance between compression and linear amplification devices.

This study was carried out to compare the performance for speech intelligibility of subjects with tolerance problems for loud sounds on linear and non-linear output limiting hearing aids. This was to validate similar studies carried out earlier (Burchfield, 1970; Villchur, 1973; Fikret-Pasa, 1993; Dreschler, 1988;Hickson, 1995).

This exercise will be useful in:

1) Determining the differences in subject's performance for various levels of non-linearity.
2) Determining if one particular setting for non-linearity was uniformly useful across all subjects.
3) Determining whether there is any relation between degree of hearing loss and discriminability at various levels of non-linearity.

Lastly, this study should be helpful to provide pointers to set up a protocol for fitting of hearing aids for persons with tolerance problems.

METHODOLOGY

The study was undertaken to evaluate the performance of subjects with aided intolerance to loud sounds on linear vs. non-linear hearing aids. The aim was to see the differences in speech identification/recognition across various levels of non-linearity and arrive at a conclusion whether a particular level setting was commonly useful across all subjects.

Subjects:

Subjects were selected for the study based on the following criteria:

1) Hearing loss: (sensorineural) hearing loss of moderate to moderately-severe degree (41 dBHL to 70 dBHL)

2) Uncomfortable level of listening: Uncomfortable listening levels (UCL) of less than or equal to 105 dBHL for pure tone and/or intolerance for loud (or softer) intensity drum beat.

3) Age: Adult, above eighteen years of age

4) Language: Subject should be verbal, and conversant with Kannada and/or English

Instrumentation/material used:

The following equipment were used for the study:

(1) Analog hearing aids: With linear amplification and with non-liner circuits were compared.

(2) Drum beats were used to check for tolerance problem. The various intensities of drum beats used were labelled as 'soft', 'moderately soft',

'moderate', 'moderately loud' and 'loud\ The stimulus was presented from a distance of five feet from the subject at 90° azimuth.

(3) A standard passage ('The Rainbow" passage in English and the 'Bangalore passage' in Kananda) read by the examiner with moderate vocal effort at a distance of five feet from the test ear at30° azimuth was used to help the subject make clarity judgements.

(4) A five point rating scale was given to the subject to make judgement for clarity of speech

(5) Aided performance was measured using the set of standard list of questions in English and Kannada, developed at AIISH.

(6) Real Ear Aided Responses was measured at each hearing aid setting using Fonix 6500C Hearing Aid Test System.

Test Environment:

A relatively quiet room was used io carry out the experimental procedure.

Test procedure;

Step 1: Sound field equalization of the Fonix 6500C Hearing Aid Test system was carried out as per instructions in the manual. This procedure was done before every subject was taken up for further testing.

Step 2: The hearing aid was used at following settings:

(a) Linear mode, ie., 0% non-linearity circuit setting.

(b) Non-linear mode at 25% setting.

(c) Non-linear mode at 50% setting.

(d) Non-linear mode at 75% setting.

(e) Non-linear mode at 100% setting.

At each of the above settings, the (following) data was collected by presenting the stimuli at a distance of five feet from the test ear at 90' azimuth.

(i) Using the drum beat, the level at which the tolerance problem or TP was reported was noted. If no tolerance problem was reported even at 'loud' stimulus intensity, it was noted as 'no tolerance problem' or NTP.

(ii) Using the speech stimulus (reading of standard passage with moderate vocal effort) the subject was asked to rate the clarity of speech along the five point rating scale. The five rates are as shown in Table3.1.

Table 31;Rating scale for clarity of speech judgements.

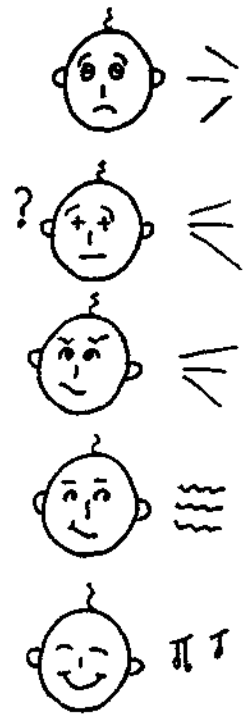
1 : Very distorted

2 : Distorted / unclear

3 : Can understand but is distorted

4 : Can understand if concentration given

5 : Clear speech



(iii) The subject was then asked five simple questions from the standard list of questions. No repetition of the question was made. The number of correct answers out of five were noted down.

(iv) The real ear aided response (REAR) was computed at each of the five settings. The following protocol of REAR measurement was used:

The sound field equalization of calibrated probe tube was carried out. The probe tube was then placed in the ear canal so that the tip of the probe tube was about 2-3 mm beyond the tip of the earmould canal. The hearing aid was fitted on the subject and the volume control wheel adjusted to a comfortable level of loudness. 70 dB SPL composite noise stimulus was used to conduct the measurement.

The data obtained from each subject was tabulated as illustrated below:

Subject no. ____________

Gender: M / F

Audiological diagnosis:__________________

Model of the hearing aid used : 1 / 2 / 3 / 4 / 5

Settings: M, H/N / L

Level of output limiting PARAMETER	Linear Amplification 0%	25%	50%	75%	100 %
1) Volume control					
2) Tolerance problem: Present at x* level/ absent					
3) Subject rating on Scale (y/5)**					
4) Performance for Questions (z/5) ***					

* x = loudness level at which tolerance problem is present.
[1 = soft sound; 2 = moderately soft; 3 = moderate;
4 = moderately loud; 5 = loud]

** y = rating for clarity of speech, given by the subject.

[1 = very distorted; 2 = distorted / unclear;
3 = can understand but is distorted;
4 = can understand if concentration given;
5 = clear speech]

***z = number of questions (out of five) answered correctly by the subject.

The data on REAR measurements at each level of output limited was noted down.

The data thus collected was subjected to statistical analysis.

RESULTS AND DISCUSSION

The test procedure was carried out on seven subjects having intolerance for loud sounds (i.e., drum beat at various intensity levels). The aim of the study was to compare the changes in speech perception ability at various levels of output limiting (0%, 25%, 50%, 75% and 100%) and determining if one particular setting of output limiting level was uniformly useful across all subjects, irrespective of degree and configuration of hearing loss. This data would then be used to develop a protocol for hearing aid selection and fitting for hearing impaired individuals with loudness recruitment.

Table4:Data obtained for subjects across different levels of **outp it limiting**

Subject No.	**Auditory** sensitivity	**Hearing aid used.**	**LINEAR 0%**				25%				S0%				73%				**100 %**			
			Vol.	T.P. AT X/5	Ratin	**Ques.** z/5	Vol.	T.P. x/5	Rating y/s	Ques z/5	Vol.	**T.P.** x/5	Rating y/5	Ques Z/5	Vol.	**T.P.** z/5	Rating y/s	Quea z/5	Vol.	**T.P.** x/y	**Rating** y/s	**Quea** x/y
1.	SNI1L Rt. 100 dD **III**. IX. 10S dUHL.	**A**	**1/2**	1	4	5	1 1/2	2	4	5	1 1/2	5	5	5	1/4	4	3	**4**	**13/4**	**3**	**2**	**3**
	Uil. Mild to pro-found S.N.H.L VCL. Rt. > 90 dBHL Lt. >100 dBHL	**B**	1	4	2	**1**	1 1/2	3	**4**	**3**	1 1/4	5	**4**	**3**	1 1/2		**4**	2	2	J	**3**	**1**
i.	**Rl. Mod. Sev. S.N.H.L.** VCL Rt. > 105 dBHL U. > **105** dBHL	**C**	2	2	5	5	**2**	**2**		5	**2**	3	5	**5**	**2**	**4**	5	**5**	**2**	**5**	**5**	**5**
4.	**Uil. Mod.** Sev. Mixed H.L. VCL Rt. 100 dUHL U. 95 dBHL	**D**	2	4	5	3	**2**	**5**	**4**	**3**	2 1/2	**3**	**5**	4	2 1/2	**3**	**5**	4	2	4	4	2
	Bil. Mod. Sev. Mixed H.L. VCL Rt.75 dBHLLt. 80 dBHL	**D**	**1**	**3**	**4**	**3**	**1**	**4**	5	**3**	**1 1/2**	**4**	5	4	2	4		3	2	3	5	2

6.	Bil. Mod. to Mod Scv. S.N.H.LMixed H.L. VCL Rt. 90 95 dBIH	E	2 1/4	3	4	3	2 1/4	3	5	2	2 1/4	4	5	3	2'/4	4	5	3	2/14	4	5	3
7.	Rt. Mild SN. 1A. Mod. Mixed III. RL. 100 dBHL 11. 105 dBIH.	F	1	;i	S	»	1	◁		'1	1 1/2	5		4	1 1/2	5	5	J	1 1/2		J	3

x loudness level at which tolerance problem is present
y rating for clarity of speech given by the subject
z number of questions (out of five) answered correctly by the subject.

Rating	Tolerance	Rating score
1.	Soft	Very distorted
2.	Moderately soft	Distorted/unclear
3.	moderate	Can understand but is distorted
4.	moderately loud	can understand if concentration given
5.	loud	clear speech

The data collected was subjected to descriptive statistics. The results of the study are discussed below:

Subject 1: (Figure 4.1.a)

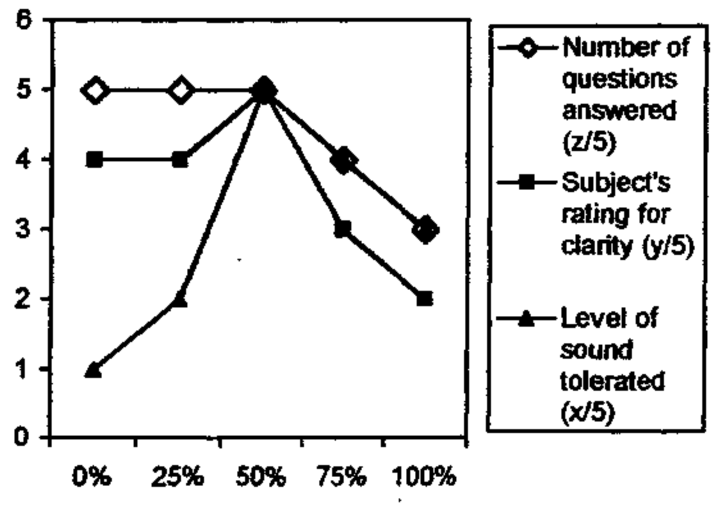


Fig. 4.1.a: Performance for subject 1.

The aided tolerance for loud sounds increased as the output limiting levels increased till 50% then decreased. As the tolerance to loud sounds increased till the 50% setting, the rating for clarity of speech also improved and then decreased for levels above 50% output limiting. For simple questions no significant difference was observed till 50% as she could answer all the five questions at the three settings. However, beyond 50% of output limiting, the performance reduced.

Therefore, for subject 1, the best setting for all the three parameters viz, tolerance, speech clarity and performance for question-answers occurs at the 50% non-linearity setting.

Subject: 2 (Figure 4.1.b)

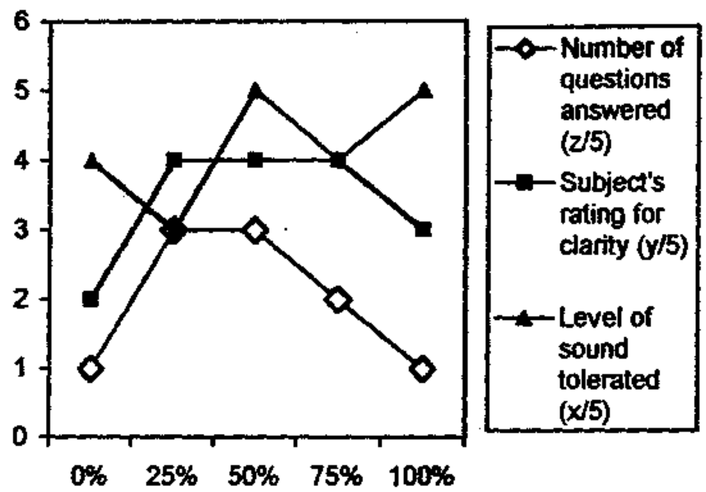


Fig. 4.1.b: Performance for subject 2.

The aided tolerance for sounds varied as a function of the output limiting levels. The maximum level of tolerance occurred at 50% and 100% settings. The rating for clarity of speech improved for non-linear settings (25%, 50%, 75% levels) as compared to the linear settings (0% level). Across different levels of non-linear output limiting, not much difference in terms of improvement in clarity was noticed except at 100% where the clarity reduced. For the task involving answering simple questions, better performance was observed at the non-linear settings (25% and 50%) as compared to the linear settings (0%). However, as the degree of non-linearity was increased to 75% and 100% there was a decrease in performance.

For this subject, performance was optimum at 50% non-linearity level. The irregular variations in aided tolerance for loud sounds at different non-linearity settings could be because perhaps the frequency specific gain characteristics at each non-linearity setting differed. It could be that these changes in gain did not fit frequency specific gain requirements of this subject.

Subject 3: (Figure 4.1 .c)

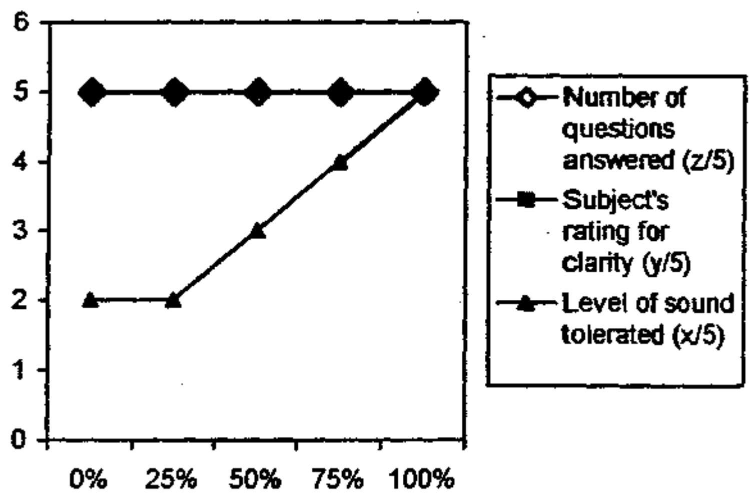


Fig. 4.1.c: Performance of subject 3.

Subject 3 was a case of bilateral conductive hearing loss with intolerance for loud sounds. A steady improvement in aided tolerance levels was observed as the degree of output limiting was increased. In this case, the subject had no difficulty in understanding speech. This is apparent form the uniformly good scores for ratings for speech clarity and performance for answering simple questions.

This subject is a good example where the non-linear circuit aid is used solely for the purpose of output limiting.

Subject 4: (Fig. 4.1.d)

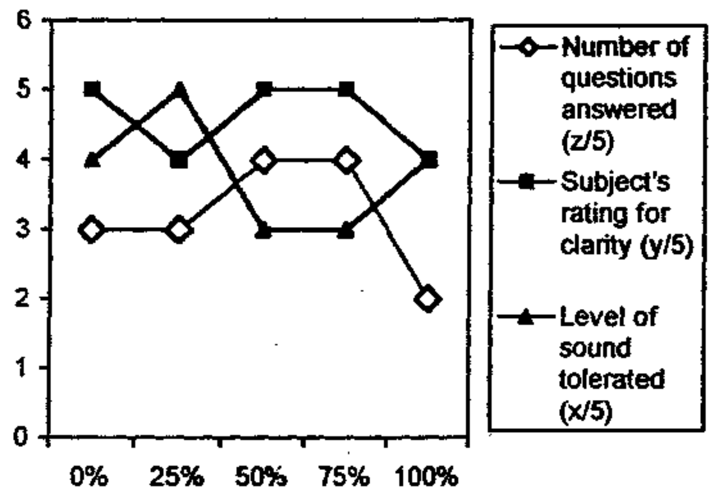


Fig. 4.1.d: Performance of subject 4.

In this case, intolerance for loud sounds increased with increase in degree of non-linearity beyond 25%, though a slight improvement in tolerance was observed at 100% output limiting. The rating for clarity of speech was best at 0%, 50% and 75% non-linearity settings, with a slight decrease in rating at 25% and 100%. Performance on question-answer task was best at 50% and 75% setting and decreased significantly at 100% non-linearity setting.

Based on this data, the 50% non-linearity setting would be recommended to the subject for daily use. Though performance across all three parameters are identical at 50% and 75% settings, on the REAR curve, the hearing aid showed peaks of intensity at 110 dB SPLs. The 50% setting had a smoother intensity-frequency response curve, and hence was selected as the optimum setting.

Subject 5: (Figure 4.1.e)

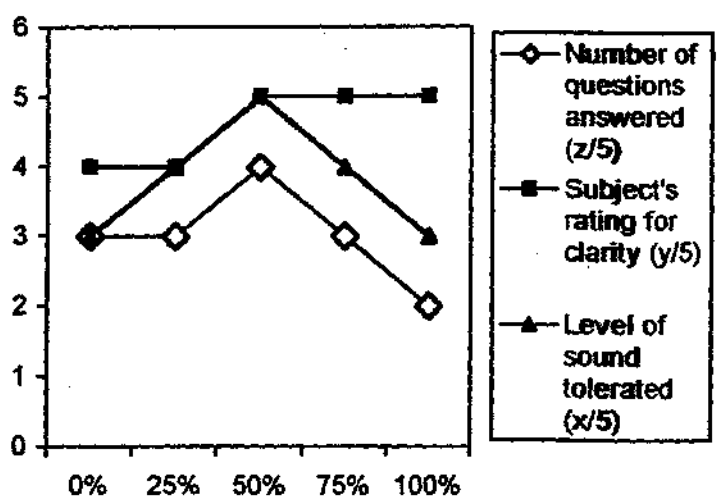


Fig. 4.1.e: Performance of subject 5.

For this subject, better ability to tolerate loud sounds was observed at 25%, 50%, and 75% of non-linearity settings, and a slight decrease in tolerance was reported by the subject at 100% non-linearity setting. Rating for clarity of speech for non-linear settings was better than for linear setting. Across different levels of non-linearity, however, there was no significant difference in rating for clarity. The performance on question-answer task showed a different picture. Performance on this task was best at the 50% non-linearity setting with decrease at both higher and lower settings (viz., 0%, 25% and 75%). At 100 % non-linearity setting there was a significant decrease in performance on this task.

Thus, performance for all three parameters was best at 50% setting as compared to the other settings. The differences in performance of this subject for the more subjective task of clarity rating and the more objective task of answering question, concurs with research findings that the quality of speech and its intelligibility do not have a one-is-to-one relationship (Moore, 1990)

Subject 6: (Fig. 4.1.f)

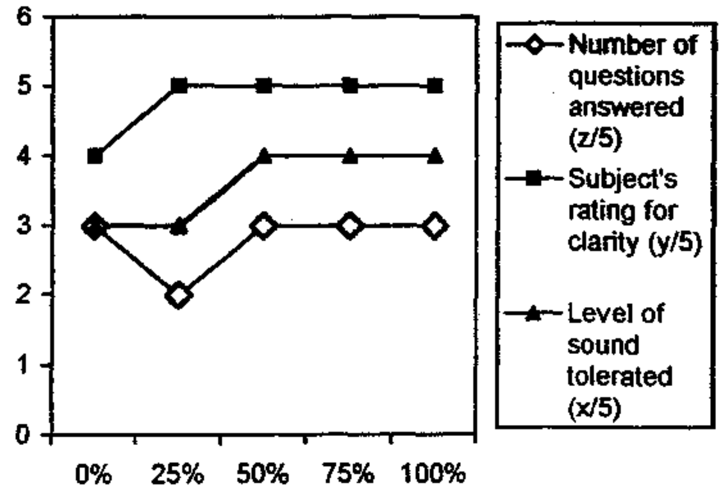


Fig. 4.1.f: Performance of subject 6.

The tolerance for loud sounds increased for the 50% non-linearity setting, and remained steady thereafter. The subjects rating for clarity was greater for the non-linear setting as compared to the linear setting. But across the different levels of non-linearity, no differences were seen. The subject gave a uniformly good rating for clarity of 5 for all the non-linear settings. The performance for question-answer task was identical for all the settings (0%, 50%, 75% and 100%) except at 25% non-linearity setting where the performance was poorer.

For this subject also, the 50% non-linearity setting would be recommended as performance on all three parameters are in concurrence with each other at this setting. Though similar values were obtained at 75% and

100% setting also, they would not be recommended as relative distortion of the signal at these settings is more when compared with the 50% level.

Subject 7: (Fig. 4.1.g)

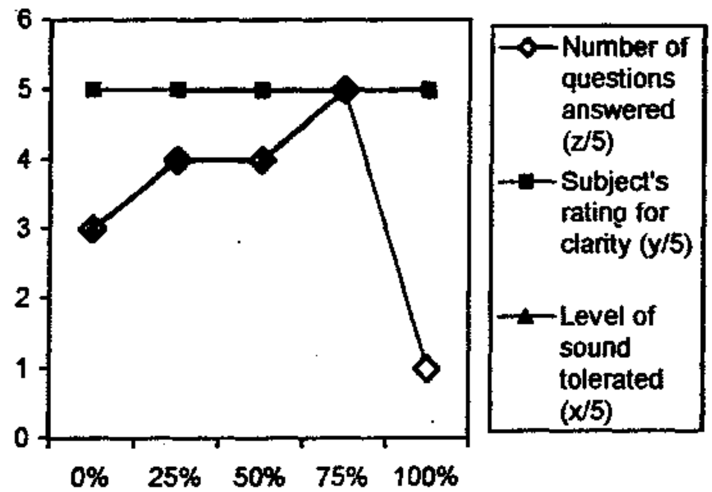


Fig 4.1.g: Performance of subject 7.

For this subject, tolerance for loud sounds steadily improved as a function of change in non-linearity setting form 0% to 50% where it reached the maximum value of 5. From 50% to 100% setting, the tolerance level remained constant at 5. The rating for clarity of speech was consistently good across all levels of non-linearity settings (0-100%). But the performance for question-answer task varied, and was not similar to the clarity ratings. The performance on this task was lowest at 0% setting and steadily improved to reach the peak at 75% setting. At 100% non-linearity setting, the performance showed a steep drop back to the performance at 0% level.

Thus the 75% sotting seems to be appropriate for this subject since performance across all three parameters reached their peak at this level.

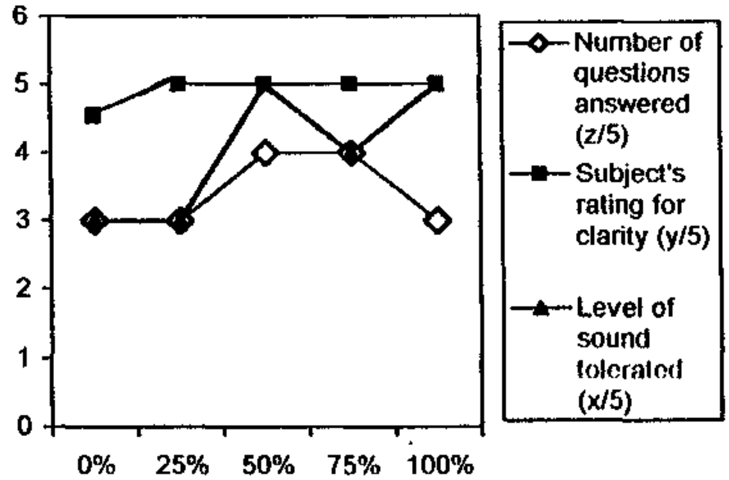


Fig. 4.2 : Modal scores all the three parameters for the seven subjects.

If the modal scores of all seven subjects on different tasks (viz., loudness tolerance, rating for clarity and performance on question-answers) are considered (as shown in Tig 4.2), the following observations could be inferred:

(1) The level of sound tolerated increases till 50% output non-linearity level and then varies. (It may increase or decrease beyond 50% setting).

(2) The subjects rating for clarity is not significantly different for linear vs. non-linear settings. Also, there is no significant difference in clarity across the different levels of non-linearity (i.e., 25%, 50%, 75% and 100%).

(3) For performance on the question-answer task, the number of questions answered increases till 50% -70% non-linearity level setting and then

decreases. This trend reveals a decrease in speech intelligibility at higher settings of non-linearity, though the rating for clarity do not show a similar trend.

The results obtained show that some subjects (subjects 2, 4, 5 and 7) showed improvement in performance for non-linear circuits as compared to linear amplification hearing aids. These results concur with the observations of researchers like Davis et. al, (1947); Hudgins (1948); Dreschler (1988), Barker et. al, (1999) and Jenstad et. al, (1999). Logically also, improved processing of the signals to accommodate the dynamic characteristics of loudness recruitment, should lead to better speech perception and discrimination. This view is corroborated by the performance data for subjects 2,4, 5, and 7.

Subjects 1 and 6 showed only slight differences in performance for linear vs. non-linear hearing aids. This results seem to agree with those obtained by Caraway and Carhart (1967), Blegvad (1974), Dreschler et. al, (1984) and Biering-Sorenson et. aL, 1999.

This variation in performance for linear vs. non-linear hearing aids could be an indication that, the linearity or non-linearity feature of signal processing is not the only important consideration. Other factors like threshold of non-linearity setting, ratio, time constants, and patient factors like degree and configuration of hearing loss, nature of loudness recruitment are also important considerations.

For many subjects, at least at two settings, there was an disagreement between their rating for clarity and their actual performance on question-answer tasks. Often, though the subject gave a high rating for clarity, their performance indicated otherwise. This result is in consonance with the opinion of Moore, 1990, Sweetow, 1994, and Mueller and Killion, 1996, Dillon, 1999 that improved speech clarity by better speech processing technologies may not necessarily lead to a better subject performance.

Also, the parameters used by the subject to rate for clarity and those required for better performance on question-answer/discrimination tasks may be different. The many parameters which may be common for these tasks lead to a positive correlation (+0.4891 on Spearman's Rank correlation) between the two tasks. But the differences in the parameters explain the absence of a one-to-one correlation.

Most of the subjects showed improved speech discrimination skills as evidenced by performance on question answer tasks only till the 50% non-linearity level. The deteriorating performance at high levels of non-linearity settings agree with research literature saying that high levels of non-linearity introduce unwanted distortion of the signal.

Based on these observations of subject performance for linear and non-linear hearing aids and also across different levels of non-linearity, a protocol for hearing aid fitting for hearing impaired persons with loudness recruitment was developed.

Protocol for hearing aid selection and fitting for hearing impaired subjects with recruitment:

Hearing impaired subjects with recruitment are the target population for this protocol. Considering the degree of their hearing loss, an appropriate class hearing aid should be selected first. The hearing aid should incorporate a non-linear circuit with adequate trimmer controls so that fine tuning could be made to fit the output spectrum into the subject's dynamic range. Fitting procedure will take into considerations three parameters:

(1) Tolerance problems.

(2) Improved loudness relations in speech leading to better speech clarity.

(3) Performance on discrimination tasks for ease of listening.

Stage 1 : Tolerance problem:

Presence of tolerance problem and the exact intensity at which it is present is noted down for each setting of the non-linearity trimmer control. Those settings of the control at which the tolerance problems are least are noted down. Loudness scales on a five point scale:

(1) Very loud.

(2) Loud.

(3) Moderate.

(4) Moderately soft.

(5) Very soft.

may be used to determine the intensity at which tolerance problem is present.

Stage 2 : Clarity rating

At each setting of the non-linearity trimmer control, changes will be introduced in the loudness relationships of speech sounds, especially when the trimmer controls change the compression ratio. Hence at each setting, the subject should be asked to rate for clarity of a speech stimuli produced with moderate effort at a distance of 5 feet from the test ear. This should be done on a five point rating scale, similar to the one used in the study.

Stage 3 : Performance on discrimination task:

At each setting of trimmer control, performance of the subject for discrimination tasks [spondee-repetition or answering simple questions] should be administered. The setting at which best performance was present is noted down.

Stage 4 : Selection of appropriate trimmer control setting :

Select that setting at which:

1) Loudness Tolerance problem is least.
2) Best clarity rating are present, and
3) Discrimination skills for speech stimuli are good.

In case, two or more settings have equally good scores on all/any of the three parameters considered, REAR data may be computed at each of those setting. That setting is finally recommended, at which the loudness levels/intensities at user volume and tone setting does not cross the discomfort levels of the subject.

SUMMARY AND CONCLUSIONS

A hearing impaired person with loudness recruitment presents with a challenge for appropriate, comfortable hearing aid fitting. A narrow dynamic range, with intolerance for loud sounds forms a characteristic feature of this type of hearing loss. The narrower the dynamic range, the greater is the recruitment and the more complicated is the fitting (Mynders, 1985).

A number of hearing aids incorporating different linear and non-linear output limiting circuits that claim to be beneficial for the recruiting ear are available. But, there are a number of variables in any fitting, that we cannot consider for a variety of reasons. This makes comparison of studies carried out on hearing aid fitting for the sensorineural hearing loss ear with loudness recruitment, difficult, if not impossible (Van Vliet, 1994). Another problem is the reliability of the stimuli used in the fitting procedure to simulate real life situations of hearing aid use. Moore et al, (1998) suggested giving more importance to the use of speech in fine tuning the fitting rather than use of tones or bands of noise.

This study was carried out to compare the effects of various levels of output limiting in a hearing aid on speech perception/discrimination, and to monitor the efficacy of the different non-linearity settings in reducing tolerance problems. Seven subjects having tolerance problems for loud sounds (using the drum beat as stimuli) Were selected for the study.

At five different settings of non-linearity (0%, 25%, 50%, 75% and 100%), data was collected for performance on three parameters:

(a) The level at which the subject experienced tolerance problem for drum beat at five different intensities (soft to loud).

(b) The subject was asked to make clarity ratings on a five point scale.

(c) Lastly, their ability to understand and answer five simple questions was checked.

The results showed that, the tolerance for loud sounds was better for the non-linear circuits than for the linear hearing-aid setting. Clarity rating for non-linear setting was slightly better than for linear settings; but across the different levels of non-linearity no differences were observed.

The subject showed an initial increase in speech intelligibility discrirnination ability with increase in non-linearity till 50% level and then showed a drop in performance. Overall, the non-linear circuits gave better performance than the linear amplification hearing aids. This was evident on all the three parameters tested i.e., tolerance for loud sounds, the clarity rating and the performance on question answer tasks.

Based on these results, an outline of the suggested protocol for fitting non-linear hearings aids for hearing impaired subjects with loudness recruitment was put forward. This protocol suggests use of the three parameters-viz. aided tolerance for loud sounds, subjective clarity rating and performance on speech discrimination tasks, to arrive at an appropriate non-linearity setting.

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