EFFECT OF SYLLABIC AND DUAL COMPRESSION ON SPEECH IDENTIFICATION SCORES

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A Dissertation submitted in part fulfillment for the final year
Masters of Science (Audiology)

University of Mysore, Mysore

All India Institute of Speech and Hearing

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Mysore – 570006.

MAY - 2005

Dedicated to

God,

My parents

&

Teachers

Certificate

This is to certify that the dissertation entitled ***"The Effect of Syllabic and Dual Compression on Speech Identification Scores"*** is the bonafide work done in part fulfillment of the degree of Master of Science (Audiology) of the student with Register No. A0390005.

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This is to certify that the dissertation entitled ***"The Effect of Syllabic and Dual Compression on Speech Identification Scores"*** has been prepared under my supervision and guidance. It is also certified that this has not been submitted earlier in any other University for the award of any Diploma or Degree.

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Declaration

I hereby declare that this dissertation entitled ***"The Effect of Syllabic and Dual Compression on Speech Identification Scores"*** is the result of my own study under the guidance of Ms. P. Manjula, Lecturer in Audiology, All India Institute of Speech and Hearing, Mysore, and has not been submitted earlier at any other University for the award of any Diploma or Degree.

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CHAPTER I

INTRODUCTION

The common observation in individuals with sensorineural hearing loss is recruitment or softness imperception, i.e., the occurrence of steeper than normal loudness growth function, together with an elevated absolute threshold. The typical means by which a hearing aid compensates for this recruitment is the use of a non-linear compressor circuit. The general aim of the compressor is to provide higher gain for softer sounds than for louder sounds (Hansen, 2001). This is achieved by manipulating the compression parameters such as compression time constants, compression ratio and compression threshold or knee point.

Compressor can react to a change in input levels within a few milliseconds, or they can be sluggish that they take many tens of seconds to fully react. This depends on the time constants – attack time, the time taken for the compressor to react to an increase in input level, and release time, the time taken for the compressor to react to a decrease in signal level.

The compression ratio describes the variation in output level that corresponds to a variation in input level and, is reflected in the slope of the curves on input-output diagram. The compression threshold is the input level above which the compressor operates. The degree to which a compressor finally reacts as signal level changes is best depicted by the compression time constants, compression threshold and compression ratio.

The optimal fitting of the hearing aid gain characteristics, both as a function of hearing loss and physical properties of the incoming sound, has been the subject of a large number of studies and has resulted in many standard prescription rules for hearing aid fitting. A few prescriptive formulae also calculate the recommended compression ratio, compression threshold and gain of the static input-output curve of a compressor as a function of hearing loss (Byrne, Dillon, Ching, Katsch & Keidser, 2001).

In contrast to the effort invested in deriving the optimal static input-output curve there has been much less focus on two other important compression parameters that control the output signal of a compressor. These parameters are its attack time and release time, which are determined by the time constants of the input level estimator. The choice of time constants in the level estimator has a large effect on the time course of the gain that is applied, and thus, on the output of the hearing aid.

Most commercially available hearing instruments have relatively short time constant ("syllabic compression") with an attack time in the range of 1 to 10 msec and a release time of less than 100 msec. The rationale for a short time constant is that the hearing aid should be fast enough to be able to follow the temporal level changes in the speech signal in order to apply necessary gain changes for all segments of the speech signal. Physiological evidences (Ruggero & Rich, 1991, cited in Hansen, 2002; Buus, 1999) indicate that if the rationale of a hearing aid is to restore the perception of loudness to normal, short time constants are required. There are studies showing that compressors with short time constants have a negative influence on the sound reproduction quality of the system or have a large amount of signal distortion.

A compression hearing aid with long release times will adjust the gain accordingly in different listening situations. On a short-time basis within one situation it will act more like a linear hearing aid, which maintains the spectral and temporal level differences of the incoming sound almost unaltered (Hickson, Dodd & Byrne, 1994).

Neuman, Bakke, Mackersie, Hellman and Levitt (1995) investigated the effect of release time on sound quality. For combinations of compressor parameters, compression ratio (of 1.5, 2 or 3.1) and release time of (60, 200 or 1000 msec), they did not find a significant main effect for the release time.

Hansen (2002) studied the influence of the compression time constant in a multichannel compression hearing aid on both subjectively assessed speech intelligibility and sound quality in realistic binaural acoustical situations for normal hearing and hearing impaired. All subjects showed a significant preference for the longest release time (4 sec) over a shorter release time (400 msec to 40 msec). The preference of a rather long release time raise the question of whether a compression system with a very short time constant is the optimal means to compensate for the sensory recruitment or softness imperception phenomenon and also, prescriptive formulae consider the gain required across different frequencies, but most of these fitting procedures do not consider the optimal compression parameters such as compression time constants (Hansen, 2002).

Although fast-acting compression systems are becoming integral components of modern hearing aids, research results have not consistently demonstrated their benefit over linear amplification. There is, however, no standard method of specifying the most effective compression strategy for a particular hearing-impaired person, mainly because of the larger number of the parameters required to specify, fit and evaluate a compression system (Olsen, Olofsson & Hagerman, 2004).

There are a large number of studies done using fast acting compression system and the results of these studies are equivocal. Van Harten-deBruijn, Van Kreveld-Bos, Dreschler and Verschuure (1997) studied to test whether multichannel syllabic compressors enhance speech perception scores and found that no negative effects were shown, on average, for the syllabic non-linear processors, but no positive effects were demonstrated either. However, Plomp (1988) has reported of a poorer scores multichannel amplitude syllabic compression over linear amplification, and also, there are a only few studies (Hansen, 2002; Moore & Glasberg, 1986) evaluating the time constants of compression. Hence, there is a need to study the effect of compression types that differ in terms of time constant on speech identification scores.

AIM OF THE STUDY

The aim of the present study was to study the effects of compression types (syllabic compression and dual compression) on speech identification scores (SIS) in the following two conditions:

(a) In quiet condition

- (i) At 45 dB HL level of speech
- (ii) At 75 dB HL level of speech

(b) In the presence of noise, with SNR of +10 dB

- (i) At 45 dB HL level of speech
- (ii) At 75 dB HL level of speech

CHAPTER II

REVIEW OF LITERATURE

The reduced dynamic range of hearing impaired listeners compared with normal hearing listeners has inspired several researchers to use full dynamic range compression to fit level-varying signals such as speech into the reduced dynamic range of hearing impaired listeners (Hickson, 1994). Hence, one approach to reduction of dynamic range of hearing is to allow gain to be a function of output over the complete range of intensities in the speech signal, rather than only when some maximum desirable output is reached (Boothroyd, Springer, Smith & Schulmann, 1988).

Even the most advanced hearing aid technology does not restore "normal hearing". The human cochlea, with the differential roles played by the outer and inner hair cells, is a magnificent non-linear sensory organ. All of our recent developments in compression technology are but tiny steps towards the goal of imitating the cochlea and its functions. Still, however, these are steps in a positive direction (Venema, 1999).

Despite the complexity, the benefits of compression can be summarized as follows: compression can make *low-level speech* more intelligible, by increasing gain, and hence audibility (Souza & Turner, 1998); compression can make *high-level sounds* more comfortable and less distorted and, in *mid level environments,* compression offers little advantage relative to a well-fitted linear aid. Once the input level varies from this, of course, the advantages of compression become evident. Its major disadvantages are a greater likelihood of feed back oscillation and excessive amplification of unwanted lower level background noises.

The review is being discussed under the following headings:

1) Rationales for compression system

2) Different approaches to compression method

   a) Output limiting

   b) Wide dynamic range compression

3) Dynamic aspects of compression

   a) Syllabic compression

   b) Dual compression

   c) Peak detection

   d) Adaptive compression

   e) Average detection

1) Rationales for compression system

The general aim of the compressor is to provide higher gain for softer sounds than for louder sounds. Because of the frequency dependence of the recruitment it is necessary to compensate for it by independent compressors in different frequency channels. Dillon (1996) has outlined the rationales of compression system. All of the following rationales include the desire to reduce the dynamic range of the signal in some way.

(i) Discomfort and distortion avoidance (Compression limiting)

(ii) Loudness normalization

(iii) Noise reduction

(iv) Reduction of signal dynamic range

(i) Discomfort and distortion avoidance (compression limiting):

If the output of a hearing aid is not limited in some way, output signals will sometimes exceed the loudness discomfort level of the aid wearer. The primary advantage expected for compression is that even if the aid wearer selects a high volume control setting to amplify weak input signals, the compressor will prevent discomfort from occurring without distortion, if a high level wanted or unwanted signal occurs.

(ii) Loudness normalization:

Because of the effects of recruitment, the equal loudness contours of a person with a high frequency sensorineural hearing loss show the greatest deviation from normality at low input levels. The principle of loudness perception is that, for any input level and frequency, the hearing aid gain should be such that it is sufficient enough for the wearer to report the loudness to be the same as that which a person with normal hearing would report.

(iii) Noise reduction:

The noise reduction rationale aims to identify frequency components that do not contribute to intelligibility or comfort, and, to attenuate those components relative to more useful components. The basic assumption is that at any given time, the signal-to-noise ratio (SNR) will vary with frequency. A second assumption is that noise in one frequency region will mask useful signals in the other frequency regions. The rationale of reducing masking by attenuating the frequency region with the poorest SNR is supported by the data of Rankovic, Freyman and Zurek (1992).

(iv) Reduction of signal dynamic range:

Because sound in various listening environments vary over a wide range of levels, and because hearing impaired people listen most effectively over a narrower range, compression can be used to translate a wide range of levels at the hearing aid input into a narrower range of levels at the aid output. This reduces or eliminates the necessity for the aid wearer to vary the volume control.

The major advantage expected for wide dynamic range compression (WDRC) is that the user has less need to vary the volume control. The major disadvantages expected are the increased gain for low-level inputs makes feedback oscillation more likely than linear amplification. The higher gain also makes internal aid noise more likely for some subjects. Finally, compression at low levels can cause pumping of background noise over a wide range of speech and noise levels.

2) Different approaches to compression method

There are actually two different philosophical approaches to compression method, not specific controls. They are:

a) Output Limiting

b) Wide Dynamic Range Compression (WDRC)

The effects of output limiting compression and WDRC can be seen in Figure1.

a) Output limiting compression or compression limiting:

The salient features of compression limiting are shown in the left most graph of Figure1. Compression limiting hearing aids have two main features, a “high” compression knee-point and a high compression ratio. Low-level sounds are amplified linearly, but the inputs from moderate to intense sounds are squashed into a narrower range of outputs (Dillon, 2001). A high compression ratio usually is defined as being greater than 5:1 (Dillon, 1988).

b) Wide dynamic range compression:

WDRC hearing aids have become extremely popular during the past several years. The function of WDRC is shown in Figure1 (right most graph). WDRC is associated with low compression knee-points (below 55 dB SPL) and compression ratios (less than 5:1). The WDRC hearing aid is almost always in compression. It can be seen that many different input levels, from very soft speech to very loud speech, will put the hearing aid in compression. Perhaps it is called “wide dynamic range compression” because of its low knee-point, which allows compression to take place over a wide range of input levels.

The two different types of WDRC are 1) Bass increases at low levels (BILL), with a low knee-point for the low frequencies and a higher knee-point for the high frequencies. 2) Treble increases at low levels (TILL), in which the knee-point is set at a high input SPL for the low frequencies and at a lower input SPL for the high frequencies (Killion, Staab & Preves, 1990).

WDRC can be incorporated either in single channel, two channel or multichannel compression hearing aids. Relative to single-channel compression, multichannel compression can increase intelligibility because it increases the audibility of speech. Unfortunately, multichannel compression also decreases some of the essential differences between different phonemes. Because compressors give less amplification to weak signals, multichannel compressors tend to decrease the height of spectral peaks and to raise the floor of spectral valleys, that is, they partially flatten the spectral shapes. Spectral peaks and valleys give speech sounds much of their identity. Spectral flattening makes it harder for the aid wearer to identify the place of articulation of consonant, (De Gennaro, Braida & Durlach, 1986; Lindholm, Dorman, Taylor & Hannley, 1988; & Lippman, Braida & Durlach, 1981), and hence offsets the positive effect of increased audibility.

Considering these opposing effects of multichannel compression, it is not surprising that some experiments have shown multichannel compression to be better than single channel compression (Kiessling & Sliffens, 1991; Moore & Glasberg, 1988; Moore & Glasberg, 1986) and some have failed to show any advantage for multichannel compression (Moore, Peters & Slone, 1999; Walker, Byrne & Dillon, 1984). Keidser and Grant (2001) have also reported similar results. They found no significant differences in speech recognition scores between one, two or four compression channels, but the field tests revealed a preference for the two-channel scheme compared to the other systems by subjects with steeply sloping hearing loss. This suggest that the use of at least two channels be available when fitting clients with sloping loss; while for clients with flat loss, the number of channels may be less important.

3) Dynamic aspects of compression

Hearing aids are not the only electrical devices that use compression and have dynamic aspects such as attack/release times. Audiovisual equipment have used input and output compression for many years.

The attack and release times are set to achieve a best compromise between two undesirable extreme. Times that are too fast will cause the gain to fluctuate rapidly and this may cause a jarring, “pumping” perception by the listener. Times that are too slow may make the compression act too slowly and cause a lagging perception on the part of the listener (Staab, 1996).

Different attack and release times, sometimes, are used to categorize different types of dynamic compression. Different methods of providing attack / release times also separate one type of dynamic compression from another (Venema, 1999).

One purpose of compression systems in hearing aids is to allow speech to be both audible and comfortable over the wide range of sound levels encountered in every day life; this range extends from about 50 dB SPL to over 90 dB SPL (Pearsons, Bennet & Fidell, 1976, cited in Sandlin, 2000). It has been shown that the goal of audibility and comfort can be achieved in two quite distinct ways.

a) Fast acting compression, sometimes called syllabic compression or dynamic compression (Hohmann & Kollmeier, 1995)

b) Slow acting compression acting on the whole speech signal (Moore, Glasberg & Stone, 1991).

The other ways are c) peak detection, d) adaptive compression and e) average detection.

a) Syllabic Compression / Fast Acting Compression:

Fast acting compression amplifiers are those with time constants sufficiently short that the gain of the amplifier changes significantly during a syllable or word (Walker, Byrne & Dillon, 1984). The attack/release time are specifically intended to be shorter than the duration of the typical speech syllable, which is about 200 to 300 msec (Hickson, 1994). This provides more uniformity in the intensity of on-going speech syllable.

Hence, fast acting compression hearing aids have short attack and release times, typically the attack time is 1 to 10 msec and the release time is 10 to 150 msec. In such aids, the gain is different for the speech maxima and the speech minima, and the shape of the frequency-gain characteristics can also vary from moment to moment depending on the short-term spectrum of the incoming sound (Moore, Peters & Stone, 1999). One reason for using a fast-acting compressor is to increase the audibility of low level speech sounds and/or to minimize the effect of forward spread of masking of low level sounds (Dillon, 2001).

Moore and Colleagues reported that, fast-acting compression, compared to linear amplification, provided small but significant benefits in hearing-impaired subjects particularly when listening to speech in a noise background that contained temporal and spectral dips (Moore, Peters & Stone, 1999). The fast acting compression was able to improve the audibility of speech in spectral and temporal dips in the noise, but it does not restore performance to normal.

However, syllabic compression is somewhat controversial and not everyone agrees with its use. Because syllabic compression compresses the peak amplitude of speech and makes the waveforms of on-going speech more uniform, noise can easily fill in the small gaps that remains (Johnson, 1998). In noisy situations, the hearing aid may amplify the noise that is situated between peaks of speech. According to Killion (1996), fast attack/release time of 50 msec can distort the waveform of speech and, thus compromises speech intelligibility. Kuk (1999) suggested that use of fast attack time (<10 msec) and short release time (<100 msec) will compromise the intensity differences between the various phonetic elements of speech. Specifically, in the time waveform of speech, the differences between the "peaks" or loud elements and "valleys" or soft elements are compromised or lessened by fast attack / release time. Such a reduction, in turn, can distort the spectral content of speech cues.

In Olsen, Olofssen and Hagerman's (2004) study, two-thirds of the listeners performed worse with fast-acting compression than with linear processing. Normal-hearing listeners showed the most benefit from compression. Similar negative results were found by Plomp (1988). He described an article which deals with the question of why multichannel-amplitude compression appears to have a negative rather than a positive effect on speech intelligibility by hearing impaired listeners. It is argued that the small time constants of amplitude compression diminish the temporal as well as the spectral contrasts in the speech signal. Some authors have suggested that the deterioration of speech through compression could be due to distortion of the temporal cues of the speech signal (Boothroyd, Springer, Smith & Schulmann, 1988; Plomp, 1988; Souza & Turner, 1998).

A potential problem is that fast compression alters the intensity relationships between different phonemes and syllables. However, if the hearing aid wearer uses the relative intensities of sounds to help identify them, altering relative intensities may decrease the intelligibility of some speech sound, even if it increases their audibility (Plomp, 1994).

Another potential problem is the effect that compression has on brief weak sounds that follow closely after sustained intense sounds. Suppose a sound of higher than average level causes the gain to be lower than would be chosen than the gap between the intense and the weak sound, then the gain will still be decreased when the brief weak sound arrives. Consequently, such weak sounds will be less audible than they would be for linear amplification. Release times of 50 msec or less may be sufficiently short to eliminate this problem (Dillon, 2001).

b) Dual compressors / automatic volume control:

A type of compression called "Automatic volume control" (AVC) is used in broadcast audiovisual equipment. A compression system can, in principle, enhance the audibility of, long-term and/or short-term speech levels. A slow acting compressor (automatic volume control, AVC) can only account for the overall level of speech and thereby increase the audibility of the long-term speech cues. AVC is known to have relatively long attack and long release times, and it contributes to the time lag or delay in loudness changes in the announcers voice relative to the sudden – onset cheers of the audience. Its release times are usually more than 150 msec and may be as long as several seconds (Hickson, 1994). Because of long attack / release times, it does not respond to rapid fluctuations of sound input and reduces the need for the listener to adjust the volume control manually hence, its name.

Neuman, Bakke, Mackersie, Hellman and Levitt (1995) studied the effect of release time in compression hearing aids and found that the longer release times, 200 msec and 1000 msec, were preferred for the higher-level noises (apartment noise, cafeteria noise). The biggest problem with slow-acting compressor is what happens when the input level varies suddenly. Suppose a person, for some time, has been listening to a softly spoken person in a quiet place. The hearing aid will react by turning up the gain appropriately. If a loud noise then occurs, or a loud talker joins the conversation, the new sound will be amplified with the high gain that was appropriate to the weaker talker. The output will thus be excessive and must be decreased with an appropriate limiter of some type, preferably a compression limiter. Sudden increases in level are very common: they will probably occur every time the aid wearer talks, because his or her mouth is probably closer to the hearing aid than is anybody else's (Dillon, 2001).

The opposite problem, sudden decrease in level, also occurs, but is not so easily fixed. If everyone at a gathering suddenly stops talking to hear what one person is saying, the wearer of an automatic volume control hearing aid may miss the important announcement if the hearing aid still has the gain appropriate to the higher input level that was present a moment before. Having a release time no longer than that necessary to avoid rapid increases in gain during brief pauses in the conversation minimizes this problem.

The system developed by Moore and Glasberg (1988) makes use of two control voltages with different time constants. This system is referred to as the 'dual front-end AGC system'. This system has got two purposes: (1) to compensate for variations in the overall level of the speech from one situation to another by slowly changing its gain; (2) to protect the user from sudden intense transients without affecting the long term gain.

This is achieved by two control voltages to determine the gain. One changes slowly as the listening situation changes. Normally this determines the operation of the system. It has an attack time of roughly a few hundred milliseconds and a recovery time of a few several seconds. The other comes into operation when there is sudden increase in sound level, but its action ceases quickly at the end of the transient. It has an attack time of roughly a few milliseconds and a recovery time of about 100 ms.

Stone, Moore, Alcantara, and Glasberg (1999) implemented four different compression algorithms in wearable digital hearing aids. They are as follows: (1) slow acting dual front-end AGC combined with appropriate frequency equalization (compression threshold of 63 dB SPL and compression ratio of 30); (2) dual front-end AGC with compression threshold of 55 dB SPL and compression ratio of 3); (3) fast acting full dynamic range compression in four channels; and (4) a combination of (2) and (3) above, where each applied less compression than when used alone. They found that speech recognition scores were high for all systems at both 50 and 80 dB SPL. But subjective preference indicated that there was slight preference for dual end AGC system with lesser compression ratio, that is, 3.

Moore, Glasberg and Stone (1991) also found that in the presence of no background sound or continuous speech shaped noise as a background, the dual front end AGC system gives significantly better performance than the adaptive compression and linear amplification.

Moore, Stainsby, Alcantara and Kuhnel (2004) found that the intelligibility of speech at a fixed level, presented in background sounds, is not markedly affected by rather sustained variations of the time constants in a multichannel compression system.

In quiet, performance was best for linear amplification and worst for slow and fast algorithm.

c) Peak detection:

Most compression hearing aids use a technique called peak detection to "track" the peak amplitude of incoming sound waves. If the peak is greater than the compression threshold, the circuit attacks and compresses the signal, which reduces the gain. Once the peak is below the knee-point, the compression releases and then, the gain increases once again. Peak detection allows for a wide variety of times that can separately be specified and assigned as attack and release times; however, these times are constant and fixed for any incoming sound intensity patterns (Armstrong, 1993). Most peak detection systems in hearing aids are adjusted to provide quick attack times and longer, slower release times.

An advantage of peak detection is that it reacts very quickly to increases in environmental sound level. But, with fixed attack/release times, the hearing aid cannot respond differently to different patterns of sound input intensities when needed (Venema, 1999).

d) Adaptive compression:

This type of compression has fixed, quick attack times, but release time vary with the duration of the intense incoming sound. For sudden, intense, transient sound inputs, the release time is short (Venema, 1999).

e) Average detection:

The average detection method looks at the average of the incoming signal over a given length of time. When the average exceeds the knee point of compression, the gain is reduced (Venema, 1999).

CHAPTER III

METHOD

Subjects

Twenty subjects who were post-lingually hearing impaired satisfying the following criteria were included in the study.

1) With mild to moderately-severe sensorineural type of hearing loss in the ear tested (symmetrical and asymmetrical hearing loss).
2) With speech identification scores above 75%.
3) Kannada speaking subjects.
4) Naive hearing aid users.
5) Age ranging from 24 to 58 years with the mean age of 41 years.

Stimulus

The Phonetically Balanced list in Kannada developed by Vandana (1998) was used. Four lists, each with twenty-five words were selected from the original test. The twenty-five words in each of the four lists were iterated thrice in order to get a total of twelve word lists. These word lists are given in the appendix.

Hearing aid description

A non-linear digital behind-the-ear hearing aid with the following features:

- Two compression channels,
- Compression threshold: from 39 to 81 dB and ‘off’,
- Compression ratio: from 1.3 to 2.7 and ‘off’,
- Gain option ranging up to 56 dB in the two channels
- With the facility to select dual or syllabic compression

Prescriptive formula used

For fitting the hearing aid to the subject, National Acoustical Laboratory Non-linear 1 (NAL-NL1) prescriptive formula was used, as this was the default fitting formula for the hearing aid.

Test Environment

Programming the hearing aid to the individual subject's ear as well as evaluating the performance of the hearing aid was carried out in sound treated room in which the ambient noise levels were within permissible limits (re: ANSI S3.1-91, cited in Wilger, 1994).

Instruments

A Pentium IV computer along with NOAH-2 and Connex (V5.0a) software, Hipro (for connecting the hearing aid with the computer) were used for programming the hearing aid.

A calibrated MA53 diagnostic audiometer was used to present the stimuli.

Procedure

The subjects fulfilling the stated criteria were included in the study.

Pure tone thresholds (from 250 Hz to 8 kHz for air conduction and from 250 Hz to 4 kHz for bone conduction) of the test ear were fed into the NOAH software.

- The subject was made to sit comfortably
- The subject was fitted with the hearing aid on the test ear using an appropriately sized ear tip.
- The hearing aid was connected to the Hi-pro that was in turn connected to a computer with the programming software.

- The hearing aid was detected by the Connex (Sifit V5.0a) software after switching the hearing aid 'on' and volume control set at '2'
- The following general settings were selected for first fit
    - Test ear (Right or left ear)
    - Acclimatization level: Two (as all the subjects were naïve hearing aid users)
    - Prescriptive formula: NAL-NL1
    - Acoustical and other parameters were set to default setting
- Frequency shaping option was selected for fine tuning
    - The first fit target gain curve was set by the software.
    - Then, depending on the subjects' need, the low-cut and high-cut gain values and the cut-off frequency values were manipulated during fine-tuning.
- Adjustment of the gain in each of the two channels was done independently.
- The subjects opinion regarding amplification was taken, that is, the subject was asked whether it sounded too loud, too soft or just sufficient, when spoken with normal vocal effort from 4 to 5 feet distance.
- Compression option was chosen. The compression threshold and the compression ratio values set by the software, i.e., default settings were unchanged. Only, the compression type, selecting either syllabic or dual compression, was changed, keeping the compression threshold and compression ratio unchanged for both syllabic and dual compressions.

In order to see whether sufficient amplification was provided to the subject, routine hearing aid evaluation of performance was carried out where the subject's task was to answer five open ended questions and to repeat five paired words in Kannada. The questions and paired words were presented at 45 dB HL through the speakers of the audiometer. For each subject, the level was constant during unaided and aided conditions. The gain provided was considered to be sufficient only if the subject answered all the five questions and repeated all the five-paired words correctly. If the subject did not give 100% correct scores, fine-tuning of the gain was made until the subject answered all five questions and repeated all five paired words. The gain was optimized for individual subjects in order to ensure adequate audibility. Hence, the effect of degree and configuration of hearing loss was overcome by optimizing the gain. In the subjects with asymmetrical hearing loss and having lesser loss in the non-test ear, the non-test ear was blocked in order to avoid its participation.

In the next step, the subject was presented PB word list having 25 words, different list in each of the following conditions, using monitored live voice.

For 50% of the subjects, syllabic compression was evaluated first and then dual compression was evaluated. For the other subjects, dual compression first and then the syllabic compression was evaluated.

To evaluate the hearing aid performance in the presence of noise condition, speech noise was presented at 10 dB SNR, i.e., the level of the speech noise was either 35dB HL with the presentation level for speech being 45 dB HL or 60 dB HL with the presentation level for speech being 70 dB HL. The performance was evaluated in the following two conditions:

(i) Unaided condition:

1) In the presence of noise, with speech level at 45 dB HL and SNR of 10 dB (here in after referred to as UN45 condition)
2) In the presence of noise, with speech level at 70 dB HL and SNR of 10 dB (here in after referred to as UN70 condition)
3) In quiet, with speech presented at 45 dB HL (here in after referred to as UQ45 condition).
4) In quiet, with speech presented at 70 dB HL (here in after referred to as UQ70 condition)

(ii) Aided conditions: Two aided conditions, i.e., syllabic and dual compression:

A) Syllabic compression condition:

1) In the presence of noise, with speech level at 45 dB HL and SNR of 10 dB (here in after referred to as SN45 condition)

2) In the presence of noise, with speech level at 70 dB HL and SNR of 10 dB (here in after referred to as SN70 condition)

3) In quiet, with speech presented at 45 dB HL (here in after referred to as SQ45 condition)

4) In quiet, with speech presented at 70 dB HL (here in after referred to as SQ70 condition)

B) Dual compression condition:

1) In the presence of noise, with speech level at 45 dB HL and SNR of 10 dB (here in after referred to as DN45 condition)

2) In the presence of noise, with speech level at 70 dB HL and SNR of 10 dB (here in after referred to as DN70 condition)

3) In quiet, with speech presented at 45 dB HL (here in after referred to as DQ45 condition)

4) In quiet, with speech presented at 70 dB HL (here in after referred to as DQ70 condition)

Half of the subjects were presented with the order of testing being unaided condition, syllabic compression and dual compression condition and another half of the subjects were presented with the dual compression condition after unaided condition and then with the syllabic condition in order to avoid the order effect.

Hence, a total of 12 PB lists, each presented at each of these 12 conditions were used. The subject was instructed to repeat the words as he/she heard and the tester noted down the responses in a response sheet.

In speech identification testing, each correct response was given the score of ‘one’ and total number of correct responses was noted down for each condition for each subject. The SIS was not converted into percent correct scores. The speech identification scores (SIS) were tabulated and subjected to appropriate statistical analysis.

Subjective preference for compression time constants was evaluated by using two-point scale. At the end of the first aided condition, i.e., either syllabic or dual condition, the subject was instructed that he/she will be fitted with the hearing aid with another setting and word lists would be presented to him or her in that particular setting. The subjects’ task was to compare the quality of speech (word lists) through the hearing aid, between the two settings. The subject either rated the syllabic or dual compression as best.

CHAPTER IV

RESULTS

The data obtained from twenty sensorineural hearing loss subjects were analyzed to investigate the effects of compression types (syllabic compression and dual compression) on speech identification scores (SIS). SPSS (version 10) for windows was used to analyze the following:

a) Effect of type of compression on SIS in quiet when

(i) Speech was presented at 45 dB HL

(ii) Speech was presented at 70 dB HL

b) Effect of type of compression on SIS in noise when

(i) Speech was presented at 45 dB HL

(ii) Speech was presented at 70 dB HL

c) Subjective preference evaluation

a) Effect of type of compression on SIS in quiet

Table 1 and Figure 2 show the mean, standard deviation (SD) and range of SIS in quiet condition, at different input levels. The results show that the variability in the scores is more for the unaided scores compared to aided scores.

Table 1: Mean, SD and range of SIS in quiet condition

CONDITIONS	MEAN	SD	RANGE
UQ45	8.2500	8.9905	0-25
UQ70	22.0500	3.7902	9-25
SQ45	19.2500	4.0636	11-25
SQ70	22.7000	2.6378	17-25
DQ45	19.3500	4.4400	10-25
DQ70	23.4000	2.3065	19-25

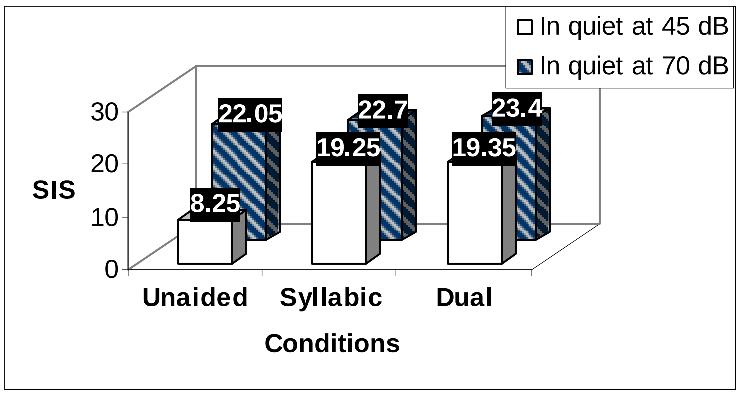

Figure 2: Mean and standard deviations of SIS obtained in quiet

(i) Speech at 45 dB HL:

Repeated measures ANOVA was carried out to study if there was any statistically significant difference between SIS in the unaided, syllabic and dual compression conditions, in quiet. The results revealed a highly significant difference between SIS in quiet, at 45 dB HL, and SIS in all other conditions at 45 dB Hl as was expected [F (2, 38)= 43.293, P<0.001]. However, there was no significant difference observed between SIS in syllabic and in dual compression conditions at 45 dB HL as given Table 2.

Table 2: Comparison of SIS at 45 dB HL in quiet with other conditions in quiet at 45 dB HL using repeated measures ANOVA

CONDITION 1	CONDITION 2	MEAN DIFFERENCE (1-2)
UQ45	SQ45	-11.000**
	DQ45	-11.100**
SQ45	UQ45	11.000**
	DQ45	-0.100
DQ45	UQ45	11.100**
	SQ45	0.100

**: Mean difference is significant at the .001 level

(ii) Speech at 70 dB HL:

At 70 dB HL, results showed no statistically significant difference [F (2, 38) = 3.175] obtained between the SIS in all the three conditions in quiet. Higher SIS was obtained at higher presentation level when compared to lower presentation level in all the conditions as shown in Table 3. However, the variation in the SIS was large for the unaided compared to the aided condition as reflected in standard deviation values in Table 1.

Table 3: Comparison of SIS between 45 dB HL signal level and 70 dB HL signal level in quiet

Pair	Paired difference	't' value
	Mean	
UQ45-UQ70	-13.8000	7.894**
SQ45-SQ70	-3.4500	4.245**
DQ45-DQ70	-4.0500	5.195**

**: Mean difference is significant at the .001 level

b) Effect of compression type on SIS in noise

Table 4 shows the mean, SD and range values, in noise, at different input levels.

Table 4: Mean, SD and range of SIS obtained in noise condition

CONDITIONS	MEAN	SD	RANGE
UN45	5.7000	7.3849	0-22
UN70	18.600	5.0095	7-24
SN45	15.1000	5.6652	6-25
SN70	21.2500	3.2747	14-25
DN45	16.1500	5.0500	7-24
DN70	19.8500	4.2953	11-25

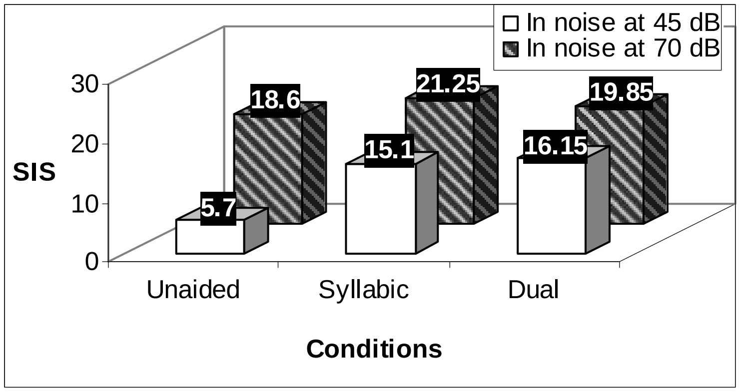

Figure 3: Mean values of SIS obtained in noise condition

(i) At 45 dB HL speech level:

There was a statistically significant difference at .001 significance level, between unaided SIS at 45 dB HL and scores obtained in all other conditions, in noise [F (2, 38) = 63.831, P< 0.001].

Table 5: Comparison between the SIS at 45 dB HL in noise with other conditions in noise AT 45 dB HL using repeated measures ANOVA

CONDITION 1	CONDITION 2	MEAN DIFFERENCE (1-2)
UN45	SN45	-9.400**
	DN45	-10.450**
SN45	UN45	9.000**
	DN45	-1.050
DN45	UN45	10.450**
	SQ45	1.050

**: Mean difference is significant at the .001 level

Results similar to quiet condition were observed, that is, at 45 dBHL, there was no significant difference observed between syllabic and dual compression types as shown in Table 5.

(ii) At 70 dB HL speech level:

Table 6: Comparison of SIS at 70 dB HL in noise with other conditions in noise at 70 dB HL using repeated measures ANOVA

CONDITION 1	CONDITION 2	MEAN DIFFERENCE (1-2)
UN70	SN70	-2.650*
	DN70	-1.250
SN70	UN70	2.650*
	DN70	1.400
DN70	UN70	1.250
	SN70	-1.400

*: Mean difference is significant at the 0.05 level

At higher presentation level also, results were similar to that in quiet condition except that there was a statistically significant [F (2, 38) = 6.309, P< 0.05] difference between unaided and syllabic condition, but no statistically significant between syllabic and dual compression as shown in Table 6.

Table 7: Comparison of SIS between 45 dB HL signal level and 70 dB HL signal level in noise

Pairs	Paired difference	't' value
	Mean	
UN45-UN70	**-12.9000**	**9.602****
SN45-SN70	**-6.1500**	**6.195****
DN45-DN70	**-3.7000**	**4.388****

**: Mean difference is significant at the .001 level

Higher scores were obtained at 70 dBHL speech level, when compared to scores obtained at 45 dBHL in both unaided and aided conditions in noise, which is statistically significant at 0.001 level of significance as given in Table 7.

However, the variations in the scores were more for unaided condition compared to scores obtained in the aided conditions as indicated by the range in Table 4.

.***Table 8: Paired comparison between SIS obtained in quiet and in noise***

Pairs	Paired difference	't' value
	Mean	
UN45-UN45	**2.5500**	**4.073***
UN70-UN70	**3.4500**	**5.969****
SN45-SN45	**4.1500**	**5.713****
SN70-SN70	**1.4500**	**3.746***
DN45-DN45	**3.2000**	**5.250****
DN70-DN70	**3.5500**	**5.957****

**: Mean difference is significant at the 0.001 level

*: Mean difference is significant at the 0.05 level

Paired comparison between SIS obtained in quiet and in noise using paired't' test Table 8. In all the conditions, SIS obtained in quiet was significantly higher compared to that in noise conditions.

Table 9: Correlation values obtained between SIS in quiet and noise condition

Pairs	Correlation
UQ45-UN45	0.960**
UQ70-UN70	0.863**
SQ45-SN45	0.826**
SQ70-SN70	0.850**
DQ45-DN45	0.843**
DQ70-DN70	0.886**

**: P< 0.001

Table 9 shows the direction and the degree of correlation between the SIS obtained in quiet and in noise. Results show that there is highly positive correlation and it is highly significant, that is, the SIS obtained in quiet and in noise conditions move in the same direction.

d) Subjective preference evaluation

This revealed that 60% of subjects showed preference for 'dual' compression option, where as only 30% subjects preferred syllabic compression, and 10% subjects could not find any difference between the two conditions

CHAPTER V

DISCUSSION

Results of the study indicate that though the SIS do not significantly differ between syllabic and dual compression, there was a slight preference towards dual compression and at higher presentation level, the SIS do not degrade. The results are discussed in detail below.

(a) Effect of compression type on SIS in quiet

In the present study, results showed that there was no statistically significant difference seen at 45 dB HL between syllabic and dual compression, in quiet condition. Moore, Glasberg and Stone (1991) found superior results for the dual time constant compressor compared to adaptive compression but the reasons for this, or whether the result was applicable only to their sample, are not clear. However, Moore, Stainsby, Alcantara and Kuhnel (2004) have reported that in quiet, the performance was best for linear amplification and worst for slow and fast algorithm. This slight poor performance was attributed to the combined effect of low level expansion (applied during the gaps between speech items) and the very long time constants of the slow compression system.

At higher presentation levels, also, there was no significant difference found between the two aided conditions, and higher scores were obtained at 70 dB HL when compared to lower presentation level, i.e., 45 dB HL. The results of the study conducted by Moore, Glasberg and Stone (1991) support the findings of the present study. They found superior results obtained for both dual and adaptive compression at higher presentation levels also. Hence, at higher presentation level, speech recognition is not degraded, and at the same time, compression does not provide a significant benefit over the unaided condition, nor does any difference exists between the performances of different compression types.

In the present study, the SIS did not reduce at higher presentation levels, in the aided condition. A non-linear hearing aid reduces the gain at high input levels, and hence the hearing aid output was still comfortable.

(b) Effect of compression type on SIS in noise

In noise condition, results similar to quiet condition were obtained except that there was a statistically significant difference between unaided and syllabic condition, at 70 dB HL, and no statistically significant difference between syllabic and dual compression. Moore, Glasberg and Stone (1991), also, found better results for dual condition compared to adaptive compression at 65 dB SPL, i.e., at lower speech presentation level in the presence of continuous noise presentation. They found that both dual compression system and adaptive compression system showed high scores at high presentation levels, also, in the presence of cafeteria noise.

Several researchers have shown that speech recognition is degraded when speech is presented at higher presentation levels in normal subjects (French & Steinberg, 1947, cited in Hornsby & Ricketts, 2001; Goshorn & Studebaker, 1994), especially, at poorer SNRs (Pollack & Picket, 1958, cited in Hornsby & Ricketts, 2001).

Paired comparison of SIS in quiet and in noise showed that scores in quiet were better than in noise, for all the conditions. Moore, Laurence and Wright (1985) measured speech reception threshold in two levels of noise (‘babble’), 60 and 75 dB SPL. Seven out of eight subjects showed an improvement in the speech reception threshold for at least one of the two noise levels, although the size of the improvement differed considerably from one subject to the other. These higher scores may be due to binaural fitting of the hearing aid in their study.

Though there was no statistically significant difference found between the two compression types (syllabic and dual), there are studies showing a significant benefit of compression over linear hearing aid systems (Moore, Glasberg & Stone, 1991; Stone, Moore, Alcantara, & Glasberg, 1999).

(c) Subjective evaluation

The present study showed that there was a preference by majority of the subjects for the dual compression type over syllabic compression type. A similar observation was made by Neuman, Bakke, Mackersie, Hellman and Levitt (1995) who found a significant interaction between the release time and the type of background noise, with the tendency of increasing preference for increasing release times in some noise types. However, single channel compression was used in the Neuman, Bakke, Mackersie, Hellman and Levitt's study and the differences in the preference were much less pronounced. Similar results were also found by several other investigators (Hansen, 2002; Stone, Moore, Alcantara & Glasberg, 1999; Neuman, Bakke, Mackersie, Hellman & Levitt, 1998). Stone, Moore, Alcantara and Glasberg (1999) implemented four different compression algorithms in wearable digital hearing aids and found that speech recognition scores were high for all systems at both 50 and 80 dB SPL. But subjective preference indicated that there was slight preference for dual end AGC system with lesser compression ratio, that is, 3.

Though there was no significant difference found in SIS between syllabic and dual condition, subjective evaluation showed a preference towards dual compression system.

These results indicate that the use of quality or preference judgments will be an useful addition to measures of speech intelligibility while prescribing hearing aids or hearing aid features. And, in the view of present finding, fast acting compression, a method for compensating for recruitment, may not be the most optimal solution for most of the sample of subjects studied.

FURUTE DIRECTIONS

1) For the detailed evaluation of subjective preference, a rating scale could be made use of.

2) The performance with syllabic and dual compression types could be compared with the linear system.

3) The effect of compression types on SIS in the presence of different types noise could be evaluated.

4) Effect of compression type on different degrees and configurationsof hearing loss could be done.

CHAPTER VI

SUMMARY AND CONCLUSION

The typical means, by which a hearing aid compensates for recruitment in a sensorineural hearing impaired, is the use of a non-linear compressor circuit. The general aim of the compressor is to provide higher gain for softer sounds than for louder sounds (Hansen, 2001). Although fast-acting compression systems are becoming integral components of modern hearing aids research results have not consistently demonstrated their benefit over linear amplification. There is, however, no standard method of specifying the most effective compression strategy for a particular hearing-impaired person, mainly because of the larger number of the parameters required to specify, fit and evaluate a compression system (Olsen, Olofsson & Hagerman, 2004). There are a number of studies (Moore, Stainsb, Alcantara, & Kuhnel, 2004; Moore, & Glasberg, 1988; Plomp, 1988) done in compression time constants and the results of these studies are equivocal. Hence, the present study aimed at investigating the effects of compression types (syllabic compression and dual compression) on speech identification scores (SIS) in quiet condition (at 45 dB HL and at 75 dB HL level of speech) and in noise condition, with the SNR of 10 dB(at 45 dB HL and at 75 dB HL level of speech).

In the present study, twenty adult subjects with mild to moderately-severe sensorineural hearing loss were tested. The SIS was measured across three different conditions (unaided, aided-syllabic and aided-dual conditions) in quiet and in noise at 10 dB SNR.

The following conclusions were drawn from the study:

(a) There was no significant difference in SIS obtained between syllabic and dual condition, in quiet and in noise, at different input levels.

(b) The higher-level presentation of signal did not degrade the speech recognition, rather, showed superior results.

(c) Majority of the subjects preferred dual compression system.

Thus, it can be inferred that though there was no statistically significant difference between the SIS in syllabic and dual compression conditions, subjective preference was for dual compression.

REFERENCES

Armstrong, S. (1993). The Dynamics of Compression: Some Key Elements Explored. *The Hearing Journal*, 46(11), 43-47.

Boothroyd, A., Springer, N., Smith, L. & Schulmann, J. (1988). Compression amplification and profound hearing loss. *Journal of speech and hearing research*, 31, 362-376.

Buus, S. (1999). Loudness functions derived from measurements of temporal and spectral integration of loudness. In Hansen, M. (2002). Effect of multichannel compression time constants on subjectively perceived sound quality and speech intelligibility. *Ear & hearing*, 23, 369-380.

Byrne, D., Dillon, H., Ching, T., Katsch, R., & Keidser, G., (2001). NAL-NL1 procedure for fitting nonlinear hearing aids: characteristics and comparisons with other procedures. *Journal of the American Academy of Audiology*, 12, 37-52.5

De Gennaro S, Braida L, Durlach, N. (1986). Multichannel compression for severely impaired listeners. Journal of Rehabilitative Research Development, 23 (1), 17-24.

Dillon, H. (1988). Compression Hearing Aids. In R.E. Sandlin (Ed.), Handbook for hearing aid amplification (Vol.1). Boston: College Hill.

Dillon, H. (1996). Compression? Yes, but for low or high frequencies, for low or high intensities and with what response times. *Ear & Hearing*, 17(4), 287-307.

Dillon, H. (2001). Hearing Aids. Sydney: Boomerang Press.

French, N. R., and Steinberg, J. C. (1947). Factors governing the intelligibility of speech sounds. In Hornsby, B. W. Y & Ricketts, T. A. (2001). The effects of compression ratio, signal-to-noise ratio, and level on speech recognition in normal hearing listeners. *Journal of Acoustical Society of America*, 109 (6), 2964-2973.

Goshorn, E. L. and Studebaker, G. A. (1994). Effects of intensity speech recognition in high and low frequency bands. *Ear & Hearing*, 15 (6), 454-460.

Hansen, M. (2002). Effect of multichannel compression time constants on subjectively perceived sound quality and speech intelligibility. *Ear & hearing*, 23, 369-380.

Hickson, L.M.H. (1994). Compression Amplification in Hearing Aids. *American Journal of Audiology*, 3, 51-63.

Hickson, L. M. H., Dodd, B. & Byrne, D. (1994). In Hansen, M. (2002). Effect of multichannel compression time constants on subjectively perceived sound quality and speech intelligibility. *Ear & hearing*, 23, 369-380.

Hohmann, V. & Kollmeier, B. (1995). The effect of multichannel dynamic compression on speech intelligibility. *Journal of the Acoustical Society of America*, 97 (2), 1191-1195.

Johnson, W.A. (1993). Beyond AGC – O and AGC – I: Thoughts on a new default standard amplifier. *The Hearing Journal*, 46(11), 37-42.

Keidser, G. & Grant, F. (2001). The Preferred number of channels (one, two or four) in NAL – NL1 prescribed Wide Dynamic Range Compression (WDRC) devices. *Ear & Hearing*, 22 (6), 516-527.

Kiessling, J. Sliffens, T.C. (1991). Clinical evaluation of a programmable three-channel Automatic Gain Control amplification system. *Audiology*, 30(2), 70-81.

Killion, M.C., Staab, W., & Preves, D. (1990). Classifying Automatic Signal Processor. *Hearing Instruments*, 41 (8), 24-26.

Killion, M.C. (1996). Compression distinctions. In Sandlin R.E. (Edn). Textbook of Hearing Aid Amplification (2nd Edn.). San Diego: Singular Publishing Group.

Kuk, F. K. (1999). Hearing aid design considerations for optimally fitting the youngest patients. *The Hearing Journal*, 52 (4), 48-55.

Lindholm, J., Dorman, M., Taylor, B. & Hannley, M., (1988). Stimulus Factors Influencing the Identification of Voiced Stop Consonants by Normal Hearing and Hearing Impaired Adults. *Journal of the Acoustical Society of America*, 83(4), 1608-1614.

Lippmann, R, Braida, L. & Durlach M. (1981). Study of multichannel amplitude compression and linear amplification for persons with sensorineural hearing loss. *Journal of the Acoustical Society of America*, 69 (2), 524-534.

Moore, B. C. J., Stainby, T. H., Alcantara, J. I. & Kuhnel, V. (2004). The effect on speech intelligibility of varying time constants in a digital hearing aid. *International Journal of Audiology*, 43 (7), 399-410.

Moore, B.C.J., Peters, R.W. & Stone, M.A., (1999). Benefits of linear amplification and multichannel compression for speech comprehension in backgrounds with spectral and temporal dips. *Journal of the Acoustical Society of America*, 105(1), 400-411.

Moore, B. & Glasberg, B. (1986). A comparison of two channel and single channel compression hearing aids. *Audiology*, 25 (4-5), 210-226.

Moore, B. & Glasberg, B., (1988). A comparison of four methods of implementing Automatic Gain Control (AGC) in hearing aids. *British Journal of Audiology*, 22(7), 93-104.

Moore, B. Glasberg, B. & Stone, M. (1991). Optimization of slow-acting automatic gain control system for use in hearing aids. *British Journal of Audiology*, 25(3), 171-182.

Moore, B. C., Laurence, R. F. & Wright, D. (1985). Improvements in speech intelligibility in quiet and in noise produced by two-channel compression hearing aids. *British Journal of Audiology*, 19 (3), 175-187.

Nabelek, I. (1983). Performance of hearing impaired using various types of amplitude compression. *Journal of the Acoustical Society of America*, 74, 314-320.

Neuman, A., Bakke, m., Mackesie, C., Hellman, S. & Levitt H. (1995). Effect of release time in compression hearing aids: Paired-comparison Judgments of Quality. *Journal of the Acoustical Society of America*, 98, 3182-3187.

Neuman, A., Bakke, M., Mackesie, C., Hellman, S. & Levitt H. (1998). The effect of compression ratio and release time on the categorical rating of sound quality. *Journal of the Acoustical Society of America*, 103 (5 pt 1), 2273-2281.

Olsen, H., Olofeson, A. & Hagerman, B. (2004). The Effect of presentation level and compression characteristics on sentence recognition in modulated noise. *International Journal of Audiology*, 43(5), 283-294.

Pearsons, K. S., Bennet, R. L. & Fidell, S. (1976). Speech levels in various exreriments. In Sandlin R.E. (Edn). Textbook of Hearing Aid Amplification (2nd Edn.). San Diego: Singular Publishing Group.

Plomp, R. (1988). The negative effect of amplitude compression in multichannel hearing aids in the light of modulating transfer function. *Journal of the Acoustical Society of America*, 83(6), 2322-2327.

Plomp, R. (1994). Noise, Amplification and Compression: Considerations of three main issues in hearing aid design. *Ear & Hearing*, 15(1), 2-12.

Pollack, I. & Pickett, J. M. (1958). Masking of speech by noise at higher sound levels. In Hornsby, B. W. Y & Ricketts, T. A. (2001). The effects of compression ratio, signal-to-noise ratio, and level on speech recognition in normal hearing listeners. *Journal of the Acoustical Society of America*, 109 (6), 2964-2973.

Rankovic, C. Freyman, R. & Zurek, P. (1992). Potential benefits of adaptive frequency – gain characteristics for speech reception in noise. *Journal of the Acoustical Society of America*, 91(1), 354-362.

Ruggero, M., & Rich, N. (1991). Furosemide alters organ of corti mechanism: Evidence for feedback of outer hair cells upon the basilar membrane. In Hansen, M. (2002). Effect of multichannel compression time constants on subjectively perceived sound quality and speech intelligibility. *Ear & hearing*, 23, 369-380.

Staab, W. J. (1996). Limiting systems in hearing aid. In Sandlin R.E. (Edn). Textbook of Hearing Aid Amplification (2nd Edn.). San Diego: Singular Publishing Group.

Stone, M.A. Moore, B., Alcantara & Glasberg, S. (1999). Comparison of four different compression algorithms in wearable digital hearing aids. *Journal of the Acoustical Society of America*, 106 (6), 3603-3619.

Souza, P. E. & Turner, C.W. (1998). Multichannel compression, temporal cues, and audibility. *Journal of Speech, Language, and Hearing Research*, 41, 315-326.

Vandana, S. (1998). Speech Identification Test for Kannada Speaking Children. Unpublished Independent project, University of Mysore, Mysore.

Van Hasten-de Bruijn, H. E., Siddone, C., Arschler, W. A. & Verschure, H. (19970. Design of two syllabic non-linear multichannel signal processes and the results of speech test in noise. *Ear & hearing*, 18, 26-33.

Venema, T.H. (1999). The Many Races of Compression. In Sandlin R.E. (Edn). Textbook of Hearing Aid Amplification (2nd Edn.). San Diego: Singular Publishing Group.

Van Hasten-de Bruijn, H.E., Sidonne, C., Arschler, W.A. & Verschuure, H. (1997). Design of two syllabic nonlinear multi-channel signal processes and the results of speech tests in noise. *Ear & Hearing*, 18, 26-33.

Walker, G., Byrne, D. & Dillon, H. (1984). The effects of multichannel compression/expansion amplification on the intelligibility of non-sense syllables in noise. *Journal of the Acoustical Society of America*, 76(3), 746-759.

Wilber, A. (1994). In Katz (1994). Handbook of clinical Audiology. Baltimore: Williams & Wilkins.

APPENDIX
TEST LISTS

List 1	List 2	List 3
¯ÉÆÃl - /lo:ta/	ªÉÄÃdÄ – /me:dzu/	vÀmÉÛ - /tate/
Kt¤ - /e:ni/	E° - /ili/	Vt¤ - /gini/
ZÁPÀÄ – /t a:ku/	¸ÀÆf – /su:dzi/	PÁUÉ – /ka:ge/
§¸ÀÄì – /basu/	vÀ¯É – /tale/	ªÀÄÆUÀÄ – mu:gu/
UÀÆ¨É - /gu:be/	Q« - /kivi/	¯Áj – /la:ri/
PÀvÀÄÛ – /katu/	¥É£ÀÄß - /penu/	©ÃUÀ – /bi:ga/
¯Áj - /la:ri/	ªÀÄgÀ – /mara/	¸ÉÃ§Ä – /se:bu/
ªÀÄ£É – /mane/	§¼É – /bale/	ªÉÄÃPÉ – /me:ke/
£À°è – /nali/	PÁ®Ä – /ka:lu/	bÀwæ – /t atri/
ªÉÄÃPÉ - /me:ke/	UÀAmÉ – /gante/	UÀÆ¨É – /gu:be/
ªÉÆ® - /mola/	¸ÀgÀ - /sara/	PÀvÀÄÛ – /katu/
PÁUÉ - /ka:ge/	ZÉAqÀÄ – /t endu/	§¸ÀÄì - /basu/
¸ÉÃ§Ä - /se:bu/	gÉÊ®Ä - /railu/	N¯É – /o:le/
©ÃUÀ - /bi:ga/	PÁgÀÄ – /ka:ru/	£À°è - /nali/
PÉÆÃ½ - /ko:li/	N¯É – /o:le/	ªÀÄ£É – /mane/
ºÀÆªÀÅ - /hu:vu/	D£É – /a:ne/	ZÁPÀÄ – /t a:ku/
ªÀÄÆUÀÄ - /mu:gu/	vÀmÉÛ – /tate/	zÁgÀ – /da:ra/
ºÀ¸ÀÄ – /hasu/	Vt¤ – /gini/	¯ÉÆÃl – /lo:ta/
ªÀÄ¼É – /male/	ºÁªÀÅ – ha:vu	ªÀÄ¼É - /male/
PÀ¥Éà – /kape/	£Á¬Ä – na:ji	Kt¤ – /e:ni/
PÀtÄÚ - /kanu/	ºÀ®Äè – halu	ªÉÆ® - /mola/
zÁgÀ - /da:ra/	PÁ¸ÀÄ – ka:su	PÀ¥Éà – /kape/
bÀwæ – /t atri/	¸ÀÆAiÀÄð - su:rja	PÁUÉ – /ka:ge/
aÃ® - /t i:la/	¤ÃgÀÄ – ni:ru	ºÀÆªÀÅ – /hu:vu/
«ÄÃ£ÀÄ - /mi:nu/	J¯É – ele	PÀtÄÚ – /kanu/

List 4	List 5	List 6
ºÁªÀÅ – /ha:vu/	PÀtÄÚ – /kanu/	¸ÀgÀ - /sara/
vÀ¯É – /tale/	ºÀÆªÀÅ – /hu:vu/	PÁgÀÄ – /ka:ru/
gÉÊ®Ä – /railu/	PÁUÉ – /ka:ge/	¥É£ÀÄß – /penu/
UÀAmÉ – /gante/	PÀ¥Éà – /kape/	¤ÃgÀÄ – /ni:ru/
¸ÀÆf – /su:dzi/	ªÉÆ® - /mola/	§¼É – /bale/
ªÉÄÃdÄ – /me:dzu/	Kt – /e:ni/	D£É – /a:ne/
aÃ® - /t i:la/	ªÀÄ¼É – /male/	ZÉAqÀÄ – /t endu/
J¯É – /ele/	¯ÉÆÃl – /lo:ta/	ºÀ®Äè – /halu/
PÁ®Ä – /ka:lu/	zÁgÀ – /dara/	ªÀÄgÀ – /mara/
PÁ¸ÀÄ – /ka:su/	ZÁPÀÄ – /t a:ku/	«ÄÃ£ÀÄ – /mi:nu/
¸ÀÆAiÀÄð – /su:rja/	ªÀÄ£É – /mane/	£Á¬Ä – /na:ji/
E° - /ili/	£À°è – /nali/	PÉÆÃ½ - /ko:li/
Q« - /kivi/	N¯É – /o:le/	Q« - /kivi/
PÉÆÃ½ - /ko:li/	§¸Àì – /basu/	E° - /ili/
£Á¬Ä – /na:ji/	PÀvÀÄÛ – /katu/	¸ÀÆAiÀÄð– /su:rja/
«ÄÃ£ÀÄ - /mi:nu/	UÀÆ¨É – /gu:be/	PÁ¸ÀÄ – /ka:su/
ªÀÄgÀ – /mara/	bÀwæ – /t atri/	PÁ®Ä – /ka:lu/
ºÀ®Äè – /halu/	ªÉÄÃPÉ – /me:ke/	J¯É – /ele/
ZÉAqÀÄ – /t endu/	¸ÉÃ§Ä – /se:bu/	aÃ® - /t i:la/
D£É – /a:ne/	©ÃUÀ – /bi:ga/	ªÉÄÃdÄ –/me:dzu/
§¼É – /bale/	¯Áj – /la:ri/	¸ÀÆf – /su:dzi/
¤ÃgÀÄ – /ni:ru/	ªÀÄÆUÀÄ – /mu:gu/	UÀAmÉ – /gante/
¥É£ÀÄß – /penu/	PÁUÉ – /ka:ge/	gÉÊ®Ä – /railu/
PÁgÀÄ – /ka:ru/	Vt – /gini/	vÀ¯É – /tale/
¸ÀgÀ – /sara/	vÀAmÉÖ – /tate/	ºÁªÀÅ – /ha:vu/

List 7 List 8 List 9

UÀÆ¨É - /gu:be/
ªÉÄÃPÉ - /me:ke/
PÉÆÃ½ - /ko:li/
PÀ¥Éà – /kape/
«ÄÃ£ÀÄ - /mi:nu/
¯ÉÆÃ l - /lo:ta/
PÀvÀÄÛ – /katu/
ªÉÆ® - /mola/
ºÀÆªÀÅ - /hu:vu/
PÀ t ÄÚ – /kanu/
K t - /e:ni/
¯Á j - /la:ri/
PÁUÉ - /ka:ge/
ªÀÄÆUÀÄ - /mu:gu/
zÁgÀ – /dara/
ZÁPÀÄ – /t a:ku/
ªÀÄ£É – /mane/
¸ÉÃ§Ä – /se:bu/
ºÀ¸ÀÄ – /hasu/
bÀwæ – /t atri/
§¸ÀÄì – /basu/
£À°è – /nali/
©ÃUÀ – /bi:ga/
ªÀÄ¼É - /male/
aÃ® - /t i:la/
Q« - /kivi/
UÀAmÉ - /gante/
N¯É - /o:le/
£Á¬Ä - /na:ji/
J¯É – /ele/
ªÉÄÃdÄ - /me:dzu/
¥É£ÀÄß – /penu/
¸ÀgÀ – /sara/
D£É - /a:ne/
ºÀ®Äè - /halu/
E° -/ili/
ªÀÄgÀ - /mara/
ZÉÆqÀÄ - /t endu/
vÀmÉÖ – /tate/
PÁ¸ÀÄ - /ka:su/
¸ÀÆ f - /su:dzi/
§¼É – /bale/
gÉÊ®Ä – /railu/
V t – /gini/
¸ÀÆAiÀÄð - /su:rja/
vÀ¯É – /tale/
PÁ®Ä - /ka:lu/
PÁgÀÄ - /ka:ru/
ºÁªÀÅ - /ha:vu/
¤ÃgÀÄ - /ni:ru/
«ÄÃ£ÀÄ - /mi:nu/
aÃ® - /t i:la/
bÀwæ - /t atri/
zÁgÀ – /dara/
PÀ t ÄÚ – kanu/
PÀ¥Éà – /kape/
ªÀÄ¼É – /male/
ºÀ¸ÀÄ – /hasu/
ªÀÄÆUÀÄ - /mu:gu/
ºÀÆªÀÅ - /ha:vu/
PÉÆÃ½ - /ko:li/
©ÃUÀ - /bi:ga/
¸ÉÃ§Ä - /se:bu/
PÁUÉ - /ka:ge/
ªÉÆ® - /mola/
ªÉÄÃPÉ - /me:ke/
£À°è – /nali/
ªÀÄ£É – /mane/
¯Á j - /la:ri/
PÀvÀÄÛ - /katu/
UÀÆ¨É - /gu:be/
§¸ÀÄì - /basu/
ZÁPÀÄ - /t a:ku/
J t - /e:ni/
¯ÉÆÃ l - /lo:ta/

List 10

J¯É - /ele/
¤ÃgÀÄ - /ni:ru/
¸ÀÆ0iÀÄð - /su:rja/
PÁ¸ÀÄ - /ka:su/
ºÀ®Äè _ /halu/
ºÁªÀÅ - /ha:vu/
£Á¬Ä - /na:ji/
Vt - /gini/
vÀmÉÖ - /tate/
D£É - /a:ne/
N¯É - /o:le/
PÁgÀÄ - /ka:ru/
gÉÊ®Ä - /railu/
ZÉ0qÀÄ - /t endu/
¸ÀgÀ – /sara/
UÀ0Mé - /gante/
PÁ®Ä - /ka:lu/
§¼É - /bale/
ªÀÄgÀ - /mara/
¥É£ÀÄß - penu/
Q« - /kivi/
vÀ¯É - /tale/
¸ÀÆ f - /su:dzi/
E° - /ili/
ªÉÄÃdÄ - /me:dzu/

List 11

ªÉÆ® - /mola/
ZÁPÀÄ - /t a:ku/
PÀvÀÄÛ - /katu/
©ÃUÀ - /bi:ga/
vÀmÉÖ - /tate/
PÀ t ÀÄ - /kanu/
K t - /e:ni/
ªÀÄ£É - /mane/
UÀÆ¨É - /gu:be/
¯Á j - /la:ri/
ºÀÆªÀÅ - /hu:vu/
ªÀÄ¼É - /male/
£À°è - /nail/
bÀwæ - /t atri/
ªÀÄÆUÀÄ - /mu:gu/
PÁUÉ - /ka:ge/
¯ÉÆÃl - /lo:ta/
N¯É - /o:le/
ªÉÄÃPÉ - /me:ke/
PÁUÉ - /ka:ge/
PÀ¥Éà - /kape/
zÁgÀ - /da:ra/
§¸ÀÄì - /basu/
¹Ã§Ä - /se:bu/
Vt - /gini/

List 12

§¼É - /bale/
«ÄÃ£ÀÄ - /mi:nu/
¸ÀÆ0iÀÄð - /su:rja/
ªÉÄÃdÄ - /me:dzu/
ºÁªÀÅ - /ha:vu/
¸ÀgÀ – /sara/
D£É - /a:ne/
£Á¬Ä - /na:ji/
PÁ¸ÀÄ - /ka:su/
¸ÀÆ f - /su:dzi/
PÁgÀÄ - /ka:ru/
ZÉ0qÀÄ - /t endu/
PÉÆÃ½ - /ko:li/
PÁ®Ä - /ka:lu/
UÀ0mÉ - /gante/
¥É£ÀÄß - /penu/
ºÀ®Äè - /halu/
Q« - /kivi/
J¯É - /ele/
gÉÊ®Ä - /railu/
¤ÃgÀÄ - /ni:ru/
ªÀÄgÀ - /mara/
E° - /ili/
aÃ® - /t i:la/
vÀ¯É - /tale/