ACOUSTIC ANALYSIS OF THE SPEECH PROCESSED THROUGH THREE AMPLIFICATION STRATEGIES AND THEIR EFFECT ON SPEECH RECOGNITION SCORES OF INDIVIDUAL WITH SEVERE HEARING IMPAIRMENT

Registration No: 05AUD008
GUNJAN CHAND

A Dissertation submitted in part fulfillment for the
degree of Master of Science (Audiology)
University of Mysore,
Mysore.

ALL INDIA INSTITUTE OF SPEECH AND HEARING
NAIMISHAM CAMPUS, MANASAGANGOTHRI
MYSORE-570006

APRIL 2007

Dedicated to

"Mummy-Papa

&

my guides"

Certificate

This is to certify that this dissertation entitled **"Acoustic analysis of the Speech processed through three Amplification strategies and their effect on Speech Recognition Scores of individual with Severe Hearing Impairment"** is a bonafide work in part fulfillment for the degree of Master of Science (Audiology) of the student Registration No. 05AUD008. This has been carried out under the guidance of a faculty of this institute and has not been submitted earlier to any other University for the award of any other Diploma or Degree.

V. Basavaraj

Dr.Vijayalakshmi Basavaraj
Director
All India Institute of Speech and Hearing
Manasagangothri
Mysore -570006.

Mysore
April, 2007

Certificate

This is to certify that this dissertation entitled **"Acoustic analysis of the Speech processed through three Amplification strategies and their effect on Speech Recognition Scores of individual with Severe Hearing Impairment"** has been prepared under my supervision and guidance. It is also certified that this dissertation has not been submitted earlier to any other University for the award of any other Diploma or Degree.

V. Basavaraj

Dr.Vijayalakshmi Basavaraj
Guide
Director
All India Institute of Speech and Hearing
Manasagangothri,
Mysore -570006.

Mysore
April, 2007

Mr Ajisn K. Abraham
Co-Guide
Reader, H.O.D Electronics
All India Institute of Speech and Hearing
Manasagangothri,
Mysore -570006.

Declaration

This is to certify that the dissertation entitled **"Acoustic analysis of the Speech processed through three Amplification strategies and their effect on Speech Recognition Scores of individual with Severe Hearing Impairment"** is the result of my own study and has not been submitted earlier to any other university for that award of any degree or diploma.

Register No. 05AUD008

Mysore

April, 2007

ACKNOWLEDGEMENT

/ sincerely thank my guide **Dr. Vijayalakshmi Basavaraj** *Director, AIISH,., for .her support and guidance. "Thank you Ma'am, for your valuable suggestions, unending patience, words of encouragement and motivation throughout the study. "*

Sincere thanks to my guide **Mr Ajish Abraham,** H.O.D Electronics dept, *AIISH. Thank you sir for everything you have done for me, you had been like a father, building my confidence. There had been lots of ups and downs.........but you were always there to support me.*

It would have been impossible to complete the study without your help Brinda...you have done a lot for me.........thanks a lot dear......all the best for your future endevours

Heartfelt thanks to ***Mummy, Papa, Monu, tonu*** *for your unconditional love, care and faith in me. Your support helps me to be strong and confident in life and deal with the hardships of life.*

Special thanks to *romil bhai ,* ***because of you today I am in AIISH__thanks for your valuable guidance and love.***

Minakshi, Sumita ,Rima,you ***have been wonderful and caring seniors to me, all the best di's.***

I wonder what I would have done without you being there to give me your unbiased suggestions and unconditional love, care, ***Rohu*** *Your warm-heartedness makes you a wonderful sis and friend who is special in a way that no one else could be...Thanks a lot for everything dear*

I wish to thank all the subjects who participated in the study.

/ express my sincere gratitude to ***all my teachers at AIISH*** *especially* ***Animesh*** *sir,* ***Asha mam, Vanaja mam,Manjula mam, Rajlakshmi mam*** *for imparting knowledge and being a source of inspiration.*

I also thank ***all my teachers at A. Y.J.N.H.H, Kolkata, especially*** *Sinha sir for laying the foundation for us and for moulding us into what* ***we*** *are today.*

A special thanks to Vasanthalakshmi Mam for all the help rendered during the statistical analysis, Chudamani mam for helping in calibration of the instuments.

I am lucky to have you as a friend ***neha***.............*thanks for all the help and support dear*.........*all the best.*

I would also like to thank my juniors especially ***priyashree, divyashree*** *,Leahfor there timely help.*

Rahana, Pri, Shruti, Sumesh, Dayal Shibasis, Yatin, Nambi, Maria, Swappu, Mitu, Niraj, Deeps, *and all my other classmates... Thank you all for being such great friends and being so willing to help me when I needed it. I hope all of you have a nice life...*

The ***library staff*** *merits special thanks, for being very kind and helpful.*

Last but not the least, thanks to ***Shivappa*** *and Prasadfor the endless Xeroxes and printouts.*

Contents

Sl No.	*Title*	*Page no.*
1.	Introduction	1-6
	Need for the study	5
	Objectives	5
	Hypothesis	6
2.	Review of literature	7-29
	Amplification strategies	8
	Characteristics of compression hearing aids	9
	Different approaches to Compression methods	10
	Rational for Compression systems	13
	Effect of Compression Strategies on Speech intelligibility	15
	Effect of Compression Strategies on Acoustic cues of Speech identification	19
	Consonant vowel ratio ; and Speech perception	21
	Technique For CVR Enhancement	24
3.	Method	30-40
	Experiment I	32
	Experiment II	34

4.	Results	41-60
	Experiment I	42
	Experiment II	44
5.	Discussion	61-65
	Speech Recognition Scores	61
	Consonant Vowel Ratio	62
	Envelope Difference Index	64
6.	Summary and Conclusion	66-69
7.	References	70-81

List of tables

SL No.	Title	Pg, No.
1.	Consonant, vowels and combination of CV stimuli used in the study	32
2.	Mean and Standard Deviation of speech recognition scores across conditions	41
3.	Mean Consonant vowel ratio and standard deviation for peak clipping	44
4.	Mean and Standard Deviation of CVR values for the processed and unprocessed stimuli for peak clipping	45
5.	Mean Consonant vowel ratio and standard deviation for Compression limiting.	47
6.	Shows the mean CVR values and Standard Deviation for unprocessed and processed stimuli for CL	48
7.	Mean Consonant vowel ratio and standard deviation for WDRC	50
8.	Mean CVR values and Standard Deviation for unprocessed and processed stimuli for WDRC	51
9.	Mean CVR values across strategies for stimuli with vowel /a/ at 65 dBSPL	52
10.	Mean CVR values across strategies for stimuli with vowel /a/ at 80 dBSPL	53
11.	Mean CVR values across strategies for stimuli with vowel /*i*/ at 65 dBSPL	54

12. Mean CVR values across strategies for stimuli with vowel *lil* at 80 dBSPL 55

13. Mean CVR values across strategies for stimuli with vowel /u/ at 65 dBSPL 55

14. Mean CVR values across strategies for stimuli with vowel /*u/* at 80 dBSPL 56

15. Mean Correlation index values and standard deviation for the stimuli with Vowel environment /a/, /i/ */u/* at input 65 and 80 dBSPL 58

List of Graphs

SI	*No.* *Title*	*Page No.*
1.	Mean Percentage Speech recognition scores across strategies	42
2.	Error graph showing 95% confidence interval for Consonant vowel ratio at 65 dBSPL for input and output processed stimuli across strategies.	57
3.	Error graph showing 95% confidence interval for Consonant vowel ratio at 80 dBSPL for input and output processed stimuli across strategies	58
4.	Average Correlation index across strategies for stimuli with vowel environment /a/, /i/, */u/* at 65 dBSPL	59
5.	Average Correlation index across strategies for stimuli with vowel environment /a/, *IV, lul* at 80 dBSPL	59

List of figures

SI No.	*Title*	*Page No.*
1.	Audiogram showing Speech spectrum	31
2.	Block diagram of the experimental setup for CVR calculation	36
3.	Block diagram of the experimental setup for EDI calculation	38
4.	Flow chart of the algorithm used to calculate EDI	39

INTRODUCTION

Sensorineural hearing loss is often associated with loudness recruitment, an abnormally rapid growth of loudness level with increasing level (Fowler, 1936; Hood, 1972; Moore, 2004). Recruitment is thought to be atleast partly related to reduced compressive nonlinearity on the basilar membrane, produced by loss of outer hair cell function (Moore, 1998; Ruggero & Ritch, 1991). The effect of recruitment is represented on the audiogram by the reduced range between hearing thresholds and uncomfortable loudness levels. In some patients with large losses, and thus small dynamic range, even the dynamics of speech signal itself causes problem, amplifying the weak parts of the speech to audible level causes the strong parts to be uncomfortably loud.

Thus individual with reduced dynamic range face difficulty in listening to discussions involving multiple speakers, each talking at different level. Listening under different acoustical conditions requires such clients to frequently readjust the volume control on their hearing aids.

Individuals with severe hearing loss are characterized by suprathreshold processing deficits primarily by dramatically reduced frequency selectivity (Faulkner, Rosen and Moore 1990) and in some circumstances by reduced temporal discrimination (Lamore, Verwiej and Brocaar 1990, Nelson and Freyman 1987, Tyler, Summerfield, Wood and Fernandes, 1982). Speech signal being a modulated spectrum, both aspects, namely, spectral information and temporal information are relevant. The severely impaired listeners rely to a greater extent on temporal information such as variation in speech amplitude (Rosen et al 1990) for perception.

Since the audiological profile differs for different degrees of hearing loss, the choice of amplification also varies. Severely hearing impaired individuals require different amplification characteristics than listeners with better hearing sensitivity. The types of amplification strategies can be broadly divided into Linear and Compression.

With linear hearing aids, the same amount of gain is applied to incoming sounds of a given frequency regardless of the level of sound entering the hearing aid, up to the maximum output level of the hearing aid. The effect of linear amplification is that at a set volume, weak sounds may have insufficient gain to be audible to the listener but intense sounds may be uncomfortably loud. One approach to the problem is to reduce the dynamic range of speech signal so as to match the dynamic range of the impaired ear. In order to do so, it is important to take into account the temporal structure of speech. Compression strategies have been used for this purpose from many years.

Different types of compression strategies are available including Automatic gain control (AGC), Wide Dynamic Range Compression (WDRC) and Compression Limiting (CL). Most hearing aid wearers are now fitted with multichannel wide dynamic range compression which gives more gain for weak sounds than for intense sounds. WDRC compresses most of the speech spectrum into the residual range, giving increased audibility and comfort and making loudness perception more similar to normal (Villchur, 1973).

There has been various studies reported in literature that compared WDRC with Linear amplification and found greatest benefits for WDRC for low level speech in quiet, conversational level speech in quiet (Souza 2002) and some studies have even shown small benefits for speech in background noises (Moore, Peters and Stone 1999). However, nearly all these data were based on listeners with mild to moderate hearing loss.

It has long been accepted that listeners with a severe loss require different linear amplification characteristics than listeners with a mild to moderate loss (Byrne, 1978; Byrne et al., 1990; Schwartz et al., 1988; Van Tasell, 1993). Because of their broader auditory filters (Faulkner et al., 1990), listeners with a severe-to profound loss may not be able to take full advantage of spectral information (eg, Erber, 1972) and must rely to a greater extent on temporal cues, which are altered by WDRC amplification (Lamore et al., 1990; Moore 1996; Van Tasell et al., 1987). For WDRC amplification, recall that one effect is alteration of the natural time-intensity variations of the speech signal. For listeners with a mild-to-moderate loss who presumably depend to a greater extent on spectral cues, these changes in time-intensity variations do not significantly offset the benefits of improved speech audibility (Souza and Turner, 1996, 1998 and 1999).

Souza P. E., and Jenstad L. M, (2005) attempted to compare speech recognition scores across different amplification strategies for listeners with severe hearing loss and found that the benefits of fast acting WDRC relative to more linear amplification may be reduced in listeners with severe loss.

In contrast, Moore and Marriage (2005) studied the effect of three amplification strategies on speech perception by children with severe and profound hearing loss and found that speech scores on close set testing for the profound group showed significant benefit for WDRC over the other two algorithms. There was a contrast in result probably because the latter study was done on children with congenital hearing loss, were the dynamic range is not reduced as in adults with sensorineural hearing loss.

Review of hearing aids for hearing impairment has shown that signal processing techniques that take the acoustic- phonetic structure of speech into account promise to be more effective in improving intelligibility than non phonetically - based methods of signal processing, provided the relevant speech features are extracted reliably. A form of signal processing which is phonetically based and which holds some promise for improving intelligibility is that of adjusting the ratio of consonant intensity to vowel intensity (C-V ratio). So the consonant vowel ratio appears promising as a good measure for selection of suitable strategy for an individual. Thus the amplification strategy that improves the consonant vowel ratio by enhancing the consonant amplitude might provide better speech perception.

Acoustic analysis of single channel syllabic compression and linear amplification has revealed that compression may result in changes in the intensity relationships between parts of the speech signal (Hickson & Byrne, 1995). It is expected that increase in the CVR could be expected to improve consonant perception for people with hearing impairment. Even research in linguistics with normal hearing subjects reveal that CVR itself is an important cue for perception of certain sounds.

Thus calculating the Consonant vowel ratio of the speech signal after signal processing through a hearing aid might help in predicting the performance through that hearing aid. Hickson & Thyer (1998) reported that it is possible to predict speech perception performance with compression by examining the acoustic characteristics of the processes speech signal.

Need for the study

The aim of the present study was to investigate acoustic changes to the speech singal (in terms of Consonant Vowel Ratio and Envelope Difference Index) that occuarred with different amplification strategies and to examine the relationship between such changes and speech perception in individuals with severe sensorineural impairment.

Objectives

> To study the effects of different amplification strategies on speech recognition scores of severely Hearing Impaired listeners,

> To objectively measure the acoustic effects of different amplification strategies on amplified speech (by calculating the consonant vowel ratio and Envelope difference index) and,

> To evaluate the relation between acoustic changes and speech recognition.

Hypothesis:

A) Hypothesis 1: There is no significant difference between the Speech Recognition scores across strategies (Peak Clipping, Compression Limiting and Wide Dynamic Range Compression) at

1. 65dBSPL
2. 80dBSPL

B) Hypothesis II: There is no significant difference in the Speech Recognition scores for the stimuli across Levels within strategies (Peak Clipping, Compression Limiting and Wide Dynamic Range Compression).

C) Hypothesis III: There is no significant difference in the Consonant Vowel Ratio values for the stimuli across strategies (Peak Clipping, Compression Limiting and Wide Dynamic Range Compression) at

1. 65dBSPL
2. 80dBSPL

REVIEW OF LITERATURE

Hearing loss involves a multifaceted loss of hearing ability. There are some problems which are common in individuals with sensorineural hearing loss. Some sounds are inaudible. Other sounds can be detected because part of their spectra is audible, but may not be correctly identified because other parts of their spectra (typically the high frequency parts) remain inaudible. The range of levels between the weakest sound that can be heard and the most intense sound that can be tolerated may be less. In addition, sensorineural impairment diminishes the ability of a person to detect and analyze energy at one frequency in the presence of energy at other frequencies; similarly, a hearing-impaired person has decreased ability to hear a signal that rapidly follows, or is rapidly followed by a different signal. This decreased frequency resolution and temporal resolution makes it more likely that noise will mask speech than would be the case for a normal-hearing person. All the above mentioned problems affect or hamper communication. Hearing aids are used to reduce or overcome these problems. Over the years there has been advancement in hearing aid technology to provide comfort and better perception to the clients. The main aim is to improve speech perception without causing discomfort.

The Speech Signal: A speech signal can be described in physical terms as a modulated spectrum. Both aspects i.e. spectral information and temporal information are relevant. The two aspects are not equally important for all parts of speech (Verschuure et al. 1993). Vowels, semi-vowels, and nasals require good spectral resolution (separate detection of Fl and F2) while there is little information in the (almost absent) modulations. Fricatives and plosives, on the other hand, are strongly modulated signals differing mainly in time structure (e.g. the gap before the plosive); only crude spectral analysis is required. The average spectrum of speech can be described roughly as having a peak around 400 Hz and falling off above 500 Hz at a rate of about 10 dB/octave.

The modulation can also be represented by a spectrum, the amplitude-modulation spectrum (Plomp, 1983). The relevant frequencies of this spectrum range roughly from 0.1 to 40 Hz. The frequency of maximum modulation is around 3 Hz and the maximum amount of modulation is found in the frequency band around 1 kHz. For the high frequency band (4 kHz), the maximum shifts somewhat toward a higher modulation frequency (5 Hz).

Amplification Strategies

The amplification strategies are broadly divided into Linear and Compression types.

Linear Amplification Strategy:

In Hearing aids linear amplification strategy provide a constant gain is applied to all input levels until the hearing aids saturation limit is reached. Since daily speech include such a wide range of intensity levels, from low intensity consonants / f / to high intensity vowels / i / and whispered speech to shouting speech, the benefit of linear amplification gets restricted when the amplification needed to make low intensity sounds audible amplifies high intensity sounds to the point of discomfort. In other words linear amplification strategies have a limited capacity to maximize audibility across a range of input intensities.

Compression Amplification Strategy:

To solve the problem of reduced dynamic range most of the hearing aids now incorporate some form of compression amplification strategy in which gain is automatically adjusted based on the intensity of the input signal. The higher the input intensity more the gain is reduced. It is expected that individual using compression hearing aids perform better than those using peak clipping aids in listening condition that include wide range of speech levels (Benson, Clark & Johnson, 1992; Moore et al, 1992, Souza and Turner, 1999). However the benefit of compression amplification strategy is yet to be established well.

Characteristics of Compression Hearing Aids: The basic characteristics of a compressor are

i) Dynamic compression characteristics

Attack and Release time: Attack time is defined as the time taken for the output to stabilize to within 2 dB (IEC 60118-2) or 3 dB (ANSI S3.22) of its final level after the input of the hearing aid increases from 55 to 80dBSPL (IEC 60118-2) or 55 to 90dBSPL (ANSI S3.22). The release time is the time taken for the compressor to react to a decrease in input level. Although attack and release time could be made to have extremely short values, the consequences are most undesirable. If the release time is too short, the gain will vary during each voice pitch periods, so the compressor will distort the waveform. If the attack time is made extremely short, and the release time is long then distortion is minimal. The attack and release times have a major effect on how compressors affect the levels of the different syllables of speech.

Several hearing aids currently available have an adaptive release time. The release time is short (e.g. 20ms) for brief intense sound, but becomes longer (e.g. 1 sec) as the duration of the intense sound increases. When an adaptive release time is combined with a short attack time, a brief intense sound will cause the gain to rapidly decrease and then rapidly increase when the intense sound ceases.

ii) Static Compression Characteristics

Compression threshold: Compression threshold is defined as the point at which the output deviates by 2dB from the output that would have occurred had linear amplification continued to higher input levels.

Compression ratio: Compression ratio is defined as the change in the input level needed to produce a ldB change in output level. The compression ratio of a linear amplifier is 1:1. Compression ratio greater than about 8:1 would be considered as Compression limiting. Compression ratio less than 1:1 corresponds to dynamic range expanders rather than Compressors.

The range of inputs over which compression occurs is called the compression range.

Different approaches to compression methods: There are two different approaches to compression methods. They are-

1. Output control
2. Input control

/. *Output control*

a) Peak clipping

b) Compression limiting

a) *Peak Clipping:* Amplifiers cannot produce signals larger in voltage than some specified maximum. If the biggest signal in the amplifier (usually the output signal) is near this maximum, and either the input signal level or the gain of the amplifier is increased, then the amplifier will clip the peaks of the signal. All amplifiers create large amounts of distortion if the signal is sufficiently peak clipped. When a complex signal is peak clipped, the distortion products occur at frequencies that are harmonics of all the frequencies in the input signal and at frequencies that are combination of all the harmonics.

b) *Compression limiting:* Compression limiting systems are characterized by a short attack time, high compression threshold and high compression ratio, typically greater than 5. The amplification is linear for most input levels and an average speech signal would trigger the compression circuitry. As a means of limiting hearing aid output it is generally considered to be superior to peak clipping, which is an 'instantaneous' limiter that generates significant harmonic and intermodulation distortion (Boothroyd et al. 1988; Braida et al. 1979; Dreschler, 1988 b; Hawkins Noidoo, 1993; Preves, 1991; Walker & Dillon, 1982).

2) Input control:

The fast acting syllabic compression or wide dynamic range compression have short attack and release limits, low compression activating threshold and low compression ratio (LS), Release times ranges from 50 to 150 msec. This type of compression is attractive from the theoretical perspective in that, in principle, weak speech sounds can be amplified well above the threshold of hearing while relatively intense speech sounds are not amplified above the listener's loudness discomfort level.

AVC or longer term compression is characterized by the long time constants used, with release time being greater than 150 msec. Compression threshold is generally low and there is a high compression ratio (greater than 5). Theoretically this type of amplification should allow good speech intelligibility and quality for a large range of input levels, and can improve intelligibility and quality of amplified speech for hearing impaired listeners with small dynamic range.

Multi-channel compression

In Multi-channel compression the incoming signal is split into different frequency band and each band of signal passes through a different amplification channel and each channel contains its own compressor. In a single channel compression hearing aid, when the compressor turns the gain down, signal components at all frequencies are decreased in level. It might not be appropriate to have signal component at frequency being activated just because there is strong signal or a limited dynamic range of hearing, at another frequency. Multi-channel compression avoids this problem.

When the degree of compression is greater in the high frequency than low frequency channel, there will be greater emphasis at low input levels than at high input levels. This characteristic has been labeled as Treble Increase at Low Level (TILL). When the degree of compression is greater in the low frequency channel than in the high frequency channels, there will be less high frequency emphasis at low input level than at high input levels. This characteristic has been labeled as a Bass increase at Low Level (BILL).

Rationales for Compression System

The general aim of the compressor is to provide higher gain for the soft sounds than is for the loud 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.

1. Discomfort and distortion avoidance (CL)
2. Loudness normalization
3. Noise reduction
4. Reduction of signal dynamic range

a) Discomfort and distortion avoidance (CL):

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 settings to amplify weak input signals the compressor will prevent discomfort from occurring without distortion, if a high level wanted or unwanted signal occurs.

b) Loudness normalization

Because of the effects of the recruitment, the equal loudness contours of the person with a high frequency sensorineural hearing loss show the greatest deviation from the 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. Use of WDRC has been proposed as a means to compensate for abnormal loudness growth, and several fitting procedures have been developed in accordance with this philosophy (e.g., Allen et al., 1990; Cox, 1995 and 1999; Kiessling et al., 1997; Kiessling et al., 1996; Ricketts, 1996). The intent of these procedures is to set compression parameters such that a listener wearing a WDRC aid will perceive changes in loudness in the same way as a normal-hearing listener (Kuk, 2000). Recent data confirms that WDRC amplification can normalize loudness growth better than linear amplification (Fortune, 1999; Jenstad et.al., 2000). Byrne (1996) argues against strict loudness normalization, pointing out that normal-hearing subject can easily adjust to situational variations in loudness. Byrne (1996) also notes that hearing-impaired listeners might do better with compression parameters that explicitly do not normalize loudness growth, such as equalizing loudness across frequency (Byrne et al., 2001).

c) Noise reduction

The noise reduction rationale aims to identify frequency component 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 (SNR) will vary with frequency. A second assumption is that noise is one frequency region will mask useful signals in the other frequencies regions. The rationale of reducing masking by attenuating the frequency regions with the poorest SNR is supported by the data of Rankovic, Fryman & Zurek (1992).

d) 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 narrow range, compression can be used to translate a wide range of levels at the hearing aid input to a range of levels at the aid output. This reduces the necessity for the aid wearer to vary the volume control.

The major advantage expected for a WDRC is that the user has less need to vary the volume control. The major disadvantage expected in the increased gains for low-level inputs make feedback oscillation 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.

The Effect of Compression Strategies on Speech Intelligibility

A major problem limiting the efficacy of acoustic amplification systems for sensorineural hearing impairments is that the dynamic range of hearing is reduced significantly. Since the vast majority of people, who are candidates for acoustic amplification, have a sensorineural hearing loss, this problem is one of considerable importance. One approach to the problem is to reduce the dynamic range of the speech signal so as to match the dynamic range of the impaired ear. In order to do so, it is important to take into account the temporal structure of speech. Temporal fluctuations in speech level can be subdivided into two broad categories - slow variations in level (on the order of seconds) associated with changes in overall voice level, and relatively rapid changes in level (on the order of tens of milliseconds) associated with the differences in level among the various sounds of speech (e.g. vowel versus consonants).

Substantial changes in a speech signal can occur as the result of signal processing by hearing aids. Different signal processing strategies results in acoustic modification of the speech signal in both spectral and temporal aspects. These changes might affect the speech perception of the individuals with hearing impairments. There have been various studies reported in literature that shows the relationship of different strategies and speech recognition performance.

Peak clipping was the commonly used strategy in hearing aids in the 1990 in US to limit maximum output. Several earlier studies reported that peak clipping actually increased the intelligibility of speech when the speech was presented in either a background of noise (Licklider & Pollack, 1948, Miller & Mitchell, 1947) or in quiet (Martin, 1950; Pollack, 1952).

But studies by Pollack & Pickett (1958) reported a slight decrease in word recognition performance as the amount of peak clipping increased. Similar result was found lately by Crain & Tasell (1994), the speech recognition thresholds increased both for normal and hearing impaired subjects with increasing level of peak clipping, with significant threshold shift occurring for clipping levels greater the 18 to 24 dB.

With the advancement of technology there has been a proliferation of advanced signal processing circuitry for hearing aids. Various forms of non linear processing came in to use. The commonly used signal processing strategies are Compression Limiting (CL) and Wide Dynamic Range Compression (WDRC).

Extensive research has compared linear and WDRC circuitry. Some of these studies have shown benefits of compression (e.g. Benson, Clark & Johnson, 1992; Moore et al, 1992, Souza and Turner, 1999). Others have reported no difference (Plomp, 1994., Crain and Yund, 1995).

Jenstad & Shantz (1999) found that the WDRC aid resulted in high and uniform speech recognition scores across the five spectra. In contrast, the linear gain aid resulted in a lower recognition scores for softer speech and shouted speech relative to that obtained with average speech level.

There are reports in literature that show that different degrees of hearing loss gets advantage with different amplification strategies, Shanks & Williams (2002) reported that significant differences favored the peak clipping and compression limiting circuits over the WDRC in mild hearing loss groups and favoured the WDRC over the peak clipping in the more severe slopping hearing loss groups.

Compressing the speech signal into a very small dynamic range using a WDRC for a severely hearing impaired individual might have detrimental effects on speech intelligibility by reducing the depth of amplitude modulation in speech by introducing distortion in temporal envelopes, and reducing spectral contrast (Plomp 1988, Stone and Moore 2002, 2004).

Marriage & Moore (2003) reported that WDRC can give significant improvement in consonant discrimination for children with moderate and severe-profound hearing loss.

Souza P.E., & Jenstad L.M. (2005) attempted to compare speech recognition scores across different amplification strategies for listeners with severe hearing loss and found that the benefits of fast acting WDRC relative to more linear amplification may be reduced in listeners with severe loss.

Moore & Marriage (2005) studied the effect of three amplification strategies on speech perception by children with severe and profound hearing loss and found that speech score on close set testing for the profound group showed significant benefit for WDRC over the other two algorithms.

Experimental studies on speech perception with multi channel compression have been characterized by their variability. Some studies have shown positive benefits of compression compared to conventional linear amplification (Benson et al. 1992; Moore et al. 1992, Souza & Turner, 1999) and others have reported no difference (Plomp, 1994, Crain & Yund, 1995). There has been a report of detrimental effects of compression on speech perception for some individuals (Crain & Yund, 1995; Hickson et al. 1995; Plomp, 1994). It has been suggested that closer examination of the acoustic properties of the compressed speech signal may help to explain some of the variation in findings (Hickson, 1994; Plomp, 1994; Hickson et al. 1999).

Effects of Compression on Acoustic Cues for Speech Identification

Speech intelligibility is determined by the listener's ability to identify acoustic cues essential to each sound. Implicit in this process is accurate transmission of these cues by the hearing aid. Certainly audibility of specific speech cues is a major factor in speech intelligibility. However, it is also important to consider whether acoustic cues are distorted or enhanced by compression amplification. The work of DeGennaro et al. (1986) provides a convincing demonstration that more than simple audibility changes are involved. These investigators began by measuring the distribution of short-term RMS levels at each frequency. They then processed speech with compression systems that placed progressively greater amounts of the range of amplitude distributions above the subject's hearing threshold. Interestingly, no subject showed a consistent improvement with compression, although from an audibility perspective some improvement would be expected as greater amounts of auditory information exceeded detection thresholds and thus became audible. It is possible that compression distorts some speech cues, offsetting the benefits of improved audibility, at least for some compression systems and for some listeners.

Recently, interest has been renewed in the importance of temporal cues for speech intelligibility (e.g. Shannon et al., 1995; Turner et al.., 1995; Van Tasell et al.., 1987 and 1992) and speculation that these cues are disrupted by fast-acting WDRC (e.g., Boothroyd et al., 1988; Dreschler, 1989; Plomp, 1988; Verschuure et al., 1996). Temporal cues include the variations in speech amplitude over time and range from the very slow variations of the amplitude envelope to the rapid "fine-structure" fluctuations in formant patterns or voicing pulses (Rosen, 1992). With regard to compression, most attention has focused on fluctuations in the amplitude envelope, in part because alteration of the amplitude envelope is the most prominent temporal effect of fast-acting WDRC. The amplitude envelope contains information about manner and voicing (Rosen, 1992; Van Tasell et al., 1992) and some cues to prosody and also the suprasegmentals of speech (Rosen, 1992). Compression alters the variations in the amplitude envelope and reduces the contrast between high-intensity and low-intensity speech sounds. Of course, the reduced intensity variation is a desirable effect of compression. However, because both normal-hearing and hearing-impaired listeners can extract identification information from amplitude envelope variations (Turner et al., 1995), it is possible that alterations of these cues could affect speech intelligibility.

The listeners who have normal or near-normal spectral discrimination ability (Moore, 1996; Van Tasell, 1993) should be able to extract sufficient spectral and contextual information to compensate for altered temporal cues. The clinical impact may be greater for listeners who depend to a greater extent on temporal cues-most obviously, listeners with a severe-to-profound loss (Lamore et al., 1990; Moore, 1996).

All sounds are not equally susceptible to distortions of temporal cues. The greatest effect is on sounds where critical information is carried by variations in sound amplitude over time. For example, important features of the stop consonants (/p/, /t/, k/, /b/, /d/, /g/) include a stop gap (usually 50 to 100 milliseconds in duration) followed by a noise burst (5 to 40 milliseconds in duration). Voiced stops (/b/, */d/,* /g/) are distinguished from voiceless stops (/p/, /t/ */k/)* by the onset of voicing relative to the start of the burst. For syllable-initial stops, voice onset time (VOT) ranges from close to 0 milliseconds for voiced stops to 25 milliseconds or more for voiceless stops (Kent and Read, 1992). Perception of stop consonants can therefore be modeled as a series of temporal cues (i.e., a falling or rising burst spectrum followed by a late or early onset of voicing).

Single-channel, fast-acting compression applied to synthetic speech increases the amplitude of the consonant burst, resulting in erroneous perception (/t/ for /p/) (Hedrick and Rice, 2000). Similarly, Sreenivas et al., (1997) noted that a two-channel syllabic compressor increases the amplitude of the consonant burst, particularly in the mid-frequency region, resulting in more errors of /g/ for /d/ (for unprocessed speech, the

peaks of /g/ are more prominent in the 1-2 kHz range, with the spectral peaks for /d/ mainly in the region of 4-5 kHz. Affricate perception is impaired in multichannel WDRC systems, and that the most common error is a stop consonant (Jenstad and Souza, 2002).

Consonant vowel ratio and *speech perception*

The Consonant vowel ratio is the measure of the difference in intensity between the consonant and vowel. Several studies have demonstrated the relationship between CVR and speech intelligibility. House et al (1965) found differences in test scores of about 3 dB shift in speech to noise ratio for the two different talkers stimuli, and analysis of the recorded stimuli showed that, although the vowel for both speakers were approximately equal in level ,consonant levels for the less intelligible speaker were 2-4 dB lower than the more intelligible speaker.

The strategies use to increase the magnitude of consonants relative to the vowels (e.g., reduce the vowel-consonant ratio), are reported to have the most favorable improvements in consonant recognition by hearing-impaired subjects.

With single channel compression, significant acoustic changes are introduced into the speech signals that are different from those seen in linear amplification system. The most commonly reported change is an increase in the consonant vowel ratio (CVR) where the intensity of vowels decreased (Hickson & Byrne, 1995). CVR changes may also occur with linear amplification that has a high frequency emphasis, and the extent of any CVR change depends on a combination of the compression characteristics and the frequency gain response.

There is evidence that CVR may be a cue for the perception of some consonant sounds for people with hearing impairment, and therefore increasing CVR is not always beneficial (Hickson et al. 1995; Hickson & Byrne, 1997; Hedrick & Rice, 2000) reported that a single channel fast acting WDRC restricted the range of differences in amplitude between consonant and vowel in discrete frequency region. The restriction of relative amplitude between consonant and vowel significantly affected labeling of phase of articulation for a continuum of synthetic voiceless stop consonants by listeners with sensorineural hearing loss.

Hickson & Thyer (1998) reported that it is possible to predict speech perception performance with compression by examining the acoustic characteristics of the processed speech signal. Hickson & Thyer(2003) r eported that compression amplification, that best preserves the acoustic properties of speech relative to linear amplification will yield the best speech perception results.

Hearing Impaired Listeners confuse unvoiced consonants more than any other type of speech phoneme, especially in background noise. Speech-enhancing hearing aids are designed to improve speech intelligibility of these difficult to perceive sounds by reducing environmental noise or by enhancing the speech signal itself. Improving consonant to vowel intensity ratio (CVR) has been explored as one technique for speech enhancement. As a result of increasing CVR, consonants may become audible, which is probably more important for good speech perception than the improvement in CVR (Freyman & Nerbonne, 1989). Consequently, it is assumed that more consonant emphasis relative to vowel amplification is desirable in a hearing aid fitting, but there have been relatively few investigations to substantiate this.

In the past, most published efforts to increase CVR have been implemented with large digital computers for research purposes rather than with technology suitable for packaging in head worn hearing aids. However, it is also possible to develop consonant enhancement algorithms with analog circuitry that are suitable for head worn, even in the ear (ITE), hearing aids. Some of these approaches have been utilized for many years in head worn hearing aids. For example, Edgardh (1952) studied a single-channel syllabic compressor hearing aid with a low enough compression threshold and reported that it significantly improves the CVR.

Regardless of whether large digital or subminiature analog approaches are used, the goal is to accomplish this CVR increase without audible artifacts so that speech and background environmental sounds are not altered in unnatural or unpleasant ways. Indeed, excessive manipulation of the speech signal in inappropriate ways can actually lead to degradation in speech recognition (e.g. Bunnell, 1990; Bustamante & Braida, 1987; Moore, 1990; Plomp, 1988).

Techniques for CVR Enhancement

Many of the first ideas for speech enhancement resulted directly from investigations conducted to determine what speech cues are important for perception. For example, because of the findings of Picheny, Durlach & Braida (1985), in which intelligibility of clearly spoken speech was superior to that for normally spoken speech in continuous discourse, attempts were made to alter some of the acoustic features of clearly spoken speech (Picheny, Durlach & Braida, 1986). This was typically accomplished by manipulating the speech waveform to alter such features as vowel duration and amplitude, consonant duration and amplitude, extent of formant transitions, voice onset time and burst amplitude for stop consonants, and amount of frication noise.

Increasing the acoustic energy of consonants relative to vowels via consonant amplification and expansion shows promise for improved consonant recognition for hearing-impaired persons. This type of processing may be useful in improving SNR compared to linear processing for low-level signals (Dillon, 1989; Villchur, 1973). Compression has been studied for use in hearing aids for about 40 years (e.g. Walker & Dillon, 1982). Niederjohn & Grotelueschen (1976) studied the effects on speech intelligibility in high levels of competing noise of high-pass filtering followed by automatic amplitude normalization (syllabic compression). They used a compressor with rapid attack and release times (8 msec) so as not to interfere with the transient characteristics of speech. They pointed out that because of its quick response time, the compressor would produce an attenuation of high-level vowel energy, which would tend to increase the CVR. They found significant intelligibility advantages for high-pass filtering followed by amplitude normalization over unprocessed speech.

Dreschler (1988) indicated that syllabic compression equalizes levels between successive sounds, thus bringing up consonant levels. Although syllable compression has considerable intuitive face validity for compensating the reduced dynamic range of hearing-impaired persons, it is not routinely used in a high percentage of hearing aid fittings.

Several researchers have shown little, if any, benefit comparing syllabic compression to linear amplification. Walker & Dillon (1982) and Dreschler (1988) speculated that the benefits of recruitment compensation may be nullified in effect by temporal distortions from the compressor attack and recovery times and their alterations of the normal intensity cues of speech. A compression limiting is only active at high signal levels; it may provide some CVR enhancement without significantly altering the dynamics of conversational speech signals compared to the effect of a syllabic compressor (Walker & Dillon, 1982).

To eliminate the problem of single-channel compressors in which gain across the entire frequency range is reduced by a low-frequency noise, a multiband compressor having independent Automatic Gain Control (AGC) circuits for each frequency band has been utilized. With multi-channel compression, low-frequency noise would theoretically cause gain reduction only in the low frequency band(s), and the weaker high frequency components of speech, critical for good speech intelligibility, would continue to.be maximally amplified (Kates, 1986). High compression ratios with multichannel AGC may degrade the relative intensity cues required to identify stops or fricatives (DeGennaro, Braida & Durlach, 1986; Plomp, 1988).

The problems of interference with the high-frequency components of the speech signal by syllabic compressors may apply even more strongly to fast-acting multichannel compressors with many independent compression bands reduce the natural amplitude contrasts in the speech signal (Plomp, 1988). Therefore, Plomp contends that longer compressor time constants should be used with multichannel AGC. However, Villchur (1989) reminds us that Licklider and Pollack (1948) showed that speech was perfectly intelligible after infinite amplitude clipping resulting in a signal with no amplitude contrasts. Villchur states that although multichannel AGC decreases the peak to valley level differences within speech, the audibility of weaker components of speech, such as consonants, may be preserved after compression. He concluded that only field experience with two-channel compression will prove the viability of multichannel AGC.

Using modified dbX compressors in two channels, Yanick & Drucker (1976) excluded that a combination of expansion and compression was superior to both compression alone and to linear amplification. They reported a 10% improvement in recognition scores with compression and expansion combined over compression alone for six hearing-impaired listeners in a signal to noise ratio (SNR) of+6 dB. Walker, Byrne & Dillon (1984) evaluated speech intelligibility for a six-channel expander/compressor for a small group of subjects. The expander operated mainly on low-level energy in the high-frequency channels. They concluded that expansion degraded low-level speech intelligibility or, at best, did not change it as compared to linear six-channel frequency shaping. In their study, expansion failed to improve intelligibility of very low-level final consonants. However, the outcome of this study might have been influenced by the degradation of the speech signal resulting from automatically manipulating the gain simultaneously in six bands.

Use of expansion for increasing perception of low-level consonants has been suggested by Kates (1984). In his implementation, the level of high-frequency energy was sensed in a number of band-pass channels. After determining the presence of a consonant from the short-term spectral shape, if the speech level in a high-frequency band exceeded a preselected threshold, a 3:1 dynamic range expansion was applied in that band. Linear amplification was performed at frequencies below 500 Hz. Gordon-Salant (1987) found that a CVR increase of 10 dB resulted in an overall 14% increase in consonant recognition percent correct for elderly persons with gradually sloping and sharply sloping, mild to moderate high-frequency hearing loss. For these tests, 19 consonants, paired with three vowels, were presented at 75 and 90 dB SPL with competing 12-talker babble at a +6 dB SNR. In the same study, increasing consonant duration by 100% and combining the amplitude and duration enhancements produced no improvement and even a decrement in consonant recognition scores for some subjects.

Using much the same protocol for consonants from the California Consonant Test amplified by 10 to 21 dB at 65 and 95 dB SPL presentation levels, an improvement in intelligibility of 10% was reported at the lower level by Montgomery and Edge (1988). In that study, this amount of consonant enhancement resulted in the consonants and vowels having equal amplitude (CVR = 0 dB), which may have been too large an increase for some of the mild to moderately hearing-impaired subjects. In searching for the optimal CVR for each subject with 3, 6, 9 and 12 dB increases in consonant level and the CVR, Kennedy & Levitt (1990) obtained a 15% improvement in NST scores for nine moderate to moderately severe hearing-impaired listeners, averaging across all subjects and conditions. Too much consonant amplification for certain consonants for some subjects resulted in an intelligibility decrement from the "ideal" amount of enhancement. The implication is that the optimal CVR varies with the particular consonants and vowels combined and the characteristics of the auditory system of the listener.

The review of literature has made it clear that there are ample studies on the amplification choices for individual with mild to moderate hearing loss but very few studies highlighting the amplification choices for severely hearing impaired individuals.

There is dearth of studies in Indian literature highlighting the amplification needs and suitable choices available for this group of population. Hence, the particular study was planned. There is a need to support the perceptual findings with some objective data to strengthen the study results. Literature available has shown that speech perception can be predicted through acoustic analysis of the processed stimuli (Hickson and Thyer, 1998). So with this objective, consonant vowel ratio and Envelope Difference

Index was calculated for the speech stimuli after processing through different amplification conditions and there effect on speech recognition scores was studied.

METHOD

The present study attempted to measure the acoustic changes in speech processed through three different amplification strategies and to study their effect on speech recognition scores of individual with severe hearing impairment.

Subjects

> 10 subjects (5 males and 5 females) between 20- 55 years of age participated in the study.

> All subjects had bilateral moderately severe to severe sensorineural hearing loss (65-90 dBHL).

> All subjects were naive hearing aid users.

> They all had normal middle ear functioning.

> They all were native Kannada speakers.

Instruments and Software used:

> MADSEN OB922- Dual channel diagnostic audiometer attached with TDH 39 Headphone, and 2 Martin Audio loudspeakers.

> GSI-Tympstar

> Phonak Supero Digital BTE

> NOAH Link Compass Version 4 programming software

> MATLAB, Wave Surfer, Wave pad

> Larsen and Davis 824 Sound level Meter

Stimuli:

- CV items word list containing nonsense monosyllabic words were recorded with a unidirectional microphone fixed at a distance of 6 inches from the speaker. The recording was done by a native Kannada speaker seated in a sound treated room.
- The speaker uttered the words thrice which were recorded through a PC sound card and stored onto the computer memory.
- The CV word list consisted of 16 consonants paired with three different vowels /a/, *IV* and *l\xl* such that most of the speech frequencies are covered. Attempt has been made to include most of the phonemes covering the full speech spectrum as shown in (Figl).

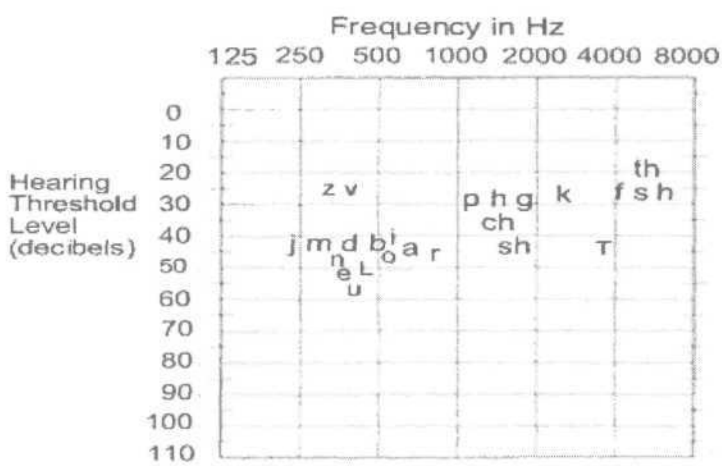


Fig 1. Audiogram showing Speech spectrum

- However the number of consonants and vowels were restricted because the study required calculation of Consonant Vowel Ratio for each stimulus which is a tedious and lengthy job. The word list consisted of 48 CV items shown in (Table 1).

> The stimulus was recorded in a sound proof room at the sampling rate of 44.1 kHz and 16 bit resolution and stored onto to the computer memory. An inter-stimulus interval of 3 sec was introduced between stimuli using Wave pad Software.

> The list was randomized and 6 different word lists were prepared. A calibration tone of 1 kHz was recorded at the same level as the speech stimuli.

Vowels	Stop						Nasal		Affricate		Fricative			Liquid		Glide
	p	b	T	d	k	g	m	n	tʃ	dz	s	ʃ	h	r	l	v
a	pa	ba	Ta	da	ka	ga	ma	na	tʃa	dza	sa	ʃa	ha	ra	la	va
i	pi	bi	Ti	di	ki	gi	mi	ni	tʃi	dzi	si	ʃi	hi	ri	li	vi
u	pu	bu	Tu	du	ku	gu	mu	nu	tʃu	dzu	su	ʃu	hu	ru	lu	vu

Table I- Consonant, vowels and combination of CV stimuli used in the study.

> All speech stimuli were played through a computer attached to an OB992 two channel diagnostic audiometer attached to two loudspeakers (Martin Audio). Speech and 1 kHz tone were generated from the same loudspeaker at a distance of around l meter from the clients head.

> Level in dBA was set using the calibration track on the computer output with the sound level meter (Larsen and Davis 824), placed in the position of the clients head without the client present.

Procedure

[A] Experiment-I: *To measure the effect of different amplification strategies on speech recognition scores.*

A routine audiological evaluation that includes pure tone audiometry, using Carhart - Jerger Modified Hughson - Westlake (1959) procedure using a calibrated (ISO-389, 1994) dual channel diagnostic audiometer (MADSEN OB922) with TDH 39 headphone was done. Speech recognition scores and Uncomfortable loudness level for speech was measured. Immitance measurements including Tympanogram and Acoustic Reflex Threshold were carried out using GSI-Tympstar to rule out any middle ear pathology. The tests were carried out in a acoustic treated room with noise level within the permissible limits (ANSI S3.1-1991 cited, Wilber 1994).

After audiological evaluation subjects were fitted with Phonak Supero 412 Digital BTE Hearing aid having the option of different signal processing strategies: Wide Dynamic Range Compression, Peak clipping and Compression limiting. The hearing aid was programmed for all the three signal processing conditions using NAL-NL1 (Dillon et al., 1998) prescriptive formula using the NOAH Link Compass Version 4 programming software.

Subjects were fitted with the programmed hearing aid coupled to a custom made ear mold and the speech recognition scores were calculated for all the different signal processing conditions. CV items were presented from the computer Sound card attached to the two channel diagnostic audiometer. The stimuli were presented via the loudspeaker at the distance of 1 m from the client. The stimuli were presented at the level of 65 and 80 dBSPL. The responses of the client were noted and scored. This procedure took around 2 hours because as the CV list containing 48 items had to be administered at two levels (65, 80 dBSPL) and at three different amplification conditions (Peak clipping, Compression limiting, Wide dynamic range compression). Hence the numbers of subjects were restricted to 10.

[B] Experiment *II: To measure the effect of different amplification strategies on speech acoustics.*

The acoustic measures used in the study were Consonant Vowel Ratio (CVR) and Envelope Difference Index (EDI) that quantifies the effect of amplification strategies on the temporal envelope of speech. The programmed hearing aid (for all the different signal processing conditions) as used in experiment I, was kept in an anechoic chamber and the following acoustic analysis was done:

1) CVR calculation: Fig shows the Block diagram for the Experimental set up for Consonant Vowel Ratio calculation. Steps involved in the calculation of CVR were as follows

> The CV items were presented at the level of 65 and 80dBSPL into an anechoic chamber through a PC soundcard. A microphone connected to the Sound level meter was placed in the anechoic chamber to record the input stimuli. The level in dBSPL was maintained by monitoring through Sound Level Meter. Levels were adjusted using preamplifier. The stimuli picked up by the microphone was routed through the SLM and stored on to the computer memory. Using the same procedure all the CV items were recorded at 65 and 80 dBSPL

> In the next step, the programmed hearing aid for each of the different conditions was kept in the anechoic chamber with the receiver output coupled to a 2cc coupler. The microphone of the SLM was coupled to the other end of the 2cc coupler. The stimuli presented in the anechoic chamber were picked up by the hearing aid microphone. The input level of the stimuli was fixed at 65 and 80 dBSPL that was monitored through the SLM (by manipulating the volume control of the preamplifier). The stimuli processed through the hearing aid were picked up by the microphone attached to SLM and stored onto the computer memory. Using the same procedure all the stimuli were processed through the hearing aid at 2 different input levels (65, 80 dBSPL) and 3 different amplification conditions.

> Recording of stimulus was a complex job, because for each stimulus the preamplifier had to be manipulated manually to keep the input level in SLM fixed at 65 and 80 dBSPL. Recording for one particular condition (288 stimuli) took around 3 hours.

> Consonant vowel ratio was calculated for the processed (output from hearing aid) and unprocessed (input to the hearing aid) stimuli using MATLAB software. Since the process is time consuming and complex we had to restrict the CVR calculation to 5 subjects.

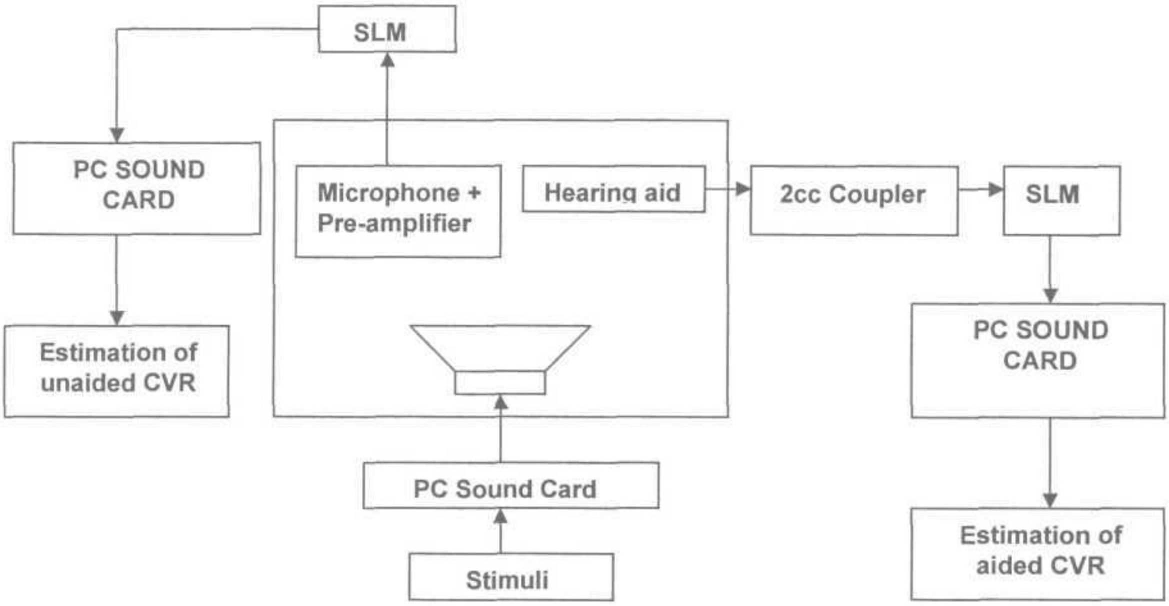


Fig. 2: Block diagram of the experimental setup for CVR calculation.

> *MATLAB* is a numerical computing environment and programming language. Created by *The MathWorks,* MATLAB allows easy matrix manipulation, plotting of functions and data, implementation of algorithms, creation of user interfaces, and interfacing with programs in other languages. Although it specializes in numerical computing, an optional toolbox interfaces with the Maple symbolic engine, allowing it to be part of a full computer algebra system. Toolboxes are comprehensive collection of MATLAB functions (M-files) that extend the MATLAB environment to solve the particular classes of problems.

> Using the MATLAB (M-files), the coding is done for analyzing the Consonant vowel ratio.

Algorithm: Matlab based algorithm for CVR calculation proceeds in the following steps:

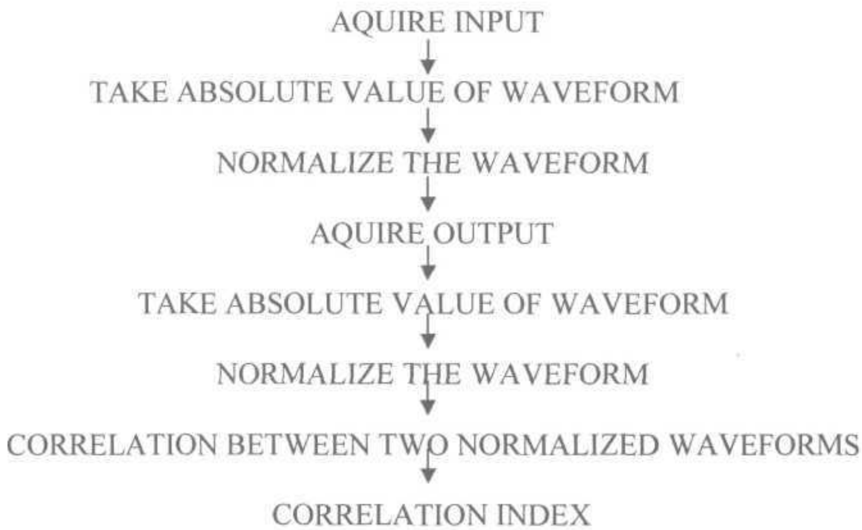
1) Acquire the speech signal (processed and unprocessed stimuli)
2) Plot the waveform
3) Apply low pass Butterworth Filter on the input signal to suppress high frequency content of the signal for e.g. for the vowel /a/ the cut off frequency was fixed at1000Hz.
4) Plot the waveform and obtain the maximum amplitude of vowel.
5) Apply high pass Butterworth Filter on the input signal to suppress low frequency content of the signal.
6) Plot the waveform and obtain the maximum amplitude of consonant.
7) CVR = maximum amplitude of consonant / maximum amplitude of vowel

2) Envelope Difference Index: This quantifies temporal changes caused by amplification, and a measure is used for comparing the temporal contrasts of the two acoustic signals called EDI. The Block diagram of the experimental set up for EDI calculation is shown in Fig 3. To calculate EDI the original input waveform is fed as input to hearing aid. The original waveform is scaled for the absolute values by squaring each value of the waveform. Since the signal had to be subtracted, the waveforms have to be scaled to a common reference point, so that temporal effects may be analyzed without contamination with amplitude variations. Scaling is accomplished by calculating the overall mean amplitude. This process scales each value of the waveform to mean amplitude of 1.0, allowing the direct comparison of the waveform. Correlation index was calculated using the formula:

5) Both the scaled waveforms are correlated using cross correlation technique

6) The CI value is calculated using the formula given above.

Fig 4: Flow chart of the algorithm used to calculate EDI



The unaided waveform (SAMPLEIn) was subtracted point by point, from the aided waveform (SAMPLE2n), and the absolute value of the difference was taken. Correlation Index (CI) was calculated as the mean of these value divided by 2. CI value ranges from 0.00 (perfect correspondence between two waveforms) to 1.00 (no correspondence between two waveforms)

- The procedure of recording was similar to that in CVR calculation. After recording the output processed stimuli for the three conditions, following steps were carried out.
- The length of the unprocessed and processed stimuli for the three conditions at 65 and 80 dBSPL were manually adjusted to be exactly equal using Wave Surfer. This was a complicated job because each time we had

to run the stimuli in MATLAB then check the length, depending on that we had to adjust the output in Wave surfer. It took around 30 min for each stimulus for making the length equal to the input.

> After this the input and processed stimuli were run in MATLAB and using the algorithm Correlation Index was calculated. The EDI calculation was done for the data of one subject.

RESULTS

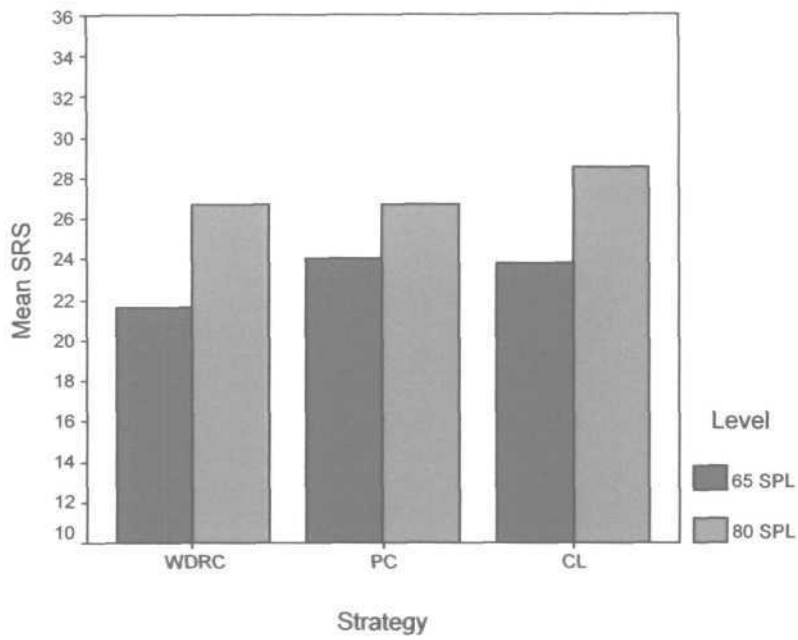
[A] Experiment-I: The speech recognition scores obtained for 10 subjects were analyzed to study the effect of amplification strategies and levels. SPSS, Statistical Package for Social Sciences (version 10) for windows was used to analyze the data. The following parameters were analyzed.

I) Effect of strategy on Speech recognition Scores: Table 2 shows the overall mean Speech recognition scores, Standard deviation for different amplification strategies at 65 and 80 dBSPL. The mean scores were better for the Peak clipping (PC) at 65 dBSPL and for Compression Limiting (CL) strategy at 80 dBSPL.

Level (dBSPL)	Strategies	Mean	SD
65	Compression Limiting	23.8	2.49
	Peak Clipping	24.0	3.83
	Wide Dynamic Range Compression	21.6	2.72
80	Compression Limiting	28.5	2.01
	Peak Clipping	26.7	4.11
	Wide Dynamic Range Compression	26.7	4.00

Table 2- Mean and Standard Deviation of speech recognition scores across conditions

Graph 1 shows the Mean percentage Speech recognition scores across strategies. The mean percentage scores were higher for the Peak Clipping at 65 dBSPL and for Compression Limiting strategy at 80 dBSPL



Graph-1 Mean Percentage Speech recognition scores across strategies

a) ***At 65 dBSPL input:***

One-way repeated measure ANOVA was performed for comparison across strategies within 65 dBSPL. The effect of amplification strategy was significant, F (2, 18) = 3.661; $p < 0.05$. Since there was a significant difference across strategies, pair-wise differences among them was tested with Bonferroni's multiple comparison. There was a significant difference between WDRC and CL at 0.05 level of significance. The mean scores were better for Compression Limiting than WDRC. The remaining pairs were not significant at 0.05 level. ***Therefore, the*** *Hypothesis* ***that*** *there is no significant difference in Speech recognition scores across strategies at 65 dBSPL was rejected.*

b) At 80 dBSPL input:

One-way repeated measure ANOVA was performed for comparison across strategy for 80 dBSPL input. The effect of amplification strategy was not significant, $F(2, 18) = 1.254$; $p > 0.05$. *Therefore, the Hypothesis that there is no significant difference in Speech recognition scores across strategies at 80 dBSPL was accepted.*

2) Effect of presentation level on speech recognition Scores

Paired t- test was done for comparison across level within each strategy. The effect of presentation level was significant for all the strategies, WDRC [$t(9) = 4.680$, $p < 0.05$], PC [$t(9) = 6.384$, $p < 0.05$], CL [$t(9) = 6.567$; $p < 0.05$]. The scores were higher at 80 dBSPL than at 65 dBSPL. *Therefore, the Hypothesis that there is no significant difference in Speech recognition scores across levels within each strategy was rejected.*

In summary it can be stated that there was performance differenece between Compression Limiting and WDRC strategies at 65 dBSPL but performance was similar for Compression Limiting and Peak clipping strategies. However, there was no difference in the performance between all the three strategies at 80 dBSPL.

[B] Experiment II

la) Effect of strategy on consonant vowel ratio

The consonant vowel ratio values obtained for 5 subjects were analyzed to study the effect of amplification strategy. The CVR obtained were divided on the basis of vowel and consonant environments. There were 3 vowel groups (/a/, /i/ */u/)* and 6 consonants groups (stops, nasals, affricates fricatives liquids and glides).

a) *Peak clipping condition'.* Table 3 shows the overall mean CVR values and Standard Deviation for 5 subjects The CVR values were for the stimuli with vowel environment /a/, /i/ */u/* divided into 6 consonants groups (stops, nasals, affricates, fricatives, liquids and glides)

Group	Stimuli	N	Mean	SD
Stop /p/, /b/, /t/,/d/, /k/, /g/	/a/65	6	0.82	0.02
	/a/80	6	0.77	0.05
	/i/65	6	0.34	0.08
	/i/80	6	0.33	0.04
	/u/65	6	0.66	0.12
	/u/80	6	0.66	0.07
Nasal /m/, /n/	/a/65	2	0.85	0.01
	/a/80	2	0.81	0.02
	/i/65	2	0.34	0.04
	/i/80	2	0.27	0.02
	/u/65	2	0.60	0.01
	/u/80	2	0.74	0.06
Affricate /tʃ/, /dz/	/a/65	2	0.79	0.02
	/a/80	2	0.77	0.00
	/i/65	2	0.43	0.27
	/i/80	2	0.33	0.05
	/u/65	2	0.73	0.04
	/u/80	2	0.72	0.11
Fricative /s/, /ʃ/, /h/	/a/65	3	0.75	0.04
	/a/80	3	0.74	0.04
	/i/65	3	0.40	0.08
	/i/80	3	0.48	0.13
	/u/65	3	0.68	0.04
	/u/80	3	0.70	0.05
Liquid /r/, /l/	/a/65	2	0.82	0.06
	/a/80	2	0.79	0.00
	/i/65	2	0.34	0.01
	/i/80	2	0.27	0.01
	/u/65	2	0.66	0.07
	/u/80	2	0.68	0.14
Glide /v/	/a/65	1	0.84	.
	/a/80	1	0.79	.
	/i/65	1	0.42	.
	/i/80	1	0.48	.
	/u/65	1	0.74	.
	/u/80	1	0.80	.

Table-3- Mean Consonant vowel ratio and standard deviation for Peak Clipping

Table 4 shows the mean CVR values and Standard Deviation of the input unprocessed stimuli and the output processed stimuli in 3 different vowel environments for 5 subjects.

Stimuli		Mean	SD
/a/ 65	Input	0.68	0.14
	Output	0.80	0.04
/a/ 80	Input	0.60	0.17
	Output	0.77	0.04
/i/65	Input	0.39	0.22
	Output	0.37	0.10
/i/80	Input	0.21	0.18
	Output	0.35	0.09
/u/ 65	Input	0.66	0.23
	Output	0.67	0.08
/U/ / 8 0	Input	0.58	0.19
	Output	0.69	0.08

Table 4- Mean and Standard Deviation of CVR values for the processed and unprocessed stimuli

Paired t test was done to compare the CVR values of the unprocessed input and processed output stimuli.

i) Stimuli with vowel environment /a/: There was significant difference between the CVR values of the unprocessed and processed stimuli at 65dBSPL [t (15) = 3.451, p<0.01] and 80 65dBSPL [t (15) =4.351, p<0.01]. The CVR values were higher for the processed stimuli as compared to the unprocessed stimuli both for 65 and 80 dBSPL

ii) Stimuli with vowel environment /if: There was no significant difference between the CVR values of the unprocessed and processed stimuli at 65dBSPL [t (15) = 0.736, p>0.01] whereas difference was seen at 80 dBSPL input [t (15) = 2.899, p<0.01]. The CVR was enhanced significantly after processing at 80 but not at 65 dBSPL.

Hi) Stimuli with vowel environment /u/: There was no significant difference between the CVR values of the unprocessed and processed stimuli at 65dBSPL [t (15) = 0.259, p>0.01] whereas difference was seen at 80 dBSPL input [t (15) = 2.940, p<0.01]. The CVR was enhanced significantly after processing at 80 but not at 65 dBSPL.

b) *Compression Limiting:* Table 5 shows the overall mean CVR values and Standard Deviation for 5 subjects. The CVR values were for stimuli with vowel environment /a/, /i/, *Iwl* divided into 6 consonants groups (stops, nasals, affricates, fricatives, liquids and glides respectively). Table 6shows the mean CVR values and Standard Deviation of the input unprocessed stimuli and the output processed stimuli in 3 different vowel environments.

Group	Stimuli	N	Mean	SD
Stop /p/, /b/, /t/,/d/, /k/, /g/	/a/65	6	0.82	0.02
	/a/80	6	0.77	0.05
	/i/65	6	0.34	0.08
	/i/80	6	0.33	0.04
	/u/65	6	0.66	0.12
	/u/80	6	0.66	0.07
Nasal /m/, /n/	/a/65	2	0.82	0.03
	/a/80	2	0.79	0.01
	/i/65	2	0.37	0.00
	/i/80	2	0.36	0.01
	/u/65	2	0.84	0.01
	/u/80	2	0.77	0.01
Affricate /tʃ/, /dz/	/a/65	2	0.79	0.00
	/a/80	2	0.83	0.01
	/i/65	2	0.52	0.26
	/i/80	2	0.36	0.01
	/u/65	2	0.73	0.07
	/u/80	2	0.71	0.05
Fricative /s/, /ʃ/, /h/	/a/65	3	0.76	0.04
	/a/80	3	0.79	0.03
	/i/65	3	0.40	0.14
	/i/80	3	0.54	0.05
	/u/65	3	0.70	0.07
	/u/80	3	0.68	0.07
Liquid /r/, /l/	/a/65	2	0.80	0.01
	/a/80	2	0.82	0.03
	/i/65	2	0.38	0.02
	/i/80	2	0.33	0.04
	/u/65	2	0.75	0.04
	/u/80	2	0.76	0.02
Glide /v/	/a/65	1	0.76	.
	/a/80	1	0.80	.
	/i/65	1	0.36	.
	/i/80	1	0.39	.
	/u/65	1	0.78	.
	/u/80	1	0.72	.

Table-5- Mean Consonant vowel ratio and standard deviation for Compression limiting

Stimuli		Mean	SD
/a/ 65	Input	0.68	0.14
	Output	0.79	0.04
/a/80	Input	0.60	0.17
	Output	0.80	0.02
/i/65	Input	0.39	0.22
	Output	0.41	0.11
/i/80	Input	0.20	0.18
	Output	0.40	0.08
/u/65	Input	0.66	0.23
	Output	0.75	0.07
/u/80	Input	0.58	0.19
	Output	0.71	0.05

Table 6: Mean CVR values and Standard Deviation for unprocessed and processed stimuli

Paired t test was done to compare the CVR values of the unprocessed input and processed output stimuli.

i) Stimuli with vowel environment /a/

There was significant difference between the CVR values of the unprocessed and processed stimuli at 65dBSPL [t (15) = 2.840, $p<0.01$] and 80 dBSPL [t (15) =5.035, $p<0.01$].

ii) Stimuli with vowel environment ///

There was no significant difference between the CVR values of the unprocessed and processed stimuli at 65dBSPL [t (15) = 0.194, $p>0.01$] whereas difference was seen at 80 dBSPL input [t (15) = 4.454, $p<0.01$].

Hi) Stimuli with vowel environment /u/

There was no significant difference between the CVR values of the unprocessed and processed stimuli at 65dBSPL [t (15) = 1.389,

$p > 0.01$] whereas difference was seen at 80 dBSPL input [$t(15) = 2.720$, $p < 0.01$].

c) *Wide dynamic range compression:* Table 7 shows the overall mean CVR values and Standard Deviation for 5 subjects. The CVR values were for the stimuli with vowel environment /a/, /i/, */u/* divided into 6 consonants groups (stops, nasals, affricates, fricatives, liquids and glides respectively).

Group	Stimuli	N	Mean	SD
Stop /p/, /b/, /t/,/d/, /k/, /g/	/a/65	6	0.70	0.04
	/a/80	6	0.73	0.04
	/i/65	6	0.48	0.10
	/i/80	6	0.34	0.05
	/u/65	6	0.52	0.04
	/u/80	6	0.62	0.06
Nasal /m/, /n/	/a/65	2	0.73	0.03
	/a/80	2	0.76	0.10
	/i/65	2	0.47	0.07
	/i/80	2	0.26	0.00
	/u/65	2	0.58	0.01
	/u/80	2	0.67	0.01
Affricate /tʃ/, /dz/	/a/65	2	0.74	0.07
	/a/80	2	0.81	0.00
	/i/65	2	0.61	0.11
	/i/80	2	0.39	0.09
	/u/65	2	0.50	0.02
	/u/80	2	0.60	0.16
Fricative /s/, /ʃ/, /h/	/a/65	3	0.72	0.08
	/a/80	3	0.72	0.06
	/i/65	3	0.42	0.09
	/i/80	3	0.41	0.08
	/u/65	3	0.62	0.08
	/u/80	3	0.71	0.04
Liquid /r/, /l/	/a/65	2	0.66	0.02
	/a/80	2	0.75	0.04
	/i/65	2	0.51	0.01
	/i/80	2	0.29	0.06
	/u/65	2	0.60	0.02
	/u/80	2	0.70	0.04
Glide /v/	/a/65	1	0.68	.
	/a/80	1	0.68	.
	/i/65	1	0.48	.
	/i/80	1	0.41	.
	/u/65	1	0.64	.
	/u/80	1	0.68	.

Table-7- Mean Consonant vowel ratio and standard deviation for WDRC

Stimuli		Mean	SD
/a/ 65	Input	0.68	0.14
	Output	0.70	0.50
/a/80	Input	0.60	0.17
	Output	0.74	0.05
/i/65	Input	0.40	0.22
	Output	0.49	0.10
/i/80	Input	0.21	0.18
	Output	0.35	0.08
/u/65	Input	0.56	0.23
	Output	0.66	0.06
/u/80	Input	0.58	0.19
	Output	0.66	0.07

Table 8: Mean CVR values and Standard Deviation for unprocessed and processed stimuli.

Paired t test was done to compare the CVR values of the unprocessed input and processed output stimuli.

i) Stimuli with vowel environment /a/: There was no significant difference between the CVR values of the unprocessed and processed stimuli at 65dBSPL [t (15) = 0.520, p>0.01] whereas there was significant difference at 80 dBSPL [t (15) =3.152, p<0.01].

ii) Stimuli with vowel environment /i/: There was no significant difference between the CVR values of the unprocessed and processed stimuli at 65dBSPL [t (15) = 1.431, p>0.01] whereas difference was seen at 80 dBSPL input [t (15) = 3.118, p<0.01].

iii) Stimuli with vowel environment /u/: There was no significant difference between the CVR values of the unprocessed and processed stimuli at 65dBSPL [t (15) = 1.488, p>0.01] at 80 dBSPL input [t (15) = 1.790, p>0.01].

In summary, the consonant vowel ratio values for the processed were higher than that of unprocessed for all the three conditions, following similar trend for peak clipping and compression limiting but differed for wide dynamic range compression.

lb) Comparison of CVR values across different amplification strategies

i) Stimuli with vowel environment /a/

a) *At 65 dBSPL:* The mean CVR values and Standard Deviation for the processed stimuli for all the 3 conditions are shown in Table 9. The mean CVR values were higher for peak clipping.

Strategies	Mean	SD
Peak Clipping	0.81	0.04
Compression Limiting	0.79	0.03
Wide Dynamic Range Compression	0.70	0.05

Table 9: Mean CVR values across strategies for stimuli with vowel /a/ at 65 dBSPL

One-way repeated measure ANOVA was done to see the effect of amplification condition. There was a significant effect of amplification condition on the CVR values [F (2, 30) = 4.946, p < 0.05]. Since there was a significant difference across strategies, pair-wise differences among them was tested with Bonferroni's multiple comparison. There was significant difference between Peak Clipping and WDRC, WDRC and Compression Limiting at 0.05 level of significance. There was no significant difference between Peak Clipping and Compression Limiting at 0.05 level. *The hypothesis that there is no significant difference between the CVR values across strategies at 65 dBSPL was rejected for the vowel environment /a/.*

b) *At 80 dBSPL:* The mean CVR values and Standard Deviation for the processed stimuli for all the 3 conditions are shown in Table 10. The mean CVR values were higher for compression limiting

Strategies	Mean	SD
Peak Clipping	0.77	0.39
Compression Limiting	0.80	0.02
Wide Dynamic Range Compression	0.74	0.05

Table 10: Mean CVR values across strategies for stimuli with vowel /a/ at 80 dBSPL

One-way repeated measure ANOVA was done to see the effect of amplification condition. There was a significant effect of amplification condition on the CVR values. [$F (2, 30) = 10.659$, $p < .05$]. Since there was a significant difference across strategies, pair-wise differences among them was tested with Bonferroni's multiple comparison. There was significant difference between Peak Clipping and Compression Limiting, WDRC and Compression Limiting at 0.05 level of significance. There was no significant difference between Peak Clipping and WDRC. *The hypothesis that there is no significant difference between the CVR values across strategies at 80 dBSPL was rejected for the vowel environment /a/.*

ii) Stimuli with vowel environment /i/

a) *At 65 dBSPL:* The mean CVR values and Standard Deviation for the processed stimuli for all the 3 conditions are shown in Table 11. The mean CVR values were higher for Wide Dynamic range Compression.

Strategies	Mean	SD
Peak Clipping	0.37	0.99
Compression Limiting	0.41	0.10
Wide Dynamic Range Compression	0.49	0.09

Table-11-Mean CVR values across strategies for stimuli with vowel *l\l* at 65 dBSPL

One-way repeated measure ANOVA was done to see the effect of amplification condition. There was a significant effect of amplification condition on the CVR values [F (2, 30) = 12.961, $p < 0.05$]. Since there was a significant difference across strategies, pair-wise differences among them was tested with Bonferroni's multiple comparison. There was significant difference between Peak Clipping and WDRC, WDRC and Compression Limiting at 0.05 level of significance. There was no significant difference between Peak Clipping and Compression Limiting. *The hypothesis that there is no significant difference between the CVR values across strategies at 65 dBSPL was rejected for the vowel environment A/.*

a) *At 80 dBSPL:* The mean CVR values and Standard Deviation for the processed stimuli for all the 3 conditions are shown in Table 12. mean CVR values were higher for compression limiting.

Strategies	Mean	SD
Peak Clipping	0.35	0.95
Compression Limiting	0.40	0.08
Wide Dynamic Range Compression	0.35	0.07

Table 12- Mean CVR values across strategies for stimuli with vowel *I'\I* at 80 dBSPL

One-way repeated measure ANOVA was done to see the effect of amplification condition. There was a significant effect of amplification condition on the CVR values [$F(2, 30) = 5.533$, $p < 0.05$]. Since there was a significant difference across strategies, pair-wise differences among them was tested with Bonferroni's multiple comparison. There was significant difference between Peak Clipping and Compression Limiting, WDRC and Compression Limiting at 0.05 level of significance. There was no significant difference between Peak Clipping and WDRC. *The hypothesis that there is no significant difference between the CVR values across strategies at 80 dBSPL was rejected for the vowel environment /i/.*

in) Stimuli with vowel environment /u/

a) At 65 dBSPL: The mean CVR values and Standard Deviation for the processed stimuli for all the 3 conditions are shown in Table 13. The mean CVR scores were higher for compression limiting.

Strategies	Mean	SD
Peak Clipping	0.67	0.07
Compression Limiting	0.75	0.07
Wide Dynamic Range Compression	0.56	0.06

Table 13- Mean CVR values across strategies for stimuli with vowel *IvJ* at 65 dBSPL

One-way repeated measure ANOVA was done to see the effect of amplification condition. There was a significant effect of amplification condition on the CVR values [$F (2, 30) = 42.301$, $p < 0.05$]. Since there was a significant difference across strategies, pair-wise differences among them was tested with Bonferroni's multiple comparison. There was significant difference between Peak Clipping and Compression Limiting, WDRC and Compression Limiting, Peak Clipping and WDRC at 0.05 level of significance. *The hypothesis that there is no difference between the CVR values across strategies at 65 dBSPL was rejected for vowel environment* /«/.

b) *At 80 dBSPL:* The mean CVR values and Standard Deviation for the processed stimuli for all the 3 conditions are shown in Table 14. The mean CVR were higher for compression limiting.

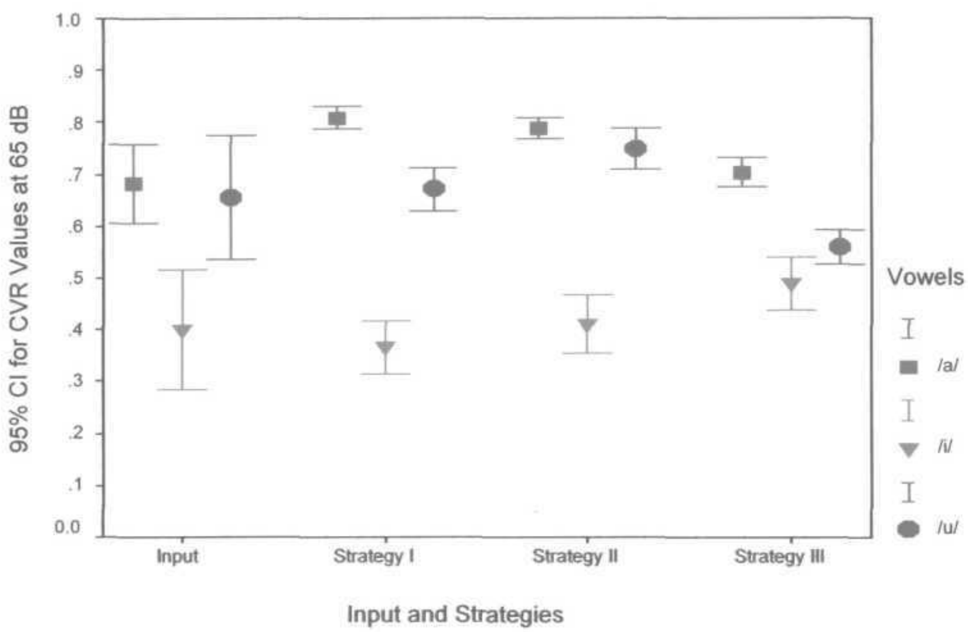
Strategies	Mean	SD
Peak Clipping	0.70	0.08
Compression Limiting	0.71	0.05
Wide Dynamic Range Compression	0.66	0.06

Table 14- Mean CVR values across strategies for stimuli with vowel /u/ at 80 dBSPL

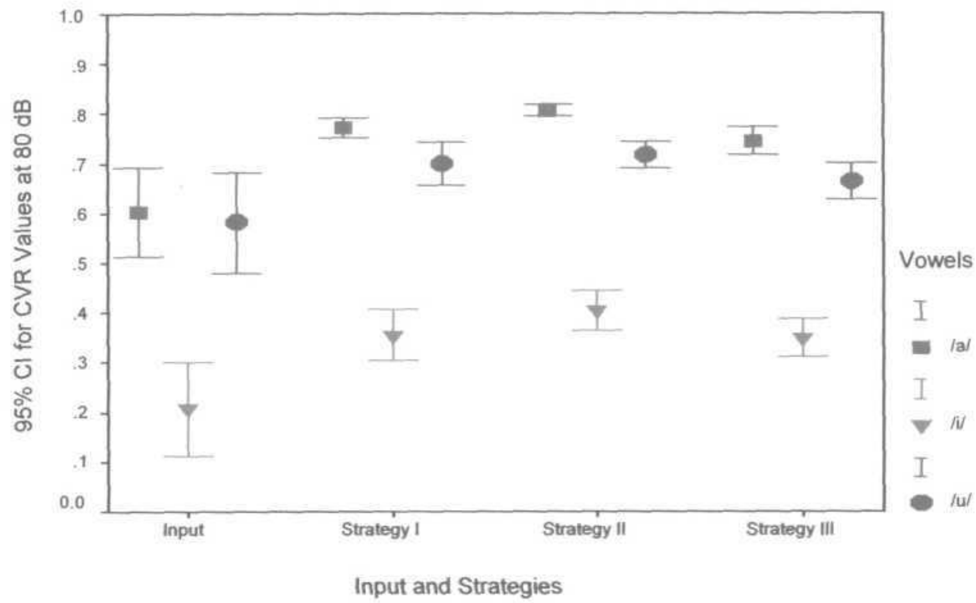
One-way repeated measure ANOVA was done to see the effect of amplification condition. There was a significant effect of amplification condition on the CVR values [$F (2, 30) = 4.194$, $p < 0.05$]. Since there was a significant difference across strategies, pair-wise differences among them was tested with Bonferroni's multiple comparison. There was significant difference between Compression Limiting and WDRC at 0.05 level of significance. There was no significant difference found between Peak Clipping and WDRC, and Compression Limiting and Peak Clipping. ***The hypothesis that there is no*** *significant difference between the* ***CVR*** *values across* ***strategies at 80 dBSPL was rejected for vowel environment/u/.***

The overall result shows that the CVR values were higher for the Compression Limiting strategy for the stimuli with vowel environment /a/ at 80 dBSPL, /i/ at 80 dBSL and ***IvJ*** at 65 and 80 dBSPL whereas it was higher for Peak Clipping strategy for stimuli with vowel /a/ at 65 dBSPL and WDRC strategy for *l\l* at 65 dBSPL.

Graph 2 & 3 shows the 95% confidence interval for Consonant vowel ratio at 65 and 80 dBSPL respectively for input and output processed stimuli across strategies. The range of CVR values was greater for the input stimuli as compared to the output stimuli at 65 and 80 dBSPL. The CVR values were higher for the stimuli with /a/ and ***IvJ*** vowel environment than for *HI* both for the input and output stimuli.



Graph 2: Error graph showing 95% confidence interval for Consonant vowel ratio at 65 dBSPL for input and output processed stimuli across strategies. Strategy I- peak clipping, strategy **II-** compression limiting, strategy **III-** wide dynamic range compression



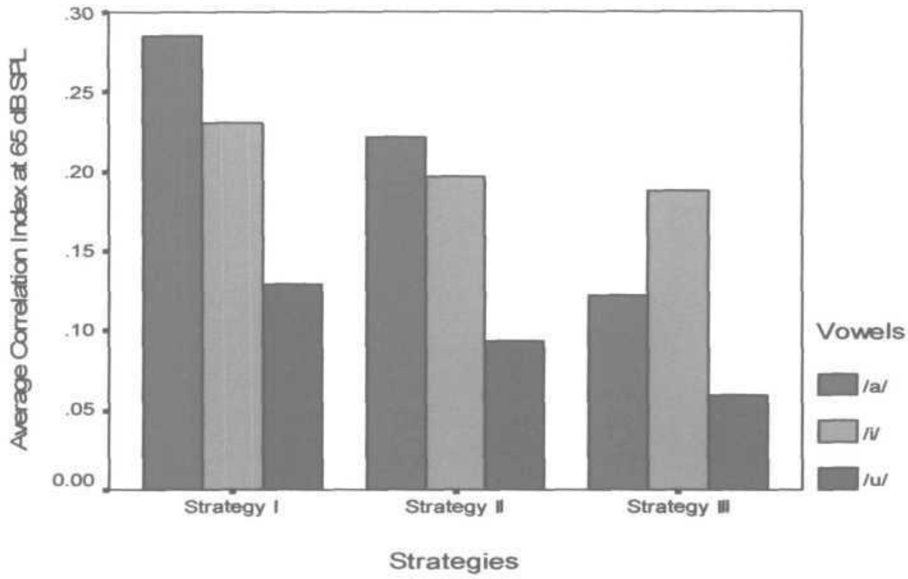
Graph 3 - Error graph showing 95% confidence interval for Consonant vowel ratio at 80 dBSPL for input and output processed stimuli across strategies. Strategy I- peak clipping, strategy II- compression limiting, strategy III- wide dynamic range compression

2) Effect of strategies on Envelope Difference Index: Table 15 shows the mean Correlation index values and standard deviation for the stimuli with vowel environment /a/, /i/, *lul* at input 65 and 80 dBSPL for one subject across three strategies

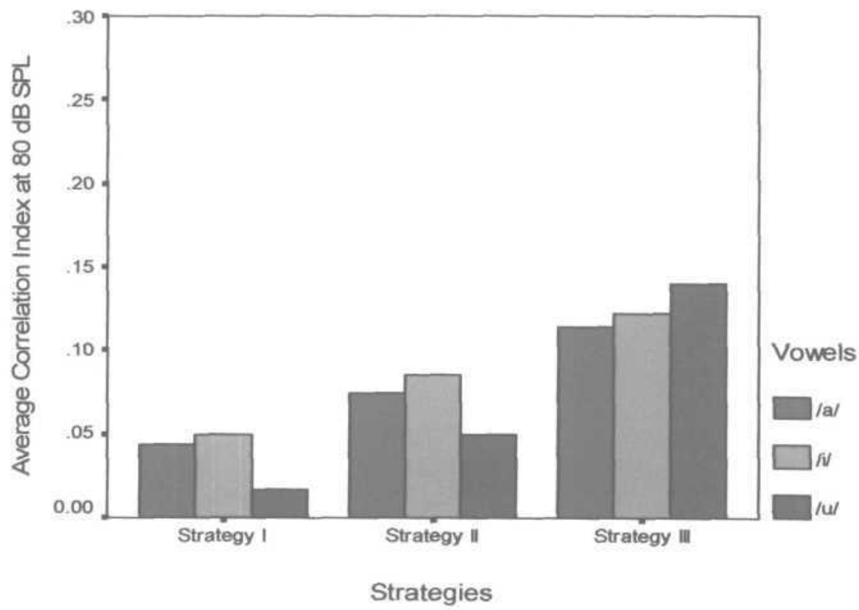
stimuli	Strategies					
	Peak Clipping		Compression Limiting		Wide Dynamic Range Compression	
	Mean	SD	Mean	SD	Mean	SD
/a/ 65	0.29	0.17	0.22	0.18	0.12	0.09
/a/ 80	0.04	0.02	0.74	0.07	0.11	0.10
III 65/	0.23	0.13	0.20	0.14	0.19	0.15
/i/80	0.05	0.05	0.08	0.13	0.12	0.14
/u/65	0.13	0.10	0.09	0.06	0.06	0.06
/u/80	0.02	0.06	0.05	0.07	0.14	0.15

Table 15- Mean Correlation index values and standard deviation for the stimuli with Vowel environment /a/, /i/, *lul* at input 65 and 80 dBSPL

Graph 4 & 5 shows the average Correlation index value across strategies at 65 and 80 dBSPL respectively.



Graph 4- Average Correlation index across strategies for stimuli with vowel environment /a/, /i/, /u/ at 65 dBSPL. Strategy I- peak clipping, strategy II- compression limiting, strategy III- wide dynamic range compression



Graph 5 - Average Correlation index across strategies for stimuli with vowel environment /a/, /i/, /u/ at 80 dBSPL. Strategy I- peak clipping, strategy II- compression limiting, strategy III- wide dynamic range compression.

To recall, an EDI value of 0 means no difference between the unprocessed and processed stimuli, and an EDI value of 1 means no correspondence between the two temporal envelopes. The EDI value was greater at 65 dBSPL and as the level increased to 80 dBSPL there was a decrease in the EDI value for all the strategies, the change more significant for Peak clipping and Compression limiting but not for Wide dynamic range compression. The EDI values were highest for Peak clipping and lowest for compression limiting at 65 dBSPL whereas at 80 dBSPL it was just the opposite, highest for WDR.C and lowest for Peak Clipping.

DISCUSSION

1. Speech recognition scores

Results of the study demonstrated significant difference in speech recognition scores across strategies at 65 dBSPL but no significant difference between strategies at 80 dBSPL. The scores were better with Compression Limiting at 65, 80 dBSPL compared to WDRC. The study is in agreement with previous study by Barker et al (2001) and Souza & Jenstad (2005). Even Hickson & Thyer (2003) found that at higher input levels there was no difference between linear and compression amplification. We know that variation in amplitude over time provides critical speech information. Some authors have suggested that severely hearing impaired listeners depend more heavily on these cues, because their broadened auditory filters prevent full access to spectral detail. As compression varies the temporal structure of speech thus degrading the cues for perception. Bishop & Souza (1999) demonstrated compression benefit was linked to the degree of hearing loss, with smaller improvements observed with pure tone thresholds exceeds 70 dBSPL.

In this study, data was collected on specific groups of stimuli from a small number of subjects. To assess the efficacy of the approach, it would be desirable to obtain similar measures from a large number of subjects across a relatively small number of phonetic contrasts. Because the listeners in this study wore the amplification systems only in a laboratory environment, one concern is that recognition might differ if subjects wore similar hearing aid in every day use for a longer period of time. Several studies have shown no acclimatization in subjects accustomed to linear amplification and a newly fit one or 2 channel WDRC aids (Keidser & Grant, 2001). However, Kuk (2001) suggested that severely hearing impaired subjects fitted with more complex systems, those that incorporate large numbers of channels, may require more time to reach maximum recognition performance. To study the effect of long term experience on recognition is beyond the scope of the present study, future work should continue to explore this issue.

2. Consonant vowel ratio

The results of the study showed that the Consonant vowel ratio was better for Compression limiting condition compared to Wide dynamic range compression for the stimuli with vowel environment at /a/ dBSPL, 80, *IV* at 80 dBSPL and /u/ at 65 and 80 dBSPL but not for /a/ and *IV* at 65 dBSPL. One of the reasons attributed is that, the benefits of recruitment compensation may be nullified in effect by temporal distortions from the compressor attack and recovery times and their alterations of the normal intensity cues of speech. As compression limiting is only active at high signal levels; it may provide better CVR without significantly altering the dynamics of conversational speech signals compared to the effect of a syllabic compressor as speculated by Walker & Dillon (1982) and Dreschler (1988b).

The finding of the study suggests that substantial changes in a speech signal can occur as a result of signal processing by hearing aids. In addition to simple changes due to frequency shaping, temporal changes, such as loss or reduction in the periodicity associated with voicing and as obscuring of the boundary between aperiodic consonant noise and the onset of voicing, can occur. In this study marked changes in Consonant vowel ratio occurred with processing. The magnitude of these changes for a given syllable however appears to be influenced by many factors including system release time, compression parameters, amplitude and duration of preceding speech sounds, the time delay between the vowel and consonant and the amplitude of the unprocessed consonant. As such the changes in the speech signal observed after processing may not be easily predicted from traditional electroacoustic measures of hearing aid performance. There is likely to be a complex interaction between the dynamic characteristics of hearing aid processing and the dynamic characteristics of the speech signal. The effect of these various acoustic alterations on speech perception for listeners with hearing loss is unclear. For listeners with normal hearing or individuals with mild to moderate hearing loss, multiple acoustic cues and linguistic experience may render these acoustic changes irrelevant. For listeners with severe to profound hearing loss, however, these acoustic changes may have a more significant effect on perception.

The result of the present study indicate a relationship between acoustic changes to the hearing aid processed speech signal and speech perception performance of severely hearing impaired individuals. The Consonant vowel ratio was higher for the Compression Limiting compared to WDRC strategy for most of the stimuli, and also the speech recognition scores were better with Compression Limiting compared to WDRC. So it is clear that the acoustic analysis of the aided speech signal is does provide indicators about the perceptual measures and thus has clinical applications. It may be possible to define acceptable level of distortion with compression amplification using acoustic analysis. It is needless to say that speech perception is one aspect of the outcome with compression amplification and those other measures (e.g.. self report of sound quality, satisfaction) cannot be underestimated.

The quantification of audibility for specific acoustic segments is a very complex issue. Ideally, one would like to predict performance from measures of audibility for the acoustic segments associated with specific phonemes or class of phonemes. Since multiple cues are often available, particularly for listeners with normal hearing, it is difficult to determine which aspects of a signal should be used to compute audibility. It is likely that redundancy of cues is reduced for listeners with hearing loss. Further acoustic analysis of a subset of the phonemes is necessary to quantify how much of the benefit was due to increased audibility and how much of the detriment was due to unacceptable changes in the temporal envelope of the signal. Stelmachowicz, Kopun, Mace, Lewis, and Nittrouer (1995) proposed that traditional Articulation Index and Speech Transmission Index measures were not suitable for quantifying audibility of individual speech sounds processed with compression. Instead, several acoustic cues need to be explored for each speech sound. Such a metric has not yet been defined, although Stelmachowicz et al.(1995) offered some suggestions and examples for quantifying the audibility of compressed stop consonants. Developing such a procedure is beyond the scope of the current investigation, but such a metric can be developed in the future.

3. Envelope difference index

The EDI value was greater at 65 dBSPL and as the level increased to 80 dBSPL there was a decrease in the EDI value for all the strategies. The change more significant for Peak clipping and Compression limiting and not for Wide dynamic range compression. Since the results were only for one subject one cannot generalize the results.

SUMMARY AND CONCLUSION

Individuals with severe hearing impairment exhibit reduced frequency resolution and temporal discrimination. Therefore, the requirements of amplification for this group of population will be different from those with lesser degree of hearing loss. Literature shows that individual with Mild to moderate hearing loss get benefit with Wide Dynamic Range Compression (Souza 2002, Moore, Peter and Stone 1999). The few studies done on severely hearing impaired individuals (Souza, Jenstad 2005), states that WDRC strategy are not beneficial, Speech perception scores are reported to be poorer with this strategy.

The aim of the present study was to study the effect of different amplification strategies on the speech recognition scores of individual with severely impaired hearing. Acoustic analysis of the speech stimuli processed through different amplification strategies was done by calculating the Consonant Vowel Ratio and Correlation Index to study the relationship with the perceptual measures.

A total of 10 subjects having moderately severe to severe hearing loss participated in the study. A CV word list of nonsense syllables was used as the stimulus, and Speech Recognition Scores were calculated at input level of 65 and 80 dBSPL for all the three strategies (Peak Clipping, Compression Limiting, and Wide Dynamic Range Compression). Consonant Vowel Ratio was calculated for the unprocessed and processed stimuli for 5 subjects at input level of 65 and 80 dBSPL for all the three strategies The CVR was calculated in Matlab using an algoritm.

Correlation Index was calculated for one subject at input level of 65 and 80 dBSPL for all the strategies in Matlab using an algorithm.

The results of the study are the following:

> There was significant difference in speech recognition scores across strategies at 65 dBSPL but no significant difference between strategies at 80 dBSPL. The scores were better with Compression Limiting compared to Wide Dynamic Range Compression at both 65, 80 dBSPL.

> The consonant vowel ratio values for the processed signal were higher than that of unprocessed for all the three conditions, following similar trend for peak clipping and compression limiting but differed for wide dynamic range compression.

> The CVR values for the processed stimuli with vowel environment /a/ at 80 dBSPL, *l\l* at 80 dBSPL and *Inl* both at 65, 80 dBSPL were higher for the Compression Limiting strategy as compared to WDRC and Peak Clipping strategy. For the stimuli with vowel environment /a/ at 65 dBSPL CVR values were higher for Peak clipping strategy and higher for WDRC strategy for *I'll* at 65 dBSPL.

> The EDI value was greater at 65 dBSPL and as the level increased to 80 dBSPL there was a decrease in the EDI value for all the three strategies. The change was more significant for Peak clipping and Compression limiting and not for Wide dynamic range compression.

> The Consonant vowel ratio for most of the stimuli was higher for the Compression Limiting strategy as compared to WDRC strategy. Similarly the speech recognition scores were also better with Compression Limiting strategy as compared to WDRC strategy. The result of the present study indicate a relationship between acoustic changes to the hearing aid processed speech signal and speech perception performance of severely hearing impaired individuals.

The study represents a step at resolving the clinical issue of how audiologists choosing the right amplification strategy while prescribing hearing aids for the severely hearing impaired individuals. The acoustic analysis is an initial step in describing and quantifying the effects of amplification strategies on phonemes. Further research needs to be done to see the interaction with other compression parameters and with different speech stimuli.

Limitation of the study and Future directions

1) In the present study Speech recognition scores were calculated in quiet condition which does not depict real life situations so further study may be done to see the effect of amplification strategies on speech perception in noise.
2) The study was done on a small group of subjects so the results cannot be generalized, hence it may be replicated on a larger number of subjects to validate the results.

3) Subject preferences for amplification strategy were not checked. Further studies can be done to see the preferences of subjects for different strategies by using a rating scale.

4) The choice of amplification strategy might change in long term use of hearing aids by the clients as reported in the literature. Such variables were not studied in the present study. Hence, studies on acclimatization effects in long term use of hearing aids and their effect on speech perception can be carried out.

5) The present study addressed to only the effect of different amplification strategies on speech perception and speech acoustics. However, there are other compression parameters such as Compression threshold, attack time / release time and Compression bands that affect speech perception. Further studies needs to be done to see the effect of these parameters on speech perception and speech acoustics.

REFERENCES

Allen, J. B., Hall, J. L., & Jeng, P. S. (1990). Loudness growth in octave bands (LGOB): A procedure for the assessment of loudness. *Journal of Acoustical Society of America*, 88, 745-753.

American National Standard Institute (1991) Maximum Permissible ambient noise for audiometric test rooms. ANSI S3.1-1991. New York: American National Institute.

Byrne, D. (1978). Selecting hearing aids for severely deaf children. *British Journal of Audiology,* 12, 9-22.

Braida, L. D., Durlach, N. I., Lippman, R. P., & Hicks, B. L. (1979). Hearing aids- a review of past research on linear amplification, amplitude compression, and frequency lowering. *American Speech and Hearing Monograph,* 19.

Bustamante, D., & Braida, L. (1987). Principal-component amplitude compression for the hearing impaired. *Journal of Acoustical Society of America*, 82, 1227-1242.

Boothroyd, A., Sringer, S., Smith, L., & Schulman, J. (1988). Amplitude compression and profound hearing loss. *Journal of Speech and Hearing Research,31,* 326-76.

Bunnel, T. (1990). On enhancement of spectral contrast in speech for speech for hearing -impaired listeners. *Journal of Acoustical Society of America,* 88, 2546-2556.

Byrne, D., Parkinson, A., & Newall, P. (1990). Hearing aid gain and frequency response requirements for the severely/ profoundly hearing impaired. *Ear and Hearing, 11,* 40-49.

Benson, D., Clark, T. M, & Johnson, J. S. (1992). Patient experience with multiband full dynamic range compression. *Ear and Hearing,* 13, 320-330.

Byrne, D. (1996). Hearing aid selection for the 1990s: Where to? *Journal of American Academy of Audiology,* 7, 377-395.

Byrne, D., Dillon, H., Ching, T., Katsch, R., & Keidser, G. (2001). The NAL-NL1 procedure for fitting non-linear hearing aids: Characteristics and comparisons with other procedures. *Journal of American Academy of Audiology,* 12, 37-51.

Crain, T. R., & Tassel, D. J. V. (1994). Effect of peak clipping on speech recognition threshold. *Ear and Hearing,* 15, 443-453.

Crain, T. R., & Yund, E. W. (1995). The effect of multichannel compression on vowel and stop consonant discrimination in normal hearing and hearing impaired subjects. *Ear and Hearing,* 16, 529-543.

Cox, R. M. (1995). Using loudness data for hearing aid selection: The IHAFF approach. *Hearing Journal,* 48, 39-44.

Cox, R. M. (1999). Five years later: An update on the IHAFF fitting protocol. *Hearing Journal,* 52, 10-18.

DeGennaro, S., Braida, L., & Durlach, N. (1986). Multiband syllabic compression for severely impaired listeners. *Journal of Rehabilitation Research Development,* 23, 17-24.

Dreschler, W. (1988). The effect of specific compression settings on phoneme identification in hearing impaired subjects. *Scandinavian Audiology,* 17, 35-43.

Dreschler, W. A. (1988). Dynamic range reduction by peak clipping or compression and its effects on phoneme perception in hearing impaired listeners. *Scandinavian Audiology,* 28, 49-60.

Dreschler, W. A. (1989). Phoneme perception via hearing aids with and without compression and the role of temporal resolution. *Audiology,* 28, 49-60.

Dillon, H. (1989). Hearing aid amplification method and apparatus. U. S. Patent.

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

Edgardh, B. (1952). The use of extreme limitation for the dynamic equalization of vowel and consonants in hearing aids. *Acta Otolaryngology,* 40, 376-386.

Erber, N. P. (1972). Speech-envelope cues as an acoustic aid to lip reading for profoundly deaf children. *Journal of Acoustical Society of America,* 51, 1224-1227.

Fowler, E. P. (1936). A method for early detection of otosclerosis. *Archives of Otolaryngology,* 24, 731-741.

Freyman, R. L., & Nerbonne, G. P. (1989). The importance of consonant vowel ratio intensity in the intelligibility of voiceless of voiceless consonants. *Journal of Speech and Hearing Research,* 32, 524-535.

Freyman, R., & Nerbonne, G. (1989). The importance of consonant vowel intensity ratio in the intelligibility of voiceless consonants. *Journal of Speech and Hearing Research,* 32, 524-535.

Faulkner, A., Rosen, S., & Moore, B. C. J. (1990). Residual frequency selectivity in the profoundly hearing impaired listener. *British Journal of Audiology,* 24, 381-392.

Freyman, R. L., & Nerbonne, G.P. (1991). Effect of consonant vowel ratio modification on amplitude envelope cues for consonant recognition. *Journal of Speech and Hearing Research,* 34, 415-426.

Fortune, T. W., Woodruff, B. D., & Preves, D. A. (1994). Anew technique for quantifying temporal envelope contrasts. *Ear and Hearing,* 15, 93-99.

Fortune, T. (1999). Aided growth of masking for speech and nonspeech signals. *Ear and Hearing,* 20, 214-227.

Gordon-Salant, S. (1987). Effects of acoustic modification on consonant recognition INF on by elderly hearing impaired subjects. *Journal of Acoustical Society of America,* 81, 1199-1202.

House, A. S., William, C. E., Hecker, M. H. L., & Kryter, K. D. (1965). Articulation - testing methods: Consonantal differentiation with a closed response set. *Journal of Acoustical Society of America,* 37, 158-166.

Hood, J. D. (1972). Fundamentals of identification of sensorineural hearing loss, *Sound,* 6, 21-26.

Hawkins, D. B., & Naidoo, S. V. (1993). Comparison of sound quality and clarity with asymmetrical peak clipping and output limiting compression. *Journal of the American Academy of A udiology,* 4, 221 -228.

Hickson, L. M. H. (1994). Compression amplification in hearing aids. *Journal of the American Academy of Audiology,* 3, 51-65.

Hickson, L., & Byrne, D. (1995). Acoustic analysis of speech through a hearing aid: effects of linear vs compression amplification. *Australian Journal of Audiology,* 17, 1-13.

Hickson, L. M. H., & Byrne, D. (1997). Consonant perception in quiet: Effect of increasing the consonant-vowel ratio with compression amplification. *Journal of American Academy of Audiology,* 8, 322-332.

Hickson, L. M. H., Thyer, N., & Bates, D. (1999). Acoustic analysis of speech through a hearing aid: consonant vowel ratio effects with two channel compression amplification. *Journal of the American Academy of Audiology,* 10,549-556.

Hedrick, M. S., & Rice, T. (2000). Effect of a single channel wide dynamic range compression circuit on perception of stop place of articulation. *Journal of Speech Language and Hearing Research,* 43, 1174-1184.

Hickson, L., & Thyer, N. (2003). Acoustic analysis of speech through a hearing aid: Perceptual effects of changes with two channel compression. *Journal of the American Academy of Audiology,* 14, 414-426.

Jenstad, L. M., Seewald, R. C, & Shantz, J. (1999). Comparison of linear and wide dynamic range compression hearing aid circuits: aided speech perception measures. *Ear and hearing,* 20, 117-127.

Jenstad, L. M., Pumford, J., Seewald, R. C, & Cornelisse, L. E. (2000). Comparison of linear gain and wide dynamic range compression hearing aid circuits II: Aided loudness measures. *Ear and Hearing,* 21, 32-44.

Jenstad, L. M., & Souza, P. E. (2005). Quantifying the effect of compression hearing aid release time on speech acousitcs and intelligibility. *Journal of Speech Language and Hearing Research,* 48, 651-667.

Jenstad, L. M., & Souza, P. E. (2005). Using multichannel wide dynamic range compression in severely hearing impaired listeners: Effects on speech recognition and quality. *Ear and Hearing,* 26, 120-131.

King, A., & Martin, M. (1984). Is AGC beneficial in hearing aids? *British Journal of Audiology,* 18,31-38.

Kates, J. (1986). Signal processing for hearing aids. *Hearing Instruments,* 36, 19-22.

Kennedy, E., Levitt,, H., Neuman, A. C, (1998) Consonant Vowel Intensity Ratios - Its effect on speech recognition by hearing impaired. *Journal of Acoustic Society of America,* 103, 1098-1114.

Kent, R. D., & Read, C. (1992). *The acoustic analysis of speech.* San Diego: Singular Publishing Group, Inc.

Kiessling, J., Schubert, M., & Archut, A. (1996). Adaptive fitting of hearing instruments by categorical loudness scaling. *Scandavian Audiology,* 25, 153-160.

Kennedy, E., Levitt, H., & Neuman, A. C. (1998) Consonant Vowel Intensity Ratios - Its effect on speech recognition by hearing impaired. *Journal of Acoustical Society of America,* 103, 1098-1114.

Kuk, F. K., & Ludvigsen, C. (2000). Hearing aid design and fitting solutions for persons with severe-to-profound loss. *Hearing Journal,* 53, 29-37.

Licklider, J.C. R., & Pollack, I. (1948). Effect of differentiation, integration, and infinite peak clipping upon the intelligibility of speech. *Journal of Acoustical Society of America,* 20, 42-51.

Lamore, P. J. J., Verweij, C, & Brocaar, M. P. (1990). Residual hearing capacity of severely hearing-impaired subjects. *Ada Otolaryngolica Supplement,* 469, 7-15.

Miller, G. A., & Mitchell, S. (1947). Effects of distortion on the intelligibility of speech at high altitudes. *Journal of Acoustical Society of America,* 19, 120-125.

Martin, D. W. (1950). Uniform speech - peak clipping in a uniform signal-to-noise spectrum ratio. *Journal of Acoustical Society of America,* 22, 614-621.

Montgomery, A., & Edge, R. (1988). Evaluation of two speech enhancement techniquews to improve intelligibility for hearing impaired adults. *Journal of Speech and Hearing Research,* 31,386-393.

Moore, B. (1990). How much do we gain by gain control in hearing aids? *Acta Otolaryngologica Supplement,* 469, 250-256.

Moore, B. J. C, Clark, T. M., & Johnson, J. S. (1992). Evaluation of a dual channel full dynamic range compression system for people with sensorineural hearing loss. *Ear and hearing,* 13,349-370.

Moore, B. C. J. (1996). Perceptual consequences of cochlear hearing loss and their implications for the design of hearing aids. *Ear and Hearing,* 17, 133-161.

Moore, B. C. J., & Glasberg, B. R. (1998).Use of loudness model for hearing aid fitting. I. Linear hearing aids. *British Journal of Audiology,* 35, 349-374.

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 Acoustical Society of America,* 105, 400-410.

Marriage, J. E., & Moore, B. C. J. (2002). New speech tests reveal benefit of wide dynamic range, fast acting compression for consonant discrimination in children with moderate to profound hearing loss. *International Journal of Audiology,* 42,418-425.

Marriage, J. E., Moore, B. C. J., & Stone, M. A. (2005). Effect of three amplification strategies on speech recognition by children with severe and profound hearing loss. *Ear and Hearing,* 26, 35-47.

Moore, B. C. J., & Stone, M. A. (2004). Side effects of fast dynamic range compression that affect intelligibility in a competing speech task. *Journal of Acoustical Society of America*, 116, 2311 -2323.

Niederjohn, R., & Grotelueschen, *J.* (1976). The enhancement of speech intelligibility in high noise levels by high pass filtering followed by rapid amplitude compression. *IEEE Transactions of Acoustics, Speech and Signal Processing*, 24, 277-282.

Nelson, D. A., & Freyman, R. L. (1987). Temporal resolution in sensorineural hearing impaired listeners. *Journal of Acousical Society of America*, 81, 709-720.

Pollack, I. (1952). On the effect of frequency and amplitude distortion on the intelligibility of speech in noise. *Journal of Acoustical Society of America*, 24, 538-540.

Pollack, I., & Picket, J. M. (1958). Masking of speech by noise at high noise levels. *Journal of Acoustical Society of America*, 30, 127-130.

Plomp, R. (1983). Perception of speech as a modulated signal. *Proceeding of the 10th International Congress of Phonetic sciences, Utrecht,* 29-40.

Picheny, M., Durlach, N., & Braida, L. (1985). Speaking clearly for a hard of hearing. II: Intelligibility differences between clear and conversational speech. *Journal of Speech and Hearing Research*, 28, 96-103.

Picheny, M., Durlach, N., & Braida, L. (1986). Speaking clearly for a hard of hearing. II: Acoustic characteristics of clear and conversational speech. *Journal of Speech and Hearing Research*, 29, 434-446.

Preves, D. A. Output limiting and speech enhancement. In: Studebaker, G., Bess, F. H., Beck, L. B., (1991). eds. The Vanderbilt Hearing Aid Report II, York Press, 35-51.

Plomp, R. (1994). Noise, amplification, and compression: considerations of three main issues in hearing aid design. *Ear and hearing,* 15, 2-12.

Rosen, S., Faulkner, A., & Smith, D. A. (1990). The psychoacoustics of profound hearing impairment. *Ada Otolaryngologica supplement,* 469, 16-22.

Ruggero, M. A., & Rich, N. C. (1991). Furosemide alters organ of corti mechanics: Evidence for feedback of outerhair cells upon the basilar membrane. *Journal of Neuroscience,* 11, 1057-1067.

Renkovic, C, Freyman, R., & Zurek, P. (1992). Potential benefit of adaptive gain characteristics for speech perception in noise. *Journal of Acoustical Society of America,* 91 (1), 354-362.

Rosen, S. (1992). Philosophical Transactions of the Royal Society of London B, 367-336.

Ricketts, T. A., & Bentler, R. A. (1996). The effect of test signal type and bandwidth on the categorical scaling of loudness. *Journal of Acoustical Society of America,* 99, 2281-2287.

Schwartz, D., Lyregaard, P. E., & Lundh, P. (1988). Hearing aid selection for severe-to-profound hearing loss. *Hearing Journal,* 41, 13-17.

Shannon, R. V., Zeng, F. G., Kamath, V., Wygonski, J., Ekelid, M. (1995). Speech recognition with primarily temporal cues. *Science,* 270, 303-304.

Stelmachowicz, P. G., & Lewis, D. E. (1995). The perception of amplified speech by listeners with hering loss: Acoustic correlates. *Journal of Acoustical Society of America*, 98, 1388-1399.

Souza, P. E., & Turner, C. W. (1996). Effect of single-channel compression on temporal speech information. *Journal of Speech and Hearing Research,* 39, 901-911.

Sreenivas, C, Fourakis, M., & Davidson, S. (1997). Effect of varying release times on speech perception using syllabic compression. Poster session presented at the annual meeting of the American Academy of Audiology, Fort Lauderdale, FL.

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

Souza, P. E., & Turner, C. W. (1999). Quantifying the contribution of audibility to recognition of compression-amplified speech. *Ear and Hearing,* 20, 12-20.

Shanks, J. E., Wilson, R. H., & Williams, D. (2002). Speech recognition performance of patients with sensorineural hearing loss under unaided and aided conditions using linear and compression hearing aids. *Ear and hearing,* 23, 280-290.

Souza, P. E. (2002). Effects of compression on speech acoustics, intelligibility, and sound quality. *Trends in amplification, 6,* 131- 165.

Shannon, R. V., Zeng, F. G, Kamath, V., Wygonski, J., Ekelid, M. (1995). Speech recognition with primarily temporal cues. *Science,* 270, 303-304.

Tyler, R. S., Summerfield, Q., Wood, E. *J.* & Fernandes, M. A. (1982). Psychoacoustic and phonetic temporal processing in normal and hearing impaired listeners. *Journal of Acoustical Society of America,* 72, 740-752.

Turner, C. W., Souza, P. E., & Forget, L. N. (1995). Use of temporal envelope cues in speech recognition by normal and hearing-impaired listeners. *Journal of Acoustical Society of America,* 97, 2568-2576.

Tassel, D. J. V., & Trine, T. D. (1996). Effect of single band amplitude compression on temporal speech information in nonsense syllables and in sentences. *Journal of Speech and Hearing Research,* 39, 912-922.

Villchur, E. (1973). Signal processing to improve speech intelligibility in perceptive deafness. *Journal of Acoustical Society of America, 53,* 1646-1657.

Van Tasell, D. J., Soli, S. D., Kirby, V. M., & Widin, G. P. (1987). Speech waveform envelope cues for consonant recognition. *Journal of Acoustical Society of America,* 32, 1152-1160.

Villchur, E. (1989). The negative effects of amplitude compression in multichannel hearing aids in the light of the modulation transfer function. *Journal of Acoustical Society of America,* 86, 425-427.

Van Tasell, D. J., Greenfield, D. G., Logemann, J. J., & Nelson D. A. (1992).Temporal cues for consonant recognition: Training, talker generalization, and use in evaluation of cochlear implants. *Journal of Acoustical Society of America,* 92, 1247-1257.

Van Tasell, DJ. (1993). Hearing loss, speech, and hearing aids. *Journal of Speech Hearing Research,* 36, 228-244.

Verschuure, H., & Dreschler, W. A. (1993). Syllabic compression and speech intelligibility in hearing impaired listeners. *Scandinavian Audiology Supplement, 22,* 92-100.

Verschuure, H., Prinsen, T. T., & Dreschler, W. A. (1994). The effects of syllabic compression and frequency shaping on speech intelligibility in hearing impaired people. *Ear and Hearing,* 15, 13-21.

Verschuure, J., & Goedegebure, A. (1996). Compression and its effect on the speech signal. *Ear and Hearing,* 17, 162-175.

Walkner, G., *&* Dillon, H. (1982). Compression in hearing aids: an analysis, a review and some recommendations (NAL Report No. 90). Australian Government Publishing Service.

Yanick, P., & Drucker, D. (1976). Signal processing to improve intelligibility in the presence of noise for person with ski-slope hearing impairment. *IEEE Transactions of Acoustics, Speech and Signal Processing*, 24, 507-512.

Yares, G. K. (1990). Basilar membrane nonlinearity and its influence on auditory nerve rate intensity function. *Hearing research*, 50, 145-162.