THE COMBINED EFFECT OF COMPRESSION AND DIGITAL NOISE REDUCTION ALGORITHMS ON SPEECH PERCEPTION AND SPEECH QUALITY

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MAY, 2013.

CERTIFICATE

This is to certify that this dissertation entitled '***The combined effect of compression and digital noise reduction algorithms on Speech perception and speech quality***' is a bonafide work submitted in part fulfilment for the degree of Master of Science (Audiology) of the student Registration No: 11AUD003. This has been carried out the under guidance of a faculty of this institute and has not been submitted earlier to any other university for the award of any diploma or degree.

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DECLARATION

This is to certify that this master's dissertation entitled '***The combined effect of compression and digital noise reduction algorithms on speech perception and speech quality'*** is the result of my own study under the guidance of Ms. Geetha. C, Lecturer in Audiology, Department of Audiology, All India Institute of Speech and Hearing, Mysore, and has not been submitted earlier to any other university for the award of any diploma or degree.

Mysore

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Dedicated to...

The Almighty

&

My Family

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TABLE OF CONTENTS

Chapter	Title	Page No.
1	Introduction	1-6
2	Review of Literature	7-15
3	Method	16-23
4	Results and Discussion	24-31
5	Summary and Conclusion	35-37
	References	38-43

LIST OF TABLES

Table	Title	Page No.
4.1	Mean and SD at three different levels for all the four conditions	25
4.2	Results of Repeated measure ANOVA at 40 dB SPL	26
4.3	Results of Repeated measure ANOVA at 65 dB SPL	28
4.4	Results of Repeated measure ANOVA at 85 dB SPL	30
4.5	Mean and SD for the six parameters of quality for three different conditions	32
4.6	Results of the Fried man test	33
4.7	Results of Wilcoxon Signed Ranks Test	34

CHAPTER 1

Introduction

Individuals with sensorineural hearing loss have been found to face many problems in their daily life, with reference to communication. The dynamic range, that is, range of levels between the threshold and the loudest sound that can be tolerated is less for a person with hearing impairment. They are also found to have reduced frequency resolution and reduced temporal resolution apart from the reduced dynamic range (Moore, 2008).

The reduced frequency resolution and temporal resolution results in poor perception, especially, in the presence of noise. Hence, Individuals with sensorineural hearing impairment require a greater SNR than those of normal hearing subjects (Nordrum, Erler, Garstecki, & Dhar, 2006). In order to overcome these difficulties, most modern digital hearing aids have been implemented with many signal processing strategies. The most commonly implemented strategies are wide dynamic range compression (WDRC) algorithm, noise reduction algorithms and directional microphone technology.

WDRC algorithms in hearing aids are implemented in order to allow the user to hear soft and loud sounds without the need to adjust the volume control (Kuk, 2002; Cox, 1999; Souza, 2002). The compression parameters can be varied to achieve maximum comfort, to maximize the intelligibility, and also to reduce noise (Dillon, 2001).

Depending on the setting of the time constants and other static parameters of compression, the spectrum of speech signal gets altered (Levitt, 2007). Moore (2008) has reported that the WDRC could provide audibility for soft sounds and solve issues of comfort, distortion and over amplification of loud sounds.

Boike and Souza (2002) assessed speech intelligibility and speech quality ratings using single channel WDRC hearing aid for different values of compression ratios. They found that quality ratings and speech intelligibility were better in linear mode of processing, and poorer with increasing compression ratios. One effect of compression is to increase noise during the gaps in the speech signal (Dillon, 1996; Moore, Peters, & Stone, 1999) at higher compression ratios. It is possible that this caused a greater masking effect for the listeners with hearing loss.

Alternatively, alteration of temporal cues at higher compression ratios may have a relatively greater impact on listeners with hearing loss and, presumably, poorer spectral discrimination ability than listeners with normal hearing. Further, higher speech intelligibility scores were found to be correlated with high quality ratings (Souza & Turner, 1998).

The results of the study carried out by Souza (2002) suggested that compression can provide speech audibility over a wide range of input levels, resulting in improved speech intelligibility, quality, and loudness comfort. However, when the hearing aid has more number of compression channels and the compression ratio is set high, it could cause reduction in speech intelligibility or speech quality. The compression ratios of WDRC are typically kept at or below 3:1 because higher compression ratios along with fast time constants are reported to degrade speech intelligibility (Souza, 2002). However, even if the parameters are optimized to minimize the negative consequences, there will still be changes in the acoustic spectrum of speech signal.

Hickson and Thyer (2003) evaluated 74 adults with mild to moderate sensorineural hearing impairment and carried out acoustical and perceptual measures of compression amplification. They concluded that compression tends to bring about significant changes in the output. It can be stated that when high compression ratio and high crossover frequency was used in the high frequency channel, it had an adverse effect on consonant perception for older people with sensorineural hearing impairment. Hence, it is clear that the compression amplification does significantly alter the speech signal.

Digital Noise Reduction algorithm (DNR) is yet another algorithm implemented to improve the acceptance of hearing aid amplification in noisy environment which is the major difficulty faced by the individuals with sensorineural hearing impairment. Noise reduction algorithms are implemented in almost all digital non-linear products. DNR algorithms are designed to take advantage of temporal separation and spectral differences between speech and noise. These basically detect modulations in the incoming signal to infer the presence or absence of speech.

Mueller, Weber and Hornsby (2006) had fitted adult hearing impaired listeners with a 16 channel digital hearing aid to assess the digital noise reduction processing. Speech intelligibility and acceptable noise level were measured when DNR was ‘off’ and when DNR was ‘on’. They found a significant improvement in acceptable noise level when the DNR was activated and no significant improvement of Hearing in noise test in either of the two conditions.

In a study carried out by DiGiovanni, Davlin and Nagaraj (2011), speech intelligibility was measured for sentences in quiet, transient noise, multi talker babble and a combination of noise on 17 hearing impaired individuals. Subjective ratings of overall speech understanding, comfort, and sound quality were obtained for transient noise reduction algorithm (TNR) activated and deactivated mode. A significant speech intelligibility improvement was found during TNR activated condition for the speech babble and for transient noise. However, no improvement in speech intelligibility was found when a combination of noise (making the noise spectrum complex) was given and the TNR activation did not see any significant improvement in any of the subjective rating.

Further, the amount of noise reduction is also found to depend on th number of noise reduction channels. Rout, Hanline and Halling (2007) evaluated multichannel noise reduction by digital behind the ear hearing instruments with multichannel noise reduction from four major manufacturers. Hearing aids with 16 channel DNR resulted in the maximum gain reduction across all frequencies. While 14 channel DNR provided greater noise reduction at higher frequencies (>2000 Hz), 16 channel DNR provided greatest noise reduction at lower frequencies (<2000 Hz). Of the tested algorithms, the noise reduction from manufacturer C (8 channel DNR) was comparatively least effective.

1.1 Need for the study

Many research studies have been carried out to evaluate the effect of different compression settings and different digital noise reduction algorithms (Hickson & Thyer, 2003; Mueller, Weber, & Hornsby, 2006; Souza, 2002). The results depend on the settings of the hearing aid and the noise conditions.

Compression amplification has been reported to alter the acoustic signal in comparison to linear amplification in a significant way (Hickson & Thyer, 2003). Dillon (2001) reported that it can also show some adverse effect such as decreased SNR for noises occurring within the gaps of speech, increased chance of feedback, disrupted intensity cues (when compression is fast acting) and attenuation of signal as well as noise.

Studies have shown that DNR algorithms may be effective in improving speech perception in noise when the speech and noise sources are not spatially separated or when the noise spectrum is different that of speech (Bray, Sandridge, Newman, & Kornhass, 2002). Bray and Nilsson (2001) reported that the algorithms with largest amount of noise reduction are showing the strongest speech distortion in unfavorable noisy situations.

Hence, it is evident from the above that the digital signal processing strategies such as WDRC and noise reduction affect the amount of amplification that has been done and as a result attenuate the individual signal components. However, the available reports evaluate only the independent effects of these algorithms. In reality, these algorithms work together either sequentially or parallel to each other (Schaub, 2004). In either case, the speech signal gets altered twice. Hence, evaluating overall effects would provide a complete view of the resultant output and their effect on perception and quality.

Chung (2006) has evaluated a WDRC hearing aid along with DNR in order to see the amount of noise reduction so that it could be implemented in hearing protectors. He has found that speech intelligibility and quality measures did show a difference when both the algorithms were ‘on’. However, the participants in his study were individuals with normal hearing sensitivity. The results of the same cannot be completely generalized to individuals with hearing impairment who have audibility and discrimination issues.

1.2 Aim of the study

Hence, the present study aims to evaluate the combined effect of compression and DNR algorithms on speech intelligibility and on self-rated quality measures.

1.3 Objectives of the study

The objectives of the study are:

- To evaluate the speech identification scores in the presence of noise, at 40, 65 and at 85 dB SPL, in the following conditions:

    1. DNR only
    2. Compression only
    3. Combined-Compression and DNR
    4. Unaided condition

- To evaluate the self-rated quality of speech in the above mentioned conditions.

CHAPTER 2

Review of literature

Individuals with sensorineural hearing loss experience abnormal perception of loudness i.e., when intensity increased just above the threshold of hearing, it can be unbearably loud for them. However, the very-low-intensity sounds cannot be heard (Nikoleta, 1990).

Everyday speech includes a wide range of intensity levels, that is, from low intensity consonants such as /f/ to high intensity vowels such as /i/, and from whisper to shouting. In these individuals with hearing impairment, the benefit of a linear hearing aid is restricted. As the linear hearing aid amplifies the low intensity sounds to make it audible, the high-intensity sounds also gets amplified to the point of discomfort. The smaller the dynamic range of the listener, the most difficult it is to make the speech audible in variety of situations (Nikoleta, 1990).

Individuals with sensorineural hearing impaired also have a great difficulty in understanding speech especially in the presence of noise. They require a greater SNR when compared to individuals with normal hearing sensitivity. This is because of broadening of auditory filters (Moore, 2008).

Presenting speech to hearing impaired today in an intelligible and comfortable form remains a major challenge in hearing aid research today. More advanced forms of digital processing have been incorporated using digital techniques in order to achieve the goals of comfort and intelligibility. These include compression algorithm, directionality and noise reduction algorithms among others (Dillon, 2001; Souza, 2002).

Each of these algorithms affects the speech in a different way. In the sections that follow, the effect of WDRC, digital noise reduction algorithm and directionality on the speech perception and quality are discussed.

3.1 Compression algorithm

Compressor's major role is to decrease the range of sound levels in the environment so as to better match the dynamic range of a person with hearing impairment. The compressor may be most active at low, mid or high sound levels or it may vary its gain across a wide range of sound levels, in which case it is known as wide dynamic range compressor (WDRC).

3.1.1 Effect of compression on speech perception and speech quality

Many studies have compared the speech intelligibility and quality of speech from linear and compression amplification in a controlled environment, using both simulated hearing aid responses or wearable hearing aids. A study done by Jenstad and Seewald (1999) incorporated five conditions representing different speech spectra at different levels and frequency responses. For both linear and WDRC conditions the same hearing aid was used, with targets generated using the same prescriptive procedure (DSL[i/o]). Sentence and nonsense syllable intelligibility, and speech loudness ratings were measured. For average speech levels, both circuits provided same loudness comfort and speech intelligibility. For high and low speech levels, the WDRC aid provided better intelligibility and loudness comfort.

Humes, et al., (1999) did a study by fitting hearing aids with linear peak clipping for 55 hearing-impaired adults (fit according to linear, NAL-R targets) and two-channel WDRC aids (according to nonlinear, DSL [i/o] targets). All subjects wore the linear aids for two months, after that by the WDRC aids for two months. At the end of every two month trial period, a set of outcome measures were done, which included word recognition in quiet and in noise at various presentation levels, judgments of sound quality and subjective ratings of hearing aid benefit. The results, showed better speech intelligibility using WDRC aid at all but high-level inputs. WDRC hearing aids give better ease of listening for low level speech in quiet as per the subject's report. They said that, these results to the greater gain at low input levels provided by the WDRC circuit and the higher DSL target gain levels for the WDRC aid. Hence, WDRC is found to be more beneficial over linear amplification. However, the WDRC by itself can affect speech differently depending on the compression parameters.

The effect of hearing aid compression parameters, such as compression time constants and compression ratio, on speech perception and the subjectively perceived sound quality of hearing aids has been investigated by several authors (Gatehouse, Naylor & Elberling, 2006; Hansen, 2002; Moore, Stainsby, Alcántara, & Kühnel, 2004; Neuman, Bakke, Mackersie, Hellman, & Levitt, 1998).

Hansen (2002) investigated subjectively perceived sound quality as a function of compression time constants and compression threshold using a 15-channel compression hearing aid. He found a strong preference for long time constants and low compression thresholds. Keidser, Dillon, Dyrlund, Carter and Hartley (2007) investigated the preferred compression ratio in a fast-acting multi-channel device by listeners with severe hearing loss. The listeners generally preferred lower compression ratios than are typically prescribed for that degree of hearing loss.

Souza in 2002 compared the outputs of unprocessed sentences to wide-dynamic range compression amplified speech with syllabic compression. In the unprocessed version, she found a significant difference between low-intensity (typically consonants) and high-intensity (typically vowels) inputs. Whereas, when the same sentence were processed by the syllabic compression circuit using a compression threshold at 45 dB SPL with an attack and release time of 3 milliseconds and 50 milliseconds respectively. The clearest effect was the decreased amplitude variation. Further, high intensity phonemes were reduced in level relative to low-intensity phonemes, which resulted in an overall smoothing of amplitude variations. This is also called amplitude smearing.

Hence, compression, though has been found to be beneficial, introduces many changes in the spectrum of the signal. However, the extent of change depends the compression ratio and the time constants.

3.2 Digital Noise Reduction Algorithm

Digital Noise Reduction (DNR) algorithms are designed to take the advantage of temporal separation and spectral differences between speech and noise. These basically detect modulations in the incoming signal to infer the presence or absence of speech. In the section below, effect of noise reduction on the speech perception and quality is discussed.

3.2.1 Effect of noise reduction on speech perception and speech quality

The introduction of digital signal processing offered a way to incorporate complex algorithms in real time. DNR basically detects modulations in the incoming signal to deduce the presence or absence of speech. Although different forms of DNR have been accessible in hearing aids for many years (Graupe, Grosspietsch, & Basseas, 1987), early improvements for speech perception in noise typically were less (Tyler & Kuk, 1989).

When spectral and temporal characteristics of the noise are expected and can be clearly characterized, DNR can be helpful for increasing the SNR. In real life, the characteristics of both the signal and noise are unknown and time varying. Hence, better speech perception through noise reduction is more difficult (Levitt, 2001).

DNR algorithms use spectral subtraction or an assessment of SNR in each band followed by gain reduction. In the spectral subtraction, the noise spectrum is captured during pauses between words and is then subtracted from the noisy speech spectrum in either the time domain or frequency domain. In the latter method of DNR, the stationary signals are interpreted as noise and modulated signals are taken as speech. After that the gain reductions are applied according to pre-determined tolerable SNR values in each band. In common, the ability of these systems to detect the presence of noise is better. To separate speech from noise without changing the speech is much harder.

Studies with adults have shown an increase in speech perception when the noise is restricted to a narrow frequency region (Van Dijkhuizen, Festen, & Plomp, 1991). When the long-term spectra of the target signal and noise are the same, there are several studies that have failed to show improvement in speech perception (Alcantara, Moore, Kuhnel, & Launer, 2003; Boymans & Dreschler, 2000; Levitt, Bakke, & Kates, 1993).

Most of the studies on noise reduction algorithms did not report any benefit for speech understanding in broad band noises, such as car noise or speech spectrum noise (Alcantara et al., 2003). The reason for this is that if the noise reduction algorithm reduces the gain at frequency channels with noise dominance, it also reduces the audibility of speech information in the channel.

Rout, Hanline and Halling (2007) have done a study on evaluation of multichannel noise reduction on a new stimuli. Three different bandwidths of steady-state noise (1/3 octave, 1 octave, 2 octave) were embedded at six different frequencies (0.25, 0.5, 1, 2, 3, and 4 kHz) resulting in 18 new stimuli. In addition, one speech shaped noise, and one ICRA noise at 0 dB SNR were included in the pool of stimuli creating a total of 20 test stimuli. Digital behind the ear hearing instruments with multichannel noise reduction from four major manufacturers were used in this study. Results showed that the proposed stimuli can adequately assess the degree of noise reduction at different frequencies in multichannel noise reduction hearing aids. Hearing aids from the manufacturer D (16 channel DNR) resulted in the maximum gain reduction across all frequencies. While manufacturer A (14 channel DNR) provided greater noise reduction at higher frequencies (>2000 Hz) and manufacturer B (16 channel DNR) provided greatest noise reduction at lower frequencies (<2000 Hz). Of the tested algorithms, the noise reduction from manufacturer C (8 channel DNR) was comparatively least effective when tested with the proposed stimuli.

From the above, it is clear that DNR does provide some amount of benefit. However, it depends on many factors such as the number of channels, spectrum of speech and noise.

3.3 Directional Microphones

Directional microphone technology is considered to take advantage of the spatial separation between speech and noise. Directional microphones are more responsive to sounds coming from the front than sounds coming from the back and the sides. The assumption is that when the hearing aid user connects in conversation, the talker is usually in front and the sounds from other directions are undesirable (Dillon, 2001).

3.3.1 Effect of Directionality on speech perception and speech quality

Directional microphones are one of the few methods for increasing SNR within the same frequency band in hearing aids and to enhancing speech intelligibility in noise. Studies show a significant improvement in speech intelligibility in noise with the use of directional microphones (Luts, Jean & Wouters, 2004; Valente, Fabry, & Potts, 1995)

Luts, Jean and Wouters (2004) aimed to evaluate the improvement in speech intelligibility in noise obtained with an assistive real time fixed end fire array of bidirectional microphone in comparison with an omni directional hearing aid microphone in a realistic environment. The microphone array was evaluated physically in anechoic and reverberant conditions. Ten individuals with normal hearing sensitivity and ten hearing aid users participated in the study. Results showed that improvement in speech intelligibility in noise obtained with array relative to omni directional microphone depends on noise situation in a subject group.

A study done by Rickettes and Dhar (1999) compared the speech recognition performance of 12 individuals with hearing impairment listeners with both directional and omni directional modes. Testing was done in both real life and anechoic room environments. Speech recognition was found out using modified forms of the Hearing in Noise Test and the Nonsense Syllable Test. Results showed significant speech recognition in noise advantage for all directional hearing aids in comparison to their omni directional counterparts. In addition, the results revealed that performance in one reverberant condition cannot be used to correctly expect performance in an environment with different reverberation.

To conclude from the above studies, there is improvement in scores for the directional microphones comparing to omni directional microphones, for speech recognition in noise.

3.4 Combined effect of WDRC and DNR algorithms

Many research studies have been carried out to evaluate the effect of different compression settings and different digital noise reduction algorithms (Hickson & Thyer, 2003; Mueller, Weber & Hornsby, 2006; Souza, 2002). The results of those studies depended on the settings of the hearing aid and the noise conditions. However, all these studies have investigated either WDRC or DNR.

Chung (2012) studied the wind noise levels between DNR-activated and DNR-deactivated conditions for linear and WDRC (with compression ratio of 3:1) conditions. He also compared the wind noise levels between directional and omni directional modes. The results showed that the WDRC increased low-level noise and decreased the gain at high-level noise and also showed that for the different microphone modes the different noise reduction algorithms provided different amounts of wind noise reduction. He concluded that the wind noise can be minimized by decreasing the gain for low-level inputs, increasing the compression ratio for high-level inputs, and activating modulation-based noise reduction algorithms.

Similarly, Keidser et al. (2006) studied the effect of these algorithms on localization of hearing aid users. Chung (2006) investigated the effects of compression and noise reduction configurations on the amount of noise reduction, speech intelligibility, and overall preferences using digital hearing aids. Sentences mixed with speech spectrum noise and white noise were processed by eight digital hearing aids.

When the hearing aids were set to 3:1 compression, the amount of noise reduction achieved was enhanced or maintained for hearing aids with parallel configurations, but reduced for hearing aids with serial configurations. In the experiments 2 and 3, speech intelligibility and perceived sound quality were tested on individuals with normal hearing sensitivity. Regardless of the configuration, the noise reduction algorithms reduced the noise level and maintained speech intelligibility in white noise. Additionally, the listeners preferred the parallel rather than the serial configuration in 3:1 conditions and the serial configuration in linear rather than compression when the noise reduction algorithms were activated.

Hence, it is evident from the above that the digital signal processing strategies such as WDRC and noise reduction affect the amount of amplification and as a result attenuate the individual signal components. However, the Chung's study was done on individuals with normal hearing sensitivity. The focus of this study was to investigate the effect of compression and noise reduction on speech intelligibility and subjective preferences on individuals with hearing impairment as the perception is very different in the hearing impaired ears and hence, the results obtained from individuals with normal hearing sensitivity may not old good on individuals with hearing impairment. Further, directionality was not included in the present study considering the time taken for assessing each individual.

CHAPTER 3

Method

The present study aims to evaluate the effects of WDRC and digital noise reduction algorithms on speech intelligibility and self-rated speech quality measures. The method consisted of the following steps:

3.1 Selection of participants

3.2 Experiment to assess the effect of compression and DNR on SIS

3.3 Experiment to assess the effect of compression and DNR on Quality rating

3.1 Selection of participants

The present study included 20 participants in age range between 18-55 years(with the mean age of 43 years. Routine audiological evaluation was carried out including pure tone audiometry, speech audiometry and Immittance evaluation along with a detailed case history to select the participants for the current study. The following inclusion and exclusion criteria were used:

3.1.1 Inclusion criteria

- Post-lingual bilateral mild to moderately severe sensorineural hearing loss, with flat audiogram configuration showing relatively little change in hearing loss (with in 10 dB rise or fall over the range) from 500 to 5000 Hz (Kennedy, Levitt, Neuman, & Weiss, 1998),

- Identification score not less than 70%.
- The difference in PTA between ears less than or equal to 15 dB (Gatehouse, Naylor, & Elberling, 2006).
- Naive hearing aid users,
- Fluent speakers of Kannada language, and
- A or As type of tympanogram.

3.1.2 Exclusion criteria

Participants who are presented with one or more of the following were excluded from the study.

- Any history or presence of middle ear disorders,
- Any history or presence of neurological involvement, and
- Any history or presence of psychological problems.

3.1.3 Instruments

- A calibrated OB-922 dual channel diagnostic audiometer was used for obtaining the pure tone thresholds, SRT and SIS. The audiometers were connected to the TDH 39 head phones housed in MX-41 AR cushion, Radio Ear B-71 Bone vibrator and two loud speakers located at 45° azimuth at 1 meter distance.
- Tympanometry and Acoustic reflex assessment were carried out with GSI-Tympstar middle ear analyzer.

- Two digital behind-the- ear hearing aids of same model with fitting range of mild to moderately severe hearing loss were used. These hearing aids had four channels with a facility to turn ‘on’ and ‘off’ the compression and DNR algorithms.
- A personal computer with NOAH-3 software connected with Hi-PRO, appropriate programming cable and hearing aid specific program were used to program the hearing aid.
- A personal computer connected to the auxiliary input to the OB 922 audiometer was used to present the stimuli for testing.

3.1.4 Test Environment

Air conditioned sound treated double room set-up was used to administer all the tests. The noise level was within the permissible limits.

3.1.5 Stimuli

- Paired words in Kannada language developed at the Department of Audiology, All India Institute of Mysore, were used for establishing the SRT.
- Recorded version of phonetically balanced word lists in Kannada language developed by Yathiraj and Vijayalakshmi (2006) were used to obtain the speech identification scores during routine evaluation.
- List of words in Kannada language developed by Manjula, Geetha, Sharath and Antony (2013) were used to obtain the speech identification scores in the actual

experiment. It consists of 21phonemically balanced lists with each list containing 25 words.

- A paragraph in Kannada language developed by Sairam (2003) containing all the speech sounds of Kannada was used for quality rating.
- The quality rating scale developed by Eisenberg and Dirks (1995) was adapted and modified for this study. Six parameters of quality were rated by the listeners using a five point rating scale. The parameters included loudness, clearness, sharpness, fullness, naturalness and the overall impression. The five point rating scale is as follows:

  0 = Very poor

  1 = Poor

  2 = Fair

  3 = Good

  4 = Excellent

3.1.6 Procedure for Routine audiological evaluation

Pure tone thresholds were obtained using the calibrated OB-922 dual channel diagnostic audiometer using modified Hughson and Westlake procedure (Carhart & Jerger, 1959). This was done across frequencies from 250 Hz to 8000 Hz for obtaining air conduction thresholds and for 250 Hz to 4000 Hz for bone conduction. Pure Tone Average (PTA) was taken as an average for the air conduction thresholds for the frequencies 500 Hz, 1 kHz and 2 kHz.

Speech recognition scores were obtained to correlate with PTA using Kannada paired words. Speech Identification Scores (SIS) were obtained at 40 dB SL (re: SRT) using the PB word lists in Kannada language developed by Yathiraj and Vijayalakshmi (2006). UCL for speech were also measured.

Immittance Evaluation was done on all the individuals. Tympanometry and Acoustic reflex assessment were carried out using standard procedures with GSI-Tympstar middle ear analyzer. Based on the results of the above tests, those participants satisfying the selection criteria were selected for further evaluations.

3.1 Experiment to assess the effect of compression and DNR on SIS

The actual experiment included hearing aid programming, routine hearing aid evaluation and then the assessment of SIS in different experimental conditions.

3.2.1 Hearing aid programming

The participants were seated comfortably in a chair and fitted with digital BTE hearing aids, which has the features given in the section 3.1.3. The hearing aids were connected to a Hi-PRO using appropriate cables. Hi-PRO was in turn connected to a personal computer with NOAH software and hearing aid specific programming software. Audiometric threshold data for each participant's ear were fed in NOAH-3 software. The following settings were used while fitting the hearing aid.

- Acclimatization level was set to 2

- Feedback cancellation was ‘off’
- Volume control was ‘off’
- Directional microphone was ‘off’
- Compression settings were set by the soft ware
- Fixed attack and release time

The hearing aids were programmed based on NAL-NL1 prescriptive formula. The initial fit was done using first fit feature of the hearing aid programming software.

3.2.2 Routine hearing aid testing

With the first fit settings, the participant was asked to identify the ling’s six sounds. The gain settings were modified till they were able to identify all the sounds. Following this, routine hearing aid evaluation was carried out by asking five unrelated open ended questions and finding out SIS for words at 40 dB HL through the audiometer. This was done monaurally. Following this, binaural testing was carried out in the same way and binaural balancing was done by asking the patient to balance the loudness between the two ears. The gain settings for a given case were constant across the testing conditions.

3.2.3 Procedure for the experiment to assess SIS in different conditions

The actual experiment was carried out by obtaining SIS for recorded words developed by Manjula, Geetha, Sharath and Antony (2013). The stimuli were presented at three different levels 40 dB SPL, 65 dB SPL, and at 85 dB SPL or just below UCL (whichever was higher) through the loud speaker placed at 45° azimuth in the presence of speech spectrum noise. The noise was presented at +5 dB SNR. A different list was presented in each condition to avoid practice effect and the order of the conditions was also different for each individual to avoid the order effect.

The listeners were instructed to repeat the words. Recorded words were played in the following conditions:

1. Unaided condition
2. Only Noise reduction- In this condition, compression was turned ‘off’ and DNR option has turned ‘on’
3. Only Compression- In this condition, compression was turned ‘on’ and DNR was turned ‘off’.
4. Combined - In this condition, both compression and DNR were turned ‘on’.

For the compression ‘on’ conditions, the settings of compression parameters were as per the software prescription and they remained same across input levels and conditions.

The tester had noted down the responses in a response sheet. The total number of words repeated correctly for each list was calculated for all the above mentioned conditions at three different levels.

3.3 Step 3: Experiment to assess the effect of compression and DNR on quality rating

The participants were asked to quantify the sound quality in three aided conditions using the five point rating scale. For this, the recorded paragraph in Kannada language developed by

Sairam (2003) was presented at most comfortable levels through the loudspeaker placed at 45° azimuth at 1 meter distance. The participants were asked to rate six parameters of quality on a five point rating scale. The participants were explained about the six parameters of quality and practice trials were given before the actual testing. The parameters included,

1) Loudness: The story given is sufficiently loud, in contrast to soft or faint.
2) Clearness: The story is clear and distinct in contrast to blurred and diffuse.
3) Sharpness: The story is able to hear with respect to its unevenness.
4) Fullness: The story is full in contrast to thin.
5) Naturalness: The story seems to be as if there is no hearing aid, and the story sounds similar to original.
6) Overall impression: The reproduction of sound with little distortion giving result very be similar to original.

3.4 Statistical analysis

The data of the present study was tabulated and statistically analyzed using the Statistical Package for Social Sciences (SPSS, version 16.0) software. The data was subjected to repeated measures ANOVA and Fried man test.

CHAPTER 4

Results and Discussion

The present study aims to evaluate the combined effect of WDRC and digital noise reduction algorithm on speech intelligibility and on self-rated quality measures. The objectives of the study was to evaluate the speech identification scores in the presence of noise, at 40 dB SPL, 65 dB SPL and at 85 dB SPL, and to evaluate the self-rated speech quality, in the below mentioned conditions:

1) Unaided (UA)
2) Digital Noise reduction only (NO)
3) Compression only (CO)
4) Combined-Compression and Digital noise reduction (BO)

The data of the present study was tabulated and statistically analysed using the Statistical Package for Social Sciences (SPSS, version 16.0) software. The data was subjected to repeated measures ANOVA and Fried man test.

4.1 Experiment to assess the effect of compression and DNR on SIS

The Table 4.1 shows the mean and the standard deviation (SD) of the number of correctly identified words in all the conditions across different levels of presentation.

It can be observed from the Table 4.1 that the number of correctly repeated words varies between the unaided and aided conditions. It is also evident that the scores are better for aided conditions than that of unaided conditions. Among the aided conditions, the mean scores for the compression only condition was found to be greater than other two conditions. Further, the scores also vary depending on the level of presentation.

Table 4.1

Mean and SD at three different levels for all the four conditions (N=20)

Condition	Level	Mean	SD
Unaided	40 dB SPL	0.00	0.00
	65 dB SPL	6.50	2.02
	85 dB SPL	8.68	2.69
DNR Only	40 dB SPL	4.23	2.38
	65 dB SPL	19.09	3.77
	85 dB SPL	22.32	3.16
Compression only	40 dB SPL	5.18	2.30
	65 dB SPL	22.23	3.01
	85 dB SPL	23.73	1.96
Both On	40 dB SPL	5.45	2.92
	65 dB SPL	21.45	3.36
	85 dB SPL	22.45	2.84

Statistical analysis was done for each level of presentation separately in order to see if there is any difference between the scores across conditions, and the results of the same are given below.

4.1.1 At 40 dB SPL

Repeated measures ANOVA was done to compare the SIS between four conditions. The results revealed a significant difference across conditions [$F(3,57) = 37.253$, $p<0.05$]. In order to see which of the conditions differed from each other, Bonferroni pairwise-comparison was done.

Table 4.2

Results of Repeated measures ANOVA at 40 dB SPL

Condition(I)	Condition (J)	Mean Difference (I-J)
UA	NO	-4.05*
	CO	-5.25*
	BO	-5.35*
NO	UN	4.05*
	CO	-1.20
	BO	-1.30
CO	UN	5.25*
	NO	1.20
	BO	-0.10
BO	UN	5.35*
	NO	1.30
	CO	0.10

UN= Unaided; NO= DNR only; CO=Compression only; BO= Both DNR and compression 'on'; Note:* p<0.05.

From the Table 4.2, it is evident that there is a significant difference between the unaided and the three aided conditions. The scores were found to be better for the aided conditions compared to that of unaided conditions. However, it can be observed that SIS in the aided conditions has not improved more than 25% when the signal is presented at 40 dB SPL. The reason for this could be that the routine hearing aid verification of gain settings (before the actual setting) was done using functional gain measurement at 40 dB HL. This represents normal conversational level. Verification of gain for soft speech was not done during the routing hearing aid verification. Probably, the participants required higher gain than that was set for softer sounds using NAL NL1 formula.

It has been reported by Kuk and Ludvigsen (2003) that the aided sound-field thresholds provide information for low-level signals. Hence, sound field threshold measurement or insertion gain measurement could have given better information on the gain settings for softer sounds. Further, the testing was done in the presence of noise which might have made the perception more difficult.

Further, in the present study, there is no significant difference between any of the aided conditions. That is, DNR and compression algorithms independently and when combined have yielded similar scores. In the present study, the compression threshold, for all the participants was more than 40 dB SPL. Hence, the hearing aid's compressor circuit would not have been activated most of the time at 40 dB SPL level of presentation. Hence, the advantage of WDRC was not felt when the signal was presented at 40 dB SPL. In addition, the audibility was not adequate which could have resulted in poor improvement in aided condition and uniform performance across different aided conditions.

4.1.2At 65dB SPL

Repeated measures ANOVA was done to compare the SIS across the conditions at 65 dB SPL and the results revealed a significant difference [$F(3,57) = 207.21, p < 0.05$]. In order to see which of the conditions differed from each other, Bonferroni pair-wise comparison was done.

Table 4.3

Results of Repeated measures ANOVA at 65 dB SPL

Condition (I)	Condition (J)	Mean Difference (I-J)
UN	NO	-11.90*
	CO	-15.20*
	BO	-14.55*
NO	UN	11.90*
	CO	-3.30*
	BO	-2.65*
CO	UN	15.20*
	NO	3.30*
	BO	.65
BO	UN	14.55*
	NO	2.65*
	CO	-.65

UN= Unaided; NO= DNR only; CO=Compression only; BO= Both DNR and compression On; Note:* $p<0.05$.

From the Table 4.3, it can be observed that there is a significant difference between the unaided and the three aided conditions. That is, the hearing aid improved the audibility and thus the SIS. Further, the noise reduction only condition significantly differed from the compression only and combined conditions, i.e., when only the noise reduction was 'on' the performance was significantly poorer when compared to the compression only condition and when both the algorithms were 'on'. Hence, it can be said that WDRC has provided substantial improvement in speech perception. These results are in concordance with the other available reports. Souza (2002) has reported that if the compression ratios were below 3:1 and slow acting, WDRC provided good benefit. In the present study, the compression ratios in all the channels were within 3:1 and the digital hearing aid had a fixed slow release time. This advantage was not present during the noise reduction only condition as the compression was not 'on' due to which the performance could have been poorer in NO condition. Further, scores in the CO and BO conditions did not differ significantly.

From the above results it can be deduced that compression has provided better benefit in the word recognition task in the presence of noise than the noise reduction only condition. Further, when both the algorithms are presented, the scores do not deteriorate significantly. This could be because of the type of noise that was used and the compression settings of the hearing aid in the present study. That is, speech noise was used which does not have too many fluctuations, and lower compression ratios, slower time constants and only four channels of processing were used.

4.1.3 At 85dBSPL

Results of repeated measures ANOVA at 85 dB SPL revealed that there was a significant difference across conditions [$F(3,57)=257.22, p<0.05$]. Bonferroni pair-wise comparison revealed that even at 85 dB SPL the unaided scores were significantly poorer than the aided scores. This is evident in the Table 4.4.

Table 4.4

Results of Repeated measures ANOVA at 85 dB SPL

Condition (I)	Condition (J)	Mean Difference (I-J)
UN	NO	-13.05*
	CO	-14.75*
	BO	-13.30*
NO	UN	13.05*
	CO	-1.70*
	BO	-.25
CO	UN	14.75*
	NO	1.70*
	BO	1.45*
BO	UN	13.30*
	NO	0.25
	CO	-1.45*

UN= Unaided; NO= DNR only; CO=Compression only; BO= Both DNR and compression ‘on’; Note:* $p<0.05$.

Further, the noise reduction only condition significantly differed from the compression only condition. That is, when only the noise reduction was ‘on’ the performance was significantly poorer when compared to the compression only condition. In addition, compression only condition yielded significantly better performance than the performance on the condition when both the algorithms were ‘on’. This indicates that when both the algorithms activated, the performance does deteriorate when compared to when only compression is activated. However, DNR alone gave the poorest performance.

Hence, the WDRC seems to have offset the negative effects of DNR in the combined condition.

This is not in agreement with the results found by Chung (2006). He found that the introduction of noise reduction offset the negative performance of the compression algorithm. It has been reported that when the WDRC system has compression ratio less than 3:1, release time longer than 100 msec and compression channels equal to or less than four channels, the compression does not deteriorate the SIS (Boike and Souza, 2000; Hansen, 2002; Humes et al., 1999; Souza & Turner, 1998). In the present study, the hearing aid had four channels with compression ratio less than 3:1 and the release time was around 800 msec.

4.2 Experiment to assess the effect of Compression & DNR on Quality rating

The six parameters of quality were rated on a five point scale. The ratings by all the twenty listeners were tabulated in SPSS. The Mean and SD for loudness, clearness, sharpness, fullness, naturalness and overall impression in three different conditions (DNR only, Compression only, Both DNR and Compression ‘on’ conditions) are given in the Table 4.5.

As it can be observed from the Table 4.5, the mean and SD are quite uniform across conditions for all the six parameters. The mean ranges between 2 to 3, which represents fair to good quality rating in the rating scale. Fried man test was done for the statistical comparison between the parameters across the conditions.

Table 4.5

Mean and SD for the six parameters of quality for three different conditions

Conditions	Parameters	N	Mean	SD
Noise reduction only	Loudness	20	2.54	0.50
	Clearness	20	2.41	0.59
	Sharpness	20	2.14	0.71
	Fullness	20	2.45	0.80
	Naturalness	20	2.45	0.67
	Overall impression	20	2.73	0.55
Compression only	Loudness	20	3.00	0.69
	Clearness	20	2.64	0.73
	Sharpness	20	2.54	0.96
	Fullness	20	2.82	0.96
	Naturalness	20	2.69	0.72
	Overall impression	20	3.00	0.62
Both On	Loudness	20	2.77	0.62
	Clearness	20	2.45	0.67
	Sharpness	20	2.41	0.73
	Fullness	20	2.50	0.59
	Naturalness	20	2.68	0.65
	Overall impression	20	2.72	0.63

The result showed that there is no significant difference between the different parameters for the three aided conditions except for the rating on loudness. This indicates that the compression and DNR algorithms, when they are activated alone and when they are activated together resulted in a similar self-perceived quality except for the loudness.

Table 4.6

Results of the Fried man test

	N	Chi-Square	df	Significance
Loudness	20	6.62	2	0.04*
Clearness	20	1.56	2	0.46
Sharpness	20	4.78	2	0.09
Fullness	20	7.97	2	0.06
Naturalness	20	1.66	2	0.44
Overall impression	20	1.66	2	0.44

Note:* $p<0.05$

Wilcoxson signed rank test was done on the ratings on loudness parameter to see which of the conditions differed from each other. The results of this are given in the Table 4.7. The results revealed that the DNR only and compression only condition differed from each other. The WDRC, when turned 'on' contributes better for self-perceived loudness. This is in accordance with SIS in compression only condition. That is the SIS is better in compression only condition and perceived loudness is also more in that condition though other parameters of quality were perceived similar.

Table 4.7

Results of Wilcoxon Signed Ranks Test

Conditions	Z Value	Significance
NO – CO	-2.517*	0.01
BO – CO	-1.291	0.19
NO – BO	-1.513	130

NO= DNR only; CO=Compression only; BO= Both DNR and compression 'on'; Note: $p<0.05$.*

Results of the study done by Souza (2002) indicated that, for speech in quiet, speech-quality ratings decreased as compression ratio increased. In the present study, the quality rating was done in quiet and the compression ratios were lesser. Further, in the present study, the testing was done binaurally which might have resulted in better SIS and better quality across the conditions because of binaural advantages.

CHAPTER 5

Summary and Conclusions

The major sequels of sensorineural hearing loss (SNHL) are tolerance problems and speech perception difficulties, particularly in the presence of noise. This can be due to reduced dynamic range, reduced frequency resolution and reduced temporal resolution. In order to overcome these difficulties, most modern digital hearing aids have been implemented with many signal processing strategies. The most commonly implemented strategies are WDRC algorithm, directionality, noise reduction Algorithms and directionality.

The digital signal processing strategies such as WDRC and noise reduction affect the amount of amplification and as a result, attenuate the individual signal components. However, the available reports evaluate only the independent effects of these algorithms. In reality, these algorithms work together and hence, the speech signal gets altered twice. Hence, the present study aims to evaluate the combined effect of WDRC and digital noise reduction algorithms on speech intelligibility and speech quality measures.

The method consisted of experiments for assessing the independent and combined effects of compression and DNR on SIS for words in the presence of noise, at three different levels (40 dB SPL, 65 dB SPL and at 85 dB SPL) and the participant were asked to rate six parameters of quality.

At 40 dB SPL, the results showed that the SIS was significantly different between the aided conditions. There was no difference between SIS across the aided conditions.

At 65 dB SPL level of presentation, aided conditions resulted in better performance than unaided. Compression only condition provided better benefit in the SIS in the presence of noise than the noise reduction only condition. Further, when both the algorithms were presented together, the scores do not deteriorate significantly. This could be because of the type of noise that was used and the compression settings of the hearing aid in the present study.

At 85 dB SPL level of presentation, when only the noise reduction was ‘on’ the performance was significantly poorer when compared to the compression only condition. In addition, compression only condition yielded significantly better performance when compared to the performance in the condition when both the algorithms were ‘on’.

The quality ratings by all the twenty listeners showed that there was no significant difference between the different parameters for the three aided conditions except for the rating on loudness. This indicates that the compression and DNR algorithms, when they are activated alone and when they are activated together resulted in a similar self-perceived quality except for the loudness. This could be because, the testing was done binaurally which might have resulted in better SIS and better quality across the conditions.

To conclude, when both compression and DNR were activated, the speech signal was not altered more than when they are active alone. However, compression brought about more positive changes than DNR. In addition, at higher presentation level, the compression offset the negative effects of DNR. These results, however may be different if the hearing aid with more number of channels, high compression ratio and faster attack and release time, and if the noise had more complex spectrum.

Future directions for research

- The effect of both compression and DNR algorithms needs to be examined using real-world noises which are likely to have more temporal fluctuations.
- The combined effect of DNR and compression can be done by varying the different compression parameters like, compression ratio, attack time and release time.

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