WARP PROCESSING Vs CONVENTIONAL PROCESSING IN DIGITAL HEARING AIDS: A COMPARATIVE STUDY

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MAY 2010

To my beloved Parents

CERTIFICATE

This is to certify that this dissertation entitled "***Warp processing vs Conventional processing in digital hearing aids : A comparative study***" is a bonafide work submitted in part fulfilment for the degree of Master of Science (Audiology) of the student Registration No: 08AUD001. 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

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Chapter 1

Introduction

Many of the individuals with hearing impairment often complain of reduced audibility and distortion of the speech, and they require more signal to noise ratio (SNR) than normals for the speech understanding (Hagerman 1984). Reduced audibility can be compensated by amplification, but the distortion is difficult to deal with. In quiet condition the performance can be improved due to the audibility. But in presence of noise, there are reports of both benefit (Alcantara, Moore, Kuhnel, & Launer 2003) as well as of no benefit (Gustafsson & Arlinger 1994) from hearing aid use in speech recognition tasks.

Digital processing provides better signal processing and performs mathematical operations that enable many of the spurious effects of the analog components to be overcome. Digital hearing aids provide a wide range of highly sophisticated signal processing systems which can enhance speech recognition in noise and create a more comfortable listening environment (Agnew 1999; Edwards 2000). Most digital instruments manipulate the input signal differently which is not possible with analog instruments (Edwards, Struck, Dharan & Hou 1998).

For implementing all these algorithms the hearing aid needs time. This delay includes group delay and frequency dependent delay. Literature shows different values for group delay that may affect the speech perception. According to Agnew and Thornton (2000) delays of 3 to 5 msec was noticeable and delays more than 10 msec was objectionable to the hearing aid users. Frequency dependent delay is another major concern. A recent investigation by Stone and Moore (2003) reported that delay of 9 ms has deleterious effects on speech perception. Digital reproduction also introduces the possibility of new forms of distortion, which arises when the analysis of sound into different frequency regions and subsequent resynthesis to create a single analog signal for presentation to the hearing aid user. The distortion introduced in these processes by processor is called non linear distortion. In designing digital hearing aids there should be a best balance between implementation of the desired processing and the time required for this processing to be carried out (Groth & Soendergaard 2004). The most common reasons for failure of using the conventional hearing aids are non-linear distortion and processing delay, which compromises speech perception in adverse listening conditions. Recent advances in digital technology in hearing aids came with overcoming these problems.

1.1 Need for the study

Conventional processing can add distortion to the processed signal, and it needs more time to process the input signal. But much of the distortion can be avoided using frequency warping. The warp processor provides frequency resolution similar to the human auditory system, with minimal delay, and a high sound quality (Groth & Nelson 2004). Warp processor uses parameters that closely correspond to the auditory bark scale (Smith & Abel 1999). Kates and Arehart (2005) showed that warp processing has less non linear distortion than that of conventional processing.

However with the advancement in technology the warp processor overcomes these drawbacks. There are few commonly available hearing aids which has warp processing, which is a recent technology which has been adopted in digital technology. Most of the studies are in terms of processing delay and they are reporting its advantages in terms of technology. However there is a scarcity of studies which have been reported regarding its benefits in terms of speech identification scores. The Most common complaint of hearing impaired people is difficulty in understanding speech in presence of noise. The most commonly involved environment is multi talker environment. As our knowledge there is no study evaluating the performance of warp processing in presence of back ground noise especially with speech babble. Literature shows that subjective preference is more for warp processing over conventional processing. Hence their performance has to be assessed on larger population to see its advantages over conventional processing. Hence the present study has been taken up with the following aims.

1.2 Aim of the study:

1. Comparison of warp processing over conventional processing on speech identification scores in individuals with sensorineural hearing loss in quiet condition.
2. Comparison of warp processing over conventional processing on speech identification scores in individuals with sensorineural hearing loss in different signal to noise ratios.
3. Compare the subjective preference between the two hearing aids with quality rating.

Chapter 2

Review of literature

Hearing loss caused by damaged to the cochlea is probably the most common form of hearing loss seen in adults. People with sensorineural hearing loss often complain of difficulty with speech communication. The extent and nature of the difficulty depends partly on the severity of the hearing loss. People with mild to moderate losses can usually understand speech in quiet conditions but generally have problem in noisy conditions. The major problems faced by people with sensorineural hearing loss include decreased audibility and decreased frequency resolution.

Decreased audibility is one of the problems faced by the people with sensorineural hearing loss. Absolute thresholds are higher than normals so most of the speech sounds will be above the speech spectrum and it will be inaudible to the subjects (Humes, Dirks & Kincaid 1987). Some researchers are argued that the difficulty in understanding speech arises partly from a reduced ability to discriminate sounds which are well above the absolute threshold (Glasberg & Moore, 1989; Dreschler & Plomp 1980, 1985).

The impaired frequency resolution is main consequence of sensorineural hearing loss. This occurs because of the impaired outer hair cells function. Psycho acoustically this shows up as flatter masking curves and tuning curves (Zwicker & Schorn 1978). People with cochlear hearing loss usually have auditory filters that are broader than normal (Glasberg & Moore 1986). Difference limen for frequency (DLF) of normal and hearing impaired group showed that DLFs were higher for hearing impaired groups compared to normal hearing group (Moore & Peters 1992) which is caused due to impaired frequency resolution. If frequency resolution is impaired relatively intense low frequency parts of speech may mask the weaker higher frequency components this will affect the perception of speech (Danaher, Wilson & Pickett 1975).

2.1 **Conventional Hearing aids**

The advancement in digital technologies in hearing aids is playing a vital role in overcoming these problems. The multi channel dynamic range compression is a basic part of all commercially available digital hearing aids and it is trying to mimic the damaged cochlea. Multichannel hearing aids split the incoming signals in to different frequency bands, and each band of signal passes through a different amplification channel.

The design of digital hearing aids involves many considerations; which includes filtering, amplification, noise reduction. The complexity of algorithm involved in filtering and further processing causes digital delay. The delays are classified as group delay, frequency dependent delay (across channel delay) (Kates & Arehart 2008).

2.2 Processing delay

Computations performed by the digital hearing aids introduce a delay between the arrival of the signal at the hearing aid microphone and delivery of the signal to the ear canal. Digital processing introduces delays about a few to several tens of milliseconds, which could lead to a range of disturbing effects (Moore 1998).

2.3 Group delay

Group delay in a digital signal processing hearing aid results in a finite time delay created while the signal passes through the hearing aid from input to output. Literature has demonstrated that group delay will have deleterious effects on speech production and perception of own voice (Stone & Moore 1999).

In this direction Stone and Moore (1999) investigated the effect of group delay on own voice in a group of 20 subjects with simulated hearing loss. Recordings of two talkers were mixed with various delays. Result showed that delays of 20 to 30 msec were rated as disturbing with the simulation of mild to moderate hearing losses. Whereas, Agnew and Thornton (2000) examined 18 subjects listening to their own voices through a digital signal processing hearing aid with a variable group delay. The subjects varied the length of the delay and determined the amounts that were noticeable and objectionable as compared to the undelayed condition, while listening to their own amplified voices. Results indicated that a delay of 3 to 5 msec was noticeable to the listeners in 76 percent of the trials, and a delay of longer than 10 msec was objectionable 90 percent of the time. The discrepancy in the results may be due to subjects involved. Agnew and Thorton examined experienced listeners whereas Stone and Moore studies naive listeners.

Stone and Moore (2002) studied the frequency dependent delay effect on speech production. The study includes 32 participants who were fitted with hearing aids binaurally. The hearing aids introduce delays of 7 to 43 msec. The participants were asked to read a passage with varying delay. Result shows that speech production is hardly affected by delays less than 30 msec.

2.4 **Effect of Frequency dependent delay on Speech Perception**

The filtering of the input signal by the hearing aid will cause more delay to the low frequency end of the spectrum than the high frequency end. This difference in the delay can cause disturbing effects (Stone & Moore 1999, 2002). Frequency dependent delay can also introduce audible artifacts when listening to speech even when the user of the hearing aid is not talking. Blauert & Laws (1978) examined the frequency dependent delay perception by presenting clicks. The clicks were passed through all-pass filters giving increased delay in narrow-frequency regions, and found that normal-hearing subjects can detect delays as short as 1 millisecond at 2 kHz, with the detection threshold increasing to 2 milliseconds at 8 kHz or 1 kHz.

Arai and Greenberg (1998) studied the effect of across frequency delay by introducing delay variations as a function of frequency in sentence materials. They found that word identification accuracy for normal-hearing subjects decreased as the delay variations increased. The subjects maintained good word identification scores with a across delay variation of 140 milliseconds. Whereas in a recent investigation by Stone and Moore (2003) examined the identification of nonsense syllables in individuals with hearing impaired by introducing the delay varying from 0 ms to 24 m sec. They demonstrated that delay of 9 ms has deleterious effects on speech perception. Even the smaller delay of 4 ms has an effect but it did not reach significance. The variation in results for effect of frequency dependent delay may be due to the type of test stimuli used and subject group.

2.5 Warp processing

Most commonly used filtering technique in hearing aids is frequency domain filtering, and less commonly used one is time domain. In current conventional hearing aids digital filtering provides constant band width across frequencies but in the human auditory system band width increases as frequency increases (Moore & Glasberg 1983; Zwicker & Terhardt 1980).

A multi channel design technique which provides logarithmic frequency representation with high efficiency is frequency warping which is a side branch type processor. Figure 2.1 represents the 17 overlapping warp compressor.

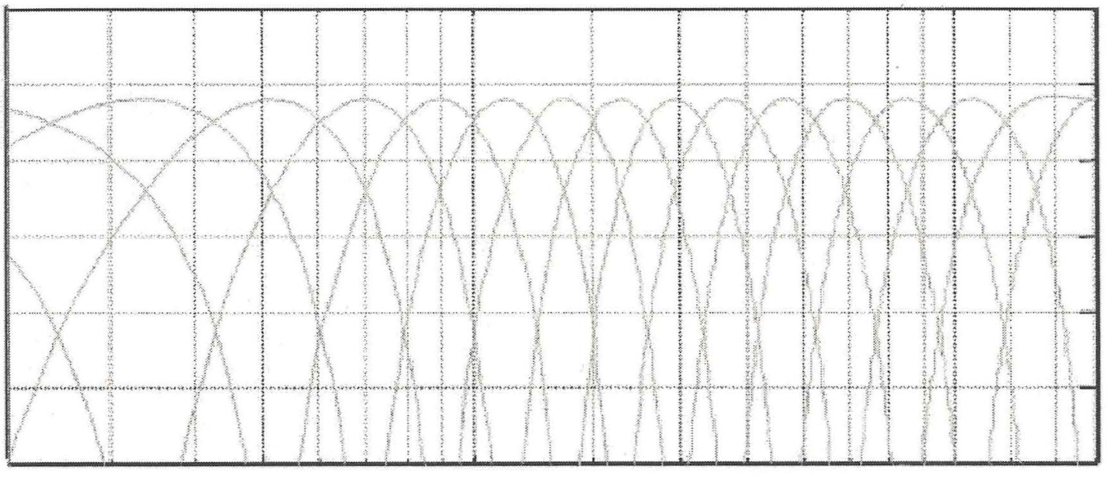

Figure 2.1 Represents the warp processor that has 17 overlapping frequency bands.

Kates and Arehart (2005) showed that warp processing has less non linear distortion than that of FFT based processor. They used spectrogram for this purpose. Results are shown in the figure 2.2.

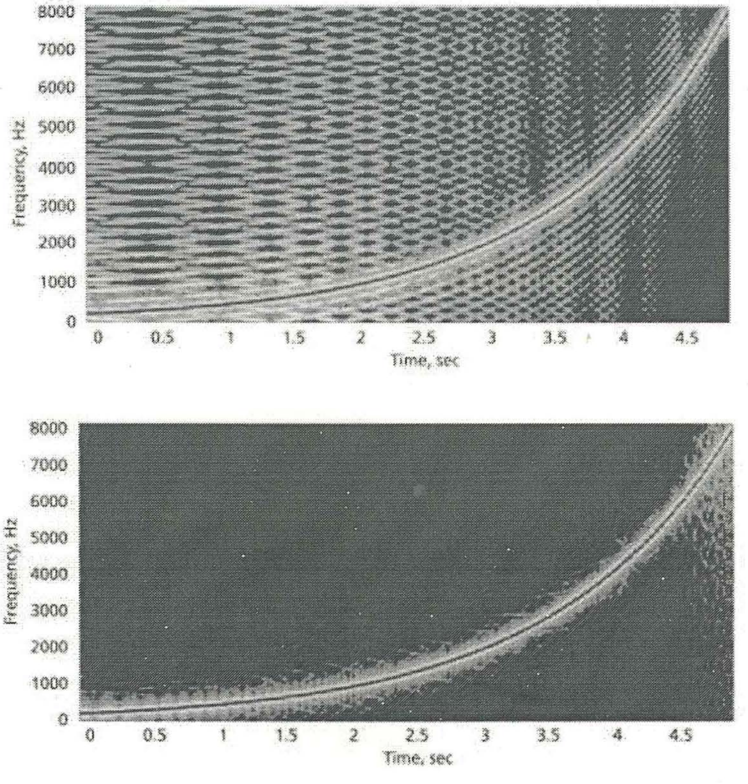

Figure 2.2 Showing the responses for a swept tone for an FFT based processor and warp processor. Above figure shows the responses with FFT and below one is with warp.

One can note from the figure 2.2 that non linear distortion is more in FFT based processor across all the frequencies while warp processor has negligible distortion.

Kates and Arehart (2005) studied detection threshold for impulsive sounds, for steady state sounds, and for continuous speech. The participants include 10 normal hearing individuals and 11 mild to severe sensorineural hearing impaired. The processed sounds were presented to the participants. Two compressors were simulated for the performance evaluation. One was FIR filter and the other was warped compressor. The test stimuli included were clicks, sentences and vowels.

The just noticeable delay was obtained in listeners using a three interval three down one up adaptive procedure (Levitt 1971). Two of the three intervals on each trial contained a standard stimulus with no frequency warped group delay. One of the three intervals on each trial contained a comparison stimulus with a frequency warped delay. The listener's task was to identify the interval with the frequency warped group delay. The result of the study shows that normal hearing subject had lower thresholds on average than the hearing impaired subjects.

The maximum delay in a practical system is just over 6 milliseconds, which is comfortably below the threshold of approximately 9 milliseconds found for audible interference when the hearing-aid user is talking. This results show that the frequency dependent group delay produced by warp compressor was inaudible for most listeners for the click stimuli and for steady-state speech sounds. Thus, a warped compressor should give a system with inaudible delay under nearly all listening conditions.

Ricketts, Dittberner & Johnson (2008) examined the relative impact of frequency warping based versus fast fourier transform based compression systems on perceived sound quality of music and speech as a function of degree of hearing loss. Sound quality comparisons for two different signals processed using two different types of compression processing fast fourier transform (FFT) and warp. Participants include 10 listeners with normal hearing and 20 listeners with mild-to-moderate hearing loss. Paired comparisons were made separately for each of two recorded sound samples. Test stimulus includes short segment of bluegrass music ("The Lucky One" as performed by Alison Krauss [AK]) and a popular movie (both voices and a ringing musical triangle from the movie Seabiscuit [SB]). Subjects task is to compare the sound quality of the two segments and pick the one you like best by telling the "number 1" or "number 2." The processing delay in the FFT-based compressor is constant across frequencies (5 ms), whereas in the warp-based compressor, there is a greater delay at low frequencies (6 ms) than at high frequencies (3 ms). All the subjects demonstrated a clear preference for the frequency warping based processing among listeners with moderate sensorineural hearing loss for all types of sounds tested.

There are several differences in processing of conventional and warp. The differences are shown in the table 2.1.

Table 2.1 *Shows the differences in warp and fast fourier transform processing.*

WARP PROCESSING	FFT PROCESSING
Provides non uniform logarithmic frequency representation similar to auditory bark scale. In the auditory system band width of frequency analysis is constant at low frequencies and increases at higher frequencies (Moore & Glasberg 1983)	The FFT technique is based on a uniform spacing of frequency components while the auditory system is based on a logarithmic spacing.
The Warp processor has 17 overlapping frequency bands. The frequency warping results in non-uniform band spacing corresponding closely to the auditory Bark scale. The degree of overlap between the bands provides a smooth, artifact-free frequency response.	An FFT-based system can provide the necessary low-frequency resolution but only at the expense of unnecessary narrow high frequency bandwidths.
Smoothly overlapping band will avoid band edge effects.	Uniform spacing of bands results in band edge effects.
The Warp processor design introduces delays that are shorter than required to ensure no disturbing perceptual effects (Groth & Soendergaard, 2004)	Processing delay is long, results in a negative user perception (e.g. an echo).
Negligible non linear distortion	Non linear distortion is more
Provides a technique for approximating the frequency resolution of the ear while minimising the time delay(Kates 2003)	Improved frequency resolution results in longer time delays.

A review of the literature revealed that earlier investigations have not documented a clear relationship between speech identification scores in quiet and speech-in-noise performances with hearing aids having warp processing. There is a lack of study showing the processing delay and the speech perception abilities of individuals with hearing loss. Therefore the present study is focusing the performance of warp processing and conventional processing in individuals with sensorineural hearing loss.

Chapter 3

Method

The aim of the present study was to evaluate the performance of a hearing aid with and without 'warp processing' in individuals with sensorineural hearing loss. The performance of the hearing aid was evaluated using both speech identification scores in quiet and noisy conditions and quality judgement.

3.1 Participants

The study consisted of 20 participants (14 males and 6 females) in the age range of 50-65 years with a mean age of 60.25 years. The participants were clinically diagnosed as having moderate to moderately severe sensorineural hearing loss, based on pure-tone average (500Hz, 1KHz, 2KHz), Tympanometry, acoustic reflexometry, Transient evoked otoacoustic emissions (TEOAEs). The mean pure tone average for the group was 51.6 dB HL. All the participants were native speakers of kannada language and naïve hearing aid users. The demographic and audiological profile of the participants is given in the Table 3.1. None of the participants had any symptoms of otological and neurological disorders. The retrocochlear pathology was ruled out by administering auditory brainstem response on all the 20 participants.

Table 3.1 *Represents subjects age, gender, test ear, pure tone average (PTA), speech identification scores (SIS) and audiogram configuration*

Subject	Age (in years)	Gender	Test ear	PTA (in dB)	SIS (%)	Audiometric configuration
1	50	M	right	45	90	Flat
2	60	M	Left	55	86	Flat
3	60	F	Left	53.3	80	Flat
4	65	F	right	51.6	96	Flat
5	67	M	right	56.6	80	Flat
6	50	M	Left	56.6	80	Flat
7	60	M	right	61.6	96	Gradually sloping
8	62	M	right	55	90	Flat
9	55	F	Left	55	80	Flat
10	65	M	right	38.3	76	Sharply sloping
11	63	M	Left	43.3	100	Gradually sloping
12	58	M	right	60	85	Flat
13	58	M	right	43.5	92	Flat
14	65	M	right	41.6	88	Precipitously sloping
15	64	F	Left	48.3	92	Flat
16	60	M	right	60	80	Flat
17	62	M	Left	66.6	80	Flat
18	62	F	Left	43.3	92	Flat
19	55	F	right	50	100	Flat
20	58	M	right	48.3	85	Flat

1. Flat : <5 dB rise or fall per octave.
2. Gradually sloping: 5-12 dB threshold increase per octave.
3. Sharply sloping: 15-20 dB threshold increase per octave.
4. Precipitously sloping: Flat or gradually sloping then threshold increasing at rate of 25+ dB per octave. (Adapted from Carhart 1945, Lloyd & Kaplan 1978).

3.2 Test environment

All the experiments were conducted in a sound treated double room situation. The ambient noise levels were within permissible limits as per ANSI S3.1 (1991).

3.3 Equipment

1) A calibrated dual channel diagnostic audiometer and two Martin (C115) free field speakers were used for evaluating hearing aid performance.
2) A computer with sound card (High definition audio device) and adobe audition (version 3) were used for playing the stimulus.
3) Two non linear multichannel digital behind the ear hearing aids, one with warp processing and the other with conventional processing (The hearing aid with FFT processing is considered as conventional processing in this study) was used in the present study. The fitting range of these hearing aids was from mild to severe degree with the frequency range of 250Hz to 6000Hz.
4) A Pentium IV computer with NOAH-3 aventa (version 2.9) software and hearing instrument programmer (hi-pro), a hardware interface was used for connecting the hearing aid to the personal computer for the programming of the hearing aid.

3.4 Test material

1) Speech Material

The speech stimuli used in the present study were taken from bisyllabic wordlists in Kannada, developed by Yathiraj and Vijaylakshmi (2005). This test contains four different word lists of equal difficulty, each containing 25 bisyllabic words, which are phonetically balanced. The words spoken in a conversational style by a female native speaker of Kannada, were digitally recorded in an acoustically treated room on a data acquisition system with a sampling frequency of 44.1 kHz and 16-bit analog to digital converter. The order of words in each original list was randomized so as to produce two lists from each original list. Thus a total of six lists were available for testing. The speech material was always presented at 45^0 azimuth to the test ear.

2) Back ground Competing Stimuli

Kannada speech babble developed by Anitha and Manjula (2005) was used as the competing stimulus in the study. The back ground competing stimuli was presented through the other loud speaker of the audiometer at 270^0 azimuth. The spectrum of the speech babble and speech material is shown in Figure 3.1.

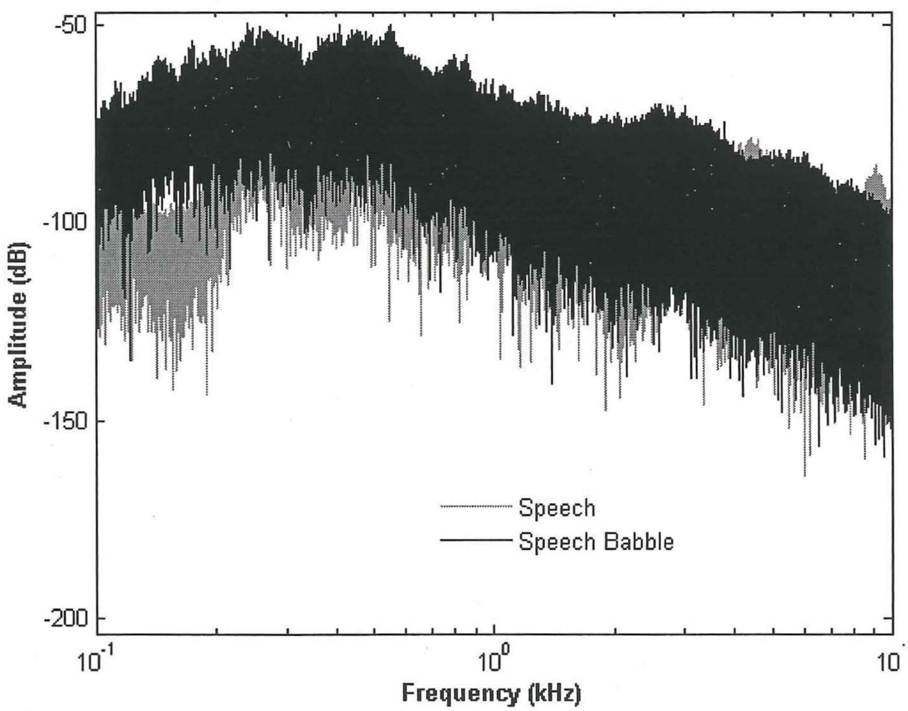

Figure 3.1 Shows the spectrum of the speech babble and the speech material.

From the above figure, one can notice that the spectrum of the speech babble is almost covering the speech stimulus, indicating that speech babble is sufficient enough to mask the speech. Only for low frequencies below 250Hz the speech babble is not covering the speech stimulus.

3.5 Procedure

Hearing aid programming and fitting

- The participants were seated comfortably on a chair and were fitted with the hearing aid on the test ear using an appropriate sized ear tip.
- Two hearing aids were selected; one was with the facility of having warp processing and another with conventional processing.
- The hearing aid was connected to the programming hardware (Hi pro) through a suitable cable and then detected by the programming software.
- The hearing aid was programmed either for the right / left ear depending on the speech identification scores and the degree of hearing loss.
- The pure tone air conduction and bone conduction thresholds of the participant's test ear was fed into the programming software and the target gain curves were obtained using the NAL1 prescriptive formula.
- The hearing aid was fine tuned according to the participant's preference by manipulating the gain of the each frequency channel at different input levels (50 dB and 80 dB).
- Other parameters of both the hearing aids (warp processing and conventional) were kept at default settings.

- The unaided speech identification scores were assessed at 40dBHL for all the participants as it was a part of regular testing.

The present study was conducted in two different phases for two aided conditions (warp processing and conventional processing). In the first phase, aided speech identification scores were obtained in both quiet and noisy conditions at different signal to noise ratios (0 dB SNR & -10 dB SNR). In the second phase, the performance of both the hearing aids was assessed through quality judgment.

Phase 1: Evaluation of the performance of the hearing aids

Participant's performance was assessed for both the hearing aids (warp processing & conventional processing) in quiet and noisy conditions (0dB &-10dB SNR). The speech stimuli were played from a personal computer (PC) at a sampling rate of 44.1 kHz and routed to a calibrated (ANSI, 1996) diagnostic audiometer (Madsen OB-922 with speaker). For quiet condition, the participants were presented with the signal from the loudspeaker of the audiometer at a distance of one meter. In all the testing conditions the signal was presented at 45^0 azimuth to the test ear. The presentation level of the stimulus was 40 dB HL.

In noisy condition, the aided performance was tested at two different signal to noise ratio (0dB & -10dB SNR). The noise was speech babble played in PC at 44.1 KHz sampling rate. The speech babble was presented through speaker which was placed at 270^0 azimuth to the test ear. Noise level was varied to obtain the required SNR. Participants were instructed to repeat the speech tokens heard by them. There were total six conditions [2 (hearing aids) × 3 (Quiet, 0 dB, -10 dB SNR)] and only four lists were available. We generated eight lists by randomising the words in each list and for the present study only six lists were selected. The order of presentation of these lists was randomized across the participants to ensure that the practice effect did not influence the results of the test.

Phase 2: Quality Judgement

The participants were asked to rate both the hearing aids in terms of its quality of speech output. For this, the recorded Kannada passage developed by Sairam 2002 was routed through the audiometer at 40 dB HL at 45^0 azimuth. The participants were instructed to rate on six parameters of quality. The parameters and rating scale used in the present study was similar to that used by Sruthi 2009 and are given below. The instructions were made simple in Kannada and it was explained to the participant and they were asked to rate the performance on a 10 point scale. The parameters and the rating scale for evaluating the quality judgment were:

- Loudness : From 0 to 10
- Clearness : From 0 to 10
- Sharpness : From 0 to 10
- Fullness : From 0 to 10
- Naturalness : From 0 to 10
- Overall impression : From 0 to 10

Each of the six parameters were rated on a 10 point rating scale. The details are as given below.

0 – very poor

2 – Poor

4 – Fair

6 – Good

8 – Very Good

10 – Excellent

The participants were asked to rate the odd numbers if they found the quality to be intermediate between two points.

3.6 Statistical analysis

A suitable statistical analysis was carried out to determine the effectiveness of warp processing in quiet condition as well as in noisy condition. Repeated measure Analysis of Variance (ANOVA) was done for finding the main effect of different aided conditions (Hearing aid with warp processing & conventional processing) in quiet and in the presence of noise. Paired sample t-test was done to check the group differences between aided conditions in quiet, and in presence of noise (0 dB and -10 dB SNR). Friedman's test was done to find the effect of the different subjective ratings of quality across the two hearing aids. Wilcoxon signed rank test was done to see the significant difference in quality rating.

Chapter 4

Results and Discussion

The present study was conducted with an aim to find out the efficacy of the warp processing over the conventional processing in digital hearing aids. The data were obtained from a group of participants having moderate to moderately severe sensorineural hearing loss which includes flat as well as sloping configuration. The data were analyzed using Statistical Package for the Social Sciences (SPSS for windows, Version 17) software. The following statistical tools were used to analyze the collected data.

a) Descriptive statistics to obtain the mean and standard deviation for speech identification scores with the two hearing aids in quiet and noisy conditions.

b) A repeated measure Analysis of Variance (ANOVA) for finding the main effect of different aided conditions (Hearing aid with warp processing & conventional processing) in quiet and in the presence of noise.

c) A paired samples t-test was done to check the group differences between aided conditions in quiet, and in the presence of noise (0 dB and -10 dB SNR).

d) Friedman's test was done to find out the effect of the different subjective ratings of quality across the two hearing aids.

e) Wilcoxon signed rank test was done to see whether the difference in quality rating was significant across the hearing aids.

A) Speech Identification Scores

The speech identification scores were obtained and tabulated for all the participants for two hearing aids in quiet and noisy conditions. The mean and standard deviation (SD) in parenthesis for speech identification scores in quiet and noisy conditions for warp and conventional processing hearing aids are shown in Table 4.1.

Table 4.1 *Mean and SD of speech identification scores for warp and conventional processing hearing aids in quiet, 0 dB & -10 dB SNR*

Condition	Quiet	0 dB SNR	-10 dB SNR
Warp	76.2% (14.88)	49% (15.56)	18.2% (9.48)
Conventional	70% (15.76)	25.6% (19.79)	13.4% (7.92)

One can note from the table that performance with warp processing is better over conventional processing in all the three conditions. But the mean difference was more at 0 dB SNR condition over other two conditions. The more variation in the data was noticed for conventional processing than warp processing.

A repeated measure of ANOVA, for with in subject factors, hearing aids (2 levels) and condition (3 levels), was done to find out the differences between the performance of hearing aid with warp and conventional processing in quiet and noisy (0 dB SNR , -10 dB SNR) conditions. Results revealed a significant main effect of hearing aids [$F_{(1, 19)} = 87.608$, $p < 0.001$] and condition [$F_{(2, 38)} = 183.396$, $p < 0.001$]. There showed a significant interaction between hearing aids and condition [$F_{(2, 38)} =32.718$, $P < 0.001$]. This interaction indicates that difference in mean scores is not same across conditions for different hearing aids. Table 4.1 clearly shows that the differences between hearing aids are more at 0 dB SNR than other conditions. Further, Bonferroni pair wise analysis indicated that the mean difference across conditions was reached significance ($P <0.001$).

All participants demonstrated improved performance with warp processing over conventional processing in all the conditions. The improved performance was higher for 0 dB SNR compared to other conditions. Very small improvement in performance was noticed for quiet and at -10 dB SNR conditions. To our knowledge there were no studies which directly investigated the performance for speech with warp processing aid. However, few earlier investigators have provided the technological difference between warp processing and conventional hearing aids. They are warp processor introduces less group delays, across channel delay than conventional processing hearing aids (Kates & Arehart 2005; Groth & Soendergaard 2004). More over the warp processor provides frequency resolution similar to the human auditory system, with minimal delay, and a high sound quality and it uses parameters that closely correspond to the auditory Bark scale (Smith & Abel 1999). Kates and Arehart (2005) showed that warp processing has less non linear distortion than that of FFT based processor. Stone and Moore (2003) studied the effect of across channel delay in conventional processing hearing aids on speech perception scores and they demonstrated that a delay of 9 ms or higher has significant deleterious effect on speech perception scores in quiet. The smaller delays did reduce identification scores, but that reduction was not significant. Adding the noise to speech signal would have exaggerated the difficulty in understanding speech even at smaller delays. This could be one of the reasons for lower scores with conventional processing in presence of noise.

The delay at different channels for the hearing aids used in the present study was assessed using B&K pulse analyzer. From the analysis it was noted that the frequency dependent delay was more for conventional processing hearing aid (3 to 4 msec) than warp processing hearing aid (1.71msec). In the present study probably the frequency dependent delay along with other factors would have contributed for the difference in performance between the hearing aids.

The performance difference was less for quiet and -10 dB SNR condition, as the less frequency dependent delay (< 9 ms) does not affect speech scores significantly in quiet (Stone & Moore 2003). The observed small improvement may be due to bark scale filtering and fewer nonlinear distortions. The identification scores are very low in the hearing aids for -10 dB SNR condition which would have caused floor effect, leading to less significant difference. Plomp (1988) demonstrated that hearing impaired group needs more signal to noise ratio than normal hearing people. More over in adverse conditions the hearing impaired performance will decrease drastically.

2. Quality Judgements

For the judgement of quality six parameters were evaluated. The participants were asked to rate these parameters on the recorded Kannada passage played to them. Friedman test was carried out to see the significant difference in ratings for all the six parameters with the two hearing aids. The results for six quality parameters for warp and conventional hearing aids are shown in the figure 4.1.

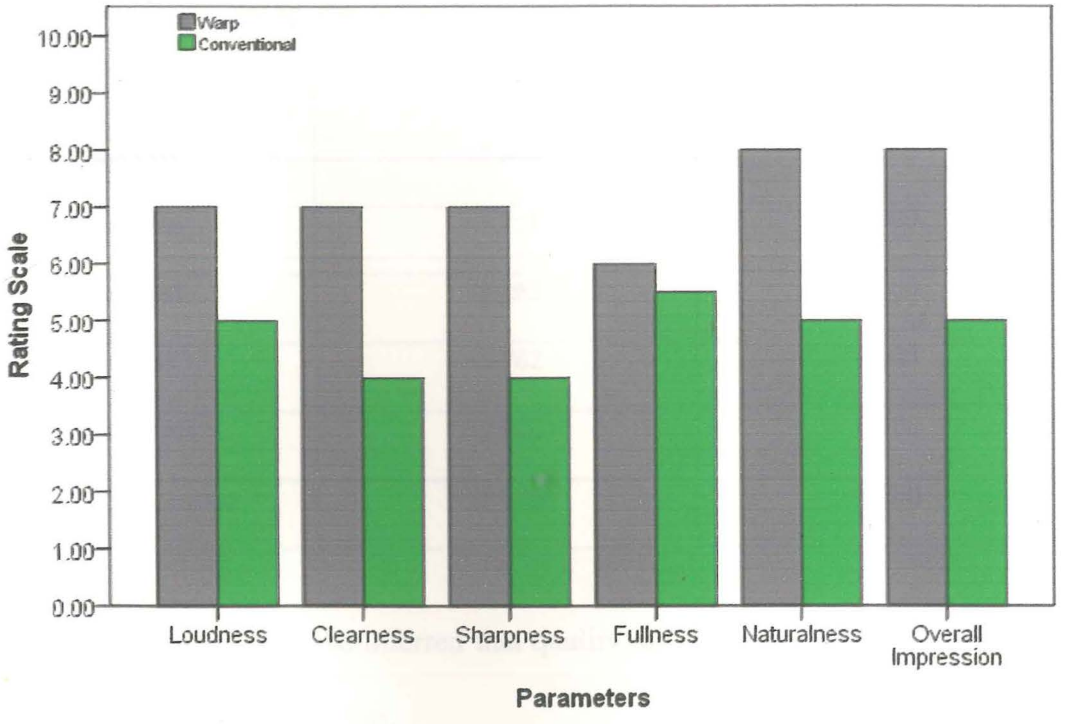

Figure 4.1 Shows the quality ratings with the two hearing aid processors for the sixparameters of quality.

From the Figure 4.1, it can be inferred that the ratings obtained for hearing aid with warp processing is higher than the conventional hearing aid. For all the six parameters the hearing aid with warp processing showed a clear preference over conventional hearing aids. Wilcoxon signed rank test was done to see if the difference of each parameter were significant. Results for all the parameters are shown in the table 4.2.

Table 4.2 *Represents the quality parameters, the Z value and the significance*

Prameters	Z value	Significance
Loudness	-3.867	0.000
Clearness	-3.758	0.000
Sharpness	-3.962	0.000
Fullness	-2.762	0.003
Naturalness	-3.902	0.000
Overall impression	-3.976	0.000

From the above table it can be inferred that quality ratings with the warp hearing aid is significant in all the conditions.

This result is consistent with the study done by Dittberner, Rickets, & Johnson 2008. He examined the relative impact of frequency warping-based versus Fast Fourier Transform based compression systems on perceived sound quality of music and speech as a function of degree of hearing loss. They demonstrated a clear preference for the warp based processing among listeners with moderate sensorineural hearing loss for all types of sounds tested. In this study also participant's preference for warp processing was higher than for conventional processing.

The warp processing hearing aids possesses the features such as reduced frequency dependent delay, and is based on logarithmic scale that is close to the bark scale. Moreover spectrogram results are also showing less non linear distortion (Kates & Arehart 2005). Probably all these features could be contributing for improvement in loudness, clearness, sharpness, fullness, naturalness, and overall impression for warp hearing aid over conventional hearing aid. From this study one can infer that hearing aid with warp processing will be useful in noisy environment. The multi talker speech babble used as back ground competing stimuli is representing almost the real life situation. Hence hearing aid with warp processing will be useful in natural environment.

Chapter 5

Summary and conclusion

Multichannel wide dynamic range hearing aids were most commonly used amplification devices for individuals with cochlear hearing loss. These hearing aids have several advantages that are reported by many investigators (Armstrong 2006, Kates & Arehart 2005). However, recent literature has demonstrated a disadvantage of these hearing aids, in terms of delay and distortion to the processed signal. Recent advances in digital technology in amplification system uses warp processing, which promises compromise for disadvantages noted in conventional hearing aids (Groth & Nelson, 2005). There is a scarcity of studies which have been reported regarding warp processing's benefits in terms of speech identification scores in quiet as well as in presence of noise. Hence their performance has to be assessed on larger population to see its advantage over conventional processing. Hence the present study has been taken up with the following aims.

1) Comparison of warp processing over conventional processing on speech identification scores in individuals with sensorineural hearing loss in quiet condition.
2) Comparison of warp processing over conventional processing on speech identification scores in individuals with sensorineural hearing loss in different signal to noise ratios.
3) Compare the subjective preference between the two hearing aids with quality rating.

The study consisted of 20 participants (14 males and 6 females) in the age range of 50-65 years with a mean age of 60.25 years. The participants were clinically diagnosed as having moderate to moderately severe sensorineural hearing loss. The present study was conducted in two different phases for two aided conditions (warp processing and conventional processing). In the first phase, aided speech identification scores were obtained in both quiet and noisy conditions at different signal to noise ratios (0 dB SNR & -10 dB SNR). In the second phase, the performance of both the hearing aids was assessed through quality judgment. The participants were asked to rate the six parameters of quality on a ten point scale.

The result shows that participants obtained better speech identification scores in quiet as well as in noisy conditions for warp processing over conventional processing. But the mean difference was significant only at 0 dB SNR condition than in quiet and -10 dB SNR. In addition it was also noted that quality rating for six parameters was higher for warp processing over conventional processing. These results suggests it is more advantageous using warp processing hearing aids for improved understanding of speech in adverse listening conditions and also the sound quality is much better than conventional hearing aids. However, there need further investigations studying in different listening conditions and different age group of subjects.

The findings of the present study have some clinical implications.

1) The study helps in better understanding of warp processing in hearing aids.
2) Hearing aids with warp processing have less processing delay so that we can implement more sophisticated algorithms to the digital signal processor.

3) Hearing aids with warp processing is useful in noisy environments.

4) Warp processing strategy can be implemented in open fit hearing aids because of its low processing delay.

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