Warp Processing vs Conventional Processing in Digital Hearing Aids: A Comparative Study

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Abstract

Recent advances in digital technology in amplification system uses warp processing that promises compromise for disadvantages noted in conventional hearing aids. 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 the presence of noise. Hence the present study aimed to assess its advantage over conventional processing in a group of individuals. The study included 20 participants with moderate to moderately severe sensorineural hearing loss. The performance of the hearing aid was evaluated using both speech identification scores in quiet and noisy conditions and, on quality judgement. Results showed that 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. For all the six parameters of quality, the hearing aid with warp processing showed a clear preference over conventional hearing aid. Hence it is concluded that it is more advantageous using warp processing hearing aids for improved understanding of speech in adverse listening conditions. Also, the sound quality is much better than the conventional hearing aid.

Key words: *Conventional processing, warp processing, digital hearing aids.*

Many of the individuals with hearing impairment often complain of reduced audibility and distortion of speech, and they require more signal to noise ratio (SNR) than normals for the speech understanding (Hagerman, 1984). 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). 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), a delay of 3 to 5 ms was noticeable and delays more than 10 ms was objectional 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 is done. 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. 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. Kates and Arehart (2005) showed that warp processing has less non linear distortion than that of FFT based processor. The study also shows 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.

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

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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 was taken up with the aim of comparing between warp and conventional processing on speech identification scores in individuals with sensorineural hearing loss in quiet condition, and at different signal to noise ratios. Subjective preference was also compared between the two types of processing.

Method

Participants: The study consisted of 20 participants (14 males & 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. 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.

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).

Equipment: A calibrated dual channel diagnostic audiometer and two Martin (C115) free field speakers were used for evaluating hearing aid performance. A computer with sound card (High definition audio device) and adobe audition (version 3) were used for playing the stimulus. 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. 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.

Speech material: The speech stimuli used in the present study were taken from bisyllabic word lists 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.

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.

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 through 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. 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, with 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.

Results and Discussion

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 1.

Table 1. Mean and SD of speech identification scores for warp and conventional processing hearing aids in quiet, 0 dB & -10 dB SNR

Condition	Quiet	0dB SNR	-10dB 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 within 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. Table1 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 the 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 ms) than warp processing hearing aid (1.71ms). 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.

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 1.

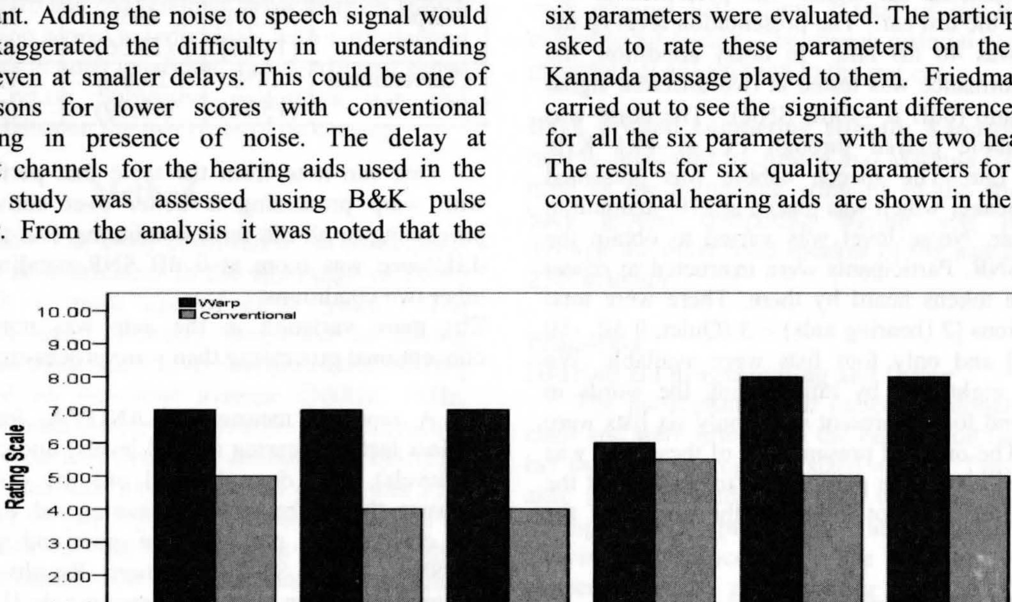

Figure 1. Shows the quality ratings with the two hearing aid processors for the six parameters of quality.

Table 2. Represents the quality parameters, the Z value and the significance

Prameters	Z	p
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 Figure 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 2. From the Table 2 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). They 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 possess 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.

Conclusions

Hearing aids with warp processing have less processing delay so that we can implement more sophisticated algorithms to the digital signal processor. Hearing aids with warp processing are useful in noisy environments and Warp processing strategy can be implemented in open fit hearing aids because of its low processing delay.

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