STUDENT RESEARCH AT A.I.I.S.H. MYSORE
(ARTICLES BASED ON DISSERTATION DONE AT AIISH)

VOLUME VI: 2007-08

PART – A

Audiology

Compiled by

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Director

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Reader in Special Education

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FOREWORD

AIISH is presenting the sixth volume of full length articles based on the dissertation work done by our post-graduate students in part fulfillment of their PG degrees in Audiology, Speech-Language Pathology and Special Education (HI). The one year PG program of M.S.Ed (HI) was started in the year 2007-08. The present volume is presenting the additional Part C covering the articles based on M.S.Ed (HI) dissertations for the first time.

This volume includes articles based on dissertations done by the post-graduate students during the year 2007-08. There are totally 45 articles. Part A comprises of 19 papers related to Audiology. A wide range of interest is yet again illustrated by our PG students in the volume as well. The section contains articles covering evoked potential testing and related issues; diagnostic tests and intervention aids; issues related to hearing aids cochlear implants aging and FM systems; auditory dys-synchrony; Vestibular Evoked Myogenic Potentials (VEMP); issues related to speech perception; speech recognition in noise. Part B comprises 20 papers in the area of Speech-Language Sciences and Pathology containing articles covering various topics covering issues related to Language development and assessment, cognition and aging; aphasia, motor speech disorders; speaker identification; traumatic brain injury; issues related to bilingualism and specific language impairment (SLI). Part C contains 6 papers in the area of Special Education. It covers topics of Segregated and inclusive education; present status of special schools; capabilities of preschoolers; manpower requirement, adaption of curriculum and teaching methodology. The dissertations were guided by the young faculty in the Department of Special Education, who were guiding PG students for the first time. I am glad that our faculty both experienced and not so experienced have enthused the students to undertake research in a variety of topics in Audiology, Speech-Language Pathology and Special Education. The titles of the articles are the titles of the dissertations. The first authors of part A and part B are the II M.Sc. students of 2007-08 while that of part C are the 1st batch students of M.S.Ed (HI) and the second authors are their respective guides (with their academic designations as on January 2010) who have supervised and guided the research work. The AIISH faculty members who have guided the dissertations have modified and edited the papers to bring it to the present shape to the best of their abilities in spite of their busy academic schedules. Dr. G. Malar, Reader, Special Education has put in efforts to procure and compile the edited articles and has herself corrected the English in many of the articles. Her sincere hard work in bringing out this publication is highly appreciated. Ms. R. Rajalakshmi has formatted the papers very well within a very short time. Her contributions and the contributions of Mr. Sujit Kumar Sinha in proof reading are sincerely acknowledged. The unattended mistakes in print and references, if any, in spite of best efforts put in by the team are regretted.

We are happy to state that seventh volume of the publication of dissertation papers is also published along with this. You may please e-mail your valuable feedback about this volume to aiish_director@yahoo.com with the subject “Student research, Volume VI A/B/C 2007-08.

Dr. Vijayalakshmi Basavaraj
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Efficacy of NAL-NL1 Prescription for the First-Time Hearing Aid Users: A Follow-Up Study

Ankit Mathur & P. Manjula*

Abstract

A prescriptive approach to hearing aid fitting is one in which the amplification characteristics are calculated from some of the hearing characteristics of an individual. NAL-NL1, one such generic formula, which aims at maximizing speech intelligibility, is widely in use. The aim of the study was to evaluate the efficacy with which NAL-NL1 prescribes the hearing aid parameters for persons with varying types and degrees of hearing loss. In addition to find out the changes in the amplification parameters preferred by the hearing aid users in terms of, after the first 6 to 8 weeks of hearing aid use. Further, the study also aimed at evaluating the extent of deviation of the amplification parameters from the target prescribed by NAL-NL1. For this, a follow-up study was undertaken, in which participants (N=20) were tested under three conditions of hearing aid program. The three hearing aid programs were one with NAL-NL1, second with preferred setting and third with fine tune setting. Both subjective (aided thresholds, Speech reception scores and uncomfortable level) and objective (real ear insertion gain and real ear saturation response) measurements were carried out to compare performance of the participants in the three conditions. The results indicated an improved performance in subjective measures with fine tune settings compared to the other two conditions. These findings were supported by the objective test results also. These findings prove that fine tune program provides better results when compared to NAL-NL1. Re-programming according to individual’s listening needs can enhance the benefit that one derives from the hearing aid. Hence, follow-up of the hearing aid users for fine tuning of the hearing aid should be considered as an integral part of hearing aid prescription procedure for greater user satisfaction and continued hearing aid use.

Key words: aided threshold, speech recognition scores, uncomfortable level, real ear measurement

Introduction

A prescriptive approach to hearing aid fitting is one in which the amplification characteristics are calculated from the hearing characteristics of an individual. This is based on the assumption that certain amplification characteristics suit certain types, degrees and configurations of hearing loss (Byrne, 1986). There are several prescriptive procedures for hearing aid selection, some for linear hearing aids such as Prescription Of Gain and Output (POGO), National Acoustic Laboratories formula-Revised (NAL-R) and others for non-linear hearing aids such as Desired Sensation Level-input/output (DSL-i/o), Figure 6 (FIG6), NAL - Non Linear 1 (NAL-NL1).

The first prescriptive formula by National Acoustic Laboratories (NAL) was published in 1976 (Byrne & Tonnison, 1976). NAL procedure is a threshold-based prescription for linear hearing aids that aims at maximizing speech intelligibility. The procedure is based on three principles, viz. preferred insertion gain at 1 kHz equals 0.46 times the loss at 1 kHz, speech bands according to the long-term average speech spectrum should be perceived equally loud, and equal loudness at a most comfortable level is modelled using the 60-phon equal loudness contour curve by listeners with normal

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hearing. As its predecessor NAL, NAL-R (Byrne & Dillon, 1986) is also a threshold-based procedure for prescribing the gain-frequency response in linear hearing aids with the aim of maximizing speech intelligibility.

NAL-NL1 evolved as a compression based method for non-linear hearing aids in 1998, from the older NAL-R method for linear hearing aids. The NAL-NL1 fitting method presents with some other interesting features, which are based in part, on the original underlying philosophy of the whole NAL “family” of fitting methods (NAL, NAL-R, NAL-RP). This includes equalization, rather than normalization (preservation) of the loudness relationships among the speech frequencies. The reason the NAL-NL1 method deviates from the approach of preserving the unaided loudness relationships among the different frequency elements of speech is because the “preserving” approach has not been shown to improve speech intelligibility (Dillon, Byrne, Ching, Katsch, Keidser and Brewer, 2000).

A question that might come to mind is whether the hearing aid users appreciate and accept the hearing aid that is programmed to NAL-NL1 targets. Can a user always accept the levels that would theoretically maximize their speech intelligibility? Killion and Revit (1993) have cautioned that even accurate calculations, carefully computed coupler transfer functions, and rigid standards of manufacturing the hearing aid might not yield the perfect results when measured on an individual probe microphone.

The present study aims at evaluating the efficacy with which NAL-NL1 prescribes the hearing aid parameters for persons with varying types and degrees of hearing loss. It also aims at finding out the changes observed in the preferred amplification parameters by the hearing aid users after the first 6 to 8 weeks of hearing aid fitment, and the extent by which they deviate from the target prescribed by NAL-NL1.

Method

Participants

The study included 20 participants in the age ranging from 45 to 80 years (mean age = 60.9 years). The participants were divided into groups based on the degree and type of hearing loss.

A. Based on degree of hearing loss in the aided ear, three groups were formed, which were:

<table>
<thead>
<tr>
<th>Groups</th>
<th>Grouping Criteria</th>
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<tr>
<td>Group A1</td>
<td>Participants with a Pure Tone Average (PTA) of 35 to 55 dB HL</td>
</tr>
<tr>
<td>(N=5)</td>
<td></td>
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<tr>
<td>Group A2</td>
<td>Participants with a Pure Tone Average (PTA) of 56 to 70 dB HL</td>
</tr>
</tbody>
</table>

Efficacy of NAL-NL1 in Hg. Aid users

(N=8)

| Group A3 (N=7) | Participants with a Pure Tone Average (PTA) of 71 to 90 dB HL |

Based on type of hearing loss in the aided ear, two groups were formed:

<table>
<thead>
<tr>
<th>Group</th>
<th>Grouping Criteria</th>
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<td>Group B1 (N=12)</td>
<td>Mixed hearing loss (Bone conduction thresholds &gt;15 dB; ABG = 15 to 40 dB HL)</td>
</tr>
<tr>
<td>Group B2 (N=8)</td>
<td>Sensori-neural hearing loss (ABG &lt; 10 dB)</td>
</tr>
</tbody>
</table>

All the participants were first-time hearing aid users and had no previous experience of hearing aid use. The data were collected after they used the hearing aid for a period of at least 45 days to six months. All the participants had post-lingual onset of hearing loss, with the duration of hearing loss not greater than five years, and they spoke Kannada fluently. All the participants of the study used only digital BTE hearing aid monaurally. The aided thresholds with the selected hearing aid programmed for these individuals were within the speech spectrum.

**Instruments used**

A calibrated two channel sound field audiometer with two loud speakers to perform the aided sound field testing. The loud speakers were located at $0^\circ$ Azimuth and $180^\circ$ Azimuth, at a distance of 1 meter from the participant. A personal computer was connected to the auxiliary input of the audiometer for presentation of speech material through a CD. Personal computer was used along with HiPro, NOAH 3 and hearing aid fitting software for programming the digital hearing aid. A calibrated hearing aid analyzer was used for performing the insertion gain measurements. A questionnaire for fine tuning of hearing aid, the ‘fine tuning questionnaire’ was used. Participant’s own hearing aid was used for the study. They either used model ‘A’ or ‘B’ or ‘C’. All the three models of the hearing aid were manufactured by the same company and had the features as mentioned below:

- A two channeled fully digital BTE hearing aid suitable for hearing loss from mild to profound degree, with three programmable memories, Automatic Gain Control - Input (AGC-I) compression and output limiting
  - or

- A two channeled fully digital BTE hearing aid suitable for moderate to severe degree of hearing loss, with three programmable memories, AGC-I compression and output limiting
  - or
A single channeled fully digital BTE hearing aid with two frequency bands, suitable for hearing loss of moderate to severe degree, with three programmable memories, AGC-I compression and output limiting

**Speech Material**

Recorded phonemically balanced Kannada bi-syllabic word lists on a CD, developed by Yathiraj and Vijayalakshmi (2005) were used. Four out of the eight lists in the test material were used. Each of the lists had 25 bi-syllabic words.

**Procedure**

The testing was carried out in a two room sound treated environment. The procedure was as given below in three phases for each participant. They are:

- Phase I: Programming the participant’s own hearing aid for three test conditions
- Phase II: Measurement using subjective tests
  - A: Aided hearing threshold
  - B: Aided Speech Recognition Score (SRS)
  - C: Aided Uncomfortable level (UCL)
- Phase III: Measurement using objective tests
  - A: Real Ear Insertion Gain (REIG)
  - B: Real Ear Saturation Response (RESR)

**Phase I: Programming the participant’s own hearing aid**

For each of the participant, his/her own hearing aid model was programmed in three different settings as three different programs. The three hearing aid programs were:

- **a. Program with NAL-NL1 setting:** This is the program which was generated as ‘first fit’ by the hearing aid fitting software. ‘First fit’ settings were obtained by using the NAL-NL1 prescriptive formula, and the participant’s hearing thresholds. This was stored in Program 1 (herein after referred to as P1).

- **b. Program with participant’s preferred setting:** This was the modified program, modified at the time of hearing aid dispensing as per the participant’s needs. The settings that were noted in the scoring sheet at the time of trial (before hearing aid prescription) were used for this purpose. This was saved in Program 2 (herein after referred to as P2).

- **c. Program with fine tune setting:** The hearing aid program was modified based on a ‘Fitting Assistant Questionnaire’, designed specifically for hearing aid fitment. This
was used to fine tune the hearing aid after at least a minimum of 45 days of hearing aid use. This program was stored in Program 3 (herein after referred to as P3).

During programming, the acclimatization level was kept at a constant value of two for the above mentioned programs, viz., P1, P2 and P3. The testing was then carried out with the other two phases with each of the above mentioned programs. Both objective and subjective measurements were performed with each of the programs for ten out of 20 participants. Only subjective testing was done for the rest of the participants (10/20). All testing were done using participant’s own hearing aid and custom ear mould.

**Phase II: Measurement using subjective tests**

The subjective tests were carried out by measuring the aided thresholds, Speech Recognition Scores (SRS), and Uncomfortable Level (UCL) in the three aided conditions (P1, P2, and P3).

**II A. Aided hearing thresholds**

The aided thresholds in sound field were measured for Frequency Modulated (5% frequency modulation) tones at 250 Hz, 500 Hz, 750 Hz, 1 kHz, 2 kHz, 3 kHz, 4 kHz and 6 kHz. The tones were presented through the loud speaker located at $0^\circ$ Azimuth and 1 meter distance from test ear of the participants. Bracketing method was used to arrive at the threshold. The participant was instructed to indicate the presence of the tone, and to respond by raising the forefinger of the right hand even to the faintest tone heard by them. The lowest level in dB HL at each frequency detected by the participant was noted and tabulated as the aided threshold, with each program setting (P1, P2, P3). This was done for each participant.

**II B. Aided Speech Recognition Scores (SRS)**

The participants were seated in the calibrated position in the sound field (as mentioned in IIA), with the speech material being presented through the loudspeaker. The recorded word list was played through windows media player in the computer and was routed through the auxiliary input of the audiometer to the loudspeaker. The VU meter deviation was monitored to ensure that it did not exceed an average deflection of 0 dB on the scale. Care was taken to ensure that there was no effect of the order of the word list on SRS.

The participant was instructed to repeat the words that he/she heard. The presentation level was kept constant at 45 dB HL. During aided testing, it was ensured that this level was within the UCL of the participant. The responses were scored on a response sheet as the number of words correctly repeated. The maximum score was 25 as
the list consisted of 25 words. The SRS was measured separately for each participant with
the hearing aid programmed in three settings (P1, P2 & P3) and tabulated.

II C. Aided Uncomfortable Level (UCL)

Speech noise was presented through the loud speaker.

The level of speech noise was increased systematically from 45 dB HL in 5 dB steps. The participant was instructed to indicate the level at which the noise presented was uncomfortably loud. The instruction was to indicate the level at which the speech noise started to become uncomfortable and no longer tolerable to the participant.

The procedure was repeated two times. The average of highest values at which he/she could tolerate the noise was noted as the UCL, for each setting (P1, P2, P3), for each participant.

Thus, the aided thresholds, SRS, and UCL were established in each program setting (P1, P2 & P3) for all the 20 participants.

Phase III. Measurement using objective tests

Real ear measurements were carried out to evaluate the following, in each of the three hearing aid programs (P1, P2 & P3) for ten participants.

III.A. Real Ear Insertion Gain (REIG)

REIG is the difference in decibels, as a function of frequency, between the Real Ear Aided Gain (REAG) and the Real Ear Unaided Gain (REUG), obtained with the same measurement point and the same sound field conditions (ANSI, 1997).

Before the actual testing started, leveling of the probe system of the hearing aid analyzer was done using the reference microphone placed above the ear to ensure a smooth frequency output from the analyzer. The REIG was obtained by subtracting the REUG from REAG. The participant was seated at one foot distance and 45° Azimuth from the loudspeaker of the real ear analyzer.

Measurement of REUG

To ensure proper insertion depth of the probe tube, the probe tube was placed in the ear canal, so that the tube rested along the bottom of the canal part of the ear mold, with the tube extending at least 5 mm (1/5 inch) past the ear mold. The target curve was created in the real ear analyzer by entering the participant’s audiometric data into the instrument and selecting the NAL prescriptive procedure.
Protocol for REUG

- Type of stimulus: International Collegium for Rehabilitative Audiology (ICRA, a temporally modulated signal) digital speech signal.
- Level of stimulus: 60 dB SPL for REUG, REAG, and 90 dB SPL for RESR.
- Reference microphone: On
- Smoothing: Off
- Output limit: 125 dB SPL
- Test type: Insertion Gain

The probe tube microphone in the ear canal picked up and measured the sound in the unoccluded ear canal. The Real Ear Unaided Gain (REUG) was measured and displayed as dB at different frequencies.

Measurement of REAG

Probe tube was placed in the ear canal, so that the tube rested along the bottom of the canal part of the ear mold, with the tube extending at least 5 mm (1/5 inch) past the canal opening as explained in the REUG measurement. The hearing aid was fitted into the participant's ear while holding the probe tube so that its position in the ear canal was not disturbed.

The hearing aid was turned ‘on’.

Protocol for REAG

- Type of stimulus: ICRA speech signal
- Level of stimulus: 60 dB SPL
- Reference microphone: On
- Smoothing: Off
- Output limit: 125 dB SPL
- Test type: Insertion Gain

For an ICRA speech signal presented at 60 dB SPL, the probe tube microphone measured the dB SPL in the ear canal as delivered by the hearing aid. The Real Ear Aided Gain (REAG) was displayed as a curve with frequency versus dB.

REIG

The real ear analyzer automatically displayed the REIG across frequencies. This was done by the instrument, by subtracting the REUG from the REAG. The values of REIG were noted down from the data table at 200 Hz, 500 Hz, 700 Hz, 1 kHz, 1.5 kHz, 2 kHz, 3 kHz, 4 kHz, 6 kHz, and 8 kHz, in each program setting (P1, P2, and P3), for each participant.
**III B. Real Ear Saturation Response (RESR)**

Location of the participant and the loud speaker of the hearing aid analyzer were the same as that for REIG. The placement of probe tube in the ear canal and the hearing aid was the same as that for REAG. The volume of the hearing aid was set to the highest position just before feedback or projected use setting.

**Protocol for RESR**
- Type of stimulus: ICRA speech signal
- Level of stimulus: 90 dB SPL
- Reference microphone: On
- Smoothing: Off
- Test type: SPL

The probe tube microphone measured the dB SPL in the ear canal as delivered by the hearing aid. The Real Ear Saturation Response (RESR) was displayed as a curve with frequency versus dB SPL. Values of RESR were noted down from the data table at 200 Hz, 500 Hz, 700 Hz, 1 kHz, 1.5 kHz, 2 kHz, 3 kHz, 4 kHz, 6 kHz, and 8 kHz for each program setting (P1, P2, and P3), for each participant.

Thus, the REIG and RESR were noted down for each of the ten participants, at three different hearing aid program settings, i.e., Program 1 (P1), Program 2 (P2), and Program 3 (P3).

**Statistics**

Statistical Package for Social Sciences, SPSS (version 15) was used for analysis of the data. To examine if there was any difference between these groups of participants an independent samples t-test was run initially. The results revealed no significant difference between performance of sensori-neural and mixed hearing loss groups, hence these two groups were considered as a homogenous group for the rest of the study.

Independent samples t-test was again run for the groups of participants with different degrees of hearing loss and the three groups were found to be statistically different for a few frequencies. Based on this result, the three sub-groups with different degrees of hearing loss have been considered separately for statistics. For all the sub-groups considered henceforth, Friedman’s test was done initially to check if there was any significant difference between the three programs (P1, P2 & P3). If a significant difference existed, then the Wilcoxon’s test was administered to know which of the three programs differed significantly from each other.

**Results and Discussion**

Comparison of the three programs (P1, P2 & P3) is done on subjective (aided thresholds, SRS and UCL) and objective (REIG and RESR) measures.
**Aided Thresholds:**

For all the groups of participants, viz. moderate, moderately severe and severe hearing loss, the aided thresholds for the lower frequencies were better and became progressively poorer at the higher frequencies in all the three hearing aid programs. This can be attributed to the greater hearing loss usually seen at higher frequencies than at the lower frequencies and also to the limited ability of the hearing aids to provide more amplification at the higher frequencies.

Fig. 1: Average aided thresholds at different frequencies for the groups with moderate, moderately-severe and severe hearing loss for P1, P2 & P3.

In all the three groups, i.e., moderate, moderately severe, and severe, P1 also showed equal or better thresholds than P2, though not statistically significant, for most of the frequencies (more so in the moderate and moderately severe groups), as can be noted from Figures 1. This goes on to prove that participants in the study, even with no previous experience, preferred a hearing aid setting that provided them with improved thresholds when compared to the NAL-NL1 setting, except at 250 Hz and 500 Hz.

Table 1: Programs which differed significantly from each other at various frequencies for the group with moderate hearing loss

<table>
<thead>
<tr>
<th>Frequency (Hz)</th>
<th>Pairs which were significant from Wilcoxon’s test</th>
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<tbody>
<tr>
<td></td>
<td>Moderate</td>
</tr>
<tr>
<td>250</td>
<td>-</td>
</tr>
<tr>
<td>500</td>
<td>-</td>
</tr>
<tr>
<td>750</td>
<td>-</td>
</tr>
<tr>
<td>1000</td>
<td>P2 &amp; P3**</td>
</tr>
<tr>
<td>1500</td>
<td>-</td>
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<tr>
<td>2000</td>
<td>-</td>
</tr>
<tr>
<td>3000</td>
<td>-</td>
</tr>
<tr>
<td>4000</td>
<td>-</td>
</tr>
<tr>
<td>6000</td>
<td>P1 &amp; P2**</td>
</tr>
</tbody>
</table>

** p< 0.05, *p<0.1
Significant differences between P2 and P3 at various frequencies across the three groups (as shown in Table 1) indicated better performance from the participants with fine tuning of the hearing instrument rather than with the NAL-NL1 setting.

**Speech Recognition Scores**

Speech Recognition Scores (SRS) were also compared for P1, P2 and P3 across the three groups. Mean and Standard Deviation for SRS are given below in Table 2. From this table it can be observed that the SRSs were highest with P3 settings, in all the three groups.

**Table 2: Mean and Standard Deviation (SD) of aided SRS across three hearing aid programs (P1, P2, & P3) for Moderate, Moderately Severe and Severe hearing loss groups**

<table>
<thead>
<tr>
<th>Program</th>
<th>Moderate</th>
<th>Moderately Severe</th>
<th>Severe</th>
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<tbody>
<tr>
<td></td>
<td>Mean</td>
<td>S.D.</td>
<td>Mean</td>
</tr>
<tr>
<td>P1</td>
<td>22.40</td>
<td>1.14</td>
<td>22.00</td>
</tr>
<tr>
<td>P2</td>
<td>22.80</td>
<td>0.84</td>
<td>21.50</td>
</tr>
<tr>
<td>P3</td>
<td>23.80</td>
<td>0.45</td>
<td>22.75</td>
</tr>
</tbody>
</table>

However, Results revealed no significant difference in SRS across the three programs, even at 0.05 level of significance, except for SRS in those with severe hearing loss. In this group, the SRS was significantly higher with P3 than with NAL-NL1. This implied comparable speech recognition provided by the three program settings used in the experiment with all the three programs in groups with moderate and moderately-severe hearing loss.

This can also be attributed to ‘ceiling effect’, i.e., the hearing aids had reached their optimum performance with the first program itself and hence no statistically significant improvement was noted in SRS, though there were changes in aided threshold across programs. For the group with severe hearing loss, significant difference (at p<0.1 level) was seen in SRS between programs P2 and P3, indicating a significant improvement in SRS from P2 to P3, again displaying improved performance of participants with the fine tuned settings over NAL-NL1 setting.

**Uncomfortable level:**

Uncomfortable level (UCL), like SRS, was compared for P1, P2 and P3 across the three groups. The level at which the loudness was uncomfortable were higher in P3 than in P2 in all the groups of participants. These differences were not statistically significant as revealed by Friedman's test, implying that comparable uncomfortable levels were
Efficacy of NAL-NL1 in Hg. Aid users

provided by the three program settings, i.e., P1, P2 and P3. The ceiling effect could have been a reason for this, as in SRS.

Also, unlike SRS, no trend in improvement of UCL was seen from P2 to P3 and the results remain inconclusive regarding the trend of UCL with the three programs, P1, P2, and P3 because the speech noise remained tolerable at the maximum limits of audiometer in all the three programs for all the participants.

Objective tests, i.e., REIG and RESR, were also carried out for 10 of the 20 participants. The Mean and SD of the REIG revealed variations across the three programs and across the tested frequencies.

Real Ear Insertion Gain (REIG):

The insertion gain provided by the hearing aid was greater at the mid frequencies than at the lowest or the highest frequencies in all the three program settings, i.e., P1, P2, and P3 for the three groups of participants, as shown in Figure 2. This shows the importance that is given to these frequencies for speech perception not just while calculating the generic formula (NAL-NL1, P2), but also by the participants themselves while selecting the tailor made program for their hearing aid (P1 and P3). This also reflects the lesser efficiency of the hearing aid in amplifying the low and high frequencies.

Fig. 2: Average REIG for groups with moderate, moderately-severe and severe hearing loss for P1, P2, and P3 at tested frequencies

Owing to the lesser number of participants in the moderate group (N=2), no statistical tool could be applied to check the influence of the three programs on REIG. However, as can be noted from Figure 2, the fine tune program gave higher insertion gain than the NAL-NL1 program through 1 kHz to 6 kHz, i.e., at frequencies, which are important for perception of speech.

For the group of participants with moderately-severe hearing loss the insertion gain again showed similar pattern, i.e., greater gain for the mid frequencies as compared to the lower and higher frequencies for P1, P2, and P3. However, in this group, no statistically significant difference was noted between P1, P2 and P3. This indicated that for the group with moderately severe hearing loss, the gain provided by the prescriptive
formula, NAL-NL1, approximated the actual gain preferred by the participant, even after some days of hearing aid use.

Friedman’s test was administered for the group of participants with severe hearing loss to examine if there was any significant difference in REIG at different frequencies. Significant difference was seen at 250 Hz and 8 kHz. Further, at 200 Hz, the Wilcoxon’s test revealed a significant difference between the preferred setting (P1) and NAL-NL1 setting (P2). Also significant difference between preferred setting (P1) and fine tune setting (P3). Similarly, at 8 kHz, significant differences were noted between P1 and P2, and also between P2 and P3.

This difference, as can be noted from Figure 2, is in reverse direction, with NAL-NL1 program giving higher gain at both these frequencies than the preferred and fine tune programs. This implies that the NAL-NL1 formula over estimated the gain needed at these frequencies. For many participants, at 200 Hz the mean REIG for P1 approximated zero as for many participants in this group, the REIG was in negative values, which nullified the mean REIG. This is because the hearing aid does not significantly amplify at very low frequencies.

At other frequencies, though statistically insignificant, greater REIG was seen for P3 than P2 which shows that the insertion gain was higher with the fine tune program than that provided by the generic NAL-NL1 formula.

**Real Ear Saturation Response (RESR):**

As can be observed from Figure 3, no general pattern or behaviour could be attributed to the RESR across the various degrees of hearing loss. As in the case of REIG, here also greater output across the three programs was observed for frequencies of interest, though the difference between low, mid and high frequencies was not as pronounced as seen in REIG. The RESR was higher with P3 compared to P2 (NAL-NL1) except at frequencies below 700 Hz.

![Fig. 3: Average RESR for group with moderate, moderately severe and severe hearing loss for P1, P2, and P3 at different frequencies](image)

The RESR for the group with moderate hearing loss showed a similar trend as that of REIG responses in the same group. That is, the RESR was higher in the mid
frequencies than at low and high frequencies, in all the three programs. The group with moderately severe hearing loss consistently showed greater RESR values across frequencies (except at 8 kHz) for P3 compared to P1, as can be noted from Figure 3.

This indicates that the hearing aids with fine tune programs are better equipped to work at higher input levels without causing discomfort to the wearer or causing feedback. Though visible from the figure, this difference was not statistically significant (on Friedman’s test) throughout the entire frequency range.

The group of participants with severe hearing loss showed a clear advantage of P3 over P2, as evident from Figure 3. From this figure, it can be observed that the RESR was higher with P3 except at 8 kHz. However, statistically significant difference from Friedman’s test in RESR across programs was revealed only at 4 kHz and 6 kHz. The fine tune setting gave the maximum values of RESR proving the efficacy of fine tuning procedure.

Comparison of aided threshold, REIG and SRS:

As can be noted from the results of REIG, the gain provided by the hearing aid was greater for P3 when compared to P1 and P2 for most of the frequencies and in all the three participant groups. This is comparable with the results for aided threshold and speech recognition scores. The performance was better for the P3 program when compared to P1 and P2, thus indicating that fine tune setting gave higher insertion gain, resulting in improved aided thresholds and better speech recognition.

Comparison of UCL and RESR:

On examining the UCL values, one can notice higher UCL for all the three participant groups for P3, though not statistically significant, which comparable to greater RESR values is for the three groups of participants for P3 when compared to P2. This proves the advantage that P3 provided over P2. The RESR values have effect on UCL, i.e., higher the RESR, higher will be the UCL, and better will be the Dynamic Range (DR).

The overall results of this study stand at odds with the study by Keidser and Dillon (2006), where they reported NAL-NL1 as being too loud for the first time hearing aid users. As can be noted from REIG and RESR results, the insertion gain as well the saturation response at the time of initial fitting, i.e., with P1, was greater than the gain prescribed by NAL-NL1 (P2), indicating an acceptance of greater loudness than that prescribed by the generic formula in question. Though this difference cannot be proved statistically, which could be attributed to the lesser number of participants in the study, it is recommended that further research be carried out to confirm these results.

However, current study gains support from the work of Arlinger, Lyregaard, Billemark, and Oberg (2000), where they found no correlation between preference and
audiological variables. The results of present study also support that reported by Stelmachowicz, Dalzell, Peterson, Kopun, Lewis, and Hoover (1998), who proved that a proprietary formula, which was a statistical summary of the gains actually used by wearers was more accurate than the generic formulae (DSL i/o, and FIG6), which either over- or under- amplified for the degree of hearing loss.

The three programming parameters across all the three groups revealed a negligible difference. However, these negligible changes in the programming parameters brought about some significant changes in the perception (as tested subjectively) and real ear measures (as measured objectively). These results are important as they show that even a minimal change in hearing aid programming parameters can either improve or adversely affect the performance of the hearing aid and can also affect the benefit that the hearing aid user receives from it. Thus, this may also have an effect on the continued use of the hearing aid. Keidser, and Grant (2001), in their study to compare the performance of NAL-NL1 and IHAFF, had reported that even when the difference between the two fittings was small, the subjects preferred and performed better with one program compared to other, proving that even an insignificant change in program is important in terms of subjective results and continued use of hearing aids.

As earlier discussed, most of the results are in favour of the fine tune setting as compared to the preferred settings and the NAL-NL1 setting. It can be safely concluded that the programming done using the ‘Fine tuning questionnaire’ for fine tuning hearing aids gives better results than the NAL-NL1 program.

Participant’s preferred program settings, at the time of first hearing aid trial, usually gave unsatisfactory results with lot of overshooting of the various parameters tested, which can be attributed to the inexperience on the part of the hearing aid user, as they tend to demand greater amplification at the time of first trial of hearing aid. This was observed as all the participants in this study were naïve hearing aid users. However, since only slight changes were present in the programming parameters and only at a few frequencies, the NAL-NL1 can still be considered as the base formula on which changes can be incorporated.

The findings of the present study prove that fine tune program provides better results when compared to NAL-NL1. Re-programming according to individual’s listening needs can enhance the benefit that one can obtain from the hearing aid. Hence, follow-up for fine tuning of hearing aid should be considered as an integral part of hearing aid prescription procedure for greater user satisfaction and continued hearing aid use. Also, these results can guide us in determining the possible changes in programming parameters, resulting in more client-oriented hearing aid setting on the first trial itself.

However, since only slight changes were present in the programming parameters and only at a few frequencies, the NAL-NL1 can still be considered as the base formula on which changes can be incorporated.
The findings of the present study have important clinical implications. The importance of follow-up and fine tuning can be emphasized for obtaining greater benefit from the hearing aid. The information on importance of fine tuning will be useful for hearing aid dispensing audiologists to enhance their knowledge on the probable changes that may occur in the programming over a period of time. Comparing and contrasting the changes occurring over time will endure continued use of the device.

The present study also has certain recommendations for future investigations. Extensive study with different types of hearing loss, and different degrees of hearing loss can help us identify the pattern of changes required in hearing aid parameters, which can be incorporated at the time of first fit itself, hence eliminating the disuse of hearing aid. Also, it is recommended to study such effects with different types of hearing aids, using different technologies. Such studies will help us to know if technology has an effect on the changes that occur in user preference with hearing aid usage for a period of time.

References


Auditory Learning Manual for Malayalam Speaking Children with Hearing Impairment

Asha Manoharan & Asha Yathiraj

Abstract

Hearing is a powerful sensory modality, which enables language development in a natural way. Auditory impairment in an individual can seriously impede their ability to communicate. The aim of the present study was to develop an auditory learning manual for Malayalam speaking children with hearing impairment, and to check its usefulness on the target population. In addition, the study aimed at comparing the performance between a group of younger and older children with hearing impairment using the developed material. The participants included two groups of children with hearing impairment aged 4 to 6 years and 6; 1 to 10 years. They were selected only if they had a language ability of at least 4-6 years and were exposed to Malayalam from early childhood. Their aided speech spectrum was within 40-50 dB HL and their listening age was 1 year or more. The study was carried out in two phases. Phase I dealt with the development of the material for the auditory learning manual and Phase II was concerned with administration of the developed material on Malayalam speaking children with hearing impairment. The results of the present study showed that the performance of the younger group was better than the older ones in all the tasks except detection which can be attributed to the ease of activity. From the study it can be considered that all the tasks developed in the manual can be carried out by children with hearing impairment, who speak Malayalam. This can be administered on children as young as 2; 6 years.

Introduction

Hearing is a powerful sensory modality, which enables language development in a natural way. Auditory impairment in an individual can seriously impede their ability to communicate. As audition plays an important role in communication, audiological rehabilitation represents an extremely important process. One component of audiological rehabilitation that has been reported in the literature is auditory verbal training/auditory verbal learning, which aims at maximum use of a child’s residual hearing. (Schow and Nerbonne, 1996).

Ling (1976); Erber (1982) reported that the auditory sense is the preferred sense to teach children with hearing impairment since it has been found to be the fastest, easiest and most direct means to acquire spoken language. Children with normal hearing develop speech and language as a result of auditory input combined with communicative experience. The auditory channel is the modality through which self monitoring of speech is done. The majority of children with hearing impairment have been noted to have some amount of residual hearing, which can be made use through auditory training. A critical factor considered in the acquisition of oral language for children with hearing impairment was the amount and quality of auditory experience.

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Ling and Ling (1978) and Sanders (1993) opined that auditory training was an integral part of language and speech training. Ling (1986) recommended the term ‘auditory learning’ rather than ‘auditory training’. This would enable those with hearing impairment to refine their perception of acoustic events that they heard naturally. Thus, the essence of auditory learning is considered to be learning through listening (Beebe, Pearson & Koch, 1984).

As reported by Ling and Ling (1978) and Pollack (1964) the focus of auditory learning is on maximizing the use of audition rather than vision, simply because audition is the most efficient and appropriate sense for speech reception and for developing functional verbal communication skills. Kretschmer (1974), Ling (1975) and Paterson (1982) suggested that auditory learning or the ability to use auditory channel to deduce meaning, must occur in the context of daily conversation and language learning.

The most effective stimulus which can be used to stimulate the auditory mechanism is speech because it serves as an effective mode of communication. Also, speech is the preferred stimulus for stimulation since it has been found by Ling (1976) that the use of non-speech stimuli do not help in the perception of speech stimuli. This was reported to happen because the acoustics of non-speech signals are different from speech. Also the two types of sounds are processed in different hemispheres of the brain; speech pattern provides a greater range of contrasts and similarities; training involving speech is a direct approach and; more precise and more durable auditory discrimination and identification skills results when a child’s speech system is employed as a part of the listening process. The acoustical stimuli used to stimulate the auditory mechanism should have a variety of acoustical characteristics such as durational and frequency characteristics. This is to ensure that the training given enables the children with hearing impairment perceive speech sounds of different acoustical characteristics.

It is preferable that training be imparted in a meaningful context rather than in nonsense syllables. Hence, it becomes essential to have language specific manuals for listening training for children with hearing impairment. While manuals have been developed in India in Tamil by Anjana (1998), in Indian English by Anitha (2002) and in Kannada by Vijayalakshmi and Yathiraj (2008), no such manuals has been developed in Malayalam.

The aim of the present study was to develop an auditory learning manual for Malayalam speaking children with hearing impairment, and to check its usefulness on the target population. In addition, the study aimed at comparing the performance between a group of younger and older children with hearing impairment using the developed material.
Method

Participants: The participants included two groups of children with hearing impairment aged 4 to 6 years and 6; 1 to 10 years. They were selected only if they had a language ability of at least 4-6 years and were exposed to Malayalam from early childhood. Their aided speech spectrum was within 40-50 dB HL and their listening age was 1 year or more. None of them had any additional disability.

Procedure: The study was carried out in two phases. Phase I dealt with the development of the material for the auditory learning manual and Phase II was concerned with administration of the developed material on Malayalam speaking children with hearing impairment.

Phase I: Development of the material for auditory learning manual

Selection of material for the manual:

Initially, the phonemes of Malayalam were classified as low, mid and high frequency depending on the energy concentration of the major perceptual cues. Meaningful words comprising of these low, mid and high frequency consonants and vowels were selected from Malayalam preschool and Grade-I books. Simple phrases, sentences and stories were also constructed. The material for the manual was also collected from the caregivers of children in the age range of 2; 6 years to 4 years.

This material was administered on the 20 typically developing children aged 2; 6 years to 4 years. This was done to check if the vocabulary was familiar to the children. Those words which were identified correctly by more than 80% of the children were considered for the construction of the manual. Using the vocabulary, phrases, sentences and stories that were familiar, the manual was developed.

Content of the manual

The manual was divided in five sections progressing from a simple level to a more complex level (Figure A). The five sections were:

Section I – Detection
Section II – Discrimination
Section III – Identification
Section IV – Comprehension
Section V – Memory and Sequencing

Each section was designed to have one or more lessons. The number of lists varied from lesson to lesson (Figure A).
Section I: Detection Tasks

Lesson I
List 1

Section II: Discrimination Tasks

Part 1                                                                 Part 2
Lesson 1  Lesson 2  Lesson 3  Lesson 4  Lesson 5  Lesson 6  Lesson 7  Lesson 8
List 1    List 1    List 1    List 1    List 1    List 1    List 1    List 1
List 2    List 2    List 2    List 2    List 2    List 2    List 2    List 2

Section III: Identification Tasks

Lesson 9  Lesson 10  Lesson 11  Lesson 12  Lesson 13
List 1    List 1    List 1    List 1    List 1
List 2    List 2    List 2    List 2    List 2
List 3    List 3    List 4    List 5    List 3
List 1    List 4    List 5    List 4    List 4
List 2    List 4    List 5    List 5    List 5

Section IV: Comprehension Tasks

Lesson 14  Lesson 15  Lesson 16
List 1    List 1    List 1
List 2    List 2    List 2
List 3    List 3    List 3
List 4    List 5

Section V: Memory and Sequencing

Lesson 17  Lesson 18
List 1    List 1
List 2    List 2
List 3    List 3

Figure A: Flow chart of the material used for training
Section-I (Detection)

**Lesson 1: Detection of verbal stimuli**

This section checks the ability of the children to perceive the presence or absence of verbal stimuli having different frequency characteristics. It has a list of 12 words, representing high, mid and low frequency phonemes.

Section-II (Discrimination)

This section had two parts:

Part I involved the discrimination between words of varying length of utterances while part II dealt with the discrimination of words varying in frequency characteristics.

Part I: Discrimination between words of varying length of utterance

It had four lessons each having two lists. The lessons had words of varying length of utterances, are arranged in hierarchal order. The initial lessons required the discrimination of utterances with larger difference in length, while this difference reduced in the later lessons. The four lessons were as follows:

- **Lesson 2** - Discrimination of bisyllabic words versus 9-10 syllables phrases/sentences
- **Lesson 3** - Discrimination of 3 - 4 syllabic words versus 7 - 8 syllable phrases
- **Lesson 4** - Discrimination of bisyllable words versus trisyllable phrases/ sentences
- **Lesson 5** - Discrimination of words which differ in vowel length

Part II- Discrimination between words based on frequency characteristics

This part has three lessons which are described below:

- **Lesson 6** - Discrimination between words which differ in frequency characteristics of both consonants and vowels
- **Lesson 7** - Discrimination between words which differ in frequency characteristics of the vowels
- **Lesson 8** - Discrimination between words which differ in frequency characteristics of consonants.

Section – III (Identification)

This section had five lessons:

- **Lesson 9** - Identification of simple words which differ in frequency characteristics
- **Lesson 10** - Identification of minimal pairs of words which differ in duration / frequency
characteristics of vowels

**Lesson 11** - Identification of minimal pairs of word which differ in frequency characteristics of consonants

**Lesson 12** - Identification of non-minimal paired key words which differ in frequency characteristics of the consonants, in a sentence context

**Lesson 13** - Identification of minimal paired key words which differ in frequency characteristics of the consonants, in a sentence context

**Section IV (Comprehension)**

The section regarding comprehension had three different lessons:

**Lesson 14** - Comprehension of related sentences

**Lesson 15** - Comprehension of commands which differ in frequency characteristics of consonants

**Lesson 16** - Comprehension of unrelated questions

Two lists of items are included under this lesson

**Section V (Memory & Sequencing)**

This section had two lessons:

**Lesson 17** - Auditory memory and sequencing of words within a lexical category, embedded in a sentence

**Lesson 18** - Auditory memory and sequencing of words between lexical categories, embedded in a sentence

**Phase II: Administering the manual on target population**

The developed manual was administered on ten children with hearing impairment in the range of 4 to 10 years. They were divided into two age groups (4 to 6 years and 6; 1 to 10 years).

Each child was evaluated independently. They were seated at a distance of 3-4 feet from the clinician in a quiet room, free from distraction. All the lessons, starting from lesson I, were administered on each participant. Prior to each lesson appropriate instructions were provided. For children who did not follow the initial instruction, the same was explained using simpler language. The instructions varied depending on the tasks.

The responses were noted for each child separately. The number of sessions for a child varied between 5 and 7, with each session lasting for duration of about 45 minutes.
A score of ‘1’ was given for every correct response and a score of ‘0’ was given for a wrong response.

**Results and Discussion**

The responses of the ten children with hearing impairment, who were administered the auditory learning manual are provided and discussed. A comparison was made between the two age groups (4 to 6 years and 6; 1 to 10 years), using SPSS version 15.0, using the statistical test used was Mann Whitney U-test. The results are discussed under five sections for all eighteen lessons.

**Section I: Detection**

**Comparison of the Group I (younger group) with Group II (older group) on the detection of verbal stimuli**

The Mann Whitney U-test indicated that there were no significant differences in the performance of the two age groups, with both the groups obtaining 100% scores. This reveals that children with hearing impairment as young as four years of age are able to do detection tasks as well as the older group with hearing impairment. This can occur as long as children with hearing impairment have aided audiogram within the speech spectrum. Much earlier Goldstein (1939) and Ling and Ling (1978) reported that when properly aided, children who are hearing impaired can detect sounds effectively if their hearing is within the speech spectrum. Similar results were also reported by Anitha (2000) and Vijayalakshmi and Yathiraj (2008).
Section II: Discrimination

Comparison of Group I with Group II on the discrimination of varying length of utterances

All the participants in the Group I obtained significantly higher scores (72% to 90%) than Group II (64% to 78%) indicating that the temporal cue were more easily perceived by the former group. In general, as the task got more complex, the performance of the participants from both groups reduced. These results are in agreement with the findings of Anitha (2002) and Vijayalakshmi and Yathiraj (2008), who had also reported that temporal cues are easily perceived by individuals with hearing impairment when compared to spectral cues.

Comparison of Group I with Group II on the discrimination of words differing in the frequency characteristics of consonants and vowels as well as vowels alone.

There was no significant difference in performance between the two age groups on the discrimination of words that differed in the frequency of vowels and consonants as well as vowels alone. Though there was no significant difference between the groups, the younger group performed slightly better. This indicates that children as young as 4 years are able to discriminate words that have gross frequency differences. Similar results were reported by Vijayalakshmi and Yathiraj (2008) who found no significant difference between children aged 4 to 5 years and 5 to 12 years on similar tasks. In contrast, Anitha (2002) reported a difference performance in children aged 4 to 5 years and 6 to 13 years on a similar task. Her younger group performed better than the older ones. She attributed the superior performance of younger group to the early stimulation received by them.

Comparison of Group I and Group II on the discrimination of words differing in the frequency characteristics of consonants

A significant difference (p < 0.05) in the discrimination of words differing in the frequency characteristics of consonants was observed between the two groups. The performance of the younger group was superior to the older group. This could be due to the early training in speech perception skills for the younger group (Group I). However, there was no significant difference between the two groups for the discrimination of high frequency consonants, with the performance of the younger children dropping. This probably occurred since the task required finer discrimination of consonants in the high frequency region which both groups found difficult. As the complexity of the tasks increased, the performance went down for Group I, but remained almost the same for Group II. Similar results were observed by Anitha (2002) and Vijayalakshmi and Yathiraj (2008).
Section III: Identification

Comparison of Group I and Group II on the identification of words which differ in frequency characteristics

In these tasks the results shows that there were significant differences in the performance by both the age groups for identification of words with low and high frequencies, with the younger children out performing the older group. However, for identifying mid frequency words, both groups performed almost equally well. On all the above three lists, the younger group (Group I) scored above 90%, whereas the performance of the older group ranged from 67% to 87%. Further, it was observed that all children required lesser time to carry out the identification tasks when compared to the earlier discrimination tasks. As mentioned by Anitha (2000) and Vijayalakshmi and Yathiraj (2008), this was probably due to the type of training provided to the children. The emphases of the training programs that the children had undergone were more on identification activities rather than discrimination tasks.

![Bar chart showing mean percentage score for Group I and Group II across three lists.](chart.png)

Comparison of Group I and Group II on identification of minimal pairs which differ in vowels

There was no significant difference in the performance between the two groups. This indicates that the younger (group I) and older (group II) children with hearing impairment perceived minimal pairs which differed in terms of the vowels equally well. However, both groups performed better on list 1 compared to list 2. List 1 differed in terms of the vowel length, while list 2 differed in terms of the frequency characteristics of the vowels. The better performance on list 1 reflects the children’s ability to perceive temporal cues with greater accuracy than the frequency cues.
Most of the children exhibited some difficulty in the perception of certain vowels especially the high frequency vowels. Similar results have been reported by Anitha (2000) and Vijayalakshmi and Yathiraj (2008).

**Comparison of Group I and Group II on identification of minimal paired words which differ in the frequency characteristics of consonants**

There was a significant difference in the performance between two groups in identifying minimal pairs which differed in the frequency characteristics of consonants. This was observed across all the lists except list 3. List 3 contained words differing in mid and high frequency. As was observed in the earlier lesson, the performance of the younger age group was superior compared to the older age group. Further, it was generally seen in the present study that when the frequency contrasts between the pairs of words decreased the performance of the subjects declined. This was true for both the age groups (Group I and Group II). Similar results were also reported by Anitha (2000) and Vijayalakshmi and Yathiraj (2008). This could have occurred due to a difficulty in perception of spectral cues. Revolie, Pickett and Spyket (2002) reported that spectral cue perception was difficult for those with hearing impairment. They noted that the recognition of consonants were more difficult than vowel recognition as the former varied more in frequency characteristics.

**Comparison of Group I and Group II on identification of non-minimal pair key words, which differ in the frequency characteristics in a sentence context**

There was a significant difference in the performance between the groups on this task. Once again, the younger age group performed better than the older group for all the lists. This was similar to the results reported by Anitha (2000) and Vijayalakshmi and Yathiraj (2008). As mentioned earlier this could be due to the training method the subjects had undergone. The younger children probably learnt to use their auditory abilities a lot better than the older children. It was observed that as the complexity of the tasks increased, the performance of the older group worsened. However, in the younger group the performance was similar for the first three lists, but dropped markedly in last two lists which had more complexity.

**Comparison of Group I and Group II on identification of minimal pair key words, which differ in frequency characteristics, in a sentence context**

On the easier tasks of Lesson 13 (list 1 and 2), the two groups obtained similar scores, which were relatively high. In contrast, on the more difficult task (list 3, 4 and 5) the younger children performed significantly better. Thus, both the age groups were able to identify the grossly different pairs with similar ease, where as it was considerably more difficult for the older children to carry out the more difficult tasks. The difficulty seen in children with hearing impairment in perceiving finer spectral changes is evident from these results. Based on these findings, it is suggested that more emphasis should be
given to auditory training using material that have subtle differences in frequency characteristics.

Section IV: Comprehension

Comparison of Group I and Group II on the comprehension of related questions unrelated commands and unrelated questions

No significant differences in performance between the two age groups were obtained on the comprehension of related and unrelated commands. Both groups obtained mean scores of above 88% for comprehension of unrelated commands. The variability in performances seen by both groups for all five lists of lesson 14 was very low. However, the performance of the younger group was superior to the older group for the comprehension of unrelated commands and unrelated questions. It was reasoned that the better performance of the older group could be due their higher language abilities. Children in the two groups showed difficulty carrying out the commands with high frequency consonants. This was similar to their difficulty in perceiving high frequency words. It is suggested that more emphasis should be given in training the children using similar material. Similar results were obtained by Anitha (2004) and Vijayalakshmi and Yathiraj (2008).

Section V: Auditory Memory and Sequencing

Comparison of Group I and Group II on auditory memory sequencing of words within a lexical category and between lexical categories, embedded in a sentence

The performance of the two age groups was observed to be poor for these tasks compared to the lessons discussed earlier. There was a significant difference in performance of the two age groups as the complexity of the task was increased. The overall performance of the younger group was better than that of the older ones. As the complexity of the tasks increased, the performance of all the children went down irrespective of the group. The poor performance in this lesson by two groups could be because the regular training they had undergone focused on sequencing isolated words and not in sequencing within sentences. Similar results had been reported by Vijayalakshmi and Yathiraj (2008) where the auditory memory and sequencing scores were poorer when compared to the other tasks carried out by them.

Conclusions

From the results of the present study it can be summarized that both the age groups performed equally well on the detection task. They performed the best on this task compared to all other tasks probably due to the ease of the activity. Discrimination of temporal cues was better perceived than discrimination of the spectral cues by both the groups. The younger children performed better on both temporal and spectral
discrimination tasks. On the task involving the discrimination of frequency characteristics of consonants and vowels, both groups displayed more difficulty in the discrimination of consonants when compared to the discrimination of vowels. The performance of both the groups was better in the identification lessons compared to discrimination lessons. Both the groups performed almost equally well on the comprehension tasks. The performances of both the group were comparatively poorer in the auditory memory and sequencing tasks when compared to all other tasks. It was observed that in most of the tasks, the performance of the younger group was superior to the older ones. This could be attributed to the kind of training program the younger children were enrolled in, which focused on listening skills. From the study it can be considered that all the tasks developed in the manual can be carried out by children with hearing impairment, who speak Malayalam. This can be done by children as young as 2; 6 years.

References


Estimation of Behavioural Threshold Using Click Evoked ALR in Normal and Hearing Impaired Population

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Abstract

Establishing frequency specific threshold has significant importance in reaching appropriate diagnosis and planning individualized rehabilitation program. Auditory evoked potentials are used to predict behavioural threshold in difficult to test population when audiologist fails to obtain frequency specific behavioural threshold. Among auditory evoked potentials ALR require lesser neural synchrony than ABR for recording, so ALR could be a better objective tool to estimate the threshold. The current study investigated the ALR as objective tool to assess behavioural threshold in normal hearing and other clinical population. ALR was recorded from 20 ears with normal hearing, 16 ears with sensorineural hearing and 20 ears with auditory dys-synchrony. ALR recording was initiated at 80 dBnHL and gradually reduced the intensity. The lowest intensity at which N1-P2 complex was observed was considered as ALR threshold. Statistical analysis revealed that a significant positive correlation between ALR threshold and pure tone average in sensorineural hearing loss individuals. However a weak positive correlation was obtained in normal hearing group and auditory dys synchrony group. It can be concluded that ALR can closely estimate the behavioural threshold in sensorineural hearing loss individuals.

Key words: ALR, Auditory dys synchrony.

Introduction

Auditory long latency response is an auditory evoked potential came in to the field since 1960s. However, it has failed to gain much popularity due to the explosion of interest in the auditory brainstem response (ABR). This could be because of its accuracy and reliability to predict the behavioral threshold. ALR has not gain the popularity to predict behavioural threshold as it is affected by several factors.

Hyde, Alberti, Matsumoto and Liyl (1980) reported that tone burst evoked ALR audiometry can be used specifically at approximating the pure tone audiogram for at-risk infants, difficult-to-test children, and adults with certain mental or physical handicaps. They found that ALR can be used to estimate behavioral threshold within 10 dB in at least 90% of cases and for those subjects, who are both awake and passively cooperative.

Alberti (1970) carried out evoked cortical response audiometry using tone burst in normal hearing and patients with abnormal hearing. All were neurologically normal. He found that nine of the ten normal hearing subject’s threshold was within 10 dB at the best conventional threshold tested between 500 Hz and 4000 Hz. The Patients with hearing loss showed a little greater spread from 250 Hz to 2000 Hz and the thresholds were within 15 dB. He also reported that cortical audiometry is valuable in detecting functional hearing loss. However, there are lesser number of studies which used click as stimulus for threshold estimation due to its lack of frequency specificity and short duration. However, use of long duration click could predict behavioral threshold within short period of time.

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The effect of hearing loss on ALR has been studied extensively by several authors. Polen (1984) studied the effect of hearing loss on ALR components and compared the findings with normal hearing group. He found that moderate to severe SN hearing loss resulted in prolongation of latencies of P₂, N₂ components of ALR. Decreased N₂ amplitude in the sensorineural hearing loss group in comparison to normal hearing group was also reported. However, there are inconsistencies among studies of ALR in hearing impaired subjects reported (Oates, Kurtzberg and Stapells, 2002).

Cortical potential like ALR requires different neural synchrony compared to the synchrony required for relatively shorter latency response (Kraus et al., 2000). It is possible that ABR or ALR which requires high synchronization may be disrupted in some subjects; where as low neural synchrony required for ALR may be intact. Auditory dys-synchrony is one such disorder characterized by abnormal or absent ABR and presence of OAE and / CM indicating normal functioning of OHC (Starr et al., 1991). In such condition, ALR recorded from auditory dys-synchrony clients could be an important tool to predict behavioral threshold.

Speech intelligibility is another problem consistent with sensorineural hearing loss and auditory dys-synchrony. Most of the affected adults with auditory dys-synchrony report perceptual difficulties for greater than, would be expected from their behavioral audiogram (Zeng et al., 2001 and Starr et al., 2003). Speech perception ability cannot be reliably estimated from behavioral audiogram in individuals with auditory dys-synchrony, ALR components may offer a means of predicting perceptual skills (Rance et al., 2002). Hence, present study aimed at finding out the relationship between

- Relationship between ALR threshold and pure tone average in individuals with normal hearing, sensorineural hearing loss and auditory dys-synchrony.
- Relationship between ALR threshold and speech identification score (SIS) in individuals with sensorineural hearing loss and auditory dys-synchrony.
- Relationship between click evoked ALR threshold and frequency specific pure tone threshold (250 Hz to 4000 Hz) in individuals with sensorineural hearing loss and auditory dys-synchrony.

Method

Participants: A total of 37 subjects were participated in the study. Participants were grouped in to three groups. Group I: consists of 20 ears with normal hearing from 14 individuals from 18 to 50 years of age with a mean age of 33.9 years. Group II included 16 ears with cochlear hearing loss from 12 individuals. The age range was between 18 to 60 years with a mean age of 36.4 years. Group III: This group consisted of 20 ears of 11 individuals with auditory neuropathy or auditory dys-synchrony with the age range of 18 - 50 years with a mean age of 30.1 years. Group II and III were further divided in to three
subgroups each depending upon their severity of hearing loss as mild, moderate and moderately severe for comparison.

**Instrumentation:** A calibrated double channel diagnostic Madsen Orbiter 922 (version.2) audiometer with TDH 39 ear phone and B-71 bone conduction vibrator was used to carried out pure tone audiometry. A calibrated immittance meter (Granson Stadler Inc. Tymp Star) was used to assess middle ear status and ILO 292 DP Echo port (Otodynamics, version 5) system was used for recording TEOAEs. Auditory brainstem responses and auditory long latency responses to click stimuli was recorded using Intelligent Hearing System (IHS smart EP, NSB version 2.39) evoked potential system. The click stimuli was delivered using ER-3A insert receiver.

**Procedure**

- Pure tone threshold for air conduction were obtained at octave frequencies between 250 Hz and 8000 Hz and from 250 Hz to 4000 Hz for bone conduction. Modified Hughson –West lake procedure (Carhart and Jerger, 1959) was used to obtain pure tone threshold. Speech identification score was obtained at 40 dB SL with reference to speech recognition threshold in each ear independently using phonetically balanced list developed by Mayadevi (1978)

  Speech in noise test was carried out at 0 dB SNR condition (both signal and noise were presented at 40 dB SL) using the same Phonetically balanced list.

- Tympanometry was carried out using 226 Hz probe tone frequency and acoustic Reflexes were also checked at 500 Hz, 1000 Hz, 2000 Hz and 4000 Hz pure tones, for both ipsi and contra laterally to rule out middle ear pathology. TEOAEs were recorded using nonlinear broad band clicks of 256 sweeps presented at around 75 dB peSPL to identify the presence or absence of cochlear pathology.

- After the appropriate placement of the electrodes ABR and ALR were recorded. ABR recording was done using alternate click stimulus at a rate of 30.1/sec at 90 dB nHL. While recording ALR and ABR, Inverting electrode was placed on the left mastoid (M1)/ right mastoid (M2), Ground was on either of the mastoid (M2/M1), Non inverting electrode was on the high forehead (FpZ). ABR was recorded to identify presence or absence of retrocochlear pathology.

- ALR Testing was initiated at 80 dBNHL using alternate click stimulus at a rate of 1.1/sec for normal hearing and sensorineural hearing loss group. Where-as, for auditory dys-synchrony group ALR was initiated at 90 dBNHL. Intensity was then gradually reduced if the observable ALR was noticed. Initially intensity was reduced by 20 dBNHL for normal hearing group and 10 dBNHL for sensorineural hearing loss and auditory neuropathy group till no response was obtained. Then
the intensity was increased by 5 dB till the response (N1-P2) was observed. ALR was recorded twice at each presentation level to check for replicability. The presence of N1-P2 complex at the lowest intensities, were considered as threshold.

**Results**

The mean, standard deviation (SD) and range were also calculated for each parameter for both sensorineural and normal hearing groups separately. This can be seen in the Table 1. The Kruskal wallis test was carried out for the comparison of latency of the P1, N1, P2 and amplitude of N1-P2 complex across the normal hearing, sensorineural hearing loss and auditory dys-synchrony groups. The latency value of N2 is not considered for this analysis as it was absent in majority of the auditory neuropathy cases. The result indicated that P1, N1 and P2 latency and N1-P2 amplitude were significantly different across the groups; Whereas P2 latency did not show any significant difference. In order to know the significant difference between the two groups, Man Whitney test was carried out. This can be seen in Table2

<table>
<thead>
<tr>
<th>Table 1- Mean, SD, range of the latency for each component of ALR (P1, N1, P2, and N2) and the amplitude of N1-P2 complex at different intensity levels</th>
</tr>
</thead>
<tbody>
<tr>
<td><strong>Intensity</strong></td>
</tr>
<tr>
<td></td>
</tr>
<tr>
<td><strong>P1 Latency</strong></td>
</tr>
<tr>
<td>Mean</td>
</tr>
<tr>
<td>SD</td>
</tr>
<tr>
<td>Range</td>
</tr>
<tr>
<td><strong>N1 Latency</strong></td>
</tr>
<tr>
<td>Mean</td>
</tr>
<tr>
<td>SD</td>
</tr>
<tr>
<td>Range</td>
</tr>
<tr>
<td><strong>P2 Latency</strong></td>
</tr>
<tr>
<td>Mean</td>
</tr>
<tr>
<td>SD</td>
</tr>
<tr>
<td>Range</td>
</tr>
<tr>
<td><strong>N2 Latency</strong></td>
</tr>
<tr>
<td>Mean</td>
</tr>
<tr>
<td><strong>N1-P2 Amplitude</strong></td>
</tr>
<tr>
<td>Mean</td>
</tr>
<tr>
<td>SD</td>
</tr>
<tr>
<td>Range</td>
</tr>
</tbody>
</table>
It can be seen from the Table 1 that as the intensity of the stimulus was reduced from 80 to 40 dBnHL, there was an increase in the latency of all the ALR components and decrease in the N1-P2 amplitude in both the normal hearing group and sensori neural hearing loss group. The SD and range of latency for different ALR component were more than that of amplitude of N1-P2 complex for both normal hearing and sensorineural hearing loss group. This indicates that the latency of ALR has limited clinical utility.

Table 2 - Depicts the Z-value and the significance level for all the parameters of ALR between the groups at 80 dBnHL.

<table>
<thead>
<tr>
<th></th>
<th>P1 Latency</th>
<th>N1 Latency</th>
<th>P2 Latency</th>
<th>N1-P2 Amplitude</th>
</tr>
</thead>
<tbody>
<tr>
<td></td>
<td>Z-value</td>
<td>Sig level</td>
<td>Z-value</td>
<td>Sig level</td>
</tr>
<tr>
<td>Normal Vs SNHL</td>
<td>2.14</td>
<td>0.032</td>
<td>3.02</td>
<td>0.002</td>
</tr>
<tr>
<td>Normal Vs AD</td>
<td>1.811</td>
<td>0.07</td>
<td>1.02</td>
<td>0.919</td>
</tr>
<tr>
<td>AD Vs SNHL</td>
<td>1.24</td>
<td>0.213</td>
<td>2.15</td>
<td>0.031</td>
</tr>
</tbody>
</table>

Table 2 show that between normal hearing and sensorineural hearing loss group, latency of P1, N1 and amplitude of N1-P2 differed significantly, whereas, the latency of P2 did not differ significantly. Between auditory dys-synchrony and sensorineural hearing loss group only the latency of N1 component showed significant difference. Apart from this other latency parameters and N1-P2 amplitude parameter did not show statistical significant difference. Between normal hearing and auditory dys-synchrony group, no significant difference was obtained for all the ALR parameters.

The ALR threshold obtained from different groups were then compared with the behavioral threshold. The mean, SD and range for pure tone threshold and ALR threshold were calculated for all the three groups. Pearson product moment correlation was also done to identify the relationship between the ALR threshold and pure tone average. This can be seen in the Table 3 and 4.
Table 3 - Depicts the mean, SD and range obtained for ALR threshold and PTA for all the three groups

<table>
<thead>
<tr>
<th>GROUP</th>
<th>ALRT MEAN</th>
<th>ALRT SD</th>
<th>ALRT RANGE</th>
<th>PTA MEAN</th>
<th>PTA SD</th>
<th>PTA RANGE</th>
</tr>
</thead>
<tbody>
<tr>
<td>SNHL Mild</td>
<td>49</td>
<td>5.48</td>
<td>40-55</td>
<td>32.96</td>
<td>3.21</td>
<td>30-38</td>
</tr>
<tr>
<td>(n=5)</td>
<td></td>
<td></td>
<td></td>
<td></td>
<td></td>
<td></td>
</tr>
<tr>
<td>SNHL Mod</td>
<td>62.5</td>
<td>5.00</td>
<td>55-65</td>
<td>50.38</td>
<td>6.01</td>
<td>60-70</td>
</tr>
<tr>
<td>(n=4)</td>
<td></td>
<td></td>
<td></td>
<td></td>
<td></td>
<td></td>
</tr>
<tr>
<td>SNHL Ms</td>
<td>65</td>
<td>5.00</td>
<td>60-70</td>
<td>62.34</td>
<td>4.29</td>
<td>58-68</td>
</tr>
<tr>
<td>(n=7)</td>
<td></td>
<td></td>
<td></td>
<td></td>
<td></td>
<td></td>
</tr>
<tr>
<td>AD Mild</td>
<td>85</td>
<td>8.37</td>
<td>70-90</td>
<td>34.63</td>
<td>4.39</td>
<td>26-40</td>
</tr>
<tr>
<td>(n=6)</td>
<td></td>
<td></td>
<td></td>
<td></td>
<td></td>
<td></td>
</tr>
<tr>
<td>AD Mod</td>
<td>83.33</td>
<td>11.54</td>
<td>70-90</td>
<td>46.07</td>
<td>7.74</td>
<td>41-60</td>
</tr>
<tr>
<td>(n=3)</td>
<td></td>
<td></td>
<td></td>
<td></td>
<td></td>
<td></td>
</tr>
<tr>
<td>AD Ms</td>
<td>75</td>
<td>21.21</td>
<td>60-90</td>
<td>58.3</td>
<td>0.00</td>
<td>58-58</td>
</tr>
<tr>
<td>(n=2)</td>
<td></td>
<td></td>
<td></td>
<td></td>
<td></td>
<td></td>
</tr>
<tr>
<td>Normal</td>
<td>35.25</td>
<td>6.78</td>
<td>20-40</td>
<td>7.34</td>
<td>2.43</td>
<td>3.3-12</td>
</tr>
<tr>
<td>(n=20)</td>
<td></td>
<td></td>
<td></td>
<td></td>
<td></td>
<td></td>
</tr>
</tbody>
</table>

Table 4 - Depicts the r-value and the significant level obtained between the ALR threshold and PTA for all the three groups

<table>
<thead>
<tr>
<th>GROUP</th>
<th>r value</th>
<th>Sig level</th>
</tr>
</thead>
<tbody>
<tr>
<td>Normal</td>
<td>ALR vs PTA</td>
<td>0.11</td>
</tr>
<tr>
<td>SNHL</td>
<td>ALR vs PTA</td>
<td>0.833</td>
</tr>
<tr>
<td>AN</td>
<td>ALR vs PTA</td>
<td>-0.394</td>
</tr>
</tbody>
</table>

It can be seen from the Table 3 that the ALR threshold increased gradually as the degree of sensorineural hearing loss increased. This trend was not observed in the auditory dys-synchrony group. It can also be seen that the difference between the ALR threshold and PTA reduced as the hearing loss is increased. The difference was maximum for normal hearing group. The range of ALR threshold was less in auditory dys-synchrony group even though the degree of hearing loss in this group varied from mild to moderately severe. The behavioral pure tone threshold obtained was lesser than the ALR threshold in all the groups and this was relatively better in auditory dys-synchrony group. Table 4 shows that a highly significant positive correlation obtained between ALR threshold and PTA in sensorineural hearing loss group. A weak positive correlation, but not significant was observed in normal hearing group. Where as in auditory dys-synchrony group weak negative correlation is observed. ALR may not be able to predict
behavioral threshold in auditory dys-synchrony individuals rather the presence or absence can give an idea about their processing ability. ALR can be a good tool to predict behavioral threshold in individuals with sensorineural hearing loss.

To understand the relation between ALR threshold and the speech identification scores in individuals with sensorineural hearing loss and auditory dys-synchrony. The mean, SD and range for these two aspects were calculated according to the degree of hearing loss. Pearson product moment correlation was also calculated. This can be seen in the Table 5 and 6.

Table 5 – Depicts the mean, SD and Range of ALR threshold and SIS obtained in sensorineural hearing loss and auditory dys-synchrony group

<table>
<thead>
<tr>
<th>Group</th>
<th>ALRT Mean</th>
<th>ALRT SD</th>
<th>ALRT Range</th>
<th>SIS Mean</th>
<th>SIS SD</th>
<th>SIS Range</th>
</tr>
</thead>
<tbody>
<tr>
<td>SN HL Mild</td>
<td>49</td>
<td>5.47</td>
<td>40-55</td>
<td>84</td>
<td>9.61</td>
<td>70-95</td>
</tr>
<tr>
<td>SN HL Mod</td>
<td>62.5</td>
<td>5</td>
<td>60-70</td>
<td>88.75</td>
<td>7.5</td>
<td>80-95</td>
</tr>
<tr>
<td>SN HL MS</td>
<td>65</td>
<td>5</td>
<td>60-70</td>
<td>68.57</td>
<td>17.25</td>
<td>40-85</td>
</tr>
<tr>
<td>AD Mild</td>
<td>85</td>
<td>8.36</td>
<td>70-90</td>
<td>47.5</td>
<td>12.14</td>
<td>25-60</td>
</tr>
<tr>
<td>AD Mod</td>
<td>83.33</td>
<td>11.54</td>
<td>70-90</td>
<td>55</td>
<td>25.98</td>
<td>40-85</td>
</tr>
<tr>
<td>AD MS</td>
<td>75</td>
<td>21.21</td>
<td>60-90</td>
<td>77.5</td>
<td>17.68</td>
<td>65-90</td>
</tr>
</tbody>
</table>

Table 6 - Depicts the r-value and the significance level between ALR threshold and SIS for sensorineural hearing loss and auditory dys-synchrony group.

<table>
<thead>
<tr>
<th>Group</th>
<th>ALRT Vs SIS</th>
<th>r value</th>
<th>Sig level</th>
</tr>
</thead>
<tbody>
<tr>
<td>SNHL</td>
<td>ALRT Vs SIS</td>
<td>-0.242</td>
<td>0.366</td>
</tr>
<tr>
<td>AN</td>
<td>ALRT Vs SIS</td>
<td>-0.646</td>
<td>0.023</td>
</tr>
</tbody>
</table>

It can be seen in the Table 5 that SD is more for speech identification scores in individuals with auditory dys-synchrony compared to individuals with sensorineural hearing loss. Table 6 shows that SIS and ALR had significantly negative correlation in auditory dys-synchrony group. SIS reduces as the ALR threshold increased. However, no significant correlation was obtained in individuals with sensorineural hearing loss.

To establish the relationship between ALR threshold and the frequency specific pure tone behavioral threshold, ALR threshold and pure tone threshold was obtained at each frequency from 250 Hz to 4000 Hz. The mean, SD and range were then computed. This can be seen in the Table 7.
Table 7 – depicts the mean, SD and range for ALR threshold and pure tone threshold at each frequencies for different subgroups of sensorineural hearing loss and auditory dys-synchrony group

<table>
<thead>
<tr>
<th>Frequency</th>
<th>Mild SN</th>
<th>Mod SN</th>
<th>Ms SN</th>
<th>Mild AN</th>
<th>Mod AN</th>
<th>Ms AN</th>
</tr>
</thead>
<tbody>
<tr>
<td>250 Hz</td>
<td>Mean</td>
<td>19</td>
<td>37.5</td>
<td>52.86</td>
<td>40.83</td>
<td>40</td>
</tr>
<tr>
<td></td>
<td>SD</td>
<td>7.42</td>
<td>10.41</td>
<td>4.88</td>
<td>8.01</td>
<td>8.66</td>
</tr>
<tr>
<td></td>
<td>Range</td>
<td>30-40</td>
<td>25-50</td>
<td>50-60</td>
<td>30-50</td>
<td>35-50</td>
</tr>
<tr>
<td>500 Hz</td>
<td>Mean</td>
<td>24</td>
<td>43.75</td>
<td>55.71</td>
<td>43.33</td>
<td>45</td>
</tr>
<tr>
<td></td>
<td>SD</td>
<td>6.51</td>
<td>9.46</td>
<td>6.07</td>
<td>8.16</td>
<td>13.23</td>
</tr>
<tr>
<td></td>
<td>Range</td>
<td>20-35</td>
<td>30-50</td>
<td>50-65</td>
<td>30-55</td>
<td>35-60</td>
</tr>
<tr>
<td>1 kHz</td>
<td>Mean</td>
<td>32</td>
<td>48.75</td>
<td>62.1</td>
<td>35.83</td>
<td>45</td>
</tr>
<tr>
<td></td>
<td>SD</td>
<td>2.73</td>
<td>2.5</td>
<td>5.67</td>
<td>3.76</td>
<td>8.66</td>
</tr>
<tr>
<td></td>
<td>Range</td>
<td>30-35</td>
<td>45-50</td>
<td>55-70</td>
<td>30-40</td>
<td>40-55</td>
</tr>
<tr>
<td>2 kHz</td>
<td>Mean</td>
<td>44</td>
<td>56.25</td>
<td>70</td>
<td>25</td>
<td>43.33</td>
</tr>
<tr>
<td></td>
<td>SD</td>
<td>2.25</td>
<td>4.79</td>
<td>10.4</td>
<td>7.07</td>
<td>5.77</td>
</tr>
<tr>
<td></td>
<td>Range</td>
<td>40-45</td>
<td>50-60</td>
<td>55-80</td>
<td>15-35</td>
<td>40-50</td>
</tr>
<tr>
<td>4 kHz</td>
<td>Mean</td>
<td>53</td>
<td>63.75</td>
<td>82.14</td>
<td>23.33</td>
<td>46.67</td>
</tr>
<tr>
<td></td>
<td>SD</td>
<td>10.37</td>
<td>13.15</td>
<td>12.20</td>
<td>2.58</td>
<td>2.89</td>
</tr>
<tr>
<td></td>
<td>Range</td>
<td>40-65</td>
<td>45-75</td>
<td>65-95</td>
<td>20-25</td>
<td>45-50</td>
</tr>
<tr>
<td>ALRT</td>
<td>Mean</td>
<td>49</td>
<td>62.5</td>
<td>65</td>
<td>85</td>
<td>83.3</td>
</tr>
<tr>
<td></td>
<td>SD</td>
<td>5.48</td>
<td>5.00</td>
<td>5.00</td>
<td>8.37</td>
<td>11.55</td>
</tr>
<tr>
<td></td>
<td>Range</td>
<td>40-55</td>
<td>55-65</td>
<td>60-70</td>
<td>70-90</td>
<td>70-90</td>
</tr>
</tbody>
</table>

It is evident from the Table 7 that the ALRT is in close approximity to mid frequency pure tone threshold in sensorineural hearing loss group. Whereas pure tone threshold observed at 4 KHz is seem to be higher than the click evoked ALR threshold. However, in auditory dys-synchrony group, ALR threshold was much higher than any frequency pure tone threshold except for the individuals with moderately severe (Ms) hearing loss. Pearson product moment correlation was done to find out the correlation between ALR threshold and each frequency pure tone threshold. The results of the Pearson product moment correlation coefficient can be seen in the Table 8
Table 8 - Depicts the r value and the significance level between the ALR threshold and pure tone threshold at different frequencies for sensorineural hearing loss group and auditory dys-synchrony group.

<table>
<thead>
<tr>
<th>Group</th>
<th>r value</th>
<th>Sig level</th>
</tr>
</thead>
<tbody>
<tr>
<td>SNHL</td>
<td>ALRT vs 250 Hz</td>
<td>0.807</td>
</tr>
<tr>
<td></td>
<td>ALRT vs 500 Hz</td>
<td>0.757</td>
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<td>ALRT vs 1 kHz</td>
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<td>ALRT vs 2 kHz</td>
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<td>ALRT vs 4 kHz</td>
<td>0.711</td>
</tr>
<tr>
<td>AN</td>
<td>ALRT vs 250 Hz</td>
<td>-0.251</td>
</tr>
<tr>
<td></td>
<td>ALRT vs 500 Hz</td>
<td>-0.457</td>
</tr>
<tr>
<td></td>
<td>ALRT vs 1 kHz</td>
<td>-0.326</td>
</tr>
<tr>
<td></td>
<td>ALRT vs 2 kHz</td>
<td>-0.384</td>
</tr>
<tr>
<td></td>
<td>ALRT vs 4 kHz</td>
<td>-0.411</td>
</tr>
</tbody>
</table>

Table 8 shows that Pearson product moment correlation for ALR threshold and each frequency pure tone threshold. The ALRT and frequency specific pure tone threshold had shown significant positive correlation in sensorineural hearing loss group. In individuals with auditory dys-synchrony no significant correlation was obtained between ALRT and pure tone threshold at any frequency.

**Discussion**

The poor agreement between the ALR threshold and pure tone average obtained in the normal hearing group and good agreement in sensorineural hearing loss group could be due to the active and passive mechanism that takes place at the cochlea. In sensorineural hearing loss group the active mechanism of inner ear is affected, so passive mechanism take part in exciting more number of auditory nerves as it excites larger area of the basilar membrane. This might have resulted in increasing amplitude of the response and passed on to higher centers. This would have given rise to better agreement between behavioral threshold and ALR threshold in sensorineural hearing loss subjects.

Where as in normal hearing group, active mechanism is intact. The presence of active mechanism results in sharp tuning leading to the excitation of a few auditory nerves. This would have resulted in lesser compound action potential. Thus, causing higher ALR threshold resulted in poor agreement between behavioral threshold and ALR threshold.

In individuals with auditory dys-synchrony poor agreement was obtained. This could be due to the reduced transmission of signal to higher centers. This could be due to the leakage of signal conduction or a conduction block due to demyelization. Thus,
resulting in reduced but broadening of compound action potential (Starr, Picton and Kim, 2001). Which could have resulted in higher ALR threshold. In auditory dys-synchrony group there is altered temporal synchrony of auditory nerve and afferent discharges (Zeng, Oba, Garde and Starr, 1999). In neuropathy particularly, a demyelinating neuropathy, nerve impulses become slow when a demyelinated segment of the axon is encountered and then regain normal speed when that segment is passed (McDonald 1980). This type of conduction change results in a slowing of nerve conduction velocity. Demyelinated axons are impaired in their ability to transmit the information to the higher cortical centers. This might have lead to poor ALR threshold compared to behavioral threshold and thus poor correlation as ALR is a far field recording potential.

Kraus et al. (2000) reported that speech evoked ALR is a good predictor of speech processing in individuals with auditory dys-synchrony. Rance et al. (2002) found that speech perception abilities of auditory dys-synchrony children can not be reliably estimated from behavioral audiogram as like sensorineural hearing impaired group. In auditory dys-synchrony group where the difficulty in perception of speech is related to impaired temporal processing. The temporal processing is important for speech perception and to elicit the auditory evoked potentials. Thus, degraded processing would have resulted in better correlation.

Conclusions

It can be concluded from the present study that click evoked ALR is not a good tool to estimate frequency specific behavioral threshold. However click evoked ALR can closely estimate the behavioral threshold in sensorineural hearing loss group and it can be used as a clinical tool with less time consumption and more effectively in some of the hearing impaired population.
References


Recognition of Oriya Monosyllabic PB Words in Presence of Temporally Variant Noise

Biswajit Sadangi & Asha Yathiraj

Abstract

The study attempted to determine the effect of temporally variant noise on recognition of monosyllabic PB words. Thirty Oriya speaking individuals with normal hearing were evaluated. Their speech identification scores were determined in the presence of speech shaped noise. The different conditions included evaluation of speech identification in the presence of continuous noise, 16 Hz interrupted noise and, 32 Hz modulated noise at two different SNRs (0 dB and -5 dB). In addition, their speech identification in a quiet condition was also obtained. No significant difference was found between the speech identification scores in the continuous noise and 32 Hz interrupted noise condition at 0 dB SNR, whereas with the 16 Hz interrupted condition a significant difference was seen.

Introduction

It has been proven time and again that the presence of background noise adversely affects the perception of speech. Kryter (1970) reported that when a speech signal is masked, either partially or completely by a burst of noise, its intelligibility changes in a complex manner. In research studies, various types of noises have been utilized to determine their influence on the intelligibility of speech. These noises can be categorized based on their temporal pattern or their frequency characteristics as well.

Assessing speech intelligibility in interrupted noise has been reported to reveal the auditory system’s temporal ability to resolve speech fragments or get ‘glimpses’ or ‘looks’ of speech between the gaps of noise and to patch the information together to identify the specific speech stimuli (Miller, 1947).

Miller and Licklider (1950) reported about the intelligibility of monosyllables as a function of the interruption rate of noise. For interruption rates below 200 Hz, the speech intelligibility increased as the rate was lowered. The maximum intelligibility was reached at about ten interruptions per second. For low rates, the speech intelligibility dropped again because complete words were masked.

It was observed by Pollack (1955) that speech intelligibility decreased as the inter-burst level was increased for an interrupted masking noise of constant burst level. He found large improvements in speech intelligibility at lower repetition rates. Similarly, earlier Miller and Licklider (1950) observed that when the rate of interruption was 4 Hz or less, there was some loss of information because entire syllables and words were eliminated from the stimulus. However, once the interruption rate reaches 8 to 10 Hz, the words became as intelligible as un-interrupted speech.

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In contrast, it has been found by Dirks and Bower (1970), Dirks, Wilson and Bower (1969) and Miller and Licklider (1950) that word intelligibility did not change significantly when continuous speech was intermittently masked by white noise and interruption rates were varied from 1 to 100 Hz. They carried out the study using a 0 dB SNR.

It has been reported by Carhart, Tillman and Greetis (1969) that when the masker was the speech from a single talker or noise modulated periodically, speech intelligibility improved compared to when un-modulated noise was used even if the modulated and un-modulated noise had equal average energy (Dirks, Wilson and Bower, 1969; Festen and Plomp, 1990).

Results from the study of Smits and Houtgast (2007) revealed that individuals with normal hearing benefited from interruptions in noise while listening to digits in noise. The masking release was reported to be higher for the 16 Hz interruption than for 32 Hz interruption. The highest digit identification scores were obtained for 16 Hz modulated noise and lowest were for continuous noise.

The masking ability of a noise has also been noted to dependent upon the relation between the intensity of the speech and noise (Fant, 1960, cited in Silman and Silverman, 1991). For satisfactory communication, the signal-to-noise ratio was estimated to be +6 dB. When the criterion was not met, speech perception was noted to drop drastically. Moore (1996) found that at a 0 dB signal-to-noise ratio, word articulation scores reached 50%.

Groen (1969) evaluated the phoneme scores for individuals with hearing impaired and normal hearing individuals in presence of three SNRs (-5, 0 and +5 dB SNR). He observed that at -5 dB SNR the scores reduced drastically whereas at +10 dB the scores were significantly higher compared to at 0 dB SNR for hearing impaired individuals. For normal hearing group he found significantly lower scores at -5 dB SNR whereas no significant difference in scores were found for 0 dB and +10 dB. On the similar lines Kamlesh (1998) studied the effect of three SNR conditions (-5, 0 and +5 dB) on hearing impaired individuals using paired word (Kannada) recognition and questions. She reported that at adverse SNR the scores were significantly lower and with increase in SNR the scores improved. In the present study the similar results were obtained with the lowest scores for -5 dB SNR and better scores for 0 dB SNR in all the masking noise conditions. It is well documented in literatures about the need of speech perception tests in presence of noise for hearing aid selection (Carhart, 1965, cited in Plomp (1994); Tillman, Carhart and Olsen, 1970; Miller, Heise and Lichten, 1951; Stuart and Phillips, 1996).

From the literature, it is evident that speech perception of normal hearing individuals varies under fluctuating as well as steady-state noise. However, there is no consensus regarding its effect. There is a need to know if the speech identification
abilities of normal hearing individuals vary depending on whether the noise is interrupted temporally with different interruption rates. Further, noise in the environment occurs at different signal-to-noise ratios and often is fluctuating and not continuous. Hence, there is a need to know how individuals would perceive modulated noise in different signal-to-noise ratios. This would provide information about how individuals perform in a real life situation. Thus, the study aimed at comparing speech identification in the presence of continuous and temporally modulated speech shaped noise at two different signal-to-noise ratios (0 dB & -5 dB). The effect of two maskers having different temporal modulations was studied.

**Method**

Thirty individuals in the age range of 20 years to 50 years were evaluated in the study. All the individuals knew the dialect of Oriya spoken in Bhubaneswar region of Orissa. The participants did not have a history of any ear disorders. As they needed to provide an oral response, it was ensured that they did not have any speech or language disorders. In addition, in order to be included into the study, the participants had to have pure-tone thresholds within 20 dB HL. The participants whose speech identification scores were above 80% using the Oriya monosyllable PB wordlist, developed by Behera and Yathiraj (2007) were selected. All of the participants had an A’ type tympanogram with reflexes present in both the ear and passed the Screening Checklist for Auditory Processing (SCAP) developed by Yathiraj and Mascarehans (2002).

A calibrated double channel, diagnostic audiometer, Orbiter 922 with TDH-39 headphone was used for the pure-tone air conduction testing and speech audiometry. A Radio Ear B-71 vibrator was used for estimating bone conduction thresholds. A calibrated middle ear analyzer, (GSI-Tymptstar) provided tympanometry and reflexometry information. The speech and noise stimuli were presented through a Pentium 4 computer. The signals from the computer were routed to the audiometer.

Monosyllabic Phonemically Balanced (PB) Words of Oriya language, developed by Behera and Yathiraj (2007) was used. The test contained four half lists having phonemically balanced words. Each half-list had 25 monosyllabic words. The words were recorded digitally by a female Oriya speaker, who was fluent in the dialect of Oriya spoken in Bhubaneswar region of Orissa. The recordings were done using a Philips unidirectional microphone, connected to a Pentium IV computer, using the Adobe Audition 2.0 software. The recorded materials were scaled so that all the words were equally loud. Further, the four lists were randomized using a randomization table to form eight lists. Prior to each list a 1 kHz calibration tone was recorded. The materials were administered on 10 normal hearing Oriya speakers to ensure that the material was clearly recorded.
Speech shaped noise was generated with the Adobe Audition 2.0 software based on the parameters given by Silman and Silverman (1991). As suggested by them a broadband noise was filtered to have a frequency range of 250 Hz to 4000 Hz. The slope of the speech spectrum was +3 dB per octave from 250 Hz to 1 kHz and 12 dB per octave from 1 kHz to 4 kHz.

The speech shaped noise was further modulated to get 16 Hz and 32 Hz modulated noises. This was done using the MATLAB 7.0 software. The noises were then mixed with the recorded speech materials using the Adobe Audition 2.0 software. The following six conditions were thus generated: Continuous speech noise + speech at 0 dB SNR; 16 Hz modulated noise + speech at 0 dB SNR; 32 Hz modulated noise + speech at 0 dB SNR; Continuous speech noise + speech at -5 dB SNR; 16 Hz modulated noise + speech at -5 dB SNR and; 32 Hz modulated noise + speech at -5 dB SNR.

An example of a waveform for the word /kar/ is provided in Figure 1. Also given in Figure 1 are the waveforms for continuous noise, 16 Hz modulated noise, 32 Hz modulated noise, the combination of the test stimulus /kar/ with all the different types of noises used at 0 and -5 dB SNR.

Figure 1: Waveforms of the (a) word /kar/, (b) continuous noise, (c) 16 Hz modulated noise (d) 32 Hz modulated noise; word /kar/ in combination with (e) continuous noise, (f) 16 Hz modulated noise and (g) 32 Hz modulated noise at 0 dB and -5 dB SNR

All the tests were carried out in a sound treated suite. The noise levels were within permissible levels as specified by ANSI S 3.1(1991).

The recorded speech materials were played on a Pentium-IV computer with the help of the Adobe Audition 2.0 software, routed through the audiometer and presented through headphones. All participants heard the speech signals at an intensity of 40 dB HL. In the two SNR conditions, the level of the signal was held constant, while the level
of the noise varied. Thus, in the -5 dB SNR condition the speech materials were presented at 40 dB HL and the noise was presented at 45 dB HL. The speech as well as noise was heard in the same ear. The choice of ear was randomized such that half the participants heard the signal through right ear and the other half through the left ear. The order in which each of the participants heard these lists were randomized to avoid any list effect. No participant heard the same list more than once.

The participants were instructed to repeat the words heard by them. The oral responses of the participants were scored. Every correct response was given a value of one and an incorrect response a score of zero. The data thus obtained on the thirty normal hearing participants were analyzed using the Statistical Package for Social Sciences (SPSS) software version 15. Repeated measure ANOVA and paired ‘t’ test was carried out to determine the effect of the different masking conditions on speech perception.

The rationalized arcsine transform, developed by Studebaker (1985) was done to convert the speech identification scores into rationalized arcsine units (rau). This was done since it has been observed by Studebaker (1985) that speech identification scores are non-linear or additive. This was found to result in the critical difference between two speech identification scores being unequal. Hence, the available scores were converted to ra u scores using the RATARC online rationalized arcsine transform program developed by Studebaker (1985). Thus, all statistical analyses were done for the word scores as well as for the ra u scores.

Results and Discussion

The impacts of the following on the speech identification scores are discussed:

- Effect of different listening conditions (quiet, two SNR conditions and three masking conditions),
- Effect of signal-to-noise ratio (0 dB and -5 dB SNR) and,
- Comparison of quiet and masking conditions (continuous noise, 16 Hz and 32 Hz modulated noises at the two SNRs).

Effect of different listening conditions

The mean and the standard deviation of the speech identification scores in quiet and in the presence of noises were calculated separately. The mean speech identification scores were better for the quiet condition compared to the masking conditions. Among the difference noise conditions, better mean speech identification scores were obtained in the presence of 16 Hz modulated speech shaped noise at 0 dB SNR. In contrast, poorer scores were obtained for the 32 Hz modulated speech shaped noise at -5 dB SNR.
Recognition of Oriya words in temporally variant noise

The mean and the standard deviation of the raw scores for the different conditions are given in Table 1 and Figure 2.

Also provided are the mean and standard deviation (SD) of the rau scores in Table 1 and Figure 3. Similar results were obtained for rau scores in all the listening condition as mentioned for the raw scores.

Table 1: Mean and Standard deviation (SD) for the speech identification (raw and rau) scores in different listening conditions

<table>
<thead>
<tr>
<th>Listening Conditions</th>
<th>SNR</th>
<th>Raw scores</th>
<th></th>
<th></th>
<th>rau scores</th>
<th></th>
<th></th>
</tr>
</thead>
<tbody>
<tr>
<td></td>
<td></td>
<td>Mean #</td>
<td>Standard deviation (SD)</td>
<td>Mean</td>
<td>Standard deviation (SD)</td>
<td>Mean</td>
<td>Standard deviation (SD)</td>
</tr>
<tr>
<td>Quiet</td>
<td>-</td>
<td>23.90</td>
<td>1.09</td>
<td>102.51</td>
<td>9.74</td>
<td></td>
<td></td>
</tr>
<tr>
<td>Continuous noise</td>
<td>0 dB</td>
<td>19.93</td>
<td>2.66</td>
<td>79.37</td>
<td>11.86</td>
<td></td>
<td></td>
</tr>
<tr>
<td>Continuous noise</td>
<td>-5 dB</td>
<td>18.33</td>
<td>1.54</td>
<td>71.86</td>
<td>6.24</td>
<td></td>
<td></td>
</tr>
<tr>
<td>16 Hz modulated speech noise</td>
<td>0 dB</td>
<td>20.67</td>
<td>1.24</td>
<td>81.90</td>
<td>5.80</td>
<td></td>
<td></td>
</tr>
<tr>
<td>16 Hz modulated speech noise</td>
<td>-5 dB</td>
<td>19.53</td>
<td>1.11</td>
<td>76.74</td>
<td>4.63</td>
<td></td>
<td></td>
</tr>
<tr>
<td>32 Hz modulated speech noise</td>
<td>0 dB</td>
<td>19.03</td>
<td>1.79</td>
<td>74.92</td>
<td>7.91</td>
<td></td>
<td></td>
</tr>
<tr>
<td>32 Hz modulated speech noise</td>
<td>-5 dB</td>
<td>18.23</td>
<td>1.70</td>
<td>71.42</td>
<td>7.33</td>
<td></td>
<td></td>
</tr>
</tbody>
</table>

# Maximum score = 25

In order to check whether there was a significant difference between the different noise conditions and also the different SNR conditions, two-way repeated measure ANOVA was done (3 noise conditions × 2 SNRs). The ANOVA results showed a significant main effect for the different noise conditions [F (2, 58) = 8.52, p < 0.01] and different SNR conditions [F (1, 29) = 52.35, p < 0.01]. It also showed a significant interaction effect between the different noise and SNR conditions [F (2, 58) = 3.68, p < 0.05].
ANOVA results for rau scores too showed a significant main effect for the different noise conditions \([F (2, 58) = 7.90, p < 0.01]\) and different SNR conditions \([F (1, 29) = 53.45, p < 0.01]\). A significant interaction effect was also seen between the different noise and SNR conditions \([F (2, 58) = 4.39, p < 0.05]\).

Since the two SNR conditions were significantly different, separate one-way ANOVAs were done for the raw scores. This was done to see the significance of difference for different masking noise conditions at each of the two SNRs. A significant main effect was seen for the 0 dB SNR \([F (2, 58) = 6.99, p < 0.01]\) and -5 dB SNR \([F (2, 58) = 9.24, p < 0.01]\).

Similarly, on analysis of the rau scores a significant main effect was seen. This was observed for the 0 dB SNR \([F (2, 58) = 6.60, p < 0.01]\) and -5 dB SNR \([F (2, 58) = 8.75, p < 0.01]\) conditions.

Further, Bonferroni pairwise comparison was done to check the significance of difference between the different masked noise conditions. At the 0 dB SNR, a significant
difference between the 16 Hz modulated speech noise condition and 32 Hz modulated noise condition was observed. However, unlike the expected findings, there was no significant difference between the continuous and the modulated speech noise conditions (Table 2). On the other hand, the pairwise comparison for the -5 dB SNR revealed a significant difference (p < 0.05) between the continuous masking condition and the 16 Hz modulated noise condition. A similar significant difference was also observed between the two modulated conditions (16 Hz & 32 Hz) (Table 3). However, like that obtained at 0 dB SNR condition, no significant difference (p > 0.05) was seen for the continuous and the 32 Hz modulated noises.

Table 2: Pairwise comparison of different listening conditions at 0 dB SNR for raw scores

<table>
<thead>
<tr>
<th>Masking noise condition</th>
<th>16 Hz modulated</th>
<th>32 Hz modulated</th>
</tr>
</thead>
<tbody>
<tr>
<td>Continuous</td>
<td>p &gt; 0.05</td>
<td>p &gt; 0.05</td>
</tr>
<tr>
<td>16 Hz modulated</td>
<td>-</td>
<td>p &lt; 0.05</td>
</tr>
</tbody>
</table>

Table 3: Pairwise comparison of different listening conditions at -5 dB SNR for raw scores

<table>
<thead>
<tr>
<th>Masking noise condition</th>
<th>16 Hz modulated</th>
<th>32 Hz modulated</th>
</tr>
</thead>
<tbody>
<tr>
<td>Continuous</td>
<td>p &lt; 0.05</td>
<td>p &gt; 0.05</td>
</tr>
<tr>
<td>16 Hz modulated</td>
<td>-</td>
<td>p &lt; 0.05</td>
</tr>
</tbody>
</table>

Similar results were found for the rau scores at both SNRs. Probably, identical results were obtained for the raw and rau scores since the values obtained in the present study did not contain scores along the entire range of scores. Most of the speech identification scores, across all conditions, were concentrated only in the upper extreme of the range.

It has been reported by Miller and Licklider (1950) and Gustafsson and Arlinger (1994) that at higher modulation rates, the release from masking was less. Hence, the speech intelligibility at higher modulation rates such as 32 Hz tended to be similar to be that observed for continuous or steady-state noise. In contrast, it has been reported that for maskers with lower modulation rates of 10 Hz (Miller and Licklider, 1950; Gustafsson and Arlinger, 1994) and 16 Hz (Smits and Houtgast, 2007), the release of masking was more, resulting in better speech perception compared to continuous noise makers.

In the present study, such a release from masking was observed only at the -5 dB SNR and not at 0 dB SNR. This indicates that only at a more adverse SNR condition, did
the release of masking occur. Unlike the expected finding, no release in masking was obtained at the 0 dB SNR condition, even when the participants with extreme scores were eliminated. Hence, participant variability could not have accounted for the lack of release from masking at 0 dB SNR. Thus, it can be construed that release from masking is not solely dependent on the modulation rate but also on the SNR.

Further, in the present study the scores obtained using continuous masking noise were lower than that obtained with the 16 Hz modulated noise at -5 dB SNR. However, at the same SNR, no improvement was seen for the 32 Hz modulation rate compared to the continuous noise. This is in agreement with the results of various previous studies, where the masking release was reported to be higher for modulated noise than for continuous noise (Miller and Licklider, 1950; Gustafsson and Arlinger, 1994). It has also been reported by Smits and Houtgast (2007) that at higher modulation rates of 32 Hz, the noise functions similar to continuous noise and the advantage from release of masking does not occur.

**Effect of signal-to-noise ratio (SNR)**

From the mean values given in Table 1, it can also be observed that the speech identification scores were higher for the 0 dB SNR condition and lower for the -5 dB SNR condition. This was observed for continuous masking noise and both modulation masking (16 Hz and 32 Hz) conditions.

Paired sample ‘t’ test was done to see if these differences in mean scores across the two SNRs were significantly different. The paired sample ‘t’ test revealed a significant difference between the speech identification scores at the two different SNRs. This significant difference between 0 dB and -5 dB SNR (p < 0.01) was present in all the three masking conditions (continuous noise, 16 Hz modulated noise & 32 Hz modulated noise). This is evident from the information given in Table 4. Similar results were seen for the rau scores too.

*Table 4: Significance of difference of different masking conditions at the two SNRs*

<table>
<thead>
<tr>
<th>Listening Conditions</th>
<th>SNR</th>
<th>Mean #</th>
<th>Standard deviation (SD)</th>
<th>‘t’ value</th>
</tr>
</thead>
<tbody>
<tr>
<td>Continuous noise</td>
<td>0 dB</td>
<td>19.93</td>
<td>2.66</td>
<td>4.94**</td>
</tr>
<tr>
<td></td>
<td>-5 dB</td>
<td>18.33</td>
<td>1.54</td>
<td></td>
</tr>
<tr>
<td>16 Hz modulated speech</td>
<td>0 dB</td>
<td>20.67</td>
<td>1.24</td>
<td>5.78**</td>
</tr>
<tr>
<td>noise</td>
<td>-5 dB</td>
<td>19.53</td>
<td>1.11</td>
<td></td>
</tr>
<tr>
<td>32 Hz modulated speech</td>
<td>0 dB</td>
<td>19.03</td>
<td>1.79</td>
<td>5.17**</td>
</tr>
<tr>
<td>noise</td>
<td>-5 dB</td>
<td>18.23</td>
<td>1.70</td>
<td></td>
</tr>
</tbody>
</table>

** Significant at p < 0.01  
# Maximum score = 25
The finding of the present study is in consonance with that of Groen (1969) and Kamlesh (1998). They too reported that at higher SNR the speech identification scores were higher and at a lower SNR of -5 dB the scores dropped. This drop in scores has been attributed to the masking that occurs at lower SNRs. Similar findings have also been noted by Olsen, Olofsson and Hagerman (2005) while using different other signal-to-noise ratio.

**Effect of quiet Vs. masking conditions**

The mean and standard values given in Table 1 clearly reveal that the speech identification scores were comparatively higher for the quiet condition than for any masking condition (continuous or modulated). Amongst the masking conditions, highest scores were obtained for the 16 Hz modulated noise at 0 dB SNR condition and the lowest scores were obtained for the 32 Hz modulated noise at -5 dB SNR condition.

To check for a significant difference between the quiet and the different masking conditions, paired ‘t’ test was done. A significant difference between the quiet and all the different masking conditions was obtained. From Table 5 it can be noted that, the scores obtained in the quiet condition were significantly higher than that obtained with any of the masking conditions. Thus, irrespective of whether the masking noise had a modulation rate of 16 Hz / 32 Hz or had a SNR of 0 dB / -5 dB, it resulted in significantly lower scores than the quiet condition. Similar results were obtained with the rau scores as well.

**Table 5: Significance of difference between the quiet and different noise conditions**

<table>
<thead>
<tr>
<th>Listening Conditions</th>
<th>Mean#</th>
<th>‘t’ value</th>
</tr>
</thead>
<tbody>
<tr>
<td>Quiet</td>
<td>23.90</td>
<td>7.67 ***</td>
</tr>
<tr>
<td>Continuous noise at 0 dB SNR</td>
<td>19.93</td>
<td></td>
</tr>
<tr>
<td>Quiet</td>
<td>23.90</td>
<td>15.42 ***</td>
</tr>
<tr>
<td>Continuous noise at -5 dB SNR</td>
<td>18.33</td>
<td></td>
</tr>
<tr>
<td>Quiet</td>
<td>23.90</td>
<td>15.60 ***</td>
</tr>
<tr>
<td>16 Hz modulated noise at 0 dB SNR</td>
<td>20.67</td>
<td></td>
</tr>
<tr>
<td>Quiet</td>
<td>23.90</td>
<td>18.79 ***</td>
</tr>
<tr>
<td>16 Hz modulated noise at -5 dB SNR</td>
<td>19.53</td>
<td></td>
</tr>
<tr>
<td>Quiet</td>
<td>23.90</td>
<td>13.13 ***</td>
</tr>
<tr>
<td>32 Hz modulated noise at 0 dB SNR</td>
<td>19.03</td>
<td></td>
</tr>
<tr>
<td>Quiet</td>
<td>23.90</td>
<td>15.09 ***</td>
</tr>
<tr>
<td>32 Hz modulated noise at -5 dB SNR</td>
<td>18.23</td>
<td></td>
</tr>
</tbody>
</table>

*** Significant at p < 0.001  
# Maximum score = 25

From the findings of the present study it can be noted that in the presence of masking noise, speech identification scores in normal hearing adults dropped drastically.
Even the least of the masking conditions (16 Hz at 0 dB SNR) was highly significantly different from the perception obtained in the quiet situation. The findings of the current study with regard to the comparison between the quiet and masking conditions are in agreement with that reported by Miller and Nicely (1955).

Though the present study was carried out with adult normal hearing participants, it can be construed that similar noise conditions would have a much more adverse affect on children. Studies in literature, comparing the performance of children with adults on speech intelligibility in noise, have shown that the former group performs poorer than the latter group (Elliott, 1979; Newman & Hochberg, 1983; Nittrouer & Boothroyd, 1990). Further, noise levels in classrooms have been found to range from +35 dB to -10 dB as reported by Nebelek and Pickett (1974). It is possible that the intermittent noise present in the classrooms would have a highly negative effect on speech perception and hence the learning abilities of children. Thus, it is essential that noise levels should be much lower than what has been utilized in the present study in order to enable children to perceive speech effectively.

From the findings of the present study, it can also be extrapolated that if individuals with normal hearing are adversely affected with masking noise, those with a hearing loss are likely to be more adversely affected. Barrenas and Wikstrom (2000), Elpern (1960), Schneider and Daneman (2007), and Ross, Huntington, Newby and Dixon (1965) have reported that those with hearing impairment are likely to have more difficulty in speech perception in the presence of noise.

Conclusions

From the findings of the study it was observed that while a release from masking occurred at the more adverse SNR (-5 dB) no such release occurred at the lower SNR (0 dB). Thus, it can be inferred that release from masking is dependent on both modulation rate as well as SNR.

Further, it was observed that there was a significant difference between the two modulated masking noise condition (16 Hz and 32 Hz) at both at 0 dB and -5 dB SNR. Higher scores were obtained for the 16 Hz modulation condition. There was also a significant difference in the performances between the two SNRs (0 dB and -5 dB), with the scores being higher for the 0 dB SNR condition.

References


Yathiraj, A. & Mascarehans, K. (2002). Effect of auditory stimulation in central auditory processing in children with APD. ARF project carried out at department of audiology in All India Institute of Speech & Hearing, Mysore.
Effect of Envelope Enhancement on Speech Reception and Late Latency Response Measures in Subjects with Auditory Dys-Synchrony

Chandrakant Vishwakarma & Vijayalakshmi Basavaraj*

Abstract

Auditory neuropathy is a disorder characterized by the impairment of the peripheral auditory function with the preservation of OHCs (Starr, et al. 1996; Berlin et al., 1998; Berlin et al., 1999). It is known fact that these individuals have problem with speech discrimination.

Aim: To compare the effect of envelope enhancement on speech perception and LLR in subjects with auditory neuropathy, with those associated with normal hearing.

Method: 11 VCV syllables were recorded using an adult male voice by using PRAAT software. These syllables were further mixed up in preset proportion of speech noise to make it in 10 dB SNR condition. In perceptual testing the subjects task was to repeat the stimuli is heard, in both for quiet as well as 10 dB SNR condition for non enhanced and enhanced stimuli, whereas objective recording was done using only one stimulus /da/, where latency, amplitude and morphology of LLR were recorded in quiet condition for non enhanced and enhanced stimulus. The testing was done for AN/AD subjects as well as age and gender matched subjects with normal hearing.

Results: analysis was done using SPSS version 15, which revealed that there is decrease in latency and better amplitude in both groups with enhancement, but it was not significant for all the peaks. Mean and standard deviation are given for each analysis.

Conclusions: The envelope enhancement did tend to decrease the latency and increase the amplitude of LLR, but the effect was not much significant for all the peaks, hence more advanced study with better control over the variables is advocated.

Introduction

Auditory neuropathy/auditory dys-synchrony (AN/AD) is a disorder characterized by the impairment of the peripheral auditory function with the preservation of the outer hair cell (OHC) integrity (Berlin et al., 1998; Berlin, 1999; Butinar et al., 1999; Starr, Sininger, Pratt, 2000). The peripheral lesion could be localized at the level of the inner hair cells (IHCs), auditory nerve fibers or the synapse in between (Starr, Picton, Sininger, Hood, Berlin, 1996; Berlin et al., 1998; Butinar et al., 1999). It is now well established that speech identification abilities of individuals with auditory dys-synchrony are disproportionate to the degree of their hearing loss (Li, et al., 2005; Starr, et al., 1996).

Physiological tests generally used in diagnosing auditory dys-synchrony are auditory brainstem response and otoacoustic emissions. Another physiologic test which has been studied widely in individuals with AN/AD is the auditory late latency responses (LLR). The synchrony required for LLR is on the order of several milliseconds that’s why the LLR is expected to be present in the individuals with AN/AD (Kraus et al. 2000). However there are equivocal findings regarding presence/absence of LLR in individuals with AN/AD (Starr et al. 1991., Starr et al., 1996).

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As AN /AD adversely affects speech comprehension, appropriate management should be considered. Communication difficulties in individuals with auditory dys-synchrony, even in those with the mild hearing loss, are much more severe compared to those individuals with cochlear hearing loss of 60 dB HL or more. Conventional amplification through hearing aids does not seem to be beneficial as this does not address the problem of neural dyssynchrony (Rance et al., 2002). Cochlear implantation is of benefit to some patients with auditory dys-synchrony (Singer & Oba, 2001). However, the usefulness of cochlear implantation seems to depend on the site of lesion and not all cases of AN /AD are suitable for a cochlear implant (Simmons and Beauchaine, 2000). Cases with lesions at the inner hair cell level or at the synapse with the auditory nerve, which are bypassed by the implant, may achieve greater benefit (Simmons and Beauchaine, 2000).

Thus, it is important to explore alternative strategies that are much less invasive than cochlear implants which may benefit individuals with AN, particularly for those who have relatively mild AN. One effective means of improving speech intelligibility is to speak clearly (Picheny, Durlach and Braida, 1985, 1986, 1989). When talkers are instructed to speak clearly, they usually produce more intelligible speech than they would when interacting in casual conversation. The higher intelligibility in clear speech than in conversational speech is likely a result of acoustic and phonetic differences between these two styles of speech. These differences include reduced speaking rate, increased energy in the 1000–3000 Hz range, enhanced temporal modulations, expanded voice pitch range and vowel space (Ferguson & Kewley-Port, 2002; Krause & Braida, 2004; Liu, Del Rio, Bradlow and Zeng, 2004; Payton, Uchanski and Braida, 1994).

Another option which can be opted for individuals with the AN/AD is the envelope enhancement of speech. A number of investigators have studied the importance of envelope enhancement on speech perception in noise for subjects with normal hearing, cochlear hearing loss and learning disability (Tallal et al., 1996; Larenzi, Berthommier, Apoux, and Bacri, 1999; Apoux, Tribut, Debrilule, & Lorenize, 2004). They have shown improvement with envelope enhancement for cochlear hearing loss and other group of individuals, but improvement observed was lesser in these groups. The rationale behind employing envelope enhancement in noise is that the noise reduces the ability to process amplitude variation in the speech signal, so enhancing amplitude variations improves speech perception. Since AN/AD subjects have impairment in processing amplitude variation of speech signal, enhancing the modulations might improve speech perception.

Zeng and Liu, (2006) have demonstrated that clear speech improved speech perception in individuals with AN/AD. The improvement observed for clear speech has been attributed to enhanced envelope in clear speech. Clear speech has certain properties; type of speaking style to facilitate better communication in adverse listening conditions, roughly 17 % more intelligible than normal conversational speech for mild to moderate hearing impaired individuals (Pichney, et al., 1985; Payton 1994).
Need of the study:

There is no consensus over the management issues of AN/AD subjects. Studies dealing with envelope enhancement have shown improvement in speech perception in persons with cochlear loss. However, there is a dearth of information regarding the usefulness of envelope enhancement of speech in improving speech perception in individuals with AN/AD. AN/AD group have been reported to have temporal deficits, and therefore difficulty in recognizing short signals effectively. Hence there is a need to study whether spectral enhancement of signal improves speech recognition in such subjects or not. It will be interesting and relevant to study the effect both through objective as well as subjective measures. Hence this study was convinced and conducted to examine the effect of envelope enhancement on speech perception in subjects with AN/AD.

Aims of the study: To compare the effect of envelope enhancement on speech perception and late latency response in subjects with auditory dys-synchrony/auditory neuropathy with those obtained in subjects with normal hearing.

Objectives:
1. To compare LLR amplitudes for /da/ syllable in non enhanced and enhanced condition in subjects with AN/AD
2. To compare LLR latencies for /da/ syllable in non enhanced and enhanced condition in subjects with AN/AD
3. To compare LLR amplitudes for /da/ syllable in non enhanced and enhanced condition in subjects with normal hearing
4. To compare LLR latencies for /da/ syllable in non enhanced and enhanced condition in subjects with normal hearing
5. To compare LLR amplitudes for /da/ in non enhanced and enhanced condition between normal and AN/AD subjects
6. To compare LLR latencies for /da/ syllable in non enhanced and enhanced condition between subjects with normal hearing and subjects with AN/AD
7. To compare the morphology of LLR for /da/ syllable in non enhanced and enhanced condition for subjects with normal hearing subjects and subjects with AN/AD on 3 point rating scale
8. To compare speech perception results obtained with non enhanced and enhanced signals in AN/AD subjects in: (i) Quite and (ii) 10 dB SNR condition
9. To compare speech perception results obtained with non enhanced and enhanced signals in normal subjects in: (i) Quite and (ii) 10 dB SNR condition
Research Design

Mixed group pretest posttest design wherein independent variables are the speech stimuli in non enhanced and enhanced condition, age and sex of the subjects in both groups and the dependent variables are the latencies, amplitude, morphology obtained for each peak as well as the responses given by the subjects for perceptual measure.

Hypothesis:
1) There is no difference in LLR amplitudes in subjects with AN/AD for syllable /da/ in the non enhanced and enhanced condition.
2) There is no difference in LLR latencies in subject with AN/AD for syllable /da/ in the non enhanced and enhanced condition.
3) There is no difference in LLR amplitudes in subjects with normal hearing for syllable /da/ in the non enhanced and enhanced condition.
4) There is no difference in LLR latencies in subjects with normal hearing for syllable /da/ in non enhanced and enhanced condition.
5) There is no difference between subjects with normal hearing and AN/AD in LLR amplitude in non enhanced and enhanced condition.
6) There is no difference between subjects with normal hearing and AN/AD in LLR latencies in non enhanced and enhanced condition.
7) There is no change in morphology in non enhanced and enhanced condition for subjects with normal hearing and AN/AD on 3 point rating scale.
8) There is no difference in speech perception results obtained for non enhanced and enhanced condition in subjects AN/AD.
9) There is no difference in speech perception results obtained for non enhanced and enhanced condition in subjects with normal hearing.

Method

The study was done in two parts.

Part I: preparation of the speech stimuli 11 VCV syllables were recorded in an adult male voice using PRATT software. All the recording was carried in a sound treated room. The stimuli chosen were /aba/, /acha/, /ada/, /adha/, /aga/, /aka/, /ala/, /ama/, /ana/, /apa/ and /ara/. The consonant chosen represents different place and manner of articulation. Stimuli were restricted to less number due to time constraint issue in the testing. These stimuli were further enhanced using PRATT software, with 16 bits sampling rate (as it gives better waveform) and 22050 sampling frequency and 2to 32 Hz modulation frequency. The stimuli were later mixed up with speech noise in preset proportion to make it 10 dB SNR condition, using MATLAB software version 6. The software calculates the root mean square for signal and noise and then does the mixing.
Once the stimuli preparation was over, the stimuli were copied over a CD, for testing, and of 11 stimuli one stimulus /ada/ was used for the LLR recording (objective testing). In this only /da/ portion was retained, to load on to the instrument (IHS) as the instrument takes stimuli up to 250 ms. This editing work was done using PRATT software and later the wave file was converted into stimulus file using waveform converter in the instrument itself. Unenhanced /da/ and enhanced /da/ were both loaded in the instrument as stimulus file for objective recording.

**Part II:** Testing the subjects with AN/AD and age/gender matched subjects with normal hearing.

**Subjects:** Subjects were divided in two groups; experimental group and control group.

**Experimental group:** Nine subjects (18 ears) diagnosed as having auditory dysynchrony were taken for the study. Inclusion criteria were as follows:
- Age ranging from 10 to 26 years old with a mean age of .19.77 years
- Normal to moderately severe hearing loss (based on the pure tone average of 500 Hz, 1 kHz & 2 kHz).
- Speech identification Score disproportionate to pure tone average of 500 Hz, 1 kHz & 2 kHz.
- “A” type tympanogram indicating normal middle ear functioning.
- Absent of both ipsilateral and contralateral acoustic reflexes.
- No history of any middle ear problems, and no misarticulations.
- Presence of Otoacoustic emissions.
- Absence of auditory brainstem responses.

**Control Group:** Control group consisted of 9 age and gender matched subjects with normal hearing sensitivity. The inclusion criteria for the control group were as follows:
- Hearing threshold <15 dB HL from 250 Hz to 8 kHz , at octaves and interoctaves.
- Good speech identification score of more than 90%.
- “A” type tympanogram with present Ipsilateral and Contralateral reflexes, and no history of middle ear problem.
- Presence of OAEs.
- Presence of ABR response.
- No history/presence of any neurological deficits.

**Instrumentation:**
- A calibrated (ANSI S3.6-1996), two channel clinical audiometer OB922 with TDH-39 headphones housed in Mx-41/AR ear cushions with audio cups were used for puretone audiometry. Radioear B-71 bone vibrator was used for measuring bone conduction threshold.
• A calibrated middle ear analyzer, (GSI tymptstar) using 226 Hz probe tone was used for tympanometry and reflexometry.

• Oto acoustic emissions were recorded using either Intelligent Hearing System Smart OAE windows USB version 2.62 or otodynamics ILO V6 OAE instrument.

• Intelligent Hearing System (Smart EP windows USB version 3.91) evoked potential system with insert ear ER-3A receiver was used for recording auditory brainstem responses and late latency responses.

• Perceptual testing for the speech reception was carried out with the help of CD which was played through Pentium IV computer, routed through OB922 audiometer with head phone output.

• Late Latency Response for speech stimulus was recorded using Intelligent Hearing System (Smart EP windows USB version 3.91) evoked potential system.

**Test environment:** All the audiological tests were carried out in an acoustically treated room (as per ANSI, 1996) with adequate illumination.

**Procedure:**

• Pure tone audiometry was done from 250 Hz to 8 kHz at octaves and interoctaves for air conduction stimuli and from 250 Hz to 4 kHz for bone conduction stimuli. All the testing was done using Modified Hughson-Westlake Method (Carhart & Jerger, 1959). Speech audiometry was also done using modified Olsen –Tillman method (1973). Inbuilt talk back system was used for speech audiometry.

• Tympanometry and reflexometry was done to check to rule out middle ear pathology. 226 Hz was the probe frequency and 85 dB SPL was the level used. Reflex eliciting signal was at 500 Hz, 1000 Hz and 2000 Hz. It was checked for ipsilateral and contralateral mode of stimulation.

• Otoacoustic emissions evoked by clicks presented at 85 dBpeSPL for the linear clicks were recorded. The probe with a tip was positioned in the external ear canal and was adjusted to give flat stimulus spectrum across the frequency range. The response was acquired using the linear averaging method. The two averaged TEOAE waveforms of each memory buffer composed of 256 accepted click trains, were automatically cross-correlated and used to determine the reproducibility of the measured TEOAEs by the software. Responses were accepted when the reproducibility was 70% or greater. A total of two responses were recorded to ensure the stability of the response. A minimum of one minute gap was given between any two recordings to reduce the influence of the one
recording over another recording. Care was taken to ensure that the position of probe was not altered.

- Auditory brainstem responses were recorded from one channel using ER-3A insert receiver. The site of electrode placement was prepared with skin preparation gel. Silver chloride disc electrode was used with a conducting gel.

In perceptual testing client had to repeat whatever was heard to them. Following were the objective protocol.

<table>
<thead>
<tr>
<th>Stimulus Parameters</th>
<th>Speech stimulus</th>
<th>/da/ Enhanced &amp; Non enhanced</th>
</tr>
</thead>
<tbody>
<tr>
<td></td>
<td>Duration</td>
<td>230 ms</td>
</tr>
<tr>
<td></td>
<td>Level</td>
<td>90 dB nHL</td>
</tr>
<tr>
<td></td>
<td>Polarity</td>
<td>alternating</td>
</tr>
<tr>
<td></td>
<td>Mode of presentation</td>
<td>Ipsilateral</td>
</tr>
<tr>
<td></td>
<td>Repetition rate</td>
<td>1.1/s</td>
</tr>
</tbody>
</table>

<table>
<thead>
<tr>
<th>Acquisition Parameters</th>
<th>Transducer</th>
<th>ER-3A insert receiver</th>
</tr>
</thead>
<tbody>
<tr>
<td></td>
<td>Analysis time</td>
<td>0-500 msec with -50 pre stimulus period</td>
</tr>
<tr>
<td></td>
<td>Filter setting &amp; Gain</td>
<td>1-30 Hz, 50,000</td>
</tr>
<tr>
<td></td>
<td>Electrode placement</td>
<td>Inverting(-ve): Test ear Noninverting(+ve): FPz Ground Non Test ear.</td>
</tr>
<tr>
<td></td>
<td>Sweeps, Artifact rejection</td>
<td>150 sweeps &amp; 40 uV</td>
</tr>
<tr>
<td></td>
<td>Electrode Impedance</td>
<td>&lt; 10 kHz</td>
</tr>
<tr>
<td></td>
<td>Inter Electrode Impedance</td>
<td>&lt; 3 kHz.</td>
</tr>
</tbody>
</table>

The following data were generated for analysis:

1) Speech identification score results in non enhanced and enhanced condition in quiet as well as in 10 dB SNR condition for subjects with AN/AD and normal hearing.
2) LLR latencies in non enhanced and enhanced condition in quiet condition for subjects with AN/AD and normal hearing.
3) LLR amplitude LLR latencies in non enhanced and enhanced condition in quiet condition for subjects with AN/AD and normal hearing.
4) Morphology status in non enhanced and enhanced condition in quiet condition for subjects with AN/AD and normal hearing.
Results and Discussion

SPSS 15 was used, mixed ANOVA, independent sample t test and paired t test were done, and Latencies were measured for all the four peaks of LLR in non enhanced and enhanced condition.

Latencies were measured for all the four peaks of LLR i.e P1, N1, P2, N2 in enhanced and non enhanced condition. Late latency responses were present in all the normal hearing individuals as well in all individuals with AN/AD. Overall mean value for the latencies for the enhanced stimulus was less compared to the non enhanced signal for all the four peaks and across two groups as shown in Table 1. Also the standard deviation (given in parenthesis) was more in AN/AD group than in the normal hearing group.

Table 1: Mean and standard deviation of latencies of LLR peaks in non enhanced (NEL) and enhanced (EL) conditions for experimental (AN/AD) and control (Normal) group.

<table>
<thead>
<tr>
<th>Groups</th>
<th>P1</th>
<th>N1</th>
<th>P2</th>
<th>N2</th>
</tr>
</thead>
<tbody>
<tr>
<td></td>
<td></td>
<td></td>
<td></td>
<td></td>
</tr>
<tr>
<td>Experimental</td>
<td>NEL</td>
<td>EL</td>
<td>NEL</td>
<td>EL</td>
</tr>
<tr>
<td></td>
<td>71.11</td>
<td>67.27</td>
<td>121.66</td>
<td>123.50</td>
</tr>
<tr>
<td></td>
<td>(27.01)</td>
<td>(23.86)</td>
<td>(36.25)</td>
<td>(41.54)</td>
</tr>
<tr>
<td>Control</td>
<td>NEL</td>
<td>EL</td>
<td>NEL</td>
<td>EL</td>
</tr>
<tr>
<td></td>
<td>72.11</td>
<td>70.11</td>
<td>129.61</td>
<td>123.00</td>
</tr>
<tr>
<td></td>
<td>(10.30)</td>
<td>(13.41)</td>
<td>(16.24)</td>
<td>(17.20)</td>
</tr>
</tbody>
</table>

Absolute amplitude for all the peaks (P1, N1, P2 & N2) were measured in non enhanced and enhanced condition for both the groups. Mean and standard deviation for amplitude of different peaks in non enhanced and enhanced stimulus within each group is mentioned in the Table 2. It was found that within the enhanced condition there was increase in the amplitude in both the groups. Again, there was more standard deviations for the AN/AD group was larger than that of the normal hearing group.

Table 2: Absolute amplitude for the peaks in LLR for both conditions (NEA & EA) in both groups. Values in parenthesis are standard deviation.
LLR sample recording of the experimental and control group showing the change in latency and amplitude for the non enhanced and enhanced conditions are shown in Figure 1 (a) & (b) and Figure 2 (a) & (b) respectively.

Figure 1: (a) & (b): LLR waveforms recordings for (a) non enhanced and (b) enhanced /da/ stimulus in control group (normal hearing).

Figure 2 (a) & (b): LLR waveform recorded for (a) non enhanced and (b) enhanced /da/ stimulus in experimental group (AN/AD).

Mixed ANOVA was done to find out (i) main effect of enhancement i.e., the difference between non enhanced and enhanced conditions when both the groups were combined, (ii) main effect of group i.e., the effect of group AN/AD and normal group when non enhanced and enhanced are compared and (iii) Interaction effect of enhancement and group for both latency and amplitude of LLR.

Table 3: Mixed ANOVA results for each parameter of peaks in terms of F (1, 34) value. .

<table>
<thead>
<tr>
<th></th>
<th>P₁</th>
<th>N₁</th>
<th>P₂</th>
<th>N₂</th>
</tr>
</thead>
<tbody>
<tr>
<td>Enhancement effect</td>
<td>1.717</td>
<td>0.585</td>
<td>1.776</td>
<td>6.174*</td>
</tr>
<tr>
<td>Group effect</td>
<td>0.094</td>
<td>0.154</td>
<td>0.551</td>
<td>1.677</td>
</tr>
<tr>
<td>Interaction</td>
<td>0.170</td>
<td>1.826</td>
<td>0.940</td>
<td>1.505</td>
</tr>
</tbody>
</table>
From Table 3 it can be seen that the latency as well as amplitude of N2 was significantly different for the effect of enhancement when both the group were combined. There were positive results of enhancement meaning to say enhancement did decrease the latency and increased the amplitude for all the peaks but it was statistically significant for N2 peak. However there was no significant difference in terms of latency and amplitude of other peaks i.e., P1, N1 and P2 between non enhanced and enhanced conditions. There was no group effect or interaction effect (between enhancement & group) found for any of the peaks parameter (latency & amplitude).

Once the overall results is calculated (main effect), further analysis was done to see the significant difference if any for the effect of enhancement in both the groups and across the groups. Independent sample t test was done to find out whether there is any significant difference between the groups in terms of latency and amplitude of P1, N1, P2 and N2 considering the non enhanced and enhanced conditions separately. The results are shown in Table 4. It was found that there is no significant difference for any peak parameters in either of the conditions between groups (p>0.05).

Table 4: Shows the “t” value for latency and amplitude parameters in both conditions when comparison was made between the groups.(Independent “t” test result)

<table>
<thead>
<tr>
<th>“t” value for LLR peaks in latency measures between both groups.</th>
<th>P1</th>
<th>N1</th>
<th>P2</th>
<th>N2</th>
</tr>
</thead>
<tbody>
<tr>
<td>Non enhanced</td>
<td>0.147</td>
<td>0.848</td>
<td>0.398</td>
<td>0.532</td>
</tr>
<tr>
<td>Enhanced</td>
<td>0.439</td>
<td>0.047</td>
<td>0.981</td>
<td>1.539</td>
</tr>
</tbody>
</table>

<table>
<thead>
<tr>
<th>“t” value for LLR peaks in amplitude measures between both groups.</th>
<th>P1</th>
<th>N1</th>
<th>P2</th>
<th>N2</th>
</tr>
</thead>
<tbody>
<tr>
<td>Non enhanced</td>
<td>0.468</td>
<td>1.375</td>
<td>1.930</td>
<td>0.713</td>
</tr>
<tr>
<td>Enhanced</td>
<td>1.363</td>
<td>0.155</td>
<td>0.872</td>
<td>0.117</td>
</tr>
</tbody>
</table>

Paired sample t test was done to find out whether there is any significant difference within group when compared between non enhanced and enhanced conditions.

Table 5: Paired t test results in AN/AD (experimental group) between non enhanced and enhanced conditions for the peaks of LLR parameters in terms of t (17) value. Shaded box represents significant results (p < 0.05 level).
“t” values for latency of LLR peaks

<table>
<thead>
<tr>
<th></th>
<th>P1</th>
<th>N1</th>
<th>P2</th>
<th>N2</th>
</tr>
</thead>
<tbody>
<tr>
<td>NEL Vs EL</td>
<td>1.207</td>
<td>0.327</td>
<td>0.214</td>
<td>0.706</td>
</tr>
</tbody>
</table>

“t” values for amplitude of LLR peaks

|       | NEA Vs EA | 2.596 | 0.578 | 0.755 | 1.428 |

It can be seen from table 5 that there is significant difference for P1 amplitude in non enhanced condition when compared with enhanced condition, but was not seen for any other peaks i.e. N1, P2 and N2. Though there was increase in amplitude for all the peaks, it was statistically significant only for peak P1 in AN/AD group. Also there was no statistically significant difference in latency parameter for any of the peaks when comparison was made though there was increase in latency.

Table 6: Paired t test results in normal’s (control group) between non enhanced and enhanced conditions for the peaks of LLR parameters in terms of t (17) value. Shaded box represents significant results (p < 0.05 level).

<table>
<thead>
<tr>
<th></th>
<th>Latency</th>
<th>P1</th>
<th>N1</th>
<th>P2</th>
<th>N2</th>
</tr>
</thead>
<tbody>
<tr>
<td>NEL Vs EL</td>
<td>0.641</td>
<td>2.383</td>
<td>2.177</td>
<td>3.808</td>
<td></td>
</tr>
</tbody>
</table>

|       | Amplitude | NEA Vs EA | 0.126 | 2.234 | 1.571 | 1.674 |

From table 6 it can be seen that the latencies for the two conditions were significantly different for peaks N1, P2 and N2 (p < 0.05). However, it was not so for P1 latency. For amplitude there was significant difference in N1 peak (p < 0.05) and there was no significant difference for peaks P1, P2 and N2.

Late latency responses was recordable in all the subjects with AN/AD and also in all the normal hearing individuals. Earlier studies have also reported the presence of late latency responses in individuals with AN/AD (Starr et al.1996; Hood, 1998; Kraus et al., 2000; Rance et al., 2002; Pearce, Golding & Dillon, 2007) and also in normal hearing subjects( Kurtzberg, Hilbert, Kreuzer, and Vaughan, 1984). The late latency responses may be present in the individuals with AN/AD due to the fact that the disruption of peripheral function which often leads to absence of ABRs, does not necessarily affect the later responses as these are not reliant on timing as the earlier evoked responses (Hood, 1998, Rapin & Gravel, 2003).

However the hit rate of LLR in the present study is higher than that reported in literature. In the present study LLR was present in all the AN/AD subjects. Rance et al. (2002) reported the presence of LLR in 50% of the AN/AD individuals. The difference
may be due to the difference between the subject selection criteria in the two studies. The subjects in the study by Rance et al (2002) were aged between 3.4 years to 9 years, who were born prematurely, whereas in the present study all the subjects were aged above 10 years, with no history of prematurity. Ponton et al. (2000) and Wunderlich and Cone-Wesson (2006) report about the absence of LLR due to maturational factors. It is possible that the auditory development was still underway in the subjects of Rance et al (2002) study and hence LLR was absent.

The results of the present study reveal that there is no difference in terms of latency of LLR in normals and the AN/AD group. This is comparable to the previous study by Starr et al. (2003). Starr et al also reported no significant latency differences in LLR between normals and the AN/AD group at higher intensities whereas there was a significant difference in latency at the lower intensities. In present study, a high intensity (90 dB nHL) presentation was used to record LLR. Starr et al. (2003) reports ‘the no significant difference’ in terms of latency between the two groups at higher intensities may be due to the fact that in AN there may be a form of ‘central recruitment’ which may accompany hearing impairment at higher intensities. Cody et al. (1968) described an abnormal growth of N100 amplitude as a function of signal intensity in individuals with ‘sensorineural’ hearing loss, and speculated as to its relationship to abnormal growth of loudness often encountered in such patients. For AN subjects, however psychoacoustic measures of intensity processes are normal in contrast to their marked abnormality of temporal processes (Zeng et al., 1999). The mechanisms underlying altered cortical excitability in AN may reside within the cortex. An animal model of AN showing increased excitability of auditory cortex did not have a corresponding excitability change of inferior colliculus (Salvi et al., 1999). The abnormal excitability of auditory cortex in AN may be likened to the central excitability changes encountered in disorders of other sensory systems following differentiation”.

The mean latencies for the LLR for non enhanced signal in the present study for the AN/AD group was 71.11 msec for P1, 121.66 msec for N1, 185 msec for P2 and 235.33 msec for N2. The latencies for LLR are lesser than reported by Rance et al (2002). Rance et al (2002) reported 140.2 msec for P1, 227.7 msec for N1 and 320.9 msec for P2. The difference in latencies may be attributable to the difference in the subject’s selection criteria and the stimulus used between the two studies. As mentioned earlier in the study of Rance et al (2002) the subjects had the history of prematurity but there was no such history of premature birth in the subjects for the present study. The stimulus used by Rance et al (2002) was 440 Hz tone burst and /daed/ whereas in the present study the speech stimulus /da/ was used. The latencies for the normal hearing group in the present study was 72.11msec for P1, 129.61msec for N1, 180.22msec for P2 and 228.44msec for N2 respectively. However the mean latencies in the study of Rance et al., (2002) for normal hearing group was 100msec for P1, 200msec for N1 and 301.5 msec for P2, while
Cunnigham, Nicol, Zecker and Kraus (2000) reported latencies for the different age groups for a synthetic syllable (CV) as follows:

Table 7: Peak Latencies (msec) in different age groups for synthetic CV syllable /GA/.

<table>
<thead>
<tr>
<th>Age groups in years</th>
<th>Peak Latencies in msec</th>
</tr>
</thead>
<tbody>
<tr>
<td></td>
<td>P1</td>
</tr>
<tr>
<td>11-12</td>
<td>88</td>
</tr>
<tr>
<td>13-15</td>
<td>80</td>
</tr>
<tr>
<td>19-27</td>
<td>64</td>
</tr>
<tr>
<td>55-78</td>
<td>68</td>
</tr>
</tbody>
</table>

Again these differences between the present study and the study reported could be due to the wide range of subjects (10 yrs to 26 yrs) which were age and gender matched when selecting for the control group. It could also be due to the stimuli used for the testing.

Cunnigham, Nicol, King, Zecker and Kraus 2002 reported that “stimulus modifications that improve the temporal precision of individual neural firing patterns can enhance neural synchrony, across a population of cortical neuron, leading to large amplitude aggregating neural response”. So if there is large amplitude due to aggregation of neural response there has to be reduced latencies, as it was seen in the present study for both the groups. However, when latency was compared for non enhanced and enhanced stimulus there was no significant difference in AN/AD group for any of the peaks recorded, while there was significant difference in normal group for N1, P2 and N2 peaks. This difference in the presence of the significant enhancement effect of the peaks in normal’s could be due to the preserved synchrony which was absent for the AN/AD group. Though the LLR was present in the AN/AD group, the enhanced condition did result in betterment of the latency but it was not significant (p > 0.05). it is possible that the reduced synchrony in subjects with AN/AD did not facilitate improvement in latency or amplitude. It is also possible that the amount of enhancement was not adequate to bring about such a change.

In the present study mean absolute amplitude for the AN/AD group was 1.23 uV for P1, -3.57uV for N1, 1.74uV for P2 and -3.41uV for N2. Rance et al (2002) reported 4.1 uV for P1N1 and 3.4 uV for N1P2. Cunnigham et al (2000) gives the baseline amplitude as follows:
Table 8: Peak amplitude (uV) in different age groups for synthetic CV syllable/GA/.
(Cunnigham et al 2000)

<table>
<thead>
<tr>
<th>Group age Years</th>
<th>Peaks (uV)</th>
</tr>
</thead>
<tbody>
<tr>
<td></td>
<td>P1</td>
</tr>
<tr>
<td>11-12</td>
<td>1.5</td>
</tr>
<tr>
<td>13-15</td>
<td>1.0</td>
</tr>
<tr>
<td>19-27</td>
<td>0.8</td>
</tr>
<tr>
<td>55-78</td>
<td>1.3</td>
</tr>
</tbody>
</table>

The difference in amplitude between the present study and the reported studies could be due to the subject selection criteria, the stimulus used the method of marking amplitude. Relative amplitude was considered by Rance et al (2002) whereas absolute amplitude was considered by Cunnigham et al., (2000) study as done in the present study.

When amplitude was compared between non enhanced and enhanced conditions there was increment in the enhanced condition for both AN/AD group as well as the Normal group. The reason for this was explained earlier as per the study by Cunnigham et al 2000 which is due to better synchrony. In the present study there was significant difference between non enhanced and enhanced stimuli for P<sub>1</sub> (p < 0.05) in AN/AD group, whereas in normal group it was N<sub>1</sub> (p < 0.05). This result may be due to the difference in the feature of synchrony preserved i.e normal hearing group had better synchrony than the AN/AD group, which could have lead to better amplitude. The significant difference is seen only for N<sub>1</sub> peak, is not explainable, more research is needed to discuss for the same.

LLR waveform morphology analysis was done by two judges (audiologist) on 3 point rating scale namely, good, average and poor. It was found that enhancement gave poorer waveform compared to non enhanced stimulus recording in 50 % of the subjects and for the rest 50 % it was similar morphology irrespective of using enhanced or non enhanced stimuli. The results are inconclusive to say regarding the changes in the waveform morphology due to enhancement.

Results on perceptual testing: Total scores shows that there is improvement in scores in quiet from 53.55 % to 60 % between non enhanced and enhanced condition whereas in 10 dB SNR condition it was from 40.36 % to 53% between non enhanced and enhanced condition. The range calculated for non enhanced and enhanced conditions clearly shows that in enhanced condition the range has reduced in both quiet as well as 10 dB SNR condition. Perceptual testing results revealed that there is less improvement in the quite condition i.e., 6.45 % whereas in 10 dB SNR condition it was 12.64 % which is almost double than in quiet condition. In control group there is improvement in scores in quiet from 97.47 % to 97.97 % between non enhanced and enhanced condition. In 10 dB SNR it is from 95.95 % to 97.47 % between non enhanced and enhanced condition.
the normal group there was more improvement in the 10 dB SNR condition i.e., 1.52% than 0.50% in quiet, but it shows that there is marginal improvement.

Summary and Conclusions

Auditory neuropathy/dys-synchrony is a disorder characterized by the impairment of the peripheral auditory function with the preservation of outer hair cell integrity (Starr, Sininger, Picton, Hood and Berlin, 1996; Berlin et al., 1998; Berlin, 1999). It is a known fact that these individuals have problem with speech discrimination. Speech identification scores of subjects with AN/AD are widely documented to be disproportionate to their degree of hearing loss. To overcome this difficulty many management options have been advocated from sign language to cochlear implant, but none of them have given 100% success. Research is underway on the management issues of subjects having AN/AD. Enhancement of the cues in the speech is reported to make speech identification better. This has been tested with subjects having normal hearing and cochlear hearing loss, wherein improvement in speech identification has been reported.

AN/AD group have been reported to have temporal deficits, and hence have difficulty in recognizing short signals. Therefore, present study was carried out to see whether enhancing the speech temporal envelope will improve the speech perception or not. This was done both objectively (LLR) and subjectively (SIS).

11 VCV syllables were recorded using an adult male voice by using PRAAT software. These syllables were further mixed up in preset proportion of speech noise to make it in 10 dB SNR condition. In perceptual testing the subjects task was to repeat the stimuli is heard, in both for quiet as well as 10 dB SNR condition for non enhanced and enhanced stimuli, whereas objective recording was done using only one stimulus /da/, where latency, amplitude and morphology of LLR were recorded in quiet condition for non enhanced and enhanced stimulus. The testing was done for AN/AD subjects as well as age and gender matched subjects with normal hearing.

Perceptual testing was done using OB 922 clinical audiometer and a Pentium IV computer to route the recorded speech stimuli. This testing was done using TDH-39 headphones at 40 dB SL to the pure tone average. In objective recording was done using Intelligent Hearing System (Smart EP windows USB version 3.91). The stimulus /da/ was loaded in the software and then the LLR testing was carried out at 90 dB nHL with repetition rate of 1.1/s and alternating polarity. 3 site electrode placements were used, and the mode of presentation of kept ipsilateral, filter setting 1-30 Hz, with a gain of 50,000. All together 150 sweeps were considered with artifact rejection at 40 uV.

The latency and absolute amplitude were noted, with comment over morphology in both non enhanced and enhanced condition. All the recording was done twice to check for the replicibility. SPSS version 15 was used for the analysis of the data obtained.
Mixed ANOVA, Independent sample “t” test and Paired “t” test was done. Results revealed that LLR was present in all the subjects taken for the study with the following effect:

1) Mean values of Latency in enhanced condition was lesser in experimental as well as control group, with a larger standard deviation in experimental group implying heterogeneity of the experimental group.

2) Also the mean value for amplitude was higher in enhanced condition for both experimental and control group with larger standard deviation in experimental group.

3) There was significant difference for latency and amplitude of peak N2 between non enhanced and enhanced condition, when both experimental and control group were combined.

4) But there was no significant difference in latency or amplitude for any other peaks of the LLR, when tested between the groups for the non enhanced and enhanced condition.

5) On comparison between non enhancement and enhancement within experimental group it was found that there is significant difference in amplitude of P1 peak only.

6) Whereas comparison between non enhancement and enhancement within control group revealed that there is significant difference in latency of N1, P2 and N2 peaks, also in amplitude of N1 peak.

7) Perceptual testing showed the improvement with enhancement, more in 10 dB SNR condition, in both experimental as well as control group, though it was very less.

8) Morphology of the waveform was degraded in 50 % of the subjects and remained same in another 50 % of the subjects over a 3 point rating scale in enhanced condition.

**Based on the results following conclusions were made:**

1) Enhancement does help in improvement of speech identification scores majorly, for AN/AD in 10 dB SNR condition.

2) Enhancement lead to the decrease in latency and increase in the amplitude of LLR peaks in AN/AD group and normal hearing group.

3) Few more studies on the similar topic are advocated taking more number of subjects and more stimuli to record the LLR further, to illustrate the effect of enhancement, so that if there is significant improvement this strategy may help subjects with AN/AD.

**Limitations of the study:**

1) The configuration of hearing loss was not controlled in the experimental group.
2) The speech identification scores also varied in the experimental group.
3) If the subjective recording were also done in 10 dB SNR condition the results would have made provision to make observation on the enhancement effect.
4) The stimulus used for objective recording (LLR) was only /da/, more number of stimuli would have given better information on the effect of envelope enhancement in subjects with AN/AD.

References


Effect of Different Signal Enhancing Technologies on Speech Recognition in Noise

Dhanya V. K. & K. Rajalakshmi*

Abstract

A major consequence of sensori neural hearing loss (SNHL) is communicative difficulty, especially in the presence of noise and/or reverberation. The purpose of this investigation was to compare three types of technologies that have been shown to improve the speech perception performance of individual with SNHL: directional microphones (DMic), digital noise reduction (DNR) and frequency modulation (FM) system. 23 adult subjects with moderate to moderately severe SNHL served as subjects. Speech identification scores and signal to noise ratio (SNR) measurements were used to understand the benefit of these technologies in the presence of noise. These measurements were carried out in four listening conditions such as unaided, DMic, DMic+DNR and FM. Results revealed that speech perception in noise was significantly better with FM technology than with the other two listening conditions (DMic and DMic+DNR) in both SNR and speech identification measurement. There was no significant difference in performance between DMic and DMic+DNR listening conditions in SNR measurement. Even though the statistical analysis showed significant difference in performance between DMic and DMic+DNR listening condition in speech identification there was only an average of 2-3% improvement in speech identification with DMic+DNR over DMic.

Introduction

With appropriate prescription and fitting, a hearing aid can significantly improve speech recognition scores for an individual with hearing impairment in quiet and non-reverberant listening environment. This benefit, however, is greatly reduced in presence of noise, especially for individuals with higher degrees of hearing loss (Killion and Niquette, 2000). Hence, one of the challenges in providing amplification for the hearing impaired population is to select the technology that will provide the maximum benefit in background noise or competing speech. The most effective ways to improve speech recognition in noise is to improve the signal to noise ratio (SNR). Frequency modulation (FM) systems and directional microphones (DMic) are two examples of such technological advances (Hawkins, 1984; Lewis, Crandell, Valente and Horn, 2004). Automatic noise reduction or automatic signal processing is also one of the technologies designed to potentially increase intelligibility in noise (Graup et al., 1986).

Directional microphones typically use a cardioid polar plot sensitivity pattern, it means that they reduce signals originating from the rear and the sides and only amplify signal arriving from the front-where the speaker will often be located. Numerous investigations have demonstrated that directional microphone technology can improve speech intelligibility in noise by as much as 3 to 8 dB (Valente, Fabry & Potts (1995); Kuk, Ludvigsen and Paludan-Muller (2002); Ricketts and Dhar, 1999; Valente, Schuchman, Potts, and Beck (2000).

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Personal FM system has also been shown to improve speech intelligibility in noise (Hawkins, 1984; Fabry 1994; Crandel and Smaldino, 2000). Past investigations have demonstrated that the utilization of FM technology can improve speech intelligibility in noise by as much as 20-25 dB (Crandel and Smaldino, 2000). With personal FM system, the speaker’s voice is picked-up via FM wireless microphone located near speaker’s mouth – where the effect of reverberation, distance, and noise are minimal. The FM system converts the acoustic signal to an electrical waveform at the microphone, and the signal is transmitted via FM signal, from the transmitter to the receiver. Both the transmitter and the receiver are tuned to the same transmitting and receiving frequency. At the receiver end, the electrical signal is amplified, converted back to an acoustical waveform and conveyed to the listener.

The term ‘Digital noise reduction (DNR) will be used to describe processing from a digital hearing aid which aims to provide less amplification for noise than speech. DNR algorithm relies on difference in physical characteristics of a signal to distinguish speech from noise (Rickets and Hornsby 2005).

Studies on the efficacy of DNR algorithms are less frequent in literature, and their conclusions are often inconsistent. Although listeners often demonstrate a strong tendency for subjective preference for DNR algorithms (Boymans and Dreschler, 2000), actual improvement in speech perception in reportedly unreliable. An implementation of DNR processing is to at least provide improved sound quality for speech in noise, in the absence of improved speech recognition (Ricketts and Hornsby 2005).

Despite the documented enhancement in speech intelligibility with directional microphone and FM technologies, only a few investigations have attempted to directly compare these two. Hawkins (1984) evaluated the speech intelligibility of children utilizing these two types of technologies (FM technology and DMic). Results demonstrated that FM technology, FM only mode provided significantly better speech recognition in noise when compared to directional microphone technology.

Lewis, Crandell, Valente and Horn, (2004) studied the speech perception ability of adults with mild to severe sensory neural hearing loss in noisy background utilizing directional microphone and FM technology. Results from this investigation indicate that FM system provides significantly improved speech intelligibility over the omni directional microphone (22.74 dB) and directional microphone (19.3 dB) listening conditions.

In practice DMic and DNR technologies are used in conjunction. Their interaction and resultant effect on speech perception in noise were studied by Nordrum and Dhar (2006). Results showed 50% of the participants performed better with both DMic and DNR activated in conjunction, while the other 50% performed better in the DMic only condition.
There are no studies which compare the effect of all the three technologies on signal enhancement in the presence of noise. Hence, main focus of the study was to compare the speech recognition in noise with DMic, DMic and DNR combined (DMic+DNR) and FM technology.

**Aims of the Study:**

The study aims to

1) Compare the speech identification scores in noise in following listening conditions,
   a) Monaural digital BTE in DMic mode (DMic).
   b) Monaural digital BTE in DMic + DNR condition (DMic+DNR).
   c) Monaural digital BTE utilized with one Microlink MLxS FM receiver in FM only mode (FM).

2) compare the Speech Recognition Threshold in noise in terms of SNR in the following conditions,
   a. Monaural digital BTE in DMic mode (DMic).
   b. Monaural digital BTE in DMic + DNR condition (DMic+DNR).
   c. Monaural digital BTE utilized with one Microlink MLxS FM receiver in FM only mode (FM).

**Method**

**Subjects:** Twenty three post-lingually hearing impaired subjects in the age range of 20 to 60 years (mean age of 51 years) served as the participants in the study. All subjects had bilateral gradually sloping moderate to moderately severe sensory neural hearing loss with a mean pure tone average of 65dBHL. Their speech identification score was greater than 60%. No indication of middle ear pathology as confirmed by tympanometry. They were native speakers of Kannada language and were experienced hearing aid users for more than 6 months.

**Instrumentation:** A calibrated dual channel diagnostic audiometer (Madsen orbiter 922) with TDH-39 head phone, bone vibrator B-71 and Martin (c115) speakers were used.

A Calibrated immittance meter (GSI-Tymstar) was used to rule out middle ear pathology.

Nonlinear digital BTE hearing aid which had options for directional microphone, digital noise reduction algorithm and FM compatibility (direct audio input) was used.

A Pentium IV computer with NOAH-3 software was used to program the hearing aid. Hi-pro was used to connect the hearing aid with computer.

A calibrated dual channel audiometer (Madsen orbiter 922) with two Martin (c115) speakers was used for the hearing aid testing. With input from a Pentium IV
computer, the channel one of the audiometer was used to deliver the recorded speech material and the channel two of the audiometer was used to deliver speech babble.

Multifrequency FM transmitter and Microlink MLxS receiver was used in the study. The FM receiver was connected to the hearing aid with an appropriate audio shoe.

**Stimulus:** The phonetically balanced list in Kannada developed by Yathiraj and Vijayalakshmi (2005) was used in the study. The speech material consists of 4 phonetically balanced wordlist and each list has 25 words. The words were spoken in conversational style by a female native speaker of Kannada and were digitally recorded in acoustically treated room; on a data acquisition system using 44.1 kHz sampling frequency and 16 bit analogue to digital converter. Kannada speech babble developed by Anitha and Manjula (2005) was used as noise in the study.

The testing was done in sound treated double room. The ambient noise level inside the test room was within the permissible limits (re: ANSI S3.1 1991, as cited in Wilber 1994).

**Procedure:** The conditions used in the study were the following:

1. Monaural digital BTE hearing aid in directional mode (DMic).
2. Monaural digital BTE hearing aid in DMic with DNR.
3. Monaural digital BTE hearing aid connected to Micro link FM receiver in the FM only mode.

The hearing aid was programmed based on the audiometric thresholds using NAL-NL1 fitting formula. The participants were seated comfortably and were fitted with hearing aid on the test ear with appropriately sized ear tips. The hearing aid was fine tuned depending on the subject’s listening needs by manipulating the low cut, high cut gain and the cut-off frequency values. Two programs were stored in the hearing aid, in the first program DMic was activated, whereas in the second program both DMic and DNR were activated. Other parameters of the hearing aids were kept at default setting.

In the present study the test hearing aid used had a 16 channel modulation based digital noise reduction system and an adaptive Wiener filter in its DNR processing scheme. The DMic used in this study has a hyper cardioid polar pattern which suppresses noise coming from one direction (rear end) while retaining good sensitivity to sound arriving from the other direction (front end).

In the third condition, in addition to the hearing aid the subject was also fitted with Microlink MLXs FM receiver. The FM receiver was attached to the hearing aid directly with the audio shoe and the "FM only" mode was selected. Synchronization of the FM transmitter and receiver was made according to protocols specified by the manufacturer. The FM transmitter was placed on a stand located 7.5 cm from the loud speaker at a height of 0.5 meters to simulate ideal user position.
The testing was carried out in two phases: Speech identification in noise measurement and Speech recognition threshold in noise measurement. Among the 23 participants 11 subjects were randomly selected for speech identification measurement and 12 subjects for speech recognition testing.

**Phase 1: speech identification in noise measurement.**

The testing was done in a sound treated double room. The participant was seated at a distance of 1 meter from the loud speakers. Recorded speech material was presented from a loud speaker positioned at 0° azimuth and noise was presented at 180° azimuth. Speech identification score was measured in two signal-to-noise ratio’s (SNR) 0 dB and +10 dB, the signal level was kept constant at 45 dB HL.

The order of listening conditions was randomized for each of the 11 participants tested. The participants were asked to repeat the words presented. The words correctly repeated were given a correct score of one; the words incorrectly repeated or missed out were not scored.

The speech identification measurements were done in the three listening conditions, namely

1. Monaural digital BTE hearing aid in directional mode (DMic).
2. Monaural digital BTE hearing aid in DMic with DNR.
3. Monaural digital BTE hearing aid connected to Micro link FM receiver in the FM only mode.

**Phase 2: Speech Recognition Threshold in Noise in terms of Signal to Nose Ratio (SNR)**

In this study, SNR is defined as the level at which the participant is able to repeat two out of three words (66.6% criterion) presented in noise. An adaptive procedure was utilized to establish the SNR. In this procedure, intensity of speech stimuli was held constant at 50 dB HL. The noise level was set 15 dB below the signal and systematically varied in 2 dB steps based on the participant’s response. The noise level was varied until the subject repeats 2 words out of the three words presented. The noise level was subtracted from the speech level to find the SNR.

The performance was evaluated in three listening conditions, namely

1. Monaural digital BTE hearing aid in directional mode (DMic).
2. Monaural digital BTE hearing aid in DMic with DNR.
3. Monaural digital BTE hearing aid connected to Micro link FM receiver in the FM only mode
Results and Discussion

The present study was carried out to compare the benefit of various hearing aid technologies (DMic, DMic+DNR and FM) designed to improve speech understanding in noise. Speech identification scores and SNR measurements were used to understand the benefit of these technologies in the presence of noise. Speech identification testing was carried out in eleven subjects and SNR measurements were carried out in twelve subjects. All subjects had bilateral gradually sloping moderate to moderately severe sensory neural hearing loss with a mean pure tone average of 65dBHL. They were native speakers of Kannada language and all were experienced hearing aid users of more than 6 months. The data was appropriately tabulated and statistically analyzed using SPSS (15.0) version. Repeated measure ANOVA was used for statistical analysis.

Speech Identification Measurement

Speech identification measurement was carried out at two SNR’s (0 and +10dB) in eleven subjects in three listening conditions namely DMic, DMic+DNR and FM. Mean and standard deviation for each of these conditions at two SNRs are depicted in Figure 1.

![Figure 1 Comparison of mean speech identification scores across the three listening conditions (DMic, DMic+DNR and FM) in 0 and 10dB SNR](image)

From figure 1, it can be observed that mean speech identification performance is higher with FM compared to other two listening conditions (DMic and DMic+DNR). FM listening condition had an average of 10 to 14% greater improvement in speech identification at 0 dB and 10 dB SNR over DMic and DMic+DNR. Among DMic and DMic+DNR listening condition, the mean speech identification score was better in DMic + DNR by 2% at 0 dB SNR and 5% at 10dBSNR.

Repeated measure ANOVA was performed to assess the difference in speech identification scores across the three listening conditions (DMic, DMic+DNR and FM) at two SNR (0 dB SNR and 10 dB SNR), with listening conditions and SNR as within group factors.

Analysis revealed a significant main effect of listening conditions (DMic, DMic+DNR) (F (2, 20) = 76.04, P<0.001) and SNR (F (1, 10) = 26.01, P<0.001). Interaction analysis revealed that there is no significant interaction between listening conditions and SNR (F (2, 20) =3.01, P=0.072). As there was significant difference
between speech identification performance in the listening conditions multiple comparison using Bonferroni's test was performed for the three listening conditions, DMic, DMic+DNR and FM. Results showed that there was significant difference between DMic and DMic+DNR (P<0.05) listening conditions, DMic+DNR and FM (P<0.001) listening conditions, and DMic and FM (P<0.001) listening conditions.

Earlier research indicates significant improvement in hearing-in-noise performance with the use of DMic and FM. However, DNR has shown improvement in listening comfort rather than improvement in speech recognition in the presence of noise (Ricketts and Hornsby, 2005). In the present study there was significant difference in speech identification scores across DMic, DMic+DNR, and FM listening conditions. This finding is in contrast to the previous studies which have showed no significant improvement in speech perception in noise when using a DNR algorithm in isolation or in conjunction with directional microphone (Walden et al., 2000; Ricketts and Hornsby, 2005).

It is difficult to compare across studies, because of the different procedures employed in estimating the benefit of these technologies in noise. Even though the statistical analysis showed significant difference in performance between DMic and DMic+DNR, there was only an average of 2-3% improvement in speech identification with DMic+DNR over DMic. Hence, this improvement cannot be considered as a drastic improvement in speech identification in the presence of noise.

Similarly, studies suggest that DNR algorithms may be effective in improving speech perception in noise when the speech and noise sources are not spatially separated (Bray et al. 2002) or when the noise field is isotropic (Bray & Nilsson, 2001).

However, Ricketts and Hornsby (2005) studied the effect of digital noise reduction (DNR) processing on aided speech recognition and sound quality measures in a commercial hearing aid. The results revealed that the presence or absence of DNR processing did not impact speech recognition in noise (either positively or negatively). Paired comparisons of sound quality for the same speech in noise signals, however, revealed a strong preference for DNR processing. These data suggest that at least one implementation of DNR processing is capable of providing improved sound quality, for speech in noise, in the absence of improved speech recognition.

Hawkins (1984) demonstrated that the FM condition provided a significant improvement in speech identification scores in the presence of noise. Nelson, LaRue and Rourk (2004) fitted subjects monaurally with a unidirectional linearly programmed hearing aid and later coupled their hearing aid to a Phonak MLx FM receiver with DAI. It was found that the improvement in word identification scores were statistically significant for the FM condition when compared to the hearing aid alone condition.

To summarize, all the three conditions showed significant improvement in speech identification scores in the presence of noise. Though, the improvement with DMic and
DMic + DNR condition was statistically significant, the improvement in scores were minimal, this could be attributed to the difference in the speed and magnitude of gain reduction for the steady state signal across channels as well as the type of the competing signal (speech babble) used in this study. The improvement with FM was statistically significant over DMic and DMic + DNR conditions, this could be attributed to the improved signal-to-noise ratio provided with the FM system as it overcomes the effect of distance, reverberation and background noise.

(b) Speech Recognition Threshold in Noise in terms of Signal to Noise Ratio (SNR)

In this study, SNR is defined as the level at which the participant is able to repeat two out of three words (66.6% criterion) presented in noise. Repeated measures ANOVA were carried out to compare the SNR across various listening conditions namely unaided condition, DMic, DMic + DNR and FM conditions. Repeated measure ANOVA revealed that there is significant difference in SNR (F (3, 33) = 329.086, p<0.001) across the three listening conditions. Bonferroni’s multiple comparison test showed that there was no significant difference in SNR across DMic and DMic+DNR listening condition (P>0.05). However, there was significant difference between FM and each of the two other listening conditions (DMic, DMic+DNR).

Figure 2 Comparison of mean SNR across different listening conditions (DMic, DMic + DNR and FM).

From the figure 2, it can be noted that the speech recognition performance (better SNR) was better with FM condition than with other listening conditions (DMic and DMic+DNR). In DMic listening condition, the subjects required an average SNR of 3.83 dB, while subjects required an average SNR of 4.33 dB in DMic+DNR condition. It can also be inferred from the figure that use of DMic resulted in an improvement of 6.75 dB over the unaided condition, while use of DMic+DNR resulted in an improvement of 6.25 dB over the unaided condition. Hence, it can be concluded that the use of DMic+DNR resulted in increment in SNR of only 0.5 dB over DMic listening condition, however this improvement in SNR was not statistically significant. In the FM condition subjects required SNR of -10.25; hence the use of the FM resulted in an improvement in SNR of
14.58 over DMic+DNR condition and 14.08 over DMic condition. This improvement in SNR with FM was statistically significant over the other two conditions.

In general, individuals with hearing impairment require the speech signal to be 4 to 18 dB higher than extraneous background noise in order to obtain speech recognition scores similar to individuals with normal hearing (Killion 1997a; Moore 1997). Similarly, Killion (1997b), suggests that individuals with pure-tone averages of 65dB (PTA of individuals in present study) require an average SNR of 7-9 dB in order to obtain 50% correct on the Speech-In-Noise (SIN) test when the signal is presented at 70 dB HL. The subjects in this study required a SNR of approximately 10.58 in the unaided listening condition, which is in accordance with the results of Killion (1997).

In DMic condition there was an improvement of 6.75 dB over unaided condition. This finding is in accordance with the study by Lurquin and Rafthy (1996), where they obtained a statistically significant difference in SNR of 6.8 dB between unaided and directional microphone condition in similar experimental set up as in the present study.

In DMic+DNR condition there was only 6.25 dB advantage over the unaided condition. However, there was no significant difference in speech recognition threshold between DMic and DMic+DNR conditions. These finding are in agreement with the past researches (Walden et al 2000, Ricketts and Hornsby 2005), where it was concluded that there was no significant difference in the speech recognition in noise threshold between DMic and DMic+DNR conditions.

The best speech identification scores and better speech recognition in noise threshold (SNR) was found when subject fitted with FM than DMic or DMic+DNR. FM provided an improvement in SNR of 20.83 dB over unaided and 14.08 over DMic condition.

The results are similar to the conclusions derived from these studies:

Hawkins (1984) concluded that the FM only condition provided a significant improvement over DMic and DMic+DNR conditions (15.3 dB).

Similarly, Lewis and Crandall (2006) reported that monaural FM resulted in an improvement of SNR of 14.2 dB over directional microphone. In these studies, the proximity of the FM transmitter to the desired signal reduces the effects of noise, distance, and reverberation in a better way than hearing aids. This could be the reason for the improved speech recognition with FM technology.

To summarize, for the assessment of benefit from the three technologies (DMic, DMic + DNR and FM conditions) two methods were employed:

1) Speech identification scores
2) SNR measurement.
In the present study, an improvement of 14.08 dB was observed with FM technology over DMic in SNR measurement, whereas only 15% improvement was observed in the speech identification measurement with FM technology over DMic. This difference in the benefits across these methods could be attributed to the variability in the measurement procedures. One other reason for this difference in benefit could be the ceiling effect observed with speech identification (1st method) scores due to which the advantage of FM system could not be completely assessed.

Overall, from the results of this investigation it can be concluded that FM technology significantly improves the speech intelligibility scores over the hearing aid conditions (DMic and DMic+DNR conditions) in the presence of noise. This data suggests that FM technology will offer significantly better communicative performance in adverse listening situations than any type of hearing aid microphone configuration or microphone with digital noise reduction configuration. Speech recognition in the presence of noise does not improve across DMic and DMic+DNR condition, this could be attributed to the DNR technology (modulation detection based noise reduction) used in the hearing aid and the type of noise used in this study.

References


The Effect of Various Modes of Excitation of Sterno-Cleido Mastoid Muscle on Vestibular Evoked Myogenic Potentials (VEMP)

Gubba Vijay Shankar & Vijayalakshmi Basavaraj

Abstract

The Vestibular Evoked Myogenic Potential (VEMP) is an inhibitory potential recorded from the Sternocleidomastoid muscle (SCM) in response to loud sounds. VEMPs are short latency Electromyograms (EMG) that are evoked by higher-level acoustic stimuli and are recorded from surface electrodes over the tonically contracted SCM muscle. VEMP testing is considered to be easy, but there are a lot of technical pitfalls such as, operator pitfalls, assuring neck muscle activation, electrical artifact, mode of neck muscle activation, sound stimulus, electrode layout etc. Research in the area of VEMP have addressed the issues of subject variables and test parameters, but literature does not provide any suggestion to the clinicians about which method/mode of excitation of sternocleidomastoid muscle should be employed. So, as patient comfort and reliability of mode of excitation of sternocleido mastoid are also the factors to be considered while testing, there is a need to study the effect of mode of excitation of SCM muscle on VEMP response. 25 normal male and 25 normal female subjects were tested for VEMP in three different body positions namely: Subject’s body rotated to one side for measurement of VEMP on the opposite side in sitting position, Subject’s body rotated to one side for measurement of VEMP on the opposite side while lying in supine position, subject pressing the Forehead against a soft surface. VEMP measures amplitude of P13-N23, latencies of P13, N23, response rate and wave morphology were analyzed along with the subjective rating on patient comfort in the three positions for all the subjects. Irrespective of gender VEMP measures such as latencies of P13, N23, amplitude of p13-n23, response rate and wave morphology were not affected by body position that is used to excite the sternocleidomastoid muscle during VEMP testing. But, the rating of comfortness was more to second position in males and to third position in females.

Introduction

The Vestibular Evoked Myogenic Potential (VEMP) is an inhibitory potential recorded from the Sternocleidomastoid muscle (SCM) in response to loud sounds.

VEMPs are short latency Electromyograms (EMG) that are evoked by higher-level acoustic stimuli and are recorded from surface electrodes over the tonically contracted SCM muscle. The neurophysiologic and clinical data indicate that the VEMPs are mediated by a pathway that includes the saccular macula, inferior vestibular nerve, the lateral vestibular nucleus, the lateral vestibulospinal tract, and the motor neurons of the ipsilateral SCM muscle (Halmagyi & Curthoys, 2000).

VEMP testing is a new diagnostic tool for professionals who are dealing with assessment of Vestibular and Auditory disorders. VEMPs were first described by Bickford, Jacobson, and Cody (1964), and recently have been proposed as a reliable clinical test of saccular or inferior vestibular nerve function (Colebatch, 2001).

Normal VEMP responses are characterized by biphasic (positive–negative) waves. In a majority of studies, the peaks and troughs are usually labeled with the mean latency

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in milliseconds preceded by the lowercase letters “p” (for positive) or “n” (for negative), as proposed by Yoshie and Okudaira (1969) to distinguish them from neutrally generated evoked potentials. The first positive–negative complex is often labeled as p13–n23 (Colebatch, Halmagyi & Skuse, 1994). Authors have also reported later serial peaks of VEMP labeled as n34 –p44. The VEMP amplitudes are large and vary from a few micro volts, depending on the muscle tension and the intensity of stimuli. (Cheng & Murofushi, 2001a, 2001b).

There are various tests that can be carried out in vestibular clinic, such as, Electronystagmography (ENG), Rotational chair test, tests for postural control Assessment that majorly focus on assessing vestibulo spinal pathways and VEMP test. Among all these tests, ENG is the most widely used test since long, ENG is not a single test but consists of various sub tests in its battery such as caloric test, Gaze test etc., All the tests mentioned above though address the issue of assessing the same problem i.e. vestibular disorder; each test has its own unique significance and hence cannot be replaced by other test. Thus, the information obtained from VEMP testing is different from the information obtained from the results of ENG tests. ENG assess the Semicircular canals, utricle and superior vestibular nerve where as VEMP assess saccule and inferior vestibular nerve.

Review of Literature

Vestibular evoked myogenic potentials (VEMP) are responses from the otolithic organs i.e. the saccule and the utricle to the high intensity acoustic stimulation. These responses can be acquired from the anterior neck muscles, specifically from the Sternocleidomastoid (SCM) muscles. There is initially biphasic positive-negative response (p13-n23) recorded from the averaged EMG which occurs at short latency and ipsilateral to the stimulated ear (Colebatch, Halmagyi & Skuse, 1994).

VEMP has been used as a clinical tool, which provides additional information about disturbances of vestibular function as a result of their dependence upon different vestibular receptors.
1. Neck muscles via the medial vestibulospinal tract (MVST).
2. The leg muscles via the lateral vestibulospinal tract (LVST). (Colebatch, Halmagyi & Skuse 1994; Murofushi, Halmagyi, Yavor, & Colebatch, 1996; Uchino et al., 1997; Kushiro et al., 2000).

Figure 1. Pathway of Vestibular Evoked Myogenic Potential.
Methods for recording VEMP

**VEMPs evoked by bone-conducted stimuli**

A bone-conducted tone burst delivered over the mastoid process via a B71 clinical vibrator (radio ear corporation, Philadelphia, PA), routinely used in audiometric testing, evokes VEMPs despite conductive hearing losses. VEMPs are often bilateral as the stimulus is transmitted via bone and activate end organs on both sides. The ipsilateral VEMP is about 1.5 times larger and occurs approximately 1 millisecond earlier. Rarely larger responses have been recorded contra lateral to the stimulated ear (Sheykholeslami, Murofushi, Kermany & Kaga, 2001; Welgampola, Rosengren, Halmagyi & Colebatch, 2003).

**VEMPs evoked by galvanic simulation**

A short duration (2 millisecond) pulsed current delivered via electrodes attached to the mastoid processes evokes a p13-n23 response on the side ipsilateral to cathodal stimulation. Similar to that evoked by sound stimuli of 4mA/2msec as used for clinical testing are well tolerated by patients. Such a current in close proximately to the recording site causes a large stimulus artifact and specific subtraction techniques are required to recover the response of interest (Watson & Colebatch 1998).

**VEMP in Clinical Use**

*Superior canal dehiscence*

Sven-Olrik., Cremer, Carey, Weg, and Minor (2001), have studied VEMP responses in subjects with Superior Canal Dehiscence and found lowered VEMP thresholds in these subjects and concluded that VEMP can be included in the test battery along with symptoms, signs and CT imaging in diagnosis of Superior Canal Dehiscence syndrome.

*Vestibular neuritis*

Halmagyi, Colebatch and Curthoys (1994) reported in their study that patients who did not have caloric responses on the affected sides indicated dysfunction of the lateral semicircular canal. Results showed VEMPs were normal in 6 patients, reduced in 5 patients and absent in 11 patients. Therefore, results not only suggested that VEMPs were not of lateral canal origin but also revealed different pathologies involved in vestibular neuritis.
Vestibular Schwannoma/Acoustic Neuroma

Itoh, A. (2001), concluded that VEMP test compromises a useful new diagnostic method for identifying lower Brainstem lesions in which ABR is present and VEMP is absent.

Diallo (2006), stated that VEMP can be included in the audio vestibular assessment in diagnosing all sizes of vestibular schwannomas. It was also reported that inter aural latency of VEMP can be a tool to diagnose vestibular schwannomas, (Yang, W., Han, D., & Wu, Z., 2002).

Meniere’s disease (Endolymphatic hydrops)

Robertson and Ireland (1995) reported that VEMPs were absent in all 3 of their patients with Meniere’s disease (MD). VEMPs in patients with MD showed that 54% of the patients had no VEMPs when clicks were used as stimuli (De waele, Hay, Diard, Freyss, & Vidal, 1999). Shojaku, Takemori, Kobayashi and Watanabe (2001), reported similar results in which 8 out of 15 patients with MD had abnormal VEMP amplitude.

Multiple sclerosis

The latencies of a vestibulo spinal reflex can be prolonged in multiple sclerosis (MS). The VEMP delay could be attributed to demyelination either of primary afferent axons at the root entry zone or secondary vestibulo spinal tract axons rather than to lesion involving vestibular nucleus. Measurement of VEMPs could be helpful in detecting sub clinical vestibulospinal lesions in suspected multiple sclerosis (Shimizu, Murofushi & Sakurai, 2000).

Conductive hearing loss

Interference of sound transmission due to some disorders such as chronic otitis media (COM) may lead to absent VEMPs (Young, Wu & Wu, 2002). Tone stimuli rarely elicit VEMP responses in patients with conductive hearing loss (Halmagyi, Colebatch & Curthoys, 1994).

Gentamycin therapy

DeWaele et al. (2002) found that the VEMPs can be used to monitor the effects of low close intra tympanic gentamycin injections used to achieve chemical labyrinthectomy, a procedure used to control debilitating vertigo in Meniere’s disease and other peripheral vestibulopathies.

Auditory Neuropathy

Sheykholeslami, Kaga, Murofushi and Hughes (2000) studied 3 auditory neuropathy patients. These patients also complained of balance disorders. Tests of battery were administered, audiometric tests (pure-tone audiometry and speech
Effect of Various Modes of Excitation Of SCM on VEMP

discrimination tests), Otoacoustic emissions, auditory-evoked brainstem responses and vestibular function tests (clinical tests of balance, electronystagmography, damped rotation tests and VEMPs). VEMP responses were absent in the affected ear. They concluded that, in patients with isolated auditory neuropathy, the vestibular branch of the 8th cranial nerve and its innervated structures may also be affected.

Reviewing the literature available on VEMP testing shows that a majority of the factors affecting the stimulus and response or parameters have been studied and reported. However, there is scanty literature on the variable of the subjects comfort with respect to the various body positions or postures used to excite the sternocleidomastoid muscle. Hence, present study aimed at fulfilling the above mentioned lacuna in literature related to VEMP testing.

Method

The present study aimed at investigating the effect of various modes of excitation of sternocleidomastoid muscle on vestibular evoked myogenic potentials.

Subjects

50 subjects in the age range of 18 to 25 years were selected for the study. This group consisted of 25 male and 25 female subjects. These were selected on the criteria that subject should be
- having bilateral hearing sensitivity within normal limits PTA<20dBHL
- having no history of conductive hearing loss
- having no history of tinnitus and vertigo
- having no history of neuromuscular problems in body & neck region
- having no history of intake of drugs that may lead to vestibulotoxicity

All the subjects underwent case history and basic hearing evaluation for the purpose of meeting subject selection criteria which included pure tone audiometry and impedance audiometry.

Research Design: “within subject design”

Procedure

Puretone thresholds were obtained using modified Hughson - Westlake procedure (Carhart and Jerger, 1959), across octave frequencies (including interoctave frequencies) from 250 Hz to 8 kHz for air conduction stimuli and from 250 Hz to 4 kHz for bone conduction stimuli. Tympanometry was done to rule out middle ear pathology. 226 Hz was the probe frequency at the intensity level of 85 dBSPL.

All the Subjects selected based on the inclusion criteria were subjected to VEMP testing in left ear using three different modes of excitation of sternocleidomastoid muscle.
Instructions were given to subjects regarding the following body postures required in testing:

A. Subject’s body rotated to one side for measurement of VEMP on the opposite side in sitting position
B. Subject’s body rotated to one side for measurement of VEMP on the opposite side while lying in supine position
C. Instructing the subject to press the Forehead against a soft surface

VEMP Testing

- As mentioned in the test protocol, VEMPs were recorded for all the subjects by an averaging of the acoustically evoked electromyogram of the sternocleidomastoid muscle. The site of electrode placement was prepared with skin preparation gel. Silver chloride disc electrodes were used with a conducting gel. Absolute electrode impedances were less than 5kohms and inter-electrode impedances were less than 2 Kohms.

- Subjects were instructed about the three body positions and were asked to maintain the same during the test run. While testing VEMP for each subject in all the three positions subjects the tonic EMG level was maintained between 30 – 50 micro volts. A visual feedback was provided to the subject so as to monitor tonic EMG level of sternocleidomastoid muscle.

- A single-channel recording of the evoked potential was obtained with a non-inverting electrode placed at the mid point of sternocleidomastoid muscle, inverting electrode placed on the sternoclavicular junction, and the ground electrode on the forehead.

- VEMPs were obtained from each subjected evoked by 500 Hz tone bursts (rarefaction onset phase, blackman gating function, two cycle rise-fall time with no plateau) presented at 95 dBpeSPL. The stimuli were presented monaurally to the ear ipsilateral to the activated (excited) sternocleidomastoid muscle via ER3A (Etymotic Research) insert earphones at a rate of 3.1Hz.

- Each run of VEMP test consisted of 75 to 250 no. of sweeps, each run lasting for 45sec to 1min of duration. Minimum of required rest period was given to the subject between each run. VEMP test was done in all the three positions and a rest period was given after testing in each position.

- Replicability of VEMP wave form was checked by performing VEMP on all clients in all modes by testing at least twice. It was also ensured that VEMP is absent at low intensity level i.e. 65 dBpeSPL by testing all subjects in all the three positions at that level. This indirectly confirmed that the VEMP wave form present at higher intensity levels were indeed reliable.
Test protocol for VEMP Testing

Table 1. Stimulus parameters

<table>
<thead>
<tr>
<th>Transducer</th>
<th>Insert ear phones</th>
</tr>
</thead>
<tbody>
<tr>
<td>Type</td>
<td>Tone burst</td>
</tr>
<tr>
<td>Frequency</td>
<td>500 Hz</td>
</tr>
<tr>
<td>Duration</td>
<td>2-0-2 cycle Tone burst</td>
</tr>
<tr>
<td>Intensity</td>
<td>95 dBpeSPL</td>
</tr>
<tr>
<td>Polarity</td>
<td>Rarefaction</td>
</tr>
<tr>
<td>Repetition Rate</td>
<td>3.1/sec Tone burst</td>
</tr>
</tbody>
</table>

Figure 2. Electrode configuration required for VEMP test

<table>
<thead>
<tr>
<th>Inverting (-) electrode</th>
<th>Stemoclavicular junction</th>
</tr>
</thead>
<tbody>
<tr>
<td>Non-inverting (+) electrode</td>
<td>Midpoint of Sternocleidomastoid muscle</td>
</tr>
<tr>
<td>Ground electrode</td>
<td>Forehead</td>
</tr>
<tr>
<td>Stimulator</td>
<td>Side being tested</td>
</tr>
</tbody>
</table>

Table 2. Acquisition parameters for VEMP test

<table>
<thead>
<tr>
<th>Analysis time</th>
<th></th>
</tr>
</thead>
<tbody>
<tr>
<td>Pre stimulus</td>
<td>10 to 20ms</td>
</tr>
<tr>
<td>Post stimulus</td>
<td>50 to 100ms</td>
</tr>
</tbody>
</table>

Filter Settings

| High pass | 1 to 30Hz |
| Low pass  | 250 to 1500Hz |
| Notch     | None       |

Amplification 5000

Sweeps 75 to 250

After VEMP testing each subject was evaluated for patient comfort during VEMP testing in all modes of SCM muscle excitation using 3 point rating scale. The subjects were asked to rate their level of comfort in all positions based on rating scale.
3 point rating scale was used to measure comfort
   1 – Intolerable
   2 – Tolerable
   3 – Comfort

Results and Discussion

The present study aimed at investigating the effect of different body positions on the following parameters of VEMP assessment. Latency of p13, Latency of n13, Amplitude p13-n23, Wave Morphology, Response rate, and Comfort of the patient. The data was subjected to statistical analysis using SPSS software (version.15.0) to test the hypothesis of the study namely: there is no effect of mode of excitation of SCM on VEMP response. The statistical analysis included descriptive statistics, repeated measure analysis of variance and independent samples t-test.

1) The effect of body position on Latency of P13

Repeated measure analysis of variance was carried out to investigate the relationship of body position on latency of p13 peak in VEMP response. Table-3 shows the descriptive statistics of latency of p13 of males and females for the three body positions namely

Table 3. Mean & S.D of latencies of p13 (in msec) in male and female subjects

<table>
<thead>
<tr>
<th>Body position</th>
<th>Males</th>
<th>Females</th>
</tr>
</thead>
<tbody>
<tr>
<td></td>
<td>Mean</td>
<td>S.D</td>
</tr>
<tr>
<td>A</td>
<td>14.1273</td>
<td>2.17566</td>
</tr>
<tr>
<td>B</td>
<td>13.6400</td>
<td>2.32750</td>
</tr>
<tr>
<td>C</td>
<td>13.9053</td>
<td>2.44892</td>
</tr>
</tbody>
</table>

Results revealed that there is no significant difference in p13 latency between any of the three body positions used in VEMP testing for both male[F (2, 30) = 1.295, p > 0.05] as well as female [F (2, 32) = 1.462, p > 0.05] subjects at 0.05 level of significance.

The mean and S.D values of p13 latency of VEMP response in present study are almost in agreement with the studies on VEMP by various authors such as Cheng & Murofushi (2001), Akin et al. (2003), Kaushal (2006), and Huei-Jun et al. (2007). The present study indicates that there is no significant difference in latency of p13 peak in VEMP response between the three body positions used. Thus the null hypothesis that there is no effect of body position on VEMP measurement with respect to p13 latency is accepted.
2) The effect of body position on Latency of n23

Repeated measure analysis of variance was carried out to investigate the effect of body position on latency of n23 in VEMP response. Table-4 shows the descriptive statistics of latency of n23 of males and females across the three body positions (A, B & C).

Table 4. Mean & S.D of latencies of n23 (in msec) in male and female subjects

<table>
<thead>
<tr>
<th>Body position</th>
<th>Males</th>
<th>Females</th>
</tr>
</thead>
<tbody>
<tr>
<td></td>
<td>Mean</td>
<td>S.D</td>
</tr>
<tr>
<td>A</td>
<td>21.7255</td>
<td>2.39769</td>
</tr>
<tr>
<td>B</td>
<td>21.4310</td>
<td>2.49932</td>
</tr>
<tr>
<td>C</td>
<td>19.7368</td>
<td>2.35613</td>
</tr>
</tbody>
</table>

Results revealed that there is significant difference in latency of n23 between position A and position C in males [F (2, 30) = 4.415, p < 0.05] and between position B and position C in female [F (2, 32) = 4.881, p < 0.05] subjects at 0.05 level of significance. In males C position yielded shorter latency of n23 than A position and in females B position yielded shorter latency of n23 than C position. Thus the null hypothesis that there is no effect of body position on VEMP measurement with respect to n23 latency is rejected.

Literature on response consistency of VEMP was reviewed by Ferber et al. (1999) based on the studies done by Cody and Bickford (1969), Townsend and Cody (1971), Colebatch et al. (1994) and Robertston and Ireland (1995). This review suggests that response consistency was not 100% in all the subjects for both the waves of VEMP. However, consistency is more for first wave p13 and less for second wave n23 of VEMP response. In the present study, the results suggest that there is significant difference between the latencies of n23 evoked by three different body positions. And this difference can be attributed to the possible poor response consistency of n23 as reported in the literature.

3) The effect of body position on Amplitude of p13-n23

Repeated measure analysis of variance was done to investigate the effect of body position on p13-n23 (peak to trough) amplitude. Table-5 shows the descriptive statistics of amplitude of p13-n23 of males and females across the three body positions (A, B & C)

Table 5. Mean & S.D of amplitudes of p13-n23 (in micro volts) in male and female subjects

<table>
<thead>
<tr>
<th>Body position</th>
<th>Males</th>
<th>Females</th>
</tr>
</thead>
<tbody>
<tr>
<td></td>
<td>Mean</td>
<td>S.D</td>
</tr>
<tr>
<td>A</td>
<td>48.3427</td>
<td>19.28879</td>
</tr>
<tr>
<td>B</td>
<td>46.4415</td>
<td>22.97675</td>
</tr>
<tr>
<td>C</td>
<td>50.8195</td>
<td>24.71197</td>
</tr>
</tbody>
</table>
Results revealed that there is no significant difference in amplitude of VEMP between any of the three body positions in both males \([F (2, 30) = 0.254, p > 0.05]\) as well as females \([F (2, 32) = 1.088, p > 0.05]\) at 0.05 level of significance. Hence, the null hypothesis that there is no effect of body position on VEMP measurement with respect to p13-n23 amplitude is accepted.

The mean amplitude values of VEMP in the present study are not consistent with the amplitude values reported in other studies. The mean amplitudes of p13-n23 as reported by Kaushal (2006) were less in comparison to the amplitudes recorded in the present study even though the level of stimulus was 10dB lower in the present study than the level of the stimulus in Kaushal’s (2006) study. The reason for this could be that the EMG level was controlled to be between 30-50 micro volts in the present study, where as it was not controlled in the Kaushal’s study. It is possible that the EMG level greater than 50 micro volts would have raised the mean amplitude value of p13-n23. This further indicates that the effect of EMG level is more than that of the intensity of stimulus (short tone burst) on amplitude of VEMP response.

Huei-Jun et al. (2007) found mean amplitudes of VEMP to be 198.53 micro volts with S.D of 64.64. Mean amplitudes of present study are around 50 micro volts with S.D of around 25. Since the level of stimulus used was same in both the studies, the difference in amplitudes can be attributed to two reasons: (a) The S.D of amplitudes of Huei-Jun et al. study was higher than the S.D of the present study indicating that the group was heterogeneous thus affecting the mean, (b) The effect of EMG level was >50micro volts in Huei-Jun et al. study and this was lower than 50 micro volts in the present study (maintained between 30-50micro volts).

4) The effect of body position on Response Rate of VEMP

Graph-1 shows the descriptive statistics of response rate of VEMP for the different body positions (A, B & C) for male subjects. X-axis represents the body positions and Y-axis represents the percentage of subjects in whom VEMP could be recorded, i.e. response rate.

<table>
<thead>
<tr>
<th>Hit Rate in %</th>
<th>Present(%)</th>
<th>Absent(%)</th>
</tr>
</thead>
<tbody>
<tr>
<td>A</td>
<td>88</td>
<td>12</td>
</tr>
<tr>
<td>B</td>
<td>80</td>
<td>20</td>
</tr>
<tr>
<td>C</td>
<td>76</td>
<td>24</td>
</tr>
</tbody>
</table>

Graph-1. Effect of body position on response rate of VEMP in Males
Graph-1 indicates that response rate of VEMP was highest (88%) in A position and minimum (76%) response rate was obtained in C position. The following are the details of male subjects in whom VEMP could not be recorded:
- Number of subjects in whom VEMP was absent in all three positions: 02 (08%)
- Number of subjects in whom VEMP was absent in only two positions: 01 (04%) (A and B positions)
- Number of subjects in whom VEMP was absent only in one position: 06 (24%) (8% in B position and 16% in C position).

Graph-2 shows the descriptive statistics of response rate of VEMP for the different body positions (A, B & C) for female subjects. X-axis represents the body positions and Y-axis represents the percentage of subjects in whom VEMP could be recorded, i.e. response rate.

Graph-2. Effect of body position on response rate of VEMP in Females

Graph-2 indicates that response rate of VEMP was highest (84%) in A position and minimum (72%) response rate was obtained in C position. The following are the details of female subjects in whom VEMP could not be recorded:
- Number of subjects in whom VEMP was absent in all three positions: 03 (12%)
- Number of subjects in whom VEMP was absent in only two positions: 02 (08%) (4% in A, C and the other 4% in B, C)
- Number of subjects in whom VEMP was absent only in one position: 03 (12%) (4% in B and 8% in C positions).

According to the present study there is not much difference in the Response rate of VEMP between the three positions and overall response rate is consistent with the response rate observed by Cody & Bickford (1969) and Townsend & Cody (1971). The number of subjects in whom VEMP could not be recorded was more for position C in both male and female subjects.
5) The effect of body position on Wave morphology of VEMP

Wave morphology of VEMP wave forms was rated using a 3 point rating scale (good, average and poor). Graph-3 & 4 shows the descriptive statistics of the rating on wave morphology of VEMP response in the different body positions (A, B & C) for male and female subjects. X-axis represents the body positions and Y-axis represents the number of subjects.

Graph-3. Effect of body position on wave morphology of VEMP in Males

Graph-4. Effect of body position on wave morphology of VEMP in Females

Graphs 3 & 4 show that position ‘A’ could evoke good wave morphology of VEMP in 48% of subjects compared to B (44%) & C (44%) positions in males. In female subjects also ‘A’ position has evoked good wave morphology of VEMP in 40% of subjects compared to B (36%) & C (32%) positions. But it should be noted that there was not much difference across 3 positions in number of subjects that had good wave morphology.
The present study indicates that there is difference in wave morphology of VEMP response evoked by different body positions and position ‘A’ evokes better wave morphology in males as well as in females.

6) Relationship between body position and Subjects rating for comfort

All the subjects were asked to rate the body positions used in VEMP testing for the comfort level experienced on a 3 point rating scale (1-Intolerable, 2-Tolerable, 3-Comfortable).

Graph-5 shows the descriptive statistics of subjective rating for comfort across different body positions (A, B & C) for male subjects. X-axis represents the body positions and Y-axis represents the number of subjects.

Graph-5. Relationship between body position and comfort in males

From the graph-5 with respect to male subjects it can be seen that:
- in position ‘A’ most of the subjects (68%) rated as tolerable, some (20%) rated as comfortable and least number of subjects (12%) rated it as intolerable
- in position ‘B’ most of the subjects (48%) rated as comfortable, some (36%) rated as tolerable and least number of subjects (16%) rated it as intolerable and
- in position ‘C’ most of the subjects (44%) rated as intolerable, some (32%) rated as tolerable and least number of subjects (24%) rated it as comfortable.

Graph-6 shows the descriptive statistics of subjective rating for comfort across different body positions (A, B & C) for female subjects. X-axis represents the body positions and Y-axis represents the number of subjects.
Graph-6. Relationship between body position and comfort in females

From the graph-6 with respect to female subjects it can be seen that:

- in position ‘A’ most of the subjects (44%) rated as comfortable, some (36%) rated as tolerable and least number of subjects (20%) rated it as intolerable
- in position ‘B’ most of the subjects (52%) rated as tolerable, some (28%) rated as comfortable and least number of subjects (20%) rated it as intolerable and
- in position ‘C’ most of the subjects (56%) rated as comfortable, some (28%) rated as intolerable and least number of subjects (20%) rated it as tolerable

Since this is the first study to investigate the comfortness of subjects for the different body positions used to evoke VEMP response, there is no literature to compare the results of the present study with that of the others. And the results of the present study indicate that not all positions are equally comfortable to all the subjects and the position preferred by males is different from the position preferred by females. Males have preferred ‘B’ position i.e. subject’s body rotated to one side for measurement of VEMP on the opposite side while lying in supine position and females have preferred ‘C’ position i.e. instructing the subject to press the Forehead against a soft surface to be comfortable for VEMP testing.

Independent samples t-test was administered to the data to investigate whether there is any significant difference between males and females for p13, n23 latencies and amplitudes of VEMP across the three positions (A, B & C). Table-6 shows the descriptive statistics of independent t-test. It shows t values for males and females with respect to p13, n23 latencies and amplitudes across the three positions.

The present study indicates that though the male and female subjects choose different body position for VEMP testing in terms of the comfort, there is no difference between them in terms of the response parameters of VEMP (p13, n23 latencies and amplitude) in any of the body position.
Summary and Conclusions

The present study was taken up to investigate the relationship between VEMP measures (latency of p13 & n23 and p13-n23 amplitude), response rate, wave morphology and subject comfort when tested in different body positions used to excite sternocleidomastoid muscle during testing. Further, it was also to see whether there is any relationship between different body positions performed by client in terms of comfort and the nature of VEMP response.

A. Subject’s body rotated to one side for measurement of VEMP on the opposite side in sitting position.
B. Subject’s body rotated to one side for measurement of VEMP on the opposite side while lying in supine position.
C. Instructing the subject to press the forehead against a soft surface.

From the results obtained the following conclusions are tenable:
1. Irrespective of gender VEMP measures such as latencies of P13, N23 and amplitude of p13-n23 are not affected by body position that is used to excite the sternocleidomastoid muscle during VEMP testing.
2. Response rates, wave morphology of VEMP are also not affected by the body position used to excite the sternocleidomastoid muscle during VEMP testing irrespective of gender.
3. Regarding the comfort of the patient during VEMP testing, position ‘B’ that is “Subject’s body rotated to one side for measurement of VEMP on the opposite side while lying in supine position” was the most preferred position while testing male subjects. And position ‘C’ that is “subject pressing the forehead against a soft surface” was reported to be more comfortable position for females.

Limitation of the study
- Very small population was taken into the study

Implication of the study
- The study reveals that clinician while testing VEMP need not prefer any body position for better VEMP measures; one may rather consider the subject comfort in choosing the prescribed position.

Further Scope
- VEMP testing may be carried out on different age groups to explore the issue of subject comfort.
- VEMP testing may be carried out on disorder population to explore the issue of subject comfort.
- The same kind of study can be carried out with some other body positions that can excite Sternocleidomastoid muscle.
VEMP testing may be carried out to study the effect of body positions for binaural VEMP.

References


Effect of Various Modes of Excitation Of SCM on VEMP

Diallo, B.K. (2006). The neurotologic evaluation of vestibular schwannomas. Results of audiological and vestibular testing in 100 consecutive cases. Rev Laryngol Otol Rhinol (Bord), 127(4), 203-9


Early Speech Perception Test Development for Malayalam Speaking Children with Hearing Impairment

Jijo, P. M. & Asha Yathiraj*

Abstract

The ‘Early speech perception test for Malayalam speaking children with hearing impairment’ was developed have two versions. The low verbal version was developed for children between the age of 2 to 3 years and the standard version for children aged 3; 1 to 5 years. The developed material was evaluated on a group of twenty Malayalam speaking children with hearing impairment, ten from each age group. The performances of the two age groups on the developed test are discussed.

Introduction

Early onset of hearing loss can impose substantial delays in communication and psychosocial development unless immediate and appropriate intervention is undertaken. Much of the impact of the sensorineural hearing loss has been noted to depend on the extent to which it affects speech perception (Boothroyd, 1988). It has also been found that those with a greater problem in speech perception are considered to have a greater communication problem than those with fewer problems in speech perception (Boothroyd, 1984).

The primary goal of management is to improve speech perception by using appropriate sensory devices and management strategies. Hence, it is very essential to assess the speech perception capabilities of a child, for the effective selection and planning of management strategies. It has been reported that pure-tones have poor predictive power of the speech perception abilities for children whose average threshold for 500, 1000, and 2000 Hz was in the range of 85 to 100 dB. Hence, it was recommended that audiologist should consider a child’s word recognition ability as well as his pure-tone threshold in making further management options (Erber, 1974).

Efforts to develop speech materials suitable for the paediatric speech audiometry dates back to at least the 1940’s. Haskins (1949) developed word material for speech audiometry in children, with limited number of test items representatives of the vocabulary of kindergarten children. Watson (1957) used the same principle of test construction to generate word and sentence for paediatric speech audiometric tests. Paediatric speech intelligibility testing was advanced by Siegenthaler (1975) by modifying the stimuli and response paradigms to confirm to the children’s interests and abilities.

Later, many tests for the evaluation of speech intelligibility in children were developed (Ross and Lerman, 1970; Erber, 1974; Elliot and Katz, 1980; Moog and Geers, *Professor of Audiology, AIISH, Mysore 570 006

It is ideal to have speech identification tests in all languages as an individual’s perception of speech is influenced by his/her first language or mother tongue (Singh, 1966; Singh and Black, 1966). It is essential that speech identification tests be available even for children with limited vocabulary. Such a speech perception test would be a tool used to determine the further line of rehabilitation for children with hearing impairment. The test score could help choose appropriate devices to be worn by the child. It can also be used to monitor the progress of children with hearing impairment who are provided auditory listening training.

Hence, the study aimed to develop a speech perception test for Malayalam speaking children with hearing impairment, having limited vocabulary. It aimed at developing two versions, one for children aged 2 to 3 years and another for children aged 3;1 to 5 years.

**Method**

The study was carried out in two phases. Phase I involved the development of the test material and checking the familiarity of the material on a group of typically developing children. Phase II dealt with the evaluation of the developed material on a group of children with hearing impairment.

**Phase I**

*Development of test material*

The test material was developed in lines similar to the ‘Early speech perception test’ (Moog & Geers, 1990). Appropriate adaptations were made regarding the stress patterns utilized and number of phonemes across the words, as present in Malayalam. The words for the test were selected from age appropriate books and caregivers of children aged 2 to 3 years and 3;1 to 5 years.

*Participants for phase I of the study*

Thiry normal hearing children were selected to check the familiarity of the test items. Fifteen of them were aged 2 to 3 years and fifteen were aged 3;1 to 5 years. Equal number of males and females were taken in both the groups. They were exposed to Malayalam from early childhood and spoke the language. None of them had any ear infection, speech and language impairment or any developmental delay. It was ensured that they did not have any illness at the time of testing.
Procedure to check familiarity

To ascertain that the word lists that were prepared was familiar to typically developing children in both the age groups (2 to 3 years & 3;1 to 5 years), they were evaluated. The testing was done in a distraction free, quiet room. Each child, who was seated facing the examiner, was tested one at a time.

Pictures representing the words were shown and each child was asked to name the item presented. A word was considered to be familiar only if 90% of the children named it correctly. From the words that were familiar a low level version (version I) and a standard version (version II) of the test were developed. Words familiar to children aged 2 to 3 years were used for version I and words familiar to children aged 3;1 to 5 years were used for version II. It was ensured that each test and subtest contained low, mid and high frequency speech stimuli. Pictures representing all the words in both of the versions were also developed. The test developed was titled ‘Early Speech Perception Test in Malayalam’. The details of version I and version II are described below.

Version I (low verbal version): This version, developed for children between the ages of 2 to 3 years, had the following two tests with the second test having two subtests:
- Syllable categorization test having 2 test items
- Word identification test
  - Bisyllabic word identification subtest having 4 test items
  - Trisyllabic word identification subtest having 4 test items.

Details of the test items are given in appendix I

Version II (standard version): Version II, which was developed for older children, had two tests. The second test had two subtests. The two tests and subtests were as follows:
- Syllable categorization test having 12 test items
- Word identification test
  - Bisyllabic word identification subtest having 12 test items
  - Vowel identification subtest having 10 test items

The syllable categorization test contained four bisyllabic words, four trisyllabic words and four polysyllabic words. Further, the word identification test had two subtests. The bisyllabic word identification subtest had twelve words, represented the phonemes of Malayalam that are used by children in the target age. The vowel identification subtest had words with the vowel varying (Appendix II).
Phase II

Evaluation of the performance of children with hearing impairment, using the constructed material, was carried out in phase II. Each child was tested independently. The details of the instrumentation, environment and procedure are given below.

Instrumentation

A clinical audiometer (Orbiter 922) with option for speech audiometry was used. The output of the audiometer was routed to a loud speaker placed 1 meter away from where the child was seated, at $0^0$ Azimuth.

Test environment

The testing was done either in a two-roomed sound treated set-up or in a quiet distraction-free room. The ambient noise levels in the sound-treated room were within the permissible limits prescribed by ANSI-S3.1-1991.

Participants for phase II of the study

Twenty children with hearing impairment, in the age range of 2 to 5 years were selected. They were divided into two groups, one group in the age range of 2 to 3 years and other in the age range of 3 to 5 years. It was ensured that the children had been exposed to Malayalam from early childhood and spoke the language. In addition, they had severe to profound hearing loss, aided audiogram within the speech spectrum at least up to 2 kHz, awareness of normal conversation with their prescribed hearing aids, no additional handicap like mental retardation or visual impairment and no illness at the time of testing.

Procedure for phase II of the study

While most the children were tested in a sound tested room, a few of them had to be tested in a non-sound treated room. The latter had to be done since some of the children refused to enter the sound treated room. All the children wore on their prescribed binaural hearing aids, which had been earlier checked to be functioning well.

Testing done in a sound treated room:

All 10 children from the older age group and 3 children from the younger age group were tested in the two room set-up. They were seated at a distance of one meter from the loud speakers which was placed at an angle of $0^0$ Azimuth. The pictures representing the test item were placed before them on a table. The words were presented one by one at a presentation level of 50 dB HL. The level of the live speech was monitored using a VU meter.
**Testing done in a quiet room situation:**

Seven children in the younger age group were tested in a quiet, distraction free room, as they did not cooperate to be evaluated in the sound treated room. They were seated at a distance of 2 feet from the examiner and the test material was placed in front of each child on a table. The stimuli were presented one-by-one by the tester at a normal conversational level (60 dB SPL).

For testing in both situations, initially the caregiver was asked whether the child was familiar with the test items. If a child was not, he/she was given training using the test items until he/she could readily carryout the activity through an audio-visual mode of presentation.

The test items were presented once with audio-visual cues and twice with only auditory cues. The items were randomized during each presentation. It was ensured that the children were attentive prior to the presentation of each stimulus. The children were required to point out to the appropriate picturised item.

While administering the low verbal version, the syllable categorization test was carried out first, followed by the word identification subtest. Likewise, for the standard version also the syllable categorization was evaluated first, followed by the word identification tests. Bisyllables were tested initially followed by the vowel identification. The entire testing was carried out in 2 to 3 sessions. The duration of each session was 15 to 20 minutes depending on the attention span of a child.

**Scoring**

Responses were recorded on a scoring sheet for each child (Appendix III & Appendix IV). For the syllable categorization test, a score of ‘1’ was given when the child identified any picture from a given category, and a score of ‘0’ if it was identified from a different category. Similarly for the identification test a correct response was given a score of ‘1’ and a wrong response a score of ‘0’. Descriptive statistics, paired sample ‘t’ test and ANOVA were used to carry out the analyses.

**Analyses**

The data obtained from children on the ‘Early speech perception test in Malayalam’, using the developed low verbal version and standard version, were analyzed. The Statistical Package for Social Sciences version 10 for Windows was used to carry out the analyses on children aged 2 to 3 years and 3;1 to 5 years respectively. A comparison was also made between the performances of children on the two versions of the test. Descriptive statistics, paired sample ‘t’ test and ANOVA were used to carry out the analyses.
Results

Results of the Low Verbal Version Test

Descriptive statistics were carried out on the responses of the younger age group on the low verbal version of the developed test. From Table 1 it is evident that mean score of the pattern perception test was higher than that of the word identification tests. Also, the standard deviation (SD) was highest for the pattern perception test. This indicates that variability in the scores obtained by the participants on the pattern perception test was greater than the bisyllabic and trisyllabic word identification subtests.

Table 1: Mean score and SD for the ‘Pattern perception test’ and ‘Word identification test’ for the low verbal version

<table>
<thead>
<tr>
<th>Tests scores</th>
<th>Mean percentage scores</th>
<th>SD</th>
</tr>
</thead>
<tbody>
<tr>
<td>Pattern perception test</td>
<td>85.00%</td>
<td>17.48</td>
</tr>
<tr>
<td>Bisyllabic word identification</td>
<td>73.75%</td>
<td>9.22</td>
</tr>
<tr>
<td>Trisyllabic word identification</td>
<td>77.50%</td>
<td>11.48</td>
</tr>
<tr>
<td>Combined word identification</td>
<td>75.62%</td>
<td>9.05</td>
</tr>
</tbody>
</table>

To compare the pattern perception test scores and word identification scores, paired ‘t’ test was performed. The results revealed a significant difference between mean percentage score of the pattern perception tests and word identification test (p < 0.05).

Bisyllabic word identification scores and trisyllabic word identification scores were also compared using paired sample ‘t’ test. The results brought to light that there was no significant difference between these two tests (p > 0.05).

Results of the Standard Version Test

The mean scores of the bisyllabic, trisyllabic and polysyllabic pattern perception test revealed that the polysyllabic pattern perception score was better than the trisyllabic and bisyllabic pattern perception scores. Further, the overall pattern perception test scores were higher than that obtained for both the word identification scores (Table 2). From the Table 2 it can also be noted that the SD was maximum for the bisyllabic word identification test. However, the variability was only marginally more than that obtained for pattern perception test. Though the variability was least for the vowel identification test, it also happened to have the lowest mean score.
Table 2: Mean scores and SD for the ‘Pattern perception test’ and ‘Word identification test’ for the standard version

<table>
<thead>
<tr>
<th>Tests</th>
<th>Mean</th>
<th>SD</th>
</tr>
</thead>
<tbody>
<tr>
<td>Bisyllabic pattern perception</td>
<td>63.75%</td>
<td>9.22</td>
</tr>
<tr>
<td>Trisyllabic pattern perception</td>
<td>73.75%</td>
<td>9.22</td>
</tr>
<tr>
<td>Polysyllabic pattern perception</td>
<td>86.25%</td>
<td>9.22</td>
</tr>
<tr>
<td>Total pattern perception test</td>
<td>74.58%</td>
<td>4.14</td>
</tr>
<tr>
<td>Bisyllabic word identification test</td>
<td>50.00%</td>
<td>9.82</td>
</tr>
<tr>
<td>Vowel identification test</td>
<td>44.00%</td>
<td>6.58</td>
</tr>
<tr>
<td>Total word identification test</td>
<td>47.27%</td>
<td>7.85</td>
</tr>
</tbody>
</table>

To determine the significance of difference between the overall pattern perception scores and overall word identification scores, paired ‘t’ test was performed. A significant difference (p < 0.01) between the two was observed with the latter test obtaining significantly lower values.

A comparison of the bisyllabic, trisyllabic and polysyllabic words within the pattern perception tests was done using a repeated measure ANOVA, in which syllable duration was taken as the independent variable and the identification scores as the dependent variable. The results showed a significant effect of syllable duration on pattern perception scores [F (2, 18) = 12.48; p < 0.001]. Boneferroni pairwise test revealed that there was a significant difference (p < 0.05) present only between the bisyllabic and polysyllabic pattern perception test.

The bisyllabic word identification scores and vowel identification scores were compared using the paired sample ‘t’ test. It was observed that the two were significantly different [t (9) = 2.90, p < 0.01]. Significantly higher scores were obtained by the children with hearing impairment aged 3 to 5 years on the bisyllabic word identification test.

**Comparison between the low verbal version and standard version test scores**

A comparison of the performance of the younger group with that of the older group was made for the pattern perception scores and the word identification scores. The responses of the two age groups are shown in Table 3 for the pattern perception and word identification scores respectively.

Table 3: Mean scores and SD for pattern perception and word identification scores

<table>
<thead>
<tr>
<th>Test</th>
<th>Mean</th>
<th>SD</th>
</tr>
</thead>
<tbody>
<tr>
<td></td>
<td>Low verbal</td>
<td>Standard</td>
</tr>
<tr>
<td>Pattern perception</td>
<td>85.00%</td>
<td>74.58%</td>
</tr>
<tr>
<td>Identification</td>
<td>75.62%</td>
<td>47.27%</td>
</tr>
</tbody>
</table>
The overall pattern perception scores were compared between the two age groups using independent sample ‘t’ test. The results revealed that there was no significant difference (p > 0.05) between mean score of the pattern perception tests between the two age groups. In contrast, for the word identification tests, there was a significant difference (p < 0.001) between the two groups, with the younger group getting higher scores.

**Discussion**

The results of the study are discussed in relation to the findings obtained for the low verbal version meant for children aged 2 to 3 years and the standard version meant for children aged 3;1 to 5 years. In addition, the comparisons between the two versions of the tests are also discussed.

**Low verbal version**

The results of the present study revealed that the younger age group found the pattern perception task significantly easier than the word identification task. The former task mainly required participants to identify suprasegmental information related to the length of the test stimuli, while the latter required them to identify segmental information also. It has been reported by many authors that suprasegmental features are better perceived than segmental features in individuals with hearing loss (Smith, 1975; Bilger & Wang, 1976; Risberg, Agelfor, 1978; Hack & Erber, 1982). The above results are in accordance with the previous studies by Begum (2000) and Tamilmani (2002). They too observed that pattern perception scores were significantly better than the word identification scores.

Zeiser and Erber (1977) reported that children with profound hearing impairment probably receive only time and intensity information (that is, vibratory patterns) through their hearing aids. Hence, one of the acoustic features of speech that seems to be available even to those children through the vibratory sense is the number of syllables in a word, phrase, or sentence. Though the children in the present study had aided audiograms within the speech spectrum up to 2 kHz, they too probably made better utility of the temporal based cues.

**Standard version**

In the standard version of the test in the current study, it was observed that the pattern perception test scores were significantly better than the word identification test scores. This was in accordance with several studies, which report that in subjects with sensorineural hearing loss, suprasegmental features are better perceived than segmental features (Smith, 1975; Bilger and Wang, 1976; Risberg, Agelfor, 1978; Hack and Erber, 1982). Better pattern perception over word identification was also reported by Moog & Geers (1990), Begum (2000) and Tamilmani (2002).
In addition, it was also found that the mean percentage score for the polysyllabic pattern perception test was significantly better than the mean trisyllabic pattern perception test. Further, the trisyllabic pattern perception test score was significantly better than the mean bisyllabic pattern perception test score. Thus, it is evident that stimuli that have a longer duration are better perceived by children having hearing impairment.

It was found in the present study that the bisyllabic word identification score was significantly better than vowel identification score. Similar findings were reported in the previous studies (Moog & Geers, 1990; Begum, 2000). Poor vowel recognition in individuals with sensorineural hearing loss was also reported by Turner and Henn (1989). They reported that poor frequency resolution commonly noted in sensorineural hearing loss can be a significant factor in the poor recognition of vowels in these subjects.

The reduced scores on vowel identification task could also be attributed to poor vowel formant discrimination ability in individual with hearing impairment. Liu and Kewley Port (2004) reported that the thresholds of vowel formant discrimination for syllables and sentences were significantly elevated for individual with hearing impairment compared to thresholds for young normal hearing listeners. Formant discrimination was found to be elevated in the F2 region by almost 100%, where the greater hearing loss occurred, rather than in the F1 region.

Liu and Kewley-Port (2007) also reported that high levels of presentation for speech signals degraded thresholds for formant discrimination for listeners with hearing impairment rather than improved performance when audibility was assured. Several factors were considered to account for the level effect on formant discrimination, including audibility, frequency selectivity, and upward spread of masking on F2. All these factors may have interacted with each other to affect formant discrimination. In the present study, decreased frequency selectivity and greater upward spread of masking on F2 at the high signal level may have contributed to the reverse level effect of formant discrimination.

Comparison between low verbal version and standard version

The results of the current study revealed that the older group performed significantly poorer than younger group in the word identification test. However, there was no significant difference between the two groups for the pattern perception test. This shows that both the age groups found the pattern perception test to be equally easy, but with increase in age word identification abilities improved.

In contrast to the present results, Begum (2000) reported that children in the older age group performed significantly better on the pattern perception test. However, she found no significant difference between the two groups on the word identification test scores. Subject variability may have accounted for the difference in findings. The kind of training received by the children in the two studies may have also influenced the findings. Though both studies evaluated children who were enrolled in the same clinical
program, the focus of training has changed over the years. At the time when Begum carried out the study, the main focus of training was through an audio-visual mode. In the last few years the focus has shifted towards a more auditory based training program. The findings of the present study, where the younger children obtained higher word identification scores than the older children, probably reflect their ability to make better use of their auditory skills. The older group probably did not use their auditory skills to the same extent.

This finding is supported by the results of the study by Meyer, Svirsky, Kirk and Miyamoto (1998). They too found that a group of children with profound hearing loss, who had enrolled for an oral communication program, obtained 25% to 40% higher scores on a speech perception test. This was in comparison to a group who had not enrolled in such an oral program, as their thresholds of hearing were higher.

**Conclusions**

From the findings of the present study it is evident that the pattern perception scores were significantly better than the word identification scores. This was observed in the low verbal version and the standard version of the test. However, there was no significant difference between the bisyllabic and trisyllabic word identification test scores in the low verbal version. Unlike the low verbal version, for the standard version the mean percentage scores for the polysyllabic pattern perception scores was significantly better than the mean trisyllabic pattern perception score which was significantly better than the mean bisyllabic pattern perception score. Further, the mean score of the bisyllabic word identification test was significantly better than that of the vowel identification test.

The comparison of the low verbal version and standard version indicated that there was no significant difference between the pattern perception test scores between the two age groups. On the contrary, the mean score of the word identification test was significantly poorer in the older group compared to the younger group.

The findings of the study indicate that the developed test material can be administered effectively on children with hearing impairment in the age range of 2 to 5 years who are exposed to Malayalam for a period of 6 months or 1 year prior to being tested. It is suggested that the low verbal version can be used to evaluate older children who have inadequate speech and or language skills to perform speech tests relevant to their age and also for those with poor attention span. The standard version of this test can be used to for children of 3-5 years age and also those younger children with higher language abilities. If required, the test can be administered after some training to evaluate the performance of the child on speech perception tasks. This would help to eliminate the disadvantage of lack of vocabulary to carry out the test. Hence, it can also be the first speech identification test administered for children with hearing impairment.
References


APPENDIX I
LOW VERBAL VERSION

Pattern Perception Test

പന്തു പന്തു പന്തു പന്തു

Word Identification Test

Bisyllabic Word Identification Subtest

കണ്ണ്

പൂച

പശു

Trisyllabic Word Identification Subtest

കസേര

തവള

APPENDIX II
STANDARD VERSION

Pattern Perception Test

Bisyllabic pattern perception test

കണ്ണ്

മാല

മുട്ട

പാബ്

Trisyllabic pattern perception test

പൂബാറ്റ

കുതിര

മത്ത

Polysyllabic pattern perception test
ESPT in Malayalam for HI children

Bisyllabic word identification subtest

കാളവ

തലമുടി

അലമാര

മുന്തിങ്ങ

മാങ്ങ

കുട

പന്നി

വായ

കയ്

പലല്

ചക്ക

പട്ടി

പൂച്ച

ചെവി

Vowel identification test

പട്ടി

പപട്ടി

പുട്ട്

പുട്ട്

പാററ

ചപാട്ട്

പിന്ന്

പീലി
Dichotic Rhyme Test in Telugu: A Normative Data on Adults

Kishore, T. & Rajalakshmi. K

Abstract

Dichotic speech tests proved to have high sensitivity in assessing binaural integration tasks that often noticed in individuals having (Central) Auditory Processing Disorder [(C) APD]. Research focusing on one of the dichotic tests i.e. Dichotic Rhyme test has been relatively scanty. The present study aimed to develop one such test and collect normative data for Telugu speaking individuals. Stimulus was developed using 18 pairs of CVCV rhyming words that differ only in initial consonant. These stimuli were made similar in total duration and imposed on to stereo tracks and aligned such that these are played dichotically with no onset disparities. Normative data was collected from 60 young adult native speakers of Telugu, including equal number of male and female subjects. Analysis of the results revealed a significant right ear advantage in male subjects than in female subjects. Double correct scores were also to be greater in male subjects than in female subjects. The results correlated with gender differences that exist for language laterality.

Key words: (C) APD, Dichotic Rhyme Test, Gender Differences, Right Ear Advantage.

Introduction

Dichotic rhyme test (DRT) was introduced by Wexler and Halwes (1983) and modified by Frank E. Musiek (1989). This test uses well aligned and is composed of simple common words. The stimuli are aligned such that, although presented with two words, patients generally report only one, with slightly more than 50% of all words recognized being those presented to right ear (Wexler and Halwes., 1983; Musiek et al., 1989). This unique pattern of performance was presumed to be the result of some type of dichotic “fusion” of the signals, which occur low within the central auditory nervous system. The rationale behind this test has come from series of experiments carried by Repp (1976). Fusion in the dichotic listening condition takes place when words with similar spectral shape (waveform envelop) are presented to the listener (Repp, 1976). The waveform envelop for words is generally determined by the low frequency energy (Perrot and Berry, 1969), which is essentially its fundamental frequency (Repp, 1976, 1977a).

Therefore if two words presented dichotically, which have similar spectral envelopes and are temporally aligned, they will fuse and will be heard as one word (Repp, 1977a). The words in DRT for the most part, are words that are perfectly or partially fused. Due to the fusion this test also called as Fused Dichotic Words Test (FDWT). Musiek, Kurdzielschwan, Kibbe, Gollegly, Baran, and Rintelmann (1989) reported normative values of 30% - 73% for right ear and 27% - 60% for left ear in a group of 115 normal hearing subjects. Bellis (2003) normative data indicated no significant effect of age or ear on the Dichotic Rhyme test. Normative values (2 standard deviations above and below the mean) were 32% - 60% per ear.

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Bode, S.D. et al., (2007) examined DRT for measuring the degree of hemispheric specialization for language in individuals who had undergone cerebral hemispherectomy. Results revealed that most syllables or words are reported for the ear contralateral to the remaining hemisphere, while few or none are reported for the ear ipsilateral to the remaining hemisphere. In the presence of competing inputs to the two ears, the stronger contralateral ear-hemisphere connection dominates-suppresses the weaker ipsilateral ear-hemisphere connection.

Musiek et al., (1989) studied the performance of normal hearing individuals and patient undergone commissurectomy on dichotically presented monosyllabic rhyme words. Data was collected from a group of 115 normal hearing individuals and 6 patients undergone commissurectomy for intractable seizures (2 weeks postoperatively). Results reveal that spilt-brain patients yielded marked left ear deficit, as seen on other dichotic speech tests and demonstrated a right-ear enhancement, producing a large inter-ear differences. This right-ear enhancement on the dichotic rhyme task (DRT) may suggest a release from central auditory competition in the left hemisphere. The dichotic rhyme task's normative data results and sensitivity to lack of callosal transmission make it worthy of further clinical and basic research.

There have been several factors that affect the performance of DRT that include both subjective and stimuli parameters. One such major subjective factor for the performance on DRT was gender differences. As right-ear advantage in dichotic listening is a reflection of the left hemisphere's dominance for speech perception and related functions (Studdert-Kennedy and Shankweiler, 1970; Kimura, 1961a, 1967). A much debated question is whether sex differences exist in the functional organization of the brain for language. A long-held hypothesis posits that language functions are more likely to be highly lateralized in males and to be represented in both cerebral hemispheres in females, but attempts to demonstrate this have been inconclusive.

From these findings one can understand that DRT is highly sensitive to the lesions that involve corpus callosam. The primary lesion for the patients with (C) APD facing auditory integration deficits was thought to be involving posterior commissural fibers (Baran et al., 1986). Thus having DRT to assist clinically in the assessment of the patients facing such difficulties. As DRT involves rhyming meaningful words, and the rhyming words across languages differ, developing DRT in different languages will help audiologists to assess patients having different languages spoken and facing problem with auditory integration deficits. Thus the aim of the present study was

A. To develop the Dichotic Rhyme test using commonly spoken words in Telugu and establishing the normative data on newly developed DRT.

B. To evaluate the gender differences on dichotic performance using dichotic rhyme test.
Method

The present study was carried out in two phases. Phase I involved construction of test material for “Dichotic Rhyme test in Telugu”. Phase II involved obtaining normative data for the newly constructed “Dichotic Rhyme test in Telugu”.

Phase I: Construction of the test material

Forty pairs of bi-syllabic words, in which each word had syllabic structure as CVCV, were selected from “Brown’s Telugu- English Dictionary” as well as from text books (below V Grade). Each word of a rhyming pair had started with one of the six consonants (/p/, /t̪/, /k/, /b/, /d̪/ and /g/). In a pair of rhyming words, the two words (here onwards referred as members of a pair) differ only in the initial consonant. Furthermore, the difference in the initial consonant was either in terms of place of articulation or in terms of voicing. Thus, it leads to have nine possible combinations with the six consonants. Of these 40 rhyming words, 25 were selected based on subjective ranking given by the native speakers of Telugu. From these 25 rhyming words, 18 were selected to include two rhyming pairs for each consonantal difference.

These thirty six words (i.e.,), eighteen rhyming pairs were given to an adult female (native speaker of Telugu), and was asked to say out these words. This voice sample was picked up with an electret condenser type Omni-directional microphone. These speech samples were recorded as mono sound using “PRAAT” software with a sampling rate of 22050Hz and 16-bit amplitude rate. Using the same software, the final CV portion of one member of each pair was then replaced with final non-distinctive portion of the other member, making the final portions of the members of each pair identical. This was a preliminary step to reduce the variation in the final portion of members in a rhyming pair.

E.g., the final CV portion in /t̪ala/ (/la/) was replaced with final CV portion in /kala/ (la), to reduce the perceptual difference in the final portion of both words.

After cross-splicing process, the two members in a pair were made identical in stimulus duration by reducing the glottal pulses and/or by reducing the steady state portion in the initial CV portion of longer durational member. By doing so, the duration of members within a pair has been kept the same. But the duration of different rhyming pairs was not maintained, as the duration of different consonants in different words varies.

The selected stimuli were then normalized to 6dB and imposed onto stereo tracks. These were aligned such that when one member of the pair was presented to one ear and at the same time other member played in the other ear. This was achieved by using the software “ADOBE AUDITION 3.0”. In addition a counter balanced design was used to decrease the ear effects, by aligning the stimuli which was reversed between ears. This leads to a total number of 36 stimuli for total 18 pairs of words. 10 seconds silence was
inserted between each stimulus, during which subject wrote their responses. All the stimuli were constructed into 4 tracks. Content of these tracks are shown in Appendix A. These were recorded on to a compact disk with initial calibration tone of 1 KHz of equal intensity in both ears as stereo tracks.

The recorded material was presented to the native speakers of Telugu, for noticing any distortion embedded in the altered signal and to observe stimulus dominance effects. Stimulus dominance is the tendency for one member of a pair to be consistently reported regardless of ear of presentation. It was made sure that all the stimuli selected for the final testing to have acceptance (without any distortion) by 80% of total 20 native speakers. Rhyming pairs with more than 60% of stimulus dominance were identified and modified by altering amplitude of some important acoustic cue (mainly the burst amplitude and voicing amplitude) in the dominant member of the pair, to reduce the effects of stimulus dominance.

**Phase II: Obtaining normative data**

**Subjects:**

The subjects for the study were 50 normal young adults (25 males and 25 females) in the age range of 17 to 25 years. All the subjects had normal hearing sensitivity in both ears with no previous otological problem and having greater than 80% of SIS scores in each ear. All subjects were right hand users (Established through verbal report and tested by comparing the writing ability of the two hands). Some of the subjects taken for normative data also participated in the stimulus dominance experiment. For these individuals, to avoid practice effect minimum of 15 days time period was given between two experimental phases.

**Instrumentation:**

A two channel diagnostic audiometer (Madsen OB 922), which was calibrated in accordance with ISO 389 (As mentioned in Madsen electronics Instrumental Manuel), was used for preliminary testing (Air conduction, Bone conduction and Speech audiometric measures) and also to present the test material. Stimuli were played from a computer that constituted the sound drives namely, “legacy audio devices”. All stimuli were presented through TDH-39 earphones mounted in MX-41/AR cushions.

**Test environment:**

The testing was carried out in a well lit air-conditioned sound treated double room and ambient noise levels within permissible limits according to ANSI S3.1 -1999 (As cited by Tom Frank, 2000).
**Procedure:**

The test stimuli were presented at a level of 60dB HL, through audiometer routed to headphones. The subjects initially had to match the loudness of the calibration tone between ears. Then the test stimuli were presented dichotically, with no lag between ears. The subjects were instructed to write down the words they heard, and also not to guess any word of the pair (if only one word was heard). Subjects were encouraged to write down both words in a pair.

**Scoring:**

Responses were scored in terms of single correct and double correct scores. A single correct score was given when the subject writes only one word presented to any one ear correctly. A double correct score was given when the subject reported the words presented to both ears correctly. From these scores, the total number of stimuli repeated from one ear (right or left) was calculated and named as ear correct score. These include the total number of responses that were correct from one ear (right or left) out of 36 (total number of stimuli) and were used for further analysis.

**Reliability measure:**

Intra subject reliability of the test results was verified, by testing 10 individuals (Constitute 20% of total population) including 5 males and 5 females, repeatedly. Further results of the reliability test measure are discussed under results and discussion chapter.

**Analysis:**

The raw data was subjected to statistical analysis from which descriptive statistics such as mean, standard deviation and range were calculated. Ear correct scores were examined for gender differences. A 2 X 2 repeated measure analysis of variance was performed with gender (2 levels: Male, Female) as between-group factor and ear (2 levels: Right and Left) correct scores as the within-group factors. As significant Ear X Gender interactions were revealed in the analysis of variance, indicating differential effects of gender on the magnitude of ear correct scores, separate planned t-tests (Paired and Independent samples) were carried out for right- and left-ear correct scores on within and between genders to explore these interactions.

**Results**

To have normative values, data collected on 25 male and 25 female subjects in the age of 17 to 25 years was subjected to statistical analysis using the software program SPSS version 10.0. Analysis was carried out to reveal information on,

I. Comparison of ear correct scores within gender
II. Comparison of ear correct score across genders
III. Double correct scores across gender
IV. Reliability measures
I. Comparison of ear correct scores within gender:

Ear correct scores were used for statistical analysis. Left ear and right ear correct scores were analyzed for differences in both males and females. The average values of raw data (ear correct score) for both males and females are depicted in the following graph:

![Graph showing comparison of ear correct scores across gender.]

Figure 1: Comparison of ear correct scores across gender.

From the graph it can be observed that, there is large difference between right and left ear correct score for males, but a less difference for the same in female subjects. It can also be observed that on right ear scores there is greater difference obtained for males and females, but for the left ear scores were similar. To explore the statistical difference on right and left ear correct scores within gender, paired samples t-test was performed. Results of paired t-test are shown in the table 3.

Table 1: Descriptive statistics for each ear correct scores & Right ear advantage

<table>
<thead>
<tr>
<th>Gender</th>
<th>Ear</th>
<th>Mean</th>
<th>SD</th>
<th>t-value</th>
<th>Significance level (2-tailed)</th>
</tr>
</thead>
<tbody>
<tr>
<td>Male</td>
<td>Right</td>
<td>24.4</td>
<td>6.18</td>
<td>3.07</td>
<td>0.005 (P* &lt; 0.01)</td>
</tr>
<tr>
<td></td>
<td>Left</td>
<td>19.52</td>
<td>5.61</td>
<td></td>
<td></td>
</tr>
<tr>
<td>Female</td>
<td>Right</td>
<td>19.96</td>
<td>3.66</td>
<td>0.33</td>
<td>0.745 (P &gt; 0.05)</td>
</tr>
<tr>
<td></td>
<td>Left</td>
<td>19.64</td>
<td>4.34</td>
<td></td>
<td></td>
</tr>
</tbody>
</table>

The maximum correct score that could be obtained for each ear is 36.

The above table indicates there exists a significant difference (P* < 0.01) between ears for males and no significance difference (P > 0.05) between ears for females. From these results, one can understand that the right ear advantage is more in male subjects than female subjects. Overall scores from right ear are higher than from left ear. This indicates that the stimulus processed through right ear has been superior to left ear. This is called as Right Ear Advantage (REA).
II. Comparison of ear correct score across genders:

Using independent samples t-test, each ear correct scores were analyzed for differences among gender. Results from independent samples t-test is shown in the following table 2.

Table 2: Comparison of ear correct score across genders.

<table>
<thead>
<tr>
<th>Ear</th>
<th>Gender</th>
<th>Mean</th>
<th>SD</th>
<th>t-value</th>
<th>Significance level (2-tailed)</th>
</tr>
</thead>
<tbody>
<tr>
<td>Right</td>
<td>Male</td>
<td>24.4</td>
<td>6.18</td>
<td>3.09</td>
<td>0.003 (P* &lt; 0.01)</td>
</tr>
<tr>
<td></td>
<td>Female</td>
<td>19.96</td>
<td>3.66</td>
<td></td>
<td></td>
</tr>
<tr>
<td>Left</td>
<td>Male</td>
<td>19.52</td>
<td>5.61</td>
<td>0.85</td>
<td>0.933 (P &gt; 0.05)</td>
</tr>
<tr>
<td></td>
<td>Female</td>
<td>19.64</td>
<td>4.34</td>
<td></td>
<td></td>
</tr>
</tbody>
</table>

The maximum correct score that could be obtained for each ear is 36.

From the table, it can be understood that, right ear scores are significantly different (P* < 0.01) among males and females( higher scores for males). There is no significant difference (P > 0.05) on left ear correct scores among males and females. This reveals that the stimulus processed through left ear is same in both male and females subjects, but the stimulus processed through right ear is superior in male subjects than in female subjects.

III. Double correct scores across gender:

When the subject repeats both stimuli presented to both ears, one double correct score was given. These double correct scores obtained in both males and females are depicted graphically as given in the figure below:

Figure 2: Gender differences on double correct scores.

From the graph, one can notice a difference in double correct scores obtained from males and females. But the amount of double correct scores is always less than either ear (Right or Left) correct score, and also constitutes a very less portion to identification score. This reflects the difficulty involved in processing two temporally equated rhyming words simultaneously.
This difficulty could be due to the precise alignment of the two members of a pair. Subjects generally report only one, although presented with two words, with slightly more than 50% of all words recognized being those presented to right ear (Wexler & Halwes, 1983; Musiek et al, 1989). The difference between males and females evaluated using independent samples t-test. The results are displayed in the following table:

Table 3: Comparison of double correct scores across genders.

<table>
<thead>
<tr>
<th>Score</th>
<th>Gender</th>
<th>Mean</th>
<th>SD</th>
<th>t-value</th>
<th>Significance level (2-tailed)</th>
</tr>
</thead>
<tbody>
<tr>
<td>Double correct score</td>
<td>Male</td>
<td>8.88</td>
<td>7.92</td>
<td>2.38</td>
<td>0.021 (P &lt; 0.05)</td>
</tr>
<tr>
<td></td>
<td>Female</td>
<td>4.08</td>
<td>6.25</td>
<td></td>
<td></td>
</tr>
</tbody>
</table>

Maximum double correct that could be obtained is 36

From the table, one can understand that, there is significant difference (P < 0.05) between males and females on double correct scores. But the variability in double correct score was high in both genders.

IV. Reliability measures:

The reliability measures for 20% of the total subjects participated, were analyzed using SPSS 10.0. Mainly alpha value between two measurements (done at different times) was considered for reliability index. The results of reliability measure are shown in the following table 4.

Table 4: Reliability measure results

<table>
<thead>
<tr>
<th>Gender</th>
<th>Alpha values</th>
<th>Right ear correct</th>
<th>Left ear correct</th>
<th>Both ear correct</th>
</tr>
</thead>
<tbody>
<tr>
<td>Males</td>
<td>0.75</td>
<td>0.7087</td>
<td>-11.68</td>
<td></td>
</tr>
<tr>
<td>Females</td>
<td>0.9636</td>
<td>0.9681</td>
<td>0.99824</td>
<td></td>
</tr>
</tbody>
</table>

The above table reveals that all the scores obtained on dichotic rhyme task at two different times, are having alpha value more than 0.7, which indicates a good reliability of the test. But for the double correct scores in male subjects’ revealed poor reliability. This can be due to large variability observed on double correct scores. Thus, it is wise to advice measuring ear correct scores rather than double correct scores in a clinical practice.

Discussion

These normative results obtained from the present study are consistent with the results of studies conducted on western population using dichotic rhyme test by Musiek et al., (1989). Musie, Kurdzielschwan, Kibbe et al., (1989) reported slightly better (more than 50% correct) scores in right ear using dichotic rhyme test. Results reveal normative values of 30% - 73% for right ear and 27% - 60% for left ear in a group of 115 normal
hearing subjects. In Indian population the similar results on REA, were reported by Rajagopal (1996), Ganguly (1996), Puranik. P. (2000), Krishna (2001) and Moumitha (2003), using dichotic Consonant-Vowel test. From the current study, it can be concluded that significant right ear advantage (REA) was present in male subjects, but not so in female subjects. These gender differences can be attributed to functional lateralization of auditory stimuli processing during dichotic rhyme test.

The gender differences observed in the present study are in correlation with the findings of Wexler and Lipman (1988). They reported that the gender differences of right ear advantage using fused dichotic word test of 120 trials. Results revealed that males showed higher right ear advantage on the first 60 trials, relative to female subjects. These results suggest that males respond to the novelty of a new task with relative left hemisphere activation while females respond with relative right hemisphere activation. These results are in correlation with the findings of present study as the number of stimuli used in the present study was 36 and thus leading to better right ear advantage in males.

The similar results were also obtained by Shaywitz et al (1995), on functional magnetic resonance imaging (fMRI) using blood oxygen level- dependent (BOLD) method. The results revealed that, for phonological task (rhyme) men showed lateralized left inferior frontal gyrus, where as women showed more diffuse neural systems that involve both right and left inferior frontal gyrus regions. The similar results on gender difference was also noticed by Ikezawa et al., (2008), using other type of dichotic stimuli. Ikezawa et al (2008), reveal that gender differences were observed on dichotic CV tasks using Mismatch Negativity(MMN). MMNs generated by pure-tone and phonetic stimuli were compared, using EEG amplitude and scalp current density (SCD) measures. The results revealed that, males exhibited left-lateralized activation with phonetic MMNs, whereas females exhibited more bilateral activity.

As right-ear advantage in dichotic listening is a reflection of the left hemisphere's dominance for speech perception and related functions (Studdert-Kennedy and Shankweiler, (1970); Kimura, 1961a, 1967), it could be concluded that males have more lateralized dominance ability for speech perception. These results are in support of findings by Clements et al (2006), where functional magnetic resonance imaging (fMRI) was used to study gender differences during phonological and visuospatial tasks. Results indicate that lateralization differences exist, with males more left lateralized during the phonological task, whereas females showed greater bilateral activity.

Conclusions

In conclusion, the findings of the present study on Indian population are consistent with the findings obtained on western population. The present study also revealed the gender differences on dichotic rhyme test, which is consistent and also proved using electrophysiological measures. The results of this study also provide normative data for adults, that is for men right ear scores ranges 60% to 74% and left ear 47% to 60% and for females right ear scores ranges 51% to 59% and left ear scores 49% to 59% (values
given were 95% confidence interval for mean). Before using for clinical assessment, the test material developed as a part of present study has to be further studied. This is due to the large variation noticed in results between genders. As females did not exhibit significant right ear advantage, the present test may have limited usefulness in testing this group.

Acknowledgements

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References


Aided Acceptable Noise Levels (ANL): A Comparison across Degree of Hearing Loss, Noise Reduction in Hearing Aid and Personality Type

Manuj Agarwal & P. Manjula*

Abstract

Acceptable Noise Level (ANL) is a measure of the willingness to accept the background noise while listening to speech and is defined as the difference between the most comfortable listening level (MCL) for running speech and the maximum background noise level (BNL) that a listener is willing to accept. The present study aimed to evaluate the relationship between the unaided and aided Acceptable Noise Levels, effect of degree of hearing loss, digital hearing aid with and without the noise reduction scheme, effect of presentation level and effect of personality type on Aided Acceptable Noise Levels. Three groups of participants based on the degree of hearing loss were assessed in unaided and two aided conditions. The two aided conditions included fitting of an appropriate digital hearing aid with noise-reduction feature turned-off and turned-on. To assess the effect of presentation level, only participants with the moderate degree were considered. The ANLs were obtained at three presentation levels, at 5 dB SL, mid-value of DR and 10 dB below the UCL. Eysenck Personality Questionnaire (EPQ) was administered to assess personality of the participant. The results indicated that: 1. ANLs obtained in the unaided and aided conditions were not significantly different. 2. The difference in ANLs across the severity of hearing loss was non-significant, indicating that ANLs are not affected by the peripheral hearing loss. 3. Digital noise-reduction feature significantly decreased the ANL by increasing the amount of tolerance to background noise. 4. When ANL was measured at different presentation levels of speech, there was a gradual increment in the ANL with increase in the presentation level. 5. On personality assessment, the higher extroverted personality obtained a lower ANL while the participant high on neuroticism obtained a higher ANL.

Key words: noise reduction, presentation levels, global ANL, personality questionnaire

Introduction

People with cochlear hearing loss frequently complain that their hearing aids are of limited benefit. Difficulty in understanding speech in the presence of noise is the most frequent complaint of adults who use hearing aids (Kochkin, 2002; Cord, Surr, Walden and Dyrlund, 2004). Nabelek, Tucker, and Letowski (1991) hypothesized that willingness to listen to speech in background noise may be more indicative of hearing aid use than speech perception scores obtained in the background noise. This hypothesis led to the development of a procedure called “Acceptable Noise Level” (ANL) which is a measure of the willingness to accept the background noise while listening to speech.

The ANL is defined as the difference between the most comfortable listening level (MCL) for running speech and the maximum background noise level (BNL) that a listener is willing to accept. The ANL measure assumes that speech understanding in noise may not be as important as is the willingness to listen in the presence of noise. It has been established that people who accept background noise have smaller ANLs and

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tend to be “good” users of hearing aids (Nabelek, Freyaldenhoven, Tampas, Burchfield and Muenchen, 2006).

Nabelek, et al., (2006) assessed the usefulness of ANL as a predictor of hearing aid use. Results indicated that ANLs were related to hearing aid use. Specifically, full-time hearing aid users accepted more background noise than part-time users or non-users. Studies on ANL have investigated the effect of age, hearing sensitivity (Nabelek, Tucker and Letowski, 1991), gender (Rogers, Harkrider, Burchfield and Nabelek, 2003), type of background noise (Crowley and Nabelek, 1996), efferent activity of the medial olivocochlear bundle (MOCB) pathway (Harkrider and Smith, 2005), middle ear characteristics or speech perception in noise performance (Nabelek, Tampas and Burchfield, 2004).

Although the utility of ANL as a clinical tool to assess the success of hearing aid use has been established, most of the studies have used listeners with mild to moderate hearing loss. Freyaldenhoven, Plyler, Thelin and Hedrick (2007) investigated the effect of hearing sensitivity on ANL, by comparing global ANL to the pure-tone average (PTA) in listeners with hearing loss. The participants of the study had hearing loss of mild-moderately sloping to moderately severe- severe degree. The results obtained were insignificant indicating that global ANL is not related to hearing sensitivity. However, the participants were not classified into separate groups based on the degree of hearing loss. There is a dearth in literature on how the ANL value varies as a function of hearing loss from moderate to severe degree of loss.

Also, the recent development of digital hearing aids opens up substantial new possibilities with respect to the use of advanced signal-processing techniques for noise reduction (Levitt, Neuman, Mills and Schwander, 1986). Mueller, Weber, Benjamin, and Hornsby (2006) studied the effect of digital noise reduction (DNR) on ANL on 22 adults fitted with 16-channel wide-dynamic range compression hearing aids with DNR processing. The results indicated a significant mean improvement for ANL (4.2 dB) for the DNR-on condition than DNR-off condition. However, it was not indicated that how this improvement with noise reduction algorithm varied as a function of degree of hearing loss and/or as a function of presentation level of the speech stimulus.

The effect of personality types on hearing aid benefit has been studied. Cox, Alexander, and Gray (1999) studied the relationship between the personality trait and self-reported hearing aid benefit in 83 individuals with mild to moderate sensorineural hearing loss. Participants completed the APHAB and measures of personality on the State-Trait Anxiety Inventory (STAI) (Spielberger, 1983), the MBTI (Myers and McCaulley, 1985), and a measure of locus of control (Levenson, 1981). The results indicated extroversion-introversion to be the best predictor of hearing aid benefit. However, there has been no literature available on the effect of personality type on ANL.
score. Since, ANL is indicated as an inherent characteristic of an individual, which does not change with age or acquired hearing loss (Nabelek, et al., 2006), it would be interesting to study its relation with the personality of an individual.

The present study aimed to evaluate the following:
1. The relationship between the unaided and aided Acceptable Noise Levels
2. The effect of degree of hearing loss on Aided Acceptable Noise Levels.
5. The effect of personality type on Aided Acceptable Noise Levels.

Method

Participants

21 participants in the age range of 15 to 65 years (mean age being 49.78 years) were included in the study. They had a speech identification scores (SIS) of ≥ 75 %. The participants had hearing loss that was acquired post-lingually, symmetrical bilaterally and either sensori neural or mixed. They were native speakers of Kannada language. All of them were naïve hearing aid users. The participants did not have any significant neurological or cognitive listening deficits. Participants who got a lie score of ≤4 on the Eysenck Personality Questionnaire (EPQ) were included. Each participant was assigned in one of the three groups based on the degree of hearing loss. Each group had seven participants. The groups were-
- **Group I** - Participants with moderate degree of hearing loss (PTA between 41 dB HL and 55 dB HL).
- **Group II** - Participants with moderately-severe degree of hearing loss (PTA between 56 dB HL and 70 dB HL).
- **Group III** - Participants with severe degree of hearing loss (PTA between 71 dB HL and 90 dB HL).

Equipment and Test Material

1. A 15 channel digital BTE hearing aid, suitable for the hearing loss of the participants, with a noise reduction algorithm.
2. Personal Computer and Hi-Pro
3. A calibrated double channel diagnostic free-field audiometer
4. Speech material - Three recorded speech passages in Kannada.
5. Eysenck Personality Questionnaire (EPQ) (Eysenck and Eysenck, 1975).
**Procedure**

The conventional ANL procedure involved the listeners to first adjust the level of a story to their most comfortable listening level (MCL). Then, a background noise was added, and the listener had to adjust the noise to a level at which they would be willing to accept or “put up with” without becoming tense or tired while following the words of the story (called “background noise level, BNL”). The ANL was calculated by subtracting the BNL from MCL. In addition, the effect of presentation level on ANLs was also evaluated. For this, one of the three groups in the study was chosen. Since, the participants in Group I had the maximum dynamic range (DR) as compared to those in Group II and Group III, participants in Group I were utilized for this purpose.

The data were collected in the following five stages:

- **Stage 1. Establishing the unaided ANL (ANL₁).**
- **Stage 2. Programming the hearing aid.**
- **Stage 3. Establishing the aided ANL with noise reduction scheme turned off (ANL₂).**
- **Stage 4. Establishing the aided ANL with noise reduction scheme turned on (ANL₃).**
- **Stage 5. Assessment of the personality through Eysenck Personality Questionnaire.**

**Stage 1. Establishing the unaided ANL (ANL₁).**

The participant was made to sit comfortably on a chair in front of the loudspeaker in the test room. The speech stimulus (recorded passage in Kannada) was initially presented through the loudspeaker at the level of speech reception threshold (SRT), determined at the time of audiological assessment. Gradually, the presentation level was adjusted in 5 dB steps up to the level of MCL and then in 2 dB steps until the participant’s MCL was established reliably. The step was repeated two times, and the average of the two levels was taken as the MCL. This level was noted down as MCL.

For establishing the MCL the following instructions were given: “You will listen to a story through the loudspeaker placed in front of you. The loudness of the story will be varied. First, the loudness will be turned up until it is too loud and then down until it is too soft. You have to indicate the level at which the loudness of the story is most comfortable for you”.

At this stage, a speech noise was introduced at 30 dB HL and its level raised to a point, in 5 dB steps, at which the participant was willing to accept the noise without becoming tired or tensed while listening to and following the words in the speech. This maximum level of speech noise at which he/she could accept the noise without becoming tired was considered as the BNL. For the purpose of establishing BNL, the instructions used were “You will listen to the story with a background noise. After you have listened to it for a few moments indicate the level of background noise that is the most you would be willing to accept or ‘put-up-with’ without becoming tense or tired while following the word.”
story. First, the noise will be turned up until it is too loud and then down until the story becomes very clear. Finally, indicate the maximum noise level that you would be willing to ‘put-up-with’ for a long time while following the story”. This level, called the BNL, was noted down. The ANL₁ (in dB) was calculated as difference between MCL (dB HL) and BNL (dB HL) for each participant.

For the participants in Group I, the participant’s dynamic range was also determined, by establishing the SRT and UCL for speech. The difference between UCL and SRT determined the dynamic range (DR). The instructions used for establishing the DR was “You will be listening to a story through the loudspeaker placed in front of you. Initially the loudness of speech will be more and gradually the loudness will reduce. You have to indicate the softest level at which you are just able to follow the story. Then, the loudness of the story will be gradually increased. You have to indicate the level at which you are not able to tolerate the loudness of the speech stimulus”. This procedure was repeated two times, and the average level was taken as DR. The dynamic range (DR) was measured for each participant in Group I. For this purpose, the softest level at which the participant was just able to follow the story was determined. For the purpose of the study, this level was noted down as SRT. Then, the level was gradually increased to a level at which the participant reported of discomfort. This level was noted down as UCL. ANL was obtained at three presentation levels of speech, i.e., at 5 dB SL, at mid value of DR and at 10 dB below UCL. The ANL₁ was calculated as difference between Presentation Level (dB HL) and BNL (dB HL). Thus, for the participants in Group I, three ANLs were obtained at three presentation levels, which were referred as ANL₁a, ANL₁b and ANL₁c respectively. This was in addition to the ANL₁ that was measured, as described earlier.

Stage 2. Programming the hearing aid

The hearing aid was connected to the programming hardware (Hi-Pro) through a cable and detected by the programming software. The hearing thresholds of the better ear of the participants were then fed into the programming software and target gain curves were obtained using the proprietary prescription formula of the hearing aid. The hearing aid gain was set to the default target gain and fine-tuned according to participant’s preference. The hearing aid chosen for the study had three programs. Program 1 of the hearing instrument had the noise reduction algorithm (dNC) turned off. In Program 2, the noise reduction algorithm (dNC) was turned on. The dNC had two degrees of noise reduction - light, and moderate. ‘Moderate dNC’ was selected for the purpose of the study. Only Program 1 and Program 2 were used in the study. These settings were saved in the hearing aid for each participant.
Stage 3. Establishing the aided ANL with noise reduction scheme turned off (ANL₂).

Only one ear, the ear with better PTA, of the participant was aided. The hearing aid was fitted in the ear with better pure tone average while it was ensured that the participation of the unaided ear was ruled out, when indicated. The hearing aid was set at Program 1. The participant was made to listen to a story passage. A different story was used to avoid any practice effect. MCL and BNL were determined in this condition following the procedure described earlier, and the ANL was calculated. This ANL was labeled as ANL₂.

For the participants in Group I, their SRT₂ and UCL₂ were established to obtain the dynamic range with the hearing aid called DR₂. The ANL was obtained at 3 presentation levels of speech, i.e., at 5 dB SL, mid value of DR₂ and 10 dB below UCL₂. The ANL₂ was calculated as difference between presentation level (dB HL) and BNL (dB HL). Thus, for the participants in Group I, ANLs were obtained at three presentation levels, which were referred to as ANL₂a, ANL₂b and ANL₂c respectively.

Stage 4. Establishing the aided ANL with noise reduction scheme turned on (ANL₃).

To activate the noise reduction scheme in the hearing aid, Program 2 was used. This program had all the settings similar to Program 1 except for the addition of dNC noise reduction scheme at moderate level. With this program setting, the entire procedure described in Stage 3 was repeated and ANL was obtained. This was labeled as ANL₃. For the participants in Group I, three ANLs were obtained at three presentation levels, which were referred as ANL₃a, ANL₃b and ANL₃c respectively.

Stage 5. Assessment of the personality through Eysenck Personality Questionnaire (EPQ).

The Eysenck Personality Questionnaire (EPQ) (Eysenck and Eysenck, 1975) was administered to each participant. He/she was instructed to read each statement and if it described him or her, or, if he or she was in agreement with the statement then to draw a circle around ‘Yes’. If the statement did not describe the participant, then a circle was drawn around ‘No’. The participant was also informed that there were no right or wrong answers and were required to give honest answers.

Scoring was done after the administration of EPQ. Each response was checked with the scoring key. Scoring was done with the help of a clinical psychologist. If the participant’s response agreed with the key, a score of 1 was given, if not a score of 0 was given. Separate scores were derived for Extroversion (E) and Neuroticism (N). Lie scale (L) was also checked and the number of responses agreeing with the key was totaled. Thus, there were three scores - E, N and L. If the L score was found to be high (5 or >5), the data were deleted and the participant was not considered for further study. In the present study, two participants had to be deleted based on this criterion.
Results and discussion

The analysis of the data was done separately for Group I, Group II and Group III.

**Group I (moderate degree of hearing loss)**

In the unaided condition, the mean ANL in group with moderate hearing loss was 8.85 dB with a standard deviation of 2.03 dB. In the first aided condition when the noise reduction was turned-off (A1), the mean ANL was also 8.85 dB with a standard deviation of 1.57 dB. In the second aided condition when the noise-reduction was turned-on (A2), the mean ANL was 7.71 dB with a standard deviation of 1.79 dB. It was also noticed that, the MCL values in all the conditions were higher than the BNL values.

**Group II (moderately-severe degree of hearing loss)**

In the unaided condition, the mean ANL in the group with the moderately-severe hearing loss was 9.42 dB with a standard deviation of 2.93 dB. In the first aided condition (A1), the mean ANL was 9.71 dB with a standard deviation of 2.43 dB. In the second aided condition (A2), the mean ANL was 8.85 dB with a standard deviation of 1.95 dB. The MCL values in all the conditions were higher than the BNL values.

**Group III (severe degree of hearing loss)**

In the unaided condition for the group with severe hearing loss, the mean ANL was 8.28 dB with a standard deviation of 2.98 dB. In the first aided condition (A1), the mean ANL was 8.57 dB with a standard deviation of 3.20 dB. In the second aided condition (A2), the mean ANL was 7.71 dB with a standard deviation of 2.69 dB. The MCL values in all the conditions were higher than the BNL values.

**The effect of digital hearing aid, with and without the noise reduction feature, on aided Acceptable Noise Levels and its relationship with the unaided ANLs**

To evaluate interactions between the ANLs obtained in the unaided and different aided conditions across the severity of the hearing impairment, mixed analysis of variance (ANOVA) was done. There was no significant interaction effect of the ANL among the three conditions across the severity of hearing impairment [F (4, 36) = 0.202, p > 0.05]. Thus, the results indicated that the ANLs obtained for all the three groups varying in severity of hearing impairment were not significantly different. However, there was a significant main effect for different unaided and aided conditions [F (2, 36) = 5.66, p<0.01]. To evaluate the significant differences in the unaided (UA) and different aided (A1 & A2) conditions, pair-wise comparison using post-hoc Bonferroni test was done. The Table 1 depicts the results of post-hoc Bonferroni analysis for the UA, A1 and A2 conditions.
Table 1: Results of post-hoc Bonferroni analysis for the unaided (UA) and aided (A1 & A2) conditions.

<table>
<thead>
<tr>
<th>Conditions</th>
<th>Significance level</th>
</tr>
</thead>
<tbody>
<tr>
<td>UA*A1</td>
<td>p&gt;0.05</td>
</tr>
<tr>
<td>UA*A2</td>
<td>p&lt;0.05</td>
</tr>
<tr>
<td>A1*A2</td>
<td>p&lt;0.05</td>
</tr>
</tbody>
</table>

Pair-wise comparisons of the UA, A1 and A2 conditions revealed a non-significant difference (p > 0.05) between the ANLs obtained in the unaided and A1 conditions, i.e., the ANL obtained in the unaided condition and in the aided condition with noise-reduction algorithm turned-off were not significantly different. This implied that the hearing aid did not make any significant difference in the ANLs measured in the UA and A1 conditions. However, the ANLs obtained in the A2 condition were significantly different (p < 0.05) from the ANLs obtained in the unaided and A1 conditions, i.e., when the noise-reduction algorithm was turned-on, the ANL values obtained were significantly different from those obtained in the unaided and the aided condition with noise-reduction turned-off conditions. Thus, the activating the noise-reduction feature in a hearing aid significantly affected the ANL score. The ANL values were always lower in the aided condition when the noise-reduction was turned-off.

The results of the present study are in agreement with those reported earlier (Nabelek, Tucker and Letowski, 1991; Nabelek, Freyaldenhoven, Thelin, Burchfield and Muenchen, 2006; Mueller, Weber and Hornsby, 2006). The participants with average thresholds in the range of mild to moderate were included in the previous studies. In the present study, the hearing loss of the participants ranged from moderate to severe degree and it was found that the ANLs were not significantly different from moderate hearing loss to severe hearing loss groups.

Harkrider and Smith (2005) investigated the individual differences in the efferent activity in medial olivocochlear bundle (MOCB) and acoustic reflex (AR) pathways to account for inter-subject variability in ANL and phoneme recognition in noise (PRN). They indicated that the amount of background noise the participants were willing to accept in monotic and dichotic listening conditions were directly related, suggesting that non-peripheral factors, beyond the level of the superior olivary complex where binaural processing first occurs, mediate ANL. Thus, in the present study even when the degree of hearing loss was varying across the groups, the ANL obtained for the participants was not dependent on the severity of hearing loss, indicating a central mediation of ANL.

The results of pair-wise comparison indicated a significant difference between the ANLs obtained in both UA or A1 conditions and the A2 condition. Also, no significant difference between UA and A1 condition was noticed. The results are in agreement with
the findings of Nabelek, Tampas and Burchfield (2004) that compared the speech perception in background noise (SPIN) with the acceptance of background noise in the unaided and the aided conditions. The results indicated that ANLs were independent of hearing aid amplification. In their study, the ANLs and SPIN scores were found to be unrelated as there was a significant improvement in the SPIN scores with amplification.

In the present study, when the noise-reduction algorithm was turned-on, the mean MCL value was found to be same as in the first aided condition, i.e., 61.4 dB, which indicates no effect of noise reduction algorithm on the MCL for speech. The mean BNL values for the moderate group in the UA, A1 and A2 conditions were 63.1 dB, 52.5 dB and 53.7 dB respectively. Thus, a reduction in the BNL was observed with the use of the hearing aid. However, when the noise-reduction is turned-on, the BNL was further increased by 1.2 dB, which is reflected as a lower ANL in the A2 condition than in A1 condition. A similar trend was seen in Group II and Group III. It should be noted that in a study by Nabelek, Tampas, and Burchfield (2004), the full-time and part-time groups had similar MCLs and the main contributor for the difference in ANL was the BNL. In contrast, in the present study, the groups were formed on basis of the degree of hearing loss and though both the MCLs and BNLs were found to vary, the ANLs were found to be similar in each group.

Within the group, the ANLs in the A2 condition were significantly different from the ANL obtained in the UA and A1 conditions. This indicates a significant effect of noise-reduction algorithm on the ANLs and specifically on the BNLs, as the MCLs were unaffected with the noise-reduction algorithm. The results of the present study are in consensus with the results of the study by Mueller Weber, and Hornsby (2006), in which a significant mean improvement of 4.2 dB was observed on ANLs with the noise-reduction turned-on. However, in the present study, the mean improvement in ANL with noise reduction algorithm turned-on was 1.1 dB in Group I, 0.9 dB in Group II and 0.8 dB in Group III, which was significant. The discrepancy between the mean improvement in the study by Mueller Weber and Hornsby (2006) and the present study could be accounted for by the differences in the hearing instrument used and the lesser number of participants in each group in the present study. It has been reported that the listeners often demonstrate a strong tendency for subjective preference for digital noise reduction (DNR) algorithms (Boymans and Dreschler, 2000), and actual improvement in speech perception is reportedly unreliable.

**Effect of presentation level on Aided Acceptable Noise Levels in moderate degree of hearing loss**

Table 2 gives the mean and standard deviation (SD) of the ANL values obtained at the three presentations levels in the unaided and two aided conditions. The results indicated that the ANL score increased as the level of presentation increased.
Table 2: Mean and standard deviation (SD) of ANLs obtained at three presentation levels, in unaided (UA) and aided (A1 & A2) conditions.

<table>
<thead>
<tr>
<th>Condition</th>
<th>Mean (dB)</th>
<th>Std. Deviation</th>
</tr>
</thead>
<tbody>
<tr>
<td>UA ANL1</td>
<td>8.86</td>
<td>2.61</td>
</tr>
<tr>
<td>UA ANL2</td>
<td>10.00</td>
<td>1.73</td>
</tr>
<tr>
<td>UA ANL3</td>
<td>11.71</td>
<td>1.38</td>
</tr>
<tr>
<td>A1 ANL1</td>
<td>8.28</td>
<td>1.70</td>
</tr>
<tr>
<td>A1 ANL2</td>
<td>10.71</td>
<td>2.98</td>
</tr>
<tr>
<td>A1 ANL3</td>
<td>11.86</td>
<td>1.95</td>
</tr>
<tr>
<td>A2 ANL1</td>
<td>7.71</td>
<td>2.50</td>
</tr>
<tr>
<td>A2 ANL2</td>
<td>9.00</td>
<td>2.89</td>
</tr>
<tr>
<td>A2 ANL3</td>
<td>9.28</td>
<td>2.43</td>
</tr>
</tbody>
</table>

The Global ANLs were also calculated by taking the sum of ANLs obtained at three presentation levels and averaging it. Table 3 shows the mean and standard deviation for the Global ANLs across the unaided and the two aided conditions.

Table 3: Mean, standard deviation and range of the Global ANL in the unaided (UA) and aided (A1 and A2) conditions

<table>
<thead>
<tr>
<th>Condition</th>
<th>Global ANL, dB</th>
</tr>
</thead>
<tbody>
<tr>
<td></td>
<td>Mean</td>
</tr>
<tr>
<td>UA</td>
<td>10.19</td>
</tr>
<tr>
<td>A1</td>
<td>10.28</td>
</tr>
<tr>
<td>A2</td>
<td>8.67</td>
</tr>
</tbody>
</table>

The results indicate that the mean Global ANL was 10.19 dB in the unaided condition (UA), 10.28 dB in the first aided condition (A1) and 8.66 dB in the second aided condition (A2). To determine the effect of presentation level on ANL under different conditions, two-way repeated measures ANOVA was carried out.

The results are in agreement with the study by Freyaldenhoven, Plyler, Thelin and Hedrick (2007) determining the effect of presentation level on ANLs. They measured the effects of speech presentation level on the acceptance of noise in listeners with normal and impaired hearing to determine whether these effects were related to the hearing sensitivity of the listener. The results demonstrated that global ANLs and ANL growth were not significantly different for listeners with normal and impaired hearing. Further, neither of the ANL measures was related to pure-tone average (PTA; i.e., average of 0.5,
1, 2, and 4 kHz) for listeners with impaired hearing. In addition, conventional ANLs were significantly correlated with both global ANLs and ANL growth for all listeners.

Freyaldenhoven, Plyler, Thelin and Muenchen (2008) evaluated the effects of speech presentation level on the hearing aid users. The participants formed different groups based on the hearing aid use were tested at eight presentation levels in the aided condition. Results indicated similar findings as reported by Freyaldenhoven, Plyler, Thelin and Hedrick (2007). However in their study too, no direct comparison was made between the unaided and aided conditions, as a function of speech presentation level. This discrepancy in results in the present study and the previous research can be explained in terms of measurement of ANL. It is possible that since the ANLs were measured at different presentation levels and not at MCLs. This would have affected the performance when the participants were aided. When the participant is aided, the dynamic range is increased and thus the perception of loudness of speech and the background noise at the three presentation levels would be different from the loudness perceived in the unaided conditions at three presentation levels.

**Effect of personality type on Aided Acceptable Noise Levels**

To investigate the effect the personality type on the ANL value, Pearson’s correlation analysis was carried out. The personality scores were obtained as Extroversion score and Neuroticism score and compared with ANLs separately. Table 4 gives the results of correlation analysis between ANL (in different conditions) and Extroversion score.

Table 4: Pearson’s correlation between ANL and Extroversion score.

<table>
<thead>
<tr>
<th>Condition</th>
<th>N</th>
<th>Pearson Correlation</th>
<th>Level of significance</th>
</tr>
</thead>
<tbody>
<tr>
<td>UA ANL</td>
<td>21</td>
<td>-0.288</td>
<td>p&gt;0.05</td>
</tr>
<tr>
<td>A1 ANL</td>
<td>21</td>
<td>-0.118</td>
<td>p&gt;0.05</td>
</tr>
<tr>
<td>A2 ANL</td>
<td>21</td>
<td>-0.288</td>
<td>p&gt;0.05</td>
</tr>
</tbody>
</table>

The results indicate negative correlation between the ANLs and the Extroversion score (p>0.05). Thus, for all the conditions, the participants with higher ANL scores achieved a lower score on Extroversion scale and vice versa. However, this correlation was non-significant. It is thus possible that, an individual scoring higher on extroversion is likely to tolerate more amounts of background noise and thus obtains a lower ANL.

The Table 5 gives the results of correlation analysis between ANL (in different conditions) and Neuroticism score. The results indicate a positive correlation between the ANLs and the Neuroticism score (p>0.05).
Table 5: Results of correlation analysis between ANL and Neuroticism score.

<table>
<thead>
<tr>
<th>Condition (N=21)</th>
<th>Pearson Correlation</th>
<th>Level of significance</th>
</tr>
</thead>
<tbody>
<tr>
<td>UA ANL</td>
<td>0.196</td>
<td>p&gt;0.05</td>
</tr>
<tr>
<td>A1 ANL</td>
<td>0.122</td>
<td>p&gt;0.05</td>
</tr>
<tr>
<td>A2 ANL</td>
<td>0.101</td>
<td>p&gt;0.05</td>
</tr>
</tbody>
</table>

For all the test conditions, the participants with higher ANL scores achieved a higher score on Neuroticism scale and vice versa. However, this correlation was also non-significant. According to Costa and McCrae (1997), individuals scoring low on Neuroticism are more relaxed and calm, and are better able to cope with stressful situations in their lives. Thus, participants who scored low on neuroticism were capable of tolerating more amount of background noise which was reflected as a low ANL score. The background noise represents a stressful condition for an individual with hearing impairment.

The results on the effect of personality, thus, indicate that the acceptance of background noise by an individual is related to his personality also. The relationship was however, non-significant, which may be because of the limited number of participants in the study. Cox, Alexander and Gray (1999) investigated the relationship between the personality trait and self-reported hearing aid benefit in individuals with mild to moderate sensori-neural hearing loss. The results indicated extroversion-introversion to be the best predictor of hearing aid benefit. More extroverted individuals reported greater hearing aid benefit on these three sub-scales of the APHAB than the more introverted individuals. In addition, individuals who reported greater anxiety also reported more problems in communication as measured on the aided condition of the ‘Ease of Communication’ sub-scale of the APHAB.

Summary of the findings of the present study:

The present study investigated the effect of digital hearing aid, with and without noise reduction, on the aided Acceptable Noise Levels and the relationship between the unaided and aided ANLs. The effect of degree of hearing loss, presentation level of speech stimuli, and the effect of personality type on the aided Acceptable Noise Levels (ANLs) were also investigated. The findings of the study indicate:

1. ANLs obtained in the unaided and aided conditions are not significantly different.
2. ANLs across the severity of hearing loss were found to be non-significant, indicating that ANLs are not affected by the peripheral hearing loss.
3. Digital noise-reduction feature significantly decreased the ANL by increasing the amount of tolerance for background noise. However, as the hearing loss increased, this decrement in ANL reduced.

4. When ANL was measured at different presentation levels of speech rather than MCL, there was a gradual increment in the ANL with increase in the presentation level.

5. The personality of the participant influenced the ANLs. Individuals with higher extroverted personality type obtained a lower ANL while those with high neuroticism obtained a higher ANL. However, this correlation was not significant.

References


Rhythm Perception with Fine Structure Cues: A Simulated Study

Ramia Narayanan & K. Rajalakshmi*

Abstract

The present study aimed at investigating the effect of adding fine structure cues in rhythm perception. All the speech coding strategies used with cochlear implants extract the envelope of the signal and code the information while missing out information on the fine structure cues. In this study, the fine structure cues were provided along with the envelope and the rhythm perception enhancement was studied. A group of 30 individuals with hearing within normal limits served as subjects for the study. The stimulus consisted of thirty rhymes which were modified by the MATLAB 6.5 software into envelope only (AM) and envelope + fine structure (AM+FM) conditions with 4, 8, 12 channels respectively. Significant improvement in the rhythm perception was seen with the addition of fine structure cues. This study focuses on the importance of adding fine structure cues in the cochlear implant processing strategy to enhance the performance of cochlear implant users in both speech and music perception.

Keywords: Envelope cues, Fine structure cues, Cochlear implants, FAME strategy, Rhythm perception.

Introduction

Cochlear implants (CI) are devices that have been successful in restoring hearing to profoundly deaf patients through electrical stimulation of the auditory nerve with fine electrodes inserted into the scala tympani of the cochlea. The performance of cochlear implants depends on the speech processor’s ability to faithfully decompose speech signals into a number of channels of narrow-band electrical signals that is used to activate the spiral ganglion cells of the auditory nerve. The number of electrodes in modern cochlear implant devices may be different from the number of channels in the speech processor, and may vary from 12 to 22 or more depending on the strategy used or with the monopolar or bipolar stimulation used. The configuration and placement of the electrodes are of importance to the overall performance of these devices.

Components of music perception:

Every person is immersed in an environment full of sound and being able to understand speech is not the only function of hearing. For most people, listening to music is also a significant and enjoyable experience. Natural speech carries abundant acoustic cues in both spectral and temporal domains (Smith, Delgutte and Oxenham, 2002), while music is very complex and wide ranging. Pitch conveys melody and is strongly related to the spectral content of the sound. For accurate musical perception, all three of these characteristics must be transmitted. Rhythm, pitch, timbre and melody identification requires good functioning of highly specific patterns of discrimination.

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**Rhythm**: temporal patterns in musical sounds along with overall variations in loudness are perceived as the rhythm.

**Melody**: the ability to recognize melody depends on highly variable factors, such as individual’s musical training, the socio-cultural experience, memory of the tune and situational contexts. The ability to perceive accurately the fundamental features of musical sounds, such as pitch and temporal patterns is a pre-requisite for melody recognition.

**Timbre**: the principal properties of the frequency spectrum and the amplitude envelope of sounds, including changes in those attributes over time and frequency are also relevant (Fragoulis, 1999).

**Music perception in Cochlear implants**

Investigators have shown that trends in the patterns of correlation between speech and music perception suggests that particular structural elements of music are differently accessible to cochlear implant users (Gfeller and Lansing, 1997; McDermott and McKay, 1997). Pitch cues can be elicited using two independent mechanisms in cochlear implants, namely place pitch and temporal pitch cues (Tong, Clark and Blamey, 1982; Shannon, 1983; McKay, McDermott and Clark, 1996). Temporal pitch cues arise when the repetition rate of stimulation on one channel changes. The temporal pitch sensation rises with increasing rate up to 300 Hz but saturates at higher rates (Shannon, 1983). Place pitch cues arise when the site of stimulation is changed while keeping the stimulation rate constant, with more basal stimulation eliciting higher pitches. It has been shown that both temporal and place pitch cues enable CI recipients to some degree perceive Fo differences of synthetic harmonic sounds with currently used sound processors (Geurts and Wouters, 2001), and in normal hearing subjects with acoustic models of CI processors (Green et al., 2004). Rhythm perception and melody perception was observed to be poorer than normal hearing counter parts due to inadequate cues provided through the cochlear implant (Gfeller and Lansing, 1991).

Music perception in CI subjects was poor compared to normal hearing subjects (Leal, 2003; Gfeller and Lansing, 1991). Pijl (1997) showed that at least part of the limited pitch performance of the CI subjects was due to ineffective sound processing as CI subjects were able to estimate musical intervals more accurately for synthetic stimuli (pulse trains) than for real musical sounds. Part of this limited Fo discrimination ability in CI recipients using their speech processors may be due to the limited transmission of the fine temporal information in current speech processors, since most current speech processors only extract the slowly varying envelope and use this to modulate a constant rate pulse (Smith, Delgutte and Oxenham, 2002).

The temporal envelope consists of frequency information in the 2-50 Hz range; periodicity consists of frequency information in the 50-500 Hz range and the fine structure consists of frequency information in the 600-10,000 Hz (Rosen, 1992).
It is commonly believed that envelope cues are represented in the auditory system as fluctuations in the short-term rate of firing in auditory neurons, while temporal fine structure (FM) is represented by the synchronization of nerve spikes to a specific phase of the carrier (phase locking) (Shannon, Zeng, Kamath, Wygonski and Ekelid, 1995).

Current signal processing for cochlear implants allow adequate speech perception in quiet environments for most users. However, their speech recognition performance in noise and music perception is severely limited. Typically, each electrode receives an amplitude modulated pulse train representing the narrow band temporal envelope of a sound from a particular frequency band. Amplitude modulations from low frequencies are delivered to the apical electrodes and amplitude modulations from high frequencies are delivered to the basal electrodes. Though amplitude modulations are sufficient to support sentence recognition in quiet (Shannon, 1995), it is not sufficient for speech recognition in noise and music perception (Dorman, Loizou and Tu, 1998; Freisen, Shannon, Baskent and Wang, 2001). The pitch of the voice can be conveyed by the temporal envelope; however this cue provides a relatively weak representation of pitch especially for the high frequencies. This indicates that pitch information is not effectively coded by the envelope modulations and hence not sufficient for rhythm perception (Green and Rosen, 2004).

**Need for the study:**

The speech processing strategy in most modern cochlear implants extracts and encodes only amplitude modulation in a limited number of frequency bands and hence the place pitch coding is impaired. To overcome the inherent limitations of temporal pitch coding in cochlear implants, techniques may be needed to provide perceptual information about the “fine structure” of the acoustic signals. The fine structure contains rapidly varying components of sounds that are not present in the envelope coding strategies that are being used. Amplitude modulations alone do not provide good rhythm perception as described above; hence it is proposed to study the importance of adding fine structure cues in rhythm perception.

**Aim of the study:**

The study aims to understand the

1) Effect of number of channels on rhythm perception with envelope cues (amplitude modulation).
2) Effect of number of channels on rhythm perception with fine structure and envelope (amplitude and frequency modulation) cues.
3) To compare rhythm perception with the fine structure and envelope (AM + FM) condition and envelope only (AM) condition across different channels.

**Method**

**A. Subjects:**

The study included 30 subjects with normal hearing in the age range of 18-25yrs (15 females and 15 males with the mean age of 21.5 years). Pure tone audiometry thresholds at all frequencies were within 20 dBHL. No indication of middle ear pathology as shown by tympanometry and presence of acoustic reflexes at 500 Hz and 1000 Hz.

**B. Instrumentation:**

- A calibrated dual channel clinical audiometer and a calibrated Immittance meter were used to recruit subjects.
- A dual channel clinical audiometer was used for the testing.
- The stimulus generated using MATLAB 6.5 software was played using an Intel Pentium processor computer.

All the testing, both for selecting subjects and for experimental purposes were conducted in an air conditioned, acoustically treated double room set-up. The ambient noise levels inside the test room were within the permissible limits (re: ANSI S3.1 1991, as cited in Wilber 1994).

**C. Stimulus and signal processing:**

The stimulus consists of thirty nursery rhymes (30 rhymes; 5 rhymes in each of the six conditions). The simulation was done using MATLAB 6.5 software. The stimulus contains cues in the frequency range (100 Hz - 10,000 Hz). The stimulus was separated into its envelope (AM) and fine structure (FM) using FAME (frequency and amplitude modulation encoder) software. Frequency modulation (FM) was used to code the instantaneous frequency, or temporal fine structure of the speech waveform, while the envelope (AM) coded the instantaneous amplitude of the signal.

The stimulus was filtered into 4, 8, 12 narrow bands with third order Butterworth filters respectively. Each of the narrow bands was then subjected to full-wave rectification to obtain the slowly varying envelope followed by a low pass cut off filter at 500 Hz (first order Butterworth filter) to control the amplitude modulation rate, to form the AM component of the synthesized signal. The envelope thus extracted was then modulated with sinusoids of 100 – 10,000 Hz frequency range to produce the AM signal. The fine structure (FM) was extracted using FAME software. This was followed by low pass filtering (500 Hz) to limit the FM depth and rate to relatively slowly varying components that can be perceived by the cochlear implant users, an optimum of 500 Hz bandwidth and 400Hz rate was used (Stickney, Nie and Zeng, 2005). The processed
signal was then band pass filtered to remove any frequency components that fall outside the analysis filters bandwidth.

Finally, the band passed signals were summed to form the synthesized signals that contain AM and FM components. The slowly varying envelope modulated with the sinusoids of 100 Hz-10,000 Hz formed the AM component of the signal.

D. Procedure:

The stimulus was presented in a sound treated room in the free field condition at 40 dBSL. The subjects were asked to tap the rhythm of the rhymes heard (Schutz and Kerber, 1993). All the thirty rhymes were presented to the thirty subjects randomly in the six conditions. The conditions include

Amplitude modulated stimulus with 4 channels,
   A. Amplitude modulated stimulus with 8 channels,
   B. Amplitude modulated stimulus with 12 channels,
   C. Amplitude modulated and Frequency modulated stimulus with 4 channels,
   D. Amplitude modulated and Frequency modulated stimulus with 8 channels,
   E. Amplitude modulated and Frequency modulated stimulus with 12 channels.

The stimulus was presented and the participants were asked to tap out the rhythm heard. The rhythm pattern tapped was scored on a sheet for all the thirty (single foot) rhymes. This pattern was then compared with a model created from the tap pattern of three speech language pathologists who were well versed in music. Each correct tap was scored one and the incorrect tap was scored zero.

Results and Discussion

The present study aimed at investigating the effect of adding fine structure cues in rhythm perception. All the speech coding strategies used with cochlear implants extract the envelope of the signal and code the information while missing out information on the fine structure cues. In this study, the fine structure cues were provided along with the envelope and the rhythm perception enhancement was studied. A group of 30 individuals with hearing within normal limits served as subjects for the study. The stimulus consisted of thirty rhymes which were modified by the MATLAB 6.5 software into envelope only (AM) and envelope + fine structure (AM+FM) conditions with 4, 8, 12 channels respectively.

The following effects were analyzed using the repeated measure ANOVA with age as the independent variable

a) The effect of using only envelope (AM) cues in rhythm perception with respect to 4, 8,12 channels.
b) The effect of using fine structure (FM) cues along with envelope (AM) cues in rhythm perception with respect to 4, 8, 12 channels.

c) Paired t-test was done to compare the effects of envelope only and envelope and fine structure cues with respect to 4, 8, 12 channels.

The results are as follows:

(a) Effect of only envelope cues on rhythm perception with 4, 8, 12 channels:

A repeated measure ANOVA was performed with the number of channels and envelope cues (AM). There was a main effect of the number of channels. As expected the 12 channel condition produced higher scores than 8 channel condition, however there was no significant difference between 4 and 8 channels.

*AM represents envelope only condition.

*AMFM represents the condition where both envelope and fine structure cues are presented.

Figure 1 comparison of percentage correct scores across gender and 4,8,12 channels in AM and AM+FM condition.

Figure 1: Depicts the mean and the standard deviation as error bars for different channels across gender. From this figure it is evident that there is significant difference between the channels (4, 8, 12 channels) in the envelope only condition.

The statistical analysis shows that there is a significant difference between 8 and 12 channels when only amplitude cues are provided, but there is no significant difference between 4 and 8 channels. The absence of significant difference between 4 and 8 channels could be due to the large standard deviation as shown in table-1. This is in consonance with the earlier studies (Freisen, 2001; Stickney, 2006) which showed that increase in number of channels improves the speech identification scores in quiet and in the presence of noise in cochlear implant users. Leal et al. (2003) demonstrated a
significant correlation between speech perception in the presence of noise and rhythm identification and discrimination.

Table 1: Pair wise comparison across channels (4, 8, 12 channels) in the AM (envelope) condition.

<table>
<thead>
<tr>
<th>(I) AM</th>
<th>(J) AM</th>
<th>Mean difference (I-J)</th>
<th>Sig.</th>
</tr>
</thead>
<tbody>
<tr>
<td>1</td>
<td>2</td>
<td>-.633</td>
<td>1.000</td>
</tr>
<tr>
<td></td>
<td>3</td>
<td>-10.290</td>
<td>.079</td>
</tr>
<tr>
<td>2</td>
<td>1</td>
<td>.633</td>
<td>1.000</td>
</tr>
<tr>
<td></td>
<td>3</td>
<td>-9.657</td>
<td>.029*</td>
</tr>
<tr>
<td>3</td>
<td>1</td>
<td>10.290</td>
<td>.079</td>
</tr>
<tr>
<td></td>
<td>2</td>
<td>9.657</td>
<td>.029*</td>
</tr>
</tbody>
</table>

* The mean difference is significant at the .05 level.

In addition, Nie, et al. (2006) demonstrated a significant improvement in all speech tests with increase in number of stimulating electrodes from 4 to 12. The improvement in the scores was more significant in the quiet than in the presence of noise.

Brill et al. (1997) and Fishman, et al. (1997) demonstrated that 4-6 channels are adequate to support high levels of speech reception in quiet, they also report that further increase in number of channels do not produce an increase in speech test scores. This is true for speech in quiet, but doesn’t hold good for rhythm perception, as a significant improvement in scores is seen with increase in number of channels. This improvement in the scores with increase in number of channels can be attributed to the increased spectral information provided through the channels.

In addition, Freisen, et al. (2001) demonstrated that though the number of channels was increased CI users weren’t able to utilize the spectral information provided by the additional channels. The reason for this limitation was quoted as the electrode interactions and possible tonotopic shifts and warping in the frequency to place mapping of spectral information. No improvement was observed in speech recognition as the number of channels was increased from 7 to 20 channels for vowel and consonant recognition and increased from 10 to 20 channels for sentence recognition. However in this study, the effect of number of channels can be attributed to the normal physiological system of the subjects due to which they are able to make use of the fine spectral and temporal cues unlike the compromised physiological system in cochlear implantees (Souza and Boike, 2006).

Gfeller and Lansing (1991) examined the performance of 18 adult cochlear implant users on rhythm perception with the “primary measures of music audition” (PMMA). The speech coding strategies used were SPEAK and F0 F1 F2 and the mean
score for both schemes on the PMMA rhythm subtest was approximately 84% correct which is very close to the average scores obtained by a control group of 35 subjects with normal hearing (Gfeller, et al., 1997).

However, Leal et al. (2003) assessed rhythm perception by 29 recipients of the nucleus 24 electrode device. Twenty subjects used ACE while nine of them used SPEAK strategy. Rhythm discrimination and identification tasks were used. 24 of the 29 subjects obtained 75%-90% scores in the rhythm discrimination test while only 12 subjects obtained good scores in rhythm identification.

The statistical analysis also showed that there was no significant difference between gender and no significant interaction between the number of channels and gender. It is evident from the above studies, that rhythm perception is average in cochlear implant users and addition of fine structure cues could result in better rhythm perception due to the additional spectral and temporal cues available.

(b) Effect of number of channels (4, 8, 12 channels) on rhythm perception with envelope and fine structure cues:

Repeated measure ANOVA was performed with the number of channels and envelope + fine structure cues (AM+FM) conditions. There was a main effect of the number of channels. As expected the 12 channel condition produced higher scores than 8 channel condition, however there was no significant difference between 4 and 8 channels.

Table: 2 Pair-wise comparison across channels (4, 8, 12 channels) in the AM+FM (envelope + fine structure) cues condition.

<table>
<thead>
<tr>
<th>(I) AMFM</th>
<th>(J) AMFM</th>
<th>Mean Difference (I-J)</th>
<th>Sig.</th>
</tr>
</thead>
<tbody>
<tr>
<td>1</td>
<td>2</td>
<td>-4.867</td>
<td>.223</td>
</tr>
<tr>
<td>1</td>
<td>3</td>
<td>-22.653</td>
<td>.000*</td>
</tr>
<tr>
<td>2</td>
<td>1</td>
<td>4.867</td>
<td>.223</td>
</tr>
<tr>
<td>2</td>
<td>3</td>
<td>-17.787</td>
<td>.000*</td>
</tr>
<tr>
<td>3</td>
<td>1</td>
<td>22.653</td>
<td>.000*</td>
</tr>
<tr>
<td>3</td>
<td>2</td>
<td>17.787</td>
<td>.000*</td>
</tr>
</tbody>
</table>

* The mean difference is significant at the .05 level.

It is evident that there is significant improvement in the scores as the numbers of channels were increased from 4 to 12 and 8 to 12 channels. But no significant differences in scores were obtained between 4 and 8 channels. There was no significant interaction between the number of channels and gender and no significant effect of gender on the scores as expected.
Similar to the previous condition, increase in the number of bands and addition of fine structure (FM) cues resulted in improved performance in rhythm perception task.

Frequency modulations derived from the temporal fine structure is important to support speech recognition in noise and other critical functions such as speaker identification, music perception, tonal language perception and sound localization (Smith, Delgutte and Oxenham, 2002). Similarly it was stated that additional FM(fine structure) cues helps the listener to better segregate the envelope of the target signal and hence helps in improved performance (Nie, et al., 2005; Zeng, et al., 2005).

In addition, Nie, et al. (2005) demonstrated improved performance in speech tests with increase in number of channels and addition of fine structure (FM cues). This improvement was significant in the presence of noise condition and hence applies for rhythm perception. Significant correlation was obtained between speech in noise and rhythm perception (Leal et al., 2003).

The results of this study are in support with the above studies which conclude that increase in the number of channels with the fine structure cues (FM) results in improved overall performance. This improvement in performance can be attributed to the increased spectral and temporal resolution provided in the envelope and fine structure (AM+FM) condition.

(c) To compare the effect of envelope cues (AM) only and envelope along with fine structure cues (AMFM) across 4,8,12 channels.

To compare the effect of additional FM cues (AM + FM) in rhythm perception with 4, 8, 12 channels paired t-test was done.

<table>
<thead>
<tr>
<th>Pair</th>
<th>Condition</th>
<th>T</th>
<th>Sig. (2-tailed)</th>
</tr>
</thead>
<tbody>
<tr>
<td>Pair 1</td>
<td>P4AM – P4AMFM</td>
<td>1.333</td>
<td>.193</td>
</tr>
<tr>
<td>Pair 2</td>
<td>P8AM – P8AMFM</td>
<td>.077</td>
<td>.939</td>
</tr>
<tr>
<td>Pair 3</td>
<td>P12AM- P12AMFM</td>
<td>2.912</td>
<td>.007</td>
</tr>
</tbody>
</table>

The statistical analysis shows that there is a significant improvement in the 12 AM and 12 AM+FM condition, this can be attributed to the increased temporal and spectral information provided in this condition.

The results of this study are in accordance with the following studies:

Nie, Stickney and Zeng (2005) demonstrated that fine structure (FM) cues are crucial for improving cochlear implant performance under realistic conditions and also helps in music perception in normal hearing subjects. The subjects obtained large improvement in sentence recognition scores in the presence of noise. Speech perception
in noise and rhythm perception are significantly correlated (Leal, et al., 2003). Hence, the improvements in the scores in this study are attributed to the better temporal and spectral representation of the signal.

Lan, Nie, Gao and Zeng (2004) demonstrated that the novel speech processing strategy with dynamic modulations of both frequency and amplitude is encouraging as it resulted in significant improvement in perception of Chinese tones, phrases and sentences. Experimental results reveal that tonal information in Chinese speech was encoded primarily in the fine details in the spectrum of the speech signal. It is inferred from this study that if the perception of tonal languages improves with the novel processing strategy, then rhythm perception which depends on the overall loudness variations of the signal will also be perceived better.

Stickney, Nie and Zeng (2005) demonstrated a significant improvement in the speech perception scores when fine structure (FM) cues were added to the envelope (AM) information especially in realistic listening environments. The addition of FM cues would provide information about pitch and formant transitions which will help in better speech and music perception, in turn rhythm perception.

Zeng, Nie, Liu, Stickney, et al. (2004) demonstrated that envelope cues are important for speech perception whereas the fine structure cues are critical for pitch perception which in turn enhances rhythm perception.

Figure 2: Comparisons of percentage correct scores across number of channels

To summarize, from the figure 2 it is evident that subjects showed improved performance with fine structure cues (AM+FM) and with increased number of channels. This increase in the scores can be attributed to the increased spectral and temporal resolution provided with 12 channels and the additional fine structure cues.

Conclusions

The present study aimed at investigating the effect of adding fine structure cues in rhythm perception. All the speech coding strategies used with cochlear implants extract the envelope of the signal and code the information while missing out information on the
Rhythm perception with fine structure cues. In this study, the fine structure cues were provided along with the envelope and the rhythm perception enhancement was studied. A group of 30 individuals with hearing within normal limits served as subjects for the study. The stimulus consisted of thirty rhymes which were modified by the MATLAB 6.5 software into envelope only (AM) and envelope + fine structure (AM+FM) conditions with 4, 8, 12 channels respectively.

The investigation was carried out to address the following research goals:

- To determine effect of number of channels on rhythm perception with envelope cues (AM).
- To determine the effect of number of channels on rhythm perception with envelope and fine structure cues (AM+FM).
- To compare rhythm perception in the AM+FM condition and AM condition across different channels.

The results obtained are given below:

- In the first condition, i.e., rhythm perception with only envelope cues (AM) with 4, 8, 12 channels, significant improvement was seen between 8 and 12 channels, however no significant difference was measured between 4 and 8 channels. Hence it can be concluded that this improvement in scores with 12 channels could be due to the increased spectral information provided. However, the absence of improvement between 4 and 8 channels could be due to the large standard deviation obtained in this condition.

- In the second condition, i.e., rhythm perception with envelope and fine structure cues (AM+FM) with 4, 8, 12 channels, significant improvement was seen between 8 and 12 channels, however no significant difference was measured between 4 and 8 channels. This improvement in scores with 12 channels could be attributed to both the increased spectral information and temporal information provided by the fine structure cues. However, the absence of improvement between 4 and 8 channels could be due to the large standard deviation as mentioned.

- In the third condition, i.e., rhythm perception with envelope cues (AM) only and envelope + fine structure cues (AM+FM) with 4, 8, 12 channels, significant difference was seen between 12 AM and 12 AM+FM condition. However, no significant difference was seen between 4 AM and 4AM+FM condition, 8 AM and 8 AM+FM condition. Thus it can be concluded that significant difference between AM and AM+FM condition is seen only in the optimal condition, i.e., the condition with the maximum spectral and temporal cues (12 AM and 12 AM+FM condition).

This study focuses on the importance of adding fine structure cues in the cochlear implant processing strategy to enhance the performance of cochlear implant users in both speech and music perception. Further studies need to be done to understand the effect of
fine structure cues on melody and timbre perception, which will provide in depth information about the music perception. Studies should also be replicated in cochlear implant users to understand if similar results are obtained in them, as many other factors determine the performance of cochlear implant users unlike a simulated study on subjects with normal hearing.

References


Effect of Degree of Loss and Age on Speech Identification With Multi Channel Hearing Aids

Rubina & K. Rajalakshmi*

Abstract

Digital hearing aids have become common, with numerous modern digital signal processing technology. Multichannel hearing aids are now widely prescribed. There is conflicting evidence on the benefit of multichannel hearing aids. The current study investigated the effect of age and hearing loss on speech identification in multi channel hearing aids. Speech identification is measured using recorded version of bi-syllabic wordlist (Vijayalakshmi and Yathiraj,2005) in quiet and noise, using five and fifteen channel hearing aids, for two groups, fifteen adult and eighteen geriatric subjects who were again subgrouped into mild to moderate and moderately severe to severe hearing loss subjects. They Analysis of the results revealed that the benefit from the increase in number of channel is inversely proportional to the degree of hearing loss. In mild to moderate hearing loss subjects up to fifteen channel would not cause any deleterious effect in performance. The overall performance decreases with increasing age, but the geriatric subjects can combine the temporal information across channel with increase in number of channels. The degree of loss found to decrease the performance greatly with increase in number of channel rather than age.

Key words: Multichannel hearing aids, Number of channels, Speech identification, Age, and Hearing loss.

Introduction

The last decade has seen numerous and significant improvements in the technology of hearing aids. With advancement of digital technology, digital hearing aids have become increasingly common. Modern digital signal processing technology includes non –linear, adaptive, multi channels / bands, speech enhancement, noise reduction feedback management etc. The issue regarding the ideal number of channels had been a hot topic, and till to date there is conflicting evidence on the benefit of increasing number of channel in digital hearing aid.

Even though multi channel hearing aids are now widely prescribed to the subjects irrespective of their age and hearing loss, due to its frequency dependent compression characteristics, there is conflicting evidence on the benefit from this hearing aid. From theoretical point of view, multi channel compression is considered to be the best remedy for recruitment in sensory neural hearing loss. This is because multi channel compression can 1) improve audibility by better matching the variation of a person’s audible range across frequency, and 2) improve the signal to noise ratio (SNR) in situations where the background noise is dominant in a restricted range of frequencies.

Some experiments have shown multichannel compression to be better than single channel compression (Moore, Lynch and Stone, 1992, Souza and Turner 1999) some have failed to show any advantage for multichannel compression (Crain and Yund, 1995;
Hickson and Byrne, 1995; Plomp, 1994) and some have found no difference in speech ineligibility using single and multi channel compression hearing aids.

The degree of loss and age of the client are the two of the factors that can also limit the degree of success from the hearing aid (Dillon, 2001). It has long been accepted that listeners with severe loss require different amplification characteristics than listeners with mild to moderate hearing loss (Vantassel, 1993). Severe loss is characterized by supra threshold processing deficits, primarily by dramatically reduced frequency selectivity and also in some circumstances by reduced temporal discrimination. When the ability to resolve auditory information is limited, it is critical to select processing techniques that do not further degrade available speech cues. Listeners with severe loss are less able to resolve spectral detail. As a result, they may need to relay to a greater extent on temporal information such as variation in speech amplitude (Rosen et al., 1990), which are altered by multi channel and wide dynamic range compression hearing aids (Moore, 1996). For listeners with mild to moderate hearing loss who presumably depend to a greater extent on spectral cues, does benefit from improved speech recognition with multi channel hearing aids. Barford (1978) also reported that there was a shift towards better performance with the multi channel hearing aids as the severity of impairment decreased.

It also has been demonstrated frequently that older listeners have more difficulty understanding speech than younger listeners (Gordon–Salant and Fitzgibbons, 1997). Some studies have found no effect of age on speech recognition when younger and older listeners were matched for hearing sensitivity (Souza and Turner, 1998). In other studies, older listeners demonstrated poorer speech recognition than younger listeners even after accounting for threshold differences (Humes and Christopherson, 1991; Humes and Roberts, 1990). In general, age deficits occur more often in complex listening situations, such as speech presented in complex listening situations, such as speech presented in a noisy or reverberant environment. Studies support that older listeners experience reduced temporal resolution.

Souza and Kitch (2001) showed that mean identification scores decreased significantly with increasing age, the presence of hearing loss, the removal of spectral information, and with increasing distortion of the amplitude envelope (i.e., higher compression ratios). There was a consistent performance gap between young and aged listeners, regardless of the magnitude of change to the amplitude envelope.

Among the reasons for disagreement in the usefulness of multichannel hearing aids, degree of loss and age are important factors. There are limited number of studies directly comparing the effect of age and degree of loss with number of channel on speech identification with multichannel compression hearing aids and understanding the influence of the these factors would be useful in prescribing appropriate amplification. Therefore the current study was aimed at investigating the effect of number of channel on younger and older listeners and also with varying degree of hearing loss on speech identification.
Aim of the study

To examine the speech identification using multichannel hearing aids across severity of hearing loss and age.

1. To study the effect of severity of hearing loss on speech identification in multichannel hearing aids in quiet and in noise (+5 dBSNR).
2. To study the effect of age on speech identification in multichannel hearing aids in quiet and in noise (+5 dBSNR).

Method

Subjects: Sixteen adults in the age range of 20-55 yrs and eighteen geriatric subjects in the age range of 60-80 years, with unilateral or bilateral gradual sloping sensory neural hearing loss served as subjects for the study. Group I consisted of adult subjects (mean age: 38 years; age range 20 -55 years) with mild to severe sensory neural hearing loss. Group II consisted of Geriatric subjects (mean age: 70 years; age range: 58-80 years) with mild to severe sensory neural hearing loss. Based on their hearing threshold, these two groups were again subdivided into, mild to moderate hearing loss and moderately severe to severe hearing loss. In group I and group II, the mean pure tone thresholds for mild to moderate hearing loss was 49dBHL and 45dB HL respectively, and the mean pure tone thresholds for moderately severe to severe hearing loss was 60dBHL and 61 dB HL respectively. Graph 1 depicts the mean audiogram of the groups. Graph 2 and 3 depicts the mean with standard deviation of the pure tone thresholds of the two groups. The Speech identification scores of the all the participants were 50% or greater. They were Naïve hearing aid users. They were native speakers of Kannada language.

Figure 1: Mean puretone average threshold for adult group.
Stimuli: The speech stimuli used in the present study was taken from bi syllabic word list in Kannada developed by Vijayalakshmi and Yathiraj (2005). The speech material consists of four word lists each with 25 bi-syllabic words which are phonetically balanced. All the four lists were selected for the present study. As there are six conditions to test, each list was randomized to get two lists out of one list. Total of eight lists were available for testing. The words were spoken in conversational style by a female native speaker of Kannada. Words were digitally recorded in an acoustically treated room, on a data acquisition system, using 44.1 kHz sampling frequency and 16 bit analog to digital converter. All the words in a list were mixed individually with speech babble (Anitha and Manjula, 2005) at +5 dB SNR. The speech babble was mixed with words based on RMS level by the program written in MATLAB 6.5 software.

Hearing Aid description: Two non-linear behind the ear digital wide dynamic range compressions hearing aids with 5 channels and 15 channels were taken for the study.

Instrumentation:

- A calibrated dual channel diagnostic audiometer (Madsen Orbiter 922) with TDH-39 head phone, B-71 bone vibrator and Martin (C115) speakers were used.
- Calibrated immittance meter (GSI –Tympstar) was used to rule out any middle ear pathology.
- One five channel and fifteen channel hearing aids were used for the comparison of performance.
- Pentium IV computer with NOAH-3 software was used to program the hearing aid. Hi-Pro was used to connect the hearing aid with computer.
- Stimuli were played from Pentium IV computer 44.1 kHz sampling rate and 16 bit software.
**Test environment:** The testing was done in sound treated double room with the ambient noise level within permissible limits as recommended by ANSI (1999).

**Procedure:** Pure tone thresholds were obtained using modified Hughson and Westlake procedure (Carhart and Jerger, 1959) across octave frequency from 250 Hz to 8000 Hz for air conduction and 250 Hz to 4000 Hz for bone conduction. Tympanometry and acoustic reflex thresholds were done in GSI-Tympstar using 226 Hz probe tone.

The hearing aids were programmed on the basis of audiometric thresholds using NAL-NL1 fitting formula with the default gain provided by the software in order to avoid any unwanted effect on result. The testing was done in a sound treated double room with ambient noise level within the permissible limits (ASHA 1999). Subjects were seated at a distance of one meter and 45 azimuth from the speaker.

First the testing was done in unaided condition and later in aided condition in quiet and in noise condition. The order of hearing aid tested was randomized for each subject during the aided condition. In case of unilateral hearing loss masking noise was given to the better ear to avoid participation of that ear. The presentation level of the stimuli was kept at constant at 45 dB and the inter stimulus interval was kept at 2 sec. The subjects were asked to repeat the words presented. The words correctly repeated were scored.

**Results**

The present study was designed to investigate the effect of severity of hearing loss and age on speech identification with multi channel hearing aids. The statistical analysis includes, mixed ANOVA (two-way repeated measure ANOVA) and independent t test, which was performed using SPSS version 15.0.

1) **Effect of degree of loss on speech identification**

One of the purposes of the present study was to investigate the effect of severity of hearing loss on speech identification with multi channel hearing aids. The analysis and results are discussed separately for group I consisted of eight mild to moderate hearing loss subjects and seven moderately severe to severe hearing loss subjects and group II consisted of nine mild to moderate hearing loss subjects and seven moderately severe to severe hearing loss subjects.

a) **Effect of degree of loss on speech identification in adults:**

The mean performance with the 15 channel (A1) and 5 channel (A2) hearing aids in different listening conditions (quiet (Q) and noise (N)) for group-I is shown in the Figure 4.

It can be noted from the Figure 4 that the mean performance with the two hearing aids was different with respect to severity i.e., identification was better for mild to
moderate hearing loss subjects than moderately severe to severe hearing loss subjects. In mild to moderate hearing loss subjects, performance improved with increase in number of channels in quiet as well as in noise. Whereas, the mean performances with the two hearing aids were almost similar with moderately severe to severe hearing loss subjects or in other words the benefit from increase in number of channel was not observed with this group in quiet condition. In background noise, moderately severe to severe hearing loss subjects showed small improvement with increasing numbers of channels. It’s also been found that for all the subjects the mean performance in quiet is better than in presence of noise.

Figure 3: Comparison of effect of severity on mean identification performance with standard error, for bi-syllabic word list presented in quiet (Q) and +5dB SNR (N) for the group I (adult) with A1 (15 channel) and A2 (5 Channel) hearing aids. Bars represent the hearing aids and condition.

Mixed ANOVA (Two way repeated measures ANOVA ) was performed to assess the significant difference across channels for different degrees of hearing loss, with number of channels ,listening conditions as within group factor and severity of hearing loss as between group factor. Results revealed that there is a significant main effect of numbers of channels

\( f(1, 26) =159.4, p<0.001 \) and listening conditions \( F(1, 13) =117.9, p<0.001 \). Even though the mean scores for severity of hearing loss is different, analysis revealed no significant difference \( F(1, 13) =0.002, p=0.967 \) in the performance as a function of severity, which could be due to more variability in the data. Interaction analysis revealed number of channels interact significantly with severity \( F(1, 26) =3.511, p<0.005 \) and listening condition \( F(1, 26) =33.089, p<0.001 \). Interaction between number of channels and severity of hearing loss indicate, improvement in performance with increasing number of channels was not same between the groups. As it can be noted from the figure 4 that moderately severe to severe group showed minimal improvement with increasing number of channels. Further, it also showed significant interaction between number of channels and listening condition, which indicate that increasing numbers of channels improved the performance in presence of noise. However, there is no interaction for listening condition and severity \( F(1, 13) =117.964, p>0.05 \) and number of channel,
condition and severity ($F_{(1, 26)} = 0.561, p>0.05$). Although there is difference in performance between mild to moderate and moderately severe to severe group in both quiet and noise conditions, as observed from the figure 4, due to the large variability observed in data, statistics did not show any significant difference.

b) Effect of degree of loss on speech identification in geriatrics:

The data obtained for group II were analyzed and the mean performance with the 15 channel (A1) and 5 channel (A2) hearing aids in different listening conditions (quiet (Q), and noise (N)) is shown in the figure 5.

![Figure 4](image)

**Figure 4**: Comparison of effect of severity on mean identification performance with standard error for bi-syllabic word list presented in Quiet (Q) and +5dB SNR (N) for Group II (geriatric) with A1 (15 channel) and A2 (5 Channel) hearing aids. Bars represent the hearing aids and condition.

It can be observed from the Figure 5 that, as the severity of hearing loss increases the mean performance decreases in all the listening conditions. Further, the mean performance was improved with increase in the number of channel for mild to moderate group, whereas moderately severe to severe group showed small deterioration in performance with increasing number of channels in quiet condition. In presence of background noise both the groups showed improved performance but, moderately severe to severe group showed very small improvement when compared to mild to moderate group with increasing number of channels.

Mixed ANOVA (two way repeated measures of ANOVA) was performed to assess the significant difference across channels for different degrees of hearing loss, with number of channels (2 levels: 15 Ch and 5 Ch), listening conditions (2 levels: quiet and noise) as within group factor and degree of hearing loss (2 levels: Mild to moderate and Moderately Severe to Severe) as between group factor. Results revealed that there is a significant main effect of numbers of channels ($f_{(1, 32)} = 194.609, p<0.001$) and listening conditions ($f_{(1, 16)} = 111.533, p<0.001$). Even though the mean performance scores differ with severity of hearing loss, analysis revealed no significant difference ($F_{(1, 16)} = 3.834, p=0.068$) in the performance as a function of severity. Interaction analysis revealed
numbers channels interact significantly with severity ($F_{(1, 32)}=10.614, p<0.001$) and listening conditions ($F_{(1, 32)}=41.586, p<0.001$). i.e., the performance with increase in number of channels was not similar with different severity of hearing loss and also in quiet and in noise. As it can be read from the figure 5 that moderately severe to severe group showed decrement in performance with increasing number of channels in quiet and a small improvement in presence of noise, as opposed to better performance seen in mild to moderate hearing loss subjects. Further, it also showed significant interaction between number of channels and listening condition, which also indicate that increasing number of channels improved the performance in presence of noise. However, there is no interaction for listening condition and severity ($F_{(1, 16)}=0.0642, p=0.435$) and number of channels, listening condition and severity ($F_{(1, 32)}=0.375, p>0.690$). Although there is difference in mean performance with increase in severity in both quiet and noise conditions, due to the large variability observed in data, statistics did not show any significant difference.

2) Effect of age on speech identification:

The second purpose of the study was to investigate the effect of age on speech identification with multi channel hearing aid. As there was no significant difference ($p>0.01$) in the performance between mild to moderate and moderately severe to severe hearing loss subject in both the groups, for further analysis data was combined in each group. The effect of age on speech identification was analyzed by comparing the performance between fifteen adult and eighteen geriatric subjects irrespective of their severity.

Independent t test was carried out for fifteen adult and nineteen geriatric subjects. The analysis revealed that there is mean performance was significantly lower ($t = 2.165, p<0.05$) for geriatric group when compared to adult group with less number of channel (5 Ch) in noise condition. Even though the mean performance is comparatively better for adult, analysis revealed no significant difference in the performance with increase in number of channel (15 Ch) in quiet ($t =0.361, p = 0.720$) and in noise ($t = 1.741, p =0.092$). There is also no significant difference in the performance ($t = 1.283, p = 0.209$) with less number of channel in quiet.

![Figure 5: The effect of age on mean speech identification performance with standard error, for bi-syllabic words presented in quiet (Q) and in noise (N) for 15 subjects.](image)
channel (A1) and 5 channel (A2) hearing aids. Bars represent hearing aids and condition.

Figure 5 shows that the mean performance is slightly better for adult subjects than geriatric subjects in both quiet and noise condition. Even though, adult subjects performed only slightly better in quiet condition than geriatric, they performed better in noise condition. Performance deteriorates with increase in age, which is more evident in noise condition, indicating that geriatric subjects have difficulty hearing in noise.

**Discussion**

1) Effect of degree of loss on speech identification

Mild to moderate hearing loss subjects showed greater improvement in performance with increasing numbers of channels when compared to moderately severe to severe hearing loss subjects in quiet and in noise. Even though statistics showed no significant effect of severity, the mean performance is different. Similar results reported by number of other investigators (Degannaro, Braida and Durlach 1986; Yund et al., 1987). According to Yund and Buckles (1995), at varying signal to noise ratio (-5 to 15), they found that hearing-loss severity and multi channel compression hearing aid (8 channel) performance were related. He found that subjects with less severe impairments showed greater improvement with the multi channel hearing aid.

Exact reason is not known for lack of improvement or deterioration in performance with increasing numbers of channels for moderately severe to severe hearing loss subjects. One reason could be, these listeners were less able to resolve spectral detail and they may rely to a greater extent on temporal information such as variation in speech amplitude (Rosen et al., 1990). Further, increasing the number of channels alters the spectro-temporal properties of speech (Plomp, 1994) which can have large impact on speech perception in these participants. On the other hand mild to moderate hearing loss subjects relay on available spectral details to recognize speech, the altered temporal variation due to increase in number of channel would have produced less deleterious effect. In other words fifteen-channel hearing aid, for a mild to moderate hearing loss, causes little information degradation and can be of great benefit for speech discrimination in noise, particularly at low S/N.

One another could be, as severely impaired participants are impaired in combining the temporal information across number of channels (Narne, Manjula and Vanaja, 2007; Souza and Boike, 2006), increasing number of channels would not provide any extra information, further it might deteriorate temporal information (Plomp, 1994) for these participants, as a result they did not show any improvement. Souza and Boika, (2006) reported that ability to combine the temporal information in speech across number of channels, was more significantly impaired because of degree of hearing loss than that of age. This could be one reason that similar pattern of performance observed between adults and geriatrics.
It should also be noted that, the performance in quiet is significantly better than the performance in noise with both the hearing aid. Although in quiet condition both the hearing aid performance is almost similar or with mean performance is little greater for 15 channel hearing aid, in noise condition the performance with 15 channel hearing aid is greater compared to 5 channel hearing aid. Theoretically, the greater the number of channels and narrower the channels the greater the likely hood that important frequency components of the signal will fall into channels which do not include higher intensity components of the noise or of the signal itself. So amplification will increase the signal level greater than the noise, which intern increases the signal to noise ratio in situations where the background noise is dominant in restricted range of frequencies (Dillon 2001). This study is in agreement with the previous studies (Yund and Buckles 1995 a, 1995 b), that with 8-16 channels in noise condition the performance with increased number of channel is beneficial.

2) Effect of age on speech identification:

The overall mean performance shows that the adult performed better than geriatric listeners did. The mean performance of geriatric group was poorer in noise than adult subjects. However, there is no significant difference in the performance with age except with 5-channel hearing aid in noisy condition. It has been demonstrated that older listeners have more difficulty understanding speech than younger listeners (Gordon-Salant and Fitzgibbons, 1997) and mean identification score decreases with increase in age (Souza and Virginia, 2001). This could because they have reduced temporal resolution (Souza, 2000; Souza and Boika, 2006) when compared to adults.

Although, the performance is lower than adults, geriatrics also demonstrated similar pattern of performance with increasing number of channels. Geriatrics participants were as good as adult subjects in utilizing and combining the temporal information across channels. In connection, Souza and Boika (2006) reported that the older listeners performed poorly than younger listeners in identifying nonsense syllables, but did not have difficulty combining temporal envelope cues across channel. The poor performance with 5-channel hearing aid in presence of noise in geriatric subjects can be attributed to the general age deficits, which occur more often in complex listening situations such as in noise. Whereas, the mean performance with 15 channel hearing aid is slightly better for adult in noise condition, but there is no significant difference in the performance between the age group. With increase in number of channel, signal-to-noise ratio improves (Yund and Buckles, 1995 a, b) which can be attributed to the improved performance with the 15 channel hearing aid.

To conclude, effect of increasing numbers of channels on speech identification majorly depends upon the degree of hearing loss than the age. That is in other words, benefit from the increase in the number of channel does not improve with severity, in speech identification in quiet and in presence of noise. The multi channel hearing aid at least up to 15-channel will not cause any detrimental effect for mild to moderate hearing
loss subjects of younger the age group, in addition they improve the perception in both groups in quiet and in presence of noise.

**Conclusions**

The present study investigated the effect of degree of loss and age on speech identification in quiet and in noise with multi channel hearing aids. Group I with fifteen adult subjects and Group II with eighteen geriatric subjects with gradual sloping mild to severe sensory neural hearing loss served as a subject. They were again sub grouped into mild to moderate hearing loss and moderately severe to severe hearing loss based on their pure tone average in each group. The stimulus consisted of recorded version of bi-syllabic wordlist developed by Vijayalakshmi and Yathiraj (2005). All the words in a list were mixed individually with speech babble (Anitha and Manjula, 2005) at +5 dB SNR, based on RMS level by the program written in MATLAB 6.5 software. Two hearing aids, one with five channels and other with fifteen channels digital hearing aids were programmed on the basis of audiometric thresholds using NAL-NL1 fitting formula with the default gain provided by the software. The investigation was carried out in a sound treated double room to determine effect of degree of loss and age on speech identification with multi channel hearing aid. The stimulus was presented at 45 dB HL in the free field condition and the subjects were asked repeat the words heard. The words correctly repeated were scored. It can be concluded from the present study that, the benefit from the increase in number of channel is inversely proportional to the degree of hearing loss. Channels upto fifteen channel would not cause any deleterious effect in performance in mild to moderate hearing loss subjects. The overall performance decreases with increasing age, but the geriatric subjects can combine the temporal information across channel with increase in number of channels. The degree of loss found to decrease the performance with increase in number of channel rather than age. It’s also found that with increase in number of channel the performance improved in noise.

**References**


Cross Language Perception of Voicing in Children With Hearing Impairment

Sandeep Ranjan & Asha Yathiraj*

Abstract

The present study was designed to evaluate the discrimination of stop consonant voicing in children with hearing impairment who were exposed to different regional languages (Kannada and Malayalam). Eighteen children with hearing impairment (10 Kannada speaking children and 8 Malayalam speaking children) with moderate to moderately severe degree of sensorineural hearing loss were evaluated. The results indicated that perception of voicing was significantly better for the Kannada speaking children with hearing impairment compared to the Malayalam speaking children with hearing impairment. This was observed irrespective of the place of articulation where the stop occurred (bilabial, dental, retroflex and palatal), the vowel along with which it was produced (a, i, and u) and the position (initial and medial).

Introduction

The voicing of consonants has been found to be an important distinction in speech communication, as indicated by statistics of phonological contrasts (Carterette and Jones, 1974). Children with hearing impairment have been noted to have difficulty in both discrimination and production of the voiced-voiceless speech sound distinction (Bennett & Ling, 1973). They often produce voiced consonants that are voiceless, and vice versa (Calvert, 1961; Preston, Yeni-Komshian and Stark, 1967). This confusion has been noted to frequently occur when a voiceless sound is surrounded by strongly voiced sounds (Levitt, 1971). This has been considered to be related to the difficulty children with hearing impairment have in discriminating the presence or absence of voicing cues.

Generally it has been observed that most of the listeners with hearing impairment use the same voiced and voiceless cues as used by the normal hearing listeners, although some of them may be unable to make full use of these cues (Pickett and Revoile, 1983). Listeners with hearing impairment were found to be less sensitive to the vowel-onset transition cues than the normal hearing listeners (Lisker and Abramson, 1972). The other cues used by them to differentiate voice-voiceless contrasts have been noted to different to what is used by normal hearing individuals.

Children with moderate-to-severe sensorineural hearing impairment are known to identify the voicing characteristics of naturally produced stop consonants accurately (Byers, 1973; Erber, 1972). These experiments establish that children with hearing impairment can readily identify signals which differ greatly in the acoustic cues that signal voicing.

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A common outcome of experiments on the identification of stop consonants by children with moderate and severe hearing losses is that the children make relatively few errors when identifying the voicing feature of the consonants, but make relatively more errors when identifying the place feature (Byers, 1973; Erber, 1972).

Studies carried out in India have reported that there exist differences in voicing contrasts in various Indian languages. Malayalam is a semi syllabic phoneme-rich language of the Dravidian language family spoken in the southern state of Kerala. Kannada belongs to the same family spoken in another southern state of Karnataka. In Malayalam and Kannada, voiced plosives are characterized by lead VOT and unvoiced plosives by lag VOT. Voiced plosives in Malayalam have longer lead VOT and shorter lag VOT compared to those in Kannada (Savithri, Sreedevi and Santosh, 2001). All these can be attributed to the differences in the phonological structure of each of these languages.

It has been reported by Ramakrishna, Nair, Chiplukar, Ramachandran and Subramanian (1962) that Malayalam has 21 stop consonants, of which 10 are grouped into five pairs each with voiced and voiceless counterparts. The alveolar stop has no voiced counterpart. It has only a voiced allophone which occurs after homorganic nasal. In Malayalam, voiced plosives have been found to be characterized by lag VOT. Voiced plosives in Malayalam have longer lead VOT and shorter lag VOT. However, no significant difference between the transition duration of voiced and unvoiced plosives have been reported. VOT has been found to be used more than transitions to contrast voicing in word-initial position in Malayalam (Savithri et al., 2001).

Studies of phonemes occurrence in conversation have indicated that the voicing feature is a very important phonological distinction (Mines, Hanson and Shoup, 1978). There is a need to evaluate the perception of voicing of stops in children with hearing impairment exposed to different regional languages. Voicing is produced differently in different languages. In Malayalam, voiceless stops have been observed to be produced with a trailing voicing in the medial position (Geethakumary, 2002). However, the voicing distinction is perceived when produced by a native as well as non-native speaker of the language. There is a need to know whether children with hearing impairment, with different exposure to voicing, are able to perceive them similar to their normal hearing counterparts. This study will help in knowing how voicing features of stops are perceived by Malayalam and Kannada speaking children with hearing impairment.

Besides knowing how children from each of the above language groups perceive voicing contrasts, there is also a need to compare the way they differ in the perception the contrasts. Such information will be helpful in developing materials to teach voicing features in children with hearing impairment, which in turn would help them to perceive the contrasts.

The aim of the present study was to see if stop voicing can be perceived equally well by two groups of children with hearing impairment, belonging to two different
languages groups. Thus, the study aimed at evaluating the voicing perception in children with hearing impairment exposed either to Kannada or Malayalam. The study also aimed at studying the effect of place of articulation, vowel environment and position of stop, on voicing perception in these two groups.

**Method**

Voice-voiceless discrimination in four places of articulation (bilabial, dental, retroflex and palatal), in two positions (initial and medial) and with three vowel environments (\(\text{a} \), \(\text{i} \), \(\text{u} \)) were evaluated. This was done on children with hearing impairment, exposed either to Kannada or Malayalam.

**Participants:**

Ten Kannada and eight Malayalam speaking children in the age range of 6 years to 13 years with moderate to severe degree of sensorineural hearing loss were studied. They all used BTE hearing aids for at least 4 years, and their aided thresholds were within the speech spectrum at least up to 2 kHz. It was confirmed that they had not undergone prior training specifically to learn voicing contrasts. They were able to communicate in sentences and had adequate speech and language skills. They had average intellectual capacity and no neurological problems or additional physical disabilities.

**Instrumentation**

A Pentium IV computer with Adobe Audition 2.0 software was used for the presentation of the stimuli. A calibrated two channel diagnostic audiometer, Orbiter 922 was used for the selection of participant and for the presentation of the stimuli. An immittance meter GSI Tympstar provided information regarding the absence of any middle ear problem. A Pentium IV computer was used for presenting the speech stimuli for the discrimination activity.

**Test environment:**

All the tests were carried out in an air conditioned sound treated two-roomed condition. The ambient noise levels were within permissible limits (ANSI S3.1, 1991). The recording of the material for the study was also done in a sound treated room.

**Procedure:**

**Material development:**

Nonsense consonants-vowel (CV) and consonant-vowel-consonant (CVC) syllables were used as test stimuli. Stops \(\text{\l, \r, \l\text{th}, \l\text{i}, \l\text{b}, \l\text{d}, \l\text{dh}, \l\text{g}}\) which are common in Kannada and Malayalam were selected. Each of these consonants was combined with the vowels \(\text{\a, \i, \u} \). Thus, a total of 72 stimuli were recorded. The stimuli were randomized to form two sets. The consonants were used in the initial and
medial positions. The final position was not used since consonants are not used in this position in Kannada.

The speech stimuli were recorded by a male speaker whose mother tongue was Kannada. A Kannada talker was used since native speakers of Malayalam tend to produce voiceless stops in the medial position with a trailing voicing (Geethakumary, 2002). The recording was done on a Pentium IV computer using the Adobe Audition Software at a sampling frequency of 16 kHz. A unidirectional microphone was used for the recording, and it was kept at a distance of 10 cm from the speaker’s mouth. The recorded material was normalized using the Adobe Audition software so that all the speech stimuli were of same intensity. A 1 kHz calibrated tone was recorded prior to the speech stimuli. An interstimulus interval of 3 sec was maintained for obtaining the responses from the participants. The recorded material was written on a CD. The recorded material was heard by ten native adult listeners of Kannada to get a goodness rating of the recorded material. The stimuli were considered acceptable only if 90% of these adults, having normal hearing, were able to identify the stimuli correctly.

**Procedures for participant’s selection:**

A preliminary pure-tone audiometry was done to determine the hearing threshold of the participants using a calibrated Madsen OB-922 diagnostics clinical audiometer. Air conduction thresholds were obtained between 250 Hz and 8 kHz and bone-conduction threshold were obtained between 250 Hz and 4 kHz. Screening tympanometry and reflex threshold testing were done using a calibrated GSI-TS impedance audiometer to rule out the presence of any middle ear pathology.

**Procedure for speech discrimination testing:**

The developed materials were presented using a Pentium IV computer. The output from the computer was routed to the tape input of the audiometer (OB-922). Prior to the presentation of the stimuli, a 1 kHz calibrated tone was presented to set the VU meter deflection of audiometer to ‘0’. Participants heard the stimuli through a TDH-39 headphone. The stimuli were presented in pairs and the child had to indicate whether the pairs were same or different. This was done by asking each child to point to pairs of drawings, one pair having similar pictures and one pair having dissimilar pictures. Prior to the actual testing, each child was trained using practice items. Instructions were given to the children orally in their respective language. It was ensured that the children followed the instruction and pointed to the correct picture-pair, on hearing the stimuli. Following the practice trial, the children were tested using the 72 test items. All stimuli were randomized to avoid any order effect. For children who showed any sign of fatigue, breaks were given. Social as well as token reinforcement were given for correct responses. A random schedule of reinforcement was used. Test-retest reliability was obtained on all of the Malayalam speaking children and 60% of the Kannada speaking children. The retest was done using a different list containing the same test materials that
were randomized. The test stimuli were administered to the better ear of each child at the most comfortable level (MCL) at 30 dB SL.

The responses of the children were noted by the experimenter. A correct response was given a score of ‘1’ and a wrong response was given a score of ‘0’. The data thus collected were subjected to statistical analyses using the Statistical Packages for Social Sciences (SPSS) software version 15.

Results and Discussion

The data obtained from the eighteen children with hearing impairment (10 Kannada speaking & 8 Malayalam speaking) were analyzed using descriptive statistics, mixed ANOVA, repeated measure ANOVA and independent ‘t’ test. Details of results are further discussed.

Comparison of voicing within and between language groups:

The mean and standard deviation for score obtained for each language group was computed. This information is provided in Figure 1. These scores represented the responses obtained to all 72 stimuli by each language group.

From Figure 1 it is evident that the Kannada speaking children with hearing impairment could correctly discriminate the 72 stimuli 81% of the time. However, the matched Malayalam speaking children with hearing impairment could discriminate the same tokens only 65% of the time. Though the Kannada speaking children obtained a higher mean value, the variability in their score was also higher.

In order to check whether the discrimination scores differed significantly between the two language groups, an independent ‘t’ test was administered. The results revealed that the scores between the two language groups were significantly different with respect to the total scores \[ t (16) = 3.899, p < 0.001 \]. The perception of voicing was significantly better for the Kannada speaking children with hearing impairment compared to the Malayalam speaking children with hearing impairment.

![Figure 1: Mean total raw score for the Kannada and Malayalam speaking groups](image-url)
Effect of place of articulation on discrimination of voicing

Discrimination of voicing as a function of place of articulation was measured separately for the Kannada and Malayalam speaking children with hearing impairment. This was done to see whether the ability to discriminate voicing was influenced by the place of articulation in each of the language groups. The mean scores for voicing as a measure for place of articulation was more for the Kannada speaking group. The mean and SD obtained for the two groups is shown in Table 1.

Table 1: Mean and standard deviation of voicing as a measure for place of articulation

<table>
<thead>
<tr>
<th>Place of articulation</th>
<th>Kannada</th>
<th>Malayalam</th>
<th>Possible maximum score</th>
</tr>
</thead>
<tbody>
<tr>
<td></td>
<td>Mean#</td>
<td>SD</td>
<td>Mean#</td>
</tr>
<tr>
<td>p-b\</td>
<td>14.10</td>
<td>2.69</td>
<td>11.13</td>
</tr>
<tr>
<td>t-d\</td>
<td>15.10</td>
<td>1.67</td>
<td>11.63</td>
</tr>
<tr>
<td>k-g\</td>
<td>15.10</td>
<td>1.92</td>
<td>12.00</td>
</tr>
<tr>
<td>th-dh\</td>
<td>13.60</td>
<td>2.46</td>
<td>11.88</td>
</tr>
</tbody>
</table>

# Maximum scores = 18

In order to determine whether there was a significant difference within and between the two groups, ANOVA was done. The ANOVA (4 place of articulation x 2 groups) revealed there was no significant main effect for the different place of articulation [F (3, 48) = 1.738, p > 0.05] and also there was no significant interaction effect found between place of articulation and language [F (3, 48) = 1.183, p > 0.05]. However, it was once again observed that a significant difference existed between the two language groups, Kannada and Malayalam [F (1, 16) = 15.199, p< 0.01].

To check the influence of place of articulation on the discrimination of voicing in each language groups, repeated measure ANOVA was performed. The results revealed that there was no significant difference between the place of articulation in the Kannada speaking children with hearing impairment [F (3, 27) = 3.172, p> 0.05]. Likewise, there was no significant difference between place of articulation in the Malayalam speaking children with hearing impairment [F (3, 21) =0.448, p> 0.05]. Thus, the place of articulation did not influence the way the two groups perceive voiced \ voiceless stops.

The effect of place of articulation on the discrimination of voicing between the Kannada and Malayalam speaking children was determined using a mixed ANOVA. The mixed ANOVA brought to light that there did exist a significant main effect of place of articulation between the two language groups [F (1, 16) = 15.199, p< 0.01]). To get a better understanding of the effect of place of articulation on voicing discrimination between the two language groups, separate independent ‘t’ tests were carried out. The results of the ‘t’ tests revealed a significant difference between the two language groups.
regarding the way they perceive voicing of bilabials (p < 0.05), palatals (p < 0.01) and velars (p < 0.01). However, there was no significant difference between the two groups regarding the way they perceived voicing of dental stops (p > 0.05). This is depicted in Table 2 and Figure 2.

Table 2: Significance of difference between mean voicing perception scores between the two language groups, for each place of articulation.

<table>
<thead>
<tr>
<th>Stimuli</th>
<th>Language groups</th>
<th>Mean scores #</th>
<th>‘t’ value</th>
</tr>
</thead>
<tbody>
<tr>
<td>\p-b\</td>
<td>Kannada</td>
<td>14.10</td>
<td>2.88*</td>
</tr>
<tr>
<td></td>
<td>Malayalam</td>
<td>11.13</td>
<td></td>
</tr>
<tr>
<td>\th-dh\</td>
<td>Kannada</td>
<td>13.60</td>
<td>1.77</td>
</tr>
<tr>
<td></td>
<td>Malayalam</td>
<td>11.88</td>
<td></td>
</tr>
<tr>
<td>\t-d\</td>
<td>Kannada</td>
<td>15.10</td>
<td>4.20**</td>
</tr>
<tr>
<td></td>
<td>Malayalam</td>
<td>11.63</td>
<td></td>
</tr>
<tr>
<td>\k-g\</td>
<td>Kannada</td>
<td>15.10</td>
<td>3.35**</td>
</tr>
<tr>
<td></td>
<td>Malayalam</td>
<td>12.00</td>
<td></td>
</tr>
</tbody>
</table>

*Significant at 0.05 level. **Significant at 0.01 level.

The results also revealed that the mean scores for all the four place of articulation were higher for the Kannada speaking children with hearing impairment compared to the Malayalam speaking children with hearing impairment. Though the difference was not significantly higher for the dentals, the Kannada group continued to perform better than the Malayalam group. The performance of the Kannada group was slightly lower for this place of articulation compared to their perception of other places of articulation. However, the Malayalam groups performed similarly for all four places of articulation, resulting in no significance of difference for the \th-dh\ contrast.
Figure 2: Mean voicing perception scores between the language groups at four places of articulation

**Effect of vowels on the perception of voicing**

Voicing perception as a function of vowel environments (\(a\), \(i\), \(u\)) was measured for both Kannada as well as Malayalam speaking children with hearing impairment. From Table 3 it is evident that irrespective of the vowel environment, the scores were better for the Kannada speaking children with hearing impairment compared to the Malayalam speaking children with hearing impairment. However, the SD for the two groups was approximately the same.

Table 3: Mean and standard deviation of voicing perception in different vowel environments

<table>
<thead>
<tr>
<th>Vowel environment</th>
<th>Language</th>
<th></th>
<th></th>
<th></th>
<th></th>
</tr>
</thead>
<tbody>
<tr>
<td></td>
<td>kannada</td>
<td>mean</td>
<td>SD</td>
<td>malayalam</td>
<td>mean</td>
</tr>
<tr>
<td>(a)</td>
<td>19.50</td>
<td>2.80</td>
<td>16.00</td>
<td>2.45</td>
<td></td>
</tr>
<tr>
<td>(i)</td>
<td>19.20</td>
<td>2.58</td>
<td>14.50</td>
<td>1.78</td>
<td></td>
</tr>
<tr>
<td>(u)</td>
<td>19.20</td>
<td>2.58</td>
<td>16.13</td>
<td>2.11</td>
<td></td>
</tr>
</tbody>
</table>

# Maximum scores = 24

Mixed ANOVA (Repeated measure of ANOVA with language as independent factor) was carried out to check the significance differences of mean perception of voicing in different vowel contexts. The 3 vowel environment X 2 group ANOVA revealed that there was no significant difference in voicing perception with different vowel combination [F (2, 32) = 1.623, p > 0.05)]. Also, there was no significant interaction effect found between different vowel combination and language [F (2, 32) = 1.170, p > 0.05), but the results revealed a significant difference between the two languages, Kannada and Malayalam [F (1, 16) = 15.199, p < 0.01)].
Separate one-way ANOVAs were performed on the Kannada speaking group and the Malayalam speaking group to determine the influence of vowels within each language group. These revealed that there was no significant difference between different vowel combination in the Kannada speaking children with hearing impairment \([F (2, 18) = 0.310, p > 0.05]\). Similarly, there was no significant difference between voicing perception with different vowel combination in the Malayalam speaking children with hearing impairment \([F (2, 14) = 1.329, p > 0.05]\). From this it can be inferred that voicing perception was not influenced by the vowel context in which it occurred.

The influence of vowels on voicing perception between the Kannada and Malayalam speaking children was evaluated using a mixed ANOVA. A significant effect of the vowel context on voicing perception between the two language groups was obtained \([F (1, 16) = 15.199, p < 0.01]\). To determine the effect of specific vowels across the two groups, separate independent sample ‘t’ tests were carried out. The results from the ‘t’ test revealed a significant difference in the way the two groups perceived voicing in all three vowel environments (Table 4).

Table 4: Significance of difference between mean voicing perception scores between the two language groups, for each vowel environment

<table>
<thead>
<tr>
<th>Types of vowel environment</th>
<th>Language groups</th>
<th>Mean scores#</th>
<th>‘t’ value</th>
</tr>
</thead>
<tbody>
<tr>
<td>(\text{\textbackslash u})</td>
<td>Kannada</td>
<td>19.50</td>
<td>2.783*</td>
</tr>
<tr>
<td></td>
<td>Malayalam</td>
<td>16.00</td>
<td></td>
</tr>
<tr>
<td>(\text{i})</td>
<td>Kannada</td>
<td>19.20</td>
<td>4.388***</td>
</tr>
<tr>
<td></td>
<td>Malayalam</td>
<td>14.50</td>
<td></td>
</tr>
<tr>
<td>(\text{\textbackslash a})</td>
<td>Kannada</td>
<td>19.20</td>
<td>2.726*</td>
</tr>
<tr>
<td></td>
<td>Malayalam</td>
<td>16.13</td>
<td></td>
</tr>
</tbody>
</table>

#Maximum scores = 24  
*Significant at 0.05 level of significance  
***Significant at 0.001 level of significance

The results also revealed that the mean scores for all the three vowel environment were higher for the Kannada speaking children with hearing impairment compared to the Malayalam speaking children with hearing impairment (Figure 3).

Figure 3: Mean voicing perception scores between the language groups at three vowel environment
**Effect of position of stop on voicing perception within and between language groups**

Perception of voicing as a function of position of stop consonants (initial and medial position) was assessed separately for the Kannada and Malayalam speaking children with hearing impairment. The data was analyzed to see whether the position of stop consonant effected the voicing perception in each of the language groups. Within each language group, it can be noted from Table 5 that the mean values were very similar for stops occurring in the initial and medial position. However, the mean scores as well as SD values were higher for the Kannada group than for the Malayalam group.

A paired sample ‘t’ test was performed to assess the effect of initial and medial position of stop consonants on voicing perception within each language group. The analysis revealed that there was no significant difference (p > 0.05) between the scores obtained in the two positions in both language groups.

Table 5: Significance of difference between in scores obtained in the initial and medial position within each language group.

<table>
<thead>
<tr>
<th>Groups</th>
<th>Position</th>
<th>Mean</th>
<th>SD</th>
<th>‘t’ value</th>
</tr>
</thead>
<tbody>
<tr>
<td>Kannada</td>
<td>Initial</td>
<td>29.0</td>
<td>4.4</td>
<td>0.63</td>
</tr>
<tr>
<td></td>
<td>Medial</td>
<td>29.6</td>
<td>3.9</td>
<td></td>
</tr>
<tr>
<td>Malayalam</td>
<td>Initial</td>
<td>23.0</td>
<td>1.7</td>
<td>0.07</td>
</tr>
<tr>
<td></td>
<td>Medial</td>
<td>23.06</td>
<td>2.8</td>
<td></td>
</tr>
</tbody>
</table>

*Maximum score = 36

A paired sample ‘t’ test was also performed to assess the effect of position of stop consonants on voicing perception between the Kannada and Malayalam groups. The results of paired sample ‘t’ test revealed there was a highly significant difference in the mean scores for each position between the language groups (Table 6). It can be noted from the table that the Kannada speaking children had higher scores in both the initial and medial position when compared to the Malayalam speaking children. Although the scores were higher for Kannada speaking children, they had more variability. This variability was higher in the initial position than in the medial position.

Table 6: Significance of difference between scores obtained in the initial and medial position between groups.

<table>
<thead>
<tr>
<th>Position</th>
<th>Group</th>
<th>Mean</th>
<th>SD</th>
<th>‘t’ value</th>
</tr>
</thead>
<tbody>
<tr>
<td>Kannada</td>
<td>Initial</td>
<td>29.0</td>
<td>4.4</td>
<td>5.1***</td>
</tr>
<tr>
<td>Malayalam</td>
<td></td>
<td>23.0</td>
<td>1.7</td>
<td></td>
</tr>
<tr>
<td>Kannada</td>
<td>Medial</td>
<td>29.6</td>
<td>3.9</td>
<td>5.95***</td>
</tr>
<tr>
<td>Malayalam</td>
<td></td>
<td>23.06</td>
<td>2.8</td>
<td></td>
</tr>
</tbody>
</table>

***p < 0.001  #Maximum scores = 36
The test retest reliability was checked by administering the coefficient alpha test. On the Kannada group, it was found that the co-efficiency was 0.96 while it was 0.90 in the Malayalam group. This brought a light that test retest reliability was high and that children belonging to both language groups gave reliable responses.

From the findings of the current study it is evident that children with hearing impairment from the two language groups perceive voicing differently, which was found to be statistically significant. This significance of difference was present irrespective of the place of articulation in which it occurred, the vowel combination with which it occurred and the position. The difference in perception of two language groups can be attributed to the difference in the way, voicing is produced in the two languages.

It has been reported by Geethakumary (2002) that voiced stops in the medial position in Malayalam are produced with a gradual trailing in voicing after homorganic nasals (nasals having the same place of articulation). However, she reported that voiced stops were noted to occur in the initial and medial position. In Kannada, no such reduction in voicing has been reported to occur in the presence of homorganic nasals. Though voiced stops do occur in Malayalam, it is possible that the perception of the Malayalam speaking children in the present study were highly influenced by the trailing in voicing that normally occurs in their language. Though no homorganic nasals were used in the present study, probably these children were inclined to not perceive the voice-voiceless distinction.

Though Geethakumary (2002) reported of a trailing of voicing in the medial position, she did not report of any such trailing in the initial position in Malayalam. Hence, it was anticipated that there would be a difference in the way voicing would be perceived in the initial and medial position in the Malayalam group, but not in the Kannada group. However, no such difference was observed. Both language groups showed no significant difference in the way they perceived stop voicing in the initial and medial position. Thus, it can be construed that the trailing voicing in the medial position
alone could not account for the difference in perception of voicing in the two language groups.

Further, it has been reported by Ramakrishna et al. (1962) that the voiceless sounds occur a lot more frequently in Malayalam when compared to the voiced speech sounds. The ratio of voiceless to voiced stops varied from 1:22 to 1:4, where \( \text{th} \) and \( \text{dh} \) had the highest ratio and \( \text{dh} \) and \( \text{dh} \) had the lowest ratio. However, in Kannada, the voiced and voiceless contrasts were reported to occur almost equally. Thus, the children exposed to Kannada were stimulated almost equally with voiced and voiceless contrasts, unlike the children exposed to Malayalam, who had unequal exposure to these contrasts. This could have influence their perception of the two groups, resulting in the significant difference observed in the present study.

**Conclusion**

The results of the study revealed that there existed a significant difference between the way voicing of stops was perceived by Kannada speaking and Malayalam speaking children with hearing impairment. The significant difference was observed at all four places of articulation that were studied (bilabial, dental, retroflex and palatal), vowel environment (\( \text{a} \), \( \text{i} \), \( \text{u} \)) and position (initial and medial). However, no significant difference in voicing perception occurred across the four places of articulation, vowel environment or position within each language group. This difference in perception between the two participant groups can be attributed to the difference in the way voicing is produced in the two languages.

**Acknowledgements**

We would like to thank the Director, All India Institute of Speech and Hearing, Mysore for permitting us to carry out this study.

**References**


Benefits of Binaural Amplification in Participants with Asymmetric Hearing Impairment

Sandhya S. Shekar & P. Manjula*

Abstract

Binaural hearing involves the integration of signals from each of the two ears into a single hearing sensation. The advantages of loudness summation, localization and discrimination enhancement especially in noise are a result of this binaural hearing process. However, when the signals to each ear are disproportionate with each other in an individual with hearing impairment, binaural integration process is less effective and binaural advantages are correspondingly diminished or even lost (Davis & Haggard, 1982). The present study was undertaken to assess the benefits of binaural amplification over monaural amplification in individuals with sensorineural hearing loss with various degrees of asymmetry. Twenty-two individuals with post-lingually acquired bilateral sensorineural hearing loss served as participants in this study. The participants were assigned to one of the three groups based on the degree of hearing loss in the ears. The most comfortable levels (MCL), speech recognition scores (SRS) and speech recognition threshold in noise (SRT) were established for each participant in the three-aided conditions—amplification to the better ear, to the poorer ear and to both ears. It was observed that lowest MCL was obtained in the binaural amplification condition compared to the monaural amplification condition in all three groups of participants. For speech recognition in quiet, it was found that providing binaural amplification did not result in significant improvement over monaural amplification for participants with bilateral symmetric hearing loss. In participants with asymmetric hearing loss (Groups II & III), providing binaural amplification resulted in improved speech recognition performance compared to monaural amplification either to better or poorer ear. It was observed that the participants obtained higher speech recognition threshold in noise (SNR) with binaural amplification than with monaural amplification condition.

Key words: Most comfortable levels (MCL), speech recognition scores (SRS), speech recognition threshold in noise (SNR)

Introduction

The loss of hearing ability characterized by decreased sensitivity to sound in comparison to normal hearing is termed as hearing loss (Silman and Silverman, 1991). Hearing loss is measured by the amount of loss in terms of decibels (dB) hearing level (HL). The magnitude of hearing loss can be equal (symmetric) in both the ears or unequal (asymmetric). Symmetric hearing loss refers to a difference of less than 15 dB in the pure tone average and less than 8% difference in the speech recognition scores between ears (Markides, 1977). Asymmetric hearing loss implies a difference of greater than 15 dB between the two ears regardless of the magnitude of hearing loss (Valente, 1994).

Asymmetric hearing impairment can be defined as interaural differences in threshold sensitivity. A working definition of asymmetric hearing loss, relative to the application of hearing aids, would be that an asymmetrical hearing loss implies a

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significant difference between ears. Regardless of the magnitude of asymmetry, use of amplification must improve hearing performance so that it can be verified by objective and subjective evaluations. Such differences can be expressed in terms of pure tone threshold, most comfortable listening levels, word recognition scores, loudness growth compensation, and positive response to amplified sound in everyday listening environments (Valente, 1994).

Individuals with asymmetric losses have been the participants of many studies in which the researchers have concluded that, because binaural redundancy advantage tends to decrease as the average threshold differences of the two ears increase, these individuals may not benefit as much from a binaural fitting. However, there are certainly other physical acoustic factors (e.g., head shadow effect) and auditory processing factors (e.g., squelch) that can contribute to audition and speech understanding in noise.

Studies have also shown that the loudness levels depend on the degree of hearing loss. Speech recognition abilities are adversely affected in individuals having sensorineural hearing impairment. The amount of degradation in the speech recognition abilities in quiet and in noise condition depends upon the degree of hearing impairment in these individuals (Sandlin, 2000). Many studies have reported a lack of amplification in adults with asymmetric sensorineural hearing impairment that leads to auditory deprivation (Arlinger, Gatehouse & Bentler, 1996).

Thus, it is evident that speech recognition abilities in an individual with hearing impairment might depend upon the amplification condition. However, there is a dearth in literature regarding the comparison of aided MCL, speech recognition scores and speech recognition threshold in noise in different amplification conditions in individuals with symmetrical and asymmetrical hearing impairment.

Need for the Study

It is evident that there are equivocal reports on the efficacy of binaural amplification in individuals with asymmetric hearing loss. Also, there is a dearth of studies in literature regarding the effect of degree of asymmetry on monaural and/or binaural amplification. Hence, the present study was undertaken to assess the benefits of binaural amplification over monaural amplification in individuals with sensorineural hearing loss with various degrees of asymmetry.

Objectives of the Study

The following objectives were evaluated in participants with bilateral sensorineural hearing loss of different degrees of asymmetry:

1. To compare the Most Comfortable loudness Level (MCL) with monaural and binaural amplification conditions.
2. To compare the Speech Recognition Scores (SRS) with monaural and binaural amplification conditions.
3. To compare the Speech Recognition in Threshold in Noise, in terms of signal to noise ratio (SNR), with monaural and binaural amplification conditions.
4. To assess the effect of asymmetry in aided monaural Most Comfortable Levels (MCL) between the two ears on Speech Recognition Scores in binaural amplification condition.
5. To assess the effect of asymmetry in aided monaural Most Comfortable Levels (MCL) between the two ears on Speech Recognition in Threshold in Noise in terms of signal to noise ratio (SNR) in binaural amplification condition.

Method

To evaluate the effect of different amplification conditions on MCL, SRS and SNR, the following procedures were administered.

Participants

Twenty-two individuals with post-lingually acquired bilateral sensorineural hearing loss served as participants in this study. The age range of the participants varied from 18 to 60 years (mean age = 48.6 years). All participants were naïve hearing aid users. All participants had Kannada as their native language. The participants had mild to severe degree of hearing loss with flat audiometric configuration, a slope of ≤ 5 dB rise or fall per octave (Lloyd & Kaplan, 1978). The speech recognition score was ≥ 60% in the both ears with the difference between the ears ranging from 5% to 40%.

The participants were assigned to one of the following three groups based on the degree of hearing loss in the ears.

1. Group I with participants having symmetrical sensorineural hearing loss (S-SN): Eight participants were included in this group. The inclusion criterion for participants in this group was that the difference between the pure tone average (PTA for 500 Hz, 1000 Hz, and 2000 Hz) of the right and left ears was within 15 dB.

2. Group II with participants having a lesser extent of asymmetrical sensorineural hearing loss (A-SN I): Seven participants were included in this group. The inclusion criterion for participants in this group was that the difference in the PTA of the right and left ears was between 16 dB and 25 dB.

3. Group III with participants having a greater extent asymmetrical sensorineural hearing loss (A-SN II): Seven participants were included in this group. The inclusion criterion for participants in this group was that the difference in the PTA of the right and left ears was between 26 dB and 35 dB.
Instrumentation

A calibrated two channel diagnostic audiometer with sound field-testing facility was used. A computer connected to the auxiliary input of the audiometer to administer the speech recognition tasks. A HiPro interface unit, personal computer (PC) with NOAH-3 and hearing aid programming softwares were also used. In addition, digital behind-the-ear (BTE) hearing aids with a fitting range from mild to severe degree of hearing loss was used. The hearing aids had single channel, single band and were programmable.

Speech Material

A bi-syllabic phonemically balanced word lists in Kannada (Yathiraj & Vijayalakshmi, 2006) was used. The test material was recorded in a female voice on a CD. It consisted of eight lists with 25 words each. In addition, a word list in Kannada for measurement of SNR (Sahgal, 2005). The list was recorded in male voice on a CD. The word list consisted 40 sets of words. Each set had three words with a combination of low-mid, low-high and high-mid frequency speech sounds. A passage in Kannada (Sairam, 2002) was used. The passage was recorded in a male voice with normal effort on a CD.

Procedure

The testing was carried out in a sound treated double-room set-up with the ambient noise levels within permissible limits. The MCL, SRS and speech recognition threshold in noise, in terms of SNR, were established for each participant in the three aided conditions. The three aided conditions were amplification to the better ear, amplification to the poorer ear and binaural amplification. In case of symmetric hearing loss, since both ears had similar audiometric thresholds, monaural amplification to the individual ears (right and left) and binaural amplification formed the three aided conditions.

The speech material was played through a computer connected to auxiliary input of the audiometer. Before the presentation of the stimuli, the level of presentation was monitored with a calibration tone. During the presentation of the stimuli also, it was ensured that the mean deflection of VU-meter of the audiometer was about 0 dB. For the speech recognition tasks participants were instructed to repeat the speech stimuli heard. For speech recognition tasks and establishment of most comfortable levels, the speech stimuli were presented through a loudspeaker located at 0° Azimuth at a distance of one meter in front of the participant. The speaker Azimuth and distance remained the same for all the three tasks. The speech and noise were routed through the same speaker. To evaluate the objectives of the study, the data were collected in the following two phases.

Phase 1

In the first phase of the study, hearing aid fitting and establishment of most comfortable levels were carried out.
1. **Hearing Aid Fitting**

   Each participant was fitted with a single channel programmable digital behind-the-ear (BTE) hearing aid in each ear. The hearing aids were programmed using a PC and a HiPro interface unit using the NAOH and the hearing aid software. For each participant the hearing aid was programmed according to the ‘first fit’ using the generic NAL-NL1 formula. The right and the left hearing aids were programmed separately and binaural balancing was done. The establishment of Most Comfortable Level (MCL) and speech recognition tasks was carried out with the programmed hearing aid, which the participant wore with an appropriately sized standard ear tip during the test.

2. **Aided Most Comfortable Level (MCL)**

   For each participant the MCLs were established for each ear separately in the aided condition only. The MCLs were established in three aided conditions, two monaural (right ear aided and left ear aided) and one binaural condition. In each aided condition, the participant was instructed to rate the loudness of a recorded Kannada passage being presented based on the seven-point rating scale (Cox, 1995) – very soft, soft, comfortable but slightly soft, comfortable, comfortable but slightly loud, loud but okay and uncomfortably loud. The recorded passage was presented in sound filed condition. The initial presentation level (PL) of the passage was 10 dB SL (re: aided speech reception threshold). The level of the recorded passage was increased in 2 dB steps if the participant judged the loudness to be below comfortable level and decreased in 4 dB steps if loudness was judged to be above comfortable level. The monaural MCL was noted down for each ear separately. In participants with asymmetrical hearing loss, the non-test better ear was masked to avoid its participation in the monaural testing of the poorer ear by providing broadband noise at 70 dB SPL through the insert earphone. The binaural MCLs were also established with hearing aids to both ears using a similar procedure with the initial presentation level being 10 dB SL (re: aided speech reception threshold of the better ear).

**Phase II**

   In the second phase of the study, two speech recognition tasks were administered, one in quiet and the other in the presence of noise:

   1. Aided speech recognition scores (SRS)
   2. Aided speech recognition threshold in noise in terms of Signal to noise ratio (SNR).

1. **Aided Speech Recognition Scores**

   The speech recognition score (SRS) gives an indication of the ability of the individual to discriminate different speech sounds (Moore, 1998). In the present study, aided SRS were measured using recorded (phonemically balanced) speech material in Kannada (Yathiraj and Vijayalakshmi, 2005) in the sound field condition. The presentation level (PL) of speech stimuli was fixed at 35 dB HL if the hearing loss in
either one or both ears was of mild degree and the level was set to 40 dB HL otherwise. The right and left ear of each participant was aided with appropriately programmed digital BTE hearing aids. The SRSs were measured in each of the above mentioned amplification conditions.

The aided SRS in each of the above mentioned aided conditions were measured by presenting one complete word-list of 25 words for each condition. The participants were instructed to repeat the words being presented. If the participant correctly repeated the word, then a score of ‘1’ was given, and if not, a score of ‘0’ was given. The total number of words correctly repeated in the list was noted for each condition. This was considered as the speech recognition score of the participant for the respective aided condition. Therefore, each participant had three SR scores, one for each aided condition.

2. Aided Speech Recognition Threshold in Noise (SNR)

One of the advantages of binaural hearing aids is that it improves speech perception in the presence of noise. For the aided Speech Recognition Threshold (SRT) in Noise, the signal to noise ratio (SNR) associated with 50% recognition performance was measured.

For the purpose of the study, signal to noise ratio (SNR) is defined as the difference between the intensity of recorded speech material and the intensity of the competing speech-shaped noise in dB when the individual correctly repeats two or more than two words in a set of three words being presented in the presence of competing speech babble.

The SNR was measured in a sound-field condition using the recorded Kannada word list developed by Sahgal (2005). The speech material and speech shaped noise were routed through the same speaker. The presentation level of the word list was fixed at 44 dB HL and the initial level of the speech noise was set at 16 dB below the speech signal and varied systematically to measure the SNR. The participant was instructed to repeat the words heard in presence of competing speech shaped noise. The participant was presented a set of 3 words at each level of noise. If the participant correctly repeated at least 2 words out of 3, then the level of noise was increased by 4 dB and if the participant failed to repeat at least 2 words, the level of noise was decreased in 2 dB steps till the participant repeated at least 2 out of 3 words. Further, the level of noise was increased in 1 dB steps till the participant repeated at least 2 out of 3 words. At this point, the difference between the intensity of speech and competing speech-shaped noise in dB was considered as the SNR.

The SNR was measured in all the three aided conditions using the above-described procedure. Therefore, each participant had three SNR values, one in each aided condition. The MCL, SRS and speech recognition threshold in noise (SNR) in the three aided conditions were obtained for each participant and tabulated for statistical analysis.
Results and Discussion

The results of the study are discussed in terms of MCL, SRS and SNR for three groups of participants (Group I, Group II and Group III) in three different amplification conditions (amplification to the better ear, amplification to the poorer ear, binaural amplification). The data obtained were subjected to statistical analysis using Statistical Package for Social Sciences (SPSS, version 14) software.

To know if there was a significant main effect of amplification on MCL, SRS and SNR in each of the three groups of participants repeated measures analysis of variance (ANOVA) was carried out. If a main effect was present, Bonferroni post-hoc analysis was carried out to know if a significant difference between the scores of the three groups in three amplification conditions was present.

I. Aided most comfortable loudness level (MCL)

Individuals with hearing impairment have altered most comfortable loudness levels compared to normal hearing individuals (Dillon, 2001).

A. Comparison of MCL in three groups of participants in three amplification conditions

The mean values revealed that MCL in binaural amplification condition (MCLbin) was lowest compared to MCL with amplification to the better ear alone (MCLb) which in turn was better than amplification to the poorer ear alone condition (MCLp) (Figure 1) in all three groups of participants.

![Figure 1: Mean and standard deviation (SD) of the MCL with A. binaural amplification (MCLbin), B. amplification to better ear (MCLb) and C. amplification to poorer ear (MCLp) in three groups of participants.](image)

The lowest MCL in binaural amplification condition could be attributed to binaural summation of loudness. The results of the present study are in consensus with the earlier studies (Verhey, Anweiler and Hohmann, 2006).
Group I (Symmetrical Hearing loss)

Repeated measures ANOVA revealed a significant main effect of the amplification conditions on MCL \([F (2, 14) = 26.27; p< 0.001]\), indicating that the mean MCLs in the three amplification conditions were significantly different from each other. Bonferroni post-hoc pair-wise analysis revealed a significant difference between MCLb and MCLbin \((p<0.01)\) and between MCLp and MCLbin \((p<0.01)\). In these pair-wise comparisons, the MCLbin was the lowest. However, there was no significant difference in the MCL between MCLb and MCLp \((p>0.05)\). This can be attributed to almost similar thresholds in both ears, as the participants in this group had symmetrical hearing loss.

Group II (Asymmetrical Hearing loss I)

Repeated measures ANOVA which revealed a significant main effect of the amplification conditions on MCL \([F (2, 12) = 128.86; p< 0.001]\). The mean MCLs in the three amplification conditions were significantly different from each other. Bonferroni post-hoc pair-wise analysis revealed a significant difference between MCLb and MCLbin \((p<0.05)\), between MCLp and MCLbin \((p<0.001)\) and also between MCLb and MCLp \((p<0.001)\).

Group III (Asymmetrical Hearing loss II)

A significant main effect of the amplification conditions on MCL was revealed by repeated measures ANOVA \([F (2, 12) = 64.81; p< 0.001]\) indicating that mean MCLs in the three amplification conditions were significantly different from each other. Bonferroni post-hoc pair-wise analysis revealed a significant difference between MCLb and MCLbin \((p<0.001)\), between MCLp and MCLbin \((p<0.001)\) and also between MCLb and MCLp \((p<0.01)\). Thus, the results for Group II and Group III were similar but were different from Group I. This indicates that in participants in Group II and Group III, binaural amplification yielded an MCL that is much lower than MCLb, thus increasing the range of comfortable loudness. The performance in the binaural condition was significantly better than either amplification to the better ear or amplification to the poorer ear; in terms of lowered MCL.

B. Comparison of MCL across the three groups of participants in binaural amplification condition.

In the literature, it has been documented that most comfortable loudness levels are affected by the amplification condition and the degree of asymmetry of hearing loss in participants with sensorineural hearing impairment (Ricketts, 2000). Table 1 summarizes the mean and SD of the MCL in binaural amplification condition (MCLbin) in the three groups of participants.
Table 1: Mean and SD of the MCL in binaural amplification condition (MCLbin) in the three groups of participants

<table>
<thead>
<tr>
<th>Groups</th>
<th>MCLbin Mean (dB HL)</th>
<th>SD</th>
</tr>
</thead>
<tbody>
<tr>
<td>I</td>
<td>54.0</td>
<td>6.5</td>
</tr>
<tr>
<td>II</td>
<td>48.85</td>
<td>1.06</td>
</tr>
<tr>
<td>III</td>
<td>48.28</td>
<td>3.14</td>
</tr>
</tbody>
</table>

One-way ANOVA revealed no significant main effect of degree of asymmetry of hearing loss on the MCLbin \([F (2, 19) =3.96; p>0.05]\). The results indicate that the MCL in binaural amplification condition did not vary as a function of degree of asymmetry of hearing loss between the two ears. Also, the results of the present study are in consensus with the earlier reports that the type of amplification affects the most comfortable loudness levels in individuals with hearing impairment (Sandlin, 2000; Dillon, 2001).

C. Correlation of degree of asymmetry of hearing loss and MCL

Studies have reported that the most comfortable loudness level depends upon the degree of hearing loss in individuals with sensorineural hearing impairment (Summers & Cord, 2007). Spearman’s correlation analysis revealed no significant correlation between the degree of asymmetry of hearing loss and MCLbin in Group I, II and III (\(\rho = 0.35; p>0.05\) for Group I, \(\rho = 0.00; p>0.05\) for Group II and \(\rho = 0.69; p>0.05\) for Group III). Therefore, binaural amplification gives equal benefit irrespective of the degree of hearing loss in both ears.

II. Speech recognition scores (SRS)

The present study analyzed the speech recognition abilities in individual with hearing impairment in three different amplification conditions.

A. Comparison of speech recognition scores in three groups of participants in three amplification conditions

The mean values of SRS indicate that performance in binaural amplification condition was better when compared to amplification to better ear condition, which in turn was better than amplification to poorer ear condition, in all the three groups of participants (Figure 2). The mean difference between SRSbin and SRSb or SRSp is highest in Group III followed by Group II and then by Group I.
Figure 2: Mean and SD of SRS A. amplification to better ear condition (SRSb), B. amplification to poorer ear condition (SRSp) and C. binaural amplification condition (SRSbin) in three groups of participants.

**Group I (Symmetrical Hearing loss)**

Repeated measures ANOVA revealed no significant main effect of the amplification conditions on SRS \[ F (2, 14) = 2.38; p> 0.05 \], that is, SRS in the three amplification conditions were not significantly different from each other. Results of the present study indicate that providing binaural amplification for participants with bilateral symmetric hearing loss does not result in significantly improved speech recognition performance since the amplification depends on the auditory thresholds of either poorer ear or better ear. There are several studies that have investigated the benefits of amplification in individuals with various degrees of hearing loss (Ching, Dillon and Byrne, 1998).

**Group II (Asymmetrical Hearing loss I)**

Repeated measures ANOVA revealed a significant main effect of the amplification conditions on SRS \[ F (2, 12) = 90.25; p< 0.001 \], indicating a significant difference between the SRS in the three amplification conditions. The Bonferroni post-hoc analysis revealed a significant difference between SRSb and SRSbin (p<0.001), between SRSp and SRSbin (p<0.001), between SRSb and SRSp (p<0.05). Further, from figure 2 it can be noted that the SRS is best in the binaural amplification condition followed by SRSb and then followed by SRSp. Hence, it can be inferred that binaural amplification is significantly beneficial compared to monaural amplification, either to better ear or poorer. The results of the present study are in consensus with the earlier reports (Gelfand, Silman and Ross, 1987).

**Group III (Asymmetrical Hearing loss II)**

Repeated measures ANOVA revealed a significant main effect of the amplification conditions on the SRS \[ F (2, 12) = 126.78; p< 0.001 \]. This indicated a significant difference between the SRS in the three amplification conditions. The Bonferroni post-hoc analysis revealed a significant difference in the SRS between amplification to better ear and binaural amplification condition (p<0.001),
between amplification to poorer ear and binaural amplification condition (p<0.001),
between the amplification to better ear and amplification poorer ear amplification
condition (p<0.001).

Hence, binaural amplification results in better speech recognition abilities
compared to monaural amplification, either to better ear or poorer ear only condition
which is in agreement with that reported by Persson, Harder, Arlinger and Magnuson
(2001) which had concluded that individuals achieved significantly better speech
recognition scores in the binaural amplification condition compared to monaural
conditions.

The results of the present study contradict that reported by Rothpletz, Tharpe and
Grantham (2004). In their study, the effect of different degrees of degradation of speech
signal on speech recognition task was investigated. There was a significant binaural
advantage (average of 7 dB) when listening to symmetrically degraded signals as
compared to when listening monaurally. Further, little or no binaural benefit was
reported, on an average, when listening to asymmetrically degraded signals. Also, the
overall performance of the adults was significantly worse when listening to binaural
asymmetrically degraded signals than when listening to monaural signals, thus
demonstrating evidence of binaural interference. However, the study considered
asymmetrical degradation of signals and did not consider participants having
asymmetrical hearing loss. The effect of asymmetrical degradation may be different from
that of asymmetrical hearing loss. This might have contributed to the differences in the
results observed in their study and the present study.

The present study indicated that providing binaural amplification resulted in
improved speech recognition performance compared to amplification to better ear only
and amplification to poorer ear only in participants with asymmetry in pure tone
thresholds across the two ears ranging from 16 to 35 dB.

B. Comparison of speech recognition scores across the three groups of participants
in binaural amplification condition.

The present study analyzed the SRS in binaural amplification condition across all
three groups of participants. Results indicated that mean SRSbin was better in Group III
when compared to Group II, which in turn was better than Group I participants (Table 2).

Table 2: Mean and SD of SRSbin in the three groups of participants

<table>
<thead>
<tr>
<th>Group</th>
<th>SRSbin</th>
</tr>
</thead>
<tbody>
<tr>
<td></td>
<td>Mean (dB HL)</td>
</tr>
<tr>
<td>I</td>
<td>21.87</td>
</tr>
<tr>
<td>II</td>
<td>22.85</td>
</tr>
<tr>
<td>III</td>
<td>23.14</td>
</tr>
</tbody>
</table>
One-way ANOVA revealed no significant effect of the degree of asymmetry on the speech recognition scores \( [F (2, 19) =1.21; p>0.05] \), suggesting that participants in each group performed similarly in quiet with binaural amplification.

In the present study, participants having symmetrical and asymmetrical hearing impairment were included. The maximum degree of asymmetry between the ears of these participants included in the study was 35 dB. It may be possible that the degree of asymmetry did not have an effect on the speech recognition abilities in binaural amplification condition since the audibility provided from the hearing aids was sufficient enough to understand speech stimuli. Previous investigators have also reported that audibility may be a major factor in speech recognition abilities in individuals with sensorineural hearing impairment (Hogan & Turner, 1998).

Hence, from the present study it can be inferred that the degree of asymmetry of hearing loss upto 35 dB between the two ears might not have an effect on SRS in binaural amplification condition. In other words, binaural amplification results in similar speech recognition performance in participants with symmetric as well asymmetric hearing impairment.

C. Correlation of degree of asymmetry of hearing loss and SRS

Spearman’s correlation analysis revealed no significant correlation between the degree of asymmetry of hearing loss and SRSbin \((\rho = 0.28; p>0.05 \text{ for Group I, } \rho = 0.38; p>0.05 \text{ for Group II and } \rho = 0.20; p>0.05 \text{ for Group III})\). This implies that, SRSbin does not vary with the degree of asymmetry between the ears in individuals with bilateral hearing loss.

D. Correlation of asymmetry in aided monaural MCL between the two ears and SRS in binaural amplification condition.

Spearman’s correlation analysis revealed a no significant correlation between the difference in aided monaural MCL of the two ears and SRS in binaural amplification condition for Group I and Group III \((\rho = 0.16; p>0.05 \text{ for Group I and } \rho = 0.32; p>0.05 \text{ for Group III})\) and a significant correlation for Group II \((\rho = 0.77; p<0.05 \text{ for Group II})\). Thus, SRSbin does not vary with the difference between aided monaural MCL of the two ears in individuals with bilateral hearing loss, that is, increase in the difference between aided monaural MCL of the two ears does not lower the SRS.

III. Speech Recognition Threshold in Noise (SNR)

Previous studies have shown that individuals with sensorineural hearing loss have reduced speech recognition abilities in quiet condition. However, the problem becomes more complicated when speech signal is presented in background noise. Hence, individuals with sensorineural hearing loss may exhibit more problems of speech recognition in noise condition. The present study analyzed the effect of amplification conditions on the speech recognition threshold in noise in terms of SNR.
A. Comparison of SNR in three groups of participants in three amplification conditions

It can be observed from Figure 3 that a lower SNR was obtained in binaural amplification condition followed by amplification to better ear condition and then by amplification to the poorer condition in all three groups of participants. Lower SNR values indicate that the participants performed well even when difference between speech and noise was very lesser.

![Figure 3: Mean and SD of SNR in A. binaural amplification condition (SNRbin), B. amplification to better ear condition (SNRb) and C. amplification to poorer ear condition (SNRp) in three groups of participants.](image)

**Group I (Symmetrical Hearing loss)**

Repeated measures ANOVA indicated a significant difference between the SRS in the three amplification conditions \(F\) (2, 14) = 10.44; \(p< 0.01\). Bonferroni post-hoc pair-wise analysis revealed a significant difference between SNRb and SNRbin \((p<0.01)\) and between SNRp and SNRbin \((p<0.05)\). However, there was no significant difference between SNRp and SNRb \((p>0.05)\). Therefore, in individuals with bilateral symmetrical hearing loss the improvement of performance in noise with binaural amplification over monaural amplification did not reach statistical level of significance.

The results of the present study are in consensus with the earlier reports. The present study revealed that the signal to noise ratio was least in the binaural amplification condition compared to monaural amplification either to the better ear or to the poorer ear. These findings suggest that individuals with varying degrees of asymmetrical hearing loss (up to 35 dB of asymmetry) might still be able to take advantage of the binaural squelch phenomenon and hence are prospective candidates for binaural amplification.

**Group II (Asymmetrical Hearing loss I)**

Repeated measures ANOVA revealed a significant main effect of the amplification conditions on SNR \(F\) (2, 12) = 63.00; \(p< 0.001\) indicating a significant difference between the SRS in the three amplification conditions. The results of Bonferroni post-hoc pair-wise comparison revealed a significant difference between SNRb and SNRbin \((p<0.05)\), between SNRp and SNRbin \((p<0.001)\) and between SNRb
and SNRp (p<0.01). This indicated that binaural amplification was better than either ear amplification alone and amplification to the better ear was better than amplification to the poorer ear alone.

**Group III (Asymmetrical Hearing loss II)**

Repeated measures ANOVA revealed a significant main effect of the amplification conditions on SNR \(F(2, 12) = 67.08; p< 0.001\) indicating a significant difference between the SRS in the three amplification conditions. The results of Bonferroni pair-wise comparison revealed a significant difference between SNRb and SNRbin (p<0.05), between SNRp and SNRbin (p<0.001) and SNRb and SNRp (p<0.05). Therefore, binaural amplification was better than either ear amplification alone and amplification to the better ear was better than amplification to the poorer ear alone.

It has been reported in the previous studies that the degree of asymmetry might have an on the speech recognition abilities in the presence of noise (Summers & Cord, 2007). Their results also suggested that amplification condition might affect subjective performance in noise and overall for listeners with varying degrees of mild to severe hearing loss when feedback was eliminated. However, in the present study, subjective preference was not considered. Inclusion of subjective preference would have provided additional information.

On the other hand, studies have reported the advantages of monaural amplification over binaural amplification condition in participants with hearing impairment. Carter, Noe, and Wilson (2001) evaluated four individuals who preferred monaural as compared with binaural amplification. For these individuals, the results of sound field testing using a speech in multitalker babble paradigm indicated that when listening in noise, there was a little difference between aided and unaided word-recognition performance, suggesting that binaural hearing aids originally fit for each individual were not providing substantial benefit when listening in a competing babble background.

Thus, the results of the present study are in consensus with the earlier reports which have inferred that, the speech recognition abilities in the presence of noise depends upon the amplification conditions, and that binaural amplification is better than monaural amplification conditions.

**B. Comparison of SNR across the three groups of participants in binaural amplification condition**

The present study analyzed the speech recognition threshold in noise in terms of SNR in binaural amplification condition across all three groups of participants (Table 3). One-way ANOVA revealed no significant main effect of degree of asymmetry on speech recognition abilities in all participants \(F(2, 19) =10.19; p>0.05\). Thus, the performance of the participants in all the three groups was comparable on speech recognition in noise task with binaural amplification.
The speech recognition in noise does not depend on the degree of asymmetry in individuals with hearing impairment. The results of the present study are in consensus with that reported in earlier study regarding effect of degree of asymmetry on speech recognition abilities in noise (Posner and Ventry, 1977).

C. Correlation of degree of asymmetry of hearing loss and SNR

Spearman’s correlation analysis revealed no significant correlation between the degree of asymmetry of hearing loss and SNR in binaural amplification condition ($\rho = 0.32; p>0.05$ for Group I, $\rho = 0.34; p>0.05$ for Group II and $\rho = 0.20; p>0.05$ for Group III). Thus, the speech recognition threshold in noise does not depend on and thus does not vary with the degree of asymmetry between the two ears.

D. Correlation of asymmetry in aided monaural MCL between the two ears and SNR in binaural amplification condition

For the analysis of the correlation between asymmetry of aided monaural MCL between the two ears and SNR in binaural amplification condition Spearman’s correlation analysis was used. It revealed no significant correlation between the difference in aided monaural MCL of the two ears and signal-to-noise ratio in binaural amplification condition ($\rho = 0.00; p>0.05$ for Group I, $\rho = 0.75; p>0.05$ for Group II and $\rho = 0.32; p>0.05$ for Group III).

The results indicate that SNR does not vary with the difference in aided monaural MCL of the two ears in individuals with bilateral asymmetric hearing loss, that is, increase in the difference in aided monaural MCL of the two ears does not increase the SNR. This indicates that asymmetry of hearing loss does not have an effect on the speech recognition in noise presented under the binaural amplification condition.

Conclusions

The study evaluated the effect of degree of asymmetry between the two ears on the Most Comfortable Loudness levels (MCL), Speech Recognition Scores (SRS) and Speech Recognition Threshold in Noise in terms of signal to noise ratio (SNR). The important findings on the three parameters studied are as follows:

Table 3: Mean and SD of SNRbin in three groups of participants

<table>
<thead>
<tr>
<th>Group</th>
<th>SNRbin</th>
<th>Mean (dB)</th>
<th>SD</th>
</tr>
</thead>
<tbody>
<tr>
<td>I</td>
<td></td>
<td>1.50</td>
<td>2.32</td>
</tr>
<tr>
<td>II</td>
<td></td>
<td>-0.28</td>
<td>2.43</td>
</tr>
<tr>
<td>III</td>
<td></td>
<td>2.00</td>
<td>1.10</td>
</tr>
</tbody>
</table>

The speech recognition in noise does not depend on the degree of asymmetry in individuals with hearing impairment. The results of the present study are in consensus with that reported in earlier study regarding effect of degree of asymmetry on speech recognition abilities in noise (Posner and Ventry, 1977).
1. Aided Most Comfortable Loudness Level (MCL)

- Lowest MCL was obtained in the binaural amplification condition compared to the monaural amplification condition in all three groups of participants. This could be attributed to the binaural summation of loudness, in all the participants irrespective of the asymmetry between the two ears.
- For participants in Group II and Group III, amplification to both the ears yielded an MCL that was significantly lower than the MCL with amplification to the better ear alone (p<0.05), thus increasing the range of comfortable loudness.
- The results indicated that the degree of asymmetry (in the pure tone average between the two ears up to 35 dB between the two ears) did not influence the benefit in terms of MCL from binaural amplification, in participants with sensorineural hearing impairment.

2. Speech Recognition Scores (SRS):

- For speech recognition in quiet, it was found that providing binaural amplification did not result in significant improvement over monaural amplification for participants with bilateral symmetric hearing loss.
- In participants with asymmetric hearing loss (Groups II & III), providing binaural amplification resulted in improved speech recognition performance compared to monaural amplification either to better or poorer ear.
- Results indicated that SRS in binaural amplification condition did not vary with the difference between aided monaural MCL of the two ears in individuals with bilateral asymmetric hearing loss, that is, increase in the difference between aided monaural MCL of the two ears did not lower the SRS.

3. Speech recognition threshold in noise, in terms of, signal-to-noise ratio (SNR):

- The present study indicated that participants obtained higher speech recognition threshold in noise (SNR) with binaural amplification than monaural amplification to either better ear or poorer ear. These findings suggested that individuals with varying degrees of asymmetrical hearing loss might still be able to take advantage of the binaural squelch phenomenon with binaural amplification.
- Participants obtained higher speech recognition threshold in noise (SNR) with binaural amplification than with monaural amplification condition.

These findings imply that individuals with varying degrees of asymmetrical hearing loss, up to 35 dB, can be considered candidates for binaural amplification.

Acknowledgements

We thank the Director, All India Institute of Speech and Hearing, Mysore for allowing us to carry out the study. We extend our thanks to the Head of the Department of Audiology for permitting to use the instruments required for the study.
References


Yathiraj, A. & Vijayalakshmi, C.S. (2006). PB word list in Kannada developed at Department of Audiology, AIISH.
Dichotic Rhyme Test in Kannada: A Normative Data on Adults

Sangamesh C. & K. Rajalakshmi*

Abstract

Dichotic speech tests proved to have high sensitivity in assessing binaural integration tasks that often noticed in individuals having CAPD. Research focusing on Dichotic Rhyme test has been relatively scanty. The present study aimed to develop one such test and to collect normative data on kannada speaking normal hearing individuals. Stimulus was developed using 18 pair of cvcv rhyming words that differ only in initial consonants. These stimuli were made similar in total duration and imposed on to stereo tracks and aligned in such a way that there was no onset delay between the two. Normative data was established on young normal hearing kannada speaking adults. Analysis of results revealed that there existed a significant right ear advantage for the dichotic stimuli. Females had greater mean ear correct scores as compared to males for both right and left ears. Females had greater mean double correct scores as compared to males. The double correct scores were found to be lower when compared to the ear correct scores.

Introduction

The concept of dichotic listening was first introduced by Broadbent in 1954. Dichotic listening occurs when different auditory stimuli are presented to each ear simultaneously. It has been used historically to assess hemispheric dominance as well as hemispheric asymmetries (Kimura, 1961a, 1961b, 1967; Zattore 1989), with diminished scores on these types of listening tasks suggesting auditory and/or cognitive dysfunction or pathology (Kimura, 1961a, 1961b).

Dichotic listening tasks have been used in the evaluation of both normal and disordered auditory processes at the cortical level (Kimura, 1961; Berlin et al. 1972). The term ‘dichotic’ refers to the simultaneous competing presentation of two different speech signals to opposite ears. Subjects are asked to repeat back what is heard in one or both ears. Generally when speech is presented dichotically to normal listeners, higher scores are obtained from the material to the right ear, than the left. This has been referred to as right ear advantage and is believed to reflect the dominance of left hemisphere for speech and language perception (Studdert-Kennedy, Shankweiler, 1970).

The early work by Kimura has been the foundation for the widely accepted theory that in man, the contralateral (or crossed) auditory pathway has more neural connections than the ipsilateral pathway and is considered the dominant pathway. On dichotic listening task individuals will generally show an ear advantage in the ear contralateral to the hemisphere dominant for language. For most individuals this will result in an right ear advantage (REA), which is believed to be the result of left hemisphere dominance for language and the auditory perception of speech stimuli (Kimura, 1967). Objective evidence for this hypothesis has come from studies of dichotic listening in subjects with surgical sectioning of the corpus callosum. Milner et al. (1968) and Sparks and

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Geschwind (1968) demonstrated complete left-ear suppression of dichotically presented stimuli following surgical sectioning of the corpus callosum.

In a series of experiments, Musiek reveals that sectioning of the posterior portion of corpus callosum, but not the anterior portion results in a suppression of left-ear scores, where as right ear performance remains at preoperative levels (Musiek, Kibbe and Baran, 1984; Musiek, et al., 1985; Musiek and Reeves, 1990; Baran, Musiek and Reeves, 1990).

Dichotic speech tasks differ from each other in terms of the stimuli utilized as well as the task required for the listener. Stimuli used in dichotic tests range from digits and nonsense syllables to complete sentences. Depending on the test itself, listeners may require to repeat everything that is heard (binaural integration) or to direct their attention to one ear and repeat what is heard in that ear only (binaural separation). Dichotic stimuli may be viewed on a continuum from least to most difficult. In general most similar and closely aligned the stimuli presented to the ears are the more difficult dichotic task will be (Bellis, 2002).

One such test using most commonly spoken words is Dichotic Rhyme Task (DRT). The dichotic rhyme task (DRT) was first introduced by Wexler and Halwes (1983) and then later modified by Musiek, et al. (1989). The DRT uses temporally aligned consonant-vowel-consonant pairs that vary only in their initial consonants. Although subjects are presented two words (one word to each ear), the precise alignment of the words, as well as the fact that the final vowel-consonant elements in each pair of words are identical, result in the subjects perceiving only one word the vast majority of the time. As a result of these test features, normal right-handed subjects tend to demonstrate test scores that are slightly greater than 50% in the right ear and slightly less than 50% in the left ear (Musiek, et al., 1989). This unique pattern of performance is presumed to be the result of some type of dichotic “fusion” of the signals, which occur low within the central auditory nervous system.

The rationale behind DRT has come from series of experiments carried out by Repp (1976). Fusion in the dichotic listening condition takes place when words with similar spectral shape (waveform envelop) are presented to the listener (Repp, 1976). The waveform envelop for words is generally determined by the low frequency energy (Perrot & Berry, 1969), which is essentially its fundamental frequency (Repp, 1976, 1977a). Therefore if two words are presented dichotically, which have similar spectral envelopes and are temporally aligned, they will fuse and will be heard as one word (Repp, 1977a). The words in DRT for the most part, are words that are perfectly or partially fused. Due to this fusion this test is also called as Fused Dichotic Words Test (FDWT).

Musiek, Kurdzielschwan, Kibbe, Gollegly, Baran and Rintelmann (1989) reported normative values of 30% - 73% for right ear and 27% - 60% for left ear in a group of 115 normal hearing subjects. Bellis (2006) normative data indicated no significant effect of
age or ear on the Dichotic Rhyme test. Normative values (2 standard deviations above and below the mean) were 32% - 60% per ear.

On dichotic tasks, speech signals are preferred to non-speech signals as they can be manipulated in more complex ways than tones or other non-speech stimuli (Berlin 1973). Speech signals that are linguistically similar and spectrally time aligned short and of similar duration are preferred to other types of speech stimuli in CANS evaluation due to their greater lesion detection capacity (Speaks, 1974).

The present study was taken to generate normative data regarding the performance of young Indian adults on a dichotic rhyme test in Kannada.

**Method**

The present study was carried out with an aim of developing the dichotic rhyme test and also to generate the normative data. The test was developed in Kannada language.

The study was carried out in two phases.

- **Phase I** - Development of test material
- **Phase II** - Establishing the normative data for the developed test material.

**Phase I: Procedure for developing test material**

**Construction of test material**

18 pairs (36 members) of Kannada rhyming words consisting of /p,th,k,b,dh,g/ in the initial position and which has a syllable structure of CVCV was taken from a standard Kannada dictionary. Members of each pair differed from each other only in the initial consonant and the members of the pair differed only on one phonetic feature (either voicing or place of articulation).

**Familiarity test**

These 36 words were given to 10 adult native speakers of Kannada (5 males and 5 females) to rate on a 5 point scale, with following rates:

- 0 – Very unfamiliar (Not heard)
- 1- Unfamiliar (Heard but not commonly used)
- 2- Quite familiar (Less commonly used)
- 3- Familiar (Commonly used)
- 4- Very familiar (Most commonly used)

The rating score of two or more was set as the criteria for inclusion in the test material. All of the words had a rating of greater than or equal to 2. So, all the 36 words were considered as familiar and were taken for the construction of test material.
Recording of test stimulus

An adult native speaker of the language was asked to produce each of these 36 words 3 times in a continuous manner and the words were recorded using “PRAAT” software with a sampling frequency of 22050 Hz and digitization of 16 bits. For the test material, the middle word of the 3 continuous words was considered to get a flat frequency spectrum. These words were analyzed using Adobe Audition 1.0 computer software.

Construction of Stimulus

The final portions of the members of each pair were made identical using cross-splicing. (i.e, the initial, distinctive portion of the one member of each pair was cross spliced onto the final, non-distinctive portion of the other member, making the final portion of the members of each pair identical).

E.g., in /pennu/ - /bennu/, the portion of /ennu/ in either /pennu/ or /bennu/ was selected and positioned in both the words, thus the portion /ennu/ was same in both the words.

After cross splicing, the total duration of rhyming words were made equal by reducing the voicing bars or by reducing the steady state portion of the vowel, of the initial CV portion of the word. Cross splicing was done to reduce the intrinsic variability among the final syllables in a rhyming pair.

Using Adobe Audition 1.0 Software, the two members of each Rhyming pair were recorded on stereo tracks with 0 millisecond delay between each member of the pair. The word pairs were 10 seconds apart on stereo tracks.

Stimuli were placed on a stereo track such that one member of the pair was routed to one ear and the other member of the pair was routed to the other ear. These 18 rhyming pairs (randomly) along with initial calibration tones were recorded onto the Compact Disk. These 18 rhyming pairs were randomly chosen again and words in each pair were counterbalanced (i.e, in the first 18 pairs if “Pennu” was presented to the right ear and “bennu” was presented to the left ear, then in the second 18 pairs the channel designations were reversed).

Thus, the list consisted of a total of 36 pairs of rhyming words.

Phase II: Establishing Normative data

Subjects:

- 50 young normal hearing adults (25 males and 25 females) with Kannada as mother tongue were taken as subjects.
- The age ranges of the subjects were 18 to 30 years.
Subject selection criteria

The subjects selected for the study had
- No history of Hearing loss
- No Chronic otological problems
- No neurological problems or Trauma to the brain
- No previous experience with dichotic listening tasks
- Right-handedness
- Pure-tone thresholds less than 15dB in both ears for both air conduction and bone conduction measurements
- Speech identification scores of 80% or greater

Instrumentation

- A calibrated two channel (ANSI S3.6-1996) diagnostic audiometer Madsen Orbiter 922 with TDH-39 headphones housed in MX-41 AR ear cushions and B71 bone vibrator was used to check the hearing threshold in all the participants.
- GSI Tympstar tympanometer was used to evaluate the status of the middle ear in all the participants.
- A CD player was used to play the compact disc. The signal from the CD player was fed to the tape input of the audiometer.

Test Environment

The testing was carried out in a well lit air conditioned sound treated double room and ambient noise levels were within 35 dB SPL (ANSI 1999).

Test procedure

1. Puretone thresholds were obtained at octave intervals between 250 Hz to 8000 Hz for air conduction and between 250 Hz to 4000 Hz for bone conduction.
2. Immittance audiometry was carried out with a probe tone frequency of 226 Hz. Ipsilateral and contralateral acoustic reflexes thresholds were measured for 500 Hz, 1000 Hz, 2000 Hz, and 4000 Hz.
3. Subjects who passed the subject selection criteria were administered the dichotic rhyme test. The VU meter was adjusted to the 1 kHz calibration tone. The 36 pairs of dichotic stimuli were presented at an intensity level of 60 dB HL. Subjects were instructed to respond on an open set answer sheet (APPENDIX-B). The task involved writing down the rhyming words heard after each presentation. All subjects were encouraged to guess when unsure of the word or words.

Scoring

The subject responses were scored in terms of
Single correct scores: Total number of correct responses for the right ear or the total number of correct responses for the left ear.

Double correct scores: Scores obtained when subject correctly responded both the stimuli presented to the two ears.

Ear correct scores: To get total ear correct scores, the double correct score was added to single correct score of respective ear and were used for analysis.

Statistical analysis

The raw data was subjected to statistical analysis where the mean, the range and standard deviation were calculated. ‘Repeated measure of ANOVA for ears with independent factor as gender’ was used to evaluate the main effect and the interaction between gender and ear. Independent and paired t-test was also used to reveal the significant difference between genders on ear correct scores and between ear correct scores with-in gender. Further details on results are discussed under results and discussion chapter.

Results and Discussion

The aim of present study was to develop Dichotic Rhyme Task in Kannada and to have normative data for the developed test. To have normative values, data collected on 25 male subjects and 25 female subjects in the age of 18 to 30 years was used. The data was subjected to statistical analysis using the software program SPSS version 10.0.

The following statistical analyses were done:

- Repeated measures of ANOVA to see the main effect and the interaction between gender and ear
- Paired sample ‘t’ test was done to see the significant difference between right and left ears. And also to see the significant difference between single correct and double correct scores.
- Independent ‘t” test was done to see the significant difference between genders on ear correct scores and double correct scores.

The results were analyzed by calculating the mean, standard deviation and the range. Analysis was done to obtain information on:

(i) Ear correct scores: Total number of correct responses for the right ear or the left ear plus the double correct scores.
(ii) Double correct scores: Scores obtained when subject correctly responded both the stimuli presented to the two ears.

I. Comparison of ear correct scores with-in gender

The mean and the standard deviations for male and females were calculated separately. As it can be seen form the table, the mean scores for the right ear were better
than the left ear scores for both males and females. Repeated measures of ANOVA revealed a significant main effect for the ears \([F(1, 48) = 34.560, p < 0.001]\) but it did not show the interaction effect for the ear and gender \([F(1, 48) = 1.840, p > 0.05]\).

Table 1: The mean values, standard deviation, the range and the t-scores along with the level of significance for the ear correct scores

<table>
<thead>
<tr>
<th>Gender</th>
<th>Mean</th>
<th>SD</th>
<th>t</th>
<th>Sig. (2-tailed)</th>
</tr>
</thead>
<tbody>
<tr>
<td>Females</td>
<td></td>
<td></td>
<td></td>
<td></td>
</tr>
<tr>
<td>Right</td>
<td>24.24</td>
<td>4.75</td>
<td>3.76</td>
<td>.001</td>
</tr>
<tr>
<td>Left</td>
<td>21.64</td>
<td>3.45</td>
<td></td>
<td></td>
</tr>
<tr>
<td>Male</td>
<td></td>
<td></td>
<td></td>
<td></td>
</tr>
<tr>
<td>Right</td>
<td>22.32</td>
<td>4.16</td>
<td>4.52</td>
<td>.001</td>
</tr>
<tr>
<td>Left</td>
<td>18.16</td>
<td>4.35</td>
<td></td>
<td></td>
</tr>
</tbody>
</table>

Maximum score = 36

As it can be seen from figure 1, the scores for right ear were better than the left ear for both males and females, which was statistically significant. As depicted in table 1, the mean scores for the right ear was 23.28 and the mean scores for the left ear was 19.90. Paired sample ‘t’ test results revealed a significant difference \((p < 0.01)\) between the left and the right ear scores for both males and females.

The results obtained from the present study are consistent with results from studies conducted on the western population by Musiek et al. (1989), Wexler and Halwes (1983) and Berlin et al. (1973). Musiek, et al. (1989) reported normative values of 30% - 73% for right ear and 27% - 60% for left ear in a group of 115 normal hearing subjects.

Berlin et al. (1973) reported a right ear advantage (REA) for dichotic speech stimuli. This REA is seen in normals because the left anterior temporal lobe is closer to the left primary speech areas than the right anterior temporal lobe. Therefore, it is postulated that there is less ‘transmission loss’ to the left posterior-temporal-parietal...
lobe on the basis of proximities within areas of the brain. Due to this proximity there is more efficient interaction between the shorter pathways (Berlin et al. (1973). Similar findings have been reported in studies conducted by Studdert-Kennedy and Shankweiler (1967). They reported of right ear superiority in the perception of speech stimuli when normal hearing listeners are stimulated dichotically with speech stimuli.

Kimura (1967) attributed this difference in ear accuracy as a function of stimulus type to bilateral asymmetry in brain function (BAF).

The BAF hypothesis suggest that
(i) The contralateral auditory neural pathways are dominant over the ipsilateral pathways during the dichotic stimulation.
(ii) Performance superiority of a particular ear is a result of that ear being contralateral to the hemisphere involved in the perception of a given type of sound.

In particular, the hypothesis implies that the left cerebral hemisphere is dominant in perception of sounds conveying language information while the right hemisphere is dominant for perception of non-speech sounds such as melodies (Kimura, 1967).

Thus, the results of the present study indicated that there existed a significant REA for the dichotic rhyme stimuli.

II. Comparison of ear correct scores and double correct scores across gender

II. A) Comparison of ear correct scores across gender

As it can be seen from table 2, the mean ear correct scores for females were better than males for both left and right ears. For the right ear, the mean score for females were 24.24 and the mean score of males were 22.32. For the left ear, the mean score for females were 21.64 and the mean scores for the males were 18.16.

Independent t-test was carried out for comparison of gender within each ear. Table-II highlights the difference in ear correct scores between males and females for both right and left ears. Independent sample ‘t’ test revealed a significant difference for the left ear (p <0.05) whereas, difference in mean scores for the right ear was not statistically significant (p>0.05).

Table 2: The Mean values, standard deviation and ‘t’ test results for the comparison across genders for the ear correct scores

<table>
<thead>
<tr>
<th></th>
<th>Gender</th>
<th>Mean</th>
<th>SD</th>
<th>t</th>
<th>Sig.(2-tailed)</th>
</tr>
</thead>
<tbody>
<tr>
<td>Right</td>
<td>Female</td>
<td>24.24</td>
<td>4.75</td>
<td>1.519</td>
<td>.135</td>
</tr>
<tr>
<td></td>
<td>Male</td>
<td>22.32</td>
<td>4.16</td>
<td></td>
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</tr>
<tr>
<td>Left</td>
<td>Female</td>
<td>21.64</td>
<td>3.45</td>
<td>3.13</td>
<td>.003</td>
</tr>
<tr>
<td></td>
<td>Male</td>
<td>18.16</td>
<td>4.35</td>
<td></td>
<td></td>
</tr>
</tbody>
</table>
As it can be seen from figure 2, females had higher scores compared to males for both right and left ears.

II. B) Comparison of double correct scores across gender

As it can be seen from table 3, the mean double correct scores were better for females as compared to males. The mean double correct scores were 11.52 for females and 7.16 for males, respectively. This difference in double correct scores across gender was statistically significant (p < 0.05).

Table 3: The mean values, standard deviation, the range and results of independent ‘t’ scores for the double correct scores

<table>
<thead>
<tr>
<th>Gender</th>
<th>Mean</th>
<th>95% Confidence Interval for Mean</th>
<th>t</th>
<th>Sig. (2-tailed)</th>
</tr>
</thead>
<tbody>
<tr>
<td></td>
<td></td>
<td>Lower Bound</td>
<td>Upper Bound</td>
<td>SD</td>
</tr>
<tr>
<td>Females</td>
<td>11.52</td>
<td>5.87</td>
<td>16.16</td>
<td>9.41</td>
</tr>
<tr>
<td>Males</td>
<td>7.16</td>
<td>4.80</td>
<td>12.51</td>
<td>7.70</td>
</tr>
</tbody>
</table>

Maximum score = 36
Elliot and Welsh (2001) concluded that gender differences found utilizing dichotic procedure may be due to differences in strategic approach to the task rather than to differences in cerebral laterality. Although the dichotic listening procedure has been used as a non-invasive neuropsychological technique for assessing laterality of speech perception, it has tended to underestimate the proportion of the right-handed population that is left-hemisphere lateralized for speech perception (Segalowitz & Bryden, 1983) and individual differences in hemispheric representation of language. These underestimations may be due to dichotic procedures being susceptible to attentional biases, order of report effects, and/or memory effects that obscure functional differences between the cerebral hemispheres.

Kalil, et al. (1994) did an exhaustive survey of auditory laterality studies from six neuropsychology journals to see if there is a sex difference in human laterality. The entire contents of six neuropsychology journals (98 volumes, 368 issues) were screened to identify auditory laterality experiments. The overall patterns of results were compatible with a weak population-level sex difference in hemispheric specialization.

Kahn, et al. (2008) studied sex differences in handedness, asymmetry of the Planum Temporale and functional language lateralization. This study was aimed to provide a complete overview of sex differences in several reflections of language lateralization: handedness, asymmetry of the Planum Temporale (PT) and functional lateralization of language, measured by asymmetric performance on dichotic listening tests (Right Ear Advantage) and asymmetry of language activation as measured with functional imaging techniques. Based on the results they concluded that there is no sex difference in asymmetries of the Planum Temporale, dichotic listening or functional imaging findings during language tasks. The observed sex effect may therefore be caused by publication bias.

Thus, gender difference seen for the left ear in the present study, can be the result of procedural variability or underestimation of this dichotic test to individual differences.
Dichotic rhyme test in Kannada for adults

in hemispheric representation of language. It is difficult to attribute this difference in scores on dichotic task between males and females to sex difference in hemispheric lateralization.

III. Comparison of ear correct scores with double correct scores

As it can be seen from Table 4, the double correct scores were lower when compared to the ear correct scores. The mean values were, 21.59 for the ear correct scores and 9.34 for the double correct scores, respectively. Standard deviation and ranges were higher for the double correct scores as compared to single correct scores.

Table 4: The Mean values, standard deviation and the ranges for the comparison between ear correct scores and double correct scores

<table>
<thead>
<tr>
<th></th>
<th>Mean</th>
<th>SD</th>
<th>95% Confidence Interval for Mean</th>
<th>Sig. (2 tailed)</th>
</tr>
</thead>
<tbody>
<tr>
<td>Ear correct scores</td>
<td>21.59</td>
<td>4.39</td>
<td>95% Confidence Interval for Mean</td>
<td></td>
</tr>
<tr>
<td></td>
<td></td>
<td></td>
<td>Lower Bound</td>
<td>Upper Bound</td>
</tr>
<tr>
<td></td>
<td></td>
<td></td>
<td>19.85</td>
<td>23.3</td>
</tr>
<tr>
<td>Double correct scores</td>
<td>9.34</td>
<td>8.55</td>
<td>5.33</td>
<td>14.33</td>
</tr>
<tr>
<td></td>
<td></td>
<td></td>
<td></td>
<td>3.37</td>
</tr>
<tr>
<td></td>
<td></td>
<td></td>
<td></td>
<td>.003</td>
</tr>
</tbody>
</table>

The double correct scores were found to be lower when compared to the ear correct scores. Paired sample ‘t’ test was done to see the difference between single correct scores and double correct scores. The difference in scores between ear correct scores and double correct scores were statistically significant (p< 0.05). This is in agreement with the previous reports by Wexler and Halwes (1983) and Musiek et al. (1989) that on a dichotic rhyme task although subjects are presented two words (one word to each ear), the precise alignment of the words, as well as the fact that the final vowel-consonant elements in each pair of words are identical, result in the subjects perceiving only one word the vast majority of the time.

The range was also calculated which showed the double correct scores to be highly variable across subjects. It is suggested that the ear correct scores be used to calculate the norms than the double correct scores because of its larger variability among subjects. This finding is in accordance with the finding by Dermody, et al. (1983) where they found that the double correct scores do not provide information about differential ear effects, when compared to the ear correct scores.

Summary and Conclusions

The purpose of the present study was to generate normative data for the dichotic rhyme test on adults with Kannada as their mother tongue. The 36 pairs of dichotic stimuli were presented at an intensity level of 60 dB HL. Subjects were instructed to
respond on an open set answer sheet (APPENDIX-B). The task involved writing down the rhyming words heard after each presentation.

The subjects taken for the study were fifty young normal hearing adults with Kannada as their mother tongue in the age range of 18 to 30 years. None of the subjects had any history of neurological involvement and were initially tested to ensure normal auditory functioning prior to administering the dichotic rhyme test. The responses were scored in terms of ear correct and double correct responses. The raw data was subjected to statistical analysis. The mean, standard deviation and range were also calculated. The results obtained from the present study were consistent with results from studies conducted on the western population by Musiek et al. (1989), Wexler and Halwes (1983) and Berlin et al. (1973).

The results from the present study are as follows:
1) There existed a significant right ear advantage for the dichotic stimuli.
2) Females had greater mean ear correct scores as compared to males for both right and left ears.
3) Females had greater mean double correct scores as compared to males.
4) The double correct scores were found to be lower when compared to the ear correct scores.
5) Since the variability is lesser for the ear correct scores as compared to double correct scores, it is recommended that ear correct scores be utilized while scoring the responses on the dichotic rhyme test.

In conclusion, the findings of the present study in Indian language context are consistent with the findings obtained on the western population.

**Future Implications**

Dichotic listening tasks can be used in the identification of cortical lesions. Hence, the dichotic rhyme test developed can be incorporated as part of the CANS evaluation battery, to evaluate central auditory processing in adults with Kannada as their mother tongue.

**References**


Effect of Low Pass Noise on Speech Perception in Individuals with Hearing Impairment

Shweta Prakash & Asha Yathiraj*

Abstract

The current study investigated the effect of low-pass noise on speech perception in 20 adults with normal hearing and 20 adults with hearing loss. Their speech identification scores in a quiet condition and in the presence of three low pass filters masking noises (250 Hz, 500 Hz and 1000 Hz) was determined. In addition to word scoring and phoneme scoring, an error analysis was also carried out. The results revealed that the word scores and phoneme scores were higher in the quiet condition in comparison to the three masking conditions. However, there was no significant difference between word identification scores and phoneme scores across the three masking conditions. The individuals with hearing impairment obtained poorer word and phoneme scores in comparison to the normal hearing individuals. The error analysis showed that place errors occurred more than voicing errors, which in turn occurred more than the manner errors.

Introduction

In everyday life many individuals with hearing impairment have difficulties in understanding speech especially in the presence of background noise or reverberation (Plomp, 1978; Duquesnoy and Plomp, 1980; Nabelek and Robinson, 1982). To study the effect of noise on speech perception, several studies have been carried out. Most of the studies have used speech spectrum noise, broadband noise or multi-talker babble (Cooper & Cutts, 1971; Dirks, Morgan and Dubino, 1982; Fallo, Trehub and Schneider, 2000; Pittman and Wiley, 2001; Hall, Grose, Buss and Dev, 2002). However, very few studies have determined the effect of low pass noise on speech perception, despite this also being a common form of noise in the environment. Many environmental signals have been noted to contain low frequency energy (Kryter, 1970). The few studies in which low pass filtered noise was used did show large difference in speech perception between normal hearing listeners and individuals with hearing impairment (Cohen & Keith, 1976; Liden, 1967; Stelmachowicz, Lewis, Kelly and Jesteadt, 1990). Cohen and Keith (1976) found that word recognition scores in normal hearing individuals and those with hearing impairment were similar in a quiet condition. However, in the presence of noise the two groups performed differently, with those with hearing impairment getting poorer scores. Similarly, Stelmachowicz, Jesteadt, Gorga and Mott (1985) reported that in the presence of low pass noise filtered at 500 HZ, large differences in performances between those with normal hearing and those with hearing impairment were observed. In contrast, with broadband-noise condition, only small differences in speech perception were present.

In 1990, Stelmachowicz, Lewis, Kelly and Jesteadt extended the study conducted by Stelmachowicz, et al. (1985). They studied speech perception with low pass filtered noise at nine different cutoff frequencies (500, 750, 1000, 1500, 2000, 2500, 3000, 4000, and 5000 Hz). This was done on five listeners with normal hearing at two speech levels (50 and 75 dB SPL) and four listeners with hearing impairment at one speech level (75 dB SPL). The

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results showed that the listeners with hearing impairment required a better S/N ratio than the listeners with normal hearing at either presentation level for all except the widest bandwidth, where their S/N ratio began to converge with the normal values. In addition, the S/N ratios for the listeners with hearing impairment plateaued at relatively narrow bandwidths (0.75 to 2.5 kHz) compared to the group with normal hearing (3.0 to 5.0 kHz).

From the above literature it is evident that low pass filter noise does effect speech perception. It has been found to have a differential effect on subjects with normal hearing and subjects with hearing impairment. Information regarding the perception of speech in noise would be helpful in the differential diagnosis of different sites of lesion. Further, understanding the influence of the spectral characteristics of noise on speech perception would be useful in counseling individuals regarding expectations of speech perception in the presence of noise. It would also be of utility in the designing of noise reduction systems in digital hearing aids. This in turn would help in the prescription of hearing aids. Also, knowledge regarding the impact of noise on speech perception would be of considerable use while fine-tuning of hearing aids. It needs to be studied whether some frequency components of noise can be retained or all frequency components of environmental noise have to be removed for optimal speech perception.

Thus, the present study aimed at determining the perception of speech in the presence of different low pass filtered noises in individuals with normal hearing and individuals with acquired hearing loss. The study also aimed at comparing the performance across the listening conditions within each participant group as well as comparing the performance between the two groups.

**Method**

**Participants:** The study was carried out on two groups of participants, one group having individuals with normal hearing and the other group having individuals with acquired sensorineural hearing loss. The age range of the two groups was 18 to 30 years and 18 to 55 years respectively. The normal hearing group had pure-tone thresholds of less than 15 dB HL in the frequencies 250 Hz to 8000 Hz and speech identification scores (SIS) of greater than 90% in both ears. Those in the clinical group had moderate to moderately-severe, flat or high frequency sensorineural (SN) hearing loss. Their speech identification scores (SIS) was greater than 80% in both ears in a quiet listening condition. All participants could read and write Kannada.

**Test Material:** For determining the speech-recognition threshold (SRT), the Kannada paired-word list developed at the Department of Audiology in AIISH, Mysore, was used. The recorded version of the Kannada phonemically balanced words developed by Yathiraj and Vijayalakshmi (2005) was used for obtaining the speech identification score. Three different low pass filtered noises with cut-off frequencies at 250 Hz, 500 Hz and 1 kHz were generated through the Adobe Audition 2.0 software. The rejection rate for the noise was approximately 20 dB per octave.

**Instrumentation:** A calibrated diagnostic audiometer was used to carry out pure-tone audiometry and speech audiometry. An immitance audiometer (GSI - Tymplestar) was utilized to find out the middle ear function. With the help of a Pentium IV computer with Adobe
Audition 2.0 software, low pass filtered noises were generated at different cut-off frequencies.

**Test Environment:** The testing for both groups was done in a sound-treated double room. The ambient noise levels were within permissible limits, as recommended by ANSI (ANSI S3.1 1991, as cited in Wilber, 1994).

**Procedure:** Initially, pure-tone thresholds were established for both groups, using a modified Hughson-Westlake procedure. SRT was obtained using the Kannada paired-word list. For speech identification testing, the recorded version of speech material (Yathiraj and Vijayalakshmi, 2005) was played through the Pentium IV computer. The output of the computer was routed through the audiometer. The 1000 Hz calibration tone of the speech identification test was used to adjust the volume unit (VU) meter deflection of the audiometer to zero. The output from the audiometer was played at 40 dB SL with reference to the participants’ SRT and delivered through TDH-39 earphones. The participants heard the noise and speech in the same ear. The choice of ear was randomized such that half the participants heard the signals through right ear and the other half through the left ear.

The speech identification score was obtained in quiet and in the presence of the three low pass filtered noises, which were presented at +5 dB SNR as recommended by Gopikrishnan (2003). This was done for both groups. The order in which the participants heard the lists was randomized to avoid any list effect. The participants were instructed to write down the words heard by them.

**Scoring:** The written responses of each participant were scored separately in the four conditions in which they were tested. Both word scores and phoneme scores were calculated. In addition, an error analysis was also carried out. The data thus obtained were subjected to statistical analyses using descriptive statistics, mixed ANOVA, repeated measures ANOVA and paired sample ‘t’ test. The results of the study discussed separately for the different scoring procedures (word scores and phoneme scores). In addition, the error analysis across the different listening conditions is also discussed.

**Results and Discussions**

**Effect of low pass filter on word scores**

The mean and standard deviation (SD) for the word scores, obtained in the four listening conditions (quiet, low pass filter of 250 Hz, 500 Hz and 1000 Hz) were computed (Table 1). This was obtained for the normal hearing group as well as the group with hearing impairment.

A comparison across listening conditions showed that the mean scores for both the groups were better in the quiet condition compared to all the three masking conditions using 250 Hz, 500 Hz and 1000 Hz low pass noise. This is evident in Table 1 and Figure 1.

A comparison of the normal and clinical groups indicated that the mean score for the group with normal hearing was higher than that obtained by the group with hearing impairment for all the four listening conditions (Table 1). Further, the SD was also lower for the normal hearing group. This shows that the variability in scores was much less for the normal hearing group, and was considerably higher for those with hearing impairment.
Table 1: Mean and standard deviation (SD) for the normal hearing group and group with hearing impairment (HI) for the word scores

<table>
<thead>
<tr>
<th>Listening conditions</th>
<th>Participant Groups</th>
<th></th>
<th></th>
</tr>
</thead>
<tbody>
<tr>
<td></td>
<td>Normal</td>
<td>HI</td>
<td></td>
</tr>
<tr>
<td></td>
<td>Mean</td>
<td>SD</td>
<td>Mean</td>
</tr>
<tr>
<td>Quiet</td>
<td>24.6 (98.2%)</td>
<td>0.76</td>
<td>18.0 (72.2%)</td>
</tr>
<tr>
<td>LP 250 Hz</td>
<td>23.6 (94.4%)</td>
<td>1.4</td>
<td>15.0 (60%)</td>
</tr>
<tr>
<td>LP 500 Hz</td>
<td>23.8 (95.95%)</td>
<td>1.0</td>
<td>14.2 (56.8%)</td>
</tr>
<tr>
<td>LP 1000 Hz</td>
<td>23.6 (94.4%)</td>
<td>0.9</td>
<td>15.5 (61.8%)</td>
</tr>
</tbody>
</table>

Note: LP 250 Hz = Low pass noise below 250 Hz
LP 500 Hz = Low pass noise below 500 Hz
LP 1000 Hz = Low pass noise below 1000 Hz

Figure 1: Mean, 95% confidence interval and the level of significance for the normal hearing group and group with hearing impairment for word scores across at four listening conditions.

Initially, to check for a main effect and interaction effect, a mixed ANOVA was carried out, with information for groups merged. The results revealed a highly significant main effect for word scores in all the listening conditions [F (3, 114) = 25.88, p < 0.001]. A highly significant interaction between the listening conditions and groups [F (3, 114) = 9.64, p < 0.001] was also obtained.

Further, the Bonferroni’s pairwise comparison test was carried out to check for any significant difference of word scores between all the conditions. It was found that there was a significant difference in word scores between the quiet condition and the three low pass filter noise masking conditions (p < 0.001). However, there was no significant difference (p > 0.05) in the word scores within the three masking conditions (250 Hz, 500 Hz and 1000 Hz low pass noise).
As the grouped data indicated that there was a significant main effect, separate one-way repeated measure ANOVAs were carried out for each group (with group as independent variable and listening conditions as dependent variable). A highly significant difference was observed between the listening conditions. This was noted for the normal group \[ F (3, 57) = 9.365, p < 0.001 \] and also for the clinical group \[ F (3, 57) = 18.989, p < 0.001 \].

To obtain a comparison across the listening conditions, Bonferroni’s pairwise comparison test was done. The results indicated that there was a significant difference \( p < 0.05 \) in the word scores between the quiet condition and all three masking conditions for the normal hearing group (Table 2) and the clinical group (Table 3). However, there was no significant difference between the speech identification scores obtained within the three masking conditions.

Table 2: Pairwise comparison between the four listening conditions for word scores in the normal hearing group

<table>
<thead>
<tr>
<th>Listening Condition</th>
<th>LP 250 Hz</th>
<th>LP 500 Hz</th>
<th>LP 1000 Hz</th>
</tr>
</thead>
<tbody>
<tr>
<td>Quiet</td>
<td>Significant ( p &lt; 0.05 )</td>
<td>Significant ( p &lt; 0.001 )</td>
<td>Significant ( p &lt; 0.001 )</td>
</tr>
<tr>
<td>LP 250 Hz</td>
<td>[ ]</td>
<td>Not significant ( p &gt; 0.05 )</td>
<td>Not significant ( p &gt; 0.05 )</td>
</tr>
<tr>
<td>LP 500 Hz</td>
<td>[ ]</td>
<td>[ ]</td>
<td>Not significant ( p &gt; 0.05 )</td>
</tr>
</tbody>
</table>

Note: LP 250 Hz = Low pass noise below 250 Hz, LP 500 Hz = Low pass noise below 500 Hz, LP 1000 Hz = Low pass noise below 1000 Hz

Table 3: Pairwise comparison between the four listening conditions for word scores in the clinical group

<table>
<thead>
<tr>
<th>Listening Condition</th>
<th>LP 250 Hz</th>
<th>LP 500 Hz</th>
<th>LP 1000 Hz</th>
</tr>
</thead>
<tbody>
<tr>
<td>Quiet</td>
<td>Significant ( p &lt; 0.001 )</td>
<td>Significant ( p &lt; 0.001 )</td>
<td>Significant ( p &lt; 0.001 )</td>
</tr>
<tr>
<td>LP 250 Hz</td>
<td>[ ]</td>
<td>Not significant ( p &gt; 0.05 )</td>
<td>Not significant ( p &gt; 0.05 )</td>
</tr>
<tr>
<td>LP 500 Hz</td>
<td>[ ]</td>
<td>[ ]</td>
<td>Not significant ( p &gt; 0.05 )</td>
</tr>
</tbody>
</table>

Note: LP 250 Hz = Low pass noise below 250 Hz, LP 500 Hz = Low pass noise below 500 Hz, LP 1000 Hz = Low pass noise below 1000 Hz

To compare the word scores of the normal group with the clinical groups a paired ‘t’ test was performed for each of the listening conditions (Table 4). It can be observed from Figure 1 and Table 4 that there was a significant difference between two groups for all four listening conditions.
Table 4: Mean and ‘t’ values for listening conditions on word scores in the normal hearing group

<table>
<thead>
<tr>
<th>Groups</th>
<th>Mean &amp; ‘t’ values for listening conditions</th>
</tr>
</thead>
<tbody>
<tr>
<td></td>
<td>Quiet Mean</td>
</tr>
<tr>
<td>Normal</td>
<td>24.6</td>
</tr>
<tr>
<td>Clinical</td>
<td>18.0</td>
</tr>
</tbody>
</table>

Note: *** = p < 0.001

The results of the present study regarding performance across listening conditions are comparable with those of Cohen and Keith (1976) and Stelmachowicz, Jesteadt, Gorga and Mott (1985). They too reported a significant difference in word recognition scores using a 500 Hz low pass masking noise in individuals with normal hearing and individuals with hearing impairment.

The current study found that individuals with normal hearing as well as individuals with hearing impairment scored significantly lower for the low pass filter noise conditions when compared to the quiet condition. In both groups the overall word scores were probably lower in the noise condition due to the masking effect of noise. Essential segmental cues were possibly not available in the presence of masking noise, making it difficult for the participant to perceive speech. Similar findings have also been noticed by Gordon-Salant and Fitzgibbon (1993) in individuals with normal hearing and by Stelmachowicz, Lewis, Kelly and Jesteadt (1990) in individuals with hearing impairment.

Within participant group comparison highlighted that both groups obtained no significant difference in word scores across the three masking conditions. Thus, irrespective of the low frequency cut off, word score dropped equally. This was observed in both groups. These findings are unlike that observed by Stelmachowicz, Lewis, Kelly and Jesteadt (1990) who reported that the required SNR to obtain 50% correct performance on nonsense syllables improved as the low pass cut off increased from 500 Hz to 1000 Hz. The contradiction in finding could be due to the difference in material used to determine speech identification. In the present study words were used while nonsense syllables used in the study by Stelmachowicz, Lewis, Kelly and Jesteadt (1990). The redundant information present in words could have help the participants hear equally well across the different masking conditions.

The comparison across participant groups in the present study indicated that the overall scores were lower in the clinical group when compared to the normal hearing group. This reduction in the speech perception scores could be attributed to certain factors. These include factors such as upward spread of masking, widened critical bands, abnormal response growth, or possible central effects, all of which might play a role while perceiving the speech in noise, as observed by Stelmachowicz, Lewis, Kelly and Jesteadt (1990). Also the strategies used by the listeners with hearing impairment to recover the spoken message, either in quiet or in noise, may have been different from those employed by listeners with normal hearing. Another reason for the poor performance in the group with hearing impairment
Effect of Low Pass Filter on Phoneme Scores

Descriptive statistics was carried out for the phoneme scores obtained at each listening conditions. This was done for both participant groups (Table 5).

The comparison across listening conditions indicates that just like the word scores, the phoneme scores were better for both the groups in the quiet listening conditions. The mean phoneme scores were lower in the three masking conditions (Table 5 & Figure 2).

Comparison of the normal group with the clinical group indicated that the former group had higher phoneme scores with lesser variability than the latter group. This occurred at all the four listening conditions (quiet, low pass filter of 250 Hz, 500 Hz and 1000 Hz).

Table 5: Mean and standard deviation (SD) for the normal hearing group and the group with hearing impairment (HI) for phoneme scores.

<table>
<thead>
<tr>
<th>Listening conditions</th>
<th>Groups</th>
<th>Normal Mean % score</th>
<th>Normal SD</th>
<th>HI Mean % score</th>
<th>HI SD</th>
</tr>
</thead>
<tbody>
<tr>
<td>Quiet</td>
<td>Normal</td>
<td>99.2</td>
<td>1.5</td>
<td>88.7</td>
<td>11.9</td>
</tr>
<tr>
<td></td>
<td>HI</td>
<td>89.7</td>
<td>11.9</td>
<td>78.7</td>
<td>11.9</td>
</tr>
<tr>
<td>LP 250 Hz</td>
<td>Normal</td>
<td>97.7</td>
<td>2.6</td>
<td>80.1</td>
<td>16.9</td>
</tr>
<tr>
<td></td>
<td>HI</td>
<td>80.1</td>
<td>16.9</td>
<td>70.1</td>
<td>16.9</td>
</tr>
<tr>
<td>LP 500 Hz</td>
<td>Normal</td>
<td>98.3</td>
<td>1.5</td>
<td>80.9</td>
<td>14.6</td>
</tr>
<tr>
<td></td>
<td>HI</td>
<td>80.9</td>
<td>14.6</td>
<td>70.9</td>
<td>14.6</td>
</tr>
<tr>
<td>LP 1000 Hz</td>
<td>Normal</td>
<td>97.8</td>
<td>1.2</td>
<td>82.2</td>
<td>17.8</td>
</tr>
<tr>
<td></td>
<td>HI</td>
<td>82.2</td>
<td>17.8</td>
<td>72.2</td>
<td>17.8</td>
</tr>
</tbody>
</table>

Note: LP 250 Hz = Low pass noise below 250 Hz, LP 500 Hz = Low pass noise below 1000 Hz.

Figure 2: Mean, 95% confidence interval and level of significance for the normal hearing group and the group with hearing impairment, for phoneme scores across at four listening conditions.
Further, mixed ANOVA was done to determine whether any main effect and interaction effect existed. This was done with the information for groups merged. The findings indicated a highly significant main effect for phoneme scores in all the listening conditions (quiet, low pass filter of 250 Hz, 500 Hz & 1000 Hz) \[F (3, 114) = 11.993, p < 0.001\]. Also a significant interaction between the listening conditions and groups was obtained \[F (3, 114) = 6.475, p < 0.001\].

To get a better understanding regarding the way the variables interacted, Bonferroni’s pairwise comparison test was done. It indicated that there was a significant difference for phoneme scores between the quiet condition and the three low pass filter noise masking conditions \(p < 0.001\). On the other hand, there was no significant difference among the three masking conditions \(p > 0.05\).

To determine whether the two groups differed in their responses in the different listening conditions, separate one-way repeated measure ANOVAs were carried out. The ANOVAs, with groups as independent variables and listening conditions as dependent variables, showed highly significant differences for the normal hearing group \[F (3, 57) = 4.338, p < 0.001\] and also for the clinical group \[F (3, 57) = 9.563, p < 0.001\].

To obtain a comparison across the listening conditions, Bonferroni’s pairwise comparison test was done in the normal hearing group. The test revealed the presence of a significant difference between the quiet and masking conditions using 500 Hz and 1000 Hz low pass noise \(p < 0.05\). However, there did not exist a significance difference between the quiet and 250 Hz low pass masking condition \(p > 0.05\). There also existed no significant difference between the three masking conditions \(p > 0.05\), which is evident in Table 6.

<table>
<thead>
<tr>
<th></th>
<th>LP 250 Hz</th>
<th>LP 500 Hz</th>
<th>LP 1000 Hz</th>
</tr>
</thead>
<tbody>
<tr>
<td><strong>Quiet</strong></td>
<td>Not significant (p &gt; 0.05)</td>
<td>Significant (p &lt; 0.01)</td>
<td>Significant (p &lt; 0.05)</td>
</tr>
<tr>
<td>LP 250 Hz</td>
<td>--------------</td>
<td>Not significant (p &gt; 0.05)</td>
<td>Not significant (p &gt; 0.05)</td>
</tr>
<tr>
<td>LP 500 Hz</td>
<td>--------------</td>
<td>--------------</td>
<td>Not significant (p &gt; 0.05)</td>
</tr>
</tbody>
</table>

Note: LP 250 Hz = Low pass noise below 250 Hz
LP 500 Hz = Low pass noise below 500 Hz
LP 1000 Hz = Low pass noise below 1000 Hz

On the other hand, for the clinical group, the phoneme scores were significantly different between the quiet condition and all three masking conditions \(p < 0.05\). However, there was no significant difference between the three masking conditions (Table 7).
Table 7: Pairwise comparison of the four listening conditions on phoneme scores in the clinical group for phoneme scores

<table>
<thead>
<tr>
<th></th>
<th>LP 250 Hz</th>
<th>LP 500 Hz</th>
<th>LP 1000 Hz</th>
</tr>
</thead>
<tbody>
<tr>
<td>Quiet</td>
<td>Significant, P &lt; 0.01</td>
<td>Significant, p &lt; 0.01</td>
<td>Significant, p &lt; 0.05</td>
</tr>
<tr>
<td>LP 250 Hz</td>
<td>Not significant, p &gt; 0.05</td>
<td>Not significant, p &gt; 0.05</td>
<td></td>
</tr>
<tr>
<td>LP 500 Hz</td>
<td>Not significant, p &gt; 0.05</td>
<td>Not significant, p &gt; 0.05</td>
<td></td>
</tr>
</tbody>
</table>

Note: LP 250 Hz = Low pass noise below 250 Hz  
LP 500 Hz = Low pass noise below 500 Hz  
LP 1000 Hz = Low pass noise below 1000 Hz

To compare the normal and the clinical groups, a paired ‘t’ test was performed for each of the listening conditions (Table 8). From Figure 2 and Table 8, it is clear that there was a significant difference in phoneme scores between the two groups for all four listening conditions.

Table 8: Mean and ‘t’ values for listening conditions on phoneme scores in the normal hearing group

<table>
<thead>
<tr>
<th>Groups</th>
<th>Mean &amp; ‘t’ values for listening conditions</th>
</tr>
</thead>
<tbody>
<tr>
<td></td>
<td>Quiet Mean ‘t’ value LP 250 Hz Mean ‘t’ value LP 500 Hz Mean ‘t’ value LP 1000 Hz Mean</td>
</tr>
<tr>
<td>Normal</td>
<td>99.2 3.9*** 97.7 4.4*** 98.3 5.5*** 97.8 3.8***</td>
</tr>
<tr>
<td>Clinical</td>
<td>88.7 80.1 4.4*** 80.9 5.5*** 82.2 3.8***</td>
</tr>
</tbody>
</table>

Note: *** = p < 0.001

The results across listening conditions show that there was a lack of difference in phoneme scores between the quiet and 250 Hz low pass masking condition, in the normal hearing group. This indicates that very low frequency masking noises does not interfere with their phoneme identification abilities. On the other hand, in individuals with hearing impairment the phoneme scores were more easily hampered with the presence of noise with a low pass cut off as low as 250 Hz. This highlights the differential effect of a similar kind of noise on the two groups (normal hearing & clinical groups).

The comparison of word scores and phoneme scores across groups in the present study showed that the phoneme scores were higher for both participant groups than the word scores. However, the difference was more marked in the group with hearing impairment. These findings are similar with those of Olsen, Tasell and Speaks (1997) and Mascarehans (2002). They too noted that phoneme scores were higher than word scores in normal hearing individuals as well as individuals with hearing impairment.

The results of the current study revealed that the performance of individuals with hearing impairment were poorer on the phoneme scores in comparison to the normal hearing
group. This was similar to what was obtained with the word scores. As mentioned earlier the poorer performance of the group with hearing impairment could be due to factors such as upward spread of masking, abnormal response growth, or central effect which affect speech perception in the presence of noise. Furthermore, the strategies used by the group with hearing impairment to recover spoken message, either in quiet or noise, have been reported to differ from those employed by the group with normal hearing (Stelmachowicz, Lewis, Kelly and Jesteadt, 1990).

**Error Analysis across different listening conditions**

The errors in the perception of place, manner and voicing were calculated for both participant groups. From Table 9 it is evident that percentage of errors was very less in the normal hearing group in comparison to the group with hearing impairment. In the group with hearing impairment place errors were maximum, followed by voicing errors. The least errors were seen in for manner of articulation (place > voicing > manner). This trend was seen for all four listening conditions.

Table 9: Feature errors for normal hearing individuals and individuals with hearing impairment (HI) in the four listening conditions

<table>
<thead>
<tr>
<th>Listening Conditions</th>
<th>Groups</th>
<th>Place (%)</th>
<th>Manner (%)</th>
<th>Voicing (%)</th>
</tr>
</thead>
<tbody>
<tr>
<td>Quiet</td>
<td>Normal</td>
<td>0.13</td>
<td>0.00</td>
<td>0.00</td>
</tr>
<tr>
<td></td>
<td>HI</td>
<td>1.09</td>
<td>0.00</td>
<td>0.13</td>
</tr>
<tr>
<td>LP250 Hz</td>
<td>Normal</td>
<td>0.13</td>
<td>0.00</td>
<td>0.13</td>
</tr>
<tr>
<td></td>
<td>HI</td>
<td>1.3</td>
<td>0</td>
<td>0.33</td>
</tr>
<tr>
<td>LP500 Hz</td>
<td>Normal</td>
<td>0.26</td>
<td>0.00</td>
<td>0.05</td>
</tr>
<tr>
<td></td>
<td>HI</td>
<td>1.3</td>
<td>0.05</td>
<td>0.28</td>
</tr>
<tr>
<td>LP1000 Hz</td>
<td>Normal</td>
<td>0.23</td>
<td>0.00</td>
<td>0.13</td>
</tr>
<tr>
<td></td>
<td>HI</td>
<td>1.56</td>
<td>0.05</td>
<td>0.26</td>
</tr>
</tbody>
</table>

Note: LP 250 Hz = Low pass noise below 250 Hz  
LP 500 Hz = Low pass noise below 500 Hz  
LP 1000 Hz = Low pass noise below 1000 Hz

Mixed ANOVA showed a significant difference between the participant groups on feature errors in the normal hearing group [F (2, 38) = 12.667, p< 0.001] and group with hearing impairment [F (2, 38) = 47.432, p< 0.001]. However, there was no significant difference for the three masking conditions using 250 Hz, 500 Hz and 1000 Hz [F (2, 38) = 0.345 p> 0.05] and interaction of feature and listening conditions [F (4, 76) = 0.360, p> 0.05].

To get information about the significance of difference between each of the three features, Bonferroni’s pairwise comparison was done. The analysis showed that there were significant differences in place errors and manner errors (p< 0.001) and manner errors and voicing errors (p< 0.001). In contrast, there was no significant difference between place errors and voicing errors (p> 0.05) for the normal hearing group, which is evident from Table 10.
Table 10: Pairwise comparison of the different feature errors in the normal hearing group

<table>
<thead>
<tr>
<th>Place</th>
<th>Manner</th>
<th>Voicing</th>
</tr>
</thead>
<tbody>
<tr>
<td></td>
<td>Significant p &lt; 0.001</td>
<td>Not Significant P &gt; 0.05</td>
</tr>
<tr>
<td>Manner</td>
<td>--------</td>
<td>Significant p &lt; 0.01</td>
</tr>
</tbody>
</table>

Also, for the clinical group, the place errors were significantly different from the manner errors and voicing errors (p < 0.001). Manner errors were also significantly different from voicing errors at the 0.01 level of significance (Table 11).

Table 11: Pairwise comparison of the different feature errors in the clinical group

<table>
<thead>
<tr>
<th>Place</th>
<th>Manner</th>
<th>Voicing</th>
</tr>
</thead>
<tbody>
<tr>
<td></td>
<td>Significant p &lt; 0.001</td>
<td>Significant p &lt; 0.001</td>
</tr>
<tr>
<td>Manner</td>
<td>--------</td>
<td>Significant p &lt; 0.01</td>
</tr>
</tbody>
</table>

Thus, it can be seen when comparing the errors across the listening conditions that they were similar. This is evident from the information given in Table 9 and the mixed ANOVA results. Hence, immaterial of the listening conditions, the pattern and number of feature errors continued to be the same.

Comparing the feature errors across the groups indicated that the percentage of feature errors were relatively more in the group with hearing impairment than the normal hearing group. This result is supported by Cohen and Keith (1976). They too observed that the errors were higher in their subjects with hearing impairment.

Also, in the current study, the percentage of place errors occurred the most, voicing errors lesser and manner errors the least (place > voicing > manner). This result is comparable to findings of Moumita (2004) and Cox (1969, cited in Revoille and Pickett, 1982) who evaluated feature analysis in listeners with hearing impairment and reported that confusion of place of articulation was more. However, they observed that the number of manner errors was more than the voicing errors. This contradicts the findings of the present study where voicing errors were relatively more than the manner errors. In the quiet situation, it was found in the current study that the normal hearing group had no voicing or manner errors. Also, those with a hearing impairment had a negligible voicing error. This could be due to the fact that in the quiet condition, listeners would get the voicing cues from the low-frequency regions of the speech signal. However, in the presence of a low pass noise, the major cues for voicing could have been masked. It has been reported by Stevens and Blumstein (1981) and Lisker and Abramson (1964) that the major cues for the perception of voicing are present in the low frequency region. In contrast, a few of the manner of articulation cues, such as fricatives and affricates, are present in the higher frequency region. These would have been audible to the listeners despite the presence of low frequency maskers. Thus, the difference in findings between the present study and that of Cox (1969)
and Moumita (2004) could be due the difference in listening conditions. While the current study evaluated the presence of feature errors in the presence of low frequency maskers, their studies did not do so. Their evaluation was carried out only in a quiet condition.

Conclusions

From the findings of the present study it can be concluded that the word and phoneme scores were significantly better in a quiet condition compared to the three masking conditions using 250 Hz, 500 Hz and 1000 Hz in individuals with hearing impairment. However, in those with normal hearing, the phoneme scores were not significantly reduced by the presence of the 250 Hz masker and was affected only by the 500 Hz and 1000 Hz maskers. In general there was no significant difference between the scores got with different types of low frequency maskers. The error analysis revealed that the place errors were maximum, followed by the voicing errors. The manner errors were the least. This pattern of errors was observed in both groups of participants.

Thus, in individuals with hearing impairment, a masking noise with a low frequency cut-off as less as 250 Hz can interfere with their speech identification abilities, unlike normal hearing individuals. Hence, it is essential that even low frequency maskers should be avoided to enable those with hearing impairment to communicate effectively.

References


Contribution of FM System for Speech Recognition in Noise in Cochlear Implantees

Sreeraj, K. & P. Manjula*

Abstract

The aim of the study was to evaluate the influence of the FM system for each participant using a cochlear implant by evaluating the speech identification performance in noise, with and without the FM system, with different durations of cochlear implant use. The study was designed using data from 12 children with pre-lingual hearing loss using cochlear implants. The data were collected in two phases: 1) Establishing SRT in noise in CI alone condition 2) Establishing SRT in noise in CI+FM condition. The results revealed that the mean scores obtained in the two test conditions, i.e.; CI alone and CI+FM, were significantly different and that the CI+FM condition gave better SNR compared to CI alone condition. The difference in mean scores was found to be 11.66 dB which was statistically significant. The results also indicated that there was no significant difference among the participants, with and without FM system, with different durations of CI use ranging from 9 months to 25 months. There is a significant improvement in the speech perception in noise when FM system is coupled to a cochlear implant. Even when the noise was 10 to 15 dB higher than the signal, the speech perception was unaltered and that the use of FM system with a cochlear implant is an effective means to improve the perception of speech in the presence of noise. FM system should be considered for children with CIs, which may be a cost-effective solution for improving speech recognition in noise. The findings of this study support the use of FM system by cochlear implantees, especially in class room situations.

Key words: SRT in noise, signal to noise ratio (SNR), CI alone, CI+FM

Introduction

Cochlear implant is one of the most significant technological achievements of the 20th century that have improved the life of individuals with severe to profound hearing loss. Listeners with cochlear implant can achieve scores of 70% to 80% in quiet but are particularly challenged by understanding speech in noise (McGuire, Carroll and Zeng, 2005). Children with cochlear implants (CIs) often experience reductions in speech recognition in noise ranging from 20% to 35% relative to quiet listening conditions regardless of the type of speech and noise stimuli (Davies, Yellon and Purdy, 2001; Eisenberg, Kirk, Martinez, Ying and Miyamoto, 2004; Litovsky, Parkinson, Arcaroli, Peters, Lake and Johnstone, 2004; Schafer and Thibodeau, 2003).

Difficulty in noise is significant because young children with cochlear implants will encounter noise in most of the situations, including school, where there is a constant level of noise in the classroom ranging from 34 to 73 dBA (Arnold and Canning, 1999; Bess, Sinclair and Riggs, 1984; Knecht, Nelson, Whitelaw and Feth, 2002). Children require speech to be sufficiently higher than the level of noise. That is, the signal to noise ratio required is higher for children than that for adults. According to Picard, and Bradley (2001) individuals with normal hearing can perform well even in 40 dBA noise and even

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when reverberation is about 0.5 seconds. Younger children, having normal speech processing in noise for their age, would require noise levels ranging from 39 dBA for 10-11 year olds to only 28.5 dBA for 6-7 year olds. In contrast, groups suspected of delayed speech processing in noise may require levels as low as only 21.5 dBA at age 6 to 7 years. As one would expect, these more vulnerable students would include the children with hearing impairment in the course of language development and also non-native listeners.

Use of a second CI (bilateral input), an hearing aid (HA) on the non-implant ear (bimodal input), and frequency modulation (FM) system input to one or both sides can improve speech recognition in noise among children with CIs (Ching, 2000; Ching, Psarros, Hill and Incerti, 2001; Davies, Yellon and Purdy, 2001; Dettman, D'Costa, Dowell, Winton, Hill and Williams, 2004; Holt, Kirk, Eisenberg, Martinez and Campbell, 2005; Kühn-Inacker, Shehata-Dieler, Muller and Helms, 2004; Luntz, Shpak and Weiss, 2005; Schafer and Thibodeau, 2003; Senn, Kompis, Vischer and Haeusler, 2005). The use of a second CI or an HA on the non-implant ear improves speech recognition in noise and may provide several binaural benefits including binaural summation, binaural squelch, reduction of the head shadow effect, and improved localization (Nabelek and Pickett, 1974).

Apart from improving speech recognition in noise an FM system provides direct access to the talker's voice through a teacher-worn transmitter and a student-worn receiver coupled to the CI speech processor. Use of an FM system also reduces the negative effects of distance from the speaker, and reverberation in the environment because of the placement of the transmitter microphone 3 to 6 inches from the mouth of the speaker. If a bilateral or bimodal input is used along with an FM system, a child may receive even greater improvements in speech recognition in noise from the combination of binaural benefits and improved signal to noise ratio.

For children using a single CI, speech recognition in noise significantly improves when using an FM system (Davies, Yellon and Purdy, 2001; Schafer and Thibodeau, 2003). There are reports of investigations where in improvement was not observed when an FM system was coupled to a CI. Crandall, Holmes, Flexer and Payne (1998) studied word recognition for eight children and ten adults with CIs and they found that there was no benefit using FM system and there was no change in the results obtained between adults and children. The lack of benefit of using the FM system in cochlear implantees can be because of the protocol used for testing, i.e., there could be a ceiling effect with maximum performance being reached with the CI alone condition. Another reason could be that of optimizing the FM parameters. Hence, this study evaluates the performance of FM system when coupled to a cochlear implant using relatively new measures - the speech recognition threshold (SRT) in noise and the signal to noise ratio (SNR) measurement which are more effective than the routine way of testing.

The aims of the present study were to study the influence of the FM system in each participant using a cochlear implant; to evaluate the speech identification performance in
noise, with and without the FM system; and to evaluate the speech identification performance in noise with and without the FM system, with different durations of cochlear implant use.

**Method**

The study was designed using data from 12 children with pre-lingual hearing loss using cochlear implants.

**Participants**

All the children were in the age range from 3 to 8 years (mean age = 6.29 years). They had severe to profound hearing loss bilaterally hearing loss of pre-lingual onset. All of them were speaking in Malayalam and were attending auditory verbal training. They were using of cochlear implant, on either right ear or left ear only. None of them used a hearing aid in opposite ear. They used the CI during all the waking hours. These children had the ability to point to the pictures of the words presented in audio mode.

**Table 1: Demographic data of the participants.**

<table>
<thead>
<tr>
<th>Participant No.</th>
<th>Ear Implanted</th>
<th>Age at implantation (in months)</th>
<th>Stable map</th>
<th>Duration of training after implant (in months)</th>
<th>Type of implant</th>
</tr>
</thead>
<tbody>
<tr>
<td>1</td>
<td>Left</td>
<td>25</td>
<td>Yes</td>
<td>25</td>
<td>CI-24M</td>
</tr>
<tr>
<td>2</td>
<td>Right</td>
<td>24</td>
<td>Yes</td>
<td>18</td>
<td>CI-24M</td>
</tr>
<tr>
<td>3</td>
<td>Right</td>
<td>34</td>
<td>Yes</td>
<td>24</td>
<td>CI-24M</td>
</tr>
<tr>
<td>4</td>
<td>Right</td>
<td>31</td>
<td>Yes</td>
<td>20</td>
<td>CI-24M</td>
</tr>
<tr>
<td>5</td>
<td>Right</td>
<td>50</td>
<td>Yes</td>
<td>9</td>
<td>CI-R(CS)</td>
</tr>
<tr>
<td>6</td>
<td>Left</td>
<td>42</td>
<td>Yes</td>
<td>9</td>
<td>CI-24M</td>
</tr>
<tr>
<td>7</td>
<td>Right</td>
<td>61</td>
<td>Yes</td>
<td>18</td>
<td>CI-24M</td>
</tr>
<tr>
<td>8</td>
<td>Right</td>
<td>23</td>
<td>Yes</td>
<td>18</td>
<td>CI-24M</td>
</tr>
<tr>
<td>9</td>
<td>Left</td>
<td>53</td>
<td>Yes</td>
<td>17</td>
<td>CI-24M</td>
</tr>
<tr>
<td>10</td>
<td>Right</td>
<td>94</td>
<td>Yes</td>
<td>11</td>
<td>CI-24M</td>
</tr>
<tr>
<td>11</td>
<td>Right</td>
<td>73</td>
<td>Yes</td>
<td>10</td>
<td>CI-24M</td>
</tr>
<tr>
<td>12</td>
<td>Right</td>
<td>74</td>
<td>Yes</td>
<td>24</td>
<td>CI-24M</td>
</tr>
</tbody>
</table>

**Equipment**

A calibrated audiometer with the facility for doing sound field audiometry was used. The children were using the cochlear implant system with body level speech processor, where in the sensitivity was set at 12 and volume at 9. An FM system - Campus S transmitter and MLxS receiver with micro link CI S adaptor was used. A Picture test of speech perception in Malayalam (Mathew, 1996), for children in the age group of 3 to 8 years, was used as the speech stimulus.
Procedure

The procedure involved measurement of speech recognition threshold in noise, i.e., SRT in noise. The speech was presented at a constant conversation level and the level of the noise was varied to obtain the SRT. For the purpose of this study SRT in noise was defined as the difference between the levels of speech and the noise when the participant repeated at least 2 out of 3 words being presented at a constant speech level.

The data were collected in two phases:

Phoenix I: Establishing SRT in noise in CI alone condition
Phoenix II: Establishing SRT in noise in CI+FM condition

Phase I: Establishing SRT in noise in CI alone condition

Prior to the evaluation, familiarization of the test words in the picture test of speech perception in Malayalam (Mathew, 1996) was ensured for all participants. It was also ensured that the speech processor sensitivity was at 12, volume was set at 9 and that the processor of the CI was working satisfactorily. The participant was seated in the test room. The loud speakers were located on the right and left side of the participant at 45° Azimuth. Distance from the centre of participant’s head to loud speakers was maintained at a constant distance of one meter throughout the evaluation as illustrated in Figure 1. The signal was delivered through the loud speaker that was closest to the implanted ear. The noise was delivered through the other loud speaker.

Participant was seated in the test room. The picture book was placed on a stool in front of the child. Each page in the book contained four pictures per stimulus word. Turning of the page in the picture book and noting the number of pictures correctly identified was done by a helper inside the test room sitting beside the child.

The participant was instructed to point to the picture which was being presented through the loud speaker by the tester. The speech was presented through monitored live voice. There were two familiarization items to make sure that the participant had understood the task correctly.

Figure 1: Illustration of the test situation in Phase I (CI alone condition).

During the test procedure, monitored live voice was used to present the speech stimulus. The intensity of the speech through the loud speaker was kept constant at 45 dB
HL. The starting level for speech noise was 30 dB HL. At this level, i.e., speech at 45 dB HL speech noise at 30 dB HL, three words were presented. The level of the noise was varied till the participant correctly identified two out of three words being presented.

**Scoring**

The speech recognition threshold (SRT) in noise and signal to noise ratio (SNR) were noted and tabulated for each participant. For the purpose of the study, the speech recognition threshold in noise was defined as the intensity of the noise at which the speech presented at a constant level of 45 dB HL was identified correctly by the participant. The level of noise at which there was correct repetition of at least two out of three words, being presented at a constant level of 45 dB HL, was noted as the speech recognition threshold in noise. For the purpose of the study, the SNR was defined as the difference between the levels of speech and noise at this point.

**Phase II: Establishing SRT in noise in CI+FM condition**

The microphone of the FM transmitter was positioned on a tripod stand at a distance of 6 inches from the speaker through which speech stimuli was presented, as represented in Figure 2. The volume control of the CI-S adaptor was kept constant at the maximum level. The volume control of the CI speech processor and sensitivity of microphone were kept constant at a level of 9 and 12 respectively across measurements.

![Figure 2: Illustration of the test situation in Phase II (CI+FM condition).](image)

Full charge of the FM system and the CI was ensured before the test. Connecting the FM system in this phase followed the steps mentioned below (as shown in Figure 3). Before testing each participant, the functioning of the FM system was ascertained by speaking into the microphone of the FM receiver from the next room and noticing the segment meter variation in the CI speech processor. This was done after setting the CI+FM in the following manner:

1. Prior to the connection the speech processor, FM receiver and transmitter were turned off.
2. The orange cable was plugged into the CI-S adaptor.
3. The FM receiver was connected to the adaptor and the setting of the receiver was kept in double green dot position which is meant for use in FM+M mode, so that both the environmental noise and the signals from FM are being received by the CI speech processor.
4. The internal gain setting of the FM receiver was set at the optimized level of 10 dB, as specified in the product specification.
5. The speech processor was then turned ‘on’ followed by turning the FM transmitter and CI-S adaptor ‘on’.
6. Synchronization of the transmitter and receiver of FM system was done.

![Diagram](image)

Figure 3: Coupling of FM receiver to the body level speech processor through the adaptor.

The speech recognition threshold in noise and SNR were measured in CI+FM condition using the procedure similar to that in Phase I.

**Scoring**

At the end of Phase I and Phase II, speech recognition threshold in noise and SNRs were obtained for each participant. The speech recognition threshold in noise and SNR for each participant was tabulated for statistical analysis.

**Results and Discussion**

Statistical analyses were done on the tabulated data using statistical package for social science (SPSS) software version 15. From the Figure 4 and Table 2 it is seen that the mean value of the speech recognition threshold (SRT) in noise obtained in the CI+FM condition was well above that obtained in the CI alone condition. This revealed that the performance with the CI+FM system was fairly higher in comparison with CI alone. On an average, the scores in CI+FM were better, by 11.66 dB compared to CI alone condition. The noise levels were much higher when the participants correctly pointed to the pictures in the CI+FM condition than in the CI alone condition, which reflects the benefit of FM system. The participants were able to point correctly even when the noise was 10 to 15 dB higher than the signal in CI+FM condition, which clearly demonstrates the FM advantage. Boothroyd and Iglehart (1998) reported that the FM benefit was present both in quiet and noise conditions, but was somewhat greater in noise. In their study, vowels were recognized more easily than consonants, and initial consonants were recognized more easily than final consonants, but the FM benefit was present for all three groups mentioned here. According to them the FM system helps individuals with severe to profound hearing loss, in both quiet and noise.

The individual variation of SRT in noise highlights the importance of determining an optimal listening arrangement on an individual basis (Figure 4). It was not only in the
mean values, but even for each of the participants tested, the SRT in noise was higher in CI+FM condition than in the CI alone condition.

![Image](image.png)

**Figure. 4:** SRT in noise of the 12 participants using CI for different durations of use.

**Table 2:** SRT in noise (dB) across different durations of implant use

<table>
<thead>
<tr>
<th>Duration of implant use</th>
<th>N</th>
<th>Mean</th>
<th>SD</th>
</tr>
</thead>
<tbody>
<tr>
<td><strong>CI alone in dB</strong></td>
<td></td>
<td></td>
<td></td>
</tr>
<tr>
<td>0-1 year</td>
<td>4</td>
<td>46.25</td>
<td>4.787</td>
</tr>
<tr>
<td>1-2 year</td>
<td>5</td>
<td>42.00</td>
<td>2.739</td>
</tr>
<tr>
<td>2-3 year</td>
<td>3</td>
<td>50.00</td>
<td>5.000</td>
</tr>
<tr>
<td><strong>CI+FM in dB</strong></td>
<td></td>
<td></td>
<td></td>
</tr>
<tr>
<td>0-1 year</td>
<td>4</td>
<td>57.50</td>
<td>5.000</td>
</tr>
<tr>
<td>1-2 year</td>
<td>5</td>
<td>57.00</td>
<td>2.739</td>
</tr>
<tr>
<td>2-3 year</td>
<td>3</td>
<td>56.67</td>
<td>2.343</td>
</tr>
</tbody>
</table>

The comparison of different durations of use was done to find out whether there was any significant effect of duration of implant use in understanding speech in noise. Kruskal-Wallis test revealed that there was no significant effect of duration on the performance across varying durations of implant use.

From the Figure 5, it is evident that the mean SNR with FM system is much higher than that without the FM system. Further, it can be observed that there was a difference in the mean SNR with different durations of use in the CI alone condition, though the difference in the mean SNRs with CI+FM condition was not much. To see if this difference was significant, Kruskal-Wallis test was administered. It was seen that there was no statistically significant difference in the performance with and without FM, for the three durations of cochlear implant use. The performance with FM showed relatively lesser variation compared to CI alone condition. The lower the performance in noise without FM system, the greater the benefit that was observed with the FM system. Earlier, Lewis, Crandell, Valente and Horn (2004) has also supported the use of FM systems in children with CIs.
As there was no significant difference between the performances in using cochlear implants for different durations, all the participants were grouped as one single group in all the future application of statistics. To examine if there was any significant difference between the two test conditions, paired t-test was performed. From this (Table 3), it can be noted that the signal to noise ratio was significantly better in the CI+FM condition than in the CI alone condition \( t (11) = 8.21, (p<0.001) \). Thus, it is evident that children with cochlear implants are able to perceive speech significantly better, even when the speech is 11 to 12 dB below the level of noise. This finding proves to be effective in justifying the use of FM system in the noisy environment. This has implication in the classrooms also. The children might be able to perceive speech of teacher more clearly when FM system is used in conjunction with CI.

Table 3: Mean and standard deviation of SNR (dB) in CI alone and CI+FM conditions.

<table>
<thead>
<tr>
<th></th>
<th>Mean</th>
<th>N</th>
<th>Std. Deviation</th>
</tr>
</thead>
<tbody>
<tr>
<td>CI alone</td>
<td>-0.4167</td>
<td>12</td>
<td>1.438</td>
</tr>
<tr>
<td>CI+FM</td>
<td>-12.0833</td>
<td>12</td>
<td>0.965</td>
</tr>
</tbody>
</table>

Thibodeau (2005) has also reported similar findings that the average speech recognition for single words presented in speech noise at +5 dB SNR was 45.5% with CI alone condition. With the addition of an FM system, the performance improved to an average of 76% from 45.5%. All the students showed improved performance with the FM system with the 95% confidence interval, ranging from 13 to 47%. Further, the lower the performance in noise without the FM, the greater the benefit that was observed with the FM system. An FM system provided direct access to the teacher’s voice through a teacher-worn transmitter and a student-worn receiver plugged into the CI speech processor. Use of an FM system reduced the negative effects of distance from the speaker, noise, and reverberation in the environment because of the placement of the transmitter microphone 3 to 6 inches from the mouth of the speaker (in this study the placement was 6 inches in front of the signal speaker).

In a study by Schafer and Thibodeau (2006), a comparison of no-FM and FM system showed that the FM system allowed for improvements in SRT in noise up to 20
dB relative to the no-FM condition. Statistically significant differences were detected among the FM-system conditions with FM-system input to the first CI or to both sides providing superior performance. It was also reported that for a child with a single CI, use of an FM system may provide more improvement in speech recognition in noise than the addition of an HA or a second CI. In their study, addition of an FM receiver to a single CI allowed for an average improvement in SRT in noise of 13.3 dB relative to the single CI alone. The large improvements are not surprising considering the ability of the FM system to reduce the deleterious effects of the noise and the distance from the talker.

Apart from the absolute value of SRT in noise, the relative measure of speech and noise as reflected in the use of signal to noise ratio (SNR) was also evaluated. It is to be noted that lower values of SNR indicates good speech recognition in the presence of noise. In Figure 6, lower SNR values indicate that the participants performed well even when the difference between speech and noise was less. Further, the negative SNR values indicate better performance in the presence of noise, even when the level of noise was higher than that of speech.

![Figure 6: Overall mean and standard deviation of SNR in CI alone and CI+FM conditions.](image)

The inter-judge reliability of the responses by the participants was validated by comparing the rating of the tester, with the two other audiologists. There was a subjective three point rating scale which described the performance of the child in terms of good, fair and poor responses. After each evaluation, three audiologists rated the performance independently on the subjective observation of the child response. The speed and accuracy of pointing with and without confusions was the key for demarcating the participants' performance as good, fair or poor. It was found that all the three judges gave the similar rating (good/fair/poor) for a particular participant.

<table>
<thead>
<tr>
<th></th>
<th>Frequency</th>
<th>Percentage</th>
</tr>
</thead>
<tbody>
<tr>
<td>Good</td>
<td>10</td>
<td>83.3</td>
</tr>
<tr>
<td>Fair</td>
<td>2</td>
<td>16.7</td>
</tr>
</tbody>
</table>

Table 4: Reliability of the response through subjective judgment
From the Table 4 it is evident that the majority of the participants were very consistent (83.3%) and two participants gave fair responses (16.7%). Thus, the results indicate that the responses of almost all the participants were consistent and reliable throughout the study.

These results suggest that an FM system should be considered for children with CIs, which may be a cost-effective solution for improving speech recognition in noise. This has implications in classroom situations.

**Conclusions**

The results of this study revealed the following:

- There was no significant effect of duration (9 months to 25 months) of CI use on the performance.
- The SRT in noise obtained in the CI+FM condition was higher than that obtained in the CI alone condition. This implies that with the CI+FM system is fairly superior to that with CI alone and that the CI user can cope with higher levels of noise in the CI+FM condition than with CI alone condition.
- The SNR with CI+FM condition was, on an average, higher by 11.66 dB compared to the CI alone condition. That is, when the noise was higher than the speech by up to 11.66 dB the participant was able to perform better in CI+FM than in CI alone condition. This implies that children with cochlear implants and the FM systems are able to perceive speech, even when the speech is 11 to 12 dB below the noise level which might prove to be an effective finding for use of an FM in the classroom environment.

From the results we can conclude that

- there is a significant improvement in the speech perception in noise when FM system is coupled to a cochlear implant. Even when the noise was 10 to 15 dB higher than the signal, the speech perception is unaltered.
- use of FM system with a cochlear implant is an effective means to improve the perception of speech in the presence of noise.

**Clinical Implications**

FM system should be considered for children with CIs, which may be a cost-effective solution for improving speech recognition in noise. The findings of this study support the use of FM system in cochlear implantees, especially in class room situations. Thus, these findings can be disseminated to the parents, school authorities and other centers to justify the need for use of an FM system. The protocol used to determine the benefit of the FM system is also found to be suitable for evaluating the benefit of FM systems.
References


Mathew, P. (1996). Picture test of speech perception in Malayalam, Dissertation submitted to the Univ. of Mysore, Mysore, as part fulfillment of M.Sc. (Sp & Hg).


The Cortical Neural Processing for Spectrally Different Speech Sounds in Individuals with Cochlear Hearing Loss

Sumitha. M.M. & Animesh Barman*

Abstract

The Auditory Late Responses (ALLRS) is reported to test the integrity of the auditory system and it also provides a tool to investigate the neurophysiological processes that underlie our ability to perceive speech (Purdy, Katsch and Sharma, & Dillon, 2001). The development of such electrophysiological measures such as ALLRS is important because they can be used to evaluate the benefits of hearing aids and cochlear implants in infants, young children, and adults those who do not cooperate for behavioral speech discrimination testing. In the current study ALLRSs were recorded using three different speech sounds, which together covered a range of frequencies across the speech spectrum. It was determined whether it could be differentiated from each other based on response latency and amplitude measures. ALLRSs were recorded from 32 ears of 16 subjects with normal hearing adults and 23 ears of 12 adult subjects with cochlear hearing loss for three different speech sounds at 40dB SL and 70 dBnHL. ALLRS waveforms were reliably elicited by each of the speech sounds in all participants in both normal hearing and cochlear hearing loss individuals. Each of the speech sound elicited different ALLRS waveforms. The results suggest that neurophysiological processes are different for different speech sounds. Longer latency for /da/ suggests that the processing at the cortical center is different depending on the frequency composition of the signal. A significant difference between the groups for latency and amplitude of ALLRSs for all the three speech sounds at each presentation level suggests that speech processing is altered in individuals with cochlear hearing loss.

Introduction

The cortical auditory evoked potentials are scalp recorded evoked potentials that occur in response to variety of stimuli (Näätänen and Picton, 1987). Cortical auditory evoked potentials can be classified into ‘obligatory’ and ‘discriminative’ potentials. Discriminative potentials are evoked by a change from frequent ‘standard’ stimulus to an infrequent ‘deviant’ stimulus. The discriminative potentials consist of mismatch negativity (MMN) and P300. The ‘obligatory’ Auditory Late Latency Responses (ALLRS) are classified in terms of their latencies or the time of occurrence after presentation of a stimulus (Hall, 1992). These responses are reported to test the integrity of the auditory system (Hall, 2007).

The auditory late latency responses have four major components. The first voltage component, P1 occurs in the 50 to 80 ms region. It is followed by, N1 between 100 and 150 ms, P2 between 150 – 200 ms and N2 between 180 to 250 ms. Early positive component in the region of 40 to 50 ms (P1) occurs less consistently than N1 and P2. The amplitude of long latency auditory evoked potentials is around 2–7 micro volts (Hall, 2007).

The ALLRSs can be used as an electrophysiological method for estimation of hearing sensitivity in infants and young children. It has been used in the evaluation of auditory processing disorders in learning disabilities and auditory neuropathy (Hall, 2007). ALLRSs have been recently used to determine the effect of phonologic and acoustic features (Crottaz-Herbette and Ragot, 2000) and to identify the cortical areas

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activated by these features (Makela, Alku and Tiitinen, 2003). This objective measure provides a tool to investigate the neurophysiological processes that underlie our ability to perceive speech (Purdy, Katsch and Sharma and Dillon, 2001; Trembley, Friesen, Martin and Wright, 2003). Furthermore, it has been used to index changes in neural processing with hearing loss and aural rehabilitation (Martin, Trembley and Stapells, 2007). The auditory late responses elicited by speech stimuli can be applied in the electrophysiological assessment to assess the representation of speech in the central auditory nervous system. Furthermore, it can be used to understand the neural encoding of speech in individuals with impaired auditory pathways (Eggermont and Ponton, 2003).

Agung, Purdy, McMahon and Newall (2006) recorded ALLRS for, /a, u, i, s, sh, m and 2 / which covered a broad range of frequencies across the speech spectrum. They observed that P1 and P2 elicited by longer duration vowels /u/, /a/, /2 / /i / decreased in latency in the order as written above. Hence, it was concluded that ALLRS wave components may provide an objective indication about the neurophysiological process of speech processing. Spectrally different speech sounds might be encoded differently at the cortical level. However, the ALLRS recording using different speech sounds may not be sufficient to measure the discrimination ability of an individual.

Need for the study

The P1-N1-P2 complex signals the arrival of stimulus information to the auditory cortex and the initiation of cortical sound processing (Hillyard and Kutas, 1983). As reported by Novak, et al. (1989), Trembley, et al. (2003) cortical potentials reflect the functional integrity of the auditory pathways involved in processing of complex speech stimuli. In general, majority of the studies have focused on recording of ALLRSs to click stimulus or more frequency specific tone bursts. But using tone burst doesn’t give much information about the processing or perception of the speech. Based on the cortical potentials recorded using speech sounds it can be possible to predict the communication abilities of an individual, and also can be used as a tool to evaluate the improvement due to treatment. ALLRS changes have been shown to occur prior to improvement seen in behavioral perception of speech sounds; physiological recordings may be helpful to predict the prognosis (Trembley, Kraus and McGee, 1998). Hence, use of speech signal has been taken in the current study.

Recording of ALLRSs using speech sounds can probe how the brain processes that underlie auditory detection and discrimination is altered in the individuals with cochlear hearing loss. To date, there is dearth of information regarding the effects of cochlear hearing loss on the ALLRSs to the speech stimuli. Hence, there was a need to study the speech processing abilities in individuals with cochlear hearing loss. As there are no normative to compare, in the Indian population, the study was also considered to include the individuals with normal hearing as a control group.

It is not sufficient to study only the processing of single frequency stimuli. Hence, there was a need to study the ALLRSs which are evoked by speech stimuli with a different spectral energy. Hence, the three different speech stimuli /ba/, /ga/ and /da/ which have spectral energy concentration in low, mid and high frequency spectral energy respectively were taken up for the study.

Speech perception of individuals with cochlear hearing loss is poorer relative to normal hearing individuals in spite of presenting stimuli at most comfortable levels. This
is because spectral and temporal cues of speech get distorted at the peripheral level before reaching the higher structures. Hence, it was hypothesized that cortical processing may be abnormal in individuals with cochlear hearing loss as cortical structures receive abnormal inputs from the lower auditory structures. Because dynamic cues like speech burst and transition are more susceptible to show abnormality. Hence, two different presentation levels were considered to record the ALLRs.

**Aims of the study**

The aims of the present study were to determine:
(i) whether the auditory late latency responses recorded for spectrally different syllables differ significantly in normal hearing adults, (ii) whether the auditory late latency responses recorded from spectrally different syllables differ significantly in hearing impaired adults, (iii) whether the ALLRSs from two groups differ significantly, and (iv) to investigate the difference in speech evoked ALLRSs between the normal hearing and cochlear hearing loss individuals when the signal presented at the same sensation level or at the same intensity level.

**Method**

**Control group**

Thirty two ears from 16 subjects with normal hearing were evaluated. The subjects had pure tone threshold within 15 dB HL at octave frequencies between 250 to 8000 Hz for air conduction and between 250 to 4000 Hz for bone conduction. ‘A’ type tympanogram with normal acoustic reflex thresholds. Speech identification scores were >90%. No history of acute or any chronic ear infection, ear ache, tinnitus, vertigo or any other otological problems. No relevant history of any medical and neurological impairment.

**Clinical Group**

Twenty three ears from 12 subjects with cochlear hearing loss were evaluated. The subjects were diagnosed as having cochlear hearing loss by an experienced audiologist. Air bone gap was within 10 dB HL. Pure tone average (PTA) ranged from 26 dB HL to 55 dB HL. ‘A’ type tympanogram with elevated or absent acoustic reflexs. Speech identification scores were proportionate to their pure tone average. No history of acute or any chronic middle ear infection, ear ache, tinnitus, vertigo or any other otological problems.

**Stimulus generation**

Syllables /ba/ /da/ and /ga/ were spoken by a male speaker and digitally recorded into a computer with the PRAAT software version 4.2.01 with a sampling frequency of 44,000 Hz and a 16 bit resolution. The voice onset time, burst portion and a little portion of the vowel was retained to make the syllable duration approximately 150 ms. The stimuli durations were 147 ms/-ba/, 150 ms/-da/ and 146 ms/-ga/.
Test procedure

**Pure tone audiometry**

A Calibrated double channel diagnostic audiometer Orbitter 922 was used. Pure tone air conduction and bone conducted thresholds for each individual was established using Modified Hughson-Westlake method (Carhart and Jerger, 1959).

**Impittance**

A Calibrated immittance meter (GSI tymp star) was used to assess middle ear status. A 226 Hz probe was used to know the type of tympanogram and acoustic reflexes were measured at 500 Hz, 1000 Hz, 2000 Hz and 4000 Hz eliciting stimulus.

**Transient otoacoustic emissions (TEOAE)**

ILO 292 system was used to record transient evoked oto-acoustic emissions (TEOAE). The transient Otoacoustic emissions were recorded for nonlinear 256 clicks presented at 85 dBpSPL. The absence of TEOAEs in the presence of hearing loss was considered as an indicator of cochlear hearing loss.

**ABR and ALLR recording**

Intelligent Hearing Systems (IHS smart EP windows USB version 3.91) evoked potential system was used to record and analyze the ABR and ALLR. TDH 49-P headphone was used to deliver the stimulus. ABR testing was done to rule out retro cochlear pathology. The recording was done at 90 dBnHL at 11.1 and 90.1 repetition rates using standard protocol for ABR. The parameters used to record ALLRS are given in Table1.

<table>
<thead>
<tr>
<th>Acquisition parameters</th>
<th></th>
</tr>
</thead>
<tbody>
<tr>
<td>Amplification</td>
<td>50,000</td>
</tr>
<tr>
<td>Analysis window</td>
<td>-100 to 500 ms</td>
</tr>
<tr>
<td>Filters</td>
<td>1–30 Hz</td>
</tr>
<tr>
<td>Notch filter</td>
<td>On</td>
</tr>
<tr>
<td>Artifact rejection</td>
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</tr>
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</table>

<table>
<thead>
<tr>
<th>Stimulus parameters</th>
<th></th>
</tr>
</thead>
<tbody>
<tr>
<td>Transducer</td>
<td>TDH-49 head phone</td>
</tr>
<tr>
<td>Type of stimulus</td>
<td>/ba/ /ga/ /da/</td>
</tr>
<tr>
<td>Duration</td>
<td>/ba/-147, /da/-150 ms, /ga/-146 ms</td>
</tr>
<tr>
<td>Intensity</td>
<td>70 dBnHL; 40 dB SL</td>
</tr>
<tr>
<td>Presentation ear</td>
<td>Monaural</td>
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<tr>
<td>Stimulus polarity</td>
<td>Alternating</td>
</tr>
<tr>
<td>No of averages</td>
<td>300</td>
</tr>
<tr>
<td>Rate</td>
<td>1.1/s</td>
</tr>
</tbody>
</table>

The recording was done twice at each presentation level, for each syllable to check for the replicability. The ALLRS peaks P₁, N₁ and P₂ were identified by 2 experienced judges other than the investigator. The latency of P₁, N₁ and P₂ and peak to
peak amplitude of $P_1$-$N_1$, $N_1$-$P_2$ were noted for /ba/, /ga/ and /da/ eliciting stimuli recorded at 70 dBnHL and at 40 dB SL.

**Results**

The latencies of $P_1$, $N_1$, $P_2$ and peak to peak amplitude of $N_1$-$P_2$ complex peaks were measured. The Mean and standard deviation (SD) were calculated for 2 groups for 3 syllables at each presentation level. Details are shown in Table 2.

It can be seen from the Table 2, that the mean latency values for the control group were shorter for all the speech sounds elicited at 70 dBnHL, compared to the clinical group. This trend was not seen at 40 dB SL level. The control group was having a mean latency values which were longer than the latency values obtained from the clinical group. The amplitude elicited was larger in the clinical group at 40 dB SL and 70 dBnHL.

Table 2: Mean, SD and t-values and significance levels for $P_1$, $N$, $P_2$ latencies and amplitude of $N_1$-$P_2$ elicited by /ba/, /da/ and /ga/ syllables at 40 dB SL and 70 dBnHL in control and clinical group

<table>
<thead>
<tr>
<th>Parameter</th>
<th>Level</th>
<th>Syllables</th>
<th>Control group</th>
<th>Clinical group</th>
<th>t values</th>
</tr>
</thead>
<tbody>
<tr>
<td></td>
<td></td>
<td></td>
<td>Mean</td>
<td>SD</td>
<td>Mean</td>
</tr>
<tr>
<td>$P_1$</td>
<td>40 dB SL</td>
<td>/ba/</td>
<td>100.00</td>
<td>19.65</td>
<td>75.95</td>
</tr>
<tr>
<td></td>
<td></td>
<td>/da/</td>
<td>110.81</td>
<td>16.88</td>
<td>78.21</td>
</tr>
<tr>
<td></td>
<td></td>
<td>/ga/</td>
<td>104.37</td>
<td>19.95</td>
<td>69.43</td>
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<tr>
<td></td>
<td>70 dBnHL</td>
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<td>87.52</td>
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<td></td>
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<td>83.72</td>
<td>17.53</td>
<td>97.17</td>
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<td></td>
<td>/ga/</td>
<td>78.72</td>
<td>15.19</td>
<td>83.17</td>
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<td>40 dB SL</td>
<td>/ba/</td>
<td>152.65</td>
<td>18.53</td>
<td>131.78</td>
</tr>
<tr>
<td></td>
<td></td>
<td>/da/</td>
<td>162.25</td>
<td>20.76</td>
<td>142.21</td>
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<tr>
<td></td>
<td></td>
<td>/ga/</td>
<td>155.15</td>
<td>22.84</td>
<td>127.17</td>
</tr>
<tr>
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<td>124.87</td>
<td>27.15</td>
<td>144.52</td>
</tr>
<tr>
<td></td>
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<td>/da/</td>
<td>137.65</td>
<td>21.69</td>
<td>160.65</td>
</tr>
<tr>
<td></td>
<td></td>
<td>/ga/</td>
<td>129.09</td>
<td>22.58</td>
<td>142.73</td>
</tr>
<tr>
<td>$P_2$</td>
<td>40 dB SL</td>
<td>/ba/</td>
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<td>25.59</td>
<td>197.39</td>
</tr>
<tr>
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<td>/da/</td>
<td>222.56</td>
<td>43.90</td>
<td>205.13</td>
</tr>
<tr>
<td></td>
<td></td>
<td>/ga/</td>
<td>225.90</td>
<td>16.81</td>
<td>202.30</td>
</tr>
<tr>
<td></td>
<td>70 dBnHL</td>
<td>/ba/</td>
<td>183.80</td>
<td>42.76</td>
<td>210.43</td>
</tr>
<tr>
<td></td>
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<td>/da/</td>
<td>205.25</td>
<td>23.30</td>
<td>227.6</td>
</tr>
<tr>
<td></td>
<td></td>
<td>/ga/</td>
<td>191.78</td>
<td>28.53</td>
<td>219.1</td>
</tr>
<tr>
<td>$N_1$-$P_2$</td>
<td>40 dB SL</td>
<td>/ba/</td>
<td>3.91</td>
<td>2.04</td>
<td>5.33</td>
</tr>
<tr>
<td></td>
<td></td>
<td>/da/</td>
<td>3.61</td>
<td>2.14</td>
<td>4.67</td>
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<tr>
<td></td>
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<td>/ga/</td>
<td>3.28</td>
<td>1.34</td>
<td>4.75</td>
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<td></td>
<td>70 dBnHL</td>
<td>/ba/</td>
<td>3.88</td>
<td>2.10</td>
<td>5.23</td>
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<td></td>
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<td>3.83</td>
<td>2.66</td>
<td>5.45</td>
</tr>
<tr>
<td></td>
<td></td>
<td>/ga/</td>
<td>3.20</td>
<td>1.83</td>
<td>4.48</td>
</tr>
</tbody>
</table>

* $p < 0.05$, ** $p < 0.01$
Comparison between the groups

Independent t-test was done to compare the latency of ALLR parameters between the groups elicited by the three spectrally different speech syllables at two different presentation levels. The t-values with significance level for $P_1$, $N_1$ and $P_2$ are given in Table 2. There was statistically significant difference in $P_1$ latency between the control group and the clinical group for all the speech sounds at 40 dB SL. At 70 dBnHL a statistically significant difference between the two groups for syllable /ba/ and /da/ was obtained but no difference was noticed for the syllable /ga/. The latency values of $N_1$ for control group were longer than the clinical group at 40 dB SL. The latency for $N_1$ wave was prolonged for /ba/, /da/ and /ga/ in the clinical group compared to control group elicited at 70 dB nHL. There was a statistically significant difference in $P_2$ latency between the control and clinical group for /ba/ and /ga/ speech sounds at 40 dB SL. The latency was significantly different for /ba/ /da/ and /ga/ at 70 dBnHL.

There was a statistically significant difference in $N_1$-$P_2$ amplitude between the control group and the clinical group for all speech stimuli at 70 dBnHL. Whereas, at 40 dB SL significant difference was obtained for /ga/ and no difference for /ba/ and /da/ at 40 dB SL.

Within group comparison

A Two-way repeated measure (3 speech sounds×2 levels) ANOVA were used to check for the effect of the speech stimuli and the level on the latency and the amplitude of ALLR parameters within the group. This was done separately for the control group and for the clinical group. Bonferroni post hoc test was administered to see the pair wise comparison for effect of syllable, when there was significant difference observed.

Control group

The results indicated a significant effect of presentation level for latency of three ALLR waves at 0.01 levels. The amplitude of $N_1$-$P_2$ did not show any significant effect due to the presentation level, whereas, ALLR eliciting syllable had significant effect only for latency of $P_1$ and $N_1$ component. Levels and syllables did not have significant interaction affect for any of the ALLR component. The results are displayed in Table 3 and 4

Table 3: F-values with significance level for $P_1$, $N_1$ and $P_2$ latency and $N_1$-$P_2$ amplitude for /ba/, /da/ and /ga/ in control group

<table>
<thead>
<tr>
<th>Parameters</th>
<th>Presentation level</th>
<th>Syllable</th>
<th>Level and syllable</th>
</tr>
</thead>
<tbody>
<tr>
<td>$P_1$</td>
<td>(1,31)=256.56**</td>
<td>(2,62)=3.87*</td>
<td>(2,62)=0.07</td>
</tr>
<tr>
<td>$N_1$</td>
<td>(1,31)=218.97**</td>
<td>(2,62)=3.57*</td>
<td>(2,62)=0.11</td>
</tr>
<tr>
<td>$P_2$</td>
<td>(1,31)=62.72**</td>
<td>(2,62)=2.38</td>
<td>(2,62)=2.13</td>
</tr>
<tr>
<td>$N_1$-$P_2$</td>
<td>(1,31)=0.015</td>
<td>(2,62)=2.355</td>
<td>(2,62)=0.160</td>
</tr>
</tbody>
</table>

* $p < 0.05$, ** $p < 0.01$
Speech elicited neural processing in coch. Hg loss

Table 4: Result of Bonferroni post hoc test for effect of syllable for $P_1$ and $N_1$ latency in control group

<table>
<thead>
<tr>
<th>Peak</th>
<th>Syllable</th>
<th>/da/</th>
<th>/ga/</th>
<th>Peak</th>
<th>Syllable</th>
<th>/da/</th>
<th>/ga/</th>
</tr>
</thead>
<tbody>
<tr>
<td>$P_1$</td>
<td>/ba/</td>
<td>10.14**</td>
<td>4.42</td>
<td>/ba/</td>
<td>11.18*</td>
<td>3.35</td>
<td></td>
</tr>
<tr>
<td></td>
<td>/da/</td>
<td>5.71</td>
<td></td>
<td>$N_1$</td>
<td>/da/</td>
<td></td>
<td>7.82</td>
</tr>
</tbody>
</table>

* $p < 0.05$, ** $p < 0.01$

Clinical group

Results indicated a significant effect of presentation level for latency of three ALLR waves. The amplitude of $N_1-P_2$ did not show any significant effect due to the presentation level, whereas, ALLR eliciting syllable had significant effect only for latency of $P_1$ and $N_1$ component but not for the $P_2$ latency and for amplitude parameters. Levels and syllables had a significant interaction effect only for $P_1$ latency, but did not have significant affect for any other ALLR component. These results are shown in the Table 5 and 6.

Table 5: F-values with significance level for $P_1$, $N_1$, $P_2$ latency and $N_1-P_2$ amplitude for /ba/, /da/ and /ga/ in the clinical group

<table>
<thead>
<tr>
<th>Parameters</th>
<th>Presentation level</th>
<th>Syllable</th>
<th>Level and syllable</th>
</tr>
</thead>
<tbody>
<tr>
<td>$P_1$</td>
<td>(1,22)=44.70**</td>
<td>(2,44)=5.72**</td>
<td>(2,44)=3.22*</td>
</tr>
<tr>
<td>$N_1$</td>
<td>(1,22)=54.69**</td>
<td>(2,44)=10.02**</td>
<td>(2,44)=1.65</td>
</tr>
<tr>
<td>$P_2$</td>
<td>(1,22)=33.76**</td>
<td>(2,44)=2.74</td>
<td>(2,44)=2.09</td>
</tr>
<tr>
<td>$N_1-P_2$</td>
<td>(1,22)=.23</td>
<td>(2,44)=.87</td>
<td>(2,44)=.91</td>
</tr>
</tbody>
</table>

$p < 0.05$, ** $p < 0.01$

Table 6: Result of Bonferroni post hoc test for effect of syllable on $P_1$ and $N_1$ latency in the clinical group

<table>
<thead>
<tr>
<th>Peak</th>
<th>Syllable</th>
<th>/da/</th>
<th>/ga/</th>
<th>Peak</th>
<th>Syllable</th>
<th>/da/</th>
<th>/ga/</th>
</tr>
</thead>
<tbody>
<tr>
<td>$P_1$</td>
<td>/ba/</td>
<td>5.95</td>
<td>5.43</td>
<td>/ba/</td>
<td>13.28**</td>
<td>3.19</td>
<td></td>
</tr>
<tr>
<td></td>
<td>/da/</td>
<td>11.39**</td>
<td></td>
<td>$N_1$</td>
<td>/da/</td>
<td></td>
<td>16.47**</td>
</tr>
</tbody>
</table>

* $p < 0.05$, ** $p < 0.01$

Table 6(a), indicates that there was a significant difference between the /da/ and /ga/ for $P_1$ latency. The mean $P_1$ value for the /da/ is longer compared to the /ga/ syllable, which could have led to this result. There is statistically significant difference for $N_1$ latency between the /ba/ and /da/ syllable and also between /da and /ga/ syllable (Table 6(b)). The /da/ latency was prolonged when compared to the /ba/ and /ga/ syllable. The /ga/ syllable had the least $N_1$ latency values.

Across syllable

One-way ANOVA was administered to find out the significance differences in the latencies of $P_1$, $N_1$ and $P_2$ peaks and the amplitude of $N_1-P_2$ across three different speech stimuli within group. A Bonferroni post hoc test was done to find out the effect of stimuli, when there was significant difference.
Control group

Table 7 shows that a significant difference was obtained for P1 latency at 40 dB SL and significant difference for P2 latency at 70 dBnHL. None of the other parameter showed significant effect across the stimuli. The results of the Bonferroni post hoc test for the effect of speech stimuli are shown in the Table 8.

Table 7: F- values with significance level for P1, N1, P2 latency and N1-P2 amplitude at 40 dB SL and 70 dBnHL for the control group

<table>
<thead>
<tr>
<th>Parameters</th>
<th>40 dB SL</th>
<th>70 dBnHL</th>
</tr>
</thead>
<tbody>
<tr>
<td>P1</td>
<td>(2.62)=3.24*</td>
<td>(2.62)=2.58</td>
</tr>
<tr>
<td>N1</td>
<td>(2.62)=2.19</td>
<td>(2.62)=2.96</td>
</tr>
<tr>
<td>P2</td>
<td>(2.62)=1.15</td>
<td>(2.62)=3.44*</td>
</tr>
<tr>
<td>N1-P2</td>
<td>(2.62)=1.30</td>
<td>(2.62)=1.37</td>
</tr>
</tbody>
</table>

*p < 0.05

Table 8: Result of Bonferroni post hoc test for effect of syllable on P1 and P2 latency in the control group

<table>
<thead>
<tr>
<th>Peak</th>
<th>Syllable</th>
<th>/da/</th>
<th>/ga/</th>
</tr>
</thead>
<tbody>
<tr>
<td>P1 at 40 dB SL</td>
<td>/ba/</td>
<td>10.81*</td>
<td>6.43</td>
</tr>
<tr>
<td></td>
<td>/da/</td>
<td>6.43</td>
<td></td>
</tr>
<tr>
<td>P2 at 70 dBnHL</td>
<td>/ba/</td>
<td>21.44*</td>
<td>7.98</td>
</tr>
<tr>
<td></td>
<td>/da/</td>
<td>13.46</td>
<td></td>
</tr>
</tbody>
</table>

*p < 0.05

It can be observed from the Table 8(a) that there was a significant difference between the P1 latency elicited by /ba/ and /da/ at 40 dB SL. The mean P1 value for the /da/ stimulus was prolonged compared to the /ba/ syllable in normal hearing group. A statistically significant difference was obtained between /ba/ and /da/ syllable for the P2 latency at 70 dBnHL which can be seen in Table 8(b), the mean P2 latency values for the syllable /da/ were longer than /ba/.

Clinical group

There was a significant effect of syllables at 40 dB SL only for N1 latency. No significant effect on P1, P2 and N1-P2 parameters observed at 40 dB SL. At 70 dBnHL, there was a significant effect of syllables on the P1, N1 and P2 latencies at a 0.01 level, but there was no effect observed for N1-P2 amplitudes.

Table 9: F- values with significance level for P1, N1, P2 latency and N1-P2 amplitude at 40 dB SL and 70 dBnHL for clinical group

<table>
<thead>
<tr>
<th>Parameters</th>
<th>40 dB SL</th>
<th>70 dBnHL</th>
</tr>
</thead>
<tbody>
<tr>
<td>P1</td>
<td>(2.44)=3.09</td>
<td>(2.44)=7.49**</td>
</tr>
<tr>
<td>N1</td>
<td>(2.44)=9.02**</td>
<td>(2.44)=8.76**</td>
</tr>
<tr>
<td>P2</td>
<td>(2.44)=.743</td>
<td>(2.44)=5.56**</td>
</tr>
<tr>
<td>N1-P2</td>
<td>(2.44)=.44</td>
<td>(2.44)=1.79</td>
</tr>
</tbody>
</table>

** p < 0.01
It is evident from the Table 10 that there was a significant difference between the N1 elicited by ba/-/da/ and /da/-/ga/ syllable at 40 dB SL. The mean values for the /da/ syllable was prolonged compared to the /ba/ and /ga/ syllable, this resulted in the significant difference.

Table 10: Result of Bonferroni post hoc test for effect of syllable on N1 latency in the clinical group at 40 dB SL

<table>
<thead>
<tr>
<th>Peak</th>
<th>Syllable</th>
<th>/da/</th>
<th>/ga/</th>
</tr>
</thead>
<tbody>
<tr>
<td>N1</td>
<td>/ba/</td>
<td>10.43*</td>
<td>4.60</td>
</tr>
<tr>
<td></td>
<td>/da/</td>
<td>15.04**</td>
<td></td>
</tr>
</tbody>
</table>

* p < 0.05, ** p < 0.01

There was statistically significant difference between the /da/ and /ga/ syllable for P1 latency at 70 dBnHL as shown in Table 11(a). The mean P1 latency values for the syllable /da/ are longer than the /ga/ stimulus in the clinical group. In the Table 11(b), there is significant difference between /ba/-/da/ and /da/-/ga/ syllables for N1 latency. The /da/ had longer latency when compared to /ga/ and /ba/ syllable. /ga/ had the least latency values. In Table 11(c), it is shown that there was a significant difference in the P2 latencies across /ba/ and /da/. /da/ had prolonged P2 latency when compared to /ba/ syllable.

Table 11: Result of Bonferroni post hoc test for effect of syllable on P1, N1 and P2 latency in the clinical group at 70 dBnHL

(a)

<table>
<thead>
<tr>
<th>Peak</th>
<th>Syllable</th>
<th>/da/</th>
<th>/ga/</th>
</tr>
</thead>
<tbody>
<tr>
<td>P1</td>
<td>/ba/</td>
<td>9.65</td>
<td>4.34</td>
</tr>
<tr>
<td></td>
<td>/da/</td>
<td>14.0**</td>
<td></td>
</tr>
</tbody>
</table>

(b)

<table>
<thead>
<tr>
<th>Peak</th>
<th>Syllable</th>
<th>/da/</th>
<th>/ga/</th>
</tr>
</thead>
<tbody>
<tr>
<td>N1</td>
<td>/ba/</td>
<td>16.13**</td>
<td>1.78</td>
</tr>
<tr>
<td></td>
<td>/da/</td>
<td>17.91**</td>
<td></td>
</tr>
</tbody>
</table>

(c)

<table>
<thead>
<tr>
<th>Peak</th>
<th>Syllable</th>
<th>/da/</th>
<th>/ga/</th>
</tr>
</thead>
<tbody>
<tr>
<td>P2</td>
<td>/ba/</td>
<td>17.26**</td>
<td>8.73</td>
</tr>
<tr>
<td></td>
<td>/da/</td>
<td>8.52</td>
<td></td>
</tr>
</tbody>
</table>

** p < 0.01

Effect of the presentation level

Paired t-test was carried out to the effect of the presentation level on each of the parameters of ALLR elicited by three different speech stimuli. Both the control group and the clinical group had a significant effect of the presentation level across the speech sounds on the P1, N1 and P2 latencies. Whereas there was no significant effect observed for the amplitude of N1-P2 in both the groups. The mean latency values for P1, N1 and P2 for the control group is shorter for all the speech stimuli at 70 dBnHL. However, the latency values were shorter in the clinical group when presented at 40 dB SL. These differences lead to the significant difference in the latencies of different ALLR waves between the two presentation levels for both the groups. The results obtained for each presentation level can be seen in Table 12.
Table 12: t-values along with significance level for control group and clinical group

<table>
<thead>
<tr>
<th>Parameter</th>
<th>Syllable</th>
<th>Control group</th>
<th>Clinical group</th>
</tr>
</thead>
<tbody>
<tr>
<td></td>
<td></td>
<td>t</td>
<td>t</td>
</tr>
<tr>
<td>P1 Latency</td>
<td>/ba/</td>
<td>8.94**</td>
<td>4.95**</td>
</tr>
<tr>
<td></td>
<td>/da/</td>
<td>9.37**</td>
<td>5.31**</td>
</tr>
<tr>
<td></td>
<td>/ga/</td>
<td>8.47**</td>
<td>5.91**</td>
</tr>
<tr>
<td>N1 Latency</td>
<td>/ba/</td>
<td>7.88**</td>
<td>5.83**</td>
</tr>
<tr>
<td></td>
<td>/da/</td>
<td>6.73**</td>
<td>5.79**</td>
</tr>
<tr>
<td></td>
<td>/ga/</td>
<td>7.41**</td>
<td>5.42**</td>
</tr>
<tr>
<td>P2 Latency</td>
<td>/ba/</td>
<td>5.33**</td>
<td>4.92**</td>
</tr>
<tr>
<td></td>
<td>/da/</td>
<td>2.17**</td>
<td>4.65**</td>
</tr>
<tr>
<td></td>
<td>/ga/</td>
<td>9.61**</td>
<td>3.91**</td>
</tr>
<tr>
<td>N1-P2 Amplitude</td>
<td>/ba/</td>
<td>.070</td>
<td>.15</td>
</tr>
<tr>
<td></td>
<td>/da/</td>
<td>.42</td>
<td>1.29</td>
</tr>
<tr>
<td></td>
<td>/ga/</td>
<td>.22</td>
<td>.63</td>
</tr>
</tbody>
</table>

** p < 0.01

Discussion

Effect of speech stimuli

The speech stimulus in the present study selected in such a way that it had low, mid and high frequency spectrum. All the stimuli selected for the study was voiced CV syllable, the vowel /a/ was kept constant. The duration of the three stimuli was approximately 150 ms (/ba/-147 ms, /ga/-146 ms and /da/-150ms).

It has been noticed that latency obtained for /da/ stimulus was longer in both normal and cochlear hearing loss group at both 40 dB SL and 70 dBnHL. However, significant difference was there for P1 latency at 40 dB SL and P2 at 70 dBnHL in control group. The speech stimuli /ba/ elicited a shorter latency in control group both at 40 dB SL and 70 dBnHL. There was significant difference for N1 latency at 40 dB SL in clinical group. The P1, N1 and P2 latency was significantly longer for /da/ at 70 dBnHL in individuals with Sensory-Neural hearing loss. The speech stimuli /ga/ elicited a shorter latency in clinical group both at 40 dB SL and 70 dBnHL. Amplitude did not show significant difference across the three speech sounds in both groups at 40 dB SL and 70 dBnHL.

These findings are in agreement with the findings of Shruti (2007). She used /i/, /m/ and /s/, and found that the latency of the high frequency content speech stimuli had a prolonged latency than the other. This also supports the results of Agung et al., (2006). They used the speech stimuli /a/, /u/, /i/, /s/, /sh/, /m/ and /כ/ which covered a broad range of frequencies across the speech spectrum. They found that the latencies of speech stimuli with high frequency content /s/ and /sh/ had significantly prolonged latencies than the other stimuli. This can be attributed to the fact that the high frequency has a less speech energy concentration when compared to the low or the mid frequency syllable. This would have resulted in longer latencies for /da/. Another reason could be the duration of the stimulus. The duration of /da/ (150 ms) stimulus was longer than the /ba/(147 ms) and /ga/ (146 ms), this difference in the latency for /da/ can also be
attributed to the duration difference. However, the slight variability in stimulus duration may not cause significant difference in latency difference.

The another physiological reasons for difference in ALLRS responses for low and high frequency stimuli was investigated using fMRI studies by Yeltin, Ronald, Christensen and Purdy (2004). It was reported that the cortical areas that respond to the low frequency auditory information are located more superficially (ie. closer to the surface of the scalp) and for high frequency deep layer of the cortical regions respond. Hence, the low frequency stimuli may activate more superficial cortical areas and produce smaller latency of ALLRS component than the high frequency speech sounds, when surface scalp electrodes are used.

**Effect of presentation level**

**At same dB SL**

All the speech sounds elicited a shorter latency in the clinical group at 40 dB SL. All most all the peak latencies differed between the groups was statistically significant at 40 dB SL. Amplitude obtained in clinical group was significantly more only for /ga/ stimuli. When the presentation level of the stimulus was 40 dB SL, the intensity level was much higher for clinical group when compared to the control group. Higher the intensity reaching the ear, broader will be the excitation of the basilar membrane which would have lead to excitation of more number of auditory nerve. Hence, this could have resulted in faster transmission and shorter latency and more amplitude of ALLRS components in clinical group.

Another reason for decrease in the latency with an increase in the stimulus intensity could be due to the progressively faster rising generator potential within the cochlea and similarly faster development of excitatory post synaptic potential (Moller, 1981). Picton and Hillyard (1974) reported that the latency of the compound action potential directly depends on how quickly the generator potential and the excitatory post synaptic potential reach the threshold for firing. Hence, this would lead to shorter latency in cochlear hearing loss group when presented at 40 dB SL as the intensity level was much higher in this group than the control group.

Increase in the amplitude parameters with the increase in the stimulus intensity may be because of the increase in the audibility of the stimulus. This is supported by Hall (1992). He said that the AEPs amplitude increases with the increase in the intensity. The amplitude of an AER is decided by the number of neurons firing for particular stimulus intensity. At higher intensities, the number of neuron beginning to fire will be more and amplitude of the compound action potential thus generated will be high. This would have resulted in higher ALLR amplitude elicited responses in cochlear hearing loss group.

In control group the presentation level would have been approximately 40 to 55 dBnHL, which was much lesser than clinical group. In normal hearing individuals the active mechanism was dominated at this intensity level, leading to sharp tuning of the basilar membrane. Thus, resulted in excitation of less number of auditory nerve, and less volume conduction, which leads to slow transmission. This might have lead to the longer latency and reduced ALLR amplitude in the control group.
At same dBnHL

At 70 dBnHL latency of all the ALLRS waves was shorter for control group. All most all the peak latencies differed between the groups was statistically significant at 70 dBnHL. Amplitude obtained in the clinical group for all speech sounds were more for all speech sounds at 70 dBnHL.

The latencies were shorter in the control group and prolonged in the clinical group. This can be supported by the fact that at 70 dBnHL both the passive as well as the active mechanism would have played an role in excitation of basilar membrane in normal hearing individuals, which leads to faster transmission and shorter latency. In the clinical group the energy reaching to the cochlea was less as they had hearing loss. The level would have reduced with the increase in severity of hearing loss. Hence, less compound action potential would have generated which would have lead to slower transmission and thus led to longer latency.

To conclude, the speech stimuli dominated by high frequency energy elicited a latency which was longer than the other sounds; this was true for both control as well as clinical group. These findings are in agreement with the findings of Agung et al. (2006) and Shruthi (2007). ALLRSs recorded for three stimuli at each presentation level differ significantly in control and clinical group. This suggests that the speech processing is altered in clinical group which leads to reduced speech perception abilities in clinical group.

The comparison between the groups at equal hearing level were done in order to see the difficulties that the hearing impaired individuals will face in day to day situation. As we know that in day to day situation both normal and hearing impaired individuals would be exposed to sounds at equal hearing levels. At equal presentation level the transmission of signal could be slower due to reduced energy at the cochlea. This suggests that in individual with cochlear hearing loss temporal processing may be affected if the signal is low.

At 40 dB SL the transmission of information is faster in clinical group, but still the processing is affected in clinical group. This can be due to the degraded frequency resolution due to broadening of the basilar membrane excitation. In sensorineural hearing loss group, speech perception ability is correlated with the pure tone threshold. The cochlear distortion effects, increases with the increase in the degree of hearing loss, which results in loss of cochlear amplifier leading to poor speech perception abilities (Moore, Poston, Eggermont and Huang, 1996).

Conclusion

It can be concluded that the ALLRS recorded by spectrally different speech sounds are different in both normal hearing and cochlear hearing loss individuals. This suggests that neurophysiological processes are different for different speech sounds. Longer latency for /da/ suggests that latency of the processing at the cortical center is also different depending on the frequency composition of the signal. A significant difference between the groups for all the parameters for all the speech sounds at each presentation level suggests that speech processing is altered in individuals with cochlear hearing loss.
References


Comparison of Click and Chirp Evoked ABR in Normal Hearing and Hearing Impaired Individuals

Vignesh S. S. & Animesh Barman*

Abstract

Auditory brainstem responses (ABR) evoked by click stimuli are most commonly evoked potentials used for threshold estimation. It has been reported that the click stimuli don't evoke a synchronous neural firing along the basilar membrane and thus resulting in reduced amplitude of ABR wave V. To overcome the reduction in amplitude, chirp stimuli have been developed to compensate the group delays in basilar membrane traveling wave. Thus the present study aimed at Comparing the ABR wave (amplitude, latency and morphology) elicited by click and chirp stimuli in 30 ears with normal hearing and 20 ears with cochlear hearing loss at 80 dBnHL and 40 dB SL at the repetition rates of 11.1/ sec and 30.1/ sec. ABR thresholds obtained by click and chirp in normal hearing and cochlear impaired individuals elicited at 30.1/ sec repetition rate were compared with behavioral thresholds. Also to know whether chirps can evoke any significant neural synchrony in individual with auditory dysynchrony chirp evoked ABR were recorded at 11.1/ sec repetition rate in 10 ears with auditory dysynchrony. The results revealed that there was a significant difference in latencies within and across groups for click and chirp stimuli but there was no significant differences observed in terms of amplitude except for wave I at 11.1/sec repetition rate at 80 dBnHL revealing cochlear processing differences across the groups. There were high correlations between click and chirp evoked with their behavioral thresholds in both normal hearing and hearing impaired subjects suggesting the application of chirp evoked ABR in threshold estimation. Chirp stimuli can evoke better synchrony then click stimuli suggesting the clinical use of chirp evoked ABR in individuals with auditory dysynchrony.

Introduction

The Auditory Evoked Potentials are the electrical responses of the auditory nervous system to auditory stimuli. Auditory evoked potential’s (AEP's) that are recorded from the scalp represents the contribution of neural events that arise from many discrete and neural generating sites along the auditory pathway. They are usually grouped in to various categories based on the time of occurrence after the onset of the stimuli and this grouping corresponds roughly to the site of generation. Short latency AEP’s like ABR are used clinically for threshold estimation and neurodiagnosis and are elicited by using click and tone bursts. The click evoked auditory brainstem response (ABR) waveform generally consists of seven peaks, all occurring within the first 10 ms after the signal onset. Of the seven peaks, wave I, III, and V are significantly robust for clinical use. The most robust peak can be elicited near threshold level is wave V.

It is generally assumed that ABR are the best evoked by stimulation with clicks. Clicks are commonly used in electrophysiological tests of the human auditory system to elicit synchronized auditory brainstem responses (ABR). Because of its abrupt onset, the acoustic click is often thought to be an ideal stimulus for eliciting a detectable ABR. Clicks or impulsive stimuli are also used under the assumption that their wide spectral spread, inherent in transient signals, elicits synchronous discharges from a large

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proportion of cochlear fibers (Kodera, Yamane and Suzuki, 1977; Gorga and Thornton, 1989).

But in cochlea the response of a click is not entirely synchronous, that is the peak of the response occurs several milliseconds later in the low frequency channels than it does in high frequency channels (Bekesy, 1960). As a consequence ABR responses are largely generated by synchronized activity of high frequency region (Dau, Wegner, Mellert and Kollmeier, 2000). Also when a transient stimuli progresses apically along the basilar membrane, single unit activity is less synchronous with the preceding activity from basal units (Tsuchitani, 1983) because of the temporal delays imposed by the traveling wave. This results in an asynchronous pattern of neural firing along the length of cochlear partition. In addition it is likely that the activity generated from single units is more synchronous at basal regions and would be out of phase with activity from some apical units. As a consequence the combination of phase cancellation and loss of synchronization bias the evoked potentials to reflect the activity from more basal, high frequency regions of cochlea (Gorga and Thornton, 1989). Thus, it suggests that the click may not be the optimal stimuli for ABR recording.

Recent studies have shown that a chirp rising in frequency, which is tailored to activate the entire cochlea concurrently, evokes a larger wave-V amplitude than a traditional click presented at the same sensation level (Dau et al., 2000; Wegner and Dau, 2002; Fobel and Dau, 2004). Rising chirp stimuli starts with low frequencies and sweeps nonlinearly in time toward high frequencies. The rising chirp theoretically produces simultaneous displacement maxima by cancelling traveling-time differences along the cochlear partition. The equations determining the temporal course of the chirp were derived on the basis of a linear cochlear model (de Boer, 1980), and were calculated to be the inverse of the delay-line characteristic of the human cochlear partition. The use of a broadband rising chirp was shown to reflect activity also from low-frequency regions.

Since studies done so far on comparison of click and chirp evoked ABR on a limited number of subjects with normal hearing it is necessary to study in large population of normal hearing and hearing impaired individuals using to elicit ABR threshold for generalization and clinical use. There is dearth of information in comparing click and chirp evoked ABR thresholds in individuals with normal hearing and cochlear hearing loss and also in correlating pure tone averages with chirp evoked ABR. Also there is a need to assess whether chirps can evoke a detectable ABR in individuals with auditory neuropathies where outer hair cells are preserved.

Thus, the purpose of the present study was to

- Establish ABR data using chirp stimuli in large number of individuals with normal hearing,
- Compare the wave parameters (amplitude, latency and morphology) of click and chirp evoked ABR in individuals with normal hearing and cochlear hearing loss.
impaired at 80 dBnHL and 40 dB SL and at repetition rates of 11.1/ sec and 30.1/ sec.

✓ Compare behavioral thresholds and ABR thresholds obtained by click and chirp in normal hearing and cochlear impaired individuals at 30.1/ sec repetition rate and

✓ To know whether the chirp stimuli can evoke any significant neural synchrony in individual with auditory dysynchrony at 11.1/ sec repetition rate.

Method

A. Subjects

To accomplish the aims three groups of subjects were taken

Group I: consisted of 19 subjects (30 ears) with normal hearing, aged from 19-40 years. Air conduction and bone conduction thresholds were less than or equal to 15 dB HL in the octave frequencies. All the subjects had ‘A’ type tympanogram and acoustic reflexes were within normal limits indicating normal middle ear function. Click evoked ABR and Transient Otoacoustic emissions were present in all the subjects. None of them had any history of otological symptoms (ear ache, ear discharge, and tinnitus or hearing loss) or neurological problems or any other general weakness.

Group II: consisted of 15 subjects (20 ears) with mild to moderate cochlear hearing loss, aged from 25 to 70 years with flat or sloping configuration. All of them had ‘A’ type tympanogram with present, elevated or absent of acoustic reflexes, indicative of no middle ear pathology. Latencies of click evoked ABR waves were appropriate to their hearing loss and did not indicate retrocochlear pathology. Transient otoacoustic emissions were absent in all the subjects, indicated cochlear involvement. Speech identification scores were proportionate to their degree of hearing loss. None of them had any history of acute or chronic ear infections (ear pain or ear discharge) or neurological problems or any other general weakness.

Group III: consisted of 5 subjects (10 ears) with auditory neuropathies, aged from 11 to 22 years. Both air conduction and bone conduction thresholds showed mild to moderate neural hearing loss with pure tone average ranged between 26 dB HL to 55 dB HL. Transient otoacoustic emissions were present in all the subjects. Absent or poor click evoked ABR morphology at 90 dBnHL which was disproportionate to the degree of hearing loss. All subjects in this group had poor Speech identification in quiet or speech in noise scores and difficulty in understanding speech in noisy condition. All of them had ‘A’ type tympanogram with absent reflexes. These subjects had no history of middle ear infections or general weakness.
B. Instrumentation

A calibrated two channel diagnostic audiometer (AC40) with TDH-39 head phone and B-71 bone vibrator was used to obtain pure tone thresholds. A calibrated immittance meter (GSI- tymptest) was used to assess the middle ear function. TEOAE’s were recorded using ILO292 DP Echoprost instrument. ABR recordings were done using Intelligent Hearing Systems (IHS) smart Evoked potential systems (version 2.39) with TDH-49 P head phones.

C. Stimuli

Click and chirp stimuli were used to record ABR. Click stimulus with duration of 100 µs was used. Flat spectrum rising Chirp stimuli of 10.31 ms duration with a frequency range of 100 Hz to 6 kHz was generated to record ABR. A chirp stimulus was generated using a program written in MATLAB using the method as described by Dau et al. (2000). The stimuli were generated with the sampling rate of 44100 Hz and 8 bit resolution. This stimulus was further loaded in IHS system and was converted to the IHS software acceptable format. No windowing were applied to the chirp stimuli presented. The temporal and spectral representation of chirp stimuli used to record chirp evoked ABR is shown in the Figure 1.

![Figure 1: Temporal (A) and spectral representation (B) of flat rising chirp used in the present study.](image)

D. Procedure

The subjects were instructed to sit comfortably and relax on a reclining chair facing away from the instrument. They were instructed to avoid movement of head, eyes, neck and limbs during testing to avoid artifacts. Click and chirp stimuli were presented with alternating polarity at 11.1/sec and 30.1/sec repetition rates. The ABRs were recorded differentially between electrodes applied to the upper forehead (FpZ) and the ipsilateral mastoid (M1 or M2). The contralateral mastoid was used as ground. Intraelectrode impedance and interelectrode impedance were maintained within 5 KΩ and 3 KΩ respectively. Scalp activity was amplified by 1,00,000 times and filtered with a pass band of 0.1–3 kHz.

ABR was recorded in 2 phases. In Phase I click evoked ABR was recorded while in Phase II chirp evoked ABR was recorded for the same subject.
**Phase I:** Click evoked ABR was initially recorded for 11.1/sec repetition rate at 80 dBnHL and then at 40 dB SL levels. Later the responses were recorded at the same intensity levels (80 dBnHL and 40 dB SL) at 30.1/sec repetition rate. For threshold estimation the intensity level were then set at 30 dB SL values above pure tone averages and ABR recordings were carried out. Once the response was obtained at 30 dBnHL, the intensity level of the click stimuli was reduced in 10 dB steps until no response was observed. Once no response was observed, the intensity was then increased in 5 dB steps till a detectable ABR could be obtained. The minimum intensity level at which a detectable ABR could be identified was considered as click ABR threshold. All recording for threshold estimation were carried out at the repetition rate of 30.1/sec.

**Phase II:** Chirp evoked ABR were also recorded at 11.1/sec and then at 30.1/sec repetition rates for the intensity levels of 80 dBnHL and 40 dB SL. The procedure adopted to estimate ABR thresholds using click stimulus was also adopted to establish chirp evoked ABR thresholds. Both Phase I and Phase II were carried out for both the individuals with normal hearing and cochlear hearing impaired.

For group III ABR recording were done using click and chirp at 80 dBnHL with repetition rate of 11.1/s. If any detectable wave V responses were observed at 80 dBnHL either for click or chirp stimuli, threshold estimation was carried out at 11.1/sec repetition rate. The step size used to estimate threshold were the same as mentioned earlier. The minimum level where a detectable ABR could be obtained was considered as click or chirp evoked ABR threshold in individuals with auditory neuropathy. ABR recordings for all the groups were repeated near or at threshold for replicability for the evoking stimuli.

**Analysis**

Absolute latencies and peak to peak amplitude were measured for each of the identified peaks. Descriptive statistics (mean and standard deviation) for wave latencies (I, III & V) and amplitude (I, III & V) parameters were computed for click and chirp evoked ABR obtained at two repetition rates (11.1/sec & 30.1/sec) and two intensity levels (80 dBnHL and 40 dB SL). Repeated measures ANOVA were applied to the above click and chirp evoked ABR wave V latency or amplitude across different intensity, repetition rate conditions and groups to see the significance level. Paired t - test were applied to compare the click and chirp evoked ABR wave I and III latency and amplitude(I & III) between 11.1/sec and 30.1/sec repetition rates recorded at 80 dBnHL. Since chirp ABR frequency specificity lies in the region of 0.5 – 1 kHz and click ABR frequency specificity between 2 – 4 kHz two pure tone averages were calculated - PTA 1 (averaged from 500 Hz, 1 kHz and 2 kHz thresholds) and PTA 2 (averaged from 1 kHz, 2 kHz and 4 kHz thresholds). The behavioral thresholds (PTA1 and PTA2) and ABR thresholds obtained at 30.1/sec repetition rate using click and chirp were correlated using Karl Pearson’s correlation test. For group III chirps evoked ABR obtained at 80 dBnHL at 11.1/s RR were discussed in terms of presence or absence of response. The chirp ABR
thresholds were correlated with behavioral thresholds. No statistical analysis was carried out. Morphology of ABR recorded using click and chirp ABR were discussed.

**Results and Discussion**

*Individuals with normal hearing*

Morphology of click and chirp evoked ABR varied with the type of the stimulus, repetition rates and level. From Figure 2 it can be observed that major peaks wave I, III and wave V were observed at higher intensity levels. When the intensity of both the click and chirp stimuli were changed to 40 dB SL the frequency of occurrence of earlier peaks wave I and III reduced. It was observed that for click stimuli at 40 dB SL wave III and wave V were the most frequently occurring peaks but for chirp stimuli at the same intensity level wave I and wave V were the most frequently occurring peaks. Near threshold levels for both click and chirp evoked ABR only wave V was observed.

![ABR Waveforms](image)

**Figure 2:** Shows click evoked ABR waveforms (left panel) and chirp evoked ABR waveforms (right panel) observed for different intensity levels at 30.1/sec repetition rate in one subject with normal hearing.

**Latency and amplitude measures**

The mean absolute latency values for click and chirp evoked ABR differed in individuals with normal hearing. Latency of the click and chirp evoked ABR wave I, III and V increased with decrease in intensity of the stimuli. Wave latencies also increased with increase in repetition rate for both the stimuli which can be seen in Table 1. The absolute latency of wave V was significantly different between the stimuli, intensity and repetition rate (p<0.05). Since wave III and I were not present in all the condition and
groups, main and interaction effects using ANOVA could not be carried out. Instead paired t-test was carried out to compare the significant difference between the rates for wave III and I latency and amplitude.

Table 1: Mean, SD and range for wave I, III and V latencies (in ms) of click and chirp evoked ABR at different intensities and repetition rates in individuals with normal hearing

<table>
<thead>
<tr>
<th>Repetition rate</th>
<th>Intensities</th>
<th>Click evoked ABR</th>
<th>Chirp evoked ABR</th>
</tr>
</thead>
<tbody>
<tr>
<td></td>
<td></td>
<td>Wave I</td>
<td>Wave III</td>
</tr>
<tr>
<td></td>
<td>80 dBnHL</td>
<td>Mean</td>
<td>1.67 (n=29)</td>
</tr>
<tr>
<td></td>
<td></td>
<td>SD</td>
<td>0.09</td>
</tr>
<tr>
<td>11.1/sec</td>
<td>40 dB SL</td>
<td>Range</td>
<td>1.55-1.90</td>
</tr>
<tr>
<td></td>
<td></td>
<td>Mean</td>
<td>2.01 (n=3)</td>
</tr>
<tr>
<td></td>
<td></td>
<td>SD</td>
<td>0.02</td>
</tr>
<tr>
<td></td>
<td></td>
<td>Range</td>
<td>2.00-2.05</td>
</tr>
<tr>
<td></td>
<td>80 dBnHL</td>
<td>Mean</td>
<td>1.72 (n=28)</td>
</tr>
<tr>
<td>30.1/sec</td>
<td>40 dB SL</td>
<td>SD</td>
<td>0.10</td>
</tr>
<tr>
<td></td>
<td></td>
<td>Range</td>
<td>1.50-2.00</td>
</tr>
<tr>
<td></td>
<td></td>
<td>Mean</td>
<td>2.43 (n=2)</td>
</tr>
<tr>
<td></td>
<td></td>
<td>SD</td>
<td>0.04</td>
</tr>
</tbody>
</table>

The mean latency for chirp evoked ABR wave V obtained at 80 dBnHL, 50 dBnHL and 30 dBnHL were computed at 30.1/sec repetition rate. The mean latency values were plotted as a function of intensities. It could be observed from the Figure 3 that as the intensity decreased the latency of chirp evoked ABR wave V increased.
From Table 2 it can be observed that there was a significant difference between 11.1/sec and 30.1/sec wave III and wave I latency for click stimulus and chirp stimulus. When stimulus latency values were compared between the type of stimuli at either 11.1/sec or 30.1/sec RR, significant difference was also observed at both the repetition rates between the stimuli.

Table 2: t – Values, degrees of freedom and significance level for wave (III & I) latency and amplitude in normal hearing individuals at 80 dBnHL

<table>
<thead>
<tr>
<th>Pairs compared (wave III)</th>
<th>latency</th>
<th>Amplitude</th>
</tr>
</thead>
<tbody>
<tr>
<td></td>
<td>t</td>
<td>df</td>
</tr>
<tr>
<td>CL at 11.1/sec - CL at 30.1/sec</td>
<td>4.225</td>
<td>29</td>
</tr>
<tr>
<td>CP at 11.1/sec - CP at 30.1/sec</td>
<td>2.221</td>
<td>21</td>
</tr>
<tr>
<td>CL at 11.1/sec – CP at 11.1/sec</td>
<td>67.229</td>
<td>22</td>
</tr>
<tr>
<td>CL at 30.1/sec – CP at 30.1/sec</td>
<td>41.050</td>
<td>24</td>
</tr>
</tbody>
</table>

<table>
<thead>
<tr>
<th>Pairs compared (wave I)</th>
<th>latency</th>
<th>Amplitude</th>
</tr>
</thead>
<tbody>
<tr>
<td></td>
<td>t</td>
<td>df</td>
</tr>
<tr>
<td>CL at 11.1/sec - CL at 30.1/sec</td>
<td>3.973</td>
<td>27</td>
</tr>
<tr>
<td>CP at 11.1/sec - CP at 30.1/sec</td>
<td>15.334</td>
<td>29</td>
</tr>
<tr>
<td>CL at 11.1/sec – CP at 11.1/sec</td>
<td>131.267</td>
<td>28</td>
</tr>
<tr>
<td>CL at 30.1/sec – CP at 30.1/sec</td>
<td>118.707</td>
<td>27</td>
</tr>
</tbody>
</table>

*Note: CL – click and CP – chirp*

The mean Peak to peak amplitude values for click and chirp evoked ABR responses did not vary between both the stimuli at higher intensity levels and higher repetition rates in individuals with normal hearing. But for 40 dB SL at 11.1/sec repetition rate the mean amplitude values of wave I and V for chirp stimuli was higher than the mean click amplitude values.
Chirp evoked ABR in HI individuals

Table 3: Mean, SD and range for wave I, III and V amplitude (in µv) of click and chirp evoked ABR at different intensities and repetition rates in individuals with normal hearing

<table>
<thead>
<tr>
<th>Repetition rate</th>
<th>Intensities</th>
<th>Click evoked ABR</th>
<th>Chirp evoked ABR</th>
</tr>
</thead>
<tbody>
<tr>
<td></td>
<td></td>
<td>Wave I</td>
<td>Wave III</td>
</tr>
<tr>
<td>11.1/sec</td>
<td>80 dBnHL</td>
<td></td>
<td></td>
</tr>
<tr>
<td>Mean</td>
<td>0.41</td>
<td>0.40</td>
<td>0.60</td>
</tr>
<tr>
<td>(n=29)</td>
<td>(n=30)</td>
<td>(n=30)</td>
<td>(n=30)</td>
</tr>
<tr>
<td>SD</td>
<td>0.16</td>
<td>0.21</td>
<td>0.22</td>
</tr>
<tr>
<td>Rang e</td>
<td>.13-.74</td>
<td>.13-.103</td>
<td>.29-.115</td>
</tr>
<tr>
<td>40 dB SL</td>
<td>Mean</td>
<td>0.15</td>
<td>0.15</td>
</tr>
<tr>
<td>(n=3)</td>
<td>(n=10)</td>
<td>(n=30)</td>
<td>(n=30)</td>
</tr>
<tr>
<td>SD</td>
<td>0.05</td>
<td>0.07</td>
<td>0.17</td>
</tr>
<tr>
<td>Rang e</td>
<td>.10-.21</td>
<td>.03-.25</td>
<td>.16-.85</td>
</tr>
<tr>
<td>30.1/sec</td>
<td>80 dBnHL</td>
<td></td>
<td></td>
</tr>
<tr>
<td>Mean</td>
<td>0.30</td>
<td>0.33</td>
<td>0.66</td>
</tr>
<tr>
<td>(n=28)</td>
<td>(n=30)</td>
<td>(n=30)</td>
<td>(n=30)</td>
</tr>
<tr>
<td>SD</td>
<td>0.14</td>
<td>0.10</td>
<td>0.22</td>
</tr>
<tr>
<td>Rang e</td>
<td>.05-.65</td>
<td>.13-.57</td>
<td>.32-.11</td>
</tr>
<tr>
<td>40 dB SL</td>
<td>Mean</td>
<td>0.10</td>
<td>0.23</td>
</tr>
<tr>
<td>(n=2)</td>
<td>(n=10)</td>
<td>(n=30)</td>
<td>(n=30)</td>
</tr>
<tr>
<td>SD</td>
<td>0.03</td>
<td>0.25</td>
<td>0.13</td>
</tr>
<tr>
<td>Rang e</td>
<td>.08-.13</td>
<td>.09-.90</td>
<td>.16-.62</td>
</tr>
</tbody>
</table>

From Table 3 it can be observed that as the intensity of the stimuli was varied from 80 dBnHL to 40 dB SL the mean amplitude of click and chirp ABR also decreased. The amplitude of click and chirp evoked ABR wave decreased with the increase in repetition rate. Wave V amplitude of click and chirp evoked ABR varied in normal hearing individuals. The repeated measures mixed ANOVA results did show no significant difference in amplitude between repetition rates and intensities in individuals with normal hearing (p > 0.05).

Results of paired t - test (Table 2) showed that there was no significant difference between 11.1/sec and 30.1/sec for wave III amplitude obtained either by chirp stimulus or click stimulus. But wave I amplitude were significantly different for click stimuli but not
for chirp stimuli between 11.1/sec and 30.1/sec repetition rates (Table 2). When wave III and wave I amplitude values were compared between the type of stimuli at either 11.1/sec or 30.1/sec RR there was significant difference observed at both the repetition rates except wave I click and chirp ABR amplitude at 30.1/sec repetition rate. Also wave I was consistently observed at and near 40 dB SL for normal hearing subjects.

**Individuals with sensory neural hearing loss**

In individuals with mild hearing loss the morphology of click and chirp evoked ABR varied. There was inter-subject variability observed in the presence or absence of earlier peaks (wave I and III). Morphology for both click and chirp evoked ABR in individuals with moderate hearing loss was poorer than individuals with mild hearing loss and normal hearing. The frequency of occurrence of earlier wave I and III were reduced with increase in degree of hearing loss. Wave V was the prominent peak observed even near the threshold levels.

**Latency and amplitude measures:**

Chirp evoked ABR latency and amplitude of wave I, III, V obtained at different intensities were calculated in individuals with mild and moderate sensory neural hearing loss. The mean latency values for chirp evoked wave V obtained at 80 dBnHL, 70 dBnHL and 50 dBnHL were computed for individuals with mild sensory neural hearing loss. For individuals with moderate sensory neural hearing loss the mean latency values were calculated at 90 dBnHL, 80 dBnHL and 60 dBnHL. Figure 4 shows the latency intensity functions for wave V in individuals with mild and moderate sensory neural hearing loss. It can be observed from Figure 4 that the latency increased with decrease in intensity for both the groups but the increase in latency was more for mild hearing loss group than normal hearing individuals and moderate hearing loss group. Wave V absolute latency was shorter in moderate than mild sensory neural hearing loss and the latencies varied with repetition rates and intensities within mild and moderate hearing loss and were significantly different from individuals with normal hearing (p < 0.05). Since individuals with mild and moderate sensory hearing loss had lesser frequency of occurrence of wave I and wave III paired t-test was not administered to compare the data for both click and chirp evoked ABR. The mean absolute latency values for click and chirp evoked ABR for all the peaks increased as the rate and intensity increased.

![Figure 4: Latency intensity functions for mild sensory neural hearing loss and moderate sensory neural hearing loss subjects for chirp evoked ABR wave V.](image-url)
Wave V amplitude was higher in individuals with mild hearing loss at 11.1/ sec than 30.1/sec repetition rates and also more for chirp evoked ABR. But in individuals with moderate hearing loss such differences between stimuli were not observed. However, the wave V, III & I amplitude values reduced with increase in repetition rates and decrease in intensity for individuals with mild and moderate sensory neural hearing loss. Wave III amplitude values were lower for individuals with mild and moderate sensory neural hearing loss for chirp stimuli than click stimuli. The amplitude differences between stimuli were almost similar but high variability was observed in amplitude within individuals with mild or moderate sensory neural hearing loss. It was also observed that the wave I amplitude in individuals with mild and moderate sensory neural hearing loss were consistently higher for chirp evoked ABR than click evoked ABR at all intensities and repetition rates.

**Between group comparisons**

The repeated measure ANOVA was done for click and chirp evoked ABR wave V latency and amplitude at different intensities and repetition rate within and across groups. Since the wave V was the most prominent peak observed in all the subjects at 80 dBnHL and at 40 dB SL intensities and at 11.1/sec and 30.1/sec repetition rate for both click and chirp stimuli a repeated measure mixed ANOVA [stimuli (2) X intensity (2) X repetition rate (2) X groups (3)] was applied to see the significant main effect. This analysis was carried out for both latency and amplitude of wave V separately.

**Latency**

Repeated measure mixed ANOVA results for latency values revealed a highly significant main effect for type of stimuli \[F (1, 47) = 8664.677, p < 0.01]\], intensities \[F (1, 47) = 55.624, p < 0.01\] and repetition rates \[F (1, 47) = 73.97, p < 0.01\]. Also latency values showed significant main effect \[F (2, 47) = 17.317, p < 0.01\].

A significant interactions between stimulus type and groups \[F (2, 47)=20.446, p< 0.01\], stimulus type and repetition rate \[F (1, 47)=15.597, p< 0.01\] and stimulus intensity and groups \[F (2, 47)=59.674, p< 0.01\] was also observed. However, significant interactions were not observed between stimulus type and intensities \[F (1, 47) = 1.377, p> 0.01\], repetition rates and groups \[F (2, 47) =0.015, p > 0.01\], intensities and repetition rates \[F (1, 47)=0.031 p> 0.01\]. Significant interaction for latency values were observed only for stimulus type, intensities and groups \[F (2, 47)=16.744, p< 0.01\]. No significant interactions were observed between stimulus type, repetition rates and groups \[F (2, 47) = 0.084, p > 0.01\], intensities, repetition rates and groups \[F (2, 47)=0.022, p> 0.01\] and stimulus types intensities and repetition rates \[F (1, 47)=0.412, p> 0.01\]. Interaction between stimulus types, intensities, repetition rates and groups were also statistically insignificant \[F (2, 47) =0.059, p > 0.01\]. Duncan’s post Hoc test was carried out between groups. From Table 4 it can be observed that there was a significant difference between the groups.
Table 4: Duncan’s post hoc test results for wave V latency across the group

<table>
<thead>
<tr>
<th>Groups</th>
<th>1</th>
<th>2</th>
<th>3</th>
</tr>
</thead>
<tbody>
<tr>
<td>Moderate hearing loss</td>
<td>10.4217</td>
<td></td>
<td></td>
</tr>
<tr>
<td>Mild hearing loss</td>
<td>10.7532</td>
<td></td>
<td></td>
</tr>
<tr>
<td>Normal hearing</td>
<td>11.0462</td>
<td></td>
<td></td>
</tr>
</tbody>
</table>

**Amplitude**

Repeated measures mixed ANOVA results for amplitude values revealed no significant main effect for the types of stimuli, intensities and repetition rates. Wave V amplitude for normal hearing group was consistently greater than mild and moderate sensory neural hearing loss group in all the conditions tested but the difference were not statistically significant (p > 0.05). For click and chirp stimuli as the repetition rate increased the amplitude of wave V decreased for both normal and hearing impaired group.

**The absolute latency of click and chirp evoked ABR differed significantly between groups.** Click ABR in individuals with normal hearing is usually dominated by the latency from high frequency regions and this activity phase cancels activity from apical, low frequency regions. But in individuals with cochlear hearing loss the activity from high frequency regions no longer phase cancels low frequency activity due to greater degree of loss in high frequency regions. Thus, the latency of click ABR will be reflecting the shift in domination of low frequency regions. As the hearing loss increases more activity is represented from low frequency regions thus the latency also increases with the increase in degree of hearing loss (Don and Kwong, 2005).

But for chirp evoked ABR, the wave V latency decreased with increase in hearing loss and was shortest for moderate hearing loss than for mild hearing loss and normal hearing group. This can be due to shorter cochlear response times in cochlear hearing loss subjects as reported by Don, Ponten, Eggermont and Kwong (1998). Cochlear filter buildup time is the time required to build up impulse response at the site of activation and depends on characteristic frequency, stimulus level and amount of hearing loss, but independent of gender. In cochlear hearing loss individuals the auditory filter becomes broadened thus the time required to build up an impulse response also decreases (Don et al. 1998). Since the response time is required to build up and impulse is reduced the time required for neural activation also decreases thereby decreasing the latency of response. So this can be reason another reason for getting earlier responses in chirp ABR with increase in degree of hearing loss. Thus chirp evoked ABR can be used as a useful indicator to reflect impaired cochlear processing in individual with sensory neural hearing loss.

The peak to peak amplitude values for click and chirp evoked ABR were not significantly different between the groups. But from the mean values individuals with normal hearing
showed higher amplitude values than individuals with mild or moderate hearing loss. This could be due to differences in cochlear processing for different types of stimuli and differences in individuals itself. There are no studies available in the literature in which they have compared amplitude for chirp stimuli between individuals with normal hearing and individuals with hearing impairment. Don et al. (1994) have reported that there are larger variations of amplitude in individuals with normal hearing using click evoked ABR. Thus, it concluded that larger variation in amplitude can be expected.

Within group comparisons

_Individuals with normal hearing_

Wave I, III and V was obtained for both click and chirp evoked ABR at 80 dBnHL levels. The results were in correlation with the study done Feobel and Dau (2004). As the intensity was reduced wave III and wave V were the most prominent peak in click evoked ABR but for chirp evoked ABR wave I and V were the most prominent peaks. Dau et al. (2000) have justified the presence of wave I at higher levels by upward spread of excitation where the basal region of the cochlea is excited by the low frequency energy of chirp when they are swept from low frequency to high frequency.

_The absolute latencies_ of click evoked ABR was shorter than chirp evoked ABR. This results were similar to the study done by Dau et al. (2000) where they has reported that these differences in absolute latencies are due to the differences in the duration of the stimuli. Generally latency of ABR is calculated from the onset of the stimuli thus if they are measured from the onset of the stimulus it is prolonged. When they are considered relative to the offset of the stimuli the latencies/ brainstem conduction time would remain same. In the present study the latency was measured from the onset of the stimuli hence the latency of chirp evoked ABR was longer than click evoked ABR.

_Peak to peak amplitude_ of click and chirp evoked ABR remained same at higher 80 dBnHL. The results were similar to the study done by Dau et al. (2000) and Wegner and Dau (2004). They have reported that chirp evoked ABR does not take the advantage of cochlear processing at higher intensity levels. There were no significant amplitude differences obtained between click and chirp ABR at equal at lower intensity levels which is contrary to the studies done by Dau et al. (2000); Wegner and Dau (2002) Feobel and Dau, (2004). This could be due to the transducers used in these studies were different and large number of subjects taken for the study and also due to some disadvantages of chirp evoked ABR which may lead to variation in chirp evoked ABR amplitude. Disadvantage is that there is significant variation from subject to subject in the cochlear response time between frequency regions. Thus this may cause amplitude differences between individuals or between cochlear regions within a given individual may reflect how well the chirp represents the true cochlear response times across and within individuals and not solely the amount of activation.
Individuals with sensory neural hearing loss

At 80 dBnHL the wave V was the prominent peak observed in all sensory neural hearing loss subjects. The frequency of occurrence of wave I and III were reduced and varied in individuals with mild and moderate sensory neural hearing loss. There is no information available in the literature where they have compared click and chirp evoked ABR in individuals with sensory neural hearing loss. Don and Kwong (2005) have reported that mid to high frequency cochlear hearing loss often results in poor or absent ABR wave I. Thus, due to hearing loss more in higher frequencies chirp evoked ABR earlier peaks could have been absent in the subjects with mild to moderate sensory neural hearing loss.

At 80 dBnHL wave V absolute latency of chirp evoked ABR were lesser in individuals with mild to moderate sensory neural hearing loss than normal hearing individuals. These latency differences could be due to impaired shorter cochlear response time which leads to decrease in latency in individuals with cochlear hearing loss (Don et al. 1998) which has been discussed earlier in group comparison.

Peak to peak amplitude of click ABR and chirp evoked ABR did not differ significantly in individuals with sensory neural hearing loss at 80 dBnHL. This can be due to neural saturation at higher amplitude levels as in normal hearing subjects. As the intensity of chirp stimuli was reduced the amplitude of chirp evoked ABR was also reduced as seen in individuals with normal hearing. The amplitude variations were higher in both the groups with hearing impairment. There were no significant amplitude differences between the stimuli. The amplitude variations within cochlear hearing impaired individuals can be due to impaired cochlear processing and variability in degree of phase cancellation taking place between higher frequency and low frequency regions. Also Wegner and Dau (2002) have reported that issue of cochlear response time varies from individual to individual. Thus the chirp might not match with cochlear response time with all the individuals. Thus this issue becomes problematic when impaired cochlear are assessed in which case cochlear filter characteristics vary as a function of the degree of damage.

Comparison of click and chirp evoked ABR thresholds with behavioral thresholds

To observe the relationship between the ABR threshold and behavioral threshold, click and chirp evoked ABR thresholds were obtained at 30.1/sec repetition rate.
Table 5: Mean, S.D and range for PTA 1, PTA 2, click and chirp evoked ABR thresholds obtained in different groups

<table>
<thead>
<tr>
<th></th>
<th>Normal hearing</th>
<th>Mild sensory neural hearing loss</th>
<th>Moderate sensory neural hearing loss</th>
</tr>
</thead>
<tbody>
<tr>
<td></td>
<td>Mean</td>
<td>S.D.</td>
<td>Range</td>
</tr>
<tr>
<td>PTA 1</td>
<td>7.25</td>
<td>3.94</td>
<td>0-15</td>
</tr>
<tr>
<td>PTA 2</td>
<td>6.64</td>
<td>4.02</td>
<td>0-15</td>
</tr>
<tr>
<td>Click - thresholds</td>
<td>19.8</td>
<td>5.79</td>
<td>10-35</td>
</tr>
<tr>
<td>Click - thresholds</td>
<td>19.0</td>
<td>5.47</td>
<td>10-30</td>
</tr>
</tbody>
</table>

The pure tone averages (PTA 1 and PTA 2) were correlated with click and chirp evoked ABR threshold. From Table 5 it can be observed that the click and chirp evoked ABR thresholds were obtained 15 – 20 dB above the behavioral thresholds in individuals with normal hearing. Whereas in individuals with mild and moderate hearing loss the click and chirp ABR thresholds were closer to their pure tone averages.

Table 6: Karl Pearson’s correlation coefficient results observed between PTA 1, PTA 2, click and chirp evoked ABR thresholds

<table>
<thead>
<tr>
<th></th>
<th>PTA 2</th>
<th>Click thresholds</th>
<th>Chirp thresholds</th>
</tr>
</thead>
<tbody>
<tr>
<td>PTA 1</td>
<td>.982**</td>
<td>.923**</td>
<td>.879**</td>
</tr>
<tr>
<td>PTA 2</td>
<td></td>
<td>.916**</td>
<td>.864**</td>
</tr>
<tr>
<td>Click dBnHL</td>
<td></td>
<td></td>
<td>.912**</td>
</tr>
</tbody>
</table>

** p < 0.01

To find out the correlation between PTA 1, PTA 2 with click and chirp evoked ABR thresholds Karl – Pearson correlation was applied. It can be observed from the Table 6 that both click and chirp ABR were significantly correlating with behavioral threshold (PTA1 & PTA2) with having high positive correlation between them. Since there was good correlation between click and chirp evoked ABR with the behavioral pure tone averages the study further compared the difference between click and chirp evoked ABR in individuals with normal hearing, mild and moderate sensory neural hearing loss.
Table 7: t - test values with significance level between of click and chirp ABR thresholds for different groups

<table>
<thead>
<tr>
<th>Groups</th>
<th>t - values</th>
<th>df</th>
<th>Significance level</th>
</tr>
</thead>
<tbody>
<tr>
<td>Normal hearing</td>
<td>.841</td>
<td>29</td>
<td>.407</td>
</tr>
<tr>
<td>Mild sensory neural hearing loss</td>
<td>.170</td>
<td>8</td>
<td>.870</td>
</tr>
<tr>
<td>Moderate sensory neural hearing loss</td>
<td>3.963</td>
<td>10</td>
<td>.003</td>
</tr>
</tbody>
</table>

Paired t - test was applied to the data to see whether there is any significant difference between the both the click and chirp evoked ABR thresholds in individuals with normal hearing, mild and moderate sensory neural hearing loss. From the Table 7 it can be observed that significant difference between click evoked ABR thresholds and chirp evoked ABR thresholds were obtained only in individuals with moderate sensory neural hearing loss and chirp evoked ABR thresholds being better in moderate sensory neural hearing loss. Thus the chirp evoked ABR was better than click evoked ABR thresholds at higher degree of hearing loss.

There are hardly any studies to state that chirp evoked ABR thresholds are better than click evoked ABR thresholds in individuals with normal hearing and sensory neural hearing loss. Most of the studies done with chirp evoked ABR have compared the amplitude of chirp evoked ABR with click evoked ABR at equal sensation levels. The differences obtained between click and chirp evoked ABR thresholds could be due to the configuration of hearing loss. Most of the subjects considered in the study had almost flat type of configuration and the differences between PTA 1 and PTA 2 were within 10 dB for individuals with mild hearing loss and within 15 dB for individuals for moderate hearing loss individuals. Since the difference between pure tone averages were greater for moderate hearing loss this differences could have lead to the differences seen in click and chirp evoked ABR threshold. Thus, from the correlation analysis it can be concluded that like click evoked ABR, chirp evoked ABR could be also used in threshold estimation and can estimate thresholds closely to behavioral thresholds in individuals with higher degree of hearing loss. However, further research in this line is required to confirm this finding.

Comparison of click and chirp evoked ABR in individuals with auditory neuropathy:

Out of 10 ears tested 3 ears showed click ABR responses at 80 dBnHL, whereas, 4 ears out of 10 ears had chirp evoked ABR responses at 80 dBnHL. Subjects who had click ABR also had chirp evoked ABR. However, those who did not have click evoked ABR also did not have ABR for chirp except one ear. Those who had ABR for click and chirp, morphology was poor for both the stimuli. Only wave V could be identified irrespective of severity of hearing loss. However, wave V latency for chirp evoked wave V was much longer in auditory neuropathy than that was observed with individuals with normal hearing and sensory neural hearing loss.
Table 8 shows the mean absolute latency and mean peak to peak amplitude for click and chirp evoked ABR responses wave V obtained at 11.1/sec repetition rate. When peak to peak amplitude was compared across the stimuli the chirp ABR had higher amplitudes compared to click evoked ABR. So paired t-test was administered to the see the significant differences between them. Paired t-test result showed no significant peak to peak amplitude difference between click and chirp evoked ABR \([t, (2) = 3.024, p > 0.05]\). Since, 4 ears of auditory neuropathy subjects had identifiable wave V at 80 dBnHL, chirp evoked ABR was recorded at lower intensity levels for threshold estimation. When intensity of chirp stimuli was reduced to 70 dBnHL detectable chirp ABR wave V was observed for 3 ears out of 4 ears. However, when the intensity was further reduced to 60dBnHL there were no responses for any of these subjects. But for click ABR wave was absent when click intensity was reduced by 10 dB.

Table 8: Mean and S.D for click and chirp evoked ABR wave V latency (ms) and amplitude (µv) obtained from individuals with auditory neuropathy

<table>
<thead>
<tr>
<th>Intensities</th>
<th>Auditory neuropathy</th>
<th>Latency</th>
<th>amplitude</th>
</tr>
</thead>
<tbody>
<tr>
<td></td>
<td>Click ABR</td>
<td>Chirp ABR</td>
<td>Click ABR</td>
</tr>
<tr>
<td>80 dBnHL</td>
<td>Mean (n=3)</td>
<td>6.68</td>
<td>0.27</td>
</tr>
<tr>
<td></td>
<td>SD</td>
<td>1.05</td>
<td>0.07</td>
</tr>
<tr>
<td>70 dBnHL</td>
<td>Mean (n=3)</td>
<td>No response</td>
<td>17.11</td>
</tr>
<tr>
<td></td>
<td>SD</td>
<td>0.22</td>
<td>0.10</td>
</tr>
</tbody>
</table>

It can be concluded from the above results that the chirp and click evoked ABR latency values for chirp ABR were prolonged compared to normal hearing ears. There are no studies available in literature using chirp evoked ABR in individuals with auditory dysynchrony. Since chirp ABR evokes synchronous firing along the cochlea it was expected to obtain better ABR responses with chirp stimuli. Even though cochlear outer hair cells are normal in auditory neuropathy they are not able to evoke significant synchronous activity in the auditory nerve with the compensation of basilar membrane delay differences between high and low frequencies. From chirp evoked threshold comparisons it can be concluded that since 3 ears have got chirp evoked ABR thresholds at lower intensities than the click evoked ABR thresholds and 1 ear have got chirp evoked ABR in the absence of click evoked ABR, chirp evoked ABR could be used for threshold estimation in auditory neuropathy. This would in turn give a better approximation to the behavioral threshold in individuals with auditory neuropathy.
Conclusions

It can be concluded from the study that the chirp evoked ABR can be used clinically for threshold estimation in individuals with normal hearing and cochlear hearing loss and auditory neuropathy. It can estimate more precise behavioral thresholds in individuals with higher degree of hearing loss and up to certain extent in individuals with auditory dysmorphism. It can also be used to study the cochlear processing such as cochlear transport time and cochlear filter responses. The chirp evoked ABR cannot be used for neurodiagnosis due to less frequency of occurrence of wave III. ABR wave I present till lower level could be of particular interest for future studies.

References


