PERFORMANCE OF HEARING AIDS WITH ARTIFICIAL INTELLIGENCE.

Register No. A0390014.

A dissertation submitted in part fulfillment for the degree of

Master of Science (Audiology),

University of Mysore, Mysore.

ALL INDIA INSTITUTE OF SPEECH AND HEARING.

Mansagangothri. Mysore. 570006.

May 2005.

In memory of my grandfather

Biswanath Panda,

for his knowledge, his illustration and his lessons.

CERTIFICATE

This is to certify that this Master's dissertation entitled *"Performance of hearing aids with artificial intelligence"* is the bonafide work done in part fulfillment of the degree of Master of Science (Audiology) of the student with register number: A0390014. This has been carried out under the guidance of a faculty of this institute and has not been submitted earlier to any other university for the award of any diploma or degree.

M. Jayaram

Prof. M. Jayaram.

Director.

All India Institute of

Speech and Hearing.

Mysore-570006.

Mysore.

May, 2005.

CERTIFICATE

This is to certify that this Master's dissertation entitled "*Performance of hearing aids with artificial intelligence*" has been prepared under my supervision and guidance. It also certified that this has not been submitted in any other university for the award of any diploma or degree.

Prof. M. Jayaram.

Director.

All India Institute of

Speech and Hearing.

Mysore-570006.

Mysore.

May, 2005.

DECLARATION

This is to certify that this Master's Dissertation entitled "*Performance of hearing aids with artificial intelligence*" is the result of my own study under the guidance of Prof. M. Jayaram, Director, All India Institute of Speech and Hearing, Mysore, and has not been submitted earlier at any University for any other diploma or degree.

Mysore.

May, 2005. Reg. No. A0390014.

ACKNOWELDGEMENTS

The list of my debt is long and endless; a happy circumstance. Some debts are so great that they cannot go unacknowledged. I have drawn on the stores of my Mentor's expertise, valued the knowledge of many others, and finally, attempted to consolidate own ideas and discernment for furtherance of the cause. I found all the gates open for me to render whatever assistance, guidance and support I needed. Much of what I have learned about the research problem has been learned from my mentor, whose ability to glimpse through to the essence of a technical problem and state it simply made many a solution appear self evident. I am indebted to my mentor Prof. M. Jayaram, Director, AIISH for permitting and guiding the project idea in every possible aspect, and for placing his trust on me. His tolerant research direction and keen insight into multidimensional hearing aid problem created a milieu eminently conducive to problem solving. Importantly, despite his demanding schedule, he provided calm and well directional advice without which I would undoubtedly not have surmounted the hurdles of the dissertation.

I feel obliged to document certain facts where trite words of gratitude would sound frolicsome. Dr. Carmen Taylor, as ever, provided me both audiological insights and well-honed editorial criticisms. I do not know how to enunciate the combined sense of my complacence and indebtedness to her endless blessings. Many facets of the hearing aid research relevant to this project became clear during the interesting rather lovely visits to House Ear Institute, California, and American Auditory Society, Arizona. An arresting fact is that, I would not have ventured to work on this particular project had not, I had received precious support and information almost every week from Dr. Mark Flynn. A host of international experts has generously contributed to my perspicacity of the research problem. I am appreciative to all of them. I also acknowledge the invaluable support I had received from Drs. K. Rajalakshmi, Anirban Biswas, and Mr. Ajish Abraham. Equally grateful, I am to the HOD and faculty members of the Audiology dept. who taught me lessons in audiology. No less thankful I am to the staffs of the Director's office, who did their best in many ways to let me accomplish the study. A note of special thanks to Dr. Venkatesan, Ms. Vasanthalaxmi, Mrs. Chudamani and Mr. Puroshotham for various technical advice and assistance.

This project is supported by two independent research grants from Oticon A/S, Denmark, and The William Demant & Wife Ida Emilie (Oticon) Foundation, Denmark. My sincere thanks to the generosity of these organizations. I take this opportunity in expressing my honest regards to the respective unanimous reviewers for my grant application, including Drs. Claus Elberling and Graham Naylor.

I do appreciate and admire all the participants for their cooperative attitude, patience and endurance with the various tests. I energetically recall and owe it to all my good friends, colleagues, juniors and seniors at AIISH, ERC and JNVN for keeping a very nice environment at all the times. Finally, at times when they might reasonably have expected to share in the fruits of a successful student career, my parents and relatives sacrificed an unreasonable amount so I could go back to school one more time. All of these debts can only be acknowledged; many of them can never be repaid.

CONTENTS

LIST OF TABLES

LIST OF FIGURES

CHAPTER 1

INTRODUCTION

Diagnostic tools continue to be developed, hearing aid circuitry continues to become more sophisticated and advanced, the hearing-impaired population continues to grow, and one step ahead, the speech perception in background noise by hearing aid users continues to be more dreadful! Census of India (2001) report indicates that 1,261,722 persons have hearing impairment. A similar report from the NIDCD (2001) suggests that over 29 million individuals in the U.S.A exhibit some degree of hearing impairment. A major consequence of even a mild sensorineural hearing loss is difficulty in understanding speech in noise and/ or reverberating conditions. In fact, this is one of the major reasons for dissatisfaction with, and rejection of hearing aids. Three 'MarkeTrak' studies by Kochkin (2001, 2002a, b) found that:

1. 80% of those who admit they need hearing aids don't buy them,
2. 40% of those who do buy hearing aids do not place themselves in the "satisfied" category,
3. better understanding of speech in noise is the most sought after improvement sought by those wearing hearing aids, and
4. improved sound quality is the second largest improvement sought by those wearing hearing aids.

Early research comparing performance of listeners with normal hearing to listeners with hearing loss have indicated that persons with hearing impairment need a better signal-to-noise ratio (SNR) than those with normal hearing (e.g., Davis, 1947; Plomp, 1978). Although reduced audibility is the most obvious explanation for the reduced speech recognition and discrimination ability, problems in the realm of frequency selectivity and temporal resolution have been implicated as well (e.g., Florentine, Buss, Scharf and Zwicker, 1980; Gelfand, 1982; Healy and Baycon, 2002). Even as the audibility deficit is resolved with the use of hearing aid amplification, the listener's reduced performance is due to the masking effects of noise (Boothroyd, 1993; Tillman, Carhart and Olsen, 1970). More specifically, Tillman, Carhart and Olsen (1970) noted a 30 dB disparity between signal-to-noise ratios necessary for 40% intelligibility of monosyllabic words embedded in competing sentence "noise" for their subjects with normal and impaired hearing. Following several studies in this area, Plomp (1978) concluded that no hearing aid could improve SNR to the levels enjoyed by normal-hearing listeners. Killion (1997) has provided data to suggest that listeners with minimal hearing loss (30 dB HL) can expect a 4 dB deficit in SNR performance, with another 1dB added for each 10 dB increase in hearing loss. His findings were based on a compilation of studies assessing SNR for 50% performance using the Speech-in-Noise (SIN) test. He noted that these estimates may be conservative, with an additional factor of distortion effects having an impact at higher-level inputs.

Artificial Intelligence and hearing aids: Sequential versus Parallel Processing

When digital signal processing (DSP) technology was introduced in hearing aids in the year 1996, it represented a spectacular shift in the core technology of amplification devices. However, no longer is "digital" synonymous with "premium". It no longer defines the highest level of hearing aid technology. It has rather become the minimum customary platform that all modern hearing aids use. Now we need to maximize the capabilities of digital technology as it applies to human auditory perception, that is, hearing. This is where artificial intelligence comes in. Artificial intelligence is the technology of simulating intelligent behavior and decision-making power in computers. It is essentially a parallel processing technology. It is akin to the task of using computers to understand human intelligence, but, artificial intelligence does not have to confine itself to methods that are biologically observable. One of the most common applications is to process large amounts of information and, by either deducing or following rules, determine an appropriate response to this input. Very recently, artificial intelligence has been incorporated in one of the commercially available hearing aids which analyzes all the responses in parallel and selects the best response. The overall purpose of the system is to progressively optimize the signal for human perception or in essence to clean the noise out of the signal and thereby maximize speech understanding (Flynn, 2004).

The processing of speech in noise to provide the optimum speech signal combined with appropriate noise management is one of the greatest challenges facing manufacturers of premium digital hearing instruments. The challenge when designing advanced hearing instruments is to select and manage the best potential of various technologies like adaptive directionality or noise reduction. It is crucial that decisions and processing must be made quickly. If the digital processing delay is greater than 10ms, then the user is likely to begin to hear echoes and other distortions (Stone and Moore, 1999, 2002). This need for fast processing has heretofore prohibited the use of parallel processing – where all outcomes are evaluated for the best solution. Instead, hearing instruments have relied on sequential processing combined with a comprehensive prediction model to select the preferred processing option (Gabriel, 2002; Kates, 1995). Because sequential processing does not actually compare different outcomes, an incorrect solution may be the result due to the unpredictability of complex communication environments. Parallel processing, however, provides significant advantage of not relying on predictive models, but instead relies on processing and comparing each outcome for the best solution. In fact, parallel processing is a prerequisite for artificial intelligence that allows multiple processing schemes to be evaluated simultaneously to ensure that the best solution is implemented. Use of artificial intelligence avoids the problem of trying to use complex models to predict the best response to the environment (Flynn, 2004). The problem with prediction-based models is that they attempt to narrow the multiplicity of communication environments into a restrictive prediction-based formula. While the use of prediction works well in the laboratory, it does not always work as effectively in real-world communication environments; it has been shown that the prediction model may select incorrect settings as often as 30% of the time (Gabriel, 2002).

A number of stimulating philosophical questions may emerge concerning artificial intelligence in hearing aids. Can machines be truly intelligent? What is the level of consciousness? Is artificial intelligence the same as "thinking"? Does it have the decision-making power equivalent to that of human brain? Honestly, we do not have answers to any such questions now!

Improving speech intelligibility through hearing aids in background noise is the most important concern of all hearing aid manufacturers and researchers. Various attempts have been made, in the realm of signal processing schemes, to enhance the speech signal in the presence of background noise for the hearing aid user. Amplitude compression with a variety of time constants, multichannel signal processing, and schemes such as consonant-vowel-consonant (CVC) enhancement (Hickson and Byrne, 1997) and speech-rate slowing (Nejime and Moore, 1998) have been evaluated in an effort to find some strategy that can successfully restore hearing ability. Additionally, analog and digital noise reduction schemes have provided “easier listening”, but have not resulted in anticipated levels of improvement for the hearing aid wearer (e.g., Alcantara, Moore, Kuhnel, and Launer, 2003; Bentler, Anderson, Niebuhr and Gette, 1993; Walden, Surr, Cord, Edward and Olson 2000). It is, therefore, not at all surprising that for many years now, the primary focus of much of research and development in hearing aids has been the development, and evaluation of technologies, features and algorithms intending to improve aided speech perception in noise.

Recently, Oticon, a hearing aid manufacturer based in Europe have incorporated the artificial intelligence technology into one of their digital hearing aids (Oticon Syncro). This hearing aid is built on a new digital platform that implements multiband adaptive directionality, noise management, and wide dynamic range compression. Artificial intelligence is the foundation of the voice priority processing (VPP) system, which oversees three signal-processing approaches: multiband adaptive directionality, ‘TriState’ noise management, and voice-aligned compression. All these systems are designed to work in synergy to optimize the signal progressively, with the priority being to supply the best possible speech understanding. The unity of this signal-processing goal, combined with decision making through parallel processing, is intended to ensure that correct decisions are being made and that all systems are working toward improving speech understanding in noise.

Two important assumptions underlying the use of artificial intelligence in hearing aids are; (i) evaluation of all outcomes simultaneously and selection of the most appropriate one (i.e., improved response to a potentially time-varying environment) and (ii) theoretically no or very less processing delay because of parallel processing. The first would improve speech perception in noise, and the second would lead to natural sound quality of the hearing aid processed speech. Eventually, hearing aid users urgently and badly require these two benefits as found by Kochkin (2001, 2002a, b).

Statement of the problem

The purpose of this study was to evaluate the performance of a hearing aid with artificial intelligence in both laboratory and field conditions. The aim was to see if hearing aids with artificial intelligence lead to speech perception as good as normal in hearing impaired listeners, as well as to compare hearing aids with artificial intelligence with other digital hearing aids.

Specifically, the performance of hearing aids with artificial intelligence was compared with normal perceptions in laboratory (under simulated conditions of alternating talkers, cutlery noise, speech babble noise and traffic noise), and in field conditions. Also, the performance of subjects wearing hearing aids with artificial intelligence was compared with other digital hearing aids under laboratory conditions.

Research goals

The purpose of the present study was to answer the following questions:

1. Can adults with sensorineural hearing loss, wearing hearing aids with artificial intelligence, understand speech-in-noise as well as normal hearing adults?
2. Can adults with sensorineural hearing loss wearing hearing aids with artificial intelligence, understand speech-in-noise as well as hearing impaired wearing other hearing aids?
3. Whether different listening conditions like cutlery noise, speech babble noise and traffic noise influence understanding of speech with these two types of hearing aids?
4. Is there a difference in acoustic comfort and sound quality between the hearing aid with artificial intelligence and other digital hearing aids used by hearing impaired individuals?
5. What is the performance of the hearing aid with artificial intelligence in real-life listening situations?

Need for the study

The introduction of artificial intelligence in hearing aids seems to be one of the most advanced hearing aid signal processing strategies. One would, expect from such a system to hold the most potential for enhancing speech perception in noisy situations. However, Killion (2004b) pointed that significant technology improvements in hearing aids may not necessarily result in consistent increase in expressed satisfaction by hearing aid users. Therefore, with any new hearing instrument, it is important to compare and benchmark its performance against that of other premium hearing aids to identify what critical differences people with hearing loss may observe in technology levels. There are no studies evaluating the efficacy of artificial intelligence in hearing aids, except a few reports from the manufacturers of such hearing aids (Flynn, 2004; Flynn and Lunner, 2005; Schum, 2004). These concerns demonstrate the strongest need in favor of an independent evaluation of this new technology. In fact, hearing aid industry, researchers, clinicians and consumers have always welcomed research reports, though for different reasons, evaluating new technologies that may improve hearing of speech in noise.

CHAPTER 2

REVIEW OF LITERATURE

Presently various signal-processing strategies (such as, multichannel compression, digital noise cancellation/ management, frequency modulation, and directional microphones, etc.) have been found to improve listening in noise by varying degrees. The following section provides a brief, yet comprehensive and most pertinent review of the extent of benefits and limitations of such features as reported in literature.

Multichannel compression

Yund and Buckles (1995a) used full-range multichannel compression hearing aids (MCCHAs) with 4, 6, 8, 12, and 16 independent frequency channels to determine the effect of the number of channels on speech discrimination with mild to moderately severe hearing-impaired subjects. Signal-to-noise ratios (SNR) from -5 to 15 dB with speech-spectrum noise (70 dB SPL) and two voices (male and female) were used. Average speech discrimination for 16 hearing-impaired subjects increased from 4 to 8 channels but did not change significantly between 8 and 16 channels. The effect of the number of channels did not vary significantly with SNR. Results indicated that a MCCHA with at least 8 (and up to 16) channels provides the mild to moderately severe hearing-impaired subjects with acoustic information that facilitates speech discrimination in speech-band noise. In a subsequent study, the same researchers found that an eight-channel MCCHA causes little information degradation to individuals with mild to moderately severe hearing loss and can be of great benefit for speech discrimination in noise, particularly at low SNRs (Yund and Buckles 1995b). In contrast, Keidser and Grant (2001) found laboratory tests showed no significant difference in speech recognition scores in noise across channels. Most subjects showed no preference for either scheme in the paired-comparison test. In the field, all subjects with a steeply sloping loss preferred the 2-channel scheme, except one subject. Most subjects with a flat loss preferred the single-channel scheme over the 2-channel scheme. Multichannel compression, prescribed according to NAL-NL1 in up to four channels, showed no adverse effects on speech recognition relative to a single-channel scheme. The field test revealed a preference for the 2-channel scheme by subjects with steeply sloping loss.

Digital noise cancellation/ management

Valente, Fabry and Potts (1995) in a multi- centered study with fifty subjects having mild to moderately severe sensorineural hearing loss measured speech recognition in noise using the Hearing in Noise Test (HINT) between two fitting algorithms ("basic" and "party") and two microphone conditions (single microphone omni-directional and dual-microphone directional). Results revealed an average improvement in SNRs of 7.4 to 8.5 dB at the two sites for the directional conditions in comparison to the omni-directional conditions. No significant improvement in SNR was measured between the two fitting algorithms. Alcantara, Moore, Kuhnel and Launer (2003) also found similar results. They evaluated the effectiveness of a noise reduction system implemented in a commercial digital multichannel compression hearing aid in eight-experienced hearing aid wearers with moderate sensorineural hearing loss. Speech recognition thresholds (SRTs) were measured in four types of background noise, including steady noise, and noises with spectral and/or temporal dips. SRTs were very similar with and without the noise reduction system; in both cases, SRTs were markedly lower than for unaided listening. SRTs were lower for noises with dips than for steady noise, especially in the aided conditions, indicating that amplification can help to 'listen in the dips'. Ratings of sound quality and listening comfort in the aided conditions were uniformly high and very similar with and without the noise reduction system.

Directional microphones

Ricketts and Henry (2002) assessed the effectiveness of adaptive directional processing for improvement of speech recognition in comparison to non-adaptive directional and omni-directional processing across different listening environments intended to simulate those found in the real world. HINT and Connected Speech Test (CST) were administered in all listening conditions on twenty listeners fitted binaurally with BTE style hearing aids. Results indicated improved speech recognition performance with adaptive and non-adaptive directional processing over omni-directional processing hearing aids across all listening conditions. While the magnitude of directional benefit in adaptive and fixed directional modes were similar in some listening environments, a significant speech recognition advantage was measured for the adaptive mode when a competing noise was presented from the listener's sides (both fixed and panning noise conditions), and was partially predictable from electroacoustically measured directional pattern data. Luts, Maj, Soede and Wouters (2004) evaluated the improvement in speech intelligibility in noise obtained with an assistive real-time fixed end-fire array of bidirectional microphones in comparison with an omni-directional hearing aid microphone in a realistic environment. Results indicated that improvements in speech intelligibility in noise obtained with array of microphones relative to an omni-directional microphone depend on noise scenario and subject group. Improvements of up to 12 dB for normal-hearing and 9 dB for hearing-impaired listeners were obtained with three active array microphones relative to an omni-directional microphone.

Recently, Bentler, Palmer and Dittberner (2004) compared hearing-in-noise between forty- eight normal hearing and forty- six hearing-impaired listeners. The results indicated that when the noise around a listener was stationary, a first- or second-order directional microphone allowed a group of mild-to-moderate hearing impaired listeners to perform similarly to normal hearing listeners on a speech-in-noise task (i.e., they required the same SNR to achieve 50% understanding). With moving noise source, only the second-order (three-microphone) system set to an adaptive directional response allowed a group of hearing impaired individuals to perform similarly to normal hearing subjects.

Arguably, performance with directional microphones is significantly better than omni-directional microphone performance. However, the extent of benefits of directional microphones found in laboratory studies depends on the number and azimuths of noise sources and signal, and the nature of reverberating conditions. Ricketts (2000) evaluated the impact of the position of noise source(s) and reverberation on the directional benefit and performance of three commercially available directional hearing aids in twenty-five subjects with symmetrical sloping sensorineural hearing loss using a modified version of the HINT. Findings suggest that both reverberation and configuration of the competing noise source(s) significantly affected directional benefit and performance. There was no significant correlation between directional benefit and directional performance. Data revealed that increased reverberation significantly decreased directional benefit and performance. Results also suggested that data collected in traditional test environments (e.g., source of a single competing noise placed at 180 degrees azimuth) cannot be used to accurately predict directional benefit or performance in the majority of other test and real-world environments.

FM hearing aids

Lewis, Crandell, Valente and Horn (2004) studied speech perception in forty-six adult subjects with slight to severe sensorineural hearing loss using HINT with correlated diffuse noise under five different listening conditions. Results revealed that speech perception was significantly better with the use of the FM system over any of the other hearing aid conditions, even with the use of the directional microphone. Additionally, speech perception was significantly better with the use of two hearing aids used in conjunction with two FM receivers than with just one FM receiver. Similarly, Boothroyd (2004) found that the expected benefits of a remote FM microphone in reducing the negative effects of distance and noise, for a single talker, can be demonstrated under both laboratory and field conditions. The effects of hearing loss, noise and FM assistance, on aided phoneme recognition, are well predicted by methods derived from Articulation Index theory. Considerable counseling, instruction and coaching is needed, however, to ensure optimal use of this technology.

Cumulative effects

There are only a few studies which have evaluated the cumulative effect of all the noise management features. Pumford, Seewald, Scollie and Jenstad (2000) compared the overall listening benefit in diffuse noise provided by dual-microphone technology in an in-the-ear (ITE) hearing instrument to that provided by dual-microphone technology in a behind-the-ear (BTE) hearing instrument in twenty-four adults with mild to moderately severe sensorineural hearing loss. Further, the study determined whether the use of the dual-microphone and the manufacturer's party response algorithm together in ITE and BTE hearing instruments provided listening benefit under diffuse noise over their respective omni-directional microphone modes. The results indicated that the dual-microphone and party response mode did provide significant benefit in diffuse noise for both the ITE [3.27 dB signal-to-noise ratio (SNR) improvement] and BTE (5.77 dB SNR) hearing instruments relative to their respective conventional omni-directional microphones. No significant difference in performance was found between ITE and BTE hearing instruments when each device was in the dual-microphone and party response mode.

In a field study, Boymans and Dreschler (2000) measured the effects of a digital hearing aid on speech recognition or reception in noise for two noise reduction concepts: active noise reduction by speech-sensitive processing (SSP) and improved directionality by a dual- or so-called twin-microphone system (TMS). A well-controlled clinical field trial was conducted on sixteen hearing-aid users, using a single-blind crossover design with speech recognition tests in background noise, paired comparisons, and self-report measurements (questionnaires) was used as performance indicators. The objective and subjective tests showed the same trend in performance. The effects of TMS were clearly positive, especially for the speech reception threshold tests and for the paired comparisons. The effects of SSP were much smaller, but showed significant benefits with respect to aversiveness and speech perception or reception in noise for specific acoustical environments. There was neither extra benefit nor degradation from the combined effect of SSP and TMS relative to TMS alone.

Laboratory testing versus real- world performance

Unfortunately, there is poor correlation between laboratory and real world performance with directional microphones. Cord, Surr, Walden and Dyrlund (2004) examined whether persons who were successful users of directional microphone hearing aids in everyday living tended to obtain a larger directional advantage in the test booth than persons who were unsuccessful users. Results revealed that the mean directional advantage in the laboratory set up did not differ significantly between patients who used the directional mode regularly and those who reported little or no benefit from directional microphones in daily living. Therefore, success with directional microphone hearing aids in everyday living, cannot be reliably predicted by the magnitude of the directional advantage obtained in the laboratory.

Some observations

Multichannel compression neither provides any significant benefit over single-channel compression in listening to speech in noise, nor does it degrade speech perception. Digital noise cancellation/ management may provide acoustic comfort in noisy environments, but it may also, at times, degrade speech intelligibility depending on the algorithm. Adaptive directional microphones provide a certain advantage over non-adaptive and omni-directional microphones to improve SNR. However, real world performance cannot be predicted from laboratory measures for all subjects. While adaptive directionality systems are impressive, they have a number of limitations. First, due to increased microphone noise they require relatively large input level to switch to the directional mode. Second, directional microphones are susceptible to wind noise (Thompson, 2003; Valente and Mispagel, 2004). Third, they cannot cancel multiple independent noise sources simultaneously as they only have one polar response across all frequencies (Thompson, 2003). Fourth, the selection of directional versus omni-directional mode is based on a prediction of which mode will provide the best solution rather than making intelligent decisions based on which actually provides the best signal. FM hearing aids provide improved speech perception in noise, but the subject may require some training to derive optimal benefit. In addition, the relative gains via FM and hearing aid microphones must be adjusted with care. Cumulative effects of different features on speech in noise performance were not extensively investigated.

Artificial intelligence in hearing aids

Recently in 2004, Oticon has introduced artificial intelligence technology into one of their digital hearing aids (Oticon Syncro). The unique features of this hearing aid are;

- Voice priority processing aims to provide the best possible SNR to the person with hearing impairment through optimal amplification of speech and management of noise. This is achieved by the combination of multiband adaptive directionality, 'TriState' noise reduction and voice-aligned compression.
- With multiband adaptive directionality, the system can remove up to four sources of noise simultaneously.

- ‘TriState’ noise management combines ‘VoiceFinder’ technology with a new modulation-based noise reduction system to provide maximum comfort and speech understanding.
- Voice aligned compression is built on a new platform of eight independent channels of compression across an expanded bandwidth. The amplification itself provides curvilinear compression comprising up to seven knee points to ensure a smooth frequency response at all input levels.
- Underlying these three systems is the decision making power provided by parallel processing which enables every possible configuration of the hearing aid’s systems to be evaluated prior to implementation so that the best solution can be selected. This parallel processing is the most robust method of evaluating and reacting to complex communication environments that rapidly change and cannot be predicted.

There are no studies conducted on the efficacy of artificial intelligence in hearing aids to improve understanding of speech. Hearing aid industry, audiologists and consumers have always welcomed research reports evaluating new technologies that may improve hearing of speech in noise. This concern demonstrates the strongest motivation for the present study.

CHAPTER 3

METHOD

The purpose of this study was to evaluate the performance of a hearing aid with artificial intelligence (Oticon Syncro BTE) on normal individuals and persons with sensorineural hearing loss. The performance of the hearing aid was evaluated in both laboratory and real-life situations.

(A) LABORATORY TESTING

Subjects

Experimental group

Twenty persons (4 females and 16 males) with mild-to- moderate sensorineural hearing impairment and an average age of 43.5 years (range 23 to 50 years) served as experimental subjects. They were native speakers of Kannada language. The subjects were all contended users of binaural digital BTE hearing aids from different manufacturers. The subjects were users of hearing aids for 7 months to 6 years (mean 3.5 years).

All subjects had bilateral, gradually sloping symmetrical hearing loss of post-lingual onset. Hearing thresholds ranged from 20 to 75 dB HL from 250 to 8000 Hz with bilateral symmetry within 5 to 15 dB at any given frequency. The mean pure tone average (PTA) thresholds averaged across 500, 1000, and 2000 Hz were 42.8 dB HL for right ear and 43.6 dB HL for left ear with an SD of 8.3 and 9.2 dB, respectively. No subject had an air-bone gap greater than 15 dB for low frequencies (250 and 500 Hz), and 10 dB at mid and high frequencies (1, 2 and 4 kHz). All subjects had gradually sloping audiogram configuration (Figure 3.1). Immitance audiometry done in all subjects ruled out the presence of conductive pathology. Routine speech audiometry revealed findings that were consistent with pure-tone and immitance audiometry. Transient evoked otoacoustic emissions (TEOAEs) were absent in all the subjects.

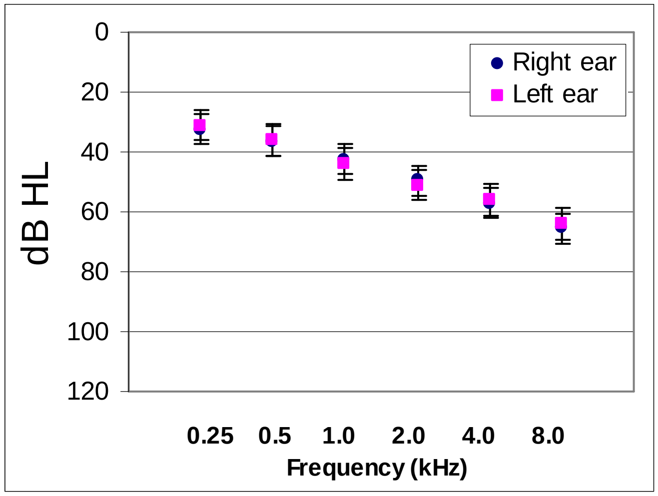

Figure 3.1: Mean thresholds of experimental subjects. Error bars indicate +/- 1 SD.

Control group

Twenty normal hearing adults (11 females and 9 males), aged 18 to 50 years with a mean age of 32.5 years, served as controls. The normal hearing subjects;

- were native speakers of Kannada,
- had air and bone conduction thresholds of 15 dB or better with an air-bone gap of not more than 10 dB,
- showed normal findings on immitance audiometry (i.e., ‘A’ type tympanogram, with presence of acoustic reflexes at normal SLs),
- had speech identification scores of not less than 80%, and
- were free from all kinds of neuro-otological and audiological disorders.

Hearing aid fitting

The Syncro BTE hearing aid manufactured by Oticon was used in this investigation. Test hearing aid in this report refers to this particular hearing aid. Test hearing aid is a fully digital multichannel instrument with no volume control. Signal processing strategies implemented in this hearing aid for noise management are multiband adaptive directionality, multichannel compression, noise management feature and artificial intelligence. It is the only commercially available hearing aid that incorporates parallel processing. The electroacoustic characteristics of this hearing aid are given in Appendix A. The important distinguishing features of the test hearing aid with reference to other premium hearing aids are its use of multi-band adaptive directionality with parallel processing or artificial intelligence. Keeping in view the potential conflict of interests, the brand names of other hearing aids are not quoted.

The subjects in the experimental group were bilaterally fitted with the test hearing aids at no cost. Shell style for a given subject was chosen according to normal audiological practice, in which cosmetic preferences and gain requirements were considered in conjunction. The hearing aids were programmed using the manufacturer's prescription and recommendations using Genie software version 5.0 (Oticon, Denmark) through NOAH 3.0 version and Hi-Pro. A specific Syncro identity was assigned for each participant (Gradual, Active or Dynamic) using the personal profile of the programming software. Real Ear Insertion Gain (REIG) measurements using Fonix FP40D (Frye Electronics, USA) with DIGSP- ANSI stimuli were conducted to verify target fittings. No fine-tuning other than adaptation manager or the feedback manager for the test hearing aids was offered to any subject during the course of the study, unless they explicitly asked for it. The purpose of not allowing fine-tuning was to keep constant the amount of audibility provided by the hearing aids. However, fine-tuning was offered to three subjects as they consistently asked for it.

For the purposes of comparisons, the subjects used their own hearing aids that were all advanced digital BTE aids currently available. Their hearing aids had been fitted and fine-tuned in accordance with specifications of the respective manufacturers. No additional fine-tuning was performed for this study. Standard electroacoustic tests for these hearing aids were conducted to ensure that the aids operated consistent with the specifications of the manufacturer.

Instrumentation and test set-up

- A calibrated dual- channel diagnostic audiometer (Madsen Orbiter 922; Version 2) with TDH39 headphones fitted into ME70 noise- excluding headset, and bone vibrator (Radio ear B71), and sound field speakers (Jamo E- 100) was used for subject selection and all other testing. A calibrated immitance meter (GSI-Tympstar) and an OAE system (ILO 292, Otodynamics Ltd.) were also used for selecting subjects.
- For hearing aid experimentation purposes, the two loudspeakers were kept at 0^0 azimuth from the subject at a distance of 1 meter (re; nose). The channel 1 of the audiometer received input from a personal computer with sound card, while channel 2 received input from a DVD player (729K, Philips). The compact discs (CDs) containing word lists were played from a personal computer with sound

card, while the noise CD was played from the DVD player. The impedance of the whole testing set-up was balanced.

- All the testings, both for selecting subjects and for experimentation purposes, were conducted in an air-conditioned, acoustically treated single or double-room set up depending on the test. The ambient noise levels inside the test room were within permissible limits (ANSI, 1999).

Test conditions

The laboratory tests included speech identification testing, and paired comparison tasks (for sound quality and acoustic comfort) in four different conditions. These conditions are as described by Dillon (2001). The four sets of conditions were as follows:

a) Alternating talkers: three quickly alternating talkers speaking in quiet at 55, 65 and 75 dB SPL.

b) Cutlery noise: speech material at 80 dB SPL with high-frequency impact sounds (e.g, cutlery noise) of 80 dB SPL in the background, that is, a SNR of 0 dB.

c) Multi-talker speech babble: speech material at 80 dB SPL with speech babble of 70 dB SPL in the background, that is, SNR of 10 dB.

d) Traffic noise: speech material at 80 dB SPL with a background simulated traffic noise of 80 dB SPL, that is, a SNR of 0 dB.

Test material preparation

Speech

A total of seventy- five words from the high frequency word lists (Subset I, II and III) in Kannada developed by Kavita (2002) were recorded for three different rapidly changing talkers randomly. The talkers included an adult male, an adult female and a child who were native speakers of Kannada. Fundamental frequency of these talkers was 135 Hz for male, 220 Hz for female and 360 Hz for child (Vaghmi software, Voice & Speech Systems). The recordings were prepared using a good quality condenser microphone in a PC through Audiolab software (Voice & Speech Systems). Scaling of the speech signals was done using the same software to ensure that the intensities of all sounds were at the same level. A 1 kHz calibration tone was recorded prior to each list. The experimenter, another audiologist and a speech-language pathologist subjectively verified the high fidelity of the recordings. The recorded speech material was then copied onto a good quality CD using a CD writer. This CD was used for presentation of words in the alternating talker condition. The original CD recorded by Kavita (2002) was used for presenting speech in all other conditions (cutlery, multi-talker speech babble and traffic noise).

Noise

The cutlery and traffic noise samples were copied onto a CD from Genie software version 5.0 (Oticon, Denmark) and were used for respective noise presentations. The multi-talker (10 speakers) speech babble CD in Kannada language recorded by Anitha (2003) was used for presenting noise in the condition of multi-talker speech babble.

Speech identification measurements

Speech identification scores (SIS) in conditions of alternating talkers, cutlery noise, multi-talker speech babble and traffic noise were measured for all the subjects with normal hearing and hearing impairment. SIS was measured twice for hearing impaired individuals, once with their own hearing aid and a second time with the test hearing aid. The order of testing with hearing aids was random. Each subject was tested with twenty-five words from any of the word subsets I, II or III. As there were four test conditions (alternating talkers, cutlery noise, multi-talker speech babble, and traffic noise) and three word subsets (I, II and III), a given word subset was chosen randomly for each subject for repeated presentation. In an attempt to avoid measurement errors related to repetition of a word subset in a new test condition, it was ensured that the repetition of any word subset for all test conditions and for all subjects was equal and random.

The CD that was specifically recorded with three talkers was used for presenting speech in the condition of alternating talkers. In all other conditions (cutlery, multi-talker speech babble and traffic noise), the original CD recorded by Kavita (2002) was used for presenting speech. It is important to note that both the CDs had same word lists. The intensity variations (55, 65 and 75 dB SPL) in alternating talkers condition for each subject were random. It was ensured that the intensity dial was manipulated only between the silent gaps between two consecutive words. As there were twenty-five words in each word subset, and stimuli had to be presented at three different intensity levels in alternating talkers condition, 8 words each were presented at each level (9 in the last level). A similar method was also followed for three different speakers in alternating talkers condition.

The subjects were instructed to write down the words they heard. Only the correct responses were considered while scoring and each correct response was assigned a score of four. In alternating talkers condition, the scores were computed in combination for all levels and all speakers. The scores were converted into percentage. In this way, the maximum possible correct SIS for a given condition would be 100. Each subject underwent speech identification test under the four test conditions in a random order.

Judgment tests for sound quality and acoustic comfort (Paired comparison tasks)

Only the subjects with hearing impairment participated in this experiment. The over all sound quality and acoustic comfort of the test hearing aid (relative to subject's own hearing aids) in four test conditions (alternating talkers, cutlery noise, multi-talker speech babble and traffic noise) were evaluated in the Paired comparison judgment tests. The paired comparison tasks were conducted blindly by sequentially running presentation of the two hearing aids in random. The subject could listen on each hearing aid for each of the conditions as many times as they liked. The subjects were instructed to identify the hearing aid that gave good overall sound quality and listening comfort. If the subject reported that both the hearing aids were equally preferable, then they were asked to select the one they would prefer for long term listening. The responses were recorded by the subject. Subjective preference results for sound quality and acoustic comfort were separately noted. Each new session was preceded by a practice run of five presentations in background noise.

(B) FIELD TESTING

Field-testing was also carried out on these subjects. Unfortunately, however, only 14 subjects were available for field testing - 7 normals and 7 hearing impaired. The mean duration of hearing aid experience for these 7 hearing impaired was 2 years. These 7 hearing impaired subjects were comparable with the larger group in terms of thresholds, age, etc. Figure 3.2 shows the mean audiometric thresholds of these 7 hearing impaired subjects.

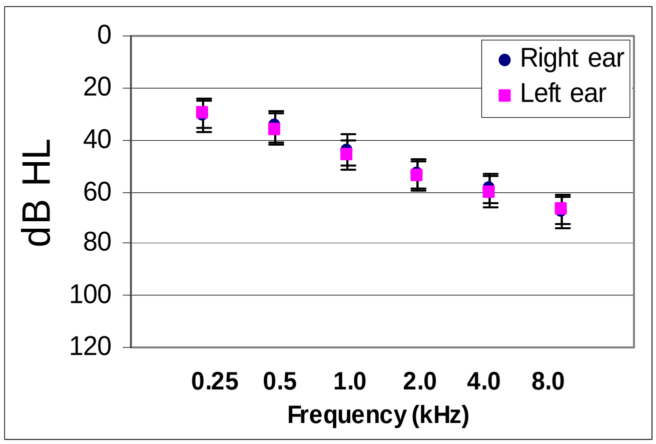

Figure 3.2: Mean thresholds (dB HL) of hearing-impaired adults who participated in the field-testing. Error bars correspond to +/-1 SD.

Instrumentation and test procedure

The SIS measurements were conducted in real-life situations of traffic and restaurant noise. Tests were carried out in a single session under each noise condition.

Traffic noise

This experiment was carried out at a busy hour between 10 am and 12 noon on a working day at the foot path of a busy traffic junction in the city of Mysore. One normal hearing and one hearing impaired subject with the test device were made to stand together facing a CD player (AZ2160V, Philips). A volunteer held the CD player approximately at the height of the subject's head. The distance between the CD player and the subject was approximately 1 meter. The output of the CD player was calibrated at 80 dB SPL. Only one word subset selected randomly from the high frequency word list (Kavita, 2002) was presented. The subjects were asked to note down the words they heard from the CD player. The experimenter who stood between the two subjects, held a compact sound level meter (Model 824, Larson & Davis) approximately at ear level and pointing away from the CD player to record the background noise SPLs. The noise levels were averaged for the whole test duration (time-weighted average i.e., TWA). Table 3.1 provides the SLM settings and noise levels data as measured in traffic and restaurant noise conditions. The spectrum of traffic noise obtained at 1/3 octave in real-life situation, for the total test duration is shown in Figure 3.3. Similarly, SIS measurements were done for other subjects. The scoring procedure for SIS was same as in laboratory tests. Figure 3.4 shows the arrangements for testing in traffic noise.

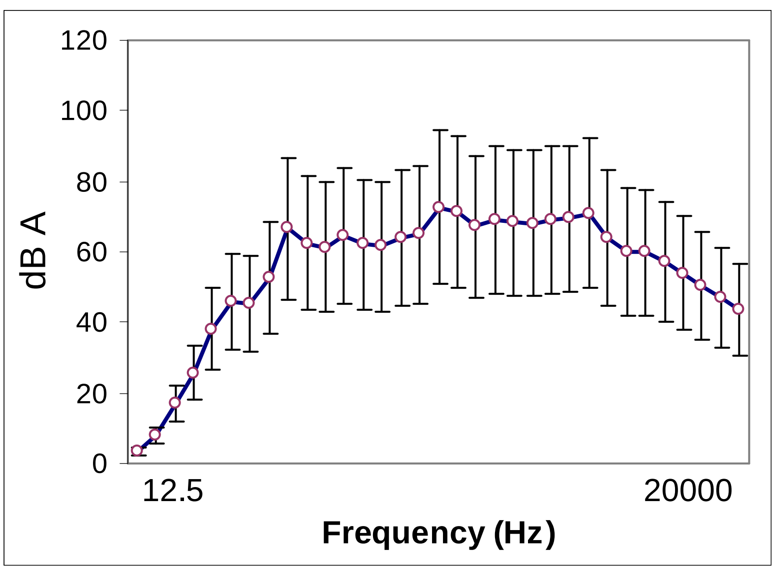

Figure 3.3: The spectrum of traffic noise measured for the whole duration of test. Error bars depict the fluctuations in level.

Table 3.1: SLM settings and measured noise levels at a busy traffic junction and a restaurant.

Parameters	Traffic	Restaurant
Auto calibration	Yes	
Transducer	Condensor	
RTA settings bandwidth	1	
Detector	Slow	
Weighing	A	
Second display	Time-weighed average (TWA)	
RTA detector	Fast	
Ln percentiles	L 1.0, 5.0, 50.0, 90.0, 95.0, 99.0	
Time interval exchange rate	3 dB	
Time spectra option	At max	
Time history units	1.0 second	
Resolution	0.1 dB	
Leq (dB A)	78.3	73.1
Peak (dB A)	116.0	111.7
Lmax; slow (dB A)	95.4	81.2
Lmin; slow (dB A)	69.5	65.1
Lmax; fast (dB A)	103.0	87.5
Lmin; fast (dB A)	68.2	64.4
Lmax; impulse (dB A)	106	91.6
Lmin; impulse (dB A)	69.6	65.0

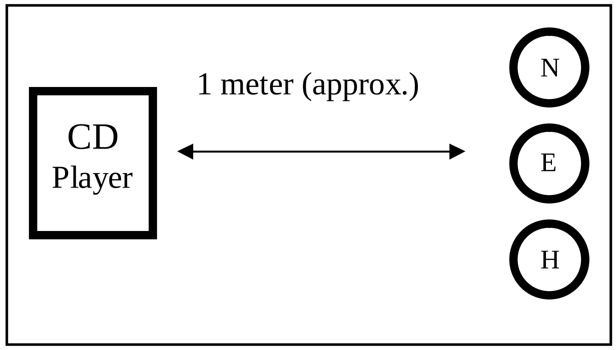

Figure 3.4: Schematic diagram of the field-testing arrangement (E = experimenter, N = normal hearing subject and H= hearing impaired subject) in traffic noise.

Restaurant noise

The measurements were carried out between 12 noon and 1pm (lunch hour) on a working day, at one of the busy eating houses in the city of Mysore. The arrangement for testing in restaurant noise was similar to that in traffic noise, except that the experimenter, and subjects were sitting and the CD player was on the dining table. One normal hearing and one hearing impaired subject with the test device were made to sit together facing a CD player (with calibrated output of 80 dB SPL), which was kept on the dining table. The distance between the CD player and the subject was approximately 1 meter. The experimenter sat in between the two subjects and held the SLM for measuring the background noise. The SLM settings and measured noise levels at the restaurant are given in Table 3.1. SIS were obtained in a similar way as in traffic noise condition. The spectrum of restaurant noise obtained at 1/3 octave, in the real- life situation for the total test duration is shown in Figure 3.5.

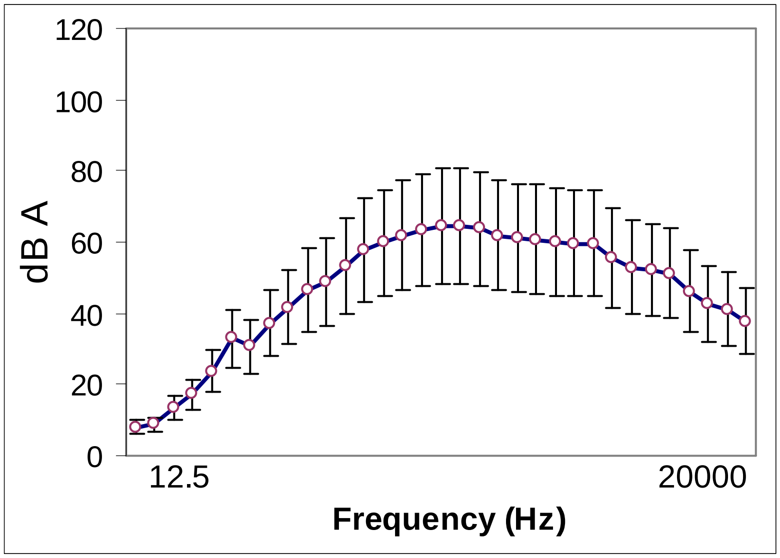

Figure 3.5: The spectrum of restaurant noise. Error bars show the fluctuations in level.

Experimental protocol

An overview of the sequence of steps in the experimental design is given below.

- Each subject in the control group and experimental group was recruited after routine audiological evaluation on pure tone audiometry, speech audiometry, immitance audiometry and TEOAEs.
- The initial tests for hearing-impaired subjects additionally included verification of their current hearing aids. The procedures for these preliminary tests were followed according to normal audiological practice.
- For laboratory tests, subjects in the control group completed speech identification testing in the test conditions of alternating talkers, cutlery noise, multi-talker speech babble and traffic noise, in a random order. Hearing impaired subjects were tested for aided speech identification and paired

comparison tests with their own hearing aids as well as test hearing aids in the four test conditions, again in a random order.

- For field testing, speech identification tests were conducted in normal hearing and hearing impaired individuals. However, unlike laboratory measurements subjects with hearing impairment were tested only with the test hearing aids.

CHAPTER 4

RESULTS

The present study was designed to evaluate the performance of a hearing aid with artificial intelligence (Oticon Syncro BTE) in persons with sensorineural hearing loss. The performance of the hearing aid was evaluated in both laboratory and real-life situations. In hearing impaired individuals, speech identification scores (SIS) were measured with the test hearing aid as well as with their own hearing aids. Paired comparison tasks were conducted to evaluate the sound quality and acoustic comfort of the test hearing aid with reference to the subject's own hearing aids.

Laboratory testing

Speech identification measurements

The mean correct SIS for the normal hearing and hearing impaired subject in various test conditions are presented in Figure 4.1. The significance of difference between mean SIS between the subject groups- normals, hearing impaired with test hearing aid, and hearing impaired with their original hearing aid- in all the four test conditions was tested by one-way Analysis of variance (ANOVA). ANOVA showed a group difference only for cutlery and traffic noise. Tukey's post hoc tests were conducted to verify the main effects for cutlery and traffic noise. The post hoc test for cutlery noise showed a significant difference in SIS between all the three subject groups ($p \leq 0.05$), that is, each subject group was significantly different in terms of SIS than the other two. In traffic noise condition, post hoc analysis ($p \leq 0.05$) showed; (i) significant difference in SIS between the test hearing aid and own hearing aid, (ii) significant difference in SIS between normals and own hearing aid, but (iii) no significant difference between normals and test hearing aid. The statistical procedures were accomplished on SPSS for Windows (Version 10.0). The summary of results of ANOVA are given in Table 4.1.

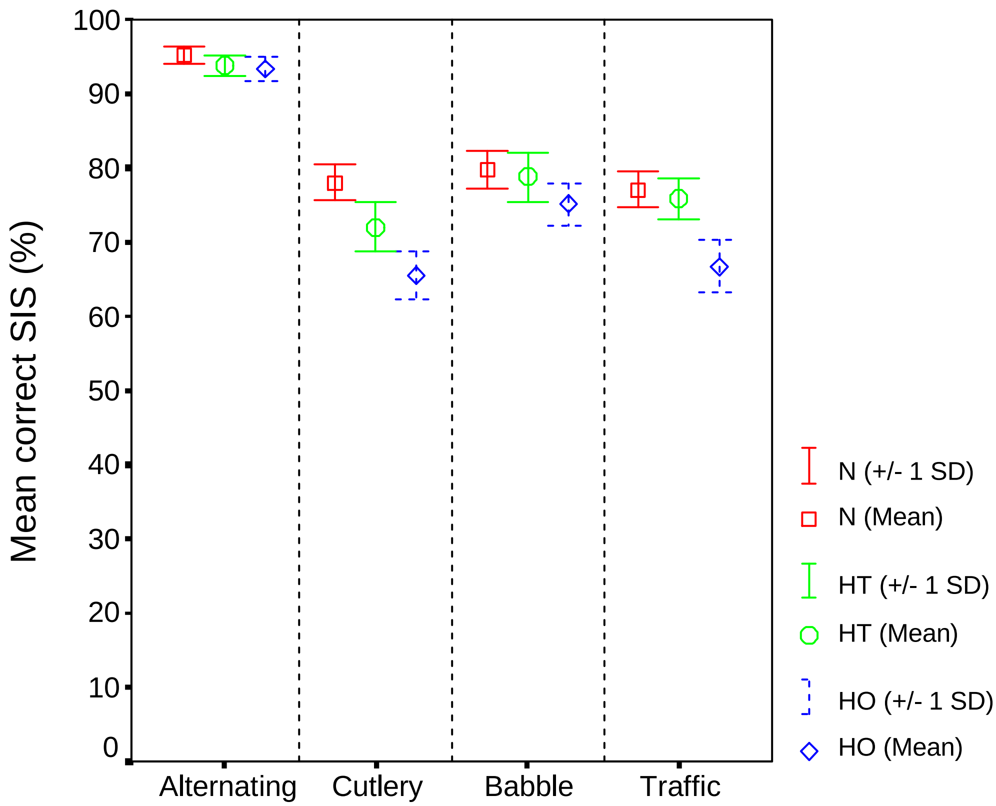

Figure 4.1: Mean and +/- 1 SD bars of SIS of subjects (N= Normal hearing subjects; HT= Hearing impaired individuals wearing the test aid; HO= Hearing impaired individuals wearing own aid) in four test conditions (alternating talkers, cutlery noise, multi-talker speech babble and traffic noise).

A few important observations from the analysis of results are as follows;

1. The mean SIS was not significantly different between the three groups in respect of alternating talkers and multi-talker speech babble condition.
2. In cutlery noise, there was a significant difference in SIS between all the groups, with best score by normals, followed by test hearing aid and own hearing aid.

3. In traffic noise, normals gave the best SIS which, however, was not significantly different from those in the test hearing aid condition. But it was significantly different from the own hearing aid conditions. Also, SIS were significantly different between the test and own hearing aid conditions.

Table 4.1: Summary of results of ANOVA

Condition	Group	Mean	SD	'F'- value	p- value
Alternating talkers	N	95.20	2.48	2.10	0.132
	HT	93.80	2.85		
	HO	93.35	3.49		
Cutlery noise	N	78.05	5.42	16.9 *	0.000
	HT	72.05	7.53		
	HO	65.55	7.22		
Multi-talker speech babble	N	79.75	5.85	2.72	0.074
	HT	78.75	7.58		
	HO	75.10	6.35		
Traffic noise	N	77.10	5.30	15.08 *	0.000
	HT	75.80	6.06		
	HO	66.75	7.84		

N= Normal hearing; HT= Hearing impaired individuals with test hearing aids; HO= Hearing impaired individuals with own hearing aid; SD: standard deviation; * Significant at 0.05 level.

Figure 4.2 depicts the percentage of hearing-impaired subjects, listening through test or original hearing aid whose SIS were within +/- 2 SDs of that of the normal hearing individuals in a given listening situation. This percentage represents individuals who fell within the range or were better than the best score of the range. It is apparent that in all the four test conditions, the percentage of hearing impaired individuals performing as well as the normal hearing listeners is lowest when they were using their original aids and highest when they were using the test aids.

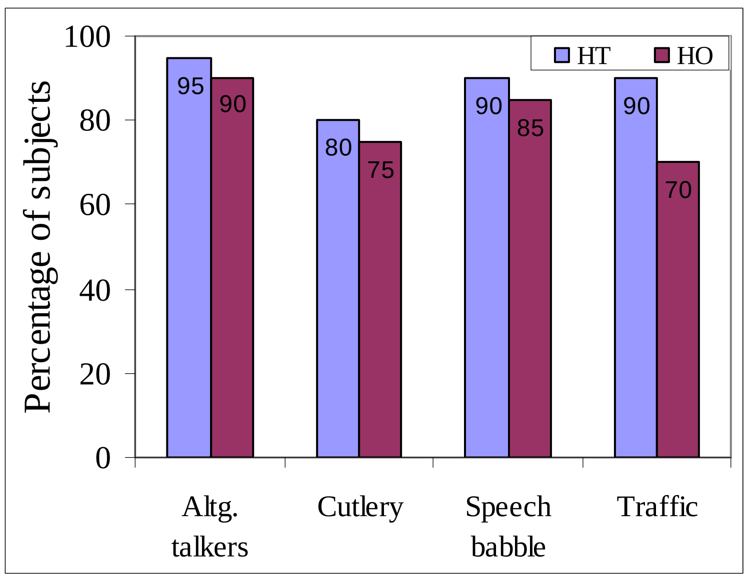

Figure 4.2: Percentage of hearing impaired individuals using a given aid (HT= Hearing impaired individuals wearing the test aid; HO= Hearing impaired individuals wearing original aid) who performed within the range (+/- 2 SDs) of normal hearing subjects.

Paired comparison tasks

The number of subjects who preferred a specific hearing aid (test aid, original aid, or equal preference) was converted into percentages. The subjective preference results for sound quality and acoustic comfort judgment tasks are separately expressed in the Pie diagrams. Figure 4.3a and 4.3b, depict the percentage of hearing-impaired individuals (N= 20) who preferred a particular device on the basis of its sound quality and acoustic comfort respectively. A definite subjective preference for the test hearing aid over original hearing aids was evident. The subjective preference for the test aid is apparent for both sound quality and acoustic comfort judgment tasks.

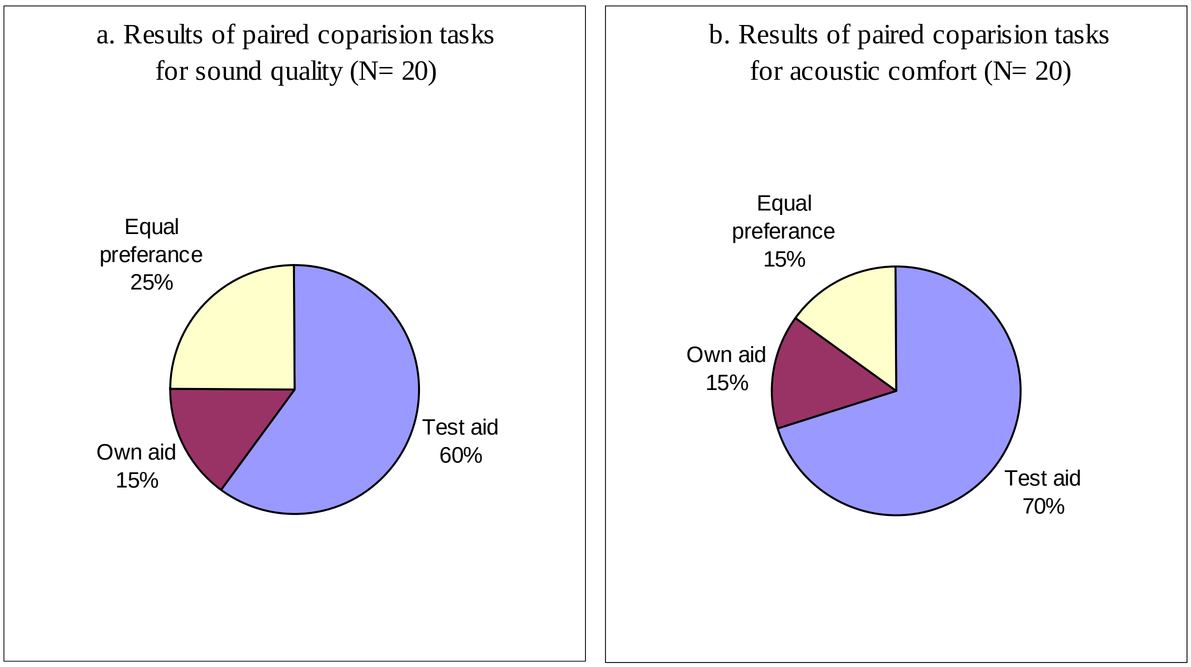

Figure 4.3: Percentage of subjects who showed a preference for a specific hearing aid on paired comparison tasks for (a) sound quality and (b) acoustic comfort.

Field testing

Normal versus hearing impaired performance

The mean correct SIS for normal hearing, and hearing- impaired listeners wearing the test hearing aid in real-life listening situations of traffic and restaurant noise are shown in Figure 4.4. As the sample size was small (N= 7), non-parametric statistical procedures were adopted to analyze the differences in SIS between normal and hearing-impaired listeners, in real-life situations of traffic and restaurant. The results of Mann-

Whitney tests are given in Table 4.2. The results suggested that in traffic, there existed a significant difference in SIS between normal listeners, and hearing-impaired individuals wearing the test hearing aids. However, there was no statistically significant difference in SIS between the two groups in real-life listening situation of restaurant noise. An individual data outlier examination suggested that 3 out of 7 and 6 out of 7 hearing-impaired subjects could perform within the range (+/- 2 standard deviations) of normal listeners in the same listening situation of traffic and restaurant noise, respectively.

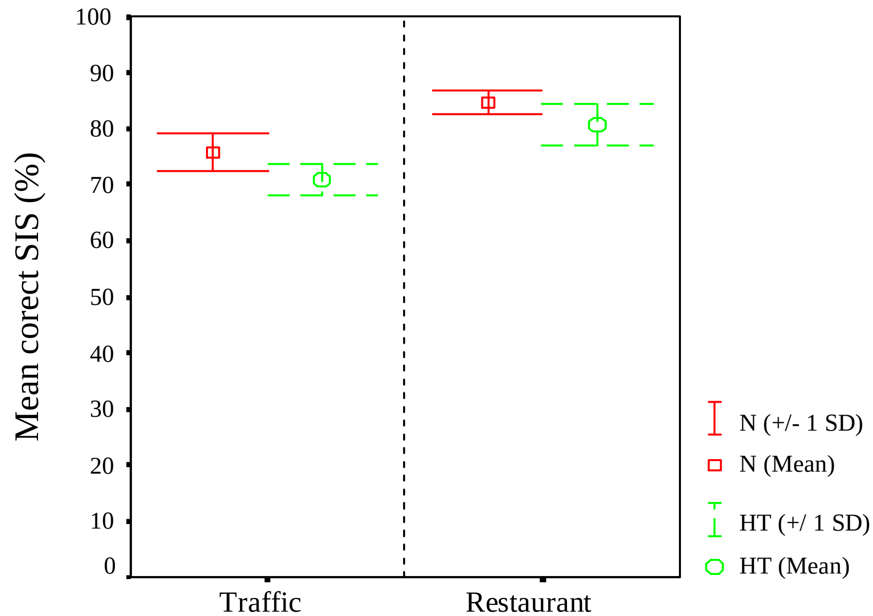

Figure 4.4: Mean and +/- 1 SD bars of SIS of subjects (N= Normal hearing subjects; HT= Hearing impaired individuals wearing the test aid) in field- testing.

Table 4.2. Summary of Mann-Whitney tests analysis for field testing.

Real- life listening situation	**Normals**		**Hearing impaired**		**'p'- value**
	Mean	S.D	Mean	S.D	
Traffic	75.71	4.57	70.85	3.76	0.024 *
Restaurant	84.71	2.87	80.06	4.92	0.084

* Significant at 0.05 level.

Laboratory versus field performance

Wilcoxon test for paired samples was performed to compare the laboratory and real-life performance of the seven hearing impaired subjects with test hearing aids in traffic noise. The statistical analysis showed no significant difference ($p= 0.219$) in SIS between laboratory (mean= 72.42, SD= 3.76) and field (mean= 70.85, SD= 4.75). The analysis indicated that the performance of the test hearing aid was as good in field situations as it was in laboratory conditions for traffic noise. A similar analysis in normals also showed no significant difference ($p=0.400$).

CHAPTER 5

DISCUSSION

Rationale for the study design

The design of the study involved both laboratory testing and field-testing for evaluating the performance of a hearing aid with artificial intelligence. The laboratory testing was undertaken because the parameters governing the test conditions can be more validly controlled. Field testing was undertaken because it is in these situations that one expects artificial intelligence (parallel processing) to deliver better speech understanding and greater comfort than other digital platforms.

A set of four test conditions, as described by Dillon (2001), were used in laboratory testing. These conditions most appropriately simulate the realistic environment in laboratory in terms of spectral variation, and are most contemporary. Therefore, SIS measurements were not conducted in different SNRs for a given test condition as done traditionally. Instead, testing was done in several conditions at a fixed SNR. For example, in real-life situations of traffic and cutlery noise, it may not be possible for a talker to raise his or her voice above the background noise level, where as, in speech babble and restaurant noise, he or she can speak over and above the level of background noise. Based on these realities, the noise levels were kept at 80 dB SPL in test conditions of cutlery and traffic noise, while, for speech babble, it was at 70 dB SPL. The presentation level for speech was kept constant at 80 dB SPL for cutlery, speech babble and traffic noise. Noise measurements in real-life situations of traffic and restaurant from the present study fulfill the recommendations of Dillon (2001). The measured Leq (dB A) for traffic and restaurant was 78.3 and 73.1, respectively. This is comparable to 80 dB SPL of traffic and 70 dB SPL for speech babble noise used in the laboratory measurements. Similarly, it may not be worth to compare modern hearing aids at normal conversation levels in quiet for a single talker. Therefore, an alternating talker condition was used in this study where there were three different rapidly changing talkers speaking at levels of 55, 65 and 75 dB SPL to simulate whispering, normal conversation and loud speech, respectively. As all the subjects had gradually sloping audiogram, high-frequency word lists were used.

The real-world environments present noise from many directions. Therefore, a dual-loudspeaker test system cannot adequately represent real-world performance of advanced hearing aids with directional microphones. A multiple-loudspeaker array would reveal accurate results in testing real-world performance of hearing aids that offer directional pickup patterns (Revit, Schulein and Julstrom, 2004). However, because of practical constraints, this study used two-loud speaker system. The speech intelligibility test was delivered from a front loudspeaker while competing noise was delivered from a second loudspeaker also placed in the front, that is, at 0^{o} azimuth (re; nose). With two-loud speaker system, this is probably a standard arrangement for testing binaural hearing aid fitting.

The assumption behind the field-testing was that a hearing aid that performs well in laboratory might not necessarily perform similarly in real-life situations. This study incorporated a novel method of field-testing to evaluate hearing aid performance in real-life situations that has not been reported in literature. This method is possibly one of the simplest and practical ways to document speech identification measures in real-life. However, several parameters such as distance, azimuth, and presentation levels were only approximating and could not be controlled to the level obtained in laboratory situations. Yet, field-testing method employed in the present study is more nearer to daily listening situations where the auditory environment is more dynamic and rapidly changing. The present method is also better than the conventional method of field studies where subjects go out into the open world and report their personal observations. However, this method of testing is time consuming and costly.

The study aimed to evaluate the performance of a hearing aid which is supposed to deliver the cumulative effect of all signal-processing strategies implemented in it. Therefore, the aim was not to test separately the benefits of individual signal processing features for noise management.

In most hearing aid evaluation studies, the performance of a given hearing aid is compared with that of other aids. The present study, however, additionally compared the performance between hearing-impaired (wearing the test aid) and normal listeners, because of two reasons; (i) normal speech understanding is assumed to be the ultimate goal of any hearing aid technology, and (ii) performance indicators dictated by normal listeners is more realistic. Thus, the overall objective of the present study design was to make the testing conditions rigorous but realistic at the same time.

Understanding of speech with the test aid: Comparison with normal perception

The results from laboratory tests indicated that in three test conditions (alternating talkers, multi-talker speech babble and traffic noise), the hearing aid with artificial intelligence allowed a group of hearing impaired listeners to perform like normal hearing subjects on speech identification measures. However, with their own hearing aids, the experimental group could perform normally in only two test conditions (alternating talkers and multi-talker speech babble).

Even when the potential benefits of a given instrument are known for the average user, the benefits for individual users may vary widely. Therefore, apart from group statistical methods, individual data outlier examination was also performed in this study, the results of which are shown in Figure 4.2.

Ninety-five percent of the hearing-impaired individuals performed within normals limits when using the hearing aid with artificial intelligence in the test condition of alternating talkers. Percentage of hearing impaired who performed within normal limits was 90 % under multi-talker speech babble condition and 90 % under traffic noise conditions. Eighty percentage of hearing impaired individuals with the test hearing aid performed normally in the cutlery noise condition, despite a statistically significant difference in SIS between normals and hearing impaired listeners with test aid. The percentage of hearing-impaired individuals who performed within normal limits, as reported by Bentler, Palmer and Dittberner (2004), was different from the present study. This could be because of the methodological differences.

Understanding of speech-in-noise: Test hearing aid versus own hearing aid

The hearing impaired had better SIS with the test hearing aid when compared with their own hearing aids in all the four test conditions. However, the difference was statistically significant for only cutlery and traffic noise. Additionally, the hearing impaired subjects performed like normals in three test conditions (alternating talkers, multi-talker speech babble and traffic noise) with test aid while they could perform like normals in only two test conditions (alternating talkers and multi-talker speech babble). It is evident from Figure 4.2 that, in every test condition, the percentage of hearing impaired individuals who performed within the normal range was higher with the test hearing aid than with their own hearing aid. These findings unequivocally suggest that adults with sensorineural hearing loss wearing test hearing aids can understand speech-in-noise better than with their own digital hearing aids. In a recent study, Flynn and Lunner (2005) also found that subjects with moderate sloping sensorineural hearing loss using hearing aids with artificial intelligence performed significantly better than a reference hearing aid in laboratory and own hearing aid in field conditions.

Understanding of speech: Influence of listening conditions

Data from the present study on SIS and percentage of hearing impaired performing with test and own hearing aids suggest that the difference in performance between the test and own hearing aid is a function of the listening situations. The spectral composition and temporal variations in noise influence aided SIS differently. The hearing impaired performed better in cutlery and traffic noise conditions with the test aid than with their own aid. In addition, hearing impaired could not perform like normals under cutlery and traffic noise conditions with their own hearing aids. This suggests that the benefits of a hearing aid with artificial intelligence were discernible in complex listening environments. It is in such challenging situations that one would expect the parallel processing to deliver better speech understanding than other digital platforms through its ability to select, from a vast array of possible choices, the solution which provides the best possible speech-to-noise ratio (Flynn, 2004). This implies that when the listening conditions were more challenging as in cutlery and traffic noise, the hearing impaired performed better with the test aid.

Acoustic comfort and sound quality judgments of test aids

With regard to sound quality and acoustic comfort evaluations in paired comparison tasks, a clear preference was noticeable for the test hearing aid. The subjective preference for the test hearing aid over their own hearing aids may have its basis in the use of parallel processing in test aid. Parallel processing virtually leads to very less or no delay for processing of input signals, unlike hearing aids that uses time-shared sequential processing (Flynn, 2004).

The results on SIS and Paired comparison tasks indicate that here is good relationship between speech understanding in noise, and sound quality and acoustic comfort. This implies that, the improvement in speech understanding from the test hearing aid did not come at the expense of other dimensions such as sound quality and acoustic comfort. Therefore, one need not to view hearing aids as being either speech-focused, comfort-focused or fidelity-focused. This corroborates with an age-old report by Killion (1979) wherein he pointed out that the best intelligibility would be achieved with the highest fidelity. Judging from the present results, the opinion of some that poor sound quality in hearing aids is not a deficiency of hearing aids as much, but a result of processing requirements necessitated by hearing loss is correct. The results also suggest that the use of artificial intelligence in hearing aids can result in multidimensional benefit to the user of such hearing aids.

Performance of test hearing aids in real-life situations

The performance of hearing impaired individuals with test hearing aids was significantly different from that of normal hearing listeners in real-life situations of traffic noise. However, in real-world restaurant condition, adults with sensorineural hearing loss wearing hearing aids with artificial intelligence could understand speech-in-noise as well as normal hearing controls. As far as the performance of the test aid itself was concerned, there was no significant difference in its performance in the laboratory and field traffic noise situations. A similar trend was also observed for normals. This implies that the test hearing aid performs equally well in the laboratory and the real-life situations. The somewhat poorer performance with test aid in real-life traffic condition could be attributed to the more rapidly fluctuating and dynamic spectral variations of the noise (as can be seen in Figure 3.3). An additional possibility could be the individual variability and small sample size (N=7).

A clinical concern encountered during the study

This section describes a few practical difficulties faced by the clinician and the subjects during the fitting of the test and other advanced hearing aids. Recent advances in technology have made it possible the incorporation of complex environmentally adaptive technology such as multi-band adaptive directionality, dynamic feedback cancellation, and complex speech detection algorithms in hearing aids. Each of these advances, while providing substantial performance benefits, present a series of challenges for the dispensing professional. First, the clinician needs to provide an explanation of the real-world benefits of these features to the client, and second, the client needs to accurately communicate, during the fitting process, how their instrument is performing in the real world. For example, when a client talks about sound quality in noise, a clinician needs to understand whether the subjects are referring to gain and compression issues, or the effect of directionality and noise management systems on a signal. In precisely understanding these requirements lies a solution to the many problems hearing aid users face with their aid.

Choice of signal processing feature for hearing in noise

Despite the problems faced in understanding speech in noise, only 20 to 30% of dispensing audiologists recommend hearing aids with directional-microphone for their patients. In contrast, most recommend the use of hearing aids with digital noise reduction algorithms even though, to date, no one has developed a digital noise reduction scheme that improves the ability to understand speech in 83 dB SPL babble (party) noise or above (Killion, 2004b). In fact, Dreschler (2002) showed that none of the popular digital noise reduction hearing aids he tested significantly improved speech intelligibility in cocktail party noise or car noise. This is not surprising because it is theoretically impossible to improve speech understanding in multi-talker babble noise condition by "noise reduction circuits", unless the noise reduction circuit can differentiate between talkers, and decide which talkers are 'noise' and which talker is the 'desired' signal. To paraphrase Villchur (1973), the noise reduction circuit would need to identify each talker in order to avoid combining syllables and words from one talker with those from another. The data, from the present study cannot be used to comment on this aspect as the test aid incorporated a holistic evaluation of several signal processing features operating simultaneously. However, it is certain from the results that the several signal processing features with artificial intelligence in the test hearing aid provided a definite benefit in terms of speech understanding, acoustic comfort and sound quality in noisy environments. Probably, artificial intelligence in hearing aids with directional microphones is the answer! All in all, the results of the present study seem to point to the potential of artificial intelligence to hearing aid users.

CHAPTER 6

SUMMARY AND CONCLUSIONS

The inadequate hearing in noise in hearing impaired listeners is a constant challenge to developers of hearing aid technology. A most recent attempt to solve this problem is to use artificial intelligence (parallel processing) in the digital hearing aids. The present study was designed to investigate the performance of a hearing aid with artificial intelligence in improving speech perception in noise for persons with hearing impairment. The aim was to see if the hearing aid with artificial intelligence leads to speech perception that is as good as normal in hearing impaired listeners, as well as to compare the test hearing aid with other digital hearing aids worn by hearing impaired. A related objective was to evaluate the sound quality and acoustic comfort provided by the test hearing aid in comparison to that provided by other digital hearing aids.

The study was designed to carry out evaluation of the performance of the test hearing aid in both laboratory and field conditions. Laboratory testing included speech identification measurements and paired comparison tasks. Twenty normal hearing subjects and twenty hearing-impaired subjects with mild-to-moderate gradually sloping sensorineural hearing loss participated in the study. Speech identification scores (SIS) were measured in normals, and hearing impaired listeners with the test hearing aid and their own hearing aids. Speech identification was tested in four different laboratory test conditions, namely, (i) alternating talkers, (ii) cutlery noise, (iii) multi-talker speech babble, and (iv) traffic noise. Paired comparison tasks involved subjective preference judgments for the test hearing aid with respect to own hearing aids in terms of sound quality and acoustic comfort. Field testing involved SIS measurements for normals, and hearing impaired with the test hearing aid in real-world situations of traffic and restaurant noise.

It was found that in three conditions, namely, alternating talkers, multi-talker speech babble and traffic noise, the test hearing aid resulted in speech perception similar to that of normal hearing listeners. Speech perception through test hearing aid was also significantly better than perception through other digital hearing aids in two conditions (cutlery and traffic noise). The performance of test hearing aid was the same as that of other premium digital hearing aids in the conditions of alternating talkers and multi-talker speech babble. Paired comparison tasks showed that the hearing impaired listeners had a clear preference for the test hearing aid over their own hearing aids in respect of sound quality and acoustic comfort. In field testing, hearing impaired subjects performed like normals through the test hearing aid in real-world listening situation of restaurant noise, but not in traffic noise. However, performance of the test hearing aid was similar in the laboratory and field conditions of traffic noise.

Importantly, the present investigation incorporated a novel method of field-testing to evaluate the performance of the test hearing aid in real-life situations that has not been previously employed by any study. The present method is simple, practical and perhaps better than the conventional method of field studies - where subjects go out into the world and report their personal observations – which can be time consuming and costly, and are associated with uncertainties inherent in self-reporting.

Limitations of the study

The investigator is conscious of some of the limitations of the study which are as follows:

- First, all the experimental subjects in this study were adults with a mean age of 43.5 years. Hence, these results cannot be generalized to other age groups. One cannot assume that children and geriatrics would obtain the same benefits from the test aid that were observed in this study.
- Second, we are also aware of the inherent limitations of simulating "real-life" listening situations in the laboratory. Although all attempts were made to simulate a real-world environment in laboratory, the laboratory test-setup of the present study was still not typical of real-world listening situations. For example, dual-loud speaker system was used and location of both the loudspeakers was at 0^0 azimuth. Unfortunately, in real-world listening environments, neither speech nor noise is always presented at 0^0 azimuth. The source and direction of both speech and noise may alter the performance of the test hearing aid.
- Another limitation, particularly relating to field-testing subjects was that the number of hearing impaired subjects was too small (N= 7). Such a sample cannot lead to definite conclusions which can be generalized.
- Finally, this investigation was conducted with a particular model of hearing aid with hearing aid fitting as per manufacturer's specifications. Therefore, the effect of fine-tuning on derived benefits is not known.

Future research

Future research can focus on such issues as the short-term and long-term benefits of hearing aid with artificial intelligence. Such studies can be carried out on subjects of different age groups, with different audiological profiles, and in a variety of listening conditions that one comes across in the real world. Other aspects of normal hearing and measures of hearing aid benefit, such as spatial hearing may also be investigated. Of course, it is also suggested that further evaluations can be done to validate or reframe the methods adopted for field testing in the present study.

The source and direction of noise and speech are highly variable in real-life situations. Therefore, a more valid approach of the performance of the hearing aid with artificial intelligence in field situations can be taken up with variable source and direction of noise and speech.

The effects or benefits of various signal processing strategies incorporated into the test aid can be investigated individually. For example, the benefits of multi-band adaptive directionality (a new directional microphone system) need to be investigated and documented for future adoptions.

In conclusion, it is said that the hearing aid with artificial intelligence has a great potential to help the hearing impaired. Perhaps, artificial intelligence (parallel processing) will serve as the benchmark technology for developing next generation of highly intelligent hearing aids, that is, "Trainable Hearing Aids". It is our optimism that trainable hearing aids will be a reality in the near future and that they will facilitate normal communication in noise for hearing impaired listeners.

REFERENCES

Alcantara, J. I., Moore, B. C. J., Kuhnel, V., & Launer, S. (2003). Evaluation of the noise reduction system in a commercial digital hearing aid. *International Journal of Audiology, 42*, 34-42.

American National Standards Institute (ANSI). (1996). Specification of hearing aid characteristics, revision of ANSI S3.22-1987. (ANSI S3.22-1996). New York: ANSI.

American National Standards Institute (ANSI). (1999). Maximum permissible ambient noise levels for audiometric test rooms. (ANSI S3.1-1999). New York: ANSI.

Anitha, R. (2003). *The effect of multi-talker babble of different languages on the speech recognition scores in Kannada*. Unpublished Independent Project. University of Mysore, Mysore.

Bentler, R.A., Anderson, C.V., Niebuhr, D., & Gette, J. (1993). A longitudinal study on noise reduction circuits. Part II: Subjective measures. *Journal of Speech and Hearing Research, 36*, 820-831.

Bentler, R.A., Palmer, C., & Dittberner A.B. (2004). Hearing-in-noise: Comparison of Listeners with normal and (aided) hearing. *Journal of the American Academy of Audiology, 15*, 216-225.

Boothroyd, A. (1993). Speech perception, sensorineural hearing loss, and hearing aids. In G., Studebaker, & I., Hochberg, (Eds), *Acoustical Factors Affecting Hearing Aid Performance* (pp. 277-299). Boston: Allyn and Bacon.

Boothroyd, A. (2004). Hearing aid accessories for adults: The remote FM microphone. *Ear & Hearing*, *25*, 22-33.

Boymans, M., & Dreschler, W.A. (2000). Field trials using a digital hearing aid with active noise reduction and dual-microphone directionality. *Audiology, 39*, 260-268.

Census of India. Data on Disabled Population- 2001. www.censusindia.net/disability (accessed December, 2004).

Cord, M.T., Surr, R.K., Walden BE, & Dyrlund, O. (2004). Relationship between laboratory measures of directional advantage and everyday success with directional microphone hearing aids. *Journal of the American Academy of* Audiolog*y, 15*, 353-64.

Davis, H. (1947). *Hearing and Deafness: A Guide for Laymen*. New York: JJ Little and Ives Company.

Dillon, H. (2001). *Hearing aids*. New York.: Thieme Medical Publishers.

Dreschler, W.A. (2002). *Active noise reduction in digital hearing aid*. Paper presented at International Hearing Aid Research Conference, California; Aug 25- 27.

Florentine, M., Buss, S., Scharf, B., & Zwicker, E. (1980). Frequency selectivity in normal-hearing and hearing-impaired observers. *Journal of Speech & Hearing Research, 23*, 646-669.

Flynn, M. C. (2004). Maximizing the voice to noise Ratio (VNR) through voice priority processing. *The Hearing Review, 11*, 54-59.

Flynn, M.C., & Lunner, T. (2005). Clinical verification of a hearing aid with artificial intelligence. *The Hearing Journal, 58*, 34-38.

Gabriel, B. (2002). Study measures user benefit of two modern hearing aid features. *The Hearing Journal, 55,* 46-50.

Gelfand, S. A. (1982). Comments on temporal distortions and noise considerations. In G., Studebaker, & F., Bess, (Eds.), *Vanderbilt Hearing Aid Report*. Pennsylvania: Monographs in Contemporary Audiology.

Healy, E.W., & Bacon, S.P. (2002). Across-frequency comparison of temporal speech information by listeners with normal and impaired speech. *Journal of Speech-Language & Hearing Research, 45,* 1262-1275.

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

Kates, J. M. (1995). Classification of background noises for hearing aid applications. *Journal of the Acoustical Society of America, 97,* 461-470.

Kavita, E.M. (2002.). *A high-frequency Kannada speech identification test (HFKSIT)*. Unpublished Master's Dissertation, University of Mysore, Mysore.

Keidser, G., & Grant, F. (2001). The preferred number of channels (one, two, or four) in NAL-NL1 prescribed wide dynamic range compression (WDRC) devices. *Ear & Hearing, 22*, 516-527.

Killion, M.C. (1979). *Design and evaluation of high fidelity hearing aids*. PhD dissertation. Northwestern University, USA.

Killion, M.C. (1997). Hearing aids: past, present, future: moving towards normal conversation in noise. *British Journal of Audiology, 31*, 141-148.

Killion, M.C. (2004a). Myths that discourage improvements in hearing aid design. *The Hearing Review, 11*, 32-40.

Killion, M.C. (2004b). Myths about hearing aid benefit and satisfaction. *The Hearing Review, 11*, 62-64.

Kochkin, S. (2001). The VA and direct mail sales spark growth in hearing aid market. *The Hearing Review, 8*, 16-24.

Kochkin, S. (2002a). 10-year customer satisfaction trends in the US hearing instrument market. *The Hearing Review, 9*, 14-17.

Kochkin, S. (2002b). Consumers rate improvements sought in hearing instruments. *The Hearing Review, 9*, 18-22.

Lewis, M.S., Crandell, C.C., Valente, M., & Horn, J.E. (2004). Speech perception in noise: directional microphones versus frequency modulation (FM) systems. *Journal of the American Academy of Audiology, 15,* 426-439.

Luts, H., Maj, J.B., Soede, W., & Wouters, J. (2004). Better speech perception in noise with an assistive multi-microphone array for hearing aids. *Ear & Hearing, 25,* 411-420.

National Institute of Deafness and Communication Disorders (NIDCD). About hearing 2001. www.nidcd.nih.gov/health/hb.htm (accessed December, 2004).

Nejime, Y., & Moore, B. C. J. (1998). Evaluation of the effect of speech-rate slowing on speech intelligibility in noise using a simulation of cochlear hearing loss. *Journal of the Acoustical Society of America, 103,* 572-576.

Plomp, R. (1978). Auditory handicap of hearing impairment and the limited benefit of hearing aids. *Journal of the Acoustcial Society of America, 63*, 533-549.

Pumford, J.M., Seewald, R.C., Scollie, S.D., & Jenstad, L.M. (2000). Speech recognition with in-the-ear and behind-the-ear dual-microphone hearing instruments. *Journal of the American Academy of Audiology, 11*, 23-35.

Revit, L.J., Schulein, R. B., & Julstrom, S.D. (2002). Toward accurate assessment of real-world hearing aid benefit. *The Hearing Review, 9,* 34-51.

Ricketts, T. (2000). Impact of noise source configuration on directional hearing aid benefit and performance. *Ear & Hearing, 21*, 194-205.

Ricketts, T., & Henry, P. (2002). Evaluation of an adaptive, directional-microphone hearing aid. *International Journal of Audiology, 41*, 100-112.

Schum, D.J (2004). Artificial intelligence: The new advanced technology in hearing aids. www.audiologyonline.com/articles/ (accessed June, 2004).

Stone, M.A., & Moore, B.C.J. (1999). Tolerable hearing aid delays. I. Estimation of limits imposed by the auditory path alone using simulated hearing losses. *Ear & Hearing, 20,* 182-192.

Stone, M.A., & Moore, B.C.J. (2002). Tolerable hearing aid delays. II. Estimation of limits imposed during speech production. *Ear & Hearing, 2002, 23,* 325-338.

Thompson, S.C. (2003). Tutorial on microphone technologies for directional hearing aids. *The Hearing Journal, 56*, 14-21.

Tillman, T.W., Carhart, R., & Olsen, W.O. (1970). Hearing aid efficiency in competing speech situation. *Journal of Speech & Hearing Research, 13,* 789-811.

Valente, M., Fabry D.A, & Potts, L.G. (1995). Recognition of speech in noise with hearing aids using dual microphones. *Journal of the American Academy of Audiology, 6*, 440-449.

Valente, M., & Mispagel, K.M. (2004). Performance of an automatic adaptive dual-microphone ITC digital hearing aid. *The Hearing Review, 11,* 42-46.

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

Walden, B.E., Surr, K., Cord, M.T., Edward, B., & Olson, L. (2000). Comparison of benefits provided by different hearing aid technologies. *Journal of the American Academy of Audiology*, *11*, 540-560.

Yund, E.W., & Buckles, K.M. (1995a). Multichannel compression hearing aids: Effect of number of channels on speech discrimination in noise. *Journal of the Acoustical Society of America*, *97*, 1206-1223.

Yund, E.W., & Buckles, K.M. (1995b). Enhanced speech perception at low signal-to-noise ratios with multichannel compression hearing aids. *Journal of the Acoustical Society of America*, *97*, 1224-1240.

APPENDIX- A.

Electroacoustic characteristics of the test hearing aid

Characteristic		Measured values
Output dB SPL	Peak	112
	1000 Hz	111
	1600 Hz	109
	HF Average (ANSI)	111
Full-on gain dB (input 50 dB SPL)	Peak	54
	1000 Hz	49
	1600 Hz	49
	HF Average (ANSI)	51
Frequency range, Hz	ANSI	130- 6900
THD % (Reference setting. Input: 70 dB SPL)	500 Hz	0.5
	600 Hz	0.5
	800 Hz	0.5
Equivalent input noise level, dB SPL	Omni: Typical/ maximum	12/ 16
	Dir: Typical/ maximum	20/ 24
Battery consumption, mA	500 Hz	1.1
Group delay (ms)		3
Attack and Release time (ms)		5; 80

Note: Electro-acoustic measurements were carried out after disabling all the adaptive features of the test hearing aid. All measurements are in the Omni-directional mode unless otherwise stated. Only important gain and output related characteristics were measured. The measurements were as per ANSI (1996).