EFFECT OF DIFFERENT SIGNAL ENHANCING TECHNOLOGIES ON SPEECH RECOGNITION IN NOISE

Register No: 06AUD005

A Dissertation Submitted in Part Fulfillment of

Final year MSc (Audiology),

University of Mysore, Mysore

ALL INDIA INSTITUTE OF SPEECH AND HEARING

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Dedicated to my beloved Achan & Amma…..

CERTIFICATE

This is to certify that this dissertation entitled is *"Effect of different signal enhancing technologies on speech recognition in noise* " the bonafide work submitted in part fulfillment for the degree of Master of Science (Audiology) of the student (Registration No.06AUD005). 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 other Diploma or Degree.

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DECLARATION

This is to certify that this dissertation entitled ***“"Effect of different signal enhancing technologies on speech recognition in noise*** ” is the result of my own study under the guidance of Dr. K. Rajalakshmi, Reader in Audiology, Department of Audiology, All India Institute of Speech and Hearing, Mysore, and has not been submitted in any other university for the award of any diploma or degree.

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April, 2008 Register No.06AUD005

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CHAPTER 1

INTRODUCTION

A major consequence of sensori neural hearing loss (SNHL) is communicative difficulty, especially in the presence of noise and /or reverberation (Needleman and Crandell, 1995). Unfortunately, conventional amplification technologies may provide little or no improvement to the signal-to-noise ratio (SNR) in adverse listening environments (Plomp, 1986). In fact, a lack of perceptual improvement in noisy listening environments is one of the major reasons why individuals with SNHL report dissatisfaction with and reject amplification (Kochkin, 1993). 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 cardiod 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 & Paludan-Muller. (2002) Ricketts & Dhar, 1999; Valente, Schuchman, Potts, & Beck (2000)).

Personal FM system has also been shown to improve speech intelligibility in noise (Hawkins, 1984; Fabry 1994; Crandel & 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 & 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 & Dreschler, 2000), actual improvement in speech perception in reportedly unreliable. An implementation of DNR processing is to atleast providing 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 omnidirectional 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 participant performed better with both DMic and DNR activated in conjunction, while the other 50% performed better in the DMic only condition.

Need for the Study:

To achieve maximum speech intelligibility in noise for listeners with Sensori Neural Hearing Loss, an audiologist must consider the benefits and limitations of different noise reduction technologies such as directional microphones (DMic), Digital Noise reduction (DNR) systems and personal FM systems. There are no studies which compare the effect of all the three technologies on signal enhancement in the presence of noise. Hence, we propose to study the effect of each of these technologies on signal enhancement in the presence of noise.

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).

CHAPTER 2

REVIEW OF LITERATURE

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.

Noise and Speech perception in individuals with Hearing Impairment

Individuals with hearing loss of cochlear origin have much greater difficulty in perceiving speech in background of noise than do listeners with conductive or mixed hearing loss. This may partly due to the spread of masking (Martin and Pickett, 1970) or abnormal widening of critical band in pathological ears (Preves, 1995).

Plomp (1994) measured the speech reception threshold for sentences (s SRT) in the presence of back ground noise. The results are expressed in terms of signal-to-noise ratio (SNR) necessary to yield 50% understanding of sentence material. Invariably individuals with cochlear hearing loss required an increase in the signal relative to the noise (2.5 dB to 7dB) for understanding the speech material. An even larger SNR (9 dB to 25 dB) was required in the presence of fluctuating noise as with single competing speaker (Baer and Moore, 1994; Eisenberg, Dirks and Bell, 1995).

Killion (1997) demonstrated the increase in SNR required for maintaining 50% intelligibility as a function of hearing loss. The study concluded that a person with 30 dB hearing loss would require 4dB increase in SNR, while an individual with an 80 dB hearing loss may need as much as a 12dB increase in SNR to maintain the same 50% level of comprehension.

The most effective way to improve speech recognition in noise is to improve the signal to noise ratio (SNR). This can be achieved by technologies such as personal frequency modulation systems (FM), directional microphone (DMic) and digital nose reduction (DNR) systems.

Personal Frequency Modulation Systems (FM)

Personal FM system is a technology designed to improve speech perception in noise. 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. In the last decade, FM systems that are smaller in size and with fewer cables and connectors are being developed to make it more cosmetically acceptable. These systems combine the microphone and transmitter into a single unit the size of a small remote control. An external antenna is necessary only for broadcasting over greater distances. The FM receiver has been miniaturized into an even smaller device that can be connected to a personal hearing aid via direct-audio- input (DAI) shoe and project less than half an inch from the hearing aid.

Benefits of FM

Personal FM systems are capable of improving the speech perception ability of individuals with Sensori Neural Hearing Loss. Hawkins (1984) compared the speech recognition scores in the presence of noise in 9 children with mild-to-moderate hearing-impairment using hearing aids and FM systems. Four hearing aid arrangements (monaural-omnidirectional, monaural-directional, binaural-omnidirectional, and binaural-directional) and a number of FM system-personal hearing aid combinations (including direct input, neck loop, and silhouette inductor--monaural and binaural--and environmental microphone on and off) were evaluated in a school classroom. Two measures of speech recognition in noise were employed. First, the signal-to-noise ratio (SNR) yielding 50% identification of spondees was determined using a simple up-down adaptive procedure. Second, word recognition scores were obtained for three amplification arrangements at two different SNR's (+6 and +15 dB). The average FM advantage over a personal hearing aid was equivalent to a 15 dB improvement in SNR. Activation of the hearing aid microphone caused most of the FM advantage to disappear. The benefit offered by the FM system decreased as the environmental SNR increased but remained significant even at +15 dB. Significant improvement was also obtained with the use of directional microphones in comparison with omni directional microphones.

Fabry (1994) evaluated a prototype frequency-modulated (FM) auditory trainer that allowed use of a remote FM microphone and/or an ear level environmental microphone (EM). For each of five subjects with moderate to severe sensorineural hearing loss, the frequency response of the EM was configured either to match that of the FM response, or to provide a high-pass filter characteristic similar to a noise reduction hearing aid. Speech recognition threshold (SRT) testing noise was measured for five test conditions: 1) FM only; 2) EM only with high-pass filter response (EM-HP); 3) EM only with "standard" response (EM-S) matched to FM; 4) FM/EM-HP combined mode; and 5) FM/EM-S mode. Results averaged across subjects indicated that SRTs for the FM only condition were 9 to 10 dB better than those for either EM only condition; data from the combined FM/EM-HP mode averaged 4 dB better than for FM/EM-S conditions.

Similarly Crandall and Smaldino (2000, 2001) demonstrated that FM technology can improve speech perception in noise by as much as 10 to 20 dB over the unaided listening condition.

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. Word identification scores were found based on full word scoring in hearing aid alone and FM + Hearing aid modes. It was found that the improvement in word identification scores were statistically significant for all FM condition when compared to the hearing aid alone condition. They also concluded that, the subject characteristic such as hearing threshold, communication style and word recognition in quiet can affect the benefit received from FM system.

In ideal listening condition an FM system connected to the hearing aid maintains the frequency out put characteristics of the hearing aid and enhances the SNR of the listening environment (Auriemmo, Keenan Passerieux & Kuk, 2005), which in turn increases the speech perception in the presence of background noise.

Directional microphone Systems (DMic)

Another technology used to improve the SNR is the use of directional microphone technology (DMic). DMic rely on spatial separation of a signal of interest (i.e., speech) and an unwanted signal (i.e., noise). DMic depends on the spatial origin of a signal and not its physical characteristics. For instance, a steady state noise or a speech source not of current interest to the listener would both be categorized as noise by a DMic provided they originated behind the hearing aid user's head (given the appropriate polar pattern of the DMic). A single DMic design consists of one microphone with two sound inlets (front and rear) that lead to separate cavities divided by a diaphragm. Sound is manipulated to ensure that it reaches the microphone diaphragm at the same time from both the inlets, through the use of an internal acoustical time delay, with the effect of cancellation of the signal (Ricketts and Dittberner, 2002).

Single directional microphone usually has a cardiod directivity pattern which can provide upto 3-4 dB enhancement of SNR in a non-reverberant test environment (Hawkins and Yacullo 1984). However, a single DMic fails to offer an omni-directional option, which is more favorable in situations such as listening to speech in diffuse noise source or listening to music (Kuehnel and Checkley, 2000). This problem was solved with the introduction of dual microphone directional design. This design uses two separate, matched omni-directional microphones that allow the user to switch manually or electronically between omni-directional and directional modes. In the directional mode an electronic internal time delay is introduced to the back microphone (Ricketts and Dittberner, 2002). The effectiveness of dual microphone systems has been well documented by many research studies (Voss 1997; Gravel, Fausel, Liskow Chobot 1999: Pumford., Jenstad 2000).

Eventhough the capability of the dual-microphone design is well documented in controlled studies, the use of directional microphone has limitations in everyday listening environments. In conditions where there are multiple and diffuse noise sources, a fixed dual microphone design will not be as effective in attenuating the noise because of the fixed directivity pattern, which may not always match the direction of noise source (Kuehnel and Checkley, 2000). Dual microphone systems were introduced to provide an adaptive directivity pattern in response to the environment such that the null is directed towards the principal noise source. In adaptive DMic, the directivity pattern is continually adjusted to keep the output level of the system at a minimum in the direction of the noise source. Studies that evaluated the effectiveness of adaptive dual microphone system indicated that the adaptive directional option can improve directivity over that of fixed directional pattern when there is a single noise source from the side of the listener.

Benefits of DMic

Numerous investigations have demonstrated that directional microphone (DMic) technology can significantly improve the speech perception ability of individuals with SNHL, particularly in noisy listening environments relative to unaided or aided listening with the use of an omni directional microphone. Before 1985 studies focused on the conventional directional microphones (single microphone). One of the earlier studies using single microphone is by Hawkins and Yacullo (1984). In their study, the SNR required to repeat correctly NU-6 words (0° azimuth) embedded in multi talker babble (180° at 65 dB SPL) was evaluated on 12 normal hearing and 11 hearing impaired subjects listening, under ear phones. The recordings were made with monaural and binaural omni directional and directional hearing aids in rooms yielding reverberation times of (Rt) 0.3,0.6 and 1.2 seconds. For the monaural listening condition, results revealed, an average advantage for directional microphone of 6.3, 4.7, and -0.6 dB for RT of 0.3 .0.6 and 1.2 respectively. For the binaural listening condition results revealed average advantage for the directional microphone of 3.6, 5.5, and 1.5 dB for RT of 0.3, 0.6, and 1.2 seconds respectively. These results revealed a directional advantage when RT was less than 1.2 seconds and the directional advantage was equal to 0.3 and 0.6 second for monaural and binaural listening. The range of the directional advantage for the hearing impaired subjects ranged from 1.6 to 7.3 dB.

Valente, Fabry, and Potts (1995) studied the speech recognition in noise with hearing aids using dual microphones. Fifty subjects with mild to moderately severe sensorineural hearing loss and prior experience with amplification were evaluated at two sites (25 subjects at each site). Speech recognition in noise scores were measured using the Hearing in Noise Test (HINT) for each subject while wearing binaural behind-the-ear hearing aids allowing switching between two microphone conditions (single microphone omni-directional and dual-microphone directional). Results revealed an average improvement in signal-to-noise ration (SNR) of 7.4 to 8.5 dB at the two sites for the directional conditions in comparison to the omni -directional conditions.

Lurquin and Rafhay (1996) evaluated 20 normal and 20 hearing impaired subjects to determine difference in signal-to-noise ratio (SNR) necessary to achieve 50% intelligibility. Then they evaluated 15 subjects with mild to moderately severe sensorineural hearing loss who used hearing aids with directional microphones to determine differences in performance between normal and unaided condition. The speech signal used was bisyllabic words presented at 0° and the competition was cocktail presented at 180°. The noise level was increased to determine 50% intelligibility of the words. The results revealed that the mean SNR required was -12.2 dB for the normal group and -4.8 for the unaided hearing impaired group. That is, individuals with hearing impairment required an average of 7.4 dB greater SNR than the normal hearing subjects to maintain the same level of intelligibility (50%). Hence it can be concluded that there was no significant difference in the SNR between the aided and the unaided condition with omni-directional microphone (-4.8 unaided versus -4.9 aided condition). However, with the dual microphone, there was mean improvement in SNR of 6.6 dB relative to the SNR required in omni directional microphone condition. A 6.8dB benefit in SNR over unaided condition was also observed in the study.

Gravel, Fausel, Liskow Chobot (1999) evaluated the advantages provided by a dual microphone behind-the-ear hearing aid on a group of twenty children (ten 4- to 6-year olds; ten 7- to 11-year-olds). For both groups, differences in the SNR necessary to repeat back words and sentences correctly 50% of the time was the dependent variable. The signal was presented at 0° azimuth and noise (multi talker babble presented at 180°) was varied in 2dB steps. SNR's were measured for the omni-directional and dual microphone positions. Results revealed an overall mean improvement of 4.7 dB for the words and sentences for the dual-microphone position in comparison to the omni-directional microphone position. For the younger group the mean advantage provided by the dual microphone was 4.6 dB for words and 5.1 dB for sentences. For older group, the improvement was 5.3 dB for words and 4.2 dB for sentences.

Ricketts and Dhar (1999) evaluated the differences in performance between three hearing aids. One of the test hearing aid was a Dual microphone BTE with analog signal processing (Phonak Audio-Zoom). The second hearing aid was a dual microphone BTE with digital signal processing (Siemens Prisma) the third hearing aid was BTE with DSP and conventional directional microphone (Senso C9). Twelve subjects with mild to moderately severe sensorineural haring loss were included in the study. Hearing in Noise Test (HINT) and nonsense syllable test were administered under anechoic and reverberant (0.6 second) conditions. For each speech test the signal was presented at 0° and the competition was presented at 90°, 180°, 225° and 270°. Result for the anechoic condition revealed an overall mean improvement of 6.5, 7.5, 5.0 dB in HINT threshold for the C9, AZ and Prisma respectively. These differences in hearing aids were not statistically different for the nonsense syllables test for the C9, AZ and Prisma hearing aids respectively. The results of this study revealed that the mean performance of a hearing aid with dual microphone analogue signal processing was equal to or better than the mean performance provided by digital signal processing hearing aid with conventional directional or dual directional microphone.

Like wise Valente Schuman, Potts and Beck (2000) studied performance of dual-microphone in, in-the-ear hearing aids. Fifty subjects with mild to moderate-severe sensorineural hearing loss and prior experience with binaural amplification were evaluated at two sites (25 subjects at each site). Signal-to-noise ratios (SNRs) were measured using the Hearing in Noise Test (HINT) after each subject wore binaural in-the-ear hearing aids programmed for omni-directional and dual-microphone performance, for 4 weeks. Both microphone conditions were evaluated under "ideal" (signal at 0^o; noise at 180^o) and "diffuse" (signal at 0^o; correlated noise at 45^o, 135^o, $225^o$and 315^o) listening conditions. Results revealed statistically significant mean improvements in SNRs between 3.7 and 3.5 dB at Site I and 3.2 and 2.7 dB at Site II for the ideal and diffuse listening conditions, respectively, for the dual-microphones in comparison to the performance provided by the omni-directional microphone.

From the review of the studies it can be noted that, the use of directional microphone technology can improve speech perception in noise by as much as 3 to 8 dB over omni directional microphone technology in the same hearing instrument depending on microphone location, type of noise, test materials, and subject related factors (Hawkins and Yacullo, 1984; Valente et al 1995; Ricketts and Dhar, 1999).

Digital Noise Reduction (DNR)

The term "digital noise reduction" can be used to describe a hearing aid processing with a general goal of providing less amplification, over a specified frequency range, for noise than for speech. DNR algorithms rely on differences in physical characteristics of a signal to distinguish speech from noise. The earliest attempts relied on the assumption that unwanted noise typically existed at the lower frequencies, and attenuated and/or compressed the output of the hearing aid at these frequencies to achieve an SNR advantage. However, such pure frequency-based algorithms are not effective under a majority of circumstances (Boymans & Dreschler, 2000; Kuk, Ludvigsen, & Paludan-Muller, 2002). Another approach is to analyze the intensity distribution of the signal; in this type of algorithm greater variability in the intensity of speech as compared with noise.

Another type of DNR referred to as "adaptive Wiener filtering". Wiener filter was first described by Nobert Weiner in 1940's. It is a theoretically derived filter that has the goal of estimating the original signal from a degraded version of the signal. The goal of modulation based digital noise reduction and adaptive Wiener filtering are similar as both intend to provide more gain for frequency range containing speech information than those containing noise.

Other similar methods attempt to identify noise by analyzing modulation depth or frequency (Kuk, Ludvigsen, & Paludan-Mulle, 2002). Thus, these algorithms identify any steady state signal as noise. When the signal in any frequency channel is detected to be predominantly noise based on the Modulation to steady state ratio (MSSR), gain is reduced for that channel, often proportionately to the level of the noise. Although this does not improve within-channel SNR, it attempts to reduce direct masking within the channel, as well as any spread of masking to adjacent channels (Kuk et al., 2002).

Second-generation of DNR algorithm is based on detection of speech in the incoming signal using the rapid analysis of multiple components of the incoming signal. One technology currently available monitors higher frequency channels for synchronous energy, such energy is suggestive of formants. Hence, better synchrony indicates the presence of speech in the signal (Chung, 2004).

The most commonly used noise reduction system in current commercial multi-channel digital hearing aids is based on identification of modulation in multiple channels allowing for an estimation of the modulated–to-steady-state ratio (MSSR). The system assumes that signals those are primarily steady-state are "noise", while signals with greater modulation are more "speech like". Gain then reduced in channel for which the MSSR indicates the incoming signal is steady state (Van Dijkhuizen et al, 1991). While modulation based digital noise reduction is implemented by several manufacturers the specific characteristic including time constant /analysis time, magnitude of gain reduction, and rules for estimating MSSR and implementing gain reduction vary significantly (Bentler,2004).

Benefits of DNR

The results of studies investigating changes in speech understanding due to implementation of DNR processing in modern hearing aids are inconclusive (e.g., Boymans, Dreschler, Schoneveld, Verschuure 1999; Boymans and Dreschler, 2000; Walden, Surr, Cord Edwards, Olson 2000; Alcantara, Moore Kuhel and Launer (2003).

Boymans and Dreschler (2000) measured the speech recognition in noise with active noise reduction and dual microphone technology. This study combined laboratory experiments with three consecutive field trials of 4 weeks each. Performance measurements (speech recognition tests in background noise), paired comparisons, and self-report measurements (questionnaires) were assessed. For all subjects, results were obtained for three different settings: no noise reduction, with noise reduction alone, dual microphone alone and both noise reduction and dual microphone combined. The effects of dual microphone were clearly positive, especially for the speech reception threshold tests and for the paired comparisons. However, the effect of digital noise reduction was much smaller but showed significant benefits with respect to aversiveness and speech perception or reception in noise for specific acoustical environments. There was no extra benefit from the combined effect of dual microphone and digital noise reduction relative to dual microphone alone.

Similarly, Alcantara, Moore Kuhel and Launer (2003) evaluated the effectiveness of a noise reduction system implemented in a commercial digital multi-channel compression hearing aid. Eight experienced hearing aid users with moderate sensorineural hearing loss were fitted bilaterally according to the manufacturer's fitting guidelines. After a 3-month period of regular use of two programs, one with and one without the noise reduction system, 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 unaided listening conditions. SRTs were lower for the noises with dips than for the steady noise, especially for 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.

Limited data suggest that on specific implementation of modulation based DNR processing may slightly improve speech recognition performance in the presence of steady state-noise. Isolated findings from a few recent studies suggest 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, Boymans et al, (1999) concluded that implementation of MDNR (modulation based digital noise reduction algorithm) processing may lead to improved general sound quality in the absence of significantly improved speech recognition.

Similarly, Ricketts and Hornsby (2005) studied the affect of digital noise reduction (DNR) processing on aided speech recognition and sound quality measures in 14 adults fitted with a commercial hearing aid. Measures of speech recognition and sound quality were obtained in two different speech-in-noise conditions (71 dBA speech, +6 dB SNR and 75 dBA speech, +1 dB SNR). 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 one of the implementation of DNR processing is to provide improved sound quality for speech in the presence of noise.

More commonly, no degradation in speech recognition or sound quality have been reported for MDNR processing implemented in commercial hearing aids (Ricketts and Dahr, 1999; Boymans and Dreschler, 2000; Walden et al, 2000; Alcantara et al, 2003).

Although the actual reasons for the discrepant findings across studies are unclear, it is assumed that differences in the speed and magnitude of gain reduction for steady-state signals as well as differences in experimental methodology (e.g., type of competing signal) play a role.

Comparison of DMic and personal FM system

Despite the documented enhancement in speech perception with directional microphone and FM technologies, to date only two investigations has attempted to compare these technologies.

Hawkins (1984) evaluated the effect of various hearing aid and FM system configurations on speech perception in noise. Nine children with bilateral mild to moderate SNHL served as study participants. These subjects used a Phonic ear 805 CD BTE hearing instrument that had the capability to switch between omni directional and d-mic modes. The phonic ear 441T microphone transmitter and the phonic ear 445R FM receiver served as the FM system. Speech perception was assessed using spondees and Phonetically Balanced Kindergarten (PB-K) words presented in class room with a reverberation time of 0.6 sec. Speech was delivered from a loud speaker located 2m from the child at 0° azimuth. Speech noise was presented from loud speaker located 4m from the child at 180° azimuth. Speech perception was assessed in the following conditions:

1) Monaural hearing aid in the omni directional mode;(2) monaural hearing aid in the directional mode; (3)Binaural hearing aids in the omni directional microphone mode; (4) Binaural hearing aid in the directional microphone mode; (5) Fm connected via neck loop to a monaural hearing aid with a directional microphone on the FM transmitter; (6) FM only connected via a silhouette inductor to a monaural hearing aid with a directional microphone on the FM transmitter; (7) FM only connected via direct audio input (DAI) to a monaural hearing aid with a directional microphone on the FM transmitter; (8) FM only connected via DAI to a monaural hearing aid with an omni directional microphone on the FM transmitter; (9) FM plus EM with no attenuation connected via DAI to a monaural hearing aid in the omni directional microphone mode; and (11) FM plus EM with no attenuation connected via DAI to binaural hearing aids in the directional microphone mode. Results of this study suggest that FM technology significantly improves speech perception in noise when compared to any of the hearing aid alone arrangements (11.8 dB to 18.4 dB improvement). Additionally, FM only condition was significantly better than any of the FM plus EM arrangements (7.9 to 16.9 dB).

Lewis et al (2004) investigated the speech-perception ability of 46 subjects in age range (24 to 84 years) with mild to severe sensori neural hearing loss, utilizing directional microphone and FM technology. Specifically, speech perception was assessed with the Hearing in Noise Test (HINT). Speech spectrum noise was utilized as the noise source, in the following listening conditions: (1) binaural BTE hearing aids in omni-directional mode; (2) binaural BTE hearing aids in the directional mode; and (3) binaural BTE hearing aids utilized with two FM receivers in the FM only mode. The speech spectrum noise was presented from four loudspeakers positioned at 45°, 135°, 225°, and 315° azimuths (diffused condition). All loudspeakers were located one meter from the subject and the noise was held constant at 65 dB (A). All subjects were fitted with Phonak Claro 311 dAZ digital, BTE hearing aids bilaterally and Phonak Microlink ML8 was used as the FM receiver. All FM receivers were evaluated in the FM only mode. Overall, preliminary results from this investigation indicate that FM utilization significantly improved speech intelligibility over the omni-directional microphone (22.74 dB) and directional microphone (19.3 dB) listening conditions. Additionally, data indicated better speech intelligibility performance with the directional microphone over the omni-directional microphone (3.4 dB). These data suggest that FM technology will offer significantly better communicative performance in adverse listening situations than any other type of hearing aid microphone configuration.

Comparison of DMic and DNR

Directional microphones are considered as one of the methods of choice in improving signal-to-noise ratio. On the other hand, digital noise reduction (DNR) algorithms, in commercially available products, are considered to provide comfort but not significant assistance in improving speech perception in noise. In practice, these 2 technologies are often used in conjunction, but a few studies have evaluated their interaction and the resultant effect on speech perception in noise.

Boymans and Dreschler (2000) evaluated the efficacy of a digital hearing aid implementing dual microphone and active noise reduction in isolation and in combination. This was conducted in a well-controlled clinical field trial in 16 hearing-aid users, using a single-blind crossover design. Sixteen subjects completed the APHAB after four week use of the directional microphone feature and four weeks use of the noise reduction feature, counter balanced across subjects. In the last week, they also performed speech recognition or reception tests in background noise with both directional microphone and noise reduction. For the speech reception threshold tests and for the paired comparisons, the effect of directional microphone was clearly positive. Although the effect of noise reduction were not significant for any of the four APHAB subscales, several questions within the subscales showed significance in favor of the noise reduction feature for loud and /or aversive situations .

However, Bray and Nilsson (2001) concluded that DNR algorithms may be effective in improving speech perception in noise when the noise field is isotropic. Twenty adults, (age ranges between 34-84 years, sixteen of the subjects were male and four were female) having bilateral, sensorineural mild-to-severe hearing loss were enrolled in the study. Speech intelligibility in noise measures were obtained in the quasi-free field with the modified Hearing in Noise Test (HINT). The testing sequence (Unaided, Omni, Omni+DNR, Dir, and Dir+DNR) and the listening environment (noise-front and noise-diffuse) were counterbalanced across subjects. For the noise-front environment, there was a mean aided benefit of 2.6 dB SNR without DNR activated and 3.5 dB SNR with DNR activated. The significant effect of the DNR condition is due to the algorithm exploiting the temporal modulation differences between the fluctuating speech and the steady-state noise. There was no significant difference between thresholds obtained in the omni-directional and directional modes in the noise-front condition. In noise diffuse condition there was benefit of 3.5dB in the Omni+DNR condition which was greater than the Omni condition (2.5 dB SNR). There was a 4.8 dB SNR in the DMic condition which was greater than the Omni+DNR condition, and benefit in the DMic+DNR condition was 6.5 dB SNR, which was greater than the DMic alone condition.

Nordrum and Dhar (2006) evaluated the performance of 16 experienced adult hearing aid users on the Hearing in Noise Test when each technology (DMic and DNR) was activated independently and then simultaneously in 4 commercially available hearing aids. Approximately 50% of the participants performed better with both DMic and DNR activated in conjunction, while the other 50% performed best in the DMic only condition. When considering statistically significant differences in performance only, a reduction or improvement in performance was observed in 17% and 14% of the conditions, respectively.

Yuen et al (2006) studied the ability of the hearing aid circuitry to reduce the effects of noise by a sentence-in-noise test in three conditions: (1) adaptive directional microphone; (2) multi-channel noise reduction system and (3) a combination of the two. In the signal-front/noise-side condition, adaptive directional microphone alone and combined adaptive directional and DNR gave better performance than DNR alone in nearly all participants, whereas in the signal-front and noise-front evaluation, the conditions revealed no significant differences.

From the literature review it is clear that many investigations have studied the benefits of each noise reduction technologies (DMic, DNR, and FM). A few investigators have compared the benefits of technologies in isolation as well as in combination (eg: DMic versus DMic+DNR and FM versus DMic). Results of those studies have shown that DMic and FM technology have significantly improved the speech intelligibility in the presence of noise and there is only limited data which suggest that implementation of digital noise reduction may improve speech recognition in the presence of steady state noise. A few studies concluded 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.

CHAPTER: 3

METHOD

The present study aimed to compare the benefits of directional microphone, directional microphone with DNR and FM system in improving speech intelligibility in background noise.

A. 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 shown by tympanometry. They were native speakers of Kannada language and were experienced hearing aid users for more than 6 months.

B. Instrumentation

- A calibrated dual channel diagnostic audiometer (Madson orbiter 922) with TDH-39 head phone, bone vibrater B-71 and Martin (c115) speakers were used.
- A Calibrated immittance meter (GSI-Tympstar) 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).
- 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 (Madson 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 used in the study. To connect the FM receiver to the hearing aid, appropriate audio shoe was used.

Figure1: Illustration of hearing aid connected with FM receiver

C. 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.

C. Test Environment:

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

D. 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 Microlink 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 a 16 channel modulation based digital noise reduction system and an adaptive wiener filter in its DNR processing scheme. This system (digital noise reduction algorithm) assumes that signals those are primarily steady-state like are "noise", while signals with greater modulation are more "speech like". Based on the MSSR(modulated–to-steady-state ratio), the hearing aid decides whether the incoming signal is steady state (noise) or modulated (speech) and then reduces the gain is in the respective channel

(Van Dijkhuizen et al, 1991). The Wiener filter was first described by Nobert Weiner in 1940’s. It is a theoretically derived filter that has the goal of estimating the original signal from a degraded version of the signal. The goal of modulation based digital noise reduction and adaptive Wiener filtering are similar as both intend to provide more gain for frequency range containing speech information than those containing noise. The DMic used in this study has a hyper cardiod 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 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 signals to noise ratio’s (SNR) 0 dB and +10 dB, the signal level was kept constant at 45 dBHL.

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 Microlink 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 dBHL. 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 Microlink FM receiver in the FM only mode

The order of conditions was randomized for each of the 12 participants. The set-up for each listening condition is illustrated in the following figures.

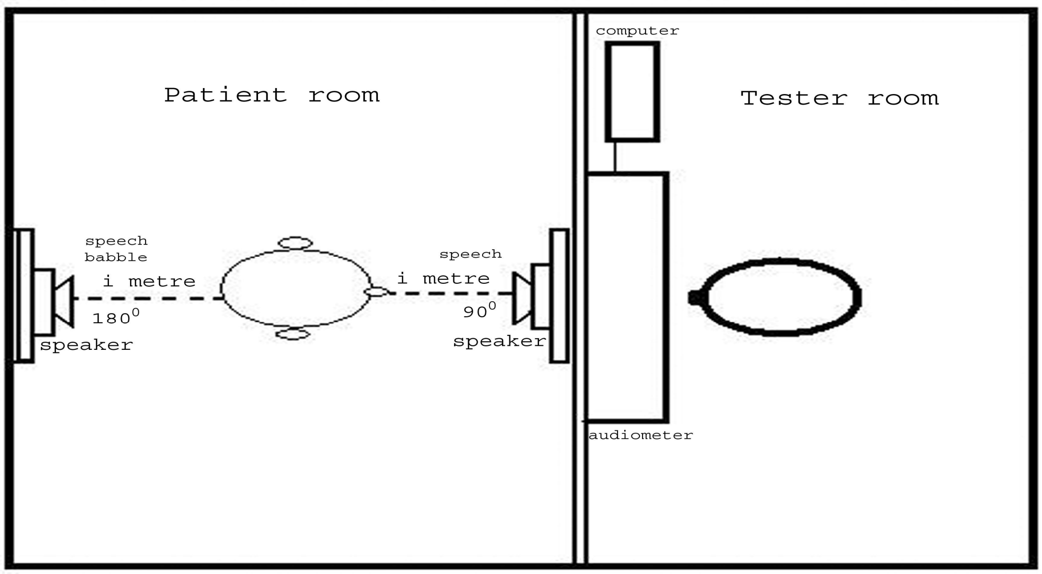

Figure: A. Illustration of the DMic and DMic+DNR listening condition.

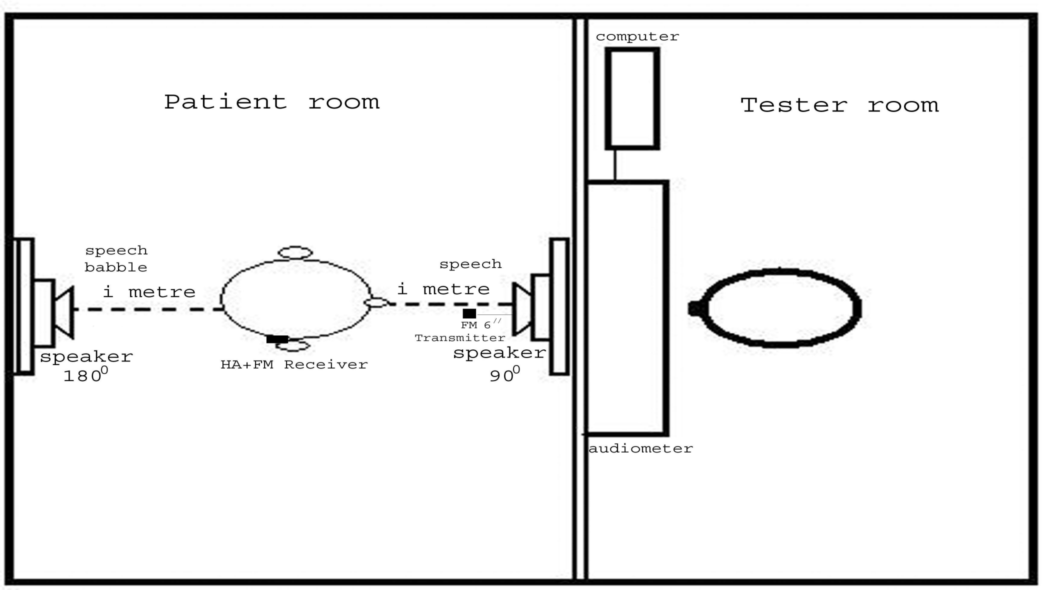

Figure: B. Illustration of the FM listening condition

CHAPTER 4

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.

(a) 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.

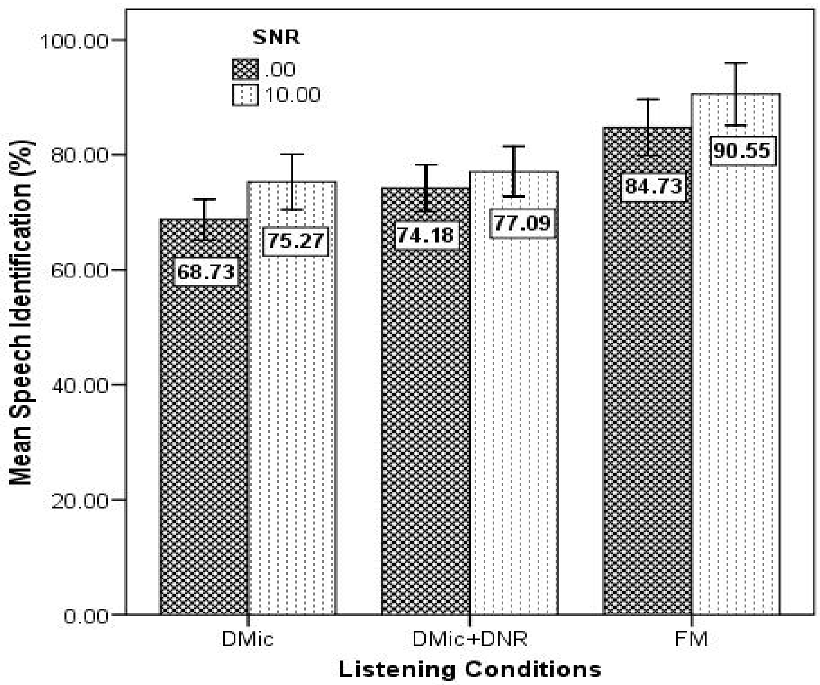

Figure: 1 Comparison of mean speech identification scores across the three listening conditions (DMic, DMic+ DNR and FM) in 0 and 10dB SNR

From figure1 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 0dB and 10dB 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 (0dB SNR and 10dB 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 Bonniferronis 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 & 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 affect 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 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. 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. Bonniferronis 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).

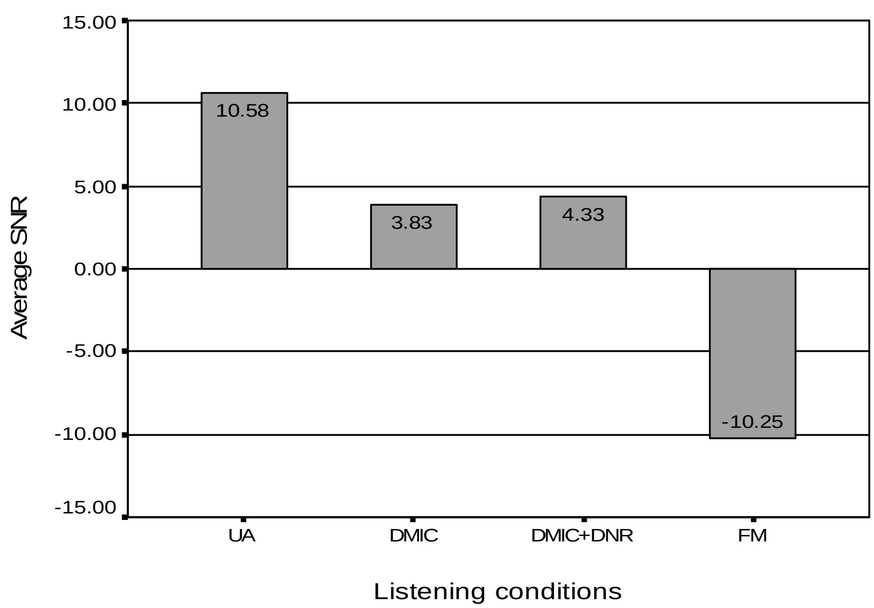

Figure: 2 Comparison of mean SNR across different listening conditions (DMic, DMic + DNR and FM).

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 they conclude 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 (1^{st} 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.

CHAPTER: 5

SUMMARY AND CONCLUSION

A major consequence of sensori neural hearing loss (SNHL) is communicative difficulty, especially in the presence of noise and /or reverberation (Needleman and Crandell, 1995). Lack of perceptual improvement in noisy listening environments is one of the major reasons why individuals with SNHL report dissatisfaction with and reject amplification (Kochkin, 1993). In the present study an attempt was made to study the effect of each of the technologies (DMic, DNR and personal FM) on signal enhancement in the presence of noise.

Benefits of these technologies in speech understanding is measured by two methods,

(a) Speech Identification in noise measurement at two SNRs (0 and 10dB),

(b) Speech recognition threshold in noise in terms of SNR necessary to repeat atleast two out of three words correctly.

Speech identification measurement was carried out in 11 subjects and SNR measurement was carried out in 12 subjects with bilateral gradually sloping moderate to moderately severe sensory neural hearing loss. The benefits of each of these technologies were examined in three listening conditions such as DMic, DMic+DNR and personal FM technology. The speech material consists of 4 phonetically balanced wordlists 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. Speech was presented at 0° azimuth and noise (Kannada speech babble) was presented at 180° azimuth.

The results of the study indicate that there was significant difference in the speech identification across DMic, DMic +DNR and FM. Even though the statistics 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 ie., subjects correctly repeated an average of 1 word more than that in DMic condition, hence it cannot be considered as a drastic improvement. However, FM technology showed greater improvement (15%) in speech identification than DMic and DMic +DNR conditions.

There was significant difference in SNR (speech recognition threshold in noise) measurement across FM and other two listening conditions (DMic, DMic+DNR). However, no significant difference in the speech identification across DMic and DMic+DNR condition was observed. With the use of FM technology SNR (speech recognition threshold in noise) reduced drastically, thus improving speech perception in the presence of noise.

The following conclusions can be drawn from this study,

- 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.
- FM technology provided an improvement of 14.08 dB over DMic in SNR measurement and only 15% improvement in speech identification measurement. This difference between the speech recognition in noise and speech identification scores could be due to ceiling effect.

- Overall, from the results of this investigation it can be concluded that FM technology significantly improved speech intelligibility compared to the hearing aid conditions, both DMic and DMic + DNR condition.

IMPLICTAIONS OF THE STUDY

This study provides documentation regarding the degree of speech recognition performance obtained with various technologies. This information is critical for Audiologists when counseling clients regarding the various signal enhancing technologies that are designed to improve speech perception in the presence of noise. This study concludes that FM system provides better speech performance in noise than the other technologies commonly used.

LIMITATIONS AND FUTURE DIRECTIONS:

Further studies have to be done with different types of noise and diffuse listening conditions to understand the effect of these technologies in real life situations. A limitation of this study is that one cannot assume that the findings in this investigation are comparable to "real-world" performance with these devices. Although attempts were made to simulate a "real-world" environment by utilizing speech babble, the conditions utilized in this study are still not typical of "real-world" listening environments and speech recognition testing was conducted in a sound-treated environment, which results in reduced effects from reverberation. Since several studies have reported a wide degree of electro acoustic variability with the use of various FM components, it should not be assumed that all brands and models of these devices would produce the same speech recognition results obtained in this study (Freeman, Sinclair and Riggs 1980; Bess; Thibeodeau and Saucedo 1991). Studies using newer technologies ADRO (advanced dynamic range optimizer), integrated signal processing will also provide in depth information about the signal enhancement in noise.

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APPENDIX

Phonemically Balanced Word List Developed by Yathiraj and Vijayalakshmi (2005).

rait̪a	tʃukki	hulu	va:tʃu
anna	hagga	su:dzi	hoṭṭe
moḷa	bat̪t̪a	rotti	d̪oni
tʃa:ku	mantʃa	gu:be	vadzra
t̪uti	bekku	akka	va:ni
me:ke	lo:ta	e:lu	t̪ale
ha:vu	ba:la	vi:ne	kat̪t̪e
kat̪t̪u	dze:bu	d̪imbu	me:dzu
bi:ga	mandi	vade	na:ji
o:du	nona	go:li	ba:lu
baḷe	male	ha:lu	ni:li
mu:ru	ṭi:vi	amma	gombe
ra:ni	d̪i:pa:	dzana	ka:ge
t̪apa:	rave	ravi	ad̪u
t̪a:ra:	mole	t̪ande	dra:kʃi
braʃu	railu	rakta	bægu
hasu	ka:ru	sut̪t̪u	kaʃta
dzade	divja	ja:va	paisa
nalli	a:ru	tʃandra	mara
kivi	pu:ri	ja:ke	hu:vu
varʃa	had̪d̪u	a:le	t̪innu
ja:ru	suʃma	aid̪u	idli
da:na	t̪a:ji	nadi	ke:lu
ʃæmpu	d̪ana	uppu	sara
ili	a:lu	kriʃna	pad̪a