PERFORMANCE MEASURES OF SPEECH INTELLIGIBILITY IN NOISE AND LOCALIZATION WITH ADAPTIVE MULTI-BAND DIRECTIONAL MICROPHONE AND FIXED DIRECTIONAL MICROPHONE IN ADULTS USING BINAURAL HEARING AIDS

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MAY, 2016

Dedicated to my

Mummy, Papa and Baba

CERTIFICATE

This is to certify that this dissertation entitled **"PERFORMANCE MEASURES OF SPEECH INTELLIGIBILITY IN NOISE AND LOCALIZATION WITH ADAPTIVE MULTI-BAND DIRECTIONAL MICROPHONE AND FIXED DIRECTIONAL MICROPHONE IN ADULTS USING BINAURAL HEARING AIDS"** is a bonafide work submitted in part fulfillment for the Degree of Master of Science (Audiology) of the student with Registration Number 14AUD019. This has been carried out under the guidance of a faculty of this institution and has not been submitted earlier to any other University for the award of any other Diploma or Degree.

Mysuru
May, 2016

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CERTIFICATE

This is to certify that this dissertation entitled **"PERFORMANCE MEASURES OF SPEECH INTELLIGIBILITY IN NOISE AND LOCALIZATION WITH ADAPTIVE MULTI-BAND DIRECTIONAL MICROPHONE AND FIXED DIRECTIONAL MICROPHONE IN ADULTS USING BINAURAL HEARING AIDS"** has been prepared under my supervision and guidance. It is also certified that this has not been submitted earlier to any other University for the award of any other Diploma or Degree.

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DECLARATION

The dissertation entitled **"PERFORMANCE MEASURES OF SPEECH INTELLIGIBILITY IN NOISE AND LOCALIZATION WITH ADAPTIVE MULTI-BAND DIRECTIONAL MICROPHONE AND FIXED DIRECTIONAL MICROPHONE IN ADULTS USING BINAURAL HEARING AIDS"** is the result of own study under the guidance of a faculty at All India Institute of Speech and Hearing, Mysuru, and has not been submitted earlier to any other University for the award of any other Diploma or Degree.

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Abstract

Adaptive directional microphones are the one which automatically changes its polar patterns and sets it null in the direction of the noise source and thus enhances signal to noise ratio. On the other hand, fixed directional microphones improves signal to noise ratio only when the desired signal comes from the front of the listeners and noise from any other direction. Hence, adaptive directional microphone may allow for better understanding of speech in noise in real life situations when compared to fixed directional mode. To demonstrate such possibilities, in the present study, SNR-50 and localization performance for 20 individuals with mild-to-moderate hearing impairment with adaptive directional microphones was compared with fixed directional mode. SNR-50 was tested under four conditions, wherein target sentences were presented from 0^0, 90^0, 180^0 and 270^0 one by one while noise came from other three speakers. For SNR-50 experiment, results of the study showed that there was significant difference between fixed and adaptive mode when target sentences were given from 0^0 azimuth speaker. Whereas when target sentences came from any other direction other than front, there was no significant difference observed between fixed and adaptive directional mode. For localization performance, there was a significant difference obtained between adaptive and fixed directional microphones. Adaptive microphone showed better performance compared to fixed directional microphone. In conclusion, hearing aids performance in improvising localization was supreme in adaptive directional mode compared to fixed mode. However, both adaptive and fixed directional modes are almost equal in performance except a marginal benefit when signal is from 0^0 azimuth in adaptive mode.

Chapter 1

Introduction

Binaural listening is required for better understanding of speech in noisy environment and it helps in localization as well, which has been reported in literature by various studies wherein binaural amplification has been proved to be advantageous (Markides, 1977; Haggard & Hall, 1982; Brooks, 1984; McKenzie & Rice, 1990; Balfour & Hawkins, 1992; Robinson & Gatehouse, 1995; Kobler, Rosenhall & Hanson, 2001). This is one of the greatest challenges faced by individuals with bilateral hearing impairment especially in noisy environment. Most of the hearing aids users usually complain poorer understanding of speech in noisy environments and this becomes one of the major reasons behind the rejection in usage of hearing aid (Kochkin, 1993). It is found that hearing impaired listeners requires more signal to noise ratio in general, to perceive speech in quiet as well as in degraded environment when compared to normal hearing individuals (Dubno, Dirks, & Morgan, 1984). One of the method for better understanding of speech and also for noise reduction is by using directional microphone, thus improving signal to noise ratio (Ricketts, 2001; Valente, 1999) as well improves wearer's satisfaction with the hearing aid in real life situations (Kochkin, 2000).

Many noise-reduction algorithms as well have been incorporated into hearing aids to overcome this issue (Thibodeau, 2014). Directionality incorporated in hearing aids includes omnidirectional, fixed directional and one of the newer technologies named adaptive directional microphones. Directional microphone improves SNR by reducing its sensitivity to sounds coming from sides and from back of the listener. Thus, assuming the signal from front and noise from other directions leading to a loophole in a situation wherein desirable signal originates from any other directions, but not from the front. Hence, leading to loss of audibility to the wearer due to reduced sensitivity by the microphones. Hearing aids with more directivity index will lead to more loss than any other hearing aid having lesser directivity index (Gravel, Fausel, Liskow, & Chobot, 1999).

Many hearing aids incorporate directional microphones in different styles of hearing aids available nowadays. These directional microphones have been designed in numerous ways, such as more number of microphones to enhance signal to noise ratio. Single microphone or dual microphone directional hearing aids works based on similar principle. Single microphones consisting of dual ports which is designed in such a way that it provides delay to the sound entering through back port reaches the diaphragm at same time thus, cancelling effect of each other and enhancing signal only from the front (Ricketts & Dittberner, 2002).

Directional hearing aids provide advantage for speech perception in noisy environment and this directional benefit varies across listeners and found to be 3.5 dB to 16.1 dB (Nilsson, Soli, & Sullivan, 1994). Cord, Surr, Walden and Olson (2002) found that directional microphone was beneficial to wearers in the presence of noise only when signal was in front of the listener and it was spoken nearer to the listener and as the noise became more diffused or with increase in reverberation choice of directional microphone seemed to be faded. Thus, in situations where noise came from any other direction other than front, wearer’s tend to switch over to omnidirectional microphone (Cord et al., 2002). In line with this study, another study conducted by the same group of people revealed that hearing aid wearers do not find any significant difference between omnidirectional and directional microphones. Walden, Surr, and Cord (2003) provided a solution to such problems by changing the directional mode to omnidirectional mode using a switch to accomplish this, thus providing better signal to noise ratio based on the situation and need of the wearer. Again, this will not be beneficial if the wearer doesn't change the switch from directional to omnidirectional or dexterity issues also limit its benefit in young and elderly populations. Automatic switches will do overcome this issue, which is now a day's implemented in hearing aids as adaptive directionality. Adaptive directional hearing aids operate by automatically varying the polar patterns, thus attenuating the unwanted noise from other direction by enhancing signal from certain other direction. It takes certain time for these hearing aids to set into its adaptive directionality and it is usually ranges from few milliseconds to more than 5-6 ms. It has been also reported that shorter time constants are required for directional polar pattern to adapt according to changing noise sources for e.g.; a hypercardoid pattern will be expected when there is a diffuse noise environment and dipole pattern would be expected when noise is located to the sides at 90^0 azimuth.

Yet another difficulty which hearing aid wearer faces in noisy situation is to locate sound in space which is important for awareness of the environment and for navigating within it. It helps in alerting individual about the potential danger in the environment by telling them about the signal source. Inter-aural level and time difference are being widely accepted as an important cue for left/right discrimination in the horizontal plane, whereas monaural spectral cues are predominant for front/back discrimination (Middlebrooks & Green, 1991). Inter-aural time difference cues is dominant for frequencies below 1500 Hz, and that intensity differences are more pronounced at frequencies above 1500 Hz (Wightman & Kistler, 1992). Several studies have suggested that it is the spectral information from 2000 to 5000 Hz that is important for front/back discrimination (Musicant & Butler, 1984). To identify and separate sounds coming from different directions, an accurate localization in the horizontal plane is necessary, which contributes to the ability to follow a conversation among a group of people. Hearing-impaired listeners frequently report having problems in such listening environments. Generally, the inter-aural difference cues are successfully used by unaided hearing-impaired listeners for left/right discrimination for sensorineural hearing loss is up to 50 dB and if the intensity level of the stimulus is adequate (Abel & Hay, 1996; Noble, Byrne, & Le Page, 1994; Noble, Byrne, & Ter-Horst, 1997; Rakerd, Vander-Velde, & Hartmann, 1998). Finer spectral features below 2000 Hz may also be important for front/back discrimination (Asano, Suzuki, & Sone, 1990). Thus, hearing aids tend to improve audibility of high frequency cues, in hearing aid wearers but do not always improve localization for listeners with hearing loss (Kobler & Rosenhall, 2002; Vanden, Klasen, Moonen, Van Deun, & Wouters, 2006). This can be maximized by using two hearing aids to improve binaural cues and this is usually reported that there is a better performance in horizontal localization experiment especially those with severe hearing loss (Byrne, Noble & Lepage, 1992; Kobler & Rosenhall, 2002). When wearers were fitted with directional microphones on both the sides auditory localization was affected, and up to 90^0 localization was poor (Keidser et al., 2006; Vanden et al., 2006). This is usually seen as directional microphone has their polar pattern which is less sensitive at the side or at the back of the head.

Thus, when directional microphone is used on both sides, the inter-aural differences are minimized and thus cut down the cues for localization which is not seen with omnidirectional microphone. This so happens because the distance between the front and the rear microphone produces a delay between the ears depending on the direction of the signal. Since, the polar patterns are shaped based on the head and location of the signal with respect to noise, it is not same for hearing aids on both the sides, thus affecting inter aural time difference (ITD) cues (Keidser et al., 2006)

Need of the study

Fixed directional microphone shows benefit over omnidirectional microphone when desired signal is presented in front and noise arises from behind the listener (Valente, Fabry, & Potts, 1995). But, in certain quiet situations wherein desired signal is present from sides and back, fixed directional microphones shows reduced sensitivity (Valente, et al., 1995) leading to elevated aided thresholds and decreased speech recognition.

Directional microphone works well than omnidirectional when speech is at 0^0 azimuth and noise arises from 90^0 or 180^0 azimuth (Valente, et al., 1995). However, in real day today life situations speech can originate from any direction at any point of time. The directional benefit may vary substantially in such conditions depending on the direction from where the speech comes from. Whereas, adaptive microphones reduces these negative effects and provided a significant 3.8 dB improvement in SNR over the omnidirectional microphone in noisy situation (Kuk, Keenan, Lau, & Ludvigsen, 2005), proving that fully adaptive microphones work like an omnidirectional microphone in quiet and directional when it is noisy background. All the previous studies which compared adaptive microphones directionality had been done only on monaural condition and in this study we attempt to see the benefit of these directional technologies in binaural condition.

Every individual's life styles are different at various environmental conditions. Keeping that in mind, it is important for an audiologist to set the directionality of the individual's hearing aids while programming according to the life style of the client.

Most of the hearing aid users are not completely satisfied with the hearing aid benefit in competing situations; programming for the appropriate directionality in hearing aids helps the hearing aid users to hear better in competing situations. Directionality also plays a main role in understanding speech in noisy situations, so setting the appropriate directionality benefits the hearing aid users and thus giving comfort while hearing better in noisy situations.

Consequently, this study is carried out to compare the benefit of adaptive multiband digital directionality vs. fixed directionality hearing aids in individuals fitted with binaural aid in competing noise situations and also to evaluate the localization ability found using the new adaptive directionality microphones. Further, the finding throw light for Audiologist's as to which directionality setting is better to adopt while programming depending on the daily life situations of the individuals.

Aim of the Study

The study aims to compare the benefit derived from adaptive multiband digital directional microphone vs. fixed directional microphones in binaural aided condition.

Objectives of the study

1. To check the speech identification in noise (SNR-50) performance with adaptive directional microphone and fixed directional microphone.
2. To check localization performance using adaptive directional microphone and fixed directional microphone.

Chapter 2

Review of Literature

2.1. Effect of hearing loss on speech perception

Understanding of speech remains one of the major complaints for hearing aid users specially when there is noise in background. Even individuals with normal hearing also face this difficulty, but the amount what hearing impaired individuals face is always more. It is reported in literature that SNR-50 of the individuals with hearing loss is found to be 30 dB higher than that of individuals with normal hearing (Baer & Moore, 1994; Dirks, Morgan, & Dubno, 1982; Duquesnoy, 1983; Eisenberg, Dirks, & Bell, 1995; Festen & Plomp, 1990; Killion, 1997; Killion & Niquette, 2000; Peters, Moore & Baer, 1998; Plomp, 1994; Tillman, Carhart, & Olsen,1970; Soede, 2000). The exact amount of SNR loss is found to vary from degree, type of hearing loss and the type of background noise (Chung, 2004).

Killion (1997) studied SNR-50 in individuals with normal hearing and comparison across different degrees of hearing loss was carried out. Speech in noise test was administered, wherein noise consist of 4-talker babble (three females and one male) and speech consisted of a recording from female talker delivering the IEEE sentences at different SNRs of 0, 5,10 and 15 dB; scoring was done for 5 key words per sentence. Results of this study showed that individual with normal hearing requires 1-2 dB SNR for 50% correct score, whereas individuals with hearing impairment with mild to moderate hearing loss requires an SNR of 4-6 dB, moderate-to-severe loss required 7-9 dB and those with severe-to-profound requires even higher SNR of 12-18 dB.

Since many years, improving signal to noise ratio has been targeted by various amplification schemes. None of the hearing aid users refuses to take any such schemes as they receive most difficulty while communicating in background noise (Rickets, 2001). It is reported by Glasberg and Moore (1989) that individuals with normal hearing required more SNR and it is approximately +6 dB for satisfactory communication. Even individuals with hearing impaired require more SNR than normal for satisfactory communication (Carhart & Tillman, 1970; Cooper & Cutts, 1971; Dirks, Morgan, & Dubno, 1982; Groen, 1969; Killion, 1997; Plomp, 1976; Schum, 1996; Suter, 1985). Hearing impaired population which is most affected by poorer signal to noise ratio are children (Crandell, 1993; Crandell & Smaldino, 2000; Finitzo-Hieber & Tillman, 1978), wherein they need good signal-to-noise ratio in educational settings. Recommended SNRs for educational settings are at least +15 to +30 dB (Berg, 1993; Bistafa & Bradley, 2000; Blair, 1990; Crandell & Smaldino, 1995).

2.2. Improvement in speech perception through hearing aid technologies

To improve speech perception in individuals with sensorineural hearing loss, hearing aids remains one of the common rehabilitation option. Omnidirectional microphones in hearing aids which is used as being default in hearing aids have shown to increase audibility of speech, but fail to give adequate signal to noise ratio irrespective of whether it is analog (Killion &Villchur, 1993; Killion, 1997; Plomp, 1978; Tyler & Kuk, 1989; VanTasell, 1993; Verschuure, Benning, VanCappellen, Dreschler, & Boeremans, 1998) or digital (Ricketts & Dahr, 1999; Walden, Surr, Cord, Edwards & Olson, 2000). It has been reported in literature that listeners who have standard microphone require more favorable signal to noise ratio when compared to normal hearing individuals in order to compensate for the SNR loss due to their poorer hearing thresholds (Rickets, 2001).

2.2.1. Compression

Individuals with sensorineural hearing loss report major difficulty tolerating loud sounds and not understanding speech if it is made soft and this difficulty is due to the loudness recruitment seen in these patients and consequently reduced dynamic range (Fowler, 1936; Moore, 2007; Steinberg & Gardner, 1937). Nowadays, most of the hearing aids have some type of compression to deal with these issues in individuals with hearing impairment which works to provide more gain for low level sounds and lesser gain for high level sounds in order to overcome the difficulty of speech understanding with soft sounds and tolerance issues with high level of sound respectively. It receives controversy whether fast acting compression will be optimum or slow acting compression will be a better option (Moore, 2007).

Verschuure et al., in 1998 studied the effect of fast compression in moderate and severely hearing impaired individuals in quiet as well as in noisy situations. Noise as a stimuli selected for the study were multi-talker babble, industrial noise, printing office and a city-noise. They found that very few participants benefited from the compression in presence of noise and those who had poorer speech discrimination scores, they were the one who benefited this technology.

Walden et al., in 2000 compared the performance of linear hearing aid with input compression and two channel wide dynamic range compression. A total of 40 participants were involved in the study wherein 21 were fitted with linear automatic gain control (AGC) with input compression and another 19 participants wore digitally programmable analog two-channel WDRC. Connected Speech test were presented in quiet as well as in degraded environment and subjective ratings were also obtained. Results showed very little benefit with WDRC over linear AGC. Thus, hearing aids having compression as a technique to improve speech intelligibility provides audibility as well as comfortable signals across a wide range of listening environments in quiet without adjusting volume controls but its benefit in noisy conditions remains questionable. Yet another technology in hearing aid to improve signal to noise ratio is directionality through hearing aid microphone.

2.2.2. Directionality

Individuals with sensorineural hearing loss show great amount of difficulties in understanding speech when the environment is noisy (Plomp 1978; Dirks et al., 1982) and they tend to require more signal to noise ratio when compared to listeners with sensorineural hearing loss (Hawkins & Yacullo, 1984; Festen & Plomp, 1986; Plomp, 1986; Peters et al., 1998). Thus, hearing aids which incorporated technologies such as directionality to attain better signal to noise ratio is clearly of interest in present scenario. Traditional hearing aids having omni-directionality doesn’t seem to improve SNR and it has been reported that in few cases it may even reduce the SNR when used in behind-the-ear (BTE) hearing aids when compared to unaided condition (Ricketts, 2000a). Rather than having omnidirectional microphones, two microphones ports opening to the same diaphragm or having multiple microphones to form a microphone array can be used as directional microphones. These directional microphones are designed in such a way that they are more sensitive to signals coming from one direction and sensitivity to sounds coming from any other direction is reduced or almost negligible. Sensitivity of these microphones is usually for the signals arriving from front than any other angles in the space. These microphones works based on the assumption that wanted signal comes from the direction of higher sensitivity of the microphone and unwanted signal arrives from the direction of lower sensitivity, then it will improve signal to noise ratio. In hearing aids, directionality has been incorporated since decades and has been found to achieve its goal in laboratory as well as simulated real world environments (Madison & Hawkins, 1983; Hawkins & Yacullo, 1984; Valente et al., 1995; Preves et al., 1999; Ricketts, 2000b; Ricketts & Hornsby, 2006; Wu, 2010).

Ricketts and Dhar (1999) compared the performance of hearing impaired individuals fit with both directional and omnidirectional modes. 12 hearing impaired individuals were taken for the study fitted with behind-the-ear hearing aids. Hearing in Noise Test and Nonsense Syllable Test were taken for the study. Results of this study revealed that speech recognition in noise scores were significantly better with directional hearing aids when compared to omnidirectional hearing aids.

Gravel et al. (1999) studied speech recognition in noise using omni-directional and dual microphone technologies in twenty children with bilateral mild to severe hearing loss. Speech signal was delivered from 0^0 whereas noise was presented from 180^0 and SNR-50 was obtained. It was seen that children obtained 4.7 dB advantages in dual microphone technology when compared to omnidirectional. Thus, concluding better performance through dual-microphone in background noise for word and sentence test over omnidirectional mode.

Ricketts and Mueller (2000) studied the effect of directional benefit across different audiogram slopes, high frequency hearing loss and aided omnidirectional performance for speech-in-noise task. 80 individuals with symmetric, mild to moderately severe sensorineural hearing loss were taken for the study. Hearing in noise test was used in the study to check the benefit obtained from directional hearing aid. Results of this study revealed that there is no correlation obtained between configurations of hearing loss or aided omnidirectional performance and scores obtained through directional hearing aid. Further, these investigators found that with use of directional hearing aid there was additional benefit seen and hence suggested that directional amplification can be given to all individuals with hearing impairment irrespective of their performance through omnidirectional amplification and configuration of hearing loss.

Ricketts, Henry and Gnewikow (2003) studied the speech recognition performance and self-assessed questionnaire benefit measure for individuals using omnidirectional mode, directional mode with low frequency compensation and user selection option of directional/omnidirectional mode. Fifteen individuals with bilateral symmetrical sloping sensorineural hearing loss using in-the-ear directional hearing aids were taken for this study. Tests administered for the evaluating the performance were, the Connected Speech Test, the Hearing in Noise Test and Profile of Hearing Aid Benefit (PHAB) and results showed that speech intelligibility in noise improved with directional mode when compared to omnidirectional mode. Overall results showed that directional hearing aid showed benefit when speech was presented from the front and no difference when it came from any other direction other than front of the speaker.

Lewis, Crandell, Valente and Horn (2004) compared performance through omnidirectional and directional mode using the Hearing in Noise sentence Test. Noise was presented from 45^0, 135^0, 225^0, and 315^0 azimuths where 46 adult individuals with slight to severe sensorineural hearing loss were taken for the study. Results of the study showed significantly better performance through directional hearing aid when compared to omnidirectional mode.

Ricketts and Hornsby (2006) compared the performance in severe to profound hearing loss using the Connected Speech Test at multiple SNRs in omnidirectional and fixed directional mode. 20 individuals with symmetrical, severe-to-profound sensorineural hearing impairment were taken for the study and tested in auditory only and audio-visual mode both. Results of the study showed significant directional benefit at all SNRs in presence of visual information. Directional benefit showed significant benefit at minimum positive signal to noise ratio. However, directional microphone showed maximum benefit when signal to noise ratio was poorest in both audio only and audio-visual presentation modes.

Though these directional hearing aids are able to improve signal to noise ratio, perception of speech in noisy situation remains the common clinical complaint by potential hearing aid users and one of the major reason of hearing aid users dissatisfaction or disuse of hearing aid (Kochkin, 2007; Takahashi et al., 2007). Modern hearing aids which incorporate directional microphones and aims at improving signal-to-noise ratio requires a listening environment not having more than moderate reverberation, individual wearing hearing aid should be in front of the signal and unwanted signal should be coming from the back or the side of the listener which makes it an ideal condition (Valente et al., 1995, Ricketts & Dhar, 1999; Hornsby & Ricketts, 2007).

Thus, when signal and noise are present in a spatially complex environment, benefit provided by the directional hearing aid will depend on the listener's head which is pointed in a correct direction. When a hearing aid listener is using directional hearing aid in situations such as restaurant or a party should orient towards the signal of interest in turn. When there is on-axis listening (i.e., with the head pointed at the signal source) with directional microphone it has been reported that there is improvement in the speech intelligibility compared to the omnidirectional microphone (Ricketts & Dittberner, 2002). This difference in performance i.e., speech intelligibility in noise benefit is termed as directional benefit. Not only this, they get cues such as lip movements which aids in improvement of speech perception performance when head is oriented towards the directional microphone (Grant & Seitz, 2000; Grant, 2001). Speech intelligibility can improve up to 30^0 off-axis orientation of head when directional microphone is being used (Ricketts, 2000; Henry & Ricketts, 2003). Whereas speech intelligibility drops when the listeners head is oriented more than 60^0 from the signal of interest based on directional pattern which is termed as directional deficit. When the angle between the listeners head and the signal becomes more than 90^0 and 135^0 results in dramatic reduction in recognition scores even in quiet (Kuk et al., 2005).

Yet another method to improve signal to noise ratio is directionality provided through microphone in hearing aids itself which overcomes portability and overhearing related issues (Rickets, 2001). Directional hearing aids improve signal-to- noise ratio takes spatial information into consideration. Improvement in SNR seen with these directional hearing aids are small when compared to FM systems and it accounts to approximately 3-6 dB which aids in better perception of speech in noisy environment when compared to omnidirectional hearing aids (Rickets, 2001).

Previous investigations have shown that individuals using directional hearing aids do better in noisy environments when compared to those who are using omnidirectional across various laboratory, simulated real world situations and real world listening environments (Hawkins & Yacullo, 1984; Killion & Christensen, 1998; Nielsen & Ludvigsen, 1978; Preves, Sammeth, & Wynne, 1999; Pumford, Seewald, Scollie, & Jestad, 2000; Ricketts, 2000a; 2000b; Ricketts, Lindley, & Henry, 2001;

Valente, Schuchman, Potts & Beck, 2000; Voss, 1997; Wouters, Litere, &VanWieringen, 1999).

Directional hearing aids came into existence in the US in late 1970's and it was first described in ITE hearing aid in 1977. In 1980s, directional hearing aid was sold almost 20% of all the hearing aids that time (Mueller, 1981). Despite its benefit was reported by several studies during that time in hearing aids; its use seemed to reduce drastically by 1980s (Hillman, 1981; Lentz, 1972; Madison & Hawkins, 1983; Mueller & Johnson, 1979; Nielsen, 1973; Nielsen & Ludvigsen, 1978; Sung, Sung & Angelelli,1975) and this may be due to many factors such as relatively large microphone size and limited ability to change from omnidirectional mode to directional mode (Preves et al., 1999; Ricketts & Dittberner, 2002). Some of the hearing aids did provide with switches which slide over to provide directionality, but was quite small which lead to poor dexterity for potential users (Christensen, 2000). That time it has not developed to the extent to provide large increase in directivity when compared to omnidirectional hearing aids and it provided directivity in free field situations not in real life situation due to limited availability of head diffraction effect (Killion & Christensen, 1998; Ricketts & Dittberner, 2002).

Directional hearings aids can be designed using one microphone and providing acoustical phase shifting network or can be designed using two microphones by providing mechanical or electronic cancellation/delay; in order to suppress the unwanted signals coming from other directions (Ricketts & Mueller, 1999; Ricketts & Dittberner, 2002) and thereby enhancing the signal to noise ratio consequently. In both the types, either single microphone technology or dual microphone, shapes of directional pattern is obtained mainly due to external and internal delay provided (Preves, 1997; Ricketts & Mueller, 1999; Thompson, 1999). In the hearing aids which uses dual microphone technology, directionality is achieved by providing internal delay additionally, which is nothing but the time taken by the sound reaching the diaphragm through the rear microphone is delayed with respect to the sound arriving from the front microphone and it calculated by the external delay which is dependent on the microphone port spacing or two microphones itself.

Boymans and Dreshler (2000) compared performance from two noise reduction technologies i.e., active noise reduction and improved directionality by a dual/twin microphone design. Speech recognition in background noise along with paired comparison and self-report measurements were also performed on 16 hearing aid users. Results of this study showed positive effects by dual/twin microphone design whereas improvement through noise reduction technology was comparatively smaller.

For most of the directional hearing aids, the amount of internal delay to be given is fixed, which accounts for predetermined directional pattern. Newer technologies implement variable polar patterns which changes according to the listening environment. Variable polar patterns are obtained by two ways: one of them uses a circuitry which allows users themselves to select the appropriate directional pattern, another method is the one which adaptively switches between polar patterns in response to listening environments (Ricketts & Henry, 2002). In the past, all directional hearing aids had fixed polar patterns i.e., nulls were kept constant. However, as we know noise in reality can come from any directions and relative direction of speech and noise vary across situations. Thus, directional hearing aids with polar patterns fixed in one location may not be appropriate to obtain optimal directional benefit across situations. Newer technologies which have come up with variable polar patterns which are available in many hearing aids, which would be suitable for real life situations to provide directional benefit and improve signal to noise ratio. These adaptive directional microphones change their azimuths of null depending on the direction and location of noise. The goal of adaptive directional microphones remains same to provide maximum sensitivity to sound coming from front and minimum sensitivity to sounds coming from the back of the listener in noisy environments (Kuk et al., 2002a; Powers & Hamacher, 2004; Ricketts & Henry, 2002). It should be understood that switchless directional microphones in some of the hearing aids are not same as adaptive directional hearing aids, these switchless directional microphones are the ones which automatically switches between omnidirectional and directional mode whereas adaptive directional hearing aids changes its polar patterns according to the situation. However, most of the hearing aid available in the market having adaptive directional microphone switches automatically between polar patterns and modes as well. Adaptive directional microphones implemented in commercially available hearing aids nowadays are first order microphones and their physical construction is same as dual directional microphones. The difference lies in the signal processing algorithm wherein output from the individual omni-directional microphones can be varied and internal delay of the posterior microphone can be manipulated. The ratio between internal and external is varied in such a way that if ratio changes from 0 to 1, polar patterns shift from bidirectional to cardoid (Powers & Hamacher, 2004). Ideally, adaptive directional microphone is supposed to adapt to a polar pattern which places its nulls in the direction wherein maximum noise is detected. For example, the adaptive directional microphone should adopt the bidirectional pattern if the dominant noise source is located at the 90° or 270° azimuths and adopt the cardoid pattern if the dominant noise source is located at 180° azimuth. Different hearing aid manufacturers use different calculation method to find the dominant noise source and vary internal delay with different ratios in directional microphones accordingly and thus the null will vary based on the calculation method used and the noise sources available in the environment.

Adaptive directional hearing aids work in such a way that its polar pattern shifts automatically with changing noise source position in order to improve signal-to-noise ratio and once they are activated user doesn't have control over specific spatial attenuation pattern. There are no data which supports whether longer time constants or shorter time constants provides better directionality but it is clear that shorter time constants would provide faster directional patterns to adapt to directional microphones wherein null is avoided in front hemisphere. Thus, a hypercardoid pattern is expected when there is a diffuse noise environment, whereas a dipole pattern is expected to be obtained if the unwanted signal is directly from the side i.e., 90^0 from the listeners head (Ricketts & Henry, 2002).

Ricketts and Henry (2002) compared performance of speech recognition through adaptive and non-adaptive directional/omnidirectional processing simulating real world situations. Twenty individuals with symmetrical sloping mild-moderately/severe sensorineural hearing loss were selected for the study fitted binaurally with behind-the-ear hearing aid and material used for the study were Hearing in Noise test (HINT) and Connected Speech Test (CST). Results of the study showed that individuals received benefit from adaptive and non-adaptive mode over omnidirectional mode. However, improvement observed with adaptive and fixed (non-adaptive) remains similar in certain listening environments and there was significant advantage obtained by adaptive in speech recognition scores few specific conditions. This advantage was prominent when competing noise was given from the sides of the listener.

Kuk et al., 2005 compared performance of speech in quiet as well as in noise in omnidirectional, fixed and fully adaptive directional mode. Seventeen hearing impaired individuals were taken for study and speech was presented from 0^0 and 180^0 in 45^0 intervals. In quiet, performance through omnidirectional and fully adaptive directional was similar whereas in noisy situation, fully adaptive microphone showed similar performance as the fixed directional microphone.

2.2.3. Noise reduction algorithms

While a directional microphone takes advantage of spatial separation between the noise and the speech, noise reduction algorithms takes advantage of spectral separation between the speech and the noise. Noise reduction algorithms aims at reducing noise which interferes with the wanted signal and its ultimate goal is to increase listening comfort and speech intelligibility (Bunnel, 1990; Cheng & O'Shaughnessy, 1991). Noise reduction algorithms detect modulations present in the incoming signal to decide upon the presence of speech in the input signal and infer on the signal to noise ratio at the microphone output. It works on the fact that speech has modulations which is of the order of 4 to 6 Hz whereas noise has a constant temporal characteristic with lesser modulations which does not lie in the range of speech modulation rate. Moreover, speech has another characteristic which makes it different from noise feature i.e., co-modulations which is generated by opening and closing of vocal folds during the voicing of vowels and other voiced constants (Rosen, 1992). Noise reduction algorithms works with two detectors in general, one which detects synchrony i.e., co-modulations in speech and another which detects slow modulations present in the speech. Modulation detectors are intended to cut down the noise interference at frequency channels where noise is dominant. Theoretically, it has been reported that these modulation detectors are very effective in noise reduction when speech and noise has spectral differences. The only major limitation faced by these schemes is that they are unable to differentiate between desired signal and unwanted signal if speech acts as a competing signal. These algorithms first classifies speech and noise from the incoming signal, it tries to match based on the different characteristics of speech and noise. These noise reduction algorithms estimates the modulation depth and modulation rate in each channel to account for amount of speech, noise or both and calculates the SNR within the frequency channel (Boymans & Dreschler, 2000; Van Dijkhuizen, Festen & Plomp, 1991; Edwards, Struck, Dharan & Hou, 1998; Fang & Nilsson, 2004; Mueller, 2002; Powers & Hamacher, 2002; Walden et al., 2000). Whereas few other noise reduction algorithms works based on other dimensions such as they detect intensity-modulation-temporal patterns within each frequency channel (Tellier, Arndt, & Lou, 2003) or spectral-intensity-temporal changes in the incoming signal across different frequency channels (Kuk, Ludvigsen, & Muller, 2002b). Some of the manufacturers detect intensity patterns in the incoming signal (Kuk et al., 2002b) and is monitored over duration of 10-15 seconds in each frequency channel. These algorithms work based on the assumption that level of noise remains relatively stable when compared speech which varies rapidly within and across frequency channels. Other than detecting level of incoming signal in each frequency channels, estimation of SNR within each frequency channels is one of the options being used by many manufacturers. Here, in this technique modulation depth and modulation rate is calculated to estimate SNR i.e., if modulation depth is high then analysis units assumes that SNR in that particular channel is high and speech is dominant in that channels, hence noise reduction is not carried out. Similarly, if modulation depth is low then it assumes that noise is dominant in the frequency channel. Thus, the amount of noise reduction being implemented is usually inversely proportional to estimated SNR in that frequency channels (Kuk et al., 2002b; Powers & Hamacher, 2002; Johns, Bray & Nilsson, 2002; Edwards et al., 1998; Latzel, Kiessling, & Margolf-Hackl, 2003; Schum, 2003; Walden et al., 2000). This approach works on the rationale that if SNR estimated is high, then speech is in quiet and then action unit let the signal pass without attenuation. While noise reduction is applied, it is taken care that the amount of gain reduction is inversely proportional to the articulation index of the frequency region i.e., frequency importance weightage for frequency region which is important for understanding of speech (Kuk et al., 2002b; Kuhnelf & Launerf, 2003; Boysmans & Dreschler, 2000). Assumption behind taking articulation index in consideration is that, if importance for particular frequency channel increases, then reduction of gain in that particular frequency channel should be less as those frequency contains more information for speech understanding (Kuk et al., 2002b). Few manufacturers use different rule for gain reduction in channels, in order to provide importance to certain frequencies (Alcantara, Moore, Kuhnelf & Launerf, 2003; Tellier et al., 2003) such as gain is reduced only in frequency channels below 1 kHz and above 2 kHz, keeping in mind that that 1 and 2 kHz are important for understanding of speech. Therefore, gain is not reduced in 1 to 2 kHz channel even if modulation depth estimated is less in the incoming signal (Kuhnelf et al., 2003). Some of them estimate sound pressure level of the incoming signal or level of noise in addition to modulation depth and speech content in the frequency channel to apply gain reduction. Thus, when level of incoming signal increases, amount of gain reduction also increases consecutively (Kuk et al., 2002b). At the end, when gain reduction is carried out, there are types of time constants which determines the effectiveness of noise reduction and if any artifact is being generated.

Attack time or adaptation time constants are the one which determines the time between the noise reduction algorithm detecting noise in the channel and the time when gain is being reduced in that particular channel. Similarly, there are three more constants, speed of gain reduction, release time and speed of gain recovery, wherein it is the time taken by the gain reduction algorithm from the beginning to the maximum gain reduction, time between noise reduction algorithm detecting absence of noise and when its starts recovering the reduction and time between starting of the gain recovery and 0 dB gain reduction respectively.

As mentioned earlier, yet another category of noise reduction algorithms which detects fast modulations of speech across frequency channels take advantage of temporal separation between speech and noise and the rationale behind using faster modulations is that energy content of speech sounds is co-modulated by the opening and closing of the vocal folds during voicing of vowels and voiced consonants whereas noise is usually not co-modulated. This rate of co-modulations is the fundamental frequency of the person's voice. This algorithm detects synchrony in speech sounds and thus gain is reduced if no speech signal is detected in the frequency channel. Once, synchrony is detected these algorithms returns back and allows the signal to pass through the channel without any further reduction (Schum, 2003; Bachler, Knecht, Launer, & Uvacek, 1995). The goal synchrony detector algorithm is to improve listening comfort in the presence of noise but it is not beneficial when speech and noise co-exist or when speech itself is a competing signal.

One of the limitations with noise reduction algorithms is that they can misclassify music also as noise as music has higher modulation rate than speech. Noise reduction algorithm may also categorize conventional test signals such as pure tones or composite noise as noise during electro-acoustic measurements and thus reduce the gain for these signals and erroneous response output when these algorithms are engaged in testing. It has been also noted in literature that higher degree of noise reduction not necessarily provide better sound quality or better understanding of speech when compared to lower degree of noise reduction (Johns et al., 2002).

Walden et al., 2000, compared performance of hearing impaired individuals for omnidirectional, directional and with noise reduction algorithms. 40 hearing impaired individuals were taken for the study and the Connected Speech Test (CST) were presented to assess the benefit. Performance improved with directional microphone compared to omni mode for CST in noise, while small benefit was seen in everyday listening condition. Noise reduction algorithm provided improved listening comfort but scores on CST did not improve much.

Nordum, Erler, Garstecki and Dhar (2006) compared performance through directional microphone and noise reduction algorithms using Hearing in Noise Test. 16 experienced hearing aid users were taken for the study and were tested for both the technologies individually and combined effects were also studied. Results indicated that 50% of the participants performed better when both directional microphone and noise reduction algorithms were on simultaneously. Whereas, another 50% of the participants performed best with directional microphone alone.

Stelmachowicz et al., (2010) studied the effect of digital noise reduction algorithms on speech perception for hearing impaired children. 16 children with mild-moderately severe hearing loss participated in the study. All the participants were using behind-the-ear hearing aids having amplitude compression and noise reduction algorithms. Monosyllabic words from Phonetically Balanced Kindergarten (PKB) and 90 meaningful sentences (BKB) were presented along with speech shaped noise at 0, +5, +10 dB signal-to-noise ratios. Results of this test revealed similar results as obtained for adults in previous studies, suggesting there is no negative effect of noise reduction algorithms on overall perception of nonsense syllables, words, or sentences for all three SNRs tested.

2.2.4. FM technology

FM technology is another option available for individuals with hearing impairment, which has also shown improvement in speech perception for individuals having sensorineural hearing loss (Hawkins, 1984; Fabry, 1994). In personal FM systems, signal from the speakers' mouth is being picked up by the microphone which is wireless, present near the mouth of the speaker. Then acoustical signal is converted to electrical signal at microphone itself and then transmitted using FM signal to the listener's receiver. Thus, ensures that only signal is being picked from near the mouth of the speaker and hence, reduces noise pick up and in turn improves signal-to-noise ratio. Earlier investigations have shown 10 to 20 dB of improvement in speech perception in noise when compared to unaided conditions (Crandell & Smaldino, 2001).

Lewis et al. 2004 compared the performance of speech recognition in noise through directional hearing aid and personal FM technology. Total of 46 individuals with slight to severe SNHL were taken and Hearing in Noise Test was carried out. They reported that there was a significantly better speech perception scores in noise with FM technology when compared to directional hearing aids. They also reported further that FM was better significantly when compared to any hearing aid conditions.

2.3. Effects of hearing loss on localization

In horizontal plane, individuals with normal hearing use binaural cues for localization task though these cues varies with frequency of the signal but generally low frequency (<750 Hz) signals cue for inter-aural time differences (ITDs) and higher frequencies (>2000 Hz) cues for interaural level differences (ILDs) . When high frequency spectral information is presented monaurally, it helps to cue for front-back/vertical resolution/localizations (Middlebrooks & Green 1991; Slattery & Middlebrooks, 1994; Blauert, 1997). Whenever there is conflict in the auditory cues available for listeners, they rely more on interaural timing cues i.e., low frequency information (Rakerd & Hartmann 1985; Wightman & Kistler, 1992). Individuals with high frequency hearing loss also rely on low frequency cues for localization due to non-availability of ILDs and monaural spectral cues (Noble et al., 1994; Bronkhorst, 2000; Neher et al., 2009). Horizontal localization is important in day-to-day situations in order to separate and identify the signal coming from different directions in space, which is important as it helps individuals to follow conversation among a group of people.

Noble, Byrne and Lepage (1994) studied the effect of configuration and type of hearing impairment on sound localization. 87 hearing impaired individuals were taken for the study and compared with 6 normal hearing impaired individuals. 66 individuals with hearing impairment had sensorineural hearing loss mostly of cochlear origin with few of them having unknown etiologies and 21 having conductive/mixed disorders. Bursts of pink noise having 150 ms duration were presented and listeners were asked to identify the number of loudspeaker through which they heard the sound. Results of the study showed that 13 individual having conductive/mixed types with sensorineural hearing loss, who were matched for degree of hearing loss showed that conductive component added the difficulty faced by these individuals in localization.

Noble, Ter-Horst and Byrne (1995) assessed disabilities and handicaps associated with hearing impairment; a questionnaire was used as a part of this study which assessed sound localization difficulties in everyday situations and deleterious effect of localization impairment. 104 individuals with symmetrical hearing loss were taken were given with the questionnaires and results showed that hearing impaired individuals rated their localization skills as significantly worse than the control group not having hearing loss.

Localization of sounds in horizontal plane is much more dependent on audibility of the signal and the psychoacoustic characteristics of the listeners (Lorenzi, Gatehouse & Level, 1999b; Noble et al., 1994; Noble et al., 1997). When sensorineural hearing loss remains moderate degree, individuals with hearing impairment in unaided condition continue getting cues for inter-aural difference for left/right discrimination, only if audibility is provided. Whereas, this doesn’t hold well with presbyacusis individuals, wherein front/back discrimination depends on their ability to process spectral cues at high frequencies (Noble et al., 1994; Noble et al., 1997).

Rakerd, Velde and Hartmann (1998) studied affect of presbycusis on localization in median sagittal plane, 8 normal hearing listeners and 25 adults with presbycusis were given three localization tests, a frontal plane test, a sagittal plane test and an elevation test. Individuals with presbycusis and normal hearing individuals performed similar on frontal plane test wherein they were supposed to judge whether sound came from left, right or overhead which requires binaural localization cues.

Whereas, they performed poorer in sagittal plane and elevation which was dependent on spectral cues since, they faced difficulty in judging source elevation.

2.4. Improvement in localization through hearing aid with directionality

It is well known that, when a sound reaches both the ears with interaural time and level differences, it helps in spatial hearing (Blauert, 1997) which in turn improves locating speech in complex environments (Hawley, Litovsky & Colbum 1999). It has been reported in literature that these naturally available cues for localization gets altered by hearing aid processing due to bilateral asynchronies in directional microphone (Keidser et al., 2006). Binaural beamformer algorithms, wherein multiple microphones are used and gives single output in order to improve SNR tends to exacerbate difficulty faced by individual in localization. Few authors have reported many potential solutions to overcome these issues, such as providing open fit by large vents, thus creating a direct sound path to the ear canal. By doing so, it helps improving direct sound transmission to ear canal but remains limited only to higher frequencies. Few others make use of closed fitting along with 'cue preserving' algorithms (Desloge, Rabinowitz & Zurek, 1997; Picou, Aspell, & Ricketts, 2014).

Best, Mejia, Freeston, VanHoesel and Dillon (2015) evaluated the performance of two binaural beamformer in complex environment be assessing localization in noise, speech intelligibility, acceptable noise level, subjective ratings and a novel dynamic range speech intelligibility measure. 27 individuals wearing bilateral BTE hearing aids with a switchable option of selecting directional versus binaural beamformer microphone modes. It was noted that binaural beamformer remains superior only when target was presented from fixed frontal plane but did not show better performance always when target locations were moving. In individuals having conductive type of hearing loss, it has been reported that, hearing aids improve localization performance in horizontal plane as it increases air-to bone conducted sound, in turn aids in interaural difference cues (Byrne et al, 1995).

Kobler and Rosenhall (2002) studied benefit obtained for horizontal localization with bilateral or unilateral amplification in 19 subjects with mild-moderate hearing loss. Listeners were instructed to point the speaker from where target signal came. 8 loudspeakers were used, one loudspeaker was used to present target and another 7 was used to present noise. It was noted that bilateral amplification did help in horizontal localization whereas their ability to localize sounds decreased with unilateral hearing aids.

Several studies have reported that behind-the-ear (BTE) hearing aid deteriorate localization performance more than in-the-ear hearing devices (Hausler, Colburn, & Marr, 1983; Westerman & Topholm, 1985). This may be noted due to the microphone location in BTE hearing aids are more likely disrupting spectral cues important for front/back discrimination. Few others have also reported that localization performance doesn’t vary much with style of hearing aid as observed by Byrne et al., 1992 wherein BTE and ITE showed similar performance. Also, Noble and Byrne (1990) noted that listeners performed better with the style of hearing aid they were experienced with.

Neher et al., (2009) studied spatial hearing in 20 bilateral hearing aid users in multi-talker situation. Speech reception threshold was assessed in three talkers serving as a competing signal separated spatially in front –back or left-right plane. There was 14 dB variation in left-right speech reception threshold whereas only 8 dB in front-back SRTs. As we know that, head angle is an important aspect to be considered for detectability of sound in space, and to achieve this it is important that listener should be able to adjust themselves in the direction of the signal. Discrimination of front/back location can be improved with use of directional microphone.

Keidser et al. (2006) studied the effect of directional microphone, WDRC, noise reduction on performance of hearing aid users in horizontal localization. 12 individuals with bilateral hearing aid, BTE users were taken for the study presented with broadband pulsed pink noise in 360^0 loudspeaker array. It was noted that directional microphone showed significant improvement in performance of horizontal localization. Cardoid pattern showed reduced front/back error over time but left/right errors increased if the microphone fitted to the two ears were different.

Van den et al., (2006) compared the localization performance in frontal plane using omnidirectional and adaptive directional hearing aids. Ten hearing impaired individuals were taken for the study and it was noted that adaptive directional microphone had a significant negative effect on localization performance. On the other hand, Chung, Neuman and Higgins (2008) studied the effect of omnidirectional and an option of changing it to directional, on localization performance. Eight normal hearing listeners and eight experienced hearing aid users were taken for the study and individuals were asked to identify the direction of sound. They noted that directional microphone did not degrade localization performance at + 10 dB SNR and moreover it helped hearing aid users to localize sounds coming from back in quiet as well as in noisy situation.

Picou et al. (2014) studied gross localization abilities across three types of directional processing. They were divided as directionality of an unoccluded ear, adaptive directionality and cue preserving bilateral beamformer. Eighteen adults with sensorineural hearing loss, each of them fitted with bilateral hearing aids with three types of directional processing. Gross localization, sentence recognition, listening effort and subjective preference were evaluated in background noise. Results indicated that gross localization abilities in adaptive directional and in cue preserving bilateral beamformer showed similar performance when visual cues were also provided. When auditory only cues were given, localization accuracy reduced with bilateral beamformer when compared to adaptive directionality for signal presented from more than 60^0 azimuths.

To summarize, the findings of studies discussed in literature, it can be seen that individuals with sensorineural hearing loss faces difficulty in understanding speech in noisy situations and overall SNR required for hearing impaired individuals remains more than the individuals with normal hearing. We also found that, individuals with hearing impairment also face difficulty in locating sound in the environment due to affected spectral cues which helps in localizing sound in vertical plane. Also, due to hearing loss at high and low frequencies, ILD and ITD cues gets affected respectively, which is an important cue for localization in horizontal plane. It was also noted that there are different technologies available to overcome these issues, such as compression in hearing aids which improves increases gain for soft sounds and reduces gain for loud sounds. It is also noted that noise reduction algorithms also reduce noise selectively using different methods and thus allows only desired signal to pass through the hearing aid output. Yet another technology we had mentioned above is the FM device which directly picks up the desired signal near the speakers mouth and unwanted signal is not being picked up and thus enhancing SNR. Directionality is yet another technology available in hearing aids which improves signal to noise ratio by forming polar patterns and placing nulls in the direction of undesired signal. There are different directionality features available, such as omnidirectionality, fixed and adaptive directionality. All of three directional microphones, work differently and it has also been reported by fewer studied that there is a difference in performance by the three directional modes in presence of noise and for locating speech in space.

Although, benefit provided by noise reduction algorithms has been studied, noise reduction algorithms tend to reduce desired signal which has spectral characteristics similar to the speech. Similarly, FM technology provides maximum benefit in understanding of speech in presence of noise, but its use again remains limited due to portability issues. Further, an extra device has to be connected always to the hearing aid, requirement of speaker having a microphone and a receiver worn by the listener to obtain its benefit. Thus, there is need to compare the differences in performance offered by directional microphones which takes advantage of spatial separation between speech and the noise and attenuated noise accordingly and can be implemented in hearing aid without any need of extra connectivity. Hence, the present study is formulated to compare the differences in performance under speech recognition in noise and localization offered by fixed and adaptive directional mode which is mentioned in Chapter 3.

Chapter 3

Method

The methodology comprised of the following Phases:

Phase1: Selection of participants

Phase2: Routine audiological evaluation and hearing aid programming

Phase 3: Experiment to check the benefit of adaptive multiband digital directionality technology and fixed directionality on speech intelligibility in noisy situations

Phase 4: Experiment to assess localization of signal using adaptive and fixed directional technology

Phase 1: Selection of Participants

Twenty hearing impaired participants, in the age range of 18-55 years who fulfilled the following criteria were included in the study. Participants with bilateral post-lingual mild to moderate flat sensorineural hearing loss. Configuration was considered flat if there was less than 5 dB rise or fall per octave (Silman & Silverman, 1991).

- The difference between right ear and left ear thresholds should not exceed 15 dB HL (Gatehouse, Naylor, & Elberling, 2006),
- Speech identification scores not be less than 75% in both ears,
- ‘A’ or ‘As’ type of tympanogram with acoustic reflex thresholds appropriate to the degree of hearing loss,
- Naive hearing aid users, and
- Native speakers of Kannada Language.

Further, participants who were presented with one or more of the following were excluded from the study:

- Any history or presence of middle ear disorders,
- Any history or presence of neurological involvement, and
- Any history or presence of psychological problems.

Instrument Used

Clinical audiometer & immittance audiometer

- A calibrated dual channel diagnostic audiometer was used for obtaining the pure tone thresholds, Speech Recognition Threshold (SRT) and Speech Identification Scores (SIS). The audiometer was connected to the TDH 39 head phones housed in MX-41 AR cushion, Radio Ear B-71 bone vibrator and two loud speakers located at 45° angle for routine evaluation.
- Tympanometry and Acoustic reflex assessment was carried out with a calibrated middle ear analyzer.

Instrumentation for SNR-50 and localization experiment

- The speech intelligibility in noise experiment were assessed through the speaker located at 0°, 90°, 180^0 & 270° azimuth at a distance of 2 meter were used for presenting the target stimuli and the noise.
- HP computer with Intel processor, windows 7 was used wherein Cubase 6 software was installed and was used for presenting the sentences and speech babble.
- Eight speakers (Genelec-8020B) arranged in a circle at a distance of 2 m covering 0^0 to 360^0 angles for horizontal localization.

- Cubase 6 software was used for presenting the noise through eight speakers selected for the experiment.

Instrumentation for programming and fitting of hearing aid

- Two Behind-the-ear hearing aids having 12 channels with the following features were selected: with the facility of selecting fixed directionality (polar pattern as per the hearing aid specifications) and adaptive multiband digital directionality (within the same hearing aid),
- With the fitting range covering mild to moderate degree of hearing loss,
- With the option of disabling/enabling the above features, and
- A personal computer with NOAH-3 software connected with Hi-PRO, appropriate programming cable and hearing aid specific program were used to program the hearing aid.

Test Environment

Air conditioned sound treated double room set-up were used to administer all these tests.

Stimuli

- Speech identification scores were assessed using the Phonetically Balanced word lists (live presentation) in Kannada language developed by Yathiraj and Vijayalakshmi (2005) in the routine hearing evaluation.
- Speech intelligibility in noise was assessed using the sentence test in Kannada language developed by Chinnaraj, Kumar, Manjula, and Pavan (2014). This test has twenty five equivalent lists with ten sentences each.
- Total of ten sentences were calibrated for a specific SNR and for a specific speaker.

- Stimulus was calibrated such that 65 dB SPL of sentence from one particular speaker and 65 dB SPL of speech babble altogether combined output from three different speakers would correspond to 0 dB SNR (For example: 65 dB SPL of sentence from 0^0 speaker = 65 dB SPL of speech babble from 90^0, 180^0 & 270^0 would correspond to 0 dB SNR. Similarly, other six SNRs were also calibrated by varying level of speech babble from three speakers)
- Speech babble included in the SNR-50 experiment was babble from 6 talkers made for the experiment.
- Broad band noise (BBN) of 250 msec was used for localization task which was generated using adobe audition version 3.0.
- BBN for localization experiment was calibrated to give an output of 70 dB SPL.

Phase 2: Procedure for Routine audiological evaluation

Step 1: Hearing Evaluation

Pure tone thresholds were obtained using the calibrated dual channel diagnostic audiometer using modified Hughson and Westlake procedure (Carhart & Jerger, 1959). This was done across frequencies from 250 Hz to 8000 Hz for obtaining air conduction thresholds and for 250 Hz to 4000 Hz for bone conduction thresholds. Pure Tone Average (PTA) was taken as an average for the air conduction thresholds for the frequencies 500 Hz, 1 kHz, 2 kHz and 4 kHz.

Speech recognition thresholds were obtained to correlate with PTA using Kannada paired words. Speech Identification Scores (SIS) were obtained at 40 dBSL (re: SRT) using the PB word lists in Kannada language developed by Yathiraj and Vijayalakshmi (2005). UCL for speech were measured.

Immittance Evaluation was done on all the individuals to rule out any middle ear problems. Tympanometry and Acoustic reflex using 226 Hz probe tone at 500 Hz, 1000 Hz, 2000 Hz and 4000Hz were assessed using standard procedures with GSI-Tympstar middle ear analyzer. Based on the results of the above tests, those participants satisfying the selection criteria were selected for further evaluations.

Step2: Hearing aid evaluation and hearing aid programming

- The participants were fitted with the digital BTE hearing aid programmed using the manufacturer specific software. Hearing aid was first connected through Hi PRO using a connecting cable and was programmed based on NAL-NL1 prescriptive formula.
- First fit was applied for most of the participants, for few of them it was manipulated based on their response on ling six sounds. A routing hearing aid evaluation was carried out by asking five questions and finding out SIS for phonetically balanced words at 40 dB HL. This was carried out for individual ears and for binaural fitting. Two speakers at 45^0azimuths were used for checking speech identification scores.

Phase 3: Experiment to check the benefit of Adaptive multiband digital directionality technology and fixed directionality on speech intelligibility in noise

Step 1: Calibration of the stimulus

Calibration was carried out in sound treated room wherein sound level meter (SLM) was kept in the center surrounded by 36 speakers. Speech babble which was routed through 3 different speakers was calibrated to get an output of 65 dB SPL for 0 dB SNR conditions. Sentence materials were routed through one of the speakers depending on the conditions mentioned below. Sentence material was routed to get an output of 65 dB SPL.

To get an SNR of +2 dB, speech babble routed through three different speakers were calibrated to get an output of 67 dB SPL. Similarly, to get different SNRs of + 4, +6, -2, -4 and -6 dB, speech babble of three different speakers were calibrated to get an output of 69, 71, 63, 61 and 59 dB SPL respectively.

Step 2: Testing Phase

Participants were seated in the sound-treated room wherein participant was facing a 0^0 speaker and the center of the head of each participant was at 2 meter away from each loudspeaker. Sound treated room has 36 speakers separated by an angle of 40^0 from each other. Four speakers were utilized for this experiment separated by an angle of 90^0 and the set up used in this experiment is represented in Figure 3.1. To carry out this experiment, two hearing aid of same model (as mentioned earlier) were taken and were programmed according to the hearing loss of the participants. Hearing aid was set to fixed directionality initially and the experiment was carried out and then it was changed to adaptive directionality with other features off. Speech intelligibility in noise was assessed using the sentence test in Kannada language developed by Chinnaraj et al. (2014). This test has twenty five equivalent lists with ten sentences each. The speech intelligibility in noise experiment was assessed in four different conditions:

Condition 1: Speech from 0^0 azimuth and noise from 90^0, 180^0 and 270^0 was presented. Sentences were presented at seven different SNRs. The levels of the speech sentences were varied and SNR-50 was found out. The participants were asked to repeat the key words in the sentences presented; the tester noted down the responses and each correctly repeated key word was awarded one point. If participant couldn't repeat the key words in the sentence at particular SNR, then SNR was increased. The SNR that resulted in 50% speech recognition scores were obtained. Before the actual test started, a practice session was given.

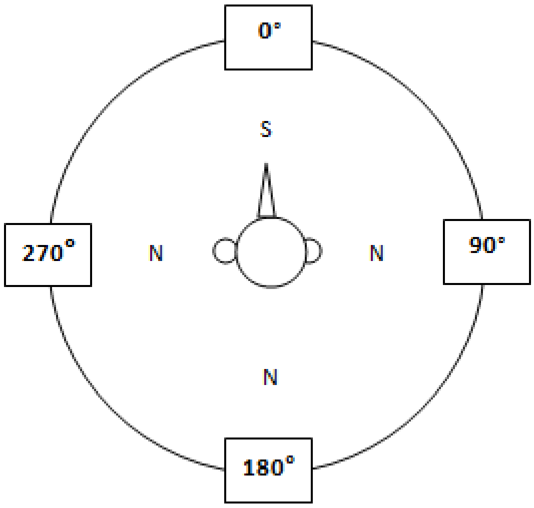

Figure 3.1: Experimental set up used for SNR-50 task.

Condition 2: Speech from 90^0 azimuth and noise from 180^0, 270^0 and 0^0 azimuths was presented.

Condition 3: Speech from 180^0 azimuth and noise from 90^0, 270^0 and 0^0 azimuths was presented.

Condition 4: Speech from 270^0 azimuth and noise from 0^0, 90^0 and 180^0 azimuths was presented and SNR-50 was calculated.

For calculation of SNR-50 across the conditions 2, 3 and 4, the procedure used in Condition 1 was repeated.

Participant's head movements were monitored to ensure that the participant was facing 0° azimuth by visual inspection. Test conditions were randomized and counterbalanced to reduce order effects. Each sentence was used only once in order to avoid practice effect. Other than directionality all other features were disabled in the hearing aid.

Step 4: Experiment to assess localization of signal using this adaptive and fixed directional technology

For localization experiment, localization lab was used and stimuli were presented through Cubase 6 software. A total of eight speakers (Genelec-8020B) were used for presentation of the target stimuli. All these 8 speakers were kept at 2 meter distance and angle of separation between each speaker was 40^0. Experimental set up for localization task is given in Figure 3.2. Broad band noise of 250 msec duration was used. Target stimuli were randomized and stimuli were presented through any of the speakers.

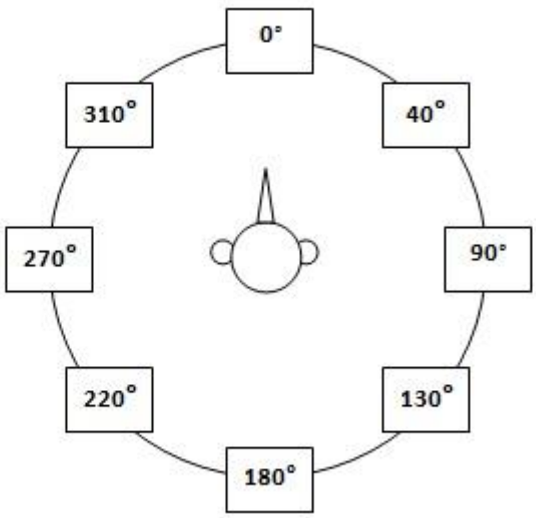

Figure 3.2: Experimental set up used for localization task

So, a total of 24 trials were given in both fixed as well adaptive directionality setting and after this degree of error was calculated using the following formula.

$$\text{RMS DOE} = \sqrt{\frac{(DOE_1)^2 + (DOE_2)^2 + \ldots\ldots\ldots + (DOEn)^2}{8}}$$

DOE_1: Degree of error of speaker no. 1

RMS: Root Mean Square

Degree of error corresponds to the difference between the angles of the speakers from where the target stimuli were presented to the speaker where the participant points to. For example, if the target stimulus was presented through fourth speaker (130^0) and participants points to 6^{th} speaker (220^0) then the DOE was 90^0.

Chapter 4

Results

The current study aimed to compare the benefits provided by two different directionality schemes in hearing aids under two objectives:

1) To obtain SNR-50 by fixed directionality and adaptive directionality and comparison of benefit obtained by these two directionality schemes.
2) To obtain localization performance and compare degree of errors provided by fixed and adaptive directionality schemes.

To fulfill the first objective mentioned above, experiment was carried out in four conditions by setting fixed and adaptive directional feature sequentially:

Condition 1: Target sentences presented from 0^0 and speech babble from 90^0, 180^0 and 270^0 speakers simultaneously.

Condition 2: Target sentences presented through 90^0speaker and speech babble from 0^0, 180^0 and 270^0 speakers simultaneously.

Condition 3: Target sentences presented through 180^0speaker and speech babble from 0^0, 90^0 and 270^0 speakers.

Condition 4: Target sentences routed through 270^0 speaker and speech babble simultaneously from 0^0, 90^0 and 180^0 speakers.

The results of the present study are explained under the following headings:

4.1. Experiment to compare the performance of adaptive multiband directionality vs. fixed directionality on speech intelligibility in noise.

4.2. Experiment to assess the localization performance and to compare the performance of using fixed and adaptive directionality feature in hearing aids.

4.1. Experiment to compare the performance of adaptive multiband directionality vs. fixed directionality on speech intelligibility in noise

SNR-50 was obtained using the sentence test in all four Conditions. Total of 20 subjects were tested for SNR-50 experiment, but data is represented only for 15 participants as data for five participants were dropped for the statistical analysis due to larger standard deviation which was making it statistically less valid. The mean and standard deviation (SD) of SNR-50 in all the four different conditions are mentioned in the Figure 4.1.

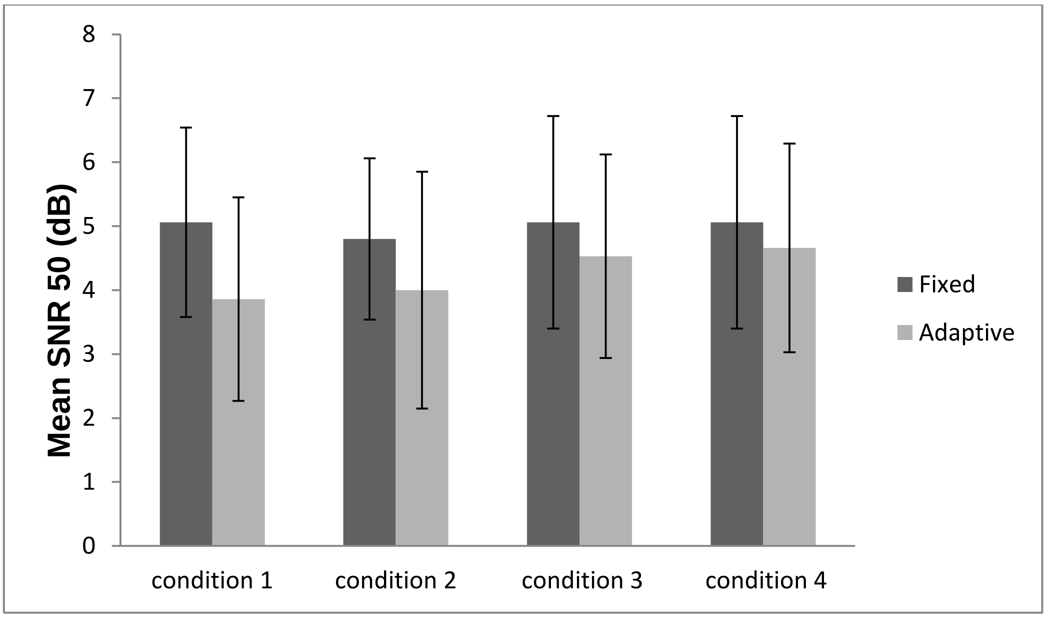

Figure 4.1: Mean and SD of SNR-50 for fixed and adaptive mode in all four conditions.

Lesser mean indicates a better performance on SNR-50 task and larger mean indicates a poorer performance. As shown in table, it can be seen that for adaptive condition mean of SNR-50 is lesser when compared to fixed condition for all degrees of target sentence presentation. Among all the four conditions, mean scores for 0^0conditions was better for adaptive directionality when compared to other conditions.

Since, the Shapiro-Wilk test showed non-normal distribution of data, non-parametric test was done. In order to check if the differences seen in the mean scores were significantly different or not, Wilcoxon signed rank test was carried out. The results of this test showed that there was significant difference between fixed and adaptive directionality only in Condition 1, i.e., 0^0 of sentence presentation, [$|z|$= -2.714, $p<0.05$], for other 3 conditions there was no significant difference i.e., Condition 2, $90^0$of target sentence presentation, [$|Z|$= -1.732, $p>0.05$], Condition 3, 180^0 of target sentences, [$|Z|$= -1.732, $p>0.05$] and Condition 4, 270^0 of target sentence [$|Z|$=-1.633, $p>0.05$].

To summarize the result of this experiment, there was significant difference between fixed and adaptive directionality when target sentences were presented from 0^0 speakers. Whereas there was no significant difference between the performance by fixed and adaptive directionality for Conditions 1, 2 and 3 wherein target sentences were presented from 90^0, 180^0 and 270^0 respectively.

4.2. Experiment to assess the localization performance and to compare the performance of using fixed and adaptive directionality feature in hearing aids

Results of localization experiment have been given under following four headings:

4.2.1 Comparison between fixed and adaptive modes for RMS degrees of error

Localization experiment was carried out using a 250 millisecond broadband signal presented through 8 different speakers placed at 0^0, 40^0, 90^0, 130^0, 180^0, 220^0,

270^0 and 310^0 azimuths. Degrees of errors were calculated for total of 20 participants involved in the study and data from 5 participants were dropped for the same reason as mentioned earlier. 15 participants were considered for obtaining the difference between fixed and adaptive directionality. Mean and standard deviation (SD) is depicted in the Figure 4.2.

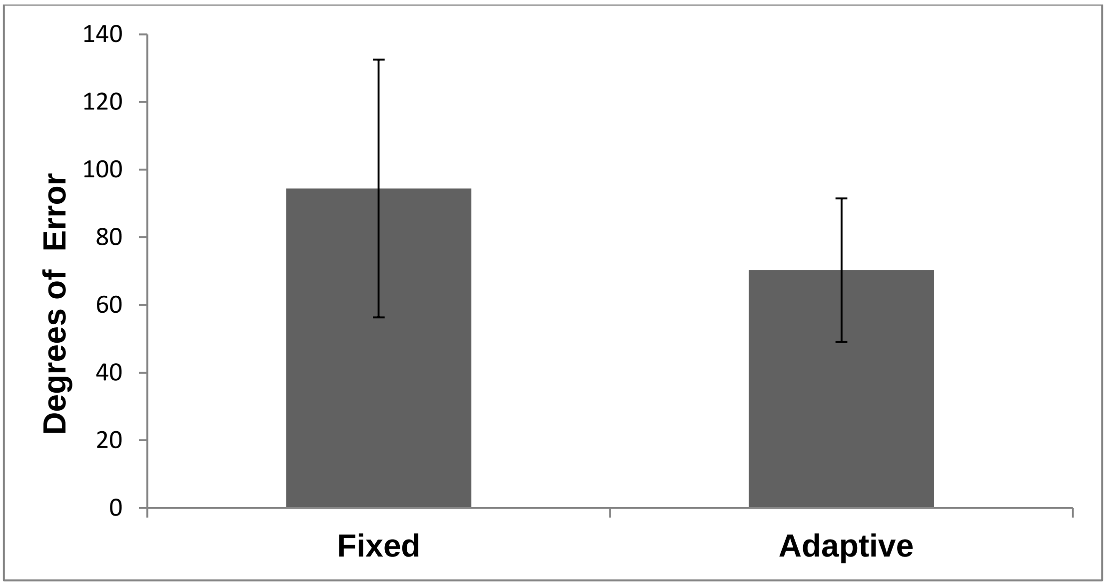

Figure 4.2: Mean and SD of degrees of error for fixed and adaptive conditions

Lesser mean values represent lesser degree of error which tells better performance in localization experiment. Adaptive directionality showed better performance when compared to fixed directionality, wherein mean scores of adaptive directionality is lesser than fixed directionality. In order to check whether the difference obtained is statistically significant or not, paired t-test was carried out as the data was within normal distribution curve. The results obtained from this test showed a significant difference between the fixed and adaptive localization experiment with [$t= 2.253$, $p<0.05$].

To summarize, the result of this section revealed that there is a significant difference between fixed and adaptive directional mode for localization.

4.2.2. Comparison between fixed and adaptive modes across degrees of stimulus presentation

Analysis was carried out to check the difference between fixed and adaptive directionality in terms of degrees of error at each degree of stimulus presentation. Mean and standard deviation was calculated, but mean was not a good representative of the data. Thus, median was calculated and is given in the Table 4.1 below.

Table 4.1. Median for localization in fixed and adaptive directionality for eight degrees taken for the study (N=15)

Azimuths	Fixed	Adaptive
0^0	0.00	0.00
40^0	0.00	0.00
90^0	40.0	0.00
130^0	0.00	40.0
180^0	40.0	0.00
220^0	40.0	40.0
270^0	50.0	50.0
310^0	40.0	40.0

In order to analyze whether there is any statistically significant difference between fixed and adaptive directionality for different azimuths, Wilcoxon Signed Rank test was carried out and results are represented in Table 4.2. There was significant difference seen in degrees of error when stimulus was presented only from 90^0 azimuths, between fixed and adaptive conditions.

Table 4.2. Wilcoxon Signed Rank Test representing Z and p values for localization experiment for fixed and adaptive conditions.

Conditions	Z value	p value
Adaptive 0^0-fixed 0^0	-.668	0.504
Adaptive 40^0-fixed 40^0	-1.379	1.68
Adaptive 90^0-fixed 90^0	-2.866	0.004*
Adaptive130^0-fixed130^0	-1.250	0.211
Adaptive180^0 -fixed180^0	-.837	0.403
Adaptive220^0 -fixed220^0	-.270	0.787
Adaptive270^0 -fixed270^0	-1.163	0.245
Adaptive310^0 -fixed310^0	-9.49	0.343

*Note:**significance level <0.05

It was found that there was statistically significant difference seen only when stimulus was presented from 90^0 speakers between fixed and adaptive conditions. But, looking into the median values (given in Table 4.1), it can be seen that there is a difference in values between fixed and adaptive modes for 90^0, 130^0 and 180^0 azimuths as can be seen in Table 4.3. Due to this, mean ranks were observed in the procedure of Wilcoxon Signed Rank Test for 90^0, 130^0 and 180^0. It was noted that differences in ranks were more for 90^0 when compared to 130^0 and 180^0 azimuths of stimulus presentation.

Table 4.3. Mean Ranks for adaptive and fixed modes for 90^0, 130^0 and 180^0

Conditions	Positive Ranks	Negative Ranks
Fixed 90^0 vs. adaptive 90^0	1.00	6.50
Fixed130^0 vs. adaptive 130^0	3.80	7.83
Fixed 180^0 vs. adaptive 180^0	6.50	5.07

To summarize, the result of this comparison, it was found that only 90^0 azimuth of stimulus presentation showed statistically significant difference when compared to other azimuths between fixed and adaptive mode.

4.2.3. Across degrees comparison for fixed mode

Comparison was done across the degrees for fixed conditions separately to check which azimuth yielded lesser degree of error and to find the difference. Friedman Test was carried out for fixed conditions and the mean ranks for the same are represented in Figure 4.4. Results of Friedman Test showed no significant difference across different azimuths, $\chi^2(7,15) = 6.106$, $p>0.05$. Thus, no further analysis was carried out for fixed mode.

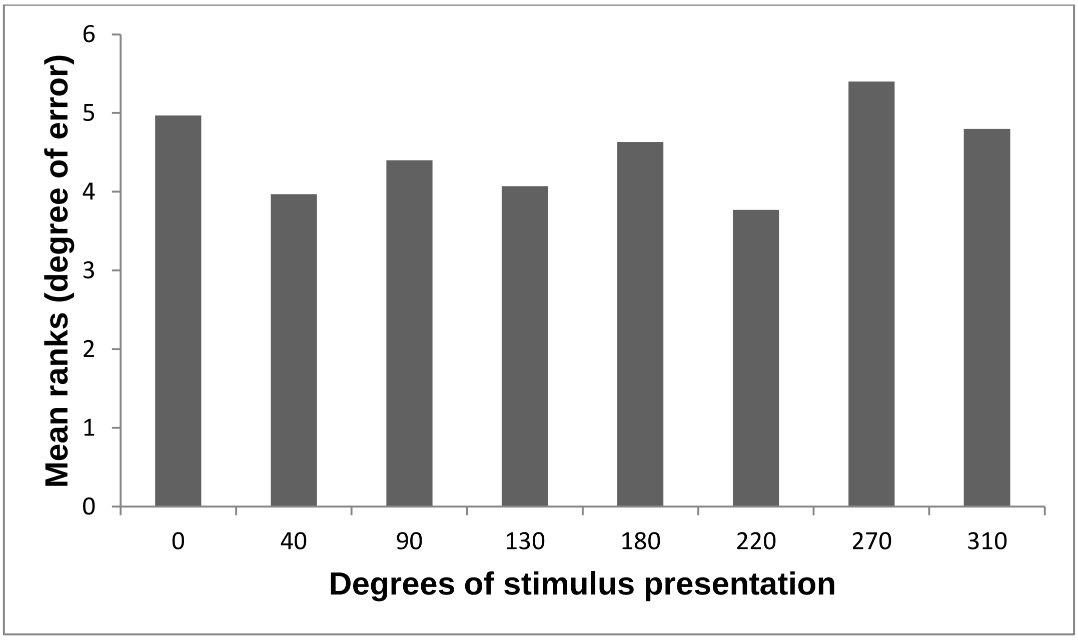

Figure 4.3: Mean Ranks of degrees of error for fixed conditions across degrees of stimulus presentation.

To summarize the result of this section, there was no significant difference observed across degrees of stimulus presentation when only fixed mode was active.

4.2.4. Across degrees comparison for adaptive mode

Further, analysis was carried out to compare the differences in degrees of error across azimuths for adaptive directionality. Friedman Test was carried out to check if there was any significant difference and the test revealed significant difference with $\chi^2(7,15) = 14.744$, $p<0.05$. Mean Ranks obtained in Friedman Test is depicted below in figure 4.4.

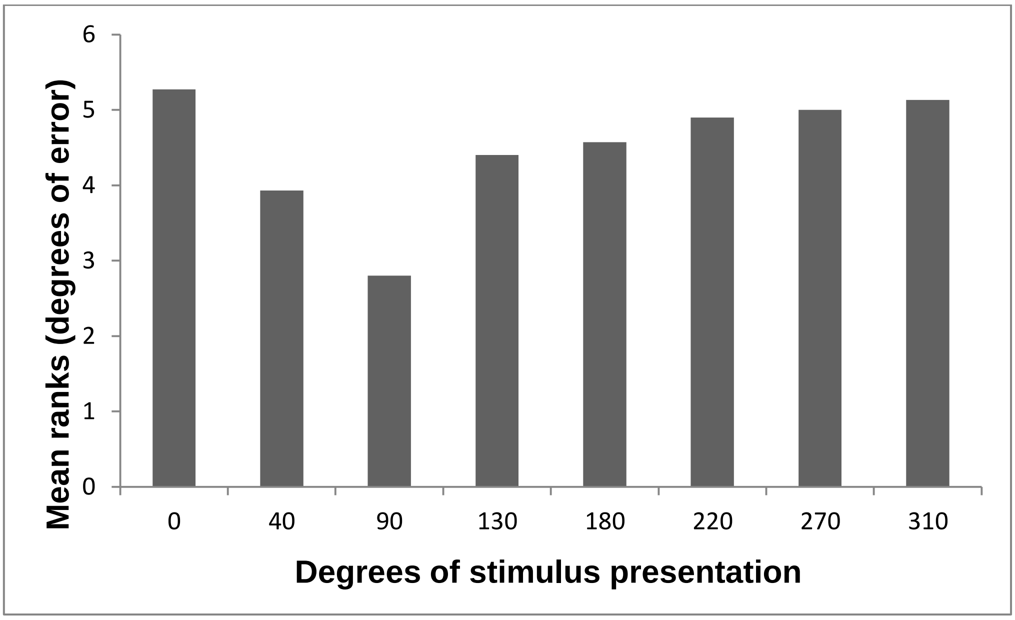

Figure 4.4: Mean Ranks of degrees of error for adaptive conditions across degrees of stimulus presentation.

Since Friedman Test showed significant difference; pair wise comparison was done using Wilcoxon Signed ranks Test in order to find which pair of degrees showed significant difference. There was a significant difference seen between the degrees of error at 0^0& 40^0 azimuths, 90^0 & 0^0 azimuth, 0^0 & 130^0 azimuth, 0^0 & 220^0 azimuth, 40^0 & 220^0 azimuth, 90^0& 130^0 azimuth, 90^0& 180^0 azimuth, 90^0 & 220^0 azimuth, 90^0

& 270^0 azimuth and 90^0 & 310^0 azimuth. Results of this Test is depicted in Table 4.4 given below.

Table 4.4.Wilcoxon Signed Rank Test for localization experiment across degree comparisons for adaptive condition

Conditions	Z value	p value	Conditions	Z value	p value
40^0- 0^0	-2.320	.020*	180^0 - 90^0	-2.047	.041*
90^0 - 0^0	-2.490	.013*	220^0 - 90^0	-2.385	.017*
130^0 - 0^0	-2.202	.028*	270^0 - 90^0	-2.495	.013*
180^0 - 0^0	-1.086	.278	310^0 - 0^0	-2.467	.014*
220^0 - 0^0	-2.005	.045*	180^0 - 130^0	-.513	.608
70^0 - 0^0	-1.891	.059	220^0 - 130^0	-1.084	.279
310^0 - 0^0	-1.892	.058	270^0 - 130^0	-1.079	.281
90^0 - 40^0	-1.754	.079	310^0 - 130^0	-1.169	.243
130^0 - 40^0	-.834	.404	220^0 - 180^0	-.051	.959
180^0 - 40^0	-1.265	.206	270^0 - 180^0	-.316	.752
220^0 - 40^0	-2.119	.034*	310^0 - 180^0	-.175	.861
270^0 - 40^0	-1.387	.166	270^0 - 220^0	-.073	.942
310^0 - 40^0	-1.030	.303	310^0 - 220^0	-.657	.511
130^0 - 90^0	-2.401	.016*	310^0 - 270^0	-.362	.717

Note: *significance level < 0.05.

To summarize, the result of this section it has been observed that there was statistically significant difference between 0^0 & 40^0 azimuths, 90^0 & 0^0 azimuth, 0^0 & 130^0 azimuth, 0^0 & 220^0 azimuth, 40^0 & 220^0 azimuth, 90^0 & 130^0 azimuth, 90^0 & 180^0 azimuth, 90^0 & 220^0 azimuth, 90^0 & 270^0 azimuth and 90^0 & 310^0 azimuths of stimulus presentations.

Chapter 5

Discussion

Under the current experimental conditions, this study demonstrated that SNR-50 required for individuals with hearing impairment remains same for fixed and adaptive directionality when target came from any other direction other than front. Adaptive directional microphone behaves like fixed directional microphone in presence of noise, i.e., when signal is from the front and noise is from any other direction, signal -to-noise ratio required is lesser. Whereas, when target sentence came from sides and back, adaptive directional microphone did not show any improvement in SNR when compared to fixed directional conditions. But, for localization experiment, adaptive directional microphone gave lesser degree of error when compared fixed directional microphones.

5.1. Adaptive vs. fixed directionality in hearing aids for SNR-50 in individuals with hearing impairment

5.1.1. Condition 1

When target sentence was presented from front i.e., 0^0 azimuth, whereas speech babble as a competing noise was presented simultaneously from 90^0, 180^0& 270^0, there was a significant difference seen in SNRs for adaptive and fixed directional settings. Wherein mean SNR obtained with adaptive directional microphone was 3.8 dB compared to 5.0 dB for fixed directionality. An SNR improvement of 1.2 dB was obtained when adaptive mode was used and the advantage obtained in terms of SNR can be attributed to the fact that adaptive directional microphones are sensitive to speech signals and places its nulls in all other angles from where noise is coming. Adaptive directional microphones can change their polar pattern accordingly and are sensitive to the direction of target signal and thus improves signal to noise ratio for the listener wearing hearings in adaptive mode. Ricketts and Henry, 2002; Fabry, 2005 and Picou et al. in 2014 also reports of similar findings. However, stimulus used and experimental conditions were not same. Ricketts and Henry (2002) reported better speech recognition performance when speech was presented from front whereas competing signal was given through 70^0 and 110^0 azimuths. Whereas, in this study we have used three loudspeakers simultaneously to present noise. Fabry (2005) also reported similar findings i.e., in adaptive directional condition hearing in noise thresholds were better irrespective of spatial separation between target signal and noise. In their experiment, noise source position changes from one sentence presentation to next and target sentence presentation was kept constant from front. Picou et al. (2014) also reported significantly better sentence recognition in adaptive directional processing condition. However, this finding is in dissonance with the finding reported by Kuk et al. (2005); they also presented target sentence from the front (0^0) and noise was presented from 45^0, 90^0, 135^0 and 180^0, but they did not find significant difference between the fixed directional and the adaptive modes.

5.1. 2. Condition 2, 3 & 4

When target sentence was presented from 90^0, 180^0 & 270^0 and competing multi-talker was presented from other three loudspeakers, mean SNR improvement obtained were 0.80 dB, 0.53 dB & 0.4 dB respectively, which is not statistically significant. This finding can be attributed to the fact that an adaptive directional microphone creates a directional pattern based on the noise source and searches for the area of most dominant noise source and places its null towards noise source and this area is usually restricted to the rear horizontal plane (Vanden et al., 2006). Hence, in Conditions 2, 3 & 4, when target sentence moves, area of noise moves in frontal plane and sides of the listeners also which is not restricted only to rear plane and thus attenuation might be lesser from the adaptive microphone. Another reason behind this finding can be the fact about adaptive directionality which works in such a way that when there is diffuse noise sources it obtains hypercardoid polar pattern when the listener turns their head towards the desirable signal, it assumes that signal coming from all other directions are undesirable and then the microphone places its null towards the direction of undesired signal (Kuk et al., 2005). Since, in the present study, listeners were instructed to face the 0^0 loudspeaker and were not allowed to move their head towards the direction of the target sentences, would have not allowed adaptive directional microphones to work to its fullest potential, i.e., if listeners would have moved their head towards the target sentences, then the adaptive microphone would have placed its null towards all other direction other than the direction in which listener has moved his head, assuming undesired signal from all other direction.

5.2. Adaptive vs. fixed directionality in hearing aids for localization in individuals with hearing impairment

5.2.1 Comparison between fixed and adaptive modes for RMS degrees of error

Localization experiment showed significant difference between adaptive and fixed directional condition. Wherein we found lesser degrees of error with adaptive mode compared to fixed directional mode. Mean degree of error obtained from adaptive directional mode is 70.2, whereas with fixed mode it is was 94.3, which is suggestive of better performance through adaptive mode compared to fixed directional mode (Picou et al., 2014). However, Vanden et al., (2006) report of better localization performance in omnidirectional mode compared to adaptive directional mode in most of the test conditions they used for the study.

5.2.2. Comparison between fixed and adaptive modes across degrees of stimulus presentation

When analysis was carried out to check which azimuth resulted in difference in terms of degrees of error between adaptive and fixed mode, it was found that 90^0 azimuth resulted in significant difference in degrees of error between the two modes. Finding of our study is incongruent with the findings reported by Vanden et al. (2006) wherein they report of negative impact on localization when stimulus was presented from $\pm 90^0$ azimuths, and reported better localization in front of the listener when compared to left and right sides for normal hearing and hearing impaired subjects. Findings of our experiment can be attributed to the fact that stimulus presented at 90^0 might show lesser degree of error due to availability of inter-aural level difference (ILD) serving as a cue when stimulus is presented from 90^0 speaker. Sound reaching to the ear will be more, as adaptive will place its polar pattern in the direction of the signal. Whereas signal coming from other seven speakers might not show improvement in localization due to unavailability of ILD cues as head will serve as an obstacles for sound to reach directly to the ear as compared to when it is given through 90^0 speaker. Results of our study is in the line with the findings of Markous and Middlebrooks (1990) and Carlile, Leong and Hyams (1997) where they observed that accuracy in localization performance was not much affected at the sides of the head when compared to frontal plane.

5.2.3. Across degrees comparison for fixed mode

When comparisons were made across degrees for fixed mode independently, results showed that there was no significant difference in localization performance across different degrees of presentation and this can be attributed to the fact that, fixed directional microphones having two microphone ports opening to the same port, are usually designed in such a way that it is more sensitive to the sounds coming from single direction and it is more often from the front (Ricketts, 2001). Hence, when fixed directional mode was active no improvement in localization performance was noted from any degrees of stimulus presentation.

5.2.4. Across degrees comparison for adaptive mode

Comparisons were made across different degrees of stimulus presentation when adaptive directional mode was active and there was a significant difference between degrees of error obtained for localization performance for different azimuths of stimulus presentation. It was observed that there was a significant difference seen between 0^0, 40^0, 90^0, 130^0 and 220^0, however degree of error was more for $0^0$when compared to other four azimuths. This finding can be endorsed to the fact that, when signal is coming from frontal plane (especially from 0^0), the inter-aural time differences (ITD) and ILD cues reaching both the ears would be same and thus creating cone of confusion in the median plane. At this point the ITD and ILD reaching to both ears are zero and listener can locate the sound anywhere in the median plane (Mills, 1972). Hence, ILD and ITD cues important for localization in horizontal plane will be affected and in turn degrading the localization performance. More degree of error noticed from 0^0 loudspeaker in our experiment could be due to the cone of confusion and thus made it difficult for listeners to locate the sound coming from the frontal plane.

Further, analysis was carried out to check the difference between 40^0 and other azimuths of stimulus presentation and it was found that significant difference was seen only between 40^0 and 220^0 wherein degree of error was lesser for 40^0 when compared to 220^0. This result of our study is in congruence with the results obtained by Vanden et al. (2006) wherein they report of more distortion in ILD and ITD cues at the sides of the listener when compared to localization performance in frontal plane for both individuals with normal hearing and hearing impaired as well hence, localization getting affected more at the sides.

It was also noted that with 90^0, there was a significant difference seen between 130^0, 180^0, 270^0 and 310^0, wherein degree of error was lesser with 90^0 when compared to other four azimuths of stimulus presentation. Head shadow effect would be minimum at 90^0 when compared to 130^0, 180^0, 270^0 and 310^0 and hence, more ILD cue obtained at 90^0, and thus better localization performance with adaptive mode.

Thus, while fitting bilateral hearing aids for individuals with hearing impairment, it should be taken care that, if they report of poorer understanding of speech in noisy environments, they should be fitted with a hearing aid having an option of adaptive mode, in order to provide them with better SNR. Also, if these individuals face difficulty in localizing sound in space, fitting them with hearing aid having an option of adaptive directionality would be better so that, localizing sounds from other directions would also be better for these individuals as adaptive mode will automatically change its polar pattern and thus keeps the microphone sensitive in the direction of the desired signal. Hence, fitting hearing aids for individuals with hearing impairment which has an option of adaptive directionality would be a better option to improve overall performance in terms of localizing sound in space and understanding speech in noisy situation.

Chapter 6

Summary and Conclusion

Adaptive directionality implemented in hearing aids tend to adapt different polar patterns based on the direction of the noise source by placing its null in direction of the unwanted signal thus said to improve signal-to-noise ratio for individuals with hearing impaired, who face difficulty understanding speech in noisy environments. However, there are no published reports evaluating effect of adaptive directionality when speech as well noise is moving. Further, there is no literature available that details about SNR-50 and localization performance in directional mode.

Hence, this study was aimed to check speech identification performance in noise (SNR-50) and localization with adaptive directional microphone and to compare it with fixed directional microphone. Fifteen hearing impaired individuals, with age range 18-55 years were participated in the study. They were evaluated using Sentence test developed by Chinnaraj et al., 2014, to assess SNR-50, where multi-talker speech babble served as a competing signal in this experiment. Broadband noise of 250 msec duration was taken for localization experiment. Experiment to assess SNR-50 was carried out under following four conditions:

Condition 1: Target sentences presented from 0^0 and speech babble from 90^0, 180^0 & 270^0 speakers simultaneously.

Condition 2: Target sentences presented through 90^0 speaker and speech babble from 0^0, 180^0 & 270^0 speakers simultaneously.

Condition 3: Target sentences presented through 180^0 speaker and speech babble from 0^0, 90^0 & 270^0 speakers.

Condition 4: Target sentences routed through 270^0 speaker and speech babble simultaneously from 0^0, 90^0 & 180^0 speakers.

Results of the first objective showed that there was a significant difference between fixed and adaptive directional mode for SNR-50, when target signal was presented from front of the listener and noise came from 90^0, 180^0 and 270^0. Whereas, there was no significant difference obtained when the target signal came from any other degree other than 0^0.

Results of the second objective revealed that localization performance improved when adaptive directional mode was used when compared fixed directional mode. It was observed that there was lesser degree of error obtained with adaptive mode than the fixed mode.

It can be concluded that adaptive directional mode helps for better speech performance in noise, but its usefulness remains questionable when speech comes from any other direction other than front and when noise occupies more of frontal and side planes in space. Further, localization performance improves using adaptive directional microphone mode compared to fixed mode. Hence, it can be concluded from the findings that, adaptive directional mode will be beneficial for speech understanding in competing situations and for locating sound present in the environment in those individuals using binaural hearing aid.

Clinical implications of the study

1. The findings of the study helps in proving that the benefits are higher from adaptive directional microphone compared to fixed directional microphone for better understanding of speech in noise and in localization. This information can be used when fitting hearing aids in clinics.

2. The finding from the present study is useful in making choices while performing clinical testing to evaluate the benefit provided by directional mode. While testing performance of speech understanding in noise, target

speech should be presented from 0^0 azimuth, in order to evaluate the benefit from adaptive mode to the fullest.

Limitations of the study

- Experienced hearing aid users would have also been taken as another group for the study, as for naïve hearing aid users understanding of speech through hearing aid would take some time (acclimatization factor). Further, the presentation of noise from three speakers at a time would have become even more challenging for the naïve hearing aid users in the present study.
- Participants should have been allowed to move head towards the direction of the target signal, so as to provide adaptive mode to work to its maximum potential.
- Number of participants taken for the study would have been more, so as to represent sufficient population of hearing aid users. In this study, experiment was carried out on 20 participants, but only data of 15 participants were included for appropriate statistical measures due to higher standard deviation.

Future directions

1. Similar, experiment can be carried out taking experienced hearing aid users, in order to check if the benefit provided by adaptive directional mode improves or remains the same.
2. Benefits obtained from adaptive mode when noise is in diffused environment, or when noise is restricted to only one direction can be studied to understand the effects of noise when sourced from various directions.
3. An experiment to compare the performance of adaptive directional microphone across different competing signal such as cafeteria noise, traffic

signal, party noise etc. could be carried out to understand its benefits under different types of competing situations.

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