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Effects of Reverberation and Amplification on Sound Localisation

By

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Abstract

Communication often takes place in reverberant spaces making it harder for listeners to understand speech. In such difficult environments, listeners would benefit from being able to locate the sound source. In noisy or reverberant environments hearing-aid wearers often complain that their aids do not sufficiently help to understand speech or to localise a sound source. Simple amplification does not fully resolve the problem and sometimes makes it worse. Recent improvements in hearing aids, such as compression and filtering, can significantly alter the Inter-aural Time Difference (ITD) and the Inter-aural Level Difference (ILD) cues. Digital signal processing also tends to restrict the availability of fine structure cues, thereby forcing the listener to rely on envelope and level cues. The effect of digital signal processing on localisation, as felt by hearing aid wearers in different listening environments, is not well investigated. In this thesis, we aimed to investigate the effect of reverberation on localisation performance of normal hearing and hearing impaired listeners, and to determine the effects that hearing aids have on localisation cues.

Three sets of experiments were conducted: in the first set (n=22 normal hearing listeners) results showed that the participants' sound localisation ability in simulated reverberant environments is not significantly different from performance in a real reverberation chamber. In the second set of four experiments (n=16 normal hearing listeners), sound localisation ability was tested by introducing simulated reverberation and varying signal onset/offset times of different stimuli – i.e. speech, high-pass speech, low-pass speech, pink noise, 4 kHz pure tone, and 500 Hz pure tone. In the third set of experiments (n=28 bilateral Siemens Prisma 2 Pro hearing aid users) we investigated aided and unaided localisation ability of hearing impaired listeners in anechoic and simulated reverberant environments. Participants were seated in the middle of 21 loudspeakers that were arranged in a frontal horizontal arc (180°) in an anechoic chamber. Simulated reverberation was presented from four corner-speakers. We also performed physical measurements of ITDs and ILDs using a KEMAR simulator.

Normal hearing listeners were not significantly affected in their ability to localise speech and pink noise stimuli in reverberation, however reverberation did have a significant effect on localising a 500 Hz pure tone. Hearing impaired listeners performed consistently worse in all simulated reverberant conditions. However, performance for speech stimuli was only significantly worse in the aided conditions. Unaided hearing impaired listeners showed decreased performance in simulated reverberation, specifically, when sounds came from lateral directions. Moreover, low-pass pink noise was most affected by simulated reverberation both in aided and unaided conditions, indicating that reverberation mainly affects ITD cues. Hearing impaired listeners performed significantly worse in all conditions when using their hearing aids. Physical measurements and psychoacoustic experiments consistently indicated that amplification mainly affected the ILD cues.

We concluded that reverberation destroys the fine structure ITD cues in sound signals to some extent, thereby reducing localisation performance of hearing impaired listeners for low frequency stimuli. Furthermore we found that hearing aid compression affects ILD cues, which impairs the ability of hearing impaired listener to localise a sound source. Aided sound localisation could be improved for bilateral hearing aid users, if the aids would synchronize compression between both sides.

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Declaration of authorship

I …Hadeel AlSaleh………. declare that this thesis entitled
“Effects of reverberation and amplification on sound localisation”
and the work presented in it are my own and has been generated by me as the result of my own original research. I confirm that:

1. This work was done wholly or mainly while in candidature for a research degree at this University;

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Effects of Reverberation and Amplification on Sound Localisation

Chapter 1

Introduction

1.1 Introduction and overview

Spatial acoustic cues are important for accomplishing many tasks, ranging from locating a sound source to detecting and understanding one source in the presence of competing sources from other locations. Sound localisation means the ability to place a sound source in space. The human auditory system uses binaural and monaural spectral cues in order to process sound for localisation and to enable it to be heard. Monaural cues, which are based on the quality of the sound as it enters the ear canal, refer to the filtering effects of external structures. Binaural cues are based on the processing of differences in intensity, time and frequency correlation between both ears in the central nervous system. Localisation can be affected by any number of factors, including age, hearing loss, gender, handedness, environmental conditions and sound variations.

An important factor in sound localisation is reverberation, which affects auditory perception and can provide listeners with a cue for sound distance. Reverberation refers to the acoustic environment that surrounds a sound, and is defined as the combined effect of multiple sound reflections within a room. Hearing loss may affect listening in reverberation by distorting the speech spectrum, therefore it is interesting to further investigate the effect of reverberation on speech localisation.

Sound localisation may also be affected by hearing aids, as these may affect the localisation cues. Sound localisation therefore can be both negatively and positively affected by hearing aids. Directional microphones may disrupt binaural cues for localisation, as they can artificially change the inter-aural level and phase differences, which are both critical cues for localisation. Group delays in hearing aids may interfere with inter-aural time difference cues, which are one of the most important cues for localisation. Conversely, hearing aids can improve localisation by providing audibility of high frequency signals. The degree to which localisation is affected therefore depends on the type of hearing aid – behind the ear, in the ear or in the canal – and whether the aiding is bilateral or unilateral. However, previous studies are contradictory as to whether bilateral or unilateral hearing aids are better when in terms of sound localisation.

The main purpose of our study was to investigate the contribution of adding reverberation on a listening environment to the abilities of normal hearing and hearing impaired listeners in localising speech and non-speech signals, and to evaluate whether it is necessary to add reverberant environments in clinical testing. Another objective was to determine if adding reverberation makes localisation more challenging for hearing impaired listeners in comparison to normal hearing listeners.

The second main purpose of our study was to understand the effect of the hearing loss and the hearing aid amplification systems on the utilization of the localisation cues in reverberant and anechoic environments. These two factors were investigated separately by comparing the hearing impaired performances with and without hearing aids in both listening environments.

1.2 Thesis Outline

Chapter 1: Introduction

Chapter 2 Background and Literature Review

This chapter introduces the concepts and literature on horizontal sound localisation in both normal and hearing impaired listeners. Background information on reverberation, and what is known on the effects of reverberation on our ability to localise sound sources in the horizontal plane, is also introduced.

Chapter 3: Experimental Design

This chapter describes the experimental set-up and apparatus used in the localisation experiments in the thesis. It will further provide some details of the stimuli, reverberation, and the methods, used in simulating different reverberant environments.

Chapter 4: KEMAR Measurements

This chapter reports the measurements used to investigate the effect of amplification and reverberation on localisation cues, which were achieved by measuring the inter-aural level and time differences on a KEMAR manikin, with, and without, hearing aids in both anechoic and simulated reverberant environments. The results from KEMAR measurements are compared to the experimental results collected in chapters 5 and 6.

Chapter 5: The effect of reverberation on localisation cues

Chapters 5, 6 and 7 address the effects of reverberation solely on the localisation performance, therefore, in order to rule out the effect of audibility, only normal hearing listeners were tested. In Chapter 5, the effect of reverberation on the inter-aural time and level differences of normal hearing listeners is measured. These results were obtained by comparing their localisation performance in both real reverberant and anechoic environments, by using speech stimuli and pure tones. For comparison, localisation performance was also measured in a simulated reverberant environment. Results from both listening environments showed no significant difference between results in real and simulated reverberation environments; this, therefore, permitted for the use of simulated reverberation throughout the thesis, and also for the testing of their localisation performance under a wider range of simulated listening environments in the Chapter 6.

Chapter 6: Comparison of localisation performances across different reverberant environments

The results from Chapter 5 suggested that the reverberation used in the previous experiment did not have a significant effect on localising speech stimuli. Therefore, a further attempt was made to investigate the effect of reverberation by varying some parameters in the listening environments and the stimuli. By measuring the localisation performance of normal-hearing listeners under different simulated reverberant environments, only the reverberation was altered. The results showed that adding reverberation to a listening environment did not seem to have an effect on the listener's ability to localise speech stimuli, even when longer reverberation times were used.

Chapters 7: Effects of the signal onset time and envelope on sound localisation

The results shown in Chapter 5 demonstrated a significant effect of reverberation on localising pure tone stimuli; however, no significant effect was found on localising speech stimuli, even after using longer reverberation times. This might be explained by the precedence effect – i.e. the ability to locate sound sources by the first wave that arrives in the listener's ears – which is thought to be used only in localising sounds with rapid onsets and transient sounds (Giguere and Abel, 1993). Therefore, this chapter describes the investigation into the effect of envelope and onset/offset times of the signal in localising a sound source. Findings from this study revealed a significant effect of envelope cues on localising the pink noise stimuli, however, there was no significant effect of signal onset/offset time.

Generally speaking, findings from Chapters 5-7 revealed that normal hearing listeners can accurately localise broadband signals under different listening environments. However, sound localisation is known to be more challenging for hearing impaired listeners, especially in the presence of background noise, therefore, in the next chapter, tests on the localisation performance of hearing impaired listeners will be described.

Chapter 8: Effects of reverberation and amplification on the localisation abilities of hearing aid wearers

This chapter shows the effect of hearing impairment and hearing aids on sound localisation, which was achieved by comparing the localisation performance for normal hearing listeners to the localisation performance of hearing impaired listeners, with and without their hearing aids, in both anechoic and simulated reverberant environments. Results suggested that older individuals with high frequency hearing loss found it more difficult to localise sound source when compared to normal hearing listeners; also, that amplification provided no benefit in both anechoic and simulated reverberant environments. It was found that amplification adversely affected the ability to localise when sounds contained mainly high frequency components – hence ILD cues were relied on – and that reverberation did have a significant impact on the localisation performance of the hearing impaired listeners when the sounds mainly contained low frequency components.

Chapter 9: Conclusions

This chapter contains a general discussion, conclusion and areas for possible future research.

The thesis outline is shown further in Figure 1.

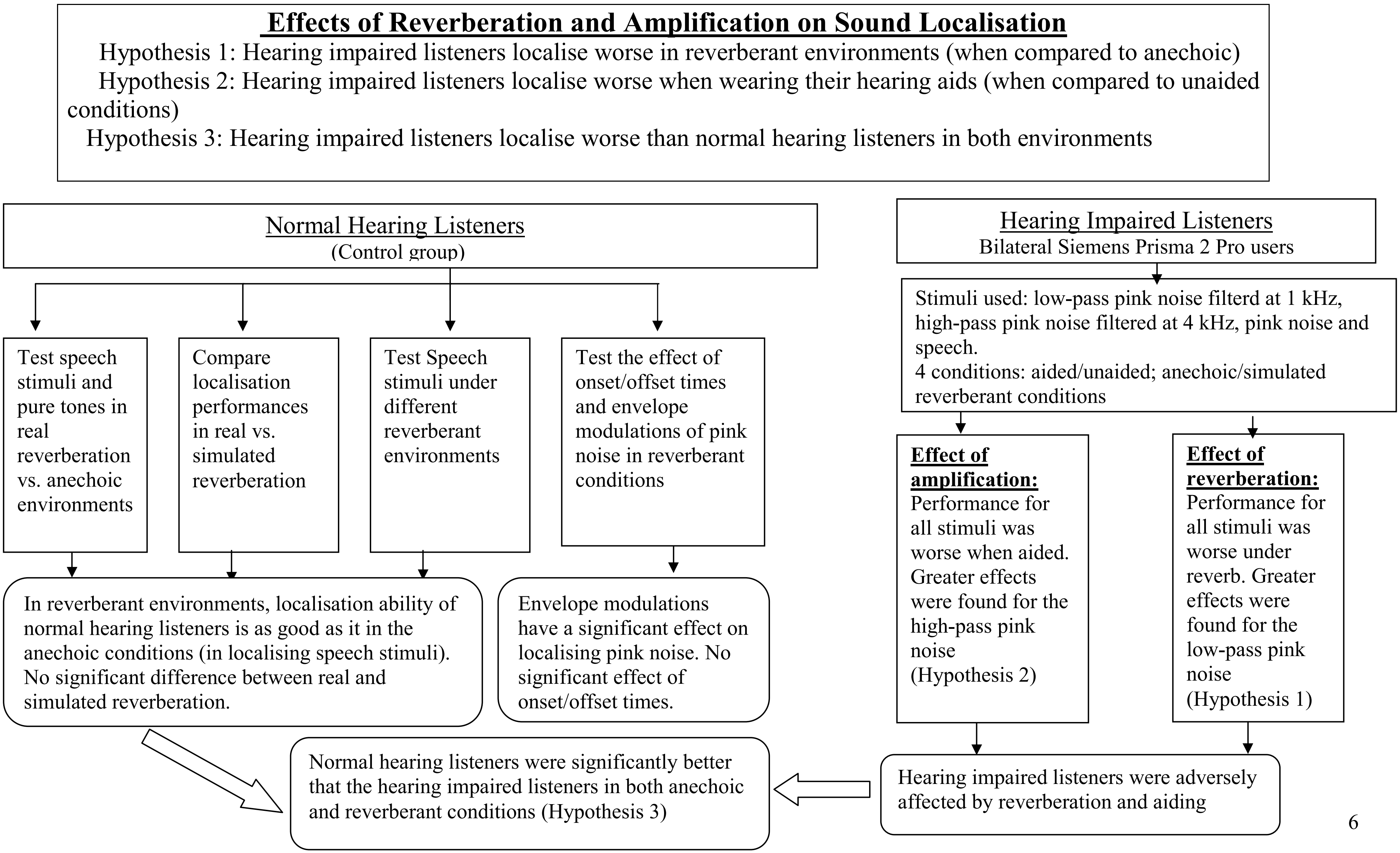


Figure 1 Thesis outline

Chapter 2

Background and Literature Review

2.1 Localisation

Localisation refers to the ability to determine the location of origin of a detected sound in space. The two mechanisms of sound localisation involve binaural and monaural cues. Binaural localisation depends on the comparison of the auditory input from two separate detectors. One of the binaural cues is the delay between the time when sound from a single source reaches the near ear, and when it reaches the far ear. This is also called the “inter-aural time difference" (ITD). The other cue is the reduction in loudness when the sound reaches the far ear, or the "inter-aural level difference" (ILD) (Raleigh, 1907 cited by Wightman and Kistler, 1993).

The location of a source is specified mainly in terms of the two parameters representing its direction - azimuth, i.e. its lateral direction with respect to the facing direction of the head, and elevation, i.e. its direction with respect to the ear-level plane. There is also one more parameter, which indicates its distance. Certain spatial tasks can be accomplished by direction alone. One task that can be accomplished by direction alone is the task of localising the direction of a threat upon which the head will turn to localise it; in this instance, distance is often essential, i.e. perceiving whether a threat is so close as to require an immediate response. For the purposes of this study, the important factor is horizontal plane localisation, which is explained below.

2.1.1 Sound localisation in the horizontal plane

Rayleigh (1907; cited by Middlebrooks & Green, 1991) suggested that, when a sound is presented from the side, the head acts like an obstacle that interrupts the path from the source to the far ear, which results in an inter-aural difference in the sound pressure level (ILD). Whereas, low frequencies have large wavelengths, allowing sound to diffract around the head; consequently, the angle of these wave fronts produces ITD cues.

2.1.2 Binaural cues for horizontal localisation

Binaural localisation, which enhances our ability to determine the direction of a sound source, is based on the processing of differences in intensity, time, and the frequency correlation between the two ears within the central nervous system. As mentioned earlier, because two ears receive input from the same sound source, the sound must travel farther to reach one of them, thereby creating a difference in the phase and level of the signal between the ears.

A common way of measuring binaural processing is by examining the masking level difference (MLD). Masking level difference is a technique of comparing threshold responses for the case when masking noise presented in phase with the test signal and the case where the masker noise and the signal are out of phase; release from masking is indicates an intact brainstem auditory pathway (Strouse et al. 1998).

2.1.3 Inter-aural Time Differences

The inter-aural time difference, or ITD, is the difference in time between a sound reaching one ear and then the other. ITD cues are mainly useful at lower frequencies, since the ITD becomes ambiguous as soon as half the wavelength of the sound equals the distance between the eardrums. ITD locations form a hyperbolic surface of rotation symmetrical about the interaural axis, and for distances that are more than a meter from the head, these hyperbolic surfaces form cones which are centred on the interaural axis. These cones are known as “cones of confusion” (Cunningham et al., 2000).

Those sounds that are located directly in front, or behind, us will reach both ears simultaneously, whereas ITDs can go up to about 700 µs for sounds originated at 90º azimuth (Kuhn, 1977). For low frequency stimuli, normal hearing listeners are sensitive to ITDs with Just Noticeable Differences (JNDs) as small as 10 μs (Mills, 1958).

Furthermore, phase locking of the auditory nerve fibres is functional up to 4-5 kHz (Moore, 2003). At high frequencies, ITD cues may still provide some localisation cues for some complex sounds, because the binaural system is able to localise high frequencies by using the ITDs extracted from the onset, offset and envelope of the signal (Middlebrooks and Green, 1991; Bernstein and Trahiotis, 2002). According to Hilbert's definition, sound can be decomposed into two components – the temporal fine structure, which can be conveyed by frequency modulations, and the envelope of the signal, which can be conveyed by amplitude modulations (Stickney et al., 2004). However, the fine structure cues are more dominant for localisation when both the fine structure and the envelope cues are available (Bernstein and Trahiotis, 2002).

2.1.4 Inter-aural Level Differences

The inter-aural level difference (ILD), which is the other cue of the binaural horizontal sound localisation, is the difference in amplitude at both ears. This can be due to the head forming a barrier to sounds coming from either side or due to pinnae effects that play a major role in high frequencies. Since the lower frequencies with longer wavelengths diffract more easily around objects, ILDs are higher for higher frequencies that have shorter wavelengths (Fitzpatrick, Kuwada & Batra, 2000). Thus, ILDs are small at low frequencies and can especially be used for the localisation of nearby sounds (Brungart, 1999). For sound sources that are more than one metre away from the listener, ILDs are considered useful cues for localisation at frequencies higher than about 1500 Hz (Moore, 1995). ILDs can be up to 35 dB at high frequencies, with the maximum occurring when sound sources lie directly to the right or left (at 90° and 270° azimuth) (Feddersen et al., 1957).

2.1.5 Spectral Cues

Monaural localisation, which is based on the quality of sound as it enters the ear canal, refers to the filtering effects of external structures (Fuzessery et al., 1990; Butler & Green, 1991). In evolved auditory systems, these external filters include the head, shoulders, torso, and outer ear or pinnae. Sounds are frequency filtered and they depend on the angle at which they strike the various external filters. The most significant filtering cue for sound localisation is the pinnae notch, a notch filtering effect resulting from where the sound hits the outer ear. The convoluted structure of the pinnae is such that sound waves, as they travel toward the ear canal, experience overlapping, cancellation, reverberation and reinforcement influences (Middlebrooks & Green, 1991). Since the cavities present in the pinnae are relatively small, pinnae effects are only present for frequencies in the range of 6 kHz and higher (Hartmann, 1999). The difference between the frequency response at the tympanic membrane, as compared to that at the centre of the head is referred to as the Head Related Transfer Function (HRTF).

Spectral cues, which are largely responsible for vertical localisation and the discrimination of elevation, play a major role in resolving horizontal source locations within the "cone of confusion", where ITD and ILD information remain constant (Hebrank and Wright, 1974; Middlebrooks et al., 1989; Langendijk and Bronkhorst, 2002). If spectral cues are omitted, several locations will have similar ITD and ILD cues. For two sound sources, if the distance which the sound has to travel to reach the far ear is kept constant, the level and time difference will be the same. This condition happens for sound locations that lie on a surface of a cone and known as a "cone of confusion" (Mills, 1972 and Wallach 1939). Nevertheless, due to the asymmetry of the head, ILD and ITD are not exactly constant along the cone of confusion area. However, acoustical differences between front and back sound sources are little, therefore the majority of cone of confusion errors are "front-back" errors (Carlile, 1994).

In addition to stationary localisation cues, dynamic cues due to head movements as well as visual cues play an important role in sound localisation. Wallach (1940) showed that head movements could aid localisation in both the horizontal and vertical planes. Lacker (1973) found that visual information prevented localisation errors for various head postures.

In our study, the main purpose was to investigate the effect of amplification and compression on ILD and ITD cues only. Therefore, movements were minimised by asking the listeners not to move their heads while the sound was presented. Practically, using a headrest or a skull clamp can physically restrict head movements; however this can cause discomfort, especially for our old hearing impaired listeners. Therefore we decided not to use them and compromised by asking the listeners not to move their heads, and to monitor them via a close circuit television (CCTV).

2.1.6 Minimum audible angle (MAA)

Sound localisation can be assessed by using the minimum audible angle (MAA), first described by Mills (1958). The MAA, which is the smallest change in the position of a sound source that can be reliably discriminated, for normally hearing listeners lies between 1° and 3° in the horizontal plane at 0° azimuth (Wightman and Kistler, 1993). The MAA requires subjects to discriminate between two sounds, rather than by localising the positions of the sound sources. Consequently, the MAA provides only limited information of the listener's localisation abilities (Wightman and Kistler, 1993).

2.1.7 Subject variables affecting localisation abilities

The ability to localise a sound source is influenced by individual variations known as subject variables, such as hearing threshold levels and age.

2.1.7.1 Age and sound localisation

Localisation abilities tend to deteriorate with aging (Granford et al., 1993) particularly the sensitivity to the ITD cues, which are reduced with aging (Strouse et al., 1998). However, ILD cues do not seem to be greatly affected by the aging process (Babkoff et al., 2002). This can be explained by the changes in the peripheral, or central auditory processing that occurs during ageing.

Abel et al (2000), who compared the horizontal localisation performance for individuals aged between 10 and 81 years, found that localisation errors tend to increase by 12-15% from the mid-thirties onwards. Most of the errors made were front-back localisation errors, suggesting that the cues were picked up by the primary organs; however, they may not be utilised by the auditory processing system. In other words, the decrease in the processing of auditory information might be caused by aging and not, by definition, a hearing impairment.

Previous studies have consistently found that older adults have a higher threshold for ITDs than young adults (Herman et al., 1977). Herman et al. (1977) suggested that the high frequency hearing loss in the elderly might be accountable for the age related decrease ability in elderly listeners to lateralize sound sources by using ITD cues.

Babkoff et al. (2002) studied lateralization of stimuli in younger and older individuals aged between 21 and 88, all of whom had normal hearing for their age. They noted that earlier studies had indicated that younger subjects required half the ITD than older subjects, but that older subjects were approximately equal in sensitivity to lateralizing ILD clicks. Their findings were consistent with their earlier findings. Using click train lateralization functions, they found that discrimination between midline-located click trains (ITD=0) from ITD-lateralized click trains became progressively worse with age, but they did not see the same with ILD-lateralized click trains. They therefore hypothesized that there was a difference between ITD and ILD processing because of a reduced ability in time cue discrimination, and that intensity and time cues are processed by two different auditory nervous system mechanisms. Therefore, when the stimulus location is based on temporal cues, aging affects the ability to perceive stimulus location changes (Babkoff et al., 2002).

2.1.7.2 Hearing loss and sound localisation

Auditory localisation functions are affected by hearing impairment. Since hearing losses are usually not flat across all frequencies, the loss of information varies depending on the frequency region. Therefore, ILD cues and spectral cues may be reduced with high frequency hearing loss, hence listeners need to rely more on ITD cues. Investigating the localisation performance of hearing impaired listeners is more challenging than it is for normal hearing listeners, which is due to the large individual differences between hearing impaired listeners, in that they include wide differences in age, type, degree, etiology, and duration of hearing loss.

Byrne and Noble (1994) conducted a study that measured auditory localisation functions between normally hearing listeners and hearing impaired listeners. Their overall research showed that hearing impairment affects sound localisation ability, and that the localisation ability is controlled by the degree of hearing loss, the signal level and frequency regions. While normally hearing listeners performed highly accurately in horizontal plane localisation, and fairly accurately in vertical plane localisation, the hearing impaired listeners had problems with localisation. They also found that hearing loss only moderately predicts localisation decrement, which suggests that some aspects of hearing impairment – e.g. distortions – in addition to attenuation, may affect the function of localisation.

Their study also investigated the effect of the type of hearing loss on the localisation abilities of hearing impaired listeners by comparing the localisation performance of a group with conductive/mixed hearing loss and a group with sensorineural hearing loss, both groups being matched for the degree of hearing loss. The results showed that conductive hearing loss has a significant effect on the horizontal plane localisation, which may be due to a higher proportion of the signal being transmitted through bone conduction when conductive loss is present, hence a reduction in inter-aural time and level difference. (Byrne et al., 1996)

Moreover, they found a relationship between the localisation abilities and the configuration of the hearing loss for listeners with sensorineural hearing loss, whereby the results showed that high frequency hearing loss has an adverse effect on median plane localisation, which may be due to the loss of spectral cues with high frequency hearing loss. However, the frontal horizontal localisation was accurate when better hearing in the low and mid frequencies was preserved. Also, front/back discrimination was improved when better hearing was available in the region of 4-6 kHz.

Several studies investigated the sensitivity of cochlear implants users to binaural cues. Direct measurements of the relative contribution of ITD and ILD cues for cochlear implant (CI) listeners were performed by Seeber and Fastl (2008). The subjects – two individuals with cochlear implants (CI) who had excellent localisation ability– underwent three different experiments that used wide band noise, low-pass pink noise and high-pass pink noise. Unlike normal hearing listeners, the results showed that the CI subjects used ILDs for localisation while ITDs played a minor role only. Their hypothesis had been that, if a single localisation cue is present and provides enough information, accurate localisation may occur. However, in more complex listening environments, the localisation ability of CI listeners might suffer, due to the unreliability of the cue. (Seeber & Fastl, 2008).

Many studies have tried to discover if there is a correlation between the audiogram and the localisation abilities. An extensive review of the literature by Durlach et al. (1981) concluded that sound localisation is degraded by the presence of unilateral hearing loss and bilateral asymmetry, and is more degraded by middle ear disorders and auditory nerve lesions than by cochlear disorders. Moreover, sound localisation is not easily predicted on the basis of audiograms. More recently, Gabriel et al. (1992) reported that, even with the same etiologies and audiograms, listeners can have different binaural hearing abilities. However, Noble et al. (1990) found a strong correlation (0.87) between the 4000 Hz threshold and the unaided localisation performance across all three groups of subjects after partialling out the age factor.

2.1.8 Non-subject variables affecting localisation abilities

Localisation abilities can be influenced by factors that are non-subject related variables and must be controlled during localisation experiments. Among these variables are environmental conditions, the precedence effect and sound variations.

2.1.8.1 Localisation and environmental conditions

The localisation accuracy of an individual is affected by the room acoustics, particularly noise and reverberation. Most of our listening environments include some reverberation. Such rooms can be evaluated by the “reverberation time”, which refers to the amount of time taken for a sound to be attenuated by 60 dB from its original value (Speaks, 1999). In small reverberant rooms, the reflected sound will add to the loudness of the original wave, since there will be little difference in the arrival time of both.

Giguere and Abel (1993) examined the ability of normal hearing listeners to localise a third octave bandwidth noise in both reverberant and absorbent conditions. They found that, for normal hearing listeners, localisation scores were consistently lower in reverberant rooms than in absorbent acoustic chambers. Their subjects were men and women aged between 22 and 46 years, all with normal hearing. The reverberant room was the size of a small office (LxWxH=3.45x2.74x2.29m), and it allowed for variation in reverberation time through the use of floor carpet and sound absorbent wall and ceiling panels. Two room with absorbent and reverberant configurations were used with reverberation times spanning 0.15 to 1.0 s. and the subjects centered in the chamber. They found that increasing the reverberation time resulted in a decrease of localisation accuracy at all frequencies (500, 1000, 2000 and 4000 Hz). They also found that reverberation time did not interact with the other variables, which suggested that the adverse effect of reverberation time was independent of the rise/decay time, speaker array and frequency. Their study suggested that room reflections disrupt the precedence effect, even when the noise stimuli had relatively brief onsets. (Giguere & Abel, 1993).

These findings were partially consistent with Hartman (1983), who investigated the localisation abilities of a continuous broadband noise in both reverberant and absorbent concert halls. The findings revealed significant deterioration of the localisation abilities in the reverberant concert hall. However, Hartman (1983) found that listeners can still manage to accurately localise various types of sound in reverberant environments due to the precedence effect.

2.1.8.2 The precedence effect

The precedence effect is a process by which humans can localise a sound source in an enclosed environment, even if the direct and reflected waves are giving conflicting directional cues (Giguere & Abel, 1993). When a sound is reproduced in a reverberant environment, it propagates in multiple directions and is subsequently reflected from nearby surfaces. The auditory system is thus faced with resolving competition between the first sound and its reflection for perception and localisation, this is phenomenon is known as the precedence effect (Litovsky & Yost, 1998). If a listener receives the same signal from multiple speakers, he will place it at the closest speaker and not in between, unless the time difference between the signals is less than about a millisecond (Litovski, 1999). An extensive review of the literature by Litovski (1999) showed that if the time difference between the signals is less than a millisecond, then the sounds from the lead source (i.e. sounds that arrive first at the eardrum) and the lag source (i.e. sounds that arrive second at the eardrum) are perceptually fused, and the lead and lag both contribute to perceiving the direction of the fused image. Several different phenomena occur as the delays increase; for relatively short delays, between one and five milliseconds (for short bursts), the sounds remain fused, but, as the delays increase, the fused sound separate into two separates sounds, with the lagging source and the lead source providing a separate auditory event. This phenomenon is called "the echo threshold". It is the first acoustic information that arrives at a listener that determines the location of a sound, and the later ones are suppressed in the interpretation process. This effect is called "the law of the first wave front" (Cremer, 1948). Localisation dominance occurs when the perceived location of a fused sound is dominated by the directional information in the lead (Livotski &

Shinn-Cunningham, 2001). The observation that, at short delays, the parameters of the lag stimulus are less able to be discriminated due to lead stimulus presence is known as "lag discrimination suppression." (Litovski, 1999). Sound fusion occurs when there are short delays (< 5 ms), as the two sounds will fuse in the listener's ears, and this breaks apart when the delays becomes longer (Livotsky & Shinn-Cunningham, 2001). When there are long delays, between 8-10 milliseconds (for short bursts), two sounds are heard 100% of the time (Livtosky & Shinn-Cunningham, 2001).

However, the precedent effect is not just based upon onsets, but, rather, persists even with the removal of the onset advantage (Dizon and Colburn, 2006). Dizon & Colburn tested this by utilizing 500 ms noise stimuli for the lead and the lag. The onset and offset time-of arrival differences in the composite stimuli were eliminated by applying a diotic time window to the composite lead and lag stimulus (Dizon and Colburn, 2006). Results indicated that the precedence continues into the ongoing part of long-duration stimuli even with the removal of the initial onsets. The researchers also showed that the precedence effect would apply to the ongoing portion of a lead-lag stimulus. This, in turn, shows that the processes that are behind the precedence effect can extract, from an ongoing stimulus, the lead and lag relationship and that this lead-lag relationship is temporally more complex than the transient-composed lead-lag stimulus (Dizon and Colburn, 2006).

2.1.9 Localisation and sound variations

Localisation ability is greatly influenced by the acoustical features of the sound being localised. A pure tone is a periodic sinusoidal wave and can be the fundamental component of more complex waves. Complex sounds are made up of sinusoids that differ in amplitude, phase and frequency (Speaks, 1999). Other sound variations that affect localisation abilities are broadband verses narrowband sounds.

2.1.9.1 Pure tones and sound localisation

Pure tones are harder to localise when compared to complex sounds (Casseday, 1973). Due to the periodic nature of the pure tones, phase leads and lags are indistinguishable which leads to ambiguity in the inter-aural phase cues. This ambiguity becomes greater as the frequency of the pure tones increases (Wightman and Kistler, 1997). At low frequencies, however, the confusion is reduced since the wavelength of the sound is larger than the head.

The localisation of pure tones can be improved by using rapid onsets, which help to reduce the phase ambiguity by introducing inter-aural differences in time of arrival of the envelope; furthermore, this allows use of the precedence effect (Rakerd and Hartmann, 1985). In a series of experiments, they (1986) varied the onset duration of pure tones in order to observe the effects on horizontal localisation accuracy of normal hearing listeners. They asked the listeners to locate a sound source from 12 sources that were arranged in the frontal horizontal plane by using a forced choice method.

In Experiment 1 they studied the effects of changing the onset duration of a 500 Hz pure tone and found that errors increase with onset duration. In Experiment 2, they changed the direction and delay by having the acoustical reflection come via a vertical reflection from the ceiling and found that the longest onset duration errors were similar to those in Experiment 1, and the shorter duration errors were fewer than in Experiment 1. Experiment 3 dealt with longer delays, where the reflection direction was the same as the standard, but the reflection delay was greatly increased, resulting in a large change in the effect of the onset of the tone. In Experiment 4, they changed the tone to a higher frequency, increasing it to 2000 Hz from 500 Hz, which they used in Experiments 1-3, but the room was configured in the same way as in Experiment 1. Here, they found that the listeners could localise these tones when they had brief onsets.

In Experiment 5, they used the same room configuration as in Experiment 4, but changed the tone to 500 Hz and reduced the peak level from 65 dB (A), which they had used previously, to 40 dB (A) since changing the peak amplitude would allow changes in the onset rate – i.e. an increase in level per unit time – which allowed for the differentiation of the effects of onset duration from onset rate. Here, they found that the peak level of a stimulus strongly affected the perceptual importance of an onset cue, showing that precedence effect depends upon onset rate. In Experiment 6, they used an empty room with no reflecting surfaces; the stimulus was a 500 Hz, 65 dBA sine tone with onset duration of 0, 5, 50, 500 and 1000 ms and found that 500 Hz tones with abrupt onsets were localised more accurately than tones with slow onsets (Rakerd & Hartmann, 1986).

The conclusions of these experiments were: (i) sound localisation was poor because of misdirection without the precedence effect; (ii) when a signal was abrupt, the precedence effect was maximally effective, (iii) the precedence effect became less accurate as the onset duration was increased, (iv) the accuracy of the precedence effect was dependent on the reflection delay time of the room; in rooms with brief reflection delay times the precedence effect was less accurate at less than 5 ms, whereas in rooms with long reflection delay times, the decrease in accuracy of the precedence effect did not begin until the onset duration reached 500 ms; (v) when the onset duration was increased beyond the point where the precedence effect began to deteriorate, localisation errors became progressively larger, until they became asymptotically large; (vi) lateral reflections appeared to be more damaging than vertical reflections to azimuthal localisation; (vii) gradual onsets led to increasing misdirection by steady-state sound field cues; (viii) the frequency of the signal affected the pattern of misdirection, however, misdirection was not affected by the signal level unless there was a change in onset rate; (ix) high frequency tones were mostly localised because of the presence of IID in the stimulus onset; (x) large individual differences were noted in the measurement for the localisation of tone pulses; and, finally, (xi) abrupt offsets yielded to measurable improvements in the localisation of long duration tones.

As mentioned earlier, the testing was carried out using pure tones; therefore, the above conclusions apply to pure tone stimuli only.

2.1.9.2 Broadband and narrowband sounds

Broadband sounds are easier to localise than narrowband sounds (Middlebrooks and Green, 1991). Due to the increased signal bandwidth, more binaural and spectral cues are available; moreover, the listeners are able to compare the ILD cues across different frequency bands, which can be a good indicator for the sound source location. Consequently, the localisation accuracy improves as bandwidth increases (Su and Recanzone, 2001; Wightman and Kistler, 1993).

Middlebrooks (1992) conducted a study, the first objective of which was to identify how inter-aural difference cues and spectral shape cues contribute to sound localisation in the vertical and horizontal dimensions. Subjects localised narrow-band sounds, varying in centre frequency and location. The second objective was to formalize a localisation model with a combination of inter-aural difference cues and spectral shape cues. The researchers in Middlebrooks (1992) found that the subjects could point to brief broadband sounds with considerable accuracy, but they made systematic vertical and front/back errors when localising narrow-band sounds; however, their horizontal localisation was accurate. This can be explained by the limited availability of the ILD and ITD cues in the narrow band noises, which provide fewer localisation cues. Furthermore, due to the limited frequency range, spectral cues are not useful in localising narrowband noises. Another factor is that they derived azimuth cues from a stimulus that is not vulnerable to narrow-band filtering and its vertical and front/back dimensions constant, which are characteristics of ILD and inter-aural delay.

2.1.10 Summary: what is known and unknown about sound localisation in the horizontal plane

To summarize, localisation is affected by several factors. The first is aging, in that localisation abilities tend to worsen with aging (Granford et al., 1993). This can be explained by the changes in peripheral or central auditory processing that occur during the ageing process. The sensitivity to the ITD cues reduces with aging (Strouse et al., 1998). However, ILD cues do not seem to be greatly affected by the aging process (Babkoff et al., 2002).

The second factor affecting localisation is hearing impairment. In their 1994 study, Byrne and Noble concluded that hearing impairment degrades sound localisation ability. However, a comparison of all localisation studies for hearing impaired listeners can be very challenging, since age, type, degree and etiology of hearing loss can be conflicting. Thus, if these factors are not separated, the results of experiments can be confounded.

An individual's localisation accuracy is also affected by the room acoustics, particularly one that contains noise and reverberation. Moreover, pure tone stimuli are harder to localise than are complex sounds (Casseday, 1973). Due to the periodic nature of pure tones, phase leads and lags are indistinguishable, leading to ambiguity in the inter-aural phase cues, and this increases with the increased frequency of the pure tones (Wightman and Kistler, 1997). At low frequencies, the confusion is reduced, since the wavelength of the sound is larger than the head. However, the localisation of pure tones can be improved by using rapid onsets, which help to reduce the phase ambiguity by introducing inter-aural differences in time of arrival of the envelope and spectral splatter, thereby allowing the use of the precedence effect (Rakerd and Hartmann, 1985).

Broadband sounds are easier to localise than narrowband sounds (Middlebrooks and Green, 1991), since, due to the increased signal bandwidth, more binaural and spectral cues are available; moreover, listeners can compare the ILD cues across different frequency bands, which can be a good indicator for the sound source location. Therefore, the localisation accuracy improves as bandwidth increases (Whightman and Kistler, 1993; Su and Recanzone, 2001).

Also, some phenomena are less understood. For instance, Byrne and Noble (1994) found that hearing loss only moderately predicts localisation decrement, which suggests that some aspects of hearing impairment, such as distortions, may affect localisation. Moreover, in reverberant environments, it is not clear yet if reverberation has an additional effect, on top of hearing aids and hearing impairment, by affecting the ITD or ILD cues, or both.

2.2 Reverberation

2.2.1 Introduction

Reverberation is defined as the combined effect of multiple sound reflections within a room. The reverberation characteristics of a room are affected by several factors: (a) the shape and size of the room, (b) the materials of which the room is constructed, and (c) the materials present, which are especially important since they determine how much sound is absorbed and how much is reflected. When sound bounces off surfaces, it creates echoes of the original signal, and these may be slightly different from the original signal in terms of frequency and temporal cues. When music is played in a concert hall, large numbers of reflections from various surfaces in the room strike the ear, producing a sensation of space to the listener (Moorer, 1979). Room reverberation introduces acoustical interference, characterised as spectral distortion as well as additional noise. Room reflections can be divided into early reverberation, which is highly correlated with the direct signal, and late reverberation which is less correlated (Palamaki et. al., 2002).

Studies of directional localisation in rooms generally show that the effect of reverberation on localisation accuracy is not very large, at least when onset information is available to the listener (Durlach et al., 2003). Early reflections – 50 to 80 ms of the direct sound – (Cunningham et. al., 2005a) are more easily localised than late ones, as our ears are sensitive to the arrival times and decreasing amplitudes of early reflections. On the other hand, late reflections tend to fuse together and create a quality of sound diffusion and our brains can no longer distinguish the arrival times or intensities of individual reflections. However, both early and late reflections can degrade sound localisation accuracy by reducing the interaural decorrelation (Hartmann, 1983; Cunningham, 2000)

Reverberation physically distorts directional cues and there is some evidence that reverberation interferes with directional perception (Kidd, Mason, & Arbogast, 2005). Reflected energy causes fluctuations in the short term values of localisation cues from moment to moment and hence distort localisation (Hartmann, 1983;

Giguere and Abel, 1993 and Devore et al., 2009). These effects on localisation cues increase as the ratio of direct to reverberant energy decrease (Cunningham et al., 2005b). The fluctuations in ITD cues caused by reverberation increase ITD variability and hence reduce the interaural coherence and decrease the reliability of ITD cues (Rakerd and Hartmann, 2010). Moreover, reverberant energy decreases ILD cues and this effect depends on the listener location in the room. With increasing source distance, reverberant energy increases the energy in the far ear – ear away from the sound source – thus decreasing the ILD (Cunningham et. al, 2005a). Since reverberation affects ITD cues and ILD cues in a different way, listeners perceptually weight low and high-frequency information differently depending on the listening environment (Ihlefeld and Cunningham, 2011).

Reverberation can provide listeners with a cue for sound source distance, since judgments of distance are thought to depend on the ratio of direct to reverberant sound energy, in that the level of the direct sound is inversely proportional to the square of the source distance, while the level of reverberation is roughly independent of source location. This 'direct-to-reverberant' energy ratio thus provides a distance cue that is independent of overall source level. Since the level of reverberation varies from room to room, the way in which listeners interpret the total signals reaching their ears must change with listening environments (Cunningham, 2000).

2.2.2 Reverberation: an overview and historical information

When we hear sounds that come from a hallway, or from the outside, far away from us, we hear reflections as well. We notice these reflections when the time delay gets longer than about the 30- to 50-ms echo threshold. Anechoic chambers are built to absorb sound energy, so that only the direct energy from the sound source reaches the ears.

In addition to natural reverberation, software simulation of reverberation may be created by using different types of instruments, which include many audio cards, synthesizers and digital audio applications in order to simulate both the natural and the artificial environment. The synthesis of reverberation by a digital signal processing algorithm is achieved by resembling the manner in which a real acoustic space works; therefore, algorithm designers simulate the early reflections, the compounding of echoes, decay time and high frequency diffusion.

Normal hearing listeners may not be affected by room reverberation significantly; however, hearing-impaired listeners can be affected by it. Poissant et al. (2006), who studied the effects of reverberation on speech intelligibility in cochlear implant situations, stated that reverberant environments are likely to contribute significantly to the difficulties experienced by cochlear implant users in their everyday lives. Their study found that reverberation and noise combination was extremely detrimental to speech perception, as subjects only understood the key words in 12% of the speech spectrum noise and 15% of the two-talker babble. They also found that, when reverberation is in a quiet place, it causes smearing of the temporal envelope of the speech signal, resulting in corrupt primary cues upon which cochlear implant users are reliant for speech understanding. The researchers, who used vocoder systems at 6, 12, and 24 channels, found that at 12 and 24 channels the processing was largely reverberation resistant, but the 6 channel system showed substantial speech degradation with increased reverberation, which shows that all reverberation, even small amounts, is detrimental to understanding speech; they also found poor speech recognition when reverberation and masking were combined.

Under the same testing conditions, and using the same stimuli, Helfer and Freyman (2005) found that a normal listener would not experience trouble, since 85% of their normal hearing listeners correctly identified their key words, even in a condition that is 8 dB lower than those used in the Poissant study. Under those circumstances, the Poissant listeners recognized only 12% of the key words, when heard through the 6 channel system. They found that even when reverberation created times were not extreme, there was a surprisingly large effect of reverberation.

2.2.3 Simulated reverberation

Reverberation may be either natural, such as the reverberation that is found in a concert hall, or simulated, which is where researchers simulate the reverberation that may be found in a natural setting. It has traditionally been difficult to simulate the reverberation found in natural environments. Some simulated reverberant conditions – where a single channel reverberation processor is used – produce a diotic sound, which means that the sound at each ear is identical. Natural reverberation, particularly its late part, is instead diffuse, which is the result of multiple sounds bouncing off of irregular walls, such as can be found in concert halls. Moorer (1979) found that diffusion is one of the reasons that simulating reverberation does not sound like real reverberation, and found that the effects of diffusion are most prominent when the walls are irregular, as opposed to flat, which is the case with most concert halls. For instance, Moorer, who cited the doublet response of the old New York Philharmonic Hall before reconstruction, noticed some distinct echoes followed by a great confusion of sounds, this confusion was caused by the multiplicity of the diffused sound sources reflecting from every irregularity in the room. Everything but the first few images were washed out due to rough surfaces in the room, which means that the geometric simulations of concert hall acoustics result in a simulated reverberant room that does not sound like a real room (Moorer, 1979).

However, Kopco and Shinn-Cunningham (2002) suggested that this diffusion depends upon where the listener is in the reverberant room. For listeners in the centre of the room, the reflective surfaces are located relatively far from the listener, which makes the sounds from all positions in the room diffuse to the listener. On the other hand, when the listener is close to a wall, early reflections are prominent, and these early reflections' magnitude and timing depends upon where the source is in relation to the listener (Kopco & Shinn-Cunningham, 2002). They also found that the reverberation effect on localisation varies dramatically depending upon where the listener is positioned in the room (Kopco & Shinn-Cunningham, 2002).

While there are certainly differences between the artificial and natural reverberant environments, and their effects upon acoustics, there are also similarities. For instance, there is some indication that the effects of 'steady-state suppression' (i.e. a pre-processing approach that suppresses steady-state portions of speech that have high energy in order to reduce overlap masking and hence improve speech intelligibility) are similar in both artificial and natural reverberant conditions. This was demonstrated by Hodoshima et al. (2005), who researched the effects of a dichotic listening condition conducted in a concert hall, and a diotic listening condition conducted using simulated reverberation. In the concert hall setting, each subject sat towards the back of the hall, with the stimuli being presented through two loudspeakers in the centre of the stage. In the simulated reverberant environment, the stimuli were the same as in the natural environment, however the stimuli were presented through headphones and the experiment was conducted in a soundproof room. The researchers found that the steady-state suppression significantly improved speech intelligibility in both environments (Hodoshima et al., 2005).

However, technical advances are closing the gap between the simulated and the real reverberant environments. For instance, Seeber et al. (2010) proposed a system to simulate and reproduce different audio-visual environments to facilitate the research on spatial hearing and listening in noise and reverberant environments. In their study, they introduced a simulated open-field environment, in which sounds are played over a wide range of frequencies and levels from multiple loudspeakers in an anechoic chamber. This enables the listeners to listen without amplification, and to allow the comparison between various listeners both with and without amplification. Room simulation software creates realistic reflection patterns and allows the simulation of different reverberant environments ranging from single echoic to more complex reverberant surfaces. Individual equalization of the loudspeaker's frequency response was achieved using custom calibration software. The sound playback system is paired by a video projection, which projects images on curtains covering the loudspeakers that may provide and receive feedback from the listeners, especially children (Seeber et al., 2010).

2.2.4 The effect of reverberation on distance estimation

Auditory distance estimation is primarily affected by sound loudness, sound spectrum, and temporal offset. All these cues require some knowledge of the original sound source and the acoustical characteristics of the environment. Their effect also depends on the expectations of the listener together with other sensory information. However, because of the complexity of conditions affecting auditory distance judgments, these judgments appear to be quite inaccurate, resulting in about 20% error or more (Moore, 1989). Furthermore, many people cannot translate perceived distance into numerical judgments. All these difficulties, which create real problems of reliability and validity of reported data, need to be considered (Scharine and Letowski, 2005).

According to a study by Mershon and King (1975), the most natural auditory distance estimation cue seems to be sound intensity, since, based on the inverse square law of sound propagation in open spaces, sound intensity decreases by 6 dB per doubling of the distance from the receiver. Therefore, by comparing the perceived intensity to the expected intensity of the original sound source at a specific distance, one cue for estimating the sound source distance in an open environment can be provided. This particular cue, however, requires some familiarity with the source of the sound. Also, the listener's movement toward, or away from, the sound source may provide another desirable cue (Ashmead et al, 1990).

In closed spaces, the decrease of sound intensity may initially follow a –6dB rule per doubling the distance, however, the decrease becomes smaller with an increase in distance, because of room reflections from nearby surfaces. This decrease continues as long as the energy of the direct sound exceeds that of the reflected sounds and a direct sound field becomes a reverberant field. The distance from a sound source, where both sound energies – direct and reflected – are equal, is called the 'critical distance', inside which sound localisation is basically not affected by sound reflections from space boundaries because of the precedence effect.

Another cue for distance estimation is the changes in the sound spectrum caused by the frequency-dependent absorption of sound energy by air. Sounds arriving at the listener from larger distances may appear as if they were low-pass filtered when compared to the original sounds. Humidity has a similar effect on absorption of high frequencies. If one has knowledge of the original sound source, the weather conditions and surrounding environment –e.g., walls– the spectral changes caused by air absorption provide useful information of distance estimation (Brungart and Scott, 2001). However, without the listener's familiarity with the sound source, changes in the sound spectrum provide some, but not all, the information required about the sound source distance (Little et al, 1992).

Reflected sound lasts longer than original sound, because, as the distance between the sound source and the listener increases, the amount of direct sound decreases and the amount of reflected energy increases (Mershon et al, 1989). The more reverberant the environment and the larger the distance between the sound source and the listener, the longer in time the sound is perceived by the listener. Consequently, reverberation is a very useful cue for distance estimation in both indoors and outdoors. However, the precise ratio of direct to reflected sound also depends on other factors, such as the direction of the sound source, the listener's hearing condition, the size of the space, and the position of the sound source relative to the listener and the surfaces (Mershon and King, 1975). In this thesis, therefore, an attempt has been made to investigate the effect of both hearing loss and reverberation on the listener's ability to localise different stimuli.

Zahorik et al. (2003) summarized this aspect of the research. They reportedthat listeners underestimate distances with regards to faraway sound sources and that acoustical and non-acoustical factors contributed to source distance perception. The acoustical factors included intensity, which decreases as the distance between a source of sound and a receiver is increased; direct-to-reverberant energy ratio, which refers to environments with sound reflecting surfaces, in which the ratio of energy reaching a listener directly to energy reaching the listener via reflecting surfaces is inversely related to the distance of the sound source; spectrum, which means that, for distances greater than 15 m, the sound absorbing properties of air significantly change the sound source spectrum; binaural cues, which play a large role in perception of sound sources near the head; and dynamic cues, which means that motion provide additional cues for auditory distance. Also, non-acoustic cues include vision and familiarity (Zahorik, 2005). Further, they found that the right temporal cortex plays a role in auditory distance perception, and that the right temporal cortex also plays a part in spatial tasks in different sensory modalities (Zahorik et al., 2003).

2.2.5 The effect of reverberation on sound localisation

Localisation ability varies to a large extent depending on the listening environment. Hartmann (1983) studied the localisation of sound sources in rooms to determine whether the ability to localise sound in a room depends on the room acoustics. Therefore, the acoustical characteristics of the listening environment were controlled by performing all the experiments in a variable-acoustics concert hall – the Escape de Projection (ESPRO) at the Institut de Recherche et Coordination Acoustique/Musique in Paris. The ESPRO, which was completely empty, had four bare walls and a bare parquet floor. Thirteen subjects were tested and results showed that the localisation of brief impulsive tones is unaffected if the reverberation time is reduced from 5 to 1 s by adding absorption. This can be explained by the precedence effect and the changes in the geometry of the room, both of which combine to reorder the sequence of reflections. Moreover, early reflections, which come from the same direction as the direct sound, reinforce the sense of localisation of the source.

In contrast to Hartmann's study, Giguere and Abel (1993) who conducted a study to assess the ability of listeners to localise one-third octave noise bands in the horizontal plane in both reverberant and absorbent conditions found that localisation was worse in the lateral array. They also found that localisation of one-third octave noise bands was adversely affected by room reverberation, although, the rise/decay time had only a small effect on performance. However, Studies of directional localisation in rooms generally show that the effect of reverberation on localisation accuracy of the normal hearing listeners is small, especially when the onset information is accessible (Hartmann, 1983).

Cunningham (2000) replicated the experimental set up of Brungart and Durlach (1999), who had studied three-dimensional localisation for sources within a meter of the listener. In Cunningham's study, subjects were seated in a reverberant room with the reverberation time set at 550 ms, a sound source at a random location was positioned within one metre from the subject, whose eyes were closed, and within his/her right hemifield. A broadband noise stimulus with a randomly set level was presented from the source. No feedback was provided.

Results showed that localisation performance improved with practice. Listeners learned to localise better in the room over time. Results from Cunningham's experiment were then compared by the author with results from Brungart's anechoic study (Brungart and Durlach, 1999). Results showed that even at the end of 5 sessions (5 hours) in the reverberant room, localisation errors were larger in the reverberant room than the anechoic room.

It can be concluded from the above studies, that localising a sound source is dependent on the environment and stimuli characteristics. Therefore, listeners may not discriminate sound in reverberant or noisy environments, since, in most cases, reverberation will have both positive and negative effects on sound localisation. The positive aspects are both better distance accuracy and a more realistic-sounding simulation. The drawbacks are that directional accuracy is worse; however, this can be overcome, at least in part, with experience, although the time required to train the subjects is large, since five hours of practice was not enough to bring reverberant performance up to par with an anechoic environment. Reverberant environments resemble daily listening environments and provide distance cues; however, directional accuracy can be impacted by reverberation.

2.2.6 Reverberation and hearing loss

The presence of noise can easily be noticed by listeners, whereas the presence of moderate reverberation can be unnoticeable. Small amount of reverberation can change the signal, but may not be identified by listeners, which result in worse speech intelligibility and leads to frustration and annoyance, since the listeners do not know exactly what is causing it (Nabelae and Robinson, 1982).

In a study by Irwin and McCauley (1987), eight normal hearing listeners and eight listeners with sensorineural hearing losses, were compared on both a gap-detection task and a speech perception task. The subjects had to detect a brief cessation in a continuous broadband noise that was band pass filtered with cutoff frequencies at 100 and 5000 Hz, to cover the frequency range included in normal speech. The minimum detectable gap (71% correct) was determined as a function of noise level, and a time constant was computed for each listener. The time constants of the hearing-impaired listeners were significantly longer than those of the normal listeners. The speech consisted of sentences mixed with two levels of noise and subjected to two kinds of reverberation (real or simulated). The speech thresholds – minimum signal-to-noise ratio for 50% correct – were significantly higher for the hearing-impaired listeners than for the normal listeners for both real and simulated reverberation. The study, which showed that the longer reverberation times produced significantly higher thresholds than the shorter times, helps in the understanding of the relationship between speech delay and hearing loss, but it was not extensive enough to detect all the possible correlations – i.e. the correlation between the type and degree of hearing loss and speech delay in reverberation.

Lutman and Payne (2002), who investigated the horizontal localisation abilities of experienced users of bilateral Phonak Claro hearing aids and compared their performances to normal hearing listeners, used the following stimuli: reverberant – total reverberation length of 402ms – and non-reverberant speech, pink noise bursts at 60 and 70 dB (A), 1 kHz pure tone bursts and a transient white noise stimulus. For the reverberant speech, reverberation was presented from four corner speakers, whereas the direct speech was presented from one of 11 speakers positioned in an arc. All testing was carried out in anechoic conditions and hearing aid users were tested with different directional microphone settings; with aids set to omni-directional (Omni-Omni), adaptive (Adapt-Adapt), and fixed (Fixed-Fixed) directional microphone settings. Moreover, the hearing aid users were tested with two asymmetric conditions, with one microphone set to the omni-directional setting and one to the adaptive directional setting (Omni L- Adapt R and Omni R –Adapt L).

Results revealed that the localisation abilities of the normal hearing listeners were more accurate in comparison to the hearing impaired listeners. One possible explanation is that their hearing is less symmetrical than that of normal hearing listeners. Another explanation is that the deficits in the auditory system of listeners with sensorineural hearing loss, i.e. reduced temporal, amplitude and frequency resolution, may have reduced their ability to use the inter-aural cues (Lutman and Payne, 2002). Also, the hearing aid itself may result in further reduced inter-aural difference and spectral cues.

When comparing performances within the hearing aid users, results revealed a significant decrease in localising the tone stimulus – 1 kHz pure tone burst – in the asymmetric microphone settings when compared to symmetric omni-directional settings. However, no significant difference was found between the fixed and adaptive directional settings.

For both normal hearing and hearing impaired listeners, the localising of stimuli presented from the midline speakers was significantly better when compared to the lateral speaker positions. These results were in agreement with the findings of Wightman and Kistler (1989).

Amongst all stimuli, the tone stimulus was significantly harder to localise by both the normal hearing and the hearing impaired listeners. This can be due to the gradual onset and offset characteristics of the stimulus. Abrupt onset stimuli – i.e. pink noise, speech and transient stimuli– have more sources of localisation information, counting spectral cues and inter-aural difference in the arrival of the envelope (Rakerd and Hartmann, 1985). Moreover, since the tone is a narrow band stimulus, which contains energy over a restricted range of frequencies, the comparison of ILD and ITD cues across different frequency regions is not possible; it therefore makes it more difficult to be localised (Wightman and Kistler, 1993).

Reverberant speech was found not to be harder to localise when compared to the non-reverberant speech for both normal hearing and hearing impaired listeners. However, only one reverberant condition was used in this experiment, which did not reflect the different reverberant conditions that listeners are exposed to in their daily lives. Consequently, the present study focused on understanding the effects of different reverberant environments on the ability of listeners to localise different stimuli.

2.3 The effect of amplification on sound localisation

2.3.1 Hearing aids: analogue vs. digital hearing aids

Sound localisation relies on balancing hearing in both ears. Good sound localisation is necessary for responding appropriately to alerting signals and for making use of binaural cues when listening to speech in noise.

Hearing aid users often report that their understanding of speech in noisy and reverberant environments is not improved by their hearing aids, which usually transmit a limited bandwidth of about 300-3000 Hz, whereas, a bandwidth of 60-8000 Hz has been recommended to be adequate (Ross, 2004). The microphone and amplification circuits affect the dynamic range of a hearing aid; the typical dynamic range of a hearing aid is about 55 dB, which is about half the dynamic range of a normal ear (Kates, 1998).

Amplification, filtering, and compression are operations that digital hearing aids perform, just like analogue ones. A microphone and receiver are also present in the digital hearing aid, along with other components, including the analogue to digital converter, a processing unit and the digital to analogue converter. In analogue hearing aids, the incoming signal has a continuously varying voltage, whereas digital hearing aids present signals as discrete levels that have been sampled from the waveform. These samples are converted from their analogue form into digital information and are represented by either a 0 or a 1, each of which is defined as a bit, with the number of bits representing the accuracy of the approximation.

Group delay is the amount of time it takes for a signal to spread through a signal processing system and is detectable at durations as short as 3-6 ms (Kates, 1998). Analogue systems analyze data continuously; therefore, they create less delay to the signal when compared with digital systems, which analyze data in discrete groups. Also, directional microphones in digital hearing aids are more likely to disrupt the binaural cues used for localisation because they can artificially change the inter-aural level and phase differences, both of which are critical cues used for sound localisation. Moreover, group delay of digital hearing aids may interfere with one of the most important cues for localisation – inter-aural time delay. More group delays are found in hearing aids that perform filtering in the frequency domain and fewer in those with time domain filtering (Stone et al, 2002).

On the other hand, hearing aids might also improve localisation by providing audibility to high frequency signals that is missing among individuals with high-frequency sensorineural hearing loss. A further purpose of this thesis is to study the influence of hearing aids and their impact on the ability to localise sound in both anechoic and reverberant environments.

2.3.2 Effect of Microphones on ITD and ILD cues

Microphones are transducers that convert sound energy into electric energy. There are different types of microphones based on their response to signals. Omnidirectional microphones respond equally to sounds from any direction in free field, whereas directional microphones are more sensitive to signals from certain directions. However, directional microphones may alter phase and intensity cues and hence affect localisation performance. Directional microphones work by introducing a small time delay to the signal caused by a phase difference between the two signals at the two microphone ports. This phase difference increases with increasing frequency. Furthermore, directional microphones are less sensitive for sounds that do not originate in front of the listener (Thompson, 2002). This reduces the audibility of signals coming from other directions, and hence increases the difficulty for the listener to detect localisation cues, especially ILD cues.

However, ITD cues maybe retained since ear mould/vent systems pass low-frequency cues without much attenuation (Byrne et al., 1998b).

In this thesis, in order to avoid complexity in analysing the results, hearing impaired listeners were tested with their hearing aids on Omni-directional microphone configuration

2.3.3 Effect of group delay on ITD and ILD cues

Digital hearing aids apply compression in separate frequency bands, which may be fitted in either the time or frequency domain. A system which implements compression in the frequency domain tends to introduce greater delay than the one that applies compression in the time domain (Stone et al., 2003). Fast Fourier Transform (FFT) method can be used in dividing the input signal into different frequency bands. The FFT method converts a waveform in the time domain into its frequency spectrum. However, as the number of channels is increased the frequency resolution required increases. This increases the delay, which can be up to 10 ms or more (Kates, 2005). Delay is detectable by listeners at durations as short as 3-6 ms (Stone et al., 2002), and an overall delays of 15-20 ms can be disturbing (Stone et al., 2002).

However, the delay can be lower if the signal is filtered in the time domain. This is usually performed by using filter banks to process the signal. There are two ways of designing filter banks, the first, by using infinite impulse response (IIR) filters and the second, by finite impulse response (FIR) filters. Filter banks using the IIR method achieve sharp attenuation between frequency bands. However, the group delay increases as the slope of the frequency response becomes steeper. In the FIR filter bank, the output of each filter is in phase with the output of every other filter in the filter bank, with the group delay equal at all frequencies. The group delay for a filter bank using the FIR method is higher than that using the IIR method (Kates, 2005).

Group delay may interfere with interaural time delay cues, which can be disrupted by the overall group delay of the signal and phase mismatches introduced by digital filters. Moreover, group delay can vary for different frequency components of the signal which may cause more disruption (Agnew and Thornton, 2000).

2.3.4 The effects of compression on ILDs and ITDs

One of the most important features in hearing aids is dynamic compression, which is used to restore deteriorated cochlear amplification where the deterioration is the result of outer hair cell loss. Dynamic compression may impact on binaural performance, such as directional hearing, since it alters the sound; this is especially important in the horizontal plane. Musa-Shufani (2006) found that compression does affect directional hearing, based on ILD differences, but it does not affect it based on ITD (Musa-Shufani, 2006).

According to Bodden (1994), conventional hearing aids may change localisation cues mostly for asymmetrical hearing losses and hearing aids with Automatic Gain Control (AGC). Hearing aids may change the ILD as a function of the absolute level of the signal so the combinations of inter-aural differences no longer fit the usual combinations the wearer is used to. This, he states, can be avoided if two hearing aids do not work independently but are controlled by a central processor, which, will then effectively control the amplification of both hearing aids (Bodden 1994).

Amplitude compression is used to protect the listener against uncomfortably loud sounds, and is also utilised for improving speech intelligibility for listeners whose dynamic hearing range is limited. Nabelek (1983) found that (i) multiband compression, with three separate bands, yielded no better results than no compression, (ii) that compression controlled by low-frequency speech components, or wideband amplitude compression (WBC), improves speech intelligibility for subjects with sensorineutral hearing impairment, especially when the compression ratio is between 5 and 10, (iii) Compression benefits depend on speech material, (iv) that WBC was not improved with words, but was improved with nonsense syllables, (v) that scores decreased when speech-shaped spectrum noise was mixed with speech, (vi) that speech recorded in reverberant condition caused the scores to decrease, with the compression decrease being the same as the linear amplification decrease, and (vii) that WBC alone, and with clipping, increased consonant scores in the initial position (Nabelek, 1983).

A study by Musa-Shufani et al. (2006) investigated the influence of dynamic compression on directional hearing in the frontal horizontal plane. Different compression types were used. Using five normal hearing listeners and seven hearing-impaired subjects, they conducted three different experiments. In Experiments I and II, measurements were performed with isolated ILDs and ITDs, respectively. In Experiment III they used HRTF stimuli, which consisted of a combination of both inter-aural cues. The influence of compression on inter-aural level differences (ILDs) and inter-aural time differences (ITDs) separately was examined in discrimination experiments. The combination of ILDs and ITDs was examined with measurements of localisation.

Results for Experiment I showed an increase of JNDs in ILD when the compression ratios were higher, explained by a reduction of ILD by dynamic compression. Experiment I also showed that attack time has an impact on directional hearing based on ILDs, in that increasing the attack time resulted in a decrease in the JNDs. The researchers assumed that the use of very long attack times would result in compression not affecting the discrimination of ILDs for both normal and hearing impaired listeners. This, the researchers explained, could be because, with longer attack times, their listeners had more time to analyze the ILDs before they are compressed. Experiment II showed that directional hearing based on ITDs was not influenced by dynamic compression. Attack time and compression ratio did not affect JNDs because compression did not affect timing of signals, just the levels of the signals.

Experiment II also showed that the JNDs were similar with all combinations of compression ratios and attack times, although they differed significantly between hearing-impaired listeners and normal hearing listeners, especially at 4000 Hz, where the hearing-impaired listeners showed 5 to 6 times higher values. Experiment III combined ITDs and ILDs, because HRTFs produce a sound source virtual direction, which could have resulted in the availability of spectral cues. However, the spectral cues on the frontal horizontal plane should be insignificant; also, they could have been crucial if multi-band compression was applied, which was not the case here. Localisation errors were mainly errors for lateral positions that were judged closer to the midline, which mirrors the deterioration of ILDs caused by compression.

In summary, the researchers demonstrated that the influence of compression ratio and attack time could be shown for ILD cues; moreover, the influence of compression decreased with prolonged attack time. The impairment of the discrimination of ITDs with the hearing impaired subjects in the high-frequency range suggests that they rely mainly on ILD cues.

Kollmeier et al. (1993) studied real-time multiband dynamic compression and noise reduction for binaural hearing aids. In their experiment, six subjects with sensorineural hearing loss were tested. They used three dummy-head recordings of classic acoustical condition – a sample of traffic noise, a loud doorbell presented in soft background noise and a sample out of string quartet by Schubert. All the listening samples were recorded with a stereophonic inserted ear-level microphone in real situations and were presented unprocessed, processed with linear frequency shaping alone, and with linear frequency shaping including compression. The sound samples were presented to the subjects via headphones. The listeners were asked to assess the subjective transmission quality within a scale of five categories ranging from bad to excellent. They found that there was no significant advantage or disadvantage of linear frequency shaping versus unprocessed speech. They also found the additional compression to be positive, which was due to the limitation of the uncomfortable acoustical components at high frequencies (Kollmeier et al., 1993).

They also evaluated noise and reverberation suppression by simulating an acoustic situation using dummy-head recordings in a reverberant room employing one target speaker and one interfering speaker. The signal to noise ratio was adjusted individually within a range of -5 dB to +2 dB. They found that the algorithm seems to work efficiently. However, the benefit obtainable for each listener from the processing strategies depended upon the hearing loss of the individual, the residual dynamic range in the high frequency region, and the signal-to-noise ratio of the test situation. They found that listeners with the smallest residual dynamic range at 4 kHz showed the least benefit from the suppression of lateral noise sources and reverberation, possibly due to the processing artifacts caused by suddenly switching on and off different frequency bands. Moreover, the algorithm seemed to operate well for high and intermediate signal to noise ratios. However, for low signal to noise ratios no benefit was obtained from the algorithm as compared to the unprocessed situation (Kollmeier et al., 1993).

2.3.5 The effect of ear moulds on localisation cues

Noble and Byrne (1990), using a normal hearing group as a control group, tested localisation performance in the frontal horizontal and vertical planes with bilateral behind the ear (BTE), in the ear (ITE), and in the canal (ITC) hearing aids fitted with omnidirectional microphone configurations. Analysis did not show significant differences between unaided and aided performance for any of the three groups. However, statistical analysis on only horizontal, or on only vertical, localisation errors was not presented in the study. They stated that, for the control group, horizontal localisation performance dropped from nearly 100% correct unaided to 73% correct when tested with BTE hearing aids. The hearing aid users did not show better hearing than unaided localisation performance in the frontal horizontal plane, except for the ITE hearing aid users when they were wearing their own hearing aids.

In a later study, Noble et al. (1998) and Byrne et al (1998), indicated that it is possible to have better performance by using open earmoulds instead of closed earmoulds for listeners who had either a moderate high-frequency and a severe low-frequency loss, or a moderate low-frequency and a severe high-frequency hearing loss. They stated that open earmoulds allow the listener to localise the sound source by using the direct sound field in the area of the moderate hearing loss. Their results showed some improvement in the vertical plane for listeners with a moderate high-frequency loss; however, for listeners with a moderate, low to mid frequency hearing loss, improvement in the horizontal plane was found in unaided performance.

Byrne and Noble (1998) also found that vertical localisation depends on the listener being able to hear sounds with high-frequency components and pinnae reflections. Closed earmoulds, having an obstructed pinnae, resulted in poor vertical localisation, regardless of hearing or signal level. The listeners with the best unaided localisation were affected the most by closed earmoulds, since they suffered the most degradation of localisation abilities; consequently, they found, these listeners benefited the most by switching to open or sleeve earmoulds. The researchers therefore found a correlation between the ability to benefit from open, or sleeve, earmould fitting and hearing at high frequencies.

They also found that, for some people, sleeve earmould-aided localisation was slightly worse than unaided localisation. They tested four people with sleeve earmoulds, and turned the aids off and on, and found that the 'off 'condition was better than the 'on ' condition, and was equal to the unaided condition, therefore they suggested that there was no degradation of localisation cues with the sleeve earmould, but amplification had a small effect. However, they also stated that listeners with good high-frequency hearing may benefit from open earmoulds, since they improve vertical localisation, and that sleeve earmoulds would be useful.

2.3.6 Sound localisation with bilateral vs. unilateral hearing aids

Improved localisation is an advantage often mentioned when bilateral fitting of hearing aids is discussed; this means that listeners with two hearing aids have superior localisation abilities.

The mechanisms of binaural hearing are based on an analysis of inter-aural differences and monaural cues. In order to support the binaural abilities of hearing-impaired listeners, it is necessary to provide them with the desired inter-aural and monaural cues. Bodden (1994) suggested that binaural hearing is more advantageous than monaural hearing because reverberation can be suppressed by using the precedence effect; moreover, the interfering sound sources can also be suppressed –the 'cocktail-party-effect'. However, present-day hearing aids enhance speech perception in quiet and non-reverberant environments. Moreover, Bodden (1994) suggested that they are not able to fully distinguish between desired and interfering signals.

Byrne et al. (1992) also measured the benefits of bilateral hearing aids. In their experiment, the aided localisation abilities of 87 hearing-impaired listeners were tested for horizontal and vertical sound localisation. Some listeners wore behind-the-ear (BTE) aids, others in-the-ear (ITE) aids; some were bilaterally fitted, others unilaterally fitted. Acclimatisation was taken into account; therefore, listeners were tested with the types and setting of hearing aids with which they were most familiar.

Results revealed the benefits of bilateral fitting for moderately and severely hearing-impaired listeners; however, for mildly impaired listeners, listeners with unilateral hearing aids performed as well as listeners wearing bilateral aids. These results were consistent with those of Vaughan-Jones et al. (1993) who found that in some of the tested listeners, localisation ability with two hearing aids was worse than with one hearing aid.

In the Byrne et al. (1992) study, the questions posed were: (i) Do two hearing aids assist localisation better than one? (ii) Does the answer to the first question depend on the degree of hearing loss, or on the signal presentation level? (iii) Does the hearing aid (BTE vs. ITE) affect localisation more effectively with the bi-lateral fitting or the unilateral fitting?

The study was performed with 87 hearing impaired listeners (44 women and 43 men), with a average age of 65 years 7 months. The participants had been using hearing aids from 3 months to 15 years, with only two having worn them for less than 6 months. The majority (53%) had worn them for over two years. All participants but three had symmetric hearing loss. The participants were divided into groups, based on type of hearing aid, type of fitting and severity of hearing impairment. The classifications, as far as hearing aid types, were: (i) Bilateral (Bil), (ii) Behind the ear (BTE), (iii) Unilateral (Uni), (iv) In the ear (ITE). All other hearing aids were Omnidirectional (Byrne et al., 1992).

The participants were presented with a signal from 20 loudspeakers in full view and were identified by number. The sound was a pink noise of 1/3 octave from 200 Hz to 12,500 Hz. The loudspeakers were arranged with two 1.22-m radius intersecting arcs – one spanning 180 degrees on the horizontal plane and the other spanning 160 degrees on the vertical plane. Two testing arrangements were used – one in which the listener faced the loudspeakers, where the two arcs intersected, and one where they faced the extreme loudspeaker of the horizontal arc, which was a sideways arrangement. Two levels of signals were used, (a) most comfortable level – MCL, and (b) ½ MCL.

The researchers found that bilateral fittings provided an overall advantage, and that the type of aid did not make a significant difference. Performance was better when the participants faced the array rather than when they were positioned sideways. Also, there were significant interactions between the categories – type of fitting (bilateral, right, left) interacted with the aid type (BTE, ITE), orientation (forward, sideways) and presentation level (MCL and ½ MCL). The type of aid (BTE, ITE) also interacted with type of fitting and orientation. All the results were affected by the degree of hearing loss. If the hearing loss for the BTE wearers, facing the loud-speakers was lower than 50 dB, bilateral fitting did not provide an advantage. However, the more severely impaired groups experienced a significant bilateral advantage in this condition, while the ITE wearers experienced little bilateral advantage which could be explained by the fact that the ITE participants, overall, had less hearing impairment than the BTE participants.

For the participants facing the sideways array, results were somewhat similar. There was no advantage for unilateral vs. Bilateral if the hearing impairment was less than 50 dB; infact, the lower than 50 dB group showed an advantage with unilateral fittings. However, across the board, the horizontal localisation results were worse for all participants in the sideways position than for those in the forward position. This could be explained by 'cone of confusion' errors in the sideways position – i.e. for a head that is stationary, the same inter-aural differences will be produced by sound sources located to one side of the listener's head, giving rise to location ambiguity. The researchers found that bilateral hearing aids showed a substantial advantage for the more severely hearing impaired listeners, except with the ITE group, when the presentation level was higher.

Byrne's study also compared bilateral and unilateral aided conditions with unaided conditions when the participants were listening to sounds loud enough to be heard unaided. For forward horizontal localisation, they found that for those whose hearing loss was under 50 dB, the performance was good for all conditions – bilateral, unilateral and unaided – although a small bias was noted towards the unaided being better than the aided, and there was no difference between the bilateral and unilateral conditions. The bilateral fitting had a slight disadvantage compared to the unaided for those whose hearing loss was over 50 dB, however the unilateral fitting had a substantial disadvantage compared to the unaided. For the sideways localisation, the over 50 dB groups showed that unilateral hearing aids had a disadvantage compared to unaided, whereas bilateral had a much less disadvantage over unaided. The same conclusions applied to the vertical forward results.

In all, a clear trend emerged for aided localisation deteriorating as hearing level increases, a trend that was more pronounced for unilateral than bilateral wearers. Moreover, for small hearing loss, unilateral and bilateral aiding came out about equally, but for larger losses, the bilateral wearer was at an advantage that increased as the hearing level increases; an advantage that first appeared at about 40 dB HTL for MCL signals, and at about 30 dB HTL for ½ MCL signals. The researchers also found that lesser impaired participants of the BTE group had experienced better performance levels when comparing the unilateral to the unaided conditions than when comparing the bilateral to the unaided conditions. However, when comparing the more impaired participants in the BTE groups, the opposite proved to be true.

For ITE, performance was better for higher hearing levels. Furthermore, when comparing the groups with left and right fittings, the more impaired participants performed better only when the usually aided ear (although, of course, not aided in the unaided trials) was away from the speakers and the signals were presented at a lower level. Most significantly, the researchers also found that the effect of the type of fitting was strongly dependent on hearing level. For moderate to severe hearing loss, bilateral fittings were better than unilateral in localisation, and, when comparing unaided to unilateral and bilateral, when the sound was loud enough to be heard unaided, unilateral fittings were at a significant disadvantage over unaided hearing, whereas bilateral fittings were not. Byrne's hypothesis was that if a listener is mildly impaired the unaided ear can process signals in order to localise sounds better, or as well as, if the ear is aided.

Another of Byrnes's hypotheses is that the ear that is closest to the signal will hear the signal louder than the other ear. Thus, when the signal is on the aided side, it is louder in the aided ear, and when the signal is on the unaided side, it is louder in the unaided ear, which is due to the signal in the aided ear being reduced by head shadow. This suggests that some individuals might consider using bilateral hearing aids, even if their hearing loss is mild to moderate, in some situations, and unilateral in others. If a person is more severely impaired, the signal is heard more loudly in the aided ear, no matter which ear is facing the source. Byrne also hypothesized that his study was different from others that found poor localisation in unilateral hearing aids, as his involved people who were experienced in using hearing aids, which suggests that, over time, they have learned to make better use of the signal in their unaided ear. He also hypothesized that bilateral hearing aids are more suitable for individuals who have more profound hearing for both horizontal and vertical performance – if the bilateral fittings improve localising horizontal sources, then vertical sources are less likely to be misread as horizontal sources, thereby diminishing the vertical error scores.

As shown above, Byrne et al's (1992) study produced different results than previous studies. Firstly, Byrne focused only on hearing aid wearers who were used to wearing them. This is an important difference, because optimization of a hearing aid is only achieved by the wearer experiencing that particular fitting. Secondly, he used vertical sourced sounds, and many of the participants performed poorly in vertical conditions. Thirdly, he added sounds that were unpredictable in level, which is important because the intensity cue provided in a real life situation was avoided. Fourthly, he permitted head movement when the signals were presented.

Vaughn-Jones et al. (1993) also studied the effects of monaural vs. binaural aids on 64 patients who were referred by their General Practitioner for the provision of hearing aids. The standard range of NHS hearing aids was used in all except for two patients, who were fitted with Pico Forte and Widex G2H aids. In their study they found better sound localisation with monaural aids, with 18% of the patients thought that unaided localisation was better than binaural-aided localisation. This result was unexpected, as it runs contrary to the assumption that sound localisation can be improved by binaural fitting since our natural auditory system is binaural. The researchers further found monaural aids were better for speech discrimination in noise, with 65% stating that they experienced an improvement. By contrast, 43% stated that speech discrimination in noise was worse with binaural aids. After making their final choice, 55% preferred initial monaural aiding, followed by the routine provision of a second aid, while 15% preferred initial binaural aiding, and 16% felt that binaural aids should only be provided on request.

The patients who preferred the binaural aids used hearing aids for eight hours more per day than those with monaural preferences, which suggests that the binaural preference group have a greater need for, or derive greater benefits from, hearing aids than the monaural preference group. Moreover, the severity of hearing loss did not influence the aiding side, or the choice of binaural aiding. The researchers also found that sequential monaural aiding, followed by the fitting of binaural aids, resulted in a 53% uptake of binaural aiding, compared to a 16% uptake where a binaural aid was tried initially, which suggests that hearing aid use requires a period of adjustment that should be done one aided ear at a time. It is important to note that Vaughn-Jones used an unbalanced design, whereby there was more exposure to unilateral conditions than to bilateral ones. For example: two groups have two unilateral fittings followed by a bilateral fit and one group has an initial bilateral fit followed, randomly, by two unilateral fittings; the final choice, therefore, might be a result of 'order-of-fit' effect (Noble, 2006).

In contrast, Köbler and Rosenhall (2002) found that localisation abilities were almost the same without any hearing aid and with bilateral hearing aids. The worst result was found for the condition with only one hearing aid. In their experiment, participants had to repeat sentences and indicate the side where they heard the sentence come from. Their results were contradictory to Byrne et al's findings which suggested that unilateral amplification with experienced users with mild to moderate hearing loss would localise at least as well as bilateral amplification.

In a study by Van den Bogaert et al. (2006), which involved ten hearing impaired subjects between 44 and 79 years old, it was found that, for hearing impaired individuals, horizontal localisation abilities were actually worse for the hearing impaired participants with bilateral hearing aids than for the hearing impaired participants without any hearing aids. All the participants were experienced bilateral hearing aid users. The subjects sat inside an array of 13 speakers, while the chair was elevated until their ear reached the level of the speakers. Hearing impaired participants with hearing aids were tested (a) without hearing aids, (b) with both hearing aids set to omni-directional microphone configurations, and (c) with both hearing aids set to adaptive directional microphone configurations

They were tested in a semi reverberant room measuring 6 m x 3 m x 3.5 m (length x width x height) and with a reverberation time of .54s. The target stimuli was a 200-ms 1/3-octave low-frequency noise band and a 200-ms 1/3-octave high-frequency noise band, centred at 3150 Hz together with a 1 s broadband telephone ringing signal. The telephone signal was tested in silence, and with a multitalker babble source located at the left and right side of the participant.

Pairwise comparisons in the Van den Bogaert et al. (2006) study showed that the participants without the hearing aid showed significantly better performance than the participants with the hearing aids, under both testing conditions – i.e. using the microphone with the omnidirectional configuration and using the microphone with the adaptive directional configuration. Moreover, there was no significant difference between the two hearing aid settings. On the low-frequency noise band, the participants without the hearing aids performed significantly better than the participants with the adaptive directional microphone hearing aids, by an average of 3.1 degrees; they also did better than the participants with the omnidirectional microphone configuration by 3.0 degrees, although no significant difference for the standard between these conditions was found. No significant differences were found on the high frequency noise between the pairwise comparisons, although large differences in the mean results between the different conditions were observed.

The 'silent' broadband telephone signal showed that the no-hearing-aid condition performed significantly better than the omnidirectional condition by an average of 3.1 degrees, and no significant difference was observed between the no-hearing-aid condition and the adaptive directional configuration condition, although the p value at 0.059 was close to being significant. Again, there was no significant difference between the adaptive directional configuration and the omnidirectional configuration conditions. Regarding the telephone ringing signal with babble, the results were different. The no-hearing-aid condition scored better than the omnidirectional configuration condition by 6.0 degrees and better than the adaptive configuration condition by 9.7 degrees. Finally, the omnidirectional configuration scored better than the adaptive configuration by 3.7 degrees.

McManus (2008) investigated the effect of amplification on the horizontal localisation by testing18 bilateral hearing aid users. The stimuli used in the experiment were: pink noise, low-pass pink noise filtered at 1 kHz and high-pass pink noise filtered at 4 kHz. Participants were tested with and without their hearing aids in anechoic conditions. Results displayed a significant increase in the localisation error for the high-pass pink noise in the aided conditions when compared with the unaided conditions. On the other hand, no significant effect of aiding was shown in localising the pink noise and the low-pass pink noise stimuli. However, the experiment was short of information on the performance of hearing aid listeners under reverberant environments. Therefore, the results from our study and the McManus (2008) study may be considered complementary in that they might lead to clarification of the effects of hearing aids on ILD and ITD cues in both anechoic and reverberant environments.

Other studies have measured the bilateral benefit of hearing aids subjectively. Boymans et al. (2009), who assessed the long term outcomes of bilateral or unilateral hearing aids after at least two years of use, sent out a questionnaire – the Amsterdam Questionnaire for Unilateral or Bilateral Fitting (AVETA) – of which 505 valid questionnaires were returned from 210 unilaterally fitted subjects and 295 bilaterally fitted subjects. The factors measured were: (a) detection, (b) discrimination, (c) speech in quiet, (d) speech in noise, (e) localisation, and (f) comfort of loud sounds.

The participants were divided into three sub-groups within the two main groups. The ‘unilateral fitting group’ was divided into three sub-groups – (i) unaided, (ii) unilateral, and (iii) bilateral. The ‘bilateral fitting group’ was also divided into three sub-groups – (iv) unaided, (v) unilateral, and (vi) bilateral. In the ‘unilateral fitting group’, the ‘unilateral’ sub-group scored significantly higher on all factors than the ‘unaided’ sub-group, except for the ‘comfort of loud sounds’. The ‘bilateral fitting group’ performed similarly, in that the ‘unilateral’ sub group showed improved results over the ‘unaided’ sub-group on all factors, apart from the ‘comfort of loud sounds’. The ‘bilateral’ sub-group scored higher than both the ‘unilateral’ sub-group and the ‘unaided’ sub-group on all factors, also except for ‘comfort of loud sounds’.

Boyman stated that the fact that the groups that did not have hearing aids experienced an increase in the comfort of loud sounds over those having one or two hearing aids could be explained by binaural summation (Boymans, 2009).

From the above findings, it can be concluded that unilateral hearing aids might be sufficient for listeners with mild to moderate hearing loss. For mild hearing impaired listeners, the gain prescribed is small; therefore, due to the head shadow effect, sound will always be louder at the ear closer to the sound source. However, for severely impaired listeners, the sound will always be louder at the aided ear since the gain prescribed is very high.

2.4 The aim of the present study and the contribution to knowledge

The main purpose of our study was to investigate the effect of reverberation on the abilities of normal hearing and hearing impaired listeners to localise speech and non-speech signals. Because hearing aids appear to exacerbate localisation performance in anechoic conditions, might this also be true in reverberant conditions? Therefore, the study also aimed to ascertain whether it is necessary to add reverberation – i.e. a more realistic environment – to clinical testing. Furthermore, in the attempt to investigate whether adding reverberation, which makes the listening environment more difficult, causes localisation to be more challenging for hearing impaired listeners than it is for normal hearing listeners, an investigation into its effects might add to an understanding of the hearing impairment mechanism.

The following are the two major aims of this research:

Firstly, to investigate the effects of reverberation in sound localisation ability for both normal and hearing impaired listeners. Secondly, to assess the effects of amplification on localisation performance of listeners wearing hearing aids. Often the management of sensorineural hearing loss is to fit hearing aids in order to improve audibility; however, aiding has been shown to affect the localisation cues, hence rendering localisation more of a challenge (Byrne et al., 1995, Lutman and Pyne, 2002).

Effects of reverberation

Giguere and Abel (1993) concluded that the ability of normal hearing listeners to localise one-third-octave band noises was more accurate in the absorbent conditions $T<0.2$ s when compared to the reverberant $0.6\text{ s}<T<1.0$ s, the difference in the percentage of correct responses being 12%. However, their study, which showed that reverberation time had a significant impact on sound localisation, focused only on normal hearing listeners.

In contrast, Beeby (2004), who investigated the localisation abilities of bilateral Phonak Claro wearers and normal hearing listeners under simulated reverberant conditions, found that, for both hearing aid wearers and normally hearing listeners, adding reverberation had no effect on localising speech, however, she found an adverse effect on localising beeps and pink noise stimuli. Moreover, feedback from the hearing aid wearers suggested that the localisation task was more challenging under reverberant than anechoic conditions. Lutman and Payne (2002) also concluded that reverberant speech was not found to be harder to localise when compared to the non-reverberant speech for both normal hearing and hearing impaired listeners. However, they had only tested speech stimulus under one reverberant condition, which does not reflect the different reverberant conditions that listeners are exposed to in daily lives. Moreover, the speech stimulus is easy to localise as it is a broadband sound, which covers a wide frequency range, hence it produces ITD and ILD cues in each frequency band as well as spectral cues in the high frequency bands.

This study focused on understanding the effect of reverberant environments on the ability of hearing impaired listeners to localise different stimuli. A crucial aim, therefore, was to understand the effect of the hearing loss, and the hearing aid amplification systems, on the utilisation of ITD and ILD cues separately in both simulated reverberant and anechoic environments. The effect of amplification and reverberation were investigated separately by comparing the hearing impaired performances with and without hearing aids in both anechoic and simulated reverberant listening environments. It was hoped to gain insight into impaired hearing and also into the normal hearing system. Our hypothesis, therefore, was that hearing impaired listeners are more affected by reverberation than normal hearing listeners.

Effect of Amplification

A previous study on the effects of hearing aids on localisation (Van den Bogaert, 2006) focused on non-speech stimuli – i.e. broadband noise, low frequency signals and high frequency signals. The conclusion was that, for hearing impaired individuals, horizontal localisation abilities were actually worse with bilateral hearing aids than without hearing aids. Marginal significant results were only found for the broadband signal, but they were not significant for either the low or high frequency signals. This suggests that the use of hearing aids may impair the ILD or ITD cues; however, the study was under-powered to show significant effect. It was not clear, therefore, as to which localisation cue was more affected by amplification.

Another limitation in the Van den Bogaert study is that he used three different types of hearing aids – Phonak, GN Resound, and Widex. As a consequence, each type may have had different effects on ITD and ILD cues; therefore, the results may be confusing. Moreover, the study did not mention if the sizes of the vents in the ear moulds were equal or not, since changing their sizes would result in varying the amount of amplified and unamplified sound at low frequencies, with the consequence of interfering with the ITD cues. Further research was suggested, therefore, in order to determine how the hearing aids utilise the ILD and ITD cues separately for different sound signals in real reverberant environments, and then to compare the results with performances in anechoic environments.

In this study, some of Van den Bogaert's limitations were addressed. For instance, more participants were recruited (28 compared to his 10) thus giving a greater power to identify the amplification effect. As with the McManus (2008) study, described earlier, the same make and model of hearing aid was used by all participants – the Siemens Prisma 2 Pro; also, the ear moulds were not vented. Further, two listening environments were used: localisation was measured in an anechoic room and a reverberant room with an array of loudspeakers at interval of 9º. Also, in addition to the broadband noise stimuli used by both Van den Bogaert (2006) and McManus (2008), speech stimulus was used, because it is the signal that is most relevant to hearing aid users – i.e., high face validity. Moreover, because of the complex nature of the speech stimulus, any interactions between frequencies were exposed.

Based on the McManus (2008) and Van den Bogaert (2006) results, we hypothesized that amplification would adversely affect the localisation performance of hearing aid users. However, it was not clear how adding reverberation would affect the ITD and ILD cues for hearing impaired listeners with and without amplification. It was interesting to find out if the effect of adding reverberation would differ in the aided conditions when compared to the unaided conditions.

Chapter 3

Experimental Design

3.1 Aims and objectives

The aim of this thesis was to answer the following questions:

1- What is the effect of reverberation in sound localisation abilities for both normal and hearing impaired listeners?

2- What effects do reverberation characteristics, e.g. reverberation time, have on localisation abilities?

3- What effects do hearing aids have on localisation cues?

3.2 Equipment and apparatus

The stimuli were presented from 21 loudspeakers, separated at 9°, positioned at a chamber with the dimensions of 4.5 m wide, 4.5 m long and 3 m high. The subject was seated 1.5 m from the speakers with the centre speaker at 0° azimuth. A further four speakers, positioned 3m from the subject at a height of 85cm, were placed in each corner of the test room to enable the presentation of reverberation. The stimuli, which had been recorded digitally and stored on computer files, were replayed using custom software via the (Creative Extigy) computer sound system digital optical output. These were then passed through the optical input of a Sony amplifier, the output of which fed one selected speaker via a computer-controlled solid-state switch box through the left channel only. The software selected the required speaker and, silently, switched to the required output. The right channel was passed directly to the four corner speakers to introduce reverberation when required. The method used in this study, was based on a method used in previous studies (Verschuur et al, 2005; McManus, 2008). Speakers set up is shown in Fig. 2

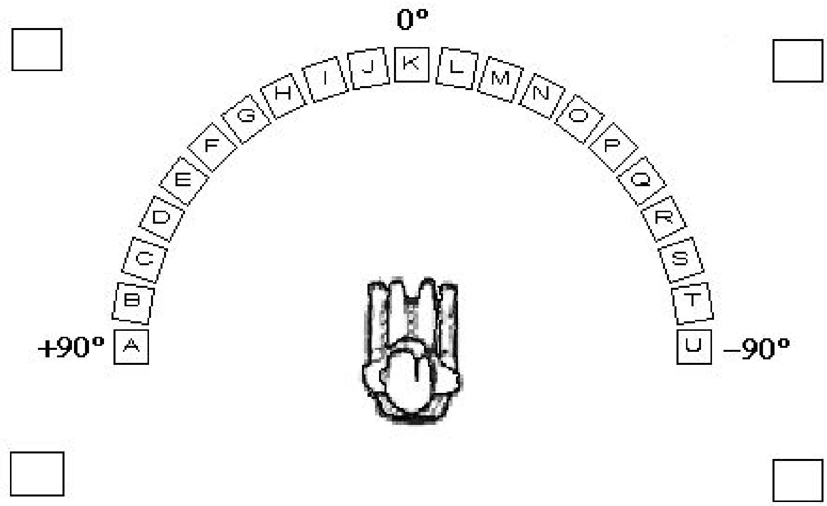


Figure 2 Arrangement of subject and loudspeakers in plan view

3.3 Stimuli

Localisation of low frequency sound relies mainly on ITD cues, whereas, localisation of high frequency sound relies mainly on ILD cues (Middlebrook and Green, 1991; Gabriel et al., 1992). Therefore, in order to discover how both ILD and ITD cues are affected by the hearing aid and reverberation, four different signals were presented at 65 dB SPL. The stimuli were chosen to include:

1- ILD, ITD and spectra cues, i.e. pink noise and speech
2- The envelope structure that is a relevant cue for localisation and realistic sounds, which give high face validity i.e. speech
3- ITD cues mainly, i.e. low-pass pink noise filtered at 1 kHz.
4- ILD cues mainly, i.e. high-pass pink noise filtered at 4 kHz.

A description of the signals is shown in Table 1

Table 1 The characteristics of the stimuli using the sound generator program

Test Stimulus	Characteristics
Speech	Recorded speech “where do I speak from” recorded using male voice (duration 1.2 s)
Pink noise	Train of five pulses of pink noise, with 150 ms pulse duration, 10 ms rise-fall times, and a 50 ms inter-pulse interval time (identical to Byrne et al., 1992)
Low-pass pink noise	The above pink noise was filtered with a low-pass filter at 1 kHz
High-pass pink noise	High-pass pink noise filtered at 4 kHz. The envelope of the high-pass pink noise consisted of one 700 ms pulse with 40 ms rise and fall times.

The stimuli were generated digitally at a sampling rate of 44.1 kHz using Adobe Audition software.

After presenting each stimulus, subjects were required to respond using a laptop computer and clicking on the letter corresponding to the label on the speaker from which they thought the sound originated. Feedback regarding the accuracy of the responses was not given. Stimuli characteristics are shown in Table 1.

The intensity of each stimulus was roved by ±5 dB. This slight change in intensity was included to prevent the subjects from using absolute level cues that might aid in localisation (Byrne et al., 1992).

3.4 Reverberation

The study was performed using simulated reverberation by four corner loud speakers in the anechoic chamber. Reverberation was created using Adobe Audition, which simulates an acoustical space by synthesising the room impulse response at a certain listening position in the space and this is convolved with the original signal. The reproduced sound signal contains both the direct sound and the reverberant sound (Väänänen, 2003). The reverberation components were added to the sound files for the three different stimuli on the right (stereo) channel, while the original (non-reverberant) signal was on the left channel. In Adobe Audition, the reverberation conditions were created by using the following parameters:

- **Total reverberation length**: the number of milliseconds after which the signal will tail off and cut out by 60 dB. Some frequencies may take longer to decay to 60 dB, whereas some will decay more quickly. The effective limit for the total reverberation length is 6000 milliseconds, i.e. a 6-second tail. Long values give longer reverberation tails, but they take longer to compute
- **Attack time**: the amount of time it takes the reverberation to gain full strength. Generally, reverberation is likely to build up over a short time span and decay over a longer one.
- **Diffusion**: this controls the way the echoes build up and how diffuse they are. High diffusion values, i.e. above 900, present smooth reverberation without distinct echoes; whereas low diffusion values give more distinctive echoes.
- **Perception**: this simulates room irregularities, i.e. objects, walls, etc, which add some qualities to the environment by making it more realistic. Higher values give the sensation of echoes coming from different directions, while smaller values create smoother reverberation, which don’t have many echoes. Low diffusion values and high perception values can produce a bouncy effect, whereas high diffusion values and low perception creates an effect like a football stadium.

- **Mixing:**

  Original signal (dry); this feature controls the level of the original signal with respect to other values in order to provide a sense of distance between the source and the listener. Generally, lower value of the original signal is needed to create a 'far distance' sensation. For this study, the original signal (dry) was kept separate from the reverberant (wet) signal, as they were presented via different speakers.

  Early reflections: gives a sense of the room size by controlling the amount of echoes that first reach the listener. The effective value would be half the volume of the original signal.

  Reverb (wet): this adjusts the volume of the dense layer of the sound associated with reverberation. A large volume of the wet component gives the impression of 'far distance' in comparison to the dry signal. However, very large volumes of the wet component may make the reverberation sounds unnatural. For the purposes of this study, the wet component was kept separate and its magnitude, relative to the dry signal, was set differently for the various simulated listening environments.

For the presentation of the reverberant stimuli, the dry component was recorded on the left waveform channel and presented from one of the 21 speakers. The wet component was recoded on the right channel and presented simultaneously from each of the four corner speakers, which had been connected in such a way to retain the same impedance.

The reverberant environments used in this thesis were chosen from Adobe Audition's preset effects; hence the parameters for each listening environment were adjusted automatically. The output stimuli were evaluated by different listeners, all of whom confirmed that the stimuli sounded realistic. The parameters used for the different listening environments are illustrated whenever reverberation was used.

As discussed in section 2.2.3, there are some differences between natural and simulated reverberation. Simulating reverberation using four corner loud speakers in the anechoic chamber does not spatialize reflections observed in natural environments; however, based on subjective observation of the reverberation effect in piloting the experiment and by listening to all tested stimuli we thought that it was of sufficient quality for the purpose of the experiment. The limitation of the study section in this thesis further describes the differences between the two environments.

3.5 Subject selection of the normal hearing listeners

For each experiment, a different number of subjects, in this case university students were recruited according to the following criteria:

- otologically normal in both ears, following an assessment by questionnaire (according to BS EN ISO 389-1) (see Appendix A)
- their pure tone audiometry air conduction thresholds had to be <20 dB HL bilaterally across the frequencies 250Hz to 8kHz (amounting to BS EN ISO 389-1) with 10 dB being the maximum difference in their hearing levels between the ears
- between 18 and 30 years
- no history of, or had undue exposure to, noise or ototoxic drugs
- in good health
- capable of understanding and performing the task
- right handed, since evidence suggests that handedness might have an effect on localisation abilities
- balanced gender mix

The following prospective subjects were excluded:

- those with central nervous system disorders, vestibular problems and those on ototoxic medication
- those who were considered to be incapable of performing the task reliably

3.6 Sources of Error and Bias

The following considerations influenced the design of the study:

Sampling effect

- The group of subjects should closely represent the general population. This was achieved by recruiting a sufficient number

Bias and compensation:

- because the subjects were university students, they did not represent the general population, this could not be compensated
- because hearing threshold levels can be affected by diurnal variation, all testing was carried out during normal working hours
- because the subjects' performances might be affected by learning and fatigue, a random order of runs were made according to a Latin square design
- the subjects were not informed about the study hypothesis in advance
- because practice in, or knowledge of, similar tasks may affect performance, prospective subjects who were music students, professional musicians or highly experienced in psychoacoustic tasks, were excluded

Measurement uncertainty

In order to reduce random uncertainly, each stimulus was presented twice from each of the 21 loud speakers, and this constituted one run. All stimuli were associated with two runs. The replication and order of the runs were counterbalanced by using the Latin square design.

3.7 Calibration

A sound-level meter, which was used to calibrate the output of the loudspeakers, was set in the frequency weighting A, fast RMS and was placed on a tripod in a position of the centre of subject's head – i.e.: the microphone was 1.2 m high and 1.5 m from each of the loudspeakers.

The calibration consisted of two stages:

Firstly, ensuring that all 21 speakers gave an equal sound intensity, which involved the presentation of each of the three stimuli from all the loudspeakers in turn, and then recording the output from each speaker by using the sound-level meter. Software adjustment enabled the outputs of the 21 speakers to be equalized within ±0.5 dB; a different set of adjustments could be used for each stimulus

Secondly, playing each of the three stimuli twice through the speaker at 0° azimuth, with the sound level adjusted via the amplifier volume control in discrete steps until the sound-level meter reached 60 dB (A) within ±0.5 dB. For each stimulus, the average of the two measurements was used throughout the test. The level of reverberant sound from the corner speakers was calibrated by using a sound-level meter also giving a reading of 60 dB (A) within ±0.5 dB.

Daily calibration

The calibration was checked daily by placing the sound-level meter as described above, and by checking its reading when playing each type of stimulus from the central speaker by using the dial settings from the initial calibration.

3.8 The questionnaire

Prior to testing, each subject was given an information sheet and asked to complete a consent form and a questionnaire, which included questions about general health, medication and auditory problems.

3.9 The test procedure

After completing the forms, each subject underwent the screening procedures described in Section 3.5, making sure that all the requirements detailed in the inclusion criteria had been met. Each subject, having been supplied with a lap-top, sat in the centre of the array of loudspeakers, labelled from A to U, and asked to concentrate on a red spot on the central speaker (K) without moving his/her head while the stimulus was presented. Each was then told that different types of stimuli would be played from the 21 speakers. After being presented with each stimulus, the subject was asked to respond by clicking the letter that corresponded to the speaker from which he/she thought the sound originated on the laptop. The subject was given the opportunity to ask any question before testing and told that he/she may terminate the test at any time. The subject was observed by close-circuit television (CCTV) and intercom.

3.10 Recording Data

The subject's responses on the laptop were automatically recorded in the software. For each run, the average absolute error in degrees, i.e. the actual speaker location minus the observed speaker location, was calculated. The average errors were compared between the two trials to check for repeatability. Furthermore, the overall average absolute error across all the speakers was calculated for each stimulus type.

3.11 The Pilot Study

It was necessary to perform a pilot study in order to:

- estimate the duration of the whole test
- ensure that the test instructions are understandable and easy for the subjects to follow
- ensure that all the equipment is working as required

One subject underwent the whole test. The screening session took about 15 minutes and a further 10 minutes were spent in reading the information sheet, and filling out the questionnaire and the consent form. The localisation test took approximately 50 minutes, amounting to about 7 minutes per run, which was considered to be acceptable. No changes needed to be made as a result of the pilot study.

Chapter 4

KEMAR Measurements: Effect of Amplification and Reverberation on Localisation Cues

4.1 Introduction

Reverberation and hearing aid amplification are known to cause some disruption to localisation cues for which there may be several reasons, such as the effects of microphone placement, the earmoulds, group delay, compression, and the noise caused by the hearing aid itself (Byrne et al., 1996).

Therefore, the effect of reverberation and hearing aids on the signals, described earlier in this chapter, was measured. The type of hearing aid used was a Siemens Prisma 2 Pro digital behind the ear (BTE), since it is the same type that was used in the Psychoacoustic Experiment described in Chapter 8.

4.2 Experiment apparatus

The equipment and apparatus used were the same as described in Section 3.2 on p.54. A KEMAR (Knowless Electronics Manikin for Acoustic Research) head and torso simulator was fitted with two small pinnae and two Brüel and Kjaer 4134 microphones in Zwislocki couplers. The microphones were connected through two Brüel and Kjaer 2619 pre-amplifiers to two Brüel and Kjaer 4231 measuring amplifiers. The KEMAR dummy was placed in the same position as the listeners, its height being adjusted with its ears 1.2 m from the floor. The signals from the KEMAR were directed to a laptop via an external sound card and recorded digitally using Adobe Audition 1.5

Each stimulus recoded by KEMAR was presented from each speaker location in four conditions: (i) anechoic unaided, (ii) anechoic aided, (iii) simulated reverberant unaided, and (iv) simulated reverberant aided. Simulated reverberation was created by using Adobe audition to resemble a large auditorium with a reverberation time of 4283 ms, a decay time of 154.1 ms and the clarity (C50) was -2.2 dB.

In the ‘aided’ conditions, two identically programmed Siemens Prisma 2 Pro digital BTE hearing aids, which were electroacoustically checked using the Auricle 2.4 Madsen HiPro module, were fitted to the KEMAR dummy’s ears. This type of aid had been chosen because all hearing impaired listeners who participated in Experiment 4 (see Chapter 8) were fitted with it. The Prisma 2 model has four compression channels and it implements its filter bank in the time domain. The program that was used for fitting the aids was the Siemens fitting software Connex on the NOAH platform and they were set for a moderate to severe sensorineural hearing loss; also, the ‘first fit’ was used so they were set to fairly accurate targets. Omnidirectional microphone configuration was used. Table 2 shows the default gain in each of the four compression channels.

Table 2 Electroacoustic characteristics of the Siemens Prisma 2 Pro digital BTE hearing aid, showing the gain and centre frequency of each compression channel

Channel	**Frequency (Hz)**	**Overall Gain (dB)**	**Compression ratio**	**Compression knee point (dB)**
1	**0-550**	**1**	**1**	**Off**
2	**550-1.4**	**17**	**2**	**45**
3	**1.4-2.8**	**22**	**2.67**	**45**
4	**2.8- >4.5**	**9**	**1.45**	**45**

4.3 Acoustic Stimuli

As described in Section 3.3 on p.55

4.4 Results and Discussion

The KEMAR measurements revealed that the effect of simulated reverberation and amplification on the speech stimulus was similar to that of the pink noise stimulus.

This result was expected, since both pink noise and speech stimuli are broad-band signals that include both ILD and ITD cues. Consequently, during the discussion of KEMAR measurements, only the pink noise stimulus was used in order to make a comparison easier with the high-pass pink noise and low-pass pink noise.

4.4.1 Measurement of ILD

Inter-aural differences were measured with, and without, the hearing aids in two different environments – anechoic and simulated reverberation. Three types of stimuli were used (i) pink noise, (ii) high-pass pink noise filtered at 1 kHz, and (iii) low-pass pink noise filtered at 4 kHz. The inter-aural level differences between both ears were calculated as the difference in the Root Mean Square (RMS) power between the right and left ears over the whole duration of the signal.

Previous experiments indicate that, as the frequency of the stimulus increases the level difference between the ears also increases (Wightman and Kistler, 1992). This may be explained by the 'head shadow' effect that mostly affects frequencies above 1500 Hz (Moore, 1995). Results from KEMAR measurements revealed that the highest ILD was for high-pass pink noise, followed by pink noise; whereas the lowest ILD was for low-pass pink noise for both anechoic and simulated reverberant conditions (Fig.3). These results corresponded with previous findings by Feddersen et al., (1957), Middlebrooks et al., (1989) and Shaw (1974), suggesting that, for frequencies lower than 1 kHz, the ILDs are generally lower than 5 dB, whereas ILD increases at higher frequencies.

Another observation was the change in the slope of the ILD function with increasing azimuth. For low frequencies, the shape of the ILD function was relatively flat; however, at high frequencies, the rate of change increased for medial speaker locations (18º to 63º) and then began to decrease and remained relatively flat for the lateral speaker positions.

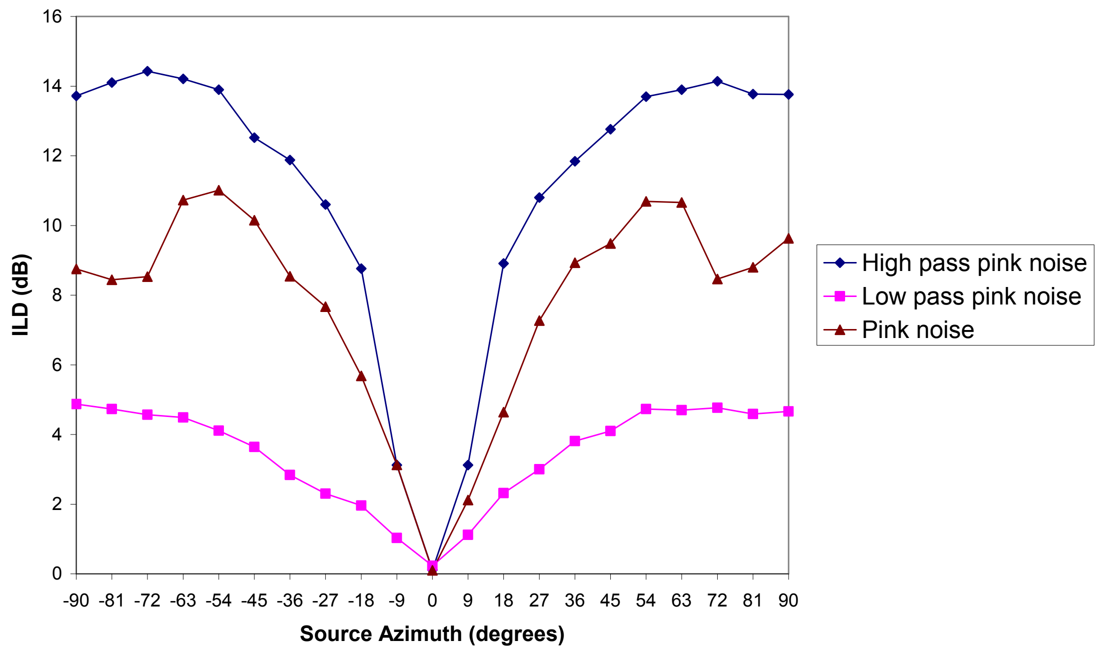


Figure 3 Measured ILDs for the high-pass pink noise, low-pass pink noise and pink noise stimuli. Measurements in this figure represent the anechoic, unaided measurements.

Adding reverberation into a listening environment showed a very slight distortion of the ILD cues for all three signals, however, the distortion was not consistent and did not follow a trend (Fig.4). The results are consistent with the findings of Cunningham (2000), as, in her experiment, ILD and ITD cues were computed using a spherical head model in both anechoic and reverberant environments and results showed that the measured mean ILD error in the reverberant condition was actually as small as the just noticeable difference (JND) reported in the literature for the anechoic conditions. Moreover, Devore and Delgutte (2010) found that directional sensitivity is significantly improved when the virtual space stimuli contain both ITD and ILD cues, suggesting that ILDs provide a more reliable localisation cue than envelope ITDs in reverberation (Devore and Delgutte, 2010).

Level differences were measured by hearing aids being placed on KEMAR and the results were compared to the ILDs measured on KEMAR without hearing aids. The 'intensity difference' for frequencies below 1 kHz did not change to a great extent with aiding (Fig. 5). However, the ILD of the high-pass pink noise was greatly suppressed by compression (Fig. 5). For the high-pass pink noise, the average decrease in ILD cues was about 4 dB and 5 dB in the anechoic and simulated reverberant conditions respectively. This most likely occurred because both hearing aids were fitted with identical compression hearing aids.

In unaided conditions, the inter-aural intensity will vary only if the direction of the sound source changes, hence it will provide a cue for localisation. However, if both ears are fitted with compression hearing aids of equivalent gain, the hearing aid on the near ear side (i.e. receiving higher input) will provide less gain than the aid in the far ear (Byrne et al. 1998a). This will result in fewer inter-aural level differences in the aided conditions when compared to the unaided conditions.

The effect of amplification on the ILD cues of the high-pass pink noise was more prominent in the simulated reverberant environment than the anechoic environment by an average of 1 dB difference. That may be explained by the fact that reverberation is not direct, hence the compression system in the two hearing aids might process reverberation differently, resulting in different S/N ratios in both ears.

Further, the change in the ILD function with increasing azimuth was less prominent in aided conditions than in unaided conditions for the high-pass pink noise (Fig. 6).

The effect of reverberation on ILD cues

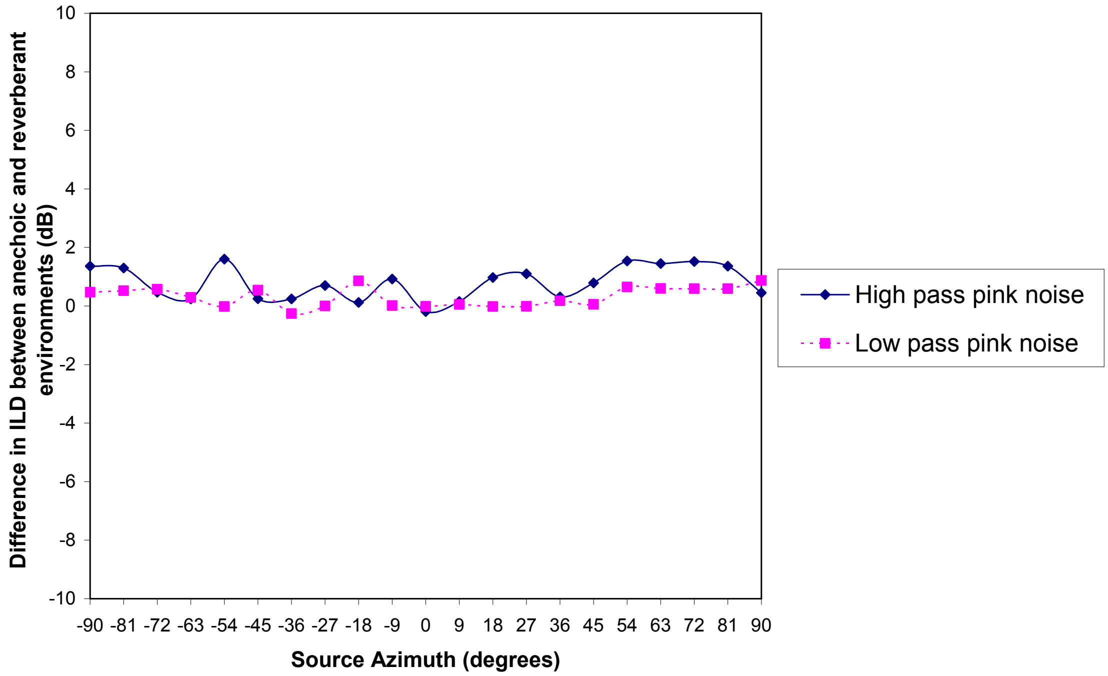


Figure 4 The difference in ILDs between the anechoic and simulated reverberant environments for the high-pass and low-pass pink noise. Positive values reflect higher ILDs in the anechoic environment when compared to the simulated reverberant environment.

The effect of amplification on ILD cues

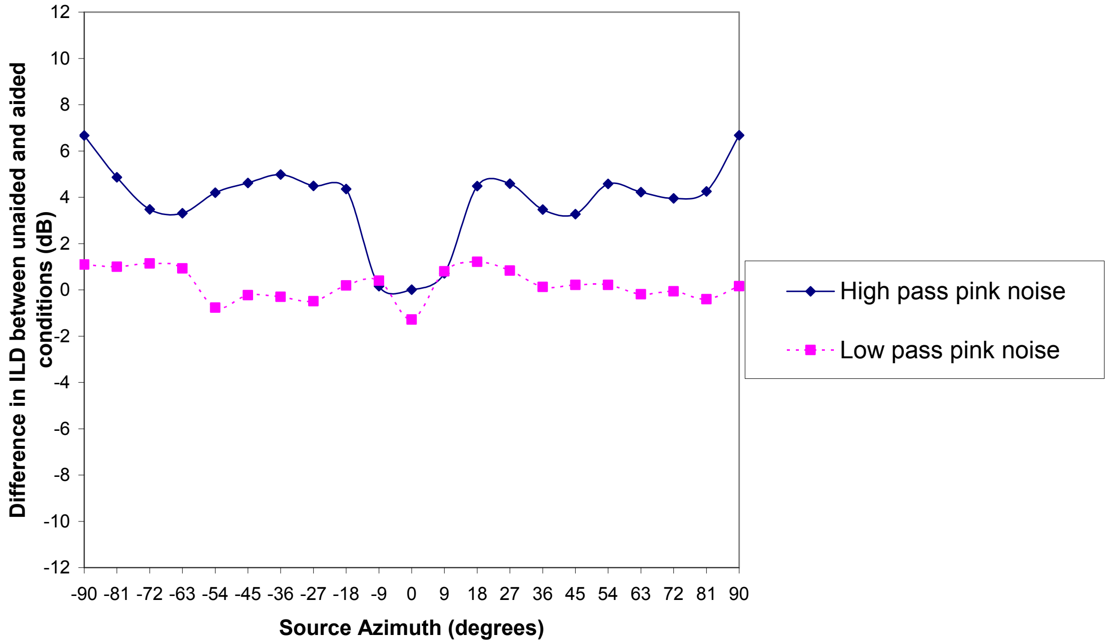


Figure 5 The difference in ILDs between the unaided and aided conditions for the high-pass and low-pass pink noise. Positive values reflect higher ILDs in the unaided condition when compared to the aided condition.

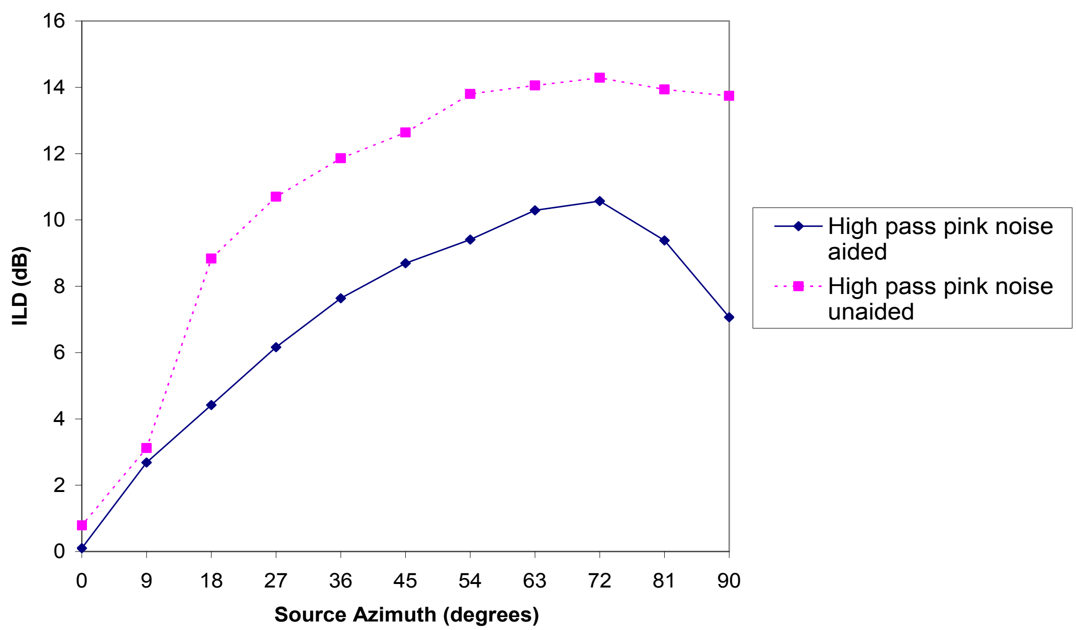


Figure 6 Measured ILDs without hearing aids (solid lines) compared to ILDs measured with hearing aids placed on KEMAR (dashed lines) for the high-pass pink noise in the anechoic environment averaged over both sides.

4.4.2 The measurement of ITD

The ITD was measured under four conditions: with the hearing aids and without the hearing aids in both anechoic and simulated reverberant environments. Hilbert transformation was implemented in order to detect envelops of the signals. The measured ITDs were determined by calculating the delay generating the maximum peak in cross correlating the signal's Hilbert envelopes from both ears using a MATLAB program.

Without amplification, and in the anechoic conditions, the ITD measured for the low-pass pink noise was higher than that measured for the high-pass pink noise and pink noise (Fig. 7), especially in the medial speaker locations (27º to 63º). This result is consistent with previous findings of McManus, (2008) and Roth et al., (1980) suggesting that ITDs are frequency dependent, where ITDs measured for low frequencies were higher than ITDs measured for high frequencies.

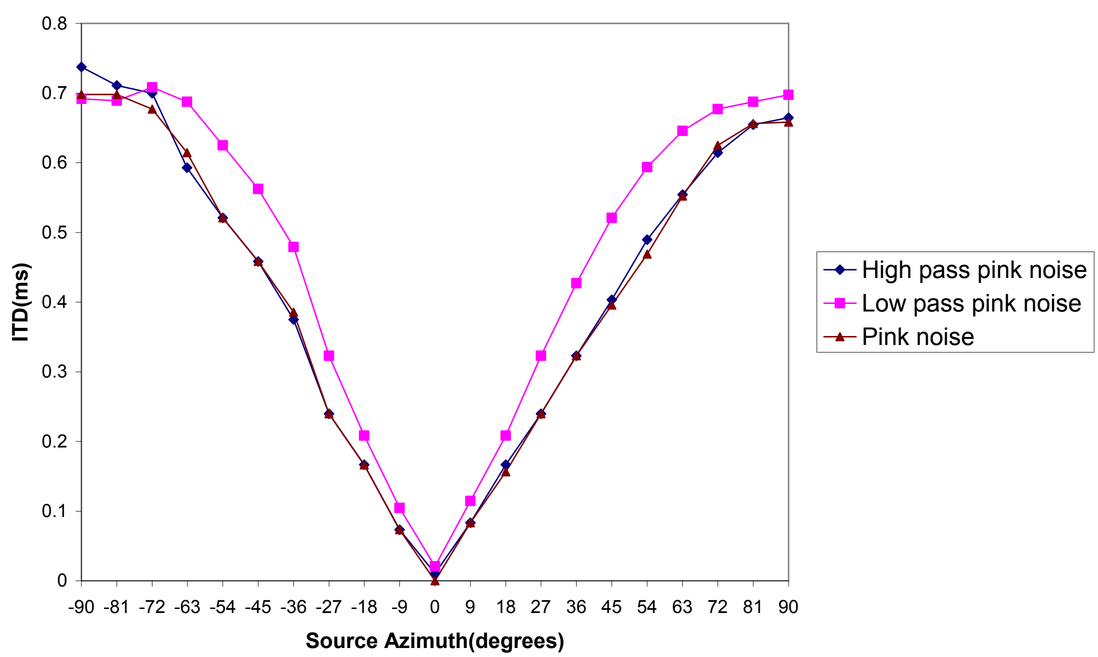


Figure 7 Measured ITDs for the high-pass pink noise, low-pass pink noise and pink noise stimuli. Measurements in this figure represent the anechoic, unaided measurements.

With amplification, the measured ITD of the pink noise did not differ when compared to the unaided measurements in both anechoic and simulated reverberant environments. However, there was a slight decrease – an average of 0.02 ms – in the ITD of high-pass pink noise with amplification, and a greater decrease – an average of 0.06 ms – in the ITD of low-pass pink noise with amplification (Fig. 8); this effect was apparent for both anechoic and simulated reverberant environments. The most likely explanation was the slight asymmetry between the two hearing aid processing systems, suggesting a possible mismatched group delay – the amount of time needed for the signal to propagate through the system – between the hearing aids (Kates, 1998). If the group delay between the two hearing aids is mismatched, the magnitude of the ITD measurements will change. Moreover, as mentioned earlier, the hearing aids used in these measurements were programmed for a moderate to severe sensorineural hearing loss, in which low frequency sounds would be reasonably audible. Therefore, with amplification, the audible signal would be a mixture of the amplified and unamplified sound. Consequently, group delay may have created a phase difference between the amplified and the unamplified signals, thereby distorting the ITD cues (Noble et al., 1998).

In reverberant environments, simulated reverberation showed a small distortion of ITD information, which was more pronounced in the low-pass pink noise to the order of 0.01 ms (Fig. 9). The distortion was very small and did not follow a trend, which suggests that it was more likely to be due to measurement uncertainty.

The effect of amplification on ITD cues

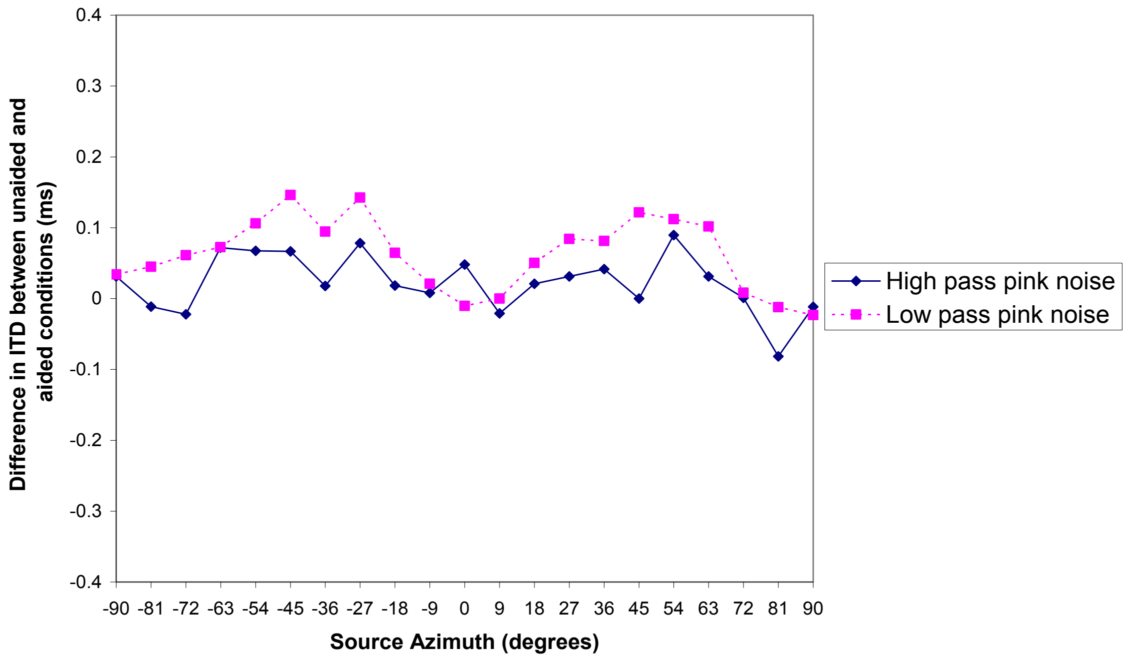


Figure 8 The difference in ITDs between the unaided and aided conditions for high-pass and low-pass pink noise. Positive values reflect higher ITDs in the unaided condition when compared to the aided condition

The effect of reverberation on ITD cues

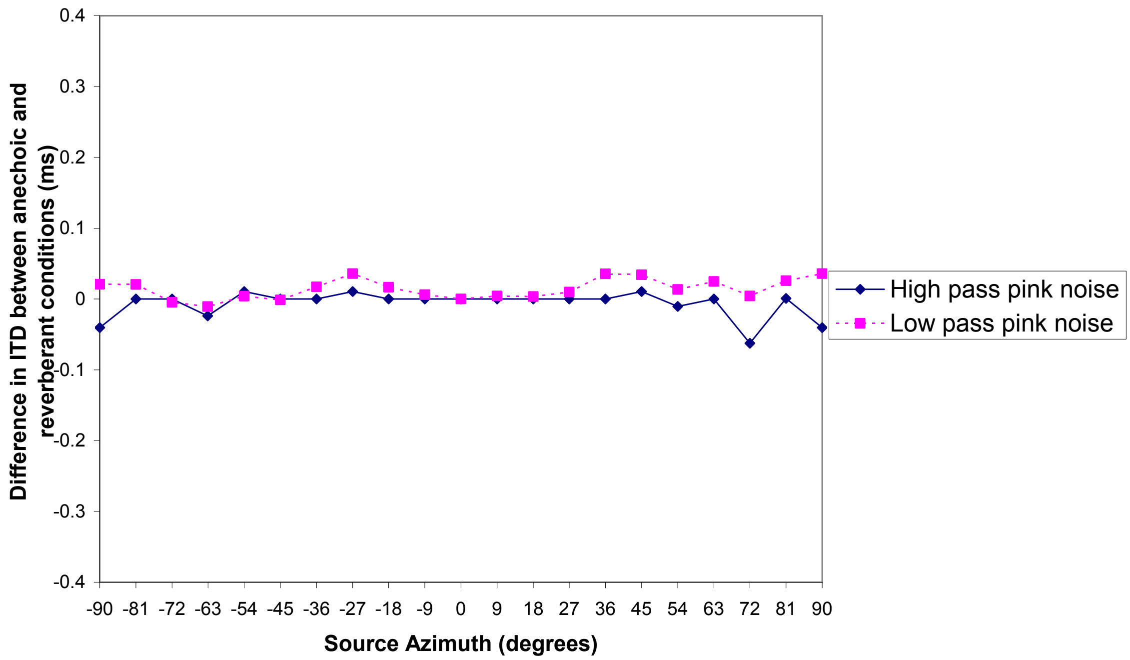


Figure 9 The difference in ITDs between the anechoic and simulated reverberant environments for high-pass and low-pass pink noise. Positive values reflect higher ITDs in the anechoic environment when compared to the simulated reverberant environment

4.5 Conclusions

ILDs were highest for high frequency pink noise and generally increased as the sound source moved laterally. Placing hearing aids on KEMAR resulted in a decrease in the inter-aural level difference of the high-pass pink noise. This could be explained by the compression of the hearing aids that resulted in a decrease in the inter-aural level differences. However, simulated reverberation did not show an impact on the measured ILD cues in either the aided or the unaided conditions.

ITD cues were highest for the low-pass pink noise. With hearing aids, measurements of the ITDs of the high-pass pink noise and low-pass pink noise were slightly decreased. This may be explained by the mismatch of the group delay between the two hearing aids. Moreover, adding reverberation did not show any significant impact on the measured ITD cues.

Chapter 5

Experiment 1: Localisation Performance in a Reverberant Environment

5.1 Introduction

Kuttruff (1991) states that, "obviously one subconsciously expects to encounter some reverberation which bears a certain relation to the size of the room". Because most listening environments – lecture rooms, churches, congress halls, theatres, etc – are, to some degree, reverberant, a certain level of reverberation is important in order to offer more realism. Moreover, the early reflections of reverberation enhance loudness, thereby giving better speech intelligibility (Begault at al. 2004).

Giguere and Abel (1993) examined the ability of normal hearing listeners to localise a 1/3-octave bandwidth noise in both reverberant and absorbent conditions. In an experiment where they changed a reverberant room – reverberation time 0.6 - 1 s – into an absorbent room – reverberation time less than 0.2 s – by adding floor carpeting and sound absorbent wall and ceiling panels, they found that performance was better in the absorbent room than in the reverberant room. Localisation accuracy decreased with increases in azimuth angle from 85% (15º) to 60% (75º). Hartmann (1983) found that adding reverberation in the environment does not affect the localisation of sounds with abrupt onsets (50 ms pulse of a 500 Hz sine tone), however, he found that reverberation reduces the ability to localise signals without attack transients (continuous broad-band).

Consequently, Experiment 1 was conducted in a real, highly-reverberant room in order to investigate the localisation performance of normal hearing listeners in the reverberant environment.

5.2 Research question and hypotheses:

The purpose of testing was to answer the following research question:

Does the localisation performance in a real reverberant environment differ from the localisation performance in an anechoic environment?

Hypothesis 1: there will be no difference to the average localisation error of the speech stimuli (high-pass speech, low-pass speech and speech) in the real reverberant room compared to that found in anechoic conditions.

Hypothesis 2: the average localisation error of the pure tones – 500 Hz and 4 kHz – in the real reverberant room will be greater than the average localisation errors in anechoic conditions.

5.3 The large reverberant room

Situated in the acoustic laboratories in the Raleigh building, the large reverberation chamber in ISVR was built in March 1968 and was designed to create a diffuse sound-field. It measures 9.15 x 6.25 x 6.10 m (348 cubic metres) and consists of non-parallel, vertical walls and a slightly inclined ceiling. In order to obtain maximum sound reflections, the surfaces are very hard and finished with gloss paint to achieve a high reflection coefficient. The room is also isolated from its surrounding building by rubber vibration isolators and the floor has a steel vibration isolation pad set into it. Reverberation times are shown in Table 3

Table 3 Third Octave reverberation times for empty large reverberant chamber in ISVR. These values were obtained from ISVR memorandum No. 267

Third-octave band centre frequency (Hz)	125	250	500	800	1 k	2 k	4 k
Reverberation Time (s)	11.7	10.4	6.8	8	7.7	5.4	3.3

5.4 Subject selection

Twenty-two normal hearing listeners were recruited to participate in the experiment. For details on inclusion criteria see Section 3.5 on p.59

5.5 Stimuli

Table 4 shows the stimuli used; these were chosen to include:

1. (a) the main localisation cues (i.e. ITD, ILD, and spectra cues), (b) the resonant nature in consonants, (c) the harmonic structure of vowels, (d) the spectral envelope, and (e) realistic sounds that give high face validity (speech)
2. ITD cues mainly (low-pass speech), cut off frequencies at 1600 Hz.
3. ILD cues mainly (high-pass speech), cut off frequencies at 1600 Hz.

Pure tone stimuli were also used to allow for the comparison between stimuli with abrupt onset/offset times (high-pass speech, low-pass speech and speech) and stimuli with gradual onset/offset times (pure tones).

Table 4 The characteristics of speech and pure tone stimuli using the sound generator program

Test stimulus	
Speech stimulus	Recorded Speech “where do I speak from” recorded using a male voice.
Low-pass speech stimulus	The same speech sentence was filtered with a low-pass filter using a cut off frequency of 1600 Hz
High-pass speech stimulus	The same speech sentence was filtered with a high-pass filter using a cut off frequency of 1600 Hz
500-Hz pure tone	40 ms onset and offset ramps. The total tone length is 1 s
4-kHz pure tone	40 ms onset and offset ramps. The total tone length is 1 s

5.6 Test Design

In order to answer the research questions, we compared the reverberant and anechoic environments:

Real reverberant room:

1) High-pass speech in real reverberant condition
2) Low-pass speech in real reverberant condition
3) Speech in real reverberant condition

4) 500 Hz pure tone in real reverberant condition
5) 4 kHz pure tone in real reverberant condition

Anechoic Room:

6) 500 Hz pure tone in anechoic condition
7) 4 kHz pure tone in anechoic condition
8) High-pass speech in anechoic condition
9) Low-pass speech in anechoic condition
10) Speech in anechoic condition

5.7 Methodology

5.7.1 Recording Data

The experiment, which was performed on 22 different participants, was conducted in two runs. In each run, sound was presented twice from each of the 21 speakers. For all experiments in this thesis, the absolute error was calculated as the difference between the response and the correct location; all differences for each speaker location were summed and then divided by the number of presentations. This average is calculated as the error.

5.7.2 Normality Test

The normality of the dependent variables was tested using the Shapiro-Wilk normality test, whereby a significance value greater than 0.05 suggests that the data distribution is normally distributed. The results indicated that the original dependent variables were not normally distributed ($p<0.05$) and were positively skewed; therefore, a log transformation was used in order to transform the variables into normal distribution. There were many error values, which were 0, and log (0) is not defined. Therefore, a constant was added to the error term in order to ascertain the log transformation. After several trials, the constant was found to be 1.5, the transformation used being (log_Spk_C=ln(SPK_C +1.5). The resulting log-transformed data sets were found to be normally distributed, consequently, the log transformed variables were used to carry out the parametric analysis.

5.8 Results

Scattergrams (Figs. 10 and 11) show the spread of the data for the anechoic and real reverberant conditions. The X-axis shows the speaker from which the sound was originated (90º to 90 º), whereas, the Y-axis shows the speaker where the listener thought the sound was coming from – i.e. Listener response. Listeners' responses were sorted into five groups – 0%-20%, 21%-40%, 41%-60%, 61%-80% and 80%-100% – each group representing the percentage of responses at each speaker location. These five groups were represented in the figures by five differently sized dots; the smallest dot represents 0%-20% responses while the largest dot represents 80%-100% responses. Listeners' responses were further averaged over locations directly and presented in the figures by the dashed line. The bold diagonal line is the zero line, where the error made by the listener is zero. If the perceived speaker location was the same as the actual speaker location, then the error would be zero and the response would lie on the zero line.

Fig. 10 shows that in the anechoic chamber and for the speech stimuli (speech, high-pass speech and low-pass speech), the average line lies mostly on top of the zero line, suggesting very minor localisation errors. For the 500 Hz pure tone, the average line was slightly shifted from zero line, especially at lateral speakers, suggesting small localisation errors. However, for the 4 KHz pure tone, responses were very scattered, suggesting that that localisation task was harder than the other stimuli.

Fig. 11 shows the spread of the data for the real reverberant environment. Listeners responses for the speech stimuli were similar to those in the anechoic chamber (mainly an average of zero error), however, for both the 500 Hz and 4 KHz pure tones, responses were more scattered in the reverberant than the anechoic environment.

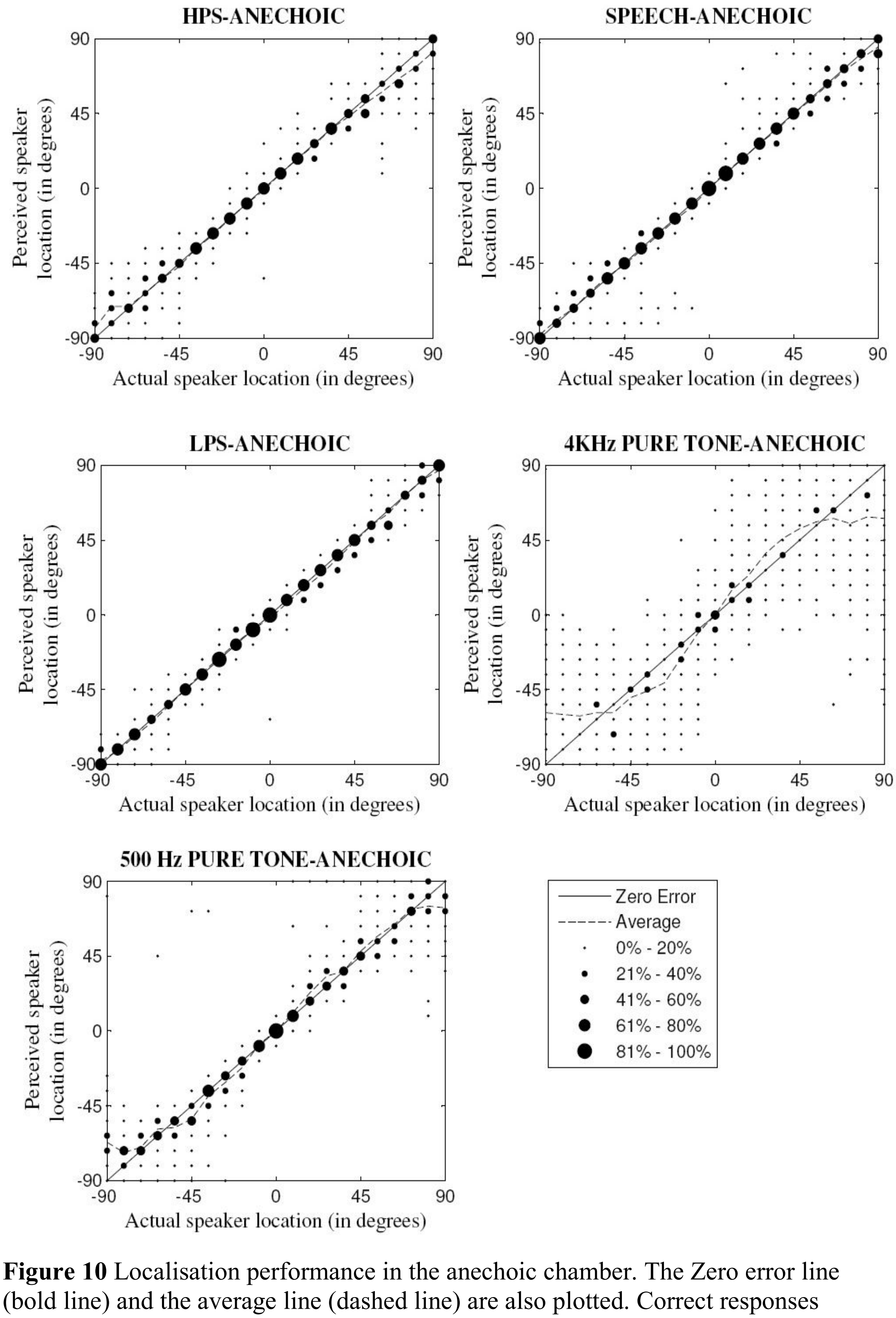


Figure 10 Localisation performance in the anechoic chamber. The Zero error line (bold line) and the average line (dashed line) are also plotted. Correct responses should lie on a diagonal line (zero error).

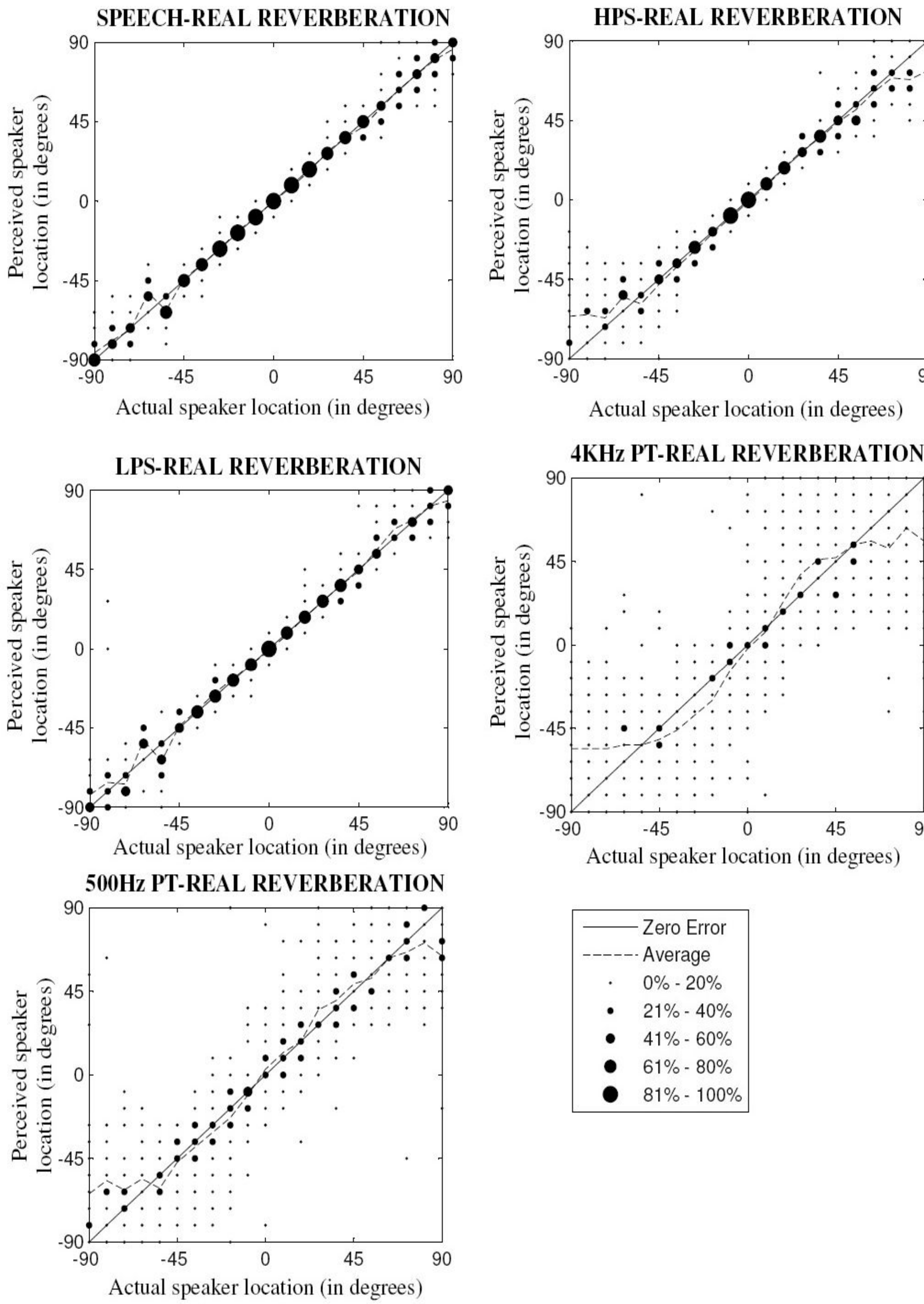


Figure 11 Localisation performance in the real reverberant room. The Zero error line (bold line) and the average line (dashed line) are also plotted. Correct responses should lie on a diagonal line (zero error)

5.9 Analysis

The 21 speakers were grouped in to three sets of speakers based on their locations. These three speakers are named as “SPK_C”, “SPK_M” and “SPK_L”.
Central speakers “SPK_C” include speakers from -27° to +27°,
Medial speakers “SPK_M” are those from -36° to -63°, and +36° to +63°, and
Lateral speakers “SPK_L” are those from -72° to -90° and +72° to +90°
An alpha level of 0.05 was used for all statistical analysis throughout the thesis.

5.9.1 Real Reverberant Room vs. Anechoic Room

A three-way ANOVA with repeated measures was performed in order to answer the research question. The dependent variable was the average localisation error across all speaker positions (log transformed), and the three independent variables were: (i) reverberation (real reverberation and anechoic environments), (ii) the stimulus (high-pass speech, low-pass speech, speech, 500 Hz pure tone and 4 kHz pure tone), and, (iii) the speaker location (central, medial and lateral). The results from the three-way ANOVA with repeated measures revealed that the main effect of reverberation ($F_{(1, 13)} = 16.035$), speech ($F_{(4, 52)} = 157.079$), and speaker location ($F_{(2, 26)} = 92.752$), were significant at $p<0.05$.

The F-test results suggested that the mean localisation error for real reverberant condition (mean= 9.72) was significantly higher than the mean localisation error for anechoic conditions (mean=7.64). Similarly, the F-test result indicated that there was a significant difference between the mean localisation errors for the different stimuli. The mean localisation error was highest for 4 kHz pure tone (mean=21.11) and lowest for speech stimulus (mean=4.83). Further, the pair-wise comparison of the mean localisation error indicated that there was a significant difference between all the pairs of 4 kHz pure tone at $p<0.05$ and 500 Hz pure tone. The only pairs of speech stimuli for which the difference was not significant were “high-pass speech-low-pass speech” and “low-pass speech-speech”. The remaining pairs of speech stimuli were statistically significant.

Likewise, the F-test result indicated that there was a significant difference between the mean localisation errors for different speaker locations. The mean localisation error was highest for lateral position (mean=17.85) and lowest for central position (mean=7). Further, the pair-wise comparison of the mean localisation error indicated that there was a significant difference between all the pairs of speaker locations. The pair-wise comparison of the mean localisation error indicates that there was a significant difference between all the pairs of speaker locations.

The pair-wise interaction effect of "reverberation-stimuli" ($F_{(4, 52)} = 3.56$) was significant. Contrasts indicated that the increase in localisation errors between the reverberant environment and the anechoic environment were not significant when comparing the high-pass speech, low-pass speech and 4 KHz with the speech stimulus. However, for the 500 Hz pure tone, the increase in localisation errors between the reverberant environment and the anechoic environment was significant when compared to the increase in the localisation error for the speech stimulus.

The pair-wise interaction effects of "reverberation-speaker location" ($F_{(2, 26)} = 6.03$), and "stimuli-speaker location" ($F_{(8, 104)} = 8.17$) were statistically significant. Similarly, the interaction effects of "reverberation-stimuli-speaker location" ($F_{(8, 104)} = 4.87$) was significant at $p<0.05$. Graphical representation of the results is shown in Figs. 13 and 14

A one-way repeated measure ANOVA was performed in order to assess which stimulus differs significantly in the real reverberant room when it was compared to the anechoic room; the dependent variable being the average localisation error across all speaker positions. The independent variable was the stimulus – (i) high-pass speech, (ii) low-pass speech, (iii) speech, (iv) 500 Hz pure tone, and (v) 4 kHz pure tone – under both reverberant and anechoic conditions – a total of ten stimuli.

The main effect of stimuli ($F_{(9, 117)} = 59.74$) was significant at $p<0.05$. Results revealed that for the three speech stimuli the localisation accuracy between the real reverberant room and the anechoic room did not differ significantly at $p>0.05$. Moreover, the localisation accuracy of the 4 kHz pure tone did not differ significantly when comparing the performance under reverberant and anechoic conditions. However, the localisation performance of the 500 Hz pure tone differed significantly in reverberant (mean=14.3) and anechoic conditions (mean=8.88). The mean values are shown in Table 5 and a graphical representation of the results is shown in Fig. 12

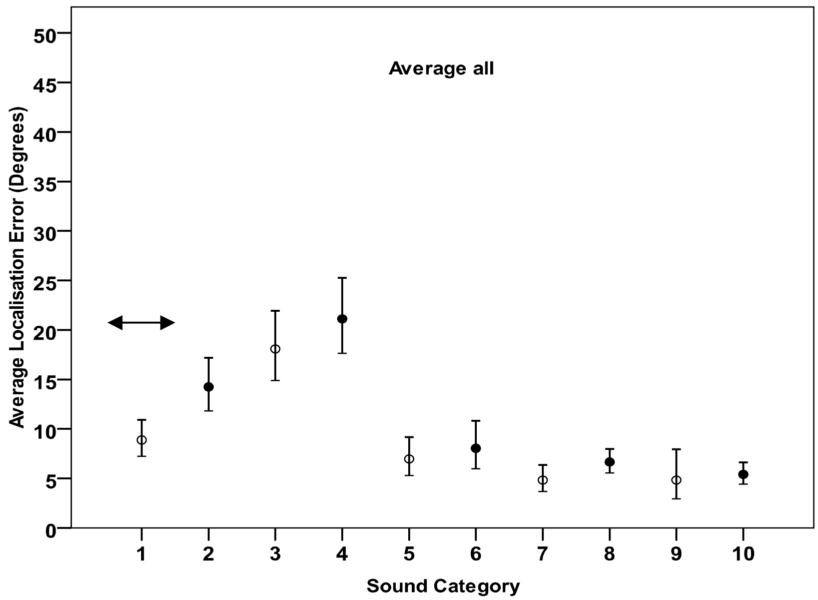
The results, therefore, supported the hypothesis that the average localisation errors for the speech stimuli in the real reverberant room would not differ significantly from that in the anechoic room.

The results also supported the hypothesis that the average localisation errors for the pure tone stimuli in the real reverberant room and the anechoic room would differ significantly.

Anechoic vs. real reverberation for all stimulus types

Table 5 Mean values of the localisation errors for different stimuli in the anechoic and real reverberant conditions

Stimuli	Real rev	Anechoic	Significance
Speech	5.40	4.83	Ns
Low-pass speech	6.66	4.84	Ns
High-pass speech	8.03	6.96	Ns
500 Hz pure tone	14.25	8.88	$p<0.05$
4 KHz pure tone	21.11	18.07	Ns



Numbers 1~10 represent the following categories of sound:

1 500 Hz pure tone in anechoic condition
2 500 Hz pure tone in real reverberant condition
3 4 kHz pure tone in anechoic condition
4 4 kHz pure tone in real reverberant condition
5 High-pass speech signals in anechoic condition
6 High-pass speech signals in real reverberant condition
7 Low-pass speech signals in anechoic condition
8 Low-pass speech signals in real reverberant condition
9 Speech signals in anechoic condition
10 Speech signals in real reverberant condition

Figure 12 The average localisation error for all speaker positions with respect to the above sound categories (log transformed data). Error Bars represent "Mean ±1 standard deviation of sample". The arrow represents a significant difference.

Anechoic vs. real reverberation for the speech stimuli averaged together

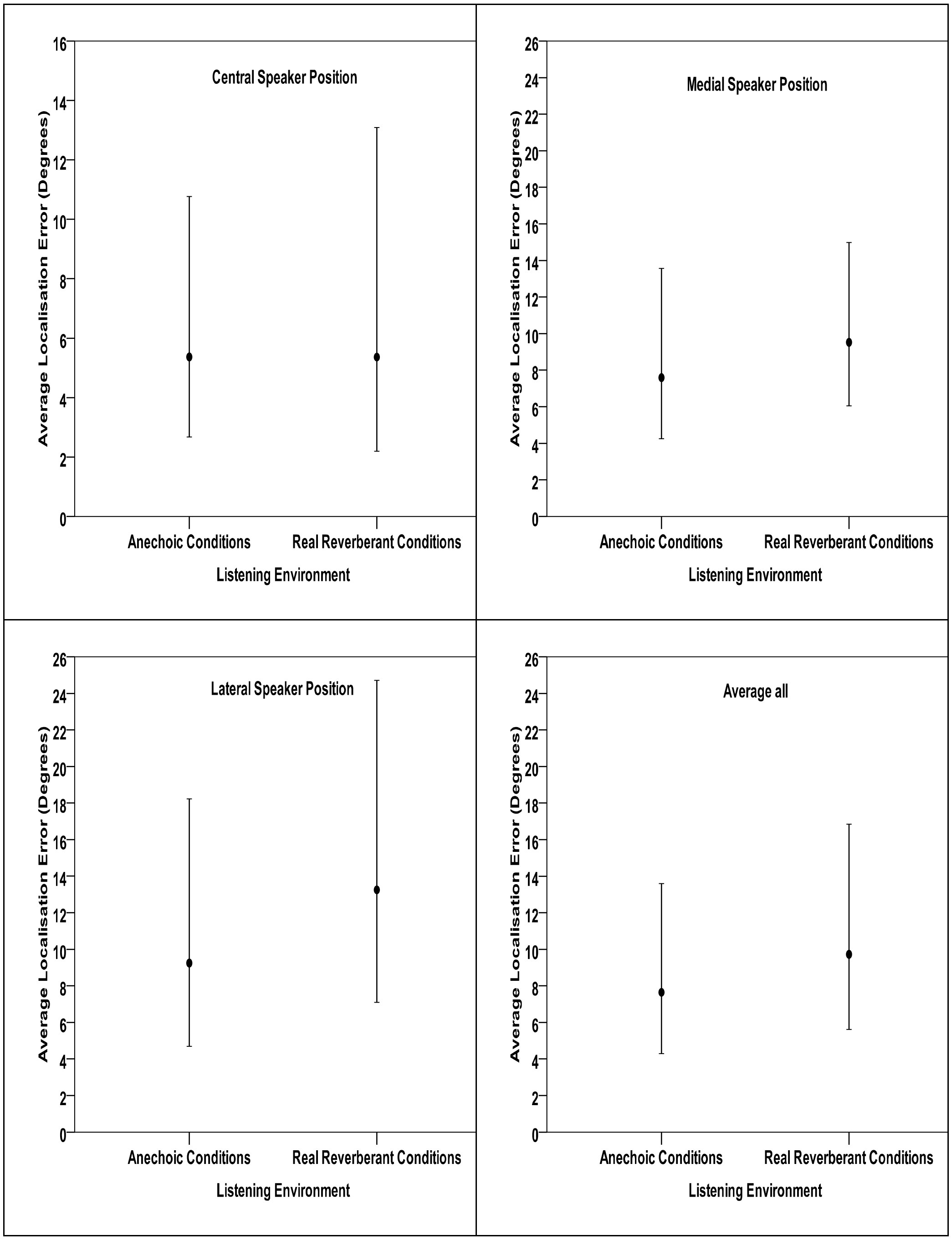


Figure 13 The average localisation error for all speaker positions for the average of all the speech stimuli in both anechoic and real reverberation (log transformed data). Error bars represent "Mean ±1 SD".

Anechoic vs. real reverberation for the pure tones averaged together

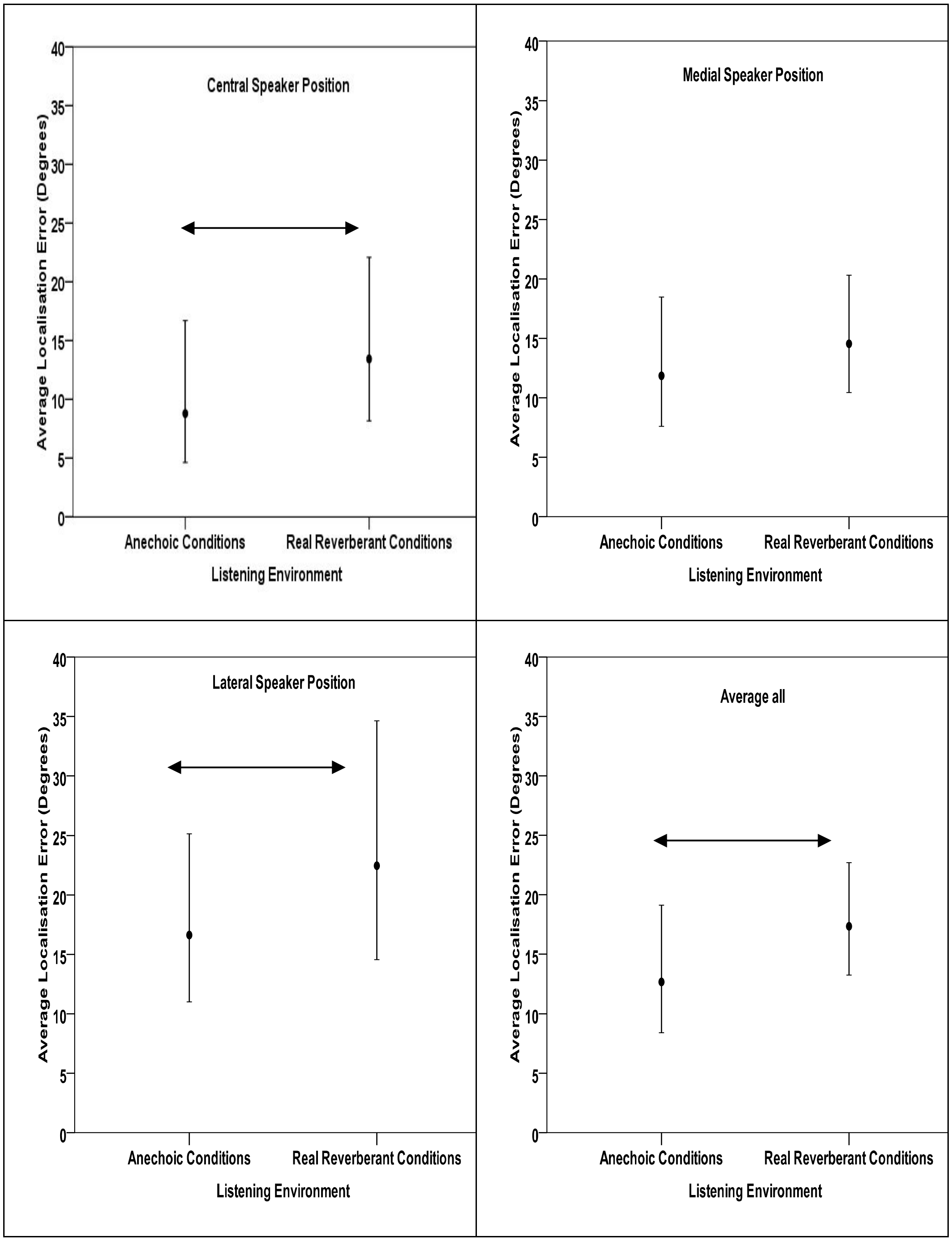


Figure 14 The average localisation error for all speaker positions for the average of pure tones in both anechoic and real reverberation (log transformed data). Error Bars Represent "Mean ±1 SD". Arrows represent significant differences.

5.9.2 Localisation of pure tone stimuli vs. speech stimuli

One other observation was made. The average localisation errors for the pure tones were greater than for the speech stimuli. Therefore, a one way repeated measures ANOVA test was conducted to assess whether means on a dependent variable (log transformed average localisation error) are significantly different among the ten different stimuli (high-pass speech, low-pass speech, speech, 500 Hz pure tone and 4 kHz pure tone). Results from this test revealed that the stimuli effect ($F_{(4, 52)} = 121.71$) was significant. Contrasts indicated that localisation performance for all the other stimuli differ significantly from the speech stimulus, except for low-pass speech. The localisation errors were the greatest for the 4 kHz pure tone, followed by the 500 Hz pure tone, whereas the lowest localisation error was found for the speech stimulus. The results are represented graphically in Fig. 15 and the mean values are shown in Table 6

The above results confirmed the third hypothesis, which suggested that the average localisation error of the pure tones (500 Hz and 4 kHz) would be greater than that of the speech stimuli (high-pass speech, low-pass speech and speech).

Table 6 Mean values of the localisation errors for the different stimuli in the real reverberant room

Stimuli	Mean
Speech	5.40
Low-pass speech	6.66
High-pass speech	8.03
500 Hz pure tone	14.25
4 KHz pure tone	21.11

Localisation of different stimulus types in the real reverberant room

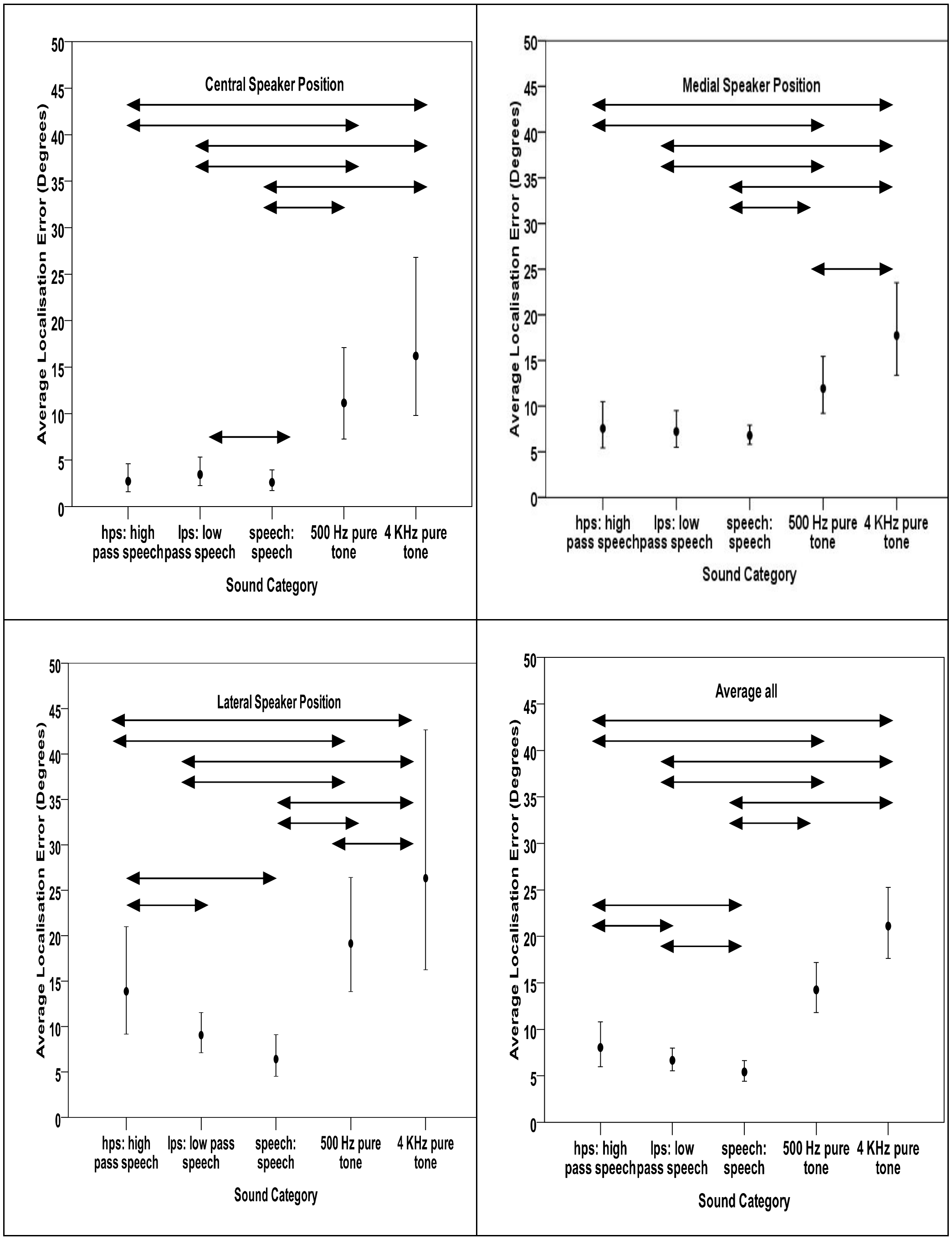


Figure 15 Average localisation errors for the different speaker positions and for all stimuli in the real reverberant room (log transformed data) . The error bars represent "Mean ±1 SD". Arrows represent significant differences.

5.10 Real reverberation vs. simulated reverberation

As mentioned earlier, the previous experiment was performed in a real reverberant chamber that differs in dimensions, construction and acoustical characteristics, from the anechoic chamber, where the control measurements were performed. Therefore, in order to provide for a better comparison with the anechoic measurements, and also to allow varying the reverberation time in future experiments in order to create different listening environments, a simulated reverberation in the anechoic chamber was performed using the same test set-up and speech stimuli. Simulated reverberation, using Adobe Audition, was added to the signals to resemble the reverberation of a church with a hundred seats. The effect was a reverberation time of 3224 ms, an attack time of 60 ms, a high frequency absorption time of 389 ms, a perception of 10 ms and clarity (C50) of -0.9 dB. Details of the simulation parameters are shown in Section 3.4 on p.57

The next experiment, therefore, was performed in order to answer the following research question:

1) Does simulated reverberation produce similar results as real reverberation - is the simulation valid?

The hypothesis was that the average localisation error of the speech stimuli in the real reverberant room will not differ significantly from the simulated reverberant room.

5.10.1 Results

Fig. 16 shows the spread of the data (average absolute localisation error) for the real (left panel) and simulated (right panel) reverberant environments. The average localisation errors for the low-pass speech and the speech stimuli were mostly zero or close to zero (i.e. less than 10º); however the average localisation errors were higher for the high-pass pink noise in both real and simulated reverberation. Moreover, the localisation performances were very similar when comparing real and simulated reverberant environments. To confirm these observations, the average performance of the respondents at 90° (average of +90° and -90°) was calculated for both listening environments individually. Further a t-test was applied for each stimulus in order to compare the localisation performance between the real and simulated reverberation at 90°. The t-test results suggested that there was no significant difference in the performance of the respondents at 90°. Table 7 shows the t-test results.

Table 7 T-test results for the localisation performance in the simulated and real reverberation environments at 90°

	Type	N	Mean	Std. Deviation	Std. Error Mean	Conclusion
HPS	Real Reverberation	192	70	16.56	t (df =302) = -0.817 p = 0.414	No significant difference
	Simulated Reverberation	112	71	16.88		
LPS	Real Reverberation	192	83	7.67	t (df =302) = 1.694 p = 0.091	No significant difference
	Simulated Reverberation	112	81	7.96		
SPEECH	Real Reverberation	192	84	7.55	t (df =302) = 1.779 p = 0.076	No significant difference
	Simulated Reverberation	112	82	10.81		

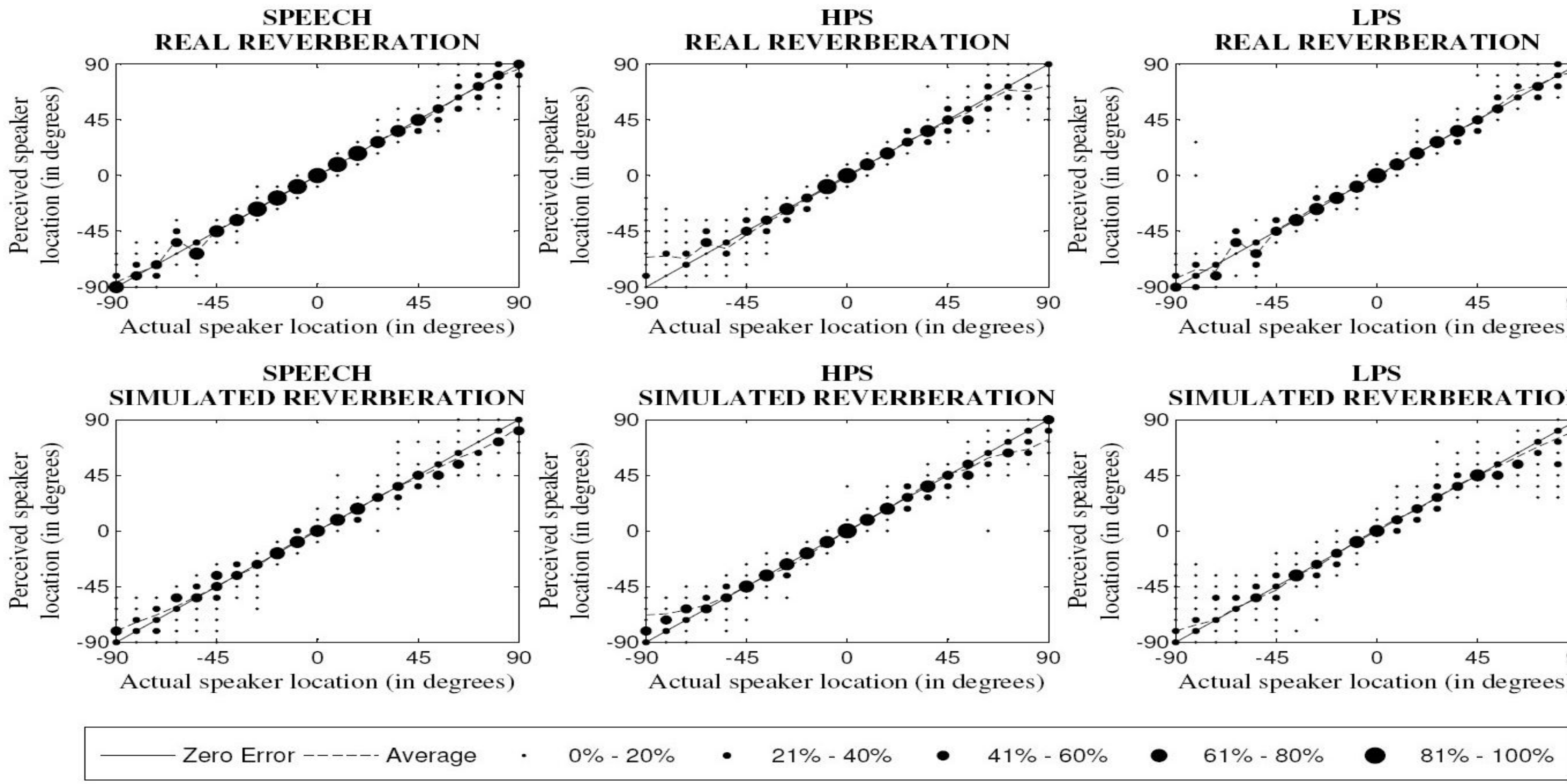


Figure 16 Localisation performance in the real and simulated reverberation. The Zero error line (bold line) and the average line (dashed line) are also plotted. Correct responses should lie on a diagonal line (zero error).

5.10.2 Analysis

A three-way ANOVA with repeated measure was performed with the dependent variable being the average absolute localisation error (log transformed) and the independent variables (i) the reverberation type (simulated and real reverberation), (ii) the speech stimuli (high-pass speech, low-pass speech and speech), and (iii) the speaker position (central, medial and lateral). The three-way ANOVA was performed with the main effects of the three factors, the two-way interactions between the factors and the three-way interaction.

Levene's test for equality of variances was significant, indicating that the assumption of variance homogeneity was violated; however, SPSS uses the regression approach to calculate ANOVA, therefore this was not an important problem.

Results from the three-way ANOVA revealed that the speech stimuli effect –i.e. high-pass speech, low-pass speech and speech – ($F_{(2, 26)} = 32.12$) was significant. Contrasts indicated that localisation performance for all three stimuli differ significantly from each other, where greater localisation errors were made for high-pass and low-pass speech (mean= 8.63, mean=6.9 respectively), when compared to speech stimulus (mean= 5.7). The main effect of the speaker position ($F_{(2, 26)} = 148.07$) was significant, where greater localisation errors were made for sound sources located medially and laterally (mean= 7.11, mean= 10.2) compared to centrally (mean= 3.8). The main effect of reverberation ($F_{(1, 13)} = 1.11$) was not significant, which implied that the localisation performance of normal hearing listeners in the simulated reverberation room was similar to their performance in the real reverberant room, as shown in Fig. 17.

The interaction effects between the speaker position and speech stimuli ($F_{(4, 52)} = 22.89$) were significant. Contrasts suggested that the increase in localisation errors of the high-pass speech and the low-pass speech, when compared to the speech stimuli, was significant when comparing the lateral to the central speaker positions.

Fig. 18 shows the greater difference amongst the speech stimuli at the lateral speaker position when compared to other speaker positions.

Based on these results, the average localisation error of the speech stimuli in the real reverberant room did not differ significantly from the simulated reverberant room.

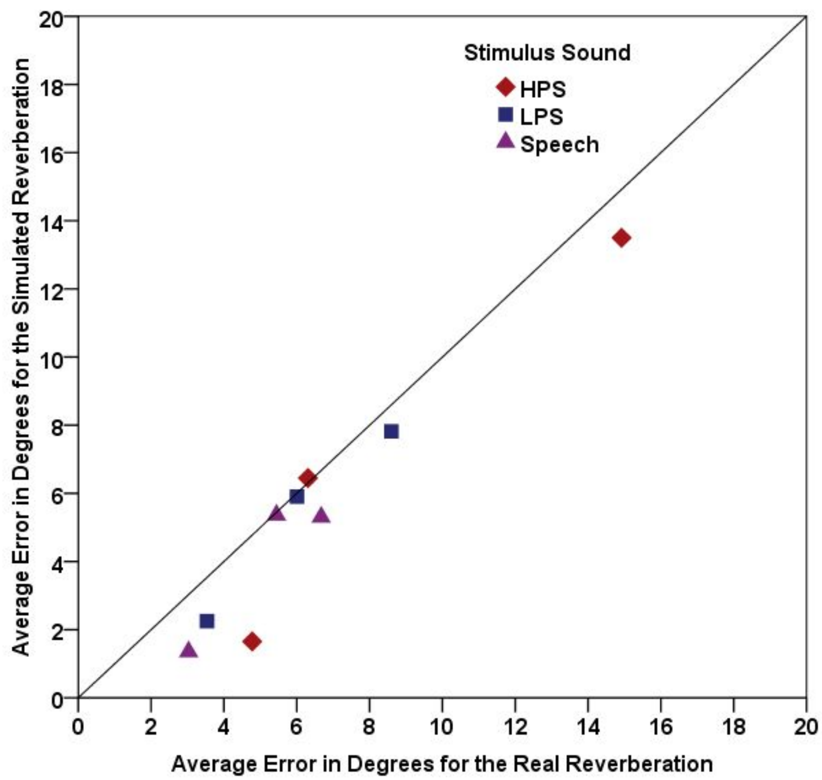
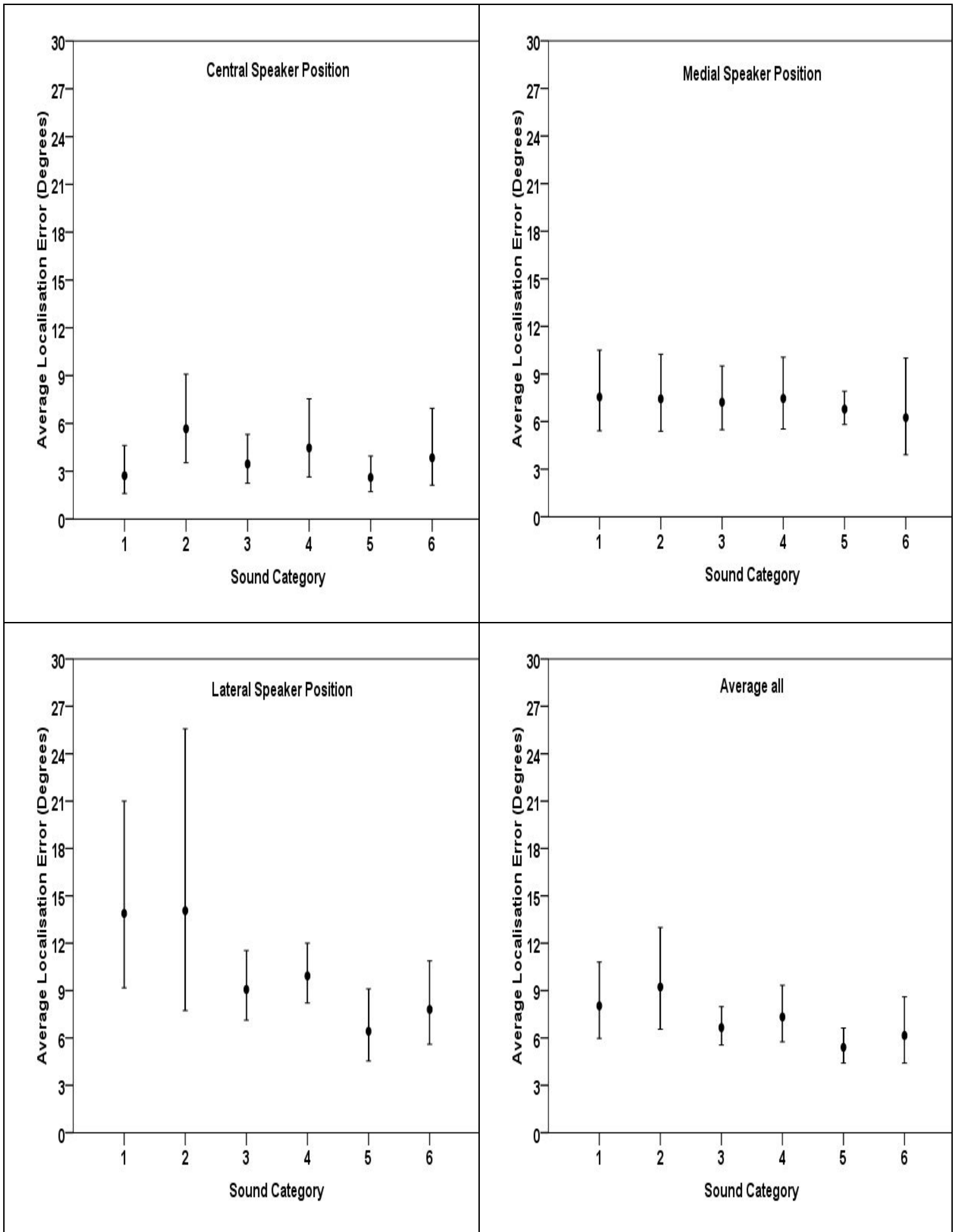


Figure 17 Scatter plot with regression line showing the relationship between the average localisation errors in real vs. simulated reverberation. The average localisation error is shown for three speaker positions – central, medial and lateral – for each stimulus.

Simulated vs. Real Reverberation



The numbers 1~6 in the above graphs represent the following categories of Sound:
1 High-pass speech signals in real reverberant condition
2 High-pass speech signals with simulated reverberation
3 Low-pass speech signals in real reverberant condition
4 Low-pass speech signals with simulated reverberation
5 Speech signals in real reverberant condition
6 Speech signals with simulated reverberation

Figure 18 The average localisation error for real and simulated stimuli (log transformed data). The error bars represent "Mean ±1 SD".

5.11 Discussion

5.11.1 Anechoic vs. real reverberation

The speech stimuli were easier to localise when compared to the pure tones in both anechoic and reverberant conditions. The results revealed significant differences in the localisation accuracy between the real reverberation and the anechoic conditions, but there was no significant difference in the localisation accuracy of the speech stimuli between the anechoic and the real reverberant conditions. As mentioned earlier, the abrupt onsets of the speech stimuli allow the use of the precedence effect in their localisation in both anechoic and reverberant environments (Giguere and Abel, 1993). Also, based on the findings of McFadden and Pasanen (1976) and Bernstein and Trahiotis (1985a), the modulations in the speech envelope provide an important cue for localisation; however, these may be suppressed by adding reverberation.

Among the speech stimuli, the low-pass speech was the most affected by reverberation (mean=6.7, mean=4.84) in the reverberant and anechoic conditions respectively. Possibly because (a) for the low-pass speech, participants relied mainly on the available ITD cues to localise the sound, and (b) two important elements that aid in localisation were missing – the spectral and the ILD cues.

The results were consistent with the findings of McManus (2008) regarding the increased errors, because of the missing ILD and spectral cues, in localising the low-pass pink noise in the anechoic environment. The results were also consistent with Cunningham's (2000) findings that the ITD error was slightly larger in the reverberant condition (70 ms) than in the anechoic condition (53 ms) and the mean ILD error in the reverberant condition was actually as small as the ITD error in the anechoic.

A significant difference between the reverberant and the anechoic conditions was found in the pure tone stimuli, however, the difference was only significant for the 500 Hz pure tone, and not for the 4 kHz pure tone, suggesting that reverberation disrupts the ITD cues more than the ILD and spectral cues. These results were consistent with both Cunningham's (2000) findings, where ITD errors were slightly larger in the reverberant condition (70 ms) than in the anechoic condition (53 ms), and with Larsen et al's (2008) conclusions that the onset/offset times of the signal have an effect on the direct to reverberant (D/R) JNDs. Larsen et al. (2008), concluded that D/R JNDs of 0 and 10 dB D/R remained the same for signals with short onset/offset times (10 ms), whereas, there was a 1 dB increase in JNDs when slow onset/offset signals (150 ms) were used.

5.11.2 The effects of stimulus type on localisation in the real reverberant room

Results from Experiment 1 showed that there is a statistically significant difference at $p < 0.05$ in the localisation performance between the speech stimuli (high-pass speech, low-pass speech and speech) and the pure tone stimuli (500 Hz and 4 kHz). The localisation accuracy for each of the speech stimuli for all speaker positions was significantly better than the localisation accuracy for the 500 Hz pure tone and the 4 KHz pure tones. Moreover, the highest localisation error occurred for the 4 kHz pure tone and the lowest error in localising the speakers occurred for the speech stimulus, (mean= 21.11, mean= 5.4) respectively.

There are a number of possible factors that make the speech stimulus easier to localise: (i) listeners hear the speech stimulus all the time, hence they localised it more accurately, (ii) speech is more natural in reverberant environments, whereas listening to other forms of stimuli is artificial in both reverberant and anechoic conditions (Flynn& Dowell, 2003), (iii) the speech envelope probably aids localisation, (iv) the abrupt onset/offset characteristics of the speech stimuli – 10 ms – rather than the gradual times of the pure tones – 40 ms – allows the precedence effect in localising the sound source to come into play (Giguere and Abel, 1993); also, because of the extra neural firing, abrupt onsets provide additional spectral cues and ITDs (Rakerd and Hartmann, 1985), (v) due to the periodic nature of the pure tones, phase leads and lags would have been impossible to differentiate, leading to ambiguity in the inter-aural phase cues, which would increase as the frequency content of the pure tones increases (Wightman and Kistler, 1993), (vi) pure tones only contain energy over a restricted frequency range, hence listeners would have been unable to compare the ILD cues across different frequency regions. Wightman and Kistler (1993) suggested that ILDs provide reliable indications of sound location if the listener compares the pattern of ILD across different frequency regions. Also, for frequencies below 1 kHz, ILDs become less effective in sound localisation (Middlebrooks and Green, 1991). However, the speech stimulus, which covers a wider bandwidth, produces ITD and ILD cues in each frequency region, as well as spectral cues in the high frequency bands (Wightman and Kistler, 1993).

There was a significant difference for the speech stimuli, at $p < 0.05$, in the localisation accuracy between the high-pass speech, low-pass speech and broadband speech stimulus. Detailed analysis revealed that the high-pass speech and the low-pass speech produced relatively similar results at the central and medial speaker positions; however, the low-pass speech was significantly easier to localise at the lateral speaker positions. The localisation accuracy of the broadband speech stimulus was better than the high-pass speech and low-pass speech, particularly, at the lateral speaker positions. The availability of all the localisation cues – ILDs, ITDs and spectral cues– is a possible explanation for the broadband speech stimulus being the easiest to localise. For the low-pass speech stimulus, ITD cues dominates and for the high-pass speech ILDs and spectral cues dominate.

These results are consistent with the findings of Wightman and Kistler (1993) and Su and Recanzone (2001), who concluded that broader bandwidth sounds are easier to localise than narrowband sounds.

For the pure tone stimuli, the localisation accuracy of the 4 kHz pure tone was lower than the localisation accuracy of the 500 Hz, for which there are two possible explanations: firstly, pure tones are periodic stimuli, therefore phase leads and lags would have been indistinguishable, making the inter-aural phase cues ambiguous for their localisation. However, at low frequencies, the confusion over which phase was leading or which was lagging is reduced, since the time period of the stimulus is longer than the maximum expected inter-aural phase difference. Consequently, as the frequency of the pure tone increases, the confusion over which phase leads and which phase lags also increases (Wightman and Kistler, 1993). The second possible explanation is the sound energy absorbed by air, since high frequency sounds are more likely to be absorbed by air than low frequencies. Therefore, more high frequency absorption occurs in larger rooms, hence the air acts as a low-pass filter (Larsen et.al, 2008; Gardner 1999 and Begault, 1992). These results are consistent with data collected in anechoic conditions, with the 4 kHz pure tone containing more localisation errors than the 500 Hz pure tone, since most absorbent materials in the anechoic rooms absorb more high than low frequencies (Larsen et al, 2008).

5.11.3 Real reverberation vs. simulated reverberation

Results from Experiment 1 revealed no significant difference in the localisation performance between the real reverberant room and the simulated reverberation used in the anechoic room. However, this was true for normal hearing listeners only who generally show little effect of reverberation.

5.12 Conclusion

The main reason behind Experiment 1 was to find out whether, by adding reverberation to the listening environment, it makes it more difficult to localise speech signals and pure tone stimuli.

The findings revealed no significant effect of reverberation on localising speech stimuli for participants with normal hearing. However, it was found that reverberation had a greater impact on localising pure tone stimuli (500 Hz and 4 kHz), with a significant increase in the average localisation error of the 500 Hz pure tone in the reverberant environment over the anechoic environment. These results were consistent with Hartmann (1983), who found that adding reverberation in the environment does affect the localisation of sounds without attack transients. Results also revealed that it was possible to simulate reverberation in the anechoic chamber, which gave indistinguishable results from the real reverberant chamber.

All the following experiments were performed using simulated reverberation in an anechoic environment since it made it possible to simulate and use different reverberant environments. Moreover, the use of simulated reverberation in the anechoic chamber creates a more controllable environment.

Chapter 6

Experiment 2: Comparison of Localisation Performances across Different Reverberant Environments

6.1 Introduction

Results from Experiment 1 in this thesis revealed no significant effect of reverberation on localising speech stimuli. There was a slight deterioration in performance between the reverberant and the anechoic conditions; not for all the participants and neither was it significant at $p<0.05$. To improve the generalisableness of the results, the effect of reverberation was further investigated by using different reverberant environments. In Experiment 2, different simulated reverberant environments were used on the grounds that it might reveal whether the reverberation time used in the previous experiment was either too large or too small to create an impact on localising speech stimuli. The aim of this experiment therefore was to answer the following question:

> Do reverberation characteristics – i.e. reverberation time, attack time and high frequency absorption time– have an effect on the ability to localise sound?

> The hypothesis was that a monotonic relationship would be discovered resulting in an increase in localisation errors with an increase in the reverberation time.

Eight subjects, who had been assessed as otologically normal, were recruited. Each underwent testing under five simulated reverberant conditions in the same anechoic chamber. In all these conditions, the same speech stimulus ‘where do I speak from’ was played; however, some reverberation was added in each condition to resemble five different environments. Thus, the independent variable, “sound”, included the five levels defined below:

Rev: 1→ no reverberation was added (control)

Rev: 2→ resembling the reverberation of a furnished living room

Rev: 3→ resembling the reverberation of a small club

Rev: 4→ resembling the reverberation of a medium concert hall

Rev: 5→ resembling the reverberation of a large auditorium

The reverberation characteristics of these environments are shown in Table 8

Table 8 the reverberation characteristics of the simulated environments

Parameters	Large auditorium	Medium concert hall	Small club	Furnished living room
Reverberation Time (s)	3.87	3.07	2.14	1.15
Clarity C50(3) (dB)	-2.2	-1.8	-1.9	-1.6
Attack time (ms)	154.1	106	72	22
High frequency absorption time (ms)	876	823	833	965
Perception (ms)	45	16	63	72
Wet (%)	16.7	31.5	46.3	52
Dry (%)	73.1	73.1	73.1	90

These parameters have been stated above since other researchers have used them in the literature and it is important to note them when comparison is required. However, it must be noted that these parameters have not been systematically varied in this study.

The reverberation time of a room – i.e. the time required for the sound energy in an enclosure to decrease by 60 dB after the source emission has stopped– was regarded as the main indicator of its acoustical properties. However, there are also other types of measurements that are needed in order to provide a complete evaluation of the acoustical properties of rooms – e.g. early/late energy ratios, lateral energy fractions and interaural cross correlation. In Table 8, 'Clarity' refers to the early to late sound energy ratio (BS EN ISO 3382-1:2009), which could either have been calculated by a 80 ms early time limit, or, as in the case of this experiment, a 50 ms time limit.

$$C_{50} = 10 \log\left(\frac{E_{0-50\ ms}}{F_{50-\infty\ ms}}\right)$$

C50 is dependent upon frequency and has been measured in Table 8 as the average of C50 values at frequency octave bands centred at 500 Hz and 1000 Hz. Clarity of the impulse response in this experiment was measured using Aurora program. The other reverberation parameters, shown in Table 8, are related to the software used in the simulation and have been explained in section 3.4 on p.57

6.2 Results

Fig. 19 shows the spread of the data in the five different simulated reverberant environments. For all listening environments, the average localisation errors of the listeners were either zero, or close to zero i.e. less than 10º. Additionally, the spread of the responses is similar in all five listening environments.

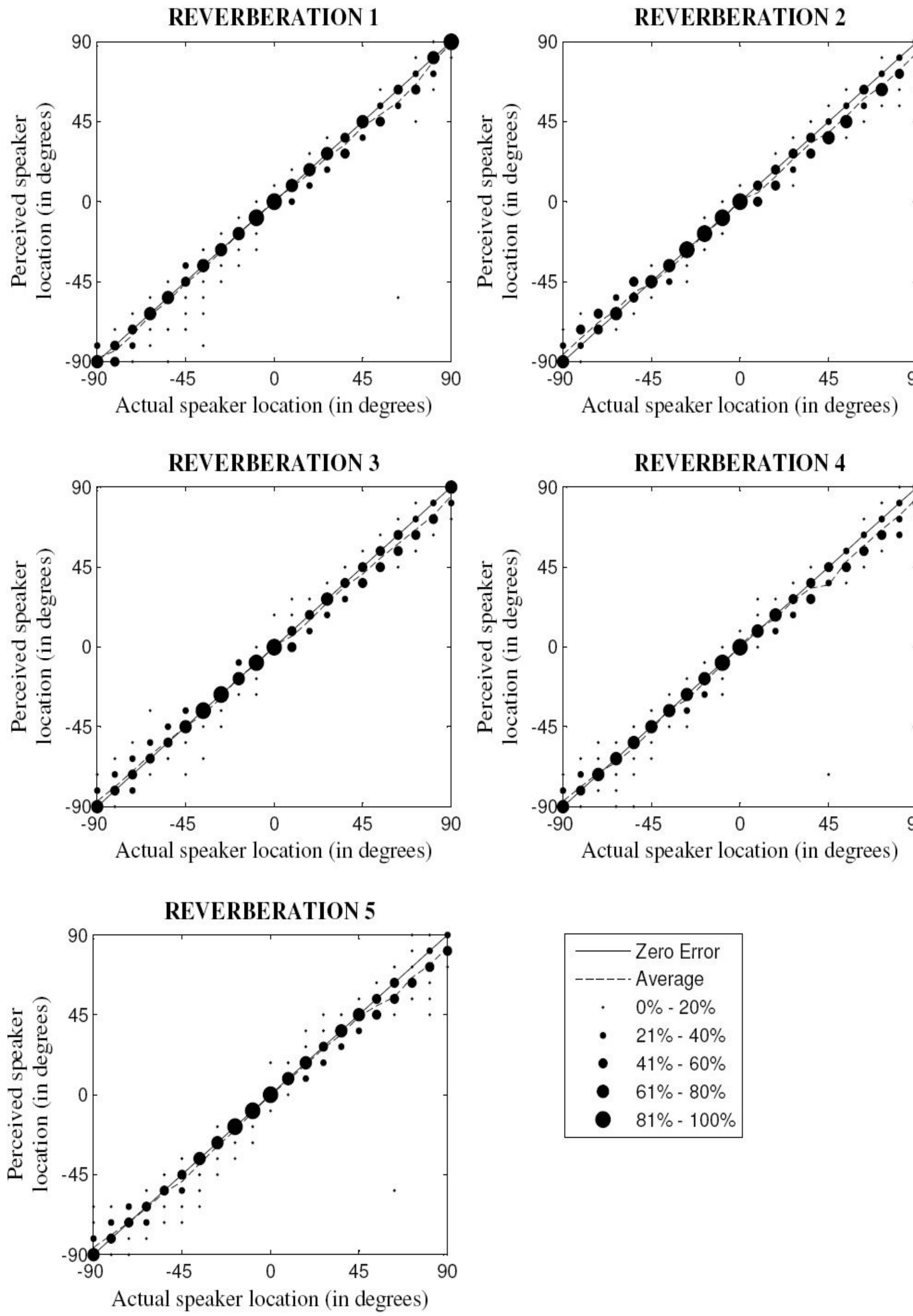
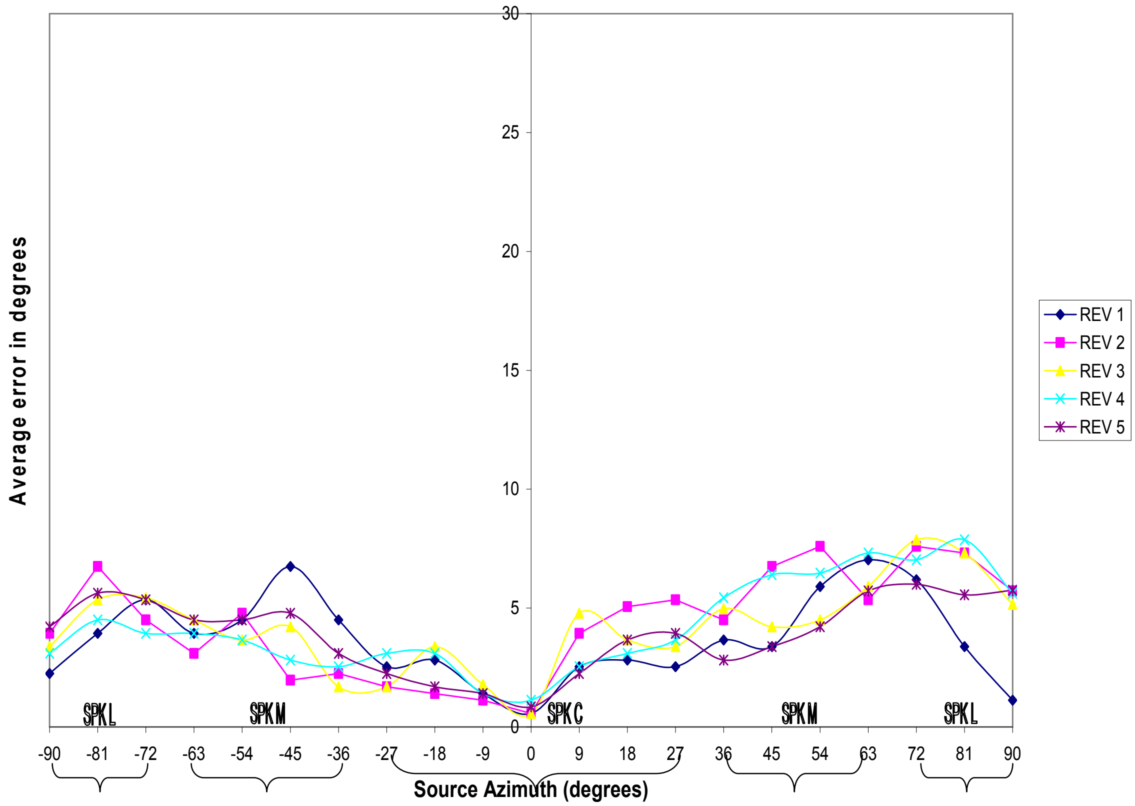


Figure 19 Localisation performance in different listening environments. The zero error line (bold line) and the average line (dashed line) are also plotted. Correct responses should lie on a diagonal line (zero error)

6.3 Analysis

Fig. 20 shows the average absolute localisation errors under the different listening environments. The graph shows that, for each speaker position, responses under the five different listening environments overlap and they are not widely spread from each other.



Rev: 1→ no reverberation was added (control)

Rev: 2→ resembling the reverberation of a furnished living room

Rev: 3→ resembling the reverberation of a small club

Rev: 4→ resembling the reverberation of a medium concert hall

Rev: 5→ resembling the reverberation of a large auditorium

Figure 20 The mean values of the errors in the localisation of a particular speaker for speech stimuli sound with five different simulated reverberant environments

A two-way repeated measures ANOVA was performed with the factors reverberation (Rev 1, Rev 2, Rev 3, Rev 4, Rev 5), and speaker location, (central, medial, and lateral), and the dependent variable was the average localisation error (log transformed). The main effect of the speaker location ($F_{(2, 14)}$ =13.53) was significant. The mean error at both medial (mean=5.1) and lateral (mean=6.1) speaker locations are significantly higher than the central speaker location (mean=2.8). There was no significant effect of reverberation ($F_{(4, 28)}$ =1.14). There was also no interaction between the speaker location and reverberation ($F_{(8, 56)}$ =1.69). Fig. 21 average localisation errors for the central, medial and lateral speaker positions as well as the localisation errors at all speaker positions averaged together.

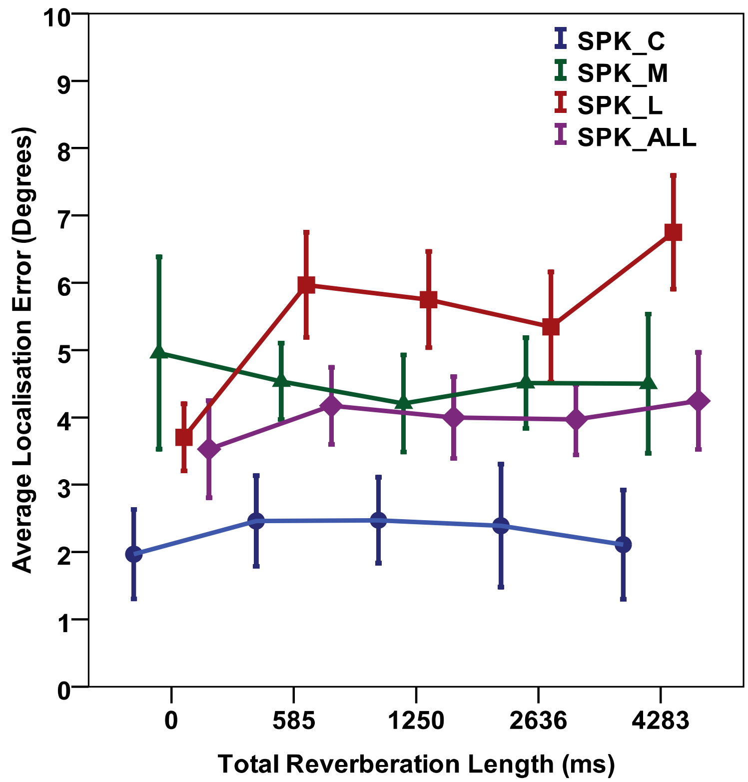


Figure 21 Means ±1 standard error in the localisation of speech stimulus at central, medial and lateral speaker positions as well as the localisation errors at all speaker positions averaged together (SPK-ALL).

6.4 Discussion

Results from Experiments 1 and 2 showed that adding reverberation did not seem to have an effect on the ability of normal hearing listeners to localise speech stimuli, even when longer reverberation times were used. This finding was consistent with Lutman and Payne's (2002) experiment, whereby the reverberant speech (reverberation time of 402 ms) was not found to be harder to localise than the non-reverberant speech, possibly due to the availability of the ILD, ITD and spectral cues in the broadband speech stimulus. According to Wightman and Kistler (1993), localisation performance improves as the stimulus bandwidth increase. Moreover, the speech stimulus contains abrupt onsets, which allows it to be localised by the precedence effect in the reverberant conditions (Rakerd and Hartman, 1985). Information at the onset of the sound where the direct sound is much larger than reflected sound is more reliable (Litovsky et al., 1999; Rakerd and Hartmann, 2004) than the later arriving acoustical information. If listeners concentrate only on the onset cues and ignore the ongoing part of a reverberant sound source, which contains both direct and reflected sound, they should localise sounds quite as well in the reverberant and anechoic environments (Devore et al., 2009)

Chapter 7

Experiment 3: Effect of the Signal Onset Time and Envelope on Sound Source Localisation

7.1 Introduction

Findings from Experiments 1 and 2 in this thesis revealed that reverberation had a significant negative effect on localising pure tone in the horizontal plane; however, no significant effect was found in localising the broadband stimuli. Furthermore, the pure tones were significantly more difficult to localise when compared to the speech stimulus in both the reverberant and anechoic conditions.

One possible explanation is the abrupt onset/offset times (10 ms) of the speech stimulus compared to the gradual onset/offset (40 ms) of the pure tones, which could result in more firing of neurons, hence providing better temporal inter-aural differences in the time of arrival of the envelope (Wightman and Kistler, 1993). Moreover, the precedence effect – i.e. the ability to localise sound sources by the first wave that arrives in the listener's ears – is thought to only be used in localising sounds with rapid onsets and transient sounds (Giguere and Abel, 1993).

Consequently, the aim of this experiment was to investigate the effect of envelope modulations and onset/offset times of the signal in localising the sound source, and whether the gradual onsets make it more difficult to localise them.

7.2 Hypotheses and Research Questions

1- **The effect of onset time**

Does increasing signal onset time create more localisation errors in the reverberant environment?

The hypothesis was that the average localisation errors will increase as the signal onset time increases.

2- The effect of envelope

Are pulsated (fluctuating) signals easier to localise when compared to continuous signals?

The hypothesis was that the average localisation errors of the pulsated signals will be lower than that of the continuous signals. Due to the pulsated nature of the signal, the envelope ITD cues might aid in localising the pulsated signals.

3- The effect of speaker location

Are lateral speaker positions harder to localise than central and medial speaker positions?

The hypothesis was that the average localisation errors of the lateral speaker positions will be greater than that of the central and medial speaker locations.

7.3 Subject selection

Eight normal hearing listeners, who were different from those who participated in Experiment 2, were recruited to participate in the experiment. For details of inclusion criteria, see Section 3.5 on p.59

7.4 Stimuli

Six signals were used:

1- Continuous pink noise with 60 ms onset time
2- Continuous pink noise with 30 ms onset time
3- Continuous pink noise with 5 ms onset time
4- Pulsated pink noise with 60 ms onset time
5- Pulsated pink noise with 30 ms onset time
6- Pulsated pink noise with 5 ms onset time

The pulsated signals consisted of four pulses of pink noise, with 150 ms pulse duration, 10 ms rise-fall times, and a 50 ms inter-pulse interval time. The envelope of the continuous signal consisted of one pulse lasting 1 s.

7.5 Reverberation

Simulated reverberation was created using Adobe audition with the parameters as shown in Table 9. Details of the parameters can be found in section 3.4 on p.57

Table 9 showing the parameter values used in Adobe Audition to simulate the reverberation of a large auditorium

Parameters	**Value**
Reverberation Time(s)	3.87
Clarity (C50) (dB)	- 2.2
Attack time (s)	154.1
High frequency absorption time	876
Perception	45
(%) Wet component	16.7
(%) Dry component	73.1

7.6 Equipment and Apparatus

The experimental setting was the same used in previous experiments in this thesis. See Section 3.2 on p.54

7.7 Test Design

To answer the above research questions, each of the participants underwent six different runs, using the six types of sounds as described in Section 7.4 above

7.8 Results

Fig. 22 shows the spread of the data (average absolute error) for all six stimuli. The average absolute localisation errors of the listeners were either zero, or close to zero – less than 10º – for both pulsated and continuous stimuli, suggesting that the used signals were easy to localise by the listeners.

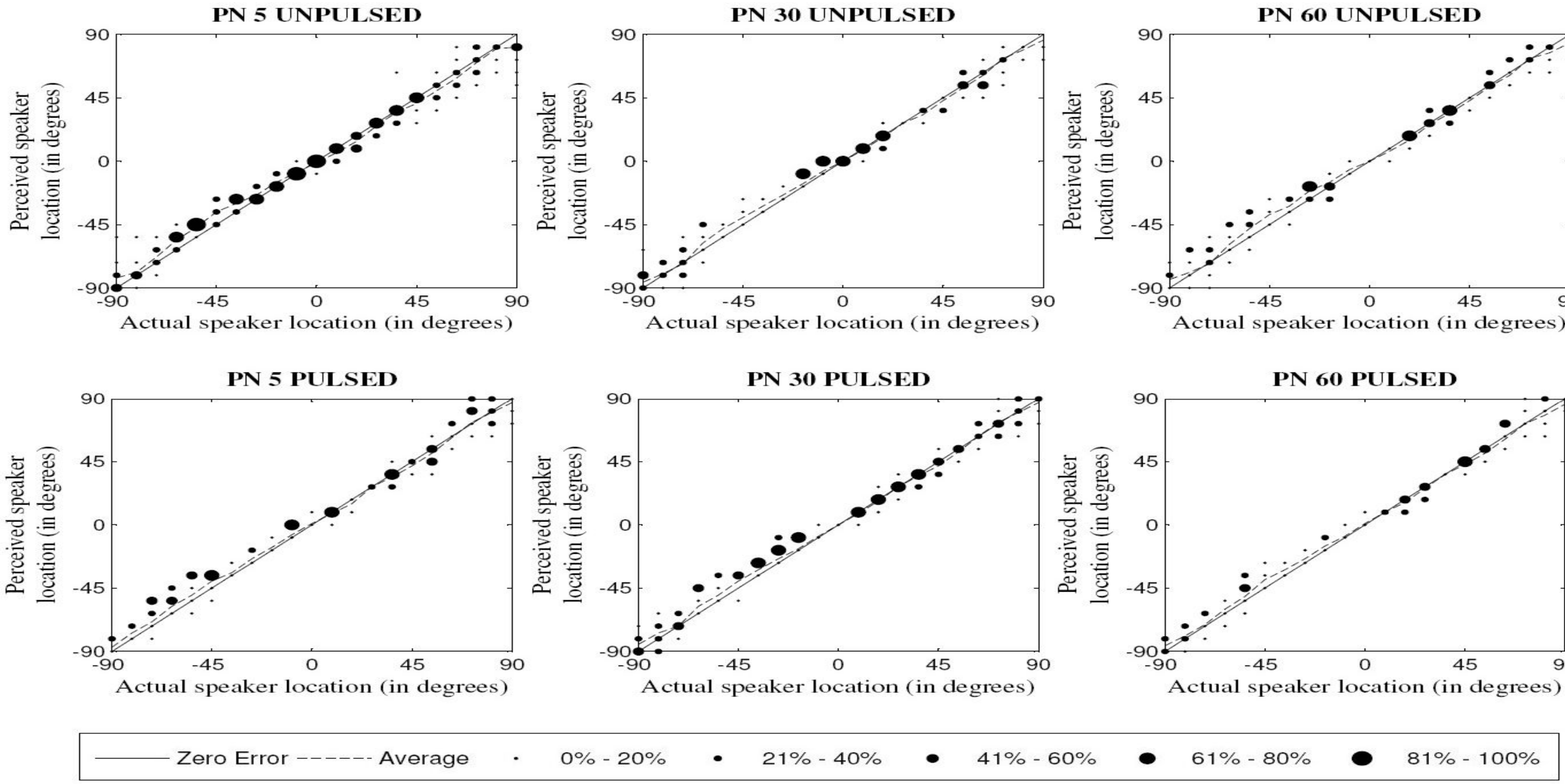


Figure 22 Localisation performance of the continuous and pulsated signals. The Zero error line (bold line) and the average line (dashed line) are also plotted. Correct responses should lie on a diagonal line (zero error)

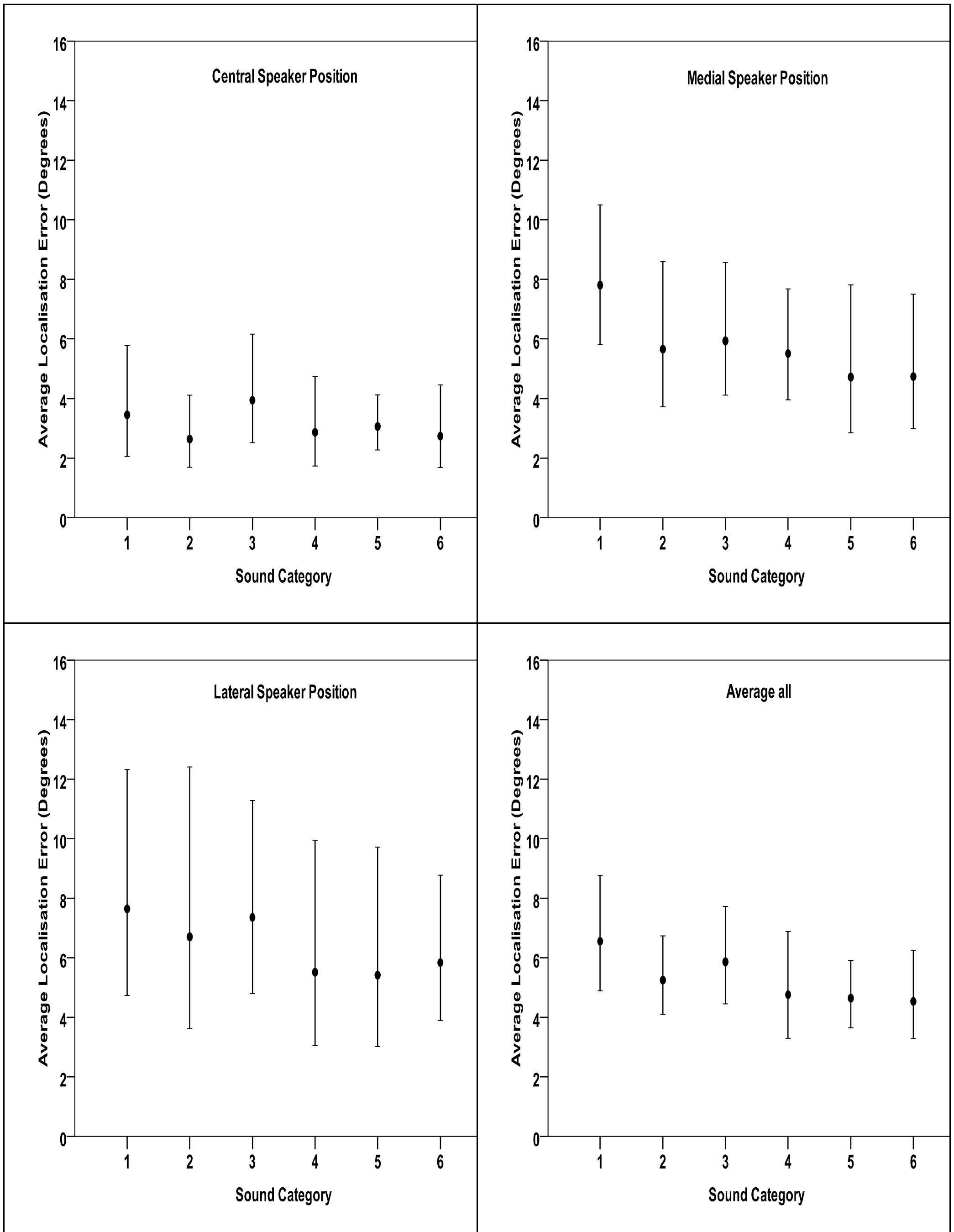
7.9 Analysis

A three-way ANOVA with repeated measures was performed, where the factors were the envelope of the pink noise (continuous and pulsated), onset time (5 ms, 30 ms, and 60 ms) and speaker location (central, medial and lateral). The average absolute error data, however, was not normally distributed, therefore it was log transformed. The results revealed a significant main effect of the speaker location ($F_{(2, 14)}$ = 15.14, p<0.001) and a significant main effect of the envelope ($F_{(1, 7)}$ = 12.5, p=0.01). However, there was no-significant main effect of the onset time ($F_{(2, 14)}$ = 3.16). Moreover, results showed a non-significant 'speaker by onset' interaction ($F_{(4, 28)}$ = 1.23), a non-significant 'speaker by envelope' interaction ($F_{(2, 14)}$ = 2.28), a non-significant 'onset by envelope' interaction ($F_{(2, 14)}$ = 2.57) and a non-significant 'speaker by onset by envelope' interaction ($F_{(4, 28)}$ = 0.64).

Simple contrasts of the main effect of the speaker location indicated that the average localisation errors for the lateral (mean=5.3) and medial (mean=5.6) speaker positions were greater than the localisation errors from the central speaker positions (mean=3.1).

The main effect of the envelope factor revealed that the localisation errors made at localising pink noise stimuli with the pulsated envelope (mean=4.64) were significantly lower than the localisation errors of the continuous pink noise stimuli with the un-pulsated envelope (mean=5.9). The results are graphically represented in Figs. 23 and 24.

The effect of signal onset time



Numbers 1~6 in the above graphs represent the following categories of Sound:
1 → continuous pink noise with 5 ms onset time
2 → continuous pink noise with 30 ms onset time
3 → continuous pink noise with 60 ms onset time
4 → pulsated pink noise with 5 ms onset time
5 → pulsated pink noise with 30 ms onset time
6 → pulsated pink noise with 60 ms onset time

Figure 23 Average localisation errors of the different sound categories. The error bars represent Mean±1

The effect of the signal envelope

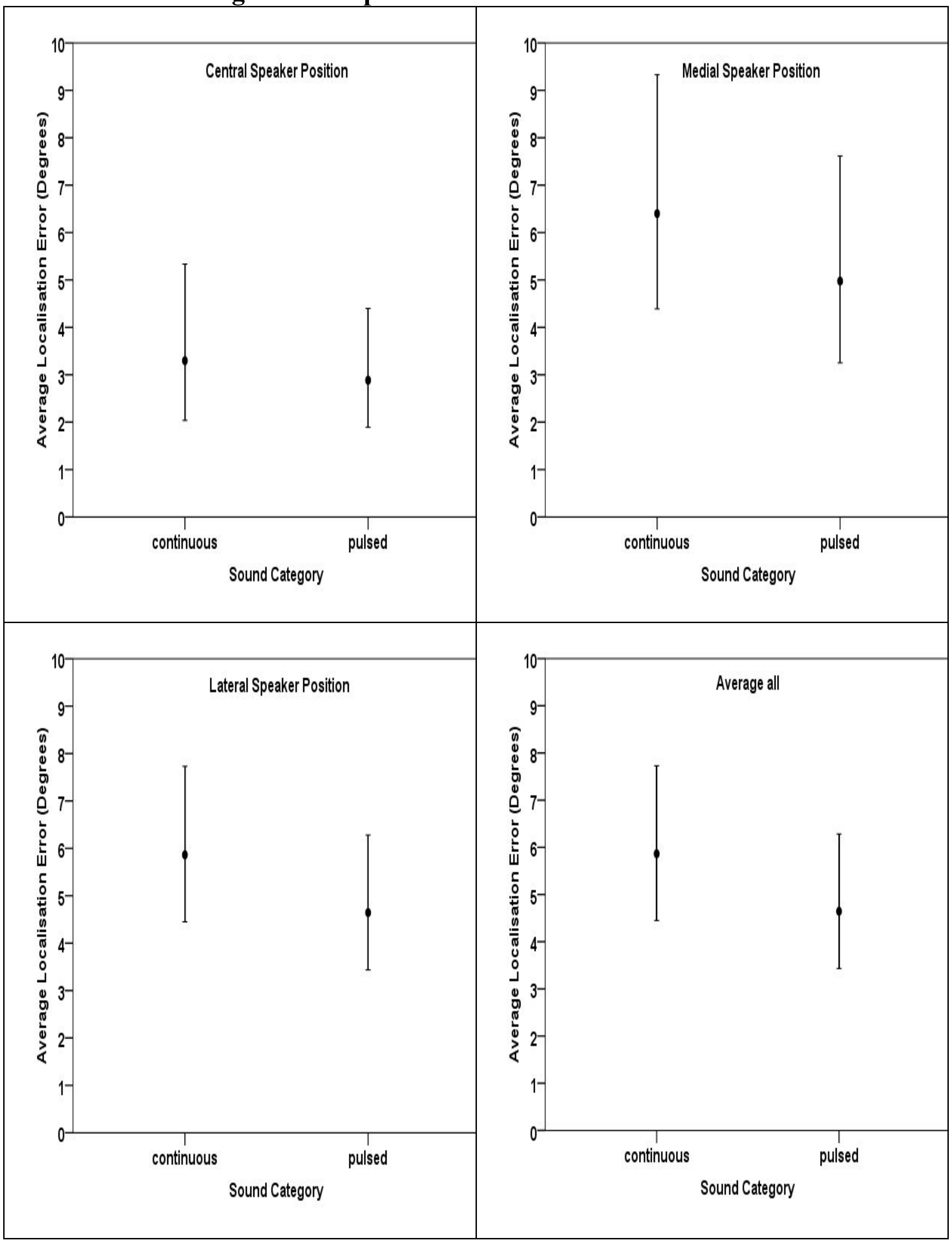


Figure 24 Average localisation errors of the continuous and pulsated signals. The error bars represent Mean ±1 SD.

7.10 Discussion

The results from this study revealed that increasing the onset time of a reverberant pink noise stimulus from 5 to 60 ms did not create more localisation errors. The ability of the participants to localise the pink noise stimulus with 30 or 60 ms onset times was as good as their ability to localise the pink noise with the 5 ms onset time.

A possible explanation for this is that the pink noise stimulus is a broadband stimulus that is easy to localise, since previous experiments revealed that participants had been very accurate in localising broadband stimuli under different reverberant conditions. Pink noise contains ITD and ILD cues at each frequency, as well as spectral cues at high frequencies, therefore reducing one of its features – rapid onsets – did not necessary reflect on the participant's ability to localise it. These results were consistent with Giguere and Abel (1993).

The results were inconsistent with the findings of Rakerd and Hartmann (1985), who found that the localisation of pure tones can be improved by using rapid onsets. However, in their experiment, pure tones stimuli were used instead of the pink noise that was used in Experiment 3.

Moreover, findings from Experiment 3 revealed that localisation of continuous pink noise signals was significantly worse than the localisation of pulsated signals. This suggests that ITDs in the modulations on the signal's envelope do aid in localisation, and that the envelope cues might be as important as the onset cues in the reverberant environments. The results were consistent with the findings of McFadden and Pasanen, 1976 and Bernstein and Trahiotis, 1985a.

7.11 Conclusion

The findings from Experiment 3 confirm the idea that envelope ITD cues contribute to the localisation abilities of pink noise stimulus in simulated reverberant environments. Varying the onset time of the pink noise stimulus however did not significantly affect the localisation performance of the listeners.

The goal of the next experiments was to understand how users of hearing aids utilise ILDs and ITDs cues in order to localise sounds in the presence of reverberation and to compare the results with and without hearing aids. A further goal was to compare the localisation performance of hearing impaired and normal hearing listeners in order to gain an insight into the auditory mechanism and how it localises speech in reverberant environments. This finding could be useful when devising and adjusting amplification devices to be used in semi-reverberant environments.

Chapter 8

Experiment 4 Effects of reverberation and amplification on the localisation abilities of hearing-aid wearers

8.1 Introduction

Binaural hearing yields advantages in many ways, such as better speech intelligibility in noise and better localisation performance. Localisation depends mainly on inter-aural timing and level differences – ITDs and ILDs – from the two ears. The ability to utilise these cues may be important not only for localising an oncoming sound, but also for understanding speech in background noise (Bronkhorst and Plomp, 1989).

The main complaint from hearing impaired listeners is that they have difficulty in understanding speech in the presence of background noise (Preves, 2000). Most of this group has sensorineural hearing loss, which results in reduced spectral, temporal and amplitude resolution and also a loss of audibility. Therefore, hearing impaired listeners require higher signal to noise ratio than normal hearing listeners, which can be achieved by using directional microphones and multi-microphone technology in digital hearing aids (Kuhnel and Checkly, 1999). The adaptive directional system reduces sensitivity to sounds coming from certain directions, and preserves sound coming from other directions. However, this feature may disrupt the inter-aural time and level difference cues, which are important for horizontal sound localisation (Lutman and Payne, 2002). Moreover, hearing impaired listeners already have large ITD thresholds, which make it more difficult for them to localise (Gabriel et al., 1992).

Most previous research on localisation performance has used stimuli that include both ITD and ILD cues, which makes it difficult to determine the weight of ITD and ILD cues separately to localisation performance (Gardner and Gardner, 1973; Butler, 1990; Byrne et al, 1992; Noble et al, 1994). Some researchers studied the effect of ILDs and ITDs separately by using high-pass vs. low-pass stimuli; however, the experiments were performed on normal hearing listeners (Wightman and Kistler, 1992).

Van den Bogaert et al. (2005), who studied the effect of bilateral hearing-aids on the horizontal plane localisation, used different types of hearing-aids to compare omni-directional and directional microphones. The stimuli they used were third-octave high frequency noise-band, third-octave low frequency noise-band and a broadband noise. Results revealed that, in the aided trials, more deterioration in localising the three signals was found than in the unaided trials, which was significant for the broadband stimulus, suggesting an adverse effect of hearing aids on the localisation cues – ILDs and ITDs.

McManus (2008), who investigated the effect of amplification on the horizontal localisation when testing 18 bilateral hearing aid users, used pink noise, low-pass pink noise filtered at 1 kHz and high-pass pink noise filtered at 4 kHz as stimuli. Participants were tested with and without hearing aids. In McManus study, the same hearing aid make and model was used; and localisation was measured in an anechoic room with an array of 21 loudspeakers at intervals of 9°. Results showed a significant increase in the localisation error for the high-pass pink noise in the aided conditions than the unaided conditions, although no significant effect of aiding was shown in localising the pink noise and the low-pass pink noise stimuli. However, the experiment lacked information on the performance of hearing aid listeners under reverberant environments.

In the Experiment 4 some of the limitations of the McManus study were addressed, in that: (a) more participants were recruited – 28 rather than 18– thereby enhancing the detection of the amplification effect, (b) it was conducted in both anechoic and simulated reverberant environments, (c) in addition to broadband noise signals, a speech stimulus was used, whereby a one-second speech sentence "where do I speak from", was played to determine the effect of the envelope structure, giving the results a high ecological validity, and (d) compensation was made for the effect of audibility by presenting stimuli at equal sensation levels in both aided and unaided conditions, whereas McManus had delivered stimuli at the same intensity in both aided and unaided conditions.

The results from both this study and the McManus (2008) study may, therefore, be considered to be complementary, which could lead to a clarification of the effects of hearing aids on ILD and ITD cues in both anechoic and reverberant environments.

The aim Experiment 4 was to determine the effect of hearing aids and reverberation on ILD and ITD cues. To this end, the localisation abilities of bilateral Siemens Prisma 2 Pro hearing aid wearers were assessed as a function of frequency content – high vs. low – under different listening environments and these were compared with and without the hearing aids. It was assumed that if poor aided localisation is mainly due to distortions of ILD cues, then aided performance for high frequency noise bands will be worse than for low frequency noise bands, pink noise and speech, because high frequency signals are localised mainly by ILD cues. However, if poor aided localisation is due to distortions of ITDs, then aided localisation for the low frequency noise bands would be worse. If aided localisation proves to be worse for high-pass and low-pass signals, then both ITD and ILD cues would be distorted by amplification. The same concept applies in determining the effect of reverberation on ILD and ITD cues independently.

8.2 Research Question and Hypotheses

The aim of Experiment 4 was to answer the following questions:

1. What effect does adding reverberation in an environment have on the horizontal localisation of bilateral hearing aid users?

   The hypothesis was that the average localisation errors for the hearing impaired participants will be greater in the reverberant environment than in the anechoic conditions in both aided and unaided trials.

2. How do digital hearing aids affect the ILD and ITD cues used in localising a sound source in the horizontal plane?

The hypothesis was that the average error in localisation will be greater in the aided trials when compared to the unaided trials. This adverse effect of aiding is assumed to be there in both the anechoic and simulated reverberant environments.

3. How does stimuli type – the availability of ITD and ILD cues – affect localisation abilities in the horizontal plane?

   The hypothesis was that the average error in localisation for all the stimuli types under all experimental conditions – i.e. aided, unaided, anechoic and reverberation– will be the least for stimuli containing both ILD and ITD cues – i.e. speech and pink noise.

4. How does the location of the sound source relative to the listener (central, medial or lateral) affect localisation abilities in the horizontal plane?

   The hypothesis was that the average error in localisation under all experimental conditions – i.e. aided, unaided, anechoic and reverberation– will the highest for lateral speaker positions.

5. What is the localisation ability of hearing aid wearers compared to normal hearing listeners?

   The hypothesis was that the average error in localisation will be greater for the hearing impaired listeners, aided and unaided, compared to normal hearing listeners.

8.3 Hearing Aid Wearers

28 hearing aid users, aged between 55 and 85, were recruited to take part in the experiment, giving a statistical power of 98%, when an effect size of 0.7 was used (based on the results of Van den Bogaert, 2006). The participants were experienced users, of six months and more, of the Siemens Prisma 2 Pro hearing aid. Hearing aids were set into ‘first fit’ in which omnidirectional microphone configuration was

used. All participants were diagnosed with bilateral symmetrical high frequency hearing loss, and they had no other complicating conditions, such as conductive hearing loss or tinnitus. The study was granted approval by the Southampton and South West Hampshire Local Research Ethics Committee. The participants were recruited through the Audiology Department at the Royal South Hants Hospital in Southampton. Description of the Siemens Prisma 2 Pro hearing aid and its parameters are shown in section 4.2 on p.63

Exclusion Criteria:

- asymmetrical hearing – difference between the average thresholds of 0.5, 1, 2 and 4 KHz is greater than 20 dB
- conductive hearing loss in either ear
- tinnitus
- ear infections

8.4 Stimuli

Details of the stimuli are shown in Section 3.3 on page 55

8.5 Reverberation

Reverberation was created using Adobe Audition to simulate the reverberation of a large auditorium. Details can be found in Section 7.5 on p.111

8.6 Test Procedure

See Section 3.9 on p.61

8.7 Design

To answer the research questions, localisation abilities of the 28 hearing aid users were assessed with, and without, their hearing aids, under simulated reverberant and anechoic conditions. To rule out the effect of audibility, tests were carried out with, and without, hearing aids at equal sensation levels. The audibility of the stimuli was adjusted until the participant confirmed that he/she was receiving equal sensation levels with and without amplification. All hearing aids were on the Omni-directional microphone configuration.

Therefore, there were three main variables:

- stimulus type – high-pass pink noise filtered at 4 kHz, low-pass pink noise filtered at 1 kHz, pink noise and speech
- simulated reverberation vs. anechoic conditions
- aided vs. un-aided

Each participant underwent 16 runs (4x 2x 2 conditions) as shown in Table 10

Table 10 the 16 test runs that each participant underwent

Stimulus	**Anechoic**		**Simulated reverberation**	
	Unaided	**Aided**	**Unaided**	**Aided**
High-pass noise	✓	✓	✓	✓
Low-pass noise	✓	✓	✓	✓
Broad band noise	✓	✓	✓	✓
Speech	✓	✓	✓	✓

Terminology

The following terminology is used throughout this chapter:

Amplification: refers to aided or unaided

Reverberation Conditions: refers to the anechoic or simulated reverberant environments

Experimental Conditions: refers to all tested conditions, i.e. anechoic or simulated reverberant, aided or unaided

8.8 Results

The spread of the data (average absolute localisation error) for all experimental conditions are shown in scattergrams (Figs. 25~28). Fig. 25 shows that in the anechoic chamber and when the listeners were not wearing their hearing aids, the average localisation error was either zero or close to zero – less than 10º – for all stimuli, i.e. high-pass pink noise, low-pass pink noise, pink noise and speech. Greater localisation errors were made when localising the high-pass pink noise when compared to the other stimuli.

Fig. 26 shows the localisation performances of the hearing impaired listeners in the simulated reverberant environment, and without wearing their hearing aids. For the high-pass pink noise, speech and pink noise stimuli, the average localisation errors tended to shift away from the zero error line at the lateral speaker positions (i.e. up to 20º error). For the low-pass pink noise, the average localisation errors deviated from the zero error line by up to 35º at the lateral speaker positions.

Fig. 27 shows the average localisation errors for the hearing impaired listeners when they were wearing their hearing aids in the anechoic environment. For the low-pass pink noise, speech and pink noise stimuli, the average localisation errors were mainly zero except for the two lateral speaker positions were it can go up to 20º. As for the anechoic-aided conditions, the average localisation errors for the high-pass pink noise were the most deviated from the zero error line (i.e. up to 30º error at the lateral speaker positions).

Fig. 28 shows the average localisation errors for the hearing impaired listeners when they were wearing their hearing aids in the simulated reverberant environment. For all four stimuli, the responses were more scattered when compared to the previous listening conditions suggesting that the localisation task was harder in the aided simulated reverberant condition. For the pink noise and speech stimuli, the average line started to deviate from the zero line at speakers _+45º to reach up to 35º average error at +_90º. The average error line for both the high-pass pink noise and the low-pass pink noise deviated from the zero line for both medial and lateral speaker positions – i.e. maximum localisation error of 20º at medial speakers and up to 30º at lateral speakers.

For all stimuli, localisation errors on the right of the midline was more or less the same in magnitude, but opposite in sign, to the error on the left of the midline

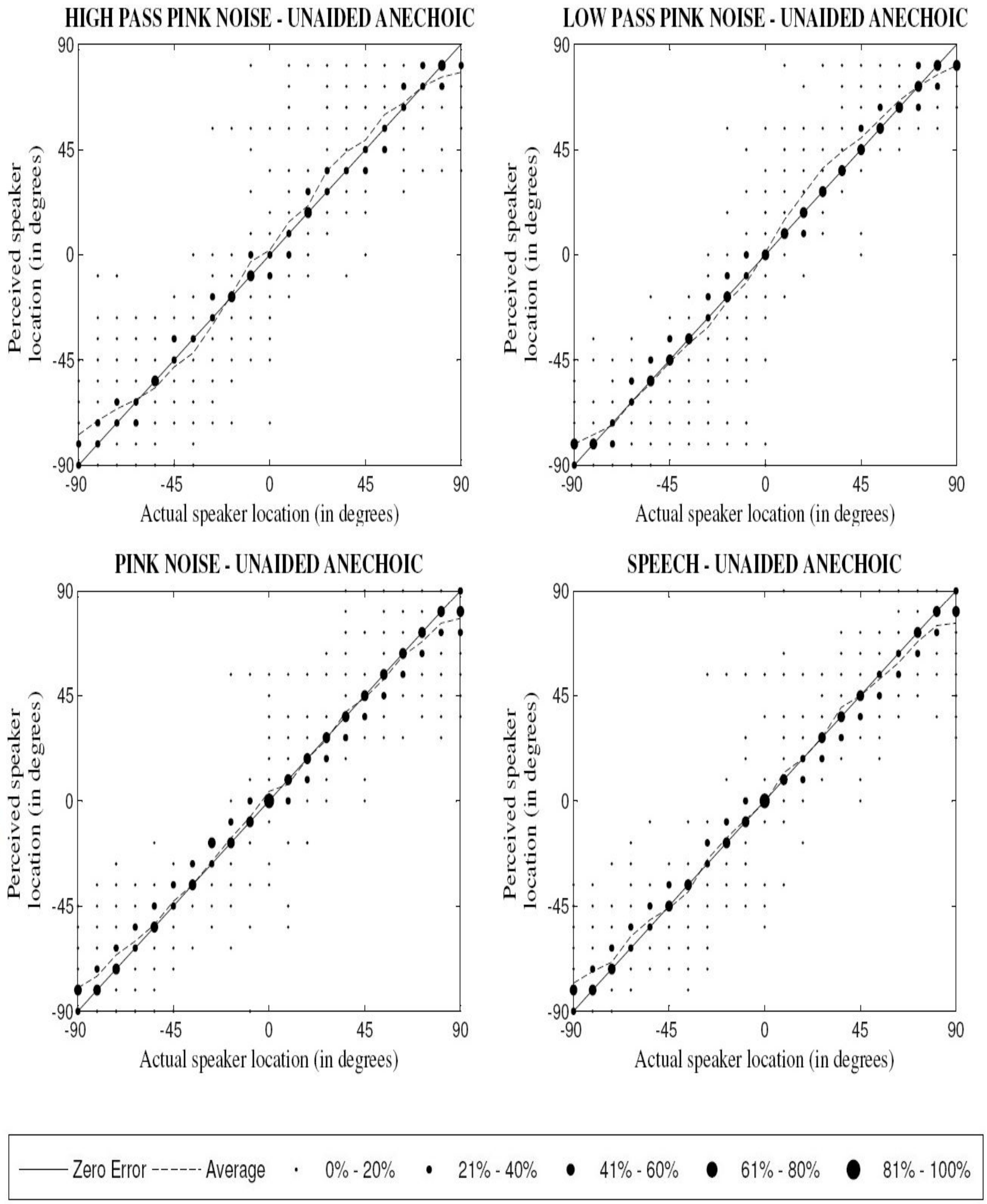


Figure 25 Localisation performance in the unaided anechoic condition. The Zero error line (bold line) and the average line (dashed line) are also plotted. Correct responses should lie on a diagonal line (zero error).

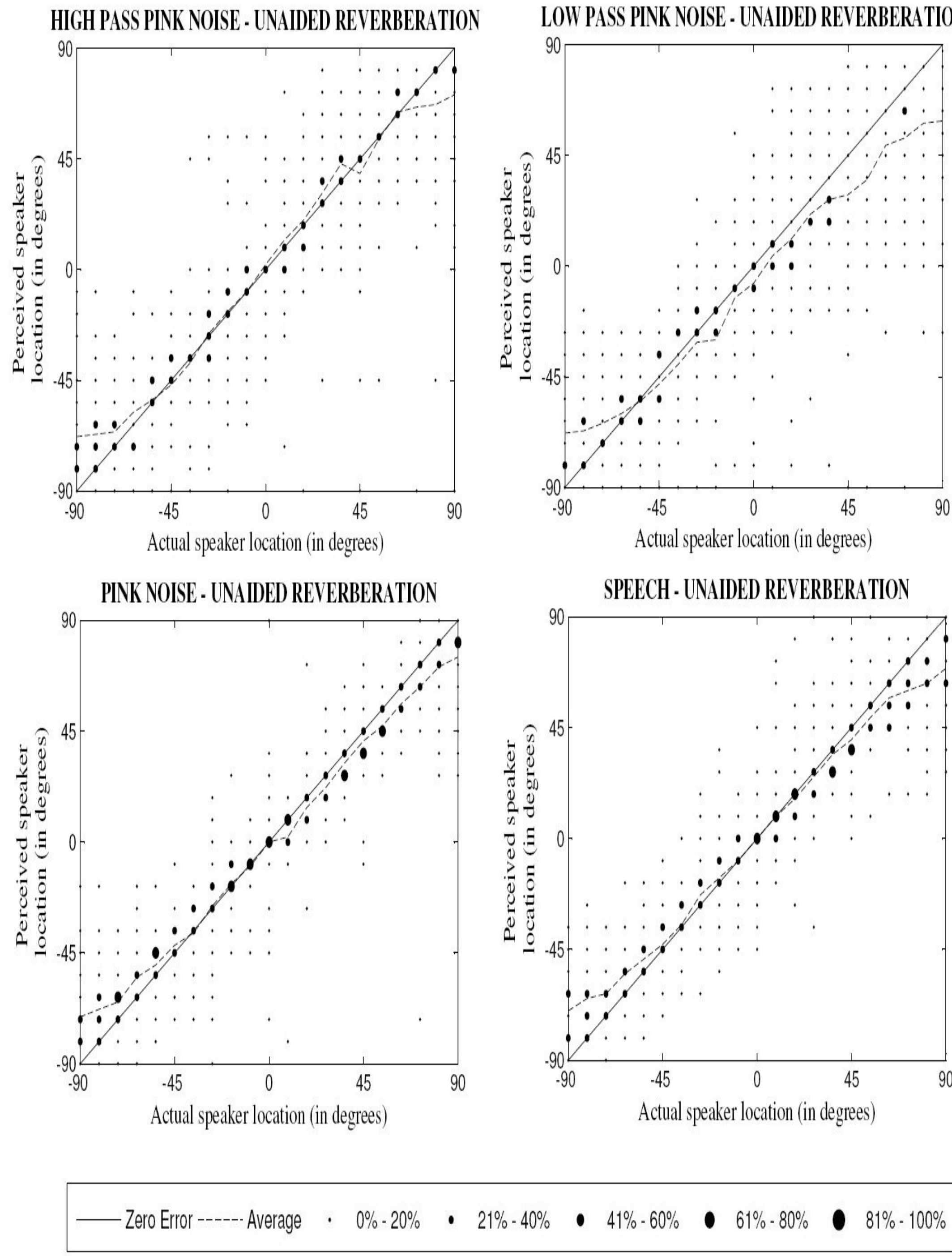


Figure 26 Localisation performance in the unaided simulated reverberant condition. The Zero error line (bold line) and the average line (dashed line) are also plotted. Correct responses should lie on a diagonal line (zero error)

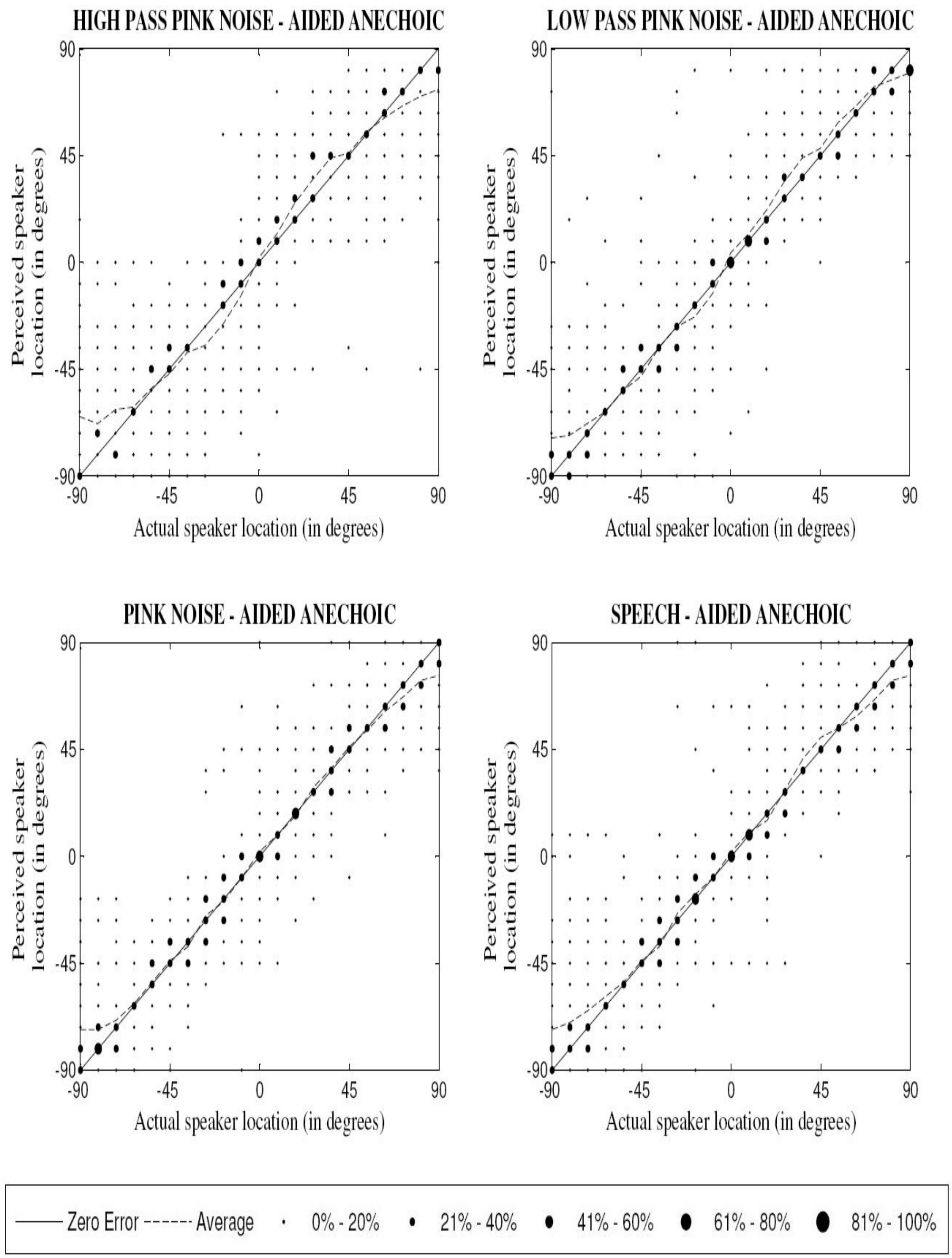


Figure 27 Localisation performance in the aided anechoic condition. The Zero error line (bold line) and the average line (dashed line) are also plotted. Correct responses should lie on a diagonal line (zero error).

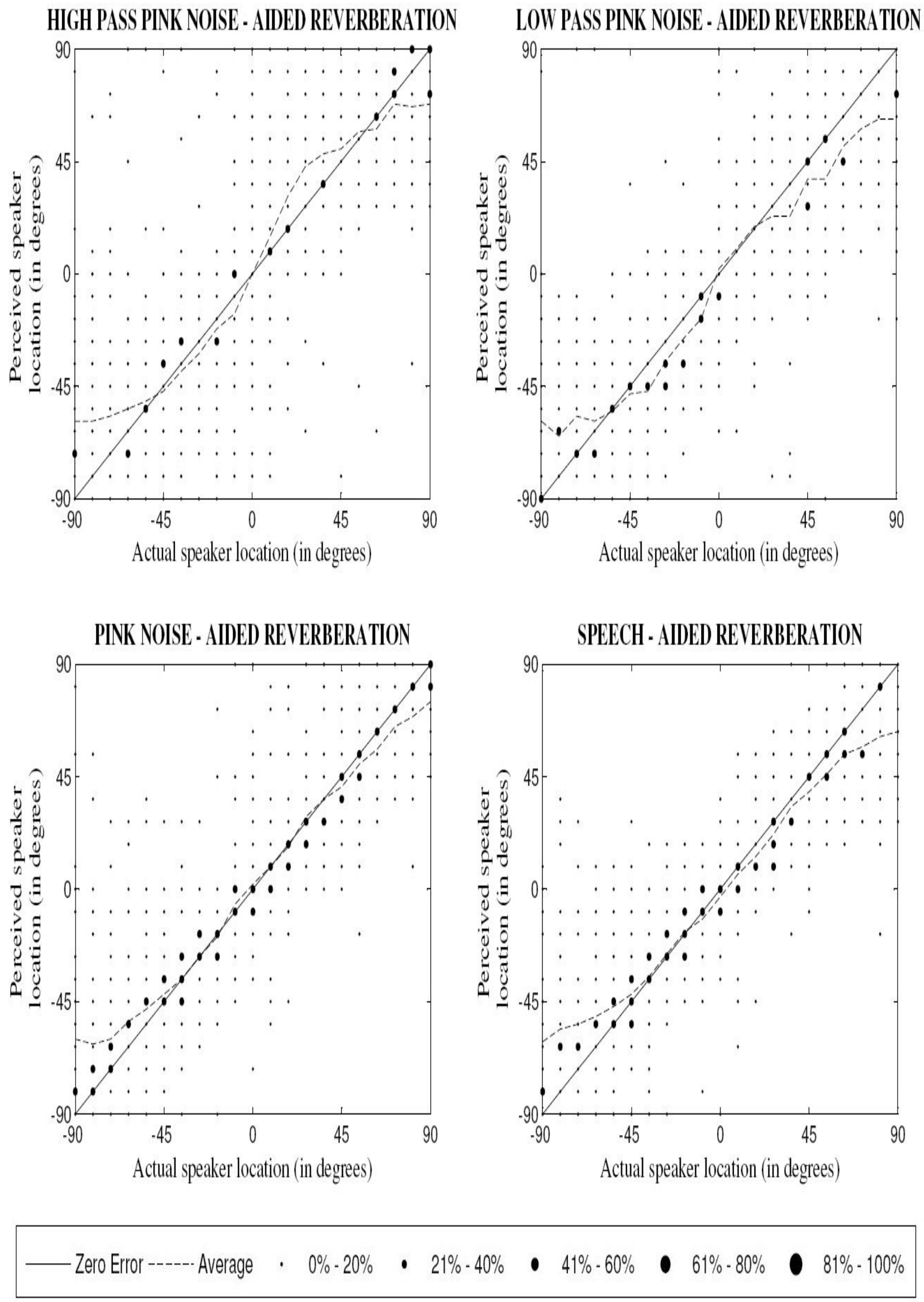
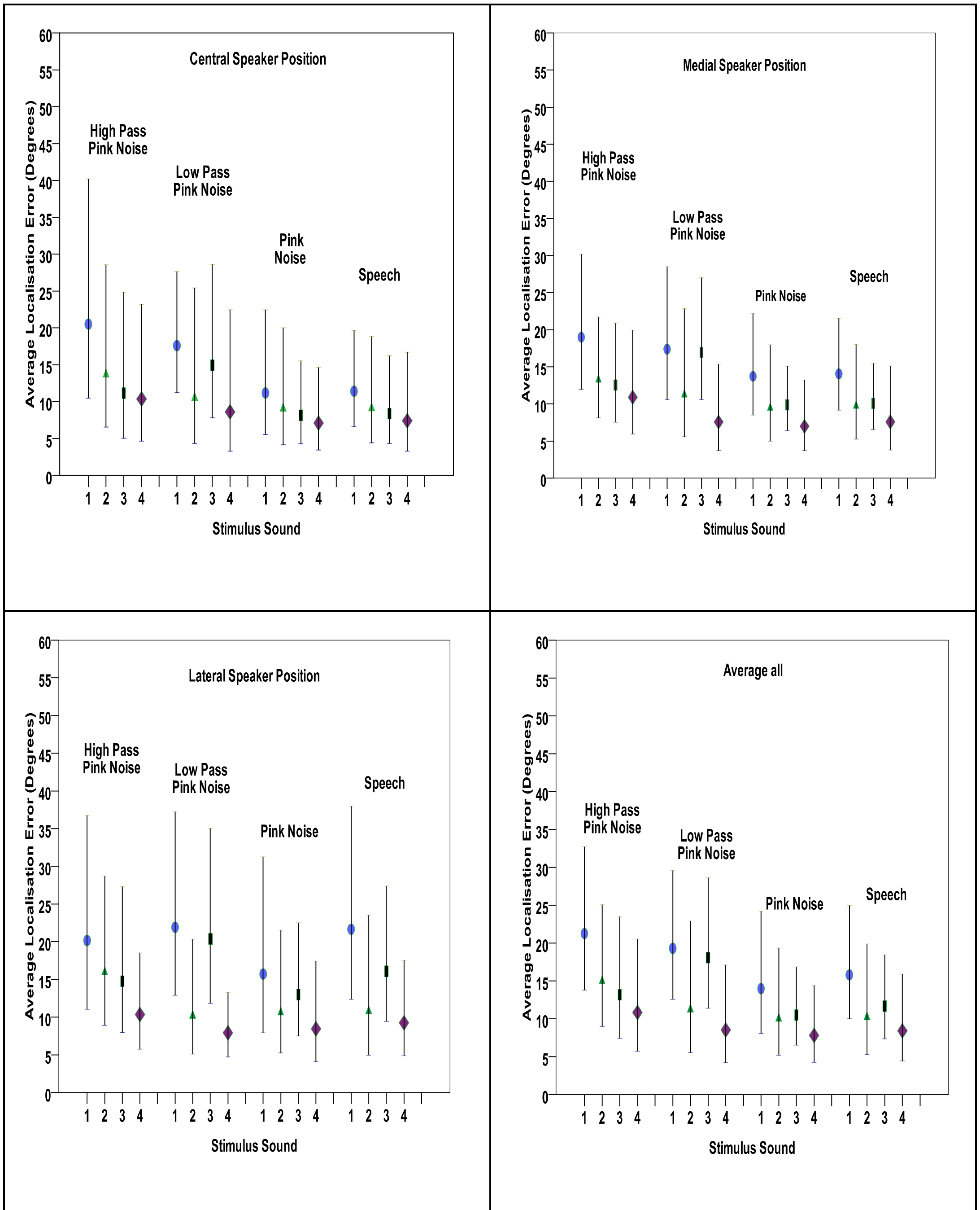


Figure 28 Localisation performance in the aided simulated reverberant condition. The Zero error line (bold line) and the average line (dashed line) are also plotted. Correct responses should lie on a diagonal line (zero error).

8.9 Data Analysis of the hearing aid wearers

A three-way repeated measures ANOVA was performed, where the factors were stimuli (i.e. pink noise, low-pass pink noise, high-pass pink noise and speech); amplification (i.e. with hearing aid, without hearing aid), and reverberation condition (i.e. anechoic, simulated reverberation). The dependent variable was the mean localisation error at all speaker positions averaged together (log transformed). Fig. 29 shows the distribution of the localisation error under different experimental conditions, i.e. by taking into consideration 'simulated reverberation or anechoic' conditions and 'aided or unaided’ conditions.



The numbers represent the following Sound Categories:

1 **Simulated reverberation (aided)**

2 **Anechoic (aided)**

3 **Simulated reverberation (unaided)**

4 **Anechoic (unaided)**

Figure 29 Average localising errors for all the stimuli types under all experimental conditions. The error bars represent "Mean±1 SD".

The repeated measures ANOVA results indicated that all the three main effects; stimuli ($F_{(3, 81)} = 15.19$), amplification ($F_{(1, 27)} = 21.67$) and reverberation condition ($F_{(1, 27)} = 56.99$) were significant at $p<0.05$. The results also indicated that the mean localisation error for different speech stimuli was highest for high-pass pink noise (mean=14.63) and lowest for pink noise (mean=10.36). Moreover, contrasts indicated that the localisation accuracy of the pink noise stimulus differed significantly from all three other stimuli (i.e. low-pass pink noise, high-pass pink noise, and speech). There was also a significant difference between the speech stimulus and all other stimuli. No significant difference was found between high-pass and low-pass pink noise (Fig. 30).

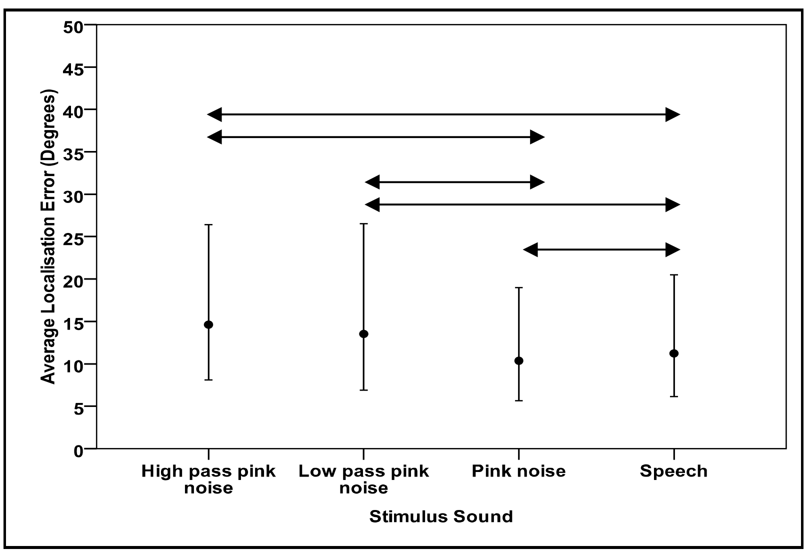


Figure 30 Average localisation errors for all speakers' positions averaged over both reverberation and amplification conditions (log transformed data). The error bars represent "Mean ±1 SD". The horizontal arrows represent significant difference at $p<0.05$

The significant effect of amplification ($F_{(1, 27)} = 21.67$) revealed that the localisation errors were greater in the aided conditions when compared to the unaided conditions (mean=14.12, mean=10.8) respectively (Fig. 31). Similarly, the significant effect of reverberation ($F_{(1, 27)} = 56.9$) indicated that the localisation errors were greater in the simulated reverberant environment when compared to the anechoic environment (mean=15.08, mean=10.06) respectively (Fig. 32).

The overall effect of amplification

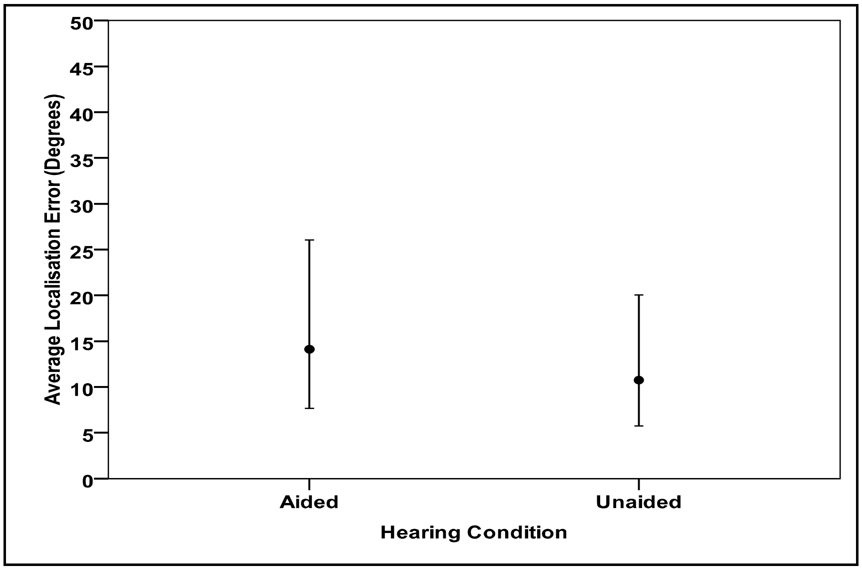


Figure 31 Average localisation errors for all stimuli sounds together, for all speakers' position together in both reverberation conditions (log transformed data). The error bars represent "Mean ±1 SD". The difference was significant

The overall effect of simulated reverberation

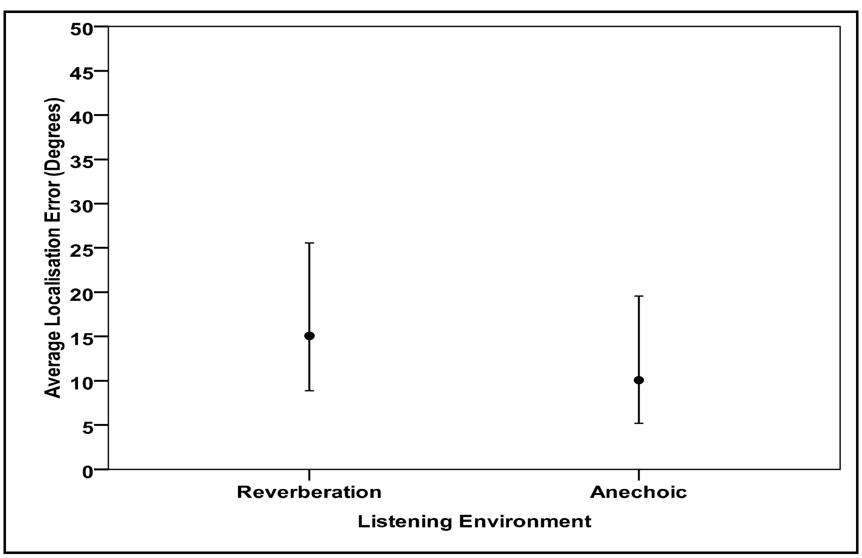


Figure 32 Average localisation errors for all stimuli sounds together, for all speakers' position together in both amplification conditions (log transformed data). The error bars represent "Mean ±1 SD". The difference was significant.

The interaction between stimuli type and reverberation condition ($F_{(1.9,\ 53.6)} = 5.08$) was significant. Simple contrasts, where all signal types were compared to pink noise, indicated that localisation performance in simulated reverberation, when compared to anechoic for speech and high-pass pink noise did not differ significantly from the localisation performance to pink noise. However, the increase in the localisation error for the low-pass pink noise in simulated reverberation, when compared to anechoic (mean=19.3, mean=11.3 respectively), was significantly different from the localisation performance to pink noise stimulus. The interaction between stimuli and amplification was not significant; however, the high-pass pink noise was most affected by aiding (mean=18.14 mean=12.03) in the aided and unaided conditions respectively. There were no other significant interactions. Graphical representation of the effect of amplification on each stimulus is shown in Fig. 33.

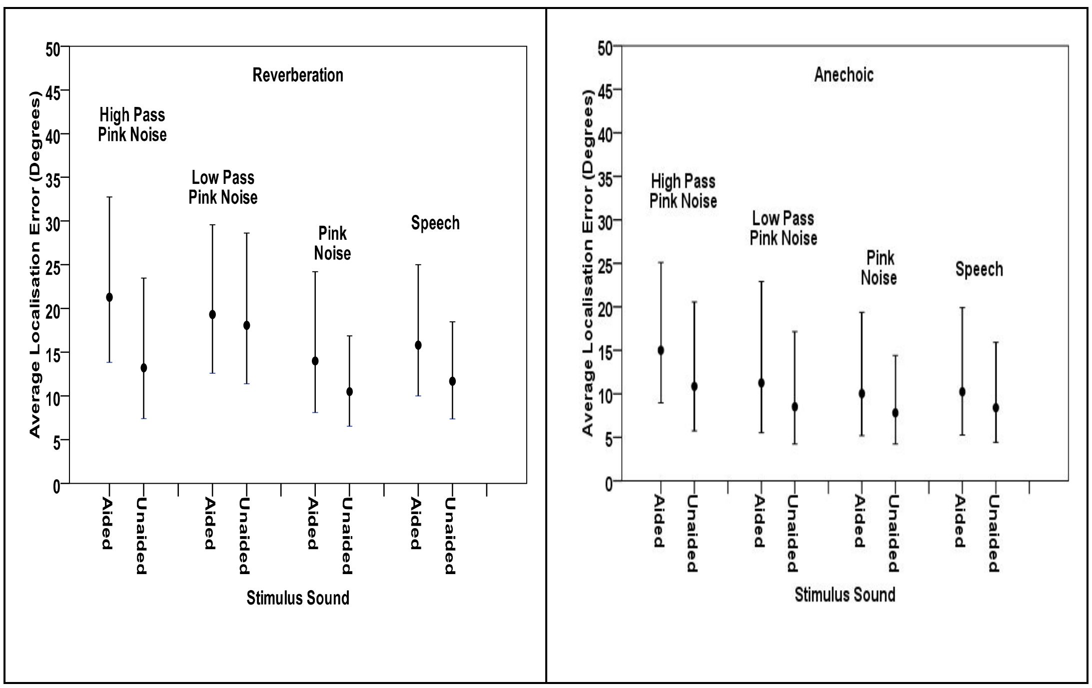


Figure 33 Localisation errors averaged over all speaker positions, for each of the stimuli, under both amplification conditions (log transformed data). The error bars represent "Mean ±1 SD".

In order to assess the effect of simulated reverberation on each of the stimuli in both aided and unaided conditions separately, a one-way repeated measures ANOVA was performed twice – once in the aided conditions and once in the unaided conditions. The tested factors were the stimulus, i.e. (i) speech-anechoic, (ii) speech-simulated reverberation, (iii) high-pass pink noise-anechoic, (iv) high-pass pink noise-simulated reverberation, (v) low-pass pink noise-anechoic, (vi) low-pass pink noise-simulated reverberation, (vii) pink noise-anechoic, and (viii) pink noise-simulated reverberation.

The results showed that in the aided conditions, the increase in the localisation error in the simulated reverberant environment, when compared to the anechoic environment, was significant for the pink noise, speech and low-pass pink noise stimuli. The results in the unaided conditions revealed that simulated reverberation produced a significant effect on localising the low-pass pink noise (mean=18.06 and mean=8.52), respectively, in the simulated reverberant and anechoic conditions. On the other hand, no significant effect of simulated reverberation was found on localising the high-pass pink noise when assessing the aided and unaided conditions separately. Graphical representation of the effect of simulated reverberation on each stimulus is shown in Fig. 34.

The effect of simulated reverberation on each stimulus, under both hearing condition

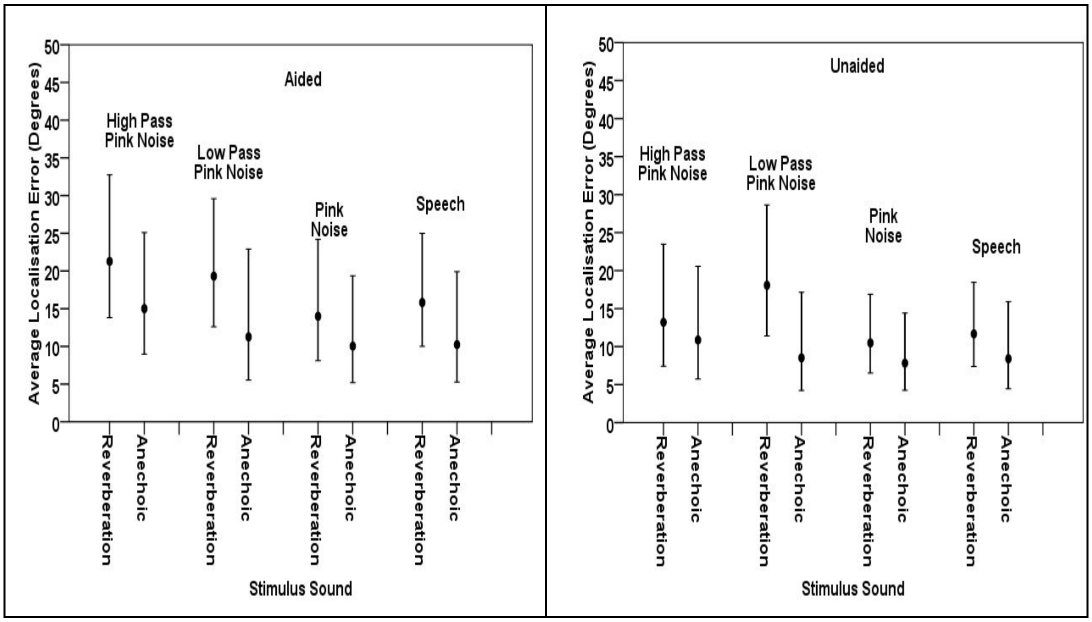


Figure 34 Localisation errors averaged over all speaker positions, for each of the stimuli, under both reverberation conditions (log transformed data). The error bars represent "Mean ±1 SD".

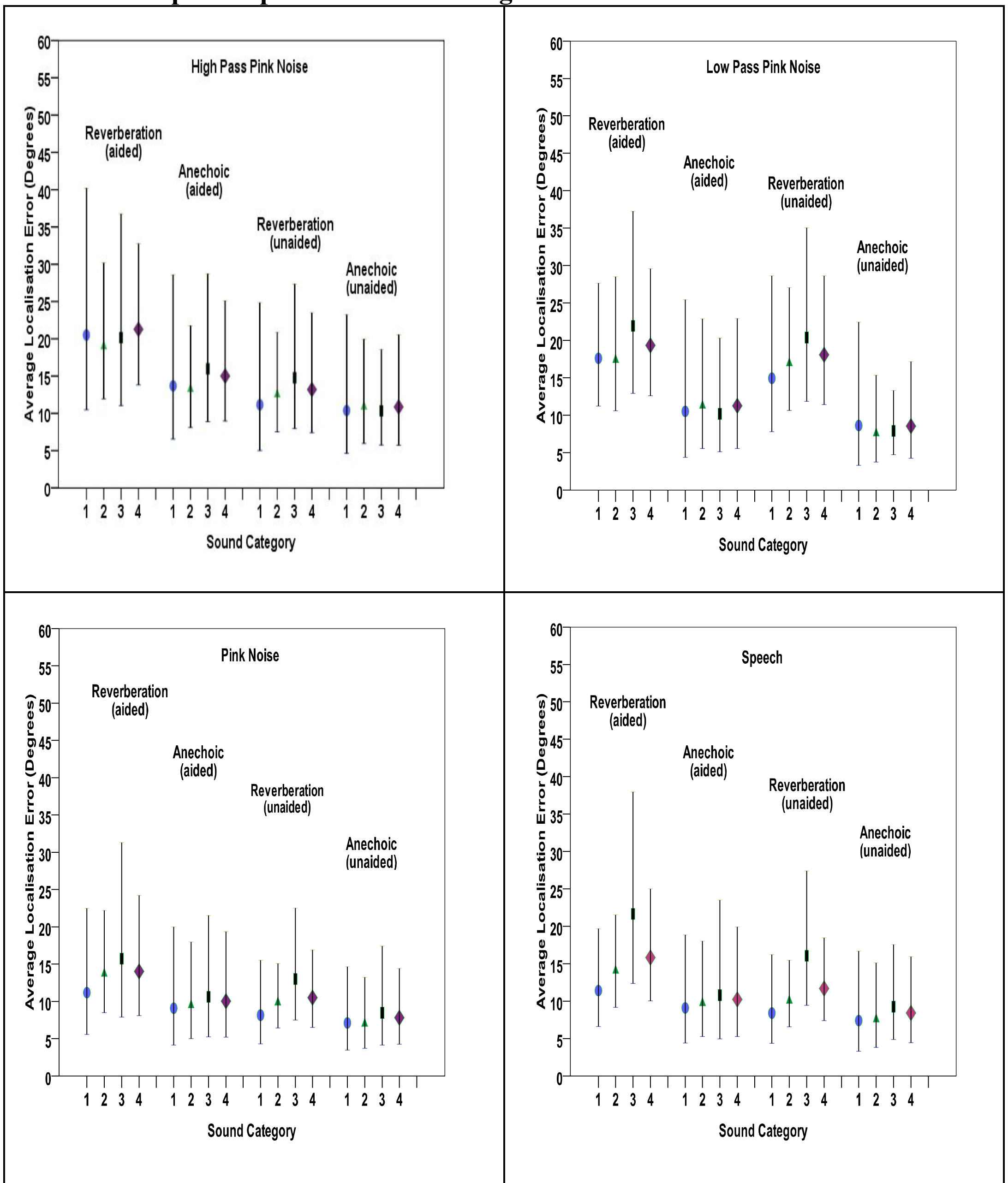
In order to determine the effect of speaker location on localisation performance, a four-way repeated measures ANOVA was performed where the factors were: (a) stimuli (i.e. pink noise, speech, low-pass noise and high-pass noise), (b) amplification (i.e. with hearing aid, without hearing aid), (c) reverberation condition (i.e. simulated reverberation, anechoic) and (d) speaker location (i.e. central, medial and lateral).

The results suggested that there was a significant effect of all the four main factors: (stimuli ($F_{(3, 81)} = 14.47$), reverberation ($F_{(1, 27)} = 60.47$), amplification ($F_{(1, 27)} = 21.20$), and speaker location ($F_{(2, 54)} = 3.82$) at $p<0.05$. Moreover, there was a significant difference in the mean localisation error for 'high-pass pink noise-pink noise', 'high-pass pink noise-speech', 'low-pass pink noise-pink noise', 'low-pass pink noise-speech', 'pink noise-speech' stimuli pairs. However, there was no significant difference in the mean localisation error for 'high-pass pink noise-low-pass pink noise'.

Furthermore, contrasts indicated that there was a significant difference between the lateral and medial speaker locations, where greater localisation errors were made for sound sources located laterally, rather than medially (mean=10.11 and mean=8.51) respectively. Fig. 35 shows the graphical representation of the effect of the speaker location.

A significant interaction effect was shown for 'stimuli-reverberation' ($F_{(3, 81)} = 5.41$), 'stimuli-location' ($F_{(6, 162)} = 5.20$), 'reverberation-location' ($F_{(2, 54)} = 4.49$), 'stimuli-location-reverberation' ($F_{(6, 162)} = 4.25$), 'stimuli-amplification-location' ($F_{(6, 162)} = 2.54$) and 'stimuli-reverberation-location-amplification' ($F_{(6, 162)} = 2.58$) at $p<0.05$.

The effect of speaker position and listening conditions



The numbers represent the following Sound Categories:

1 Central Speaker Location

2 Medial Speaker Location

3 Lateral Speaker Location

4 Average all

Figure 35 Average localisation errors for each stimulus at different speaker positions, under all experimental conditions (log transformed data). The error bars represent "Mean ±1 SD".

8.9.1 Localisation Bias

Localisation bias may be considered as the displacement of the listeners' internal map of speaker location, which can be caused by visual cues, such as the ventriloquism effect – where discordant multisensory cues are perceived as fused (Wallace et al., 2004) or acoustical cues, such as the arrangements of speakers in the room (see 'end effect', below) and sound reflections (Hartmann, 1983). In reverberant environments, listeners underestimate sound coming from lateral speakers and perceive it as closer to midline than to its actual speaker position (Cunningham et al., 2005b; Devore et. al., 2009). Generally, bias can be minimized, but not eliminated (Hartmann et al., 1998).

The spread of data in Figs. 25~28 suggest that there were distinct patterns in the localisation errors at the lateral speaker positions. The consistency of these errors reflects a perceptual bias towards the middle speakers, which was calculated by separating the random error from the localisation bias. At every speaker location, localisation errors – the difference in degrees between the perceived angle and the source location – for all listeners, were averaged together, the resulting mean error being an estimation of the localisation bias.

Figs. 36 and 37 – scattergrams – show the localisation bias and the SD in localising speech stimuli in the simulated reverberant environment for normal hearing and hearing impaired listeners respectively. The figures show that for both normal hearing and hearing impaired listeners, judgments for speech stimuli coming from more lateral speakers became more biased toward the median plane, an effect that was greater for the hearing impaired listeners. A possible explanation for showing a greater bias in the hearing impaired listeners is that they were less accurate, therefore there was a greater chance that their misperception exceeded the difference between adjacent speaker locations, hence producing a measurable error.

Localisation bias at lateral speaker positions was further tested for each stimulus type in all four experimental conditions for the hearing impaired listeners – i.e. unaided anechoic, unaided simulated reverberation, aided anechoic and aided simulated reverberation– and the t-test results show that for all listening conditions the bias was significantly different from zero at $p<0.05$. The t-test results in Table 11 suggest that there was a medial bias when localising the lateral speaker positions, which was significant for aided and unaided conditions in both anechoic and simulated reverberant environments. Speaker configuration may have been a possible explanation for a localisation bias in the responses (Perrett and Noble, 1995). If the speakers were set up in such a way that the span was 180º, then listener responses would be limited to a range of +/_90º which would yield an increase bias due to the 'end effect' (Hartmann et. al, 1998), since any signal that is perceived to originate to the left of the most extreme left-hand speaker will always be assigned to the furthest left-hand speaker, hence the listener's response choices will be more restricted at the ends of the speaker arc. However, the same response method was used for all experiments; therefore, any bias caused by the limited allowable response range would affect all tested conditions similarly to a first approximation.

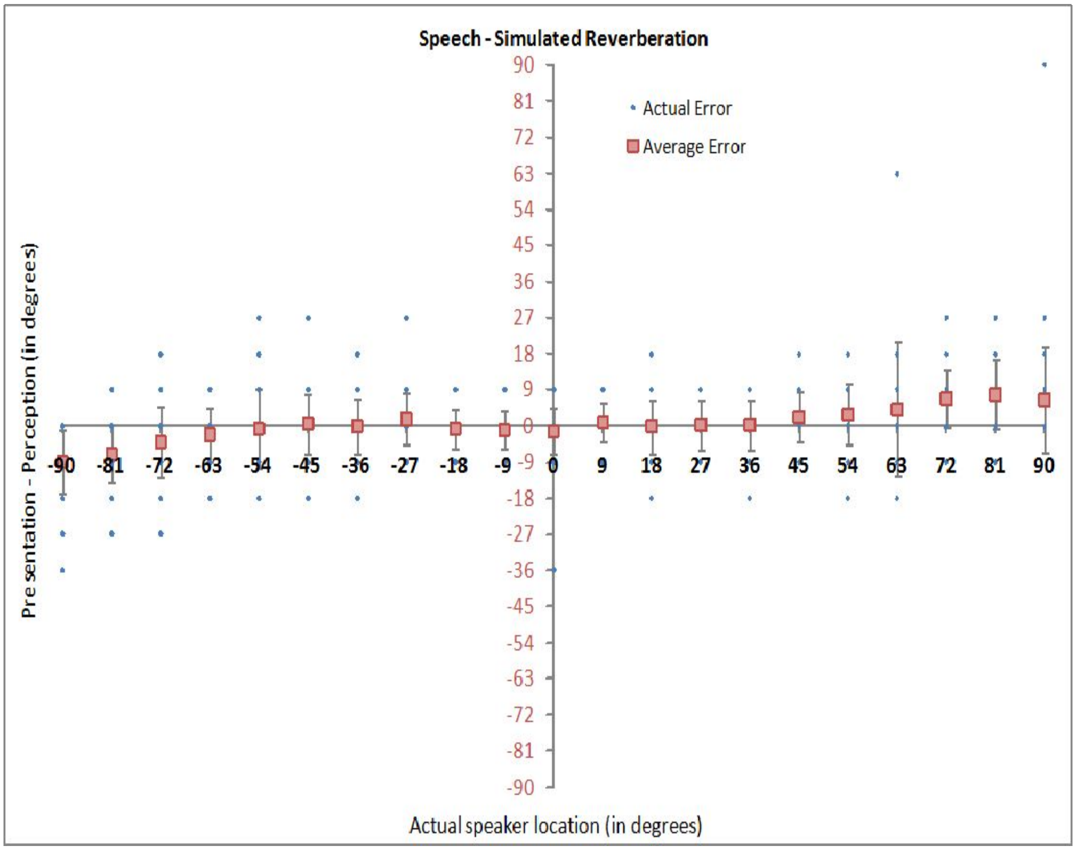


Figure 36 Localisation performance of normal hearing listeners for speech stimulus in a simulated reverberant environment. The Y-axis represents the localisation error (presentation – perception) and the X-axis represents the actual speaker location. The blue dots represent the listeners' localisation errors and the red dots represent the average error for each speaker location. The error bars represent "Mean ±1 SD"

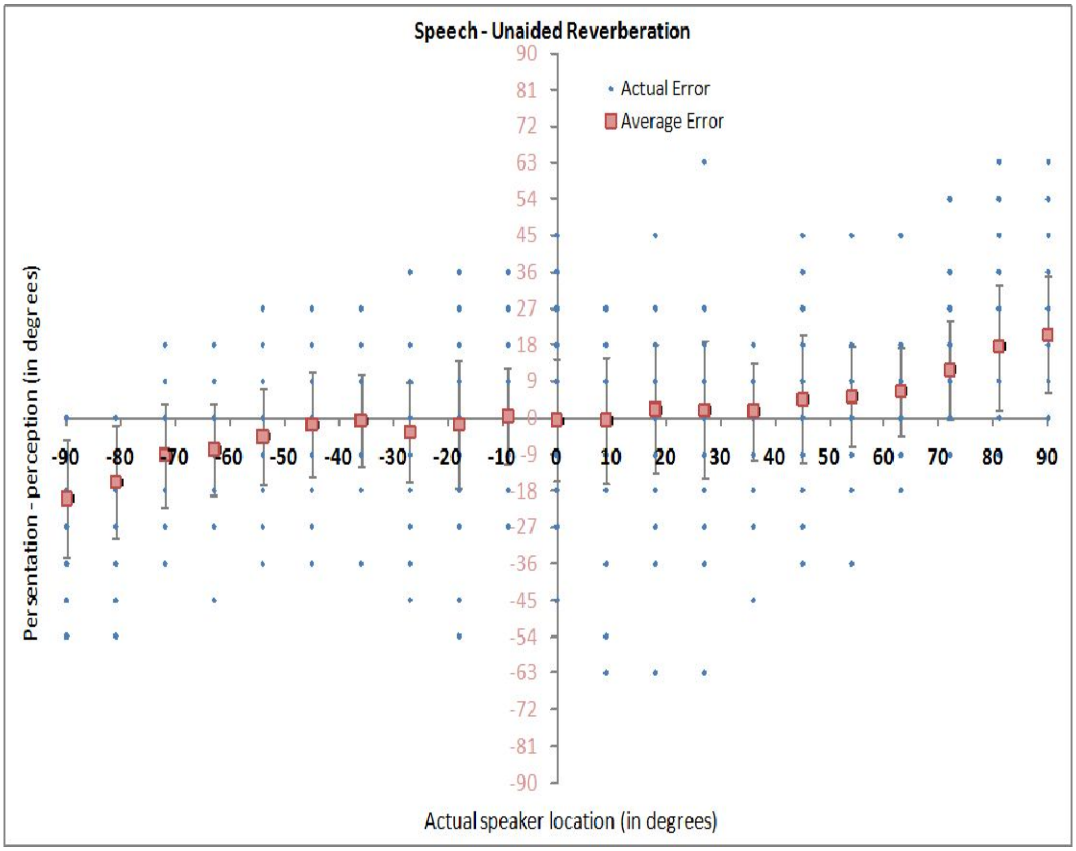


Figure 37 Localisation performance of hearing impaired listeners for speech stimulus in a simulated reverberant environment, without wearing their hearing aids. The Y-axis represents the localisation error (presentation – perception) and the X-axis represents the actual speaker location. The blue dots represent the listeners' localisation errors and the red dots represent the average error for each speaker location. The error bars represent "Mean ±1 SD"

Table 11 One sample t-test for the localisation bias averaged at speakers 90° and 81° for different conditions for hearing impaired respondents

	t	df	Sig. (2-tailed)	Mean Difference	Conclusion
HPPN Aided Simulated reverberation	-11.59	223	≤0.00	-22.30	Bias ≠0
LPPN Aided Simulated reverberation	-13.11	223	≤0.00	-23.62	Bias ≠0
PN Aided Simulated reverberation	-11.22	223	≤0.00	-19.08	Bias ≠0
SPEECH Aided Simulated reverberation	-15.44	223	≤0.00	-25.63	Bias ≠0
HPPN Aided Anechoic	-10.87	223	≤0.00	-16.67	Bias ≠0
LPPN Aided Anechoic	-7.66	223	≤0.00	-9.48	Bias ≠0
PN Aided Anechoic	-10.63	223	≤0.00	-11.53	Bias ≠0
SPEECH Aided Anechoic	-9.96	223	≤0.00	-12.33	Bias ≠0
HPPN Unaided Simulated reverberation	-11.17	223	≤0.00	-17.28	Bias ≠0
LPPN Unaided Simulated reverberation	-15.03	223	≤0.00	-22.58	Bias ≠0
PN Unaided Simulated reverberation	-14.12	223	≤0.00	-14.38	Bias ≠0
SPEECH Unaided Simulated reverberation	-19.00	223	≤0.00	-18.48	Bias ≠0
HPPN Unaided Anechoic	-10.73	223	≤0.00	-9.96	Bias ≠0
LPPN Unaided Anechoic	-9.52	223	≤0.00	-6.27	Bias ≠0
PN Unaided Anechoic	-10.68	223	≤0.00	-8.44	Bias ≠0
SPEECH Unaided Anechoic	-11.73	223	≤0.00	-9.88	Bias ≠0

8.9.2 Correlation between horizontal localisation abilities and hearing thresholds

Table 13, below, shows the correlation coefficients between the localisation performances averaged over all participants in different listening conditions, and their auditory thresholds. These were calculated using the average of hearing levels from both ears and the average of the localisation errors, under anechoic and simulated reverberant conditions, separately.

To find hearing thresholds in the sound field SPL at 0 azimuth, audiometric thresholds obtained using headphones were converted to minimum audible field measurements by adding the MFA values obtained from ISO 389-7 (Table 12).

Table 12 The MAF values obtained from ISO 389-7.

Freq (khz)	Difference (dB)
0.25	15.5
0.5	4.0
1	1.5
2	−1.5
3	−6.0
4	−4.0
6	5.0
8	12.5

Table 13 Correlation coefficients between the localisation performances averaged over all participants at different listening conditions, together with their auditory thresholds

	Simulated reverberation				**Anechoic**			
Hearing levels	**High Pass Pink noise**	**Low Pass pink noise**	**Pink noise**	**Speech**	**High Pass Pink Noise**	**Low Pass pink noise**	**Pink noise**	**Speech**
250 Hz	.264 .175	.411 .030	.205 .295	.119 .547	.271 .162	.223 .255	.180 .358	.234 .231
500 Hz	.310 .109	.384 .044	.323 .094	.204 .298	.423 .025	.327 .090	.273 .160	.398 .036
1 KHz	-.110 .577	.024 .904	-.171 .383	-.205 .295	-.133 .501	-.122 .538	-.111 .573	-.143 .468
2 KHz	.239 .221	.266 .172	.345 .072	.310 .108	.397 .037	.272 .162	.384 .043	.481 .010
4 KHz	.202 .304	.122 .536	.348 .069	.251 .198	.180 .360	.075 .705	.195 .319	.181 .358
6 KHz	.133 .498	.207 .290	.216 .270	.145 .463	.114 .562	.028 .887	.130 .508	.041 .835
8 KHz	.435 .021	.404 .033	.298 .123	.240 .219	.403 .034	.256 .188	.434 .021	.340 .077
PTA Low	-.060 .762	.095 .631	-.126 .524	-.178 .365	-.072 .718	-.073 .711	-.071 .718	-.087 .658
PTA high	.326 .091	.322 .095	.339 .078	.254 .191	.294 .129	.167 .396	.321 .095	.245 .209
PTA ALL	.046 .816	.198 .312	-.007 .970	-.083 .676	.036 .856	-.005 .980	.043 .827	.017 .931

The upper values represent Pearson's correlation coefficient. The lower values represent p-value of a two-tailed test.

All potential correlation coefficients were computed, resulting in 80 different correlations (Table 13). Since independence of measurements can not be assumed, a Bonferroni correction to control for inflation of Type I errors resulted in a (conservative) critical p-value of less than .000625 (.05/80). Only at this level significance may be assumed. However, the p-values in Table 13 are so far away from the threshold, that none of the correlations show significant correlation. These results – the absence of correlation between hearing level and localisation performance – are consistent with the findings of Durlach et al. (1981), who reviewed the literature extensively (prior to 1981) and concluded that localisation is not predictable on the basis of listeners' audiograms. The results correlations are also in agreement with the findings of Gabriel et al., (1992), who found that the pattern of localisation performance show no relation to the audiometric pattern.

8.10 Comparison of hearing impaired and normal hearing participants

The data obtained from this experiment were compared with the data collected from Experiment 1, in which twenty-two normal hearing participants, with ages ranging from 18 to 35, were tested with the speech stimulus: "Where do I speak from", in both anechoic and reverberant environments.

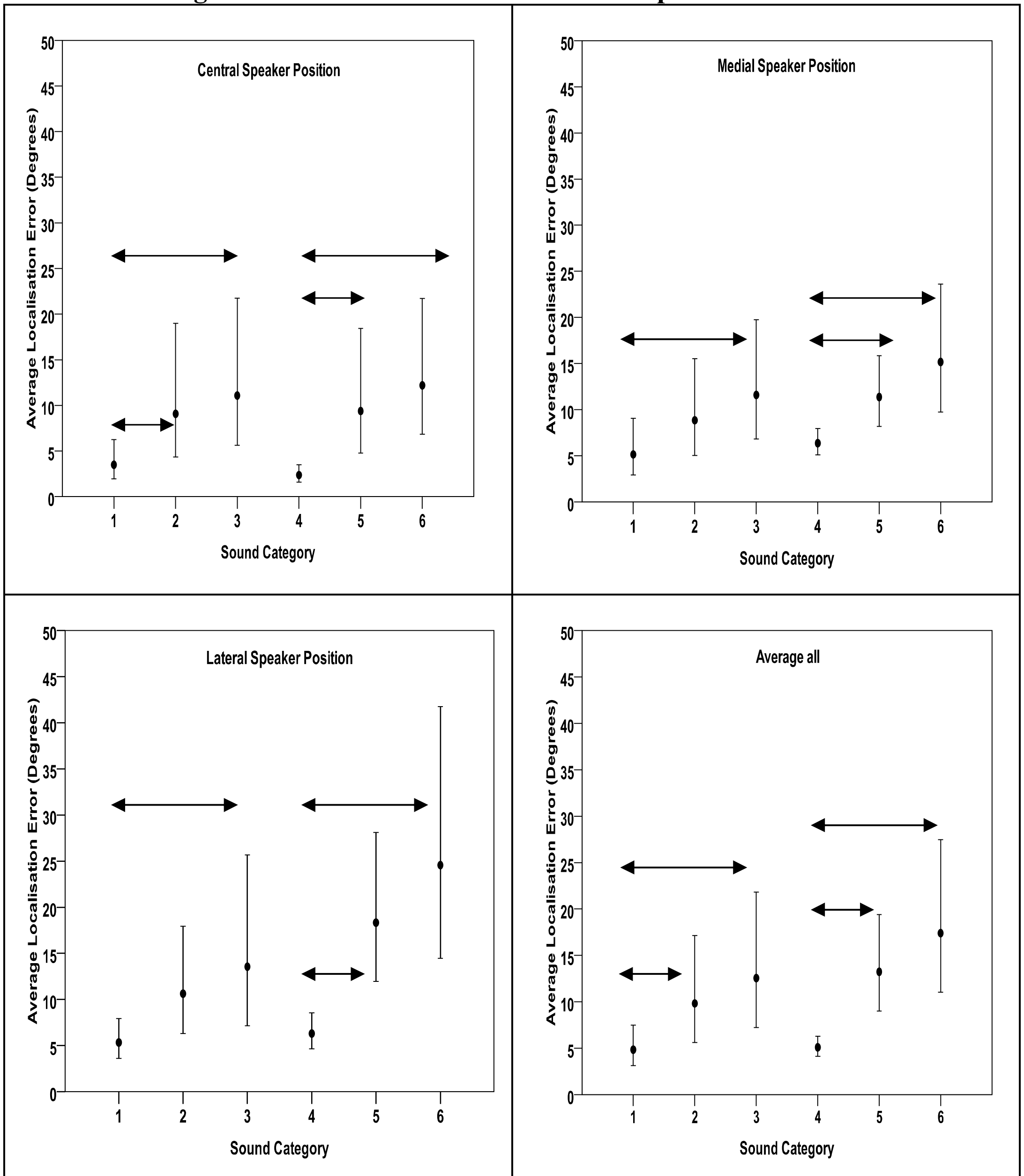
8.10.1 Analysis

For the hearing impaired participants, the better condition (unaided) was used to compare with the normal hearing participants. A one-way ANOVA test was conducted to assess whether means on the dependent variable, i.e. the mean localisation error for the speech stimulus averaged at all speaker positions, were significantly different between the two hearing category groups. The results showed that there were significant differences in the means of average localisation errors between the hearing impaired and the normal hearing participants, the main effect on the hearing categories was significant at $p<0.01$. Another observation was made, which is the larger consistency in performance between the normal hearing listeners when compared to the hearing impaired listeners. Fig. 38, below, shows the average errors in localising speech stimulus by both normal hearing and hearing impaired participants under different speaker positions.

Fig. 39 is a graphical representation of the difference between normal hearing and hearing impaired listeners in localising the pink noise stimulus in the anechoic environment.

The above results confirmed Hypothesis 5, which suggested that the average error in localisation would be greater for hearing impaired participants, both aided and unaided, than normal hearing listeners.

Effect of hearing conditions on the localisation of the speech stimulus



The numbers 1~6 in the above graphs represent the following categories of sound:

1→ speech, normal, anechoic
2→ speech, hearing impaired, anechoic (unaided)
3→ speech, hearing impaired, anechoic (aided)
4→ speech, normal, simulated reverberation
5→ speech, hearing impaired, simulated reverberation (unaided)
6→ speech, hearing impaired, simulated reverberation (aided)

Figure 38 Average localisation errors for the speech stimulus under both reverberation conditions, for all hearing conditions (normal, aided and unaided) and for different speaker positions. The error bars represent "Mean ±1 SD". The horizontal lines represent the pair of stimuli that differ significantly in the same reverberation condition - anechoic or simulated reverberation.

The effect of hearing conditions – i.e. normal, unaided, aided– on the localisation of pink noise stimulus

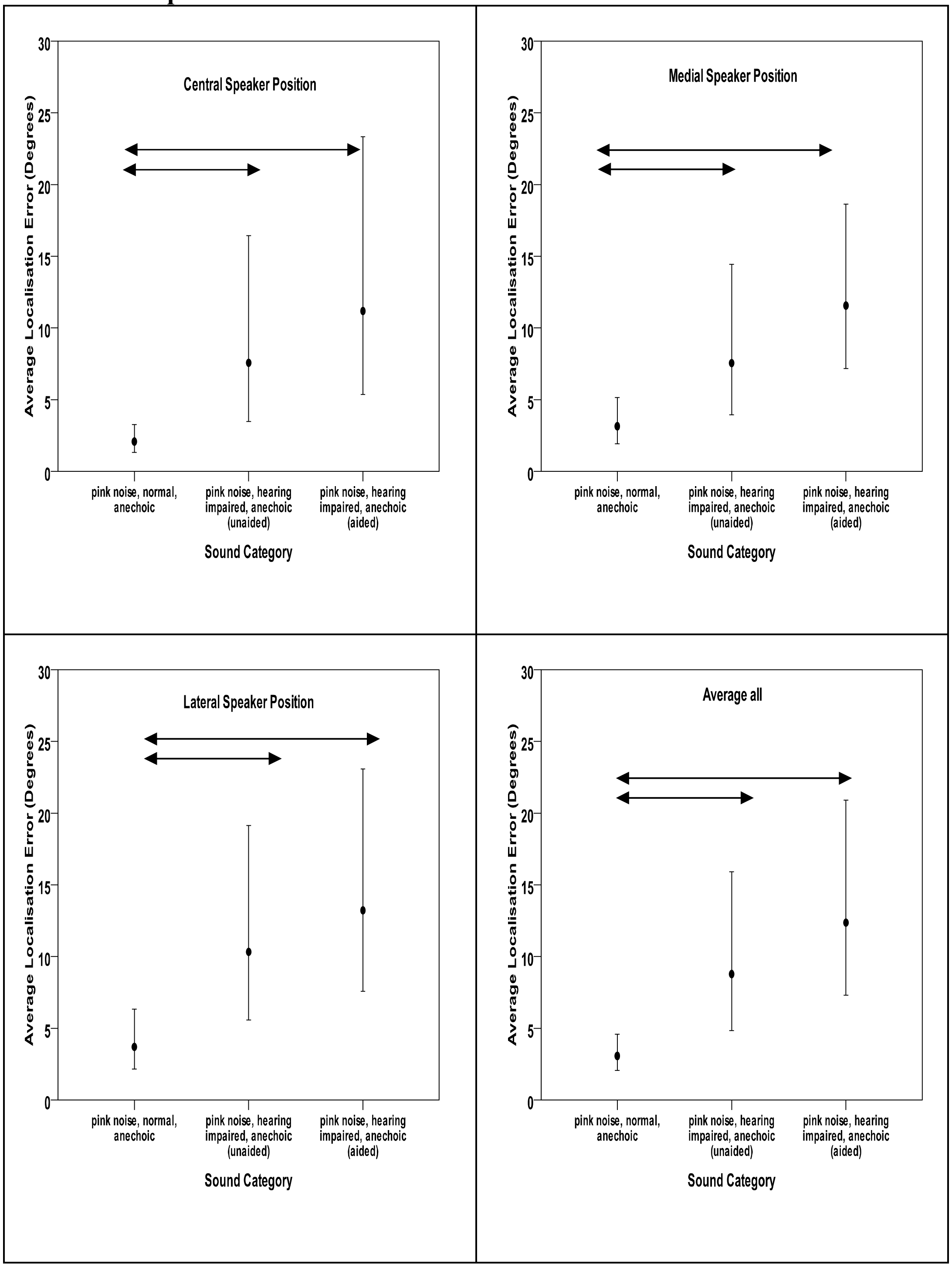


Figure 39 Average localisation errors for the pink noise stimulus in the anechoic environment, for all hearing conditions and for different speaker positions (log transformed data). The error bars represent "Mean ±1 SD". The horizontal lines represent significant difference at p<0.05

8.11 Summary of findings

The aim of this chapter was to investigate the effect of the listening environment and amplification on the localisation abilities of the hearing impaired participants. And to compare their performance with the normal hearing participants.

1. **The effect of simulated reverberation**

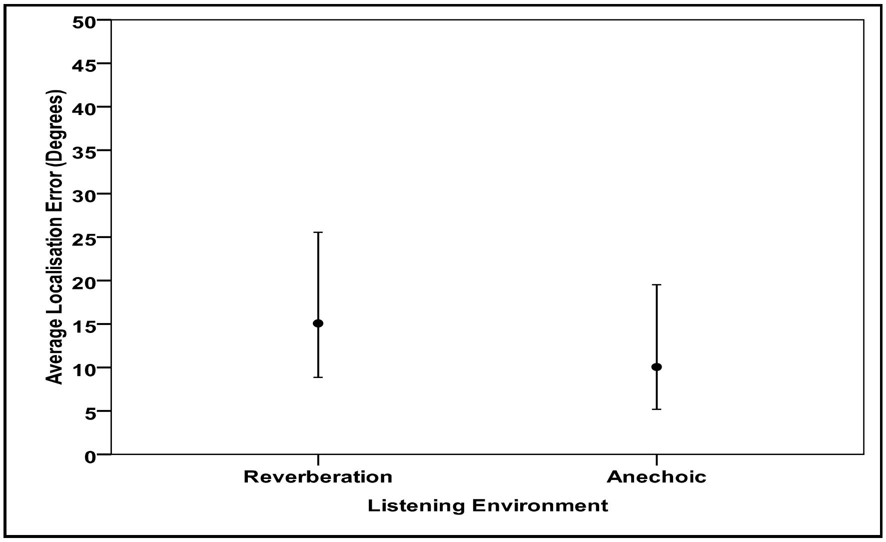


Figure 40 The localisation errors for all stimuli and for all speaker positions when combining both aided and unaided trials (log transformed data). The error bars represent "Mean±1 SD".

Fig. 40 shows that the average localisation errors for hearing impaired participants were greater in simulated reverberant than in anechoic conditions when combining aided and unaided trials. This difference was significant at $p<0.05$.

2. **The effect of amplification**

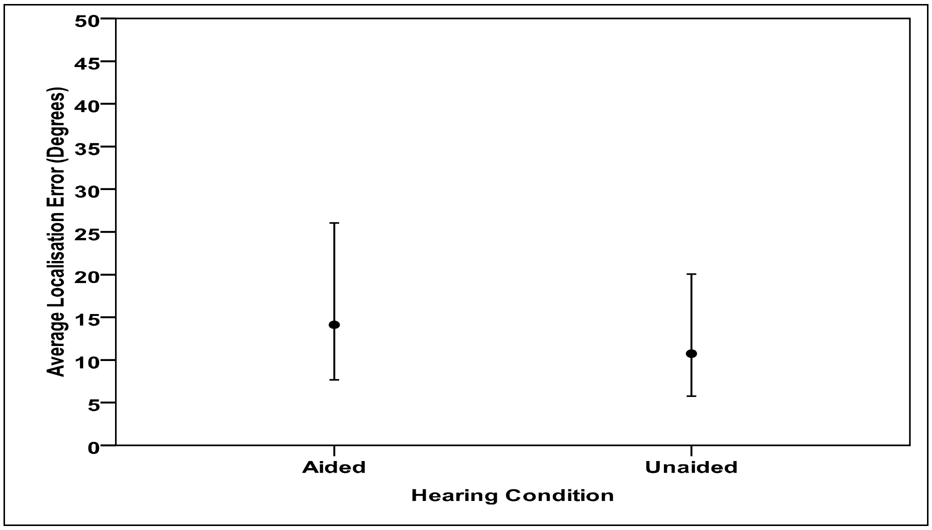


Figure 41 The localisation errors for all stimuli, and for all speaker positions, when combining both reverberation conditions (log transformed data). The error bars represent "Mean ±1 SD".

Fig. 41 shows that the average error in localisation was greater in the aided trials than in the unaided trials when combining anechoic and simulated reverberant environments. This difference was significant at $p<0.05$.

3. **The effect of stimuli type**

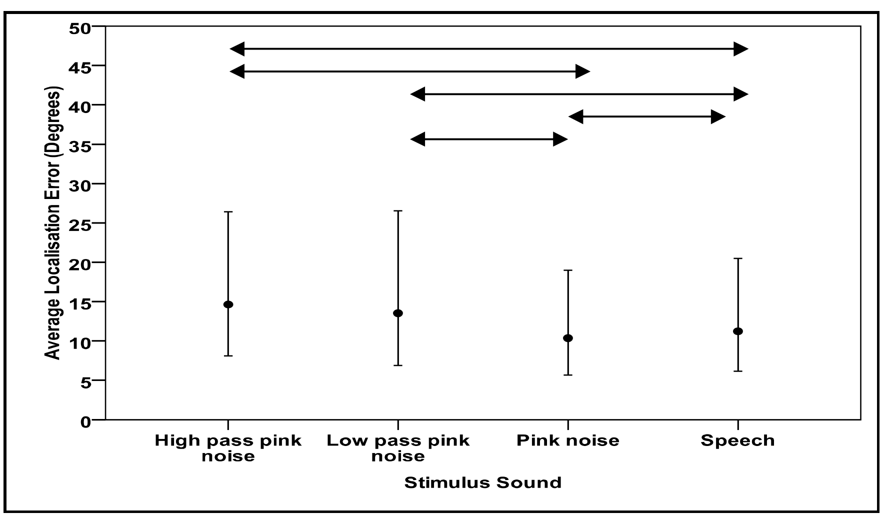


Figure 42 The localisation errors for the four 'stimuli sounds' for all speaker positions, when combining both reverberation conditions and both amplification conditions (log transformed data). The error bars represent "Mean ±1 SD". Horizontal lines represent significant difference at $p<0.05$.

Fig. 42 shows that the average error in localisation for all the stimuli types under all experimental conditions, i.e. aided, unaided, anechoic and simulated reverberation, was the least for stimuli containing both ILD and ITD cues, i.e. speech and pink noise. The effect of stimuli was significant at p<0.05.

4. **The effect of speaker location relative to the participant**

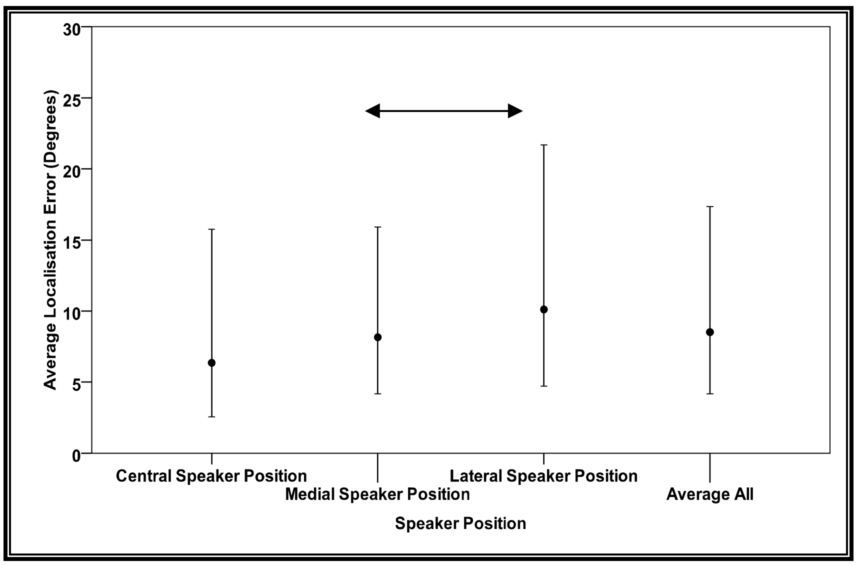


Figure 43 The localisation errors at different speaker positions, taking all 'stimuli sound' and combining all experimental conditions together (log transformed data). The error bars represent "Mean ±1 SD". The horizontal lines represent significant difference at p<0.05.

Fig. 43 shows that the average error in localisation for all stimuli, and under all experimental conditions, i.e. aided, unaided, anechoic and simulated reverberation, was the highest for lateral speaker positions. The effect of speaker location was significant at p<0.05.

5. **Normal hearing vs. hearing impaired participants**

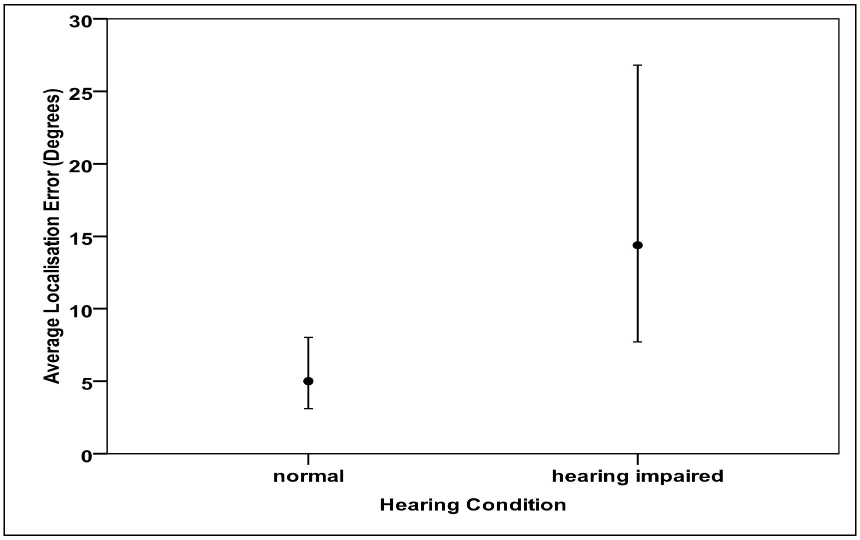


Figure 44 Localisation performance of normal vs. hearing impaired participants taking all 'stimuli sound' and all experimental conditions together (log transformed data). Error bars represent "Mean ±1 SD".

Fig. 44 shows that the average error in localisation was greater for the hearing impaired participants, both aided and unaided, than for normal hearing participants. The difference was significant at $p<0.05$.

8.12 Discussion

8.12.1 Normal hearing participants

The results from Experiments 1 and 2 in this thesis revealed that normal hearing participants could localise ‘speech, high pass speech, low pass speech and pink noise’ accurately under both anechoic and reverberant conditions. This may be explained by the ‘precedence effect’, as all tested stimuli have abrupt onset/offset characteristics that allow them to be localised using this effect (Yost, 2000). Moreover, the speech stimulus was the easiest to localise by normal hearing participants. One possible explanation is that speech is a broadband stimulus in which the listener can use the ILD contours available in several frequency bands as a good estimator of the sound source (Wightman and Kistler, 1993). Another explanation is the availability of ILDs, ITDs and spectral cues in speech stimulus, which makes it easier to be localised. Furthermore, the complex speech envelope is one more advantage that is missing in the other stimuli. Also, speech is frequently used in everyday life, therefore it would be more familiar to the participants, making it easier for them to localise.

8.12.2 Hearing impaired participants

A significant difference was found between normal hearing and hearing impaired participants in their localisation performances. Hearing aid wearers performed worse than normal hearing participants in both aided and unaided conditions. Fig. 38, above, (p.146) shows that the localisation abilities of the normal hearing participants were significantly better than the hearing impaired participants, even without amplification. This deterioration in localisation may be explained by both age and hearing loss, since, due to the changes in the peripheral, or central auditory processing that occurs during the ageing process, localisation abilities tend to worsen with age. Additionally, sensitivity to ITDs also tends to decrease with increasing age (Strouse et al., 1998). However, both hearing impaired and normal hearing participants followed the same trend, with more localisation errors towards the lateral than towards the central and medial speaker positions.

Moreover, the effect of simulated reverberation on localising speech stimulus by the unaided hearing impaired participants, was comparable to the effect of simulated reverberation on the normal hearing participants, whereby no effect was found in localising sound from central and medial speaker positions. However, at lateral speaker positions, adding reverberation resulted in an increase in the average localisation error of the unaided hearing impaired listeners by 8º, whereas no effect was found on the normal hearing listeners. This may have been due to the loss of spectral cues with the high frequency hearing loss, since they probably explain reasonable performance at lateral speaker positions, where ILD and ITD cues tend not to be accurate. Although the effect of audibility was compensated for in the unaided conditions by increasing the intensity level of the stimuli, it is still possible that high frequencies, which hold spectral cues, were at very low sensation levels. Moreover, it is possible that hearing impaired participants were unable to resolve spectral peaks and notches, due to the loss of frequency selectivity associated with their hearing loss.

8.12.3 The effect of wearing hearing aids

Hearing aid wearers were more accurate in their performances without their hearing aids than with their hearing aids. The difference was greater when localising the high-pass pink noise in the simulated reverberant conditions (mean= 13.2) without the hearing aids and (mean= 21.3) with the hearing aids. Generally, for all four stimuli, the most localisation errors occurred in the aided and simulated reverberant condition, whereas the lowest angle of error occurred in the unaided and anechoic conditions.

Analyses of the differences in localisation errors of the hearing aid wearers, with and without the hearing aids, revealed that the least difference in accuracy in anechoic conditions was for speech stimulus (1.8°), and for the low-pass pink noise (1.0°) in simulated reverberant conditions. Whereas the greatest difference in accuracy in both anechoic and simulated reverberant conditions were for the high-pass pink noise (4° and 7° respectively). This may have been the result of the greater familiarity of the speech stimulus, as opposed to the high-pass pink noise. Furthermore, while localising the high-pass pink noise, participants relied on the ILD and spectral cues, both of which were adversely affected by the hearing aids. The spectral cues can be disrupted by closed ear moulds; however, open ear moulds can improve localisation performance (Byrne et al. 1995). Also, compression in the hearing aids may reduce inter-aural intensity level difference cues (Byrne et al. 1996).

This deterioration effect of the hearing aids in the horizontal plane localisation abilities is consistent with results from previous studies (Byrne et al 1995; Van den Bogaurt et al 2006; and McManus 2008). However, results from this study, reported in this chapter, revealed that adding reverberation had an additional adverse effect on top of hearing aids by affecting ITD cues. The average localisation errors for all four stimuli under aided-simulated reverberant conditions were greater than the average localisation errors under aided-anechoic conditions. Table 14 shows the mean values of the localisation errors for the different stimuli under both aided-simulated reverberation and aided-anechoic conditions.

Table 14 Mean values of the localisation errors for the different stimuli in aided conditions

Stimuli	**Mean**	
	Aided-anechoic	**Aided-simulated reverberation**
High-pass pink noise	15.00	21.27
Low-pass pink noise	11.26	19.31
Broadband noise	10.03	13.99
Speech	10.24	15.81
All stimuli	11.63	17.59

8.12.3.1 Aiding and stimulus frequency

The results for all stimuli for the hearing aid wearers were adversely affected by aiding in both simulated reverberant and anechoic conditions. However, the increase in localisation errors with aiding, as opposed to non-aiding, was the greatest for high-pass pink noise. This implies that the use of hearing aids affects ILDs, as shown by McManus (2008).

ILD cues may be reduced by the compression of the hearing aids, especially when the compression settings on both ears are equal. This will result in the far ear being more amplified than the near ear, hence reducing the level difference between the two ears (Byrne, 1998a).

Additionally, spectral cues can be disrupted by closed earmoulds, which cause significant deterioration in localisation (Byrne at al., 1996). The disruption of the spectral cues could result in poorer performance for the pink noise, speech and high-pass pink noise. In this study, such an effect was not observed in the anechoic conditions, since the impact of amplification was significant only in localising high-pass pink noise. However, in the simulated reverberant environment, the increase in the localisation error caused by amplification was significant for the speech, pink noise and high-pass pink noise stimuli. This suggests that spectral cues may be further disrupted by reverberation. The adverse effect of amplification on localising the high-pass pink noise was double the effect seen on localising the broadband stimuli (an average error of 4° and 2° respectively) in the anechoic conditions and (7° and 3.5°) in the simulated reverberant conditions.

The results were consistent with the findings of the KEMAR measurements, suggesting that ILD cues were more adversely affected by aiding than were the ITD cues. The hearing aids used in the KEMAR measurements were programmed for a hearing loss similar to that encountered by participants in this localisation experiment. Moreover, the same hearing aid make and model – the Siemens Prisma 2 Pro– was used by all participants and for the KEAMR measurements. In the KEMAR measurements, amplification had a slight adverse effect on ITD cues, explained by the mismatch group delay between both hearing aids. However, results from Experiment 4 showed that the low-pass pink noise was the least affected by amplification. The increase in the localisation errors for the low-pass pink noise in the aided conditions, as opposed to the unaided conditions, was 3° and 1° in the anechoic and simulated reverberant environments respectively. These results suggest that it is unlikely that the group delay of the hearing aids had a significant effect on ITD cues.

In the anechoic environment, the localisation errors – original data – measured in this study for the unaided conditions were generally comparable with those measured by McManus (2008). This was expected, as, in both studies, the same pink noise, high-pass pink noise and low-pass pink noise stimuli were used. However, the mean localisation errors in this study were smaller than they were in the McManus study, which gave mean errors for the pink noise of 11.8° unaided and 12.9° aided, whereas the respective values for the pink noise of this study were 8° and 10°. The low-pass pink noise in the McManus study gave mean errors of 13.1° unaided and 14.1° aided, while the respective values of the same in this study were 9° and 11°. The high-pass pink noise in the McManus study gave mean errors of 15.9° unaided and 24° aided, while the respective values of the same in this study were 11° and 15°.

As mentioned earlier, our study compensated for the effects of audibility by presenting stimuli at equal sensation levels, whereas in the McManus study, the same intensity was used to present stimuli in both aided and unaided conditions. The lack of compensation in the McManus study would be expected to cause poorer localisation in the unaided conditions when compared to our study, especially for the high-pass pink noise. However, both studies showed clearly that the use of ILD cues is greatly affected by amplification, whereas the use of ITD cues is unaffected by amplification.

8.12.4 The effect of the listening environment

The results from Experiments 1 and 2 in this study for the normal hearing participants showed that adding reverberation did not have a significant effect on their ability to localise broadband stimuli. However, for the hearing impaired participants, adding reverberation reduced their ability to localise, since they were less accurate in the simulated reverberant environment than in the anechoic one in both aided and unaided trials. However, the increase in localisation error in simulated reverberation was more significant in the aided trials (5.9°) than in the unaided ones (2.8°). Therefore, it appears that, generally, reverberation has more impact on the localisation performance when hearing aid participants are using their hearing aids .One possible explanation is the temporal distortions that occur in hearing aids, since temporal envelope cues are reduced both by reverberation and compression. Moreover, as mentioned earlier, this may be due to the disruption of spectral cues caused by reverberation and amplification.

8.12.4.1 The listening environment and the stimulus frequency

A significant interaction in this study was found between stimulus type and reverberation condition. The analysis of the relationship between the stimulus frequency and reverberation revealed that simulated reverberation reduced the localisation accuracy of hearing aid wearers; however, the increase in the localisation errors in simulated reverberation was greater in the aided than in the unaided conditions. The increase in localisation errors in simulated reverberation was the greatest for low-pass pink noise. No significant difference was found between simulated reverberant and anechoic performance for the high-pass pink noise, under either aided or unaided conditions.

The increase in the localisation errors between the anechoic and the simulated reverberant conditions for speech and pink noise stimuli was only significant in aided conditions. Given that localisation performance appeared worse for the low-pass pink noise than for the speech and pink noise stimuli, listeners could have reduced their reliance on ITD cues when localising stimuli containing both ILD and ITD cues in reverberant environments to improve localisation accuracy. These results are also consistent with the findings of Devore and Degutted (2010), suggesting that, when the virtual space stimuli contain both ITD and ILD cues at high frequencies, ILDs provide more reliable localisation cues than envelope ITDs do in reverberation. Devore and Degutted (2010) suggested that, in reverberant environments, listeners will rely on ILD cues more than ITD envelope cues when localising broadband stimuli. Consequently, the significant effect of simulated reverberation on the speech and pink noise stimuli in Experiment 4, in aided conditions only, might be accounted for by ILD cues that have been distorted by aiding.

The increase in localisation error in simulated reverberation was significant and was at its greatest for the low-pass pink noise in both aided and unaided conditions (an average of 8.1° and 10.5° respectively). The finding that low-pass pink noise is more affected than the other stimuli suggests, therefore, that reverberation has a greater impact on the ITD cues than on the ILD cues. A possible explanation may be the reduced temporal resolution associated with hearing loss and the hearing aid itself, which could affect the ability of the hearing aid users to utilise ITD cues and the precedence effect in the reverberant environments.

Although listeners in general rely mainly on low-frequency ITD cues in localising sound sources in anechoic conditions (Wightman and Kistler, 1992), they are likely to use other cues in more challenging listening conditions involving reverberation. They may use ITD cues that are available in the amplitude envelopes of high frequency carriers, since these can be as effective as ITDs in the waveform fine structure of low-frequency sounds (Bernstein and Trahiotis, 2002). In normal hearing listeners, the lowest thresholds of ITD cues are 10–20 μs, which are based on ITD fine structure cues of low-frequency sounds (Zwislocki and Feldman, 1959), whereas ITDs in the low-frequency envelopes of high-frequency carriers need to be greater to be detected, at least 100 μs (Bernstein and Trahiotis, 1985). By looking at the KEMAR measurements (Fig. 7, p.70), it can be seen that an ITD of 200 μs corresponds to an azimuth of about 18º centrally, and much more laterally, which implies that listeners using only envelope ITDs will make relatively large errors. Looking at the top right hand panel in Fig. 35 on page 137, the average errors – anechoic unaided – are about 5-10º; this is too good for ITD envelope cues only, therefore it must have been due to using ITD fine structure cues. The anechoic aided performance may also have been a little too good for only envelope cues. However, the results for the aided and unaided simulated reverberation conditions could be accounted for by ITD envelope cues. A possible interpretation is that hearing aids reduce, but do not eliminate, ITD fine structure cues, whereas reverberation pretty much destroys the usefulness of fine structure and forces the listener to use envelope cues.

ITD cues are the main cues in localising low-pass pink noise; therefore any disruption of them will dramatically affect the low-pass pink noise localisation performance. Unlike speech, pink noise and the high-pass pink noise stimuli, in which ILD and spectral cues are accessible. The results from this study were consistent with the findings in Cunningham (2000), which revealed that the average error in ITD and ILD, in anechoic conditions, was slightly larger than the just-noticeable difference (JND) in headphone experiments (20–50 μs and 0.5-1.0 dB respectively). However, in reverberant conditions, they found that the ITD error was slightly larger than the JND reported in the literature (70 μs), whereas, the mean error in ILD was as small as the JND values.

8.12.5 The effect of loudspeaker location

Normal hearing participants and hearing aid wearers in this study followed the same pattern in their localisation performance, where less accurate performance was observed when the stimuli was presented from the lateral speaker positions. These results, therefore, were consistent with the findings of Wightman and Kistler (1989), Lutman and Payne (2004) and McManus (2008), in which subjects were more accurate in localising stimuli presented from speakers toward the midline at 0°, ±18°, ±36° azimuth, than when they were presented from speakers away from the midline at ±45°, ±72°and ±90°. One explanation for this is that, for the most central speaker positions, small changes in the sound source would have resulted in large changes in the inter-aural difference cues. However, as the sound source moves toward the more lateral speakers, the changes in the inter-aural difference cues become smaller (see Figs. 3 and 7, p.66 and 70). Another possible explanation is the way the speakers were set up. Having a limited number of response choices at the end speakers may have introduced an increase bias due to "end effect". Moreover, the centralizing tendency at the lateral speaker positions might well be an effect of the diotic presentation of reverberation.

In this study, a significant interaction was found between reverberation condition and speaker position. Analysis of the relationship between simulated reverberation and speaker location in the hearing aid wearers revealed that, generally, for all tested stimuli, simulated reverberation had a greater effect when the sound came from the lateral speakers than from the central and medial speakers, in both aided and unaided conditions. As mentioned earlier, the changes of the inter-aural cues in the cone of confusion area –i.e. the lateral speaker positions – are small, and adding reverberation may have made it even more difficult for the participants to detect these small changes in inter-aural cues. Therefore, other cues, such as spectral cues, can be used to resolve this issue (Middlebrooks et al., 1989).

Since spectral cues are monaural cues in which no comparison between the sounds at both ears is required, the participants may have relied on the spectral cues information from their HRTFs to localise signals when they found the binaural cues to be unreliable. However, pinnae effects are only present for frequencies at 6 kHz and higher (Hartmann, 1999), therefore, spectral cues are mostly important at high frequencies and require the listener to analyse steady-state portions of the signal to an ear. Unlike the precedence effect, which relies mainly on a short segment of the direct signal arriving before the reverberant components (similarly, ITDs can be extracted in this segment), spectral cues are much more likely to be disrupted by reverberation. The use of spectral cues probably explains the reasonable performance at lateral positions where ILD and ITD cues are not accurate. As a consequence, it might be expected that reverberation interferes with localisation performance at lateral speaker positions, which would explain the poor performance observed in our study when participants were localising sound from the lateral speaker positions in the simulated reverberant environment – results that were consistent with Giguere and Abel (1993). However, the effect of reverberation was similar across all speaker positions for the aided high-pass pink noise. A possible explanation is that high-pass pink noise is mainly localised by ILD and spectral cues, which are both disrupted by amplification. Therefore, we might expect amplification to interfere with localisation performance of the high-pass pink noise at all speaker positions for both anechoic and simulated reverberant conditions.

A significant interaction between amplification and speaker location was not found in the study, however, for both simulated reverberant and anechoic conditions, the aided average localisation error was greatest at the lateral speaker positions. Because spectral cues are removed, or are seriously disturbed, by hearings aids and simulated reverberation, it was expected that hearing aid wearers would be poor at localisation from the lateral speaker positions.

Chapter 9 Limitation of the Study and General Conclusions

9.1 Limitation of the study

In most sound localisation experiments, the response of the subject is limited to a number of discrete positions. In this study 21 loudspeakers were located in the frontal half of the horizontal plane only. Thus, this study did not investigate vertical localisation or the front-back confusions on the horizontal plane. Moreover, as the speakers were set on the frontal horizontal 180º, which have well defined ends (i.e. terminated span), any signal heard to originate from the left of the far left speaker, or from the right of the far right speaker, will always be assigned as originating from the far left or far right speaker, which is known as the "end effect". Consequently, the end speakers might be considered to introduce an increased bias (Hartmann et al., 1998). However, adding dummy speakers at the ends of the array might avoid the end effect (Hartmann et al., 1998; Bosman et al. 2001). Another solution is the use of a wrapped span, in which the speakers are all around the listener forming a complete circle, although, in this case, complicated effects have to be addressed, since the multidimensional character of the task has to be dealt with by considering front-back confusions as qualitatively distinct from azimuthal uncertainty. This means that using the frontal horizontal arc only is suitable for measuring azimuth errors (Wightman and Kistler, 1989). Even if the use of a terminated span might possibly have introduced a localisation bias, the same setup was used in all experiments in this thesis; consequently, all of our tested conditions would likely affected similarly.

Another set-up concern is that, although the participants were asked not to move their heads, their heads were not physically fixed, which would produce slight head movements; thereby creating a change in ITD and ILD cues which would yield additional information about the location of the source. Wightman and Kistler (1999) investigated the effect of head movements by presenting their listeners with stimuli where the listeners' own HRTFs were added over head phones. Head movements were restricted in the first condition and encouraged in the second condition. Results revealed that head movements improved localisation performance and reduced the rate of front back errors. However, in our study, listeners, who were asked not to move their heads, were monitored via CCTV; consequently, the results are valid.

The reverberant conditions used in this study were mostly simulated in the anechoic room, therefore they lack the properties found in natural reverberant environments. In the reverberation simulations in this study, the same reverberant component was transmitted from the four corner speakers, which makes the reverberation sound coherent and can form standing wave patterns, thereby introducing nodes around the listener's head at certain frequencies. This means that head movement can affect the stimulus conditions, therefore, we ensured that the subjects' heads were not moving during sound delivery. By contrast, in natural reverberant environments, the reverberation is the combined effect of reflections from different directions in the room, therefore the standing wave patterns are diffuse.

Moreover, the lack of de-correlation in the signals in the simulated reverberation can result in interferences between the sound waves, which can result in spectral cancellation. Therefore, the spatial attributes of the reverberation field is not a perfect simulation of natural reverberation.

In conclusion, although in our study, both simulated and real reverberant environments showed similar negative results for the normal hearing listeners– in that reverberation had no significant effect in localising speech stimuli – it can not be assumed that they were identical and would produce similar results in other aspects or with hearing impaired listeners who generally show more effect of reverberation when compared to normal hearing listeners.

Another limitation of our experimental design was that the generality of the conclusions from the hearing impaired participants in the study is limited, since they all used one type of hearing aid –Siemens Prisma 2 Pro – which, although it is broadly used and represents a wide range of digital hearing aids, it has its own distinctive algorithms that distinguish it from the other hearing aids, thereby potentially affecting localisation cues differently. However, all digital hearing aids encounter compression as one of their main features, thus most, if not all, will cause a similar impairment in ILD cues. On the other hand, because all participants used the same aid, the power and consistency of the results were probably improved. By contrast, Van den Bogaert et al.'s participants used a variety of instruments, which makes interpretation more difficult.

On the other hand, though, the fact that all participants used the same aid probably improved the power and consistency of the results. By contrast, Van den Bogaert et al's participants all used a variety of instruments, which makes interpretation more difficult.

One more "hearing-aid issue" was that, in the study, the hearing aids were not re-programmed prior to testing. Consequently, if there was a small change in the audiogram, the fitting would no longer be optimal. The argument against making changes to the settings was that acclimatisation to the hearing aids was shown to have an important effect on sound localisation (Noble and Byrne, 1990 and Gatehouse, 1992), however, the extent to which their optimal fitting has an effect on localisation has not been discussed in the literature. Therefore, using optimal setting might only have a small effect.

9.2 Applicability of the findings

The primary purpose of this study was to investigate the effect of reverberation, which represents a realistic listening environment, on the ability of bilateral hearing aid wearers to localise speech and non-speech stimuli. Another purpose was to investigate the effect of hearing aids, in this case the Siemens Prisma 2 Pro, in utilising the ILD and ITD cues in both anechoic and reverberant environments. The following are the two main findings of clinical significance arising from the results;

Firstly, simulated reverberation has a negative impact on the ability of hearing aid users to localise different kinds of stimuli, in particular, low-pass pink noise. The effect of simulated reverberation on speech stimulus was significant in the aided conditions. For daily life requirements, speech localisation might be considered to be the most important aspect of sound source localisation in reverberant environments; therefore, based on the above findings, it can be concluded that the effect of simulated reverberation on hearing aid wearers impacts on their daily lives. Thus, in clinical settings, adding reverberation to simulate more realistic environments might be helpful in certain clinical measurements. However, another factor to be considered is the practical costs associated with including reverberation in clinical settings. Therefore, one must take into account the goal of the clinical tests and the resources available in order to determine whether to include reverberation or not.

Additionally, in reverberant public places, such as theatres and public halls, one might consider designing absorbent walls, which will reduce the reverberation time of the low frequencies.

Secondly, results showed that the use of hearing aids has a detrimental effect on, (a) localising sound sources in both anechoic and simulated reverberant environments and, (b) that aided localisation was worse in the simulated reverberant environments than in the anechoic environment. More detailed analysis revealed that high-frequency stimuli are strongly affected by aiding, hence ILD and spectral cues are disrupted the most. A possible explanation for the reduced intensity differences between both ears is the compression systems of hearing aids, therefore, if their compression systems can be linked in such a way that the ILD remains unaffected, localisation performance of high-frequency and broadband signals might be expected to improve.

9.3 Areas for possible future research

Because hearing aid users in this study received distorted binaural localisation cues, this could lead to degraded speech perception in noisy environments, therefore, more work could be done to preserve binaural acoustical cues in digital hearing aids. For example, in order to make general improvements resulting from the findings from this study, future studies could evaluate the effects of different compression systems. Moreover, further research could be done to discover a setting in which the compression of both aids are coordinated in such a way that ILD cues remain consistent between the two ears.

Because of the focus on ITD and ILD cues of hearing impaired listeners, this study has been limited to the frontal horizontal arc. Therefore, it would be beneficial for future research to investigate the influence of reverberation on front–back confusions, or vertical localisation, which are more closely related to spectral cues, thereby offering an insight into the effects of microphone placement.

Finally, in order to reduce the impact of reverberation on listening environments, further research into developing de-reverberation algorithms would prove well worthwhile. It would also be of benefit if such algorithms could be implanted in hearing aids in order to improve their performance in reverberant environments.

9.4 Conclusions

1) The effects of amplification and reverberation :

    - Hearing impaired listeners perform worse when they localise sounds in simulated reverberant environments than in anechoic environments.
    - Hearing impaired listeners' performance deteriorates further when they use bilateral hearing aids
    - objective physical and subjective psychoacoustical measurements in this study demonstrate that specifically localisation performance of high frequency sounds is impaired with amplification, because ILD cues are compromised
    - localisation ability of bilateral hearing aid users could be improved by linking the compression systems in both hearing aids via a central processor, so that ILD cues would remain consistent between both ears
    - hearing impaired listeners experience a deterioration in localisation performance in the simulated reverberant environment when sounds mainly contained low frequency components, suggesting that ITD cues are compromised by reverberation

2) The effects of reverberation and hearing loss on localisation performance:

    - unaided hearing impaired listeners show decreased performance in simulated reverberation when compared to normal hearing listeners, specifically, when sounds come from lateral directions.

3) Normal hearing listeners:

- normal hearing listeners are not much affected by reverberation when localising speech, however they are moderately affected when localising pure tones
- In reverberant environments , sound localisation is probably aided by envelope modulations of the broadband noise

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Appendices

Appendix A: Health questionnaire for normal hearing listeners

Personal details:

1	Your surname	
2	Your forenames	
3	Your date of birth	
4	Your age	
5	sex	
6	Your contact phone number	
7	Your e-mail Address	

Are you right handed or left handed?...

General health:

1	Have you ever had any serious illness or operation? If yes, what?	Yes/no
2	Are you presently receiving treatment involving any medicine or treatment? If yes, what?	Yes/no

Ear and hearing:

1	Do you think you have difficulty hearing in either ear?	Yes/no
2	Do you wear or have you ever been advised to wear a hearing aid?	Yes/no
3	Does your hearing fluctuate other than you have a cold?	Yes/no
4	Have you ever had surgery to either ear?	Yes/no
5	Do you suffer from tinnitus (noises, such as ringing, whistling or shushing in the ears)?	Yes/no
6	Do you have trouble with your balance or do you get vertigo?	Yes/no
7	Are you experiencing or have you recently had any of the following: □ı Pain in either ear □□ Discharge (running) from either ear □□ Inflammation in either ear □□ A blockage in either ear □□ A injury to either ear □□ A cold or flu	Yes/no

8	Have you ever had a head injury requiring a stay in hospital?	Yes/no
9	Have you been exposed to loud noise in the past 2 days?	Yes/no

History of listening tests:

1	Do you have any previous experience with listening or hearing tests?	Yes/no
2	Do you have any previous experience with localisation tests in particular?	Yes/no

Best time for participating in the experiment:…………………………………………………………………………… ………………………………………………………………………………………… ……………………

Name of participant: ……………………………………………………………...
Date of completion;………………………………………………………………

Appendix B: Hearing impaired listeners

B.1 Consent Form

Centre Number:
Study Number:
Patient Identification Number for this trial:

Title of Project: Effects of amplification and reverberation on sound localisation

Name of Researcher: Hadeel AlSaleh

Please initial box

1. I confirm that I have read and understand the information sheet dated .... (version ....) for the above study and have had the opportunity to ask questions.

2. I understand that my participation is voluntary and that I am free to withdraw at any time, without giving any reason, without my medical care or legal rights being affected.

3. I understand that sections of any of my medical notes may be looked at by responsible individuals from University of Southampton ISVR or from regulatory authorities where it is relevant to my taking part in research. I give permission for these individuals to have access to my records.

I agree to take part in the above study.

Name of Patient | Date | Signature

Name of Person taking consent (if different from researcher) | Date | Signature

Researcher | Date | Signature

1 for patient; 1 for researcher; 1 to be kept with hospital notes

B.2 Information sheet for research participants

Dear Sir or Madam,

You are being invited to take part in a research study entitled 'Effects of amplification and reverberation on sound localisation' which has been approved by the Southampton and South West Hampshire Research Ethics Committee (B) (Project Reference number: 08/H0504/2)

The study is a part of a PhD I am doing at the Institute of Sound and Vibration Research at the University of Southampton. Before you decide it is important for you to understand why the research is being done and what it will involve. Please take time to read the following information carefully and discuss it with others if you wish. Ask us if there is anything that is not clear or if you would like more information. Please take time to decide whether or not you wish to take part.

Thank you for reading this.

Purpose of the trial

The purpose of the study is to understand how the hearing aids you are currently using affect your ability to pinpoint where a sound is coming from in normal reverberant environments (rooms with high levels of echo). Different hearing aids will subtly change certain characteristics of the sounds you listen to. The study will take place over the course of six month, however you will be asked to visit the University only once for a two hour appointment (which will include breaks).

Why have I been chosen?

You have been invited along with 60 others to participate in this study based on certain factors, which include the duration of your experience with your hearing aids and the degree of your hearing loss.

Do I have to take part?

Taking part in this study is entirely voluntary. It is up to you to decide whether or not to take part. If you decide to take part you will be asked to sign a consent form. A copy of your signed consent form and this information sheet will be given to you to keep. Even after deciding to take part, you are still free to withdraw at any time and without a reason. ***A decision to withdraw at any time, or a decision not to take part, will not affect the standard of care you receive from the NHS***

What will happen to me if I take part and what will I have to do?

If you decide to participate in this study, please complete the attached form indicating the times when you are generally available. A stamped addressed envelope is provided. I will arrange an appointment at a time suitable to you. Completing the form does not commit you to the trial.

You will be asked to attend the University for one session, which will last two hours at the most (including breaks). All assessments will be carried out at the Hearing and Balance Centre of the Institute of Sound and Vibration Research, at the University of Southampton. An experienced audiologist will undertake the assessment. Travel expenses will be reimbursed. Arrangements will be made for a taxi to collect you and return you home. Alternatively, you may make your own arrangements and the costs will be reimbursed.

During your appointment at the clinic, you will be seated within an arc of 21 speakers. A sound will be emitted from a speaker by random, and you will be asked to indicate on a handheld computer, which speaker you think the sound came from. There are a range of different sounds to represent everyday noises, however, none of the tests are at all unpleasant or uncomfortable.

What are possible disadvantages and benefits to taking part?

Participation in this research study will not require any lifestyle restrictions, and there are no disadvantages or risks of taking part. The information we get from this study may help us treat patients better. You will receive a letter of the research outcome if you want to.

What if something goes wrong?

If you are harmed by taking part in this research project, there are no special compensation arrangements. If you are harmed due to someone's negligence, then you may have grounds for a legal action but you may have to pay for it. Regardless of this, if you wish to complain, or have any concerns about any aspect of the way you have been approached or treated during the course of this study, the normal National Health Service complaints mechanisms should be available to you.

Will my participation in this study be kept confidential?

All information which is collected about you during the course of the research will be kept strictly confidential. Your name and address will be removed from any information about yourself which leaves the hospital so that you cannot be recognized from it

What will happen to the results of the research study?

The results of this research study will be published in peer reviewed scientific journals which are accessible to the research participants. A brief report will be made available if you so request. You will not be identified in any report/publication.

Who is organising and funding the research, and who had reviewed the study?
The University of Southampton, Institute of Sound and Vibration Research is funding this research, and it has been reviewed by the Institute of Sound and Vibration Research Human Experimentation Safety and Ethics Committee and the Southampton and South West Hampshire Research Ethics Committee (B) (*Project Reference number*: 08/H0504/2)

Contact for further information
If you require any further information or clarification, please do not hesitate to contact me. My contact details are as follows:

Hadeel AlSaleh
University of Southampton, ISVR

Tel: 02380592921
Fax: 02380593190
Email: has@isvr.soton.ac.uk

Thank you for taking the time to read this information, and I greatly appreciate your time and contribution to this study. I am looking forward to hearing from you.

Yours sincerely,

Hadeel AlSaleh

B.3 Research protocol in non-technical language

Introduction and aims:

Our ability to detect where a sound is coming from in the horizontal plane relies on two different sources of information: the difference in time of arrival of the sound at the two different ears, and the level differences at the two ears. Theses time and level differences can be considered as cues to localise sounds, and they are called inter-aural time difference (ITD) cues and inter-aural level difference (ILD) cues. The term intereaural can be thought of as "between ears". These cues can be modified by different amplification devices and might be altered by reverberation. In reverberant environments listeners receive conflicting localisation cues: Some sound waves travel directly from the sound source to the ear, where others are reflected off the walls, ceiling and surfaces of the room. The purpose of this study is to consider the effects of various amplification devices on the listener's localisation performance in the horizontal plane in reverberant environments, to understand the contribution and salience of ITD and ILD cues to the localisation performance of various amplification device users. There is the possibility that hearing aids distort ITD and ILD cues in reverberant environments.

Subjects:

A total of 60 subjects will be tested. Sixty experienced hearing aid users will be recruited from the regular throughput of the Audiology Department at the Royal South Hants Hospital, Southampton and the Audiology Department at the Royal Hampshire County Hospital in Winchester. Of those subjects, 20 subjects will be linear, analogue hearing aid users and the remaining 40 will be digital hearing aid users. Each subject will be asked to attend the University for one session only and will be tested with the localisation rig at the ISVR at University of Southampton.

Method:

Different signals will be tested. Some that contain mainly ILD cues, some that contain mainly ITD cues and others that contain both .Of those signals, some will contain reverberation as well.

Localisation ability of subjects will be tested using above stimuli presented at a normal listening level (60 dBSPL). Twenty one speakers will be arranged in a 180 degree arc, such that the angle of separation between each speaker is 9 degrees. Subjects will be seated in the centre of this array, 1.5m away from the speakers. Stimuli will be played from a signal speaker selected at random. The subject's task will be to identify the source of the stimulus.

Additional data on normal hearing people has already been collected at ISVR and will be compared with that obtained from hearing aided subjects from the present study.

Appendix C

Table 1: The age, gender and hearing thresholds of hearing impaired participants. Hearing thresholds were measured in dB HL

Subject	Gender	Age	Ear	250	500	1000	2000	4000	6000	8000
1	M	77	R	30	35	60	65	75	65	75
			L	35	35	35	75	75	70	70
2	M	79	R	40	40	45	60	65	65	95
			L	45	55	65	65	70	70	80
3	M	80	R	35	50	60	60	50	55	65
			L	35	55	55	60	60	65	65
4	M	87	R	30	30	45	60	80	70	75
			L	30	40	65	70	80	75	70
5	F	70	R	35	40	45	40	55	85	85
			L	30	35	40	50	65	65	85
6	F	60	R	15	15	25	60	55	60	75
			L	15	15	25	60	50	60	70
7	M	82	R	25	35	50	50	55	75	85
			L	30	30	45	65	65	70	90
8	M	75	R	15	30	40	60	65	70	100
			L	30	20	35	60	70	90	80
9	F	63	R	10	10	35	45	50	55	60
			L	15	25	45	55	55	65	65
10	F	65	R	30	45	55	40	55	60	50
			L	25	25	25	35	45	50	45
11	M	80	R	35	35	55	75	70	70	70
			L	35	45	60	65	80	75	75
12	f	81	R	25	30	40	55	60	65	70
			L	30	30	35	45	50	60	60
13	F	81	R	35	35	35	50	55	65	80
			L	40	35	30	35	55	75	85
14	M	72	R	20	15	20	20	55	75	75
			L	35	30	25	55	60	105	95
15	M	71	R	20	20	20	25	50	70	70
			L	15	20	35	45	60	75	75
16	M	74	R	40	40	40	50	70	70	70
			L	30	30	35	55	65	65	75
17	M	67	R	15	15	30	55	70	60	65
			L	20	30	45	65	60	80	75
18	F	73	R	30	35	40	50	65	65	70
			L	45	45	55	60	65	85	85
19	M	75	R	35	45	60	75	70	80	90
			L	20	35	40	70	80	85	85
20	f	63	R	30	30	30	45	60	80	55
			L	30	30	35	40	70	90	60
21	M	80	R	15	30	45	55	70	75	80
			L	20	35	65	75	90	95	100
22	F	55	R	25	15	25	60	75	75	70
			L	20	15	30	60	60	80	70
23	M	80	R	30	45	45	65	70	95	110
			L	30	35	50	60	70	105	100
24	F	83	R	65	65	65	65	75	80	80
			L	70	70	60	70	75	90	90
25	GF	71	R	65	60	55	65	65	80	105
			L	35	40	55	65	70	80	90
26	M	80	R	25	25	25	40	90	105	95
			L	20	15	20	40	85	105	90
27	M	81	R	20	20	30	50	75	75	75
			L	30	20	25	50	75	85	85
28	M	87	R	30	40	55	70	75	75	90
			L	30	30	45	65	75	70	85