Effect of Reverberation on Speech Identification Using Digital Hearing Aids

Sreela P. K.[1] & Devi N.[2]

Abstract

The major consequence of sensory-neural hearing loss is communicative difficulty; particularly in noisy and reverberant listening environments. The acoustical variables that affect the perception of speech include level of background noise, reverberation time (RT), distance from the speaker to the listener and the interaction among these variables. The aim of the study was to compare the speech identification scores using reverberated and non-reverberated speech stimuli and to compare them between analog and digital hearing aid with Echostop feature. The speech stimuli were reverberated in three different RTs such as 0.6secs (R1), 1.2secs (R2) and 2secs (R3). Twenty normal hearing individuals and twenty hearing impaired subjects were selected in the age range of 40-70years. Moderate to moderately severe sensory-neural hearing loss subjects were participated in the hearing impaired group. The results suggest that in subjects with hearing impairment the better scores were obtained with digital hearing aid (with Echostop feature) than with the analog hearing aids across conditions i.e., in non-reverberant and reverberant conditions. Also the normal hearing individuals got the better scores in three reverberant conditions and in non-reverberant conditions. So it can be concluded from the study that in adults with sensorineural hearing loss, the use of a digital hearing aid (with Echostop feature) in reverberant conditions will help in better speech identification than with a analog hearing aid.

Key words: *Reverberation, speech identification, digital hearing aid, echostop.*

The major consequence of sensori-neural hearing loss (SNHL) is communicative difficulty particularly in noisy or reverberant listening environments (Crandell, Henoch, & Dunkerson, 1991; Needleman & Crandell, 1995). Due to deleterious effects of SNHL on communication, research has indicated that individuals with hearing impairment may exhibit reduced psychosocial function, such as increased feelings of isolation depression, loneliness, anger, fear frustration and disappointment (Crandell, 1988; Bess et al, 1989; Christian et al, 1989). Because of broad range of potential psychosocial and physical difficulties associated with hearing loss, it is important that the audiologist not limit intervention to hearing aids alone.

In addition to the acoustical environment, speech perception in a classroom can also be decreased by reductions in the hearing sensitivity or auditory processing abilities of the child. To understand why children experience speech perception difficulties in the classroom, it is important that disciplines working in the educational setting (such as audiologists, speech-language pathologists, reading specialists, regular and special education teachers, teachers of the deaf and hearing impaired, and psychologists) have an adequate knowledge base concerning the acoustical variables that can compromise the perception of speech. These acoustical variables include; level of the background noise, level of the speech signal relative to the level of the background noise, Reverberation Time (RT), distance from the speaker to the listener and the interaction among these variables.

Background room noise is another important factor. Background noise refers to any auditory disturbance within the room that interferes with what a listener wants to hear (Crandell & Smaldino, 1995). Background noise sources in the classroom includes, external noise, internal noise and room noise. Background noise in a room can compromise speech perception by masking the acoustic and linguistic cues that are available in the teacher's the message. The spectral energy of consonants is less intense than vowel energy. Consequently, background noise in the classroom predominately reduces consonant perception. The capability of classroom noise to mask the teacher's speech depends on a number of acoustical parameters (Nabelek, 1982; Nabelek & Nabelek, 1994). These parameters include the long-term spectrum of the noise, intensity fluctuations of the noise over time, and the intensity of the noise relative to the intensity of speech.

Speech perception ability is highest at favorable SNRs and decreases as a function of reduction in SNR (Crum, 1974; Finitzo-Hieber & Tillman, 1978; Nabelek & Pickett, 1974a, 1974b). The range of SNRs for classrooms has been reported to be from approximately +5 dB to -7 dB (Sanders, 1965; Paul, 1967; Blair, 1977; Markides, 1986; Finitzo Hieber, 1988). An additional acoustical variable that can detrimentally affect speech perception in the classroom is reverberation. Reverberation refers to

[1] e-mail: sree.audio@gmail.com, [2] Lecturer in Audiology, AIISH; email: deviaiish@gmail.com

the persistence or prolongation of sound within an enclosure as sound waves reflect off of hard surfaces (Lochner & Burger, 1964; Nabelek & Pickett, 1974a, 1974b; Siebein, 1994; Siebein, Crandell, & Gold, 1997). This prolongation of sound is usually considered the most important acoustical consideration that defines the acoustical climate of a classroom.

All rooms have some reverberation. Audiometric test booths usually exhibit RTs of approximately 0.2 seconds. Living rooms and offices often have RTs between 0.4 and 0.8 seconds (Nabelek & Nabelek, 1994). RTs for classrooms usually range from 0.4 to 1.2 seconds (Bradley,1986; Crandell, 1992; Crandell & Smaldino,1995; Finitzo-Hieber,1988), while auditoriums, churches and assembly halls may have RTs in excess of 3.0 or 4.0 seconds (Crandell,1992; Nabelek & Nabelek,1994; Siebein,1997).

Reverberation degrades speech perception through masking of the directly transmitted sounds. Reverberant speech energy reaches the listener some time after the direct sounds (Bolt & McDonald, 1949; Nabelek & Pickett, 1974a; Nabelek, 1982). This results in a "smearing" or masking of the directly transmitted speech signal. Speech perception tends to decrease as RT increases (Finitzo-Hieber & Tillman,1978; Gelfand & Silman,1979; Nabelek & Pickett,1974). Speech perception in adults with normal hearing is not compromised until RT exceeds 1.0second (Crum, 1974; Gelfand & Silman, 1979; Nabelek & Pickett, 1974). Listeners with SNHL, however need considerably shorter RTs (0.4 to 0.5 seconds) for maximum speech perception (Crandell, 1991, 1992; Crandell & Bess, 1986; Finitzo-Hieber & Tillman, 1978; Olsen, 1988).

In the classroom setting, noise and reverberation combine synergistically to affect speech perception (Crandell & Bess, 1986; Crum, 1974; Finitzo-Hieber & Tillman, 1978; Nabelek & Pickett, 1974a; 1974b). These synergistic effects appear to occur because when noise and reverberation are combined, reflections fill in the temporal gaps in the noise, making it more steady state in nature (Crandell & Bess,1986; Crandell & Smaldino,1995; Crum,1974; Finitzo-Hieber & Tillman,1978; Nabelek & Pickett,1974).

Another factor that influences speech perception in the classroom is the distance from the teacher to the student. At distances relatively close to the child, the direct sound field predominates in the listening environment. In this sound field, sound waves are transmitted from the teacher to the child with minimal interference from room surfaces. The indirect sound field originates at the "critical distance" of the room. The critical distance of the room refers to the point in the room where the level of the direct sound and the level of the reverberant sound are essentially equal.

At slightly longer distances, early sound reflections reach the listener. These are those sound waves that arrive at a listener within a very short time after the direct sound. Most early reflections strikes minimal room surfaces on their path from speaker to listener. Early sound reflections usually combine with the direct sound and may increase the perceived loudness of the sound, thereby improving speech perception in listeners with normal hearing (Bradley, 1986; Lochner & Burger,1964; Nabelek & Nabelek,1994).

The methods to improve speech perception in a room include, acoustic modification of the room, using 'clear' speech procedures, reduction of speaker-listener distance, optimizing visual communication and personal and group amplification systems.

Hearing aids often offer little or no benefit in noisy or reverberation environments (Crandell, 1991). This is not surprising because traditional amplification by itself does little to improve the SNR of the listening environment. This limitation of hearing aids may has changed, because several SNR-enhancing options for hearing aids have recently been introduced that may help the listener in noisy or reverberant environments.

As we are aware that there are various variables that effect the speech perception ability of an individual. There are also various ways to overcome these variables in terms of room settings, personal settings, hearing aids etc. Hearing aid is one of the best ways which are very helpful for good perception of speech, but still there are various parameters that are to be considered while selecting the appropriate amplification device. A hearing aid that would provide maximum benefit using the following measures should be sought (Carhart, 1946). Effective gain (using SRT stimuli), best speech discrimination (using PB-50 word lists), tolerance limits (under headphones) and efficiency in background noise.

The digital noise reduction methods that are used in digital hearing aids to reduce the background noise, but does these digital noise reduction methods improves perception of speech in reverberation? So there is a need to study the digital noise reduction method that is used to reduce the effect of reverberation on speech identification.

The aim of the present study is: (1) to find out the effect of reverberation on speech identification in adult hearing aid users, (2) to compare the speech identification scores using analog and digital hearing aid (with Echostop feature) for reverberated and non-reverberated speech stimuli, (3) to compare the effect of different reverberation time in normal individuals, in analog hearing aid users & in digital hearing aid users.

It is easy for the professional to fit the hearing aid at normal hearing situation. But under degraded situation in terms of increased ambient noise, poor SNR and reverberant conditions, selection and fitting is a challenging factor. With advancement in technology there are different types of digital noise reduction methods used in digital hearing aids to reduce the background noise and to increase the speech-in-noise performance.

The ultimate goals for noise reduction algorithms are to increase listening comfort and speech intelligibility. Most of the high-performance hearing aids have some type of noise reduction algorithms; whereas, only a few have speech-enhancement algorithms.

Depending on the type of modulation detection used, noise reduction algorithms are divided into two categories: multichannel adaptive noise reduction algorithms that detect the slow modulation in speech, and synchrony-detection noise reduction algorithms that detect the co-modulation in speech.

Most of the noise reduction algorithms in commercial hearing aids use the multichannel adaptive noise reduction strategy. These algorithms are intended to reduce noise interference at frequency channels with noise dominance. In theory, multichannel adaptive noise reduction algorithms are the most effective in their noise reduction efforts when there are spectral differences between speech and noise. The major limitation of these noise reduction algorithms is that they cannot differentiate between the desired signal and the unwanted noise if speech is the competing noise.

The second category of noise reduction algorithms detects the fast modulation of speech across frequency channels and takes advantage of the temporal separation between speech and noise. The rationale is that the energy of speech sounds is co-modulated by the opening and closing of the vocal folds during the voicing of vowels and voiced consonants (i.e., the fast modulation of speech). Noise, on the other hand, is rarely co-modulated. The rate of co-modulation is the fundamental frequency of the human voice, which ranges from 100 to 250 Hz for adults and up to 500 Hz for children.

One strategy for improving speech perception in noise with hearing aids is to shape the frequency response of the hearing aid by emphasizing the high frequencies and reducing the lows. Such frequency response shaping may reduce the upward spread of masking effects of low frequency noise. In a room with abundant steady-state low-frequency noise, this shift in frequency response of the amplification system can effectively improve the SNR.

This is to enhance the SNR of the listening environment. The ASP strategies alter the electro-acoustic characteristics of the hearing aid in response to a specific environment. Commonly used ASPs include (a) conventional single channel, automatic gain control; (b) adaptive compression; (c) multichannel compression; (d) bass increases at low levels (BILL); and (e) treble increases at low levels (TILL). These strategies benefit some listeners in noise but not others.

The differential sensitivity of a directional microphone can provide a 2 to 4dB improvement in SNR if the desired signal is originating from a less-sensitive location.

The advantages of a directional microphone, however, may be compromised in a reverberant room because both the desired sound and reflections from the background noise can arrive simultaneously at the microphone's most sensitive azimuth. Dual microphones use two closely matched omni-directional microphones connected by an electronic delay. This can improve SNR by about 6.5dB to 8dB compared with an omni-directional microphone. Beamforming refers to two microphones whose outputs are digitally compared in time and intensity. Beam formers can be sensitive to a wide variety of azimuths, and it is possible to optimize the microphone response for specific listening situations and environmental conditions. Because of the flexibility of the beam formers, they are the most promising innovation for improving SNR in a variety of difficult listening environments.

A microphone or group of microphones with more than one entry port is often referred to as a beamforming array or as a microphone array. Microphone arrays can be classified into two broad types: those that have directional characteristics that do not vary from moment to moment (fixed arrays), and those that adapt to the environment in such a way as to minimize the pick-up of noise coming from particular directions (adaptive arrays).

Adaptive filtering works extremely well under certain circumstances. In particular where there is no only one noise source, no reverberation, and a very poor SNR, the adaptive filter can change it's characteristics so that the directivity pattern of the array has an almost perfect null in the direction of the noise. Under these favourable circumstances, the SNR can be improved by as much as 30dB.

Reverberation greatly decreases the effectiveness of adaptive arrays. Unless the wanted talker is very close, reverberation will cause significant speech energy to arrive from all directions. Consequently, the noise reference signal will contain speech as well as noise. This mixture makes it difficult for the filter to adapt, thus reducing the effectiveness of the noise cancelling. The beamformer can be modified in various ways to minimize, but not totally avoid, these difficulties. In one of these modifications, a speech/non-speech detector, based on the more pulsatile nature and/or overall level of speech signals, is used to stop the adaptive filter from changing its response whenever speech is believed to be present. In another modification, the noise canceller is preceded by another adaptive filter that is used to remove as much speech as possible from the noise. This other adaptive filter adapts only when speech is present. By creating such a reference signal that has as little speech in it as possible, the main adaptive filter has a much better chance of removing noise.

The presence of reverberation also means that the echoes from a single sound will arrive at the hearing aid for some time after the direct wave arrives. These echoes can be removed only if the adaptive filter is sufficiently complex to store and combine sounds that arrived perhaps many hundreds of milliseconds before. Such complex filters take a longer to adapt.

Echostop: This is a feature used in digital hearing aids to combat problems caused by room reverberation and echo. It automatically activated whenever the instrument is in directionality mode, it analyzes and adapts to the properties of the room where the patient is in. It makes sure patient can hear every syllable of the conversation, no matter how poor the acoustics are.

EchoStop can even monitor and cancel sound reflections caused by the body. The sounds that hear are reflected by shoulders, hair, and other parts of the body. EchoStop monitors and removes these uncomfortable and confusing reflections, thereby improving the overall directional performance of hearing aid. This further improves performance in challenging sound environments.

Reverberation in rooms with moderate reverberation time upto 0.5sec does not impair speech understanding for normal – hearing listeners (Crum & Tillman, 1973), but reduces speech understanding for hearing impaired subjects (Finitzo-Hieber & Tillman, 1978). A combination of noise and reverberation has a substantial effect on speech perception for both normal hearing and hearing impaired subjects (Nabelek & Pickett, 1974a). The combined effects are greater than the effect of reverberation and noise measured separately (Finitzo-Hieber & Tillman, 1978). Plomp (1976) reported great variability in the perception of reverberated speech within groups of normal hearing and hearing impaired subjects.

Johnson (2000) studied 'Children's phoneme identification in reverberation and noise'. They assessed the consonant and vowel identification abilities of listeners aged 6 years through young adulthood in four listening conditions of (a) control (no reverberation ,no noise), (b) reverberation only (i.e.,1.3s), (c) noise only (i.e., +3db S/N), and (d) reverberation plus noise. The results of this study show that children's consonant identification abilities may not reach adult- like levels of performance until the mid-to-late teenage years in listening situations involving reverberation plus noise.

It has been suggested that reverberation be treated as noise when modeling the effects of room acoustics on speech perception. However, reverberation and noise produce different kinds of listening effects or distortions. Whereas noise masks less intense cues, reverberation causes a prolongation or smearing of sounds (both within and between phonemes). In addition, reverberation smoothes the envelope of the speech signal and increases low frequency energy that can mask speech information. It is not therefore surprising that reverberation and noise produce different error patterns (Nabelek, Letowski, & Tucker, 1989; Takata & Nabelek, 1990).

Some data suggest that the multiplicative nature of simultaneous distortions also exists for the combination of reverberation and noise. That is, the actual degradation that occurs when reverberation and noise are combined is greater than would be predicted by the sum of the decline in scores produced by each of these distortions in isolation (Moncur & Dirks, 1967; Helfer, 1992).

Speech in a sound field with reflections is perceived differently than is speech in a free field condition without reflections. Reverberation in rooms with moderate reverberation time, T upto 0.5secs does not impair speech understanding for normal hearing listeners (Crum & Tillman, 1973) but reduces speech understanding for hearing impaired subjects (Finitze-Hieber & Tillman, 1978).

Plomp (1976) reported great variability in the perception of reverberated speech within groups of normal hearing and hearing-impaired subjects. Hodgson (1986) in reviewing several studies relating to reverberation and intelligibility summarized them as, speech intelligibility generally decreases as reverberation time increases. Hearing-impaired persons perform more poorly in reverberant settings than normal-hearing persons. Reverberation disrupts intelligibility in noise more than in quiet. Binaural performance is better than monaural in a noisy reverberant room.

Studies evaluating the effects of reverberation show that an increase in reverberation time distorts the speech signal, resulting in a significant decrease in speech perception (Nabelek & Pickett, 1974a; Duquesnoy & Plomp, 1980; Helfer, 1991).

Reverberation appears to act as a masking noise in reducing speech intelligibility. The reflected energy apparatus by overlapping the direct (original) speech signal so that perceptual cues are masked. Gelfand and Silman (1979) demonstrated that reverberation results in rather systematic error patterns reminiscent of those associated with masking by noise and low-pass filtering, thus supporting the concept that reverberation act as a masker.

The decrease in speech perception depends on the amount of reverberation, the distance between talker and listener, and the level and type of noise in the room. Even in a quiet room, speech intelligibility for normal hearing people gradually decreases with an increase of reverberation time (Nabelek & Pickett, 1974a; Helfer & Wilber, 1990). People with hearing loss, normal hearing children, and the elderly cannot tolerate as much reverberation as can young adults with normal hearing (Nabelek & Robinson, 1982; Loven & Collins, 1988).

People with normal hearing show some decline in speech perception, but this decline is not as large as that received by people with a sensori-neural hearing loss (Nabelek & Pickett, 1974a). The explanation may be that people with a hearing impairment are less capable of integrating the direct & reverberant sounds (Nabelek & Pickett, 1974a). Tests conducted with people with a hearing impairment show a decrease in the perception of vowels, consonants (Helfer & Wilber, 1990), non-sense words (Danhauer & Johnson, 1991), words with carrier phrases (Nabelek & Pickett, 1974; Nabelek & Robinson, 1982), words (Finitze-Hieber & Tillman, 1978; Hawkins & Yacullo, 1984;) and sentences Duquesnoy & Plomp, 1980; Yacullo & Hawkins, 1987; Pekkarinen & Viljauen, 1990;) with reverberation time. The commonality is that despite differences in design all studies report that an increase in reverberation time leads to a reduction in speech perception ability.

Harris and Swenson (1990) concluded that the negative effects of reverberation were greater for those with poorer sensitivity such as observed in people with a severe to profound hearing impairment.

Method

Present study was designed to compare the speech identification scores of individuals with normal hearing & individuals with hearing impairment using analog hearing aid & digital hearing aid for reverberated and non-reverberated speech stimuli.

Participants: Two groups of subjects participated in the study. (1) Group I included 20 normal hearing individuals. (2) Group II included 20 individuals with hearing impairment.

Participant selection criteria for Group I: Twenty normal hearing individuals satisfying the following criteria were included in this group in the age range – 40 to 70 years. Their mother tongue was Kannada having pure tone average of $\leq$15dB. Air conduction thresholds of $\leq$15dB & Bone conduction thresholds of $\leq$10dB. No Otological & Neurological indications.

Participant selection criteria for Group II: Twenty individuals in the age range of 40-70 years having mother tongue as Kannada. Their thresholds were moderate to moderately-severe sensori-neural hearing loss in the test ear. Speech identification scores of 75% in unaided condition. They are naive digital BTE hearing aid users. Prior permission was taken from the subjects who participated in the study.

Test environment: The testing was carried out in sound treated room with the ambient noise levels were within permissible limits (ANSI S3.1-1991).

Instrumentation: Calibrated two channel diagnostic audiometer (OB922) was used for estimation of pure tone thresholds. Calibrated GSI-tympstar middle ear analyzer was used for immittance measurements. An analog hearing aid and a digital hearing aid (with "Echostop" feature) were used for the purpose of comparison of performance. Hearing aids were programmed with NOAH (version 3) based software. Hearing aids were connected with the computer using HiPro. The stimuli were played from a CD through computer and routed through the OB922 two channel audiometer to the sound calibrated Martin audio C115 speaker which is placed at zero degree azimuth.

Stimuli: The test stimulus contains four word lists, each with 25 bi-syllabic words, which was phonetically balanced.

Procedure

The procedure included 2 phases;

Phase I - Development of phonetically balanced word list in Kannada.

Phase II - Compare speech identification scores of individuals using analog and digital hearing aids (with Echostop) for reverberated and non-reverberated speech stimuli.

Phase I: In order to develop the word list, initially 600 bi-syllabic Kannada words was selected. The selected words were given to 30 adult native Kannada speakers in the age range of 18-40years. The subjects were asked to rate the familiarity on a three point scale according to their usage in daily living situation;

0-50 - Unfamiliar, 50 – 75 % - Familiar & greater than 100 % - Highly familiar.

Ninety highly familiar and 10 familiar words were selected and phonemically balanced using Ramakrishna, Nair, Chiplunkar, Atal, Ramachandran and Subramanian (1965). Four word lists were made from the 100 words which were phonetically balanced. Each word list contained 25 bi-syllabic words.

The words were spoken in conversational style by a female native speaker of Kannada. They were digitally recorded in a sound treated room, on Adobe audition version 3.0, using 44100 sampling frequency and 32 bit analog to digital converter. These recorded word lists were normalized. Then each of the four word lists was reverberated in three reverberation times of 0.6sec, 1.2sec and 2 sec using Adobe audition version 3.0. The recorded word lists were then transferred to 4 CDs in order to randomize the order of presentation. Each of the CD contains one list of non - reverberated speech stimuli and three lists of reverberated speech stimuli, each of which were reverberated in three different reverberation times.

Phase II: Pure tone thresholds were obtained using modified Hughson & Westlake procedure (Carhart & Jerger, 1959), across octave frequencies from 250 to 8000 Hz for air conduction and 250 to 4000 Hz for bone conduction.

Tympanometric measurements were done using 226Hz probe tone. This was done to rule out conductive hearing loss due to middle ear pathology. Appropriate tips were used to obtain seal and comfortable pressure for the subject. The parameters documented were types of tymapanogram and acoustic reflex thresholds agreeing with ear canal volume, acoustic admittance and the tympanometric peak pressure. The volume of analogue hearing aid was kept constant at two. The digital hearing aid was programmed on the basis of audiometric thresholds with the target default gain provided by the software. While programming the following features were choosen appropriately.

The noise management option was switched off in order to avoid any unwanted effect on result. The hearing aid was switched to omni - directional microphone mode as there was no need of noise reduction during the testing. Echostop in digital hearing aid gets on automatically.

Test was done in acoustically treated room with noise within permissible limits as per ANSI (1991) specification. Subjects were seated at a distance of one meter and at 0 degree azimuth from the speaker. The testing was done using non-reverberated speech stimuli and reverberated speech stimuli and the speech identification scores were found out. The presentation of stimuli was randomized. During testing hearing aids were also selected randomly for fitment and testing. The intensity level was maintained at 40 dB SL throughout the testing and inter-stimulus interval was kept constant at 5 seconds. The responses were scored by an adult Kannada speaker.

In Group I the speech identification scores were found out using non-reverberated and reverberated speech stimuli. In Group II, the aided speech identification scores were found out using non-reverberated & reverberated speech stimuli and with both analogue and digital hearing aids. Appropriate statistical method was used to analyze the results.

Results & Discussion

The data obtained was analysed using Statistical Package for the Social Sciences (SPSS) version 16 software.

Comparison of the SIS between control and experimental group (analog and digital hearing aid users) across different reverberant conditions: Mixed ANOVA was carried out to find out the difference between SIS obtained using experimental and control group.

Comparison of SIS between analog hearing aid & control group: The mean & standard deviation of SIS obtained for those with analog hearing aid & in control group in different conditions are shown in the following Table 1. The Mean and standard deviation indicates that there is difference in SIS obtained in both control & experimental groups and also in different conditions.

Comparison of SIS between digital hearing aid & control group: The mean & standard deviation obtained using digital hearing aid (with Echostop) hearing aid & in control group in different conditions are shown in the following Table 1. The Mean and standard deviation indicates that there is difference in SIS obtained in both control & experimental groups and also in different conditions. The mean and SD of SIS shows that the SIS obtained using digital hearing aid is better scores than the SIS obtained using analog hearing aid and in control group.

The mean SIS indicates that there is a significant difference in the scores obtained using the digital hearing aid and the control group. This also indicates that as reverberation time increases the speech identification scores obtained by the both groups are decreasing.

Table 1. Mean and SD of SIS for the experimental groups (analog HA & digital HA) and for control group

Group	Conditions		Mean	SD
Control	Non-reverberant		99.60	1.23
	R1 (0.6 sec)		73.40	8.24
	R2 (1.2 sec)		56.00	10.62
	R3 (2 sec)		44.40	12.71
Experimental	Non-reverberant	Analog HA	77.25	16.32
		Digital HA	93.40	5.84
	R1 (0.6 sec)	Analog HA	15.40	13.81
		Digital HA	29.00	20.02
	R2(1.2 sec)	Analog HA	7.40	10.32
		Digital HA	22.60	14.64
	R3 (2 sec)	Analog HA	5.60	8.35
		Digital HA	18.40	15.32

Note: HA – Hearing aid.

Independent t - test was carried out to compare the mean and SD of SIS between control & two experimental groups (i.e. using analog and digital hearing aid). It was found that there is a statistically significant difference in the scores obtained in control and experimental groups. In all the conditions control subjects scored better than the other two experimental groups.

Figure 1 shows that the control group has got the better SIS than the other two experimental conditions (Analog and digital). The control group (normal hearing individual) got the better scores and the hearing impaired subjects got the poor scores in non-reverberant and also in three reverberant conditions (i.e. in 0.6s, 1.2s & 2s). As suggested by Nabelek and Pickett (1974a) that people with normal hearing show some decline in speech perception, but this decline is not as large as that received by people with a sensorineural hearing loss

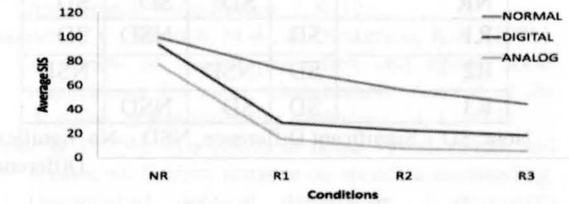

Figure 1. Comparison of average SIS between control and experimental groups.

Comparison of SIS in different reverberation conditions for analog hearing aid and digital hearing aid users & in control group: Repeated measure ANOVA was carried out to find out the average SIS obtained in control & experimental groups in non-reverberant condition & three reverberant conditions as follows.

Comparison of SIS for analog hearing aid across conditions: The results shows that there is a statistically significant difference between the scores obtained by the analog hearing aid across conditions ($F = 202.164, P<0.05$). This result is shown in Table 2. This also shows that there is no statistically significant difference between the scores obtained in the third and fourth conditions (i.e. RT of 1.2sec & RT of 2secs respectively).

Table 2. Comparison of SIS obtained by analog hearing aid across conditions

ANALOG (I)	ANALOG (J) Mean difference (I - J)			
	NR	R1	R2	R3
NR	-	SD	SD	SD
R1	SD	-	SD	SD
R2	SD	SD	-	NSD
R3	SD	SD	NSD	-

Note: SD – Significant Difference, NSD – No Significant Difference

Comparison of SIS for digital hearing aid across conditions: The results shows that there is a statistically significant difference between the scores obtained by the digital hearing aid (with Echostop) across conditions ($F = 314.883$, $p<0.05$). This also shows that there is no statistically significant difference between the scores obtained in the second & third conditions (i.e. RT of 6sec & RT of 1.2secs respectively) and also between third & fourth (i.e. RT of 1.2sec & RT of 2secs respectively.

Table 3. Comparison of SIS obtained by digital hearing aid across conditions

DIGITAL (I)	DIGITAL (J) Mean difference (I - J)			
	NR	R1	R2	R3
NR	-	SD	SD	SD
R1	SD	-	NSD	SD
R2	SD	NSD	-	NSD
R3	SD	SD	NSD	-

Note: SD – Significant Difference, NSD – No Significant Difference.

Comparison of SIS in normal hearing individuals across conditions: The results (Table 4) shows that there is a statistically significant difference in speech idenfication scores obtained in control group across conditions ($F = 181.202$, $p<0.05$).

Pair wise comparison of SIS between analog and digital hearing aid across conditions: Paired t-test was carried out to compare the SIS obtained by the digital and analog hearing aid in different conditions. The results (Table 5) shows that there is a statistically significant difference between the scores obtained using digital and analog hearing aid in each of the conditions such as non-reverberant and in three different reverberation times (i.e. 0.6s, 1.2s & 2s).

Table 4. Comparison of SIS obtained in control group across conditions

NORMAL (I)	NORMAL (J) Mean difference (I - J)			
	NR	R1	R2	R3
NR		SD	SD	SD
R1	SD		SD	SD
R2	SD	SD	-	SD
R3	SD	SD	SD	-

Note: SD – Significant Difference, NSD – No Significant Difference

Comparing the SIS between both groups, the scores are good for non-reverberated condition than in the other three reverberant conditions. And it was also found out that the SI scores decreased with an increase in reverberation time. The poorer scores obtained with a RT of 2secs.

Few studies have involved adults with a severe or greater hearing impairment. They have reported that an increase in reverberation time resulted in a significant reduction in speech perception (Harris & Reitz, 1985). Listeners with SNHL, however, need considerably shorter RTs (i.e., 0.4 to 0.5sec) for maximum speech recognition (Finitzo-Hieber & Tillman, 1978; Crandell & Bess, 1986; Olsen, 1988; Finitzo-Hieber, 1988; Crandell, 1991, 1992).

Table 5. Pair wise comparison of SIS between *digital and analog hearing aids across conditions*

Pair	t	p
ANR-DNR	4.122	0.001
AR1-DR1	-4.289	0.000
AR2-DR2	-4.601	0.000
AR3-DR3	-3.547	0.002

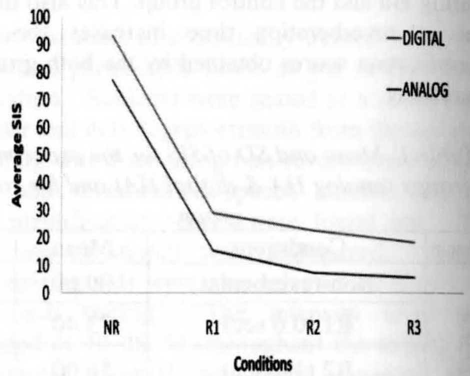

Figure 4. Comparison of average SIS obtained by analog and digital hearing aid.

Figure 2 indicates the average SIS obtained by analog and digital hearing aid. This shows that the digital hearing aid (with Echostop) has got the better scores across conditions i.e. in non-reverberant and in reverberant conditions.

In subjects with hearing impairment the better scores were obtained with digital hearing aid (with Echostop feature) than with the analog hearing aids. This may be because of the Echostop feature in the digital hearing aid got automatically activated in the reverberant situation. The Echostop feature helps to remove the reflections that interfere with the speech identification.

Nabelek and Pickett (1974 b) studied the effect of noise and reverberation on speech discrimination through analog hearing aids. He found that hearing impaired subjects performance was decreased than the normal hearing subject's performance.

Conclusions

The study compared the difference in SIS between control and experimental group (analog and digital hearing aid users) across different reverberant conditions and found out that they are significantly different. The study also compared the SI scores in different reverberant conditions and also in non-reverberant condition for analog and digital hearing aid (with Echostop feature). It was found that the scores are better with digital hearing aid (with Echostop feature) than with the analog hearing aid.

The SIS were found to be better in non-reverberant condition than in the reverberant condition in the control and experimental group. It was also found out that the SIS were poor as the reverberation time increases in both the control and in the experimental group. So from the study it can be concluded that the SIS in non-reverberant condition and in reverberant conditions is better in normal hearing individuals than in sensorineural hearing loss cases.

The better scores are obtained in non-reverberant condition than in reverberant conditions. With the digital hearing aid (with Echostop feature) the speech identification scores are better than with analog hearing aid.

As the reverberation time increases the SI scores became poorer. This is in the order of NR > R1 > R2 > R3, Where, NR – Non-reverberant condition. R1-Reverberation time of 0.6sec, R2 – Reverberation time of 1.2sec and R3 – Reverberation time of 2s.

So it can be concluded from the study that in adults with sensorineural hearing loss, the use of a digital hearing aid (with Echostop feature) in reverberant conditions will help in better speech identification than with an analog hearing aid.

References

American National Standards Institute (1991). *Maximum permissible ambient noise levels for audiometric test rooms*. ANSI S3.1, New York: American National Standards Institute.

Blair, J. (1977). Effects of amplification, speech reading, and classroom environment on reception of speech. *Volta Review, 79*,443-449.

Bolt, R., & McDonald, A. (1949). Theory of speech masking by reverberation. *Journal of the Acoustical Society of America, 21*, 577–580.

Bradley, J. (1986). Speech intelligibility studies in classrooms. *Journal of the Acoustical Society of America, 80*, 846–854.

Carhart, R. (1946).Tests for the selection of hearing aids. *Laryngoscope*, 56:780-794.

Carole E. Johnson. (2000). Children's Phoneme Identification in Reverberation and Noise. *Journal of Speech, Language and Hearing Research,* 43,144-157.

Crandell, C., & Bess, F. (1986). Speech recognition of children in a "typical" classroom setting. *American Speech-Language and Hearing Association, 29*, 87.

Crandell, C. S. (1988). Social contagion of binge eating. *Journal of Personality and Social Psychology*, 55, 588-598.

Crandell, C. (1991). Classroom acoustics for normal-hearing children: Implications for rehabilitation. *Educational Audiology Monographs, 2*, 18–38.

Crandell, C. (1992). Classroom acoustics for hearing-impaired children. *Journal of the Acoustical Society of America, 92*, 2470.

Crandell, C., & Bess, F. (1986). Speech recognition of children in a "typical" classroom setting. *American Speech-Language and Hearing Association, 29*, 87.

Crandell, C., & Smaldino, J. (1995). An update of classroom acoustics for children with hearing impairment. *Volta Review, 1*, 4–12.

Crandell, C.C., Henoch, M.A., & Dunkerson, K.A. (1991). A review of speech perception and aging: Some implications for aural rehabilitation. *Journal of the Academy of Rehabilitative Audiology, 24*, 121-132.

Crum, D. (1974). The effects of noise, reverberation, and speaker to- listener distance on speech understanding. Unpublished doctoral dissertation, Northwestern University, Evanston, IL.

Crum, M. A., & Tillman, T. W. (1973). Effects of speaker-to-listener distance upon speech intelligibility in reverberation and noise. Paper presented at the American Speech and Hearing Association Convention, Detroit.

Danhauer, J. L., & Johnson, C. E. (1991). Perceptual features for normal listeners' phoneme recognition in a reverberant lecture hall. *Journal of the American Academy of Audiology*, 2, 91-98.

Duquesnoy, A., & Plomp, R. (1980). Effect of reverberation and noise on the intelligibility of sentences in cases of presbycusis. *Journal of the Acoustical Society of America, 68, 537-544.*

Finitzo-Hieber, T. (1988). *Classroom acoustics. In R. Roeser (Ed.), Auditory disorders in school children* (221–233) (2nd ed.). New York: Thieme-Stratton.

Finitzo-Hieber, T., & Tillman, T. (1978). Room acoustics effects on monosyllabic word discrimination ability for normal and hearing-impaired children. *Journal of Speech and Hearing Research, 21*, 440–458.

Gelfand, S., & Silman. S. (1979). Effects of small room reverberation upon the recognition of some consonant features. *Journal of Acoustical Society of America; 66(1)*:22-29.

Harris, R. W., & Reitz, M. L. (1985). Effect of room reverberation and noise on speech discrimination by the elderly. Audiology, 24, 319-324.

Harris, R. W., & Swenson, D. W. (1990). Effects of reverberation and noise on speech recognition by adults with various amounts of sensorineural hearing impairment. *Audiology, 29, 314-321.*

Hawkins, D., & Yacullo, W. (1984). Signal-to-noise advantage of binaural hearing aids and directional microphones under different levels of reverberation. *Journal of Speech and Hearing Disorders, 49,* 279–286.

Helfer, K. S. (1991). Everyday speech understanding by older listeners. *Journal of the Academy of Rehabilitative Audiology,* 24, 17-34.

Helfer, K. S. (1992). Aging and the binaural advantage in reverberation and noise. *Journal of Speech and Hearing Research,* 35, 1394-1401.

Helfer, K. S., & Wilber, L. A. (1990). Hearing loss, aging, and speech perception in reverberation and noise. *Journal of Speech and Hearing Research*, 33, 149-155.

Hodgson, W. R. (1986). Special cases of hearing aid assessment. In W. R. Hodgson (Ed.), *Hearing aid assessment and use in audiologic rehabilitation* (191-216). (3rd Ed.). Baltimore, Williams & Wilkins.

Leavitt, R., & Flexer, C. (1991). Speech degradation as measured by the Rapid Speech Transmission Index (RASTI). *Ear and Hearing, 12, 115–118.*

Lochner, J., & Burger, J. (1964). The influence of reflections in auditorium acoustics. *Journal of Sound and Vibration, 4, 426–454.*

Loven, F. C., & Collins, M. J. (1988). Reverberation, masking, filtering, and level effects on speech recognition performance. *Journal of Speech and Hearing Research, 31, 681-695.*

Markides, A. (1986). Speech levels and speech-to-noise ratios. *British Journal of Audiology, 20, 115–120.*

Moncur, J., & Dirks, D. (1967). Binaural and monaural speech intelligibility in reverberation. *Journal of Speech and Hearing Research,* 10, 186-195.

Nabelek, A. (1982). Temporal distortions and noise considerations. In G. Studebaker & F. Bess (Eds.), *The Vanderbilt hearing aid report: State of the art research needs* (1242–1248). Upper Darby, PA: Monographs in Contemporary Audiology.

Nabelek, A. K., & Robinson, P.K. (1982). Monaural and binaural speech perception in reverberation for listeners of various ages, *Journal of Acoustical Society of America.* 71, 1242-1248.

Nabelek, A. K., Letowski, T. R., & Tucker, F. M. (1989). Reverberant overlap-and self-masking in consonant identification. *Journal of the Acoustical Society of America,* 86, 1259-1265.

Nabelek, A., & Nabelek, I. (1994). Room acoustics and speech perception. In J. Katz (Ed.), *Handbook of clinical audiology* (624–637) (4th Ed.). Baltimore, MD: Williams & Wilkins.

Nabelek, A., & Pickett, J. (1974a). Monaural and binaural speech perception through hearing aids under noise and reverberation with normal and hearing-impaired listeners. *Journal of Speech and Hearing Research, 17,* 724–739.

Nabelek, A., & Pickett, J. (1974b). Reception of consonants in a classroom as affected by monaural and binaural listening, noise, reverberation, and hearing aids. *Journal of the Acoustical Society of America, 56,* 628–639.

Needleman & Crandell. (1995). Speech recognition in noise by hearing impaired & noise-masked normal hearing listeners. *Journal of American Academy of Audiology, 6(6):* 414-424.

Olsen, W. (1988). Classroom acoustics for hearing-impaired children. In F. Bess (Ed.), *Hearing impairment in children* (266–277). Parkton, MD: York Press.

Paul, R. (1967). *An investigation of the effectiveness of hearing aid amplification in regular and special classrooms under instructional conditions.* Unpublished doctoral dissertation, Wayne State University, Detroit, MI.

Pekkarinen, E., & Viljanen, V. (1990). Effect of sound absorbing treatments on speech discrimination in rooms. *Audiology,* 29, 219-227.

Plomp, R. (1976). Binaural and monaural speech intelligibility of connected discourse in reverberation as a function of azimuth of a single competing sound source (speech or noise). *Acoustica,* 34, 200-211.

Ramakrishna, B. S., Nair, K. K., Chiplunkar, V. N., Atal, B. S., Ramachandran, V., Subramanian, R. (1965). Transcription efficiencies of Indian scripts. In *Some aspects of the relative efficiencies of Indian languages,* Catholic Press, Ranchi, 34-35.

Sanders, D. (1965). Noise conditions in normal school classrooms. *Exceptional Children, 31,* 344-353.

Siebein, G. (1994). *Acoustics in buildings: A tutorial on architectural acoustics.* New York: Acoustical Society of America.

Siebein, G., Crandell, C., & Gold, M. (1997). *Principles of classroom acoustics: Reverberation.* Educational Audiology Monographs, *5,* 32–43.

Takata, Y., & Nabelek, A.K. (1990). English consonant recognition in noise and in reverberation by Japanese and American listeners. *Journal of the Acoustical Society of America, 88,* 663-666.