EFFECTS OF COMPRESSION RELEASE TIME IN HEARING AID ON ACOUSTIC AND BEHAVIOURAL MEASURES OF SPEECH

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May 2012.

Dedicated To,

My Lovely Parents ☺

&

Teachers

CERTIFICATE

This is to certify that this dissertation entitled **Effects of compression release time in hearing aid on acoustic and behavioural measures of speech** is the bonafide work submitted in part fulfillment for the Degree of Master of Science (Audiology) of the student with Registration No.: 10AUD003. This has been carried out under the guidance of a faculty of this institute and has not been submitted earlier to any other University for the award of any other Diploma or Degree.

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DECLARATION

This is to certify that this Master's dissertation entitled **Effects of compression release time in hearing aid on acoustic and behavioural measures of speech** is the result of my own study under the guidance of Prof. P. Manjula, Professor in Audiology, Department of Audiology, All India Institute of Speech and Hearing, Mysore, and has not been submitted earlier in other University for the award of any Diploma or Degree.

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CHAPTER I

INTRODUCTION

A cochlear hearing loss implies a poorer functioning of the compressive non-linearity mechanism. As a result, the quality of the auditory neural information deteriorates. Although the amount of neural activity can be increased by using hearing aids, the loss of quality cannot be compensated. Auditory dysfunction due to a loss in cochlear non-linearity can be described in several psycho-acoustical terms, such as a reduced spectral and temporal resolution, a disturbed loudness perception and an increased temporal and spectral masking (Oxenham & Bacon, 2003).

Individuals with outer-hair cell damage or loss of cochlear non-linearity suffer from an increased growth of loudness or loudness recruitment. This is mainly because of the reduced dynamic range of individuals with hearing impairment. A loss of cochlear non-linearity also results in increased temporal masking effects (Oxenham & Bacon, 2003). In addition, a properly functioning cochlear non-linearity sharpens the auditory filters. A loss of non-linearity results in a broadening of the auditory filters (Patterson & Moore, 1988), resulting in reduced spectral processing. Therefore, it becomes more difficult for the ear with impairment to resolve the spectral information of a broad-band signal and to extract a signal from a background noise.

As the amount of cochlear hearing loss increases, problems such as reduced dynamic range, spectral resolution, temporal masking also increases. Hence, to combat such problems caused due to cochlear hearing loss, different kinds of amplification and signal processing techniques have been employed in hearing devices. Amongst them, dynamic range compression has been introduced in hearing aids as a possible solution to a few of the above mentioned problems. With the introduction of digital techniques in hearing aids, compression has become a standard processing technique in modern hearing aid design. Its main function is to provide sufficient amplification at low input levels without overloading the auditory system at high input levels. In this way, listening comfort is ensured as the hearing aid compensates for the reduced dynamic range.

In general, compression systems are not particularly designed to compensate for a reduced frequency resolution and upward spread of masking (USOM). Noise suppression and spectral enhancement (Lyzenga, Festen, & Houtgast, 2002) are more appropriate signal-processing techniques to compensate for this loss. Nevertheless, compression in at least two separate frequency channels may suppress high-level low-frequency sounds and simultaneously increase the level of weak high-frequency cues. This might help to compensate for increased USOM.

Different types of compression strategies are available including, Automatic Gain Control (AGC), Wide Dynamic Range Compression (WDRC), and Compression Limiting (CL). Most hearing aids have WDRC which gives more gain for weak sounds than for intense sounds. The WDRC compresses most of the speech spectrum into the residual range giving increased audibility and comfort and making loudness perception more similar to normal (Villchur, 1973).

Substantial changes in a speech signal can occur as the result of signal processing by hearing aids. Different signal processing strategies result in acoustic modification of the speech signal, in both spectral and temporal aspects. These changes might affect the speech perception in individuals with hearing impairment.

Compressing the speech signal into a very small dynamic range using a wide dynamic range compression (WDRC) for an individual with severe hearing loss might have detrimental effects on speech intelligibility by reducing the depth of amplitude modulation in speech by introducing distortion in temporal envelopes, and by reducing spectral contrast (Plomp, 1988; Stone, & Moore, 2004). Thus, different compression settings in a hearing aid are known to cause different effects on the signal. Such parameters of compression include knee-point or compression threshold, compression ratio, attack time, and release time.

Research so far, however, has revealed equivocal findings in providing guidelines on the correct setting of release time. Literature reviews on compression provide no compelling answers with regard to release time (Dillon, 1996; Hickson, 1994). Two approaches have been used to study the release time. They are studying the effects of this parameter on speech intelligibility, and on quality or user preference.

With regard to intelligibility, a few available studies conflict with each other showing either no difference among settings of different release times (Bentler & Nelson, 1997) or that moderate values (between 60 and 150 ms) provide better intelligibility than very short or very long release times (Walker & Dillon, 1982).

Studies of user preference showed mixed results. Some show no user preferences for any release time (Bentler, & Nelson, 1997; Neuman, Bakke, Mackersie, Hellman, & Levitt,1995), while others rated longer release times (anywhere from 200 to 4,000 ms) to be more pleasant than shorter release times (Hansen, 2002; Neuman, Bakke, Mackersie, Hellman, & Levitt, 1998).

Two primary factors may contribute to the expected benefit from changing release time. They include audibility and distortion. Some researchers have hypothesized that short release times should be beneficial for speech recognition as it increases the level of low-intensity consonants, thereby making them more audible and less susceptible to upward spread of masking by high-intensity vowels (Dreschler, 1988; Gatehouse, Naylor, & Elberling, 2003; Verschuure, Mass, Stikvoort, Jong, Goedegbure & Dreschler, 1996). Other researchers have hypothesized that the reduction in intelligibility with short release time compression to be due to an alteration of level differences within the speech utterance (Freyman, Nerbonne, & Cote, 1991; Hedrick, & Rice, 2000; Hickson, & Byrne, 1997). Temporal parameters are also subjected to changes in terms of temporal envelope and consonant-vowel ratio (Van Tasell, 1993). Shorter compression time constant causes continuous and rapid fluctuations in the gain and this would lead to spread of energy to the adjacent frequency components thereby, inducing spectral distortions (Wang, 2001).

To address the issue of an optimal release time for hearing aid fittings, it is necessary to undertake a comprehensive analysis of the acoustic effects of compression on speech and the resulting effects on speech intelligibility and quality. The acoustic analysis must focus on the aspects of speech that are mostly likely to be affected by varying the release time.

Different acoustic measures are available to quantify the effect of changes in the acoustic properties of the signal. They are Envelope Difference Index (EDI), CVR (Consonant-to-Vowel Ratio) and Spectral Distance/Distortion Measure (SDM).

The EDI is an index of change in the temporal envelope between two signals. The value ranges from 0 to 1, where 1 represents completely dissimilar envelopes and 0 represents perfectly similar envelopes. This technique has been developed for precisely quantifying the temporal contrasts that exists between two sound samples. This technique is based on envelope subtraction, and generates an Envelope Difference Index that may help to clarify whether alteration of the natural speech envelope via amplification improves or degrades speech intelligibility. The Envelope Difference Index method may also be used to assess hearing aid saturation, and may have other applications as well. The technique is applicable whenever a precise quantification of the difference between two temporal envelopes is required, regardless of stimulus duration. Fortune, Woodruff, and Preves (1994) have demonstrated in their experiment the EDI value for syllables processed through linear and non-linear amplifiers. EDI values were smaller in linear amplifier as compared to non-linear.

SDM produce measurements of dissimilarities seen in two speech spectra. Distance measures can be used in order to evaluate the quality of speech. One such distance measure could be the Log-likelihood Ratio (LLR). The LLR is a spectral distance measure which mainly models the mismatch between the formats of the original and enhanced signals (Rao, Murthy, & Rao, 2011). Spectral Distance Measures have been applied to speech processing and speech recognition fields. Jeon and Lee (2008) have used LLR for objectively analyzing the spectral distortions caused due to spectral enhancement technique used in hearing aids. As the weightage of the enhancement increased, LLR values also increased.

1.1 Need for the Study

As mentioned earlier, compression hearing aids have often adjustable release time. Research till date does not provide a consensus on how to choose or set release time. Two approaches have been used to study the effect of release time i.e., studying the effects of release time on speech intelligibility and on quality or user preference and results of these approaches have lead to inconclusive decisions about the appropriate release times. There is a dearth of studies on acoustic effects of compression release time.

Jenstad and Souza (2005) measured the effect of short- and long- release times on two acoustic measures on individual syllables - the envelope difference index (EDI) and CV ratio (CVR). These measurements allowed for quantification of the short-term amplitude characteristics of the speech signal and the changes to these amplitude characteristics caused by compression. They correlated the acoustic changes to the speech intelligibility perceived by listeners. They found that, inspite of significant changes observed in acoustic measurements due to varying the release times (12 ms short release time and 800 ms long release time), it did not correlate with the perceived intelligibility of speech. Hence, it can be inferred that the listeners could have used information other than temporal cues to identify the non-sense syllables.

Studies have used different release times with only a single input level (Neuman, Bakke, Mackersie, Hellman, & Levitt, 1995; Novick, Bentler, Dittberner, & Flamme, 2001). Because compression is triggered by changes in input level, the advantages or disadvantages of varying the release times of compression can only be explored by altering the input level to the hearing aid.

Studies have revealed that not all sounds are equally susceptible due to distortions of temporal cues. The greatest effect is on sounds where critical information is carried by variations in sound amplitude over time, such as stops which can be modeled as a series of temporal cues with falling or raising burst spectrum, onset of voicing, and air release.

The dynamic range for individuals with hearing loss varies in concordance with their degree of hearing loss, with lesser dynamic range for individuals with severe hearing loss. As the degree of hearing loss increases, the amount of spectral resolution decreases. Hence, individuals with hearing loss may rely more on the temporal information for speech recognition. Thus, there is a need for different settings in compression parameters which maintains the temporal cues for speech recognition, for individuals with different degrees of loss. The listeners who have normal or near-normal spectral discrimination ability should be able to extract sufficient spectral and contextual information to compensate for altered temporal cues (Van Tasell, 1993). The clinical impact may be greater for listeners who depend to a greater extent on temporal cues, most obviously, listeners with a severe-to-profound loss (Lamore, Verweij, & Brocaar, 1990; Moore, 1996).

1.2 Aim of the study

The present study was undertaken to investigate the acoustic and behavioural effects of compression release time.

1.3 Objectives of the study

1) To investigate the effect of compression release time (40 ms, 640 ms and 1280 ms) on temporal and spectral characteristics of the VCV tokens in aided condition, at three different presentation levels of 30, 45 and 65 dB HL.
2) To evaluate the effect of different compression release times (40 ms, 640 ms & 1280 ms) on the speech intelligibility at three different presentation levels (30, 45 and 65 dB HL), in individuals with hearing impairment.
3) To study the perceived quality judgments of speech through the hearing aid set at different compression release times (40 ms, 640 ms & 1280 ms) presented at three different levels (30, 45 and 65 dB HL), in individuals with hearing impairment.

CHAPTER II

REVIEW OF LITERATURE

The purpose of the study is to measure the effects of hearing aid compression release times on acoustic and behavioural measures of speech. The acoustic measures considered in the study are to measure temporal envelope distortion using Envelope Difference Index and spectral distortion using Log-likelihood Ratio. The behavioural measures considered in the study are speech identification scores and perceptual rating of quality for four parameters (loudness, clarity, naturalness and overall impression). The relevant literature for the study have been categorized under the following headings

2.1 Amplification strategies.

2.2 Characteristics of compression hearing aids.

2.3 Different forms of compression.

2.4 Rationales for use of compression.

2.5 Benefits and disadvantages of different compression systems.

2.6 Effects of compression on speech intelligibility, speech quality and speech acoustics.

2.1 Amplification strategies

A sensorineural hearing loss is usually associated with a phenomenon called recruitment. Individual with recruitment cannot hear soft sounds and may have difficulty in hearing medium level sounds, and yet hears loud sounds just as loudly as a normal-hearer. If a hearing aid applies the same gain to all input sounds, regardless of whether they are soft, medium or loud, medium sounds may be audible and comfortable, but soft sounds may still be inaudible and loud sounds may be uncomfortably loud. To overcome this, it is thus necessary to reduce the large range of input levels a hearing aid wearer encounters in everyday life into the smaller range of levels which they find audible and comfortable, i.e., the dynamic range. There are different kinds of amplification strategies to overcome with the loss of dynamic range.

The amplification strategies are broadly classified into linear and non-linear type.

2.1..1 Linear Amplification Strategy

In hearing aids, linear amplification strategy provides a constant gain to all input levels until the hearing aid's saturation limit is reached. Since speech includes a wide range of intensity levels, from low intensity consonants /f/ to high intensity vowels /a/ and whispered speech to shouting speech, the benefit of linear amplification gets restricted when low intensity sounds are made audible and meanwhile the high intensity sounds are amplified to the point of discomfort (Souza, 2002). In other words, linear amplification strategies have a limited capacity to maximize audibility across a range of input intensities.

2.1..2 Non-linear or Compression Amplification Strategies

To solve the problem of reduced dynamic range most of the hearing aids incorporate some form of compression amplification strategy in which gain is automatically adjusted based on the intensity of the input signal. As the input intensity increases, the amplification provided will be tapered. It is expected that an individual using a compression hearing aid perform better than those using peak clipping aids in a listening condition that includes wide range of speech levels (Benson, Clark, & Johnson, 1992; Moore, Johnson, Clark, & Pluvinage, 1992;). However, the benefit of compression amplification strategy is yet to be established well.

2.2 Characteristics of compression hearing aids

There are two main characteristics of compression hearing aids. They are:

1) Dynamic compression characteristics
2) Static compression characteristics

The dynamic aspects of compression hearing aid performance are known as “attack” and the “release” times. Attack time and release times are the time it takes for a compression circuit to respond to changes in the intensity of an input SPL. When the input SPL exceeds the knee-point of compression, the hearing aid attacks the sound by reducing the gain. Once the input sounds fall below the knee-point of compression, the hearing aid releases from compression and restores the gain.

Attack time is defined as the time taken for the output to stabilize to within 2 dB (IEC 60118-2) or 3 dB (ANSI S3.22) of its final level after the input of the hearing and increases from 55 to 80 dB SPL (IEC 60118-2) or 55 to 90 dB SPL (ANSI S3.22). The release time is the time taken for the compressor to react to a decrease in input level. Although attack and release times could be made to have extremely short values, the consequences are most undesirable. If the release time is too short, the gain will vary during each voice pitch periods, so the compressor will distort the waveform. If the attack time is made extremely short, and the release time is long, then distortion is minimal. The attack and release times have a major effect on how compressors affect the levels of the different syllables of speech. Depending on the time constants, compression can also be termed as slow acting and fast acting compression systems.

Static compression characteristics are compression threshold, compression ratio and compression range. Compression threshold is defined as the SPL above which hearing aid begins compressing. Compression ratio describes indirectly, how much the gain is decreased. Compression ratio describes indirectly, how much the gain decreases. Compression ratio is defined as the change in input level needed to produce a 1 dB change is output level. Compression range refers to the range of inputs over which compression occurs (Dillon, 2001).

2.3 Different forms of compression

Compression can occur at several stages and there are different types of compression.

2.3.1 Input stage and output stage:

The compression controlled from a point on the input side of the volume control, this arrangement is referred to as input stage compression. The volume control affecting the signal before the signal reaching the compressor is referred to as output stage compression.

2.3.2 Multichannel compression:

Multichannel hearing aids splits the incoming signal into different frequency bands and each band of signal passes through a different amplification channel, each channel contains compression circuit. This type of compression benefits individuals who have better hearing in some frequencies than other frequencies. Individuals who have good low frequency hearing although might require a lot more amplification in the high frequencies. Multi channel compression systems allows for the appropriate amount of compression to be applied at different frequencies.

2.3.3 Different types of compression:

Hearing aids with low compression thresholds, and low compression ratios are referred to as wide-dynamic range compression (WDRC). Hearing aids with high compression thresholds and high compression ratios are termed as compression limiting. Compression limiting is most often implemented in an output compression circuit. Compression ratios for WDRC aids are typically low (<5:1), while compression ratios for compression limiting aids are usually high (>8:1) (Walker & Dillon, 1982).

2.4 Rationales for use of compression

The following section outlines several theoretical reasons why compressors should be beneficial and the need to include compression in hearing aids.

2.4.1 Avoiding discomfort, distortion and damage:

As the input to the hearing aid increases, the signal cannot be amplified proportionally because this causes discomfort due to reduced dynamic range of the individual. Providing the gain proportionally at higher input levels also causes damage to the residual hearing capacities. The use of peak clipping to taper the output at higher levels causes distortions. Hence compression limiting , WDRC with appropriate time constants provides the audibility in the comfortable range with much lesser distortion as compared to peak clipping

2.4.2 Reducing inter-syllabic intensity differences

Among speech sounds, vowels tend to be more intense than consonants, and hence the vowel and the consonant will vary in the range of 30 dB. For individuals with hearing impairment, the weaker phonemes will be masked out by the intense ones. By introducing compression amplification, the weaker phonemes are amplified and the intense sounds such as vowels are compressed by providing lesser gain. Hence, maintaining the ratio of vowels and consonants, there by avoiding the temporal masking phenomenon. This can be well achieved by fast acting compression systems.

2.4.3 Normalizing loudness

Normalizing loudness is based on the equal loudness contour, that all the frequency sounds should sound equally loud. This is a potential factor required for individuals with sensorineural hearing loss in whom the perception of loudness is greatly affected. Normalizing the perception of loudness is possibly the most popular rationale for using compression. The most common way of achieving loudness normalization is with separate compressors located in each channel of a multichannel hearing aid.

2.4.4 Maximizing intelligibility

Multichannel Compression can be used to achieve in each frequency region the amount of audibility required to maximizes intelligibility.

2.4.5 Reducing noise

The noise usually has its energy concentration in the lower frequency region. Presence of noise can cause upward spread of masking there by masking high frequency components of speech. Compression in the low frequency channels can be used to reduce the amount of gain provided to these low frequency components, thus indirectly reducing the masking and increasing intelligibility of speech.

2.5 Benefits and disadvantages of different compression systems

The technical name for an automatic volume control that operates over a large range of input levels is Wide Dynamic Range Compression (WDRC). WDRC improves the audibility of soft speech sounds for the hearing aid wearer by applying more gain (or amplification) to soft sounds. This means loud sounds that are uncomfortable for the hearing aid wearer, are reduced. This also reduces the chance of damaging the wearer’s hearing if the hearing aid volume is set higher than the recommended level. As soft level sounds are made louder, the hearing aid wearer may find that many low level environmental noises such as air conditioners, fridges and computer noises are now audible to them (Humes, Christensen, Thomas, Bees, Williams, Bentler, 1999).

As recruitment gets worse as the hearing loss gets more severe, it would seem that people with severe/profound hearing loss in particular could benefit from WDRC. Applying WDRC to this population with the aim of presenting a large range of input level into the very narrow dynamic range of hearing would require the use of fairly high compression ratios. Unfortunately, the use of high compression ratios tends to severely distort important spectral and temporal cues of speech (Souza, Jenstad, & Folino, 2005).

Compression limiting is to serve as an output limiter to prevent discomfort-or hearing damage from high-level signals while limiting saturation distortion. Thus, this type of compression is an alternative to peak clipping. Electroacoustically, there are marked differences in distortion levels between linear peak clipping and compression limiting aids once input level plus gain exceeds the saturation threshold. The greater the amount of saturation, the stronger is the preference for compression limiting over peak clipping (Hawkins & Naidoo, 1993; Stelmachowicz, 1999). Therefore, for most listeners, compression limiting should be used rather than peak clipping. One potential exception is for listeners with a severe-to-profound loss. These listeners, who require maximum power and often are accustomed to wearing high gain linear aids in saturation, may report (at least initially) that compression limiting aids are not loud enough (Dawson, Dillon, & Battaglia, 1991).

2.6 Effects of compression on speech intelligibility, speech quality and speech acoustics

Speech perception in individuals with hearing impairment is hampered by two major factors, deficits in auditory processing as described in the previous section and a loss of audibility. A loss of audibility implies that certain low-level speech sounds cannot be perceived as they are received at sub-threshold levels. A good model has been developed to relate audibility of speech to speech understanding, called the Speech Intelligibility Index (SII, ANSI S3.79, 1998).This model predicts an almost linear relationship between both factors. A weighing function is used to incorporate the relative contribution of each frequency band to speech perception. Using this model, a loss of audibility of relevant speech signals will generally result in poorer speech intelligibility. Although the SII gives a good prediction on average, individuals may behave differently due to differences in supra-threshold processing capacities. For the more severely impaired ears an extra distortion factor is needed, which is also related to a poor processing quality. Especially for severe high-frequency losses, speech intelligibility does not always improve when increasing the audibility at high frequency components (Ching, Dillon & Byrne, 1998; Turner & Cummings, 1999). This implies that next to audibility there is a second important supra-threshold factor that influences speech intelligibility. It is assumed that this factor is related to deficits in auditory processing caused by the cochlear damage i.e., temporal masking, reduced spectral resolution (Moore, 1996).

Dynamic range compression can be used to compensate for each of the two factors. Compression is ideally suited to increase audibility without making the high-level sounds uncomfortably loud. In addition, compression can also be used to compensate for deficits in auditory speech perception. This is one of the most challenging targets. Speech sounds vary widely in intensity level even at normal vocalization. By applying same gain to all speech components (vowels and consonants), weaker speech sounds may be masked by the higher intensity sounds in a forward or backward temporal masking (Festen & Plomp, 1983; Moore, 1985) which may result in decreased speech intelligibility. Compression hearing aids with high gain and a low compression threshold (45 dB SPL) would ensure that the more intense vowel sounds receive less gain than the less intense consonant sounds. This could reduce the intensity difference between the more intense vowels and the less intense consonants and thus decrease the consonant-to-vowel ratio.

It is also possible that compression distorts some speech cues, offsetting the benefits of improved audibility. Recently, interest has been renewed in the importance of temporal cues for speech intelligibility with the speculation that these cues are disrupted by time constants of WDRC. Temporal cues include the variations in speech amplitude over time and range from the very slow variations of the amplitude envelope to the rapid "fine-structure" fluctuations in formant patterns or voicing pulses. With regard to compression, attention has been focused on fluctuations in the amplitude envelope, in part because alteration of the amplitude envelope is the most prominent temporal effect of time constants in WDRC. The amplitude envelope contains information about manner and voicing (Rosen, 1992; Van Tasell, 1992) and some cues to prosody and also the suprasegmentals of speech (Rosen, 1992). Compression alters the variations in the amplitude envelope and reduces the contrast between high-intensity and low-intensity speech sounds. Of course, the reduced intensity variation is a desirable effect of compression. However, because both individuals with normal-hearing and those with hearing impairment listeners can extract identification information from amplitude envelope variations (Turner, Souza & Forget., 1995), it is possible that alterations of these cues could affect speech intelligibility.

Compressing the speech signal into a very small dynamic range using a wide dynamic range compression (WDRC) for an individual with severe hearing loss might has detrimental effects on speech intelligibility by reducing the depth of amplitude modulation in speech by introducing distortion in temporal envelopes, and reducing spectral contrast (Plomp 1988; Stone & Moore, 2004).

According to Moore (1987), the overall level of speech varies over a range of 30 dB. For speech at a constant average level, the levels of individual acoustic elements of the speech vary over a range of 30 dB. Reduced dynamic range in individuals with hearing impairment causes the range of speech levels to indeed reduce by using compression amplification. The reduction of the speech range depends on the compression parameters of the amplification system, most notably the number of compression channels, compression threshold, compression ratio, attack time, release time.

Since individuals with hearing impairment have lesser dynamic range, broadened auditory filters, they suffer with reduced spectral resolution or impaired frequency selectivity. Improving the audibility does not compensate for reduced spectral resolution or impaired frequency selectivity. In addition, as the level of the signal is enhanced, the spectral resolution becomes more impaired and this also leads to spread of masking. Hence, it can be concluded that individuals with hearing impairment, rely more on temporal cues than spectral cues. Hence it is necessary to maintain the temporal cues of the hearing aid processed speech in order to compensate for their loss. In the WDRC compression system the major factors influencing the temporal information of speech, such as envelope are compression time constants i.e., attack and release time (Venn, Souza, Brennan, & Stecker, 2009)

With respect to compression time constants, attack and release times can be varied form fast to slow time constants. If attack time and release time have been made to have extremely short values, the consequences on envelope and spectral contents are most undesirable. If the release time is too short, then the gain will vary during each voice pitch period, so the compressor will distort the waveform. If the release time is made longer, rapid gain fluctuations will be reduced and thus the distortions would be minimal. However, for the intense signal which is of brief duration, short attack time causes rapid reduction in gain. But, long release time has its undesirable effect by having the gain to remain low for a longer time after the brief sound has gone. Attack times in hearing aids are commonly around 5 ms, release times are rarely less than 20 ms, and may be much longer (Kuk, 1996).

Findings of research so far, however, have been inconclusive in providing guidelines on the correct setting of release time. Literature reviews on compression provide no compelling answers with regard to release time (Dillon, 1996; Hickson, 1994).

Two approaches have been used to study release time of compression in hearing aids. They are behavioural approach for studying the effects of this parameter on speech intelligibility, quality or user preference and acoustic approach for studying the effects of acoustics cues in speech.

2.6.1. Effects of time constants on speech intelligibility

With regard to intelligibility, equivocal findings have been reported in literature. Specifically, studies on speech intelligibility for different release times show either no difference among settings or that moderate values (between 60 and 150 ms) provide better intelligibility than very short or very long release times (Walker & Dillon, 1982).

Nabelek and Robinette (1977) compared seven different combinations of time-constants ranging from (attack/release) 6/30 to 130/580 ms on ten participants. Faster compression yielded higher percent correct on a word recognition task. Similar study of varying the release time and its effects on speech recognition was carried out on nineteen listeners by Schweitzer and Causey (1977). The release time was varied between 15 and 320 ms. Best word recognition performance was noted with mid release time of about 90 ms, than compared to short and long release times. The effect was more pronounced for lower intensity levels. These authors concluded that, since there was no background noise and due to increase in audibility for lower intensity sounds, the fast and moderate compression provided better recognition scores.

Moore (1993) varied the release times between 5 and 320 ms in the high frequency channel. For all the five listeners, SRTs in steady state noise and babble improved with shorter release times, being on average about 4 dB lower with the 5 ms release time than with the 320 ms release time.

A study by Moore, Stainsby, Alcantara, and Kuhnel (2004) found no significant differences of different combinations of attack (8 to 500ms) and release times (32 to 500ms) at low and high frequency channels on speech intelligibility of nonsense syllable. The authors concluded that, since the study was done only at single input level i.e., 65 dB SPL, the time constants does not markedly affect the performance.

These studies have shown that, fast release times have positive effect by improving the speech recognition scores and few studies have shown no significant effect of release time on speech recognition. In summary, release time has varied effects on speech intelligibility in quiet and noisy conditions. No one common trend is followed.

2.6.2. Effect of time constants on speech quality and subjective preferences

The selection of compression time constants revolves around the question on which is more important for speech recognition, improved audibility as provided by fast release times or well-maintained temporal characteristics as provided by slower release times. Thus, it may be important to consider perceptions of the listeners of the resulting speech signal and influence of these compression parameters on their preferences.

King and Martin (1984) studied the effect of release times of 50 to 1000 ms, measuring ‘speech tracking’ performance and preference with speech in babble at 7 dB SNR on 10 participants. They noted a weak trend towards preference for the longer release time. The participants also reported that the ‘the speaker’s voice stood out better from the background’. Similarly, Neuman, Bakke, Mackersie, Hellman, and Levitt (1998) studied the effects of compression release times of 60, 200, and 1000 combined with the compression ratios of 1.5:1, 2:1, and 3:1ms on the categorical rating of sound quality such as clarity, pleasantness, background noise, loudness, and the overall impression of speech-in-noise (Ventilation, Apartment, Cafeteria). Increasing release time caused ratings of pleasantness to increase, and ratings of background noise and loudness to decrease even at compression ratio 3:1.

Hansen (2002) varied the attack times from 1 ms to 100 ms, and release times from 40 ms to 4000 ms in all 15 channels and studied their effects on sound quality and speech intelligibility subjectively rated via paired comparisons for six participants. They reported significant preference for the longest release time, particularly for signals containing speech. With the longest release time, the effects of attack times were least.

Muller, Harris and Ellison (2004) studied for the differences in the gain setting preferred by 18 subjects as the release time was varied (40, 160, or 640 ms) in a two-channel device. No interaction was found between release time and gain required by the participants.

On a concluding note, majority of the studies have shown that longer release times are more preferable and that it increases the quality of speech in quiet and in presence of noise.

Two primary factors may contribute to the expected benefit from changing release time. They are audibility and distortion. Some researchers have hypothesized that short release times should be beneficial to speech recognition because a short release time should increase the level of low-intensity consonants, thereby making them more audible and less susceptible to upward spread of masking by high-intensity vowels (Dreschler, 1988; Gatehouse, Nayler, & Elberling, 2003; Verschuure, Maas, Stikvoort, de Jong, Goedegbure, & Dreschler, 1996). Other researchers have hypothesized that the reduction in intelligibility with short release-time compression is due to an alteration of level differences within the speech utterance (Freyman, Nerbonne, & Cote, 1991; Hedrick & Rice, 2000; Hickson & Byrne, 1997). For example, listeners may use the level difference between the consonant and the vowel to identify a syllable, for some classes of consonants (Balakrishnan, Freyman, Chiang, Nerbonne, & Shea, 1996; Freyman, Nerbonne, & Cote, 1991). Alteration of the level-difference cue, via short release times, may reduce the listener's ability to identify the syllable. Much work has been done to investigate whether the CV level difference is important in recognizing a syllable. Alteration of the level cue often leads to errors in the perception of place of articulation (Hedrick & Rice, 2000; Hedrick & Younger, 2000). The results are equivocal partly, because some investigators have controlled for audibility of the consonant (Sammeth, Dorman, & Stearns, 1999), while others have made the audibility of the consonant vary (Freyman & Nerbonne, 1989). With wearable compression hearing aids, audibility of the consonant will vary.

2.6 c Effects of time constants on acoustics of speech

The attack and release times have a major effect on how compressors affect the levels of the different syllables of speech, which in turn have effects on the envelope and the spectral contrast of the syllables of speech.

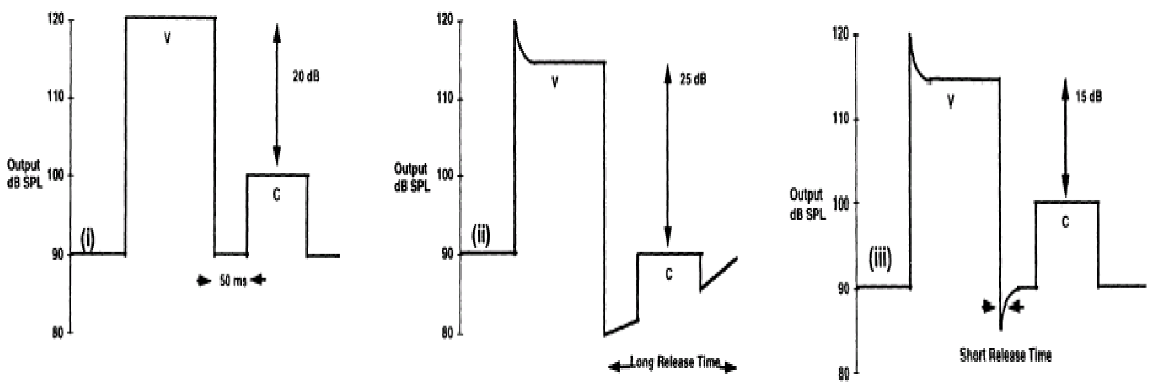


Figure 2.1 Effects of long and short release times on the output waveform (“V”- Vowels, “C”- Consonants)

Figure 3.1 depicts the alteration in the envelope of the output signal for short and long release times. The figure 3.1 (i) is a simplified illustration of the envelope of a vowel-consonant (VC) syllable with short intra-syllabic interval when it is amplified linearly. Figure 3.1 (ii) represents the output when a long release time is used. In this case, the hearing aid is still in the recovery phase when the consonant occurs. This suggests that the consonant will receive less than the full linear gain. This may render the consonant inaudible. Figure 3.1 (iii) represents the output when the release time is short. A shorter release time will return the hearing aid to full linear gain before the consonant.

In order to address the issue of an optimal release time for hearing aid fittings, it is necessary to undertake a comprehensive objective analysis of the acoustic effects of compression on speech and the resulting effects on speech intelligibility. The acoustic analysis must focus on the aspects of speech that are most likely to be affected by varying release times.

Van Tasell (1993) described one aspect of the speech signal that is affected by compression, i.e., the temporal envelope, which is the slow time-intensity modulation in speech. The envelope carries information about syllable structure and rhythm, which provide cues for identification of voicing and manner of articulation of the consonants.

According to Wang (2001) during compression activation or de-activation, due to the brief attack or release time intervals, each instantaneous output level changes continuously until the stabilized output level is reached. Hence the spread of energy and the distortions induced are inevitable. Thus, the attack and release times are the moments filled with spread of energy and the induced distortions, contributing to the temporal and spectral artifacts.

Different acoustic measures are available to quantify the effect of changes in the acoustic properties of the signal. They are Root Mean Square (RMS), Envelope Difference Index (EDI), CVR (Consonant-to-Vowel Ratio), Spectral Distance/Distortion Measure (SDM).

Since individuals with reduced dynamic range are prescribed with compression hearing aid, the benefits of amplitude compression may be limited by distortion resulting from rapid gain adjustment. As these individuals also suffer from impaired frequency resolution and they rely more on temporal cues. In order to evaluate temporal distortions, it is convenient to quantify distortion using a metric that is sensitive to the changes in the processed signal that decrease consonant recognition, such as the Envelope Difference Index (EDI). According to Jenstad and

Souza (2005) as the EDI value increased, representing higher amounts of temporal envelope distortion, SRS decreased significantly.

The EDI is an index of change to the temporal envelope between two signals. To quantify temporal changes caused by amplification, EDI serves as a technique for comparing the temporal envelopes of two acoustic signals. This technique is based on temporal envelope subtraction. The procedure given by Fortune, Woodruff, & Preves (1994) to compute EDI is described below.

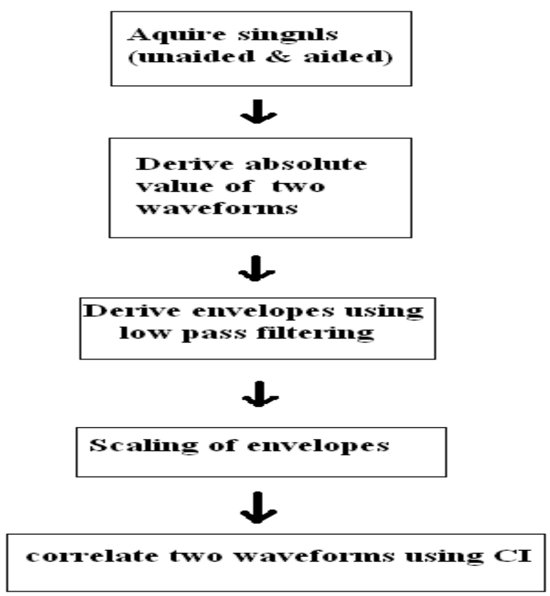
1) Once the waveform is acquired and the signal processing is initiated by taking the absolute value of this waveform, which allowed the envelope to be described as a continuous, positive-amplitude function, relative to direct current (DC).
2) The envelope of this waveform will be derived by digital low-pass filtering (20 Hz).
3) Since the EDI is based on envelope subtraction, all envelopes will be scaled to a common reference point, so that temporal effects could be analyzed without contamination by amplitude variations that usually occur between an unaided condition and aided condition.
4) Scaling was accomplished by calculating the overall mean amplitude of an envelope, and dividing each sampled point of the envelope by the mean amplitude.
5) Both unaided and aided waveforms were scaled.
6) These two scaled envelopes will be correlated to find out correlation index (CI) using cross correlation technique.
7) EDI was calculated using the formula

$$EDI = \left(\sum_{n=1}^{N} \left| Env1_n - Env2_n \right| \right) / 2N$$

The formula shows that the unaided envelope (Env2) is subtracted, point by point, from the aided envelope (Envl), and the absolute values of the differences are taken. The ED1 is calculated as the mean of these absolute values, divided by 2. Division by 2 places the ED1 on a scale that ranges from 1.00 to 0.00

These steps allow the overall temporal difference that exists between the two waveforms to be calculated.

Figure 2.2: Block diagram to calculate EDI



ED1 on a scale ranges from 1.00 which represents no correspondence between envelopes to 0.00 which represents perfect correspondence between envelopes (Fortune, Woodruff, & Preves, 1994).

As individuals with hearing impairment have impaired frequency resolution, alteration of spectral content in the amplified speech can cause additional decrement in the speech recognition performance. Hence to study the spectral distortions in the amplified signal, spectral distance or distortion measures can be used. Spectral Distance or Distortion measures are objective measures to evaluate the speech quality. They are broadly divided into time domain measures and spectral domain measures. Time domain measures include estimation of Signal to noise ratio (SNR), Segmental SNR (SNRseg). Spectral domain measures include Log-likelihood Ratio (LLR) measures by Itakura, (1975), Crochiere, Tribolet, & Rabiner (1980), the Linear Predictive Coding (LPC) parameter distance measures (Barnwell & Voiers, 1979), the cepstral distance measures by Gray and Markel (1976), Tohkura (1987) and the weighted slope spectral distance measure by Klatt (1976). Measures have been applied to speech processing and speech recognition fields. SDM produce measurements of dissimilarities seen in two speech spectra.

The LLR is a spectral distance measure which mainly models the mismatch between the formats of the original and enhanced signals (Rao, Murthy, & Rao, 2011).

LLR is inherently positive and a low value indicates a close agreement between the spectral magnitudes of a test signal and a reference signal. A zero distance means that the spectral magnitudes of both signals are identical and a large spectral distance means the signals are significantly different. Differences in phases do not affect this spectral distance measure. Hence, the measure is insensitive to delays between the original signal and its coded version.

Jeon and Lee (2008) used LLR to objectively quantify the quality of the spectral contrast enhanced speech. They found that, the smaller LLR values were associated with lesser distortion. Loizou (1998) designed COLEA, a MATLAB software tool for speech analysis, to compute the LLR. It is one of the objective, quality - based, spectral domain measures and is based on signal based models. The LLR distance measure is based on the difference between the speech production models such as all pole linear predictive coding (LPC) models of the original and distorted speech signal. Studies carried out by Jeon and Lee (2008), Cote, Turbin, and Moller (2008) in objectively assessing speech quality, have used the following procedure for computing LLR.

1) Acquire the waveforms, unaided and aided.
2) Both the waveforms are time aligned.
3) Each of the speech segments is represented by p-th order all pole linear predictive coding model.
4) The speech waveforms are windowed to form frames of 10 ms (used in the study).
5) LPC co-efficient vector, and the autocorrelation matrix of the waveforms were computed.
6) LLR values for a particular time frame is computed using the formula

$$d_{LLR} = \log_{10}\left(\frac{a_y R_s a_y^T}{a_s R_s a_s^T}\right)$$

   Where *as* and *Rs* are the linear prediction coefficient vector and autocorrelation matrix of the original (clean) speech frame respectively, and *ay* is the linear prediction coefficient vector of the enhanced speech frame.

7) The mean LLR value was obtained by averaging the individual frame LLR values.

Van Tasell (2000) showed that a long release time should have a minimal effect on the envelope and that a short release time (shorter than 200 ms) should smooth out the envelope by increasing the CV ratio (CVR). The CVR is a measure of the difference in intensity between the consonant and the vowel (Freyman & Nerbonne, 1989; Hickson & Byrne, 1997).

Ellison, Harris, and Muller (2003) studied the interactions of hearing aid compression release time and fitting formula and their effects on speech acoustics. NAL-NL1, DSL I/O, FIG.6 or ASA2p with release times of 40 and 640 ms were the condition. Recordings were made through KEMAR for running speech and analyzed to determine the long-term-average-speech spectra, consonant-to-vowel ratios and the RMS amplitude of 32 phonemic units. Aided and unaided results were compared. Within each prescriptive formula, changes in release time affected all of the speech measures subsequent to programming the instrument to a static-composite signal. The short release-time condition produced the greatest alteration to the speech signal.

Jenstad, and Souza, (2005) studied the effect of release time of compression hearing aid on speech acoustics and intelligibility. The release times under study were 12, 100, and 800 ms. The stimuli were processed through a hearing aid simulator at three input levels (50, 65 & 80 dB SPL). Two acoustic measures were made on individual syllables: the envelope-difference index and CV ratio. The acoustic analysis revealed statistically significant effects among the 3 release times. The size of the effect was dependent on characteristics of the phoneme. Twelve listeners with moderate sensorineural hearing loss were tested for their speech recognition for the same stimuli. The acoustic measurements reflecting the changes due to release time were significant predictors of phoneme recognition. Increased temporal-envelope distortion was predictive of reduced recognition for some individual phonemes.

As mentioned earlier, two approaches have been used to study release time. They are studying the effects of release time on speech intelligibility and quality or user preference. These approaches lead to inconclusive decisions about the appropriate release times. Recently the focus is on the acoustic effects of compression time constants. Very few acoustic parameters have been targeted to measure the effects caused by compression. The few available studies on acoustic effects have majorly concentrated on the temporal aspects of the signal and not much of focus has been on spectral distortions caused by the compression release time. There is a dearth of studies on acoustic effects of compression release time. Hence the present study was taken up to evaluate the effects of compression release time on acoustic and behavioural parameters of speech.

CHAPTER III

METHOD

The aim of the study was to evaluate the acoustic and behavioural effects of different release times of compression in a hearing aid. The following procedure was used.

3.1 Participant selection criteria

The criteria to select the participants for behavioural and acoustic measurements are described below.

3.1.1 Participant selection criteria for behavioural measure

Thirteen male and seven female participants were considered in the study. Their age range was 20 years to 55 years, with a mean age of 42.9 years and a standard deviation of 8.76 years. All the 20 participants were diagnosed to have flat or gradually sloping sensorineural hearing loss (SNHL) ranging from moderate to severe degree. The participants were divided into two groups based on the degree of hearing loss: Group A consisted of ten participants having moderate to moderately severe SNHL. Their Pure Tone Average (PTA) ranged from 41 dB HL to 70 dB HL, with a mean PTA of 60.9 and a standard deviation of 7.89. Group B consisted of 10 participants having moderately severe to severe SNHL. Their PTA ranged from 71 dB HL to 90 dB HL, with a mean PTA of 75.4 and a standard deviation of 4.57. Their speech recognition thresholds were proportionate to their hearing thresholds. They also had Speech Identification Scores proportionate to the hearing loss, i.e., greater than 60 %. All the participants had normal middle ear status confirmed through tympanogram and acoustic reflexes. OAEs were absent in all the participants. The participants had post-lingually acquired hearing loss with age adequate speech and language. They did not have any previous history of neurological and cognitive dysfunction. Out of twenty participants, twelve participants had more than 3 months of experience with the hearing aid usage; the other eight participants were naïve hearing aid users. The participants were native speakers of Kannada language or were exposed to Kannada since childhood, and were proficient in speaking Kannada.

3.1.2 Participant selection criteria for acoustic measurement

Five participants having moderately-severe to severe hearing loss were considered. Their age ranged from 20 to 30 years with a mean age of 24.2 years and a standard deviation of 2.48. All the participants had their pure tone average (PTA) being greater than 56 dB HL.

Acoustic measures were performed on these five participants with the hearing aid programmed for the mean audiometric data of Group A (mean PTA 60.9) and Group B (mean PTA of 75.3 dB HL).

3.2 Equipment and material used

The following instruments and material were used for data collection.

3.2.1 Instrumentation

A calibrated sound field diagnostic audiometer for behavioural and acoustic data collection, a calibrated immittance meter to rule out middle ear pathology in participants, a personal computer (PC) to play the stimulus for behavioural and acoustic data collection, Fonix 7000 real ear measurement system, and behind the ear hearing aid with the option for varying its compression release time, attack time, feedback detector, phase cancellation technique, two listening programmes, two channels and seven bands.

3.2.2 Test material: Phonemically balanced Kannada bi-syllabic word list (Yathiraj & Vijayalakshmi, 2005), a Kannada story passage and VCV syllables were used in the study.

3.3 Preparation of speech stimulus

VCV tokens and a Kannada story passage for the acoustic and behavioural data collection were prepared using the following procedure.

3.3.1 Preparation of VCV tokens:

Three female speakers whose mother tongue was Kannada (Dravidian language widely spoken in Karnataka, South India) were chosen to utter the Vowel-Consonant-Vowel (VCV) tokens. The phonemes in Kannada with greater than 0.5% of frequency of occurrence (Ramkrishna, Nair, Chiplunkar, Atal, Ramachandran, & Subramanian, 1962) were selected and paired with a vowel combination. The recognition of consonants in the /a/ context tended to be high according to Gordon-Salant (1987). Greater effects of altering the acoustic characteristics can be seen with either /i/ or /u/. Since there were 21 consonants and the procedure and acoustic analysis were time consuming, only a high short front vowel /i/ at initial and final position was used. The 21 VCV syllables from each speaker were recorded on to a computer, using the Adobe Audition software, via the recording microphone (Ahuja, AUD-101XLR) placed at a distance of 10 cm from the lips of the speaker (Winholtz & Titze, 1997). The stimulus was recorded in an air conditioned sound treated room. The recorded stimulus was digitized using a 32-bit processor at 44,100 Hz sampling frequency. Goodness test was performed in order to select one set of test stimuli. These test stimuli were then concatenated with an inter-stimulus interval of 3 sec. These stimuli were preceded by 1 kHz calibration tone and were written on to a compact disc. Table 3.1 gives the speech sounds used in the study, classified based on manner of articulation.

Table 3.1: *Classification of consonants based on manner of articulation.*

Stops	***Nasals***	***Fricatives***	***Affricates***	***Liquids***	***Glides***
/ipi/, /iti/, /iki/, /ibi/, /idi/, /igi/, /iti/, /idi/	/ini/, /imi/, /ini/	/isi/, /Iʃi/, /ihi/	/Itʃi/, /idzi/	/iri/, /ili/, /ili/	/ivi/, /iji/

In order to decide about the appropriate duration of the stimulus, a pilot study was carried out on a subject. One speech sound was selected from each of the six classes of speech sounds. The duration of stimuli selected ranged from 650 ms to 900 ms to 1200 ms. EDI was computed for each of the stimuli across release time at 65 dB HL. For shorter duration stimulus (650 to 900 ms), the EDI values for 640 ms and 1280 ms release time were almost equivalent. Hence, depending on the observation of EDI values, decision was made to prepare the stimuli with duration ranging between 800 to 1200 ms.

3.3.2 Kannada story passage

An adult female native speaker of Kannada was chosen to record the Kannada passage. The passage read out by the speaker was recorded on to a computer, using the Adobe Audition software, via the recording microphone (Ahuja, AUD-101XLR) placed at a distance of 10 cm from the lips of the speaker (Winholtz & Titze, 1997). The stimulus was recorded in an air-conditioned, sound treated room. The stimulus was preceded by 1 kHz calibration tone and was written on a compact disc.

3.4 Procedure

All the measurements were done in air conditioned sound treated room. The hearing aid was then programmed, acoustic and behavioural measurements were carried out.

3.4.1 Hearing aid programming

A digital behind the ear hearing aid was connected to the HiPro which was in turn connected to a personal computer in which the NOAH and hearing aid specific softwares were installed. The following steps were involved in programming the hearing aid. Audiometric pure tone thresholds (from 250 Hz to 8 kHz for air-conduction & from 250 Hz to 4 kHz for bone-conduction) of the participant were fed into the NOAH software in the personal computer, using the audiogram module. These data were saved. Through the hearing aid programming software, the hearing aid was detected and programmed using the NAL-NL1 prescriptive formula, with an acclimatization level of 2. The hearing aid settings were optimized specifically for each individual, using the Ling's six sound test (/a/, /i/, /u/, /s/, /sh/ and /m/). Each time, for data collection, the compression release time was set to one of the three conditions, i.e., 40 ms, 640 ms and 1280 ms. The compression type selected was Wide Dynamic Range Compression (WDRC) and the compression threshold was set to 50 dB SPL. Other than release time, all other parameters were kept constant across the three aided measurement conditions. Electroacoustical measurements were performed to verify the parameters set in the hearing aid.

3.4.2 Acoustic and behavioural measurements

Acoustic (unaided and aided) and behavioural (aided) measurements were carried out with the release time of hearing aid set to 40 ms, 640 ms and 1280 ms at three different input levels. The data were collected in two phases.

Phase I: Measurement of acoustic parameters - temporal and spectral parameters.

Phase II: Measurement of behavioural parameters - speech identification scores and speech quality ratings.

Phase I: Measurement of acoustic parameters - temporal and spectral

The acoustic measures used in the study were Envelope Difference Index (EDI) and Spectral Distance / Distortion Measure (SDM) called Log-likelihood Ratio (LLR). The following steps were involved in measuring these parameters.

Step I a: Set-up for acoustic measurement

A personal computer containing the VCV tokens was connected to the auxiliary input of the portable diagnostic audiometer. The output of the audiometer was routed to the loudspeaker of the real ear measurement system, Fonix 7000. The input from the Fonix 7000 was disabled and the input from the audiometer was routed to the loud speaker. The probe tube microphone was inserted into the unoccluded ear canal of the participant. The output from the Fonix 7000 system was routed to the microphone inlet of the PC containing Praat software. The participants were seated on a comfortable chair at the distance of 12 inches away from the loudspeaker of the real ear measurement system and at 45 degree azimuth. Figure 3.1 shows the set-up of the equipment used for recording.

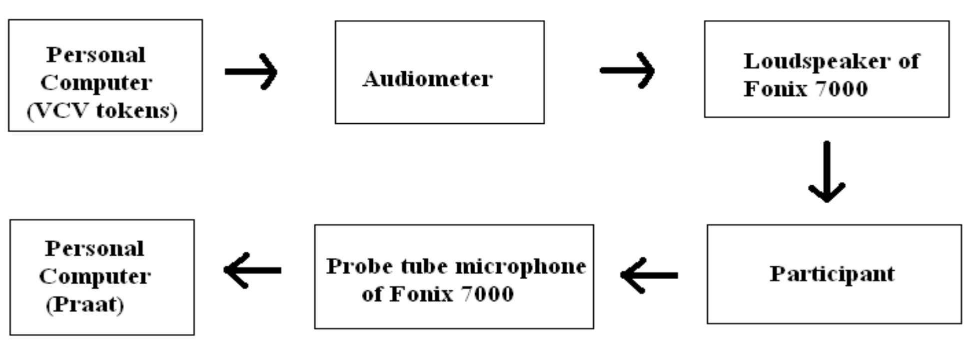


Figure 3.1: Block diagram depicting the set-up of equipment for acoustic data collection.

Step I b: Unaided measurement for acoustic analysis.

The participants were seated on a comfortable chair at the distance of 12 inches away from the loudspeaker of the real ear measurement system and at 45 degree azimuth direction. The probe tube microphone of the real ear measurement system was inserted in the ear canal. The length of the probe tube inserted in the ear canal was 3-5 mm beyond the length of the ear mould of the participant. The VCV tokens were presented to the participant. This was picked up from the ear canal of the participant and recorded in the PC using Praat software, for further analysis. This procedure was repeated at three presentation levels, i.e., 30, 45 and 65 dB HL.

Step I c: Aided measurement for acoustic analysis

The hearing aid was programmed as mentioned earlier. Hearing aid was set to one of the three release times. Later, the hearing aid was fitted to the participant using custom ear mold. The participants were seated 12 inches away from the loudspeaker of the real ear measurement system, at 45 degree azimuth. The probe tube microphone was inserted such that the tube was 3 to 5 mm beyond the ear mould. The recorded VCV tokens were presented through the audiometer routed to the loudspeaker of the real ear measurement system. The signal was picked up by the hearing aid; the output of the hearing aid in the ear canal was in turn picked up by the probe tube microphone of the real ear measurement system. The signal from the probe tube microphone was recorded in Praat software in the computer for further analysis. This procedure was performed for different compression release time (40 ms, 640 ms & 1280 ms). Each time the release time was varied, it was made sure that all the other parameters were kept constant across the measurements. The procedure was carried out at three input levels (30, 45 & 65 dB HL). Thus, the aided measurements were done on five participants with profound hearing loss, the pure tone average data of two groups of participants (Group A and Group B of behavioural measurement).

For both unaided and aided measurements, the microphone sensitivity of the personnel computer was monitored by the deflections seen in the Praat software and the sensitivity was adjusted whenever required to limit peak clipping.

Step I d: Calculation of Envelope Difference Index (EDI) and Log-likelihood Ratio (LLR)

MATLAB was used for computing EDI and LLR. Using the MATLAB functions, different algorithms or codes were generated to calculate the EDI and LLR values. To calculate EDI and LLR, the speech VCV tokens (unaided and aided) which have to be compared were edited with respect to the common reference point shared by them. The edited VCV tokens were then fed into the MATLAB software containing particular algorithm for the calculation of EDI and LLR values.

For the computation of EDI, the method used by Fortune, Woodruff, and Preves (1994) was used. The aided and unaided speech tokens were fed into MATLAB and the difference between the temporal envelopes of the two waveforms in terms of correlation index was obtained.

For computation of LLR, COLEA, a software tool for speech analysis developed by Loizou (1998) was used. The procedure adopted to compute LLR was same as the one used by Jeon and Lee (2008), Cote, Turbin, and Moller (2008). The aided and unaided speech tokens were fed into MATLAB and the mismatch between the formant peaks between these two waveforms were computed.

Phase II: Measurement of behavioural parameters - Speech Identification Scores and speech quality rating.

Behavioural measures used in the study were speech identification scores (SIS) and speech quality judgment. For both the measurements, the hearing aid was programmed as mentioned earlier. During the presentation of the stimulus in each of conditions level adjustments was done for the calibration tone such that the VU-meter deflections averaged to 0. At each presentation level, the deflection of the VU meter was monitored. The non-test ear was masked with speech noise presented through the insert receivers at 65 dB HL whenever required.

Step II a: Measurement of aided speech identification scores.

Participants were seated on a comfortable chair 1 meter away from the loudspeaker of the audiometer, at 45 degree azimuth. They were tested for speech identification using bi-syllabic words (Yathiraj & Vijayalakshmi, 2005). This word-list includes 5 lists and each list comprising of 20 words which are phonemically balanced and equally difficult. The words were presented in live mode through the loud speaker of the sound field diagnostic audiometer. This was carried out for different release time (40 ms, 640 ms & 1280 ms) at three input levels (30, 45 & 65 dB HL) on two groups of participants. In each condition, the number of words correctly identified was noted.

Step II b: Measurement of aided speech quality judgment

To assess the speech quality, a recorded Kannada story passage was presented via recorded mode through the loud speaker of the sound field diagnostic audiometer. The participants were instructed to rate the story for its quality on a 10-point rating scale. The parameters for evaluating quality judgment in the present study were loudness, clarity, naturalness and overall impression. The rating scale ranges from 0 to 10 wherein, 0 indicates very poor 2 indicates poor 4 indicates fair 6 indicates good, 8 indicates very good and 10 indicates excellent. The participants were instructed to use odd numbers for rating when the perception was between any the two points. The instructions given to the participants for quality rating of the parameters are described below.

- ✓ *Loudness*- how loud the speech is, in contrast to too loud and faint.
- ✓ *Clarity*- how clear the speech sounded with respect to intelligibility, in contrast to distorted or blurred speech.
- ✓ *Naturalness*- how similar is the speech perception via hearing aid when compared to the experience of normal and natural listening of speech.
- ✓ *Overall Impression*- how good the speech perception is.

This rating procedure was carried out for different release time (40 ms, 640 ms & 1280 ms) at three input levels (30, 45 & 65 dB HL) on two groups of participants.

Thus, the following data were collected.

I. Acoustic Measurement: Unaided and aided data were obtained for the following conditions -

1) Unaided: Twenty-one VCV tokens were recorded at three input levels (30, 45 & 65 dB HL). The recordings in each of these conditions were obtained on five participants

2) Aided: Twenty-one VCV tokens were recorded at three release times (40 ms, 640 ms & 1280 ms) when the stimulus was presented at three input levels (30, 45 & 65 dB HL) for the mean audiometric data of Group A and B. The recordings in each of these conditions were obtained on five participants.

II. Behavioural Measurement: Behavioural measurement was carried out with three release times (40 ms, 640 ms & 1280 ms) when the stimulus was presented at 30, 45 and 65 dB HL. The data on the following were collected to evaluate the effect of release time and presentation levels was measured in Groups A and B

1) SIS and
2) Quality ratings (i.e., loudness, clarity, naturalness and overall impression) for participants.

The data thus collected were tabulated and subjected to appropriate statistical analysis.

CHAPTER IV

RESULTS

The present study was carried out with the aim of evaluating the effects compression release times (RT) of hearing aid on acoustic and behavioural measures of speech. The acoustic and behavioural measures were studied across three release times, i.e., 40 ms, 640 ms and 1280 ms at three input levels, i.e., 30 dB HL, 45 dB HL, and 65 dB HL. The behavioural measures were performed on two groups of participants, viz., Group A and Group B. Group A consisted of ten participants with sensorineural hearing loss whose pure tone average ranged from 40 to 70 dB HL and Group B consisted of ten participants with sensorineural hearing loss whose pure tone average ranged from 70 to 90 dB HL. The behavioural measures used in the study are SIS and perceptual ratings on quality. The acoustic measures were performed on five participants using the mean audiometric thresholds of the participants of Group A and mean audiometric thresholds Group B participants. The acoustic measures used in the study are Envelope Difference Index (EDI) and Log-likelihood Ratio (LLR). Using these two measures, the following data were obtained, tabulated and used for analysis.

1) EDI for six classes of speech sounds namely stops, nasals, affricates, fricatives, liquids and glides for nine conditions, i.e., three release times (40 ms, 640 ms and 1280 ms) and three input levels (30 dB HL, 45 dB HL, and 65 dB HL).
2) LLR measures for six classes of speech sounds in nine conditions as mentioned above.
3) SIS at three release times (40 ms, 640 ms & 1280 ms) across three input levels (30 dB HL, 45 dB HL, & 65 dB HL) for Group A and Group B.

4) Perceptual quality ratings on four parameters of quality. They are loudness, clarity, naturalness and overall impression. The quality ratings for all these four parameters across nine conditions for the Group A and Group B were tabulated.

In order to subject the data to statistical analysis, all the above data were tabulated in the SPSS software (version 16). The independent variables in the study were groups (i.e., Group A and Group B). The dependent variables in the study were SIS, perceptual parameters of quality rating (Loudness, Clarity, Naturalness & Overall impression), EDI and LLR for six classes of speech sounds. These dependent variables were obtained with nine aided conditions i.e., three release times (40 ms, 640 ms & 1280 ms) and at three input levels (30 dB HL, 45 dB HL & 65 dB HL).

Descriptive statistics was used to compute the mean and standard deviation for all the measures and across all the conditions. Two way repeated measure ANOVA was performed to evaluate the intercation between the groups and conditions. If two way repeated measure ANOVA revealed any significant interactionand/or main effect, pair-wise comparison using Bonferroni's multiple comparison test was performed to evaluate significant difference between the pairs.The results of the present study are discussed under following headings **–**

4.1 Acoustic measures

The effect of release time and input level on acoustic measures of speech were investigated using Envelope Difference Index and Log-Likelihood Ratio.

4.1.1 Envelope Difference Index (EDI)

Descriptive statistics was performed to compute the mean and standard deviation (SD) of EDI values for six classes of speech sounds viz., stops, nasals, affricates, fricatives, liquids, and glides, at three release times and at three input levels. Non-parametric statistics was used for the analysis as the measurement was done on five participants. Man Whitney U test, a test of two independent samples was performed in order to compare between groups. In order to compare across conditions two related samples test i.e., Wilcoxon Signed rank test was performed. Table 4.1 depicts the mean and SD of EDI for six classes of speech sounds across three release times and across three input levels of data for Groups A and B.

Table 4.1: *Mean and S.D of EDI for six classes of speech sounds at three input levels across three release times for mean audiometric thresholds of Group A and Group B*

		EDI for mean audiometric threshold of Group A						***EDI for mean audiometric threshold of Group B***					
Release time (ms)	***Input Level dBHL***	***Stops***	***Nasals***	***Affricates***	***Fricatives***	***Liquids***	***Glides***	***Stops***	***Nasals***	***Affricates***	***Fricatives***	***Liquids***	***Glides***
		Mean (S.D)	***Mean (S.D)***	***Mean (S.D)***	***Mean (S.D)***	***Mean (S.D)***	***Mean (S.D)***	***Mean (S.D)***	***Mean (S.D)***	***Mean (S.D)***	***Mean (S.D)***	***Mean (S.D)***	***Mean (S.D)***
40 ms	30 dB	0.47 (0.05)	0.47 (0.03)	0.47 (0.06)	0.46 (0.05)	0.46 (0.04)	0.47 (0.04)	0.51 (0.04)	0.53 (0.05)	0.47 (0.07)	0.47 (0.04)	0.50 (0.05)	0.52 (0.06)
	45 dB	0.58 (0.06)	0.57 (0.02)	0.56 (0.03)	0.51 (0.02)	0.54 (0.03)	0.64 (0.05)	0.64 (0.03)	0.61 (0.06)	0.61 (0.07)	0.58 (0.08)	0.56 (0.09)	0.60 (0.05)
	65 dB	0.72 (0.06)	0.72 (0.06)	0.73 (0.03)	0.64 (0.03)	0.66 (.02)	0.74 (0.05)	0.75 (0.06)	0.75 (0.03)	0.68 (0.05)	0.66 (0.11)	0.64 (0.11)	0.72 (0.03)
640 ms	30 dB	0.46 (0.04)	0.46 (0.02)	0.45 (0.06)	0.45 (0.04)	0.47 (0.06)	0.48 (0.05)	0.51 (0.04)	0.50 (0.03)	0.48 (.073)	0.45 (0.04)	0.49 (0.07)	0.49 (0.05)
	45 dB	0.56 (0.05)	0.52 (0.03)	0.54 (0.05)	0.51 (0.02)	0.53 (0.03)	0.59 (0.03)	0.63 (0.01)	0.58 (0.02)	0.56 (0.05)	0.55 (0.03)	0.53 (0.05)	0.58 (0.04)
	65 dB	0.69 (0.05)	0.67 (0.07)	0.67 (0.05)	0.61 (0.04)	0.58 (.05)	0.70 (0.05)	0.68 (0.05)	0.68 (0.02)	0.62 (0.04)	0.62 (0.08)	0.58 (0.09)	0.65 (0.02)
1280 ms	30 dB	0.46 (0.04)	0.48 (0.03)	0.45 (0.05)	0.45 (0.04)	0.45 (0.03)	0.48 (0.04)	0.506 (0.05)	0.51 (0.04)	0.46 (0.08)	0.47 (.05)	0.51 (0.06)	0.53 (0.05)
	45 dB	0.56 (0.08)	0.52 (0.04)	0.52 (0.07)	0.51 (0.03)	0.50 (0.05)	0.56 (0.05)	0.61 (0.01)	0.55 (0.03)	0.53 (0.04)	0.55 (0.03)	0.53 (0.03)	0.57 (0.01)
	65 dB	0.64 (0.09)	0.60 (0.05)	0.60 (0.06)	0.61 (0.05)	0.60 (.08)	0.63 (0.08)	0.65 (0.06)	0.61 (0.07)	0.58 (0.02)	0.58 (0.07)	0.56 (0.08)	0.59 (0.02)

Table 4.1a: *Significance of effect of release time for six classes of speech sounds at three input levels for Group A mean audiometric data*

Group A		***Stops***		***Nasals***		***Affricates***		***Fricatives***		***Liquids***		***Glides***	
Input level	***Release time***	***Z***	***Sig***	***Z***	***Sig***	***Z***	***Sig***	***Z***	***Sig***	***Z***	***Sig***	***Z***	***Sig***
30 dB HL	*40 ms & 640ms*	-1.4	0.157	-.81	0.414	-1.7	0.078	-1.2	0.197	-.13	0.892	-.73	0.461
	40 ms & 1280ms	-.81	0.414	-1.3	0.180	-1.7	0.078	-.96	0.336	-1.2	0.197	-1.8	0.059
	640 ms & 1280ms	-1.0	0.276	-1.3	0.174	-.44	0.655	-.13	0.892	-.73	0.465	-1.0	0.276
45 dB HL	*40 ms & 640ms*	-1.7	0.078	-2.0	0.041*	-1.2	0.223	-.36	0.715	-.40	0.684	-2.0	0.042*
	40 ms & 1280ms	-2.0	0.043*	-2.0	0.043*	-1.8	0.066	-.13	0.893	-1.8	0.066	-2.0	0.042*
	640 ms & 1280ms	.00	1.00	-.13	0.890	-1.5	0.131	.00	1.00	-1.6	0.102	-2.0	0.041*
65 dB HL	*40 ms & 640ms*	-2.0	0.043*	-2.0	0.043*	-2.0	0.043*	-2.0	0.042*	-2.0	0.042*	-1.8	0.068
	40 ms & 1280ms	-2.0	0.043*	-2.0	0.043*	-2.0	0.043*	-.94	0.343	-1.2	0.225	-2.0	0.043*
	640 ms & 1280ms	-1.7	0.080	-2.0	0.043*	-2.0	0.043*	-.67	0.498	-.67	0.498	-1.8	0.068*

*Note:- *: significant difference at $p<0.05$*

Table 4.1b: *Significance of effect of release time for six classes of speech sounds at three input levels for group B mean audiometric data.*

Group B		***Stops***		***Nasals***		***Affricates***		***Fricatives***		***Liquids***		***Glides***	
Input level	Release time	Z	Sig	Z	Sig	Z	Sig	Z	Sig	Z	Sig	Z	Sig
30 dB HL	40 ms & 640ms	-.67	0.498	-1.4	0.144	-.36	0.713	-1.7	0.078	-.13	0.893	-1.2	0.223
	40 ms & 1280ms	-.73	0.465	-1.2	0.225	-1.6	0.102	.00	1.0	-.81	0.416	-.13	0.893
	640 ms & 1280ms	-.36	0.715	-.13	0.892	-1.0	0.273	-1.4	0.144	-.36	0.715	-1.4	0.144
45 dB HL	40 ms & 640ms	-0.73	0.465	-1.6	0.104	-2.0	0.043*	-1.0	0.273	-1.0	0.279	-1.7	0.080
	40 ms & 1280ms	-1.7	0.078	-2.0	0.042*	-2.0	0.043*	-.92	0.357	-.73	0.465	-1.4	0.138
	640 ms & 1280ms	-1.8	0.059	-1.8	0.066	-2.0	0.043*	.00	1.0	-.13	0.892	-.27	0.785
65 dB HL	40 ms & 640ms	-2.0	0.043*	-2.0	0.041*	-2.0	0.042*	-2.0	0.043*	-2.0	0.042 *	-2.0	0.043*
	40 ms & 1280ms	-2.0	0.042*	-2.0	0.042*	-2.0	0.043*	-2.0	0.043*	-2.0	0.042 *	-2.0	0.043*
	640 ms & 1280ms	-1.2	0.225	-2.0	0.043*	-2.0	0.043*	-1.8	0.066*	-1.8	0.066	-2.0	0.043*

*Note:- *: significant difference at p<0.05*

Table 4.1c: *Significance of effect of release time on EDI for six classes of speech sounds at three input levels for group A and Group B mean audiometric data.*

EDI for Group A									
Input level	***30 dB HL***			***45 dB HL***			***65 dB HL***		
Release time	40 and 640	40 and 1280	640 and 1280	40 and 640	40 and 1280	640 and 1280	40 and 640	40 and 1280	640 & 1280
Stops					*		*	*	
Nasals				*	*		*	*	*
Affricates							*	*	*
Fricatives								*	
Liquids								*	
Glides				*	*	*		*	*
Group B									
Input level	***30 dB HL***			***45 dB HL***			***65 dB HL***		
Release time	40 and 640	40 and 1280	640 and 1280	40 and 640	40 and 1280	640 and 1280	40 and 640	40 and 1280	640 & 1280
Stops							*	*	
Nasals					*		*	*	*
Affricates				*	*	*	*	*	*
Fricatives							*	*	*
Liquids							*	*	
Glides							*	*	*

Note:- *: *significant difference at* $p<0.05$

4.1.1.a Effects of release times on six groups of speech sounds

Descriptive statistics was performed to compute the mean and standard deviation (SD) of EDI values for six groups of speech sounds (stops, nasals, affricates, fricatives, liquids & glides), at three release times and at three input levels. To evaluate the effect of release time on EDI, two related sample test, i.e., Wilcoxon Signed rank test, was performed to find out the significant difference between pairs of the nine conditions for six groups of speech sounds. This test was administered separately for the data of Group A and B. The results of comparison across three release time conditions are discussed below at each input level.

a) At 30 dB HL

Descriptive statistics was done to compute the mean and SD at 30 dB HL input level for six classes of speech sounds. Table 4.1 depicts the mean and SD of EDI for six classes of speech sounds across three release times, at 30 dB HL input level, for data of Group A and B. The effects of three release times on EDI at 30 dB HL for six classes of speech sounds are described below. The effects are discussed for each Group (Group A & B) individually.

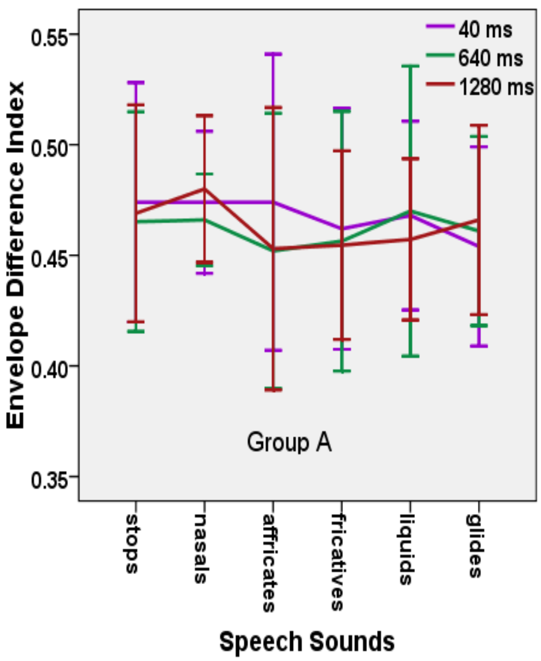


Figure 4.1

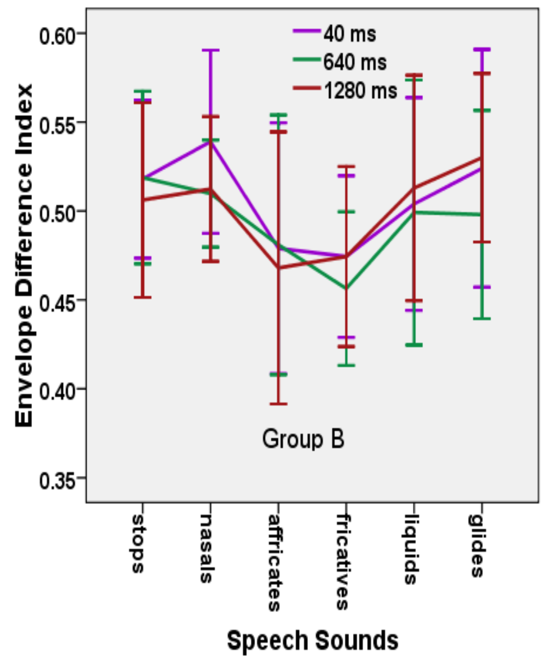


Figure 4.2

Figure 4.1and 4.2: Effect of three release times at 30 dB HL input level on EDI of six classes of speech sounds for Group A and Group B mean audiometric data respectively.

Group A: There is no systematic trend seen for the mean EDI values as the release time increased. The SD is more for affricates than other classes of speech sounds (Table 4.1).

Mean EDI values across three release times at 30 dB HL are depicted in Table 4.1 and Figure 4.1 for six groups of consonants, stops, nasals, affricates, fricatives, liquids and glides. There is no systematic trend seen for the EDI values, as the release time increased. As depicted in Table 4.1a, Wilcoxon Signed Rank test was performed for comparison across release times. Results revealed no significant difference between release times ($p>0.05$). A similar trend was noticed for all the six groups of speech sounds.

Group B: As depicted in Table 4.1, the mean EDI values decreased as the release time increased for all six classes of speech sounds. The SD is higher for speech sounds in the order of affricates, liquids, glides, stops, fricatives and nasals.

Mean EDI values are depicted in Table 4.1 and figure 4.2 for six groups of consonants, stops, nasals, affricates, fricatives, liquids and glides. There is no systematic trend seen for the EDI values, as the release time increased. As depicted in Table 4.1b, Wilcoxon Signed Rank test was performed for comparison across release times. The results revealed no significant difference between release times. A similar trend was seen for all the six speech sound groups.

b) At 45 dB HL

Descriptive statistics was done to compute the mean and SD at 45 dB HL input level for six classes of speech sounds. Table 4.1 depicts the mean and SD for EDI of six classes of speech sounds across three release times at 45dB HL input levels for data of Group A and B. The effects are discussed for each Group (Group A & B) individually.

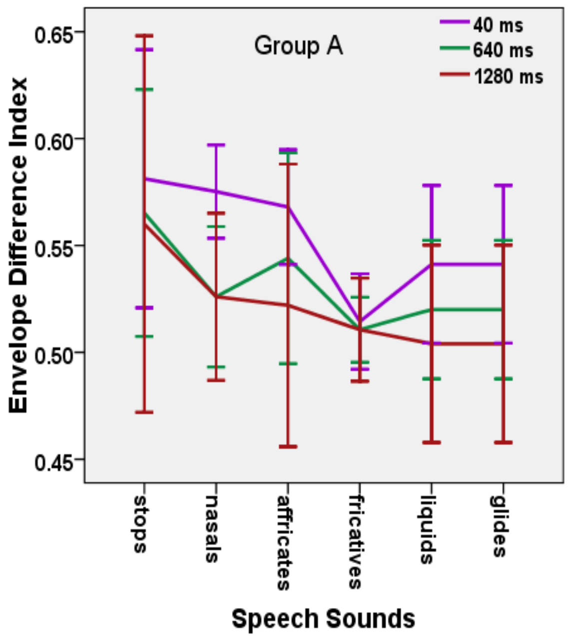


Figure 4.3

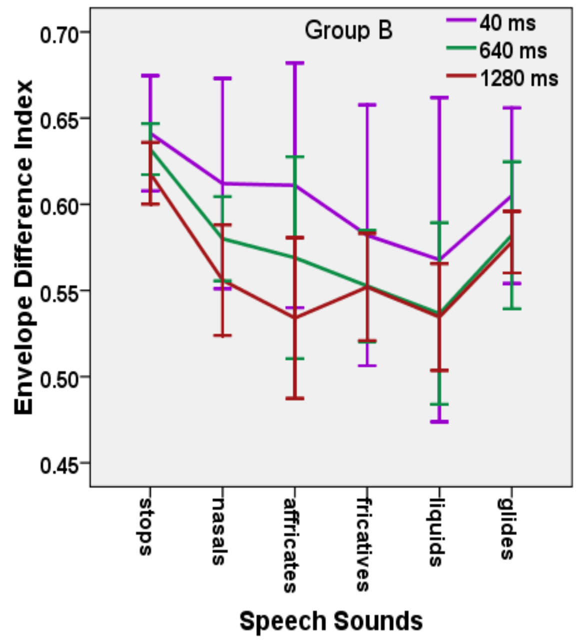


Figure 4.4

Figure 4.3 and 4.4: Effects of three release time at 45 dB HL input level on EDI of six classes of speech sounds for Group A and Group B mean audiometric data respectively.

Group A: As depicted in Table 4.1 and Figure 4.3, the mean EDI values decreased as the release time increased for all six classes of speech sounds. The SD is higher for speech sounds in the order of stops, glides, affricates, nasals, fricatives and liquids. On observation, the EDI values at 45 dB HL increased in comparison to 30 dB HL for all the conditions.

As depicted in Table 4.1a, the significant effect of release time on EDI at 45 dB HL, as revealed by the results of Wilcoxon Signed Rank test, are described for each class of speech sound.

Stops: Wilcoxon Signed Rank test, performed for comparison across release times, revealed that the RT pair of 40 and 1280 ms was significantly different.

Nasals: Wilcoxon Signed Rank test was performed for comparison across release times. This revealed that the RT pairs 40 and 640ms. as well as 40 and 1280 ms were significantly different at 0.05 level of significance.

Affricates, fricatives and liquids: Wilcoxon Signed Rank test revealed no significant difference between the three release times at 0.05 level of significance.

Glides: Wilcoxon Signed Rank test revealed a significant difference between all the three release times at 0.05 level of significance.

Group B: As depicted in Table 4.1 and figure 4.4, the mean EDI values decreased as the release time increased for all six classes of speech sounds. The SD is higher for speech sounds in the order of liquids, fricatives, affricates, glides and stops. On observation, the EDI values at 45 dB HL increased in comparison to 30 dB HL for all the conditions.

As depicted in Table 4.1b, the significant effect of release time on EDI at 45 dB HL, as revealed by the results of Wilcoxon Signed Rank test, are described for each class of speech sound.

Stops, fricatives, liquids and glides: Wilcoxon Signed Rank test revealed no significant difference between release times at 0.05 level of significance.

Nasals: Wilcoxon Signed Rank test revealed a significant difference between release times of 40 and 1280 ms at 0.05 level of significance.

Affricates: Wilcoxon Signed Rank test revealed significant difference between all the three release times at 0.05 level of significance.

c) At 65 dB HL

Descriptive statistics was done to compute the mean and SD at 65 dB HL input level for six classes of speech sounds. Table 4.1 depicts the mean and SD for EDI of six classes of speech sounds across three release times at 65 dB HL input levels for data of Group A and B. The effects are discussed for each Group (Group A & B) individually.

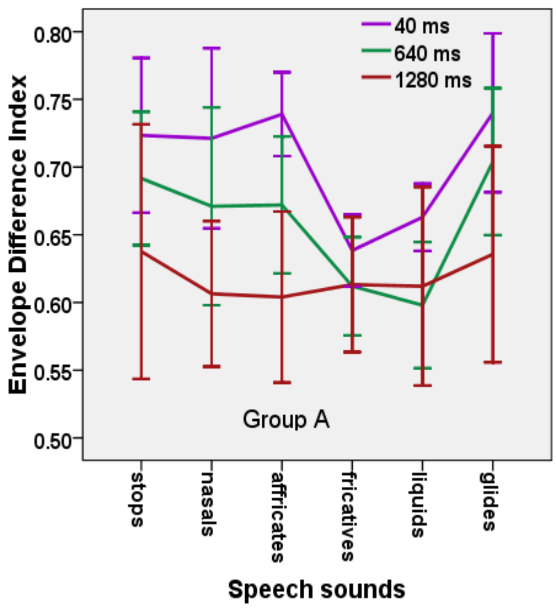


Figure 4.5

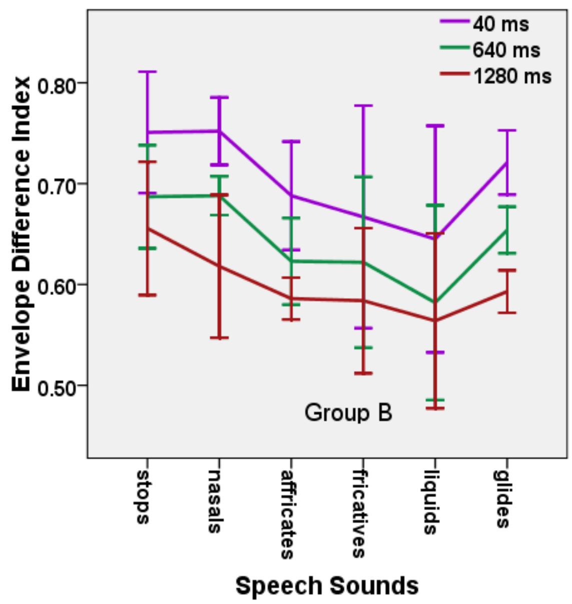


Figure 4.6

Figure 4.5and 4.6: Effect of three release time at 65 dB HL input level on EDI of six class of speech sounds for Group A and Group B mean audiometric data.

Group A: As depicted in Table 4.1 and Figure 4.5, the mean EDI values decreased as the release time increased for all six classes of speech sounds. The SD is higher for speech sounds in the order of stops, nasals, glides, liquids, affricates and fricatives. On observation, the EDI values at 65 dB HL increased in comparison to 30 and 45 dB HL for all the conditions.

As depicted in Table 4.1a, the significant effect of release time on EDI at 65 dB HL as revealed by the results of Wilcoxon Signed Rank test are described for each class of speech sound.

Stops: Wilcoxon Signed Rank test revealed that pairs 40 and 640ms. 40 and 1280 ms RT were significantly different at 0.05 level of significance.

Nasals and affricates: Wilcoxon Signed Rank test revealed significant difference between all the three release times at 0.05 level of significance.

Fricatives and liquids: Wilcoxon Signed Rank test revealed a significant difference between 40 and 640 ms RT at 0.05 level of significance.

Glides: Wilcoxon Signed Rank test revealed significant difference between 40 and 1280 ms RT at 0.05 level of significance.

Group B: As depicted in Table 4.1 and Figure 4.5, the mean EDI values decreased as the release time increased for all six classes of speech sounds. The SD is higher for speech sounds in the order of liquids, fricatives, stops, affricates, nasals and glides. On observation, the EDI values at 65 dB HL increased in comparison to 30 and 45 dB HL for all the conditions.

As depicted in Table 4.1b, the significant effect of release time on EDI at 65 dB HL as revealed by the results of Wilcoxon Signed Rank test are described for each class of speech sound.

Stops, fricatives, liquids: Wilcoxon Signed Rank test revealed that the RT pairs of 40 and 640ms, as well as 40 and 1280 ms were significantly different at 0.05 level of significance.

Nasals, affricates, glides: Wilcoxon Signed Rank test revealed significant difference between all the three release times at 0.05 level of significance.

4.1.1b Between Group comparison

Table 4.1 depicts the mean and SD of EDI values for six classes of speech sounds across three release times and across three input levels for Group A and B. Man Whitney U test, a test of two independent samples were performed to compare EDI values across all nine conditions (six classes of speech sounds) for the data of two groups. Results revealed significant difference between very few pairs. The EDI for the data of two groups differed under the conditions of, nasals at 30 dB HL with both 40 and 640 ms release time, nasals at 45 dB HL with 640 ms release time, fricatives at 45 dB HL with 640 ms release time and glides at 30 dB HL with 40 ms release time.

4.1.2 Log- likelihood Ratio (LLR)

Descriptive statistics was performed to compute the mean and standard deviation (SD) of LLR values for six classes of speech sounds (stops, nasals, affricates, fricatives, liquids & glides) at three release times and at three input levels. Since the measurement was done on five participants, non-parametric statistics was used for the analysis. Man Whitney U test, a test of two independent samples was performed in order to compare between groups. In order to compare across conditions two related samples test was performed. Table 4.2 depicts the mean and SD for LLR of six classes of speech sounds across three release times and across three input levels for data of Group A and B.

Table 4.2*: Mean and S.D of LLR for six classes of speech sounds at three input levels across three release times for Group A and Group B*

		LLR for Group A						***LLR for Group B***					
Release time (ms)	***Input Level dB HL***	***Stops***	***Nasals***	***Affricates***	***Fricatives***	***Liquids***	***Glides***	***Stops***	***Nasals***	***Affricates***	***Fricatives***	***Liquids***	***Glides***
		Mean (S.D)	***Mean (S.D)***	***Mean (S.D)***	***Mean (S.D)***	***Mean (S.D)***	***Mean (S.D)***	***Mean (S.D)***	***Mean (S.D)***	***Mean (S.D)***	***Mean (S.D)***	***Mean (S.D)***	***Mean (S.D)***
40 ms	30 dB	1.18 (0.27)	1.05 (0.38)	1.29 (0.48)	1.32 (0.47)	1.27 (0.61)	1.41 (0.42)	1.31 (0.32)	1.14 (0.40)	1.23 (0.45)	1.49 (0.67)	1.59 (0.73)	1.35 (0.44)
	45 dB	1.54 (0.45)	1.36 (0.34)	1.63 (0.56)	1.65 (0.66)	1.77 (0.78)	1.77 (0.60)	1.74 (0.81)	1.58 (0.64)	1.78 (0.88)	1.87 (0.91)	2.09 (1.19)	1.83 (0.87)
	65 dB	1.70 (0.52)	1.49 (0.29)	1.86 (0.53)	1.75 (0.62)	1.80 (0.61)	1.50 (0.51)	2.01 (0.74)	2.03 (0.59)	2.08 (0.80)	2.11 (0.85)	2.25 (1.1)	1.98 (0.78)
640 ms	30 dB	1.00 (0.52)	1.08 (0.29)	1.25 (0.35)	1.24 (0.24)	1.22 (0.44)	1.32 (0.25)	1.27 (0.22)	1.09 (0.31)	1.24 (0.38)	1.362 (0.50)	1.526 (0.61)	1.27 (0.34)
	45 dB	1.51 (0.37)	1.31 (0.32)	1.61 (0.54)	1.59 (0.52)	1.66 (0.64)	1.47 (0.32)	1.56 (0.60)	1.37 (0.54)	1.62 (0.70)	1.66 (0.74)	1.82 (0.95)	1.47 (0.61)
	65 dB	1.52 (0.46)	1.28 (0.35)	1.61 (0.45)	1.56 (0.61)	1.60 (0.63)	1.33 (0.49)	1.87 (0.66)	1.83 (0.50)	1.96 (0.70)	1.98 (0.80)	2.02 (0.95)	1.78 (0.71)
1280 ms	30 dB	1.13 (0.25)	0.958 (0.27)	1.21 (0.33)	1.24 (0.42)	1.22 (0.58)	1.26 (0.45)	1.14 (0.24)	.930 (0.31)	1.10 (0.22)	1.25 (0.39)	1.33 (0.51)	1.23 (0.38)
	45 dB	1.20 (0.30)	1.13 (0.38)	1.38 (0.48)	1.39 (0.49)	1.30 (0.52)	1.18 (0.18)	1.26 (0.38)	1.19 (0.29)	1.48 (0.53)	1.53 (0.54)	1.54 (0.68)	1.29 (0.43)
	65 dB	1.37 (0.48)	1.16 (0.37)	1.53 (0.48)	1.40 (0.55)	1.52 (0.58)	1.20 (0.49)	1.74 (0.47)	1.64 (0.46)	1.76 (0.52)	1.81 (0.69)	2.00 (0.95)	1.69 (0.56)

Table 4.2a: Significance of effect of release time on LLR for six classes of speech sounds at three input levels for Group B mean audiometric data.

		Stops		***Nasals***		***Affricates***		***Fricatives***		***Liquids***		***Glides***	
Input Level	***Release time (ms)***	***Z***	***Sig***	***Z***	***Sig***	***Z***	***Sig***	***Z***	***Sig***	***Z***	***Sig***	***Z***	***Sig***
30 dB HL	40 ms & 640ms	-.40	0.684	-1.0	0.285	-.40	0.684	-1.2	0.223	-.40	0.684	-.36	0.713
	40 ms & 1280ms	-2.0	0.042*	-1.5	0.131	-1.2	0.221	-2.0	0.042*	-1.7	0.078	-.94	0.343
	640 ms & 1280ms	-1.7	0.078	-2.0	0.042*	-1.6	0.102	-.94	0.343	-1.7	0.078	-.67	0.498
45 dB HL	40 ms & 640ms	-1.2	0.223	-2.0	0.042*	-1.2	0.223	-2.0	0.042*	-1.2	0.223	-2.0	0.04*
	40 ms & 1280ms	-2.0	0.042*	-2.0	0.042*	-1.2	0.223	-1.2	0.223	-1.2	0.221	-2.0	0.042*
	640 ms & 1280ms	-2.0	0.041*	-1.2	0.223	-.40	0.684	-.67	0.498	-1.6	0.109	-2.0	0.042*
65 dB HL	40 ms & 640ms	-1.7	0.078	-2.0	0.042*	-1.6	0.109	-2.0	0.042*	-2.0	0.042*	-2.0	0.042*
	40 ms & 1280ms	-1.2	0.223	-2.0	0.042*	-1.6	0.109	-2.0	0.042*	-2.0	0.042*	-2.0	0.042*
	640 ms & 1280ms	-1.2	0.223	-2.0	.042*	-1.0	.285	-2.0	.042*	-.67	.498	-1.2	.221

*Note:- *: significant difference at p<0.05*

Table 4.2b: Significant effect of release time on LLR for six classes of speech sounds at three input levels for Group A mean audiometric data

		Stops		**Nasals**		**Affricates**		**Fricatives**		**Liquids**		**Glides**	
Input Level	***Release time***	***Z***	***Sig***	***Z***	***Sig***	***Z***	***Sig***	***Z***	***Sig***	***Z***	***Sig***	***Z***	***Sig***
30 dB HL	40 ms & 640ms	-.94	0.345	-1.4	0.686	-.27	0.786	-.94	0.345	-.67	0.500	-.40	0.686
	40 ms & 1280ms	-.67	0.500	-2.0	0.138	-.67	0.500	-1.7	0.080	-1.0	0.279	-.67	0.500
	640 ms & 1280ms	-.13	0.893	-1.8	0.043*	-.67	0.500	.00	1.00	.00	1.0	-.40	0.686
45 dB HL	40 ms & 640ms	-.67	0.500	-1.4	0.068	-.67	0.500	-.67	0.500	-1.0	0.273	-1.7	0.080
	40 ms & 1280ms	-2.0	0.042 *	-.94	0.138	-2.0	0.043 *	-1.4	0.138	-2.0	0.043 *	-2.0	0.043 *
	640 ms & 1280ms	-2.0	0.043*	-2.0	0.345	-1.0	0.279	-.40	0.686	-2.0	0.043 *	-2.0	0.043 *
65 dB HL	40 ms & 640ms	-1.7	0.080	-2.0	0.043*	-2.0	0.043 *	-2.0	0.042*	-2.0	0.043 *	-1.4	0.138
	40 ms & 1280ms	-1.7	0.080	-1.4	0.043*	-2.0	0.043 *	-2.0	0.043*	-2.0	0.043 *	-2.0	0.043 *
	640 ms & 1280ms	-2.0	0.043*	-1.4	0.138	-1.7	0.080	-2.0	0.042*	-1.0	0.273	-2.0	0.043 *

*Note:- *: significant difference at p<0.05*

Table 4.2c: Significance of effect of release on LLR time for six classes of speech sounds at three input levels for group A and Group B mean audiometric data.

Group A									
Input level	***30 dB HL***			***45 dB HL***			***65 dB HL***		
Release Time	***40 ms and 640 ms***	***40 ms and 1280 ms***	***640ms and 1280 ms***	***40 ms and 640 ms***	***40 ms and 1280 ms***	***640 ms and 1280 ms***	***40 ms and 640 ms***	***40 ms and 1280 ms***	***640 ms and 1280 ms***
Stops		*	*						*
Nasals			*					*	*
Affricates					*		*	*	
Fricatives							*	*	*
Liquids					*	*	*	*	
Glides					*	*	*	*	*
Group B									
Input level	***30 dB HL***			***45 dB HL***			***65 dB HL***		
Release Time	***40 ms and 640 ms***	***40 ms and 1280 ms***	***640ms and 1280 ms***	***40 ms and 640 ms***	***40 ms and 1280 ms***	***640 ms and 1280 ms***	***40 ms and 640 ms***	***40 ms and 1280 ms***	***640 ms and 1280 ms***
Stops		*			*	*			
Nasals			*	*	*		*	*	*
Affricates									
Fricatives		*		*			*	*	*
Liquids							*	*	
Glides				*	*	*	*	*	

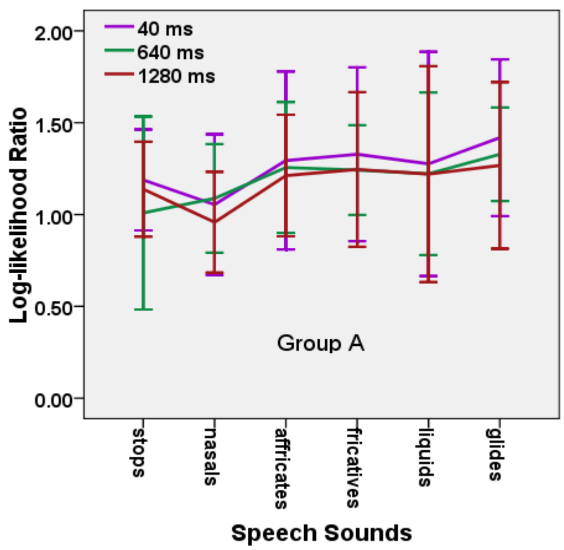
*Note:- *: significant difference at $p<0.05$*

4.1.2a Effect of release time on Log-likelihood Ratio

Table 4.2 depicts the mean and SD for LLR of six classes of speech sounds across three release times and across three input levels for data of Group A and B. To evaluate the effect of release time on LLR, two related sample test i.e., Wilcoxon Signed rank test was performed to find out the significant difference between pairs of the nine conditions for six groups of speech sounds, depicted in Table 4.2a. This test was administered separately for the data of Group A and B. The results of comparison across three release time conditions are discussed below at each input level.

a) At 30 dB HL

Descriptive statistics was done to compute the mean and SD at 30 dB HL input level for six classes of speech sounds. Table 4.2 depicts the mean and SD for LLR of six classes of speech sounds across three release times at 30 dB HL input levels for data of Group A and B. The effects are discussed for each Group (Group A & B) individually.



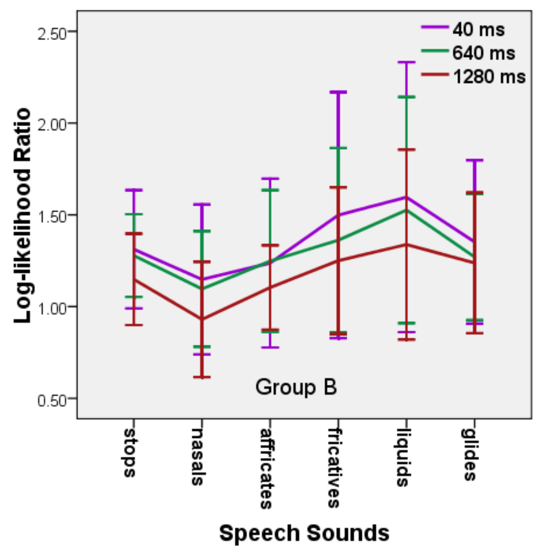


Figure 4.7 ***Figure 4.8***

Figure 4.7 and 4.8: Effect of three release time at 30 dB HL input level on LLR of six classes of speech sounds for Group A and Group B mean audiometric data.

Group A: As depicted in Table 4.2 and Figure 4.7, the mean value of LLR for these speech sounds did not vary systematically as the release time was increased. The SD is higher for speech sounds in the order of liquids, glides, affricates, fricatives, stops and nasals. As depicted in Table 4.2 b, the significant effect of release time on LLR at 30 dB HL as revealed by the results of Wilcoxon Signed Rank test are described for each class of speech sound.

Stops, affricates, fricatives, liquids and glides: Wilcoxon signed rank test did not reveal significant difference across release time conditions.

Nasals: Wilcoxon signed rank test revealed significant difference for the RT pair of 640 and 1280 ms, at 0.05 level of significance.

Group B: As depicted in Table 4.2 and Figure 4.8, the mean value of LLR for all speech sounds decreased as the release time was increased. The SD is higher for speech sounds in the order of liquids, fricatives, glides, affricates, stops and nasals.

As depicted in Table 4.2b, the significant effect of release time on LLR at 30 dB HL as revealed by the results of Wilcoxon Signed Rank test are described for each class of speech sound.

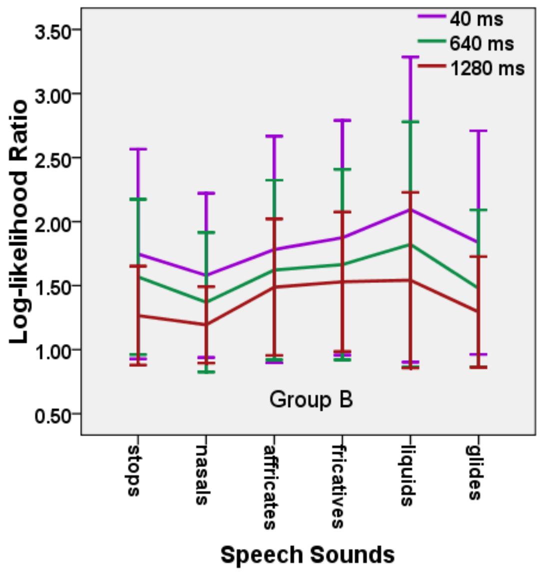
Stops and fricatives: Wilcoxon signed rank test revealed a significant difference for the RT of 40 and 1280 ms, at 0.05 level of significance.

Nasals: Wilcoxon signed rank test revealed significant difference for the RT pair of 640 and 1280 ms, at 0.05 level of significance.

Affricates, liquids and glides: Wilcoxon signed rank test did not reveal a significant difference across release time conditions.

b) At 45 dB HL

Descriptive statistics was done to compute the mean and SD at 45 dB HL input level for six classes of speech sounds. Table 4.2 depicts the mean and SD of LLR for six classes of speech sounds, across three release times at 45 dB HL input level, for the data of Groups A and B. The effects are discussed for each Group (Group A & B) individually.



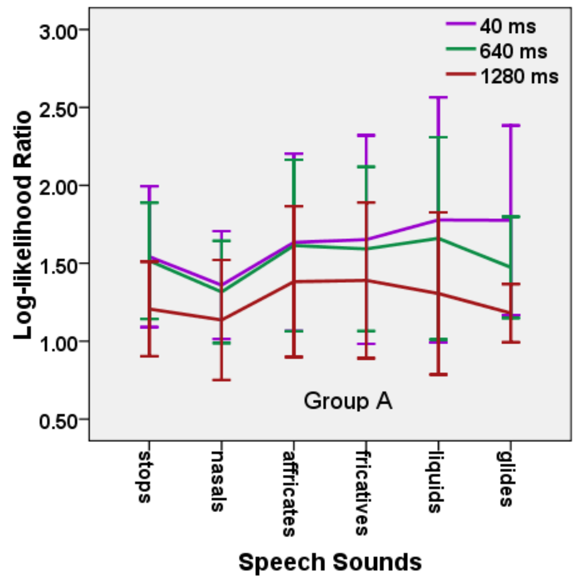


Figure 4.9 ***Figure 4.10***

Figure 4.9 and 4.10: Effects of three release time at 45 dB HL input level on LLR of six class of speech sounds for Group A and Group B mean audiometric data.

Group A: As depicted in Table 4.2 and Figure 4.9, the mean value of LLR for all the speech sounds decreased as the release time was increased. The SD is higher for speech sounds in the order of liquids, fricatives, affricates, glides, stops and nasals. On observation the LLR values at 45 dB HL increased in comparison to 30 dB HL.

As depicted in Table 4.2a, the significant effect of release time on LLR at 45 dB HL as revealed by the results of Wilcoxon Signed Rank test are described for each class of speech sound.

Stops: Wilcoxon signed rank test revealed significant difference for RT pairs of 40 and 640 ms, as well as 40 and 1280 ms, at 0.05 level of significance.

Nasals and fricatives: Wilcoxon signed rank test revealed no significant difference across release time conditions.

Affricates: Wilcoxon signed rank test revealed significant difference for 40 and 1280 ms release time conditions at 0.05 level of significance.

Liquids and glides: Wilcoxon signed rank test revealed significant difference for 40 and 1280 ms, 640 and 1280 ms release time conditions at 0.05 level of significance.

Group B: As depicted in Table 4.2 and Figure 4.10, the mean value of LLR for all the speech sounds decreased as the release time was increased. The SD is higher for speech sounds in the order of liquids, fricatives, affricates, glides, stops and nasals. On observation the LLR values at 45 dB HL increased in comparison to 30 dB HL.

As depicted in Table 4.2b, the significant effect of release time on LLR at 45 dB HL as revealed by the results of Wilcoxon Signed Rank test are described for each class of speech sound.

Stops: Wilcoxon signed rank test revealed significant difference for 40 and 1280 ms, 640 and 1280 ms release time conditions at 0.05 level of significance.

Nasals: Wilcoxon signed rank test revealed a significant difference for 40 and 640 ms, 40 and 1280 ms release time conditions at 0.05 level of significance.

Affricates and Liquids: Wilcoxon signed rank test revealed no significant difference across release time conditions.

Fricatives: Wilcoxon signed rank test revealed significant difference between 40 and 640 ms release time conditions at 0.05 level of significance.

Glides: Wilcoxon signed rank test revealed significant difference across all three release time conditions at 0.05 level of significance.

c) At 65 dB HL

Descriptive statistics was done to compute the mean and SD at 65 dB HL input level for six classes of speech sounds. Table 4.2 depicts the mean and SD for EDI of six classes of speech sounds across three release times at 65 dB HL input levels for data of Group A and B. The effects are discussed for each Group (Group A & B) individually.

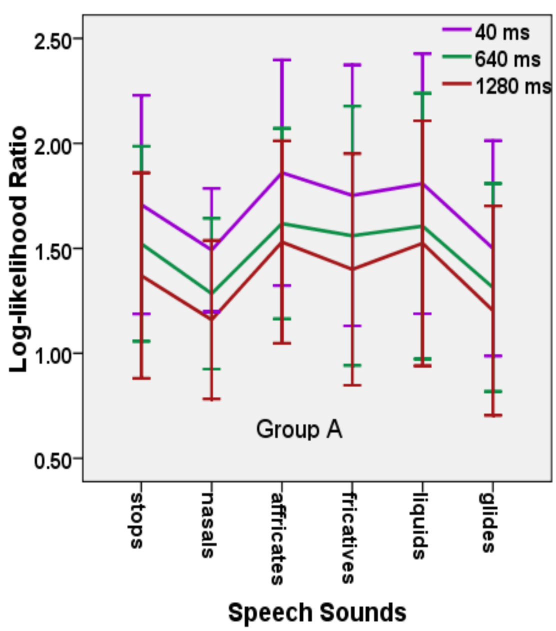


Figure 4.11

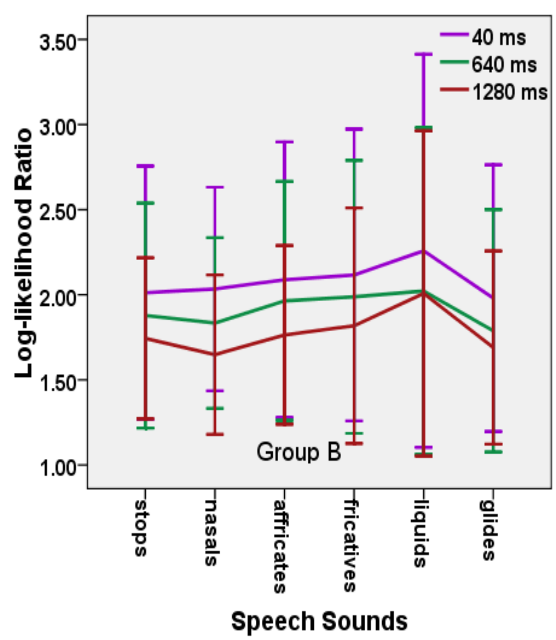


Figure 4.12

Figure 4.11and 4.12: Effect of three release time at 65 dB HL input level on LLR of six class of speech sounds for Group A and Group B mean audiometric data

Group A: As depicted in Table 4.2 and Figure 4.11, the mean value of LLR for all the speech sounds decreased as the release time was increased. The SD is higher for speech sounds in the order of liquids, fricatives, affricates, glides, stops and nasals. On observation the LLR values at 65 dB HL increased in comparison to 30 and 45 dB HL.

As depicted in Table 4.2a, the significant effect of release time on LLR at 65 dB HL as revealed by the results of Wilcoxon Signed Rank test are described for each class of speech sound.

Stops: Wilcoxon signed rank test revealed a significant difference in the pair of RT for 40 and 1280 ms, at 0.05 level of significance.

Nasals, affricates and liquids: Wilcoxon signed rank test revealed significant difference for 40 and 640 ms, 40 and 1280 ms release time conditions at 0.05 level of significance.

Fricatives: Wilcoxon signed rank test revealed a significant difference across all the three release time conditions at 0.05 level of significance.

Glides: Wilcoxon signed rank test revealed significant difference for 40 and 1280 ms, 640 and 1280 ms RT at 0.05 level of significance.

Group B: As depicted in Table 4.2 and Figure 4.12, the mean value of LLR for all the speech sounds decreased as the release time was increased. The SD is higher for speech sounds in the order of liquids, fricatives, affricates, glides, stops and nasals. On observation the LLR values at 65 dB HL increased in comparison to 30 and 45 dB HL.

As depicted in Table 4.2b, the significant effect of release time on LLR at 65 dB HL as revealed by the results of Wilcoxon Signed Rank test are described for each class of speech sound.

Stops, affricates and fricatives: Wilcoxon signed rank test revealed no significant difference across release time conditions.

Nasals: Wilcoxon signed rank test revealed significant difference across all the three release time conditions at 0.05 level of significance.

Liquids and glides: Wilcoxon signed rank test revealed significant difference between the RT pairs 40 and 640 ms, as well as 40 and 1280 ms, at 0.05 level of significance.

4.1.2b Between Group comparison

Table 4.2 depicts the mean and SD of LLR values for six classes of speech sounds across three release times and across three input levels for Group A and B. Two independent sample test was performed to compare LLR values across all nine conditions (six classes of speech sounds) for the data of two groups. Results revealed significant difference between the LLR for the data of two groups for nasals at 65 dB HL with 640 ms release time condition.

4.2 Behavioural measures

The behavioural measures, SIS and quality ratings, were obtained across three release times (40 ms, 640 ms & 1280 ms) and across three input levels (30 dB HL, 45 dB HL & 65 dB HL) for Group A and Group B.

4.2.1 Speech identification scores

The speech identification scores (SIS) were obtained across three release times (40 ms, 640 ms & 1280 ms) and across three input levels (30 dB HL, 45 dB HL & 65 dB HL) for Group A (N=10) and Group B (N=10) participants.

Descriptive statistics was done to get the mean and standard deviation. Repeated Measure ANOVA was administered to find out the overall interaction of the conditions (within subject factors) and groups (between subject factors). Conditions being SIS across three release times and across three input levels; and groups being Group A and Group B. Bonferroni's multiple pair-wise comparison was done, when indicated, to determine the pairs that are significantly different from each other.

4.2.1a Effects of Release Time on SIS

Descriptive statistics was performed to compute the mean and standard deviation (SD) of SIS. Table 4.3 depicts the mean and SD of SIS, across three release times and across three input levels, for Group A and Group B.

Table 4.3: *Mean and SD of SIS across three input levels and across three release times for Group A and Group B.*

Input Level (in dB HL)	*Release time (ms)*	*Group A*	*Group B*	*Between Groups*
		Mean (SD)	*Mean (SD)*	*p*
30	40 ms	14.70 (0.68)	11.90 (0.88)	0.00 **
	640 ms	15.70 (0.82)	11.60 (1.08)	0.00**
	1280 ms	15.50 (1.08)	12.50 (0.71)	0.00**
45	40 ms	17.50 (0.53)	15.00 (0.82)	0.00**
	640 ms	18.00 (0.67)	15.00 (0.82)	0.00**
	1280 ms	18.10 (0.74)	15.70 (0.68)	0.00**
65	40 ms	19.10 (0.32)	16.80 (0.92)	0.00**
	640 ms	19.10 (1.10)	16.90 (1.4)	0.01*
	1280 ms	19.10 (1.19)	17.20 (0.92)	0.01*

*Note:- **: significant difference at $p<0.01$, *: significant difference at $p<0.05$*

As depicted in Table 4.3, the mean SIS across all the nine conditions are greater for Group A compared to Group B. Two way repeated measure ANOVA was performed to determine the interaction between group (Group A & Group B) and conditions (SIS across three release times and across three input levels). Results revealed that the interaction between group and conditions are statistically significant [F (1, 64.14) =3.283, p<0.05]. In addition, significant main effect of conditions [F (8, 144) =119.56, p<0.05] was also present. Bonferroni's multiple comparison test was administered to find out the pairs that are significantly different. The results are discussed below at each input level. Results are discussed individually for Group A and B.

Group A:

The mean SIS varied across release times, and the effect of release time on SIS scores were different across input levels. The results are discussed under each of the input levels.

a. At 30 dB HL input level

Table 4.3 represents mean and SD of SIS. At an input level of 30 dB HL, the mean SIS increased as the release time increased. Lowest SIS scores were obtained for short release time of 40 ms. Mean SIS increased for 640, and the scores were almost similar for release time of 640 and 1280 ms. The SD was higher at 1280 ms followed by 640 ms and least for 40 ms release time condition.

Repeated measure ANOVA was performed for comparison across release times within a particular input level. The SIS for different release times were significantly different [F(2, 18) 7.132: p<0.05]. Pair-wise differences with Bonferroni's multiple comparison showed a significant difference between 40 and 640 ms, at 0.05 level of significance.

b. At 45 dB HL input level

The mean SIS at 45 dB HL increased as the release time increased. Lowest SIS were obtained for shorter RT of 40 ms. For the mid and longer RT, the mean SIS were almost equivalent. The SD was higher at 1280 ms followed by 640 ms and least for 40 ms release time condition. Repeated measure ANOVA was performed for comparison across release times within a particular input level. The SIS for different release times were significantly different [F(2, 18) 7.154: $p<0.05$]. Pair-wise differences among RT were tested with Bonferroni's multiple comparison. The results showed a significant difference between 40 and 640 ms and also between 40 and 1280 ms at 0.05 level of significance.

c. At 65 dB HL input level:

The mean SIS scores obtained were similar across three release times. The SD was higher at 1280 ms followed by 640 ms and least for 40 ms release time condition. Statistical analysis of repeated measure ANOVA showed no significant differences across the three release time conditions.

Group B:

The mean SIS slightly increased as the release time increased at each input level. Repeated ANOVA measures showed no significant difference across RT conditions in any of the input levels. The SD was higher at 640 ms followed by 40 ms and least for 1280 ms release time condition at 30 dB HL. The SD was higher at 640 ms and 40 ms and least for 1280 ms release time condition at 45 dB HL. The SD was higher at 640 ms and least for 40 ms and 1280 ms release time condition at 65 dB HL input level..

On observation, for both the groups, the SIS increased as the input level increased from 30 to 45 to 65 dB HL.

4.2.1c Between Group comparison

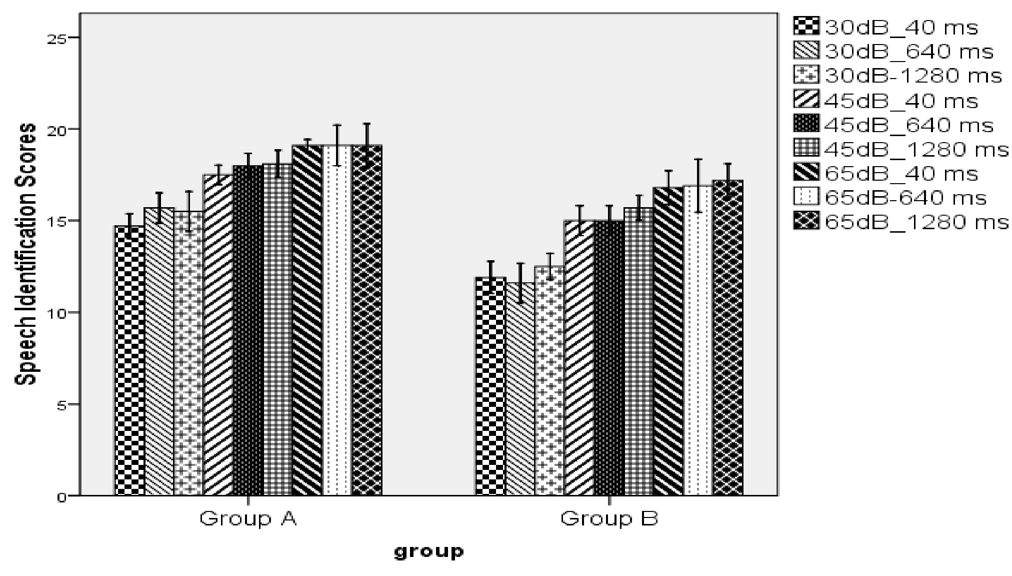


Figure 4.13: SIS at three release time and three input levels for Group A audiometric data

Table 4.3 depicts the mean and SD of SIS across three release times and across three input levels, for the Groups A and B. Repeated measure ANOVA results revealed that all the conditions were significantly different between the two groups [F (8, 144) =18, p<0.05].

4.2.2 Perceptual quality ratings (loudness, clarity, naturalness & overall impression)

Descriptive statistics was done to compute the mean and standard deviation. Repeated Measure ANOVA was administered to find out the overall interaction of the conditions and groups. Bonferroni's multiple pair-wise comparison was done to determine the pairs that are significantly different from each other.

4.2.2a Effect of release time on perceptual quality ratings (loudness, clarity, naturalness & overall impression)

Descriptive statistics was performed to compute the mean and standard deviation (SD) of perceptual rating of loudness, clarity, naturalness and overall impressionparameters. Table 4.4 depicts the mean and SD for loudness, clarity, naturalness and overall impression parameters across three release times and across three input levels for Group A and Group B. Two way repeated measure ANOVA was performed to reveal significant difference across nine conditions. If the test revealed significant difference across nine conditions, Bonferroni's multiple pair-wise comparison was done to reveal the pairs which are significantly different. The effects of release times for each of the parameter are discussed at each input level for two groups individually.

Table 4.4: *Effect of three release time across three input levels on perceptual ratings of clarity for Group A and Group B*

		Group A – Quality ratings				***Group B – Quality ratings***			
Release time (ms)	***Input Level dBHL***	***Loudness***	***Clarity***	***Naturalness***	***Overall Impression***	***Loudness***	***Clarity***	***Naturalness***	***Overall Impression***
		Mean (S.D)	***Mean (S.D)***	***Mean (S.D)***	***Mean (S.D)***	***Mean (S.D)***	***Mean (S.D)***	***Mean (S.D)***	***Mean (S.D)***
40 ms	30 dB HL	5.20 (1.31)	5.00 (1.33)	4.60 (1.21)	5.20 (1.13)	4.20 (0.76)	4.50 (1.08)	4.65 (0.88)	4.60 (0.69)
	45 dB HL	6.90 (1.28)	6.50 (1.26)	6.00 (1.41)	6.90 (1.37)	6.30 (1.39)	5.80 (1.54)	5.65 (1.29)	6.50 (1.17)
	65 dB HL	8.50 (1.08)	7.60 (1.50)	6.30 (1.41)	8.30 (0.91)	8.70 (0.94)	7.20 (1.39)	6.65 (1.29)	8.00 (0.47)
640 ms	30 dB HL	4.90 (1.30)	5.50 (1.17)	5.20 (0.63)	5.50 (1.64)	4.00 (0.91)	4.70 (0.63)	5.10 (0.56)	5.25 (0.97)
	45 dB HL	6.60 (1.56)	6.90 (1.10)	7.20 (1.03)	7.40 (0.96)	6.65 (0.88)	6.50 (1.08)	6.90 (0.73)	7.30 (0.91)
	65 dB HL	8.15 (2.10)	8.20 (1.31)	7.40 (1.07)	8.60 (1.42)	8.80 (0.78)	7.55 (1.77)	7.15 (1.41)	8.35 (1.20)
1280 ms	30 dB HL	5.10 (1.10)	5.80 (1.22)	5.90 (0.99)	6.00 (0.94)	4.00 (0.93)	4.85 (0.66)	5.45 (0.72)	5.75 (0.85)
	45 dB HL	6.40 (1.26)	7.80 (1.39)	7.40 (0.96)	8.10 (0.96)	6.30 (0.94)	7.20 (0.42)	7.25 (0.54)	7.45 (0.76)
	65 dB HL	8.60 (1.76)	8.80 (1.31)	8.20 (0.91)	9.20 (1.00)	8.40 (0.69)	7.65 (1.70)	7.60 (1.17)	8.40 (0.93)

Table 4.4a*: Significance of effect of three release time across three input levels on perceptual ratings of clarity for Group A and Group B*

Input Level	*Release time*	*Group A*				*Group B*			
		Loudness	*Clarity*	*Naturalness*	*Overall impression*	*Loudness*	*Clarity*	*Naturalness*	*Overall impression*
		p	*p*	*p*	*p*	*p*	*p*	*p*	*p*
30 dB HL	40 ms & 640 ms	0.987	0.157	0.153	0.837	0.733	1.000	0.153	0.040
	40 ms &1280 ms	1.000	0.261	0.006	0.009	0.638	0.397	0.006	0.000*
	640 ms &1280 ms	1.000	1.000	0.029	0.532	1.000	1.000	0.029	0.045*
45 dB HL	40 ms & 640 ms	1.000	1.000	0.153	0.287	0.726	0.199	0.153	0.040*
	40 ms & 1280 ms	0.896	0.172	0.038	0.126	1.000	0.075	0.038	0.005
	640 ms & 1280 ms	0.638	0.012	1.000	0.062	0.726	0.134	1.000	0.580
65 dB HL	40 ms & 640 ms	1.000	0.153	0.020	0.287	1.000	0.814	0.020	0.965
	40 ms & 1280 ms	1.000	0.175	0.007	0.126	0.837	0.759	0.007	0.630
	640 ms &1280 ms	0.613	0.334	0.067	0.62	0.504	1.000	0.067	1.000

*Note:- *: significant difference at $p<0.05$*

1) Loudness

Table 4.4 depicts the mean and SD of loudness ratings across release times for each of the input level. The mean ratings of loudness did not vary systematically across release times at a particular intensity. The SD did not vary systematically across release times and across each input level for both the groups. In order to evaluate the effect of release time, two way repeated measure ANOVA was performed. The results (Table 4.4a) revealed a significant interaction between the conditions and groups [F (8, 144)3.789: $p<0.05$]. The results did not reveal across nine conditions for both Group A and Group B.

2) Clarity:

Table 4.4 depicts the mean and SD of clarity ratings across release times for each of the input level. The mean ratings of clarity increased as the release time increased. This pattern was observed at all the input levels and for both the groups (Group A and Group B). The SD did not vary systematically across release times and across each input level for both the groups. In order to evaluate the effect of release time, two way repeated measure ANOVA was performed. The results did not reveal a significant interaction between the conditions and groups. The results revealed a significant main effect across nine conditions [F (8, 144)=44.30: $p<0.05$]. Bonferroni's multiple comparison was administered to reveal significant difference between pairs (Table 4.4a). The effect of release times are discussed for both the groups at each input level.

Group A:

At all the input levels, the mean ratings of clarity increased as the release time was increased. The significant effect of release time at each input level are depicted in Table 4.4a.

a) At 30 dB HL

At 30 dB HL, the mean ratings of clarity increased as release time increased (Table 4.4). This effect is not statistically significant as revealed by the Bonferroni's pair-wise comparisons.

b) At 45 dB HL

The mean ratings of clarity increased systematically as the release time increased from 40 to 1280 ms. Pair-wise comparison using Bonferroni's multiple comparison revealed that 640 and 1280 ms pair are significantly different.

c) At 65 dB HL

The mean ratings of clarity remained the same as the release time increased. This effect is not statistically significant as revealed by the pair-wise comparisons.

Group B:

At all the input levels, the mean ratings of clarity increased as the release time was increased. The significant effect of release time at each input level are depicted in Table 4.4a.

a) At 30 dB HL, at 45 dB HL & at 65 dB HL

The mean ratings of clarity increased systematically as the release time increased from 40 to 1280 ms. This effect is not statistically significant as revealed by the results of Bonferroni's multiple comparison.

3) Naturalness:

Descriptive statistics was performed to compute the mean and standard deviation (SD) of perceptual rating of naturalness parameter. Table 4.4 depicts the mean and SD for naturalness parameter across three release times and across three input levels for Group A and Group B. The mean ratings of naturalness increased as the release time increased. This pattern is observed at all the input levels and for both the groups (Group A and Group B). The SD did not vary systematically across release times and across each input level for both the groups. In order to evaluate the effect of release time, two way repeated measure ANOVA was performed. Results revealed that the interaction between group and conditions was not statistically significant. The results revealed significant difference across nine conditions [$F(8, 144)=41.09$: $p<0.05$]. Bonferroni's multiple comparison was administered to measure the significant difference between the pairs. The effect of release times are discussed for both the groups at each input level.

Group A:

At all the input levels, the mean ratings of naturalness increased as the release time was increased. The significant effect of release time at each input level are depicted in Table 4.4a.

a) At 30 dB HL

The mean ratings for naturalness increased as the release time increased. Results of pair-wise differences of different release times were tested with Bonferroni's multiple comparison. Results revealed significant difference between the RT of 40 and 1280, as well as 640 and 1280 ms, at 0.05 level of significance.

b) *At 45 dB HL*

The mean ratings for naturalness increased as the release time increased. Results of pair wise differences of three release time pairs at each of the input level was tested with Bonferroni's multiple comparison. Results revealed significant difference between 40 and 1280 release time conditions, at 0.05 level of significance.

c) At 65 dB HL

The mean ratings for naturalness increased as the release time increased. Results of pair-wise differences were tested with Bonferroni's multiple comparison. Results revealed significant difference between 40 and 640 ms, 40 and 1280 release time conditions, at 0.05 level of significance. Remaining pair, 640 and 1280 ms, was not significantly different.

Group B:

Group A: At all the input levels, the mean ratings of naturalness increased as the release time was increased. The significant effect of release time at each input level are depicted in Table 4.4a.

a) At 30 dB HL

The mean ratings for naturalness increased as the release time increased. Results of pair-wise differences were tested with Bonferroni's multiple comparison. Results revealed significant difference between 40 and 640, as well as 40 and 1280 ms release time conditions, at 0.05 level of significance.

b) At 45 dB HL

The mean ratings for naturalness increased as the release time increased. Results of pair-wise differences were tested with Bonferroni's multiple comparison. Results revealed significant difference between 40 and 640, as well as 40 and 1280 ms release time conditions at 0.05 level of significance.

c) At 65 dB HL

The mean ratings for naturalness increased as the RT increased. Pair-wise comparison was tested with Bonferroni's multiple comparison. The results revealed significant difference between 40 and 1280 release time conditions at 0.05 level of significance.

4) Overall impression:

Descriptive statistics was performed to compute the mean and standard deviation (SD) of perceptual rating of overall impression parameter. Table 4.4 depicts the mean and SD for overall impression parameter across three release times and across three input levels for Group A and Group B. The SD did not vary systematically across release times and across each input level for both the groups. In order to evaluate the effect of release time, two way repeated measure ANOVA was performed. Results revealed that the interaction between group and conditions was not statistically significant. The results also revealed significant difference across nine conditions [$F(8, 144)=68.4$: $p<0.05$]. Bonferroni's multiple comparison was administered to reveal significant difference between pairs. The effect of release times are discussed for both the groups at each input level.

Group A:

At all the input levels, the mean ratings of overall impression increased as the release time was increased. The significant effect of release time at each input level are depicted in Table 4.4a.

a) At 30 dB HL

The mean ratings of overall impression increased as the release time increased. Bonferroni's multiple comparison test revealed a significant difference of the RT pair 40 and 1280 ms at 0.05 level of significance.

b) At 45 dB HL

The mean ratings of overall impression increased as the release time increased. Bonferroni's multiple comparison tests did not reveal any significant difference across release times.

c) At 65 dB HL

The mean ratings of overall impression increased as the release time increased. Bonferroni's multiple comparison tests did not reveal any significant difference across release times.

Group B:

At all the input levels, the mean ratings of overall impression increased as the release time was increased. The significant effect of release time at each input level are depicted in Table 4.4a.

a) At 30 dB HL

The mean ratings of overall impression increased as the release time increased. Bonferroni's multiple comparison test revealed significant difference for all the pairs.

b) At 45 dB HL

The mean ratings of overall impression increased as the release time increased. Bonferroni's multiple comparison test revealed significant difference between the RT of 40 and 640, as well as 40 and 1280 ms.

c) At 65 dB HL

The mean ratings of overall impression increased as the release time increased. Bonferroni's multiple comparison test did not reveal any significant difference across release times.

4.2.2c. Between group comparison

Table 4.4 depicts the mean and SD of perceptual quality ratings of four parameters across three release times and across three input levels for Group A and B. For each of the parameters, repeated measure ANOVA was administered to rule out the interaction between groups and conditions. Bonferroni's pair-wise test of multiple comparisons was administered to find out the significant difference between groups across conditions. The results are discussed for each of the parameters individually.

1) Loudness

As depicted in Table 4.3, the mean rating of loudness parameter across most of the conditions are greater for Group A than Group B. Two way repeated measure ANOVA was performed to determine the interaction between groups (Group A & Group B) and conditions (three release times and three input levels). The results revealed that the interaction between group and conditions are statistically significant. Since there existed significant interactions, Bonferroni's multiple comparison test was administered to find out the pairs which were significantly different. Results revealed loudness rating at 30 dB HL of 1280 ms release time condition was significantly different between two groups at 0.05 level of significance.

2) Clarity:

As depicted in Table 4.4, the mean rating for clarity parameter across all of the conditions are greater for Group A than compared to Group B. Two way repeated measure ANOVA was performed to determine the interaction between group (Group A & Group B) and conditions (clarity ratings across three release times and across three input levels). Results revealed that the interaction between group and conditions are not statistically significant.

3) Naturalness:

As depicted in Table 4.4, the mean rating of naturalness parameter across most of the conditions are greater for Group A than compared to Group B. Two way repeated measure ANOVA was performed to determine the interaction between group (Group A &

Group B) and conditions (naturalness ratings across three release times and across three input levels). Results revealed that the interaction between group and conditions are not statistically significant.

4) Overall Impression:

As depicted in Table 4.4, the mean rating of overall impression parameter across all of the conditions are greater for Group A than compared to Group B. Two way repeated measure ANOVA was performed to determine the interaction between group (Group A & Group B) and conditions (overall impression ratings across three release times and across three input levels). Results revealed that the interaction between group and conditions are not statistically significant.

CHAPTER V

DISCUSSION

In order to evaluate the effect of release time on acoustic and behavioural measures of speech, the data was collected, tabulated and analysed using appropriate statistical analysis. The results of the study are discussed in this section.

5.1 Acoustic measures

The effects of release time on acoustic measures of speech were investigated using Envelope Difference Index and Log-likelihood Ratio.

To recall in brief, ED1 on a scale ranges from 1.00 which represents no correspondence between envelopes to 0.00 which represents perfect correspondence between envelopes.

LLR is inherently positive and, a low value indicates a close agreement between the spectral magnitudes of a test signal and a reference signal. A zero distance means that the spectral magnitudes of both signals are identical and a large spectral distance means the signals are significantly different.

5.1.1 Envelope Difference Index (EDI)

Table 4.1 depicts the mean and SD for EDI of six classes of speech sounds across three release times and across three input levels for data of Group A and B. To evaluate the effect of release time on EDI, two related sample test i.e., Wilcoxon Signed rank test was performed to find out the significant difference between pairs of the nine conditions for six groups of speech sounds Table (4.1a & 4.1b). This test was administered separately for the data of Group A and B. The results of the EDI are summarized below.

Table 5.1: *Summary of results of EDI with Group A mean audiometric data*

Group A	***30 dB HL***	***45dB HL***	***65 dB HL***
Stops	640 ms	1280 ms *	1280 ms *
Nasals	640 ms	640ms & 1280 ms*	1280 ms *
Affricates	640 & 1280 ms	1280 ms	1280 ms *
Fricatives	1280 ms	1280 ms	640 & 1280 ms *
Liquids	1280 ms	1280 ms	640 ms *
Glides	40 ms	1280 ms *	1280 ms *

*Note:- *: Significantly reduced temporal envelope distortion at particular release time compared to other release times (p<0.05).*

Table 5.2: *Summary of results of EDI with Group B mean audiometric data*

Group B	***30 dB HL***	***45dB HL***	***65 dB HL***
Stops	1280	1280	1280 *
Nasals	640 & 1280	1280 *	1280 *
Affricates	1280	1280 *	1280 *
Fricatives	640 & 1280	1280	1280 *
Liquids	640 & 1280	1280	1280 *
Glides	1280	1280	1280 *

*Note:- *: Significantly reduced temporal envelope distortion at particular release time compared to other release times (p<0.05).*

Tables 5.1 and 5.2 summarize the effect of three release time on EDI for six classes of speech sounds. As described in the above Tables 5.1 and 5.2, at each input level, and for each class of speech sound, the release time which creates significantly lesser distortion than the others are mentioned. Although the other release times are significantly different than each other as revealed by Wilcoxon Signed Rank test, the summary will be just focused on the release time which induces lesser distortions.

5.1.1a Effect of release times on six groups of speech sounds

The results of comparison across three release time conditions are discussed below at each input level.

1) At 30 dB HL

The results are in consonance with the study by Jenstad and Souza (2005). According to the study, at lower input levels of 50 dB SPL, the EDI values undergone minimal changes across release times used (12 ms, 100 ms & 800 ms).

This can be explained due to the fact that, at lowest input level used in the study, 30 dB HL, the portions of the signal would have fallen below the compression threshold. If the signal falls below the compression threshold, that particular signal would be amplified linearly rather than non-linearly. Hence, the effect of release times at this low input level is not significant and also shows larger variability.

2) At 45 dB HL

The results of the present study are in agreement with the study carried out by Jenstad and Souza (2005). Longer release time had significantly reduced temporal envelope distortions compared to short release time. This can be due to the fact as explained by Kuk (1996) at short release time (40 ms used in the current study), the gain will vary during each voice pitch period, and hence the compressor will distort the waveform. If the release time is made longer, rapid gain fluctuations will be reduced and thus the distortions would be minimal.

The results also revealed variable effects of release time on different consonants. Few consonants had significantly reduced EDI and other consonants did not vary significantly. Although not significant for few of the speech sounds, the mean EDI values decreased as the release time increased from 40 ms to 640 ms and to 1280 ms.

On a concluding note, for all the speech sounds the temporal envelope distortion was more at 40 ms release time condition. The amount of temporal envelope distortion reduces as the release time increases. This is again in consonance with the findings of Jenstad and Souza (2005). They found similar pattern of reduced temporal envelope distortion at longer release time of 800 ms.

On general observation of mean EDI values in many of the conditions, the temporal envelope distortion of consonants are higher in the order of glides, stops, affricates, nasals, liquids and fricatives. This is because, the greatest effect is on sounds where critical information is carried by variations in sound amplitude over time, such as stops which can be modeled as a series of temporal cues with falling or raising burst spectrum, onset of voicing, and air release. Speech sounds like stops, affricates and glides are known to have faster temporal variations, such as sharp rise time in burst of stops and affricates, faster transition in glides in contrast to semivowels, as compared to other classes of speech sounds such as fricatives which are high frequency hiss and are longer in duration and they vary slowly in terms of temporal parameters (Savithri, 1989).

3) At 65 dB HL

As depicted in Table 4.1a and 4.1b, the effects of release time on temporal envelopes are significant for most of the speech sounds. This implies that, at higher input levels, the compression is more effective (Henning & Bentler, 2008), i.e., the more the input intensity, more the reduction in the gain provided. The effect of release time reveals that, shorter release time (40 ms) induces more distortion in the temporal envelope than longer release times as explained by Kuk (1996). An illustration of the effect of release time on a VCV (ichi) is depicted in the figure 5.1. The consonantal portion /ch/ in the VCV /ichi/ is more distorted temporally at 40 ms release time followed by 640 ms and 1280 ms release time. The temporal envelope is less distorted at long release time of 1280 ms.

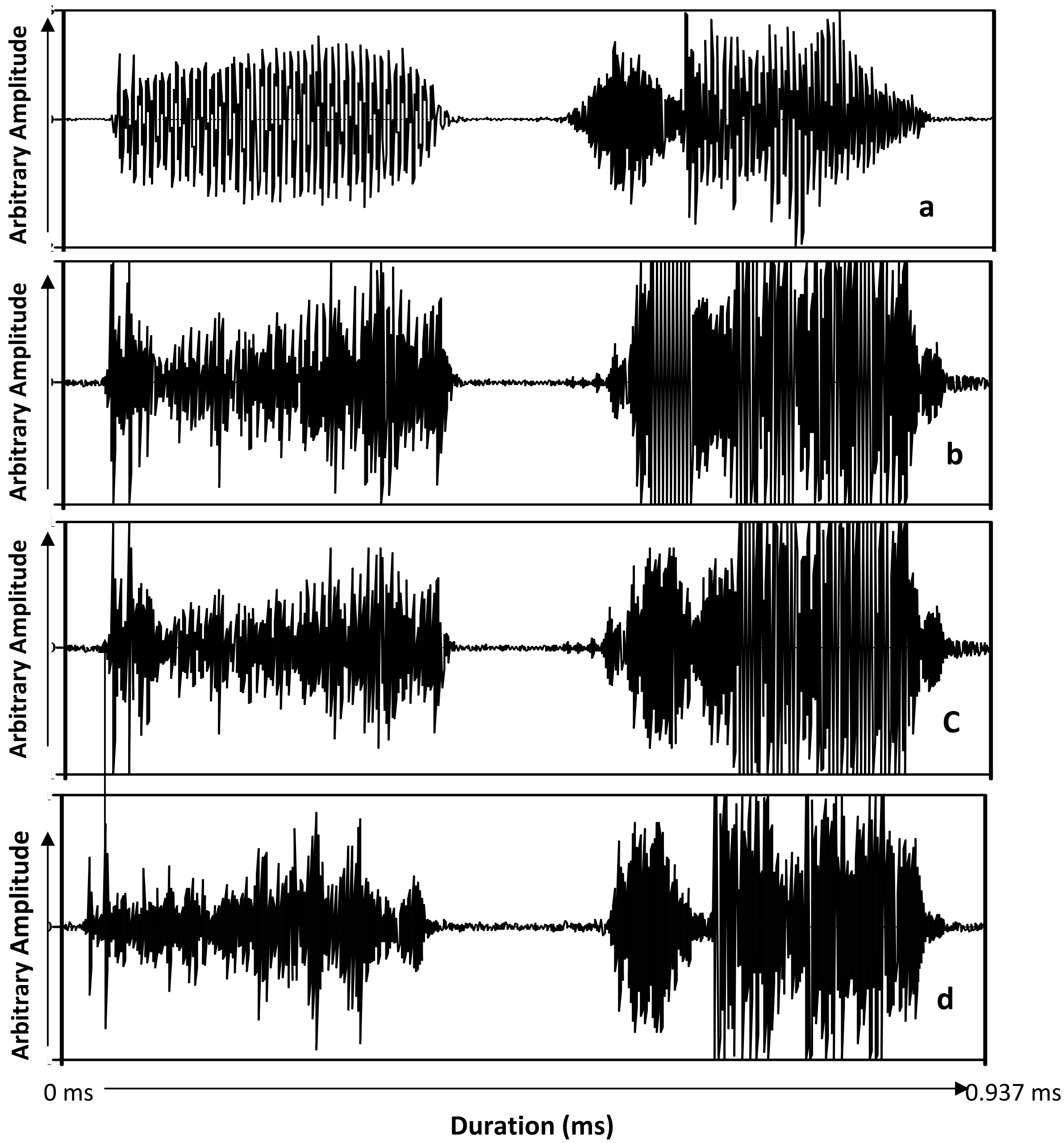


Figure 5.1: Hearing aid output waveforms of /ichi/ VCV token at 65 dB HL input level.

(a) Unaided, (b) Aided with 40 ms release time, (c) Aided with 640 ms release time, and (d) Aided with 1280 ms release time.

At 65 dB HL input level, the significant effect of short and long release times on temporal envelope distortion is more evident as compared to 45 dB HL input level. As described by Neuman et al (1996), as the compression increases, the longer release time play a major role to offset the effects of increased compression. Neuman et al (1996) reported that at higher compression ratio (3:1) longer release time of 1000 ms was found to be more useful on the categorical rating of sound clarity.

As discussed earlier, on general observation of mean EDI values as given in Table 4.1, the temporal envelope distortion of consonants are higher in the order of glides, stops, nasals, affricates, liquids and fricatives. This is due to the same fact as explained earlier for effect of release time at 45 dB HL.

Hence, it can be inferred from the findings that low level speech sounds seldom undergo compression, as they are below the compression threshold. As the input intensity increases the compression becomes more effective. The significant effect of short and long release times on temporal envelope distortion is thus more evident at higher input levels.

5.1.1c Between Group comparison

On observation of the mean EDI values, the difference between the two groups across the conditions may be attributed to the fact that, the audiometric data of Group B was moderately severe to severe hearing loss. Hence, the dynamic range with in which the signals have to compressed are less than when compared to Group A. Since, the dynamic range is lesser; the compression ratio to fit the entire intensity range within the dynamic range was set to be higher. Hence at higher compression ratio almost all sounds will undergo effective compression and the role of shorter and longer release times are more evident under this condition (Henning & Bentler, 2008; Neuman et al 1996). The fact that the two groups are not statistically significant is because the mean audiometric thresholds of Group A and Group B differs only by 10 to 15 dB HL.

5.1.2 Log-likelihood Ratio

Table 4.2 depicts the mean and SD for LLR of six classes of speech sounds across three release times and across three input levels for data of Group A and B.

Table 5.3: *Summary of results of LLR for Group A mean audiometric data*

Group A	***30 dB HL***	***45dB HL***	***65 dB HL***
Stops	640	1280	1280 *
Nasals	1280 *	1280 *	1280 *
Affricates	1280	1280	1280 *
Fricatives	1280	1280 *	1280 *
Liquids	1280	1280	1280 *
Glides	40	1280	1280 *

*Note:- *: Significantly reduced spectral distortion at particular release time compared to other release times (p<0.05).*

Table 5.3: *Summary of results of LLR for Group B mean audiometric data*

Group B	***30 dB HL***	***45dB HL***	***65 dB HL***
Stops	1280 *	1280	1280
Nasals	1280 *	1280	1280*
Affricates	1280	1280 *	1280
Fricatives	1280 *	640 * and 1280	1280
Liquids	1280	1280 *	640 &1280*
Glides	1280	1280	1280*

*Note:- *: Significantly reduced temporal envelope distortion at particular release time compared to other release times (p<0.05).*

The Table 5.3 and 5.4 summarizes the effect of three release time on LLR for six classes of speech sounds. As described in the above Table 5.3 and 5.4, at each input level, and for each class of speech sound, the release time which provides better temporal envelope or the one which induces significantly lesser spectral distortions compared to other two release times are mentioned. Asterisks mark indicates that the particular release time is significantly better than the rest of the release times.

5.1.2 Effect of release time on Log-likelihood Ratio

The results of comparison across three release time conditions are discussed below at each input level.

The results follow the same trend as seen in the effects of release times on EDI. The spectral distortion is significantly higher at 40 ms release time condition and as the release time increases from 640 to 1280, the spectral distortions tend to decrease. This is true for both 45 and 65 dB HL input levels. But, as depicted in the table, the effect of release time on LLR is significantly greater at 65 dB HL than at 45 dB HL condition. Also, the majority of speech sounds shows significant effect of release times at 65 dB HL, compared to 45 dB HL input level. As was seen for EDI, irrespective of the input level and RT, not all speech sounds are affected to the same extent. There exists a lot of variability.

In general, the overall results of LLR are similar to that of EDI, in the manner that, there are higher spectral distortions seen at short (40 ms) release time. In addition, spectral distortions significantly reduces as the release time increases from 640 ms to 1280 ms.

This can be explained based on the fact that, alterations in one of the domains (temporal/spectral) nearly always have corresponding effects in the other one. That is, changes to the waveform envelope produce corresponding spectral changes, and vice versa (Van Tassel, 1993).

The result of shorter release time having greater spectral distortions can be attributed to the reason described by Wang (2001). When the release times are made shorter, they are associated with the broadest and strongest distortion across the frequency spectrum due to the spread of energy. On the other hand, as the release times are made longer, they are associated with the narrowest and weakest distortion on the signal's frequency spectrum.

On close observation of mean LLR values, it was noted that the spectral distortion of consonants can be placed in order of higher to lower amount of distortions. They are as follows, liquids and fricatives, affricates, glides, stops and nasals. The similar trend is followed for both the mean audiometric group data. Unlike EDI which is time and intensity based measure, the LLR compares formants of speech tokens at every fixed time frame and displays one single value. Hence, both the measures tap different aspects in a particular signal. The speech sounds which are confounded majorly by temporal variations tend to be vulnerable to changes detected in the EDI than in LLR and vice versa.

The order in which the consonants have more spectral distortion can be due to the fact that, the formant frequencies of consonants are in range from low to high frequencies. For example, in Kannada language, formant frequencies of velar stops range from 400 Hz to 1.9 kHz, liquids have their formants from 450 Hz to 2.9 kHz, nasals have their formants from 300Hz to 1.5 kHz, formants of fricatives ranges from 500 Hz to 4.9 kHz etc. The compression amplification used in the study is a two channel device and each of the channel vary in terms of compression ratios. Hence, each portion of the speech signal might have undergone through varied degrees of compression and hence the effects of spectral distortion remains diffused among the six classes of speech sounds (Savithri, 1989)

Hence, the magnitude of the temporal/spectral artifacts during compression activation/deactivation is not constant, but constantly changing. Specifically, the broader and stronger distortion across frequency spectrum occurs within the shorter attack/release times whenever the input signal level rises to or falls from slightly above the compression threshold (Wang, 2001). In simpler words, at higher levels of input, maximum compression takes place and hence more spectral distortions is noted at high levels in contrast to low levels.

5.1.2c Between Group comparison

On observation of the mean LLR values, the difference between the two groups across the conditions may be attributed to the fact that, group B had lesser dynamic range and hence more compression ratio caused more spectral distortions. Hence at higher compression ratio almost all sounds will undergo effective compression and the role of shorter and longer release times are more evident under this condition (Henning & Bentler, 2008; Neuman et al 1996). The fact that the two groups are not statistically significant is because the mean audiometric thresholds of Group A and Group B differs only by 10 to 15 dB HL.

Few of the varied results noted with respect to speech sounds of a particular category across release times may be because of the reason that, within each class, the speech sounds constituted of both voiced and voiceless consonants. These effects of release time for voiced sounds might be different from that of a voiceless speech sound. Hence, it would be important to study the effect of release time on speech sounds categorized based on place of articulation, manner of articulation and based on voicing characteristics. However, due to time constraints, the study was carried out only based on categorization of manner of articulation.

5.2 Behavioural measures:

The behavioural measures, SIS and quality ratings were obtained across three release times (40 ms, 640 ms & 1280 ms) and across three input levels (30 dB HL, 45 dB HL & 65 dB HL) for Group A and Group B.

4.2.1 Speech identification scores:

The speech identification scores (SIS) were obtained across three release times (40 ms, 640 ms & 1280 ms) and across three input levels (30 dB HL, 45 dB HL & 65 dB HL) for Group A (n=10) and Group B(n=10) participants.

Table 5.5: *Summary of results of SIS for Group A and Group B*

	30 dB HL	***45dB HL***	***65 dB HL***
SIS for Group A	640 *	1280 *	40/640/1280
SIS for Group B	1280	1280	1280

*Note:- * significantly better SIS at that particular release time compared to others at $p<0.05$*

Table 5.5 summarizes the effect of three release time on SIS for Group A and Group B. As described in the above Table 5.5, at each input level, the release time which provides better SIS compared to other two release times are mentioned. Asterisks mark indicates that the particular release time is significantly better than the rest of the release times.

5.2.1a Effects of Release Time on SIS.

Table 4.3 depicts the significant effects of release times across three release times and across three input levels for both Group A and Group B.

Group A:

The mean scores are higher for longer release times, at both 30 dB HL and 45 dB HL. The high speech intelligibility scores obtained at longer release times and significantly low SIS at short (40 ms) release time is because, short release time causes un-natural alteration of vowel and consonant ratio (Freyman, Nerbonne, & Cote, 1991).

At higher levels, there was no improvement in SIS as the release time increased. A study by Vanaja and Jayaram (2006) reported the mean SIS for different degrees of hearing. According to the study, listeners with moderate hearing loss would have mean SIS of around 85 % (12.47 SD) and listeners with moderately severe hearing loss would have mean SIS of around 77.5 % (13.89 SD). Hence for listeners in the current study, the audibility was compensated through hearing aid, and more audibility was provided at the higher input level of 65 dB HL. The SIS at this level in the current study reached its maximum performance, and the effect reached plateau across release time. This could be attributed to the reason of ceiling effect of SIS at 65 dB HL.

Group B:

Although not significant, the mean SIS increased for longer release times (1280 ms) at 45 and 65 dB HL. This could be attributed to the fact that Group B listeners had very narrow dynamic range and hence the signal had to be compressed to a larger extent. When there is more compression due to high gain, high compression ratio or higher input level, longer release time can offset the possible temporal distortions to some extent compared to shorter release times (Henning & Bentler, 2008). The finding of insignificant difference can be due to the reason that, listeners used other cues like contextual cues, rather than depending upon only temporal (Jenstad & Souza, 2005)

5.2.2 Perceptual quality ratings (loudness, clarity, naturalness & overall impression)

Descriptive statistics was done to compute the mean and standard deviation. Repeated Measure ANOVA was administered to find out the overall interaction of the conditions (within subject factors) and groups (between subject factors). Bonferroni's multiple pair wise comparison was done to determine the pairs that are significantly different from each other. Multivariate Analysis Of Variance (MANOVA) was carried out to compare between the two groups.

5.2.2a Effect of release time on perceptual quality ratings (loudness, clarity, naturalness & overall impression)

The above Table 4.4, depicts the mean and SD for loudness, clarity, naturalness & overall impression parameter across three release times and across three input levels for Group A and Group B. Two way repeated measure ANOVA was performed to reveal significant difference across nine conditions. If the test revealed significant difference across nine conditions, Bonferroni's multiple pair wise comparison was done to reveal the pairs which are significantly different. The results are described below for each of the parameter. The effects of release times for each of the parameter are discussed at each input level for two groups individually.

Table 5.6: *Summary of results of perceptual quality ratings on four parameters for Group A and Group B*

	Group A			**Group B**		
Parameters	***30 dB HL***	***45 dB HL***	***65 dB HL***	***30 dB HL***	***45 dB HL***	***65 dB HL***
Loudness	40	40	1280	40	40	640
Clarity	1280	1280 *	1280	1280 *	1280 *	1280 *
Naturalness	1280 *	1280 *	1280 *	1280 *	1280 *	1280 *
Overall Impression	1280 *	1280	1280	1280 *	1280*	1280

*Note:- *: significantly better ratings at that particular release time at p<0.05*

Table 5.6 summarizes the effect of three release time on SIS for Group A and Group B. As described in the above Table 5.6, at each input level, the release time which provides better ratings compared to other two release times are mentioned. Asterisks mark indicates that the particular release time is significantly better than the rest of the release times.

1) Loudness:

The Table 4.3 depicts the mean ratings to be slightly higher at short (40 ms) release time. Although the effect is not seen at all the conditions, this is in consonance with study by Neuman, Bakke, Hellman, and Levitt (1998).

At shorter release times, the audibility will be maintained as the speech signal will be released from compression very quickly and the gain is applied for weaker part of the signal. The longer release times reduces the gain which needs to be provided for weaker sounds such as consonants. Hence, the perceived loudness at short release times is more due to the factor of audibility. This explanation is in agreement with that given by Moore (1996).

2) Clarity:

Although the effect is not significant at many conditions, the mean ratings of clarity increased as the release time increased. This is in agreement with the study by Neuman et al. (1998). At shorter release times, the gain fluctuates very rapidly giving the sensation of pumping sounds (Wang, 2001; Moore, 1996). Hence, at longer release times the clarity of the speech sounds will be preserved to some extent in comparison with short release times.

3) Naturalness:

The results are in agreement with the study by Neuman et al. (1998). At shorter release times, the gain fluctuates very rapidly giving the sensation of pumping sounds (Wang, 2002; Moore, 1996). The longer release times prevents some amount of temporal and spectral distortions as revealed by the current study and also by Jenstad and Souza (2005). Since the distortions are controlled to some extent by using longer release times, the speech might sound pleasant and more natural.

4) Overall impression:

The results are in agreement with the study by, Neuman et al. (1998). Due to the positive effects revealed by the longer release times on clarity, naturalness and other parameters, the overall impression will be maintained to be higher for longer release times.

5.2.1c.Between group comparison

The Table 4.4 depicts the mean and SD of perceptual quality ratings of four parameters across three release times and across three input levels for Group A and B.

In general, the perceptual ratings of quality are lower for Group B, though not significant. This is because, the Group B participants had reduced dynamic range and the amount of compression taking place will be high and the amplified signal of Group B would contain more temporal and spectral distortion. Hence, the quality ratings of the parameters are lesser in Group B.

CHAPTER IV

SUMMARY AND CONCLUSIONS

The purpose of the present study was to measure the effects compression release times (40 ms, 640 ms and 1280 ms) of the hearing aid on acoustic and behavioural measures of speech. The effects of each of the release time were investigated at three input level (30, 45 & 65 dB HL).

The acoustic measures considered in the study were

1. Envelope Difference Index EDI), a temporal envelope distortion
2. Log-likelihood Ratio (LLR), a Spectral Distortion Measure (SDM).

The behavioural measures considered in the study were

1. Speech Identification Scores (SIS)
2. Perceptual rating of quality on four parameters (loudness, clarity, naturalness and overall impression).

Behavioural measurements were performed on two groups of participants. Group A consisted of 10 adult participants who had flat or gradually sloping sensorineural hearing loss (SNHL) ranging from moderate to moderately severe degree. Group B consisted of 10 adult participants having flat or gradually sloping SNHL ranging from moderately severe to severe degree. The acoustic measures were performed on five participants using the hearing using programmed with the mean audiometric thresholds of the participants of Group A (mean PTA, 60.9) and Group B (mean PTA of 75.3 dB HL).

The study was carried out in two phases. Phase I involved measurement of acoustic parameters namely, temporal and spectral measures using Envelope Difference Index (EDI) and Log-likelihood Ratio (LLR) respectively, on six classes of speech sounds. Phase II involved measurement of behavioural parameters namely, SIS and perceptual quality ratings. Under both the phases the measurements were done at three release time and across three input levels.

The statistical analysis was carried out on the data to evaluate the objectives of the study. Appropriate statistical analysis was performed. The results of the analysis are summarized.

The acoustic measure EDI was done to quantify the temporal envelope distortions. The effect of release times on six classes of speech sounds at 30 dB HL was not significant. At higher input level i.e., at 45 and 65 dB HL, the shorter release time (40 ms) revealed significantly higher temporal envelope distortions. The longer release time 640 ms and 1280 ms release time revealed significantly lower temporal envelope distortion in comparison to 40 ms short release time. At shorter release time, the gain of the amplifier varies rapidly and hence distorts the waveform envelope. The amount of temporal envelope distortions was found to be increasing with increase in the input intensity.

The effects of release times and the amount of temporal envelope distortion varied across speech sounds. Not all the speech sounds were equally affected. The speech sounds whose major acoustic characteristics are temporal based such as sharp rise time release burst, rapid transitions etc. were found to have higher mean EDI, i.e., have more temporal envelope distortions.

The effect of release time on EDI was not statistically significant between groups across majority of the conditions. This can be attributed to the programming constraints of the hearing aid. Although not significant, the mean EDI was found to be higher for the data of Group B, due to reduced dynamic range and more effective compression.

The acoustic measure LLR was used to quantify the spectral distortions. The effect of release times for six classes of speech sounds at 30 dB HL was not significant. At higher input level i.e., 45 and 65 dB HL, the shorter release time (40 ms) revealed significantly higher spectral distortions due to the fact that shorter release time causes the gain of the amplifier to fluctuate rapidly and hence the energy within and across frequency spreads over the other formant regions.

The amount of spectral distortions (LLR) was found to be significantly higher as the input level increased. This LLR is significantly lesser at lower input levels. The effects were found to be significant at all three input levels for the speech sounds across most of the release time conditions.

The effects of release times and the amount of spectral distortion varied across speech sounds. The amount of spectral distortions of consonants can be placed in the order of higher to lower i.e., liquids and fricatives, affricates, glides, stops and nasals. In contrast to EDI, the LLR measures distance between formants of two signals and hence the amount of distortion on speech sounds varies for EDI and LLR measures.

The effect of release time on LLR was not statistically significant between groups across majority of the conditions. This can be attributed to the programming constraints of the hearing aid. Although not significant, the mean LLR was found to be higher for the data of Group B, due to reduced dynamic range, increased compression ratio and more effective compression.

The effects of release times on the behavioural measures of speech i.e., SIS and perceptual quality ratings were performed on two groups of participants. In Group A the effect of release time on SIS was statistically significant only at longer release time (640 and 1280 ms) in comparison to the short release time (40 ms). These effects were seen at low (30 dB HL) and at conversational levels (45 dB HL) of speech. This lower performance at short release time is due to increased temporal and spectral distortions. At 65 dB HL there was no effect of release time due to the fact that, at higher presentation levels for word recognition the ceiling effect had occurred. In Group B participants, the mean scores (although not significant) increased as the release time was made longer. The Group B participants had very narrow dynamic range and broader auditory filters which causes reduced spectral resolution and they rely more on temporal resolution, hence the high scores were obtained at longer release time which could be able to lessen the amount of envelope distortion.

The Group A performed significantly better on SIS in comparison to Group B. This could be due to reduced dynamic range, broadened auditory filters; and more spectral and temporal distortions through the hearing aid.

The effects of release time on perceived rating of quality were measured on four parameters, viz., loudness, clarity, naturalness and overall impression. The effect of release time of loudness was not significant and the mean rating of loudness showed a weaker trend of increased ratings at short (40 ms) release time condition. At short time constants, the hearing aid is able to provide gain to softer or low level consonants, thereby increasing consonant to vowel ratio, which in turn leads to perception of speech a little louder. The rating of clarity, naturalness and overall impression increased as the release time was made longer. The effect of release time was significant for clarity and naturalness parameters under most of the conditions. The 40 ms short release time had significantly lesser rating in comparison with 640 ms and 1280 ms release time condition. The effects of release time on perceptual quality rating are similar for both the groups.

From the present study it can be inferred that, 40 ms short release time induces significantly more temporal and spectral distortions when compared to 640 ms and 1280 ms release time. With respect to SIS and perceptual quality ratings, the performance was poorer with 40 ms short release time than when compared with 640 ms and 1280 ms release time. For both the acoustic and behavioural measures, as the input level increases or as the dynamic range reduces, the temporal envelope and spectral distortions also increases and longer release times are required to offset the effects of reduced temporal envelope and spectral distortions.

5.1 Clinical Implications

The information from the present study helps the audiologists to gain knowledge so as to how the compression release time can have various effects on temporal and spectral distortions. The current study helps to choose the appropriate release time depending on the knowledge of dynamic range of the individual with hearing impairment. The present study also helps to set the appropriate release time when there is interaction between compression ratio and input level. Since the current study was carried out using hearing aid on real ear, the realistic amount of distortions was computed. This throws light on the selection of the components or programmes or signal processing techniques in a hearing aid which would minimize the distortions.

5.2 Future research direction

The study can be carried in more number of participants. The study can be carried by including other variables such as compression ratio, compression threshold and study their interactions with the release time on both acoustic and behavioural measures. The study can be done under the difficult-to-hear listening situations. The effects of release time on acoustics and behavioural measures can be studied for running speech, sentences and words of varying length. The study can be done on speech sounds categorized on place of articulation and voicing characteristics.

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