COMPARISION OF SYLLABIC AND DUAL COMPRESSION ON NON LINEAR HEARING AID PROCESSED KANNADA CHIMERIC SENTENCES

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CERTIFICATE

This is to certify that the dissertation entitled "**COMPARISION OF SYLLABIC AND DUAL COMPRESSION ON NON LINEAR HEARING AID PROCESSED KANNADA CHIMERIC SENTENCES**" is the bonafide work submitted in part fulfillment for the degree of Master of Science (Audiology) of the student (Registration No. 16AUD035). 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.

Mysuru,
April 2018.

Dr. S. R. Savithri
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CERTIFICATE

This is to certify that the dissertation entitled "**COMPARISION OF SYLLABIC AND DUAL COMPRESSION ON NON LINEAR HEARING AID PROCESSED KANNADA CHIMERIC SENTENCES**" has been prepared under my supervision and guidance. It is also certified that this has not been submitted earlier in any other University for the award of any Diploma or Degree

Mysuru
May, 2018.

Dr. Devi N
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DECLARATION

This is to certify that this dissertation entitled "**COMPARISION OF SYLLABIC AND DUAL COMPRESSION ON NON LINEAR HEARING AID PROCESSED KANNADA CHIMERIC SENTENCES**" is the result of my own study under the guidance of Dr. Devi N, Reader in Audiology Department of Audiology, All India Institute of Speech and Hearing, Mysuru, and has not submitted earlier in any other University for the award of any Diploma or Degree.

Mysuru
May 2018

Register No: 16AUD035

குறள் 411:

**செல்வத்துட் செல்வஞ் செவிச்செல்வம் அச்செல்வம்
செல்வத்து ளெல்லாந் தலை.**

திருவள்ளுவர்

விளக்கம் 1:

செவியால் கேட்டறியும் செல்வம், செல்வங்களுள் ஒன்றாகப் போற்றப்படும் செல்வமாகும், அச் செல்வம் செல்வங்கள் எல்லாவற்றிலும் தலையானதாகும்.

Meaning:

Wealth (gained) by the ear is wealth of wealth; that wealth is the chief of all wealth.

“கற்றது கைமண் அளவு, கல்லாதது உலகளவு”

ஔவையார்

Meaning:

"What you have learned is a mere handful; what you haven't learned is the size of the world"

Avvaiyaar

Content

List of Tables

List of Figures

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Chapter 1

Introduction

Hearing is considered to be one of the foremost senses that are important for distant warning as well as communication. The human ear as a whole is a delicate but a complex structure. The human ear is an excellent transducer, changing sound pressure in the air into a neural-electrical signal that is translated by the brain as speech, music, noise, etc (Dobie & Van, 2004). The external ear, middle ear, inner ear, brainstem, and brain each have a specific role in this transformation process. The external ear includes the pinna, which helps collect sound in the environment whereas the ear canal channels sound to the tympanic membrane (eardrum), which separates the external and middle ear. The tympanic membrane and the three middle ear bones, or ossicles (malleus, incus, and stapes), assist in the transfer of sound pressure in air into the fluid- and tissue-filled inner ear. The inner ear is responsible for performing spectral and temporal acoustic analyses of incoming sound. The graded stiffness, graded mass and graded width of the basilar membrane altogether makes it an excellent frequency analyser. For speech recognition, spectral information is assumed to play an important role as many phonetic features are characterized by their frequency spectrum particular frequency known as the characteristic frequency (CF) (Bekesy, 1960). Sensory receptors, known as hair cells, transduce the mechanical energy into neural activity (spikes), which are elicited on Auditory Nerve (AN) fibers innervating those hair cells. The neural code in the Auditory Nerve fibers is then conveyed to higher nuclei in central auditory system for further analysis and processing.

It is difficult to understand speech for people with sensorineural hearing loss especially in background speech and also difficult for them to resolve the individual frequency components of complex sounds. Complex sound like speech can be characterized as the sum of number of amplitude-modulated signals representing the outputs of an array of narrow frequency bands. Temporal information at the output of each band can be separated into temporal fine structure (TFS), the rapid oscillations close to the center frequency and temporal envelope (ENV), slower amplitude modulations superimposed on the TFS. ENV cues are represented in the auditory system as fluctuations in the short-term rate of firing in auditory neurons, while TFS is represented by the synchronization of nerve spikes to a specific phase of the carrier (phase locking). Fine structure is important for lexical tone perception for normal-hearing (NH) listeners but the listeners with sensorineural hearing loss (SNHL) have an impaired ability to use Temporal Fine Structure. However, there are reported which has evidenced TFS for lexical tone perception. This could be due to the reduced frequency resolution. (Joris & Yin, 1992; Rose, Brugge, Anderson, & Hind, 1967)

According to Smith, Delgutte, and Oxenham, (2002) the speech with 4 – 16 frequency bands are essential to identify words represented by the envelope cues than the fine structure. But in the case of hearing impaired individuals studies have revealed that cochlear hearing loss in adults affects the ability to encode and/or use TFS cues rather preserving the ability to encode the E cues (Lorenzi, Gilbert, Carn, Garnier, & Moore, 2006). On the other hand in individuals with central damage and specific language acquisition disorder studies report that there is impairment in the perception of envelope cues (Giraud et al., 2000; Lorenzi, Dumont, & Fu¨llgrabe, 2000). The effect of hearing aid compression parameters, such as compression time constants and compression ratio, on speech perception and the subjectively perceived sound quality of hearing aids has been investigated in several studies (Neuman, Bakke, Mackersie, Hellman, & Levitt, 1995)

Dreschler (1989) evaluated twelve individuals with hearing loss using linear and compression hearing aids. Compression aid was superior to a linear aid with peak clipping for phoneme perception in quiet. Presumably, different compression systems may alter the acoustic characteristics of the speech signal, which in turn may result in changes in phoneme perception. There is only tentative support for the use of syllabic compression which is fast acting compression system for those with hearing loss. Commercial single-channel aids were no better for speech intelligibility than linear aids (Florentine, Buus, Scharf, & Zwicker, 1980). Hearing impaired individuals who do not have severely reduced dynamic range but have an optimum level for speech discrimination can benefit from Automatic volume control (AVC).Above the optimum level for speech discrimination the performance of the individuals starts to decline (King & Martin, 1984). Wide range of input levels are accepted by Automatic volume control (AVC) and are not continuously adjusting the gain as in linear hearing aid. The combination of both the syllabic compression and automatic volume control is dual compression. It holds the advantage of both syllabic and Automatic volume control (AVC) (B. C. J. Moore, Glasberg, & Stone, 1999).

Need for the study

The effect of the envelope and fine structure cues in normal hearing individuals especially in foreign tonal and non-tonal languages (Mandarin Chinese & English) was studied by Heinz and Swaminathan, (2009). The results reveal that in tonal languages TFS plays important role in the perception. The non-tonal language was governed by the envelope (E) cues however tonal language was governed by the temporal fine structure (TFS) cues in the normal hearing individual (Smith et al., 2002). In Mandarian tone recognition young normal hearing listeners performed good speech recognition in quiet using only envelope cues whereas good performance were achieved for voice pitch perception in noise. The ability to process temporal fine structure information may be adversely affected by cochlear hearing loss. This has been demonstrated using tasks involving frequency discrimination (Buss, Iii, & Grose, 2004; Hopkins, Moore, Hopkins, & Moore, 2013; Moore, Glasberg, & Hopkins, 2006), lateralization, and speech perception (Gnansia, Péan, Meyer, & Lorenzi, 2009; Hopkins, Moore, & Stone, 2008; Lorenzi et al., 2006; Mar et al., 2010)

Indu and Devi (2015) have reported south Indian languages such as Malayalam using chimeric sentences. In Malayalam, at lower frequency bands speech stimulus were better identified with Temporal Fine Structure (TFS) cues whereas at higher bands speech stimulus were better identified by Envelope (E) cues. However, Envelope (E) cues were majorly used for speech identification when Kannada chimeric words and sentences were used (Naveen & Devi, 2017). The elevation of hearing threshold in hearing impaired individuals can be compensated by means of amplification and the reduction of uncomfortable loudness level can be combined with appropriate limiting techniques (peak clipping & compression). Automatic gain control system in hearing aid is intended to adjust the gain automatically for different listening situations. The reduced frequency and temporal resolution in hearing impaired individuals cannot be compensated so it is important that hearing aid should not affect the spectro-temporal information of the stimulus. It is possible that fast acting compression system disturb the temporal structure. Hence, there is a need which focuses on better understanding of the perception of hearing aid processed speech with syllabic and dual compression in normal as a representative of hearing impaired individuals.

Aim

The aim of the study was to compare the effect of syllabic compression and dual compression on non linear hearing aid processed Kannada chimeric sentences.

Objectives:

- To study the influence of envelope cues on hearing aid processed Kannada chimeric sentences identification using syllabic compression and dual compression.
- To study the influence of fine structure cues on hearing aid processed Kannada chimeric sentences identification using syllabic compression and dual compression.
- To compare between the different compressions.
- To find out the number of frequency bands required to clearly differentiate between the envelope and fine structure cues.

Hypothesis

The following null hypotheses were framed for each main objective of the study.

- There is no significant effect of envelope on hearing aid processed Kannada chimeric sentences identification using syllabic compression and dual compression.
- There is no significant effect of fine structure on hearing aid processed Kannada chimeric sentence using syllabic and dual compression.
- There is no significant effect between the two compressions in the hearing aid
- There is no significant effect of identification of hearing aid processed Kannada chimeric sentences across the frequency bands.

Chapter 2

Review of literature

When a sound is received by the cochlea, the frequency content of the signal is mapped into a pattern of excitation along the basilar membrane. The excitation pattern code the spectrum information of the acoustic stimulus, which is concerned to as ‘Spectral’ or ‘Place’ information (Dallos, Popper, Fay, & Dallos, 1996). It is assumed that spectral information plays an important role in language recognition as many phonetic features are characterized by their frequency spectrum. Basilar membrane is a bank of band pass filter which has the property of frequency selectivity, with every filter corresponding to an exceptional place on the basilar membrane (Palmer & Russell, 1986). The output signal from these band pass filters carries important temporal information as easily. It is noticed as a gradually varying envelope modulation superimposed on fast oscillations or fine structure (TFS) in the wave form. This temporal information is relayed to the sensory nerve (Pfingst, Arbor, & Pfingst, 2003; Smith et al., 2002; Zeng et al., 2004)

Auditory nerve (AN) fibers firing rate changes are associated to the signal envelope and the times between spikes, which reflect the TFS or temporal fine structure information (Young & Sachs, 1979). The relative envelope magnitude across channels carries information that can be used in the auditory system to identify the signal spectral shape and its slow short-term spectral changes. The TFS conveys cues regarding the central frequency of the sound and about its short-term spectrum.

The auditory nerve (AN) fibers phase locking property helps to encode the TFS information. The phase locking property of the AN fibers gets weak at high frequencies and its absent for frequencies above 4-5 kHz for mammalians (Palmer &

Russell, 1986). Hence, it is usually assumed that 'TFS information is not used for frequencies above that boundary'. For robust speech identification envelope or ENV information is significant in silent situation even when provided in as few as four frequency bands (Flanagan, 1980; Shannon, Zeng, Kamath, Wygonski, & Ekelid, 1995; Smith et al., 2002). But in the presence of background noise more frequency bands are required for the ENV speech generation process. As the results reveal that envelope ENV cues are enough to provide intelligibility in silent, but the identification performance is slightly degraded in the presence of fluctuating noise. The robust speech identification in quiet from ENV cues from relatively small frequency bands is the reason that current cochlear implants provide ENV information over a small number (8 to 16) of electrodes (Bartels et al., 2008). On the other hand, TFS or temporal fine structure is associated with perception of pitch for both gross and complex feelings as well as sound localization (Jinarlene, Carneydavid, Carney, & Nelson, 2004; Oxenham & Bacon, 2003; Qin & Oxenham, 2005; Smith et al., 2002; Stone, Füllgrabe, & Moore, 2008)

Liu and Zeng, (2006); Smith et al., (2002); Xu and Pfingst, (2003)had studied the comparative roles of Speech envelope (E) and temporal fine structure (TFS) cues in speech identification.. One means to accomplish this is through the role of noise or tone vocoders. Vocoded speech is generated by separating out a broadband signal into a number of frequency bands, pulling up the E from every set to modulate a noise or tone carrier and combining the resulting signals from all frequency bands. Of late, various works have aimed out a potential contribution of TFS cues in language perception.

Xu and Zheng, 2007examined the comparative assistance of spectral and temporal cues to phoneme recognition. The spectral cues were altered by altering the number of channels in the vocoder processing, and the temporal cues were altered by changing the cutoff frequency of the envelope (E) extractor low-pass filter. Study included tests for both vowel and consonant recognition. The result revealed that there was a tradeoff between the spectral and temporal cues in phoneme identification, where enhanced spectral cues can make up for reduced temporal cues and vice versa.

Nie, Barco, and Zeng, (2006)studied on individuals with normal hearing and hearing impairment for identification of spectral and temporal cues. Number of channels and pulse rate were changed in order to alter the amount of spectral and temporal cues. The results revealed that the TFS cues are more efficiently used by the normal hearing individuals than the hearing-impaired individuals. This is because of the reduced capability of hearing impaired persons to understand speech in fluctuating background sound (Hopkins & Moore, 2007; Lorenzi et al., 2006; Mar et al., 2010; Moore & Sęk, 2009). This might be due to reduced phase locking ability in hearing impaired subjects. Alternatively, the reduced ability of hearing impaired subjects to profit from FS cues might be induced by reduced ability to decode the TFS case where it is indicated that this process involves cross-correlation of the yields of two spots on the basilar membrane (Shamma, 1985). Lastly, the broader tuning of the auditory filters in hearing impaired persons (Moore & Glasberg, 1987) may have a substantial purpose in their miserable performance in understanding TFS information. This is imputable to the limited frequency selectivity of the cochlear filters which has difficulty in decoding the complex and rapidly variable TFS information (Moore, 2008).

Lorenzi et al., 2006 evaluated the contribution of TFS in speech identification, consonant identification in nonsense processed vowel-consonant-vowel (VCV) stimuli was used. The individuals with hearing impairment and normal subjects were tested with TFS-only stimuli generated from nonsense VCV words by separating out the original signal into 16 contiguous frequency bands, computing the E and FS in every band using the Hilbert transform, and combining the TFS signals from the different frequency bands to construct the last stimulus. Results revealed that normal hearing subjects show significant intelligibility for TFS cues, where up to 90% recognition is reported after some training. Stone et al., 2008 explained the need for training to achieve high recognition scores by the possibility that the auditory system is not applied for processing TFS cues in isolation from ENV cues or that TFS cues in processed stimuli are distorted compared to intact speech and hence training is needed. In a similar experiment it has been shown that children with normal hearing aged 5 to 7 are able to make use of TFS cues. Results revealed that normal hearing children can use both ENV and TFS cues at the same level as adults, which means that tests for the sensitivity to TFS cues can be performed at this very youthful age for the early spotting of any potential problems in the TFS process.

A different approach to measure the ability of normal and hearing- impaired persons to benefit from FS has been embraced. Processing of TFS cues is assessed by measuring changes in the speech recognition threshold (SRT). SRT is the least hearing level for spoken communication at which an individual can recognize 50% of the speech material. Hopkins and Moore, 2007 quantified the importance of TFS cues by varying the number of frequency channels containing TFS information with the ease of the channels being noise or tone vocoded to suppress any TFS information. To examine the theory that hearing-impaired subjects can create usage of TFS cues only at low frequencies. Hence, removing TFS from low channels would affect the performance while removing TFS from high channels should not hold much meaning. Their answers demonstrate that hearing impaired subjects have less power to constitute use of TFS cues at medium and high frequency when listening in a competing talker background.

Hopkins and Moore, 2009 measured the SRTs in normal hearing subjects while varying the cutoff channel which is the frequency band below which the stimulus is left intact, while TFS information is murdered from all bands above it. It was established that the SRT declined significantly as the value of the cutoff channel increased, which suggests that TFS has an significant part in understanding speech in fluctuating background noise. Hopkins and Moore (2009) measured the SRTs for speech processed to contain varying amounts of FS cues. The speech signals were filtered using 30 1-ERBN filters and treated to keep ENV only information or left unprocessed to preserve both ENV and TFS cues. It was noticed that when there are more channels containing TFS cues, SRT were decreased, showing benefits from the introduced cues. Redundancy in TFS information as adding TFS in some channels does not always improve the threshold was also observed. Another experiment was carried out where filtering of the speech signal through 5 6-ERBN channels and brought forth a tone vocoded signal in four of the available five channels. The fifth channel was either absent or was unprocessed. Normal hearing subjects benefitted from the added TFS cues over a spacious range of frequency, while the benefit was less in hearing-impaired subjects.

Gnansia et al., 2009 studied the effects of spectral smearing and degradation of TFS cues on masking release, which is the capability to listen in the dips of the background noise. A stimulus was processed using a spectral smearing algorithm or a tone vocoder technique. The spectral smearing algorithm computes the short-term spectrum using fast Fourier transform, and then the spectrum is smeared by a divisor of two or four using a smearing matrix for 2-ERBN or 4-ERBN auditory filters. It was discovered that the fundamental frequency information was more degraded by the vocoder than the spectral smearing algorithm. Masking release was reduced more with the tone vocoder than spectral smearing. Conclusion of study showed that both frequency selectivity and TFS cues are significant for the ability to listen in the dips. Gilbert, Lorenzi, Gilbert, & Lorenzi, 2006 assessed the comparative use of ENV and TFS cues in reconstructing missing information in interrupted speech. In their subject field, four types of sentences processed into 32 frequency bands and information in 21 bands were removed or processed so that the final stimuli have different amounts of ENV and TFS cues. Four types of sentences were used; reference, partially empty, vocoded and partially vocoded. The resulting sentences were still understandable but the intelligibility significantly deteriorated after adding a silence gap. TFS cues have a significant part in reconstructing the broken sentences. The TFS is not sufficient alone, but is practiced along with ENV to understand interrupted speech.

A significant concern regarding the results for TFS contribution to speech understanding is that these effects may be influenced by possible ENV cues in signals. These ENV cues may be due to inefficient signal processing techniques applied to separate TFS from ENV, which is not an easy job given that the TFS and ENV are not totally independent (Ghitza, 2001). Some other significant constituent is the possible recovery of ENV cues by the human auditory filters from a correctly processed signal having only TFS cues. For example, narrow-band filtering can recover the signal ENV from the fine- structure information (Voelcker, 1966). This is especially important in humans because of the sharp cochlear tuning (narrow filters), which facilitate the retrieval of the slow amplitude variations (ENV) from the TFS signal (Ghitza, 2001; Heinz & Swaminathan, 2009; Zeng et al., 2004). Gilbert et al., (2006), it is argued that recovery of ENV cues from TFS-only signals has minimal contribution to speech recognition when the vocoder analysis filters, which are utilized to generate the TFS-only stimulus, have bandwidth less than 4 ERBN. According to them, using 16 frequency channels should be sufficient to prevent the use of recovered ENV cues. Heinz and Swaminathan, (2009), nevertheless, presented physiological evidence for the presence of recovered ENV in chinchilla AN response to chimeric speech. It was computed that 'Neural cross-correlation coefficients' to measure the similarity between ENV or TFS to quantify the similarity between ENV (or TFS) components in the spike train responses.

Sheft, Ardoint, Lorenzi, and Sheft, 2008 presented different ways to reduce the fidelity of ENV reconstruction from TFS signals. The TFS signal can be filtered by an all- pass filter with a random phase response. This is founded on the assumption that ENV and the instantaneous phase are connected, so that processing the TFS signal to produce a mismatch with the original ENV signal will reduce the fidelity of ENV recovery (Schimmel & Atlas, 2005). The other method to reduce the chances of meaningful ENV recovery from TFS cues is to increase the number of analysis filters. When the bandwidth of the analysis filter is narrower than 4 times the normal auditory filter, some studies argued that the role of recovered ENV cues in speech perception is negligible (Lorenzi et al., 2006).

The last method proposed by Sheft et al., (2008)is to limit the bandwidth of the extracted TFS signal of the analysis filter bandwidth in order to degrade ENV reconstruction. The results show that TFS stimuli, processed to reduce chances of intelligibility from recovered ENV cues, were still highly intelligible (50%– 80% correct consonant identification).

Naveen and Devi, (2016) investigated speech identification through Kannada chimeric sentences in normal adult individuals across 8 frequency bands. Results have shown that for Kannada language perception, envelope structure cues are important.

Chapter 3

Method

The present study aimed to determine the performance on the identification of hearing aid processed speech across compression with the use temporal fine structure cues and temporal envelope cues in auditory chimeric Kannada sentences by the normal hearing individual.

3.1 Participants

Participants of the study included 30 normal hearing individuals, aged between 18 years to 30 years, (Mean = 24.9, SD = 3.15), with 15 males and 15females.

3.1.1 Inclusion criteria:

The following are the inclusion criteria for selection of the participants of the study:

- Air conduction pure tone hearing thresholds less than or equal to 15 dB HL in both ears at octave frequencies from 250 Hz to 8000 Hz as measured from pure tone audiometry using modified Hughson-Westlake procedure (Carhart & Jerger, 1959).
- Normal middle ear functioning as indicated by ‘A’ type tympanogram (Margolis & Heller, 1987).
- Ipsi-lateral and contralateral acoustic reflex thresholds within 100 dB HL at 0.5 kHz, 1 kHz and 2 kHz.

- No history of speech and language disorder, neurologic disorder or any cognitive deficits.
- Native speakers of Kannada

Written consent was taken from the participants for their willingness to participate in the study

3.2 Instrumentation

The following equipments were used in the study

- A calibrated two channel diagnostic audiometer Grason –Stadler model GSI-61 coupled with acoustically matched TDH 39 headphones housed in MX-41/AR and Radio ear B-71 bone vibrator were used to estimate pure-tone threshold, speech recognition thresholds, speech identification thresholds and uncomfortable levels.
- Calibrated middle ear analyzer GSI tympstar version 2 was used for tympanometry and reflexometry.
- Otoacoustic measurements were carried out using ILOV 6 equipment.
- A digital BTE hearing aid with accessibility to use 2 programs was used.
- KEMAR (Knowles Electroacoustic Manikin for Auditory Research) was used for recording the processed speech from the hearing aid.
- Bruel & Kjær (BZ-5503) Sound level meter was used for checking the presentation level of the chimeric sentence for recording the processed stimuli and also to pick up the output of the hearing aid.

A laptop, loaded with the following softwares:

- Hilbert transform using MATLAB software [MATLAB 7.12.0 (R2011a)]
- Cubase (version 2.0.2) was used for the presentation of the chimeric sentence

- *Bruel & Kjær (BZ-5503) software* was used to record the processed stimulus.
- Adobe Audition (version 3.0) was used for the presentation of the stimulus
- PRAAT (version 6.0.39) software was used for recording the response of the subject
- A personal computer installed with NOAH 3.0 was used for programming the hearing aids.

3.3 Environment

The tests, including routine audiological evaluations and presentation of chimeric sentences done in a sound treated double room where the noise levels are within permissible limits (ANSI S3.1-1999).

3.4 Material

The sentences for preparing chimeric list were selected from "Sentence identification test in Kannada" Geetha et al. (2014). The sentences were selected such that the total number of syllables in each sentence is limited to eight-nine syllables and each word in sentences was not have more than three syllables. Total of eighty pairs of sentences were taken to prepare speech – speech chimera across eight frequency bands which includes one, four, six, thirteen, sixteen twenty-four, thirty-two, and sixty-four.

3.5 Procedure

The study was carried out in four phases.

- Phase I - Preparation of the Kannada Chimeric speech stimuli.
- Phase II - Programming of the hearing aid.
- Phase III - recording of the hearing aid processed speech stimuli using dual and syllabic compression.

- Phase IV - Administration of the processed speech stimuli

Phase I: Preparation of the Kannada Chimeric speech stimuli

- This involved selection of sentences in Kannada and recording hearing aid processed chimeric sentences as stimuli. Each chimeric stimulus was divided into different frequency bands and were grouped for the envelope and fine structure extraction for the preparation of speech to speech chimeras.

- The selected eighty pairs of sentences were processed using Hilbert transform to extract the temporal cues such as envelope and fine structure. Hilbert transform is mainly used to derive envelope function or instantaneous amplitude of a signal. It mainly represents a filter without affecting the gain (Yost & Fay, 2007).

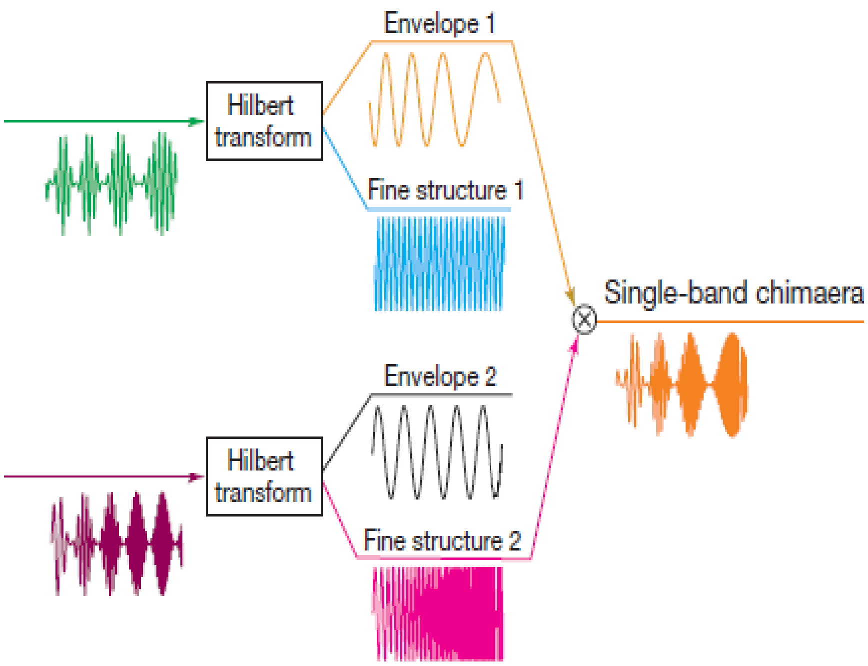

Figure 3.1. Diagrammatic representation of preparation of chimeric stimuli.

Note: Reprinted from Perception of auditory chimeras by Smith, Oxenham and Deglutte, 2001, retrieved from http://research.meei.harvard.edu/chimera/More.html, Copyright 2001 by Association for Research in Otolaryngology

Hilbert transform was computed in a following steps:

- First, calculate the Fourier transform of the given signal x(t).
- Second, reject the negative frequencies.
- Finally, calculate the inverse Fourier transform, and the result will be a complex-valued signal where the real and the imaginary parts form a Hilbert-transform pair.

For example: When x(t) is narrow-banded, |z(t)| can be regarded as a slow-varying envelope of x(t) while the phase derivative $\partial t\ [\tan-1(y/x)]$ is an instantaneous frequency. Thus, Hilbert transform can be interpreted as a way to symbolize a narrow-band signal in terms of amplitude and frequency modulation (Shi, Lee, Liu, Yang, & Wang, 2011). After obtaining envelope and fine structure for each sentence, these temporal cues were exchanged with each other in order to make speech-speech auditory chimeric sentences. For example, envelop of sentence one is combined with fine structure of sentences two to make one chimeric sentence. Likewise, cues were exchanged between all sentences and 80 chimeric sentences were made.

Phase II: Programming of the hearing aid

A digital non linear BTE hearing aid was chosen and programmed for hypothetical flat sensorineural hearing loss with air conduction threshold of 40 dB HL at all audiometric frequencies. A flat hearing loss was used so that the compression characteristic, when tested, remained same across all the frequencies. An acclimatization level of 2 was used while programming. The program 1 of the hearing aid was programmed for syllabic compression. The program 2 of the hearing aid was programmed for dual compression.

Phase III: Recording of hearing aid processed Kannada chimeric sentences using syllabic and dual compression

` The prepared 80 Kannada chimeric sentences were played through speaker at an azimuth of 90 degree in front of KEMAR fitted with non linear hearing aid. The non linear hearing aid was programmed for syllabic and dual compression as program 1 & 2 respectively. The output from the KEMAR was connected to the sound level meter and the recordings were stored in it as .wav format. The final output was stored in personal computer individually for both syllabic and dual compression. The below figure 3.2 shows the graphical representation of the recording of hearing aid processed Kannada chimeric sentences

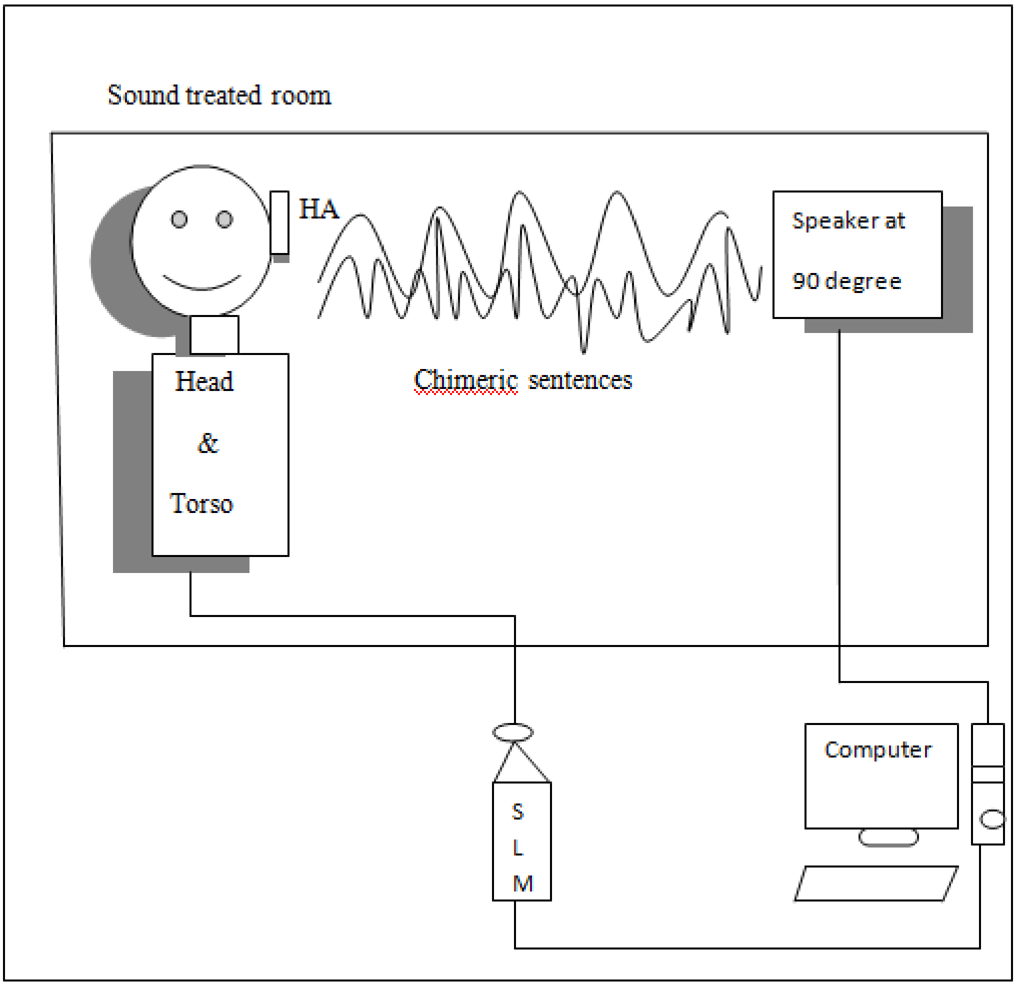

Figure 3.2: Recording of hearing aid processed chimeric sentences

Phase IV: **Administration of hearing aid processed Kannada chimeric stimuli**

All chimeric stimuli having frequency bands of 1, 4, 6, 13, 16, 24, 32 and 64 were given through the headphone TDH-39 at the most comfortable level 65dBHL. For each participant practice trial using three chimeric stimuli was given prior to testing. Each participant was provided with 160 chimeric sentences which were processed in two programs respectively. Instruction provided includes 'Listen carefully to each word and repeat back the sentences'. Speech identification scores were considered based on the percentage of number of correct key words identified from each auditory chimeric sentence. The responses were recorded. The recorded sentences were subjected to further analysis in the presence of a native Kannada speaker were speech identification scores were measured for each sentence. Based on the number of keywords repeated the score was given from 0 to 4 (0 being no words repeated and 4 being all the 4 words repeated correctly). After scoring these scores were converted to a percentage to estimate the speech identification scores.

Chapter 4

Results

The present study aimed at determining the influence of number of frequency bands on the perception of hearing aid processed (dual and syllabic compression) Kannada chimeric sentences on normal hearing individuals. The data was collected on 30 participants for assessing the speech identification for fine structure and envelope cues across 8 frequency bands (1, 4, 6, 13, 16, 28, 32 & 64). The collected data was tabulated and subjected to statistical analysis using SPSS software version 21.

Results of Shapiro-Wilks test showed no normal distribution across the participants. So, non-parametric tests were carried out for further statistical analysis.

The outcomes of the study are explained under the following:

1. Identification of hearing aid processed Kannada chimeric sentences across frequency bands in syllabic and dual compression
2. Identification of envelope cues on hearing aid processed Kannada chimeric sentences identification using syllabic compression and dual compression.
3. Identification of fine structure cues on hearing aid processed Kannada chimeric sentences identification using syllabic compression and dual compression.
4. Comparison between the syllabic and dual compression across frequency bands

The participants were presented with eight list of hearing aid processed Kannada chimeric sentences representing different frequency bands containing 10 chimeric sentences (4 key words in each). Each correct identification of key words was awarded with 1 score. The mean raw score and standard deviation scores of speech identification for chimeric words and chimeric sentences across different frequency bands were calculated using the descriptive statistical analysis as indicated in table 4.1**.**

Table 4.1 Raw Mean, Median and Standard Deviation of Chimeric sentences across frequency bands and different compressions.

Stimulus	Mean (%)	SD	Median
Du 1	0.08	0.45	0.00
Du 4	5.58	5.32	5.00
Du 6	10.08	4.97	11.25
Du 13	49.08	17.98	50.00
Du 16	100.00	0.00	100.00
Du 24	100.00	0.00	100.00
Du 32	100.00	0.00	100.00
Du 64	100.00	0.00	100.00
Sy 1	0.1667	0.63	0.0000
Sy 4	8.58	5.02	10.00
Sy 6	11.66	5.14	12.50
Sy 13	59.83	17.33	60.00
Sy 16	100.00	0.00	100.00
Sy 24	100.00	0.00	100.00
Sy 32	100.00	0.00	100.00
Sy 64	100.00	0.00	100.00

Note:Du = Dual compression, Sy = Syllabic compression, 1-64 indicates number of frequency bands.

Above results of table 4.1 shows an improvement in the hearing aid processed Kannada chimeric sentence identification as the number of frequency band increases both in syllabic and dual compression.

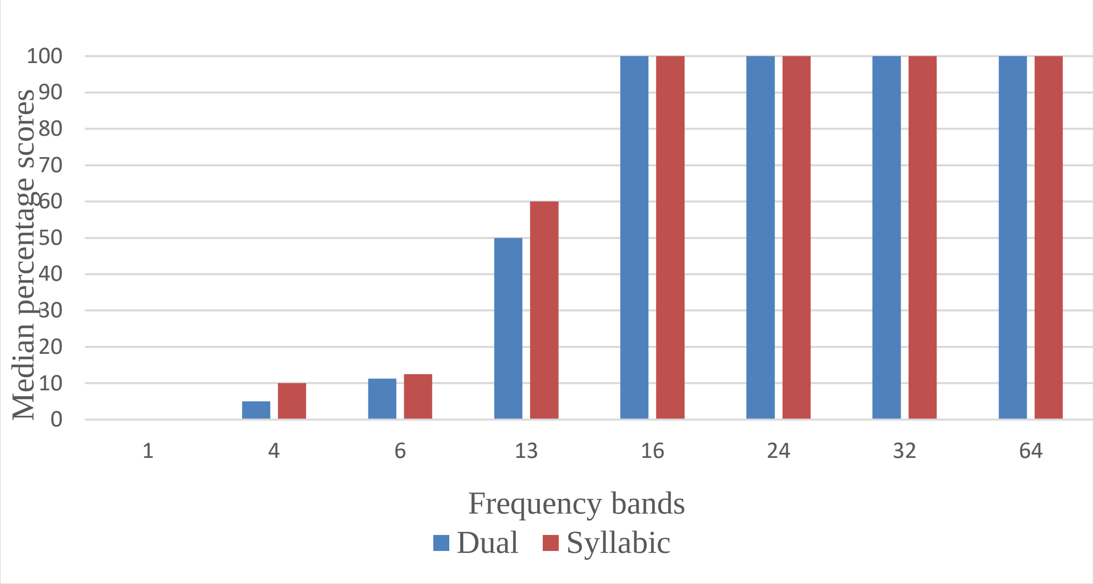

Figure 4.1: Median percentage scores for hearing aid processed chimeric sentence identification in both syllabic and dual compression.

Above figure 4.1 shows percentage score of hearing aid processed chimeric sentence identification across the 8 frequency bands which shows an improvement in word identification as frequency band increases from 6 to 13 and above that it reaches maximum score of 100 % till the band 64.

4.1. Identification of envelope cues and fine structure cues on hearing aid processed Kannada chimeric sentences using syllabic compression and dual compression across frequency bands

The data were calculated for the frequency of participants correctly identified any-one cues were measured across conditions and it is tabulated in table 4.3.

Table 4.2 Representation of participant frequency (%) for identification of fine structure and envelope cues in dual and syllabic compression system across frequency bands for chimeric sentences.

	Dual		**Syllabic**	
Frequency band	**Fine Structure (%)**	**Envelope (%)**	**Fine Structure (%)**	**Envelope (%)**
S 1	0	3.33	0	6.66
S 4	0	73.33	0	90.0
S 6	0	93.33	0	90
S 13	0	100	0	100
S 16	0	100	0	100
S 24	0	100	0	100
S 32	0	100	0	100
S 64	0	100	0	100

Note: S = Chimeric sentence list, 1- 64 indicates number of frequency bands

The above table 4.2 depicts the frequency of participants who identified the fine structure and envelope in both dual and syllabic compression. As all the participants in the study identified using temporal envelope upcoming statistical analysis were focused on temporal envelope. Further Friedman test was administered to check if there is any difference across bands for hearing chimeric sentences.

Table 4.3: Result of Friedman test for **c**omparison of envelope cues between dual and syllabic compression on chimeric sentence identification.

Condition	**χ^2**
Dual compression	207.357*
Syllabic compression	207.766

*Note: * indicates p <0.01*

Since there was a significant difference further pairwise comparison across different frequency bands was done separately for hearing aid processed Kannada chimeric sentences in dual and syllabic compression.

4.2 Identification of envelope cues on hearing aid processed Kannada chimeric sentences identification using syllabic compression and dual compression.

Since the Friedman test revealed significant differences across bands, Wilcoxon signed rank test was done to find out pairwise comparison across different bands of hearing aid processed chimeric sentences and results are shown in table 4.4

Table 4.4 Result of Wilcoxon signed rank test for **c**omparison across frequency bands within dual compression in envelope cue on chimeric sentence identification

	1 Du	4 Du	6 Du	13 Du	16 Du	24 Du	32 Du	64 Du
1 Du		4.128**	4.651**	4.790**	5.396**	5.396**	5.396**	5.396**
4 Du			2.876**	4.787**	4.803**	4.803**	4.803**	4.803**
6 Du				4.793**	4.806**	4.806**	4.806**	4.806**
13 Du					4.790**	4.790**	4.790**	4.790**
16 Du						.000	.000	.000
24 Du							.000	.000
32 Du								.000
64 Du								

*Note: * indicates p <0.01, Du = Dual compression, 1 to 64 bands*

The results of table 4.4 reveals that there is a significant difference across pairwise comparison except for comparison of bands 16 -24, 16-32, 24-32, 24-64, and 32-64. Similarly the pairwise comparision was done for the syllabic compression across bands with is depicted in Table 4.5

Table 4.5 Result of Wilcoxon signed rank test for **c**omparison across frequency bands within syllabic compression in envelope cue on chimeric sentence identification.

	1 Sy	4 Sy	6 Sy	13 Sy	16 Sy	24 Sy	32 Sy	64 Sy
1 Sy		4.567**	4.566**	4.795**	5.324**	5.324**	5.324**	5.324**
4 Sy			3.337**	4.789**	4.802**	4.802**	4.802**	4.802**
6 Sy				4.787**	4.810**	4.810**	4.810**	4.810**
13 Sy					4.792**	4.792**	4.792**	4.792**
16 Sy						.000	.000	.000
24 Sy							.000	.000
32 Sy								.000
64 Sy								

*Note: * indicates p <0.01, , Sy = Syllabic compression, 1 to 64 bands*

The above table 4.5 reveals that in syllabic compression across different frequency bands there was significant difference except between the pairs of 16-24, 16-32, 16-64,24-32,24-64,32-64.

Futher pair wise comparision was done between dual and syllabic compression in envelope cue as depicted in table 4.6

Table 4.6: Result of Wilcoxon signed rank test for **c**omparison between dual and syllabic compression in envelope cue between frequency bands on chimeric sentence identification.

	1 Du	4 Du	6 Du	13 Du	16 Du	24 Du	32 Du	64 Du
1 Sy	0.564							
4 Sy		0.005*						
6 Sy			0.058					
13 Sy				0.000*				
16 Sy					1.000			
24 Sy						1.000		
32 Sy							1.000	
64 Sy								1.000

*Note: * indicates p <0.01, Du = Dual compression, Sy = Syllabic compression, 1 to 64 bands*

From the above table in bandwise comparision between syllabic and dual compression, it is evident that there is significant difference in 4, and 13 bands.

Chapter 5

Discussion

Sound can be mathematically factored into the product of a slowly varying envelope (also called modulation), and a rapidly-varying fine time structure (also known as carrier). Our aim was to find out which of the two factors (envelope or fine structure) were most important for auditory perception of Kannada sentences. To do so, novel stimuli was synthesized which has the envelope of one sound and the fine structure of another sound. Those stimuli are called "auditory chimeras". In order to prepare the chimeric sentences Hilbert method was used in the current study. When a channel signal is manipulated by the Hilbert envelope, i.e. extracting the envelope cue using Hilbert transform leaving the signal without any envelope fluctuations, corresponding to fine structure of the physical signal, sometimes called as fine structure speech (Lorenzi, 2006).

Problem with this filtering using Hilbert transform is that, after filtering on the basilar membrane envelope cues are reintroduced which contains useful information (Ghitza., 2001; Zeng et al., 2004; Gilbert & Lorenzi,2006). This phenomenon is addressed as Envelope recovery or reconstruction which supports the test findings in the current study. Envelope perception was found to be predominant from 4th band to 64th band for speech chimeric identification for sentences. On contrary the temporal fine structure cues were not utilized in identification of sentences both in syllabic and dual compression in Kannada language. The participants in the study were only representatives of hearing impaired being envelope and fine structure processing unaffected biologically. But in the case of hearing impaired individuals studies have revealed that cochlear hearing loss in adults affects the ability to encode and/or use

TFS cues rather preserving the ability to encode the E cues (Lorenzie et al, 2006). The contrasting results of TFS warrants us about the processing takes place in the hearing aid. Studies shown that fast acting compression system disturb the temporal structure

Syllabic compression and dual compression had same effect on speech identification scores (SIS) for both quiet and in noisy conditions but the hearing aid user preferred to use dual compression (Geetha & Manjula, 2005).In the present study only significant difference was observed in 4,and 13 bands between syllabic and dual compression for the perception through envelop cues. Also 50 % of scores were achieved in dual compression whereas 60 % of scores were achieved in syllabic compression at 13 bands. The scores above 13 bands reached 100% both in syllabic and dual compression. The participant frequency (%) for identification of temporal envelope both in syllabic and dual compression clearly tells us 13 bands are required to achieve 100 % identification. Naveen and Devi (2016) investigated speech identification through Kannada chimeric sentences in normal adult individuals across 8 frequency bands. Results have shown that for Kannada language perception, envelope structure cues are important.

The use of sentences also provides the contextual clue to a certain extent but use of frequency bands (1, 4, 6, 13, 16, 24, 32, and 64) limits the contextual clues. The present study provides the details about processing of temporal envelope and temporal fine structure cues in non linear hearing aids.

Chapter 6

Summary and Conclusion

The fine structure cues of the speech have an important role in the perception of tonal language (Mandarin, Chinese) and envelope cues are important for the perception of English language. In Kannada language, normal hearing listeners predominantly used the envelope cues for speech identification. In case of hearing impaired listeners hearing thresholds are affected, frequency resolution and temporal resolution are affected and also have reduced uncomfortable loudness level. Hearing threshold can be compensated by appropriate amplification, reduced uncomfortable loudness can be compensated by using limiting techniques but the affected frequency and temporal resolution cannot be compensated. The hearing aid should provide good spectrotemporal information of the stimulus. In compression systems it is possible that it disturb the temporal structure of the stimulus. Hence the aim of the study was to compare the syllabic and dual compression on non linear hearing aid processed kannada chimeric sentences.. The objectives aimed were to study the influence of envelope cues on hearing aid processed Kannada chimeric sentences identification using syllabic compression and dual compression and to compare between the different compressions.

. The study started with the null hypothesis that there is no significant effect of envelope on hearing aid processed Kannada chimeric sentences identification using syllabic compression and dual compression. The study was carried out in two phases where phase 1 included the preparation of chimeric stimuli and phase 2 was programming the hearing aid. Phase 3 was recording of hearing aid processed Kannada chimeric sentences. Phase 4 was the administration of hearing aid processed Kannada chimeric sentences to normal hearing individuals. Descriptive statistical analysis was carried out on SPSS software (version 20). The results revealed that envelope cues are predominantly used both in syllabic and dual compression for speech identification whereas temporal fine structure cues are not used for speech identification. And also significant difference was observed between syllabic and dual compression only in 4 bands and 13 bands.

Clinical implications

The results of the study could have some clinical implication in the modification of amplification devices. The current study was a preliminary attempt to investigate temporal cues on the hearing aid processing using syllabic and dual compression. To gain in depth knowledge regarding the same, a detailed study across and hearing-impaired individuals, in different languages to be carried out.

The present study's results reveal that envelope cues are important for the perception of hearing impaired listeners in syllabic and dual compression and also the temporal fine structure is affected in processing of hearing aids. Hence there is a scope of improvement in hearing aid technology to process fine structure information of the speech signal.

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