Perception of Hearing Aid Processed Speech in Adverse Listening Condition

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This Dissertation is submitted as part fullfillment for the Degree of Master of Science in Audiology University of Mysore, Mysore

MAY 2015

ALL INDIA INSTITUTE OF SPEECH AND HEARING

Certificate

This is to certify that this dissertation entitled **"Perception of Hearing Aid Processed Speech in Adverse Listening Condition"** is a bonafide work in part fulfillment for the degree of Master of Science (Audiology) of the student with Registration No. 13AUD010 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 dissertation entitled "**Perception of Hearing Aid Processed Speech in Adverse Listening Condition"** is the result of my own study under the guidance of Dr. Sandeep M., Reader in Audiology, Department of Audiology, All India Institute of Speech and Hearing, Mysore and has not been submitted earlier to any other University for the award of any other Diploma or Degree.

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Dedicated to My Parents, Sister,
Friends
And
My Guide Dr. Sandeep M.

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Chapter 1: Introduction

For rehabilitation of the hearing impaired individuals, hearing aid fitting becomes an essential factor. However at present, none of the technologies used in these hearing aids can exactly mimic or fully supplement natural hearing. Hearing aids can account for reduced audibility with amplification, however as the signal passes through the processing stages of the hearing aid, it undergoes additional temporal and spectral changes leading to distortion of speech signal and thus affecting speech perception (Van Tasell, 1993).

Hedrick and Rice (2000) found that hearing aids with syllabic compression give more amplification to burst and less amplification to vowel part of the syllable, thus altering the consonant to vowel ratio (CVR). As CVR is an important cue for identifying place of articulation, change in CVR can lead to increased error in the identification of place of articulation Dreschler (1989) reported that compression circuits used in hearing aids modify the temporal characteristics of the speech signal in various levels but these temporal distortions introduced by the hearing aids has no significant effect on speech perception of individuals with degraded temporal resolution capacity as in cases of sensorineural hearing loss.

Stelmachowicz, Kopun, Mace, Lewis, and Nittrouer (1995) reported that the hearing aid processed speech of both linear and non-linear hearing aids shows absence of first formant frequency of the input signal on the spectrogram due to frequency roll off used in these hearing aids to avoid the upward spread of masking by the ambient noise present in the environment. The periodicity of the vowel, which is an important cue for pitch identification in adverse listening conditions is also lost due to amplified background noise and noise generated by the hearing aids.

In natural speech, the vowel will be more intense than the consonant part of the syllable and the consonants will have energy in the high frequency region which will be generally affected in individuals with sensorineural hearing loss. Thus it is easy for these individuals to identify vowels better than consonants. However, when the speech signal passes through the hearing aid the spectrogram of the output signal will show absence of F1 frequency which can further degrade speech perception in individuals with sensoineural hearing loss.

Chundu (2012) investigated the acoustic parameter of the speech signal processed by a compression hearing aid. In this study he used four vowel consonant vowel (VCV) syllables /iki/, /ipi/, /isi/ and /iʃi/ processed by Starkey BE39FX dual channel analogue wide dynamic range compression hearing aid and reported that, the hearing aid output lacked high frequency burst information which is a major cue for the perception of place of articulation for /p/. The study also reported that the hearing aid output lacked friction amplitude, burst duration and clear spectral peaks which are important cues for identifying consonants.

1.1 Justification for the Study

The review of literature has shown that hearing aid modifies spectral and temporal characteristics of the speech input other than amplifying the signal. As both spectral and temporal characteristics are known to play an important role in the speech identification of consonants as well as vowels, hearing aid processing is likely to affect speech perception. These spectral and temporal distortions may not negatively influence the speech perception to a larger extent in a quiet environment due to rich redundancy present in the speech signal. However, if the redundancy is reduced, the negative influence of these spectral and temporal distortions induced by the hearing aids on speech perception is expected to be evident. That is, in the adverse listening conditions, speech processed through the hearing aids shall lead to poorer speech identification compared to the one that is not processed through the hearing aids. In the literature the influence of hearing aid processing on speech with such a methodology is not available. From the results derived from such a methodology, important inferences about practical utility of hearing aids in the adverse listening conditions can be drawn. Therefore the present study was taken up.

1.2 The Objectives of the Study

1. To compare the identification scores of nonsense syllables processed through the hearing aid with that of unprocessed syllables in different signal to noise ratios
2. To compare identification scores across different signal to noise ratios in unprocessed and hearing aid processed speech
3. To compare the identification scores of unprocessed and hearing aid processed speech between normal and SNHL group

Chapter 2: Literature Review

2.1 Speech Perception in Individuals with Hearing Loss

Review of literature reveals that there is a definite relation between hearing sensitivity and speech perception (Alexander & Masterson, 2015; Beasley & Rosenwasser, 1950; Crain & Yund, 1995; Mullins & Bangs, 1957; Souza, Jenstad, & Boike, 2006). Majority of these studies have concentrated on establishing the relation between puretone sensitivity and word recognition (Beasley & Rosenwasser, 1950; Mullins & Bangs, 1957), but there are also ample of studies in literature which have explored the relation between puretone sensitivity and consonant recognition (Alexander & Masterson, 2015; Crain & Yund, 1995; Dubno & Levitt, 1981; Hillock-Dunn, Buss, Duncan, Roush, & Leibold, 2014; Marriage & Moore, 2003; Souza et al., 2006). The findings of these studies have showed similarities in phoneme and feature recognition abilities among hearing impaired population regardless of the nature or severity of hearing loss. With increasing severity of hearing loss, the mean scores of consonant recognition decreased, however the pattern of confusion has been reported to remain similar to that of normal hearing individuals. For example, irrespective of degree or configuration of hearing loss, voicing and manner features are perceived better by these individuals compared to place of articulation. Eventhough initially it is safe to assume that individuals with different degree and configuration should perform differently in consonant identification task, majority of the studies have failed to show difference in performance (Dreschler, 1989; Hillock-Dunn et al., 2014).

2.2 Effect of Channels and Compression on Speech Perception

Multi-channel processing can modify speech spectrum by altering spectral composition, perhaps reducing spectral contrasts which plays major role in discrimination of certain speech sounds. In multi-channel Wide Dynamic Range Compression (WDRC), the acoustic signal is filtered into several frequency channels, the bandwidth and center frequency of each channel are decided based on the fitting formula used and short term levels in each of these channels are estimated and gain is selected accordingly. Studies have shown that increasing the number of channels upto certain level can improve speech recognition (Alexander & Masterson, 2015; Crain & Yund, 1995; Yund & Buckles, 1995). However, if number of channels are increased beyond 8 channels speech recognition is known to get degraded (Crain & Yund, 1995). This could be due to the fact that when the number of channels are increased, greater gain is applied to the channels which have valleys in the spectrum and less gain to those channels which have peaks, resulting in flat spectral envelope especially in noise (Dubno & Levitt, 1981)

Marriage and Moore (2003) reported that, wide dynamic range compression (WDRC) used in most of the modern hearing aids distort the temporal envelope and reduce the modulation depth of the signal. When WDRC is used with multichannel compression, which is known to distort the spectral envelope of the signal, the combined effect of these two (WDRC and multichannel compression) can reduce the benefit of improved audibility. The dynamic properties of compression (attack and release time) can alter temporal properties of the input signals at various levels (Abraham, 2012; Kuk, Ludvigsen, & Paludan-Müller, 2002).

Marriage and Moore (2003) investigated the effect of WDRC on consonant recognition in fourteen children with moderate to profound hearing loss. These children were fitted with either a linear or a single channel WDRC hearing aid and a counter-balancing design was used. All the children were given two sets of tasks, the first being the Royal National Institute for Deaf people (RNID) Consonant Confusion Task (CCT) which is a closed set task, and the second, was an open set task which consisted 5-8 lists of Manchester Junior Word List (MJWL) and an additional 8 lists of monosyllabic words from the Bench-Kowall- Bamford (BKB) sentences. Both the tasks were administered in quiet and in noise (10dBSNR). The results showed that WDRC gave significant benefit in both quiet and in noise.

Hickson, Thyer, and Bates (1998) reported CVR changes with respect to 12 different compression settings and linear gain settings. The study revealed increased CVR for most of the compression settings compared to linear setting. Also CVR was found to increase with increasing crossover frequency. The study also reported that for stop consonants, higher compression ratio of 3 resulted in maximum CVR. Affricates had positive CVRs for most of the compression settings, and averaged around 2.56 dB for an input of 65dB. Thus, the study showed change in CVR of the input signal with different compression settings could affect speech perception in individuals with moderate to moderately severe sensorneural hearing loss.

Hedrick and Rice (2000) reported that hearing aids with syllabic compression give more amplification to burst part and less amplification to vowel part of the syllable thus altering the consonant to vowel ratio (CVR). As CVR is an important cue for identifying place of articulation, it leads to increased error in the identification of place of articulation.

Dubno and Levitt (1981) studied nonsense syllable perception, both in quiet and noise (5dB SNR). These syllables were presented to 6 normal hearing individuals, and the responses were obtained using a closed set response sheet. Acoustic analysis and consonant confusion pattern analysis were carried out. For acoustic analysis, 11 parameters were selected for acoustic analysis in both quiet and noise conditions. These included consonant and vowel spectral peaks (formants), overall consonant-noise bandwidth, total consonant and vowel energy, duration of consonant, vowel and stop closure, and three measures of the second formant transition (origin frequency, direction, and magnitude). Three additional measurements were made in noise condition only. These were consonant to noise ratio, vowel to noise ratio, and effective consonant bandwidth in noise. The results showed that the average score was significantly higher for affricates compared to stops, nasals and fricatives. Performance on consonants formed near the front of the mouth (/ p, f, b, v, m/) was significantly better than middle or back productions in quiet and in noise. They also reported that place errors are the most common in both conditions (manner of articulation identified correctly). The large increase in place/manner errors in noise may be a result of an increase in random guessing in the more difficult listening conditions.

Dreschler (1989) reported that compression circuits used in hearing aids modify the temporal characteristics of the speech signal at various levels, but these temporal distortions introduced by the hearing aids have no significant effect on speech perception of individuals with degraded temporal resolution capacity as in cases of sensorineural hearing loss.

Chundu (2012) investigated the acoustic parameter of the speech signal processed by a compression hearing aid. In this study he used four vowel consonant vowel (VCV) syllables /iki/, /ipi/, /isi/ and /iʃi/ processed by Starkey BE39FX dual channel analogue wide dynamic range compression hearing aid, the compression ratio of the hearing aid was varied (1:1, 2:1, 3:1 & 4:1). The hearing aid was placed on the ear of KEMAR and the target speech stimuli were presented at 0° azimuth and hearing aid output was recorded for each condition, using a microphone placed in the ear canal of the KEMAR. Similarly, for unaided condition the speech stimuli were recorded without the hearing aid. Both hearing aid processed and unprocessed signal were subjected to acoustic analysis which revealed significant changes in the spectral and temporal characteristics of the hearing aid processed speech which were also dependent on the compression ratios used. There was a decrement in burst amplitude and frication amplitude, and the burst amplitude for /ipi/ decreased significantly compared to that of /iki/, and the frication amplitude of /isi/ was reduced significantly compared to that of /iSi/. No significant difference observed in the burst duration across all four compression ratios. The study also reported significant difference in spectral peak for both /isi/ and /iSi/ between linear and 2:1 compression condtion.

Hickson, Dodd, and Byrne (1995) studied consonant perception through linear versus compression amplification in quiet and noise in 15 individuals with mild to moderate sensorineural hearing loss. Nonsense Syllable Test (NST), speech material was presented through a speaker, and the hearing aid output was recorded in quiet and two noise conditions (four talker babble and sharp intermittent noise), for both linear and compression amplification settings (compression ratios: 1.3 & 1.8). The results revealed no significant difference in consonant perception, between linear and compression amplification settings in quiet and sharp intermittent noise condition. But for four talker babble condition, linear amplification showed better scores compared to two different compression amplifications used in the study. Compression ratio of 1.3 yielded the poorest score in the study which was statistically different from linear amplification condition. This poor performance with the compression circuit was partially attributed to adaptive recovery time circuit used in the compression circuits.

Abraham (2012) studied the effect of frequency channels (2, 4, 6 & 16 channels) in a hearing aid on Voice Onset Time (VOT), and its effect on speech perception. Speech stimuli used in the study consisted of six stop consonants, three voiced (b, d, g) and three unvoiced (p, t, k) in three meaningful voice cognate word pairs, such as BEES / PEAS, DIME / TIME and GOAT / COAT. Speech stimuli were presented through a calibrated loud speaker, and output from four hearing aids which differed in terms of number of channels (2, 4, 6 & 16 channels) were recorded. Hearing aid processed and unprocessed speech were presented to six individuals with normal hearing sensitivity, and they were asked to judge whether there was any difference between the two. VOT was measured using Bruel and Kjaer Pulse Time Edit and Analysis software. VOT measurement of the hearing aid processed and unprocessed speech stimuli revealed a significant reduction in VOT in the processed stimuli, which ranged between 14% and 20% reduction relative to the VOT of the unprocessed stimuli, with a mean of 17ms for both lead and lag VOT. Perceptual analysis revealed 11% reduction in speech intelligibility when lead VOT was reduced by 16%, and 19% reduction in speech intelligibility when lag VOT was reduced by 18%.

Alexander and Masterson (2015) studied effect of WDRC, release time, number of channels (4, 8, & 16) on speech recognition in noise (steady state & modulated noise) at three different SNRs (-5, 0 & +5dB). Twenty four individuals, with mild to moderately severe SNHL with median age of 62 years participated in the study. The study used 720 sentences from the Institute of Electrical and Electronics Engineers (IEEE) sentence data base. These sentences were mixed with a modulated noise using the procedure recommended by International Collegium of Rehabilitative Audiology (ICRA), and these stimuli were processed by a hearing aid simulator, designed using MATLAB to obtain absolute control over attack and release time. The attack time was kept constant at 5 ms and release time was varied between 40 to 640 ms in octave steps. The output from the simulator were scaled and filtered into 4, 8, and 16 channels and the center frequencies were as per the recommendation of Desired Sensation Level Input/Output (DSL I/O). Individualized prescription were generated separately for all the subjects and presented through a calibrated head phone. The study reported that at -5dBSNR, speech recognition was significantly higher in modulated noise condition compared to steady masker condition. But at +5dBSNR, speech recognition was better in steady state condition compared to modulated noise condition. The study also reported that maximum speech recognition scores were obtained for 8 channels with long attack time compared to any other conditions considered in the study. For 4 channels, speech recognition was significantly better for short attack time and for a 16 channel hearing aid, opposite was seen where speech recognition was significantly better with long attack time and for 8 channels there was no significant difference between short and long attack time.

Souza et al. (2006) studied the acoustic effect of compression amplification on speech in noise. The study measured the effect of single channel and two channel compression on output SNR (using inversion technique which allows separation of speech and noise from the hearing aid output), effect of fast acting compression WDRC on speech dynamic range, and effect of WDRC on amplitude envelope. For measuring effect of WDRC on SNR, ten sentences taken from connected speech test (Cox, Alexander, & Gilmore, 1987) were mixed with steady state noise at four SNRs (-2, +2, +6 & +10dB). These sentences were amplitude compressed using a compression algorithm implemented in C code which mimicked input compression of a single channel and two channel WDRC amplification hearing aid. For single channel compression, compression ratio of 2:1, attack time of 5ms, release time of 50 ms, and compression threshold of 45dBSPL was used. For two channel compression, signals were compressed in two channels with a crossover frequency of 1200-Hz and other parameters were similar to single channel hearing aid. The output from above algorithm was subjected to inversion technique to obtain compressed speech and compressed noise separately, using which output SNR and effective compression ratio can be measured. To measure the effect of varying compression parameters on amplitude envelope of speech, ten sentences from speech perception in noise (SPIN) test (Bilger, Nuetzel, Rabinowitz, & Rzeczkowski, 1984) were used. The stimuli were recorded in quiet and at 8dBSNR. These files were amplitude compressed using a two channel compression algorithm with a crossover frequency of 1200Hz keeping the compression threshold and attack time constant while varying the compression ratio (2:1, 4:1, 10:1) and release time (12, 100, 800, 1600 ms). To determine the effect of compression parameters EDI was calculated. Results showed that the output SNRs were poorer after compression, compared to linear amplification at higher SNRs input levels. But at lower input SNRs like 10 dB, the output SNRs did not vary between linear and compression amplification. This is attributed to the fact that, at lower input SNR the noise level is below compression knee point and does fail to trigger gain for pauses in speech stimuli leading to better SNRs. The study also showed that the output SNRs were better for two channel compression compared to single channel compression. The measurements of effective compression showed that, it decreases with increasing input SNRs. That is at higher noise level, the effective compression ratio is negligible and thus leading to no advantage over linear amplification. EDI measurements showed that EDI was more in quiet condition compared to noise in background. The EDI also varied with the compression parameters. When the compression parameters were set to extreme levels, the EDI showed lesser values suggestive of altered output envelope. Therefore, authors suggested that clinicians should be careful when selecting compression parameters.

Verschuure, Prinsen, and Dreschler (1994) studied the effect of syllabic compression and frequency shaping on speech intelligibility in 6 moderate to moderately severe sensorineural hearing loss. The study used 20 CVC word lists developed by TNO-IZF Institute in Soestberg. The words were embedded into five different carrier phrases. The world list was processed through a prototype digital processor developed by Philip Research Laboratories. The processor was programmed to process the signal using 5ms attack time, peak hold time of 15ms, a delay of 9ms. The compression threshold used was

60dBSPL. Four different compression ratios were selected in the study (1:1, 2:1, 4:1 & 8:1). The output from the digital processor was saved and played back to the subjects using TDH-39 headphone routed through a calibrated audiometer to control the presentation level. The study reported that the consonant recognition scores were significantly better for 4:1 compression ratio which showed an improvement of almost 8-10% and poorer scores were obtained with linear amplification.

Walden, Grant, and Cord (2001) studied the effect of amplification and speech reading on consonant recognition in 25 individuals with mild to moderate sensorineural hearing loss. All the participants were hearing aid users, at least for a duration of 10 weeks. The test stimuli comprised of 14 Vowel-Consonant-Vowel (VCV) syllables, with /p,t,k,f,s,S,b,d,g,v,z,Z,m,n/ as consonants and /a/ being the vowel. Stimulus was presented in two modes, which were auditory alone and auditory + vision for both aided and unaided conditions. The subjects were instructed to identify the syllable presented and responses were subjected to statistical and SINFA analysis. Results showed significant benefit from hearing aid in both audtiory only and auditory + vision condition. Amplification showed benefit for all the three features (voicing, place & manner of articulation). However maximum benefit was seen for identification of place of articulation.

Crain and Yund (1995) investigated the effect of multichannel compression on vowel and stop consonant discrimination in normal and in individuals with sensorineural hearing loss. Eight vowels and two set of VCV syllables were synthesized using KLSYN88a Laboratory Speech Synthesizer. The first set of the VCV set consisted of three consonants /b/, /d/, and /g/ in two vowel contexts (/i/ & /u/) and second set consisted of a VCV continuum that changed from /idi/ to /iti/ by varying VOT. Both vowels and syllables were filtered into 2, 4, 8, 16 and 32 bands. Two types of compression were used in the study, the first was flat compression where the compression ratio were set constant in all the channels and in second compression ratio were set to suit the audiometric configuration of all the participants. The processed stimuli were then presented to participants and responses were obtained. Results revealed that both consonant and vowel recognition and discrimination scores decreased as the number of channels increased above 8 channels and also as the compression ratio was set high. The best scores were obtained when the compression ratios were set to suit the audiometric configuration of each individuals.

Garvita and Sandeep (2010) investigated the effect of hearing aid on spectral and temporal characteristics of the speech stimulus /da/, and their effect on brainstem potentials. Syllable /da/ of 541ms and 47ms duration were processed through two hearing aids; a multichannel WDRC hearing aid and a trimmer based analog hearing aid. Hearing aid processed /da/ (541 and 47ms) and unprocessed /da/ were subjected to subjective analysis and acoustic analysis using PRAAT. The stimuli were presented to 20 subjects with normal hearing senisitivity at a comfortable level and the participants were asked to judge the naturalness of the stimulus presented. The study found effect of hearing aids on both temporal and spectral charecteristics of the speech signals. Among spectral characteristics, fundamental frequency did not vary between natural and hearing aid processed speech, while first, second and third formant frequencies were different between the two. Among temporal measures, VOT, vowel duration and total syllable duration changes were seen. Among these, VOT changes were reported to be more evident between hearing aid processed and natural speech.

2.3 Speech Perception in Noise

Although most individuals with sensorineural hearing loss report difficulty understanding speech in the presence of noise, the magnitude of problem varies from individual to individual with same degree and configuration of hearing loss (Plomp, 1994).

Hornsby and Ricketts (2001) Studied the effect of SNRs, speech presentation level and compression ratios on consonant recognition in 9 subjects with normal hearing sensitivity. The study used 44 CV and 42 VC syllables of the nonsense syllable test (NST). The syllables were mixed with speech shaped noise at two SNRs (0 & +6 dB) and the stimuli were fed into compression algorithm buy varying the attenuation level to mimic different presentation levels. The compression threshold was set at 25 dB below the presentation level. Compression ratio was varied (1:1, 2:1, 4:1 & 6:1) the compression ratio of 1:1 was assumed to be similar to that of unprocessed stimuli. The attack and release time were kept constant at 5 and 20ms. The final 86 stimuli for each condition were presented to the subjects at 65 dB SPL randomly using a calibrated insert earphone (Etymotic ER4) monaurally and the subjects were made to select appropriate syllable that appeared on the computer screen, the data was subjected to SINFA and statistical analysis. Results revealed main effect of SNR, presentation level and compression ratio. Consonant recognition scores decreased at poor SNRs, high presentation level and also when compression ratio was increased and a systematic trend was observed in all the conditions. An average of 4% decrement was reported when the compression ratio was increased from 1:1 to 6:1. SINFA revealed decreasing feature transformation with increasing compression at low input levels but not at high input levels. Place, manner and frication demonstrated this trend at both SNRs while voicing feature was resistant to the negative effects of level and compression ratios.

Hillock-Dunn et al. (2014) investigated the effect of non-linear frequency compression in children aged between 9 and17 years. Seventeen children with bilateral sensorineural hearing loss participated in the study. All children were regular users of hearing aids. In this study all children were fitted with Noida V SP AMD hearing aid was programed according to the audiometric threshold of each child. In experiment 1, eleven syllables /b, s, d, h, k, m, n, p, t, v, z, ʃ/, were presented to the children twice, first with compression switched on, second with compression switched off. The child responses were inputted on to a computer using a user interface developed on MATLAB environment. In experiment 2, 25 spondee words were mixed with speech shaped noise and two talker noise separately and presented to children for identification using a loud speaker. Four alternative forced choice procedure was used to obtain the response. Results showed that main effect of hearing aid condition was significant indicating similar performance for children with and without compression activated. Feature analysis revealed place of articulation was affected most, followed by manner or voicing feature. High identification score for /s/ & /S/ were reported in the study. Experiment two showed that spondee identification were better with speech shaped noise compared to two talker babble, and no significant difference was reported for compression off and on condition.

Gordon-Salant (1987) investigated the consonant recognition and confusion patterns in older adults with sensorineural hearing loss. Ninteen consonants were paired with three vowels. The stimuli were presented to the subjects with a babbling noise at SNR of 6dB. The responses were recorded and subjected to statistical and feature analysis. The results revealed that, even though the consonant recognition in noise scores decreased in older adults with hearing impairment. The pattern of confusion did not vary significantly. The previous studies have also reported that even though the elderly individuals perform poorer compared to young adults, the pattern of confusion is generally same in the two population.

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Chapter 3: Method

The study aimed to investigate the effect of hearing aid processing on consonant identification in quiet and different signal to noise ratios (Quiet will be considered as one of the SNR in further reportings). It tested the null hypothesis that there is no significant effect of hearing aid processing on consonant identification scores in 4 SNRs in normal individuals and individuals with sensorineural hearing loss (SNHL).

3.1 Participants

A total of 30 adults participated in the study. All the participants were native speakers of Kannada and were in the age range of 20 to 50 years and were geographically from in and around Mysore. All the participants had normal middle ear functioning, ensured using immittance evaluation. None of the participants had any history of psychological or neurological problems. A written consent for participation was taken from each of them prior to their inclusion in the study. Based on their hearing sensitivity, they were divided into two groups; Normal and SNHL groups.

The SNHL group consisted of 10 participants, diagnosed as having moderate to moderately severe SNHL with flat audiogram configuration. All the participants had type 'A' tympanogram and acoustic reflexes proportional to hearing loss. Participants were also screened using speech in noise test and only those individuals who had more than 60% speech recognition at 0 dB SNR were included in the study.

Normal hearing group consisted of 20 individuals who were age and gender matched with clinical group. Participants of this group had their air conduction and bone conduction thresholds within 15 dBHL with an air bone gap of less than 15 dB across 250 Hz to 8 kHz. They had type 'A' tympanogram with normal acoustic reflex thresholds in both ears. These participants were also screened using speech in noise test and only those individuals who had speech identification scores of greater than 70 % were included in the study,

3.2 Instrumentation

Several technical equipments were used in the present study for preliminary audiological evaluation and for experimental procedure. They included

- A calibrated two channel diagnostic audiometer, GSI-61 (Grason-Stadler Incorporation, USA) with Telephonics TDH 39 supra aural headphones and Radio ear B-71 bone vibrator calibrated as per ANSI S-3.6, (2004), was used for hearing threshold estimation and speech audiometry
- A calibrated GSI-tympstar clinical immittance meter, was used for tympanometry and reflexometry
- ILO 292 DP Echo port system was used to record transient evoked otoacoustic emissions
- A double channel equipment (IHS), with SmartEP software was used to record and analyze auditory brainstem responses
- MicroBook MOTU instrument with AHUJA AUD-101XLR dynamic unidirectional microphone was used for recording the test stimuli
- MATLAB- 7 (Language of Technical computing, USA) was used to generate speech noise and mix the generated noise with CV syllables at different signal to noise ratios. The same was used to prepare graphic user interface for response elicitation, and for stimulus presentation

- Adobe Audition v5 software was used to normalize the recorded CV syllables and for preliminary editing
- Sony Vaio laptop computer was used for presenting the stimuli

The stimulus from the laptop was routed through a calibrated audiometer (MA-53) in order to control the intensity of the stimulus. Sennheisser HAD 200 headphones was connected to the audiometer for the output to be delivered.

3.3 Test Stimuli

A total of 10 syllables were used in the present study. Consonant in these syllables was one of the /p/,/t/./k/,/b/./d/,/g/,/s/,/ʃ/,/ʤ/,/ ʧ/, and were always combined with vowel /a/. The aim while choosing the consonant was to cover voiced and unvoiced counter parts of stop consonants, fricatives and affricates of different places of articulation.

3.3.1 Recording of Speech Stimuli

The target syllables were uttered by a male native speaker of Kannada and was recorded using a unidirectional microphone kept at 6 inches from the mouth. Each syllable was uttered 3 times in a neutral tone at conversational loudness. A laptop computer with adobe audition v5 was used for the purpose. The recorded utterances were digitized with a sampling frequency of 44,100 Hz and 16 bit resolution. The recording of the stimuli took place in a sound treated room where the noise levels were within permissible levels (Frank, 2000).

Three samples of each syllable was independently analyzed in terms of its clarity of sound, spectrogram and waveform. The sample that was best among the three was selected as the test stimulus. The ten such syllables were made into a single list and were RMS normalized using adobe audition V5 to maintain similar intensity across the syllables.

3.3.2 Recording of the Unprocessed and Hearing Aid Processed Speech

All ten syllables /pa/, /ta/. /ka/,/ ba/, /da/, /ga/,/ sa/, /ʃa/, /ʧa/ and /ʤa/ were processed through Hansaton salto super power, a 4 channel digital wide dynamic range compression hearing aid. This hearing aid was selected due to positive feedback received from hearing aid users fitted with this same hearing aid at AIISH. To record the hearing aid processed signal, the wave files of the ten stimuli were fed into a computer. The audio output of which was routed into a calibrated diagnostic audiometer. The syllables were then played at 70dBSPL, 5dB above the compression knee point of the hearing aid, through the loudspeaker. The hearing aid was programed for mild, flat sensorineural hearing loss using NAL-NL1 fitting formula. The directional microphone and noise reduction algorithm of the hearing aid were switched off. The hearing aid was placed on the pinna of the KEMAR positioned at zero degree azimuth and at a distance of one meter from the loudspeaker. The receiver of the hearing aid was connected to the ear simulator of the KEMAR. The output (processed speech) from the KEMAR was saved onto a computer using adobe audition V5. To record the unprocessed stimuli, similar procedure was followed, where the KEMAR output was be recorded without the hearing aid. Figure 3.1 shows the equipment setup for recording hearing aid processed speech.

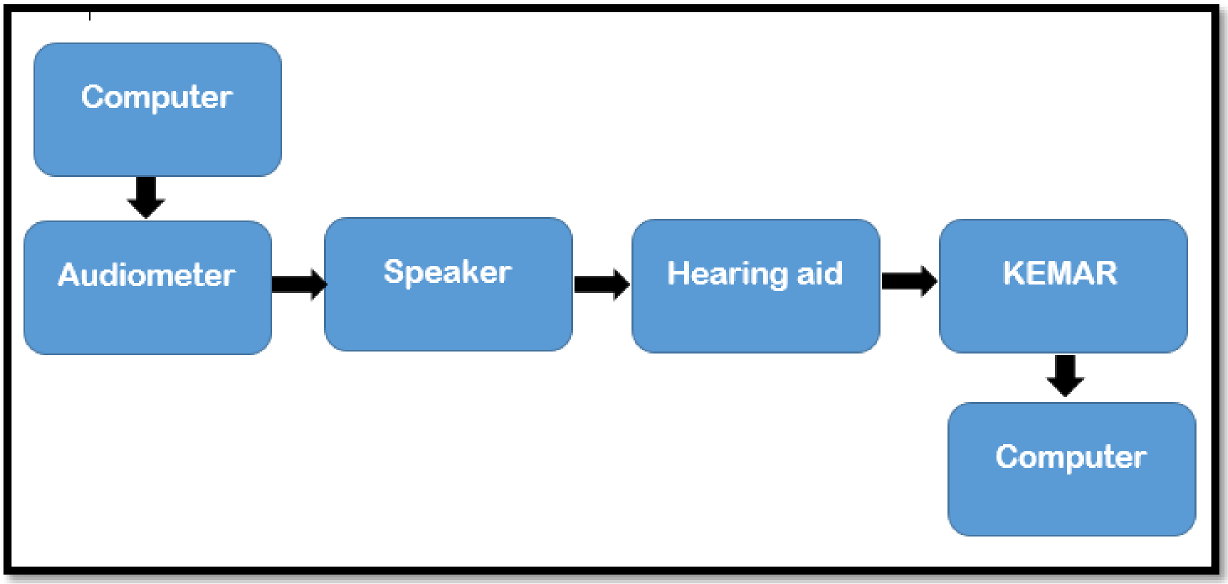

Figure 3.1 Equipment setup used for recording hearing aid processed speech.

Both hearing aid processed stimuli and unprocessed stimuli were group normalized to -15 dB, using adobe audition software version 5 and were saved in the laptop computer.

3.3.3 Mixing of Noise to the stimulus

Speech noise was generated using a MATLAB code developed by G. Nike Gnanateja. The .wav files of the test stimuli were fed into the MATLAB code which generated speech spectrum shaped noise, based on long-term average spectrum of all the stimuli that was fed in. To add speech noise to the stimuli, another MATLAB code developed by Gnanateja and Pavan (2012) was used. This code kept the RMS of the speech at a constant level while the RMS of the noise was varied to generate 3 different SNRs (0, 5, &10) and the original processed stimuli served as quiet stimuli. The total number of stimuli generated were 80 stimuli, (40 hearing aid processed stimuli and 40 unprocessed). These files were saved in .wav format in a laptop computer.

3.4 Test Procedure

3.4.1 Preliminary Audiological Evaluation for Selection of the Participants

Air conduction threshold were measured at octave frequencies, between 250 Hz and 8 kHz while, bone conduction thresholds were measured at octave frequencies between 250 Hz to 4 kHz, using a calibrated GSI-61 audiometer. The thresholds were measured using modified Hughson and Westlake procedure (Carhart & Jerger,1959).

Phonetically balanced (PB) word lists developed by Yathiraj and Vijaylakshmi (2005) was used for assessing speech perception abilities. Half list, consisting 25 words were presented at most comfortable level using a calibrated audiometer and the participants were instructed to repeat back the words heard. Ear specific % correct responses were obtained.

Impedence evaluation included administration of both tympanometry and reflexometry A probe tone of 226 Hz was used for this and tympanogram was measured using decending pressure sweep. Static admittance, peak pressure and external canal volume were noted for each tympanograms. Ipsi-lateral and contralateral reflex thresholds were measured at 0.5 kHz, 1 kHz, 2 kHz, and at 4 kHz.

To test speech in noise perception, both noise and speech were presented monotically, at 0 dB signal to noise ratio (SNR) at 40 dBSL. A total of 25 PB words from Yathiraj and Vijaylakshmi (2005) word list were presented to each ear. The participants were instructed to listen carefully and repeat back the words heard. Percentage correct responses were calculated for each ear individually.

Dual channel ABR recordings using protocol recommended by (Hall, 2007) was done in an electrically shielded room. and OAES were recorded using standardized protocol to note down its SNR, at octave frequencies

3.4.2 Experimental Procedure

All the listening tests were conducted in a quiet room and the stimuli were played using a personal laptop routed through a headphones of calibrated audiometer. The participants were instructed to carefully listen to the stimulus and point to the respective syllable shown on a computer screen (close set identification). Each syllable at each SNR and in each condition (natural and hearing aid processed) was randomly played 3 times to each participant. This resulted in a total of 240 presentations per participant. An inter stimulus interval of 4 second was used and separate list of syllable list for each SNR was prepared and each stimulus list was played randomly.

A graphical user interface was developed in MATLAB platform by G. Nike. Gnanateja and Kiran (2015) (available at MATLAB file exchange) which is shown in Figure 3.2. This was displayed on a laptop screen in front of the participants and were instructed to click on the respective syllable using the computer mouse.

The responses were registered into an excel sheet, where the X-axis was the response of the client and Y axis was the stimulus presented, to form a 30*30 matrix at each SNR and in quiet condition.

Figure 3.2 Snap shot of graphical user interface used in the study.

3.4 Response Analysis

The participant's response for each stimulus presented was automatically saved by GUI on clicking a specific syllable displayed on the computer screen. Each correct response was scored 'one' and incorrect response was scored 'zero'. The responses obtained was further subjected to error analysis and sequential information transfer analysis (SINFA) to derive the place, manner and voiced features transformed.

3.5 Acoustic analysis

All 80 syllables were analyzed using praat software. The F0, F1, F2 and F3 formant frequencies were extracted by carefully selecting 40-50ms sample of the spectrogram which showed stable formants.

The individual data thus obtained was tabulated and the group data was statistically analyzed to derive the effect of hearing aid processing, effect of group and effect of SNR on consonant identification score.

Chapter 4: Results

The primary aim of the current study was to compare the identification scores of consonants processed through a hearing aid with that of the unprocessed consonants, in different signal to noise ratios. The secondary aim was to investigate place, manner and voicing feature transmitted through the hearing aid in different signal to noise ratios. The results in the present study are reported under following broad headings.

1. Results of speech identification scores
2. Results of Sequential Information Transfer Analysis (SINFA)
3. Results of acoustic analysis

4.1 Results of Speech Identification Scores

The consonant identification scores obtained for the two stimulus conditions (hearing aid processed & unprocessed) at 4 SNRs in normal and SNHL group were subjected to suitable statistical analysis to test the hypothesis of the study.

The consonant recognition scores were subjected to Kolmogorov-Smirnov test of normality and results showed that the data significantly deviated from the normal distribution with p value greater than 0.05 in all the conditions. Hence, it was chosen to use non parametric test for the statistical analysis in the present study.

Mann-Whitney U test, Friedman test and Wilcoxon sign rank tests were used in the current study. Mann-Whitney U test was used to compare consonant identification scores between the two groups, Friedman test was used for within-group comparison of consonant identification scores across SNRs (0, 5, 10dBSNR & quiet), 10 consonants. In instances where Friedman test showed significant main effect of variable, Wilcoxon sign rank test was used for pair-wise comparison. Wilcoxon sign rank was also used to compare consonant identification scores between two stimulus conditions (hearing aid processed and unprocessed). The results of the current study are reported under the following headings.

1. Effect of Stimulus Condition
2. Effect of SNRs
3. Effect of group

4.1.1 Effect of Stimulus Condition on Consonant Identification Scores

Mean and standard deviation of consonant identification scores of normal and SNHL group in 2 stimulus conditions (HP & UP), in 4 SNRs (0dB, 5dB, 10dBSNR & quiet) are given in Table 4.1. An observation of Table 4.1 reveals that the mean identification scores for two stimulus conditions (UP and HP) were different for both normal and SNHL groups. The mean scores were higher for UP speech compared to HP speech in all SNRs and in quiet. To investigate whether these mean differences were statistically significant between two conditions, Wilcoxon sign rank test was used. The two stimulus conditions were compared separately in each group and in each SNR. The test results of Wilcoxon sign rank test are given in Table 4.1 and Table 4.2.

Wilcoxon sign rank test for combined data revealed no significant difference between the two stimulus conditions in any of the SNR, in both normal and SNHL groups.

Table 4.1 *Mean and standard deviation of consonant identification scores of normal and SNHL group in 2 stimulus conditions (HP & UP), in 4 SNRs (0dB, 5dB, 10dBSNR & quiet)*

SNR	**Condition**	**Group**	**M (raw -scores)**	**SD**
Quiet	UP	Normal	2.78	0.66
		SNHL	2.25	1.14
	HP	Normal	2.75	0.67
		SNHL	2.31	1.05
10dB	UP	Normal	2.81	0.66
		SNHL	2.11	1.22
	HP	Normal	2.74	0.66
		SNHL	2.24	1.12
5dB	UP	Normal	2.58	0.89
		SNHL	2.04	1.28
	HP	Normal	2.66	0.78
		SNHL	2.07	1.17
0dB	UP	Normal	2.58	0.89
		SNHL	2.06	1.25
	HP	Normal	2.50	0.91
		SNHL	1.99	1.25

Note :Maximum individual score is 3

Table 4.2: *Results of Wilcoxon sign rank test comparing between the two stimulus condition (UP & HP), in the four different SNRs, in the 2 groups. The table shows the results for the pooled data of all the syllables*

Group	Quiet		10dBSNR		5dBSNR		0dBSNR	
	z	p	z	p	z	P	Z	p
NL	-0.567	0.571	-1.205	0.228	-1.052	0.293	-0.987	0.324
HL	-0.466	0.641	-1.134	0.228	-0131	0.896	-0.496	0.62

Table 4.3: *Results of Wilcoxon sign rank test comparing between the two stimulus condition (UP & HP), in the four different SNRs in the 2 groups. The table shows the phoneme –wise results*

Consonant	Group	Quiet		10dBSNR		5dBSNR		0dBSNR	
		Z	p	Z	p	Z	p	Z	p
/pa/	NL	2.156*	0.031	1.089	0.276	1.444	0.149	1.196	0.232
	HL	1.342	0.180	0.000	1.000	0.000	1.000	0.736	0.461
/ta/	NL	1.289	0.039	2.060	0.084	1.730	0.084	2.326*	0.020
	HL	0.184	0.854	1.342	0.180	1.414	0.157	1.342	0.180
/ka/	NL	1.857	0.063	2.070*	0.038	2.414*	0.016	3.219*	0.001
	HL	1.807	0.071	2.549*	0.011	1.823	0.068	1.983*	0.047
/ba/	NL	1.344	0.180	1.732	0.083	0.175	0.861	2.858*	0.004
	HL	0.577	0.564	1.730	0.084	0.791	0.429	1.725	0.084
/da/	NL	0.816	0.414	1.633	0.102	1.225	0.221	1.473	0.141
	HL	0.175	0.564	2.558	0.317	2.414	0.655	2.280	0.317
/ga/	NL	1.000	0.317	0.000	1.000	2.511*	0.012	2.066*	0.039
	HL	0.000	1.000	0.966	0.334	1.890	0.059	1.604	0.109
/sa/	NL	0.378	0.705	0.378	0.705	2.277*	0.023	2.251*	0.024
	HL	0.707	0.480	0.000	1.000	0.000	1.000	1.069	0.285
/ʃa/	NL	1.000	0.317	0.000	1.000	1.000	0.317	1.000	0.317
	HL	1.000	0.317	1.000	0.317	1.414	0.157	1.732	0.083
/ʧa/	NL	0.000	1.000	1.732	0.083	1.000	0.317	0.000	1.000
	HL	0.577	0.564	1.000	0.317	0.447	0.655	1.000	0.317
/ʤa/	NL	1.807*	0.011	1.807	0.071	2.850*	0.004	2.850*	0.004
	HL	1.633	1.000	1.633	0.334	1.633	0.059	1.841	0.109

*Note: *indicates p<0.05*

Inspection of the results in Table 4.3 reveals that, in 15 instancess out of 80 comparisions, there was a significant difference between unprocessed and processed consonants in terms of their identification. The probality of finding a significant difference between the 2 conditions was more at 0dBSNR followed by 5dBSNR and then 10dBSNR. Most of the times (13 out of 15), the significance of difference was observed in the normal group. All the three classes of sounds (stop consonant, fricatives &

affricates), did show significant decrease in identification of the hearing aid processed speech compared to unprocessed speech.

4.1.2 Effect of SNR on Consonant Identification

To derive the effect of SNR on consonant identification, identification scores across the 4 SNRs were compared. This was done for normal and SNHL group independently for hearing aid processed and unprocessed consonants. Mean and standard deviation of consonant identification scores in the 4 SNRs, in the 2 groups, in 2 stimulus conditions are given in Table 4.1. In the table one can observe that both normal and hearing loss group showed difference in mean identification scores across the 4 SNRs. In both the groups, a general trend was that the mean identification scores decreased with increasing noise level. Similarly, in both hearing aid processed and unprocessed speech, speech identification scores were lower at lower SNRs and vice-versa.

The statistical significance of observed mean differences was tested using Friedman’s test and the results are given in Table 4.4. Results of the Friedman’s test on the pooled data (combined data of 10 syllables) showed a significance main effect of SNR on the speech identification scores for both unprocessed and hearing aid processed speech in the normal group. However in the SNHL group the main effect of SNR was present only for the data of hearing aid proccessd speech.

Subsequent to the observed main effect, pair-wise comparision across the 4 SNRs was done using Wilcoxon sign rank test. The results of Wilcoxon test (Table 4.5), showed that, in the normal group there was significant difference across all the pairs (6 pairs), in both unprocessed and hearing aid processed speech. The exception for this was the comparison between 5 dBSNR and 0dBSNR in unprocessed speech. On the other hand, in SNHL group, significant differences were found only in the hearing aid processed speech. In this, quiet condition differed significantly from 0dB and 5dB and, 10 dBSNR differed significantly from 0dB.

Table 4.4: *Results of Friedman test, comparing consonant identification scores across SNRs in the two groups and in the 2 stimuli conditions*

Group	Stimulus	Chi-square	p
Normal	UP	41.837	0.000
	HP	24.637	0.000
SNHL	UP	3.188	0.364
	HP	11.483	0.009

Table 4.5: *Results of Wilcoxon sign rank test for combined data used for pair-wise comparisons across SNR for UP and HP.*

Group	Stimulus condition	Q-SNR10	Q-SNR5	Q-SNR0	SNR10-SNR5	SNR10-SNR0	SNR5-SNR0
Normal	UP	0.618*	3.492*	3.492*	3.932*	3.932*	0.000
	HP	0.124*	1.781*	3.985*	3.985*	4.101*	3.110*
SNHL	UP	1.096	1.551	1.444	0.592	0.407	0.707
	HP	0.690	2.195	2.543	1.723	2.147*	0.622

Note:* indicates significance of less than 0.05

Furthermore, it was of interest to study the effect of SNR in the syllable-wise data. Friedmann test was used for this purpose. Result of Friedman's test for syllable-wise data (Table 4.6) showed that there is significant main effect of SNR on the identification scores of stop consonants both in unprocessed and hearing aid processed speech. Identification of affricates and fricatives did not differ significantly across SNR in both the groups for unprocessed as well as hearing aid processed consonants. The effect of SNR was seen in more instances of hearing aid processed speech compared to unprocessed speech.

Table 4.6**:** *Friedman test results for UP and HP for normal and hearing loss group across SNRs.*

Group	Stimulus	Chi-Square									
		/pa/	/ta/	/ka/	/ba/	/da/	/ga/	/sa/	/ʃa/	/ʧa/	/ʤa/
Normal	UP	1.28	7.72	3.00	2.15	2.41	22.38*	2.64	3.00	3.00	6.80
	HP	4.05*	11.90	13.35*	12.68*	0.52	8.60*	6.35	7.20*	6.00	6.75
SNHL	UP	3.35	1.286	5.36	6.14	16.52*	1.43	0.66	2.00	1.28	1.73
	HP	0.69	10.76*	0.14	10.85*	1.80	7.57	4.28	3.70	3.00	4.27

*Note: * indicates statistical significance of less than 0.05*

In instances where Friedman test showed statistically significant main effect of SNR, a subsequent Wilcoxon sign rank test was done. The results of Wilcoxon sign rank test is given in Table 4.7. Results showed that 0dBSNR significantly differed from quiet and 10dBSNR in both unprocessed and hearing aid processed speech in the two groups. This was the major trend observed. There were also instances wherein quiet condition differed from 10dBSNR (/da/ of UP & /ta/ of HP in the HL group), 10 dBSNR differed from 5 dBSNR (/ga/ of UP in normal, /da/ of UP & /ta/ of HP in SNHL group), and 5dBSNR differed from 0dBSNR (/ba/ of HP in normal and /ba/ of UP in SNHL group).

Table 4.7: *Results of Wilcoxon sign rank test showing pair-wise comparison across SNRs in the syllable-wise data*

Group	Stimulus	Consonant	Z					
			Q-SNR10	Q-SNR5	Q-SNR0	SNR10-SNR5	SNR10-SNR0	SNR5-SNR0
NL	UP	/ga/	1.000	2.511*	2.511*	2.511*	2.724*	0.000
	HP	/ta/	0.541	1.513	2.565*	1.200	2.040*	1.508
		/ka/	0.264	1.725	2.832*	1.552	2.719*	1.925
		/ba/	0.707	0.000	2.901*	0.816	2.631*	2.842*
		/ga/	0.000	1.000	2.00*	1.000	2.000*	1.342
HL	UP	/da/	2.165	2.588*	2.456	1.633*	1.473	1.000
	HP	/ba/	0.750	0.520	2.549*	0.000	2.136	2.266
		/ta/	2.121	1.000	0.447	2.251*	1.070*	0.816

*Note: * indicates p<0.005*

4.1.3 Effect of Group on the Consonant Identification

Mean and standard deviation of consonant identification scores of normal and SNHL groups in 2 stimulus conditions (HP & UP), in 4 SNRs (0dB, 5dB, 10dBSNR & quiet) can be referred from Table 4.1. Comparison of the mean scores between the two groups showed difference in mean consonant identification scores between normal and SNHL group, in both the conditions and in all 4 SNRs. The normal hearing group obtained higher mean consonant identification scores compared to SNHL group, for both hearing aid processed and unprocessed consonants, in all the 4 SNRs. Statistical signifcance of this mean difference was tested using Mann-Whitney U test. The results of the Mann-Whitney U test are given in Table 4.8.

Table 4.8: *Mann-Whitney U test results comparing between normal and SNHL groups, in the unprocessed and hearing aid processed speech. This result is for the data combined across the syllables*

SNR	Condition	Z	p
Quiet	UP	-4.897	0.000
	HP	-4.346	0.000
10dB	UP	-6.192	0.000
	HP	-4.777	0.000
5dB	UP	-3.766	0.000
	HP	-4.315	0.000
0dB	UP	-3.83	0.000
	HP	-3.55	0.000

Note: maximum individual score is 3

Results of Mann-Whitney U test revealed that the mean consonant identification scores were significantly different between normal and SNHL groups in all the SNRs and in both the stimulus conditions (UP and HP). Because there was a significant difference between the two groups in the combined data (identification scores of all the syllables combined), it was of interest to test the significance of difference between the two groups in each of the target consonants separately. Results of the Mann-Whitney U test for the syllable-wise data are given in Table 4.9.

Table 4.9: *Mann-Whitney U test results comparing between the two groups, separately in 4 SNRs, in Hearing aid processed and Unprocessed Speech stimuli groups. This is for the syllable-wise data*

Syllable	Condition	Z			
		Quiet	10dBSNR	5dBSNR	0dBSNR
/pa/	UP	-3.182*	-1.899	-1.391	-1.391
	HP	-0.41	-1.788	0.000	-0.082
/ta/	UP	-2.081	-1.936	-1.047	-0.991
	HP	-1.659	-0.539	-1.058	0.000
/ka/	UP	-2.979*	-2.034*	-1.936	-1.973*
	HP	-3.100*	-2.94*	-1.336	-0.776
/ba/	UP	-1.044	-3.487*	-1.798	-2.242*
	HP	-0.695	-0.719	-1.118	-2.031*
/da/	UP	-2.465*	-3.413*	-4.439*	-4.331*
	HP	-2.962*	-2.08*	-3.06*	-2.146*
/ga/	UP	-1.31	-2.535*	-0.049	-0.049
	HP	-2.987*	-2.981*	-3.447*	-2.98*
/sa/	UP	-1.544	-2.179*	-0.783	-0.783
	HP	-1.281*	-2.158*	-2.103*	-2.511*
/ʃa/	UP	-1.414	-1.414	-0.707	-0.707
	HP	-0.707	-2.035*	-0.543	-2.539*
/ʧa/	UP	-2.035*	-1.414	-0.559	-0.509
	HP	-1.414	-1.269	-1.414	-0.707
/ʤa/	UP	-0.716	-0.765	-0.095	-0.095
	HP	0	-0.952	-2.535*	-0.407

*Note : * Significant difference at p<0.05. Negative sign indicates greater mean scores in the normal group*

Results of Mann-Whiney test on syllable-wise data showed that there is significant difference between the groups in all the consonants except for /ta/. Voiced stop consonants were affected more than the unvoiced stop consonats, and within the voiced stop consonants /da/ was affected maximally. Identification scores of voiced stop consonants were significantly different between the 2 groups in all the 3 places of articulation. But only the velar place of articulation showed group difference among the unvoiced consonants.

4.2 Results of Acoustic Analysis

The results of the acoustic analysis of both hearing aid processed and unprocessed speech signals are reported in the Table 4.10. The results of the acoustic analysis revealed that there were differences in the F0 and formant frequencies between unprocessed and hearing aid processed speech. To treat as a considerable difference an operational criteria of 10 Hz and 50Hz for F0 and formants respectively was fixed. That is, any shift in the F0 in either upward or downward direction, of more than 10 Hz was treated as a considerable shift. Similarly any such sift in formants of more than 50 Hz was treated as considerable. Results showed there was no considerable shift of fundamental frequency (F0) between processed and unprocessed speech stimuli across all the SNRs. The change in F1 frequency between, hearing aid processed and unprocessed speech stimuli was evident in all SNRs. Both decrease and increase in formant frequencies were almost equally seen across consonants. The F1 frequency of the consonant /da/ showed a consistent finding in which, the formant frequencies decreased across all the SNRs considered in the study. At 0dBSNR the trend was more of increasing in all the formant frequencies except for F1 of /da/ and F2 of /pa/ which showed decrease in formant frequeny. The F2 and F3 frequencies showed a similar trend except for consonant /ʤa/, in which there was a relative increase in frequency in the hearing aid processed speech in all the SNRs. For consonant /ʤa/ the trend was reversed where both F2 and F3 frequencies decreased in

Table 4.10: *Formant frequencies of hearing aid processed and unprocessed speech stimuli. The Color shading depicts the relative increase or decrease in F0 and formats in hearing aid processed speech compared to unprocessed speech*

SNR	Consonant	F0(Up)	F0(Hp)	F1(Up)	F1(Hp)	F2(Up)	F2(Hp)	F3(Up)	F3(Hp)
Quiet	/ba/	112	111	769	774	2268	2545	2519	3175
	/da/	112	112	943	769	1597	2000	2375	2979
	/ga/	114	114	697	801	2131	2346	2413	2623
	/pa/	114	113	799	764	1900	2017	2469	2586
	/ta/	113	111	800	764	1849	2003	2559	2866
	/ka/	115	113	863	854	2293	2219	2393	3174
	/sa/	113	115	764	768	1575	1812	2554	2649
	/ʃa/	110	110	771	773	1798	1959	2588	3065
	/ʧa/	113	113	811	735	1594	1970	2514	2964
	/ʤa/	110	111	758	891	1899	1475	2562	2362
10dB	/ba/	112	111	806	763	2318	2543	2513	3107
	/da/	112	112	924	909	1586	2042	2373	2894
	/ga/	114	115	677	774	2156	2371	2412	2604
	/pa/	113	112	780	870	1934	1854	2471	2640
	/ta/	112	110	811	791	1885	1973	2546	2942
	/ka/	115	113	843	862	2312	2205	2373	3176
	/sa/	114	116	765	787	1562	1820	2559	2631
	/ʃa/	110	110	774	842	1898	2030	2444	2953
	/ʧa/	114	113	861	718	1582	1979	2502	2910
	/ʤa/	112	109	707	980	1957	1983	2550	2789
5dB	/ba/	112	111	763	857	2411	2515	2506	2925
	/da/	112	112	992	856	1648	2064	2575	2912
	/ga/	114	114	722	724	2088	2328	2420	2708
	/pa/	113	114	745	661	1931	2321	2470	2808
	/ta/	112	112	828	735	1926	2020	2526	2809
	/ka/	115	113	806	845	2323	2047	2350	3192
	/sa/	112	113	790	795	1507	1949	2546	2650
	/ʃa/	110	110	737	781	1897	1998	2473	2965
	/ʧa/	113	113	831	722	1582	1979	2513	2891
	/ʤa/	112	109	751	980	1964	1120	2546	2091
0dB	/ba/	112	111	704	796	2182	2537	2514	3002
	/da/	112	112	977	851	1763	2022	2699	2930
	/ga/	114	114	713	816	2142	2362	2407	2595
	/pa/	113	113	732	853	2012	1779	2475	2592
	/ta/	113	111	814	843	1845	1918	2569	3111
	/ka/	115	115	770	749	2273	2232	2356	3076
	/sa/	115	116	753	788	1598	1827	2562	2632
	/ʃa/	110	110	835	842	1672	2027	2447	2970
	/ʧa/	112	117	704	928	1556	2043	2532	2909
	/ʤa/	111	108	748	900	1785	1929	2551	2735

Note: Green indicates relative increase in formants of Hearing aid processed speech (HP) compared to that of unprocessed speech(UP) and yellow indicatesrelative decrease in formants of Hearing aid processed speech compared to that of unprocessed speech.`

quiet. Also F2 frequencies of /ka/ decreased across all the four conditions considered in the study.

4.3 Results of Sequential Information Transfer Analysis (SINFA)

SINFA was carried out on the group data of consonant confusion matrices in each condition. It was aimed to derive manner, place and voicing features from SINFA. The results of SINFA are given in terms of bits of information transmitted. The variance in the data attributable to each feature was calculated by keeping the variance of the other features constant, and then this process was iterated till addition of new features did not change the variance of the whole data. The information transmitted by each of the features are calculated using this procedure and the proportion of information transmitted to the proportion of information per feature actually present in the stimuli were calculated. This was termed as conditional information (TI) transferred. The total information transmitted by pooling across information transmitted by all the features was termed the Total Transmitted Information (TTI). The results of SINFA across SNRs for both SNHL and normal hearing groups for two stimulus conditions (UP & HP) are given in Table 4.11.

4.3.1 Results of SINFA in Normal Hearing Group

In the normal hearing group, the total information transmitted (TTI) and the condition information transmitted per feature (TTI) varied across SNRs in both the groups. The TTI in the normal hearing group was higher for the unprocessed syllables than the hearing aid processed syllables in quiet and in 0dBSNR conditions. However, the hearing aid processed syllables had higher TTI than the unprocessed syllables in the

5dBSNR and 10 dBSNR conditions. In the SNHL group, TTI was higher for the hearing aid processed syllables than the unprocessed syllables in the Quiet and 10dBSNR condition.

Table 4.11: *Results of Sequential Information Transfer Analysis*

SNR	Group	Stimulus	TI (Bits)			TTI(Bits)
			Manner	Place	Voicing	
Quiet	NL	UP	1.000	0.934	0.909	3.162
		HP	0.990	0.879	0.725	3.090
	HL	UP	0.793	0.584	0.503	2.241
		HP	0.899	0.579	0.483	2.373
10dBSNR	NL	UP	0.978	0.887	0.758	2.969
		HP	0.988	0.922	0.796	3.053
	HL	UP	0.906	0.526	0.448	2.278
		HP	0.765	0.573	0.596	2.247
5dBSNR	NL	UP	0.990	0.789	0.549	2.715
		HP	0.990	0.935	0.655	2.946
	HL	UP	0.793	0.531	0.543	2.178
		HP	0.615	0.574	0.416	2.005
0dBSNR	NL	UP	0.990	0.767	0.555	2.696
		HP	0.966	0.739	0.444	2.575
	HL	UP	0.791	0.535	0.508	2.184
		HP	0.693	0.519	0.389	1.995

The detailed SINFA results and consonant confusion matrices are given in Appendix 1.

The conditional information transmitted per feature (TI) was higher for the unprocessed syllables in quiet and 0dBSNR conditions. However, at 10dBSNR and 5dBSNR conditions TI was higher in the hearing aid processed syllables compared to the unprocessed conditions for all features except manner feature at 5dBSNR, which was equal in both the conditions. The information transmitted by the manner feature was highest followed by place and voicing in all the SNR conditions for the hearing aid processed syllables as well as the unprocessed syllables.

4.3.2 Results of SINFA in SNHL Group

In the SNHL group, TTI was greater for the hearing aid processed syllables than the unprocessed syllables in quiet and 10dBSNR conditions. At 0dBSNR and 5dBSNR however, the unprocessed syllables had greater TTI than the hearing aid processed syllables.

In quiet, the TI for the place and voicing features were higher for the hearing aid processed syllables than the unprocessed syllables while TI for manner was vice versa. At 10dBSNR, TI of all features were higher for the hearing aid processed syllables than the unprocessed syllables. At 0dBSNR and 5dBSNR however, TI were higher in the unprocessed syllables than the hearing aid processed syllables except TI for place at 5dBSNR which was greater for the hearing aid processed syllables than the unprocessed syllables. The information transmitted by the manner feature was highest followed by place and voicing in all the SNR conditions for the hearing aid processed syllables as well as the unprocessed syllables.

4.3.3 Comparison of Results of SINFA between the Normal Hearing and SNHL Groups

Across all SNRs, normal hearing group obtained higher TTI for both conditions compared to hearing loss groups. Also TTI for both normal hearing and hearing loss group was higher for unprocessed speech stimulus compared to that of processed speech stimulus. For both normal and hearing loss group, manner was least affected followed by place and voicing for both UP and HP speech stimulus across all SNRs and in quiet condition. The results of transmitted information for each feature reveals that, in normal and hearing loss group, TI for manner was similar across SNRs for both UP and HP speech stimuli, except for 10dBSNR of UP speech stimuli which had a better TI compared to that of quiet condition in hearing loss group. TI for place and manner feature reveals that, in normal hearing group the TI decreased with increasing noise level for both UP and HP condition and TI was greater for UP speech compared to HP speech, across all SNRs. but in hearing impaired group, TI for both place and manner features were higher for HP compared to that of UP at 5 and 10 dBSNR.

Chapter 5: Discussion

In a normal auditory system precise processing of spectral and temporal features of the speech signal ensures accurate perception of speech. However when these processes are altered due to pathologies in the auditory system, it is likely to deflect interms of speech perception. In ears with sensorineural hearing loss, both temporal processing and spectral processing would be affected (Moore, 2008) leading to poorer speech perception.

The most common rehabilitation option for individuals with sensorineural hearing loss (SNHL) is the use of hearing aids. Hearing aids amplify speech energy and thereby improve the aided hearing thresholds in individuals with sensorineural hearing loss. Apart from amplifying the sound, hearing aid processing also results in certain inadvertent changes in the acoustic characteristics of input speech due to micropone, filtering and compression affects which are inevitable components of a hearing aid. The alteration in the speech acoustics due to hearing aid processing can affect the relevant cue important for speech perception.

With the above background, in the present study it was of interest to probe into the effect of hearing aid processing on consonant acoustics and its eventual perception. Results of the present study showed that there was an effect of hearing aid processing, the effect of sensorineural hearing loss and effect of signal to noise ratios on the identification of te consonants. The findings and the reasons for them are discussed in detail under the following headings.

1. Hearing aid induced changes in the acoustics of speech
2. Perceptual changes due to hearing aid processing

3. Perceptual difference normal hearing and SNHL group

5.1. Hearing Aid Induced Changes in the Acoustics of Speech

The primary aim of the present study was to determine whether the basic processing of the hearing aid influences the acoustics of speech (specifically consonants) and eventually its perception. In the present study it was found that hearing aid processing evidently changes the formant frequencies. Most of the time hearing aid processing increased the F1, F2 and F3. In this study F1, F2 and F3 were derived from the steady state. However, one can speculate that a similar change would be present in the formant transitions. Formant transitions are the primary cues for the perception of place of articulation (Fujimura, Macchi, & Streeter, 1978; Kewley-Port, Pisoni, & Studdert-Kennedy, 1983; Stevens & Blumstein, 1978), among which F1 helps in distinguishing voiced from unvoiced consonants (Jiang, Chen, & Alwan, 2006; Kluender, 1991; Lisker, 1975; Stevens & Klatt, 1974), while F2 and F3 serve to demarcate different places of articulation (Kewley-Port, 1982; Sussman, McCaffrey, & Matthews, 1991). The increase in the formant frequencies observed in the present study were likely to result in altering the perception of place, and voicing features of consonants under study.

Alterations in the acoustics of syllables, secondary to primary hearing aid processing are also reported in couple of the earlier studies. Garvita and Maruthy (2010) had recorded the effect of hearing aid processing syllable /da/ and had found differences in VOT, following vowel duration and total syllable duration. Prabhash (2011) reported the effect of hearing aid processing on spectral and temporal characteristics of speech, and also how these changes affected the brainstem responses. It was found that the hearing aid processing had a significant effect on spectral characteristics where the formant frequencies were different between processed and unprocessed speech stimuli. Both these earlier studies had concentrated only on one consonant. Whereas, in the present study changes due to hearing aid processing is documented for 10 consonants covering voiced and unvoiced counterparts of stop consonants of different places of articulation, fricatives and affricates. Irrespective of the place of articulation, manner of articulation and voicing feature, formants increased secondary to hearing aid processing in majority of cases. However there were few instances where F1 decreased in frequency by more than 50 Hz as in /da/, /ʃa/, /ʧa/ in quiet, /pa/, /ta/ and /ʧa/ in 5 dBSNR and /ʧa/ in 0dBSNR.

These observations of change in acoustics was present at quiet as well as in different signal to noise ratios. In the presence of noise the primary consonantal cues get masked and the perception depends on the co-articulation cues such as formant transition and vowels (Revoile, Pickett, Holden, & Talkin, 1982). Therefore, one can expect tthat the increase or decrease int the formant frequencies induced by the hearing aid shall alter the perception of consonants more in the presence of noise. Furthermore, individuals with cochlear hearing loss were likely to have poorer perception compared to normal individuals as hearing aids alters the external redundancy by changing the formants.

5.2. Perceptual Changes due to Hearing Aid Processing

In disagreement to what was expected from the changes in acoustics of speech secondry to hearing aid processing, the combined data (scores of all the test stimulus combined together), showed that there was no change in the perception of any of the consonants even after processing through the hearing aid. Specifically this meant that neither the perception of place of articulation, manner of articulation nor the voicing is affected due to hearing aid processing. This was true even in the degraded stimulus conditions.

However, it was possible that the effect was present in some of the consonants which should have got nullified in the combined data. Therefore, the effect of hearing aid processing was analysed in each consonant seperately. Results showed that in all the consonants except for /da/, / ʃa/ and /ʧa/ perception decreased when processed through the hearing aid, although not at all SNRs. As speculated the perception deterioration was more in the degraded conditions and was directly related to the degree of degradation.

An important unexpected finding was that the effect of stimulus condition was seen more in the normal group compared to SNHL group. In the previous section of discussion, it was speculated that individuals with cochear hearing loss would have greater perceptual deterioration compared to normal hearing individuals. The present unexpected finding could be justified through close inspection of the mean data. The mean data showed that individuals with cochlear hearing loss had poor speech perception scores leading to floor effect and therefore, no statistical differences were seen between the stimulus conditions in the SNHL group. On the other hand, normals had high scores in the unprocessed condition, and any subtle differences in the perception caused by the hearing aid processing were observable.

Results of SINFA complimented the findings from the combined data. The Trnsmision Index (TI) of the hearing aid processed speech was comparable to that in unprocessed speech and in fact, in 5 and 10 dB SNR TI was higher for hearing aid processed speech.

5.3. Perceptual Difference between Normal Hearing and SNHL group

Results of the present study showed that there was poorer perception of consonants in SNHL group compared to normal hearing group. This was true in each of the signal to noise ratio both in unprocessed and processed speech consonants. The difference between the groups was present in all the consonants except for /ta/. Voiced stop consonants were affected more than the unvoiced stop consonats, and within the voiced stop consonants, /da/ was affected maximally. The poor perception in the SNHL group could be attributed to poor frequency selectivity (Glasberg & Moore, 1990; Moore, 1985, 2007), spectral resolution (Kluk & Moore, 2005) and temporal resolution (Hopkins & Moore, 2009). The greater effect in stop consonats and the maximum effect in /da/ could be attributed to their dynamic characterstics. The one that is more dynamic is more likely to get affected in the presence of deficient spectral and temporal processing. Results of SINFA complemented with the speculation that SNHL group would have poorer perception compared to normal group. Based on the observation in the acoustics of speech, it was speculated that in SNHL group will have poorer perception compared to normal group and this would increase with greater degrees of stimulus degradation. However, results did not support this speculation. The group effect was similar across all the 4dBSNR conditions used in the study.

The present finding of poorer perception of stop sconsonats compared to normal group is in agreement with the studies in the literature.

Chapter 6: Summary and Conclusions

The primary aim of the current study was to compare the identification scores of nonsense consonants processed through the hearing aid with that of the unprocessed consonants, in different signal to noise ratios. The secondary aim was to investigate place, manner and voicing feature transmitted through the hearing aid in different signal to noise ratios.

A total of 30 adults participated in the study. All the participants were native speakers of Kannada and were in the age range of 20 to 50 years and were geographically from in and around Mysore. Based on their hearing sensitivity, they were divided into two groups; Normal and SNHL groups. The SNHL group consisted of 10 participants, diagnosed as having moderate to moderately severe SNHL with flat audiogram configuration. Normal hearing group consisted of 20 individuals who were age and gender matched with SNHL group. Participants of this group had their air conduction and bone conduction thresholds within 15 dBHL with an air bone gap of less than 15 dB across 250 Hz to 8 kHz.

A total of 10 syllables were used in the present study. Consonant in these syllables was one of the /p/,/t/,/k/,/b/,/d/,/g/,/s/,/ʃ/,/ʤ/,/ ʧ/, and were always combined with vowel /a/. The aim while choosing the consonant was to cover voiced and unvoiced counter parts of stop consonants, fricatives and affricates of different places of articulation. All ten syllables /pa/, /ta/, /ka/, / ba/, /da/, /ga/, /sa/, /ʃa/, /ʧa/ and /ʤa/ were processed through Hansaton salto super power, a 4 channel digital wide dynamic range compression hearing aid. Hearing aid processed and unprocessed consonants were presented to the participants and their responses were recorded in an excel sheet using a graphical user interface developed on MATLAB. The data obtained was subjected to Non-parametric statical analysis due to lack of normality and sphericity in the data. Mann-Whitney U test was used for between-group comparisons while Friedman followed by Wilcoxon sign rank test was used for within group comparisions.

The results of the study showed that hearing aid processing altered the acoustics of the speech signal, leading to reduction in speech perception. This was also true across 4 SNRs considered in the study. Main effect of group was obsereved where normal hearing group had better consonant identification scores compared to SNHL group for hearing aid processed as well as unprocessed consonants across SNRs. This poor perception in the SNHL group could be attributed to poor frequency selectivity, compressive non-linearity and temporal resolution in SNHL. Contradictory to what was expected from the changes in acoustics of speech secondary to hearing aid processing, the combined data (scores of all the test stimulus combined together), showed that there was no change in the perception of any of the consonants even after processing through the hearing aid. Specifically this meant that neither the perception of place of articulation, manner of articulation nor the voicing is affected due to hearing aid processing. This was true even in the degraded stimulus conditions. However, it was possible that the effect was present in some of the consonants which should have got nullified in the combined data. Therefore, the effect of hearing aid processing was analysed in each consonant seperately. Results showed that in all the consonants except for /da/, / ʃa/ and /ʧa/ perception decreased when processed through the hearing aid, although not at all SNRs.

As speculated, the perception deterioration was more in the degraded conditions and was directly related to the degree of degradation.

Implications of the Study

Based on the findings of the study, there is a need to develop speech sound specific processing statergies to improve perception in the presence of noise.

The findings of the study emphazise the need for speech enhancement statergies in hearing aids as the primary hearing aid processing was not sufficient to improve speech perception in quiet as well as in noise.

The results of the current study can be used to explain realistic expectation from hearing aids by patients with cochlear hearing loss.

Chapter 7: References

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Appendix A:Report of Sequential Information Feature Analysis

Reports of SINFA in Normals

Unprocessed consonants

Effective Feature Matrix(Unprocessed quiet)
========================

	b	c	d	g	j	k	p	S	s	t
voicing	+	-	+	+	+	-	-	-	-	-
place	bil	pal	alv	vel	pal	vel	bil	pal	alv	alv
manner	plo	aff	plo	plo	aff	plo	plo	fri	fri	plo

Features held constant in order of maximum proportion info. transmitted.

Contents of stimulus response matrix
====================================

	b	c	d	g	j	k	p	S	s	t
b	60	0	0	0	0	0	0	0	0	0
c	0	60	0	0	12	0	0	0	0	0
d	0	0	60	0	0	0	0	0	0	0
g	0	0	0	60	0	0	0	0	0	0
j	0	0	0	0	48	0	0	0	0	0
k	0	0	0	0	0	60	0	0	0	0
p	0	0	0	0	0	0	60	0	0	0
S	0	0	0	0	0	0	0	60	13	0
s	0	0	0	0	0	0	0	0	47	0
t	0	0	0	0	0	0	0	0	0	60

Constituent SR matrices : addnUQ.txt

TRANSMITTED INFORMATION = 3.162
Proportion of correct responses = 0.958

Iteration 1 synopsis

FEATURE	INPUT	TRANS	%TRANS	TRANS/TI	%CORRECT
voicing	0.958	0.843	0.880	0.267	0.980
place	1.969	1.856	0.943	0.587	0.978
manner	1.371	1.371	1.000	0.434	1.000

Iteration 2 synopsis

Constant: manner

FEATURE	INPUT	TRANS	%TRANS	TRANS/TI
voicing	0.794	0.722	0.909	0.228
place	1.144	1.069	0.934	0.338

Iteration 3 synopsis

Constant: manner place

FEATURE	INPUT	TRANS	%TRANS	TRANS/TI
voicing	0.794	0.722	0.909	0.228

SINFA SUMMARY

Transmitted information = 3.162

ITERA-TION	FEATURE	TRANS	TRANS/ INPUT	TRANS/ TI
1	manner	1.371	1.000	0.434
2	place	1.069	0.934	0.338
3	voicing	0.722	0.909	0.228
		3.162		1.000

Effective Feature Matrix (10dBSNR)

	b	c	d	g	j	k	p	S	s	t
voicing	+	-	+	+	+	-	-	-	-	-
place	bil	pal	alv	vel	pal	vel	bil	pal	alv	alv
manner	plo	aff	plo	plo	aff	plo	plo	fri	fri	plo

Features held constant in order of maximum proportion info. transmitted.

Contents of stimulus response matrix

	b	c	d	g	j	k	p	S	s	t
b	54	0	0	0	0	0	2	0	0	0
c	0	60	0	0	17	0	0	0	0	2
d	0	0	48	0	0	0	0	0	0	0
g	0	0	5	60	0	0	0	0	0	0
j	0	0	0	0	43	0	0	0	0	0
k	0	0	0	0	0	60	0	0	0	0
p	3	0	0	0	0	0	58	0	0	0
S	0	0	0	0	0	0	0	60	14	0
s	0	0	0	0	0	0	0	0	46	0
t	0	0	7	0	0	0	0	0	0	58

```
Constituent SR matrices : addnU10.txt

TRANSMITTED INFORMATION = 2.969
Proportion of correct responses = 0.916

Iteration  1 synopsis
---------------------

FEATURE  INPUT  TRANS %TRANS TRANS/TI %CORRECT
voicing  0.939  0.706  0.752  0.238   0.951
place    1.969  1.784  0.906  0.601   0.965
manner   1.379  1.349  0.978  0.455   0.997


Iteration  2 synopsis
---------------------
Constant:  manner
FEATURE  INPUT  TRANS %TRANS TRANS/TI
voicing  0.783  0.577  0.737  0.194
place    1.135  1.006  0.887  0.339


Iteration  3 synopsis
---------------------
Constant:  manner place
FEATURE  INPUT  TRANS %TRANS TRANS/TI
voicing  0.772  0.585  0.758  0.197


SINFA SUMMARY

Transmitted information = 2.969

ITERA-  FEATURE  TRANS  TRANS/  TRANS/
 TION                   INPUT   TI

  1     manner   1.349  0.978   0.455
  2     place    1.006  0.887   0.339
  3     voicing  0.585  0.758   0.197
                 2.941          0.991
```

Effective Feature Matrix (5dBSNR)
========================

```
             b   c   d   g   j   k   p   S   s   t
voicing      +   -   +   +   +   -   -   -   -   -
place       bil pal alv vel pal vel bil pal alv alv
manner      plo aff plo plo aff plo plo fri fri plo
```

Features held constant in order of maximum proportion info. transmitted.

Contents of stimulus response matrix
====================================

	b	c	d	g	j	k	p	S	s	t
b	52	0	0	2	0	0	2	0	0	0
c	0	59	0	0	24	0	0	0	1	0
d	0	0	52	1	0	0	0	0	0	12
g	1	0	8	46	0	1	2	0	0	0
j	0	1	0	0	36	0	0	0	0	0
k	0	0	0	11	0	59	1	0	0	0
p	7	0	0	0	0	0	55	0	0	0
S	0	0	0	0	0	0	0	59	14	0
s	0	0	0	0	0	0	0	1	45	0
t	0	0	0	0	0	0	0	0	0	48

Constituent SR matrices : addnU5.txt

TRANSMITTED INFORMATION = 2.715
Proportion of correct responses = 0.852

Iteration 1 synopsis

FEATURE	INPUT	TRANS	%TRANS	TRANS/TI	%CORRECT
voicing	0.943	0.503	0.534	0.185	0.900
place	1.973	1.676	0.850	0.617	0.948
manner	1.371	1.357	0.990	0.500	0.998

Iteration 2 synopsis

Constant: manner

FEATURE	INPUT	TRANS	%TRANS	TRANS/TI
voicing	0.778	0.393	0.505	0.145
place	1.141	0.899	0.789	0.331

Iteration 3 synopsis

Constant: manner place

FEATURE	INPUT	TRANS	%TRANS	TRANS/TI
voicing	0.750	0.412	0.549	0.152

SINFA SUMMARY

Transmitted information = 2.715

ITERA-TION	FEATURE	TRANS	TRANS/INPUT	TRANS/TI
1	manner	1.357	0.990	0.500
2	place	0.899	0.789	0.331
3	voicing	0.412	0.549	0.152
		2.668		0.983

Effective Feature Matrix (0dBSNR)
========================

	b	c	d	g	j	k	p	S	s	t
voicing	+	-	+	+	+	-	-	-	-	-
place	bil	pal	alv	vel	pal	vel	bil	pal	alv	alv
manner	plo	aff	plo	plo	aff	plo	plo	fri	fri	plo

Features held constant in order of maximum proportion info. transmitted.

Contents of stimulus response matrix
====================================

	b	c	d	g	j	k	p	S	s	t
b	53	0	0	3	0	0	2	0	0	0
c	0	58	0	0	23	0	0	0	1	0
d	0	0	51	1	0	0	0	0	0	10
g	1	0	9	47	0	1	2	0	0	0
j	0	2	0	0	37	0	0	0	0	0
k	0	0	0	9	0	53	1	0	0	0
p	6	0	0	0	0	0	49	0	0	0
S	0	0	0	0	0	0	0	53	14	0
s	0	0	0	0	0	0	0	1	39	0
t	0	0	0	0	0	0	0	0	0	44

Constituent SR matrices : addnU0.txt

TRANSMITTED INFORMATION = 2.696
Proportion of correct responses = 0.849

Iteration 1 synopsis

FEATURE	INPUT	TRANS	%TRANS	TRANS/TI	%CORRECT
voicing	0.961	0.526	0.547	0.195	0.904
place	1.971	1.648	0.836	0.611	0.942
manner	1.370	1.355	0.990	0.503	0.998

Iteration 2 synopsis

```
---------------------
Constant:  manner
FEATURE  INPUT  TRANS %TRANS TRANS/TI
voicing  0.790  0.410  0.519  0.152
place    1.128  0.865  0.767  0.321


Iteration  3 synopsis
---------------------
Constant:  manner place
FEATURE  INPUT  TRANS %TRANS TRANS/TI
voicing  0.762  0.422  0.555  0.157


SINFA SUMMARY

Transmitted information = 2.696

ITERA-  FEATURE  TRANS  TRANS/  TRANS/
 TION                   INPUT   TI

  1     manner   1.355  0.990   0.503
  2     place    0.865  0.767   0.321
  3     voicing  0.422  0.555   0.157
                 2.643          0.981
```

Hearing Aid Processed Speech

Effective Feature Matrix(quiet)
========================

	b	c	d	g	j	k	p	S	s	t
voicing	+	-	+	+	+	-	-	-	-	-
place	bil	pal	alv	vel	pal	vel	bil	pal	alv	alv
manner	plo	aff	plo	plo	aff	plo	plo	fri	fri	plo

Features held constant in order of maximum proportion info. transmitted.

Contents of stimulus response matrix
====================================

	b	c	d	g	j	k	p	S	s	t
b	54	0	0	0	0	0	11	0	0	0
c	0	60	0	0	0	0	0	1	0	0
d	0	0	59	0	0	0	0	0	0	8
g	2	0	1	60	0	8	0	0	0	0
j	0	0	0	0	60	0	0	0	0	0
k	0	0	0	0	0	52	0	0	0	0
p	4	0	0	0	0	0	47	0	0	0

```
S      0  0  0  0  0  0  0  59 13 0
s      0  0  0  0  0  0  0  0  47 0
t      0  0  0  0  0  0  2  0  0  52
```

Constituent SR matrices : addnHQ.txt

TRANSMITTED INFORMATION = 2.950
Proportion of correct responses = 0.917

```
Iteration  1 synopsis
---------------------

FEATURE  INPUT  TRANS %TRANS TRANS/TI %CORRECT
voicing  0.989  0.709  0.717  0.241   0.948
place    1.968  1.792  0.911  0.607   0.970
manner   1.371  1.357  0.990  0.460   0.998


Iteration  2 synopsis
---------------------
Constant:  manner
FEATURE  INPUT  TRANS %TRANS TRANS/TI
voicing  0.793  0.564  0.711  0.191
place    1.143  1.005  0.879  0.341


Iteration  3 synopsis
---------------------
Constant:  manner place
FEATURE  INPUT  TRANS %TRANS TRANS/TI
voicing  0.785  0.569  0.725  0.193


SINFA SUMMARY

Transmitted information = 2.950

ITERA-  FEATURE  TRANS  TRANS/  TRANS/
 TION                   INPUT   TI

  1     manner   1.357  0.990   0.460
  2     place    1.005  0.879   0.341
  3     voicing  0.569  0.725   0.193
                 2.930          0.993
```

Effective Feature Matrix (10dBSNR)
========================

```
             b   c   d   g   j   k   p   S   s   t
voicing      +   -   +   +   +   -   -   -   -   -
place       bil pal alv vel pal vel bil pal alv alv
```

```
manner     plo aff plo plo aff plo plo fri fri plo

Features held constant in order of maximum proportion info.
transmitted.

Contents of stimulus response matrix
====================================

      b  c  d  g  j  k  p  S  s  t
b     57 0  0  0  0  0  3  0  0  0
c     0  60 0  0  1  0  0  0  0  0
d     0  0  60 0  0  0  0  0  0  8
g     2  0  0  60 0  7  0  0  0  0
j     0  0  0  0  59 0  0  0  0  0
k     0  0  0  0  0  53 0  0  0  0
p     1  0  0  0  0  0  56 0  0  0
S     0  0  0  0  0  0  1  60 10 0
s     0  0  0  0  0  0  0  0  50 0
t     0  0  0  0  0  0  0  0  0  52

Constituent SR matrices : addnH10.txt

TRANSMITTED INFORMATION = 3.053
Proportion of correct responses = 0.945

Iteration  1 synopsis
---------------------

FEATURE  INPUT  TRANS %TRANS TRANS/TI %CORRECT
voicing  0.984  0.785  0.797  0.257   0.967
place    1.968  1.837  0.933  0.602   0.978
manner   1.374  1.357  0.988  0.445   0.998


Iteration  2 synopsis
---------------------
Constant:  manner
FEATURE  INPUT  TRANS %TRANS TRANS/TI
voicing  0.794  0.627  0.789  0.205
place    1.144  1.055  0.922  0.346


Iteration  3 synopsis
---------------------
Constant:  manner place
FEATURE  INPUT  TRANS %TRANS TRANS/TI
voicing  0.790  0.629  0.796  0.206


SINFA SUMMARY
```

Transmitted information = 3.053

ITERA-TION	FEATURE	TRANS	TRANS/ INPUT	TRANS/ TI
1	manner	1.357	0.988	0.445
2	place	1.055	0.922	0.346
3	voicing	0.629	0.796	0.206
		3.041		0.996

Effective Feature Matrix (5dBSNR)

	b	c	d	g	j	k	p	S	s	t
voicing	+	-	+	+	+	-	-	-	-	-
place	bil	pal	alv	vel	pal	vel	bil	pal	alv	alv
manner	plo	aff	plo	plo	aff	plo	plo	fri	fri	plo

Features held constant in order of maximum proportion info. transmitted.

Contents of stimulus response matrix

	b	c	d	g	j	k	p	S	s	t
b	57	0	0	0	0	1	11	0	0	0
c	0	60	0	0	0	0	0	1	0	0
d	0	0	60	0	0	0	0	0	0	17
g	0	0	0	60	0	16	0	0	0	0
j	0	0	0	0	60	0	0	0	0	0
k	0	0	0	0	0	43	0	0	0	0
p	3	0	0	0	0	0	47	0	0	0
S	0	0	0	0	0	0	0	59	6	0
s	0	0	0	0	0	0	0	0	54	0
t	0	0	0	0	0	0	1	0	0	43

Constituent SR matrices : addnH5.txt

TRANSMITTED INFORMATION = 2.946
Proportion of correct responses = 0.907

Iteration 1 synopsis

FEATURE	INPUT	TRANS	%TRANS	TRANS/TI	%CORRECT

```
voicing  0.998  0.632  0.634  0.215   0.920
place    1.969  1.878  0.954  0.638   0.987
manner   1.372  1.358  0.990  0.461   0.998


Iteration  2 synopsis
---------------------
Constant:  manner
FEATURE  INPUT  TRANS %TRANS TRANS/TI
voicing  0.775  0.495  0.639  0.168
place    1.147  1.073  0.935  0.364


Iteration  3 synopsis
---------------------
Constant:  manner place
FEATURE  INPUT  TRANS %TRANS TRANS/TI
voicing  0.770  0.504  0.655  0.171


SINFA SUMMARY

Transmitted information = 2.946

ITERA-  FEATURE  TRANS  TRANS/  TRANS/
 TION                   INPUT   TI

  1     manner   1.358  0.990   0.461
  2     place    1.073  0.935   0.364
  3     voicing  0.504  0.655   0.171
                 2.935          0.996
```

Effective Feature Matrix (0dBSNR)
========================

	b	c	d	g	j	k	p	S	s	t
voicing	+	-	+	+	+	-	-	-	-	-
place	bil	pal	alv	vel	pal	vel	bil	pal	alv	alv
manner	plo	aff	plo	plo	aff	plo	plo	fri	fri	plo

Features held constant in order of maximum proportion info. transmitted.

Contents of stimulus response matrix
====================================

	b	c	d	g	j	k	p	S	s	t
b	35	0	0	6	0	0	9	0	0	0
c	0	57	0	0	6	0	0	0	0	0
d	0	0	56	0	0	0	2	0	0	23
g	16	0	2	54	0	30	0	0	0	0

j	0	3	2	0	54	0	0	0	0	0
k	0	0	0	0	0	30	0	0	0	0
p	9	0	0	0	0	0	46	0	0	0
S	0	0	0	0	0	0	1	60	9	0
s	0	0	0	0	0	0	0	0	51	0
t	0	0	0	0	0	0	2	0	0	37

Constituent SR matrices : addnH0.txt

TRANSMITTED INFORMATION = 2.575
Proportion of correct responses = 0.800

Iteration 1 synopsis

FEATURE	INPUT	TRANS	%TRANS	TRANS/TI	%CORRECT
voicing	0.999	0.449	0.449	0.174	0.863
place	1.963	1.612	0.821	0.626	0.933
manner	1.379	1.332	0.966	0.518	0.995

Iteration 2 synopsis

Constant: manner

FEATURE	INPUT	TRANS	%TRANS	TRANS/TI
voicing	0.754	0.312	0.414	0.121
place	1.136	0.840	0.739	0.326

Iteration 3 synopsis

Constant: manner place

FEATURE	INPUT	TRANS	%TRANS	TRANS/TI
voicing	0.702	0.312	0.444	0.121

SINFA SUMMARY

Transmitted information = 2.575

ITERA-TION	FEATURE	TRANS	TRANS/ INPUT	TRANS/ TI
1	manner	1.332	0.966	0.518
2	place	0.840	0.739	0.326
3	voicing	0.312	0.444	0.121
		2.484		0.965

Report of SINFA in SNHL Group

Unprocessed consonants(Quiet)

Effective Feature Matrix (Quiet)
========================

	b	c	d	g	j	k	p	S	s	t
voicing	+	-	+	+	+	-	-	-	-	-
place	bil	pal	alv	vel	pal	vel	bil	pal	alv	alv
manner	plo	aff	plo	plo	aff	plo	plo	fri	fri	plo

Features held constant in order of maximum proportion info. transmitted.

Contents of stimulus response matrix
====================================

	b	c	d	g	j	k	p	S	s	t
b	57	0	0	0	0	0	3	0	0	0
c	0	60	0	0	1	0	0	0	0	0
d	0	0	60	0	0	0	0	0	0	8
g	2	0	0	60	0	7	0	0	0	0
j	0	0	0	0	59	0	0	0	0	0
k	0	0	0	0	0	53	0	0	0	0
p	1	0	0	0	0	0	56	0	0	0
S	0	0	0	0	0	0	1	60	10	0
s	0	0	0	0	0	0	0	0	50	0
t	0	0	0	0	0	0	0	0	0	52

Constituent SR matrices : addnH10.txt

TRANSMITTED INFORMATION = 3.053
Proportion of correct responses = 0.945

Iteration 1 synopsis

FEATURE	INPUT	TRANS	%TRANS	TRANS/TI	%CORRECT
voicing	0.984	0.785	0.797	0.257	0.967
place	1.968	1.837	0.933	0.602	0.978
manner	1.374	1.357	0.988	0.445	0.998

Iteration 2 synopsis

Constant: manner

FEATURE	INPUT	TRANS	%TRANS	TRANS/TI
voicing	0.794	0.627	0.789	0.205
place	1.144	1.055	0.922	0.346

```
Iteration  3 synopsis
---------------------
Constant:  manner place
FEATURE  INPUT  TRANS %TRANS TRANS/TI
voicing  0.790  0.629  0.796  0.206


SINFA SUMMARY

Transmitted information = 3.053

ITERA-  FEATURE  TRANS  TRANS/  TRANS/
 TION                   INPUT   TI

  1     manner   1.357  0.988   0.445
  2     place    1.055  0.922   0.346
  3     voicing  0.629  0.796   0.206
                 3.041          0.996
```

Effective Feature Matrix (10 dBSNR)
========================

```
             b   c   d   g   j   k   p   S   s   t
voicing      +   -   +   +   +   -   -   -   -   -
place       bil pal alv vel pal vel bil pal alv alv
manner      plo aff plo plo aff plo plo fri fri plo
```

Features held constant in order of maximum proportion info. transmitted.

Contents of stimulus response matrix
====================================

	b	c	d	g	j	k	p	S	s	t
b	57	0	0	0	0	0	3	0	0	0
c	0	60	0	0	1	0	0	0	0	0
d	0	0	60	0	0	0	0	0	0	8
g	2	0	0	60	0	7	0	0	0	0
j	0	0	0	0	59	0	0	0	0	0
k	0	0	0	0	0	53	0	0	0	0
p	1	0	0	0	0	0	56	0	0	0
S	0	0	0	0	0	0	1	60	10	0
s	0	0	0	0	0	0	0	0	50	0
t	0	0	0	0	0	0	0	0	0	52

Constituent SR matrices : addnH10.txt

TRANSMITTED INFORMATION = 3.053
Proportion of correct responses = 0.945

Iteration 1 synopsis

FEATURE	INPUT	TRANS	%TRANS	TRANS/TI	%CORRECT
voicing	0.984	0.785	0.797	0.257	0.967
place	1.968	1.837	0.933	0.602	0.978
manner	1.374	1.357	0.988	0.445	0.998

Iteration 2 synopsis

Constant: manner

FEATURE	INPUT	TRANS	%TRANS	TRANS/TI
voicing	0.794	0.627	0.789	0.205
place	1.144	1.055	0.922	0.346

Iteration 3 synopsis

Constant: manner place

FEATURE	INPUT	TRANS	%TRANS	TRANS/TI
voicing	0.790	0.629	0.796	0.206

SINFA SUMMARY

Transmitted information = 3.053

ITERA-TION	FEATURE	TRANS	TRANS/ INPUT	TRANS/ TI
1	manner	1.357	0.988	0.445
2	place	1.055	0.922	0.346
3	voicing	0.629	0.796	0.206
		3.041		0.996

Effective Feature Matrix (5dBSNR)
========================

	b	c	d	g	j	k	p	S	s	t
voicing	+	-	+	+	+	-	-	-	-	-
place	bil	pal	alv	vel	pal	vel	bil	pal	alv	alv
manner	plo	aff	plo	plo	aff	plo	plo	fri	fri	plo

Features held constant in order of maximum proportion info. transmitted.

Contents of stimulus response matrix
====================================

	b	c	d	g	j	k	p	S	s	t
b	20	0	2	1	0	0	0	0	0	0
c	0	27	0	0	12	1	2	0	0	0
d	0	0	1	0	0	0	0	0	0	2
g	1	0	20	22	0	0	0	0	0	0
j	0	0	4	2	18	1	0	0	0	0
k	1	0	0	2	0	24	1	0	0	6
p	8	0	0	0	0	1	22	0	0	0
S	0	3	0	0	0	0	0	30	12	0
s	0	0	0	2	0	0	0	0	18	0
t	0	0	3	1	0	3	5	0	0	22

Constituent SR matrices : addpu5.txt

TRANSMITTED INFORMATION = 2.178
Proportion of correct responses = 0.680

Iteration 1 synopsis

FEATURE	INPUT	TRANS	%TRANS	TRANS/TI	%CORRECT
voicing	0.897	0.504	0.562	0.232	0.893
place	1.935	1.079	0.558	0.496	0.780
manner	1.430	1.134	0.793	0.521	0.950

Iteration 2 synopsis

Constant: manner

FEATURE	INPUT	TRANS	%TRANS	TRANS/TI
voicing	0.747	0.406	0.543	0.187
place	1.029	0.464	0.451	0.213

Iteration 3 synopsis

Constant: manner voicing

FEATURE	INPUT	TRANS	%TRANS	TRANS/TI
place	0.920	0.488	0.531	0.224

SINFA SUMMARY

Transmitted information = 2.178

ITERA- TION	FEATURE	TRANS	TRANS/ INPUT	TRANS/ TI
1	manner	1.134	0.793	0.521

2	voicing	0.406	0.543	0.187
3	place	0.488	0.531	0.224
		2.029		0.932

Effective Feature Matrix (0 dBSNR)
========================

	b	c	d	g	j	k	p	S	s	t
voicing	+	-	+	+	+	-	-	-	-	-
place	bil	pal	alv	vel	pal	vel	bil	pal	alv	alv
manner	plo	aff	plo	plo	aff	plo	plo	fri	fri	plo

Features held constant in order of maximum proportion info. transmitted.

Contents of stimulus response matrix
====================================

	b	c	d	g	j	k	p	S	s	t
b	19	0	2	1	0	0	0	1	0	0
c	0	29	0	0	12	1	2	0	0	0
d	0	0	2	0	0	0	0	0	0	3
g	1	0	21	22	0	1	0	0	0	0
j	0	0	4	2	18	2	0	0	0	0
k	2	0	1	2	0	23	1	0	0	4
p	8	0	0	0	0	1	22	0	0	0
S	0	1	0	0	0	0	0	30	12	0
s	0	0	0	2	0	0	0	0	18	0
t	0	0	0	1	0	2	5	0	0	23

Constituent SR matrices : addpu0.txt

TRANSMITTED INFORMATION = 2.184
Proportion of correct responses = 0.684

Iteration 1 synopsis

FEATURE	INPUT	TRANS	%TRANS	TRANS/TI	%CORRECT
voicing	0.914	0.459	0.503	0.210	0.884
place	1.932	1.039	0.538	0.476	0.774
manner	1.431	1.132	0.791	0.518	0.950

Iteration 2 synopsis

Constant: manner

FEATURE	INPUT	TRANS	%TRANS	TRANS/TI

```
voicing  0.752  0.382  0.508  0.175
place    1.017  0.454  0.447  0.208
```

```
Iteration  3 synopsis
---------------------
Constant:  manner voicing
FEATURE  INPUT  TRANS %TRANS TRANS/TI
place    0.914  0.489  0.535  0.224
```

```
SINFA SUMMARY

Transmitted information = 2.184

ITERA-  FEATURE  TRANS  TRANS/  TRANS/
 TION                   INPUT   TI

  1     manner   1.132  0.791   0.518
  2     voicing  0.382  0.508   0.175
  3     place    0.489  0.535   0.224
                 2.003          0.917
```

Hearing Aid Processed Consonants

Effective Feature Matrix (Quiet)
========================

	b	c	d	g	j	k	p	S	s	t
voicing	+	-	+	+	+	-	-	-	-	-
place	bil	pal	alv	vel	pal	vel	bil	pal	alv	alv
manner	plo	aff	plo	plo	aff	plo	plo	fri	fri	plo

Features held constant in order of maximum proportion info. transmitted.

Contents of stimulus response matrix
====================================

	b	c	d	g	j	k	p	S	s	t
b	25	0	0	0	0	0	3	0	0	0
c	0	29	0	0	0	0	0	0	0	0
d	1	0	20	0	0	2	3	0	0	7
g	0	0	5	24	0	6	0	0	0	0
j	0	1	1	4	30	2	0	0	0	0
k	0	0	1	2	0	12	0	0	0	1
p	4	0	0	0	0	0	21	0	0	1
S	0	0	0	0	0	0	0	30	13	0
s	0	0	0	0	0	0	0	0	17	0

```
t     0  0  3  0  0  8  3  0  0  21

Constituent SR matrices : addpHQ.txt

TRANSMITTED INFORMATION = 2.373
Proportion of correct responses = 0.763

Iteration  1 synopsis
---------------------

FEATURE  INPUT  TRANS %TRANS TRANS/TI %CORRECT
voicing  0.992  0.486  0.490  0.205   0.887
place    1.926  1.279  0.664  0.539   0.850
manner   1.405  1.263  0.899  0.532   0.977


Iteration  2 synopsis
---------------------
Constant:  manner
FEATURE  INPUT  TRANS %TRANS TRANS/TI
voicing  0.771  0.370  0.479  0.156
place    1.079  0.625  0.579  0.263


Iteration  3 synopsis
---------------------
Constant:  manner place
FEATURE  INPUT  TRANS %TRANS TRANS/TI
voicing  0.745  0.360  0.483  0.152


SINFA SUMMARY

Transmitted information = 2.373

ITERA-  FEATURE  TRANS  TRANS/  TRANS/
 TION                   INPUT   TI

  1     manner   1.263  0.899   0.532
  2     place    0.625  0.579   0.263
  3     voicing  0.360  0.483   0.152
                 2.248          0.947
```

Effective Feature Matrix (10 dBSNR)
========================

	b	c	d	g	j	k	p	S	s	t
voicing	+	-	+	+	+	-	-	-	-	-

```
place      bil pal alv vel pal vel bil pal alv alv
manner     plo aff plo plo aff plo plo fri fri plo
```

Features held constant in order of maximum proportion info. transmitted.

Contents of stimulus response matrix
====================================

	b	c	d	g	j	k	p	S	s	t
b	22	0	0	0	0	0	3	0	0	0
c	0	30	0	0	3	6	0	1	3	0
d	1	0	22	0	0	0	0	0	0	2
g	1	0	2	20	0	4	0	0	0	0
j	0	0	3	5	27	0	0	0	0	0
k	0	0	0	1	0	12	1	0	0	0
p	6	0	1	0	0	0	23	0	0	1
S	0	0	0	0	0	1	0	28	14	0
s	0	0	0	0	0	0	0	1	13	0
t	0	0	2	4	0	7	3	0	0	27

Constituent SR matrices : addpH10.txt

TRANSMITTED INFORMATION = 2.247
Proportion of correct responses = 0.747

Iteration 1 synopsis

FEATURE	INPUT	TRANS	%TRANS	TRANS/TI	%CORRECT
voicing	0.953	0.546	0.573	0.243	0.913
place	1.884	1.161	0.616	0.517	0.820
manner	1.435	1.098	0.765	0.489	0.937

Iteration 2 synopsis

Constant: manner

FEATURE	INPUT	TRANS	%TRANS	TRANS/TI
voicing	0.793	0.429	0.541	0.191
place	1.007	0.577	0.573	0.257

Iteration 3 synopsis

Constant: manner place

FEATURE	INPUT	TRANS	%TRANS	TRANS/TI
voicing	0.723	0.430	0.596	0.192

SINFA SUMMARY

Transmitted information = 2.247

ITERA-TION	FEATURE	TRANS	TRANS/ INPUT	TRANS/ TI
1	manner	1.098	0.765	0.489
2	place	0.577	0.573	0.257
3	voicing	0.430	0.596	0.192
		2.105		0.937

Effective Feature Matrix (5dBSNR)
=======================

	b	c	d	g	j	k	p	S	s	t
voicing	+	-	+	+	+	-	-	-	-	-
place	bil	pal	alv	vel	pal	vel	bil	pal	alv	alv
manner	plo	aff	plo	plo	aff	plo	plo	fri	fri	plo

Features held constant in order of maximum proportion info. transmitted.

Contents of stimulus response matrix
====================================

	b	c	d	g	j	k	p	S	s	t
b	23	0	0	1	0	0	2	0	0	0
c	0	29	2	0	4	4	0	0	3	0
d	0	0	17	0	0	0	0	0	0	7
g	0	0	3	15	0	4	0	0	0	3
j	1	0	3	10	25	7	1	0	0	0
k	0	0	1	2	0	14	2	0	0	1
p	6	1	0	0	1	0	22	0	1	1
S	0	0	1	1	0	0	0	26	8	0
s	0	0	0	1	0	0	0	4	18	0
t	0	0	3	0	0	1	3	0	0	18

Constituent SR matrices : addpH5.txt

TRANSMITTED INFORMATION = 2.005
Proportion of correct responses = 0.690

Iteration 1 synopsis

FEATURE	INPUT	TRANS	%TRANS	TRANS/TI	%CORRECT
voicing	0.975	0.360	0.369	0.180	0.847
place	1.889	1.001	0.530	0.500	0.783

```
manner   1.478  0.910  0.615  0.454   0.877


Iteration  2 synopsis
---------------------
Constant:  manner
FEATURE  INPUT  TRANS %TRANS TRANS/TI
voicing  0.757  0.290  0.383  0.145
place    0.974  0.560  0.574  0.279


Iteration  3 synopsis
---------------------
Constant:  manner place
FEATURE  INPUT  TRANS %TRANS TRANS/TI
voicing  0.719  0.299  0.416  0.149


SINFA SUMMARY

Transmitted information = 2.005

ITERA-  FEATURE  TRANS  TRANS/  TRANS/
 TION                   INPUT   TI

  1     manner   0.910  0.615   0.454
  2     place    0.560  0.574   0.279
  3     voicing  0.299  0.416   0.149
                 1.769          0.882
```

Effective Feature Matrix (0dBSNR)
========================

	b	c	d	g	j	k	p	S	s	t
voicing	+	-	+	+	+	-	-	-	-	-
place	bil	pal	alv	vel	pal	vel	bil	pal	alv	alv
manner	plo	aff	plo	plo	aff	plo	plo	fri	fri	plo

Features held constant in order of maximum proportion info. transmitted.

Contents of stimulus response matrix
====================================

	b	c	d	g	j	k	p	S	s	t
b	10	0	0	4	0	0	2	0	0	0
c	0	30	4	0	3	3	0	1	4	1
d	2	0	18	0	0	0	0	0	0	7
g	2	0	2	16	0	7	0	0	0	2
j	0	0	2	8	26	6	0	0	0	0

k	1	0	1	1	0	13	0	0	0	0
p	12	0	0	0	0	0	25	0	0	0
S	0	0	0	0	0	0	0	27	12	0
s	0	0	0	0	0	0	0	2	14	0
t	3	0	3	0	1	2	3	0	0	20

Constituent SR matrices : addpH0.txt

TRANSMITTED INFORMATION = 1.995
Proportion of correct responses = 0.663

Iteration 1 synopsis

FEATURE	INPUT	TRANS	%TRANS	TRANS/TI	%CORRECT
voicing	0.958	0.300	0.313	0.150	0.823
place	1.877	1.040	0.554	0.521	0.783
manner	1.457	1.009	0.693	0.506	0.900

Iteration 2 synopsis

Constant: manner

FEATURE	INPUT	TRANS	%TRANS	TRANS/TI
voicing	0.786	0.247	0.314	0.124
place	0.980	0.508	0.519	0.255

Iteration 3 synopsis

Constant: manner place

FEATURE	INPUT	TRANS	%TRANS	TRANS/TI
voicing	0.705	0.274	0.389	0.138

SINFA SUMMARY

Transmitted information = 1.995

ITERA-TION	FEATURE	TRANS	TRANS/INPUT	TRANS/TI
1	manner	1.009	0.693	0.506
2	place	0.508	0.519	0.255
3	voicing	0.274	0.389	0.138
		1.792		0.898