Relationship between Envelope Difference Index, Sentence Recognition and Speech Quality in Individuals with Hearing Impairment

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Abstract

The speech signal consists of temporal envelope and temporal fine structure. Studies have reported of alterations in the envelope of the signal when processed through different digital signal processing (DSP) algorithms in the hearing aid. Hence, the present study aimed at investigating the combined effects of the DSP algorithms used in digital hearing aids on speech recognition scores (SRS), quality perception of speech and envelope difference index (EDI), and to assess the correlation between the subjective and the objective measures. A total of 25 individuals with mild to moderate sensorineural hearing loss in the age range of 18-55 years were included in the study. Experiments were carried out to measure SRS, quality perception and EDI in the unaided condition, in linear mode (all algorithms deactivated) and nonlinear mode (all algorithms activated) at 55 dB SPL, 65 dB SPL and 80 dB SPL. The testing was done in quiet and at +10 dB SNR using speech shaped noise. The results revealed that the SRS and the quality ratings were significantly better at higher presentation level and in linear mode in quiet. However, the EDI was very similar across different aided conditions. The correlation obtained between the objective and the subjective measures was limited to few test conditions with only a moderate level of correlation. Hence, it is clear that, an insignificant change in the temporal envelope of the signal can also lead to a significant change in the perception of speech. That is, the DSP algorithms brought about negative changes in the output as reflected in the SRS, quality perception and subjective preference.

Key words: DSP, SRS, quality perception, EDI, linear, nonlinear

Introduction

For proper communication, hearing mechanism remains to be an important link in the speech chain. Hence, impairment in hearing sensitivity leads to communication breakdown. Individuals with sensorineural hearing loss have poor audibility, poor frequency and poor temporal resolution due to widened auditory filters and loss of non-linearity in the cochlea (Moore, Glasberg, & Simpson, 1992).

The individuals with hearing impairment are fitted with hearing aids to overcome the problem of inaudibility. In addition, the digital hearing aids are available with various algorithms like wide dynamic range compression (WDRC), digital noise reduction (DNR) and directionality algorithms in order to improve speech intelligibility. These algorithms were devised to improve the perception of speech in the presence of noise as speech recognition in noise is one of the common complaints of individuals with hearing impairment.

The speech signal consists of temporal envelope and temporal fine structure. The temporal envelope of a speech signal refers to slower amplitude modulations superimposed on the temporal fine structure. Temporal envelope cues have been reported to contribute for speech recognition (Dorman, Marton, & Hannley, 1985; Gordon-Salant & Fitzgibbons, 1993; Healy & Warren, 2003; Price & Simon, 1984; Shannon, Zeng, Kamath, Wygonski, & Ekelid, 1995) and quality (Anderson, 2011). It is reported that depending on the amount of modification in the envelope of the signal, the quality of the signal is also degraded (Anderson, 2011). Hence, when there is a modification of envelope cues, the quality of perception of speech is reported to be affected along with speech recognition (Anderson, 2013).

Thus, temporal envelope is an important aspect of speech signal. Nevertheless, the digital signal processing algorithms have been reported to alter these cues. There are few studies available to explain the effects of these algorithms on the temporal envelope of the speech signal.

The WDRC used in digital hearing aids have been reported to alter the temporal envelope of the speech signal (Stone & Moore, 2007; Stone & Moore, 2008). Several studies have reported a negative effect of WDRC on speech intelligibility and quality depending on the compression settings (Gatehouse, Naylor & Elberling, 2006; Hansen, 2002; Moore, Stainsby, Alcántara, & Kühnel, 2004; Neuman, Bakke, Mackersie, Hellman, & Levitt, 1998).

DNR used for improving the speech perception in the presence of noise has also been reported to alter the temporal envelope of the incoming speech signal (Levitt, 2001) and to affect the speech perception in individuals with sensorineural hearing loss (Alcantara, Moore, Kuhnel, & Launer, 2003; Boymans & Dreschler, 2000; Levitt, Bakke, & Kates, 1993).

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Directionality in hearing aid is used for better speech perception in noise when the source of the noise and the signal are spatially separated. In the presence of noise that is spatially separated, directionality has been reported to help for better perception of speech (Luts, Jean & Wouters, 2004; Valente, Fabry & Potts, 1995).

Though the primary aim of these algorithms is to improve speech perception, due to the alteration of the temporal envelope of the signal, the intelligibility and quality of the signal could be affected. It is important that these changes are quantified subjectively and objectively. There are different objective measures that are available to quantify the changes in the temporal envelope. The Envelope Difference Index (EDI) in one of them. This was originally developed by Fortune, Woodruff, & Preves (1994). EDI quantifies the temporal changes between two acoustical signals and provides it in a numerical form ranging from 0 to 1. Here '0' denotes no difference and '1' denotes the maximum difference between the processed and unprocessed signal. It has gained its popularity as it can be compared directly with the subjective performance (Fortune et al., 1994).

There are reports of good correlation of EDI and speech recognition with WDRC algorithm. Jenstad and Souza (2005, 2007) used various compression ratios and release times to create WDRC amplified signals with a range of EDI values (relative to unprocessed versions of the same tokens). At high EDIs, speech recognition decreased monotonically. For example, an increase in EDI from 0.25 to 0.34 decreased recognition by about 10% for easy speech materials and about 20% for more difficult (rapidly spoken) speech materials.

Jenstad and Souza (2007) argued that equivalent EDIs derived with any combination of compression parameters should result in similar recognition scores. This contention is supported by the data of Jenstad and Souza (2007) in combination with Walaszek (2008). The resulting sentences were recognized correctly 65%–70% of the time by Jenstad and Souza's participants and 65% of the time by Walaszek's participants. Hence, EDI can be used to quantify the temporal changes caused by amplitude compression of hearing aids (Souza et al., 2012).

A study was carried out by Geetha and Manjula (2014), to evaluate the use of EDI to quantify the independent and interactive effects of WDRC, DNR and directionality on the temporal aspects of sentence in noise, and to assess the perceived quality in individuals with normal hearing sensitivity. The temporal cues were reported to be more distorted when all the algorithms are activated simultaneously and the quality rating of loudness and clarity was lower for some combinations of the algorithms.

Many research studies have been carried out to evaluate the effect of different algorithms used in hearing aids on speech perception. (Hickson & Thyer, 2003; Muller, Weber, & Hornsby, 2006; Souza, 2002). The effects of the algorithms, like compression, DNR and directionality have been found to depend on the settings of the hearing aid and the noise conditions tested.

Temporal envelope cues have been reported to be important for speech recognition (Dorman, Marton, & Hannley, 1985; Gordon- Salant & Fitzgibbons, 1993; Healy & Warren, 2003; Price & Simon, 1984; Shannon, Zeng, Kamath, Wygonski, & Ekelid, 1995) and for quality perception (Anderson, 2011). The advanced signal processing algorithms have been reported to alter these temporal envelope cues (Venn, Souza, Brennan, & Stecker, 2009). EDI has been used to quantify the distortions induced by the hearing aid algorithm on the temporal envelope of the signal. There has been a good correlation reported between the EDI and the speech recognition (Jenstad & Souza, 2005; Souza, Hoover & Gallun, 2012). Though, EDI is a most accepted measure and has been found to correlate well with speech recognition scores, the studies on EDI have used words or syllables as stimuli, sentences and passages have not been tested. It is known that the hearing aid's behavior for a sentence can be different from that of syllables and words.

Further, most of these studies have studied the independent effects of these algorithms. In real life, these algorithms may work simultaneously, depending on the environment. Hence, it is important to quantify the alterations in the temporal envelope when all of these algorithms work together and their effect on speech recognition.

Studies have reported that, in a particular setting of a hearing aid, the presence of good speech intelligibility scores need not always indicate good quality perception (Geetha & Manjula, 2005; Punch & Beck, 1980). Hence, quality judgment is a useful addition to speech intelligibility while fitting hearing aids. Though the quality measurement has been reported to be an important tool for the fitting of amplification devices (Anderson, 2013), the correlation of this with the EDI has not been studied. Geetha and Manjula (2014) attempted to correlate EDI with the quality perception. However, statistical correlation was not carried out between EDI and quality perception of speech, as there was only a single measure of EDI obtained. Further, the study was performed on individuals with normal hearing sensitivity.

In addition, the quality measurements are used very limitedly in the clinical setting for hearing aid fitting. The main reason for this could be the lack of time as assessment of quality is time consuming and requires usage of longer stimuli. If the EDI was found to be correlated with the quality, the results could be helpful in the fitting hearing aids with reference to the quality of output. Hence, this study will focus on evaluating the clinical use of EDI by correlating it with Speech Recognition Scores (SRS) for sentences and perceptual quality rating in quiet as well as in the presence of noise.

The aim of the present study was to investigate the effect of DSP algorithms on EDI, SRS with sentences and quality. Another aim of the study was to investigate the correlation between EDI and speech recognition with sentences, and the correlation between EDI and quality rating in individuals with sensorineural hearing loss fitted with hearing aids.

The objectives of the studies were:

1. To obtain SRS for sentences at 55, 65 and 80 dB SPL.
2. To obtain quality judgment at 55, 65 and 80 dB SPL.
3. To record the output of the hearing aid at 55, 65 and 80 dB SPL, and to obtain EDI at each level.
4. To find the correlation between the objective and the subjective measurements across different input levels.

The above was done in the unaided and in the aided condition with all the DSP algorithms (WDRC, DNR and Directionality) enabled (known as nonlinear mode) in quiet and in noise at +10 dB SNR, and with all DSP algorithms disabled (known as linear mode) in quiet and in noise at +10 dB SNR.

Method

Participants: The present study included 20 participants in the age range of 18 - 55 years (mean= 42.16; SD= 11.88). Along with a detailed case history, a routine audiological evaluation was carried out, including pure-tone audiometry, speech audiometry and Immittance evaluation to select the participants for the current study.

Individuals having post-lingual mild to moderate sensorineural hearing loss, with flat audiogram configuration (within 10 dB rise or fall over the range of frequencies) in the frequency range of 500 Hz to 5000 Hz (Kennedy, Levitt, Neuman, & Weiss, 1998); speech identification scores greater than or equal to 70%; having ‘A’ or ‘As’ type of tympanogram with acoustic reflex appropriate to the degree of hearing loss were included in the study. All the participants were naive hearing aid users and fluent speakers of Kannada language. Individuals having any history or presence of the middle ear pathology, neurological involvement and psychological problems were not included in the study.

Instrumentation:

1. Pure tone thresholds, speech recognition threshold (SRT) and SRS were obtained using a dual channel calibrated diagnostic audiometer. TDH 39 headphones, housed in the MX - 41 AR cushions, Radio Ear B - 71 Bone vibrator. Two loudspeakers located at 0° azimuth and 180° azimuth at 1 meter distance were used for the actual experiment.
2. Tympanometry and acoustic reflex assessment were carried out using GSI-Tympstar middle ear analyzer.
3. Fonix® 7000 real ear analyzer along with the probe microphone was used for recording the output in the ear canal.
4. A four channel wide dynamic range compression digital behind- the- ear hearing aid with the fitting range of mild to severe degree of hearing loss and with an option to enable or disable DNR and directionality.

Stimuli: Paired words in Kannada language developed at the Department of Audiology, All India Institute of Mysore, were used for obtaining SRT. SRS was obtained using a recorded version of phonemically balanced word lists in Kannada language developed by Yathiraj and Vijayalakshmi (2005) during routine evaluation. The SRS in the actual experiment were obtained using the sentences in Kannada language developed by Geetha, Kumar, Manjula and Pavan (2014). This test consists of 25 homogeneous lists with ten sentences under each list. The number of key words in each list was 40. A recorded version of the paragraph in Kannada language developed by Sairam (2003) was used for quality rating.

Test environment: Air conditioned, sound treated double room set-up was used to administer all the tests. The noise level was within the permissible limits ANSI S3.6 (1999).

Procedure

Routine Audiological evaluation: Using the modified Hughson and Westlake procedure (Carhart & Jerger, 1959), pure tone thresholds were obtained with the help of a calibrated dual channel diagnostic audiometer. This was done at octave frequencies from 250 Hz to 8000 Hz for obtaining air conduction thresholds and from 250 Hz to 4000 Hz for bone conduction thresholds. Air conduction thresholds obtained at 500 Hz, I kHz, 2 kHz and 4 kHz were averaged to obtain the Pure Tone Average (PTA).

Paired words in Kannada were used to obtain the SRT. The obtained scores were correlated with PTA. PB words in Kannada developed by Yathiraj and Vijayalakshmi (2005) were used to obtain the Speech Identification Scores at 40 dB SL.

Immittance Evaluation was done on all the individuals. Tympanometry and Acoustic reflex assessment were carried out using standard procedures with GSI-

Tympstar middle ear analyzer. Participants satisfying the selection criteria based on the results of the above tests, were selected for further evaluations.

Programming the hearing aid: The hearing aid was connected to a personal computer with NOAH-3 software connected through a NOAH link with appropriate programming cable. The programming was done based on the NAL-NL1 formula. Acclimatization level was set to 2 for all the participants. The gain-frequency response of the hearing aid was modified depending on the listener's response to Ling's six sounds. The routine hearing aid trial was carried out with the programmed settings. Then, the hearing aid was programmed for the aided conditions, linear and non-linear. For programming the hearing aid in linear mode, the compression, directionality and noise reduction algorithms were disabled. For programming the hearing aid in nonlinear mode, the compression was enabled and the compression setting was kept as prescribed by the software. The directionality and noise reduction algorithms were also enabled in the nonlinear mode.

Experiment for subjective measurements

a. Measurement of SRS for sentences

The programmed hearing aid was fitted to the participants. The actual experiment was carried out by obtaining SRS for sentences using recorded sentences developed by Geetha et al., (2014). The test set up is given in Figure1. The stimulus was presented at 55, 65 and 80 dB SPL through the calibrated audiometer to the loudspeaker placed at 0 azimuth and the speech shaped noise was presented through loud speakers placed at 180 azimuths. The noise was presented at +10 dB SNR. The listeners were instructed to repeat the sentences. The responses were noted down in a response sheet. The total number of key words repeated correctly for each list was calculated to obtain the SRS scores. The maximum number of key words in each list was 40. The same procedure was done in quiet condition with the hearing aid, and also in the unaided condition. The assessments with the hearing aid were done in linear and nonlinear condition.

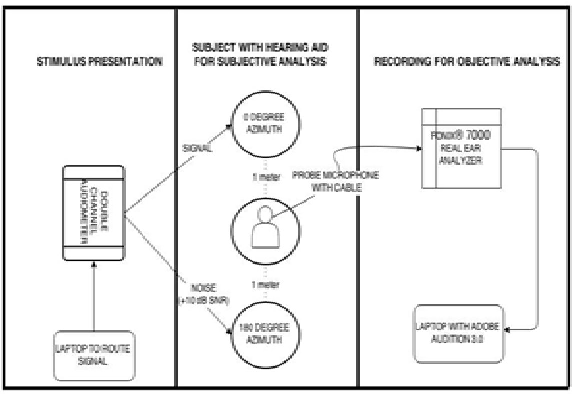

Figure 1: Pictorial representation of the test setup.

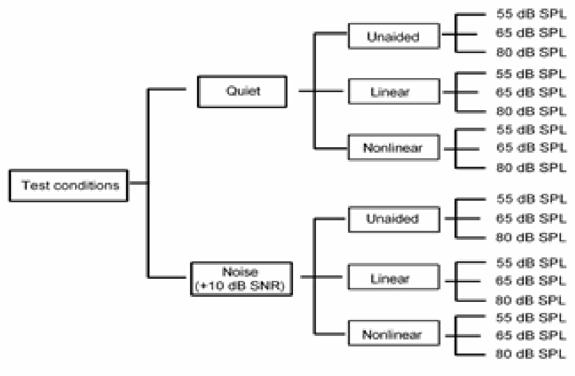

Figure 2: Test conditions used in the experiment.

The list of test conditions included in the study is as shown in Figure 2. Different sentence lists were presented for different test conditions in order to avoid the practice effect. The order of the lists and the order of the test conditions were randomized in order to avoid the order effect.

b. Measurement of quality:

A paragraph in Kannada language developed by Sairam (2003) containing all the speech sounds of Kannada language was used for quality rating. The stimuli were presented through the loud speaker placed at 0 azimuth and noise at 180 azimuths. Quality rating of the test stimuli was made by all the participants. The experiment was carried out in the conditions mentioned in Figure 2.

The participants were instructed to listen to the passage and rate the same on four parameters of quality. The quality rating scale developed by Eisenberg and Dirks (1995) was adapted and modified for this study. The five point rating scale was as follows: 0 = Very poor; 1 = Poor; 2 = Fair; 3 = Good; 4 = Excellent.

Four parameters of quality, i.e., loudness, clarity, naturalness and overall impression were rated by the listeners on a five point rating scale. Practice trials were given before the actual testing.

c. Subjective preference:

In addition, informal subjective preference with reference to comfort, ease of listening and quality of linear vs. nonlinear condition was assessed. However, no statistical analysis of this was done as this was not a part of the objectives of the present study.

Experiment for objective measurement

Otoscopic examination was done to ensure that the ear canal is devoid of impact wax and other infection. The output of the hearing aid was recorded in the same testing conditions and aided conditions as above, for both sentences and Kannada passage. The same settings were used for objective analysis as well. All the participants were fitted with hearing aid along with the probe microphone, to record the output of the hearing aid. The probe microphone was positioned in the ear canal using acoustic positioning procedure given by ANSI (1994) standards, by presenting 65 dB SPL of composite noise through the Fonix® 7000 system. Once the probe tube was inserted, the loudspeaker of the Fonix® 7000 system was disabled.

Kannada sentences and the Kannada passage were routed through a calibrated diagnostic audiometer to the loudspeakers. The test setup was as given in Figure 3.1. The speakers were kept at a 0 º angle for the presentation of speech and 180 º angles for the presentation of noise at a distance of 1 meter from the hearing aid. The output of the programmed hearing aid was obtained with the probe microphone of Fonix® 7000 real ear system and recorded with Adobe Audition 3.0 software by connecting Fonix® 7000 to an HP laptop. All the measurements were done in the conditions given in Figure 2.

The recorded stimuli were then edited with reference to a common reference point. The unaided stimulus was first loaded and then the aided stimuli in the EDI software (Jenstad & Lister, 2007). The temporal envelope differences between the unprocessed (unaided) and linearly processed signal; unprocessed and nonlinearly processed stimuli were obtained using the software. The EDI was measured between each processed and unprocessed (for sentences and passage) under the quiet environment as well as in noise for all the presentation levels.

Results

The aim of the present study was to investigate the effect of hearing aid algorithms on EDI, SRS and quality perception. The study also aimed to assess the correlation between EDI with SRS and with quality rating. The EDI, SRS and quality ratings were tabulated and statistically analyzed using Statistical Package for Social Sciences (SPSS, version 21.0).

The data were statistically analyzed for normality using the Shapiro-Willk's test. The measure of EDI obtained for sentences followed normal distribution ($p>0.05$). Hence, the data were further subjected to parametric tests. Whereas, the EDI obtained for passage, SRS and quality ratings, did not follow normal distribution ($p<0.05$). Hence, these measures were subjected to non-parametric tests.

Measurement of SRS

The results of SRS for sentences are presented in Table 1. The maximum possible SRS was 40. As observed in the table, the SRS increased with increase in presentation level. The SRS was greater for the quiet condition when compared to noise condition, and was observed to be greater for the linear condition when compared to the nonlinear condition.

Table 1: *Median and SD of SRS*

Test Condition	Presentation level (dB SPL)	Unaided	Linear	Nonlinear
		Median (SD)	Median (SD)	Median (SD)
Quiet	55	0.00 (2.72)	15.00 (12.29)	14.00 (11.66)
	65	11.00 (8.11)	28.00 (6.80)	26.00 (7.63)
	80	20.00 (8.51)	36.00 (2.64)	37.00 (2.64)
Noise	55	0.00 (1.13)	10.00 (12.30)	8.00 (11.85)
	65	8.00 (9.56)	19.00 (9.60)	17.00 (10.25)
	80	15.00 (9.01)	32.00 (6.81)	30.00 (5.69)

For statistical analysis of SRS across presentation levels and across aided conditions, the Friedman's test was carried out, as there were three variables. This was followed by Wilcoxon signed rank test. Whereas, for the analysis of SRS in quiet and noise condition, Wilcoxon signed rank test was carried out directly as there were only two variables.

The results of Friedman's test across presentation levels and across aided conditions revealed a significant difference ($p= 0.000$). Further, Wilcoxon signed rank test was done to carry out pair-wise comparison of SRS across presentation levels and across aided conditions as shown in Table 2, and pair-wise comparison of SRS in quiet and noise condition as shown in Table 3. The results of Wilcoxon signed rank test across presentation levels for SRS revealed that the SRS obtained at each of the presentation level was significantly different from each other at all aided conditions. However, there was no significant difference for SRS obtained between linear and nonlinear condition, in any of the testing conditions. In addition, the comparison of SRS in quiet and noise conditions revealed a significant difference at all the presentation levels and aided conditions.

Measurement of quality rating

In the quality rating, '0' meant very poor and '4' meant excellent. In the present study, overall, the quality rating did not go beyond '3' in any of the conditions. The median and standard deviation of quality ratings (for loudness, clarity, naturalness and overall impression) for passage can be seen in Table 4.

Table 2: *Results of Wilcoxon Signed Ranks Test of SRS across presentation levels and aided conditions*

	Conditions		¦Z¦
Across levels	Quiet Unaided	55 dB vs. 65 dB	4.377**
		55 dB vs. 80 dB	4.376**
		65 dB vs. 80 dB	4.382**
	Quiet linear	55 dB vs. 65 dB	4.324**
		55 dB vs. 80 dB	4.375**
		65 dB vs. 80 dB	4.188**
	Quiet nonlinear	55 dB vs. 65 dB	4.291**
		55 dB vs. 80 dB	4.288**
		65 dB vs. 80 dB	3.632**
	Noise unaided	55 dB vs. 65 dB	3.679**
		55 dB vs. 80 dB	4.375**
		65 dB vs. 80 dB	4.287**
	Noise linear	55 dB vs. 65 dB	4.201**
		55 dB vs. 80 dB	4.375**
		65 dB vs. 80 dB	3.584**
	Noise nonlinear	55 dB vs. 65 dB	4.291**
		55 dB vs. 80 dB	4.288**
		65 dB vs. 80 dB	3.632**
Across aided conditions	Quiet 55 dB	Unaided vs. Linear	4.199**
		Unaided vs. Nonlinear	4.109**
		Linear vs. Nonlinear	2.097*
	Quiet 65 dB	Unaided vs. Linear	4.376**
		Unaided vs. Nonlinear	4.375**
		Linear vs. Nonlinear	2.960
	Quiet 80 dB	Unaided vs. Linear	4.200**
		Unaided vs. Nonlinear	4.245**
		Linear vs. Nonlinear	0.264
	Noise 55 dB	Unaided vs. Linear	3.653**
		Unaided vs. Nonlinear	3.469**
		Linear vs. Nonlinear	2.233*
	Noise 65 dB	Unaided vs. Linear	4.374**
		Unaided vs. Nonlinear	4.174**
		Linear vs. Nonlinear	2.744**
	Noise 80 dB	Unaided vs. Linear	4.376**
		Unaided vs. Nonlinear	4.203**
		Linear vs. Nonlinear	1.619

Note: * $p<0.05$, ** $p<0.01$.

It can be observed in the Table 4 that with increase in the presentation levels, the rating for loudness, clarity, naturalness and overall impression also increased. The rating was higher for quiet condition when compared to noise condition, and higher for linear condition when compared to nonlinear condition. Statistical analysis was carried out to see if these observations were statistically significant.

Table 3: *Results of Wilcoxon Signed Ranks Test of SRS between quiet and noise conditions*

	Conditions		¦Z¦
Between quiet and noise	Unaided 55 dB	Quiet vs. Noise	1.926*
	Linear 55 dB	Quiet vs. Noise	3.619**
	Nonlinear 55 dB	Quiet vs. Noise	4.152**
	Unaided 65 dB	Quiet vs. Noise	2.895**
	Linear 65 dB	Quiet vs. Noise	4.015**
	Nonlinear 65 dB	Quiet vs. Noise	3.106**
	Unaided 80 dB	Quiet vs. Noise	4.358**
	Linear 80 dB	Quiet vs. Noise	4.232**
	Nonlinear 80 dB	Quiet vs. Noise	4.382**

Note: * $p<0.05$, ** $p<0.01$.

Similar to SRS, for the analysis of quality rating across presentation levels and across aided conditions, the Friedman's test and Wilcoxon signed rank test were carried out, as there were three variables. Whereas, for the analysis of quality across quiet and noise conditions, Wilcoxon signed rank test was carried out directly as there were only two variables. The results of Friedman's test revealed a significant difference across presentation levels and across aided conditions ($p= 0.000$). Further, Wilcoxon signed rank test was carried out for pair wise comparison of subjective rating across presentation levels and aided conditions.

Wilcoxon signed rank test across presentation levels revealed that the rating obtained at 55 dB SPL was significantly lesser from that obtained at 65 dB SPL and 80 dB SPL in all the aided conditions. The rating obtained was also significantly lesser at 65 dB SPL when compared to the rating obtained at 80 dB SPL in all the aided conditions. In quiet and in the presence of noise, for all presentation levels, there was no significant difference between the linear and nonlinear condition in the majority of conditions. In few conditions, where there is a difference, the rating obtained for linear mode was higher. Between quiet and noise conditions, the quality rating for most of the conditions were significantly better in quiet condition as given in Table 5 & Table 6.

Table 4: *Median and SD of quality rating*

Quality Parameter	Test Condition	Presentation level (dB SPL)	Unaided		Linear		Nonlinear	
			Median	SD	Median	SD	Median	SD
Loudness	Quiet	55	0.00	0.70	2.00	0.69	2.00	0.78
		65	2.00	0.71	3.00	0.40	3.00	0.61
		80	3.00	0.88	4.00	0.92	4.00	0.49
	Noise	55	0.00	0.62	2.00	1.04	2.00	1.19
		65	1.00	0.94	3.00	0.65	3.00	065
		80	2.00	0.83	3.00	0.65	3.00	0.49
Clarity	Quiet	55	0.00	0.56	1.00	1.03	1.00	0.76
		65	1.00	0.73	2.00	0.60	2.00	0.69
		80	2.00	0.75	3.00	0.55	3.00	0.66
	Noise	55	0.00	0.40	1.00	0.83	1.00	0.93
		65	1.00	0.78	2.00	0.77	2.00	0.92
		80	1.00	0.91	2.00	0.66	2.00	0.70
Naturalness	Quiet	55	0.00	0.60	1.00	0.95	1.00	0.81
		65	1.00	0.82	2.00	0.65	2.00	0.73
		80	2.00	0.79	3.00	0.73	3.00	0.83
	Noise	55	0.00	0.44	1.00	0.98	0.00	1.05
		65	1.00	0.84	2.00	0.76	2.00	0.85
		80	1.00	0.77	3.00	0.80	2.00	0.64
Overall impression	Quiet	55	0.00	0.50	1.00	0.71	1.00	0.57
		65	1.00	0.64	3.00	0.58	2.00	0.64
		80	2.00	0.67	3.00	0.74	3.00	0.69
	Noise	55	0.00	0.40	1.00	0.93	1.00	0.84
		65	1.00	0.86	2.00	0.79	2.00	0.83
		80	2.00	0.66	3.00	0.65	2.00	0.71

Table 5: *Wilcoxon signed rank test of quality rating across presentation levels and aided conditions*

Comparison		Conditions	¦Z¦			
			Loudness	Clarity	Naturalness	Overall impression
Across levels	Quiet Unaided	55 dB vs. 65 dB	4.490**	3.542**	4.021**	4.564**
		55 dB vs. 80 dB	4.414**	4.491**	4.320**	4.497**
		65 dB vs. 80 dB	4.200**	3.069**	3.380**	3.755**
	Quiet linear	55 dB vs. 65 dB	4.508**	3.542**	4.018**	4.315**
		55 dB vs. 80 dB	4.053**	4.455**	4.106**	4.346**
		65 dB vs. 80 dB	2.402*	4.041**	2.493*	3.626**
	Quiet nonlinear	55 dB vs. 65 dB	4.021**	2.631**	3.087**	3.501**
		55 dB vs. 80 dB	4.365**	4.276**	3.812**	4.187**
		65 dB vs. 80 dB	4.065**	4.242**	3.038**	3.578**
	Noise unaided	55 dB vs. 65 dB	3.938**	3.690**	3.217**	3.758**
		55 dB vs. 80 dB	4.434**	4.137**	4.428**	4.562**
		65 dB vs. 80 dB	4.001**	2.615**	3.380**	3.578**
	Noise linear	55 dB vs. 65 dB	3.906**	3.557**	3.704**	4.564**
		55 dB vs. 80 dB	4.192**	4.215**	4.101**	4.497**
		65 dB vs. 80 dB	3.252**	3.448**	2.765**	3.755**
	Noise nonlinear	55 dB vs. 65 dB	3.487**	2.786**	3.430**	3.749**
		55 dB vs. 80 dB	3.973**	3.945**	3.796**	4.104**
		65 dB vs. 80 dB	3.520**	3.038**	2.357*	3.300**
	Quiet 55 dB	Unaided vs. Linear	4.507**	3.745**	3.707**	4.244**
		Unaided vs. Nonlinear	4.326**	3.288**	3.752**	4.284**
		Linear vs. Nonlinear	1.414	1.217	1.604	0.707

Across aided conditions	Quiet 65 dB	Unaided vs. Linear	4.443**	3.729**	4.211**	4.714**
		Unaided vs. Nonlinear	4.118**	2.738**	3.266**	3.227**
		Linear vs. Nonlinear	1.890	2.673**	3.500**	3.051**
	Quiet 80 dB	Unaided vs. Linear	3.226**	4.221**	4.284**	3.969**
		Unaided vs. Nonlinear	3.345**	3.758**	3.474**	3.255**
		Linear vs. Nonlinear	0.866	2.500*	2.066*	2.840**
	Noise 55 dB	Unaided vs. Linear	4.091**	3.704**	3.817**	3.729**
		Unaided vs. Nonlinear	3.878**	3.256**	3.213**	3.750**
		Linear vs. Nonlinear	0.447	0.302	2.476*	0.535
	Noise 65 dB	Unaided vs. Linear	4.363**	3.380	4.008**	4.099**
		Unaided vs. Nonlinear	4.253**	3.201**	2.980**	3.660**
		Linear vs. Nonlinear	0.433	0.333	2.134*	1.414
	Noise 80 dB	Unaided vs. Linear	3.775**	3.704**	3.834**	3.963**
		Unaided vs. Nonlinear	3.804**	3.827**	2.777**	3.392**
		Linear vs. Nonlinear	1.414	0.832	3.095**	2.714**

Note: * $p<0.05$, ** $p<0.01$.

Table 6: *Wilcoxon signed rank test of quality rating between quiet and noise condition*

Comparison	Conditions		|Z|			
			Loudness	Clarity	Naturalness	Overall impression
Between quiet and noise	Unaided 55 dB	Quiet vs. Noise	2.236*	2.449*	1.732	1.732
	Linear 55 dB	Quiet vs. Noise	1.897	3.051**	0.258	2.309*
	Nonlinear 55 dB	Quiet vs. Noise	1.127	1.667	2.309*	2.530*
	Unaided 65 dB	Quiet vs. Noise	2.309*	2.324*	3.000**	2.714**
	Linear 65 dB	Quiet vs. Noise	3.162**	3.419**	2.828**	3.606**
	Nonlinear 65 dB	Quiet vs. Noise	1.387	0.775	1.807	1.184
	Unaided 80 dB	Quiet vs. Noise	3.771**	3.207**	2.814**	3.000**
	Linear 80 dB	Quiet vs. Noise	1.165	4.234**	2.392*	2.683**
	Nonlinear 80 dB	Quiet vs. Noise	2.333*	2.486*	2.874**	2.673**

Note: * $p<0.05$, ** $p<0.01$.

Further, the results of informal assessment of individual preferences revealed that 23 listeners of the 25 listeners preferred linear mode over nonlinear mode across all the test conditions.

EDI

a. EDI for sentences:

The EDI ranges between '0' and '1'. An EDI score of '1' represents 100% difference and a score of 'o' indicates 0% difference between unprocessed and processed signal. Table 7 represents the mean and standard deviation of EDI for sentences. It is given across aided conditions and across presentation levels in quiet and noise.

As observed in the table, overall, the EDI ranged between 0.23 to 0.31 indicating a difference of 23% to 31% between unprocessed and processed signal, and the EDI was highest at the higher presentation level. However, EDI is similar between aided conditions in quiet and noise. In order to see if these observations were statistically significant, the data were subjected to parametric analysis using three-way repeated measures ANOVA. EDI obtained across presentation levels showed a significant difference ($F = 3.457$, $df = 2$, $p< 0.05$). The other conditions did not reveal a statistically significant difference.

Table 7: *Mean, median and SD of EDI for sentences*

Sti mu lus	Test Condit ion	Present ation level (dB SPL)	Linear		Nonlinear	
			Mean	SD	Mean	SD
Sentences	Quiet	55	0.23	0.10	0.24	0.06
		65	0.27	0.12	0.24	0.12
		80	0.26	0.16	0.28	0.13
	Noise	55	0.24	0.08	0.23	0.08
		65	0.27	0.10	0.25	0.12
		80	0.31	0.13	0.27	0.13

Further, Bonferroni pair-wise comparison was carried out to find out the pairs of presentation levels that were significantly different from each other. The results revealed a significant difference between the EDI obtained at 55 dB SPL and 80 dB SPL at a significance level of less than 0.05.

b. EDI for passage

The EDI for passage is tabulated in Table 8 with the mean, median and standard deviation across different aided and presentation conditions. As it can be observed in the table, the EDI ranged from 0.11 to 0.44 indicating a similarity of 11% to 44% between the unprocessed and the processed signal, and the EDI increased with increase in presentation level. EDI was observed to be greater for nonlinear condition, mostly when compared to linear condition. In addition, for most of the parameters, the EDI was greater for the quiet condition when compared to noise condition.

Table 8: *Mean, median and SD of EDI for passage*

Stimulus	Test Condition	Presentation level (dB SPL)	Linear		Nonlinear	
			Median	SD	Median	SD
Passage	Quiet	55	0.33	0.13	0.23	0.11
		65	0.35	0.13	0.38	0.18
		80	0.40	0.11	0.44	0.09
	Noise	55	0.28	0.09	0.29	0.11
		65	0.37	0.11	0.36	0.13
		80	0.38	0.12	0.37	0.08

For analysis of EDI across presentation levels, the Friedman's test was carried out, as there were three variables. This was followed by Wilcoxon signed rank test. Whereas, for analysis of EDI across aided conditions and across quiet and noise condition, Wilcoxon signed rank test was carried out directly as there were only two variables.

The results of Friedman's test across presentation levels revealed a significant difference ($p<0.01$). Further, wilcoxson signed rank test was done to carry out for the pair-wise comparison of EDI across presentation levels, aided conditions in quiet and in noise. The results of this are as shown in Table 9.

Table 9: *Results of wilcoxon signed ranks test of EDI for passage*

Parameter	Conditions		|Z|	Significance
Across levels	Quiet linear	55 dB vs. 65 dB	2.274	0.023*
		55 dB vs. 80 dB	3.000	0.003**
	Quiet nonlinear	55 dB vs. 65 dB	2.435	0.015*
		55 dB vs. 80 dB	2.462	0.014*
	Noise linear	55 dB vs. 80 dB	3.700	0.000**
		65 dB vs. 80 dB	2.408	0.016*
	Noise nonlinear	55 dB vs. 65 dB	2.153	0.031*
		55 dB vs. 80 dB	2.556	0.011*
Between quiet and noise	Nonlinear 80 dB	Quiet vs. Noise	2.005	0.045*

Note:* $p<0.05$, ** $p<0.01$.

The results of Wilcoxon signed rank test across presentation levels revealed that at different testing conditions, the EDI obtained at 55 dB SPL was significantly different from that obtained at 65 dB SPL and 80 dB SPL. Whereas, there was no significant difference obtained between the EDI obtained at 65 dB SPL and 80 dB SPL. The comparison of EDI obtained across the aided conditions did not reveal a significant difference. The comparison of EDI in quiet and noise revealed a significant difference only at 80 dB SPL for nonlinear condition, with EDI in quiet being higher.

Correlation of EDI and subjective findings

a. Correlation of EDI and SIS:

The EDI was correlated with SIS using Pearson's correlation. This is shown in Table 10. The results revealed a significant positive correlation between EDI and SRS for nonlinear condition at 55 dB SPL in quiet and in noise. That is, as EDI increased speech recognition scores also increased significantly. A negative correlation was also obtained in quiet linear condition at 65 dB SPL. That is, as EDI increased, speech recognition scores decreased. The scatter plot for the same is given in Figure 3. There was no correlation found between EDI and SRS in other conditions.

Table 10: *Results of Pearson's correlation between EDI and SRS*

Conditions			N	r	Significance
Level	Aided condition	SNR			
55 dB	Nonlinear	Quiet	25	0.422	0.036*
		Noise	25	0.551	0.004**
65 dB	Linear	Quiet	25	-0.472	0.017*

Note: * $p<0.05$, ** $p< 0.01$.

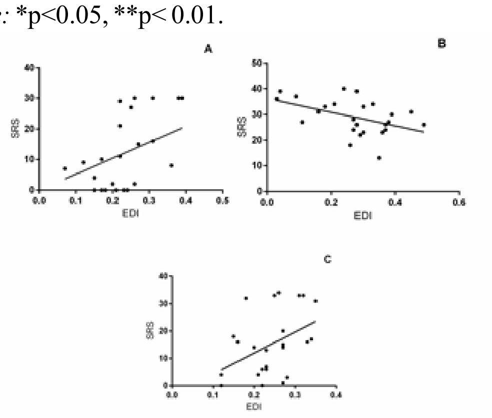

Figure 3: Scatter plot between EDI and SRS. A) Correlation of EDI and SIS in nonlinear condition at 55 dB SPL in noise; B) Correlation of EDI and SIS in linear condition at 65 dB SPL in quiet and; C) Correlation of EDI and SIS in nonlinear condition at 55 dB SPL in quiet.

b. Correlation of EDI and quality rating:

The EDI was correlated with quality rating using Spearman's correlation as shown in Table 11. The results revealed a significant correlation between EDI and quality rating at 55 dB SPL in quiet for loudness rating in linear and nonlinear condition, and also for overall impression in nonlinear condition. Whereas, at 65 dB SPL and 80 dB SPL the correlation was present in noise condition for linear algorithm under clarity, naturalness and overall impression. In addition, the correlation obtained for EDI with naturalness obtained at 65 dB SPL in noise linear condition alone was negative.

Table 11: *Results of Spearman's correlation between EDI and quality rating*

Level	SNR	Conditions Aided condition	Quality parameter	ρ	p valu
55 dB	Quiet	Linear	Loudness	0.462	0.020*
		Nonlinear	Loudness	0.505	0.010*
			Overall impression	0.484	0.014*
65 dB	Noise	Linear	Clarity	0.411	0.041*
			Naturalness	-0.478	0.016*
80 dB			Overall impression	0.514	0.009*

Note: *p<0.05, **p< 0.01.

Discussion

The aim of the present study was to find the EDI, sentence recognition scores and quality rating, and to investigate the correlation among these measures in individuals with mild to moderate sensorineural hearing loss fitted with hearing aids.

Measurement of SRS

The statistical analysis, among the aided conditions in quiet, revealed that the SRS obtained for the linear condition was significantly better when compared to SRS obtained in the nonlinear condition. Studies have reported a similar negative effect of compression algorithm (Hedrick & Rice, 2000; Marriage et al., 2005; Shanks et al., 2002) and noise reduction algorithm (Aswathi & Geetha, 2013) on speech perception when compared to linear algorithm. The probable reason given for this was the alterations in the envelope of the signal due to the digital signal processing algorithms. Shanks et al (2002) also reported a better performance in linear condition than nonlinear condition for the participants having moderate degree of hearing loss. They reasoned that this is due to the increase in gain for lower presentation levels by WDRC. This might have amplified even the low level noise and thus, reducing the performance.

In the present study, similar results were found even in the presence of noise. The presence of DNR and directionality should have enhanced the SRS in noise when tested with nonlinear condition. This did not happen as the signal along with the residual noise might be equally emphasized by the WDRC algorithm resulting in poor speech perception (Shanks et al., 2002), and hence, would have resulted in no significant improvement in SRS in noise when compared to linear condition. Further, the DNR algorithm would act on the signal to separate the signal from the noise. In the present study, a speech shaped noise was used. The speech shaped noise has a similar spectrum as that of the speech signal and hence, this would have been difficult for the algorithm to differentiate. Earlier studies using speech shaped noise have reported of a negative effect of DNR on speech perception (Aswathi & Geetha, 2012). Whereas, the studies using white noise have reported of a positive effect of DNR on speech perception (Gustafson et al., 2015; Oliveira, Lopes & Alves, 2010). This indicates that the type of noise used in the experiment plays a major role and the speech noise used in the present might have led to poor SRS.

The results of the present study reported that the SRS obtained in the quiet condition was significantly better than that obtained in the presence of noise in both linear and nonlinear conditions. This finding is in support of the results found in the earlier studies (Aswathi & Geetha, 2012; Blamey et al., 2006). This result is due to the presence of external redundancy in the quiet condition. This suggests that even the presence of all the advanced signal processing strategies cannot restore the speech in noise to its original condition.

Another finding in the present study was that a significant difference was observed across all the presentation levels in all the conditions. The scores obtained at lower levels were significantly poorer than that obtained at a higher level. Similar results have been reported in the literature (Arpitha & Manjula, 2012; Aswathi & Geetha, 2013; Jenstad & Souza, 2005). As presentation level increases, the audibility also increases and thus, the SRS.

Measurement of quality rating

The results of the rating on loudness, clarity, naturalness and overall impression revealed that there were no significant differences observed between linear and nonlinear mode in most of the conditions except few conditions. However, only at higher presentation levels, the naturalness and overall impression were rated to be significantly better in the linear mode than nonlinear mode. There have been reports of poor quality perception due to the combined effect of the algorithms at higher presentation levels (Aswathi & Geetha, 2013; Rosengard et al., 2005). At higher levels, all of these algorithms would have been activated, resulting in changes in the envelope of the speech signal.

The results also revealed that the rating for loudness, clarity, naturalness and overall impression increased significantly with an increase in the presentation level. These results are similar to that obtained for SRS. A similar kind of trend has also been reported in the earlier studies (Boike & Souza, 2000; Geetha & Manjula, 2014). According to Chasin (2007), the upper limit of the dynamic range in the hearing aid should not exceed 110 to 115 dB SPL. If this range is exceeded, the studies have reported a distortion in the perception. However, in the present study, the maximum power output used in the hearing aid was around 115 dB SPL. Hence, the hearing aid used in the present study had a good output limiting system, which, irrespective of the algorithms, would have made the signal comfortable to the listeners. This would have led to the good perceived quality at higher levels.

Further, there was no significant difference obtained in loudness rating between quiet and noise in all the conditions. However, the rating for clarity, naturalness and overall impression was higher in quiet than in the presence of noise. Irrespective of the aided conditions, the perception of loudness of speech in the presence of noise was preserved. However, in the presence of noise, the other features of quality which are essential for good quality perception were not preserved. This may be due to the distortion of the speech signal in the presence of noise. As noise decreased the external redundancy and thus, the naturalness of the perceived stimuli would have decreased. The rating on overall impression would have been the result of the rating observed across the other parameters of quality. This implies that, the quality perception in the presence of noise remains to be degraded even when the DSP algorithms are activated. This finding is also in support of the findings obtained in other studies (Aswathi & Geetha, 2012; Souza, 2002). The same trend was also observed even in SRS.

Further, the individual preferences showed that 23 out of 25 listeners preferred linear mode. This indicates that the activation of DSP algorithms does not bring about positive changes with reference to comfort, ease of listening and quality.

EDI

The EDI for sentences and passage revealed no significant difference between the linear and the nonlinear conditions. That is, the differences in the temporal envelope of the signal with (nonlinear) and without (linear) the algorithms activated were negligible. However, the earlier studies have reported a significant difference in EDI obtained between different compression parameters like time constants and compression ratio (Jenstad & Souza, 2005; Jenstad & Souza, 2007; Souza et al., 2012). These studies report of higher EDI for higher compression ratio of 10:1 and a shorter release time of 12 msec. Hence, higher compression ratio with a shorter release time yields higher EDI. However, in the present study, the compression ratio was less than 2:1 and release time was around 800 msec with a lesser compression knee point. Further, the same hearing was programmed differently for linear and nonlinear condition by enabling and disabling the algorithms. This could have resulted in similar EDI between the linear and the nonlinear condition.

Further, the processing of sentences through hearing aids is different from the processing of the shorter stimuli. The envelope variations observed for a shorter duration stimuli is more when compared to the variations observed for sentences (Jenstad & Souza, 2007; Van Tasell & Trine, 1996). This could have also been a probable reason for the differences obtained between the present study and the earlier study.

In addition, the EDI was higher at higher presentation levels, indicating that, at higher presentation levels, more changes are observed in the temporal envelope. The comparison of EDI between quiet and noise revealed no significant change. Whereas, the results of the study done by Souza et al. (2012), and Alexander and Masterson (2015) have shown a significant difference between the EDI obtained in quiet and in the presence of noise. The type of noise used in this study was International Collegium of Rehabilitative Audiology (ICRA). Whereas, a speech shaped noise was used in the present study. The speech shaped noise has spectrum similar to that of speech, and hence would have acted as an effective masker. The ICRA noise contains modulations that are repetitive at a particular frequency. The DNR algorithm used in the current was modulation based. Hence, the amount of noise reduction would have been negligible due to the similarity in the spectra of the speech signal and the noise used. Further, the alterations in the envelope of the signal could be due to the presence of noise, and hence resulting in the distortion of the original signal even in the unprocessed condition (Jenstad & Souza, 2005).

Correlation between EDI and subjective findings

The SRS and quality ratings were correlated with the EDI obtained for sentences and passage. The results revealed a significant moderate level of correlation (with 'r' ranging from 0.4 to 0.5). Further, the correlation was obtained only across few parameters among which two parameters had a positive correlation. However, the earlier studies have reported of a strong negative correlation between EDI and subjective perception (Hoover et al., 2012; Jenstad & Souza, 2005). The reason for this could be due to the variations in the type of stimuli used and the type of algorithms activated. The algorithm used in the earlier studies was only restricted to WDRC (across different compression ratios and compression time constants) (Alexander & Masterson, 2015; Hoover & Souza, 2012; Jenstad & Souza, 2005; Jenstad & Souza, 2007). Whereas, in the present study, a combination of WDRC, DNR and directionality were together activated in the nonlinear condition, and together were deactivated in the linear condition.

The stimuli used in the earlier studies were nonsense syllables (Hoover & Souza, 2012) and sentences (Alexander & Masterson, 2015; Jenstad & Souza, 2007; Walaszek, 2008). However, in the present study, sentences and passage were used. It has been reported that the envelope variations observed for a shorter duration stimuli is more when compared to the variations observed for sentences (Jenstad & Souza, 2007; Van Tasell & Trine, 1996). However, this has not been proven scientifically using EDI. Hence, the variations in the method would also have resulted in the variations in the results of the present study when compared to the earlier studies.

In the earlier studies, the processed signal was recorded using a 2cc coupler (Geetha & Manjula, 2014) and ear simulator (Walaszek, 2008). The ear canal dimension varies from individual to individual. In the present study, for obtaining more realistic outcomes, the output of the hearing aid was recorded across all the conditions for each individual. Further, the obtained EDI was used to correlate with the subjective perception. Hence, the results obtained in the present study could be generalized better. In our day to day life, we are more exposed to continuous discourse, and hence, having knowledge on the acoustical correlate (EDI) with the subjective quality perception at the discourse level is more realistic. However, the present study revealed no correlation between the EDI and quality, which may question the use of EDI to substitute the use of quality perception at the given testing conditions.

Conclusions

The presence of DSP algorithms did not bring about much change in EDI. This indicates that, the combined activation or deactivation of algorithms did not result in a quantifiable change in the envelope of the signal. This result remained same across sentences and passage. However, the presence of DSP algorithms did bring about negative changes in the perception as reflected in the SRS, quality perception and subjective preference. For all the three measures, a similar trend was observed and the performance was better with the linear mode. This indicates that even a very small change in the temporal envelope affects the perception to a greater extent. This could have led to the poor correlation between the EDI and subjective measures. The poor correlation may be due to the stimulus length and activation of many algorithms together as compared to previous research studies. Further, this study also highlights the importance of the subjective preference in hearing aid verification.

Future directions for research

1. The effect of DSP algorithms needs to be examined using multi-talker babble as the results could be easily generalized to real life.
2. The correlation of EDI and speech perception for the combined effect of the algorithm can be done using different lengths of stimuli.
3. Participants in the current study had mild to moderate sensorineural hearing loss. Individuals with greater degree of hearing loss may rely much more on the temporal cues. Hence, there is a need to quantify the changes in the temporal envelope and its effect on perception in individuals having a higher degree of hearing loss.

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