MULTI-BAND FREQUENCY COMPRESSION FOR SENSORINEURAL HEARING IMPAIRMENT

P. N. Kulkarni[1], P. C. Pandey[2], and D. S. Jangamashetti[3]

[1, 2] Department of Electrical Engineering
Indian Institute of Technology Bombay, Powai, Mumbai 400 076, India.
[3] Department of Electrical and Electronics Engineering
Basaveshwar Engineering College, Bagalkot 587 102, Karnataka, India.
E-mail: pnkulkarni@ee.iitb.ac.in, pcpandey@ee.iitb.ac.in, dsj1869@rediffmail.com

ABSTRACT

Sensorineural hearing loss is associated with widening of the auditory filters, leading to increased spectral masking and degraded speech perception. Multi-band frequency compression can be used for reducing the effect of spectral masking. The speech spectrum is divided into a number of bands and spectral samples in each of these bands are compressed towards the band center, by a constant compression factor. In the present study, we have investigated the effectiveness of the scheme for different compression factors, in improving the speech perception. Evaluation of the scheme using the modified rhyme test showed maximum improvement in recognition scores for compression factor of 0.6: about 17 % for the normal-hearing subjects under simulated hearing loss, and 6-21 % for the subjects with moderate to severe sensorineural hearing loss.

Index Terms— Sensorineural hearing loss, spectral masking, frequency compression.

1. INTRODUCTION

In cases of sensorineural hearing loss, the auditory filter bandwidth generally increases and frequency selectivity gets reduced due to increased spectral masking [1] – [3]. Hence persons with sensorineural hearing loss have difficulty in speech perception. Normal hearing persons face similar difficulty under adverse listening conditions, e.g. using a mobile phone in a noisy environment.

Earlier studies [4], [5] have shown that binaural dichotic presentation, using auditory critical bandwidth (ACB) based spectral splitting with perceptually balanced comb filters, helps in reducing the effect of spectral masking for persons with moderate bilateral sensorineural hearing impairment. For monaural hearing, effect of spectral masking may be reduced by multi-band frequency compression. In this technique, the speech spectrum is divided into a number of analysis bands, and the spectral samples in each of these bands are compressed towards the band center by a constant compression factor. Thus, the speech energy is presented in relatively narrow bands.

Earlier investigations [6] – [12] on frequency compression have shown mixed results. The main objective of these studies was to compress the speech spectrum along the frequency axis, to improve the speech perception for listeners with high frequency hearing loss. Turner and Hurtig [6] investigated proportional frequency compression, preserving the ratio between the spectral samples: each spectral sample was compressed by a constant compression factor; pitch was scaled by the same compression factor while temporal envelope and duration of the stimuli were not significantly altered. In listening tests using nonsense syllables, the recognition scores for normal-hearing subjects dropped significantly for compression factors below 0.7. Tests on 16 hearing-impaired subjects showed an average improvement of 8 % for female voice and 4.7 % for male voice. In a similar study by McDermott and Dean [7], involving 6 hearing-impaired subjects with steeply sloping high frequency loss and tests with monosyllabic word list, no significant difference in recognition scores was observed.

Sakomoto et al. [8] investigated frequency compression using PARCOR analysis-synthesis [9] and [13]. The processing involved: (i) extraction of LPC, pitch, excitation power, and voicing information (ii) nonlinear transformation of frequency scale and (iii) linear compression. The processing allowed a separate adjustment for frequency

Acknowledgement: This research is supported by TTSL – IIT Bombay Center for Excellence in Telecom (TICET).

compression (in the range of 10-90 %) for voiced and unvoiced speech. A moderate improvement in recognition was observed for 5 out of 11 subjects with severe-to-profound hearing impairment. In the study by Simpson et al. [10], frequencies above 1.6 kHz were subjected to nonlinear frequency compression. In monosyllabic word recognition tests on 17 subjects with moderate to severe sensorineural hearing loss, there were statistically significant improvements for 8 subjects, while one subject showed significant decrease. Another study [11] showed limited benefit for listeners with steeply sloping hearing loss. In the study by Reed et al. [12], the speech spectrum was compressed to fit into the reduced hearing range of the impaired ear. The technique involved segmentation, warping, dilation and time aliasing, and resynthesis. With bandwidths of 1250 Hz and 2500 Hz, the best performance obtained for frequency compressed speech was similar to the performance obtained by low pass filtering with equivalent bandwidth. Fraga et al. [16] reported improvement in recognition scores, using piecewise linear frequency compression of fricative consonants, for hearing-impairment with high frequency dead regions.

The frequency compression technique reported in [14] and [15] is based on auditory critical bands. In this technique, speech signal was compressed towards the center of each critical band along the frequency axis. The input speech was divided into frames by a Hamming window and FFT was computed on each frame. Magnitude spectrum was then compressed towards the center of each critical band along the frequency axis. Compression in the range of 0.1 to 0.9 was used. Magnitude spectrum, after piecewise frequency compression, was combined with the original phase spectrum. Speech signal was resynthesized using overlap-add method. Listening tests were conducted on hearing-impaired subjects, with fifty VCV syllables uttered by a male speaker as the test material. The best performance was obtained for compression of 0.2 – 0.4 with a modest improvement in the recognition score: 38.3 % for the processed set as against 35.4 % for the unprocessed set.

Multi-band frequency compression concentrates spectral energy towards the band centers in order to partly compensate for the increased spectral masking. The quality and intelligibility of the speech signal obtained after multi-band frequency compression depend on the bandwidth, the frequency mapping scheme, the segmentation used for analysis-synthesis, and the compression factor. The objective of our investigation is to select the most appropriate scheme of frequency mapping, bandwidth, and segmentation for analysis-synthesis, and to evaluate the effect of compression factor.

Three different frequency mapping schemes are investigated: sample-to-sample mapping, spectral sample superimposition, and spectral segment mapping. Three different bandwidths considered for investigation are (1) constant bandwidth (CB) with number of bands varying

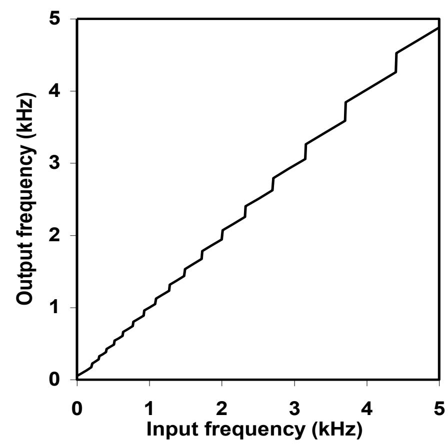

Fig. 1 Frequency mapping for multi-band compression, with auditory critical bandwidths and compression factor of 0.6

from 2 to 18, (2) 1/3 octave bandwidth, and (3) based on auditory critical bandwidth (ACB) [17]. These schemes for multi-band compression are investigated using (i) fixed-frame analysis and (ii) pitch-synchronous analysis. Optimal combination of these processing parameters is obtained by evaluating the quality of the processed speech on normal-hearing subjects under simulated hearing loss. Effectiveness of the technique with different compression factor is evaluated by conducting modified rhyme test (MRT) on normal-hearing subjects under simulated hearing loss and subjects with moderate to severe sensorineural hearing loss.

2. SIGNAL PROCESSING

Multi-band frequency compression involves three steps: (1) segmentation and spectral analysis, (2) spectral modification, and (3) resynthesis. In our scheme, the multi-band compression is carried out on the complex spectrum. The set of spectral samples in each of the predefined frequency bands are compressed towards the center of the band, by a constant compression factor as shown in Fig. 1. Since the spectral samples in each band are compressed towards the center of the band by a constant factor, and the same compression factor is used for all the bands, the processing approximately preserves the harmonic structure in case of voiced speech and randomness in case of unvoiced speech.

The frequency compression scheme was implemented using two types of segmentations for analysis-synthesis: (1) fixed-frame of 20 ms with 50 % overlap, and (2) frame length corresponding to two local pitch periods with an overlap of one pitch period. Pitch-synchronous segmentation involves voicing decision followed by determination of glottal closure instants (GCI). The GCIs are detected using Childers and Hu’s algorithm [18]. For voiced segment, analysis is carried out using analysis frame spanning from

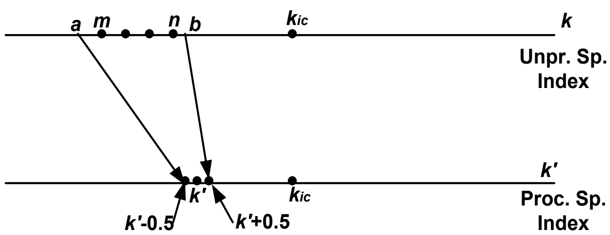

Fig. 2 Spectral segment mapping

the previous GCI to the next GCI. For unvoiced segment, analysis frame width is same as that of the last voiced frame.

The input speech signal, sampled at 10 kHz, is divided into segments with 50 % overlap. Each windowed speech segment is zero padded to the length of N and then N-point FFT is computed on it. The frequency scale is then divided into different analysis bands. The spectral samples falling in each of the bands are compressed by a constant compression factor towards the center of the corresponding band. The modified complex spectrum is converted back to time domain by N-point IFFT, and modified speech is resynthesized by overlap-add method. Investigations involving different FFT sizes showed that N = 1024 was adequate for various compression factors.

The quality and intelligibility of frequency compressed speech depend on the frequency mapping scheme employed. We have investigated three different frequency mapping techniques: (a) sample-to-sample mapping, (b) superimposition of spectral samples, and (c) spectral segment mapping [19]. Out of these three techniques, the spectral segment mapping resulted in frequency compression without any irregular variation in the spectrum. As shown in Fig. 2, the spectral segment from a to b in the unprocessed spectrum $X(k)$ contributes to the spectral sample k' in the compressed spectrum $Y(k')$. The values of a and b are given as

$$a = k_{ic} - \left[\left(k_{ic} - (k'-0.5)\right)/\alpha\right] \tag{1}$$

$$b = a + 1/\alpha \tag{2}$$

where α is the compression factor and k_{ic} is the center frequency of the i th band. Let m and n be the FFT indices of the first and the last spectral samples, respectively, falling in the segment from a to b. Index m is the lowest integer higher than a and index n is the highest integer lower than b. The processed spectrum is then given by

$$Y(k') = (m-a)X(m) + \sum_{j=m+1}^{n-1} X(j) + (b-n)X(n) \tag{3}$$

The quality and intelligibility of the resynthesized speech also depend upon the bandwidth used for segmentation of the frequency axis. With an objective of finding the optimum bandwidth for multi-band frequency compression, we have investigated three different bandwidths: constant bandwidth with number of bands varying from 2 to 18, 19 bands of 1/3-octave bandwidth in the frequency range of 70 Hz to 5 kHz, and 18 bands based on auditory critical bandwidth [17]. The spectral samples falling in each of these bands are compressed towards the band center by a constant compression factor.

A compression factor of close to one is not likely to reduce the effect of spectral masking significantly. However, a low value of compression factor is likely to introduce distortion which may offset the effect of compression. Thus we need to study the effect of (i) type of frequency mapping, (ii) bandwidth for segmentation of frequency scale, (iii) type of segmentation used for processing, and (iv) compression factor for multi-band compression.

In our earlier investigation [19], the scheme of multi-band frequency compression was optimized with respect to the frequency mapping. For assessing the effect of bandwidth and the type of processing (fixed-frame, pitch-synchronous), we conducted further listening tests on six normal hearing subjects. Mean opinion score (MOS) test was used for evaluating the quality of the frequency compressed speech. It was observed that multi-band frequency compression using spectral segment mapping based on auditory critical bandwidth and pitch-synchronous processing achieved desired compression with minimal perceptual distortion. Based on these earlier results, we selected this combination of processing parameters to investigate the effect of compression factor.

3. LISTENING TESTS

Intelligibility of the processed speech was evaluated using modified rhyme test (MRT) [20]-[23]. Two experiments were conducted. In the first experiment (Exp. I), listening tests were conducted on six normal-hearing subjects, under simulated hearing loss. Hearing loss was simulated by adding broadband noise as a masker to the processed speech, with the noise scaled for maintaining a constant SNR on a short time (10 ms) basis. In the second experiment (Exp. II), MRT was conducted on 11 subjects with moderate to severe sensorineural hearing loss. During listening tests, subjects did not wear their hearing aids and the speech signal was presented through headphones.

The test material included 50 sets of monosyllabic words of consonant-vowel-consonant (CVC) form. Each set consisted of six words with the same vowel in the middle. Either initial or final consonant remained the same, while the other consonant was different in each word. Each of the words was preceded by a carrier phrase "would you write ------". All the 300 words (i.e. 50 sets × 6 words in each set) were arranged in 6 test lists (1x, 1y, 2x, 2y, 3x, 3y) of 50 words each. The words in each test list were selected by a two level randomization process: (i) the set level 1, 2, 3, and (ii) the word level x, y within a set, in such a way that every word gets presented once.

All the sentences in the test material were recorded from a male speaker in an audiometry room, using B&K microphone model No. 2210, at 10 k samples/s with 16-bit quantization. The test was conducted using an automated test administration setup. The subject, seated in front of the computer screen, clicked the "play" button on the test window and listened the presentation from the selected test list. After each presentation, a closed set of 6 response choices was displayed on the screen and the subject selected the best match. The subject clicked the "next" button to initiate the next presentation. This procedure was repeated for all the 50 words in the selected order. The arrangement of response choices on the screen was also randomized to eliminate position bias. The response data were analyzed to get the percentage of the correct response scores for the subject under the selected processing condition.

Each subject with normal hearing responded for a total of 10800 presentations (300 words × 4 compression factors × 9 SNR values). Subjects with hearing loss were tested without adding masking noise and each of them was given a total of 1200 presentations (300 words × 4 compression factors). The test was conducted with one or two sessions (approximately one hour) per day for each subject depending on the availability and willingness of the subject. Test sessions for both the groups of subjects were spread over a span of about one month.

TABLE I

EXP. I: AVERAGE. (ACROSS THE SIX SUBJECTS) PERCENTAGE RECOGNITION SCORES (R.S.) FOR UNPROCESSED (Unp.) SPEECH AND FREQUENCY COMPRESSED SPEECH WITH DIFFERENT COMPRESSION FACTORS. STANDARD DEVIATIONS ARE IN PARENTHESES

SNR (dB)	Unp.	Compression factor		
		0.8	0.6	0.4
∞	95.7 (2.1)	94.3 (3.1)	88.4†† (3.0)	83.7†† (2.1)
6	90.8 (3.4)	89.0† (3.0)	82.2†† (3.7)	77.4†† (4.5)
3	86.8 (5.1)	83.4 (4.9)	79.5† (2.9)	77.9†† (3.8)
0	74.9 (5.8)	78.8 (4.3)	84.2†† (3.7)	77.6 (4.1)
-3	71.8 (3.4)	74.7 (6.3)	81.6†† (4.5)	75.1† (5.0)
-6	66.1 (3.6)	69.9 (7.3)	78.8†† (4.3)	73.0† (5.3)
-9	59.6 (4.6)	65.1† (6.5)	76.1†† (4.3)	68.0†† (4.4)
-12	52.2 (5.3)	58.4* (5.6)	69.4†† (3.7)	60.4†† (5.1)
-15	44.8 (4.2)	52.7† (3.8)	62.0†† (2.1)	48.7* (5.7)

* $p < 0.05$, † $p < 0.01$, †† $p < 0.005$

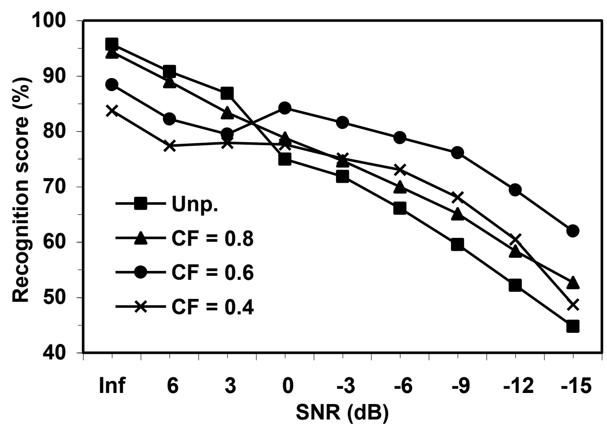

Fig. 3 Exp. I: Recognition score (%) as a function of SNR

4. RESULTS AND DISCUSSION

Results of Experiment I, conducted on six normal-hearing subjects, with hearing loss simulated with different levels of broadband masking noise, are summarized in Table I. It shows the average (across the six subjects) recognition score (%) for unprocessed speech and the speech processed with the three compression factors. The standard deviations are given in parentheses. Statistical significance of the increase in the scores due to processing was tested by paired (processed vs unprocessed) one-tailed t-test and the significance levels are also indicated in the table. Figure 3 gives a plot of the averaged percentage recognition score as a function of SNR for unprocessed speech and speech processed with the three compression factors. The plots for compression factor of 0.6 and 0.4 exhibit non-monotonicity, but the deviations from monotonicity are not statistically significant. It can be seen that even though a moderate decrease in the recognition score was observed at SNR values of ∞, 6, and 3 dB, an increase in recognition score was observed for SNR values lower than 0 dB. The improvements with compression factor of 0.8 and 0.4 are relatively smaller than those with the compression factor of 0.6.

It can be observed from Table I that for compression factor of 0.6 the improvement in the recognition scores for SNR values less than 3 dB are highly significant ($p < 0.005$). For SNR < -6 dB, there is an average (across the subjects) increase of 17 % ($p < 0.005$) in recognition scores and an average relative improvement of 33 %. For unprocessed speech, the recognition score is about 60 % for SNR of -9 dB. For the speech signal processed with compression factor

TABLE II

EXP. II: PERCENTAGE RECOGNITION SCORES (R.S.) FOR ELEVEN HEARING-IMPAIRED SUBJECTS AND THEE COMPRESSION FACTORS. AVG. R.S .= AVERAGE RECOGNITION SCORE (%), S.D. = STANDARD DEVIATION, R.I. = REL. IMPROVEMENT, p = SIGNIFICANCE LEVEL (ONE TAILED) FOR PAIRED T-TEST (UNPROCESSED VS. PROCESSED, n = 11, AND df = 10).

Sub.	Unp.	Comp. factor		
		0.8	0.6	0.4
KNR	63.3	64.7	78.3	58.3
MNR	51.0	56.0	69.7	67.0
PKR	46.7	54.7	65.7	60.3
PAL	64.3	69.0	85.0	67.0
PAR	62.3	64.7	80.7	65.7
PPR	70.7	72.3	80.0	67.3
PEL	70.3	62.5	91.7	63.7
RGR	79.7	48.7	55.7	46.7
RJL	67.7	70.0	81.3	72.7
SSL	49.0	53.3	61.7	54.0
SSR	63.3	60.3	77.7	63.0
Avg. R.S.	59.8	61.5	75.2	62.3
s.d.	9.0	7.6	10.8	7.2
p		0.121	0.000	0.142
Avg. R.I.		-4.7	22.6	-0.5
Min. R.I.		-11.1	12.1	-9.5
Max. R.I.		17.1	40.7	31.4

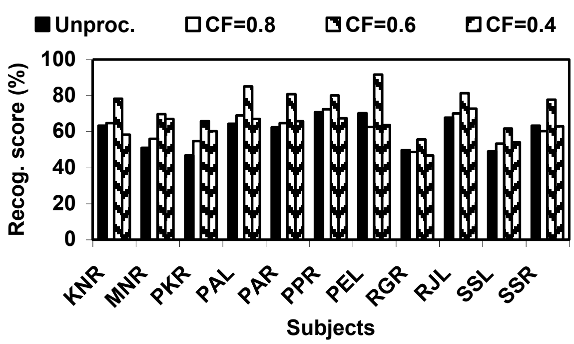

Fig. 4 Recognition score (%) for unprocessed and frequency compressed with different compression factors.

of 0.6, the same recognition score is observed for SNR of -15 dB. Thus we can say that for recognition score of about 60 %, the compression factor of 0.6 give an SNR advantage of approximately 6 dB.

Two-factor analysis of variance (ANOVA) was conducted on the recognition scores, with subjects (n = 6, df = 5) and processing (unprocessed and processed with the three compression factors, n = 4, df = 3) as the sources of variation. These results show a statistically significant effect ($p < 0.001$) for compression factor. The effect of subject as the source of variation was significant ($p < 0.02$) for all SNR values except -12 and -15 dB. Similar analysis on percentage recognition score (averaged across the six subjects) with compression factor and SNR as the sources of variations showed significant effect for both the sources of variations with $p < 0.001$ for SNR and $p < 0.03$ for compression factor.

In Experiment II, the listening tests were conducted on 11 subjects with moderate to severe sensorineural hearing loss. Table II shows the percentage recognition score for each subject. Average (across the 11 subjects) percentage recognition score and standard deviation, average percentage relative improvement with respect to the unprocessed, and p value for paired one-tailed t-test are also given in the Table II. A plot of the recognition scores for the individual subjects is shown in Fig. 4.

For compression factor of 0.8, eight out of eleven subjects showed moderate improvement in the recognition scores in the range of 2 – 8 %. For compression factor of 0.4, six out of eleven subjects reported an improvement in the recognition score in the range of 3 – 16 %. For compression factor of 0.6, all the subjects showed an improvement in the recognition score in the range of 6 – 21 % ($p < 0.001$), and relative improvement in the range of 12 – 41 %.

Two-factor analysis of variance (ANOVA) was conducted on the recognition scores with subjects (n = 11, df = 10) and compression factors (n = 4, df = 3) as the sources of variation. These results showed a statistically significant effect ($p < 0.001$) for both the sources of variation.

5. CONCLUSION

Listening tests were conducted to assess the usefulness of the multi-band frequency compression scheme, using (i) spectral segment mapping, (ii) auditory critical bandwidth based segmentation of frequency axis, and (iii) pitch-synchronous processing. MRT was conducted on six normal-hearing subjects, with hearing loss simulated by adding broadband noise as a masker to the processed speech signal. Best results were obtained for speech processed with compression of 0.6. For SNR values below -6 dB, there was an average improvement of 17 % ($p < 0.005$) in recognition scores and an average relative improvement of 33 %. For recognition score of 60 %, the processing showed an SNR advantage of about 6 dB, indicating that the processing helps in improving the speech intelligibility in the presence of masking noise.

Further evaluation was carried out for speech intelligibility on 11 subjects with moderate to severe sensorineural hearing loss. Even though only small improvements in recognition score were observed for compression factor of 0.8 and 0.4, an improvement of 6 – 21 % in the recognition score, and a relative improvement of 12 – 41 %, were observed for compression factor of 0.6.

It may be concluded that the processing of the speech signal with multi-band compression improved speech perception and the maximum improvement was observed for compression factor of 0.6. The technique needs to be further evaluated using different types of speech material and a larger number of subjects.

6. REFERENCES

[1] B. R. Glasberg and B. C. J. Moore, "Auditory filter shapes in subjects with unilateral and bilateral cochlear impairments," *J. Acoust. Soc. Am.*, 79(4), pp. 1020-1033, 1986.

[2] A. E. Carney and D. A. Nelson, "An analysis of psychophysical tuning curves in normal hearing and pathological ears," *J. Acoust. Soc. Am*., 73(1), pp. 268-278, 1983.

[3] J. R. Dubno and D. D. Dirks, "Auditory filter characteristics and consonant recognition for hearing-impaired listeners*," J. Acoust. Soc. Am*., 85(4), pp. 1666-1675, 1989.

[4] A. N. Cheeran and P. C. Pandey, "Speech processing for hearing aids for moderate bilateral sensorineural hearing loss," *Proc. IEEE ICASSP*, 2004, pp. IV-17-20.

[5] P. N. Kulkarni, P. C. Pandey, and D. S. Jangamashetti, "Perceptually balanced filter response for binaural dichotic presentation to reduce the effect of spectral masking," *J. Acoust. Soc. Am*., 120(5), pp. 3253, 2006.

[6] C. W. Turner and R. R. Hurtig, "Proportional frequency compression of speech for listeners with sensorineural hearing loss," *J. Acoust. Soc. Am.*, 106(2), pp. 877-886, 1999.

[7] H. J. McDermott and M. R. Dean, "Speech perception with steeply sloping hearing loss," *Brit. J. Audiology*, 34(6), pp. 353-361, 2000.

[8] S. Sakamoto, K. Goto, M. Tateno, and K. Kaga, "Frequency compression hearing aid for severe-to-profound hearing impairment," *Auris Nasus Larynx*, 27, pp. 327-334, 2000.

[9] S. Sekimoto, and S. Saito, "Nonlinear frequency compression speech processing based on PARCOR analysis-synthesis technique," Ann *Bull Rilp*, 14, pp. 65-72, 1980.

[10] A. Simpson, A. A. Hersbach, and H. J. McDermott, "Improvements in speech perception with an experimental nonlinear frequency compression hearing device," *Int.J. Audiology,* 44, pp. 281-292, 2005.

[11] A. Simpson, A. A. Hersbach, and H. J. McDermott, "Frequency-compression outcomes in listeners with steeply sloping audiograms", *Int. J. Audiology,* 45: 619-629, 2006.

[12] C. M. Reed, B. L. Hicks, L. D. Braida, and N. I. Duriach, "Discrimination of speech processed by low pass filtering and pitch invariant frequency lowering," *J. Acoust. Soc. Am*., 74(2), pp. 409- 419, 1983.

[13] F. Itakura, S. Saito, Y. Koike, H. Sawabe, and M. Nishikawa, "An audio response unit based on partial correlation," *IEEE Trans. Communication,* 20(4), pp. 792-797, 1972.

[14] K. Yasu, K. Kobayashi, K. Shinohara, M. Hishitani, M., T. Arai, and Y. Murahara, "Frequency compression of critical band for digital hearing aids," *Proc. China-Japan Joint Conf. Acousticstics*, 2002, pp. 159-162.

[15] T. Arai, K. Yasu, and N. Hodoshima, "Effective speech processing for various impaired listeners," *Proc. 18th Int. Congress Acoustics* (*ICA)*, 2004, pp. 1389 - 1392.

[16] F. J. Fraga, L. P. C. S. Prates, M. C. M. Iorio, "Frequency compression/transposition of fricative consonants for the hearing impaired with high-frequency dead regions," *Proc. Interspeech 2008*, pp. 2238-2241.

[17] E. W. Zwicker, "Subdivision of audible frequency range into critical bands (Freqenzgruppen*)," J. Acoust. Soc. Am*., 33(2), pp. 248, 1961.

[18] D. G. Childers and T. H. Hu, "Speech synthesis by glottal excited linear prediction," *J. Acoust. Soc. Am*., 96(4), pp. 2026-2036, 1994.

[19] P. N. Kulkarni and P. C. Pandey "Frequency mapping for multi-band frequency compression for improving speech intelligibility," *Proc. 14th National Conference on Communications,* Mumbai, India, 2008, pp. 437-441.

[20] A. S. House, C. E. Williams, M. H. L. Hecker, and K. D. Kryter, "Articulation testing methods: consonantal differentiation with closed-response set," *J. Acoust. Soc. Am*. 37, pp. 158-166, 1965.

[21] K. J. Kreul, J. C. Nixon, K. D. Kryter, D. W. Bell, J. S. Lang, and E. D. Schubert, "A proposed clinical test of speech discrimination," *J. Speech Hearing Res*., 11, pp. 536-552, 1968.

[22] W. Yang and M. Hodgson "Auralization study of optimum reverberation times for speech intelligibility for normal and hearing-impaired listeners in classrooms with diffuse sound fields," *J. Acoust. Soc. Am*., 120, pp. 801-807, 2006.

[23] ANSI, Methods for measuring the intelligibility of speech over communication systems. *Revised Standards Report ANSI S3.2-1989*, American Standards Association, New-York, 1989.