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## RESEARCH ARTICLE **CONSERVERS** OPEN ACCESS

# **Evaluation of Wavelet based filter algorithms to improve the performance of Hearing with simulated hearing loss**

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# **ABSTRACT**

Traditionally this experimentation is accomplished on hearing impaired subjects but they may be time consuming and may cause fatigue in elderly people. So, preliminary evaluations of the developed algorithms were conducted on five normal people with simulated hearing loss, by adding a broad band noise to the speech signal with signal to noise ratio of 3dB, 0dB, -3dB,-6dB and -9dB.The objective of the experimental analysis was to assess the usefulness of the developed algorithms. The wavelet based filter algorithms using Daubechies, Symlet and Biorthogonal wavelet families were developed in MATLAB. Further to randomise the tests, 15 syllables without meaning were considered in vowel consonant vowel (VCV) order. Results indicated that response time for processed speech signal was significantly lower than that for unprocessed one, the recognition scores decreased with decrease in signal to noise ratio, the relative information transmitted was near perfect with unprocessed speech and improved with processed speech for higher value of signal to noise ratio.

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*Keywords* **-** Sensorineural, VCV, Dichotic, Binaural, Spectral masking.

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#### **I. INTRODUCTION**

The sensorineural loss is characterized by an increase in threshold of hearing, compression in dynamic range of hearing, degradation of temporal resolution with increase in temporal masking and degraded frequency selectivity with an increase in spectral masking [3],[7].One of the possible reasons for the malfunction in sensorineural loss is due to spread of spectral masking along the cochlear partitions [6],[7].

It is believed to take place on account of filtering inside the cochlea, also denoted as critical bandwidth. The auditory filters are the ones that separate one sound from another. The auditory filters are broader than normal in increased spectral masking. [1]

The splitting of information in speech signal for presenting signals to the ears, in some sort of complimentary fashion, to provide the relaxation for sensory cells of the basilar membrane, may help in reducing the effect of increased masking and

thereby improve the speech reception in cases of bilateral sensorineural hearing impairment [1], [2].

Human ear portrays as utilizing wavelet transform while analyzing sound, at least in the very first stage [11]. The wavelet transform, utilized in signal processing has an ability to co-exist in timefrequency (or time-scale) representation of signals as it uses a variable-width window (narrow at high frequencies and wide at low frequencies) [14].

During wavelet investigation, a bank of band pass filters is considered. Abhjit Karmarkal et al.[12] have suggested a criterion to choose the optimal wavelet packet created on the Zwicker's model critical band structure [9]. Researchers acquired optimal wavelet-packet(WP) tree for various sampling frequencies and results were correlated with other CB motivated WP trees. M. T. Kolte et. al. [3] suggested modified wavelet packets algorithm utilizing Symlet family. The creator's guaranteed that recognition scores for processed scheme of wavelet packets were improved by 3.33% to 22.23%.and the response time were reduced. The proposed system was evaluated utilizing VCV

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speech material. Author concluded that outcomes got by proposed strategy were better than comb filter results.

In present work, wavelet based filter algorithms of Daubechies, Symlet and Biorthogonal wavelet families (db4, sym9 and bior2.4) were developed in MATLAB. In these, eight bands are made based on auditory filters of quasi octave bandwidth. Four alternate bands are combined for even-odd dichotic presentation. The inverse wavelet packet transform was utilized to synthesize speech components from the wavelet packet representation. To synthesize the speech component, wavelet coefficients were used. Listening tests on five normal people were conducted using fifteen syllables without meaning (i.e. vowel consonant vowel (VCV) order).

The paper is prepared into five sections. Section I presents the need of the proposed system and also reviews the various techniques proposed by the different researchers to overcome different problems related to the hearing impaired using wavelet transform. Section II discusses the methodology used to design the wavelet based filter algorithms. Section III includes listening tests. Section IV shows experimental results and discussion. Section V concludes the paper.

## **II. METHODOLGY**

The logarithmic frequency resolution is obtained from discrete wavelet transform (DWT). Low frequencies have narrow bandwidth while high frequencies have wider bandwidth. The segmentation of higher frequencies into narrower bands is allowed by wavelets packets [4],[15]. For speech analysis, wavelet packets (WP) prove to be an efficient tool. The selection of decomposition tree followed by selection of filters for every decomposition level of the tree is involved in the designing of wavelet packets. After the selection of decomposition tree, the immediate step involves selection of appropriate wavelet filters for every decomposition level of the tree. Different timefrequency resolution exists at each level. In our work, we have not used down sampling operation during the decomposition. We have used a frequency range criterion to select the most suitable decomposition of a given speech signal. This means we look at each node of the decomposition tree and quantify the frequency range of the information to be gained by performing each split.

In our work, we use discrete wavelet transform and wavelet packet at various levels of decomposition to develop optimized wavelet packets as per the frequency criterion as shown in Figure 1 and Figure 2. MATLAB software was used to

develop codes for wavelet filter algorithms using Daubechies, Symlet and Biorthogonal wavelet families. Figure 2 shows wavelet filter tree structure. The processing algorithms were developed as spectral splitting with this wavelet filter structure based on eight frequency bands as the performance by hearing-impaired subjects saturated around eight channels, while performance by normal-hearing subjects sustained to 12–16 channels with higher background noise [10].



**Figure 1:** Decomposition tree structure upto 4 level for a)DWT b)WP



**Figure 2:** Decomposition tree structure for Wavelt filter.

Table 1 shows the entire eight bands in alternate fashion for even-odd index with center and pass band frequency for each band in KHz.

Wavelet filters were developed using the pseudo algorithm as follows:

- Read speech signal  $x(n)$  (input) having length N.
- Decompose  $x(n)$  up to level 4 using respective wavelet packet as directed in Figure 1.
- Construct the optimized wavelet packet tree by rejoining 11, 12, 13, 14 and 9, 10, 5, 6 nodes. Thus, optimized tree will have only eight nodes as shown in Figure 1.

- Reconstruct the optimized wavelet tree (selectively) to obtain two output signals - one for right ear and other for left ear.
- In optimized tree, the approximate coefficients nodes numbered 15, 17, 9, and 5 are made zero while keeping the detail coefficients nodes as it is.

## **III. LISTENING TESTS**

Five normal hearing people (VG: M 36, SA: M 22, MF: M 19, MJ: M 36, BB: M 71) participated in the listening tests. These listening tests were conducted in an acoustically isolated room. The loss was simulated by adding broadband noise to the speech stimuli with five different SNRs. SNR conditions used were 3dB, 0dB, -3dB, -6dB and -9dB. The wavelet based filter algorithms (db4, sym9 and bior2.4) were implemented in MATLAB. Listening tests were carried out for finding the confusion among the set of fifteen English consonants, without meaning were considered in vowel consonant vowel (VCV) order. To make these tests user friendly MATLAB based GUI was developed with a provision to manually enter the SNR condition. The final results were collected in confusion matrix to evaluate the response times, recognition scores and information transmission analysis.

## **IV. LISTENING TESTS RESULTS**

This section includes results of listening tests for response time, recognition score, relative improvement in recognition score and information transmission analysis results for unprocessed and processed speech with all SNR conditions.

#### 4.1 RESPONSE TIME

Response time is the time interval between speech materials presented dichotically to subjects and the response given by subjects. The response time for the processed and unprocessed speech signal for various SNR conditions is presented in the Table 2 and its graphical presentation is provided in Figure 3. Table 3 shows the average response time of three algorithms for the processed and unprocessed signals. From Table 2, it is observed that as SNR reduces, the response time increases for unprocessed and processed signals. For SNR conditions at -9dB, -6dB, -3dB, 0dB, 3dB the corresponding decrease in average response time were 4.29sec, 3.73sec, 3.72 sec, 3.22 sec and 3.05sec for processed signals and similarly for unprocessed signal the values were 4.40 sec, 3.83 sec, 3.78 sec, 3.60 sec and 3.28 sec.The average response time increased with increase in the level of masking noise for all subjects. It was observed that response time

for processed speech was significantly lower than that for unprocessed one.



**Figure 3:** Response time

## 4.2 RECOGNITION SCORE

The recognition scores of individual subjects, average of recognition scores and average relative improvement across five subjects at different SNR conditions is shown in Table 4. The improvements in processing with all the three algorithms were highly significant when compared to unprocessed ones. The average relative increase for db algorithm was 1.38%, 1.83%, 2.14%, 3.51%, and 4.23% with SNR conditions 3, 0, 3, -6 and -9dB respectively. Similarly for bior algorithm, the average relative increase were 2.60%, 1.27%, - 0.37%, 0.43% and 3.28% with SNR conditions 3, 0, -3, -6 and -9dB respectively and for sym algorithm, these values were 1.93%, 1.60%, -0.26%, 1.11% and 3.28% with SNR conditions 3, 0, -3, -6,and -9dB respectively.

For all the subjects, as the SNR decreases (masking noise level increases), the recognition score generally decreases. It is observed that the improvements due to algorithms are more for higher levels of masking noise and was found for db algorithm at -9dB SNR condition.

## 4.3 INFORMATION TRANSMISSION ANALYSIS

The combined confusion matrix with consistency in score was subjected to information transmission analysis for every SNR condition. Table 5 (a to h) gives the relative information transmitted for the different consonantal features along with averaged over five subjects and average relative improvement across five subject with different SNR conditions.

For the overall feature with decrease in SNR condition related to unprocessed speech there was degradation in the overall information transmitted. For SNR conditions of 3, 0, -3, -6 and - 9dB the values of overall information transmitted were for db algorithms 97.30%, 97.52%, 96.44%, 94.42%, 91.07% similarly, the values were 98.33%, 97.56%, 96.74%, 95.23% and 93.05% for bior and

97.65%, 97.475, 96.36%, 95.30% and 92.72% for sym respectively.

From the analysis, it was observed that the algorithms improved the perception of most of the consonantal features with different SNR conditions. The relative information transmitted is near perfect with unprocessed speech and improves with processed speech for higher value of SNR. With lower values of SNR, the relative information transmitted with unprocessed speech decreases and improvements are observed with the processed speech. However, most of the subjects indicated the maximum improvement for the duration, frication and manner features. For lower values of SNR, relatively better improvement was observed for place feature. The reception of the place feature is related to frequency resolving capacity of the auditory processing so it can be implied that the algorithms have reduced the effect of spectral masking.

Overall: With decrease in SNR condition related to unprocessed speech there was degradation in the overall information transmitted. For SNR conditions of 3, 0, -3, -6 and -9dB the values of overall information transmitted were for db algorithms 97.30%, 97.52%, 96.44%, 94.42%, 91.07% similarly, the values were 98.33%, 97.56%, 96.74%, 95.23% and 93.05% for bior and 97.65%, 97.47%, 96.36%, 95.30% and 92.72% for sym respectively.

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# **V. CONCLUSION**

The proposed scheme shows to be the desired alternative to test the applicability of developed algorithm for hearing aid without participation of hearing impaired persons.

The results of listing tests show that the recognition scores decreased with decrease in SNR.

It was observed that for a particular level of masking noise, the score for processed speech was significantly higher than that for the unprocessed one. It was also observed that the relative improvements due to processing were more for higher levels of masking noise (- 9 dB SNR), i.e. higher levels of simulated sensorineural loss. The average response time increased with increase in the level of masking noise for all subjects. It was observed that response time for processed speech was significantly lower than that for unprocessed one. The relative information transmitted is near perfect with unprocessed speech and improves with processed speech for higher value of SNR. However, many subjects indicated the maximum improvement for the duration, frication and manner features. Relatively better improvement was observed for place feature. The reception of the place feature is related to frequency resolving capacity of the auditory processing so it can be implied that the algorithms have reduced the effect of spectral masking. Finally, it can be concluded that in noisy environment the load on speech perception process is reduced and speech perception by normal people gets improved by wavelet based algorithms having binaural dichotic.

# **REFERENCES**

- [1]. B. C. J. Moore, An Introduction to Psychology of Hearing, London, Academic, 4th ed. 1997, pp. 89-140
- [2]. L. R. Rabiner and R. W. Schafer, Digital Processing of the Speech Signals, Englewood Cliffs, NJ: Prentice Hall, 1978, pp.35-53.
- [3]. Mahesh T. Kolte, D.S.Chaudhari, Evaluation of speech processing schemes to improve perception of sensorinural hearing impaired, Current Science, Vol.98, No 5, 2010.
- [4]. Jayant J. Chopade and Dr. N. P. Futane, Wavelet based scheme to improve performance of hearing under noisy environment, International Journal of Computer Applications (0975 – 8887) Volume 130– No.6, Nov 2015.
- [5]. D. S. Chaudhari and P. C. Pandey, Dichotic presentation of speech signal using critical filter bank for bilateral sensorineural hearing impaired, Proceedings of 16th International Conf. on Acoustics, Seattle, Washington, 1998, vol. 1, pp. 213-214.
- [6]. Chopade, J.J. and Futane, N.P., 2016. Design of Optimized Wavelet Packet Algorithm to Improve Perception of Sensorineural Hearing Impaired. Journal of Signal and Information Processing, 7(01), p.18.

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- [7]. P. N. Kulkarni and P.C. Pandey, Optimizing the comb filters for spectral splitting of speech to reduce the effect of spectral masking, International conference on signal processing, Communication and Networking, Chennai, India, Jan 2008 pp.69-73.
- [8]. Lunner, T., and Hellgren, J. A digital filter bank hearing aid- design, implementation, and evaluation, Proc. in IEEE ASSP, vol. 5, 3661–3664.
- [9]. E. W. Zwicker, Subdivision of audible frequency range into critical bands (Freqenzgruppen), J. Acoust . Soc. Am.,Vol. 33, pp. 248, 1961.
- [10]. Baskent and Deniz, Speech recognition in normal hearing and sensorineural hearing loss as a function of the number of spectral channels, J. Acoust. Soc. Am., 2006, vol. 120(5), pp. 2908–2925.
- [11]. Daubechies, I., Ten Lectures on Wavelets, Philadelphia: SIAM, CBMS- NSF, Regional Conference in Applied Mathematics 61.
- [12]. Abhijit Karmarkal, Arun Kumar and R.K.Patney (2007) Design of Optimal Wavelet Packet Trees Based on Auditory Perception Criterion IEEE SIGNAL PROCESSING LETTERS, VOL. 14, NO. 4
- [13]. J. R. Dubno and A. B. Schaefer, Comparison of frequency selectivity and consonant recognition among hearing-impaired and masked normal-hearing listeners, J. Acoust. Soc. Am., 1992, vol. 91(4), pp. 2110–2121.
- [14]. Nivin Ghamry, FPGA Implementation of Hearing Aids using Stationary Wavelet-Packets for Denoising, International Journal

of Computer Applications (0975 – 8887) Volume 69– No.15, May 2013.

- [15]. Rao, R. M. and Bopardikar, A. S., Wavelet Transform Introduction to Theory and Applications (Addison Wesley Longmman Pte. Ltd., Delhi).
- [16]. Miller, G. A., and Nicely, P. E. "An analysis of perceptual confusions among some English consonants," J. Acoust. Soc. Am. vol. 27(2), 338–352, 1955.
- [17]. CHABA, Speech-perception aids for hearingimpaired people: Current status and needed research, J. Acoust. Soc. Am. vol. 90, 637– 683, 1991.





	<b>US</b>					<b>PS</b>					
Wavel ets	SNR: 3dB	SNR: 0dB	$SNR: -$ 3dB	$SNR: -$ 6dB	<b>SN</b> $R: -$ 9dB	SNR: 3dB	SNR: 0dB	$SNR: -$ 3dB	$SNR: -$ 6dB	$SNR: -$ 9dB	
db	3.18	3.80	3.24	3.18	4.2 $\Omega$	2.95	2.90	2.95	3.58	4.00	
bior	3.18	3.80	3.24	3.18	4.2 $\overline{0}$	3.20	3.13	2.68	2.50	3.66	
sym	3.18	3.80	3.24	3.18	4.2 $\theta$	3.22	2.35	3.62	3.93	4.30	
db	2.70	3.23	3.48	4.02	4.5 $\mathbf{Q}$	2.35	3.12	4.61	3.70	4.57	
bior	2.70	3.23	3.48	4.02	4.5 9	2.70	3.24	3.36	3.65	4.32	
sym	2.70	3.23	3.48	4.02	4.5 9	2.12	3.20	3.38	3.65	4.56	
db	2.88	3.20	3.73	3.17	3.6 3	2.20	2.27	2.61	2.66	3.46	
bior	2.88	3.20	3.73	3.17	3.6 3	2.13	2.36	4.43	2.51	4.20	
sym	2.88	3.20	3.73	3.17	3.6 3	2.21	2.66	3.27	3.51	3.45	
db	3.38	3.45	3.71	3.80	4.1 3	3.30	3.52	3.62	3.68	3.84	
bior	3.38	3.45	3.71	3.80	4.1 3	3.11	3.25	3.72	3.78	3.92	
sym	3.38	3.45	3.71	3.80	4.1 3	3.71	3.82	3.65	4.20	3.90	
db	4.24	4.33	4.73	4.97	5.4 $\overline{4}$	4.09	4.12	4.35	4.80	5.10	
bior	4.24	4.33	4.73	4.97	5.4 $\overline{4}$	4.17	4.26	4.72	4.92	5.40	
sym	4.24	4.33	4.73	4.97	5.4 $\overline{4}$	4.26	4.13	4.85	4.92	5.60	

**Table 2** Response time for Normal People for different SNR

## **Table 3:** Average Response time





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$SNR: -3dB$	US	100.00	98.66	100.00	92.00	77.33	93.60	
	Pr-db	100.00	98.67	100.00	100.00	80.00	95.73	2.14
	Pr-bior	100.00	96.67	100.00	100.00	69.49	93.23	$-0.37$
	Pr-sym	100.00	96.67	100.00	100.00	70.00	93.33	$-0.26$
	US	98.66	97.33	97.00	93.33	66.66	90.60	
$SNR: -6dB$	Pr-db	98.67	98.67	98.67	98.53	76.00	94.11	3.51
	Pr-bior	100.00	96.61	96.67	98.53	63.33	91.03	0.43
	Pr-sym	100.00	95.00	96.67	98.53	68.33	91.71	1.11
	US	90.66	93.33	96.00	94.66	58.67	86.66	
$SNR: -9dB$	Pr-db	93.24	94.67	97.33	93.24	76.00	90.90	4.23
	Pr-bior	96.67	100.00	96.67	93.06	63.33	89.94	3.28
	Pr-sym	98.33	95.00	98.33	93.06	65.00	89.94	3.28

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**Table 5.** Relative information transmitted (in %) for (a). overall, (b). continuance, (c). duration, (d). frication, (e). manner, (f). nasality, (g). place and (h).voicing

	Subject		VG	<b>SA</b>	MF	MJ	<b>BB</b>	Avg	Avg. R.I.
a) Overall		<b>US</b>	100.00	100.00	97.65	96.21	88.55	96.48	
	SNR: 3dB	Pr-db	100.00	100.00	100.00	100.00	86.52	97.30	0.82
		Pr-bior	100.00	100.00	100.00	100.00	91.63	98.33	1.85
		Pr-sym	100.00	100.00	100.00	100.00	88.24	97.65	1.17
	SNR: 0dB	<b>US</b>	100.00	97.34	100.00	98.67	82.26	95.65	
		Pr-db	100.00	97.07	100.00	100.00	90.54	97.52	1.87
		Pr-bior	100.00	100.00	100.00	100.00	87.82	97.56	1.91
		Pr-sym	100.00	100.00	100.00	100.00	87.34	97.47	1.82
	$SNR: -3dB$	<b>US</b>	100.00	98.67	100.00	92.63	81.24	94.51	
		Pr-db	97.26	93.04	100.00	100.00	91.89	96.44	1.93
		Pr-bior	100.00	96.92	100.00	100.00	86.76	96.74	2.23
		Pr-sym	100.00	96.92	100.00	100.00	84.86	96.36	1.85
	$SNR: -6dB$	<b>US</b>	100.00	98.46	100.00	96.79	79.18	94.89	
		Pr-db	98.54	92.26	100.00	98.60	82.72	94.42	$-0.46$
		Pr-bior	100.00	97.03	96.92	98.60	83.58	95.23	0.34
		Pr-sym	100.00	95.54	96.92	98.60	85.45	95.30	0.42
	SNR: -9dB	<b>US</b>	91.99	94.69	97.08	95.10	75.96	90.96	
		Pr-db	93.16	88.52	98.46	93.78	81.44	91.07	0.11
		Pr-bior	97.65	100.00	96.92	93.78	76.92	93.05	2.09
		Pr-sym	98.46	95.38	98.46	93.78	77.51	92.72	1.76





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