Combined Effect of Adaptive Wiener Filter and Spectral Splitting of Speech Signal in Improving Speech Intelligibility for Hearing Impaired People

Rajani S. Pujar

***Abstract*—The goal of this research is to look into the combined effect of an adaptive Wiener filter, frequency splitting, and amplitude compression on speech intelligibility for hearing impaired people in adverse listening environments. The adaptive Wiener filter is based on the adaptation of the filter frequency response from sample to sample depending on the voice signal statistics (mean and variance).The filter bank approach was modified to perform spectral splitting of the voice signal in order to lessen the influence of spectral masking. To address the issue of hearing loss, amplitude compression with a compression factor of 0.6 was used. As a result, the combined strategy aids in enhancing speech intelligibility for hearing-impaired individuals in noisy environments. To measure the intelligibility of processed speech, hearing-impaired subjects (moderate SNHL) were given the modified rhyming Test (MRT). The test procedure consists of 300 phrases. Each phrase is composed of consonant-vowel-consonant words (CVC). When compared to an adaptive Wiener filter, the results suggest that processing with a composite scheme improved speech intelligibility. With a lower SNR value of -6 dB, a maximum improvement in speech recognition score of 32.935 % was recorded.**

***Index Terms*—Adaptive weiner filter, amplitude compression, frequency splitting, modified rhyme test.**

I. Introduction

As the signal-to-noise ratio (SNR) decreases and listening effort increases, speech intelligibility decreases. Hearing impaired people are more sensitive to noise in low-SNR environments [1]. For them, voice intelligibility is a significant issue. This requires more listening effort than normal hearing individuals. As a result, the purpose of hearing aid signal processing is to raise the SNR so that sounds are easier to understand rather than just amplify them and make them audible again. The research of noise reduction in hearing aids is thus critical, and numerous ways have been proposed in the past [2].The purpose of noise reduction strategies is to minimise background noise and increase speech intelligibility or listening comfort in difficult acoustical situations by boosting SNR. Noise reduction strategies for hearing devices include spectral subtraction, wiener filter, and beam former techniques [3]-[5]. The study [6] focused on the impact of noise reduction on speech intelligibility as a function of SNR. The effects of SNR on speech intelligibility for three de – noising techniques were explored: spectral subtraction (SS), minimum mean square error spectral estimation (MMSE), and subspace enhancement algorithm (SSA). Work presentedin [7] discusses an adaptive noise filtering technique. The algorithm was created for a BTE hearing aid provided with two similar forward faced directional microphones arranged in an end fire configuration. The processing was done on PC with DSP processor. Speech intelligibility testing in a realistic scenario was performed on users with normal hearing and hearing loss (moderate hearing loss). Open set words and sentences are included in the test materials. A significant 5 dB improvement in speech reception threshold (SRT) was found for an input SNR of 0 dB.Speech Distortion Weighted Multichannel Wiener filter (SDW-MWF) for reducing noise is explored [8]. To achieve noise - reducing capability while retaining localization cues, MWF technique for binaural hearing devices is presented [9]. The effectiveness of monaural and binaural spectral subtraction methods is compared with a conventional directional microphone [10]. Input signals are recorded under typical hearing settings. The directional microphone reduced noise by 6.8 decibels. Monaural spectral subtraction method (SSM) lowers noise in the same way to directional microphones (DM). Binaural spectral subtraction (SSB) reduces noise by 8.2 decibels. The research presented in [11]-[14] describes how amplitude compression of speech signals helps in improving speech perception for hearing impaired in noisy conditions.The research presented in [15] describes an algorithm for processing speech signals in order to improve speech intelligibility in noisy conditions.The adaptive DRC method includes amplification based on time and frequency as well as DRC phases. Experiments were carried out in a vehicle noise (stationary) and a restaurant noise (non-stationary). An extended speech intelligibility index (ESII) and short time objective intelligibility (STOI)was used for objective evaluation.For stationary noise at -16 dB input SNR, adaptive DRC has a speech intelligibility score of 90%, compared to roughly 85% for time-and-frequency-dependent amplification. Speech intelligibility scores for non-stationary noise at -10 dB input SNR are 80 % for the Adapt DRC algorithm, 60 % for the time and frequency dependent amplification algorithm, and 40 % for the reference signal. The Adapt DRC method performs better in objective computations. At low SNRs, the adapt DRC algorithm improves ESII by around 0.3 and STOI by about 0.7 when compared to the unprocessed reference signal ESSI (0.1) and STOI (0.5). The work presented in [16] describes an integrated strategy for noise reduction

Manuscript received May 27, 2022, revised July 10, 2022.

Rajani S. Pujar is with Electronics and Communication Engineering Department, Basaveshwar Engineering College (Autonomous) Bagalkot-587102, Karnataka, India (e-mail: rajanisaachi@gmail.com).

doi: 10.18178/ijfcc.2022.11.3.589

(Generalized Side lobe Canceller (GSC)) and dynamic range compression to improve SNR. The goal is to manage DRC without compromising noise reduction performance. In the serial concatenation of noise reduction and dynamic range compression, remaining noise after noise reduction is enhanced by the DRC, resulting in a signal-to-noise-ratio (SNR) degradation. As a result, an approximation of the speech and noise dominant section is calculated in the technique. The noise reduction is made less active for the speech dominant part, whilst the noise reduction is made more active for the noise dominant part. The simulations were carried out using a two-mic BTE device. The sources of speech and noise are located at 0^0 and 120^0, respectively. Speech signals contain sentences from HINT database. Noise signal contains of multi-talker babble noise at 0 dB input SNR. Input value is set to 65 dB SPL in hearing aid microphones. The intelligibility-weighted SNR is used to assess noise reduction. SNR improvement of 10 dB was observed for the integrated technique when compared to serial concatenation SNR improvement of 6 dB for compression ratio of 2 and input SNR of 0 dB.In [17], binaural dichotic hearing techniques for decreasing temporal and spectral masking are described. Adjustable gain filtering with a 3 dB gain variation, spectral splitting with perceptually balanced comb filters with 9 pass bands, temporal splitting with trapezoidal fading with a 70% duty cycle, and combined splitting with cyclically swept comb filters with sweep cycles of 20, 40, 80, 120, and 160 ms were all used. For hearing impaired people, listening tests involving recognition of arbitrarily given words were employed to assess performance. Using spectral splitting, the majority of the subjects increased their recognition scores the most. At a switching rate of 20 ms, temporal splitting improved speech, whereas combined splitting improved speech at a sweep cycle of 80 ms. The speech signal was divided using binaural dichotic processing with time-varying comb filters. The comb filter is a FIR filter with 256 coefficients and a time period of 20 ms. The magnitude response of filters is complimentary to one another. The speech components that will mask each other are alternately given to the left and right ear [18]. Listening tests were carried out on people with normal hearing. To simulate hearing loss, broad band noise (at varying SNR levels) is supplied. As test samples, VCV syllable are employed. The recognition score for processed and unprocessed speech was greater for shifts of 8 and 4. The improvement in recognition scores, response time, and feature transmission indicates that the influence of spectral and temporal masking has been reduced. Effect of Noise Reduction Algorithms on Temporal Splitting of Speech Signal to Improve Speech Perception for Binaural Hearing Aids is discussed [19].

In our proposed work, we studied the application of a combination structure of adaptive Wiener filter and filter bank approach. The filter bank approach divides frequencies spanning from 0-5000 Hz into 18 bands depending on auditory critical bands. Spectral splitting of speech is accomplished by mixing odd and even numbered filters and presenting it to the two ears for binaural dichotic listening [20]. Adaptive wiener filter is based on the filter frequency response being adapted from sample to sample based on the characteristics of speech stream. Rest of the paper is organized as follows. Signal processing is explained in section II. Result analysis is given in section III. Spectrographic analysis is discussed in section IV. Finally, section V concludes the work.

II. Signal Processing

Our proposed approach took into account the combined setup of an adaptive Wiener filter and the filter bank method [21]as shown in Fig. 1. Spectrally splitting the speech signal helps in lowering an effect of spectral masking [22] [23]. The adaptive wiener filter is based on the adaption of the filter frequency response, which aids in the improvement of hearing comfort for the deaf.

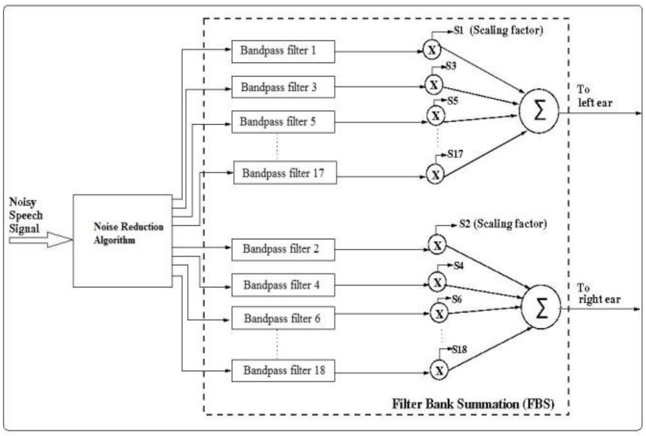

Fig. 1. Proposed block diagram.

A. Adaptive Wiener filter

The proposed work focuses on using the Wiener filter in an adaptive way to improve speech. According to the statistics of the voice signal, the Wiener filter [24] adapts the filter frequency response from sample to sample (mean and variance). The technique uses the approximated speech signal's mean and variance. It is assumed that noise $v(n)$ is additive white gaussian noise (AWGN) with a variance of σ_v^2 and zero mean. As a result, the noise power spectrum can be represented as shown in equation (1)

$$P_v(w) = \sigma_v^2 \quad (1)$$

When the signal x (n) is assumed to be stationary during a short period of the speech signal, the signal $x(n)$ can be represented as shown in equation 2

$$x(n) = m_x + \sigma_x w(n) \quad (2)$$

where m_x and σ_x are the local mean and standard deviation

Within this small section of speech, transfer function of the Wiener filter can be estimated as shown in equation 3.

$$H(w) = \frac{P_S(w)}{P_S(w)+Pv(w)} = \frac{\sigma_{s^2}}{\sigma_{s^2}+\sigma_v^2} \quad (3)$$

Enhanced speech signal $\widehat{S(n)}$ can beexpressed as shown in equation 4

$$\widehat{S(n)} = m_x + \frac{\sigma_{s^2}(n)}{\sigma_{s^2}(n)+\sigma_{v^2}}(x(n) - m_x) \quad (4)$$

After updating m_x and σ_s, the enhanced speech signal$\widehat{S(n)}$is written as given in equation 5.

$$\widehat{S(n)} = m_x(n) + \frac{\sigma_{s^2}(n)}{\sigma_{s^2}(n)+\sigma_{v^2}}(\, x(n) - m_x(n)) \qquad (5)$$

Local mean m_x (n) and $(x$ (n) - m_x $(n))$ are altered independently from frame to frame, and the outputs are then merged. If σ_s^2 is substantially greater thanσ_v^2, the output signal$\widehat{S(n)}$ will be mostly attributable to$x$ (n), and the input signal x (n) will be unaffected. If σ_s^2 is less than σ_v^2, the filtering effect is applied. σ_s^2 (n) is calculated from x (n) as shown in equation 6

$$\sigma_s^2 \ = \sigma_x^2 \ \ (n) - \sigma_v^2 \ \ ; \text{If } \ \sigma_x^2 \ \ (n) > \sigma_v^2 \qquad (6)$$
$$0 \text{ ; otherwise}$$

B. Filter Bank Method (FBM)

The proposed filter bank contains 18 band pass filters. The band pass filters were constructed using an iterative frequency sampling technique with a sampling frequency of fs = 10 KHz. Band pass filters (BPF) based on auditory critical bands (ACB18) divide the incoming audio into 18 bands (as shown in Table I). Odd-numbered BPF outputs were combined to form the signal for the left channel, and even-numbered BPF outputs produced the signal for the right channel. This aids in signal spectral splitting and as a result, lowers the spectral masking effect. Since speech synthesis occurs at the center level of the auditory system, while frequency component masking actually took place at the periphery. As a result, complementary spectral information in speech cues is separated into two parts and sent to both ears for binaural dichotic listening. This eliminates the detrimental effect of frequency masking. Fig. 2 demonstrates the frequency response of odd numbered band-pass filters (ACB18): band 1, band 3… and band 17. Fig. 3 indicates the frequency response of even numbered band-pass filters (ACB18): band 2, band 4…and band 18 respectively.

TABLE I: LIST OF AUDITORY CRITICAL BANDS (ACB18) ALONG WITH THEIR CENTRE FREQUENCIES [25]

Sl. No.	Center Frequency (in Hz)	Frequency Range (in Hz)
1	130	10-200
2	250	200-300
3	350	300-400
4	450	400-510
5	570	510-630
6	700	630-770
7	840	770-920
8	1000	920-1080
9	1170	1080-1270
10	1370	1270-1480
11	1600	1480-1720
12	1860	1720-2000
13	2160	2000-2320
14	2510	2320-2700
15	2920	2700-3150
16	3420	3150-4000
17	4050	3700-4400
18	4700	4400-5000

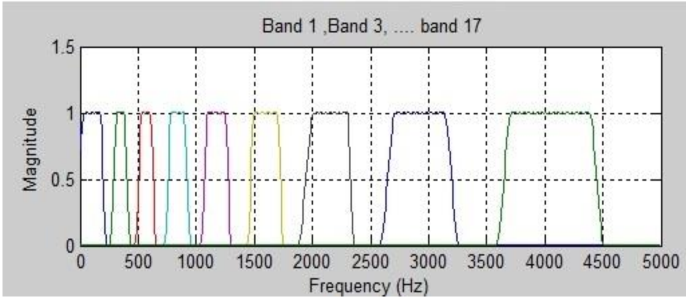

Fig. 2. O verlapped frequency response band pass filters (ACB18) (Odd numbered filtered bands)

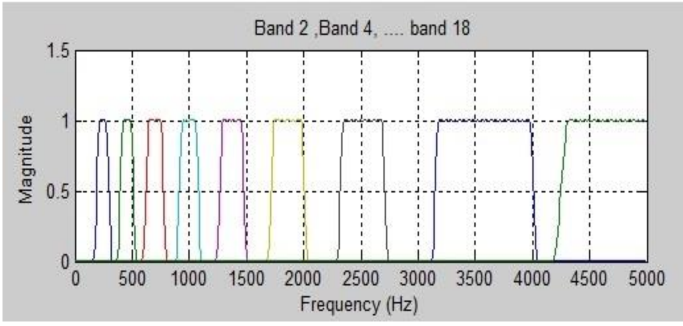

Fig. 3. Overlapped frequency response of band pass filters (ACB18) (Even numbered filtered bands)

III. LISTENING TESTS

Speech intelligibility tests were performed on six hearing-impaired patients with moderate sensorineural loss using the modified rhyme test (MRT) in the presence of wideband noise to examine the effectiveness of the suggested approach. Table II shows the audiometric thresholds of the participants. The processed voice signal was delivered via headphones.

A. Speech Intelligibility Assessment Using Modified Rhyme Test (MRT)[26]

The input speech contains monosyllabic consonant-vowel-consonant (CVC) words. For the analysis, 50 CVC word groupings were used. Each set has six words, for a total of 300 phrases. Each CVC word began with the statement "Would you write ———" These 300 CVC words were divided into six sample sets of 50 words each (1x, 1y, 2x, 2y, 3x, and 3y). Each word is chosen at random from the group levels 1, 2, 3 and the word levels x, y inside the group, and is repeated once. All phrases delivered by a male speaker are captured at 10 KHz and quantized at 16 bits. A computerised simulation environment was used to conduct the experiment. The user sat in front of the monitor and clicked the 'play' button on the test screen, responding to each presentation. During each display, a closed array of six possible replies was displayed on the monitor, and the subject chose the best match. The user selected the 'next' icon to begin the next presentation. This procedure was followed for all fifty words in the specified order. The response choices were randomly generated. The subject's responses and response time were recorded in the experiment setting. The responses were evaluated in accordance with the processing requirements to determine the percentage of correct response ratings for the subject.The experiment was conducted for about two hours per day for each subject, based on the availability of participants.

IV. RESULTS

The findings of the auditory analysis conducted utilising MRT for hearing-impaired subjects for the proposed approach are reported in this section.Table 3 shows the speech recognition scores (SRS %) and table 4 shows response time. Improvement in speech recognition scores were 0.639, 30.074, 30.401, 31.563, 32.28, 32.935 and reduction in response times were 0.068, 1.01, 1.279, 1.329, 1.41and1.581seconds for SNR values of ∞ dB, 6 dB, 3 dB, 0 dB, -3 dB, -6 dB respectively compared to unprocessed speech. The overall results revealed that at lower SNR values, processing with the proposed methodology improved intelligibility and resulted in decreased response times. The findings also showed that lower SNR values (-6 dB) result in the highest increase in speech recognition score (32.935%) and the lowest response time (1.581s).

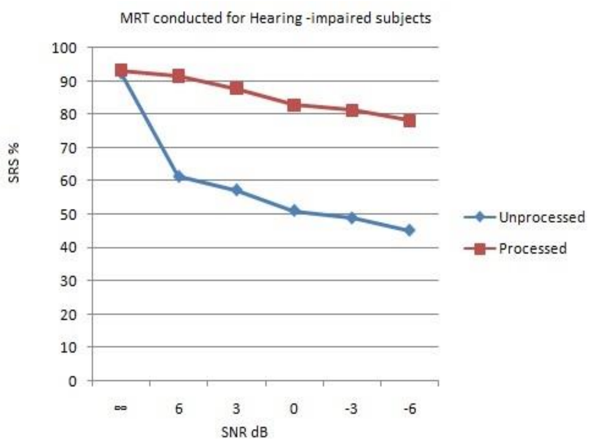

Fig. 4. Speech Recognition Score (Averaged across the six hearing – impaired subjects) Vs. SNR for unprocessed speech and Processed speech.

A. Graphical Analysis

Fig.4 shows the speech recognition scores (SRS %) Vs. SNR for unprocessed speech and speech signal processed with proposed scheme, employing adaptive wiener filter as noise reduction techniques. Experiment was conducted on hearing-impaired subjects. From these figure, we observed that processing improves speech intelligibility and reduces perception load in recognizing speech at lower SNR values.

B. Spectrographic Analysis

Below Figs. 5, 6, 7 and 8 shows the wide band spectrograms of the unprocessed and processed speech “would you write back” for SNR value of 3dB. Processed speech spectrograms shows that background noise is mostly reduced and compression of speech does not affect the harmonic structure.

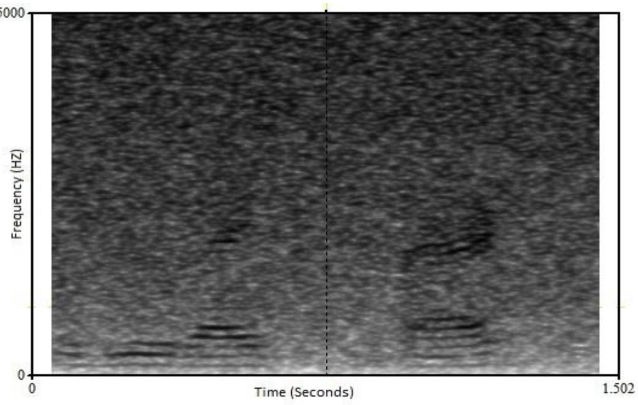

Fig. 5. Wide band Spectrogram of unprocessed speech “would you write back” for SNR value of 0dB

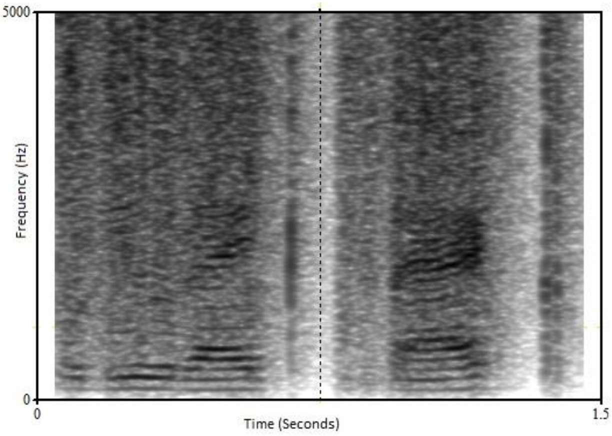

Fig. 6. Wide band Spectrogram of processedspeech“would you write back” for SNR value of 0dB

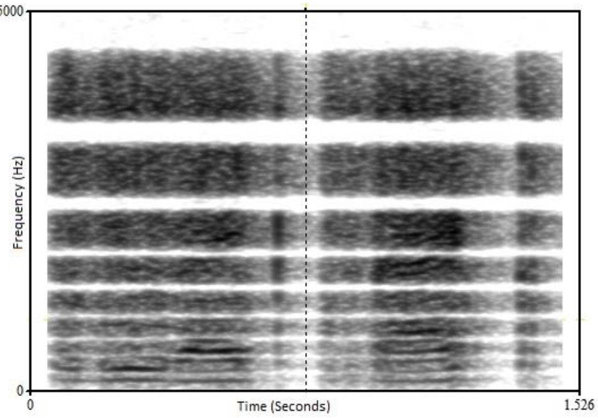

Fig. 7. Wide band Spectrogram (left ear)of processed speech “would you write back” for SNR value of 0dB

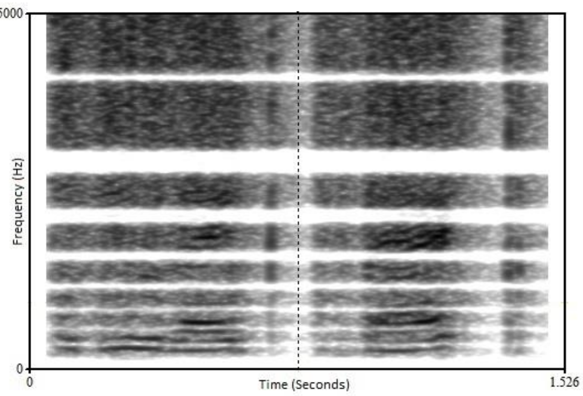

Fig. 8. Wide band Spectrogram (right ear) of processed speech “would you write back” for SNR value of 0 dB

TABLE II: AUDIOMETRIC THRESHOLD FOR THE SUBJECTS WITH MODERATE SNHL

Subject (Sex, age)	Ear	Hearing threshold (dB HL) Frequency(KHz)						
		0.25	0.5	1.0	2.0	4.0	6.0	8.0
SSP (Female, 63 Y) S1	RIGHT	38	38	40	40	50	60	70
	LEFT	28	28	30	31	40	45	50
RAY (Male, 52 Y) S2	RIGHT	55	52	55	60	68	72	80
	LEFT	50	50	50	50	60	62	75
BAY (Male, 50Y) S3	RIGHT	60	60	62	69	75	80	90
	LEFT	50	50	50	52	58	65	70
SA (Male,45 Y) S4	RIGHT	60	60	58	58	69	60	65
	LEFT	62	60	58	58	60	64	68
MJS (Female,42 Y) S5	RIGHT	28	28	29	30	38	45	45
	LEFT	40	38	40	40	48	60	60
PSB (Male,36 Y) S6	RIGHT	48	50	50	55	60	65	70
	LEFT	60	58	60	60	65	70	80

V. CONCLUSIONS

We have proposed combined configuration of adaptive wiener filter and frequency division of speech signal for binaural dichotic presentation. Proposed algorithm reduces masking effect as well as back ground noise effect. Hence improves speech intelligibility for hearing impaired. To assess the intelligibility of processed speech, Modified Rhyme Test (MRT) was conducted on subjects with hearing impaired (moderate SNHL).Theinput speech consists of 300 sentences. Each sentence contains CVC word.

Improvement in speech recognition scores of 0.639, 30.074, 30.401, 31.563, 32.28, and 32.935 respectively for SNR values of ∞dB, 6dB, 3dB, 0dB, -3dB, -6dB compared to unprocessed speechwas obtained. Results also indicate thatsignificant improvement in speech recognition score of 32.935% for lower SNR value of-6dBwas observed.

TABLE III: SPEECH RECOGNITION SCORES (SRS %) FOR HEARING IMPAIRED SUBJECTS, FOR 6 SNR CONDITIONS WHEN ADAPTIVE WIENER FILTER IS CASCADED WITH SPECTRAL SPLITTING USING FBM

Subject	Speech Recognition Score (SRS %) SNR(dB)											
	∞dB		+6dB		+3dB		0dB		-3dB		-6dB	
	Un Processed	Processed	Un Processed	Processed	Un Processed	Processed	Un Processed	Processed	Un Processed	Processed	Un Processed	Processed
S1	95	94.8	62	95	59.3	94.3	51.6	87.2	50	80	45.6	80.0
S2	95.66	95.0	56.33	88	55.66	83.6	50.33	84	46.66	82.78	42.33	78
S3	87.33	86.35	53.33	84. 5	53	88	48.66	83.98	47	80.32	41.66	77.6
S4	88.66	90.2	56.66	87.7	52.33	83	51.66	84	50.33	80.3	44.33	78.9
S5	96.66	96.00	74	93.8	71.66	88.46	56.66	77.8	54	79.56	52.66	76.8
S6	92	96.8	66	92.8	52	89	48	79.3	45.33	79	44.6	77.45
Mean	92.55	93.1916	61.386	91.46	57.325	87.72	51.15	82.713	48.88	81.16	45.19	78.125
% Improvement	0.639		30.074		30.401		31.563		32.28		32.935	

CONFLICT OF INTEREST

The author declares no conflict of interest.

REFERENCES

[1] B. W. Edwards, "Beyond amplification: Signal processing techniques for improving speech intelligibility in noise with hearing aids," *Seminars in Hearing*, vol. 21, no. 2, pp. 137–156, 2000.

[2] K. Chung, "Challenges and recent developments in hearing aids part i: Speech understanding in noise, microphone technologies and noise reduction algorithms," *Trends In Amplification*, vol. 8, no. 3, pp. 83–124, 2004.

[3] L. Yang and C. L. Philipos, "A geometric approach to spectral subtraction," *Speech Communication*, vol. 50, no. 1, pp. 453-466, 2008.

[4] B. Cornelis, M. Moonen, and J. Wouters, "A VAD-robust multichannel wiener filter algorithm for noise reduction in hearing aids," in *Proc. IEEE International Conference on Acoustics, Speech, and Signal Processing*, pp. 281–284, 2011.

[5] M. Zhang and M. H. Er, "Adaptive beam forming by microphone array,"in *Proc.IEEE Global Telecommunications Conference*, vol. 1, pp. 163-167, 1995.

[6] G. Hilkhuysen, N. Gaubitch, M. Brookes, and M. Huckvale, "Effects of noise suppression on intelligibility: Dependency on signal-to-noise ratios," *Journal of Acoustical Society of America*, vol. 131, no. 1, pp. 531-539, 2012.

[7] V.J eff, B. W. Jan, "An adaptive noise canceller for hearing aids using two nearby microphones," *Journal of Acoustical Society of America*, vol. 103, no. 6, pp. 3621-3626, 1998.

[8] B. Cornelis, M. Moonen, and J. Wouters, "Binaural cue preservation in binaural hearing aids with reduced-bandwidth multichannel Wiener filter based noise reduction," *International Workshop on Acoustic Echo and Noise Control*, 2010.

[9] S. Doclo and M. Moonen, "On the output SNR of the speech-distortion weighted multichannel wiener filter," *IEEE Signal Processing Letters*, vol. 12, pp. 809–811, 2005.

[10] H. Volkmar, "Comparison of advanced monaural and binaural noise reduction algorithms for hearing aids," *IEEE International Conference on Acoustics, Speech, and Signal Processing*, pp. 4008–4011, 2002.

[11] P. B. Sid, "Auditory compression and hearing loss," *Acoustics Today*, vol. 2,no. 2, p. 30, 2006.

[12] L. M. Cormack and V. Valimaki, "FFT based dynamic range compression," in *Proc. 14th Sound and Music Computing Conference*, Espoo, Finland, 2017, pp. 42-49.

[13] E. Villchur, "A critical survey of research on amplitude compression," *Scandinavian Audiology*, pp. 305-314, vol. 6, 1978.

[14] Dillon, "Compression? Yes, but for low or high frequencies, for low or highintensities, and with what response times?" *Ear and Hearing*, pp. 287-307, vol. 17, no. 4, 1996

[15] H. Schepkera, J. Rennies, and S. Doclo, "Speech-in-noise enhancement using amplification and dynamic range compression controlled by the speech intelligibility index," *Journal of the Acoustical Society of America*, pp. 2692-2706, vol. 138, no. 5, 2015.

[16] K. Ngo, S. Doclo, A. Spriet, M. Moonen, J. Wouters, and S. H. Jensen, "An integrated approach for noise reduction and dynamic range compression in hearing aids," in *Proc. 16th European Signal Processing Conference, Lausanne*, Switzerland, 2008.

[17] A. N. Cheeran and P. C. Pandey, "Speech processing for hearing aids for moderate bilateral sensorineural hearing loss," in *Proc. IEEE International Conference on Acoustic Speech Signal Processing*, pp. 17-20, Montreal, Quebec, Canada, 2004.

[18] D. S. Jangamashetti, P. C. Pandey, and A. N. Cheeran, "Time varying comb filters to reduce spectral and temporal masking in sensorineural hearing impairment," in *Proc. International Conference on Biomedical Engineering*, pp. 1-7, Bangalore, 2001

[19] R. S. Pujar and P. N. Kulkarni, "Effect of noise reduction algorithms on temporal splitting of speech signal to improve speech perception for binaural hearing aids," in *Proc. 13th Western Pacific Conference on Acoustics (WESPAC-2018)*, 2018.

[20] R. S. Pujar and P. N. Kulkarni, "Cascaded structure of wiener filter with FBS based spectral splitting and dynamic range compression for listeners with sensorineural hearing loss," in *Proc. 15th IEEE India Council International Conference (INDICON 2018)*, 2018.

[21] R. S. Pujar and P. N. Kulkarni, "Wiener filter based noise reduction algorithm with perceptual post filtering for hearing aids," *Int. J. Image, Graphics, and Signal Processing*, vol. 11, no. 7, pp. 69-81, 2019.
[22] R. S. Pujar and P. N. Kulkarni, "Cascaded structure of noise reduction and multiband frequency compression of speech signal to improve speech perception for monaural hearing aids," in *Proc. 16th IEEE India Council International Conference*, 2019.
[23] R. S. Pujar and P. N. Kulkarni, "Frequency compression of speech for improving speech perception in sensorineural hearing loss: FBS approach," in *Proc. International Conference on Wireless Communications, Signal Processing and Networking*, March 22-24, 2017, SSN College of Engineering, Chennai, India
[24] M. A. A. El-Fattah, M. I. Dessouky, S. M. Diab, and F. E. S. A. El-Samie, "Speech enhancement using an adaptive wiener filtering approach," *Progress in Electromagnetics Research*, pp. 167-184, vol. 4, 2008.
[25] E. W. Zwicker, "Subdivision of audible frequency range into critical bands (Freqenzgruppen)," *Journal of Acoustical Society of America*, vol. 33, no. 2, pp. 248-248, 1961.
[26] P. N. Kulkarni, P. C. Pandey, and D. S. Jangamashetti, "Binaural dichotic presentation to reduce the effects of spectral masking in moderate bilateral sensorineural hearing loss," *International Journal of Audiology*, pp. 334-344, vol. 51, no. 4, 2012

Rajani S. Pujar received her B. E. degree from Karnataka University, the M. Tech degree in digital communication and the Ph.D. degree from Visvesvaraya Technological University, Belgaum, Karnataka, India. She has experience of 20 years in teaching. Presently she is working as assistant professor, E&C Engg., Dept., Basaveshwar Engineering College (Autonomous), Bagalkot, Karnataka. Her research interests are: speech processing, information theory and coding and wireless networks. She has published many papers in national/international conferences and journals.