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Spectral splitting of speech signal using time varying recursive filters for binaural hearing aids

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ABSTRACT

Speech perception in noisy environments is reduced in human with sensorineural hearing loss (SNHL) due to masking. Moderate SNHL cannot be cured medically hence masking effects should be reduced to enhance speech perception. To reduce masking, processing delay and hardware complexity the present paper is proposed a scheme to partition the voice signal into two signals which are complementary to each other by using the filter-bank summation method (FBSM) with a set of time-varying recursive band pass filters. Performance of the filter is evaluated with following measures: perceptual evaluation of speech quality (PESQ), mean opinion score (MOS) for speech quality and modified rhyme test (MRT) for speech intelligibility. The test signals used for the evaluation of quality are a syllable and a word and for the evaluation of intelligibility 300 monosyllabic words are used. The results demonstrated an increase in the quality and intelligibility of processed speech in a noisy environment. As a result, there is an enhancement in perception of processed speech in a noisy environment.

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1. INTRODUCTION

There are different kinds of hearing loss; sensorineural hearing loss (SNHL) conductive hearing loss and mixed hearing loss [1]. SNHL arises owing to the broadening of auditory filters of the cochlea inevitably leading to the masking of frequency components of adjacent auditory filter bands. As a result, speech perception diminishes in human with SNHL. Due to this, the sensory system's capacity of frequency resolution decreases and speech detection in noisy environments becomes challenging [2], [3]. Earlier studies demonstrated that the use of a pair of comb filters and time-varying comb filters (finite impulse response (FIR)) to partition the voice signal into two complementary spectra for binaural hearing aid was effective in decreasing the impact of frequency masking [4]–[8]. Although it was discovered that the earlier methods were successful in tackling frequency masking, the step alter in the magnitude responses with a fixed amount of separate time shifts affects the ability to identify gaps. Further, since FIR filters used in the earlier studies generally require higher order, thereby increasing the processing delay and hardware complexity thus making it difficult for real-time implementation. Hence, we proposed the implementation of the spectral splitting using the filter-bank summation method (FBSM) with a set of time-varying infinite impulse response (IIR) band-pass filters for binaural hearing aids, which generally require much less order and hardware complexity [9].

The objective of this paper is to reduce the hardware complexity of binaural hearing aids and improve speech perception in SNHL persons by reducing masking as this loss cannot be medically cured. This is achieved using a bank of time-varying IIR band-pass filters to partition voice signal into two complementary spectra for binaural dichotic hearing aid. Using modified rhyme test (MRT) [10] for intelligibility and perceptual evaluation of speech quality (PESQ) [11], and mean opinion score (MOS) [12]–[14] for quality the efficiency of the approach in enhancing speech perception was assessed. To minimize frequency masking, it has been proposed to use a set of time-varying band-pass (IIR) filters with FBSM that sweeps continuously in magnitude responses, which ultimately enhances speech perception.

According to Sangeetha and Kannan [15], various digital filters are discussed. The input signal is the voice signal. With the input voice signal, additive white Gaussian noise (AWGN) is added. Using sampling rate conversion, the noisy speech signal spectrum is down-sampled into many sampling rates. The noisy audio signal and its sub-bands are subjected to a number of transforms including the fast fourier transform (FFT), fast walsh hadamard transform (FWHT), and discrete wavelet transform (DWT). The rectangular, hanning, hamming, blackman, and kaiser windows are used in the design of FIR filters, whereas butterworth and chebyshev filters are used in the design and implementation of IIR filters. The filter coefficients are then subjected to quantization. Lastly, the signal to quantization ratio (SQNR) is used to evaluate the performance of the filters. According to the performance measures, DWT applied signals filtered by the chebyshev high pass filter (HPF) give high SQNR in IIR filtering, and DWT applied signals filtered by the blackman window provide high SQNR in FIR filtering. The assessment of the performance of the filter coefficients is based on the SQNR.

The researchers in [16], [17] compared of binaural hearing aid with the monaural hearing aid is discussed. Amini *et al.* [18] proposed a noise reduction method that preserved the binaural spatial cues of point sources and simultaneously determined the best rate allocation and the optimal estimation weights across all sensors and frequencies. To retain the binaural spatial cues of point sources, the suggested technique takes into account both linear and rate constraints. The inter-aural level difference (ILD) error, the inter-aural time difference (ITD) error, and the averaged binaural SNR were used to assess the efficiency of the suggested approach. The results demonstrated that the suggested approach performed better than spatially accurate noise reduction techniques utilizing random rate allocation schemes. The development of several efficient speech processing algorithms for portable devices is facilitated by the current trend of low power and area consumption [19]. High performance is not the main area of emphasis in biological applications like hearing aids. Power and vicinity are equally important factors. The market penetration is affected by a variety of things. The system can thus operate as slowly as possible to trade speed for power and vicinity. An effective folded IIR filter is developed. The folded design employs the idea of temporal multiplexing, and all operations are handled by just one multiplier and one adder. Its effectiveness is evaluated in comparison with a typical IIR filter implemented in direct form-2 structure. In vivado high-level synthesis (HLS) a FIR filter is implemented and instantiated in the block design. The look up table (LUT) utilization is reduced from 0.13 to 0.09. The LUT utilization in IO and global clock buffer (BUFG) remains the same.

2. METHODOLOGY

In cochlea spectral components of nearby bands are likely to overlap leading to frequency masking. For this reason, odd bands are presented to the left ear and even bands to the right ear to lessen masking. To present speech signal dichotically (to the left and right ears) with complementary spectral components, a FBSM with a set (bank of 22) of time-varying band-pass (recursive) filters of order 25 with complementary magnitude responses is used. In this technique, speech signal is processed with bank of band-pass filters (with different bandwidths alternately summed) to present signal to SNHL listeners. The magnitude response $|H_m(f)|$ and impulse response $h_m(n)$ of a time-varying band-pass filters are dynamic and are functions of time 'm'. The proposed approach employs a set of time-varying band-pass IIR filters with magnitude responses that are swept along the frequency axis with respect to time to partition the speech into two signals with complementary spectra for binaural dichotic hearing aid. Figure 1 shows a block schematic of the proposed work's methodology. The voice signal is divided into two portions with complementary spectral components using FBSM with a set of time-varying band-pass recursive filters of order 25 with complementary magnitude responses. This allows the speech signal to be presented dichotically. The recursive filter in the proposed scheme uses less order hence decreases processing delay and hardware complexities compared to hearing aids with FIR filter. The Yule-Walker approach [20] is utilized to develop the filters, and the MATLAB programming language is used to implement them. Each band-pass filter is based on 1/3 octave bands (total 22 bands) [21]. The Yule-Walker method of designing IIR filters is used because arbitrary magnitude response can be designed with this method. The system function of IIR filter is given as in (1).

$$H(Z)=\frac{\sum_{0}^{N-1} b_q Z^{-q}}{1+\sum_{1}^{N-1} a_p Z^{-p}} \quad (1)$$

Here N is order, a and b are respectively poles and zeros of the filter. $|H_m(Z)|$ is the magnitude response of time-varying IIR filter which is dynamic. Figure 2 is shows a set of time-varying band-pass IIR filters' complementary magnitude responses for Figure 2(a) left ear and for Figure 2(b) right ear are depicted at various points of time. To reduce phase distortion introduced by the filter all pass filter is used [22], [23].

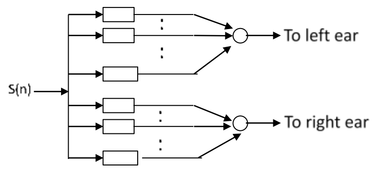
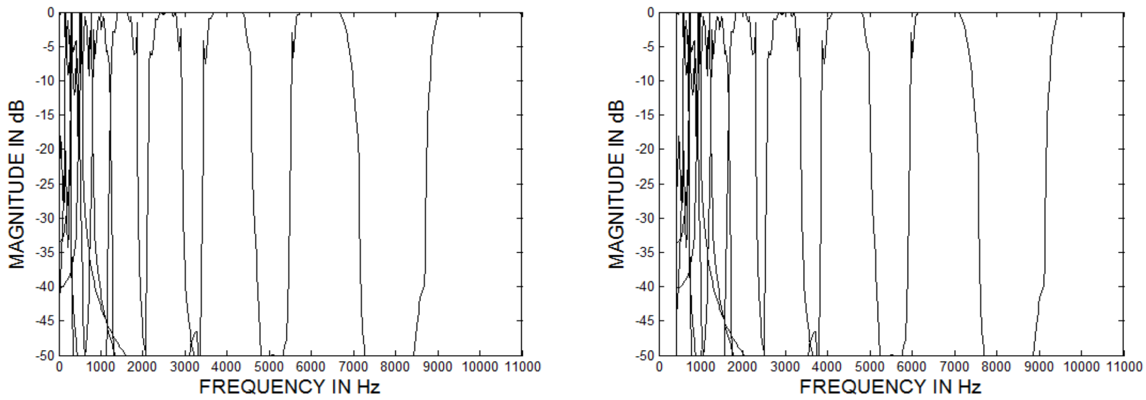
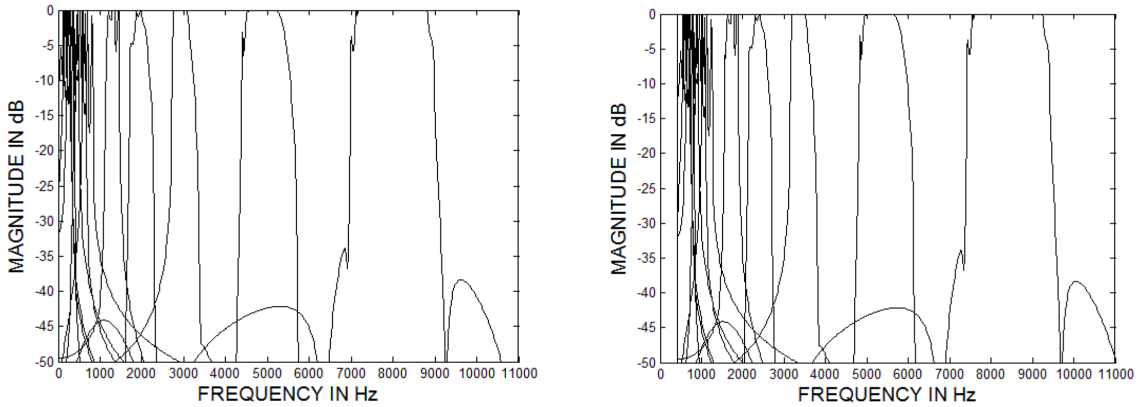


Figure 1. Schematic representation using FBSM



(a)



(b)

Figure 2. Set of time-varying band-pass IIR filters' complementary magnitude responses for: (a) left ear and (b) right ear at various points of time

3. RESULTS AND DISCUSSION

In this approach, a speech signal undergoes spectral separation through the application of a set of time-varying band-pass filters with continuous sweeps in magnitude responses. The time shifts between these filters are kept smaller than the just noticeable difference (JND), ensuring seamless transitions between spectral components. The resulting signals, characterized by complementary spectral components, are then evaluated for quality of voice signal using PESQ and MOS metrics, while intelligibility is assessed through the use of the MRT measure.

3.1. Speech quality

3.1.1. Perceptual evaluation of speech quality

The PESQ model, which employs the PESQ algorithm to evaluate speech quality, is represented in Figure 3. An International Telecommunication Union standard (ITU-T P.862) for evaluating speech quality is called PESQ. PESQ values are generated by an algorithm that resembles a human sensory system. To determine the PESQ values of the processed signal, the PESQ algorithm uses a reference signal i.e. an unprocessed signal, and a processed signal i.e. test signal as inputs. Study materials the VC syllables /aa-b/ and word /grey/ are considered to assess the quality of the speech. The PESQ values (unprocessed, processed) of the /aa-b/ and /grey/ signals at various SNRs are given in Tables 1 and 2, respectively. These findings are acquired using the PESQ algorithm, which calculates PESQ values using both unprocessed (reference signal) and processed (test signal) signals. On comparing the PESQ values of processed and unprocessed signals, processed signals show improvement at lower SNRs. There is an improvement in PESQ values for /aa-b/ for SNR values between 6 dB and -6 dB in steps of 3 dB are 0.0927,0.1261, 0.1836, 0.4636, 0.3979 respectively, and for word /grey/ are 0.0025, 0.26, 0.4699, 0.8348, 1.1259 respectively. At low values of SNR, improvements in PESQ values are observed to be more.

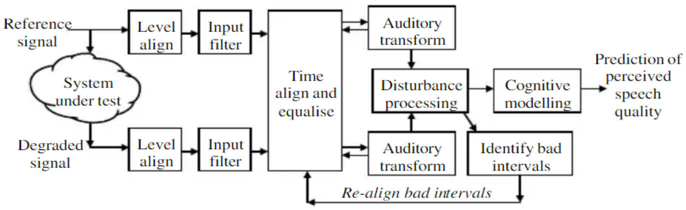


Figure 3. PESQ model [24]

Table 1. Scores (PESQ) for VC syllable /aa-b/

SNR values in dB	/aa-b/ signal		
	Unprocessed	Processed	Improvement
∞	4.5	2.3598	-2.1402
6	2.0573	2.15	0.0927
3	2.0694	2.1955	0.1261
0	2.0055	2.1891	0.1836
-3	1.9839	2.4475	0.4636
-6	1.9129	2.3108	0.3979

Table 2. Scores (PESQ) for word /grey/

SNR values in dB	/grey/signal		
	Unprocessed	Processed	Improvement
∞	4.5	1.66	-2.84
6	1.8264	1.8289	0.0025
3	1.597	1.857	0.26
0	1.4449	1.9148	0.4699
-3	1.3101	2.1449	0.8348
-6	1.2243	2.3502	1.1259

3.1.2. Mean opinion score

An assessment of speech quality is also made using the MOS measure on three male and three female subjects aged between 21 to 45 years with normal hearing. With initial training in the assessment method, participants were asked to provide their opinion (on a scale 0-5, bad-1, poor-2, fair-3, good-4, and excellent-5 [25]) about the quality of unprocessed and processed materials at various SNR levels. The first column of both Tables 3 and 4 lists different subjects along with their age and sex. The following columns present their opinions based on the perceived quality of both unprocessed and processed materials at various SNR levels (infinity, 6, 3, 0, -3, and -6 dB). By determining the average of opinion scores, MOS values are obtained. Improvement can be seen in MOS values of processed test materials at lower SNR on comparing the MOS of unprocessed signal with processed. An enhancement in MOS values for the VC signal /aa-b/ after processing is 0.58, 0.92, 1.00, 1.08, and 1.17 for SNR 6 to -6 in steps of 3 dB respectively. Similar to this, MOS is used to examine the voice quality of the word /grey/ on the same participants, and the results showed an improvement for SNR 6 to -6 in steps of 3 dB is 0.67, 0.92, 1.08, 1.17, and 1.25 respectively (processed speech). For both test signals, processed signals have higher MOS scores compared to unprocessed signals.

Table 3. Scores (MOS) for unprocessed and processed /aa-b/ signal

Subjects with sex, age	SNR in dB ∞		6		3		0		-3		-6	
	*Unpr	**Pro	Unpr	Pro	Unpr	Pro	Unpr	Pro	Unpr	Pro	Unpr	Pro
S1, [M,20]	4.5	3	2.5	3	2.5	3	2.5	3	2.5	3.5	2	3.5
S2, [F,25]	4.5	2.5	2.5	3.5	2	3	2	2.5	2	3	1	2
S3, [F,21]	4	3	3	3	1.5	2.5	2.5	3.5	2.5	3.5	2	3
S4, [F,33]	4	3	2.5	3	2	3.5	2	3.5	2	3	1	2
S5, [M,45]	4	3.5	2	3.5	1.5	2.5	1	2.5	1	2.5	1	2.5
S6, [M,27]	4	2.5	3	3	2	2.5	2	3	2	3	2	3
MOS	4.17	2.92	2.58	3.17	1.92	2.83	2.00	3.00	2.00	3.08	1.50	2.67
Improvement	-1.25		0.58		0.92		1.00		1.08		1.17	

*Unpr: Unprocessed speech, **Pro: Processed speech

Table 4. Scores (MOS) for unprocessed and processed /grey/ signal

Subjects with sex, age	SNR in dB ∞		6		3		0		-3		-6	
	Unpr	Pro	Unpr	Pro	Unpr	Pro	Unpr	Pro	Unpr	Pro	Unpr	Pro
S1, [M,20]	4.5	3	2.5	3	2	2.5	2.5	3	2.5	3	2	3
S2, [F,25]	4	2.5	2	3.5	2	3	2	2.5	2	3.5	1	2.5
S3, [F,21]	4	3	3	3.5	1.5	2.5	2.5	3.5	2.5	3.5	1	2.5
S4, [F,33]	4	3	2.5	3	2	3	2	3.5	2	3	2	2.5
S5, [M,45]	4.5	3	2	3.5	1	2.5	1	2.5	1	3	1	3
S6, [M,27]	4	2.5	3.5	3	1.5	2	2	3.5	2	3	2	3
MOS	4.17	2.83	2.58	3.25	1.67	2.58	2.00	3.08	2.00	3.17	1.50	2.75
Improvement	-1.33		0.67		0.92		1.08		1.17		1.25	

3.2. Speech intelligibility

The MRT is used to assess speech intelligibility. 300 consonant-vowel-consonant (CVC) words made up the study materials for MRT. Each word is preceded by "would you write". The words were uttered by a guy in an audiometry room and recorded at 10 kilo samples per sec. with 16-bit quantization using a B&K microphone model No. 2,210. Six test sets of 50 words each were made from all 300 CVC words. All test words were chosen using a two-level randomization process to overcome biasing. For listening tests to determine intelligibility, subjects with moderate bilateral SNHL (six in number, 3 men and 3 women, ages 32 to 51) participated. The subjects were first given directions on the testing process and stimuli. Once they were used to the testing process, the CVC words were presented to them. The whole testing process was done using an automated test administration setup.

In the testing process, 1,800 words (300 words x 6 levels of SNR) were presented to each participant to reply to. In the MRT test window, 300 MRT words were loaded and created a random file to store the recognition score of participants. Each subject was instructed to sit in front of the computer screen, and to click the test window's "play" button. The six options presented in the test window, the subject chose the best match after hearing the presentation. Response time (sec) and speech recognition score (%) were recorded in a random file. This procedure was done for all MRT words with different SNR levels. Depending on the participants' availability and willingness, the test was run for around a month. The audiometric threshold for the participants with bilateral moderate SNHL is given in Table 5.

Table 5. Threshold level for the humans with SNHL (moderate bilateral) in audiogram

Subject	Ear	Hearing threshold level (dB HL) Frequency 250 Hz	500 Hz	1000 Hz	2000 Hz	4000 Hz	8000 Hz
S1 [M,42]	*R	60	45	55	40	60	65
	**L	40	55	55	50	60	60
S2 [M,51]	R	50	55	50	45	55	60
	L	55	50	45	55	50	55
S3 [M,43]	R	20	30	60	40	50	50
	L	20	30	60	50	40	50
S4 [F,35]	R	65	60	55	50	40	45
	L	60	55	55	50	45	50
S5 [F,32]	R	25	30	40	50	60	90
	L	20	30	40	50	60	85
S6 [F,40]	R	35	45	55	60	65	70
	L	25	40	45	50	60	60

*R: Right ear, **L: Left ear

The results of listening tests conducted on participants using MRT materials are shown in Table 6, with the average audio recognition score (%) for both unprocessed and processed speech at various SNR levels. The average recognition scores of processed speech have improved by -1.78%, 15.06%, 18.22%, 20.62%, 24.17%, and 26.61% respectively, when compared to the unprocessed recognition scores with SNR levels (∞ dB, 6 dB to +6 dB in increments of 3 dB). Table 7 shows the average response times of the same subjects for the specified SNR levels, and it can be seen that response time is decreased for processed speech.

Table 6. Speech recognition scores (%) for SNHL (bilateral) subjects

Subjects with sex, age	SNR Values in dB											
	∞		6		3		0		-3		-6	
	Unpr	Pro	Unpr	Pro	Unpr	Pro	Unpr	Pro	Unpr	Pro	Unpr	Pro
S1 [M,42]	92.00	91.67	65.33	81.33	57.00	75.00	53.60	77.00	46.67	67.00	42.67	71.00
S2 [M,51]	92.33	90.33	65.33	80.67	57.67	77.00	52.40	74.00	49.33	78.00	43.67	71.33
S3 [M,43]	94.67	94.33	66.33	78.67	58.67	79.00	55.67	76.33	49.33	73.67	45.33	72.00
S4 [F,35]	94.00	92.67	66.33	81.67	61.00	76.33	56.00	76.40	47.67	72.33	47.00	72.67
S5 [F,32]	92.00	86.33	66.33	80.00	58.67	78.00	55.33	72.67	47.67	73.00	45.33	72.33
S6 [F,40]	92.67	91.67	65.33	83.00	61.00	78.00	58.33	78.67	50.33	72.00	45.33	69.67
AVG.	92.94	91.17	65.83	80.89	59.00	77.22	55.22	75.84	48.50	72.67	44.89	71.50
Improvement	-1.78		15.06		18.22		20.62		24.17		26.61	

Table 7. Response time in seconds for bilateral SNHL subjects

Subject	Response Time SNR (dB)											
	∞dB		+6dB		+3dB		0dB		-3dB		-6dB	
	Unpr	Pro	Unpr	Pro	Unpr	Pro	Unpr	Pro	Unpr	Pro	Unpr	Pro
S1	4.543	4.585	4.865	4.465	4.476	4.981	4.754	4.876	4.318	4.743	5.762	4.766
S2	4.456	4.645	4.543	4.454	5.435	4.768	4.665	4.546	4.761	4.324	5.762	4.723
S3	4.879	4.765	4.854	4.786	5.987	4.543	5.943	4.985	5.872	4.198	5.128	4.181
S4	4.865	4.954	4.543	4.435	4.345	4.984	4.874	4.754	4.821	4.643	4.984	4.241
S5	4.542	4.543	4.943	4.765	4.872	4.567	4.876	4.897	4.843	4.765	5.845	4.941
S6	4.786	4.345	4.342	4.543	4.765	4.894	4.654	4.634	4.654	4.435	5.755	4.954
Mean	4.679	4.640	4.682	4.575	4.980	4.790	4.961	4.782	4.878	4.518	5.539	4.634
Reduction	0.04		0.11		0.19		0.18		0.36		0.91	

4. CONCLUSION

The technique proposed in this paper involves dividing the voice signal into two partitions with complementary spectral components for a binaural dichotic hearing aid. This is carried out by using a set of time-varying band-pass (IIR) filters with one-third octave bands having continuous sweep in magnitude responses with time shift chosen less than JND. The method decreases the impact of spectral masking and enhances the ability of people with SNHL to detect gaps in a voice signal. The scheme is evaluated using measures: PESQ, MOS test for quality, and MRT for speech intelligibility. The results of the PESQ and MOS studies indicated an improvement in the PESQ value of 0.3979 and MOS test score of 1.17 for /aa-b/ at SNR -6 dB on comparing unprocessed speech with processed speech. Similarly, improvement in PESQ value of 1.1259 and MOS test score of 1.25 for /grey/ at-6 dB SNR on comparing unprocessed speech with processed speech. The MRT result showed 26.61% improvement in recognition score and a decreased response time of 0.91 sec in the case of bilateral SNHL persons when processed speech is compared with that of unprocessed speech at the same value of SNR. The quality and intelligibility of processed speech (processed using the proposed scheme) have improved more at lower SNR implies speech perception increases in SNHL humans. Hence the proposed method is more effective under adverse listening conditions. In future more number of participants and multiband compression require to be considered.

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