STM JOURNALS

Current Trends in Signal Processing
ISSN: 2277-6176 (Online), ISSN: 2321-4252 (Print)
Volume 7, Issue 3
www.stmjournals.com

Effective FPGA Implementation of Comb Filters to Improve Perception of Sensorineural Hearing Impaired

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Abstract
In this paper, effective method for implementation of comb filter on FPGA platform has been proposed. This work is intended for hearing aid to be used for people suffering from sensorineural hearing loss. The algorithmic implementation on FPGA is intricate during the design process. Here we propose effective and simplified approach for implementation of comb filter with 512 coefficients using Spartan-6 FPGA for dichotic presentation. The design of comb filter uses spectral splitting based on auditory critical bandwidths. Magnitude response of these filters are optimized for low perceived spectral distortion having high stop band attenuation and low pass band ripples. This method was evaluated by performing listening tests experimentation on hearing impaired people having different degree of sensorineural hearing loss. The speech material used consists of fifteen vowel consonants vowel syllables. The performances of the listening test have been analyzed with qualitative assessment, response time, recognition score and information transmission analysis. Results indicate significant reduction of response time, improvement in recognition score and transmission of all consonantal features. The latency was analyzed and found to be 8x10^{-3} seconds (average), noticeable to subjects.

Keywords: *Sensorineural Hearing Loss, Comb filter, FPGA, Dichotic Presentation, Latency*

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INTRODUCTION

Committee on Hearing, Bioacoustics and Biomechanics (CHABA) [1], National Research Council, Washington has presented statistics of percentage of hearing impaired at each decade and percentage of impairments according to age group. CHABA concluded that in future, hearing impairment will continuously grow due to adverse environmental effects caused by unavoidable human activities. Acoustic hearing aids are and will remain the most effective aids for majority of hearing impaired persons. Hence, there is still a need for further research in this area; specifically, sensorineural hearing loss, and a large variety of hearing impairments fall under this category as it deals with the most delicate part of human ear [22]. Problem of sensorineural hearing loss can be better solved by developing efficient algorithms for hearing aid.

Signal processing based on comb filter is an essential component in hearing aid to improve the perception of sensorineural hearing-impaired subjects. Perceptually balanced comb filter based on critical bandwidth helps in reducing the spectral masking. Spectral splitting approach improves the perception of hearing impaired subjects having moderate bilateral loss [20]. It is obtained by using a pair of linear phase FIR comb filter with complimentary magnitude response for binaural dichotic presentation [2, 3].

The optimization of the comb filters was analyzed with respect to the number of bands and bandwidth. 1/3rd octave band and critical band based comb filters are superior to constant bandwidth filters as discussed by Kulkarni and Pandey [4]**.** Higher recognition scores and smaller response times for normal-hearing subjects using comb filters based on the auditory critical bandwidth experiments were observed by Kulkarni et al. [5]. The experimental results on the hearing-impaired subjects showed an average increase of 22% in recognition scores and a substantial decrease in response times [5]. Comb filters using 512 coefficients and its algorithms were implemented on MATLAB by Mahesh and Chaudhari [6]. An offline experiment was carried out on hearing impaired subjects. For comb filter relative improvement in recognition scores varied from 1.78% to 4.44%. An 8-channel digital filter bank was implemented on DSP processor by Lunner and Hellgren [7]. Filters implemented on the DSP processor were complementary interpolated linear phase FIR filters having 700 Hz of constant bandwidth. However, constant bandwidth technique degrades the performance of the linear phase FIR filter particularly for sensorineural hearing problems. An idea was proposed to decrease the effect of spectral masking using complementary comb filters with 128 coefficients for hearing impaired subjects [2]. A real-time implementation using two TI/TMS320C50 DSP processors was carried out. The 128 coefficients enforced limitation on the use of comb filters.

Wavelet packets algorithm for noise reduction for hearing impaired were developed by Ghamry [8]. The proposed algorithm uses eight bands implemented on Digilent XUP II Virtex II Pro system FPGA, containing audio codec LM4550. The algorithm for de-noising the rise of speech signal by bionic wavelet transform on FPGA Xilinx (Virtex 5 XC5VLX110T) platform was implemented by Salaheddine and Bachir [9]. A 128-th order FIR filter with 8 band pass filters was used. The experimentation showed an improvement in SNR. A design for a prosody modification of speech (PMS) on field programmable gate array (FPGA) platform was developed by Bendaouia et al. [10]. A new scheme was implemented, which combined the design of discrete wavelet transform (DWT) and overlap - add method (OLA) techniques to analyze the input data. Experimentation showed efficient real time results compared to classical designs. The binary mask based algorithm was proposed by Valerie and Kofi [11] and subsequently implemented the spectral analysis stage with 28 channel bank using 8-th order band pass filters. The algorithm was implemented on the Spartan-3A FPGA and the test results show the improvement in intelligibility of speech by 85% in presence of noise.

Earlier literature survey shows the use of DSP processors to implement FIR filter banks [2,7]. Several investigations have proved outperformance of FPGA over DSP chips [12,13,23,24]. So, FPGA designs are mostly useful for experimental platform.

In this work, we have implemented the comb filter on FPGA platform. A pair of complementary comb filters from the filter sets has been used as a scheme for spectral splitting. As per eighteen critical bands of auditory filters, nine pass bands are formed in complementary manner [14]. The direct form-I structure is designed using the principle of convolution. The HDL coder is used to convert MATLAB code into synthesizable VHDL code for FPGA implementation [15]. To reduce conversion complexity in proposed method we used only two-input adder along with adder tree to add results of all multiplications. The HDL code is optimized for hardware implementation and simulation results have been observed.

The next section discusses the comb filter design and its implementation on FPGA platform. The subsequent section deals with simulation and experimental results of FPGA implementation, listening tests results and conclusion.

DESIGN OF COMB FILTER AND ITS HARDWARE IMPLEMENTATION

Critical band based comb filters were designed using 512 coefficients. Increasing the number of coefficients results in improvement in the ripples in the pass band and attenuation in the stop band. Comparatively small pass band ripples may result in perceptual distortion, and relatively small stop band attenuation may give adequate separation between bands. The FIR filter is designed using frequency sampling technique [16,17].

The processing scheme used in this paper is based on the spectral splitting with comb filters based on auditory critical bandwidths. The filters are optimized for low perceived spectral distortion with pass band ripple less than 1 dB, minimum stop band attenuation 50 dB and crossover gain -5 dB to -7 dB. The magnitude response of comb filter pair is shown in Figure 1. As per eighteen critical bands of auditory filters, nine pass bands are formed in complementary manner. The auditory critical bandwidth is up to 5 kHz.

This filter cannot be synthesized on FPGA using MATLAB HDL coder. Therefore, a novel approach has been proposed for designing the comb filter for synthesizable FPGA implementation. In this approach, the direct form–I filter with the same coefficients are used for odd and even filter banks.

The basic adder has only two inputs so an adder tree is used to add results of all multiplications. The filter structure is shown in Figure 2. This approach makes the MATLAB code synthesizable on FPGA.
If, X (z) and Y(z) are the filter input and output, respectively, then H (z) is given as below:

$$H(z) = \frac{Y(z)}{X(z)} = \sum_{i=0}^{N} a_i z^{-i} \qquad (1)$$

where, a_i is a filter coefficient, N is order of filter. The length of filter is N+1. The difference equation for odd filter bank is given by Eq. (2).

$$Out_h = h(0)u(0) + h(1)u(n-1) + \cdots \ldots + h(512)u(n-512) \qquad (2)$$

here the length of filter is 513 and order of filter is 512. $h(0), h(1) \ldots \ldots \ldots .. h(512)$ are odd filter bank coefficients Similarly the difference equation for even filter bank is given by Eq. (3).

$$Out_g = g(0)u(0) + g(1)u(n-1) + \cdots \ldots \ldots \ldots \ldots + g(512)u(n-512) \qquad (3)$$

where, $g(0), g(1) \ldots \ldots \ldots .. g(512)$ are even filter bank coefficients.

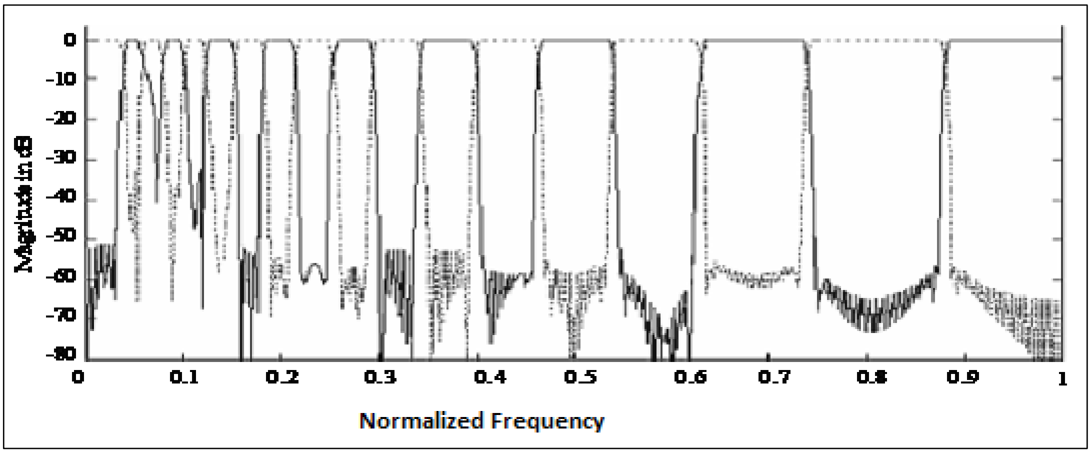

Fig. 1: *Magnitude Response of Comb Filter.*

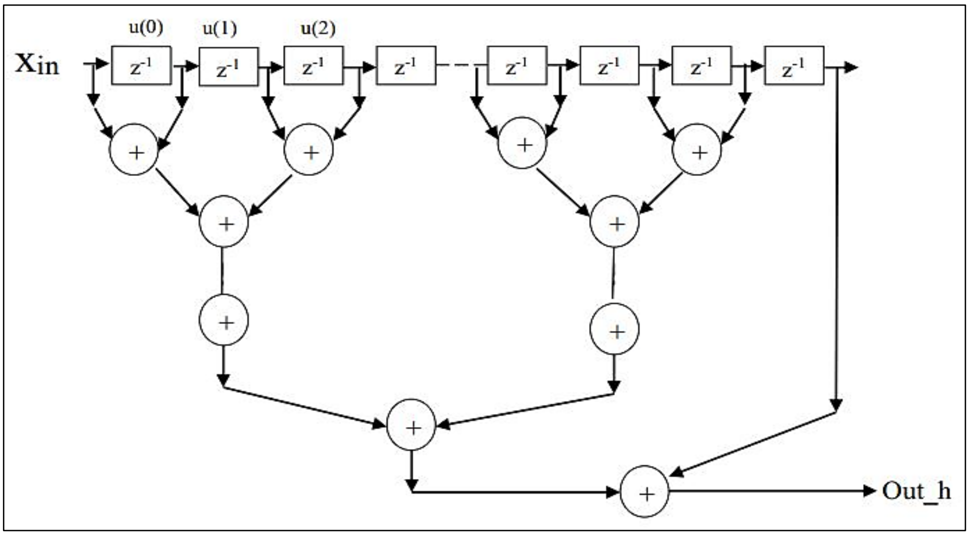

Fig. 2: *Filter Structure.*

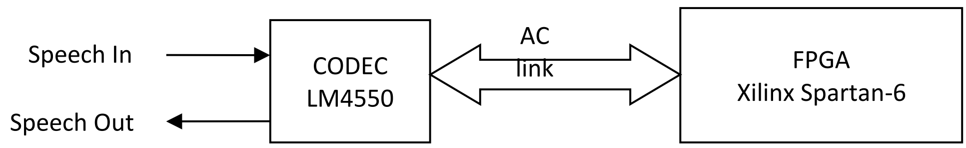

Fig. 3: *Flow of Unprocessed and Processed Signal using FPGA.*

The algorithm of comb filter for synthesizable FPGA implementation is as follows:

1) Read input speech signal ***x(n)*** (with noise) of length *N*.
2) h1, g1 are the coefficients of comb filter pair, design using FIR filter (Frequency sampling technique.)
3) For *i=1 to N* repeat following:

right_out(i)=conv(x(i), h1);
left_out(i)=conv(x(i), g1);
where *conv(x, f): step 1: multiplication*
for k=1 to M (length of comb filter f)
*m(k)=u(k)*f(k);*
step 2: addition(adder chain)
Repeat till last two terms m_1,m_2
for j=1 to M/2
add m(j),m(j+1);
$out=m_1+m_2$;
step 3: Shift input sample x(i) in persistent variable u

The unprocessed speech signal was used as input and given through audio-in of Atlys Circuit board. The VHDL compatible MATLAB code was used to program the Xilinx 6 FPGA. The input unprocessed speech signal being continuous in nature had to be converted into digital form while the processed digital signal so obtained had to be converted back to continuous signal. So, we used an on-board AC'97 Codec (LM4550) for the same.

Figure 3 shows the interconnection of FPGA "Xilinx Spartan-6 XC6SLX45CSG324C" and the onboard codec "LM4550" used to implement the filter.

The input/output range of audio codec is $1V_{RMS}$. It is programmed using AC Link Serial Interface Protocol(SIP) for 18-bit analog-to-digital and digital-to-analog conversion. The filtering operation takes place in FPGA. For interfacing codec, AC link protocol is implemented in FPGA. The resulting 18-bit is sent to Line Out through DAC.

For our Spartan 6 XCLS45, the Atlys circuit board consists of 18 bit codec. The unprocessed continuous speech signal given through mic-in of the circuit board is converted into 18 bit digital code. This code is processed through our algorithm. The output is converted back to continuous processed speech signal using 18bit DAC.

In this paper, for Spartan LS45 the incoming signal from SDATA_IN gets split in 9 bits. Each 9-bit data are stored in two registers for each channel. The compatible 9 bit VHDL code (comb filter code) is implemented on the FPGA so as to process each register. Processed data are sent on SDATA_OUT line. The data are then converted to continuous analog signal using 18 bit DAC. The codec used here processes data in serial fashion while the FPGA processes data in parallel manner. Hence we have also written the codes for serial in parallel out (SIPO) and parallel in serial out (PISO). Figure 4 shows schematic of FPGA implementation.

The experimental setup for comb filter implementation on FPGA platform is shown in Figure 5. For the implementation Xilinx Spartan6 XC6SLX45 CSG324 FPGA from Digilent was used.

SIMULATION AND EXPERIMENTAL RESULTS

The results obtained after deploying the developed algorithm on FPGA platform are discussed in following subsection:

Simulation Results

The input speech signal is fed to the system in VCV context. The filter bank splits the input signal into even and odd bands. The magnitude response for critical band filter using MATLAB code for the test input signal "ada.wav" is shown in Figure 6.

After the MATLAB simulated results are verified, the HDL coder is used to convert MATLAB code into VHDL code. The generated VHDL code is synthesized and the RTL schematic is generated using XILINX 14.2 for verification. The generated RTL

Current Trends in Signal Processing
Volume 7, Issue 3
ISSN: 2277-6176 (Online), ISSN: 2321-4252 (Print)

schematic for comb filter is shown in Figure 7. It shows the implemented filter structure in FPGA has 18 bit input signal called xin and 18 bit output signal g_out (even band) and h_out(odd band). The comb filter is implemented on Xilinx Spartan-6 FPGA. Figure 8 shows the ModelSim simulated test bench waveform of the same.

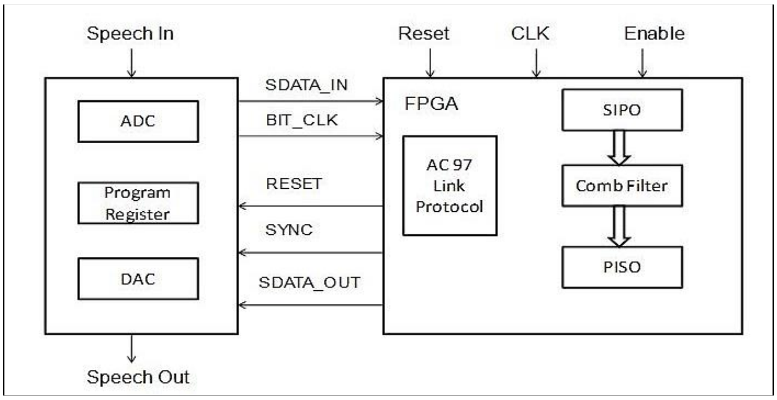

Fig. 4: *FPGA Implementation Schematic.*

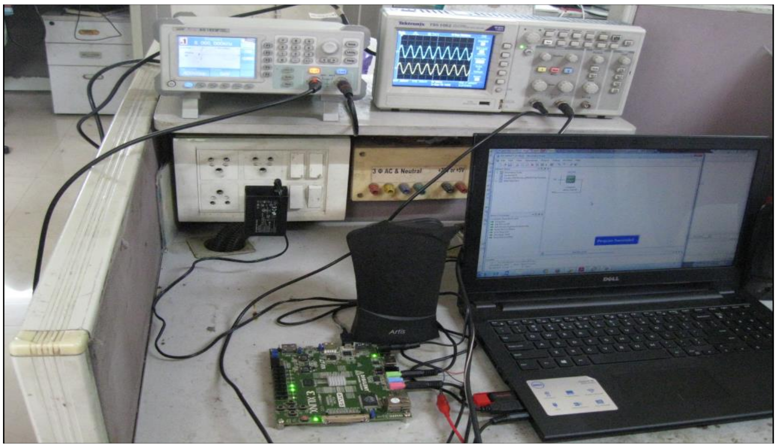

Fig. 5: *Experimental Setup for Comb Filter Implementation.*

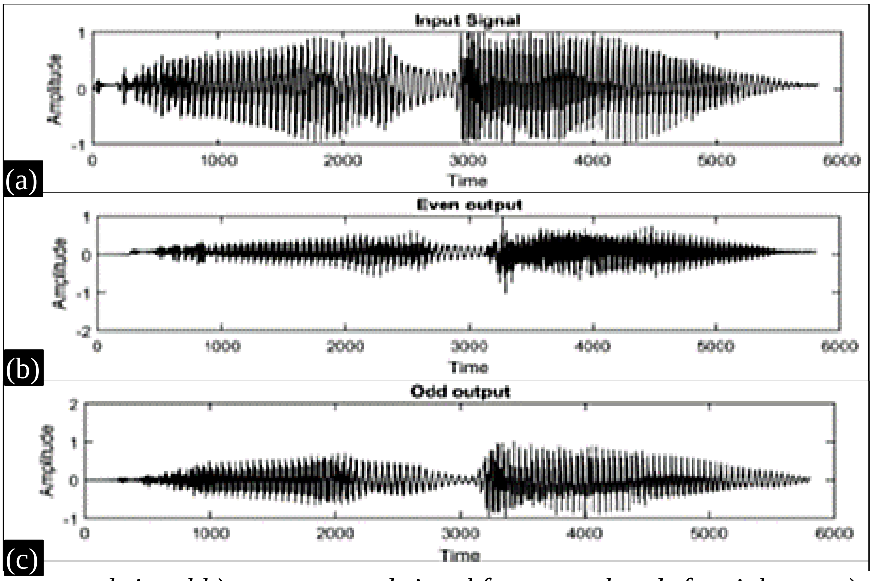

Fig. 6: *a) unprocessed signal b) reconstructed signal from even bands for right ear c) reconstructed signal from odd bands for left ear.*

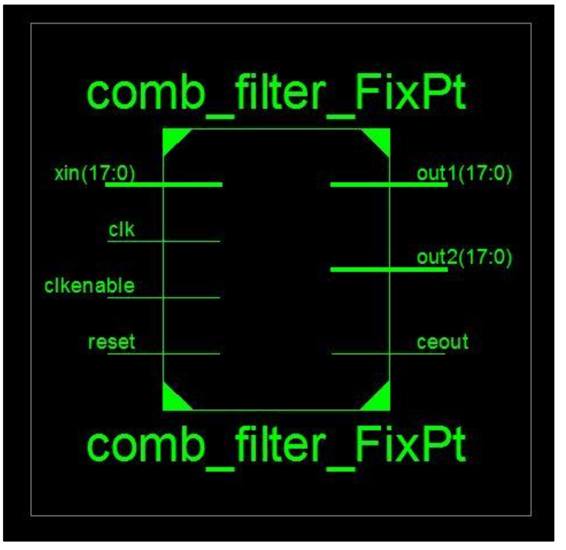

Fig. 7: *RTL Schematic for Critical Band Filter.*

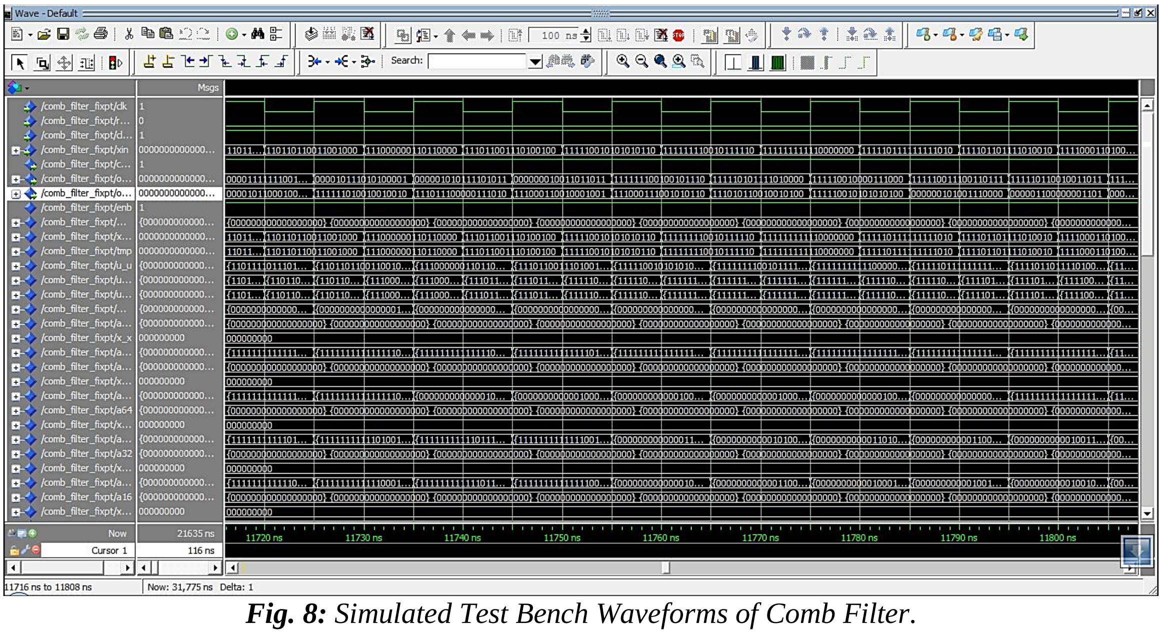

Fig. 8: *Simulated Test Bench Waveforms of Comb Filter.*

Experimental Results

In this work, for realization of comb filter on FPGA the input and output data lines splits into 9 bits each. Due to this reason, the FPGA implementation of 512-coefficient comb filter utilize fraction of the resources available on FPGA spartan6 XC6SLX45 CSG324, i.e., (10% of total slice registers, 31% of slice LUTs, 25% of fully used LUT-FF pairs and 6% of bonded IoBs') with scope for implementation of other processing blocks of the hearing aid.

The latency of our algorithm was measured using speech signals (VCV context). Here, latency is defined as the time delay between the unprocessed signal deployed on FPGA and processed signal fed to ear. This delay was measured by giving the unprocessed signal and processed signal to Digital Storage Oscilloscope Textronix TBS1062 so that it could simultaneously record both the signals with high sampling frequency. The recorded signals were then analyzed in MATLAB. We tested various VCV context speech signals having different frequency compositions. The latency obtained varied from 1.6 $\times 10^{-3}$ seconds (minimum) to 14.08 $\times 10^{-3}$ seconds (maximum). The average value of latency of our algorithm for different frequencies was found to be 8×10^{-3} seconds, noticeable to subjects [18].

LISTENING TEST RESULTS

The experimentation was carried out and evaluated by performing listening tests on seven sensorineural hearing-impaired subjects. (APL: F 16, KUR: M 57, KSH: F 12, BHL: F 10, PDH M 74, PNK: M 45, BHV: M 12) Subjects KSH, BHL and BHV have severe and symmetrically hearing impairment. Subject KUR has mild to moderate and asymmetrical high frequency impairment. Subject PNK has mild and symmetrically sloping high frequency impairment. Subject APL has moderately severe and symmetrical hearing impairment while subject PDH has moderately severe and asymmetrical low frequency hearing impairment. The speech material used for the listening test consisted of a set of fifteen nonsense syllables in vowel consonant vowel (VCV) context with consonants / p, b, t, d, k, g, m, n, s, z, f, v, r, l, y / and vowel / a / as in 'farmer' [19] ,[21]. A confusion matrix was tabulated for the responses obtained along with the recorded response time. These matrices were used further to calculate recognition scores and relative transmitted information. The following subsection includes results of listening tests for speech quality, response time, recognition score and information transmission analysis. To find out the contribution of the features in improvement of speech perception, these tests were conducted for consonantal features.

Qualitative Assessment

To compute qualitative assessments, five times pre-recorded test material in VCV context was heard by subjects. On the basis of quality of sound, they were asked to give the rating as 'Outstanding', 'Good', 'Fair', 'Average' and 'Below Average'. These ratings were indexed from 5 to 1; 5 being 'Outstanding' and '1' being 'Below Average'. Averages of ratings were computed to find mean opinion score of each subject graphically represented in Figure 9. From the graph, six subjects APL, KSH, BHL, PDH and PNK graded the quality of processed signal as higher than the unprocessed signal. Subject KUR has ranked the unprocessed signal same as processed signal.

Response Time

The time interval between input speech materials presented dichotically to subjects and the responses obtained from them is known as response time. The response time for seven subjects is shown in Figure 10. It shows variation from 4.08 to 8.6 sec for unprocessed signal and 4.68 to 7.95 sec for processed signal. There was reduction in response time for processed signals. This relative decrease is from -7.35% to 33%. Relative decrease in response time is significant for the subjects KUR and KSH as shown in Figure 11.

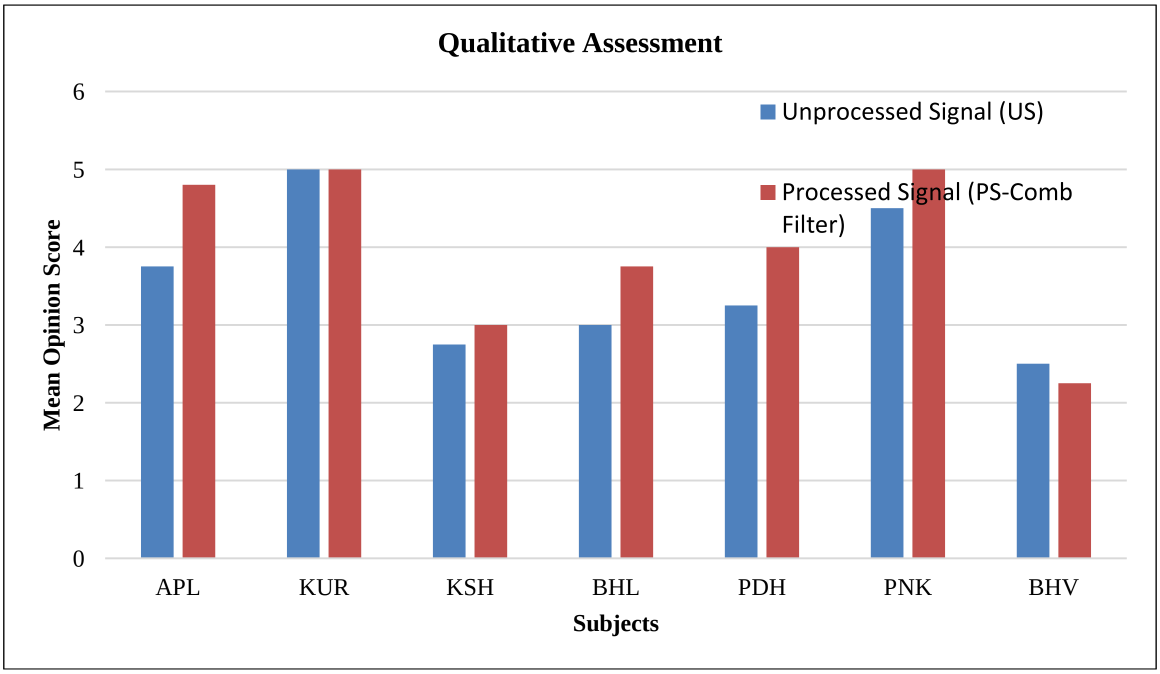

Fig. 9: *Mean Opinion Score.*

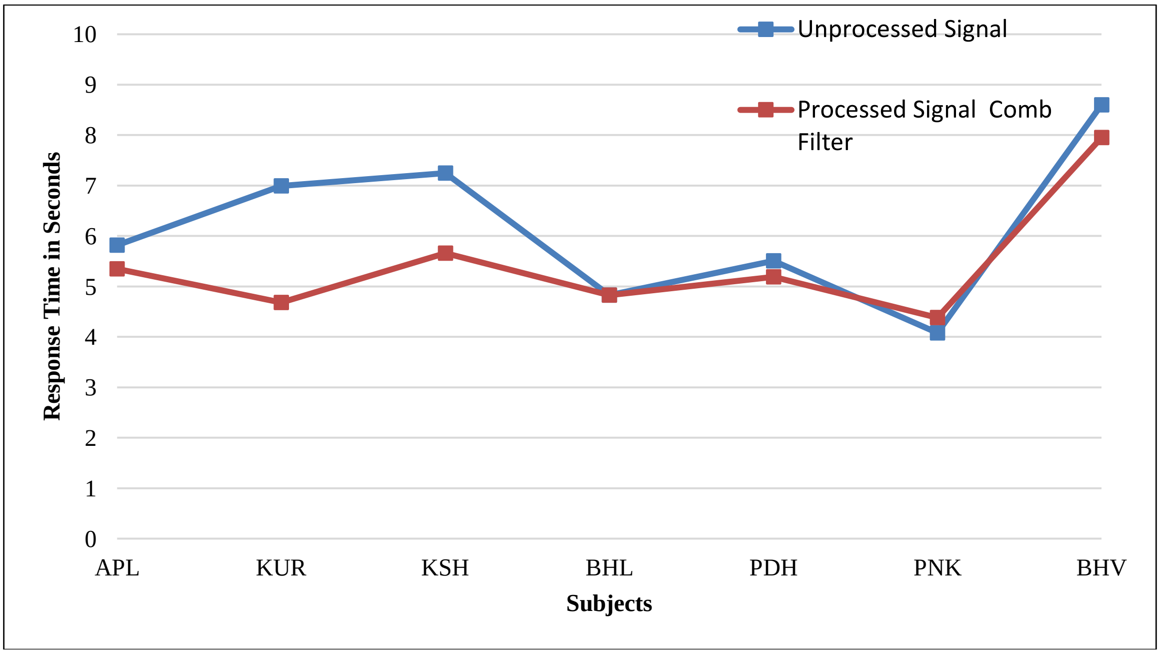

Fig. 10: *Response Time in Seconds.*

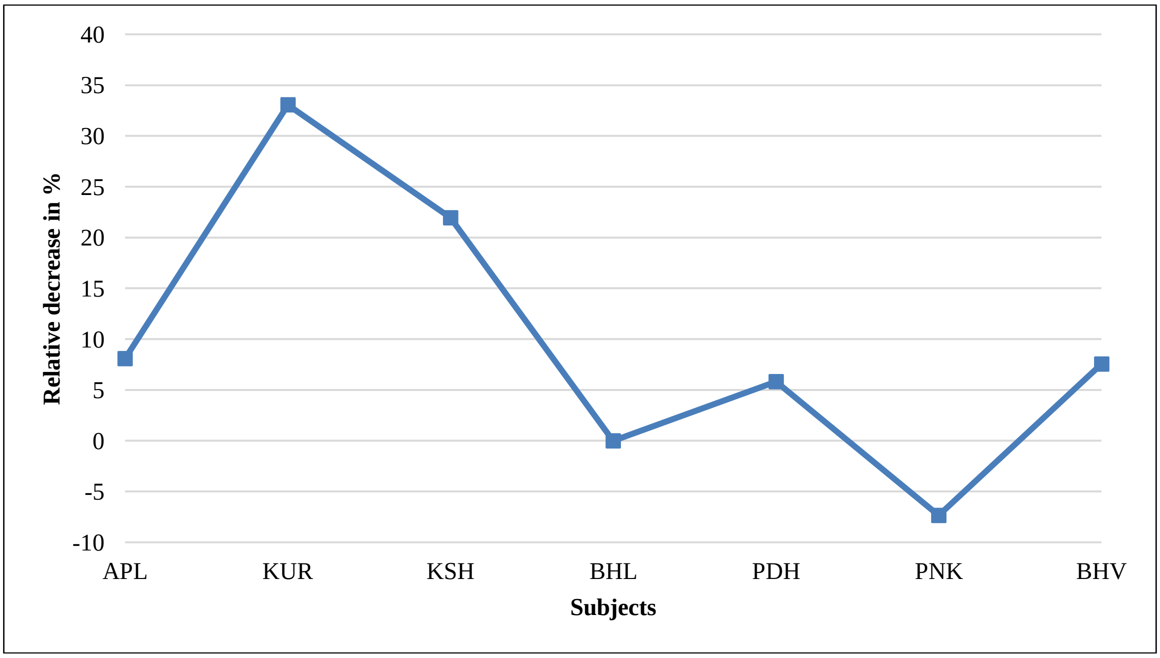

Fig. 11: *Relative Decrease in Response Time.*

Recognition Scores

The recognition scores obtained from the confusion matrix and relative improvement (%) are shown in Figures 12 and 13, respectively. For the impaired subjects, recognition score is observed in the range of 48.33% to 90% for unprocessed signal while for processed signal the recognition score increases from 60% to 96.66%. Three subjects (BHL, APL, and KUR) having mild to severe frequency impairment have shown maximum relative improvement and one subject (PNK) with symmetrical high frequency hearing impairment have shown maximum decrease in recognition score.

Information Transmission Analysis

To find out the contribution of the features in improvement of speech perception, information transmission analysis tests were conducted for consonantal features. The consonants were grouped according to the articulatory features

[19] and the contribution of different features was analyzed. Combined confusion matrices of each subject were used to evaluate these features. Table 1 shows the relative information transmitted for consonantal features and are plotted in Figures 14 to 21.

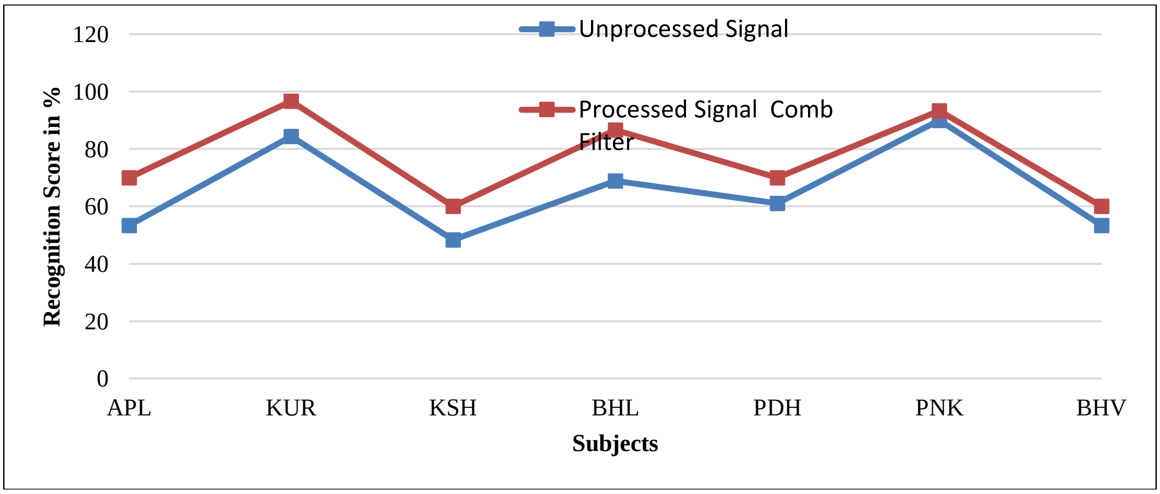

Fig. 12: *Recognition Score in Percentage.*

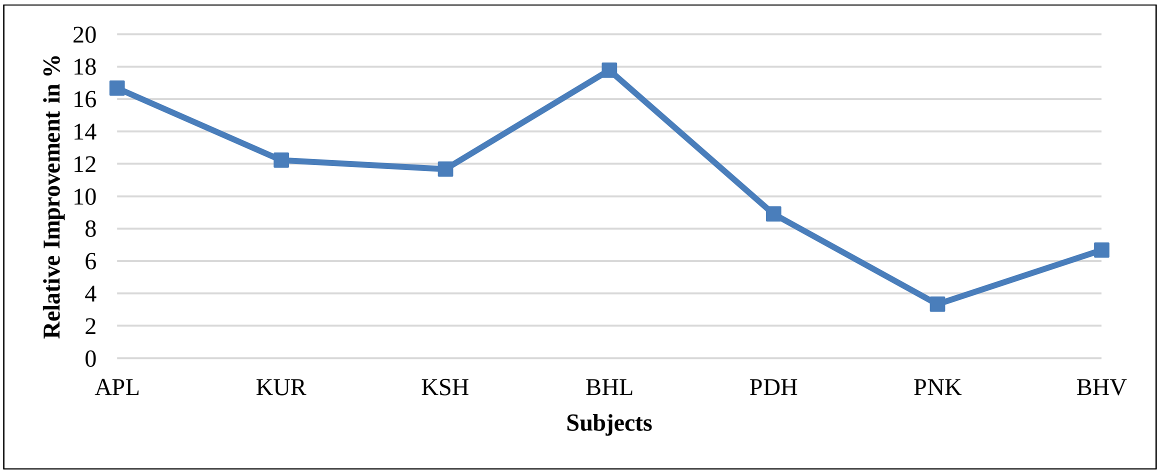

Fig. 13: *Relative Improvement in Recognition Score.*

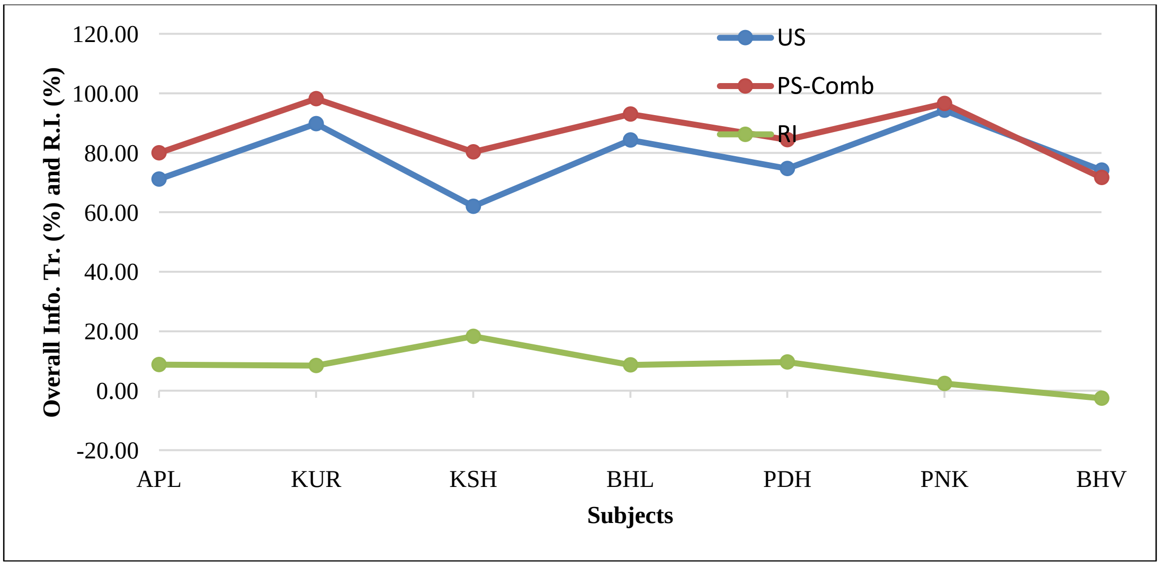

Fig. 14: *Relative Information Transmitted for Overall.*

Table 1: *Information Transmission Analysis for VCV context. US: Unprocessed speech, PS-Comb: Processed Signal using Comb Filters and RI: Relative Improvement (%) with respect to Unprocessed Signal for Different Features.*

Feature: Overall					**Feature: Continuance**			
Subject	US	PS-Comb	RI		Subject	US	PS-Comb	RI
APL	71.15	79.95	8.80		APL	33.15	44.43	11.28
KUR	89.77	98.16	8.39		KUR	41.12	86.49	45.37
KSH	62.00	80.28	18.28		KSH	26.02	44.43	18.41
BHL	84.30	92.95	8.65		BHL	39.84	76.35	36.52
PDH	74.69	84.34	9.65		PDH	42.53	35.61	-6.92
PNK	94.24	96.59	2.35		PNK	44.43	60.42	15.99
BHV	74.19	71.62	-2.57		BHV	11.42	37.59	26.16
Averages	78.62	86.27	7.65		Averages	34.07	55.04	20.97
Feature: Duration					**Feature: Frication**			
Subject	US	PS-Comb	RI		Subject	US	PS-Comb	RI
APL	41.12	62.95	21.83		APL	26.46	38.31	11.85
KUR	58.00	100.00	42.00		KUR	53.77	86.94	33.16
KSH	41.29	24.72	-16.57		KSH	21.17	53.35	32.18
BHL	41.16	100.00	58.84		BHL	38.72	78.94	40.22
PDH	76.74	100.00	23.26		PDH	57.73	64.90	7.16
PNK	76.35	100.00	23.65		PNK	48.96	64.90	15.93
BHV	44.43	58.26	13.83		BHV	10.52	52.43	41.91
Averages	54.16	77.99	23.83		Averages	36.76	62.82	26.06
Feature: Manner					**Feature: Nasality**			
Subject	US	PS-Comb	RI		Subject	US	PS-Comb	RI
APL	40.45	38.48	-1.97		APL	57.10	35.78	-21.32
KUR	70.73	91.43	20.70		KUR	100.00	100.00	0.00
KSH	40.37	57.88	17.51		KSH	71.40	63.68	-7.72
BHL	50.54	74.25	23.71		BHL	67.57	67.57	0.00
PDH	64.02	66.16	2.14		PDH	74.54	67.57	-6.96
PNK	68.05	78.42	10.37		PNK	100.00	100.00	0.00
BHV	15.55	57.50	41.95		BHV	20.52	67.31	46.78
Averages	49.96	66.30	16.34		Averages	70.16	71.70	1.54
Feature: Place					**Feature: Voicing**			
Subject	US	PS-Comb	RI		Subject	US	PS-Comb	RI
APL	27.64	38.70	11.07		APL	100.00	78.94	-21.06
KUR	61.11	100.00	38.89		KUR	71.86	100.00	28.14
KSH	18.52	24.48	5.96		KSH	63.98	82.45	18.47
BHL	52.58	88.27	35.69		BHL	86.94	100.00	13.06
PDH	79.67	100.00	20.33		PDH	37.01	71.68	34.67
PNK	89.30	100.00	10.70		PNK	100.00	100.00	0.00
BHV	25.06	30.95	5.89		BHV	21.80	26.75	4.95
Averages	50.55	68.91	18.36		Averages	68.80	79.97	11.18

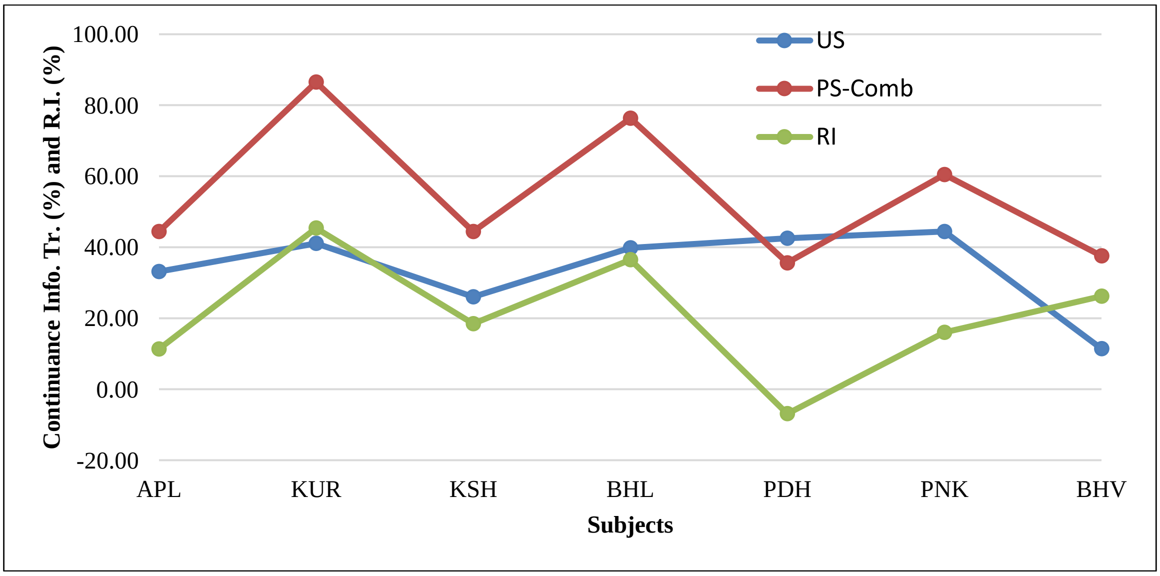

Fig. 15: *Relative Information Transmitted for Continuance.*

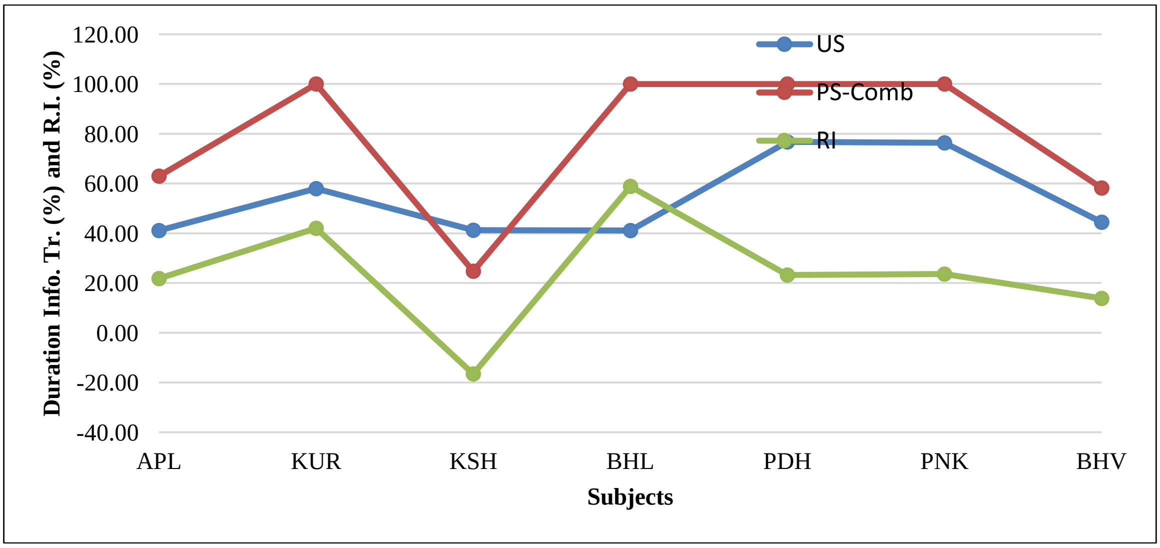

Fig. 16: *Relative Information Transmitted for Duration.*

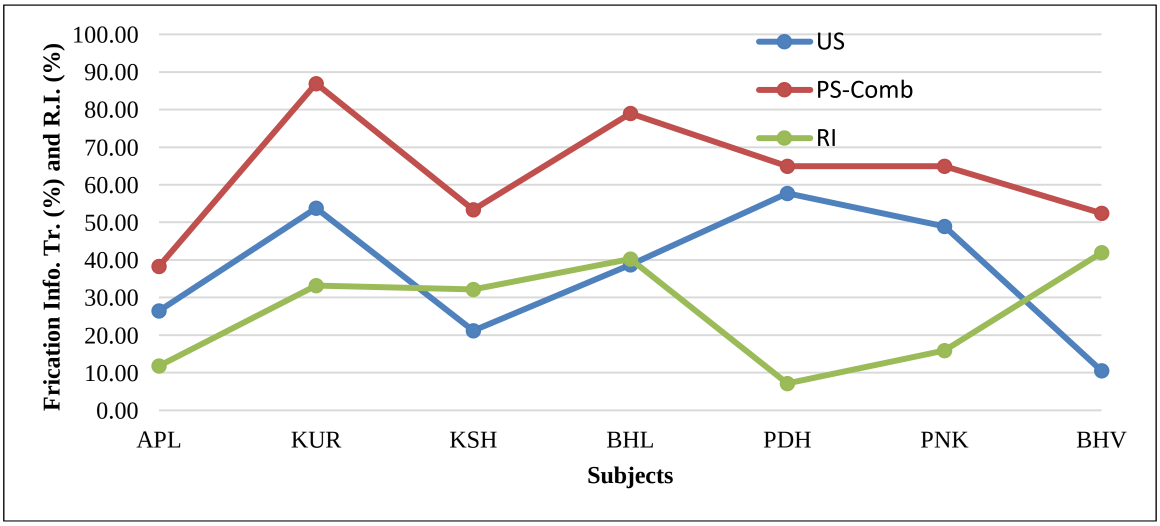

Fig. 17: *Relative Information Transmitted for Frication.*

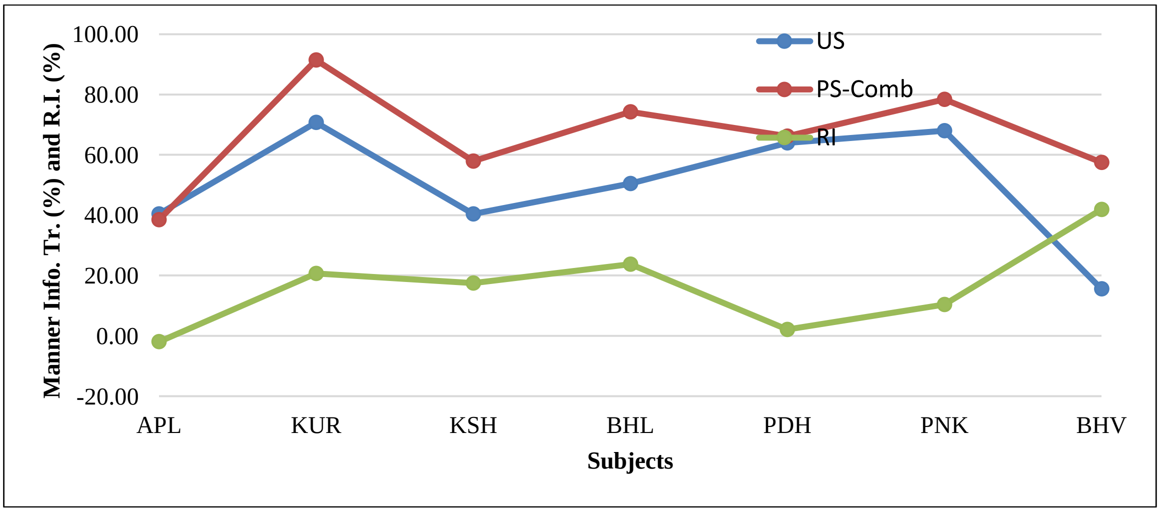

Fig. 18: *Relative Information Transmitted for Manner.*

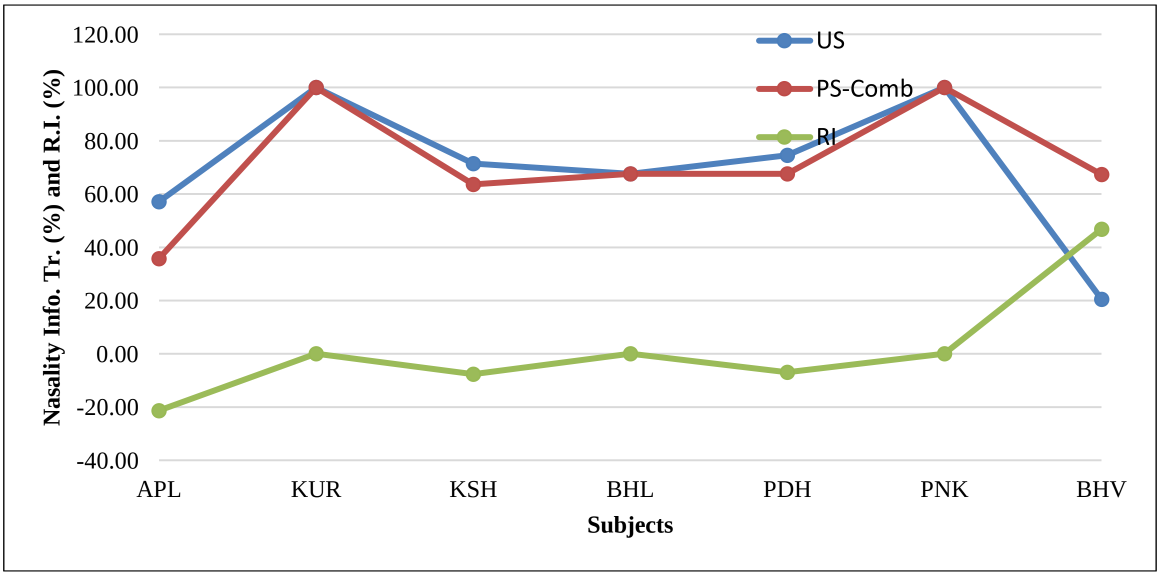

Fig. 19: *Relative Information Transmitted for Nasality.*

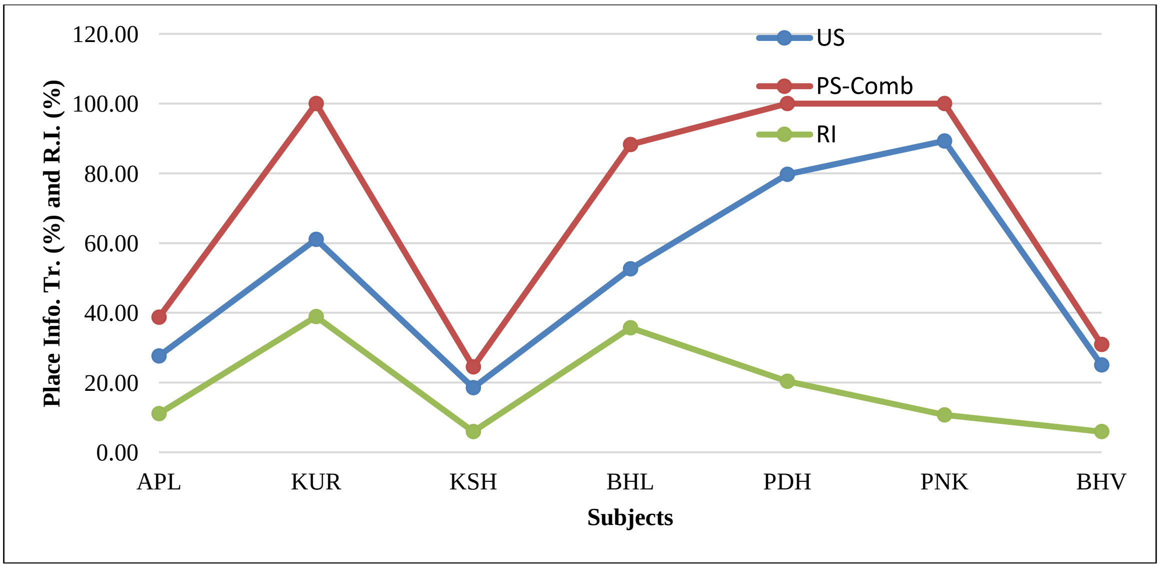

Fig. 20: *Relative Information Transmitted for Place.*

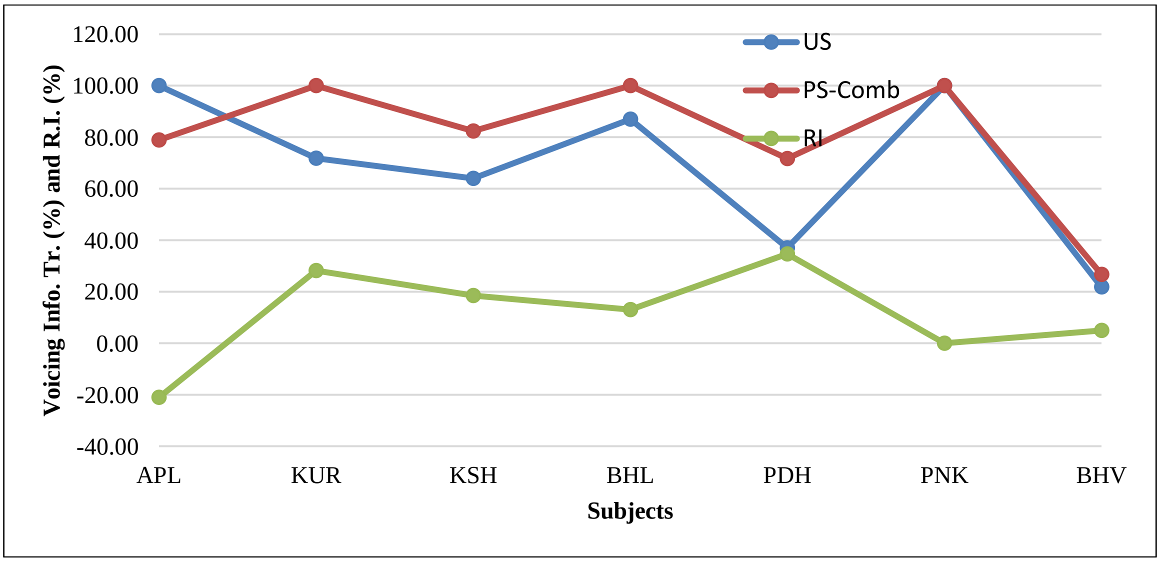

Fig. 21: *Relative Information Transmitted for Voicing.*

From Figure 14, the overall relative information transmission varied from 62.00% to 94.24% for unprocessed speech signal while for processed speech signal it ranged from 71.62% to 98.16%. The relative improvement was varied from -2.57% to 18.28% for all the subjects. The relative information transmission for continuance varied from 11.42% to 44.43% for unprocessed speech signal while it ranged from 37.59% to 86.49% for processed speech signal as shown in Figure 15. For the processed signal, the relative improvements were high for KUR and low for PDH. Figure 16 shows the relative information transmitted for duration feature. It varied from 41.12% to 76.35% for unprocessed speech signal while for processed speech signal it ranged from 24.72% to 100%. The relative improvements for the subjects BHL and KUR are high for the processed signal and were 58.84% and 42%, respectively. The relative information transmitted for frication feature is shown in Figure 17. For unprocessed speech signal varied from 10.52% to 57.73% while for processed speech signal it ranged from 38.31% to 86.94%. Subject PDH showed relative improvement of 7.16% (minimum) while subject BHL showed relative improvement of 40.22% (maximum). Figure 18 gives graphical representation of the relative information transmitted for manner feature. It varied from 40.37% to 70.73% for unprocessed speech signal while for processed speech signal it ranged from 57.50% to 91.43%. The relative improvement was varied from -1.97% to 41.95%. Figure 19 shows the relative information transmitted for nasality feature. For unprocessed signal, information transmitted varied from 20.52% to 100%. Its relative improvement varied from -21.32 to 46.78%, for processed signal. From Figure 20, the relative information transmission of place feature varied from 18.52% to 89.30% for unprocessed speech signal while for processed speech signal it ranged from 24.48% to 100%. For subjects KUR and BHL the relative improvements were high and for all subjects it varied from 5.89% to 38.89%. The relative information transmission for voicing feature is shown in Figure 21. It varied from 21.80% to 100% for unprocessed speech signal while for processed speech signal it ranged from 26.75% to 100%. The highest relative improvement was observed for subject PDH and lowest was observed for subject APL.

CONCLUSION

A comb filter based on auditory critical bandwidth having 512 coefficients with novel a simplified approach is designed. This approach requires the set of filter coefficients to obtain synthesizable VHDL code. This new approach of design helps to implement comb filter on FPGA Spartan6 XC6SLX45 CSG324, with no conversion complexity which is a significant improvement over state-of-the-art FPGA based techniques. Interestingly, the magnitude response of these filters is optimized for low perceived spectral distortion by having high stop band attenuation and low pass band ripple, hence useful for 'mild' to 'moderate' sensorineural hearing-impaired subjects. The usage of spectral splitting scheme was prominent as relative reduction in response time was observed. Results also show relative improvement in recognition score by 3.33% to 17.78% indicating improvement in consonantal identification. Dichotic presentation improves the reception of tough consonantal features like voicing, manner, and place for all subjects. Among these, place feature showed maximum improvement as the information of place feature is related to frequency resolving capacity of auditory process, the effect of spectral masking has been reduced. The average latency obtained matches to the desirable value. The low resource requirement of FPGA design shows the significance of the method, which can be effective for development of binaural hearing aids.

ACKNOWLEDGEMENTS

We express heartfelt gratitude to the Dr. Abuzar Shabbir Indorewala and Dr. Dnyandeo. K. Chopade for giving access to their esteemed hospitals. We would also thank audiologists Mr. Ramlal Singh and Mrs. Krupa for locating the hearing-impaired subjects and making the audiogram available to us and sincere thanks to all hearing-impaired subjects who actively participated in the listening test.

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Cite this Article
Chopade JJ, Futane NP, Kharate GK. Effective FPGA Implementation of Comb Filters to Improve Perception of Sensorineural Hearing Impaired. *Current Trends in Signal Processing*. 2017; 7(3): 15–29p.

