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Design & Performance Evaluation of Scalable Linear Frequency Transposition technique in Digital Hearing Aid

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ABSTRACT
High frequency hearing loss is growing issue in the state of Maharashtra (India). Significant numbers of Hearing Aid person are observed who's with Marathi mother tongue. Many researchers study frequency Transposition technique on different languages however unique phonology of Marathi language will make the difference. Hearing aid customers favored to use low price hearing aid bought by own or given through NGO's, which is no longer a successful to enhance Marathi speech attention in exceptional environments. Currently two methods are used to reduce high frequency hearing loss, frequency Compression (FC) & Frequency transposition (FT) significantly by hearing aid manufactures. Both methods are not able to assure HA user's need in different speech background environment. This paper includes design of linear frequency transposition with scalable performance parameters. Scalable LFT algorithm requires average gain 42 dB with lower average processing time of 0.46 sec for Marathi alphabets. Testing of proposed algorithm is carried on 4 Marathi language hearing aid users. Simulation & Testing results show that proposed method statically improves recognition rate of Marathi vowels, consonants, syllables in all type of environments by all hearing aid users.
Keywords - Hearing Aid, Hearing loss, Frequency compression (FC), Frequency transposition (FT), Marathi Alphabets.

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I. INTRODUCTION

High frequency hearing loss is a challenging issue in a society. It may be due to diseases effecting Middle ear or inner ear, may be congenital, age related or due to noise exposure. Various social and economic losses incurred by such individuals, many programs were conducted to help such persons by government or NGO's. In India, 63 million people (6.3%) suffer from significant hearing loss. The National Sample Survey (NSS) 58th round (2012) surveyed disability in Indian households and found that hearing disability was second most common cause of disability and top most cause of sensory deficit. In urban areas, loss was 9% of all disability and in rural areas it was up to 10%. Depending upon the extent of a person inability to properly, the degree of hearing disability was ascertained it was estimated that the number of person with hearing disability per 100000 persons was 291; it was higher in rural compared with urban regions (236). In the same survey, about 32% of the people had profound (person could not hear at all or could hear only loud sounds) and 39% had severe hearing disability (person could hear only shouted words). The survey results revealed that about 7% of people were born with a hearing disability. About 56% and 62% reported the onset of hearing disability at ≥ 60 years of age in the rural and urban areas, respectively.

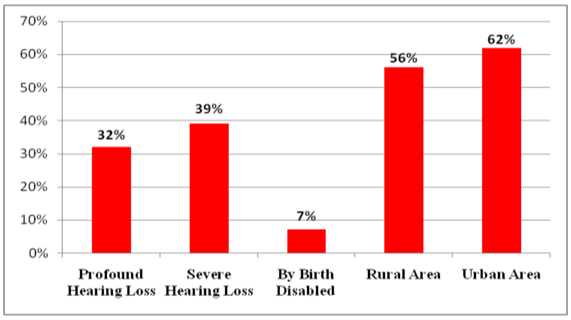

Figure1: Hearing Disabled Statistics

The incidence of hearing disability during that year was reported to be 7 per 100000 populations. Hearing loss is often associated with aging, hearing loss is clearly present in new born, children, teenagers, young adults and adults. Healthy human ears can perceive an enormous range of sounds in terms of pitch and loudness. Hearing is the primary sense through which we learn speech and language. The ability to hear clearly from birth is extremely important with regard to normal development of speech and language skills, auditory processing skills.

II. PROBLEM DEFINITION

A survey is done for four districts of North Maharashtra Region where 1647 persons found who were suffered from high frequency hearing loss. In Marathi language there are 15 vowels & 35 consonants in which phonology, pronunciation is quite different with English & mandarin language but phoneme inventory is similar to Indo Aryan language. In Marathi language vowels are structured as front, central & back to tongue movement, while there is more number of labial, dental, retroflex & velar consonants which became difficult to understand by all hearing aid users. A Marathi consonant conveys high information in words & plays an important role in words which are used in daily communication with hearing aid user. Hearing aid user is unable to understand many of Marathi consonants & vowels in all different speech background conditions. A survey on performance of own hearing aid by user was taken at deaf school in Shirpur (Maharashtra) which is run by NGO, it is found that many of students aged between 12- 25 was not satisfied with the performance of own hearing aid.

All of them face problems while recognizing many of Marathi words particularly when word structured from confusing group of consonants. In Marathi out of 35 consonants 23 to 25 consonants are difficult to understand by HA user, these consonants having same pronunciation, lip movement. By considering this we need to improve recognition rate of vowels, consonants for all Hearing aid user (students).These group of students suffered from moderate high frequency hearing loss so they got difficulty to understand to hear high pitched Marathi consonants, vowels.

Traditional hearing aid is just amplifying sound without knowing the requirement of HA user. To design frequency transposition scheme in Computer based simulation approach, we need to decide a set of parameters which is fitted in FT algorithm. These set of parameter plays vital role in performance of algorithm.

Where f s is the sampling frequency, N is the number of FFT points. Transposing higher-frequency components to a lower-frequency band will result in some distortion. To reduce the

III. FREQUENCY TRANSPOSTION SCHEME

Frequency transposition in which information specified high-frequency band is shifted downward by amplitude modulation or nonlinear distortion. Higher frequency (f H) is shifted to lower frequency (f L) by a fixed frequency value, it does not reduce bandwidth. In this method strong distortion occurs when shifting frequency is greater than signal frequency. These two FC & FT algorithms compress and shift the high-frequency contents in the same manner, but relocate them with original low frequency contents in a different way. In this method compressed high frequency contents are overlapped & mixed with original content.
This method works on linearity principal so called as L-OFT (Linear Overlapped frequency transposition).while designing frequency transposition method care should be taken to decide target band & source band, both should be designed as per requirement of HA user.

The target band in FT should be carefully determined. If the target band is too low, transposition might disturb the existing low-frequency components and negatively affect hearing for users with residual hearing at low frequencies. If the target band is too high, transposition is not meaningful since it will not preserve high-frequency information at sufficiently low frequencies that hearing-impaired listeners might be able to access. We set the target band to be 1000–2000 Hz. 2000 Hz is the cut-off frequency in our experiments to simulate a severe degree of hearing loss. 1000 Hz is chosen so that much of the Marathi vowel &consonants information and the natural pitch of the speaker are preserved since frequencies below 1000 Hz are left unchanged.

$$\mathrm{M1} = \sum_{i=0}^{N/2} \text{Avarage of Segment (l)} = [\mathrm{N} * \frac{\mathrm{Fs}}{l}] \tag{1}$$

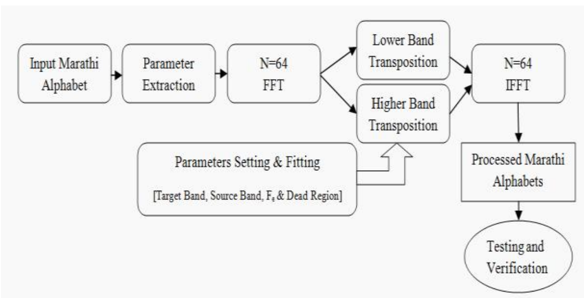

Figure2: Proposed Algorithm for Frequency Transposition

A fixed source band in FT may not adequately represent different speech signals with varying frequency components. Thus, we propose an adaptive source band on a frame-by-frame basis. It was shown that the first spectral moment M1, which can represent the centroid of the spectrum, well characterizes the consonants. M1 is calculated by equation (1) distortion in the hearing experience, we propose the concept of equal-loudness contour in our algorithm. Equal loudness is a psychoacoustics concept that represents the fact that the human auditory system has different sensitivity to different frequencies. We use the equal-loudness contour to calculate the difference in the sound pressure level (SPL) between the source frequency and the target frequency, assign a set of weights to compensate this difference. The weighted coefficients to the proposed algorithm is calculated by using equation (2)

$$\mathrm{WC}\,(\mathrm{l}) = 0.6[\Delta\,\mathrm{sH} - \Delta\,\mathrm{sl}] \quad (2)$$

The speech signal was sampled at 16 kHz and quantized in16 bits. The speech signal was divided into frames of 4 m sec (64 sample points).The FT-processed consonant frames are transformed back to the time domain by the Inverse FFT (IFFT) and overlap-added with the frames (64 points overlap between frames) . The spectral features (frequency based features) obtained after converting time domain to frequency Domain using Fast Fourier Transform (FFT). Fundamental frequency, frequency components, spectral centroid, spectral flux, spectral density, spectral roll-off are key features used in speech processing during frequency domain. Spectral analysis by using FFT is given by

$$P(k) = \left\| X(k)^2 \right\| \quad (3)$$

After processing frequency transposition algorithm processed speech file is ready to test for recognition in different environment by different hearing aid user.

Technique to reduce the speech degradation caused due to variation in vocal tract shape of speakers, a frequency warping approach is used to speaker speech parameters investigation. For Minimum Distortion at low frequency warping factor should

$$wf(a) = \left[\frac{(CR\text{-}1)}{(CR+1)} \right] \quad (4)$$

It is observed that to produce Compressed output frequency F_{out} =1000 Hz for incoming speech at different frequency warping factor limit will be 0.35, Then overlapping will starts at lower frequency speech band which will degrade speech quality. The spectral centroid is related with the measure of the brightness of a sound in spectrogram. This is obtained by evaluating the center of gravity using the Fourier transform's frequency and magnitude information. N point FFT leads to increase processing time in consonants and vowel. Computational time in FFT is challenging issue during processing.

IV. PERFORMANCE PARAMETERS IN TRANSPOSTION METHOD

All the fitting and testing processes were done in a silent class room, Crowdie class room in a School, with the laptop & wireless ear set (Bluetooth -10 meter range) in front of the listeners.

The testing sounds were broadcasted at 60 dB without considering the level of background noise.

The objective to design hearing aid is to reduce moderate hearing loss for 6 HA Users, who has Difficulty to hear high frequency& SPL to reach up to 60 dB, Steps are as follows

- The Marathi Vowels & Consonants (With & without Noise) has been input as wave file.
- De-noise function has been used to remove noise from speech signal.
- Initialized the frequency vector for frequency range 1000 to 2,000 Hz.
- Time domain signal was converted to frequency domain by using 64 point FFT function.
- The minimum and maximum limit of desired gain is 60 & 90 dB respectively.
- Set the gain for second range of frequencies
- The pointer K would be modified in between1000 to 2000 Hz.
- Processed Marathi Speech Samples are randomly play backed & listened to HA user by using Laptop & Wireless headset in different environments.

Multiple experiments were conducted to each listener for improvement & decision to decide set of needed parameter for testing algorithm.

Parameters Fitting

- Transposed coefficient is designed from 30 to 70 % for all HA users & Comfort Level is taken at 90 % of maximum of HTL (Pain).
- Threshold gain of all vowels & consonants is constant of 60 dB (for all background Condition)

By implementing & fitting these parameters in proposed algorithm, we try to calculate processing time calculation for all Marathi vowels & consonants. Processing time is challenging issue to design new algorithm.

To maintain synchronization in lip reading & spoken vowel, consonants we need to reduce this as less as possible. In this algorithm 64 Point FFT Where N=64 decides processing time of each vowel & consonant. To design high point FFT is promising issue as value of N increase the processing time also increases.

Vowel	SPL in dB	Gain in dB	Processing Time in Sec	Vowel	SPL in dB	Gain in dB	Processing Time in Sec
अ	17.239	42.760	0.4800	ओ	17.385	42.614	0.6002
आ	17.317	42.682	0.1741	औ	17.375	42.624	0.5929
इ	17.074	42.925	0.2915	अ	17.438	42.562	0.5108
ई	17.198	42.801	0.4262	अ	17.416	42.583	0.0894
उ	17.006	42.993	0.5130	अ	17.559	42.440	0.4003
ऊ	17.192	42.807	0.5798	ऑ	17.464	42.535	0.5176
ए	17.535	42.464	0.6842	ऋ	17.380	42.619	0.6501
ऐ	17.570	42.429	0.587539217	**Average**	**17.343**	**42.656**	**0.4612**

Table 1: Gain & Processing Time for Frequency Transposition Algorithm (All vowels).

	Ava. SPL in dB	Gain Needed	Ava. Processing Time in Sec
All Consonants	17.294	42.701 dB	0.51567546

Table 2: Gain & Processing Time for Frequency Transposition Algorithm. (All Consonants)

We record 4 Male & 2 female speakers' spoken vowels, consonants in different speech background, then average of speaker's Sound pressure level (SPL in dB) taken to decide gain needed to add in algorithm before processing. Average gain required to add is nearly same for all vowels & consonants but consonants need slightly more time to process than all vowels.

V. TESTING & RECOGNTION RESULTS

Marathi vowels & Consonants in a speech are different from English speech in many Aspects. Unlike English, Marathi has no allophone of consonants. Moreover, Marathi has its own distinct features such as intonation, syllable, light tone, and retroflex finals. Therefore, a more detailed recognition test in the phoneme level was taken in this study. The averaged comparative results of phoneme recognition rates between the proposed algorithm and the listeners' own hearing aids are shown in Fig.3(a,b,c,d &e) where 35 consonants and 15vowels were tested in this test. Most HA people from North Maharashtra had difficulty in distinguishing consonants. To recognize these consonants correctly by own Hearing Aid in different Conditions is challenging issue for all user. Fig 3.e shows recognition difference between Own & Proposed Method, in which most of confusing Consonants are clearly understood by group of 4 Hearing aid users.

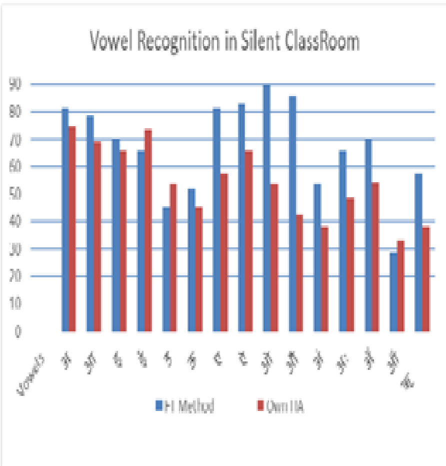

Figure 3a

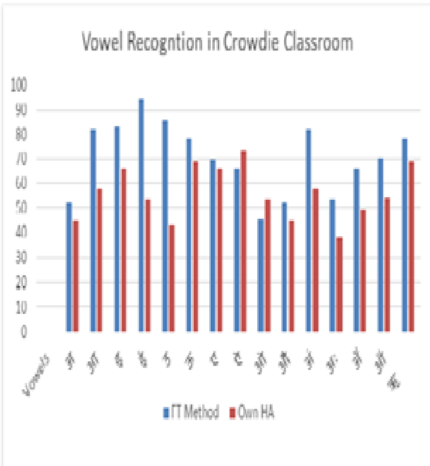

Figure 3b

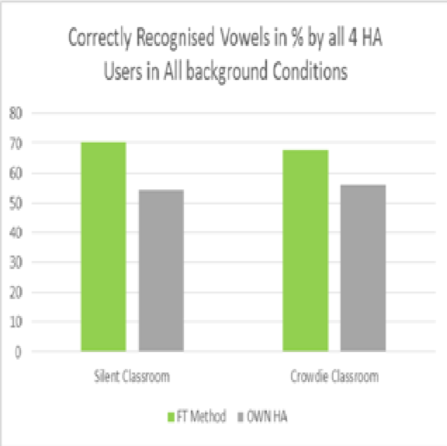

Figure 3.c

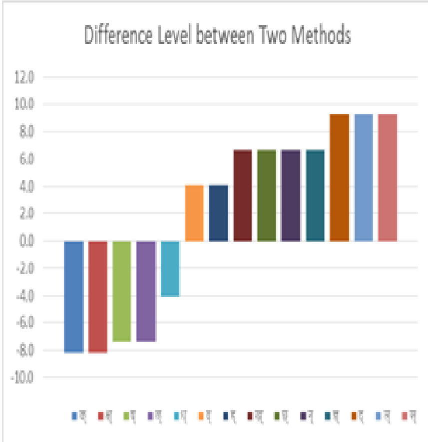

Figure 3.d

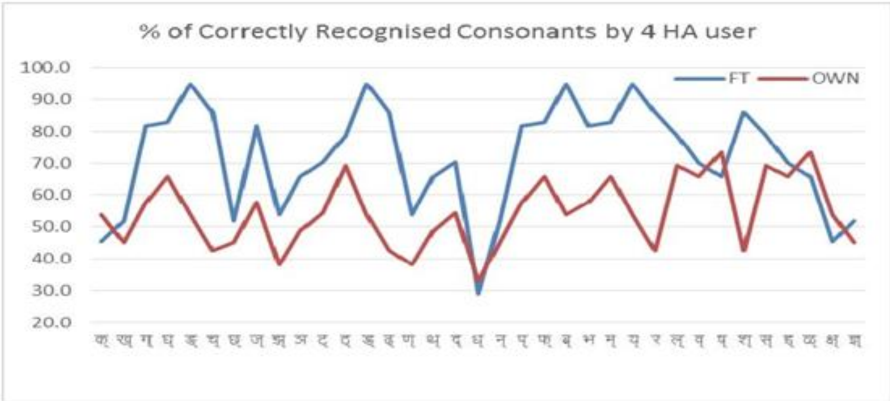

Figure 3.e

Figure 3.a: Vowel recognition test (Silent Room) for FT algorithms & own HA.
3. b: Vowel recognition test (Crowdie Room) for FT algorithms & own HA.
3. c: Vowel recognition test in % Comparison for FT algorithm & own HA.
3. d: Recognition rate difference between FT & own HA.
3. e: Correctly recognized consonants in % by all 4 HA user.

VI. CONCLUSION

MATLAB based frequency Transposing Method is more capable for fine-processing the speech with less distortion in the quality. Proposed Method is an approach towards satisfaction of hearing aid user; it is possible to refine the sound signal. For instance by transposing the higher frequency on lower band of frequency occurs few overlap in speech signals, but aim to retain higher frequency speech signal was satisfied. In addition, by using this approach the transposition will be done only at the range of frequencies which user needs to modify. This will eliminate the problems with unwanted processing of comfort zone frequency band. These method shows that 4 HA users are satisfied with configurable approach but take it needs lots of experiment to find suitable set of parameters. Gender wise every listener need to set different parameters which is difficult & time consuming process. To overcome this condition in future we extend towards Neural network & Genetic Algorithm based Transpose where manual adjustment will done by optimal set of solution.

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