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Uniform Frequency Sub-Band filtering Approach for Marathi Alphabets Processing

Prashant G Patil*[1], Patil Sheetal N[2], Dr. T. H. Jaware[3]
[1]Research Scholar, Department of Electronics Engineering, MIET Gondia (MS) India.
[2,3]Assistant Professor, R C Patel Institute of Technology, Shirpur (MS) India.

***Corresponding author**
Prashant G Patil

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Abstract: Speech enhancement aims to improve speech quality by using various algorithms. The objective of enhancement is improvement in intelligibility using audio signal processing techniques. The choice of the filter-bank has a significant influence on the performance of such systems in terms of signal quality, computational complexity and signal delay. Accordingly, the filter-bank design has to fulfill different, partly conflicting requirements in dependence of the considered application. In this paper we tried to design Low Pass, High pass & band pass Filters to process Natural Spoken Marathi alphabets. We also processed Segmented & overlapped Alphabets to compare our algorithm performance. Major perspective to design filter bank is to select particular filter band to process respective vowel & consonant.
Keywords: Frequency Sub band; Speech; Marathi; Uniform Filtering; Formant Frequency

INTRODUCTION

In speech processing, a filter bank is an array of filters that separates the input signal into multiple components, with each one carrying a single frequency sub-band of the original signal. In Filter bank we can attenuate the components differently and recombine them into a modified version of the original signal [2]. The choice of the filter-Bank has a significant influence on the performance of such systems in terms of signal quality, computational complexity and signal delay [5].

Accordingly the filter-bank design has to fulfill different, partly conflicting requirements in dependence of the considered application.

Proposed Methodology

In this paper we record Marathi Vowels & Consonants in different background condition. Which are processed through a bank of filters (Low Pass, High Pass & band Pass Filters) [1]. Formant Frequency like F1, F2, and F3 of each speech sample is calculated to select appropriate sub band filter for processing .Formant frequency plays vital role in speech processing [9]. A formant is a concentration of acoustic energy around a particular frequency in the speech wave. There are several formants, each at a different frequency, roughly one in each 1000 Hz band. Or, to put it differently, formants occur at roughly 1000 Hz intervals. Each formant corresponds to a resonance in the vocal tract. Frequencies of formants change only within 15% between female and male speakers. Recorded speech is sampled at 20 KHz rate with 16 bit numbers a second.

We divide the frequency range of interest (say 100-8000Hz) into N bands and measure the overall intensity in each band. This could be done using hardware or digital filters directly from the incoming signal, or be computed from a spectral analysis (again derived using hardware or software such as the Fast Fourier Transform) [3]. In a uniform filter bank, each frequency band is of equal size. For instance, we used 8 ranges; the bands might cover the frequency ranges.

BandI-100Hz-1000Hz...................BandII-1000Hz-2000Hz……………..VIII Band-8000Hz.

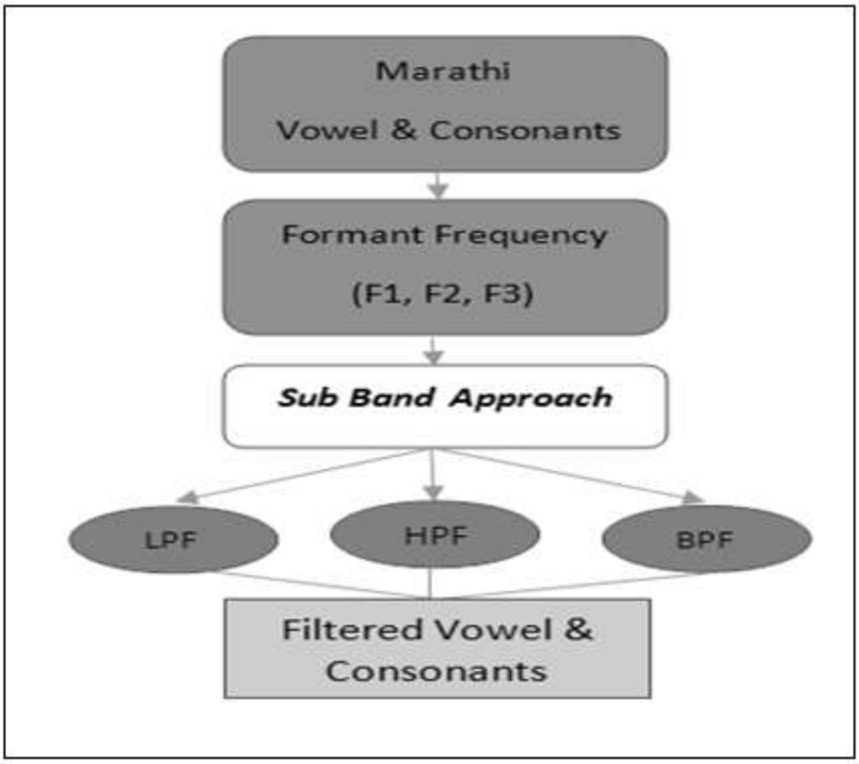


Fig-1: **Marathi Alphabet processing algorithm.**

Filter Designing Criteria

The speech signal is applied to filter bank and these filters in the bank are discrete time domain filters. The filters can be represented by following difference Equation [10]:

$$\boldsymbol{y(n) = -\sum_{k=1}^{N} a(k).\, y(n-k) + \sum_{k=0}^{M} b(k).\, x(n-k)} \quad (1)$$

From this equation, note that y(n-k) represents the outputs and x(n-k) represents the inputs, ak; k = 1; 2:::N , bk; k =1; 2:::M are called the filter coefficients [4,7]. The value of N represents the order of the difference equation and corresponds to the memory of the system being represented. The filtering problems stated above differ according to what frequency range is desired relative to the frequency ranges that need to be attenuated, as listed below.

- If it is required to allow frequencies up to a certain cutoff limit and suppress frequency components higher than the cutoff frequency, such a filter is called a Low-Pass Filter. For example, in many systems most of the noise or harmonics may be concentrated at higher frequencies, so it makes sense to perform Low-Pass Filtering [6].
- If it is required to allow frequencies beyond a certain cutoff limit and suppress frequency components lower than the cutoff frequency, such a filter is called a High-Pass Filter [8].

Formant Frequencies of Marathi Alphabets

Consider a filter bank representation of artificially generated spectra similar to that for the Marathi vowel as shown in Figure 2. We can measure the intensity in each band by computing the “area” under the curve. With a discrete set of sample points, we could simply add up all the values in range, or compute a “power” measure by summing the squares of the values. With the signal shown in Figure 2, following are formant frequencies of Marathi vowel & consonant (F1, F2, F3…) .According to these frequencies they are passed to sub band filters for processing purpose. A better alternative is to organize the ranges using a logarithmic scale, and this actually agrees better with human perceptual capabilities as well. We set the first range to have the width W, and then subsequent widths are W.

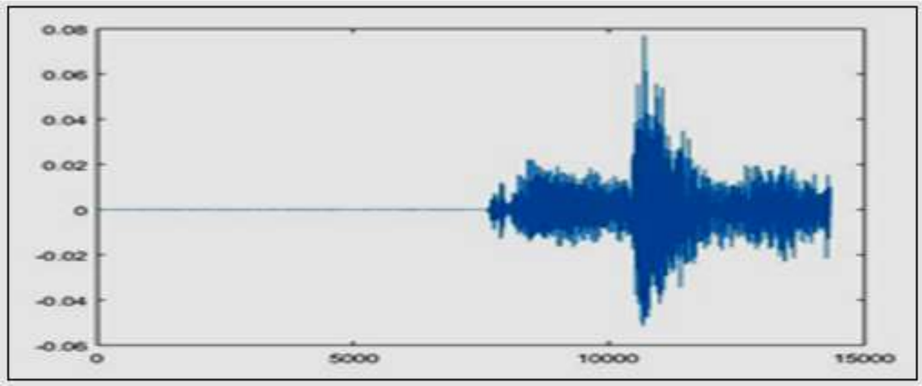


Fig-2: **Marathi consonant "अ" frequency spectrum.**

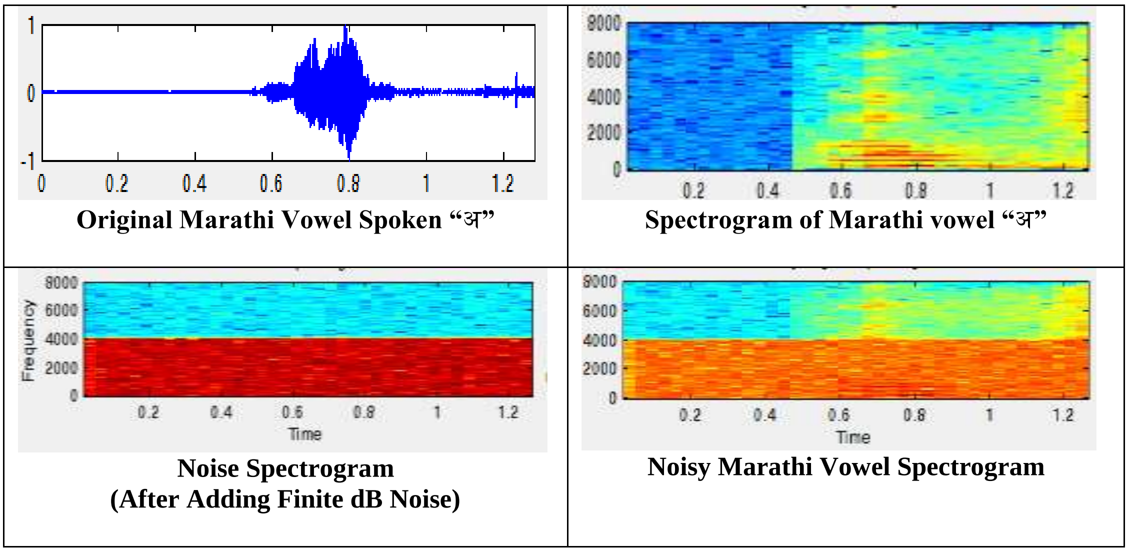
If a = 2 and W is 200 Hz, we get widths of 200 Hz, 400 Hz, 800 Hz, 1600 Hz, and so on. According to F1 we can select a particular sub band for processing this corresponding Vowel. Selection of particular sub band filter plays a vital role; in our method we have LPF, HPF, and BPF with wide range of cutoff frequencies & noise insertion dB range.

Table-1: **Formant frequency F1, F2, F3& F4 for Male & Female Speaker**

Vowel	F_1 in Hz (M)	Vowel	F_1in Hz (F)		Vowel	F1	F2	F3	F4
अं	966	अं	990		अ	757	1760	3031	4173
अ	921	अ:	923		आ	920	1532	3063	4185
आ	831	आ	920		इ	885	2093	3348	4332
ऑ	828.7	अॅ	893		ई	857	2135	3386	4338
अं	793.2	इ	885		उ	715	1892	3171	4287
अ:	721	ऋ	874		ऊ	751	1928	3378	4440
ओ	640	ई	857		ए	689	2227	3156	4119
औ	640	ऑ	863		ऐ	590	2009	3049	4074
ऊ	571	ओ	782		ओ	782	2044	3447	4344
ऐ	559	अ	757		औ	711	1944	3421	4230
उ	548	ऊ	751		अं	990	2007	3470	4311
ए	540.7	उ	715		अ:	923	1705	3460	4299
ऋ	538	औ	711		अॅ	893	1791	3278	4217
ई	491	ए	689		ऑ	863	2045	3544	4329
इ	479	ऐ	590		ऋ	874	2098	3508	4421

SIMULATION RESULTS & DISCUSSION

Figure 2 shows original Marathi vowel with its spectrogram, amount of noise is added in spoken speech & noise included vowel spectrogram. After that we have bank of filter selecting LPF, HPF & BPF speech details with frequency variation are given in figure 3.



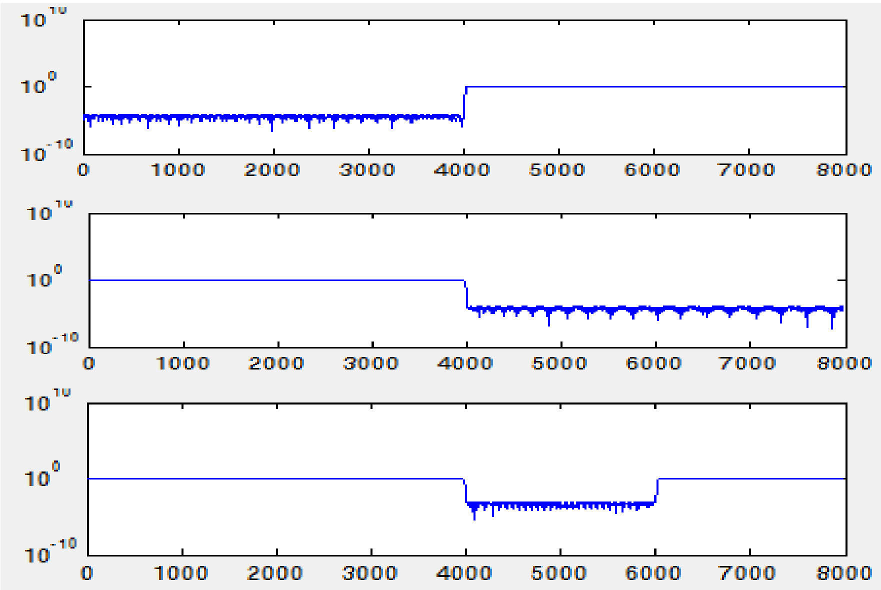
Original Marathi Vowel Spoken "अ"

Spectrogram of Marathi vowel "अ"

Noise Spectrogram
(After Adding Finite dB Noise)

Noisy Marathi Vowel Spectrogram

Fig-3: **Marathi vowel "अ" frequency spectrum & Noise Spectrogram, Noisy Speech Spectrogram**



Low Pass Filter @ 4000 Hz

High Pass Filter @ 4000 Hz

Band pass Filter @ 4000 to 6000 Hz

Fig-4: **Filtered speech response on Marathi Vowel "अ".**

CONCLUSION

In order to verify the speech enhancement capability of the designed filter bank, several experiments were carried out. For this purpose a single word speech signal is taken & noise is added with different levels (in db).The speech signal is then passed through the filter bank. Spectrograms are calculated for each of the filtered speech. Spectral values corresponding to the high gain regions are extracted from the respective spectrograms. The extracted values are vertically concatenated to form the final spectrogram. For the noised words, the spectrograms obtained through the proposed method are more informative comparing to the conventional spectrograms. The spectrograms generated by the conventional method and by the proposed filter method for single vowel effectively processed .Followed by that, 0db noise is added to the vowel & consonants and spectrograms are generated by the conventional method and by the filter bank method. The spectrograms observed by proposed method shows remarkable improvement in noise removed level in the vowel & consonants.

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