Frequency Compression of Speech for the Hearing Impaired: An FBS Approach

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Abstract: - Hearing impairment leads to loss of social connection. It has become the most common of disabilities in the present world scenario. Numerous things can damage the complex and delicate structure of the cochlea causing hearing deficit called sensorineural hearing loss (SNHL). Of the two major types of hearing losses, conductive hearing loss is often corrected with surgery. On the other hand, hearing aids are the only way to compensate for SNHL. Hearing impairment affects many aspects of audition such as audibility, dynamic range, spectral and/or temporal resolution, etc., in particular if background noise is present.

Enhancing spectral information of hearing-impaired people can compensate the spectral masking, which occurs due to SNHL Frequency compression is one of the more appropriate signal processing techniques to compensate for spectral masking. The primary goal of this paper is to develop an efficient technique of frequency compression using a Filter Bank Summation (FBS) method to enhance spectral information of hearing impaired people.

Key-Words: - Sensorineural hearing loss, hearing impairment, spectral masking, speech enhancement, frequency compression and filter bank summation.

1 Introduction

SNHL exhibits widening of auditory filters due to increased spectral masking, resulting in severe smearing of spectral envelope. Glasberg and Moore [1] measured an auditory filter of Hearing Impaired (HI) and Normal Hearing (NH) people and reported that HI subjects have a wider auditory filter than NH. Broader auditory filters produce a more highly smoothed representation of the spectrum than the normal auditory filters. This causes poor performance of speech recognition by HI listeners. Normally vowels are characterized by formant frequency cues, which are widely separated from each other; hence their perception is not much affected. However, perception of consonants is severely degraded, since it requires discrimination of sub phonemic segments like formant transitions and noise bursts [2].

Various methods of processing speech have been proposed for reducing the spectral masking. A scheme of binaural dichotic representation [3] was proposed by D. S. Choudhari and P. C. Panday. In this scheme, a speech signal was split into 18 critical bands corresponding to the auditory filters and a set of odd-numbered bands was presented to the subject's right ear, while the rest was presented to the left ear. In spectral splitting, sensory cells of the two ears corresponding to alternate bands are always relaxed. The scheme was implemented in real time processing for use as a binaural hearing aid. This technique was found to be useful in understanding the speech in noise, for the people with sensorineural hearing impairment with residual hearing in both the ears. Another method was proposed by Keiichi Yasu and others [4] in which critical-band was compressed along the frequency axis in light of the shape of the auditory filters of hearing-impaired people

The aim of present study is to develop an efficient method for the hearing aids to reduce spectral masking using FBS method. Here we have split a signal into subbands using an "analysis" filter bank that consists of 15 frequency bands.Then re-assembled subbands using a "synthesis" filter bank that again consists of 15 frequency bands but with reduced ranges based on frequency compression. The method for signal decomposition uses a set of bandpass filters (BPFs) to partition the signal spectrum into one-third-octave subbands. These subband signals are then combined by synthesis filter bank that accomplishes the frequency compression with given specifications.

2 Filter Bank Under Investigation

In this section, we specify the design of the proposed filter bank considered for investigation. We list the frequency range of the individual 15 bands of the non-uniform filter bank and state the specifications of the filterbank design. Table 1 gives the required specifications of the filter bank. These 15 bands are taken as one-third octave. Table 2 gives the frequency range for the 15 non-uniform bands of the analysis filter bank and Fig.1 shows the frequency responses of the same. Tables 3 gives the frequency range for the 15 non-uniform bands of the synthesis filter bank to accomplish frequency compression with compression factor (CF) 0.3 and Fig.2 shows the corresponding frequency response of individual frequency bands. We limit the upper frequency range to 5744 for the analysis filter bank.

The filter bank for filter bank summation (FBS) method is designed and simulated using MATLAB. All the frequency bands are designed so that they tolerate a maximum passband ripple within 1dB and minimum attenuation of 40 dB. These frequency bands are realized using Kaiser Window method with different filter orders. It has been tried to make the order of the filters as less as possible to make it comfort for real time implementation.

The simulation is run further. The procedure is as described follows. First, a simulink model is developed using Digital Filter Design Analysis Tool (FDATool) and other necessary blocks. The block parameters are set according to the specifications required. Then a sample speech signal is sent to the filter bank for frequency compression. The output signal is obtained after undergoing frequency compression.

Table 1 Specifications of the filterbank

	Specifications
Sampling frequency	12kHz
Maximum passband ripple	< 1 dB
Minimum stopband attenuation	> 40 dB

Table 2 Frequency range for the 15 Non-uniform bands of the analysis filter bank

Band	Lower Frequency (Hz)	Higher Frequency (Hz)	Bandwidth (Hz)
1	180	226	46
2	226	284	58
3	284	358	74
4	358	452	94
5	452	570	118
6	570	718	148
7	718	904	186
8	904	1140	236
9	1140	1436	296
10	1436	1810	374
11	1810	2280	470
12	2280	2872	592
13	2872	3618	746
14	3618	4558	940
15	4558	5744	1186

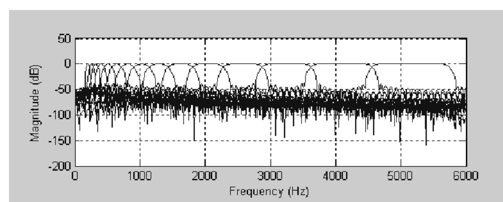

Fig. 1 Frequency responses of the 15 frequency bands of the analysis filter bank

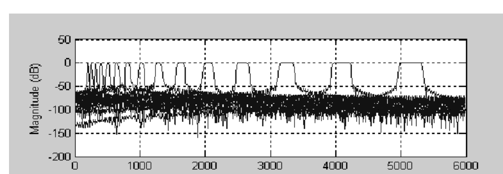

Fig. 2 Frequency responses of the 15 frequency bands of the synthesis filter bank to accomplish frequency compression of 0.3

Table 3 Frequency range for the 15 Non-uniform bands of the synthesis filter bank to accomplish CF=0.3

Band	Lower Frequency (Hz)	Higher Frequency (Hz)	Bandwidth (Hz)
1	196	210	14
2	246	264	18
3	310	332	22
4	391	419	28
5	493	529	36
6	622	666	44
7	783	839	56
8	987	1057	70
9	1244	1332	88
10	1567	1679	112
11	1974	2116	142
12	2487	2665	178
13	3133	3357	224
14	3947	4229	282
15	4973	5329	356

3 Results and Discussions

Spectra of the unprocessed and the processed speech signals were studied using spectrograms. The narrowband spectrograms for both the signals are shown below. The spectrograms for the unprocessed broadband noise and a VCV (Vowel-Consonant-Vowel) syllable /aba/ are shown in Fig. 3 and Fig.4, respectively. The spectrograms for the processed broadband noise and the processed speech signal are shown in Figs. 5 to 10. By comparing the spectrograms of both the unprocessed and the processed signals we can observe the spectral notches produced in the processed signal due to frequency compression. This clearly illustrates the frequency compression.

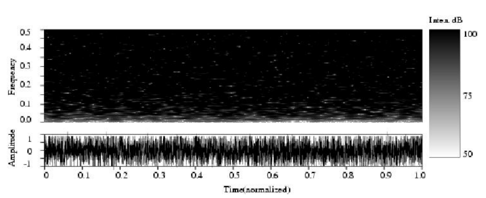

Fig. 3 Narrowband spectrogram of unprocessed noise

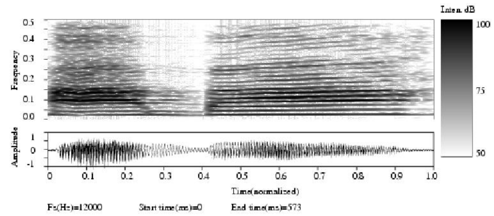

Fig. 4 Narrowband spectrogram of unprocessed VCV syllable /aba/

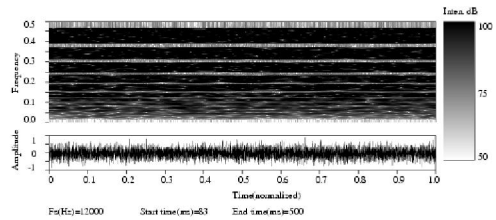

Fig. 5 Narrowband spectrogram of processed broadband noise with CF=0.7

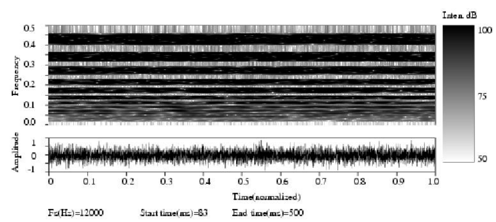

Fig. 6 Narrowband spectrogram of processed broadband noise with CF=0.5

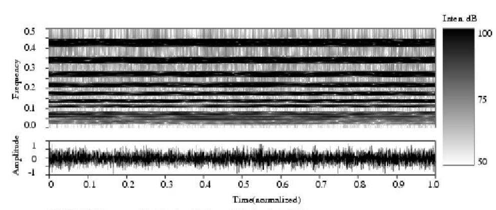

Fig. 7 Narrowband spectrogram of processed broadband noise with CF=0.3

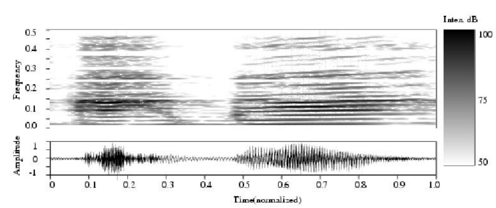

Fig. 8 Narrowband spectrogram of processed /aba/ with CF=0.7

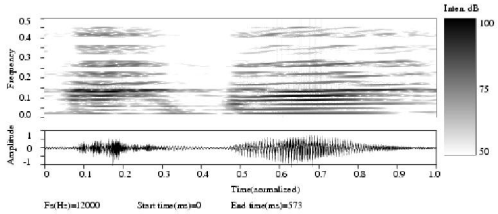

Fig. 9 Narrowband spectrogram of processed /aba/ with CF=0.5

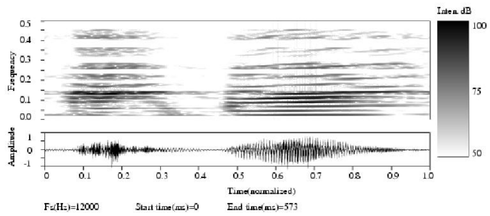

Fig. 10 Narrowband spectrogram of processed /aba/ with CF=0.3

Further we have gone for listening test to compare the quality of the unprocessed and processed signals under simulated hearing loss (i.e., by adding broadband noise). These tests showed an improvement in speech perception of processed signal under simulated hearing loss. The perception is improved with the higher frequency compression. The improvement in speech intelligibility was observed for all the compression factors ranging from 0.7 to 0.3. No further improvement was observed when the compression factor was reduced below 0.3. SNR of +3, 0, -3, -6 are considered in the experiment.

4 Summary and Future Directions

The complex multifaceted problems associated with SNHL presents a challenge for hearing aid development and it is difficult to treat all aspects of the problem to satisfaction. Improving the intelligibility of speech in noise is one of the most challenging tasks for researchers.

The frequency compression technique investigated in this project makes use of a simple straightforward filter structure. Several tradeoffs have to be made to make a practical filter bank that can meet the requirement. Informal listening tests were conducted to assess the usefulness of the algorithm, in improving the intelligibility of speech in noise. The study shows a clear improvement in intelligibility at 0, -3, and –6dB SNR. The compression factor was ranging from 0.7 to 0.3. Higher compression is found to be more useful.

The effort put here can be a step toward developing a real-time frequency compression system, using a DSP for the benefit of the hearing impaired.

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