|| Volume 4 || Special Issue 9 || ICCEME 2019-2020 || ISSN (Online) 2456-0774
INTERNATIONAL JOURNAL OF ADVANCE SCIENTIFIC RESEARCH
AND ENGINEERING TRENDS

ADJUSTABLE FILTER BANK FOR SPEECH

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Abstract: Filter is essentially just a design to extract different frequency bands or subbands from a given audio signal. So any Application with audio processing that depends on the information content present in different bands is most likely make use of filters. Main drawback of fixed filter bank is that, it gives fixed output after processing the audio input. It does not have variability in its audio processing system and hence there are some limitations on usage of fixed filter bank.

This design is applicable in hearing aid and speech processing domain, as approximately 10% of world's population suffers from this type of hearing loss. Yet only small percentage of this stastictics uses a hearing aid. Thus it is necessary to divide signal in non-uniform subbands in runtime. So according to user's requirement the number of subbands are to be adjustable that user can select the number of subbands in which signal is divided non uniformly. Selection of number of subbands is done by user and user have to give control parameter. This is the motivation for designing adaptive filter bank.

Keywords: *DSP,LMS*

I INTRODUCTION

Audio signal processing is a subfield of signal processing that is concerned with the electronic manipulation of audio signals. Audio signals are electronic representations of sound waves longitudinal waves which travel through air, consisting of compressions and rarefactions. The energy contained in audio signals is typically measured in decibels. As audio signals may be represented in either digital or analog format, processing may occur in either domain. Analog processors operate directly on the electrical signal, while digital processors operate mathematically on its digital representation. Speech signal is also a form of audio signal. Speech processing is the study of speech signals and the processing methods of signals. The signals are usually processed in a digital representation, so speech processing can be regarded as a special case of digital signal processing, applied to speech signals. Aspects of speech processing includes the acquisition, manipulation, storage, transfer and output of speech signals. Digital signal processing (DSP) is the use of digital processing, such as by computers or more specialized digital signal processors, to perform a wide variety of signal processing operations. The signals processed in this manner are a sequence of numbers that represent samples of a continuous variable in a domain such as time, space, or frequency. Digital signal processing and analog signal processing are subfields of signal processing. Digital signal processing applications include audio and speech processing, sonar, radar and other sensor array processing, spectral density estimation, statistical signal processing, digital image processing, signal processing for telecommunications, control systems, biomedical engineering, seismology, among others. The application of digital computation to signal processing allows for many advantages over analog processing in many applications, such as error detection and correction in transmission as well as data compression. DSP is applicable to both streaming data and static (stored) data.

Filter bank systems have made their appearance in the speech signal processing and communication communities nearly at the same period of time. Indeed, in the early 1970s, Schafer includes them among the recent and useful tools for speech spectrum analysis, whereas Bellanger and Daguet propose a multiplexer, i.e., the dual of an analysis–synthesis filter bank system, for telephony applications (for conversions between Time-Division-Multiplexing (TDM) and FDM based transmission schemes). However, a few years later, digital signal processing clearly became the most prominent application for filter banks. The applications behind were most often related to subband coding for source signals like speech, audio, image, and video.

II PROPOSED SYSTEM

Block diagram for proposed adjustable filter bank is shown in fig. 1 which mainly contains the input stage, low pass filter and manipulation system. It is representation of all components used for design of adjustable filter bank. Fig. 2 is extended block diagram of manipulation system. Manipulation system block diagram contains noise reduction, frequency shaper and amplitude shaper filters.

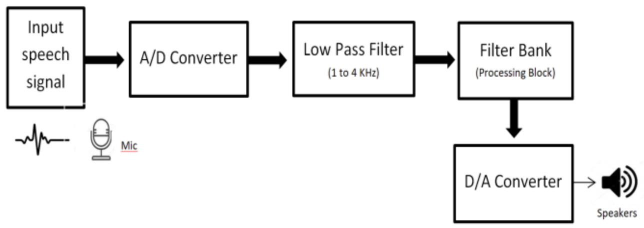

Figure 1: General Block Diagram

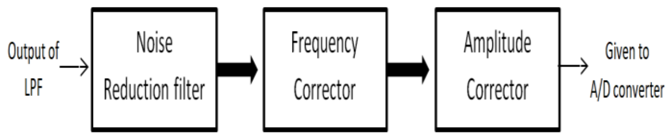

Figure 2: Manipulation System

The block diagram is divided into two parts as shown in Fig. 1 and fig. 2 namely general block diagram and manipulation system respectively. General block diagram contains input stage, low pass filter and manipulation system. Manipulation system block diagram contains processing tools as noise reduction filter, frequency shaper and amplitude shaper. Signal processing is done on board and thus output is given to speakers or headphones to analyze results.

Input: Mic(through which input speech signal is taken) or Audio file.

If there is a speech signal then we have to use mic for recording the input. Also if input is an audio file then there is no need of mic. We can directly give the input to the board

Low pass filter: Low pass filter is used to remove the noise content or high frequency parts from the input signal. It attenuates all high frequency components from signal and allows to pass low frequency signals.

III MANIPULATION SYSTEM

Manipulation system is an series of processing blocks as noise reduction filter, frequency shaper and amplitude shaper. Output of low pass filter is given to manipulation system.

1. Noise reduction filter: This filter uses LMS (least mean square) algorithm to reduce the noise. It separates background unwanted interference from speech signal.

2. Frequency shaper: This block applies gain more than 1 for hard to hear frequencies and modifies gain for other specified range. The frequency shaper is designed to correct for loss of hearing at certain frequencies. Frequency shaper applies a certain gain based on the users specific hearing loss.

3. Amplitude shaper: It checks bit by bit that the output power does not exceed a given saturation level. Amplitude shaper also helps in removing noise at low power levels.

Output: Speakers or Headphone

Output is an analog audio or speech file which is processed output as per users requirement. Speakers and headphones are used to analyze the output result by hearing it.

IV RESULTS

In this first we are taking an audio signal as input. After that the signal is passed through low pass and high pass filters. The code for plotting the high pass signal, low pass signal and their magnitude responses are shown in fig. 3which is MATLAB 2017 window

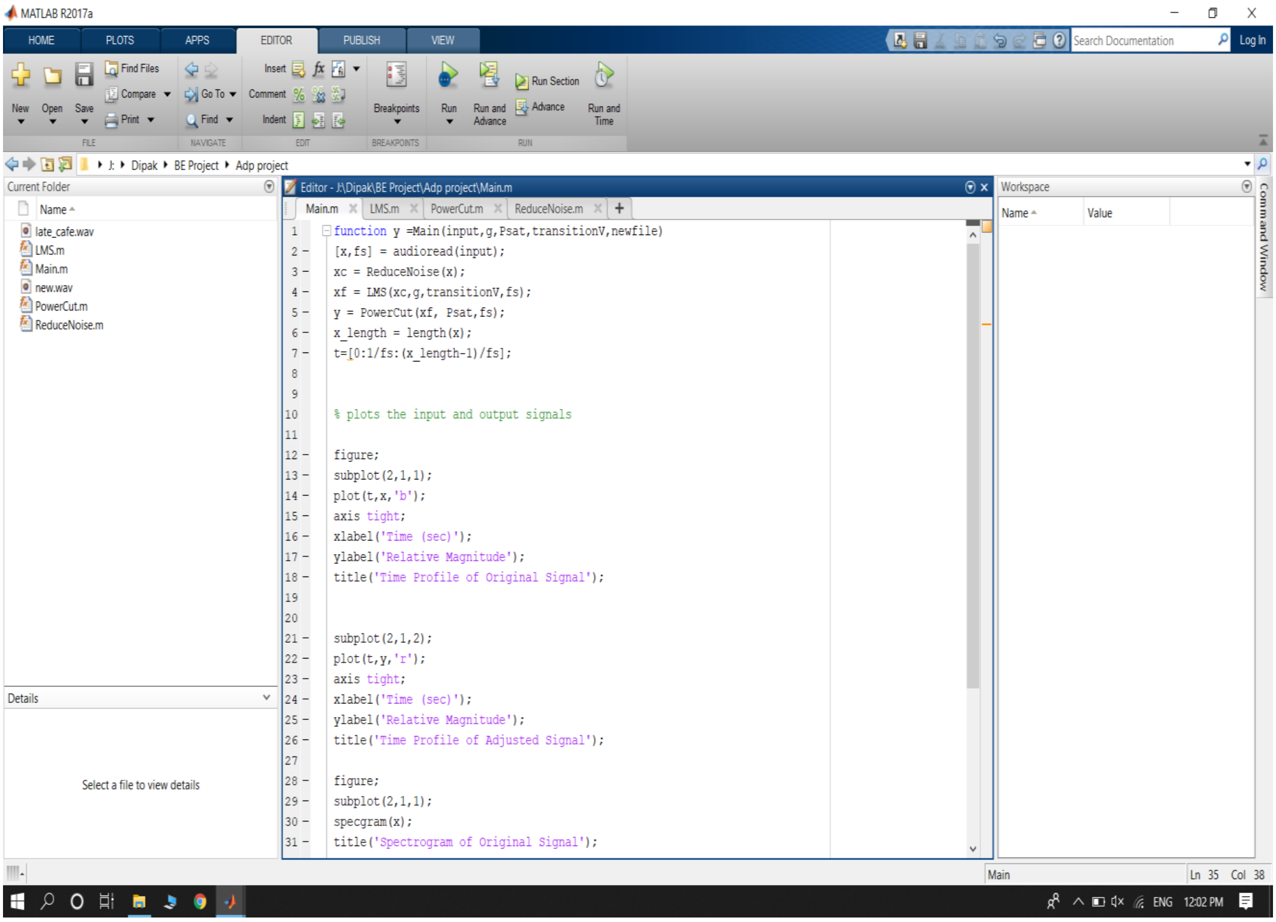

Figure 3 Project code on MATLAB 2017

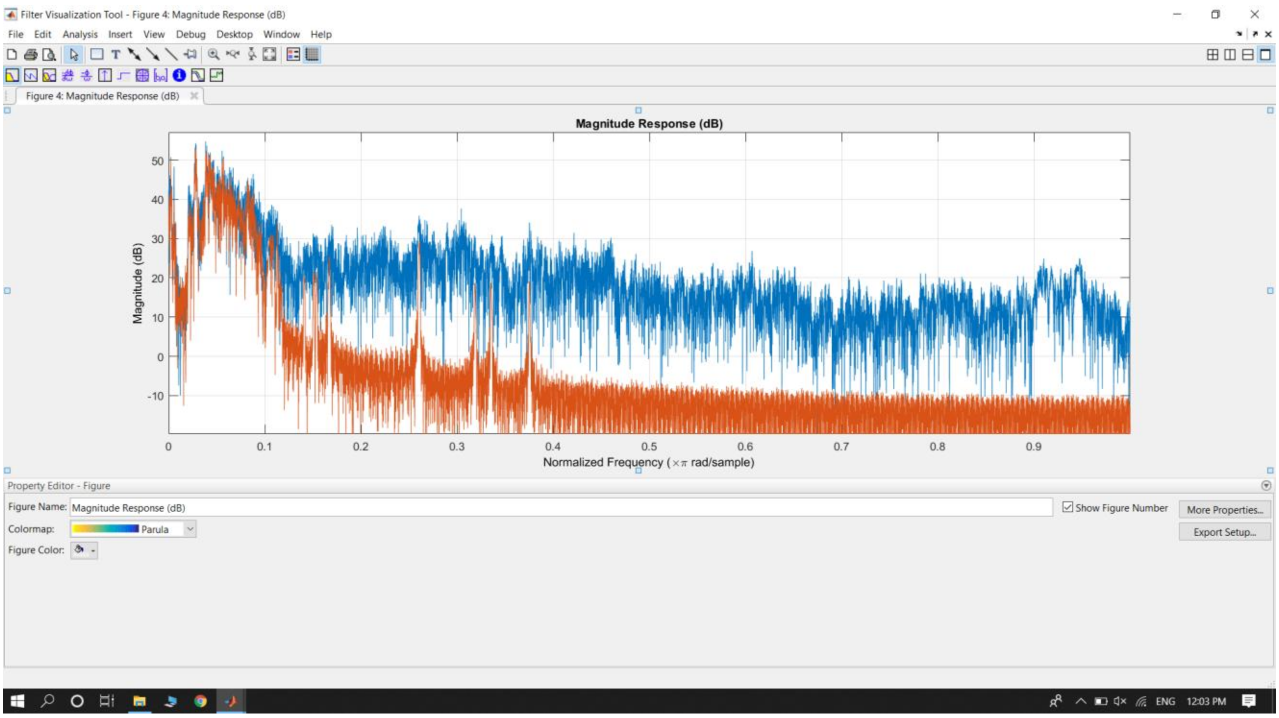

Figure 4: Magnitude Response

From the fig 4 We are not able to differentiate between the highpass filtered signal and lowpass filtered signal as it is the plot of signals in time domain. Only change seen that the noise in both the filtered output get reduced. Subplots of signals in time domain are as,

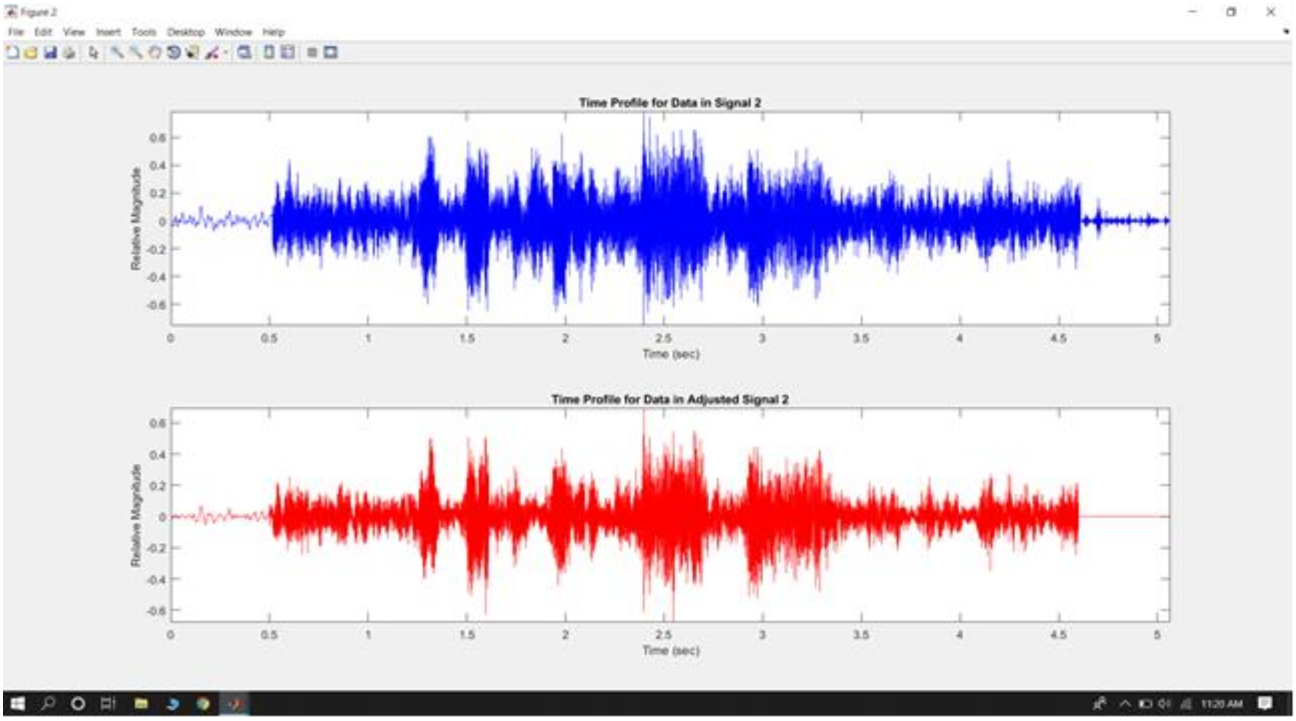

Figure 5: Original input and processed output signal

Above Fig. 5 shows two signals in time domain. The signal in blue colour is the audio file that we have taken for example. The signal in red colour represents time domain representation of processed signal. We cannot differ too much of change in these two signals but can only see the reduced noise level in red signal..

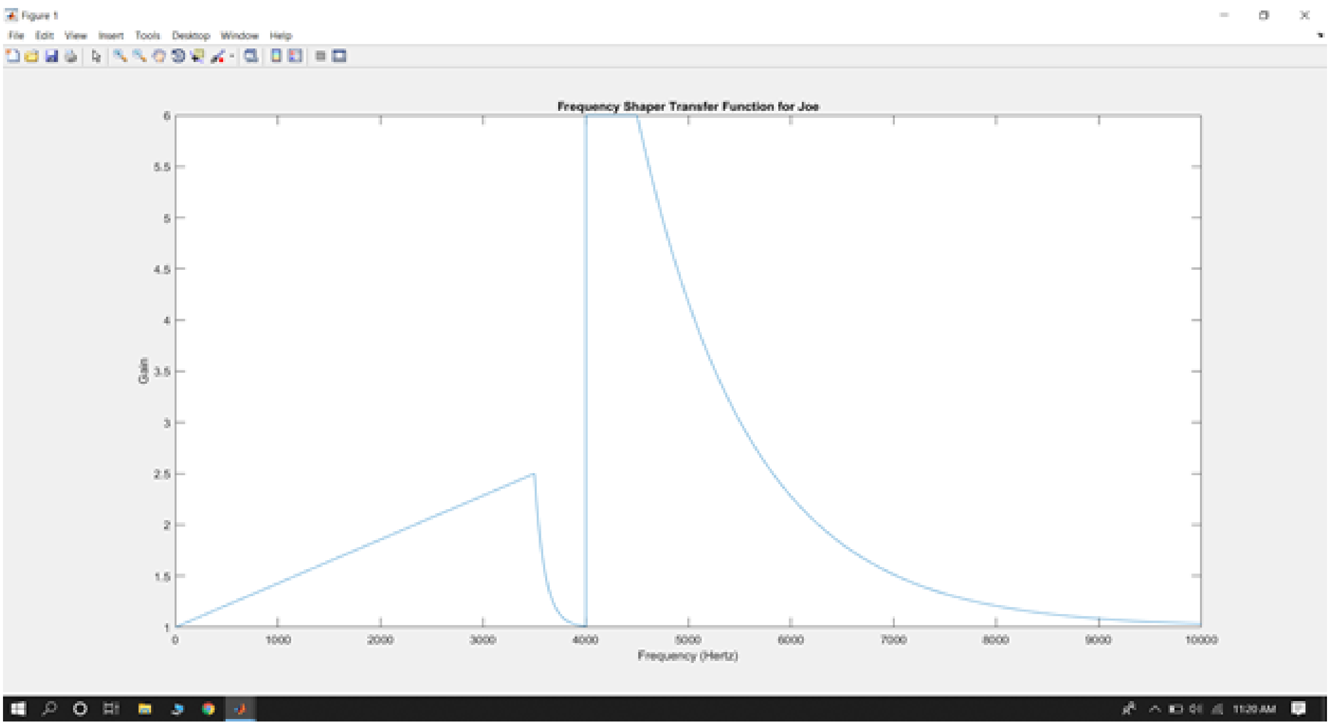

Figure 6: Transfer function

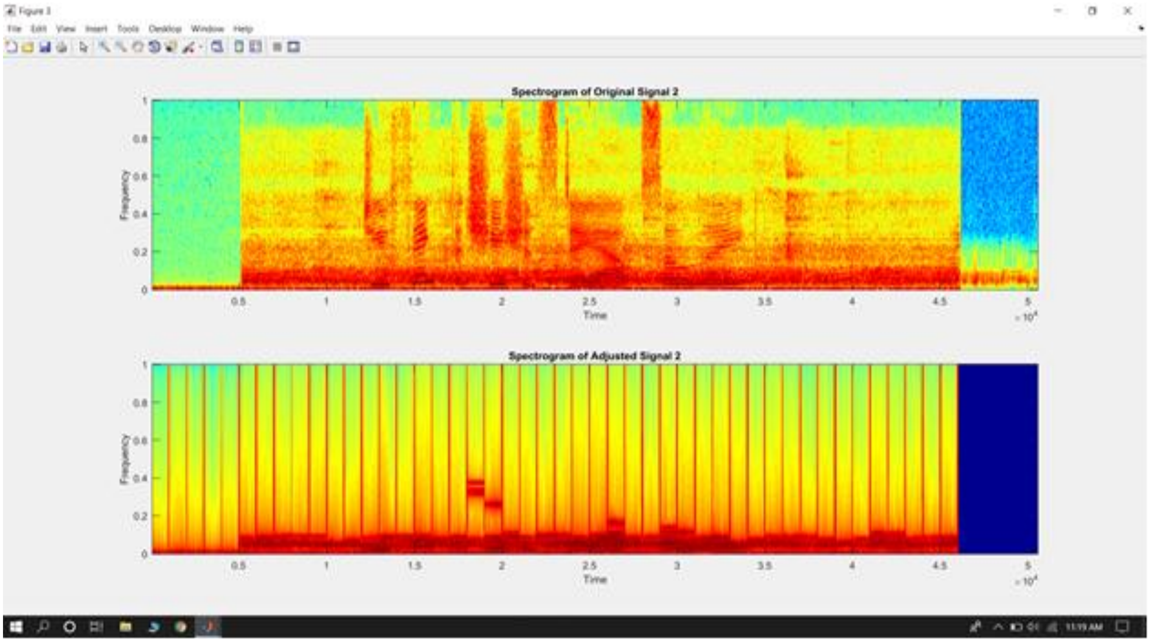

Figure 7 : Spectrograms of original & processed signal

V. Advantages & Applications

Advantages

1. This adjustable filter bank is advantageous as it gives better flexibility than fixed filter bank due to its ability to use different transfer function.
2. Proposed filter bank is applicable for processing of non-stationary speech signals.
3. The design is able to distribute subbands flexibly based on user requirements.
4. Advantageous as proposed filter bank design have very low complexity.

Applications

1. Adjustable filter bank design is used for hearing aids solutions. Rconfigurable filter banks are used in hearing impairments.
2. Proposed design is applicable to health and hygiene domain.
3. Filter banks are used for analysis of non-stationary signals such as speech signal.
4. This design can also be used in audio communication purpose.
5. In audio signal processing, noise reduction and signal reconstruction can be done with the help of proposed adaptive filter bank.

VI CONCLUSION

The adjustable filter bank is be designed for 5 to 8 subbands. Based on the transfer function it is enable to distribute sub bands flexibly. The user have flexibility to choose transfer function as per requirement. Thus the proposed adjustable filter bank with non-uniform frequency distribution is provide more flexibility as compared to fixed filter bank and can be good choice for variety of application in speech domain.

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