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Noise Removal by Spectral Subtraction & Subband Amplification for Hearing Aid- By Using Matlab & Simulink

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Abstract: This article deals with the removal of noise signal in the audio signal with the spectral subtraction of the noise and sub-band amplification for the improving the intelligibility of speech in signal for the hearing aid so that the person experiences the good quality of the speech signal over the analog hearing aid in the market
Keywords: Noise; Spectral; Inteligibility; Hearing Aid;Model Based Design;Sub-band Amplifaction.

I. INTRODUCTION

With the improvement of the technology it is possible to put more and more algorithm and functionality on the small chip which is energy efficient and also fast enough to compute the real time operation with negligible latency. Hearing aid is used for the persons with the hearing impairment to improve the signal quality and amplification of it so that the person can hear in this paper the noise in the speech(audio) signal can be removed with the short-time FFT and the gain is estimated for the sub-band audio spectrum as the cochlea of the human have characteristics that it response the audio signal with non-linear spectrum so that the gain is achieved for those mid frequency and every band is separately amplified or attenuated depending on the user audiogram given by the Doctor[1]-[2], this gain can be change with the standard deviation as per required and ten band equalizer is used for changing the gain.

II. DESIGN CONCIDERATION

Noises are signal which is different from that of the signals which are expected as we dealing with the hearing aid so the signal of interest is the speech signal we want to listen through it so our final goal is that the signal which has the different characteristics other than that of the speech is noise for us and the characteristics that make any signal speech is the vibration of the vocal cord (internal organ of the mouth responsible for the speech production) vibrates with 10 to 25ms for the one word so the system with the windowing is having the 10 to 25ms for no loss of the information or intelligibility[2]. Noise is other signal than these characteristics i.e. the signal which changes fast than that is known as noise and those signal which changes slow than this also not intelligible so design is such that it is capable of removing the unwanted signal(noise) The limitation of the human auditory system is that it can hear the sound from the 20Hz to 20kHz, but the good intelligibility of the signal is present from the 250Hz to 10kHz [1] with the more sensible at particular band of frequency so putting for the gain at that frequency for good intelligibility of speech(audio) [3]-[4] audio band mid frequency are

Octave Band Centre Frequency(Hz)	One Third Octave Band Center Frequency(Hz)
31.5	25, 31.5, 40
63	50, 63, 80
125	100, 125, 160
250	200, 250, 315
500	400, 500, 630
1000	800, 1000, 1250
2000	1600, 2000, 2500
4000	3150, 4000, 5000
8000	6300, 8000, 10000
16000	12500, 16000, 20000

Table- Octave Band Centre Frequency & One third Octave Band Centre Frequency for Hearing.

The above table contain the octave band center frequency with one third octave band frequency, for intelligibility of speech the frequency is from 200Hz to 10kHz so band gain is at 200Hz to 8kHz octave band center frequency with lower frequency at 200Hz and higher frequency at the 10kHz this band contain the 95% of the human speech[5] , higher than this frequency is useful for music and melody of the sound but our aim is of the intelligibility of the speech so that the communication between hearing impaired[6] and normal person can communicate without the loss of the intelligibility of the speech

III. MODEL FOR NOISE CANCALATION PROPOSE

Noise from the audio signal is removed with the spectral noise cancelation the proposed model which remove the noise from the audio spectrum with the help of the estimation of the gain for noise suppression by using the parameters like noise standard deviation decimation and weighting alpha.

Following is the basic model for the noise cancelation & Sub band Amplification.

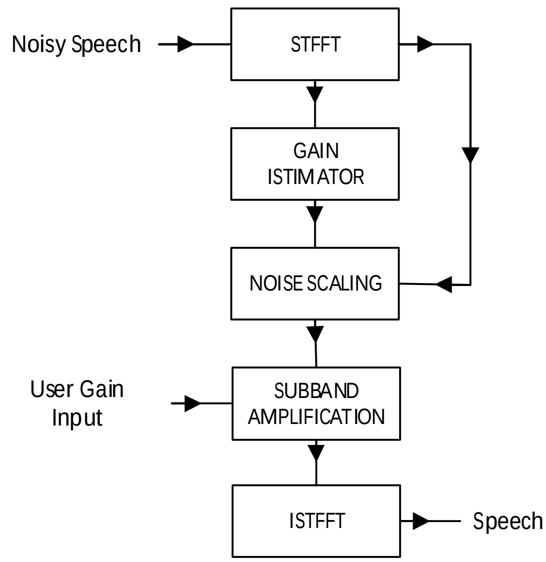

Fig. Model for Noise Cancelation & Sub-band Amplification.

The windowing function for the STFFT is hamming window for calculation which is best for the audio / speech processing

IV. SIMULATION RESULT

Following graphs contains the input noise signal which is added to the speech signal for model validation

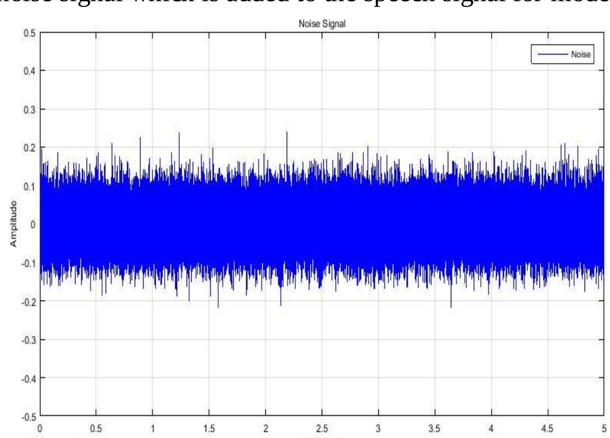

Fig. Noise Signal

Following graphs shows the noisy signal ie. speech signal and noise signal which is applied to the model for the noise removal, this noisy signal decreases the intelligibility of the speech

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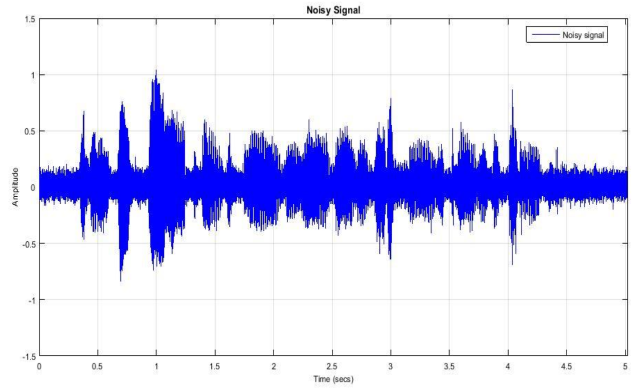

Fig. Noisy Speech Signal

above graph contains the speech signal with the noise in it And its spectrogram which shows the noisy signal with time and frequency relation, in above figure noise is low amplitude as given above

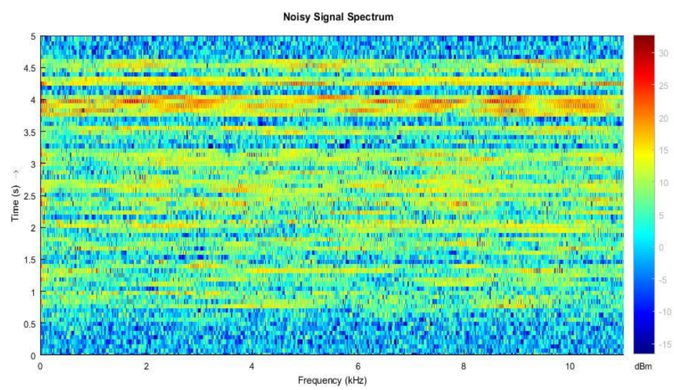

Fig. Noisy Speech Spectrogram

The output of the system after applying the model are show below it is the speech signal without noise

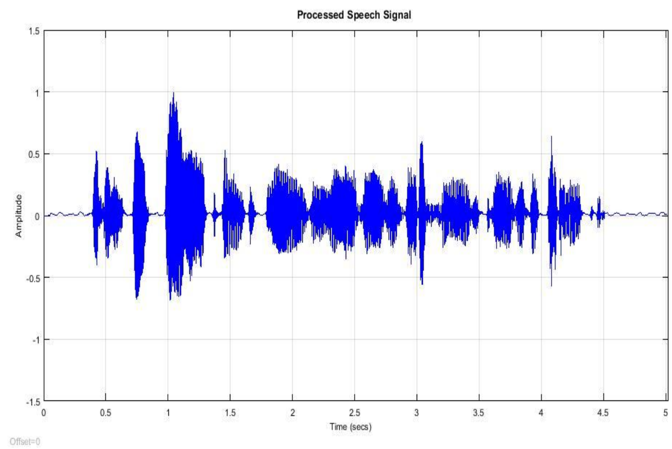

Fig. Processed Speech Signal

Above figure shows the all noise is removed from the speech signal after passing through the proposed model. Improvement of this signal with the hearing imperial person by giving the sub-band gain as per the patient hearing loss at what frequency the patient is uncomfortable to hear by octave filter with center frequency at 31.25 ,62.5 …up to 16kHz.Magnitude response of the Octave filter is as shown

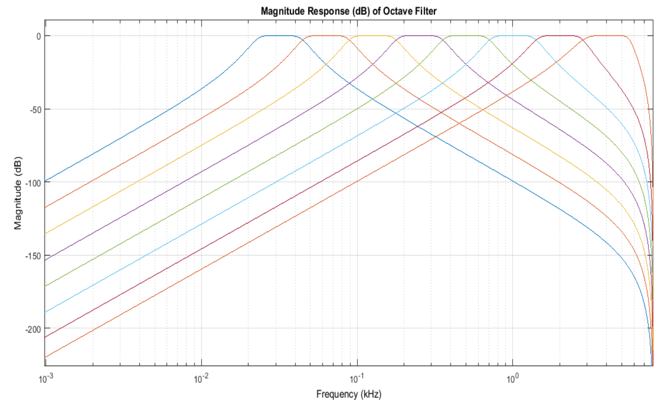

Fig. Magnitude Response of the Octave Filter

Octave filter design by the IIR filter with filter order six impl- imentation cost is less, with stable behavior

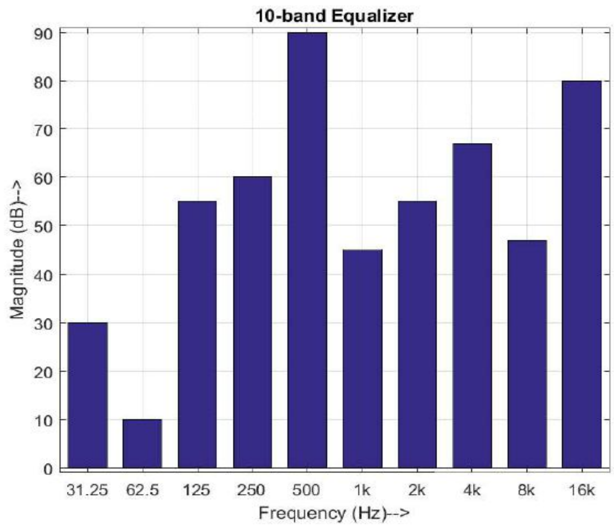

Fig. Octave band filter gain

After applying this gain the output is more intelligible to the person with the particular hearing loss, the time domain graph of the signal with amplitude is as below it can be different with the different patient with different hearing loss

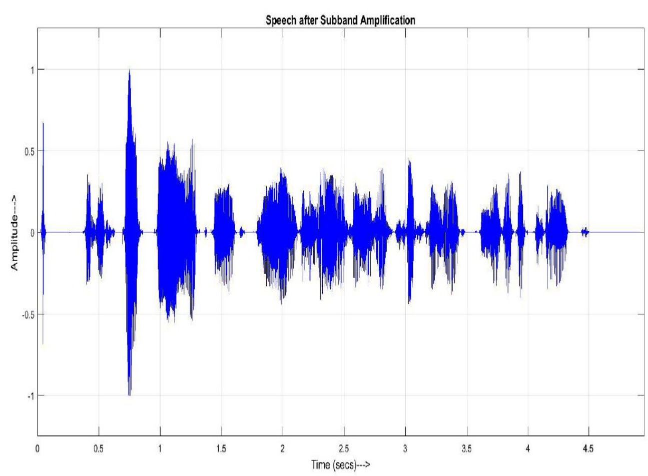

Fig. Processed Speech Signal with user Gain

Addition of the gain is more observable in the frequency domain than the time domain as the amplification of the signal which is done in the frequency domain, which is observable in the spectrogram given below.

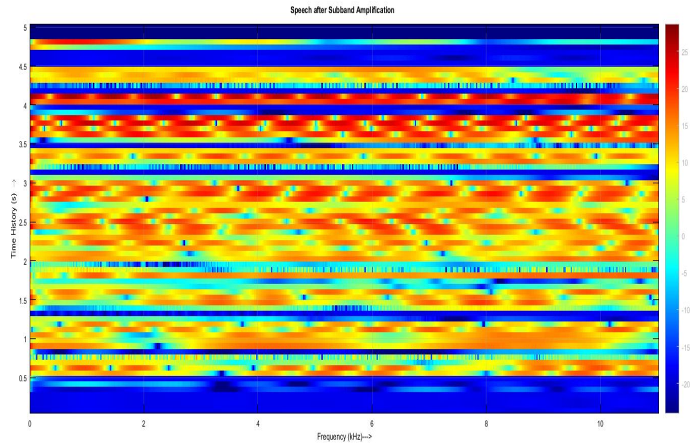

Fig. Speech Spectrogram with removed noise and Sub-Band Amplification.

V. CONCLUSION

It is seen that the noise is removed and the intelligibility of the speech is improved with this proposed model and audio is without Noise as Noise is removed by proposed model it improve hearing of the person with hearing problem.

The implementation cost is reduced by the use of octave band filter design by using IIR, as each octave band Filter required

A. Number of Multiplier :
B. Number of Adders: 9
C. Number of States: 6
D. Multiplication per input Sample:
E. Additions Per Input Sample: 9

for the octave band filter design with Centre frequency of 31.5,63,125,250,500,1000,2000,4000,8000 and 16000 Hz.

there are 9 filter so total implementation cost is given by just multiplying the above with 9.

F. Total Number of Multiplier : 81
G. Total Number of Adders: 81
H. Total Number of States: 54
I. Total Multiplication per input Sample: 81
J. Total Additions Per Input Sample: 81

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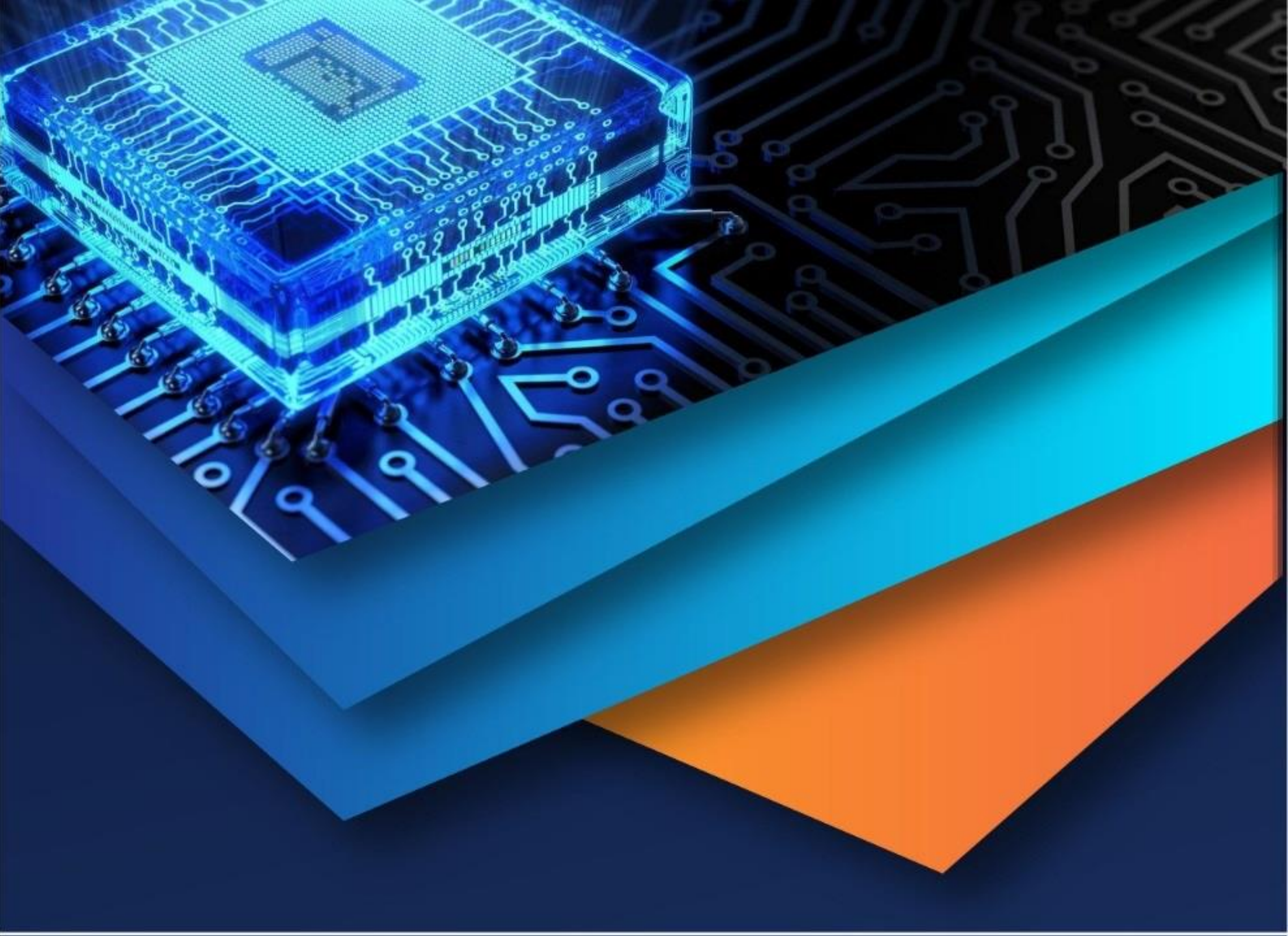

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