MODEL BASED DESIGN FOR NOISE REMOVAL BY SPECTRAL SUBTRACTION 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.

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

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

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

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.

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.
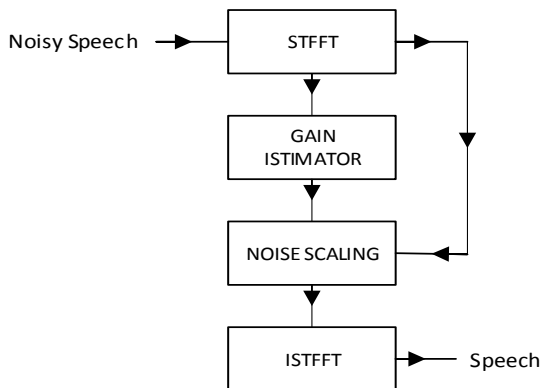
Following is the basic model for the noise cancelation

Fig. Model for Noise Cancelation

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

III. SIMULATION RESULT:

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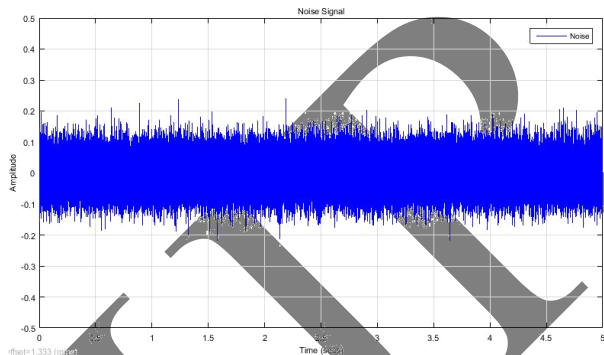

Fig. Noise Signal

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
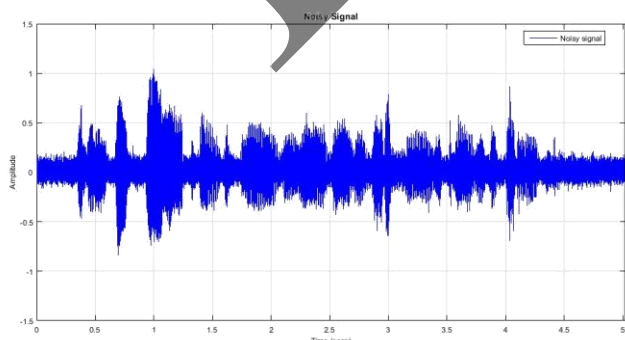

Fig. Noisy Speech Signal

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
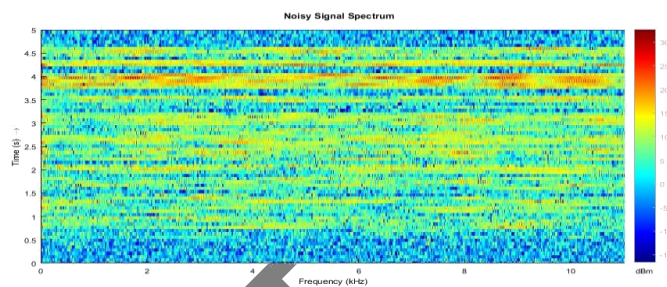

Fig. Noisy Speech Spectrogram

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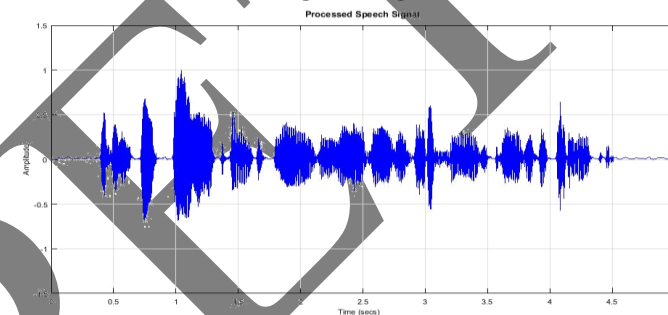

Fig. Processed Speech Signal

Above figure shows the all noise is removed from the speech signal after passing through the proposed model.

IV. 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.

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