International Research Journal of Engineering and Technology (IRJET) e-ISSN: 2395 -0056
Volume: 04 Issue: 3 | Mar -2017 www.irjet.net p-ISSN: 2395-0072

Review Paper on Noise Reduction Using Different Techniques

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Abstract – *Nowadays there are many people affected by hearing impairments. The main complaint of people with hearing loss is low ability to deduce speech in a noisy environment. Hearing aid is a device, which can acquire, process and feedback acoustic signal in real time. In this matter various impedance matching algorithm, various filter bank techniques, signal processing algorithm and echo cancellation are discussed. The purpose of this work is to develop the digital signal processing based platform for digital hearing aid application, which is for the people with hearing impairment using the low cost orange pi model. To Perform this Application FFT algorithm is used which is quite easy to implement and required less computation. The algorithms are performed using python language which gives the best clarity and functionality over MATLAB.*

Key Words: ***Hearing aid, Noise Reduction, SNR, Fast Fourier Transform***

1. INTRODUCTION

The intelligibility of human speech plays an important role in communication. It is both a measure of comfort and comprehension. The quality and intelligibility of the speech are not only determined by physical characteristics of the speech itself but also by communication conditions and information capacity, the ability to get the information from context, mimics and gestures. When discussing intelligibility it is important to understand the difference between a real and recorded speech. During a real conversation a person can recognize the surrounding sounds and concentrate on the speech of another person thus filtering the desired information out of various audio environments. Therefore the ability of a human to recognize and filter sounds significantly increases the intelligibility and comprehension of the speech even if a communication takes place in a noisy environment, situation or condition [2].

Hearing aid is a small electronic instrument which makes sounds louder and makes speech easier to hear and understand, it is designed to pick up sound waves with a tiny microphone, change weaker sounds into louder sounds, and send them to the ear though a tiny speaker, so that it can help patients with hearing loss to perceive sounds again, and then improve their listening level. With the microchips available today, hearing aids have gotten smaller and smaller and have significantly improved quality.

Digital hearing aids have a series of advantages: they can acquire a high signal-to-noise ratio, can change the gain dynamically, adjust resistance to electromagnetic interference adaptively, eliminate feedback, and have been widely concerned worldwide. But in the real environment, a variety of noises are encountered, the performance of voice system under noise environment would drop dramatically or even completely fail, therefore the noise reduction performance of a voice system is critical to evaluate the quality of a hearing aid.

Improving the speech comprehension under the noise environment has been the bottleneck of enhancing the performance of hearing aids. At present, improving methods mainly include two categories: directional microphone and noise reduction algorithm [2]. The former is designed based on the differences of speech and noise in the space, and utilizes directional microphones or beam forming technology to enhance the speech signal characteristic in the specific direction. But the improvement of this method is limited by the number or size of microphones, and does not apply to complete in the canal (CIC) hearing aids. The second way is separating the speech from noise by using differences of time and frequency spectrum between noise and speech. However, the speech and noise may overlap in time and frequency spectrum, so the noise reduction effect remains to be furtherly researched. Noise reduction performance is directly related to whether the wearer can hear really useful speech, even affects the physical and mental health of hearing-impaired patients [1].

Digital hearing aids are committed to minimize the negative impact of the noise, basically have the function of smart noise reduction. However the noise reduction effect of various types of hearing aids differ in thousands ways, which requires the establishment of a complete measurement system to evaluate the noise reduction performance of hearing aids, eventually help hearing-impaired persons to choose suitable hearing aids.

2. LITERATURE SURVEY:

Arun Sebastian, James T.G – has presents a low complex design of a non-uniformly spaced digital finite impulse response (FIR) filter bank for digital hearing aid application. The author described FRM technique in digital filter bank for hearing aid application. Frequency response masking (FRM) technique is used for the implementation of 8 non-uniformly spaced subband filters, with a single half-band filter as a prototype filter. With FRM technique and half-band filter, a drastic reduction in the number of multipliers and adders in linear phase FIR filter can be achieved. Further complexity-effective design can be achieved by producing masking filter from the prototype filter. FRM technique is achieved by cascading different combinations of prototype filter and its interpolated filters to produce subbands. The simulation results shows that, the proposed filter bank gives 120 *dB* attenuation with 13 multipliers only. The FRM technique based filter bank can be used for the audiogram matching. The proposed filter bank design is only applicable for the audiogram with sharp variation of hearing loss at mid frequency regions. But sharp variation of hearing loss occurs at low and high frequency range, the proposed design may not be applicable [5].

P .Rajesh and K. Umamaheswari – have introduce an adaptive filtering technique based on NLMS (Normalized Least Mean Square) Algorithm and RLS (Recursive Least Mean Square) Algorithm for cancelling the noise signal in hearing aids. These algorithms have been implemented using MATLAB. The author studied that the background noise is adversely affecting the speech intelligibility of the people with hearing loss. This method used to cancel the internal noise or error signal in digital hearing aids caused due to acoustic coupling between the microphone and the speaker. The main idea of this method is to replace the receiver input signal with a synthesized signal, which sounds perceptually similar to the original signal. These methods only reduce the internal noise of the hearing aids [3].

Li Zhang, Xiaomei Chen and Bo Zhong – have studied a lot about the effect of hearing loss and about hearing aids. The key technology that influences the effect of hearing is the noise reduction technology. The performance of noise reduction seriously affects the intelligibility of speech, even the physical and mental health of the people who have diminished or defective hearing. In this method first acoustic signal will be acquired though the experiment system which can simulate real working conditions, then signal-to-noise ratio (SNR) and segmental signal-to-noise ratio (segSNR) of signal will be calculated after aligning the output signal and input signal to evaluate the noise reduction performance of hearing aids. The simulation results show that the evaluation method proposed here can evaluate hearing aids automatically and conveniently. But different noise has different characteristic and technology limitation of hearing aids, it can't achieve the best effect to all noises, some results in this experiment may be not very obvious, and the system with only two evaluation indices, the evaluation results may be not very comprehensive and specific; so in the future, we should improve the system performance by adding more indices, evaluate noise reduction performance in other domains so that results are more accurate [1].

M. Poornima and E. Rajinikanth – have discussed about analog model of hearing aids. They simulate filters and reduced white noise and increased the gain of frequencies which were unable to hear and shaped the amplitude to prevent any of the frequencies from becoming loud. By cascading different combinations of prototype filter to produce more no of sub bands and amplitude shaper so that the speech signals can be improved to reduce the noise, where as the original first stage filter-bank system can still be used for compression and amplification. They also consider the algorithmic delay added when implementing filter-bank. This first stage can be applied to lower frequency bands only, so that the harmonics of speech especially for low content can be resolved. The author also represents a systematic description of the cascade filter-bank stage, discusses its influence on the processed signals in detail and further presents the results which indicate the improved performance of signal to noise ratio, computational complexity compared to the original single-stage filter-bank system[4].

3. PROBLEM DEFINATION:

Hearing is one of the five senses along with vision, taste, smell and touch. The ear serves as a receiver of incoming sounds. Hearing loss most commonly occur because of damages of the ear, rather than the central auditory system. The audio frequency range which is capable to hear is generally between- 20Hz to 20kHz.The human ear is only sensible to hear the frequency range between 1kHz to 4kHz. So below 1 kHz, ear will not respond and above the 4 kHz, it may damage the hearing capability.

Hearing loss is usually reserved for people who have relative insensitivity to sound in the speech frequency range. The main complaint of person with hearing loss is low ability to deduce speech in a noisy environment. In hearing aids the sound is processed by hearing aid and reaching to the ear. Normally it is made of three parts; Microphone, Processor units, receiver module. Recently available digital hearing aids are not compatible with environment.

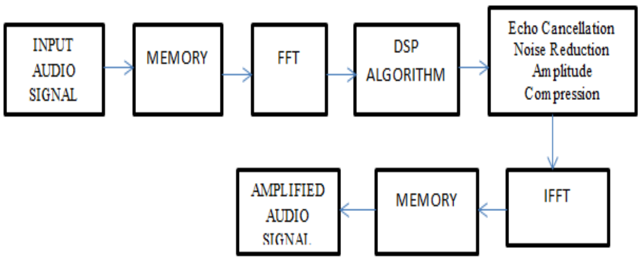
3.1 Objective of work:

To develop an efficient algorithm to reduce the noise which is used as a application in hearing aid.

To Implement a System using a new technique which is better than the Existing Various Techniques study as in literature review.

To develop a system which is compatible with different environmental places like traffic, music show, meeting, class room, theatre etc. And also compare the results of different selective modes.

4. METHODOLOGY:



The basic flow of the implementation is shown in the figure. To design the digital hearing aid, the Orange pi based module is used that is low cost comparative to DSP kit. For that purpose the FFT algorithm is used to convert the signal in to the frequency domain from the time domain and also used to split the frequency band in to various bands. Then noise reduction algorithm is discuss for different modes of the environment such as, for traffic, music, noise in dialog speech etc. Also perform the echo cancellation algorithm using NLMS adaptive filter. Finally frequency shaping and amplitude compression function are performed to smooth the signal.

5. CONCLUSION

The developed DSP platform for digital hearing aid using Orange pi is worked in to different atmosphere by manually selecting the mode of nearby atmosphere. The outputs of this system shows that the original speech signal that is somehow corrupted with environmental noise is get back through the digital hearing system with suppressed noise signal and with compressed amplitude. The algorithms are performed using Python language which gives better clarity and functionality. The system also shows that the used of Orange pi to performs the algorithms for digital hearing aid is better compare to DSP kit because the Orange pi run at high frequency so the speed of convergence is high and also the price too much low compare to DSP kit.

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