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Review Paper on Innovation of Advanced Noise Reduction Technique for Better Audibility Process

Dr. Anita Pati[1], Dr. Rohit Kumar[2], Ms. Mani Dublish[3], Ms Shweta Singh[4]
[1, 2, 3, 4]Asst. Prof IMS Noida

Abstract: Now a days there are many people affected by hearing loss that make them disabled as they cannot communicate properly .The main complaint of people with hearing loss is low ability to deduce speech in a noisy environment. Hearing aid is a delicate instrument, which can acquire, process and feedback realistic signal in real time. In this matter various apparent opposition matching algorithm, various filtering methods, digital signal processing algorithm and echo cancellation are developed and implemented. The purpose of this object is to develop the digital signal processing based platform for digital hearing aid technique, which is for the people with hearing impairment using the low cost fuzzy orange pi model. To Perform this Application fuzzy algorithm is used which is quite easy to implement and required less operative computation. The algorithms are performed using MATLAB language which gives the best clarity and simulated functionality over MATLAB.
Keywords: Speech Recognition, Noise Reduction, SNR, Fuzzy Masking Technique

I. INTRODUCTION

The intelligibility of human speech plays an important role in communication and transmitting ideas. 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, immitations 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 identify 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 . Hearing aid is a small electronic device makes sounds louder and makes speech easier to hear and understand, it is designed to pick up sound waves with a tiny and soft 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 revive properly.

To recover 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 or machines 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 drastically 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 . 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.

II. LITERATURE SURVEY

Arun Sebastian, James T.G – has presented a low complex design of a non-uniformly spaced digital finite impulse response (FIR) filtering technique for digital hearing aid application. The author described FMT technique in digital filter bank for hearing aid application. Fuzzy masking (FMT) technique is used for the implementation of 8 non-uniformly spaced subband filters, with a single halfband filter as a prototype filter. With FMT 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. FMT 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 140 dB attenuation with 15 multipliers only. The FMT technique based filter bank can be used for the audiogram matching. The proposed filter bank design is only applicable for the audiogram with sharp ,moderate and mild 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 adoptable in many cases.

P .Rajesh and K. Umamaheswari – have introduced an adaptive filtering technique based on NLMS (Normalized Least Mean Square) Algorithm and RLS (Recursive Least Mean Square) Algorithm for eliminating or capability the noise signal in hearing aids devices. The author have also applied double derivative technique and fuzzy logic filters on 2 dimensional noise while working on 2 dimensional image processing. Later author also applied advanced fuzzy logic filters on 2 dimesional pixels to recover originality of the images taken by hand at initial level.These algorithms have been implemented using MATLAB(ver 7.0). The author studied that the background noise is adversely affecting the speech capability of the people with hearing loss or disability..This method used to cancel the internal noise or error signal in digital hearing aids caused due to acoustic probability 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 as when it is unplugged from its battery till then it creates unnecessary sound which is a signal to user for its alertness.

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 probable signal will be acquired though the experiment system which can simulate real practical conditions, then signal-to-noise ratio (SNR) and segmental signal-to-noise ratio (SNR) of signal will be calculated after aligning the output signal and input signal to evaluate the noise degradation 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 which will enhance the hearing ability to process faster .

M. Poornima and E. Rajinikanth – have discussed about analog simulated model of hearing aids. They simulate filters and reduced salt and pepper noise and increased the gain of frequencies which were unable to hear and shaped the amplitude to prevent any type of the frequencies from becoming loud. By masking different combinations of prototype filter to produce more no of sub bands and amplitude efficiency shaper so that the speech signals can be improved to reduce the noise, which will become effective as well as meaningful for the user. where as the original initial stage filter-bank system can still be used for compression and amplification of signals. 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 harmonic mean of speech signal values especially for low content can be resolved. The author also represents a systematic description of the cascade filter-bank stage, which explains its influence on the processed signals in detail and further presents the results which indicate the improved performance of signal to noise ratio(SNR), computational complexity compared to the original single-stage filter-bank system.

A. Problem Definition

Hearing is one of the important sense organ among lively bodies among all 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 1.5kHz to 4kHz. So below 1 kHz, ear will not respond and above the 4 kHz, it may damage the hearing capability which lead to use hearing aid machines. Hearing loss is usually reserved for people who have relative insensitivity to sound in the speech frequency range.

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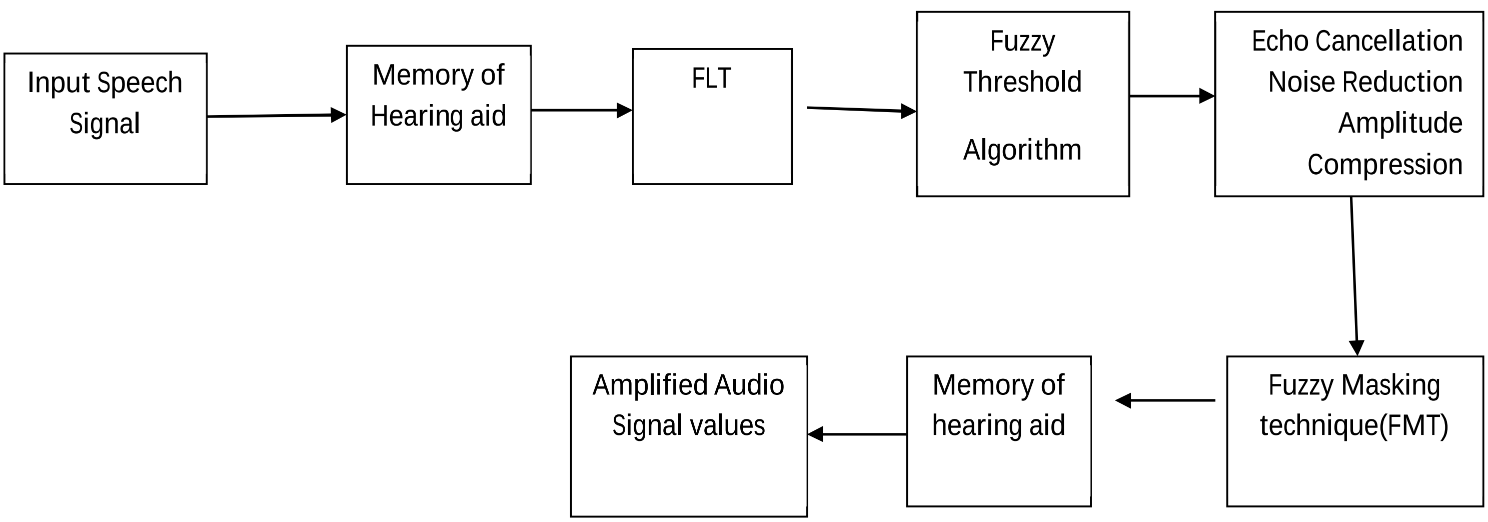
The main complaint of person with hearing loss is low ability to deduce speech in a noisy environment,so as a result people cannot express properly. In hearing aids the sound is processed by hearing aid and reaching to the ear. Normally it is made of three parts; Microphone, Processor units, receiver module. Recently available digital hearing aids like starkey, Ear solutions are not much compatible with environment especially with kids but Indian companies like Alps , Hear clear companies have adopted proper mechanisms and became popular gradually.

B. *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.

III. 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 as it is low cost comparative to DSP kit. For that purpose the FMT 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 multiple bands. To catch the proper signals. Then noise reduction algorithm is discuss for different modes of the environment such as, for traffic, music, noise in dialog speech, noise in wind or air, clouds etc. The signals also perform the echo or hazyness in voice cancellation algorithm using FMT adaptive filter. Finally frequency shaping and amplitude compression using fuzzy function are performed to smooth the signal.



IV. CONCLUSION

The developed DSP platform for digital hearing aid using Orange pi is worked in to different domain by manually selecting the mode of nearby environment. 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 values. The algorithms are performed using Matlab 7.0 which gives better audibility and functionality which leads better clear to the disabled in listening power and that gives rise to increase in audibility power in sensory organs. The system also shows that the used of FMT to performs the algorithms for digital hearing aid is better compare to DSP kit because the FMT is executed at high frequency so the speed of convergence is high and also the prices are too much less compare to DSP kit.

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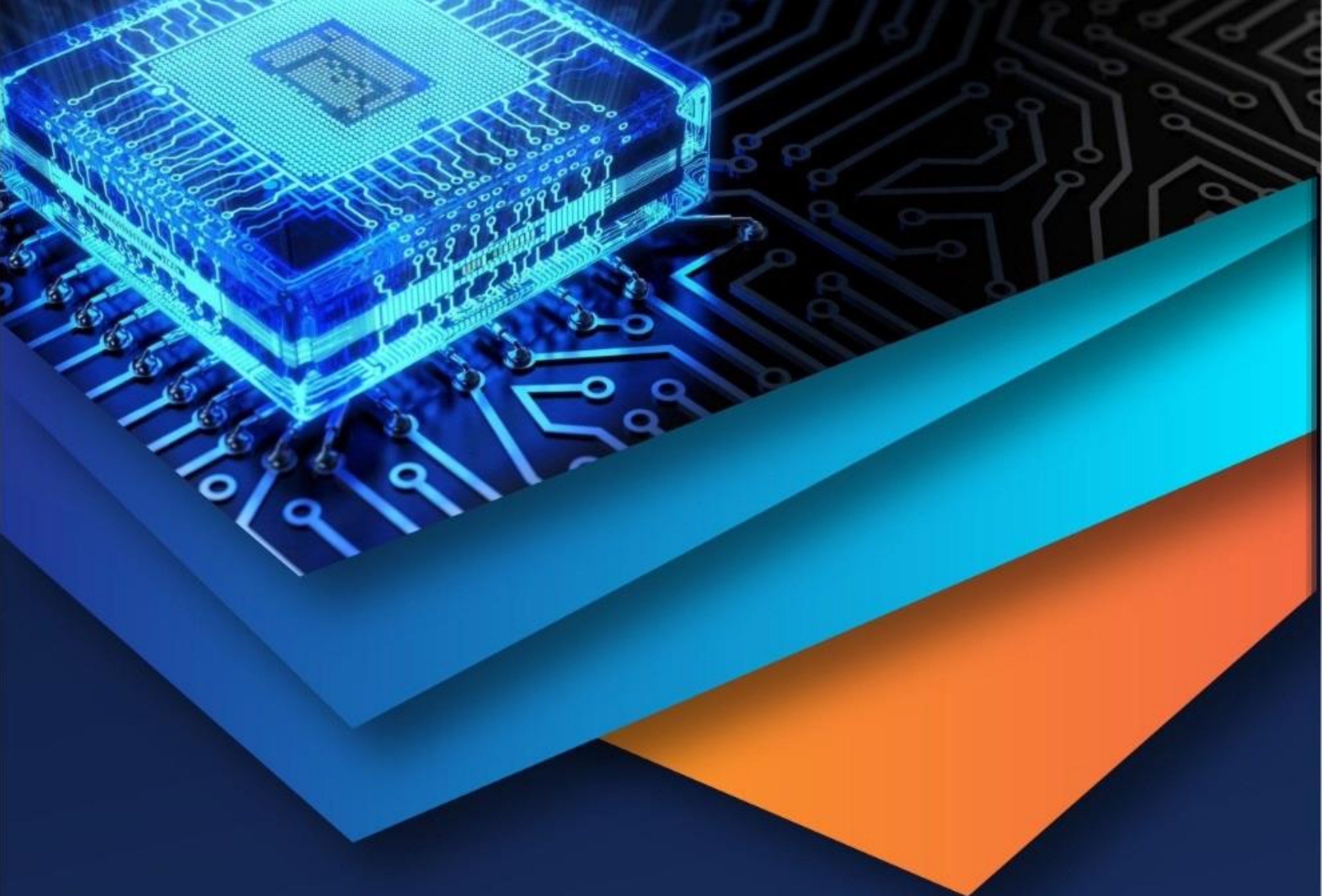

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