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LOW COST HARDWARE IMPLEMENTATION FOR DIGITAL HEARING AID USING RasPi

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Abstract

The main complaint of person with hearing loss is low ability to deduce speech in a noisy environment. So, by using the Digital Signal Processing (DSP) enhance the possibility of performing Signal-to Noise ratio. In this various impedance matching algorithms, signal processing algorithms, and echo cancellation are discussed. The purpose of this paper is to develop the digital signal processing based platform for digital hearing aid using the low cost Raspberry Pi model. The algorithms are performed using the python language which gives the best clarity and functionality over the MATLAB. The result of this work gives the faster speed of convergence and best noise suppression and compressed amplitude to the power of the signal.

I. INTRODUCTION

Hearing aids are gadgets utilized by hearing hindered persons to balance the hearing loss. They can't totally beat the perceptual bends brought on by a hearing loss however support the user to decode speech.The sound is processed by the hearing aid and reaching to the ear. Essentially it is made of three parts, Microphone, Processor unit, receiver module. Electrical Impulse is converted by use of Microphone. Not it is converted in suitable form from electrical impulse by Processor unit; receiver converts the impulses in sound using a decoder. The audio frequency range is generally between 20 Hz to 20 kHz which capable to hear. The human ear is only sensible to hear the frequency range between 1 kHz to 4 kHz[1]. So below 1 kHz, ear will not respond and above the 4 kHz, it may damage the hearing capability.

System Overview

To design the digital hearing aid, the Raspberry pi module is usedthat is low cost comparative to DSP kit. This paper describe the FFT algorithm to convert the signal in to the frequency domain from the time domain and also used tosplit 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. The basic flow of software implementation is shown in to the fig 1.

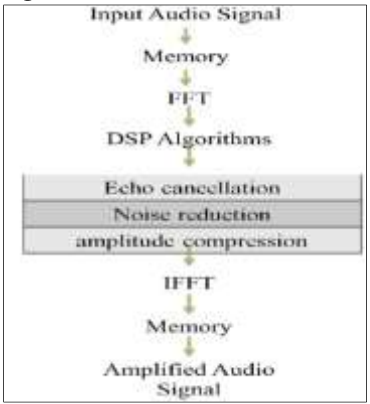

Fig 1 software implementation flow

II. MATERIALS AND METHODS

Fast Fourier Transform

For easier to implement and fewer computation FFT is the choice instead of the DCT.The signal coming from the microphone is first digitalized and fed to the FIFO memory.To synchronization of the data transfer is done by the First Input First Output (FIFO). The received samples are then divided in to the 32 samples block. The FFT calculation is then apply to the 32 sample block which transform information in to the frequency domain for the time domain.Then various signal processing algorithm are used to improve and enhance the hearing quality, which includes the noise reduction, echo cancellation, adaptive noise cancellation etc. After the noise reduction and smoothing the signal, IFFT is used to get back the signal in to the time domain. The flow chart of the FFT to IFFT is shown in to the fig 2 [2].

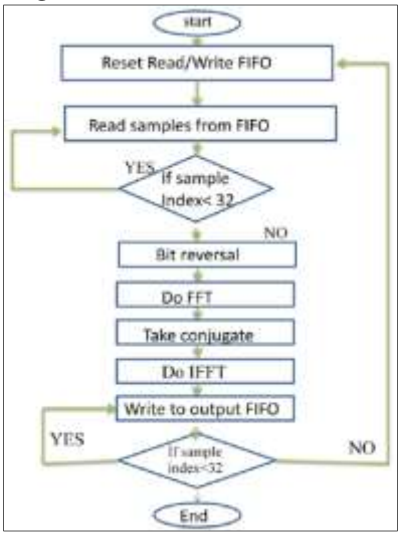

Fig 2 Flow chart of the FFT

Noise Reduction Algorithm

In this system various modes are selected according to the environmental changes and through the selected mode the noise reduction algorithm is applied. If the Frequency of the noise is very high then select the high noise factor. The basic flow of the noise reduction algorithm is shown in to the fig 3.

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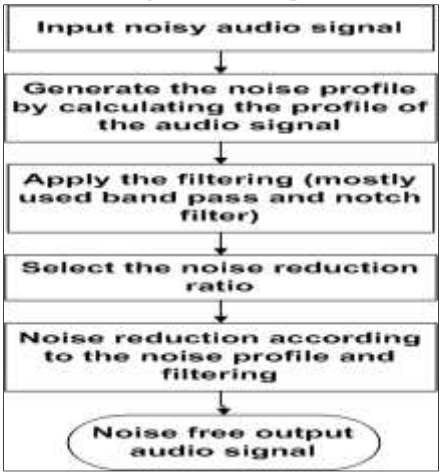

Fig 3 basic noise reduction algorithm

Echo Cancellation

Echo suppression is the techniques to enhance speech quality through creating or removing the echo if it is already present. Now to enhance subjective quality these procedure expands the limit accomplish through quite suppression, by preventing the echo from traversing a system.The goal of the echo suppression is to preventing a person's speech from listening to echo.This system developed to suppress the acoustic echo using normalized least mean square (NLMS) algorithm into the white noise background [3].

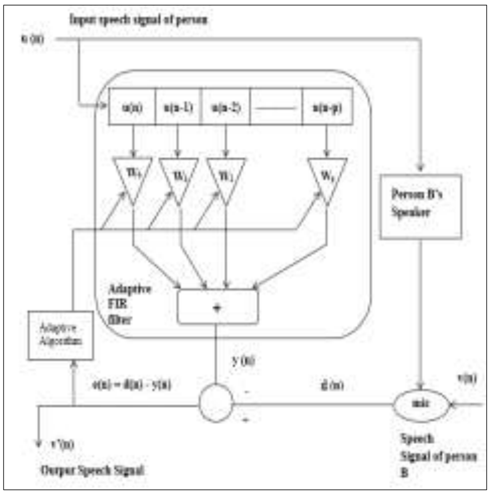

Fig 4 system overview of an adaptive echo suppression

Fig 4 illustrate the example of echo suppression system of two person's conversation on internet talking. In this example the person A's speech signal is creates his own echo when person B's speech signal coming to his so the original voice is corrupted. The speech signal from person A to person B is the input signal for this system. Firstly the system detect the signal from B to A and make an effort to model and synthesis an imitation of person A's echo. This imitation is used to subtract and counterbalance the echo of person A from the speech signal of person B. In this system

$u(n)$ - speech signal of person A

$u'(n)$ - echo signal of person A

$v(n)$ - speech signal of person B

$d(n)$ - speech signal of person B with acoustic feedback signal , $v(n) + u'(n)$

To synthesis the echo used FIR adaptive filter over IIR filter because IIR filters have stable operation and it's practically difficult with adaptation. The estimated output of the filter for the echo signal is calculated by the following equation

$$y(n) = \sum_{k=0}^{p-1} w_k(n)\, u(n-k)$$

Where, w_k - weight varying co-efficient of an FIR filter, and $y(n)$ – Estimated echo of A from B to A. After the suppression of echo the generated error signal is

$$e(n) = d(n) - y(n)$$

$$e(n) = v(n) + u'(n) - \sum_{k=0}^{p-1} w_k(n)\, u(n-k)$$

This error signal is minimized by adapting the co-efficient of echo suppressor using the NLMS algorithm. The weight of the filter is updated until the error signal is goes near to the zero or when the $v(n)$ is nearly equal to the $u(n)$.

The weight is updated using the following equation.

$$w(n) = w(n-1) + \frac{\mu\, e(n)}{\|u(n)\|^{2}}\, u(n)$$

Where, $w(n) = [u(n), \ldots, u(n-1)]$ - vector of input signal

$w(n) = [w_0(n), \ldots, w_{p-1}(n)]$ - Vector of co-efficient

$\|u(n)\|^{2}$ – Normalized input signal

μ - step size of adaptation used for stability, steady state error and to control the convergence speed.

Amplitude Compression

After the noise suppression the system also needs to compress the amplitude in real time. The main function of amplitude compression is to control the overall gain of speech amplification signal. Amplitude compression is accomplished through apply the gain that is less than the previously determined threshold of exceeded power. Amplitude compression is taking into the account of normal power within the signal. The constant time of power estimation is utilized to adjust the attack or discharge time of the compression algorithm [5].

Suppose s(n) is the input signal and its evaluated power is p(n) given by the following equation,

$$p(n) = \beta\, p(n-1) + (1-\beta)\, s^2(n) \quad \ldots\ldots (a)$$

If gain is applied to the s(n) then output signal is

$$d(n) = g\, s(n)$$

The relationship between the input output powers is

$$P_{out} = g^2 P_{in}$$

$$P_{out}dB = 20log_{10}(g) + P_{in}dB$$

$$P_{out}dB = gdB + P_{in}dB$$

In real time, input power P(n) is not available so, equation (a) is used.

III. EXPERIMENTAL SETUP

System Overview of Digital Hearing Aid

Fig 5 shows the basic block diagram of the digital hearing aid. Raspberry pi board is used to provide the platform of Digital Signal Processing. The test set up is also shown into the fig 6.

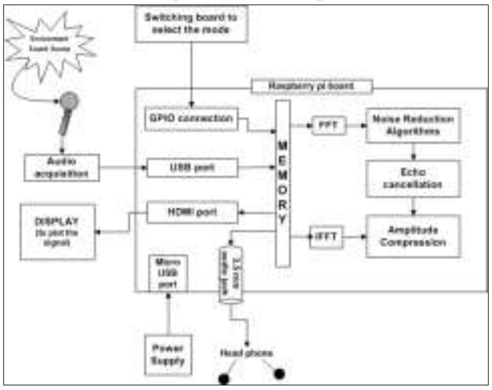

Fig 5 systematic diagram of digital hearing aid

Fig 6 Test setup of the system

Description

In this system the environmental sound signal is pick up through the web came that is used as mic to connect with the USB connection port of the Rasp PI.Switching board is used to select the different modes of environment such as music, traffic, and normal conversation. Display is used to show the graph of the noisy input signal and the noise free output signal.Raspberry pi is connected to the PC display using the HDMI port. The small sized PI board provide the PC based of CPU to the display. All the algorithms are developed in the python language. The flow of the hardware set up to the display is shown in to the fig 7. And finally to listen the noise free smoothing signal, headphone is used that is connected to the pi board using 3.5mm audio out jack port.

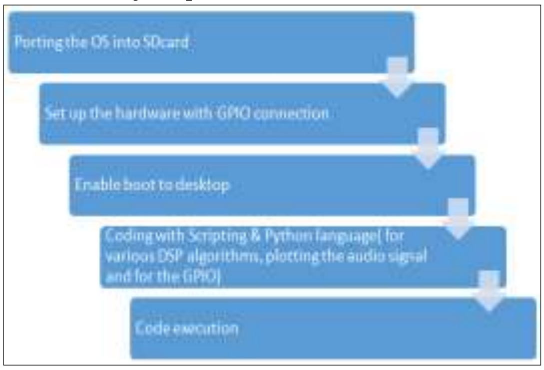

Fig 7 flow of the hardware set up

IV. RESULTS AND DISCUSSION

The results of algorithms for different selective modes such as noise reduction in different-different type environmental places like music show, class room, dialog speech conversation in noisy atmosphere are shows in below figures. There is also shows the result of echo cancelation. All the results and output audio file shows that the output signal is with compressed amplitude and smoothing signal.

Result of Noise In Music

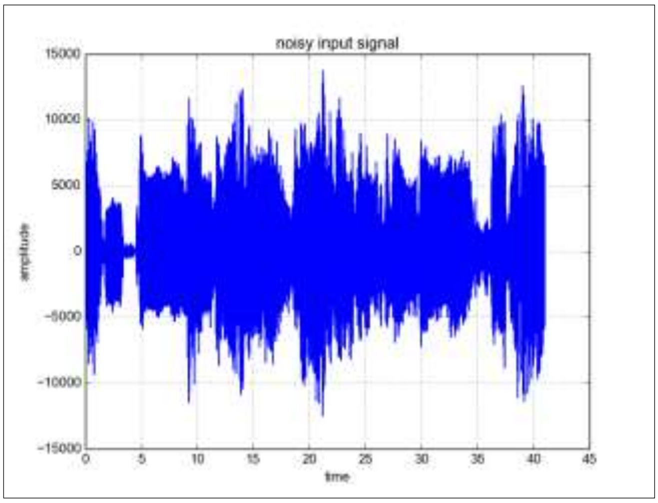

Fig 8(a) input noisy signal of musical mode

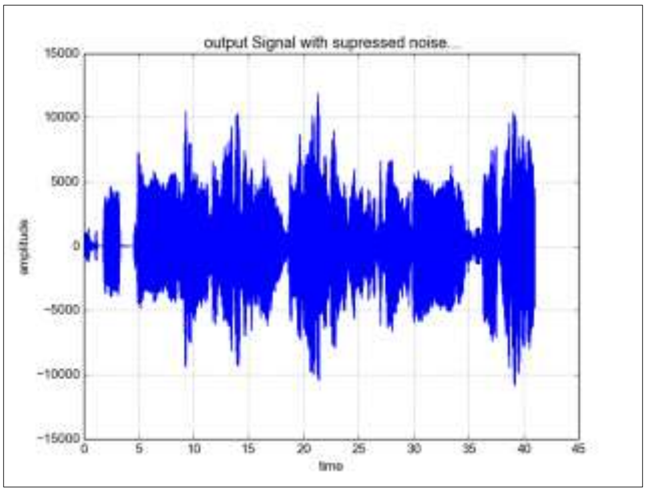

Fig 8(b) noise free output signal of musical mode

Result of Noise In Dialog Speech Signal

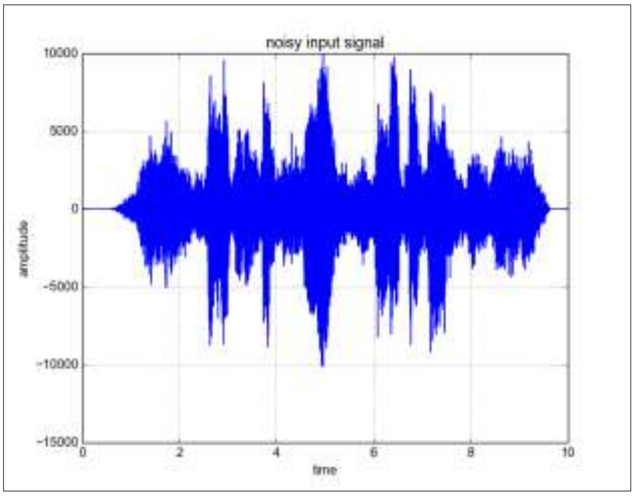

Fig 9(a) input noisy signal of dialog speech

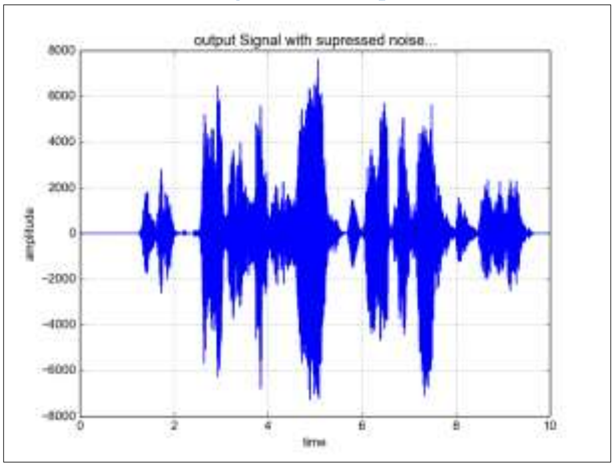

Fig 9(b) noise free output signal of dialog speech

Result of Noise in Traffic

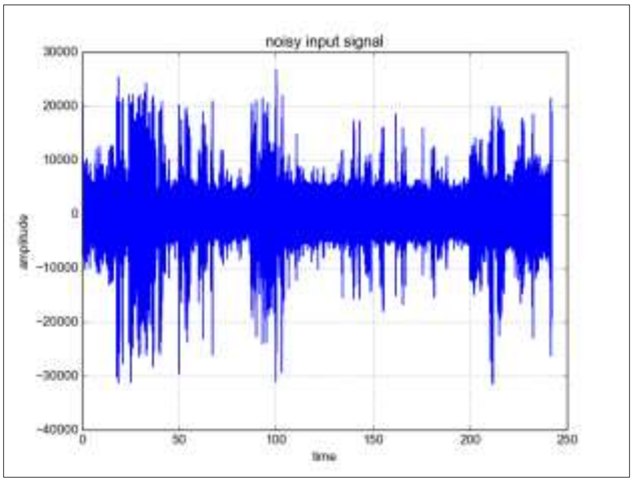

Fig 10(a) input noisy signal of traffic

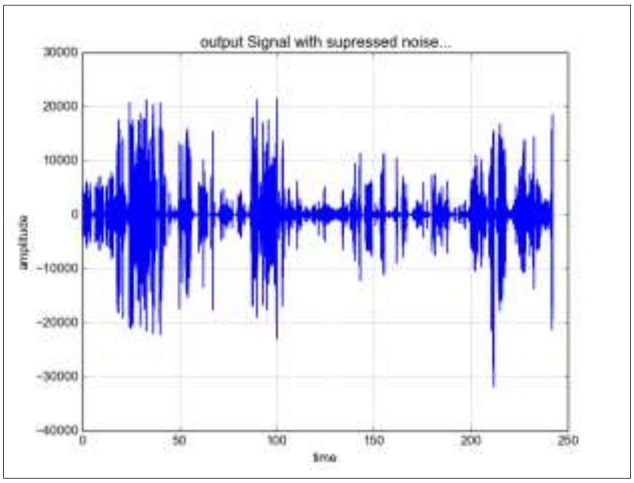

Fig 10(b) noise free output signal of traffic

Result of Echo Cancellation

Fig 11 shows the result of situation that is discussed in to the topic 1.4. The fig 11(d) shows the comparison of the original and evaluated coefficients. That proves that the error signal is minimized.

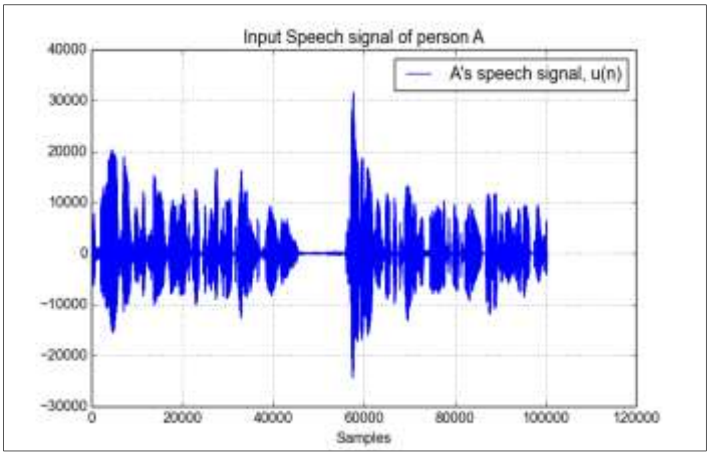

Fig 11(a) input speech signal

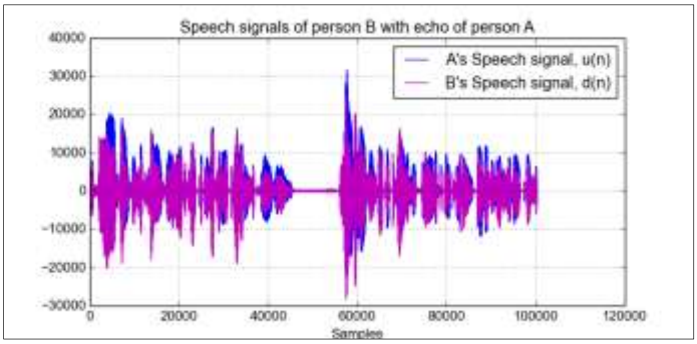

Fig 11(b) Speech signal with echo

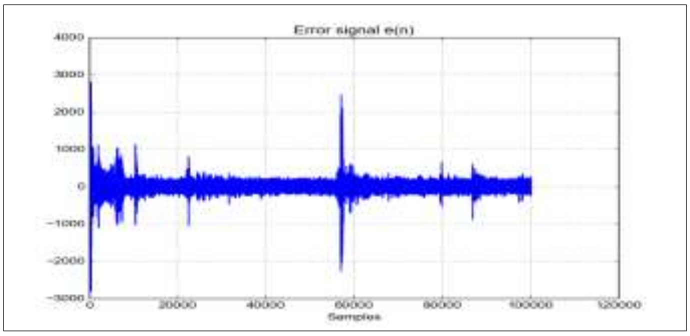

Fig 11(c) error signal

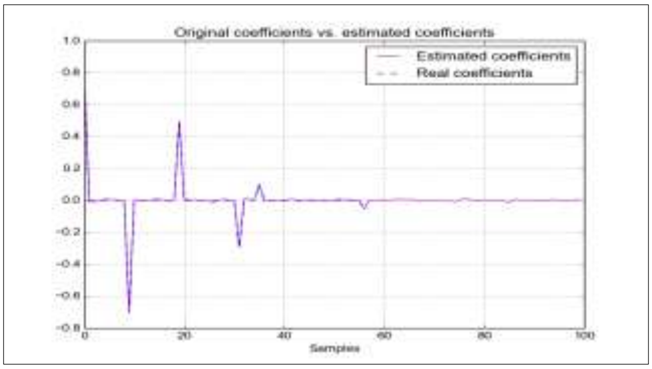

Fig 11(d) coefficient comparison

V. CONCLUSION AND FUTURE WORK

The results here by conclude that the developed DSP platform for digital hearing aid using Raspberry 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 system also shows that the used of raspberry pi to performs the algorithms for digital hearing aid is better compare to DSP kit because the raspberry pi run at 900 MHz frequency so the speed of convergence is high and also the price too much low compare to DSP kit.

The problem with this system is that the patient has to select mode manually, there is no automatic classification of signals. So in future, design the artificial neural network in the python for the automatic classification of the speech signal when patient go from one atmosphere to the other atmosphere.

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