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Adaptive Speech Enhancement using DSP Processor

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Abstract

Speech and text is the main medium for human communication .Digital signal processing plays a central role in the development of modern communication and information processing systems. In audio processing background noise is removed and required noise is enhanced using by adaptive filter and LMS algorithm. The filtered signal is compared to the original noise-free speech signal in order to highlight the level of attenuation of the noise signal. The result shows that the noise signal is successfully canceled by the developed adaptive filter. The difference of the noise-free speech signal and filtered signal are calculated and the outcome implies that the filtered signal is approaching the noise-free speech signal upon the adaptive filtering. The LMS adaptive filter algorithm shows significant noise cancellation at lower frequency range.

Keywords:-*TMS320C6713 DSP Processor, LMS algorithm, Filter.*

I. INTRODUCTION

In this project we aim at presenting an algorithm, which enhances speech by attenuating any kind of noise. To achieve this we design an adaptive filter, which would adapt itself depending on the nature of noise. we compared with LMS Algorithm and proposed blind technique, since we do not know the parameters of the noise signal. The primary differentiation that we make between speech and noise is that speech is highly non-stationary in a time interval of 250ms whereas noise stationary. This characteristic is used to derive a cost/ functional using which we may achieve speech enhancement. Adaptation takes place by changing two sets of weights. Apart from hearing aids this algorithm also finds application in mobile phones (Hands free kit). The above algorithms are simulated with the help of mat lab program and to implement using TMS320C6X DSP processor for real time application.

Rather, hearing deficiency can increase sensitivity and reduce tolerance to certain sounds while diminishing sensitivity to others. For instance ,digital technology can tell the difference between the speech and background noise allowing one in while filtering out other .Approximately 10% of the world's population suffers from some type of hearing loss, yet only a small percentage of this statistic use a hearing aid. The stigma associated with the hearing aid, customer dissatisfaction with hearing aid performance, and the cost associated with a high performance solution are all causes of low market penetration. Through the use of digitalsignal processing, digital hearing aid now offers what the analog hearing cannot offer.The primary problem faced during noise reduction pertaining to speech, is that no parameters are known about the characteristics of noise. Previous methods involve the usage of an anti-phase signal to cancel the primary source signal. This technique has been used successfully in many industrial applications to reduce noise levels. However such a technique is useless in the case of speech enhancement. Also usage of filters for noise reduction will also be useless owing to the uncertain nature of noise.

Hence an adaptive approach is an apt solution to this problem. Further a blind technique is well suited in this case wherein no prior assumptions are made regarding the properties of speech and noise. However the spectral properties of human speech and its phoneme structure are used to model the system. The existing technique used to enhance the speech quality is subject to both degradations due to surrounding environment All these tasks must be achieved with a single VLSI chip in order for the system to be both cost-effective, power efficient and widely accepted. Hence the goal of this paper is to prove that this technique can better the existing one, using VLSI technology the same can be implemented and realized. This paperproposes:

1. Adaptive signal processing .
2. Noise and echo analysis and cancellation
3. The simulation is done using MATLAB.
4. For real time application,TMS320C6X DSP processor is used.

II.PROPOSED SYSTEM

Given below is the proposed system on which the further work will be conducted.

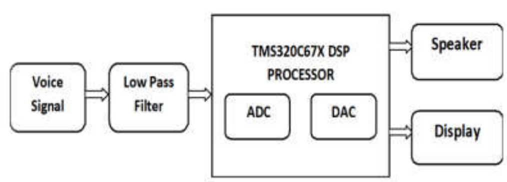


Fig.1 Block diagram

A. DSP Processor:

The TMS320C67 devices come with program memory which, on some devices, can be used as a program cache. The devices also have varying sizes of data memory. Peripherals such as a direct memory access (DMA) controller, power-down logic, and external memory interface (EMIF) usually come with the CPU, while peripherals such as serial ports and host ports are on only certain devices.DSP Processor is used to process on the noisy sound signal and obtain cleare,noise free signal at output side.

B. Frequency Filter:

· Applies gain > 1 for hard-to-hearfrequencies
· Modifies gain for other specifiedranges

The frequency shaper is designed to correct for loss of hearing at certain frequencies. The filter applies a gain greater than one to the frequencies that the user has difficulty hearing. As one of its parameters, the filter takes in avector of frequencies detrmined by an audiologist that define user's hearing characteristics.Thus,it is configurable to any user.

C. Amplitude Filter:

Once the signal has been passed through the Noise Reduction Filter and the Frequency Filter, it will be passed through our Amplitude Filter. The dynamic range of hearing is measured in terms of sound pressure, in decibels. A normal hearing range extends from approximately 0 dB to 120 dB, where 0 dB is the Threshold of Hearing and 120 dB is the Threshold of Pain.

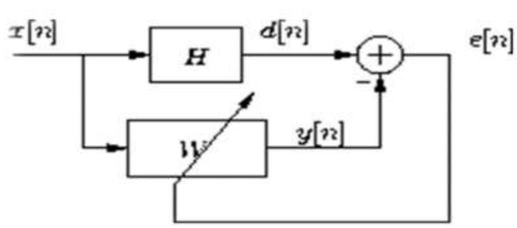
TABLE I

Classification of hearing loss	Hearing level
Normal Hearing	-10dB - 26dB
Mild hearing loss	27dB - 40dB
Moderate hearing loss	40dB - 70dB
Severe hearing loss	70dB - 90dB
Profound hearing loss	> 90dB

III.ALGORITHM

A. LEAST MEAN SQUARE ALGORITHM:

The objective is to change (adapt) the coefficients of an FIR filter, *W*, to match as closely as possible the response of an unknown system, *H*. The unknown system and the adapting filter process the same input signal $x[n]$ and have outputs $d[n]$ (also referred to as the desired signal) and $y[n]$.The adaptive filter, *W*, is adapted using the least mean-square algorithm, which is the most widely used adaptive filtering algorithm.



Fig(2):LMS Algorithm

First the error signal, $e[n]$, is computed as

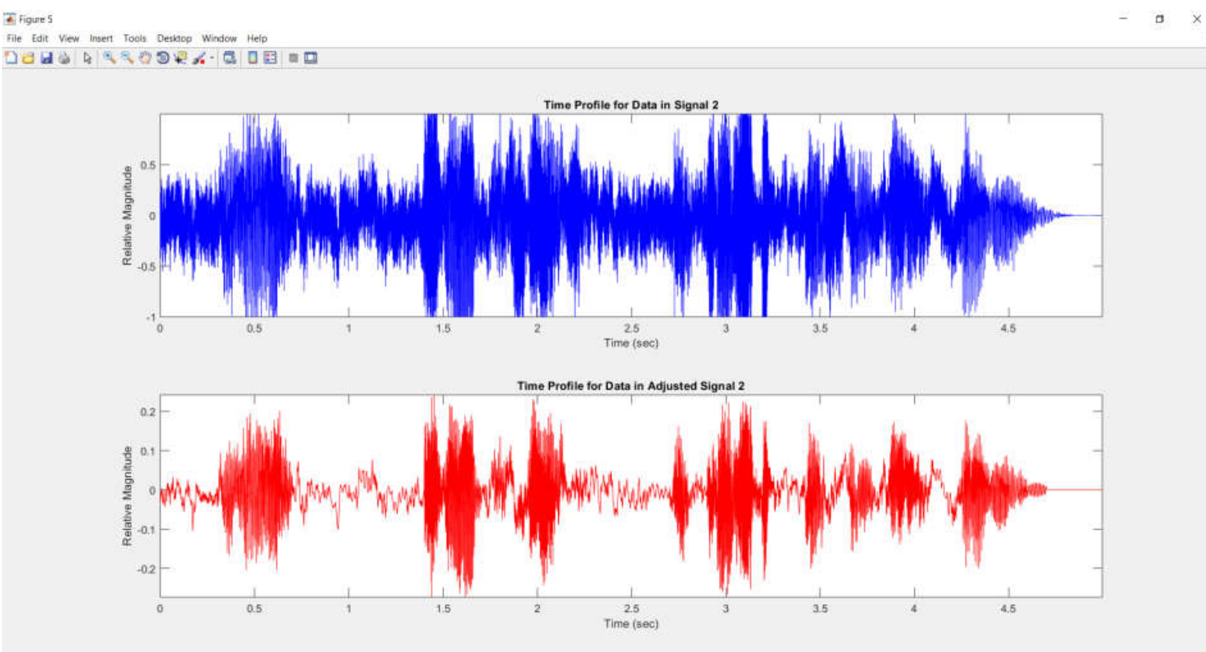
$e[n]=d[n]-y[n]$ ----------- (1),

this measures the difference between the output of the adaptive filter and the output of the unknown system. On the basis of this measure, the adaptive filter will change its coefficients in an attempt to reduce theerror.

LMS coefficient update,

$h_{n+1}[i]=h_n[i]+\mu e x[n-i]$ -------(2)

The step-size μ directly affects how quickly the adaptive filter will converge toward the unknown system. If μ is very small, then the coefficients change only a small amount at each update, and the filter converges slowly. With a larger step- size, more gradient information is included in each update, and the filter converges more quickly; however, when the step-size is too large, the coefficients may change too quickly and the filter will diverge. (It is possible in some cases to determine analytically the largest value of μ ensuring convergence.)



Fig(3): MATLAB SIGNAL REPRESENTATION

The above figure depicts the MATLAB output waveforms for noisy and noise-free signal. The blue coloured waveform contains high frequency noise signals which are removed in MATLAB using some inbuilt functions and the original voice signal is produced as shown in red colour. The output is obtained in both ways – audio signal and spectrograph representation.

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CONCLUSION

This paper mainly aims at proposing this methodology for noise cancellation, which has potential applications for impaired people as hearing aid. For a person with hearing impairment, the hearing aids are the only device which make them audible to the outside environment. The DSP based Hearing Aid system which employs compression system algorithm. The design was carried out using MATLAB and the implementation was based on the DSP development board.

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