ISSN (Online) : 2319 - 8753
ISSN (Print) : 2347 - 6710

International Journal of Innovative Research in Science, Engineering and Technology

Volume 3, Special Issue 3, March 2014

2014 International Conference on Innovations in Engineering and Technology (ICIET'14)
On 21st & 22nd March Organized by

K.L.N. College of Engineering, Madurai, Tamil Nadu, India

Noise Cancellation of Speech Signal by Using Adaptive Filtering With Averaging Algorithm

Vajrala Mangamma [#1], Saravanan.V [#2]
TIFAC-CORE in Pervasive Computing Technologies, Velammal Engineering College, Chennai, India.[#1]
TIFAC-CORE in Pervasive Computing Technologies, Velammal Engineering College, Chennai, India.[#2]

ABSTRACT—In any communication system noise always had been a major area of concern. Noise signals affect the transmitted signals during transmission. Every signal that we acquire at the receiver end of any communication system is somehow affected by noise. Noise is the unwanted part of the signal. Signals can be of various types in communication systems depending upon the requirement of the system. Signals carrying information during transmission of voice or speech or image or video. Each of these signals gets affected during transmission. So for de-noising of these signals of deferent types we need different noise cancellation mechanism. We have adopted a noise cancellation mechanism for signals carrying information in voice communication. In voice communication the recorded speech data signal transmitted will be useless until and unless proper noise cancellation mechanism is adopted at the receiver end. There are so many mechanisms for such noise cancellation. In this paper, we have adopted a new adaptive filtering with averaging algorithm for noise cancellation. This algorithm, we have proposed is based by using averages of both data and correction terms to find the updated values of the tap weights of the ANC controller of the speech signal.

KEYWORDS— adaptive filter, noise cancellar, voice transmisson, averaging algorithm and communication system.

I. INTRODUCTION

Speech is a very basic way for humans to convey information to one another with a band width of only 4 kHz; speech can convey information with the emotion of a human voice. The speech signal has certain properties: It is a one-dimensional signal, with time as its independent variable, it is random in nature, it is non-stationary, and i.e. the frequency spectrum is not constant in time. Although human beings have an audible frequency range of 20Hz to 20 kHz, the human speech has significant frequency components only up to 4 kHz.

It is controversial how far human speech is unique in that other animals also communicate with vocalizations. While none in the wild have compatibly large vocabularies, research upon the nonverbal abilities of language trained apes such as Washoe and kanji raises the possibility that they might have these capabilities. The origins of speech are unknown and subject to much debate and speculation.

In this modern world we are surrounded by all kinds of signals in various forms. Some of the signals are natural, but most of the signals are man-made. Some signals are necessary (speech); some are pleasant (music), while many are unwanted or unnecessary in a given situation.

In an engineering context, signals are carriers of information, both useful and unwanted. Therefore extracting or enhancing the useful information from a mix of conflicting information is a simplest form of signal processing. More generally, signal processing is an operation designed for extracting.

The distinction between useful and unwanted information is often subjective as well as objective. Hence signal processing tends to be application dependent. In contrast to the conventional filter design techniques, adaptive filters do not have constant filter coefficients and no priori information is known. Such a filter with adjustable parameters is called an adaptive filter.

II.RELATED WORKS

A. Design and Implementation of Adaptive Filtering Algorithm for Noise Cancellation In Speech Signal On Fpga

In this paper, Diggikar, A. B., and S. S. Ardhapurkar (2012), deals with Adaptive noise canceller is used in the design and implementation of adaptive filtering algorithm for noise cancelation in speech signal. The output of the speech signal is applied to the field programmable gate array (FPGA).In this for the execution of the speech signal code VHDL is used is performed on the basis of Signal to Noise ratio (SNR) and Mean Square Error (MSE). This paper investigates the applicability of a FPGA system for real time audio processing systems. In recent years acoustic noises become more evident due to wide spread use of industrial equipments. An Active (also called as Adaptive) noise cancellation (ANC) is a technique that effectively attenuates low frequencies unwanted noise whereas passive methods are either ineffective or tends to be very expensive or bulky. An ANC system is based on a destructive interference of an anti-noise, which have equal amplitude and opposite phase replica of primary unwanted noise. Following the superposition principle, the result is noise free original sound.

B. Fpga implementation of an adaptive noise canceller with low signal distortion

In this paper, Vijaykumar et al (2007), project analyzes the Modified Adaptive Filtering Algorithm for Noise Cancellation in Speech, Speech signal is applied to the Adaptive filter and different types of noise signals are applied to the adaptive filter. Difference between the speech signal output and noise signal output are compared. One relationship between the strength of the speech signal and the masking sound is called the signal-to-noise ratio, expressed in decibels. Ideally, the S/N ratio is greater than 0dB, indicating that the speech is louder than the noise. Just how much louder the speech needs to be in order to be understood varies with, among other things, the type and spectral content of the masking noise. The most uniformly effective mask is broadband noise.

Although, narrow-band noise is less effective at masking speech than broadband noise, the degree of masking varies with frequency. High-frequency noise masks only the consonants, and its effectiveness as a mask decreases as the noise gets louder. But low-frequency noise is a much more effective mask when the noise is louder than the speech signal, and at high sound pressure levels it masks both vowels and consonants

C.De-Noising/Noise Cancellation Mechanism For Sampled Speech/Voice Signal

In this paper, Pramanik et al (2012), we have showed a de-noising or noising cancellation mechanisms using Mean Power Calculation (MPC) for voice communication. Our algorithm is applicable for voice communication only. For testing purpose we tested it with real time recorded voice communication only. Later on we have thought of implementing de-nosing mechanisms for image and video transmission.

We thought of testing these noise cancellation techniques with real time voice signals keeping our laboratory limitations and initial testing procedures in our mind Therefore, we selected voice frequency (VF) or voice band initially for the testing of our De-noising or noise cancellation mechanisms because voice frequency (VF) or voice band is one of the frequencies within part of the audio range, used for the transmission of speech. In telephony, the usable voice frequency band ranges from approximately 300 Hz to 3400 Hz. It is for this reason that the ultra-low frequency band of the electromagnetic spectrum between 300 and 3000 Hz, which is also referred to as voice frequency because of being the electromagnetic energy that represents acoustic energy at baseband. The bandwidth allocated for a single voice-frequency transmission channel is usually 4 kHz including guard bands allowing a sampling rate of 8 kHz to be used as the basis of the pulse code modulation system used for the digital PSTN.

Noise Cancellation or De-noising using Mean power calculation***:*** To determine the maximum power located in a time series we take frame wise average power of the samples. In the frames where the power is much higher than the other frames determines the position of the signal in the time series. The position of a speech signal in a recorded time sequence can be determined by calculating the frame wise power as in the time series where the speech is located the frames will give higher power. This power detection will help us in the algorithm.

III.IMPLEMENTATION AND RESULTS

A. Block diagram

As Shown In Figure 3.1, This System Consists Of Filter, Adaptive Filter with Averaging Algorithm and DSP Processor and its description is given below.

- Applying Adaptive filter to the noise signals by using adaptive filtering with averaging algorithm
- For the output of the Adaptive filter, execute the code of the adaptive filter with averaging algorithm in the MATLAB
- The output of the MATLAB is applied to the DSP processor.

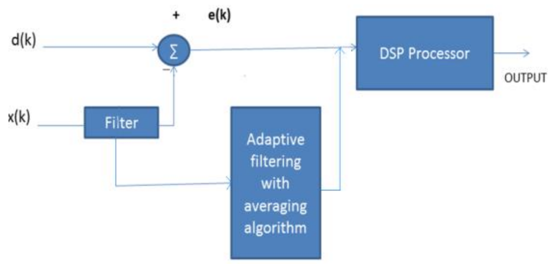

Figure 3.1 Block Diagram of Proposed System

B.Filter

The usual method of estimating a signal corrupted by additive noise is to pass it through a filter that tends to suppress the noise while leaving the signal relatively unchanged i.e. direct filtering.

The design of such filters is the domain of optimal filtering, which originated with the pioneering work of Wiener and was extended and enhanced by Kalman, Bucy and Others.

B. Types of filters used in noise cancellation

Filters used for direct filtering can be either Fixed or Adaptive.

1. Fixed filters
2. Adaptive filters

1. Fixed filters - The design of fixed filters requires a priori knowledge of both the signal and the noise, i.e. if we know the signal and noise beforehand, we can design a filter that passes frequencies contained in the signal and rejects the frequency band occupied by the noise.

2. Adaptive filters - Adaptive filters, on the other hand, have the ability to adjust their impulse response to filter out the correlated signal in the input. They require little or no a priori knowledge of the signal and noise characteristics. (If the signal is narrowband and noise broadband, which is usually the case, or vice versa, no a priori information is needed; otherwise they require a signal (desired response) that is correlated in some sense to the signal to be estimated.) Moreover adaptive filters have the capability of adaptively tracking the signal under non-stationary conditions.

Noise Cancellation is a variation of optimal filtering that involves producing an estimate of the noise by filtering the reference input and then subtracting this noise estimate from the primary input containing both signal and noise.

It makes use of an auxiliary or reference input which contains a correlated estimate of the noise to be cancelled. The reference can be obtained by placing one or more sensors in the noise field where the signal is absent or its strength is weak enough. Subtracting noise from a received signal involves the risk of distorting the signal and if done improperly, it may lead to an increase in the noise level. This requires that the noise estimate nˆ should be an exact replica of n.

If it were possible to know the relationship between n and nˆ, or the characteristics of the channels transmitting noise from the noise source to the primary and reference inputs are known, it would be possible to make nˆ a close estimate of n by designing a fixed filter. However, since the characteristics of the transmission paths are not known and are unpredictable, filtering and subtraction are controlled by an adaptive process. Hence an adaptive filter is used that is capable of adjusting its impulse response to minimize an error signal, which is dependent on the filter output.

The adjustment of the filter weights, and hence the impulse response, is governed by an adaptive algorithm. With adaptive control, noise reduction can be accomplished with little risk of distorting the signal. In fact Adaptive Noise Canceling makes possible attainment of noise rejection levels that are difficult or impossible to achieve by direct filtering.

C. Adaptive Filtering Algorithms

In this modern world we are surrounded by all kinds of signals in various forms. Some of the signals are natural, but most of the signals are man-made. Some signals are necessary (speech); some are pleasant (music), while many are unwanted or unnecessary in a given situation.

In an engineering context, signals are carriers of information, both useful and unwanted. Therefore extracting or enhancing the useful information from a mix of conflicting information is a simplest form of signal processing. More generally, signal processing is an operation designed for extracting, enhancing, storing, and transmitting useful information. The distinction between useful and unwanted information is often subjective as well as objective. Hence signal processing tends to be application dependent.

In contrast to the conventional filter design techniques, adaptive filters do not have constant filter coefficients and no priori information is known. Such a filter with adjustable parameters is called an adaptive filter.

The basic idea of an adaptive noise cancellation algorithm is to pass the corrupted signal through a filter that tends to suppress the noise while leaving the signal unchanged. This is an adaptive process, which means it does not require a priori knowledge of signal or noise characteristics. Although both FIR and IIR filters can be used for adaptive filtering, the FIR filter is by far the most practical and widely used. The reason being that FIR has adjustable zeros, and hence it is free of stability problems associated with adaptive IIR filters that have adjustable poles as well as zeros. However the adaptive FIR filters are not always stable and their stability depends critically on the algorithm.

Many computationally efficient algorithms for adaptive filtering have been developed within the past twenty years. They are based on either a statistical approach, such as the least-mean square (LMS) algorithm, or a deterministic approach, such as the recursive least-squares (RLS) algorithm. The major advantage of the LMS algorithm is its computational simplicity. The RLS algorithm, conversely, offers faster convergence, but with a higher degree of computational complexity.

D.Hardware requirement

TMS320C31 DSP Processor

1) **Key Features -**The TMS320C31 includes the features normally associated with a general-purpose embedded controller, so designing with it is very similar to designing with RISC or CISC devices. But the ’C31 is distinguished by many many high-perfor- mance features not found on processors in its price range:

2) **High performance**
   - 50-ns instruction cycle
   - 20 MIPS (million instructions per second)
   - 40 MFLOPS (million floating-point operations per second)
   - 220 MOPS (million operations per second)
   - 80-Mbytes/second I/O bandwidth
   - 0.200 µs interrupt response
   - 60-ns and 74-ns devices also available

3) **Register-based, pipelined CPU**
   - Parallel multiply and arithmetic/logical operations on integer or floating-point numbers in a single cycle
   - Eight extended-precision registers
   - 24-bit address space
   - Two address generators with eight auxiliary registers, two index registers, and two auxiliary register arithmetic units
   - 32-bit barrel shifter

4) **Powerful instruction set**
   - Single-cycle instruction execution
   - System control and numeric operations
   - Two and three operand instructions
   - Zero-overhead looping
   - Single-cycle branching
   - Conditional calls and returns
   - Flexible addressing modes including circular addressing and auto-increment/decrement modes allow high-speed data accesses
   - Single-cycle parallel math and memory operations
   - Interlocked instructions for multiprocessing support

5) **Integrated peripherals**
   - DMA controller for concurrent I/O and CPU operation
   - Two-way set associative instruction cache maximizes performance while minimizing system cost
   - Flexible serial port for 8/16/24/32-bit transfers which can be configured for general-purpose bit I/O plus two 16-bit timers
   - Two 32-bit timers which can also be configured for bit I/O
   - Extensive internal busing and parallelism for extremely fast data-movement capability
   - 8K bytes of single-cycle dual-access internal RAM support two accesses per machine cycle—can act as program memory, data memory, cache to external memory, or register file extensions
   - Memory interface optimized for single-cycle SRAM accesses and static-column decode DRAMs for high-speed external memory access while maintaining low system cost.
   - Boot loader to load/execute programs from other processors or inexpensive EPROMS
   - On-chip emulation for true nonintrusive visibility and control during debug
   - 132-pin plastic quad flat pack (PQFP) package

- Low price

TMS320C31 performance:

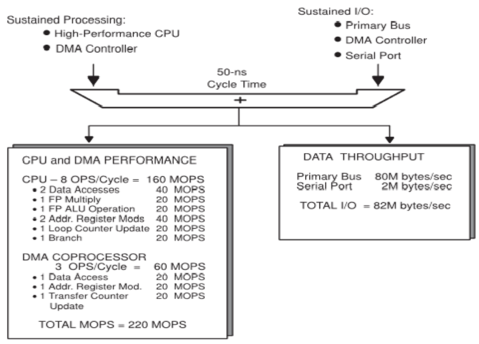

Figure 3.2 Performance Measure

6) Speech Recognition with TMS320C31 and SPOX

Voice Processing Corp. (VPC) of Cambridge, Massachusetts, a leader in speech recognition technology, develops and markets proprietary technology for speaker-independent continuous and discrete word recognition. VPC has taken an approach to speech recognition that is particularly adept for handling voices over the telephone. Telephone transactions is one area in which speech recognition technology has a compelling market need.

VPC has been supplying speech recognition technology to telecom system manufacturers and over-the-phone service providers for several years, allowing these firms to replace human operators. VPC recognizers are being used in a wide array of applications, such as credit card verification, operator intercept, telephone order entry, and voice-mail.

D. Simulation results

The proposed method for the noise cancellation in speech signal is implemented in MATLAB simulation software and its output is shown in the figure 3.3.

In this figure 3.3, the reference signal and primary signal are the input signals, cancelled wave is the noise cancellation wave from the two input signals finally desired wave is the output obtained from the two input signals after noise cancellation.

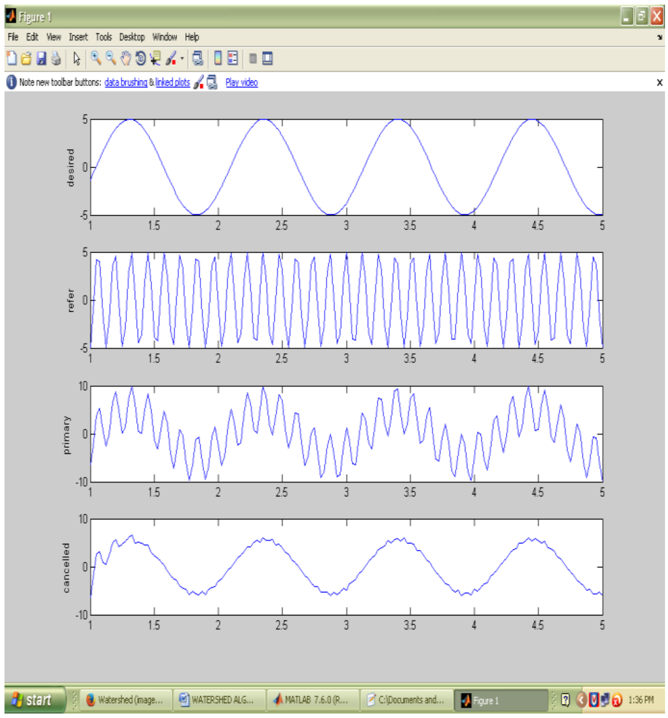

Figure 3.3 Signals and Noise cancellation

IV.CONCLUSION AND FUTURE WORK

The master work is to develop a new featured algorithm for noise cancelation in speech signal by using Adaptive filtering with averaging algorithm Work carried out in this phase to produce noise cancellation in speech signal by using Adaptive Filtering with Averaging Algorithm is implemented in MATLAB. From the comparative study of literature survey a best algorithm method is chosen and it is implemented in order to cancel noise in speech signal and moreover this analysis is done based on the performance measure. Further the future works consists of implementation of this algorithm in MATLAB Simulink and in DSP processor to get efficient result. Thus, noise in speech signal is cancelled easily.

REFERENCES

[1] Diggikar, A. B., and S. S. Ardhapurkar (2012) "Design and Implemtaention of Adaptive filtering algorithm for Noise Cancellation in speech signal on FPGA." In Computing, Electronics and Electrical Technologies (ICCEET), 2012 International Conference on, pp. 766-771. IEEE.

[2] Subramaniam, Vijay K., Visshwanth M. Reddy, and Sathyanarayana S.Rao(1999) "FPGA implementation of an adaptive noise canceller with low signal distortion." In Signals, Systems, and Computers, 1999. Conference Record of the Thirty-Third Asilomar Conference *on*, vol. 2, pp. 1396-1400. IEEE

[3] Vijaykumar, V. R., P. T. Vanathi, and P. Kanagasapabathy(2007) "Modified adaptive filtering algorithm for noise cancellation in speech signals." Electronics and Electrical Engineering.–Kaunas: Technologija.
[4] Pramanik, Arnab, and Rajorshee Raha (2012) "De-noising/noise ca for sampled speech/voice signal." Wireless and Optical Comr (WOCN), 2012 Ninth International Conference on. IEEE.
[5] Akhtar, Muhammad Tahir, Masahide Abe, and Masayuki Kawamata (2004) "Adaptive filtering with averaging-based algorithm for feedforward active noise control systems." Signal Processing Letters, *IEEE* .
[6] Malik, Garima, and Amandeep Singh Sappal. "Adaptive Equalization Algorithms: An Overview(2011)" International Journal of Advanced Computer Science and Applications.
[7] Embree, Paul M. *C*(1994) algorithms for real-time *DSP*. Prentice Hall PTR.