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ECHO Cancelation using LMS Filters

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Abstract: The common adaptive algorithms that have found widespread application are the Least Mean Squares (LMS) and the Recursive Least Squares (RLS). The LMS is the most efficient in terms of computation and storage requirements. It does not suffer from numerical instability problem inherent in the other algorithm. One of the major features of adaptive noise cancellation schemes is the identification of external noise and cancellation of the external noise signal. Continuous adaption to changing environment requires the use of adaptive systems for the identification purposes.

Keywords: ECHO, LMS filters, adaptive systems

1. INTRODUCTION

The field of digital signal processing has developed so fast in the last three decades that it can be found in the graduate and undergraduate programs of most universities. This development is related to the increasingly available technologies for implementing digital signal processing algorithms. The tremendous growth of development in the digital signal processing area has turned some of its specialized areas into fields themselves. If accurate information of the signals to be processed is available, the designer can easily choose the most appropriate algorithm to process the signal. When dealing with signals whose statistical properties are unknown, fixed algorithms do not process these signals efficiently. The solution is to use an adaptive filter that automatically changes its characteristics by optimizing the internal parameters. The adaptive filtering algorithms are essential in many statistical signal processing applications. The adaptive filter has the property that its frequency response is adjustable or modifiable automatically to improve its performance in accordance with some criterion, allowing the filter to adapt to changes in the input signal characteristics. Because of their self adjusting performance and in- built flexibility, adaptive filters are used in many diverse applications such as echo cancellation, radar signal processing, navigation systems, and equalization of communication channels and in biomedical signal enhancement. Adaptive filters are used, when it is necessary for the filter characteristics to be variable, adapted to changing conditions, when there is spectral overlap between the signal and noise or if the band occupied by the noise is unknown or varies with time. Norbert Wiener studied the theory of adaptive system in the early 1959. At the time of this work, only a handful of people were interested in adaptive systems and the development of multi weight adaptive filter was just the beginning. The common adaptive algorithms that have found widespread application are the Least Mean Squares (LMS) and the Recursive Least Squares (RLS). The LMS is the most efficient in terms of computation and storage requirements. It does not suffer from numerical instability problem inherent in the other algorithm.

II. MOTIVATION

Noise cancellation technology is a growing field that capitalizes on the combination of disparate technological advancements. The purpose of the technology is to cancel or at least minimize unwanted signals. The main aim of adaptive noise cancellation technology is to remedy the excess noise that is experience in communication links or channels. Noise canceling technologies first rely on small microphones that detect the sounds in a given environment.

III. ADVANTAGE OF DIGITAL SIGNAL PROCESSING

A more appropriate definition for DSP as it applies toward the computer industry can be derived from the name digital signal processing itself–DSP is the processing of analog signals in the digital domain. Real-world signals, such as voltages, pressures, and temperatures, are converted to their digital equivalents at discrete time intervals for processing by the CPU of a digital computer. The result is an array of numerical values stored in memory, ready to be processed.

DSP is useful in almost any application that requires the high-speed processing of a large amount of numerical data. The data can be anything from position and velocity information for a closed loop control system, to two-dimensional video images, to digitized audio and vibration signals. This application note describes DSP from an application point of view to demonstrate the many different and effective uses for digital signal and array processing. common factor in all of these applications is the need to do extremely high-speed calculations on large amounts of data in real time.

Modern computational power has given us the ability to process tremendous amounts of data in real-time. DSP is found in a wide variety of applications such as filtering, speech recognition, image enhancement, data compression, neural networks as well as functions that are unpractical for analog implementation, such as linear-phase filters.Signals from the real world are naturally in analog form, and therefore must first be discretely sampled for a digital computer to understand and manipulate.

The signals are discretely sampled and quantized, and the data is represented in binary format so that the noise margin is overcome. This makes DSP algorithms insensitive to thermal noise. Further, DSP algorithms are predictable and repeatable to the exact bits given the same inputs. This has the advantage of easy simulation and short design time.

IV. ADAPTIVE FILTERS

Adaptive filters, which aim to transform information-bearing signals into “cleaned up” or “improved” versions, adjust their characteristics according to the signals encountered. They form the simplest examples of algorithms within the field of machine learning. Adaptive filters are often preferred over their fixed-characteristic counterparts, which are fundamentally unable to adjust to changing signal conditions. The convenient autonomous adaptability of adaptive filters explains their widespread application in signal restoration, interference cancellation, system identification, and medical diagnostics, to name just a few areas.

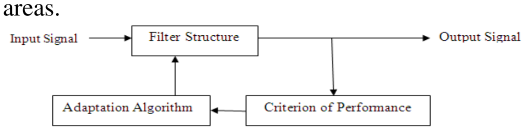

Figure 1. Adaptive Filter Structures

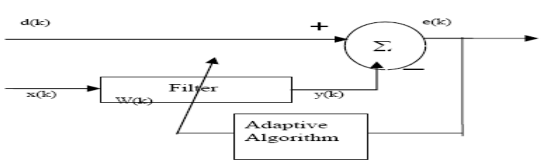

Figure 2. General set up of Adaptive Filter

V. BASIC CONCEPT OF THE NOISE CANCELLER

Noise cancellation makes use of the notion of destructive interference. When two sinusoidal waves superimpose, the resulting waveform depends on the frequency, amplitude and relative phase of the two waves. If the original wave and the inverse of the original wave encounter at a junction at the same time, total cancellation occur. The challenges are to identify the original signal and generate the inverse without delay in all directions where noises interact and superimpose.

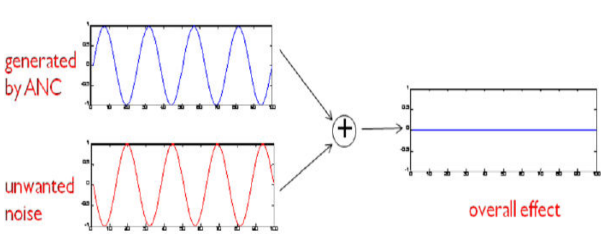

Figure3. Signal Cancellation of two waves 180° out of phase

VI. ADAPTIVE NOISE CANCELLER

As noted in the introduction, the more one knows about the speech and noise signals, the more effectively one can extract speech from noise. If, for example, the noise waveform is known exactly, then extracting the speech is a trivial problem. All that is necessary is to simply subtract the known noise waveform from the speech-plus-noise waveform and be left with speech only.

There are situations in which the noise waveform can be identified exactly. Consider the case of a single noise source in a typical room. It is possible to place a microphone at the location of the noise source so as to pick up noise only. A second microphone elsewhere in the room (e.g., on a hearing aid) will pick up both speech and noise. In order to subtract the noise from the speech plus noise picked up by the hearing-aid microphone, it is necessary to take into account the fact that there will be reflections of the noise off the walls of the room; i.e., by the time the noise gets to the hearing-aid microphone, the noise waveform will have changed. It is possible to process the noise waveform so as to correct for these reflections. A special-purpose filter can be used for this purpose. If the filter is designed properly, subtracting the filtered noise from the speech plus noise picked up by the hearing-aid microphone will effectively cancel the noise with only the speech remaining . A system of this type is shown in figure 2. Since people typically move around in a room, the pattern of reflections will change, and so it is necessary for the filter to keep adjusting itself. This method of noise reduction is known as adaptive noise cancellation.

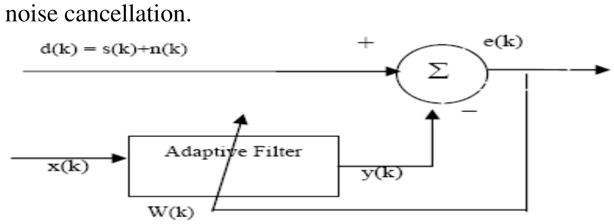

Figure 4. Adaptive Filters as a Noise Canceller

VII. APPLICATIONS OF ADAPTIVE FILTERS

Because of their ability to perform well in unknown environments and track statistical time-variations, adaptive filters have been employed in a wide range of fields. However, there are essentially four basic classes of applications for adaptive filters. These are: Identification, inverse modeling, prediction, and interference cancellation, with the main difference between them being the manner in which the desired response is extracted. These are presented in figure 5, 6,7 and 8 respectively. The adjustable parameters that are dependent upon the applications at hand are the number of filter taps, choice of FIR or IIR, choice of training algorithm, and the learning rate. Beyond these, the underlying architecture required for realization is independent of the application.

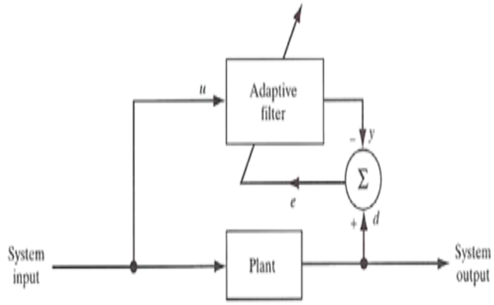

Figure 5 System Identification

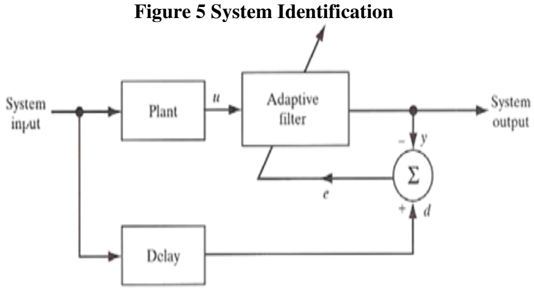

Figure 6 Inverse Modeling

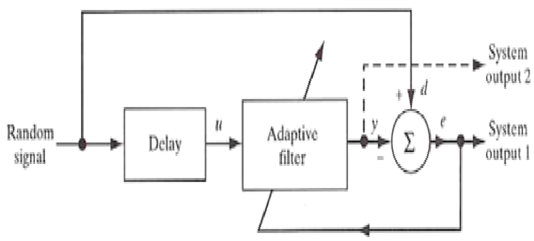

Figure 7 Prediction

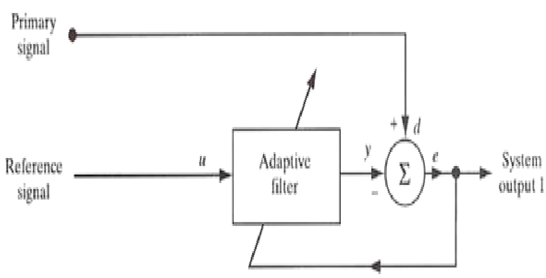

Figure 8 Interference Cancellation

VIII. CONCLUSION

This noise was much more difficult to eliminate since there exists little correlation between the reference noise and the noise in primary, given that they are both completely random.

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