Selectable algorithms for adaptive noise cancellation systems are promoted in this book. The book deals with digital signal processing for signal enhancement, it offers a source of information related to the algorithm development and implementation of adaptive noise cancellation systems. These systems can be very useful in various real-time applications such as telecommunication, headphones, hearing aid, voice controlled system, teleconferencing and many others. The book targets DSP designers, researchers and postgraduate students in the field of DSP for mobile communication systems. The book provides analysis of the results obtained from subjecting the developed system to environmental noise.

Dr Roshahliza received her B.Eng in Information System (2008) and M.Eng in Computer Science & Electrical Engineering (2010) from Polytechnic University of Japan. She then received her PhD in Electrical, Electronics & System Engineering from National University of Malaysia (UKM) in 2015. She is now a Senior Lecturer at Universiti Malaysia Pahang.

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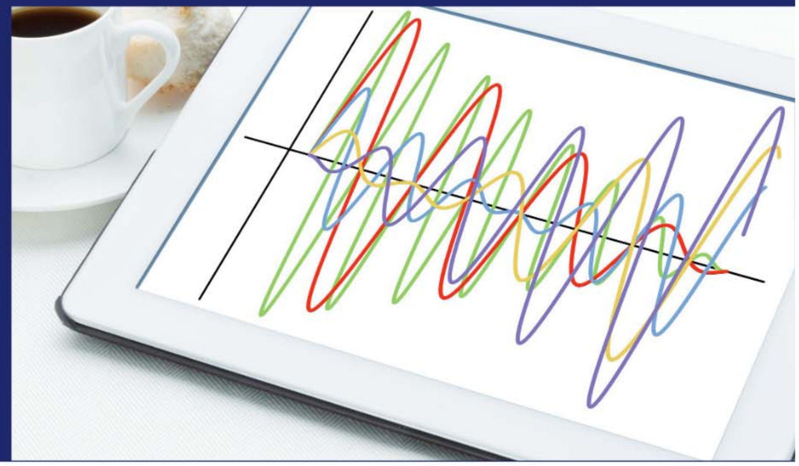
Selectable Algorithm for Noise Canceller

M. Ramli, Abdul Samad



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ADAPTIVE NOISE CANCELLATION SYSTEMS USING SELECTABLE ALGORITHMS IN VARIABLE NOISE ENVIRONMENTS

ROSHAHLIZA BINTI M RAMLI

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ABSTRACT

Adaptive noise cancellation systems in varying noise environments are developed and analysed in this study. These systems reduce noise embedded in a signal adaptively to improve the signal-to-noise ratio (SNR). The main objectives of this study are to give solutions on the slow convergence and high computational complexity incorporated with the use of real-life background noise in speech signal applications. Besides the conventional Adaptive Noise Canceller (ANC) system which utilizes multiple sensors in its configuration, this study also examines a single sensor version of ANC called the Adaptive Line Enhancer (ALE). The initial arrangement of the ALE is based on using the Affine Projection (AP) algorithm as a substitute for the conventional Least Mean Square (LMS) algorithm. Performance of the AP based ALE is examined with white, colored and real-life background noise signals. Results showed a relatively faster convergence rate compared to conventional ALE. However, this comes at the expense of a higher computational complexity when the projection order is set to a high value. Therefore, the AP algorithm is then replaced by a developed version of set-membership filtering called the Dynamic Set-Membership Affine Projection (DSM-AP) algorithm to reduce the computational complexity as well as to improve the convergence performance when colored input signals are used as test signals. Using this version, the filter coefficients of an adaptive filter are updated only if the output estimation error is higher than a prescribed quantity. This algorithm showed a faster convergence rate and a larger error deterioration compared to other adaptive algorithms such as the LMS, the Recursive Least Square (RLS) and the classical AP. However, the computational complexity of the developed algorithm is still high. With the aims of reducing computational complexity and retaining convergence to satisfactory levels, a Smart Noise Canceller (SNC) system is proposed by applying a suitable adaptive algorithm according to the characteristics of the input noise. The selection criterion is based on the measurement of the eigenvalue spread of the autocorrelation of the input noise. Experimental results show the capability of the SNC to apply a suitable algorithm according to the characteristics of noise, and removes noise from several types of real-life noisy speech signals. The SNR performance of the SNC show improvements up to 30 dB at the system output and the convergence performance outperform the conventional noise cancellers. Moreover, the computational complexity is reduced to almost 65% of that of the ANC using the RLS algorithm. When the DSM-AP algorithm is deployed in the SNC scheme, named as Improved Smart Noise Canceller (ISNC), the results show an overall better performance in convergence and SNR compared to other single adaptation noise cancellers. In the last development stage, the DSM-AP algorithm is also adopted in ALE with algorithm selection mechanism which is called the Smart Adaptive Line Enhancer (SALE). The SALE exhibited faster convergence rates and superior SNR compared to the ALE with a single adaptation algorithm. The filtered signals are then evaluated subjectively using the Mean Opinion Score (MOS) questionnaire in order to obtain the perception of the listeners on the quality of the processed speech signals. The result showed the speech processed by SALE has the highest scores in the audio test. This result proves that the speech processed by this method is the most desirable output compared to others. The MOS acceptance rate showed 93% for this method, followed by 83% for the speech processed by SNC.

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LIST OF ILLUSTRATIONS

LIST OF SYMBOLS

$A(z)$	Unknown channel
d	Desired signal
$\mathbf{d}$	Desired signal vector
δ	Regularization parameter
Δ	Prediction distance delay
e	Error signal
$E[\cdot]$	Expectation operator
$\sum$	Summation
f	Frequency
f_s	Sampling frequency
$\mathbf{g}$	Error-bound vector
H	Constraint set
λ	Eigenvalue
λ_{max}	Largest eigenvalue
λ_{min}	Smallest eigenvalue
i	Time instant
$\mathbf{I}$	Identity matrix
$\mathbf{k}(n)$	Filter gain vector
k	Natural mode
M	Projection order
N	Filter length
n	Time index
Θ	Solution set / feasibility set
$\mathbf{P}$	Inverse correlation matrix
$\mathbf{Q}$	Eigenvector matrix
$\mathbf{R}$	Input autocorrelation matrix
$\mathbf{R}^{-1}$	Inverse matrix of $\mathbf{R}$
$\mathbb{R}$	Real number set
S	Data space
$s(n)$	Input signal

$(\cdot)^T$	Matrix transpose
τ_k	Time constant at k-th iteration
μ	Adaptation gain parameter
v_k	Tap-weight difference at k-th iteration
$x(n)$	Input noise
$\mathbf{X}$	Input signal vector
$\hat{x}(n)$	Estimate of x
$x(n-\Delta)$	Delayed version of input signal
$\chi(\mathbf{R})$	Eigenvalue spread of $\mathbf{R}$
$y(n)$	Output signal
γ	Prescribed quantity
Ψ	Exact membership set
ψ	Intersection of the constraint sets
z^{-1}	Unit delay
$\|\cdot\|$	Norm
σ^2	Variance of measurement noise
$\{\}$	Set
$\forall$	For all elements
$\in$	Set-membership
$\cap$	Intersection
$\subset$	Subset

LIST OF ABBREVIATIONS

ALE	Adaptive Line Enhancer
ANC	Adaptive Noise Cancellation/Canceller
ANN	Artificial Neural Network
AP	Affine Projection
DSM-AP	Dynamic Set-Membership Affine Projection
DSP	Digital Signal Processor
ECG	Electrocardiogram
EEG	Electroencephalogram
FFT	Fourier Fast Transform
FIR	Finite Impulse Response
FPGA	Field Programmable Gate Array
IFFT	Inverse Fourier Fast Transform
IIR	Infinite Impulse Response
ISNC	Improved Smart Noise Canceller
ITU	International Telecommunication Union
LMS	Least Mean Square
MOS	Mean Opinion Score
MSE	Mean Square Error
NALE	Neural Network based Adaptive Line Enhancer
NLMS	Normalized Least Mean Squares
$O(N^2)$	Order of N^2
OBE	Optimal Bounding Ellipsoids
RLS	Recursive Least Square
SALE	Smart Adaptive Line Enhancer
SM-AP	Set-Membership Affine Projection
SMI	Set-Membership Identification
SNC	Smart Noise Canceller
SNR	Signal to Noise Ratio
SPL	Sound Pressure Level
SSM-AP	Simplified Set-Membership Affine Projection
VAD	Voice Activity Detection

CHAPTER I

INTRODUCTION

1.1 SIGNAL AND NOISE

According to Haykin (2003) and Vaseghi (2008), a signal is defined as a function of one or more variables that conveys information. In most cases, signals are real-world parameters such as temperature, pressure, sound, light, and electricity. These signals are then being processed or analysed, by a time-varying parameters that simply represent their physical quantity. Then these signals could be used by humans or machine for various applications such as in telecommunication, control, home appliances and also in medical appliances. In many applications, signal can be implemented either in analog or digital approach. The analog approach or continuous-time signal processing was dominant for many years and relies on the use of analog circuit elements such as resistors, capacitors, inductors, amplifier and diodes. By contrast, the digital signal processing or discrete-time approach relies on three basic digital computer elements such as adders, multipliers and memory for arithmetic operations and data storage.

The digital signal processing approach has two advantages over analog approach whereby firstly, it can be implemented on the same digital machine or called as digital signal processors (DSPs) by making changes on program software for different operations of interest. Meanwhile, the analog machine has to be redesigned every time operations of interest are changed. Secondly, operations in digital approach can be repetitive. In contrast, analog systems suffer from parameter variations that can be influenced by changes in the supply voltage or room temperature. In this thesis, digital signal processing approach is to be applied for the techniques proposed.

However, signals in the real world could not be captured clean enough without unwanted signals known as interference or noise. The target signal is prone to errors due to noise that tends to disturb the operation of a system which we have incomplete control. In communication system, noise can cause transmission errors and may even disrupt a communication process. Hence, noise cancelling process is an important part of modern telecommunication and signal processing systems in order to deliver the information stored in signals. There are potential sources of noise affecting the operation of a communication system. In particular, the noise can be defined in two categories; internal sources and external sources.

The internal source of noise mainly occurs from spontaneous fluctuations of the current or voltage signal in electrical circuits which commonly referred as electrical noise. This kind of noise can be detected in signals of analog circuits such as random fluctuations of the electric current in an electrical conductor. On the other hand, the examples of external sources of noise include atmospheric or environmental noise and human-made noise. These noises maybe interfering signals picked up by the receiver of the communication system due to spectral characteristics of the interference lying inside the operating frequency range for which the system is designed. In speech communication, the interfering environmental noise can disturb the quality or intelligibility of the signal for listeners. If the noise becomes too dominant, then the signal quality quickly degrades. Therefore, in the past few decades, various techniques have been developed to remove or at least reduce the effect of these unwanted signals.

In this thesis, the removal of background noise from speech signal is considered as typical application for the proposed techniques. Speech is one example of sound signal that can be processed digitally and its amplitude varies with time. Speech is basically generated when sound pressure waves originating from air pushed control by the lungs are channelled or restricted in various ways by manual control of the human vocal system components (Sinha 2009).