Ain Shams University

Faculty of Engineering

Electronics and communication Engineering Department

Robust Adaptive Signal Processing to Improve the Digital Receiver Performance

A THESIS

Submitted in Partial Fulfillment of the Requirements for the Degree of Master of Science in Electrical Engineering

By

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Statement

This dissertation is submitted to Ain Shams University for the degree of Master of Science in Electrical Engineering (Electronics and Communications Engineering).

The work included in this thesis was carried out by the author at the Electronics and Communications Engineering Department, Faculty of Engineering, Ain Shams University, Cairo, Egypt.

No part of this thesis was submitted for a degree or a qualification at any other university or institution.

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Abstract

Noise cancellation in a signal is an important core area of the digital signal processing. In this work, a novel algorithm for cancelling noise from the speech signal in real time environment is proposed. In many applications of noise cancellation, the characteristics of signal may change quite fast. This requires the usage of adaptive algorithms, which converge rapidly. One of the most popular adaptive noise cancellers that often used to recover signal corrupted by additive noise is Least Mean Square (LMS) algorithm and that is due to its simplicity in implementation. But it has limitation when the desired signal is strong, that the excess mean-square error is linearly increased while increasing the desired signal power.

This results in downgraded performance when the desired signal exhibits large power fluctuations. In the proposed algorithm we use the benefits of both variable step size (VSS) LMS algorithm and Normalized Differential LMS (NDLMS) algorithm to handle this situation. One more addition of this algorithm is that it uses the concept of decomposing the long adaptive filter into low order multiple sub-filters to relieve the effect of slow convergence of that long adaptive filter. Finally, the proposed (P-VSSNDLMS) algorithm yields faster convergence with minimum mean square error in simulations which performed using real speech signal with different noise power levels.

Keywords

Adaptive Noise canceller (ANC), mean square error (MSE), VSSNDLMS, multiple sub-filters

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C	Speed of Light
d(n)	Desired Signal
e(n)	Error Signal
f_d	Doppler Shift
J	Cost function
K	Gain vector
L	Filter Length
μ	Step Size
μ_{var}	Variable Step Size
n	Additive Noise
N	Number of taps for a filter
P_s	Average Power of Speech Signal
P_n	Average Power of Noise Signal
Rb	Bit Rate
S(n)	Transmitted Signal
S	Power Spectral Density
T_c	Coherence time
T_s	Symbol time
ν	Mobile Speed
w(n)	Weight vector
y(n)	Output of the filter
Δ	Time delay

∇	Gradient
δ	Impulse or Regularization parameter
σ_τ	Path delay or RMS delay spread
λ	Eigen value of correlation matrix or forgetting factor