بسم الله الرحمن الرحيم

شبكة المعلومات الجامعية

التوثيق الالكتروني والميكروفيلم

شبكة المعلومات الجامعية

جامعة عين شمس

التوثيق الالكتروني والميكروفيلم

قسم

نقسم بالله العظيم أن المادة التي تم توثيقها وتسجيلها

علي هذه الأفلام قد أعدت دون أية تغيرات

يجب أن

تحفظ هذه الأفلام بعيدا عن الغبار

في درجة حرارة من ١٥-٢٥ مئوية ورطوبة نسبية من ٢٠-٤٠%

To be Kept away from Dust in Dry Cool place of 15-25- c and relative humidity 20-40%

بعض الوثائق

الاصلية تالفة

بالرسالة صفحات

لم ترد بالاصل

BΣVΥ9

MENOUFIA UNIVERSITY
Faculty of Electronic Engineering, Menouf
Department of Electrical
Communications Engineering

Digital Processing of Speech Signals

By

Samir Abd El-ghaffar Abd El-Daim

A Thesis Submitted to The
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In Partial Fulfillment of the Requirements for the Degree of
Master of Science

In
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S. El-S...
14-6-2004

To

My Parents

To

My Wife

To

My Lovely Children

بسم الله الرحمن الرحيم

وقل رب زدني علماً

صدق الله العظيم

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Abstract

Abstract

This thesis represents an efficient method for reducing the quantization noise for coding speech signals at low bit rates. This method has gained interest due to its ease of implementation and channel-noise robustness compared to other digital Coders. Adaptive Differential Pulse Code Modulation (ADPCM) is one of the most efficient schemes for digital encoding of speech. Different algorithms have been proposed for increasing the dynamic range for a given value of Signal-to-quantization Noise Ratio (SNR). The predictor is a vital part of the overall system. This thesis also investigates the use of second-order prediction to increase the SNR for several ADPCM systems at low bit rates spechialy Jayant Adaptive Quantizer (JAQ) and Icremental Adaptive Quantizer (IAQ). Inaddition a study and comparsion has been done between the different adaptation systems. Moreover computer simulations have been performed using three different types of input signals, sinusoidal signal, random signal and real speech signal. Inaddition, a unit step function is used to show that the IAQ is the more suitable to follow the variation for speech signal. It concludes that the second order prediction gives better performance than the first order predication. Four decibels improvement in SNR has been achieved in the case of real speech signal and consequently the reduction bit rate has been obtained. Furthermore the hardware implementation of this system is constructed to realize the predictive coding with good digital accuracy which reduced word size.

List of Publications

A paper extracted from the research work of the MSc thesis:

M.E.M. Nasr, S.E. Diab, M.A. Zin El-Deen and S. Abd El-Daim. "Digital processing of speech using ADPCM with 2nd order predictor" Faculty of Electronic Engineering. Bulletin No. 22, July 2001, P.P. 61-66.

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