الاستهلال

قال تعالى:

" إِنْ أُرِيدُ إِلاّ الإِصْلاَحَ مَا اسْتَطَعْتُ وَمَا تَوْفِيقِيَ إِلاّ بِاللّهِ عَلَيْهِ تَوَكّلْتُ وَإِلَيْهِ أُنِيبُ "

صدق الله العظيم

سورة هود

الآية (٨٨)

DEDICATION

To

Endless love

Our mothers

To

Our fathers

To

Our brothers and Sisters

To

Our teachers &our colleagues

AKNOWLEDGMENT

First we need to thankfully our god (Allah) that without his blessing this work will not complete.

Then all thanks for our supervisor **Dr.FathElrahman** to his patience with us and countless hours and valuable efforts to guide and advise us to complete the work in his fair way.

We also give our thanks to Dr.Omer Abdalrazig and engineer. Abdullah Mustafa who gave us all what we need to accomplish this work.

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ABSTRACT

Speech coding has been and still is a major issue in the area of digital speech processing. Reducing the transmission bandwidth and achieving higher speech quality are primary concerns in developing new speech coding algorithms.

The standard PCM produce voice signal with data rate of 64Kbps which reduces the efficiency of bandwidth utilization, increases the cost and complicates to use in security applications.

The goal of this thesis is to develop a system for encoding good quality speech at low bit rate.To implement this, used very efficient speech analysis technique called Linear Predictive Coding (LPC).

The basic idea behind linear predictive analysis is that the speech sample can be approximated as a linear combination of past samples.

In this project, the simulation of the LPC vocoder was done using matlab. The output voice quality was tested by using perceptual evaluation signal quality (PESQ) and it was 60%.Then LPC was implemented in DSK C6713 chip, and the Mean Opinion Score (MOS) was used to evaluate the quality, the average rate of the MOS was 3 which means the quality is fair.

المستخلص

ترميز الصوت من القضايا المهمة في مجال معالجة الاشاره الرقمية ،الهدف الأساسي من تلك المعالجة تمثيل الاشارة بأقل عدد من البتات مع المحافظة على مستوى مقبول من الجوده ، ترميز التعديل النبضي يقوم بإنتاج إشارة بمعدل 64 كيلوبت في الثانيه وهذا المعدل يقلل من كفاءة استخدام عرض النطاق الترددي ويؤدي لزيادة التكلفة ومن الصعب استخدامه في التطبيقات الامنيه.

الهدف الاساسي من هذا البحث هو تطوير نظام لترميز مستوي جيد من الصوت باقل عدد ممكن من البتات، لتطبيق ذلك تم تقديم تقنية ذات كفاءة عاليه تسمى (الترميز التنبؤي الخطي)، الفكرة الأساسية في هذه التقنية في ان نماذج الكلام يمكن أن يتم تقديرها كدمج خطي للنماذج السابقة.

في هذا البحث تم محاكاة تقنية الترميز التنبؤي الخطي باستخدام الماتلاب واختبار جودة خرج الصوت بواسطة التقييم الادراكي لجودة الاشارة وكانت 60%.وبعد ذلك تم تطبيق تقنية الترميز التنبؤي الخطي في دائرة معالج الاشاره الرقمية C6713 وتم اختبار جودة صوت الخرج بواسطة متوسط نقاط الاراء وكان المتوسط 3 ويعني منصف.

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ABBREVIATIONS

A/D	Analogue to Digital Convertor
ADPCM	Adaptive Differential Pulse Code Modulation
AIC	Analogue Interface Circuit
AMDF	Average Magnitude Difference Function
BPF	Band Pass Filter
CELP	Code Exited Linear Predictor
CLK	Clock
CPLD	Complex Programmable Logic Device
D/A	Digital to Analogue
DIP	Dual Interface Packaging
DSK	Digital signal processor Starter Kit
FDATOOL	Filter Design and Analysis Tool
FEC	Forward Error Correction
FIR	Finite Impulse Response
FS	Federal Standard
GUI	Graphical User Interface
HPF	High Pass Filter
Hz	Hertz
IDE	Integrated Development Environment
IIR	Infinite Impulse Response
ITU	International Telecommunication Union
JTAG	Joint Test Action Group
Kbps	Kilo bit per second
LED	Light Emitting Diode

LPC	Linear Predictive Code
LPF	Low Pass Filter
LTI	Linear Time Invariant
MATLAB	Matrix Laboratory
McBSPs	Multi channel Buffered Serial Ports
MELP	Mixed Exited Linear predictive code
MDF	Magnitude Difference Function
MOS	Mean Opinion Score
PARCOR	Partial Autocorrelation
PC	Personal Computer
PCM	Pulse Code Modulation
PESQ	Perceptual Evaluation of Speech Quality
PSTN	Public Switched Telephone Network
RCs	Reflection Coefficients
SNR	Signal to Noise Ratio
SYNC	Synchronization
V/UV	Voiced / Unvoiced
ZC	Zero Crossing