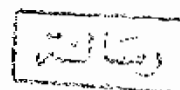


VARIABLE SAMPLING RATE TECHNIQUES

A THESIS
Submitted to the Faculty of Engineering
In partial fulfilment of the requirements
for the degree of
Master of science
In the Department of Electronics and
Communication Engineering
Ain Shams University



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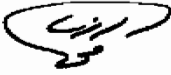
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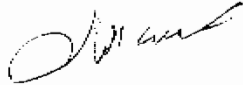
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AIN - SHAMS UNIVERSITY

THE APPLICATION OF VARIABLE SAMPLING RATE TECHNIQUE
TO SPEECH SIGNALS

Student : K. T. Bakir Supervisor : Prof. Dr. S. H. El-Ramly
M.SC. Thesis presented to Electronics and Communication
Engineering Department (1993) .

ABSTRACT

The main purpose of this thesis is to apply the variable sampling rate technique to speech signals for the reduction of the transmission rate and evaluation of the resulted speech quality . This thesis contains the basic principles of the sampling theorem and its various extensions . A review of speech signal properties and speech uniform and nonuniform quantization techniques was introduced . The concept of the proposed variable sampling rate technique and the reduction of the speech effective bandwidth by reducing speech spectral energy was introduced and its mathematical analysis was conducted . The results of the application of the variable sampling rate technique to speech signals were discussed and a comparison between the variable sampling rate technique and log PCM and ADPCM was introduced . An implementation technique for the variable sampling rate along with DSI technique was proposed .

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INTRODUCTION

Signal coding is the process of representing an information signal in a way that realizes a desired communication's objective such as analog to digital conversion , low bit rate transmission , or message encryption .

This thesis is concerned with the study of the application of the proposed variable sampling rate technique to pulse code modulated speech signals .

The process of digital coding of speech by means of a straight forward approximation of the time waveform is presented , in particular , discrete time discrete amplitude signal representation is discussed .

Chapter one reviews digital coding techniques of speech signals , and the process of amplitude discretization (quantization) is discussed for uniform and nonuniform cases [1,2,3,37] .

In chapter two the time discretization process (sampling process) was reviewed , starting from the origin of the sampling theorem , the Shannon's sampling theorem , the generalized sampling theorem and its extensions [42,43] .

In chapter three speech spectrum estimation by the use of the Discrete Fourier Transform (DFT), and the Fast Fourier Transform

(FFT) was reviewed [42] .

In chapter four the concept of variable sampling rate technique is presented and its related formulas deduced .

In chapter five the conclusions and results of the application of the variable sampling rate technique to speech signals is presented and discussed . A comparison between adaptive quantization techniques like log PCM and ADPCM and the variable sampling rate PCM is discussed . An implementation scheme for the VSRPCM is introduced , in conjunction with DSI and the VSRPCM alone .

CHAPTER ONE

DIGITAL PROCESSING TECHNIQUES OF SPEECH SIGNALS

1.1 Introduction :

This chapter presents a review of speech signal characteristics and speech generation , followed by discussion of digital processing techniques used for communication of speech signals . The digitization (discretization) of speech signal's magnitude will be reviewed , and two standrads of speech companding techniques will be discussed .

1.2 Speech coding :

To reduce signals to a minimum of digital information a wide range of source coding methods is available and these can be divided into three groupings [1].

a) The first grouping has as its objective the nearly exact replication at the receiver of the instantaneous wave shape submitted to the transmitter . It includes Pulse Code Modulation (PCM) , differential PCM (DPCM) , Delta Modulation (DM), and adaptive versions of the later two ,Adaptive Differential PCM (ADPCM) and Adaptive Delta Modulation (ADM) .

b) The second grouping comprises methods which remake the signal at the receiver using parametric models intended to represent the