VARIABLE SAMPLING RATE TECHNIQUES

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AIN - SHAMS UNIVERSITY

THE APPLICATION OF VARIABLE SAMPLING RATE TECHNIQUE TO SPEECH SIGNALS

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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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CONTENTS

PAGE

COPYRIGHT....................I
ACKNWLDGMENT....................II
ABSTRACT....................III
CONTENTS....................IV
LIST OF FIGURES....................VI
LIST OF TABLES....................VIII
INTRODUCTION....................IX

1- DIGITAL PROCESSING TECHNIQUES OF SPEECH SIGNALS..........1

1.1 Introduction....................1
1.2 Speech signals....................1
1.3 Human vocal mechanism....................2
1.3.1 Formants....................5
1.3.2 Vocal cavity model....................5
1.3.3 Phonemes....................7
1.4 Processing techniques of speech signals..............9
1.5 PCM (Pulse Code Modulation)....................13
1.5.1 Uniform quantization....................13
1.5.2 Nonuniform quantization....................18
1.5.2.1 A law companding....................20
1.5.2.2 μ law companding....................21
1.6 PCM standards....................23

2- SAMPLING THEOREM....................27

2.1 Introduction....................27
2.2 Shannon sampling theorem....................28
2.3 The generalized sampling theorem....................30
2.4 Extensions of the sampling theorem....................35
2.4.1 Sampling with nonuniformly spaced sampling points....................35
2.4.2 Implicit sampling....................39
2.4.3 Sampling for time varying systems with time varying bands....................41

3- SPEECH SPECTRUM ESTIMATION....................44

3.1 Introduction....................44
3.2 Discrete Fourier Transform....................45
3.3 Fast Fourier Transform....................47

4- THE PROPOSED VARIABLE SAMPLING RATE PCM TECHNIQUE.........51

4.1 Introduction....................51

4.2 Variable sampling rate bit rate reduction technique...52
4.3 Variable sampling rate algorithm......................64
4.4 Coded speech signal to noise ratio calculation........65
4.5 Suggested applications for the algorithm..............67
4.5.1 Applications to speech signals..................67
4.5.2 Applications to image signals...................71
4.6 Conclusion..72

5- RESULTS AND CONCLUSIONS...................................73

5.1 Introduction..73
5.2 Variable sampling rate PCM............................73
5.3 Computer simulation of the variable sampling rate algorithm..80
5.4 A proposed implementation scheme for the VSRPCM.......95
5.4.1 Implementation of the VSRPCM in conjunction with DSI...95
5.4.2 Implementation of the VSRPCM alone..............97
5.4.2.1 The VSRPCM encoding process............100
5.4.2.2 The channel assignment process.........102
REFERENCES...105

¢╝¢╛

LIST OF FIGURES

FIGURE PAGE

1.1 Sound spectrogram....................................4
1.2 Schematic digram of the vocal apparatus............6
1.3 A plot of second Formant frequency versus first Formant frequency for vowels by a wide range of speakers..8
1.4 Uniform quantization...............................15
1.5 SNR versus number of bits per sample for uniform quantization.......................................17
1.6 SNR versus number of bits per sample for A law companding and uniform quantization................22
1.7 SNR versus number of bits per sample for μ law companding and uniform quantization................24
2.1 Physical interpretation of Shannon's sampling theorem...31
2.2 A more practical system function for a filter of a sampling expansion..............................37
3.1 Two point DFT (FFT Butterfly)....................50
3.2 8 points FFT.......................................50
4.1 High pass and low pass cutoff frequency versus energy rejection threshold.........................54
4.2 Transmission rate versus cutoff frequency R_d,R_o....57
4.3 Decimation factor versus cutoff frequency (4.6).....59
4.4 Decimation factor versus cutoff frequency (4.11)....60
4.5 A plot of equ. (4.8) and (4.9) for transmission rate versus cutoff frequency.......................61
4.6 A plot of equ. (4.9) and (4.11) for transmission rate versus cutoff frequency O_r= 0.6 ,O_k= 0.2 62
4.7 Segmental SNR versus cutoff frequency...............68
4.8 (a) Segmental SNR versus transmission rate..........69
4.8 (b) Segmental SNR + Band width gain versus transmission rate....................................70
5.1 Average low pass and high pass cutoff frequency versus energy rejection threshold..................75
5.2 Transmission rate R_d ,R_o versus cutoff frequency at O_r = 0 , O_k = .3 78
5.3 Transmission rate R_d ,R_o versus cutoff frequency at O_r = .5 , O_k = .9 79
5.4 Transmission rate R_d ,R_o versus cutoff frequency at O_r = .2 , O_k = .6 81
5.5 A sample filter response at time and frequency domains...84
5.6 A sample speech segment before and after filtering at 2 KHz..85
5.7 A sample speech segment after decimation and after interpolation at F_c = 2 KHz.........................86

ቀ፱

5.8 Segmental SNR versus energy rejection threshold.....87
5.9 Segmental SNR versus cutoff frequency...............88
5.10 Segmental SNR , SNR + B.W.G versus cutoff frequency.89
5.11 Segmental SNR + B.W.G versus cutoff frequency.......90
5.12 Segmental SNR + B.W.G versus transmission rate......91
5.13 SNR of ADPCM , log PCM and ASRPCM versus transmission rate..................................94
5.14 VSRPCM + DSI.......................................95
5.15 System gain versus energy rejection threshold.......98
5.16 Energy rejection threshold versus transmission rate.99
5.17 A block diagram of the VSRPCM encoder..............101
5.18 An example for the channel assignment information..104

LIST OF TABLES

TABLE PAGE

2.1 Facsmile encoders.................................12

2.2 Parametric encoders...............................12

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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