Ain Shams University
Faculty of Engineering
Electronics and Communication Eng. Dept.

# A Method For Nonuniform Sampling of Signals with Application to Speech Encoding

By

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A Thesis

Submitted to the Faculty of Engineering in partial fulfillment of the requirements of Master Degree in Electrical Engineering
[Electronics & Communications Engineering. Dept.]



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## Statement

This thesis is submitted to Ain Shams University in Partial fulfillment of the degree of M.Sc. in Electrical Engineering.

The Work included in this thesis was carried out by the author in the Department of Electronics and Communication Engineering, Faculty of Engineering, Ain Shams University.

No part of this thesis has been submitted for a degree or qualification at any other university or institute.

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## Abstract

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In digital communication a uniform sampling of signals is usually used where the inter-samples intervals are constant and equal to the inverse of the Nyquist sampling frequency which is calculated to be twice the maximum frequency component in the signal. Some signals have the nature of being nonstationary, where its statistical characteristics is varying with time, and one of these characteristics is the maximum frequency component in the signal.

Speech signal is a nonstationary signal and its maximum frequency component is varying with time. The conventional uniform sampling rate of speech signal is equal to 8 kHz where the maximum frequency component in the speech signal is considered to be 4 kHz. This maximum frequency component, in fact, occurs in short intervals of time and during most of the speech signal interval the maximum frequency component is less than this value. So during these intervals of low maximum frequency component ( low bandwidth ) the 8 kHz sampling frequency is considered to be over-sampling frequency and results in a high amount of redundant information which degrades the efficiency of the speech coder. many efforts are done to overcome this problem by using a variable or nonuniform sampling rate.

In this thesis a survey of nonuniform sampling rate methods is introduced and a simple technique for continuous variable sampling rate is proposed. The sampling rate of the proposed method is continuously varied according to the amount of information (not only the rate of change) in the speech signal. The proposed method is studied using a sinusoidal test signal and a real speech signal. The quality of the decoded speech signal is compared with that of the uniform sampling method using a sampling frequency equal to the resulting average nonuniform sampling frequency in order to maintain the same average bit rate.

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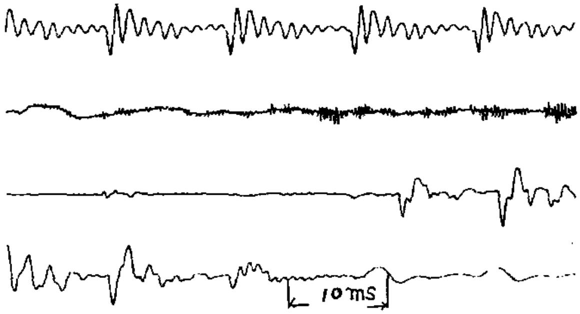
# List of Abbreviation

| | |
|---|---|
| **A-b-S** | Analysis-by-Synthesis |
| **acf** | Auto-correlation function |
| **ADM** | Adaptive Delta Modulation |
| **ADPCM** | Adaptive Delta Pulse Code Modulation |
| **APB** | Adaptive Prediction Backward |
| **APF** | Adaptive Prediction Forward |
| **AWGN** | Additive Gaussian Noise |
| **BER** | Bit Error Rate |
| **CCITT** | International Telegraph and Telephone Consultative Committee |
| **CELP** | Code Excited Linear Prediction |
| **CVSD** | Continuously Variable Slope Delta Modulation |
| **DAC** | Digital to Analog Convener |
| **DFT** | Discrete Fourire Transform |
| **DPCM** | Differential Pulse Code Modulation |
| **DPCM-AQB** | Differential Pulse Code Modulation-Adaptive Quantizer Backward |
| **DPCM-AQF** | Differential Pulse Code Modulation-Adaptive Quantizer Forward |
| **DSI** | Digital Signal Interpolation |
| **DSP** | Digital Signal Processing |
| **FFT** | Fast Fourire Transform |
| **ISDN** | International Service Digital Network |
| **MP-LPC** | Multi-Pulse Linear Predictive Coder |
| **MSE** | Mean square Error |
| **PCM** | Pulse Code Modulation |
| **RAM** | Random Access Memory |
| **RELP** | Regular Excited Linear Prediction |
| **RMS** | Root Mean Square |
| **ROM** | Read Only Memory |
| **SBC** | Sub-Band Coders |
| **SEGSNR** | Segmental Signal to Noise Ratio |
| **SNR** | Signal to Noise Ratio |
| **TDMA** | Time Division Multiple Access |
| **VBR** | Variable Bit Rate |
| **VQ** | Vector Quantization |

The signals shown in figure(1-1) illustrate the great variety of the character of speech wave. Sometimes periodic or quasi periodic, other times a mixture of periodic and random-like signals, and some times the waveform appears like random noise. A 10 ms time interval is shown in figure(1-1), a speech coder operating for example, at 4kb/s must be able to describe any such 10 ms segment (80 samples ) using only 40 binary digits in such a way that the segment will be reproduced with an accuracy sufficient to ensure that it will sound very close to the original[20].




Figure(1-1): Example of speech waveform [20]

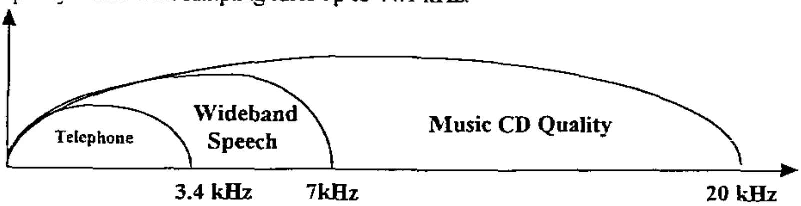
## 1.3-Compression of Speech signal

Compression of speech has been an ongoing area of research for several decades. In the last several years, there has been an interest and activity in this area with numerous applications in telecommunication and storage. High fidelity audio compression has also advanced rapidly in recent years , accelerated by the commercial success of consumer and professional digital audio products. The surprising growth of compression techniques is driven by the insatiable demand for voice communication , by the new generation of technology for cost effective implementation of digital signal processing algorithms, by the need to conserve band width in both wired and wireless telecommunication networks, and the need to conserve disk space in digital voice storage systems. As shown in figure(1-2) most of these efforts are focused on:

1-The usual telephone bandwidth of roughly 3.2 kHz (200 Hz to 3.4 kHz) with sampling rates of 8 kHz.

2- Wideband speech (7 kHz ) for audio in video teleconferencing with sampling frequency of 16 kHz

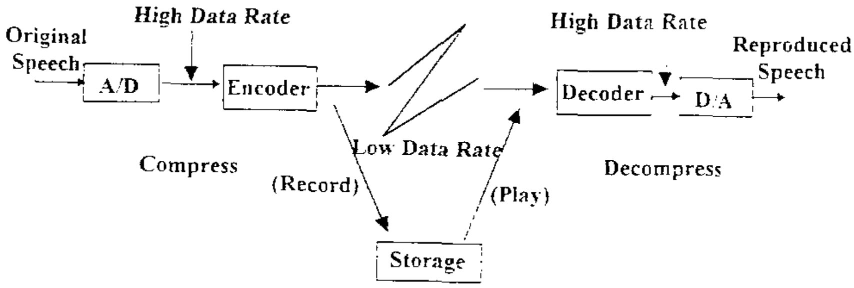
3-Wideband audio signal (typically 20 kHz bandwidth) for transmission and storage of CD quality music with sampling rates up to 44.1 kHz.




Figure(1-2) Types of signals

Another new area of application is multimedia in personal computing where voice storage is becoming a standard feature.

Virtually all work in speech and audio compression involves ***lossy compression*** where the numerical representation of the signal samples is never recovered exactly after decoding (decompression). There is a wide range of tradeoffs between bit rate and recovered speech quality that are of particular interest in the coding of telephone speech , where users are accustomed to certain degrees of degradation. On the other hand , for wide band audio compression the quality is close to that of the compact disk (CD). Thus research in speech compression includes studies for distortion-rate tradeoffs motivated by different applications with different quality objectives.




Figure(1-3) Compression and Decompression of speech

Figure(1-3) is a schematic diagram illustrates the purpose of the encoding process which enables low data transmission rates and less storage capacity.

## 1.4-Waveform coders versus voice coders (vocoders)

Speech coding algorithm can be divided into two main categories

1- Waveform coders.

2- Vocoders (voice coders)

In ***waveform coders***, the data transmitted from encoder to decoder specify a representation of the original speech as a waveform of amplitude versus time, so that the reproduced signal approximate the original waveform and, consequently, provides an approximate recreation of the original sound

Waveform coders (non-parametric) are used to compress telephone, speech and music bandwidth signal. It is designed to efficiently quantize any audio waveform by quantizing only new and unpredictable information. Such coders are robust for diverse signal sets, but don't accomplish high compression

In contrast, ***Vocoders*** do not reproduce an approximation to the original waveform, instead, parameters that characterize individual sound segments are specified and transmitted to the decoder, which then reconstructs a new and different waveform that will have a similar sound ***Vocoders*** are sometimes called ***parametric coders***, while ***waveform coders*** called ***nonparametric coders***. Often these parameters characterize the short term spectrum of the sound. Alternatively the parameters specify a mathematical model of human speech production suited to a particular sound. In either case the parameters do not provide sufficient information to regenerate an