



**AIN SHAMS UNIVERSITY**  
**FACULTY OF ENGINEERING**  
**Electronics and Communications Department**

## **Development of Audio Codec for Mobile System WCDMA**

A Thesis submitted in  
partial fulfillment of the requirements for the Degree of  
Master of Science in Electrical Engineering.

Submitted by

**Seham Ibrahim Abd El karem**

B.Sc. in Electrical Engineering  
(Electronics and Communications Engineering, zagazige University, 2002)

Supervision by

**Prof. Dr. Abd El Halim Abd EL Naby Zekry**  
Department of Electronics and Communications Engineering  
Faculty of Engineering, Ain Shams University

**Dr. Mohamed Ibrahim Shedeed**  
Chairman of the Board of Directors of  
The Center for Scientific and Technological Excellence

**Dr. Mohamed Nabel Mostafa**  
Department of computer science Engineering  
Faculty of Engineering, Ain Shams University

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**AIN SHAMS UNIVERSITY**  
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**JUDGEMENT COMMITTEE**

Name: Seham Ibrahim Abd El Karem

Thesis: Development of Audio Codec for Mobile System WCDMA

Degree: Master of Science in Electrical Engineering

**NAME, TITLE, AND AFFILIATION**

**SIGNATURE**

**Prof. Dr. Talaat Abd El Latif El Garf**

Higher Technological Institute  
Tenth of Ramadan City

.....  
(Examiner)

**Prof. Dr. Abd El Moniem Abd El Zahar Wahdan**

Computer and System Engineering Department  
Faculty of Engineering, Ain Shams University.

.....  
(Examiner)

**Prof. Dr. Abd El Halim Abd El Naby Zekry**

Electronics and Communications Engineering Department (Supervisor)  
Faculty of Engineering, Ain Shams University.

.....  
(Supervisor)

**Prof. Dr. Mohamed Ibrahim Shedeed**

Chairman of the Board of Directors of  
The Center for Scientific and Technological Excellence

.....  
(Supervisor)

Date: .../.../.....

Communication Department  
Faculty of Engineering  
Ain Shams University  
Cairo - 2009



**AIN SHAMS UNIVERSITY  
FACULTY OF ENGINEERING  
Electronics and Communications Department**

**STATEMENT**

This Dissertation is submitted to Ain Shams University in partial fulfillment of the Degree of Master of Science in Electrical Engineering (Electronics and Communications Engineering).

The work included in this thesis was received by author at the Department of Electronics and Communications Engineering, Faculty of Engineering, Ain Shams University, Cairo, Egypt.

No part of this thesis was submitted for a degree or qualification at any other university or institution.

Name: Seham Ibrahim Abd El karem

Signature: .....

Date: ...../...../.....



**AIN SHAMS UNIVERSITY**  
**FACULTY OF ENGINEERING**  
**Electronics and Communications Department**

## **CURRICULUM VITAE**

Reasearcher Name :Seham Ibrahim Abd El karem  
Date of birth : 20/7/1980  
Place of birth : Egypt, benha.  
Nationality : Egyptian  
First university degree: B.Sc. in Electrical Engineering  
(Electronics and Communications  
Engineering, zagaziage University, 2002)  
Certification date : May, 2002.

Name: Seham Ibrahim Abd El karem

Signatur:.....

Date: ...../...../.....

## List of Abbreviations

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MP3	MPEG Audio Layer 3
WCDMA	Wide Band Code Division Multiple Access.
PCM	Pulse Code Modulation.
ISO	International Standards Organization.
MPEG	Motion Picture Expert Group.
DSP	Digital Signal Processor.
DVB	Digital Video Broadcasting
ARIB	Association of Radio Industry and Business
FPGA	Field Programmable Gate Array
CRC	Cyclic Redundancy Check
CBR	Constant Bit Rate
VBR	Variable Bit Rate
FFT	Fast Fourier Transform
MDCT	Modified Discrete Cosine Transform
SMR	Signal-To-Mask Ratio
VLC	Variable Length Coding
IMDCT	Inverse Modified Discrete Cosine Transform
MFLOPS	1,000,000 Flops
IDE	Integrated Development Environment
DSK	DSP Starter Kit
COFF	Common Object File Format
DSPLIB	Digital Signal Processing Library
AIFF	Audio Interchange File Format

## ***Thesis Abstract***

Seham Ibrahim Abd EL Karem. Development of Audio Codec for Mobile System WCDMA. Master of Science in electrical engineering, Ain Shams University 2009

### **Abstract**

The Wide-Band Code Division Multiple Access is the technology of the third generation mobile systems. It has many advantages over the second generation mobile systems. It has better bandwidth efficiency. In addition, wide-band services such as videophone and Internet services are provided including video streaming. Recently new digital audio applications have been used for network, broadcasting and multimedia which face constraints such as channel bandwidth. The audio of CD quality using sampling frequency 44.1 KHZ, 16 bits, 2 channels requires 1.411.200 bits per second that means to be able to play an audio file with CD quality, we need more than 1.4 Mbit/s in band width. So different compression methods have been used to decrease bandwidth and keep the same quality of audio signal. The international standard ISO 11172-3 defines three different methods of increasing complexity and compression efficiency for perceptual coding of audio signal.

This thesis deals exclusively with the third method also known as MP3. It has become very popular for compressing CD quality music with almost no audible degradation. MP3 depends on psychoacoustic model which used in encoder only to decide which parts of the signal are irrelevant and which parts are not, and remove the inaudible parts.

In chapter 1, introduction to WCDMA system has been mentioned and different audio formats have been discussed.

In chapter 2, we studied the characteristics of the human ear and its ability to differentiate between sounds. Only a range of sound frequency is perceptible by human beings. The audible range is between 20 Hz and 20 kHz. MP3 coding depends on the masking phenomena. There are two types of masking: simultaneous masking and temporal masking. The simultaneous masking is achieved if two different signals are close to each other. The strong signal (masker) will mask the weaker signal (maskee). Temporal masking is achieved if a loud sound and a quiet sound is played simultaneously. By placing a sufficient delay between the two sounds the softer sound will be heard. Then we studied the MP3 encoder. In the encoder, the input audio signal is passed through a polyphase filter bank. This filter bank divides the input signal into 32 equally-space subbands. The samples in each subband are still in the time domain. A Modified Discrete Cosine Transform (MDCT) is then used to convert the samples in each subband to frequency domain. In the same time, input signal after FFT transformation passes through a psychoacoustic model that determines the ratio of the signal energy to the masking threshold (SMR) for each subband. The distortion control block uses the SMR from the psychoacoustic model to decide how to assign the total number of code bits available for the quantization of the subband samples to minimize the audibility of the quantization noise. Then the quantized subband samples are coded with the lossless Huffman coding. Finally the end block takes the Huffman coded samples and side information into a packed bitstream according to the MPEG/Audio standard.

This chapter ended with the MP3 decoder. The decoder is simpler than encoder because it only has to reconstruct the bit stream and does not need to be concerned about psychoacoustic model or the quality of the encoded data.

In chapter 3, the MP3 encoder/decoder reference software from ISO was discussed and implemented in different environments like on PC and on Digital Signal Processor. The DSP used in this thesis is TMS320C6713.

The implementation of the codec was done in several steps:

First, we implemented the program on PC by using different languages like C and matlab. Our goal was obtaining an encoded file from encoder with the same quality as original file and high compression ratio then send the encoded file to the decoder to get the original file.

Second, On DSP the source code was implemented and optimized to reach the real time requirements.

Finally, Chapter 4 concludes the thesis by providing a summary of the results of this work and identifying directions for future work.

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