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 Ibrahem 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

EGYPT, CAIRO
2009

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 Ibrahem 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

EGYPT, CAIRO
2009

AIN SHAMS UNIVERSITY
FACULTY OF ENGINEERING
Electronics and Communications Engineering Department

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   (Examiner)
Tenth of Ramadan City

Prof. Dr. Abd El Moniem Abd El Zahar Wahdan   ……………….
Computer and System Engineering Department   (Examiner)
Faculty of Engineering, Ain Shams University.

Prof. Dr. Abd El Halim Abd El Naby Zekry   ……………….
Electronics and Communications Engineering Department   (Supervisor)
Faculty of Engineering, Ain Shams University.

Prof.Dr. Mohamed Ibrahem Shedeed   ……………….
Chairman of the Board of Directors of   (Supervisor)
The Center for Scientific and Technological Excellence

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

<u>CURRICULUM VITAE</u>

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

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

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 then 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.

Acknowledgements

I would like to thank and express my appreciation and gratitude to the following people:

Prof.Dr. ABDEL-HALIM ZEKRY for accepting me as his student despite the fact that he is already overloaded, for helping me in understanding a lot of things and for giving me too many favors.

Dr. Mohamed Nabel for helping me a lot during the implementation part.

Dr. Mohamed Ibrahim Shadeed for accepting me as his student and for supporting during the two years master project.

I would like to thank my little family, my husband, and my son for their support and encouragement during finishing this work .

.

CONTANTS

List of Tables