M.Sc. Thesis

A new proposed eight bit hybrid mash-efm multi-order sigma delta modulator for audio dac

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DECLARATION

I, Khalid Ijaz, solemnly declare that the work presented in this thesis is my own and nothing is reproduced to make contempt of plagiarism.

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DEDICATION

I dedicate this research work to the Beloved Holy Prophet Muhammad (PBUH), members of his holy family and his highly loyal companions (may Almighty Allah be pleased with all of them) because He (PBUH) is the one who opened all the doors of knowledge benevolence.

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ABBREVATIONS

SF-DIF	: Sony Philips Digital Interface
ASRC	: Asynchronous Sample Rate Converter
SDM	: Sigma Delta Modulator
DAC	: Digital- to- Analog Converter
BMC	: Biphase Mark Coding
DAT	: Digital Audio Tape
RCA	: Radio Corporation of America
FOH	: First Order Hold
ZOH	: Zero Order Hold
AFE	: Audio Front End
ABE	: Audio Back End
DR	: Dynamic Range
LPCM	: Linear Pulse Code Modulation
PLL	: Phase Locked Loop
DUT	: Device under Test
FFT	: Fast Fourier Transform
ADC	: Analog- to -Digital Converter
LPF	: Low Pass Filter

DDSM	: Digital Delta Sigma Modulator
STF	: Signal Transfer Function
NTF	: Noise Transfer Function
SQNR	: Signal-to-Quantization Noise Ratio
SNR	: Signal-to-Noise Ratio
SFDR	: Spurious Free Dynamic Range
ENOB	: Effective Number of Bits
OSR	: Oversampling ratio
IBN	: In band quantization Noise
SQ-DSM	: Single Quantizer -Delta Sigma Modulator
RC SQ-DSM	: Reduced Complexity Single Quantizer-Delta Sigma Modulator
DNSL	: Digital Noise Shaping Loop
SNDR	: Signal-to-Noise plus Distortion Ratio
CT-SDM	: Continuous Time-Sigma Delta Modulator
MASH	: Multistage Noise Shaping
SMASH	: Sturdy MASH
FFSDM	: Feed forward SDM
CLFSDM	: Cascaded Low order Feed forward Sigma Delta Modulator
CFFSDM	: Cascaded Feed forward Sigma Delta Modulator
ELD	: Excessive Loop Delay
DEM	: Dynamic Element Matching

SAR	: Successive Approximate Register
EFM	: Error Feedback Modulator
CIFB	: Cascade-of-integrators feedback
CRFB	: Cascaded resonators with distributed feedback
CIFF	: Cascade-of-integrators feed forward

PREFACE

In this research work a new proposed architecture (Hybrid MASH-EFM) of SDM is simulated to obtain very high SNR. The results of two already existing architectures i.e. MASH and EFM are compared with the results of proposed architecture. The chapters are divided and characterized in the following order:

Chapter no 1 deals with the introduction to the development of Audio technology and study of typical D/A system which involves hardware components plus the coding system. Then, a brief historic review of Digital Audio system is presented. The last portion of this chapter includes the background of a well known problem occurred in D/A system called Jitter.

Chapter no 2 discusses the fundamental theory of Digital Signal processing which involves the Sampling phenomena. This chapter also presents the basic operation of SDM with some definitions of its performance parameters. Some problems like Idle tones and Limit cycles which degrade the performance parameters are also discussed in this section.

Chapter no 3 briefly describes the working of conventional architectures of DAC's which were used in Telecommunication industry for many years before the use of SDM DAC.

Chapter no 4 discusses different topologies of SDM DAC.A lot of SDM architectures had been presented by the researchers in the past. This chapter deals with the study of all these architectures along with their results obtained in recent works. The review literature of all these existing architectures is the main part of this thesis which led to the development of new proposed architecture called Hybrid MASH-EFM.

Chapter no 5 discusses how the simulations of two existing architectures (MASH and EFM) and proposed architecture have been done. This chapter includes the comparison between the performance parameters of above two existing architectures. The output waveform and Power spectral density of these architectures are also shown. The performance parameters are recorded in a tabular form to make comparison at the end.

Chapter no 6 describes the structure of Hybrid MASH-EFM architecture. First order and second order Hybrid MASH-EFM architecture is implemented in this work. It describes how the output is analyzed in z-domain. It's STF and NTF is computed in mathematical form from the output analysis. NTF shows that it provides better Noise Shaping. The results show that a proposed architecture is much better than the above two existing architectures. After that final

conclusion has been made in the tabular form. At the end, the conclusion is made and depicts the advantages of proposed architecture. The future work is also discussed in the last.

ABSTRACT

Digital-to-Analog (DAC) Converters manifest a very imperative role in many applications where data is interfaced with real analog world. The occurrence of some problems like jitter, quantization errors, integral nonlinearity and conversion time reduces the performance of DAC. One of the most efficacious ways to combat these problems is the use of Sigma Delta Modulation (SDM) in DAC. This work emphasizes on the designing of Multi-order SDM to achieve high resolution DAC consuming low power and high SNR. The increase of oversampling ratio (OSR) requires the high speed DAC, overloads the quantizer, eventually reduces the DAC speed and increases the power consumption. Similarly, the complexity of the transistor circuitry in VLSI technology implementation increases as the order of SDM increases. Hybrid MASH-EFM architecture is used to overwhelm the above two conditions.

MATLAB simulation will be used for comparison of different Multi-order SDM architectures to study the effect on DAC performance. Different performance parameters are recorded at different OSR's of different architectures (MASH, EFM and Hybrid MASH-EFM).1^{st} order Hybrid MASH-EFM SDM achieves high SNR equal to 131.7 db and ENOB of 16 bits at OSR of 5. These results reveal that the proposed architecture (Hybrid MASH-EFM) is superior to two existing architectures.

Chapter 1

Introduction

1.1 Audio Technology Development

The Phonograph was first invented by Edison in the 1870s [1].Probably, he had no idea of the importance of recording and reproduction of sound and music. Audio technology progressed very rapidly with the time which enhances the performance as well as an opportunity of reproduced sound.

The awareness and command on the subject of a human auditory system and its properties led to the advancement of audio technology. Fletcher and Munson explored the new world by describing the response of human ear at different frequencies versus different pressure levels. He introduced the “Fletcher-Munson curves”, related to “equal-loudness contour” [2].This curve was then modified by Robinson and Dadson in 1956 and named as “Robinson-Dadson curves” [3] which turned into the basis of an ISO226 equal loudness standard.