FPGA Implementation of Signal Processing Algorithm for Audio Application

Renuka Satpute[1], Gauravi Talegaonkar[2], Rutuja Bhosale[3], Prof. Anjali Shrivastav[4]

[1-4]*Electronics and Telecommunication Dept., At Pimpri Chinchwad College of Engineering Pune*

---***---

Abstract – *With the growth of multimedia systems and the World Wide Web, audio processing techniques, such as filtering, equalization, noise suppression, compression, addition of sound effects and synthesis. A matlab-based implementation of an audio compressor was investigated. By varying the input parameters of the compressor, results were obtained which demonstrated the typical behaviour and benefits of compressor use. The parameters however, did not correspond to those found on typical hardware or software compressors. Adjustment to the matlab code could be made to make the controls more understandable to audio practitioners. In this Project We will develop efficient algorithm for audio application. Dynamic range compression (DRC) or simply compression is an audio signal processing operation that reduces the volume of loud sounds or amplifies quiet sounds, thus reducing or compressing an audio signal's dynamic range. DRC is Primary aspect used in audio application to reduce Hearing losses in different Hearing System. The Purpose of this algorithm is, using Dynamic Range Compression (DRC) to reduce human being hearing aid from audio application.*

Key Words: **Audio, Dynamic Range Compression(DRC), FPGA Board, MATLAB with TooBox.**

1. INTRODUCTION

Well, compressors and limiters are actually a tool that was created for use in the broadcast industry. When we transmit audio over the air to radio. So originally, the compressor/limiter was created to reduce the dynamic range of our audio signal, or to protect that audio signal from over-modulating the radio wave. Well, it turns out this also had a pretty cool effect on our audio. It made things sound louder and punchier, and it reduced the need for engineers to constantly be riding the faders to make audio that was high in dynamic range easily reproduced on everybody's sound system. When this all started catching on, it really started to become a standard tool in almost every recording studio. Engineers could start recording stronger signals, protect their audio equipment from accidentally clipping, and make their job riding the faders in the console much easier, and end up with more powerful recordings that translated better across different sound systems. Then we started associating that compressed sound with the radio sound. And history went on from there. Now without a doubt, if you've done any research on compression, you've heard of the term, "The Loudness Wars." As of late, the thought of overusing compression has certainly given compression a bad stigma in the industry. The reason why is because it reduces the dynamic range of an audio signal. This in turn increases the RMS, or average loudness level of that audio signal.

2. Proposed System

After doing lot of research, Literature Survey and study some papers found out to be valuable resources for development of the project.All contains different methodologies and techniques which are used in FPGA Implementation of Signal processing algorithm for Audio Application.

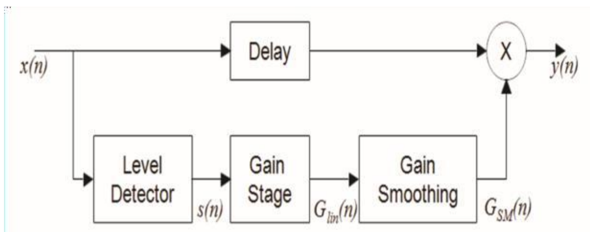

Fig 2.1: Blocks in Dynamic Range Compression.

The proposed system that is FPGA Implementation of Signal processing algorithm for Audio Application which compress audio signal. In that we are using Following blocks,

Level detector: The level detector measures the incoming signal frequency and translates it to a logarithmic scale.

Gain Stage: The gain step compares the approximate signal level with the predefined threshold value, Compression Threshold (CT) and measures the compression gain by the static gain curve which shown in fig b.

Gain Smoothing: A smoothing stage is implemented to reduce the effect of ripples produced by filtering.

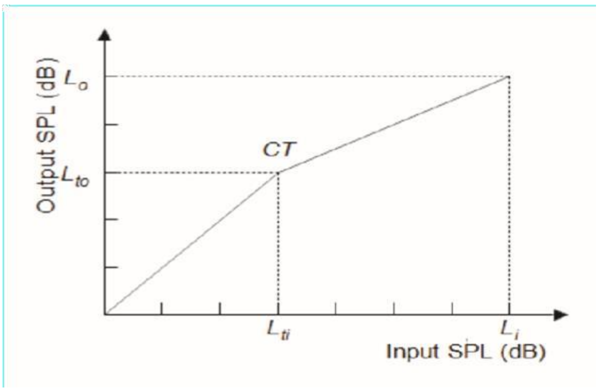

Fig 2.2: Static Compression Curve

3. Methodology

If we pass the strong or weak audio signal to this Dynamic range Compressor, then it compress the audio signal with the help of Static Compression Curve.

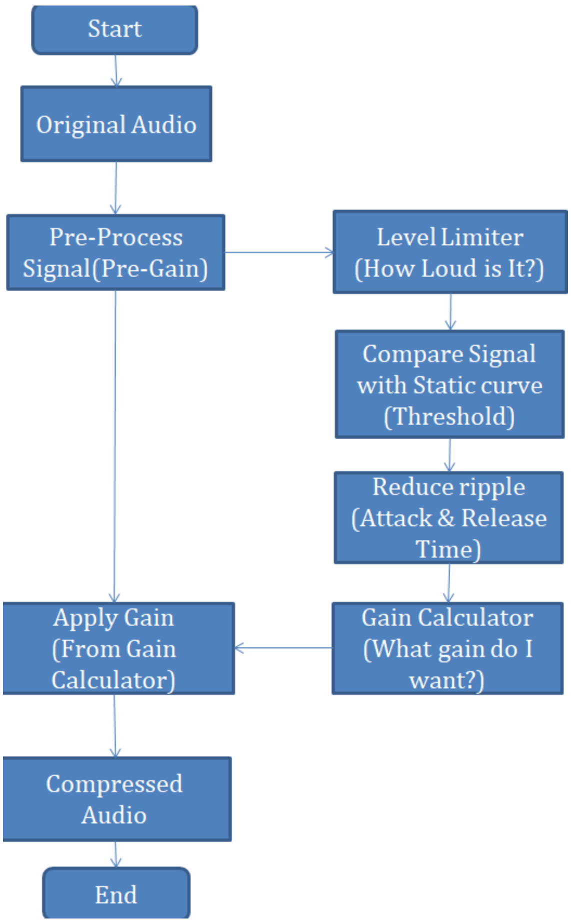

Fig no 3.1: Flow Chart 1

4. Specifications

4.1 Spartan 6 FPGA

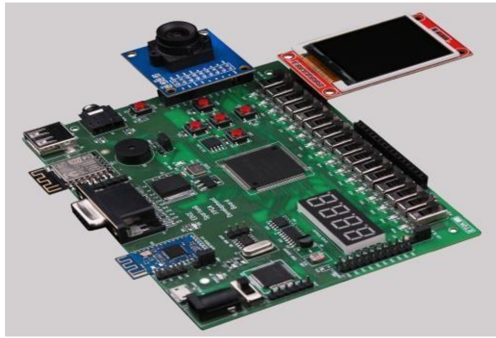

Fig 4.1: Spartan-6 FPGA

A Field-programmable gate array (**FPGA**) is an integrated circuit designed to be configured by a user or designer after manufacturing, hence it is a "Field-programmable". So Xilinx Spartan-6 FPGA offers advanced power management technology & It has on board **Codec**. Also it has Flash memory: 16 Mb SPI flash memory (M25P16), 100MHz CMOS oscillator, USB 2.0 interface for On-board flash programming, FPGA configuration via JTAG and USB.

4.2 MATLAB

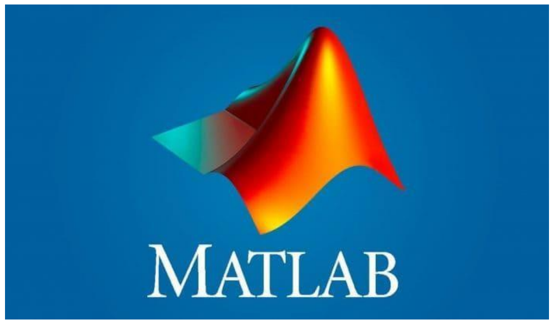

Fig 4.2: MATLAB

MATLAB is a proprietary multi-paradigm programming language and numeric computing environment developed by MathWorks. MATLAB allows matrix manipulations, plotting of functions and data, implementation of algorithms, creation of user interfaces, and interfacing with programs written in other languages.

4.3 Audio Toolbox

Audio Toolbox™ provides tools for audio processing, speech analysis, and acoustic measurement. It includes algorithms for processing audio signals such as equalization and time stretching, estimating acoustic signal metrics such as loudness and sharpness, and extracting audio features, With Audio Toolbox you can import, label, and augment audio data sets.

4.4 Signal Processing Toolbox

Signal Processing Toolbox provides functions and apps to analyze, preprocess, and extract features from uniformly and nonuniformly sampled signals. The toolbox includes tools for filter design and analysis, resampling, smoothing, detrending, and power spectrum estimation. The toolbox also provides functionality for extracting features like changepoints and envelopes, finding peaks and signal patterns, quantifying signal similarities, and performing measurements such as SNR and distortion. You can also perform modal and order analysis of vibration signals.

5. RESULT

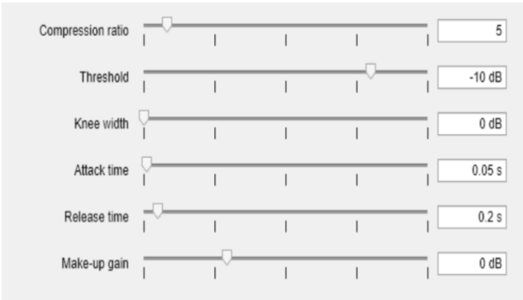

Fig 5.1: Tune parameters of the compressor

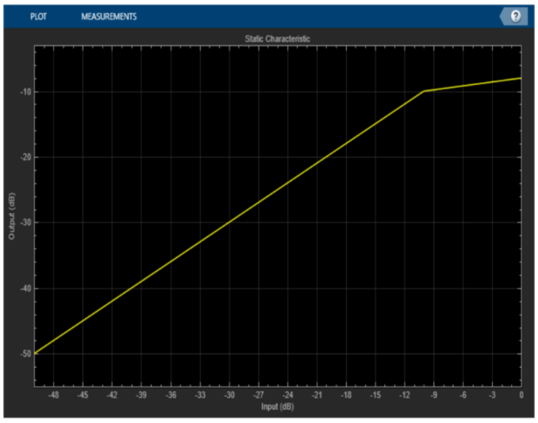

Fig 5.2: Static characteristic of the compressor

Fig 5.3: Compressed signal

6. CONCLUSION

A matlab function was used to simulate the behaviour of an audio compressor. While the compressor lacked the control commonly found on traditional software or hardware compressors. Further work could be taken to improve the compressors functionality. So, we are improving audio application by implementing Dynamic range compression algorithm on MATLAB.

7. REFERENCES

a. Journal /Article /Paper:

[1] Fan-Gang Zeng, Stephen Rebscher, William Harrison, Xiaoan Sun and HaihongFeng, "Cochlear Implants: System Design, Integration, and Evaluation", in proc. IEEE Reviews in Biomedical Engineering, vol. 1, pp. 115 – 142, 2008.

[2] P.Mahalakshmi, M. R. Reddy, "Signal Analysis by Using FIR Filter Banks in Cochlear Implant Prostheses", in proc. International Conference on Systems in Medicine and Biology, IEEE, pp. 253 - 258, December 16-18, 2010.

[3] Lunner T., Hellegren J., Arlinger S., Elberling C., "Non-Linear Signal Processing in Digital Hearing Aids" in proc. ScandAudiol IEEE , pp. 40 – 49, 1998.

[4]X. Jiang, Y. Bao., "An Design of the 16-order FIR Digital Filter Based on FPGA" in proc. International Conference on Information Science and Engineering (ICISE), IEEE, pp. 489-492, 2012.

[5] N. S. Lawand, W. Ngamkham, G. Nazarian, P. J. French, W. A. Serdijn, G. N. Gaydadjiev, J. J. Briaire and J. H. M. Frijns. "An improved system approach towards future cochlear implants", IEEE Conference on EMBS, vol. 13, pp. 5163-5166, July 2013.

[6] NikoletaGarini, Compression Technique for Digital Hearing Aids, M. tech Dissertations, University of Patras, Department of Computer Engineering and Informatics, September 2009.

[7] Eric W. Healy, Sarah E. Yoho, An algorithm to improve speech recognition in noise for hearing-impaired listeners, Acoustical Society of America, pp.3029-3038, Feb 2013.

b. Book

[1] U. Zolzer (Ed), DAFx – Digital Audio Effects. John Wiley and Sons, New York, 2002.

[2] U. Zolzer (Ed), Digital Audio Signal Processing. John Wiley and Sons, New York, 2008.