Digital Guitar Effects

Project Report

Group 17gr641

Aalborg University
Electronic Engineering and IT

This report is compiled in LaTeX. Additionally is Mathworks MATLAB, Adobe Illustrator, Inkscape, Lucidcharts.com, Altium Designer, and Xfig used to draw figures and charts.

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STUDENT REPORT

Title:
Digital Guitar Effects

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Signal processing

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Abstract:

The paper deals with the creation of different sound effects for an electric guitar on the a Digital Signal Processor. Some of these effects are the reverb, the flanger and the equalizer. The report includes a thorough explanation of each of the effects followed by the used design approach. Simulations on MATLAB were done to verify the design. All the effects have been coded in assembly for the DSP implementation. The Assembly code works with the TMS320C5515 DSP from Texas Instruments. In order to make the DSP usable on a variety of electric guitars, a preamplifier was built. All details relating to the design and the implementation of this component are included in the paper as well.

Preface

This report is composed by group 17gr641 during the 6th semester of Electronic Engineering and IT at Aalborg University. The general purpose of the report is the development and implementation of a digital guitar effects which is a part of the overall theme *Signal processing*.

For citations, the report employs the Harvard method. If citations are not present by figures or tables, these have been made by the authors of the report. Units are indicated according to the SI standard.

This project uses the Assembly language for the TMS320C5515 processor, and furthermore, the C programming standard C99.

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Glossary

dB Decibel. 3

Hz Hertz. vi

SPL Sound pressure level. 3

Chapter 1

Introduction

The directivity of a loudspeaker is a daily issue for sound events, because the loudspeaker produce sound outside the voluntary audience area. The sound from a speaker can therefore be defined as two main subject, sound for voluntary audience and sound for non voluntary audience, which is perceived the sound as noise. The following Figure 1.1 visualise the problem.

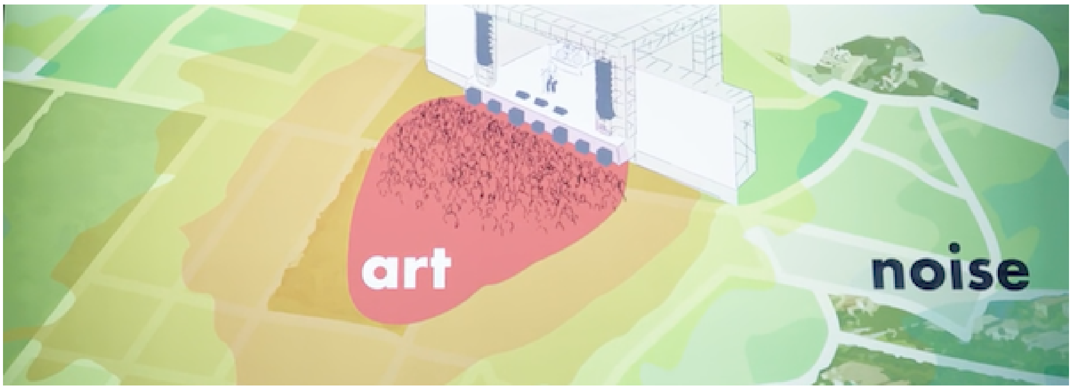

Figure 1.1: Normalized Sound pressure level (SPL) in colour, red is high SPL where blue is low SPL

Figure 1.1 shows the total sound pressure level SPL in Decibel (dB) from 20 Hz to 20 kHz for the voluntary audience and the non voluntary audience doing a concert. The voluntary audience condensing area is defined as participants area and is limited by the red area in figure Figure 1.1. The non voluntary audience will be exposed by sound in a limited frequency area, when they are located outside but still in the neighbourhood of the red area. This area is defined as the neighbourhood area. The noise is cause by omnidirectionality of the line source module and have a limited frequency range.

The directivity control of mid- and high frequency have a known solution which have been applied for many years. The solution is a horn design which is designed for a defined radiation pattern. Due to the long wave length in low/mid- and low frequency range, the horn principle will become a heavy solution for long wave length. Therefore other space saving solution have been analysed and implemented for about one decade. The commonly space saving solution is one subwoofer pointing forward and one subwoofer pointing backward with signalprocessing applied. This solution is both made by one box and two boxes.

This project aims to apply a new principle from D&B audiotechnik SL-series, where the low/mid frequency directivity is controlled by signal processing tree speaker unit. One unit arranged in the front and two arranged on the side of the line source module, one on each side.

1.1 preliminary problem statement

The following questions are made with the intention of gathering the necessary knowledge, to be able to answer a later stated problem statement. The preliminary questions, which will be answered in the analysis, are:

- In which frequency area do the line source speaker behave omnidirectional?
- Which known technique is used to do the speaker cardioid?
- Can a simulation be made which support D&B audiotechnik claim?

Part I

Analysis & Requirements

Chapter 2

Analysis of single source

2.1 Single speaker source

This section aims to introduce and analyse the fundamental for a single source, by analyse the behaviour of a line source shaped as the diaphragm. The pressure around the speaker will be analysed analytically, to determine the radiation of a single speaker from 60 Hz and upwards. The 60 Hz lower limit enable the simulation to be validated by measurement in the AAU anacoid chamber and is a used lower limit for the low/mid driver in some line source array. The analyse shall end out with a limited frequency range, where the directivity have to be controlled.

2.1.1 Pressure analysis around a single source

To characterised the directions properties of a speaker unit, the source will be modulated as three line source shaped as the diaphragm of a speaker. To modulated the diaphragm, a continues line source will be analysed and explained in this section. The analysis of a continues line source built on a thin cylindrical source of length L, and have radius a. The line source is considered as many small sources which vibrate radially, where the vibration speed is complex and modelled as

$$u = \mathcal{U}_0 \cdot exp(j\omega t) \tag{2.1}$$

Where:

	$\mathcal{U}_0$ is the Amplitude	[Ω]
	R_F is a resistor in the feedback circuit	[Ω]

?? shows the model of the speaker unit.

$$p(r,\theta,t) = \frac{j}{2} * \rho_0 c \mathcal{U}_0 \frac{a}{r} kL \left(\frac{sin(\frac{1}{2}kLsin(\theta))}{\frac{1}{2}kLsin(\theta)} \right) e^{j(\omega t - kr)} \tag{2.2}$$

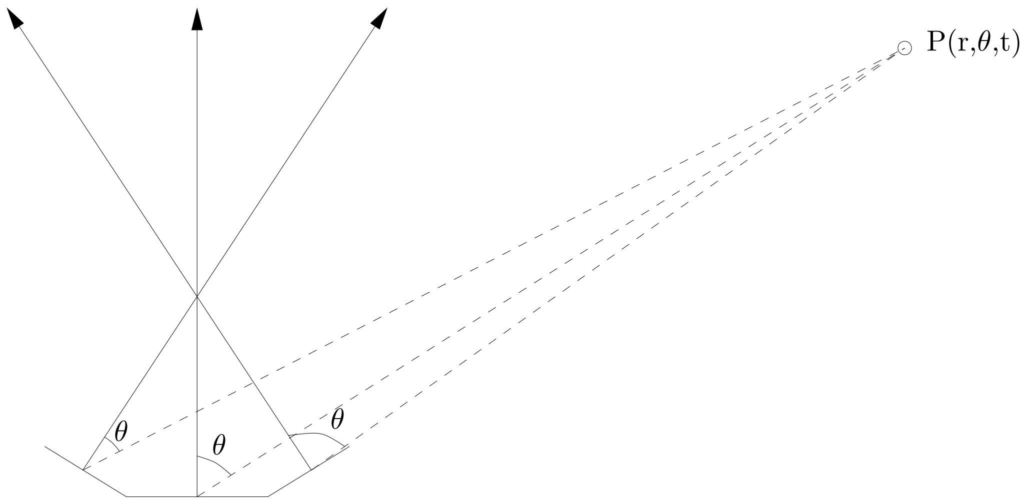

Figure 2.1: The model of a speaker unit

2.2 The 300 Hertz (Hz)

2.3 The dimension limit

Chapter 3

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Part II

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