Digital Guitar Effects

Project Report

Group 17gr641

Aalborg University
Electronic Engineering and IT

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AALBORG UNIVERSITY
STUDENT REPORT

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Digital Guitar Effects

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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

Hz Hertz. vi, 10

SPL Sound pressure level. 3, 4

Chapter 1

Introduction

The directivity of a sound sources is an issue that has an impact on many situations of our daily life, e.g. at live music venues. Voluntary listeners, namely the audience, enjoy comparatively high sound pressure levels when gathering around the stage. Non-voluntary listeners, generally neighbours, tend to perceive the sound emitted by the stage as a disturbing noise. This problem might be minimized with directivity control of the sound sources. The high sound energy will then be emits towards the audience, and less towards the neighbours.

There are several effects that make this difficult, because commonly used loudspeaker contraptions for low frequency playback tend to act like omnidirectional sound sources [Crocker, 1998, p. 1391 f.]. First, the dampening of sound in the air has significantly less influence on sound towards the lower frequency of the human hearing range than it has towards the higher frequency. Secondly, because of the way most houses are built, low frequency sound penetrates through walls and windows much more than high frequency sound. All of these effects lead to neighbours being disturbed by *the low frequency* from nearby live music venues.
The difference in the perception of the sound is visualised in Figure 1.1.

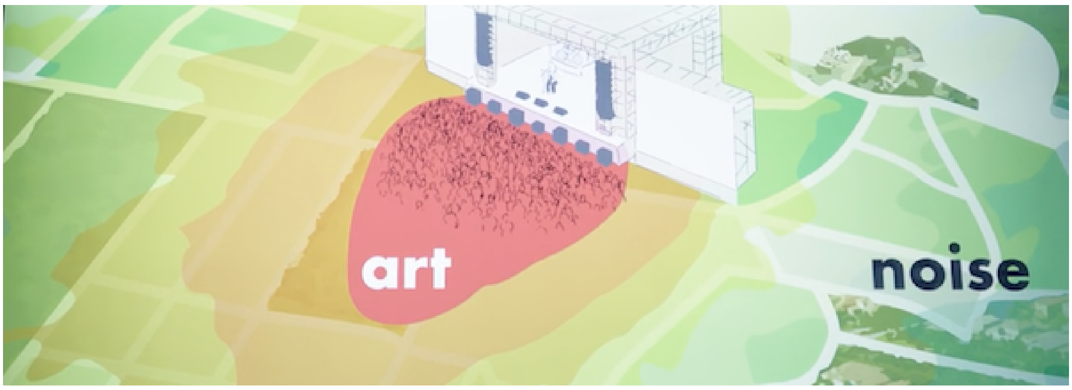

Figure 1.1: Normalized Sound pressure level (SPL) in colour, red is high SPL where blue is low SPL

Figure 1.1 shows a qualitative drawing of a near-ideal sound pressure distribution in the vicinity of a stage during a concert. The high-SPLareas are highlighted by red color, and is the area where the voluntary listeners condense. This area is define as the **participants' area** The non-voluntary listeners are located in the area around the participants' area, that we define as **the neighbourhood**.

While a SPL distribution as depicted in Figure 1.1 is easier to achieve the higher the frequency gets, towards low frequencies the SPLdistribution might look more like depicted in autoref(fig:problem2).

The directivity control of mid- and high frequency has a known solution which has been applied for many years. In general, horns are used, which are designed for a particular radiation pattern. Due to the long wave length in the low/mid- and low frequency range, the horns that are required to direct those wavelengths are not feasible for practical applications due to their size and weight. Therefore other, more space saving solutions have been developed and implemented in the last decade. It is possible to achieve a cardiod emission pattern by arranging subwoofers in a particular manner. Two or three subwoofers are pointed towards the participants' area and one subwoofer is pointed the opposite way. The signal for the subwoofer pointing away from the audience is processed to manipulate the phase.

This project aims towards applying a principle that has been put into commercial use in the D&B audiotechnik SL-series, where the low/mid frequency directivity is controlled by signal processing four speaker unit. Two units are arranged in the front of the line array module and the other two arranged on each of the sides of the line array module.

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

Problem Analysis

Chapter 2

Directional characteristics of a loudspeaker

2.1 Origin

Because this project is about shaping the directional characteristics of loudspeakers arrangements towards a particular direction, it is important to thorougly investigate the directional characteristic of a single loudspeaker cabinet. This serves as a baseline for comparison and also is essential, because the investigated cabinet will be used to form the speaker array later on.
In general, loudspeakers tend to display different directional behaviour depending on the frequency emitted. At low frequencies they can be viewed as omnidirectional sound sources. At higher frequencies the main direction of sound emission is in line with the motion direction of the voice coil. [Crocker, 1998] Depending on the ratio of the emitted wavelength to the diameter of the speaker, a radiation pattern with side lobes can occur. An analytic approximation to the behaviour can be made when looking at a vibrating piston in an infinite baffel. However, this only takes into account the front side of the speaker. It is difficult to incorporate the effects of an enclosure into this model.
There are possibilities to numerically model the sound field around a speaker in a cabinet. However in the context of this project, conducting a measurement seems to be the most favourable approach towards quantifying the sound emitted by loudspeaker in the cabinet at numerous frequencies. The knowledge gained through the measurements can then be used in order to designate a feasible frequency range for the beamforming.

Chapter 3

Analytical descriptions

3.1 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.

3.1.1 Pressure analysis around a single source

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

$$u = \mathcal{U}_0 \cdot exp(j\omega t) \tag{3.1}$$

Where:

u is the complex speed of the line source	[1]
$\mathcal{U}_0$ is the Amplitude	[1]
j is the imaginary unit	[1]
ω is the angular velocity	[1]
t is the time	[1]

Each small sources is treated as an unbaffled simple source with length dx and the source strange can be modulated as following

$$dQ = \mathcal{U}_0 2\pi a \cdot dx \tag{3.2}$$

Where:

dQ is the simple source strange [1]
$\mathcal{U}_0$ is the Amplitude [1]
a is the radius for cylinder [1]
dx is the length for the simple source [1]

The following Figure 3.1 shows an example of the continues line source where one of the small source is showed with length dx.

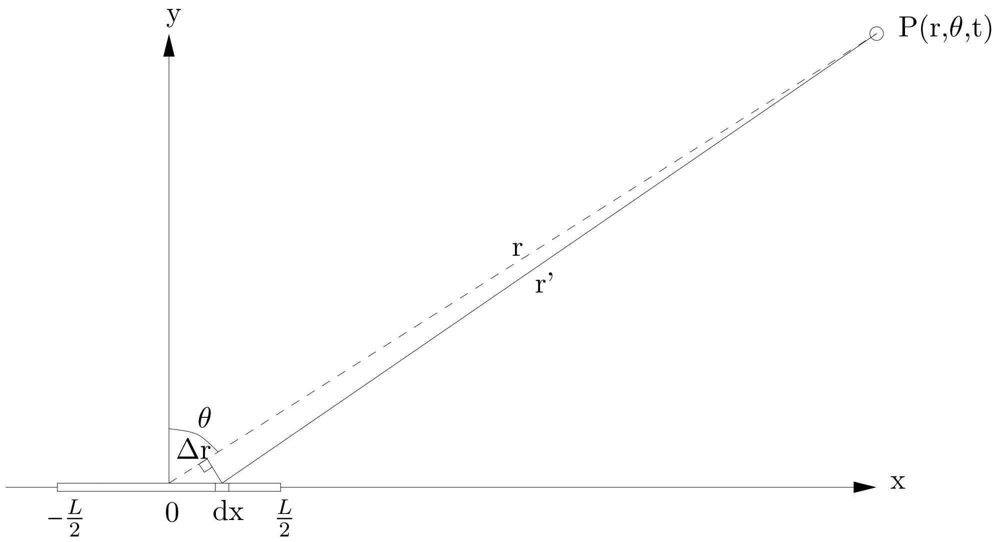

Figure 3.1: The model of a continues line source (ref the book)

Figure 3.1 shows the concept with the
?? shows the model of the speaker unit.

$$p(r,\theta,t) = \frac{j}{2}\rho_0 c\mathcal{U}_0 \frac{a}{r} kL \left(\frac{sin(\frac{1}{2}kLsin(\theta))}{\frac{1}{2}kLsin(\theta)} \right) e^{j(\omega t - kr)} \tag{3.3}$$

3.2 The 300 Hertz (Hz)

3.3 The dimension limit

Chapter 4

Numerical Simulation

Part II

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Chapter 5

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Chapter 6

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Appendix

Bibliography

Crocker, M. J. e. *Handbook of Acoustics.* John Wiley & Sons, New York, . edition, 1998. ISBN 978-0-471-25293-1.