Low/mid frequency beamforming

Project Report

Group 18gr872

Aalborg University
Electronic Engineering and IT

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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.

Aalborg University, April 18, 2018

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Part I

description of the selected scientific computing problem

Chapter 1

Numerical Simulation

1.1 The Finite-Difference Time-Domain (FDTD)

The aim of this section is to outline the basics of numerical simulation by using the FDTD method. With this method, sound propagation from a speaker array can be analysed. The principles this kind of numerical simulation will be described, such that the method can be adapted to one or more speakers in a sound field. The approach of FDTD is to solve the wave equation by a finite-difference approximation for both time and space derivatives. This makes it possible to easily simulate the sound pressure and particle velocity of a speaker at any time step. One advantage of using FDTD is that the impulse response of a specific loudspeaker can be applied to the simulation and therefore it is possible to simulate the speaker that is used in this project. For using FDTD with a specified speaker, all simulations have to be done in a narrow frequency band, in order for the simulation to give a good approximation. The FDTD can not include the whole hearing band in a single simulation. A second advantage of using FDTD is that the calculation is performed in time domain, which benefits from that the pressure and the particle velocity at a specified time step can be analyzed directly by solving two coupled equation.[C. Kleinhenrick and Karhe, 2009]. This section will end out with a FDTD model of a 3 dimensional space, where the speaker box is set up not to scatter the simulation.

1.1.1 FDTD wave equation

In FDTD simulation, there are two equations, which need to be solved. The first formula is the Euler Equation 1.1, which describes the relation between the gradient for the pressure p and the derivative of the particle velocity $\vec{v}$ with respect to time.

$$\frac{\partial \vec{v}}{\partial t} = -\frac{1}{\rho}\vec{\nabla} p \tag{1.1}$$

Where:

ρ is the density of the medium	[kg/m^3]
∂t is an infinitesimal time step	[s]
p is the pressure	[Pa]
$\vec{v}$ is the particle velocity	[m/s]

The Equation 1.1 is only valid with small variation of pressure. The second Equation 1.2 is the linear continuity equation. The equation describes the relation between the derivative of the pressure p with respect of time and the velocity gradient $\triangledown\vec{v}$. They are related trough the density of the medium and the speed of sound.

$$\frac{\partial p}{\partial t} = -\rho c^2 \vec{\triangledown}\vec{v} \tag{1.2}$$

Where:

ρ is the density of the medium	[kg/m^3]
∂t is an infinitesimal time step	[s]
p is the pressure	[Pa]
c is the speed of sound	[m/s]
$\vec{v}$ is the particle velocity	[m/s]

By use of the derivative, both equations are approximated linearly at every point in a three dimensional cartesian grid. This is done with discrete time steps.

1.1.2 FDTD using Cartesian grid

Using a Cartesian grid for FDTD approximation is a well known technique (see [Botteldoore, 1995]) and will be used in this project. The Cartesian grid is set up using the sound pressure Equation 1.2 and the particle velocity Equation 1.2 as the unknown quantities, which have to be solved for in every point in space. A small grid is visualized in Figure 1.4

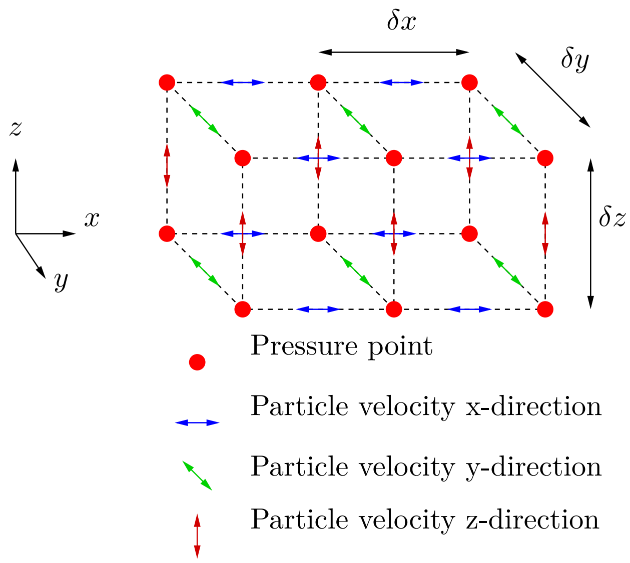

Figure 1.1: A 3 dimensional example of a Cartesian grid.

The grid points are built of positions that are described as $(i\,\delta x, j\,\delta y, k\,\delta z)$ at a time $t = [l]\delta t$. The time step is visualized in Figure 1.2.

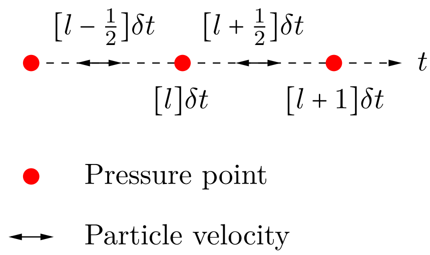

Figure 1.2: Transient definition points of sound p pressure and particle velocity $\vec{v}$

$\delta x, \delta y, \delta z$ are the spatial discretization steps as shown in Figure 1.4 and δt is the time spatial discretization step as shown in Figure 1.2. i, j, k are the discrete indices for the points in the grid and l is the discrete time index. For every axis, the corresponding particle velocity has to be determined at a position in Equation 1.3 at an intermediant time $t = [l \pm \frac{1}{2}]$.

$$\vec{v} = \begin{bmatrix} v_x\big[(i \pm \frac{1}{2})\,\delta x, j\,\delta y, k\,\delta z\big] \\ v_y\big[i\,\delta x, (j \pm \frac{1}{2})\,\delta y, k\,\delta z\big] \\ v_z\big[i\,\delta x, j\,\delta y, (k \pm \frac{1}{2})\,\delta z\big] \end{bmatrix} \tag{1.3}$$

The pressure is determined at position $p^{[l+1]}_{(\mathrm{i,j,k})}$. It can be chosen to start with either pressure or velocity arbitrarily. The time step δt can be regarded as a scaling factor for time, because MATLAB and Python only work with integer indices. This means

δt is implemented in the formulas and not in the iteration step. The time $\pm\frac{1}{2}$ is also changed to a integer with adding $\frac{1}{2}$. This scalar is only relevant in the implementation and is disregarded in the rest of this section. The same applies for the step sizes δx, δy and δz.

Substituting **??** into **??** and solving for $(v_x)^{[l+\frac{1}{2}]}_{(\mathrm{i}+\frac{1}{2},\mathrm{j},\mathrm{k})}$ leads to following three Equations 1.4:

$$(v_x)^{[l+\frac{1}{2}]}_{(\mathrm{i}+\frac{1}{2},\mathrm{j},\mathrm{k})} = (v_x)^{[l-\frac{1}{2}]}_{(\mathrm{i}+\frac{1}{2},\mathrm{j},\mathrm{k})} - \frac{\delta t}{\rho_0 \delta x}\left(p^{[l]}_{(\mathrm{i}+1,\mathrm{j},\mathrm{k})} - p^{[l]}_{(\mathrm{i},\mathrm{j},\mathrm{k})}\right) \tag{1.4a}$$

$$(v_y)^{[l+\frac{1}{2}]}_{(\mathrm{i},\mathrm{j}+\frac{1}{2},\mathrm{k})} = (v_y)^{[l-\frac{1}{2}]}_{(\mathrm{i},\mathrm{j}+\frac{1}{2},\mathrm{k})} - \frac{\delta t}{\rho_0 \delta y}\left(p^{[l]}_{(\mathrm{i},\mathrm{j}+1,\mathrm{k})} - p^{[l]}_{(\mathrm{i},\mathrm{j},\mathrm{k})}\right) \tag{1.4b}$$

$$(v_z)^{[l+\frac{1}{2}]}_{(\mathrm{i},\mathrm{j},\mathrm{k}+\frac{1}{2})} = (v_z)^{[l-\frac{1}{2}]}_{(\mathrm{i},\mathrm{j},\mathrm{k}+\frac{1}{2})} - \frac{\delta t}{\rho_0 \delta z}\left(p^{[l]}_{(\mathrm{i},\mathrm{j},\mathrm{k}+1)} - p^{[l]}_{(\mathrm{i},\mathrm{j},\mathrm{k})}\right) \tag{1.4c}$$

Substituting **??** intro **??** and solve for $p^{[l+1]}_{(\mathrm{i},\mathrm{j},\mathrm{k})}$ leads to Equation 1.5

$$p^{[l+1]}_{(\mathrm{i},\mathrm{j},\mathrm{k})} = p^{[l]}_{(\mathrm{i},\mathrm{j},\mathrm{k})} - \rho_0 c^2 \delta t \Bigg(\frac{(v_x)^{[l+\frac{1}{2}]}_{(\mathrm{i}+\frac{1}{2},\mathrm{j},\mathrm{k})} - (v_x)^{[l+\frac{1}{2}]}_{(\mathrm{i}-\frac{1}{2},\mathrm{j},\mathrm{k})}}{\delta x} + \frac{(v_y)^{[l+\frac{1}{2}]}_{(\mathrm{i},\mathrm{j}+\frac{1}{2},\mathrm{k})} - (v_y)^{[l+\frac{1}{2}]}_{(\mathrm{i},\mathrm{j}-\frac{1}{2},\mathrm{k})}}{\delta y} + \frac{(v_z)^{[l+\frac{1}{2}]}_{(\mathrm{i},\mathrm{j},\mathrm{k}+\frac{1}{2})} - (v_z)^{[l+\frac{1}{2}]}_{(\mathrm{i},\mathrm{j},\mathrm{k}-\frac{1}{2})}}{\delta z} \Bigg) \tag{1.5}$$

1.1.3 FDTD grid boundary conditions

The meaning of boundary conditions in the given context is the condition near and at the boundary surfaces (walls), that occur at the ends of the grid. The sound waves react differently on the walls opposed to propagation in free field conditions. Wall will act like either a reflecting surface, an absorbing surface or both. This boundary behaviour from the wall is described as an frequency dependent impedance. It is necessary to analyse and implement this in the simulation because the sound field will show a different behaviour compared to sound field without any boundaries. Within this project, the frequency dependent boundary conditions will only be a good approximation and not accurately represent free field conditions. An accurate frequency dependent model would require heavy calculations with convolution at each boundary point and at each time step [Botteldoore, 1995]. This kind of calculation has a high time consumption and therefore an approximation will be used.
The approximation in this project will be based on the impedance approach (see [Jeong and Lam, 2000]) as described above. The impedance approach can be used

when walls is present in simulation and does not contain a perfect matched layer. Therefore the impedance approach can not be used for free field simulation unless the simulation is stopped just before the wave hits the boundary. Because of the way the pressure spreads along the grid, the stopped simulation will lead to some areas, typically every corner, to which the wave has not yet spread. This results in a circular shape on the simulation grid if the sound sources are placed in the middle. In this project a large room is used and the simulation will be stopped just before the boundary to simulate free field condition. Afterwards the data are cropped such that only simulating data within the area to which the sound had already spread are used.

The impedance approach is usable at low frequency, meaning in a frequency range, in which the sound sources behave approximately omnidirectional [Jeong and Lam, 2000]. Two kinds of absorbing boundaries are common in real life and therefore also in simulation. These boundaries are as follows:

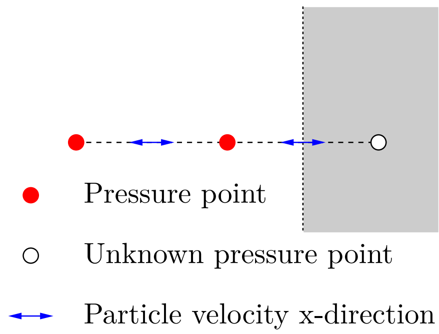

Figure 1.3: The figure visualized a boundary plan through particle velocity plan x in 1 dimension.

For solving the problem visualized in Figure 1.3, an asymmetric finite-difference approximation for the space derivative is used [Botteldoore, 1995]. **??** shows the asymmetric finite-difference approximation.

$$\frac{\partial p}{\partial x}\Big|^{[l]}_{(\mathrm{i}_0+\frac{1}{2},\mathrm{j},\mathrm{k})} = \frac{2}{\delta x}\left(p^{[l]}_{(\mathrm{i}_0+\frac{1}{2},\mathrm{j},\mathrm{k})} - p^{[l]}_{(\mathrm{i}_0,\mathrm{j},\mathrm{k})}\right) \tag{1.6}$$

The advantage of Equation 1.6 is that it only requires knowledge of one nearest pressure point, but it is only valid within δx. **??** can then be used to express **??** as function of v_x. Using the same procedure as in Equation 1.4 just with plugging in Equation 1.6 instead, the particle velocity at the boundary is approximated as Equation 1.7

$$(v_x)^{[l+\frac{1}{2}]}_{(\mathrm{i}_0+\frac{1}{2},\mathrm{j},\mathrm{k})} = (v_x)^{[l-\frac{1}{2}]}_{(\mathrm{i}_0+\frac{1}{2},\mathrm{j},\mathrm{k})} - \frac{2\delta t}{\rho_0 \delta x}\left(p^{[l]}_{(\mathrm{i}_0+\frac{1}{2},\mathrm{j},\mathrm{k})} - p^{[l]}_{(\mathrm{i}_0,\mathrm{j},\mathrm{k})}\right) \tag{1.7}$$

The only unknown in Equation 1.7 is $p^{[l]}_{(\mathrm{i}_0+\frac{1}{2},\mathrm{j},\mathrm{k})}$, but it can be found by using **??**, where v_n is changed with $(v_x)^{[l]}_{(\mathrm{i}_0+\frac{1}{2},\mathrm{j},\mathrm{k})}$ and becomes Equation 1.8.

$$(v_x)^{[l+\frac{1}{2}]}_{(\mathrm{i}_0+\frac{1}{2},\mathrm{j},\mathrm{k})} = (v_x)^{[l-\frac{1}{2}]}_{(\mathrm{i}_0+\frac{1}{2},\mathrm{j},\mathrm{k})} - \frac{2\delta t}{\rho_0 \delta x}\Bigg(Z_0 (v_x)^{[l]}_{(\mathrm{i}_0+\frac{1}{2},\mathrm{j},\mathrm{k})} + Z_1 \frac{\partial (v_x)^{[l]}_{(\mathrm{i}_0+\frac{1}{2},\mathrm{j},\mathrm{k})}}{\partial t} + Z_{-1}\int_{-\infty}^{t} (v_x)^{[l]}_{(\mathrm{i}_0+\frac{1}{2},\mathrm{j},\mathrm{k})}(\tau)d\tau - p^{[l]}_{(\mathrm{i}_0,\mathrm{j},\mathrm{k})}\Bigg) \tag{1.8}$$

The integral in the **??** is replaced with a sum from minus infinity to l in Equation 1.9.

$$(v_x)^{[l+\frac{1}{2}]}_{(\mathrm{i}_0+\frac{1}{2},\mathrm{j},\mathrm{k})} = (v_x)^{[l-\frac{1}{2}]}_{(\mathrm{i}_0+\frac{1}{2},\mathrm{j},\mathrm{k})} - \frac{2\delta t}{\rho_0 \delta x}\Bigg(Z_0 (v_x)^{[l]}_{(\mathrm{i}_0+\frac{1}{2},\mathrm{j},\mathrm{k})} + Z_1 \frac{(v_x)^{[l+\frac{1}{2}]}_{(\mathrm{i}+\frac{1}{2},\mathrm{j},\mathrm{k})} - (v_x)^{[l-\frac{1}{2}]}_{(\mathrm{i}+\frac{1}{2},\mathrm{j},\mathrm{k})}}{\delta t} + Z_{-1}\delta t \sum_{\mathrm{m}=-\infty}^{l}\left((v_x)^{[m+\frac{1}{2}]}_{(\mathrm{i}+\frac{1}{2},\mathrm{j},\mathrm{k})}\right) - p^{[l]}_{(\mathrm{i},\mathrm{j},\mathrm{k})}\Bigg) \tag{1.9}$$

The last unknown variable is the particle velocity v_x at time $t = [l]$. To find a solution for v_x at time $t = [l]$, a linear interpolation between v_x at time $t = [l \pm \frac{1}{2}]$ is used [Botteldoore, 1995]. The resulting particle velocity will be expressed as Equation 1.10

$$(v_x)^{[l+\frac{1}{2}]}_{(\mathrm{i}_0+\frac{1}{2},\mathrm{j},\mathrm{k})} = \alpha (v_x)^{[l-\frac{1}{2}]}_{(\mathrm{i}_0+\frac{1}{2},\mathrm{j},\mathrm{k})} + \beta \frac{2\delta t}{\rho_0 \delta x}\Bigg(Z_0 (v_x)^{[l]}_{(\mathrm{i}_0+\frac{1}{2},\mathrm{j},\mathrm{k})} - Z_{-1}\delta t \sum_{\mathrm{m}=-\infty}^{l}\left((v_x)^{[m+\frac{1}{2}]}_{(\mathrm{i}+\frac{1}{2},\mathrm{j},\mathrm{k})}\right) - p^{[l]}_{(\mathrm{i},\mathrm{j},\mathrm{k})}\Bigg) \tag{1.10}$$

Where:

$\alpha = \dfrac{1-\frac{Z_0}{Z_{\mathrm{FDTD}}}\frac{2Z_1}{Z_{\mathrm{FDTD}}}\delta t}{1+\frac{Z_0}{Z_{\mathrm{FDTD}}}\frac{2Z_1}{Z_{\mathrm{FDTD}}}\delta t}$ [1]

$\beta = \dfrac{1}{1-\frac{Z_0}{Z_{\mathrm{FDTD}}}\frac{2Z_1}{Z_{\mathrm{FDTD}}}\delta t}$ [1]

$Z_{\mathrm{FDTD}} = \frac{\rho_0 \delta x}{\delta t}$ [Nsm^{-3}]

1.1.4 FDTD grid cell size

The choice of grid cell size for FDTD is a critical parameter, which must fulfill some problem specific constraint [Kunz and Luebbers, 1993]. The grid cell size has to be small enough to contain data for all specified simulated frequencies, which means that the grid cell size has to be smaller than the smallest wavelength λ. As the frequency rises the wave length is decreasing. This means the grid cell size constrain is determined by the highest frequency of interest in the FDTD simulation. Opposed to that, the grid cell size should be relatively large, in order to keep the computational power down, that is required to run the simulation. The grid cell size therefore has to be chosen intelligently, for which [Kunz and Luebbers, 1993] displays a solution. After the grid cell size is chosen the Courant stability condition determines the maximum time step. The maximum time step size, which will be calculated based on the grid cell size, will be the used. A smaller time step size does not improve the accuracy in general.

The boundary for the smallest grid cell size is the Nyquist rate, which states that the wavelength shall at least be twice as big as the grid cell size δ. Since δx, δy and δz have the same size only δ will be used to represent the grid step size. The Nyquist rate is the lower boundary, but since the simulation is an approximation and is not exact and the smallest wavelength is not precise, δ has to be more than two samples per wavelength. To find a optimal grid size the grid dispersion error which relates to the wave propagation speed through the grid will be taken intro account. The error occurs because the wave propagates with different slightly speeds through the grid. This error is depending on the relative direction of the wave. The grid dispersion error is propertional to the grid cell size, which means that the error is decreased with smaller δ[Kunz and Luebbers, 1993]. Often if $\delta \leq \frac{1}{10}\lambda_{\text{min}}$ the formerly mentioned constraint is met. The solution therefore a good compromise between computation resource and approximation error. The grid cell size in this project is therefore set according to Equation 1.11.

$$\delta x = \delta y = \delta z \leq \frac{1}{10}\frac{c}{f_{\text{max}}} \tag{1.11}$$

Where:

δ is the grid cell size	[1]
x, y and z is the direction	[1]
c is the speed of sound	[m/s]
f_{max} is the maximum frequency in the simulation	[Hz]

1.1.5 FDTD time step size and stability

The time step size for FDTD follows from the Courant condition [Kunz and Luebbers, 1993]. The aim of the project is not to analyse the condition of Courant. This section will therefore only give a short overview on the most important aspects of the condition and on how to use the condition to calculate the time step size. Considering a plane wave, the Courant condition states that, in one time step, any point on the wave must not pass through more than one cell. During one time step the wave can propagate only from one cell to its nearest neighbors [Kunz and Luebbers, 1993]. To determine the time step size the time step δt can therefore be determined by the speed of sound and the grid cell size as in Equation 1.12.

$$\delta t \leq \frac{1}{\sqrt{\frac{1}{(\delta x)^2} + \frac{1}{(\delta x)^2} + \frac{1}{(\delta x)^2}} \cdot c} \tag{1.12}$$

Where:

δ is the step size	[1]
t is the time indicator	[s]
c is the speed of sound	[m/s]

Making the time step size smaller than stated by Equation 1.12 will not improve the result, in fact the equation calculates the time step size where the grid dispersion error is minimal [Kunz and Luebbers, 1993]. Unless the dispersion error is minimized, the time step might even be smaller because of the stability condition. A stable simulation is only guaranteed under certain conditions. Because Equation 1.10 is applied to different conditions e.g. in corner or flat walls, a stable simulation is not possible in general, only under certain conditions which depend on time and grid cell size. It has been shown, that the simulation is stable if Z_0 and Z_1 are positive for all simulation regions and if Equation 1.13 is satisfied [Botteldoore, 1995].

$$\delta t \leq \sqrt{\frac{2}{3}} \left(\frac{1}{\sqrt{\frac{1}{(\delta x)^2} + \frac{1}{(\delta x)^2} + \frac{1}{(\delta x)^2}} \cdot c} \right) \tag{1.13}$$

If Z_{-1} is nonzero the time step shall furthermore satisfy Equation 1.14

$$c\delta t \leq \delta x \left(\frac{1 + \frac{2Z_1}{\rho_0 \delta x}}{1 + \frac{2Z_{-1}\delta x}{\rho_0 c^2}} \right)^{\frac{1}{2}} \tag{1.14}$$

1.1.6 FDTD sound source

As described in **??**, the acoustical center of the loudspeaker is about 17 cm in the front of the speaker cabinet. It also has to be noted that the FDTD sound source is at the acoustical center and not at the position of the loudspeaker itself. Therefore the FDTD sound source is modelled as a transparent source. The speaker cabinet may have a different effect when building a speaker array in the real world, but not at the position of the sound source in FDTD. Therefore the speaker cabinet will not be included in the FDTD sound source model. The following section will shortly explain the three most common way of implementing sources in [Saarelma, 2013] and will explain the use of a transparent source.

There are two simple methods to implement a FDTD sound source and one more advanced way to implement a FDTD sound source. The simple ways of implementing at sound source are the hard- and the soft sources. The problem with implementing a hard source is, that the hard source overwrites the update step in the source point and therefore effectively scatters any incident field. This might correspond to a real scenario if the speaker cabinet was at the acoustical center and very reflective, but it is not and therefore this kind of source is not suitable in the given context. Secondly, the soft source is set up in a way, that the pressure from the source is added to the pressure source point, which means, that this source does not scatter. The problem with this method is that the actual excitation does not match the time function of the source. To make a source that acts like a hard source but does not scatter, the transparent source is used according to [John B. Schneider and Broschat, 1997]. The transparent source will briefly be explained in this section.

A transparent source is achieved by measuring the impulse response I of the grid and using it in Equation 1.15.

$$p^{[l+1]}_{(\mathrm{i_s},\mathrm{j_s},\mathrm{k_s})} = p^{[l]}_{(\mathrm{i_s},\mathrm{j_s},\mathrm{k_s})} - \rho_0 c^2 \delta t \left(\frac{(v_x)^{[l+\frac{1}{2}]}_{(\mathrm{i_s}+\frac{1}{2},\mathrm{j_s},\mathrm{k_s})} - (v_x)^{[l+\frac{1}{2}]}_{(\mathrm{i_s}-\frac{1}{2},\mathrm{j_s},\mathrm{k_s})}}{\delta x} + \frac{(v_y)^{[l+\frac{1}{2}]}_{(\mathrm{i_s},\mathrm{j_s}+\frac{1}{2},\mathrm{k_s})} - (v_y)^{[l+\frac{1}{2}]}_{(\mathrm{i_s},\mathrm{j_s}-\frac{1}{2},\mathrm{k_s})}}{\delta y} + \frac{(v_z)^{[l+\frac{1}{2}]}_{(\mathrm{i_s},\mathrm{j_s},\mathrm{k_s}+\frac{1}{2})} - (v_z)^{[l+\frac{1}{2}]}_{(\mathrm{i_s},\mathrm{j_s},\mathrm{k_s}-\frac{1}{2})}}{\delta z} \right) + f^{l+1} - \sum_{\mathrm{m}=0}^{l} \left(I^{l-m+1} f^m \right) \tag{1.15}$$

Where:

(i_s, j_s, k_s) are the source grid positions	[1]
I is the impulse response of the grid	[1]

As it can be seen in Equation 1.15, the source is implemented like a soft source but with a correction part $-\sum_{\mathrm{m}=0}^{l} \left(I^{l-m+1} f^m \right)$. The correction part includes the impulse response of the FDTD grid and has to be measured. To measure the impulse response of the grid, a Kronecker delta function is used as the sound source with unity gain. The Kronecker delta source emits an impulse with unity gain only at time one, and zero otherwise. The Kronecker delta source is implemented as a hard source [John B. Schneider and Broschat, 1997].

The impulse response is measured at the source point. As stated before, the Kronecker source is implemented as a hard source. When measuring at the source point, it is therefore to be expected, that the measurement behaves as the Kronecker source signal that is sent out, because of the hard implementation of the source. To get around this, the measured pressure is not actually recorded from the source point, but calculated back from the particular velocity surrounding the source. The impulse response measurement function is therefore described by Equation 1.16:

$$I^{[l]} = p^{[l-1]}_{(\mathrm{i_s},\mathrm{j_s},\mathrm{k_s})} - \rho_0 c^2 \delta t \left(\frac{(v_x)^{[l-\frac{1}{2}]}_{(\mathrm{i_s}+\frac{1}{2},\mathrm{j_s},\mathrm{k_s})} - (v_x)^{[l-\frac{1}{2}]}_{(\mathrm{i_s}-\frac{1}{2},\mathrm{j_s},\mathrm{k_s})}}{\delta x} + \frac{(v_y)^{[l-\frac{1}{2}]}_{(\mathrm{i_s},\mathrm{j_s}+\frac{1}{2},\mathrm{k_s})} - (v_y)^{[l-\frac{1}{2}]}_{(\mathrm{i_s},\mathrm{j_s}-\frac{1}{2},\mathrm{k_s})}}{\delta y} + \frac{(v_z)^{[l-\frac{1}{2}]}_{(\mathrm{i_s},\mathrm{j_s},\mathrm{k_s}+\frac{1}{2})} - (v_z)^{[l-\frac{1}{2}]}_{(\mathrm{i_s},\mathrm{j_s},\mathrm{k_s}-\frac{1}{2})}}{\delta z} \right) \tag{1.16}$$

The Kronecker delta source will be placed in the middle of the grid, and there is only one Kronecker delta source. The particle velocity matrices for each dimension have therefore the following symmetrical $(v_x)^{[l-\frac{1}{2}]}_{(\mathrm{i_s}-\frac{1}{2},\mathrm{j_s},\mathrm{k_s})} = -(v_x)^{[l-\frac{1}{2}]}_{(\mathrm{i_s}+\frac{1}{2},\mathrm{j_s},\mathrm{k_s})}$. The impulse response measurement function can therefore be shortened to Equation 1.17:

$$I^{[l]} = p^{[l-1]}_{(\mathrm{i_s},\mathrm{j_s},\mathrm{k_s})} - 2\rho_0 c^2 \delta t \left(\frac{(v_x)^{[l-\frac{1}{2}]}_{(\mathrm{i_s}+\frac{1}{2},\mathrm{j_s},\mathrm{k_s})}}{\delta x} + \frac{(v_y)^{[l-\frac{1}{2}]}_{(\mathrm{i_s},\mathrm{j_s}+\frac{1}{2},\mathrm{k_s})}}{\delta y} + \frac{(v_z)^{[l-\frac{1}{2}]}_{(\mathrm{i_s},\mathrm{j_s},\mathrm{k_s}+\frac{1}{2})}}{\delta z} \right) \tag{1.17}$$

There is a stability condition that has to be satisfied in order to make the transparent source stable in FDTD simulation. The stability condition is Equation 1.18 [John B. Schneider and Broschat, 1997].

$$\frac{c\delta t}{\delta d} \leq \frac{1}{\sqrt{N}} \tag{1.18}$$

Where:

N is the number of dimensions [1]
δd is the smallest grid step of δx, δy and δz [1]

1.2 FDTD simulation

The aim of this section is to make a simulation of the zero order gradient speaker, first order gradient speaker and the solution proposed in **??** using FDTD. All simulation across all three different methods of using speakers at low frequency will be compared in both free field and inside a room. The result will be discussed with respect to directionality and efficiency.

1.2.1 FDTD step size

This section will determine the the step size for both grid size and the time step size. Because the time step size is depending on the grid step size, the grid step size will be determined first. Recalling Equation 1.11 states, that all three dimention will have the same step size in this project. The calculation is as follows inEquation 1.19.

$$\delta d = \delta x = \delta y = \delta z \leq \frac{1}{10}\frac{c}{f_{\max}} \tag{1.19a}$$

$$\delta d = \frac{1}{10}\frac{343\,\mathrm{m/s}}{300\,\mathrm{Hz}} \tag{1.19b}$$

$$\delta d = 0.11\,\mathrm{m} \tag{1.19c}$$

Where:

deltad is the maximum grid cell size	[m]
c is the speed of sound at a temperature of 20 ° °C	[m/s]
$f_{\max}$ is the maximum frequency in the simulation	[Hz]

Since the maximum grid cell size is 11 cm, the grid cell size is in this project will be less or equal to 10 cm, to be sure that a simulation does not suffer from a limiting grid cell size.

From the optimization **??** it is determined that the optimal distance between the side speaker is 40 cm and to the front speaker it is 40 cm. When arranging the sources in a triangular shape, the greatest common divisor for putting them into a grid is 20 cm. However this is bigger then than the formerly stated 11 cm, which makes it unsuitable for simulation. Instead, it is decided to set the grid step size at 5 cm in all simulations, to ensure flexibility during development. Next the time step has to be determined. The condition for the time step is stated in Equation 1.13 and dictates the following Equation 1.20.

$$\delta t \leq \sqrt{\frac{2}{3}}\left(\frac{1}{\sqrt{\frac{1}{(\delta x)^2}+\frac{1}{(\delta x)^2}+\frac{1}{(\delta x)^2}}\cdot c}\right) \tag{1.20a}$$

$$\delta t \leq \sqrt{\frac{2}{3}}\left(\frac{1}{\sqrt{\frac{1}{(0.05\,\mathrm{m})^2}+\frac{1}{(0.05\,\mathrm{m})^2}+\frac{1}{(0.05\,\mathrm{m})^2}}\cdot 343\,\mathrm{m/s}}\right) \tag{1.20b}$$

$$\delta t \leq 687\,\mathrm{\mu s} \tag{1.20c}$$

This corresponds to a sampling frequency of 14 552 Hz. To be sure that a simulation does not suffer from a limiting time step size, the sampling frequency will be rounded up to 14 560 Hz.

1.2.2 FDTD source

The acoustic centers of the speakers will be represented as transparent sources, since a speaker cabinet FDTD model will not be incorporated in this project. The source has a stability condition which has to be fulfilled and this includes the number of dimensions. In section 1.1 the FDTD is prepared to dimensions, but all simulations in this section will only be in two-dimensional, since the speakers are not offset in the third dimension. The stability condition is stated in Equation 1.18 and the stability achieved as shown in Equation 1.21.

$$\frac{c\delta t}{\delta d} \leq \frac{1}{\sqrt{N}} \tag{1.21a}$$

$$\frac{343 \cdot 687\,\mu\text{s}}{5\,\text{cm}} \leq \frac{1}{\sqrt{2}} \tag{1.21b}$$

$$0.471 \leq 0.707 \tag{1.21c}$$

As shown in Equation 1.21 the transparent sound source is stable with the calculated time and grid size.

1.2.3 FDTD implementation

The aim of this section is to convert the grid to matrices. Since the grid is in three dimensions, where the time is a fourth dimension, the whole matrix system for pressure and all particle velocity matrices will built on four dimensional matrices. The following Figure 1.4 shows a simple grid in a corner with entry notation of h as the hight of the room, w as the width of the room and l as the length of the room.

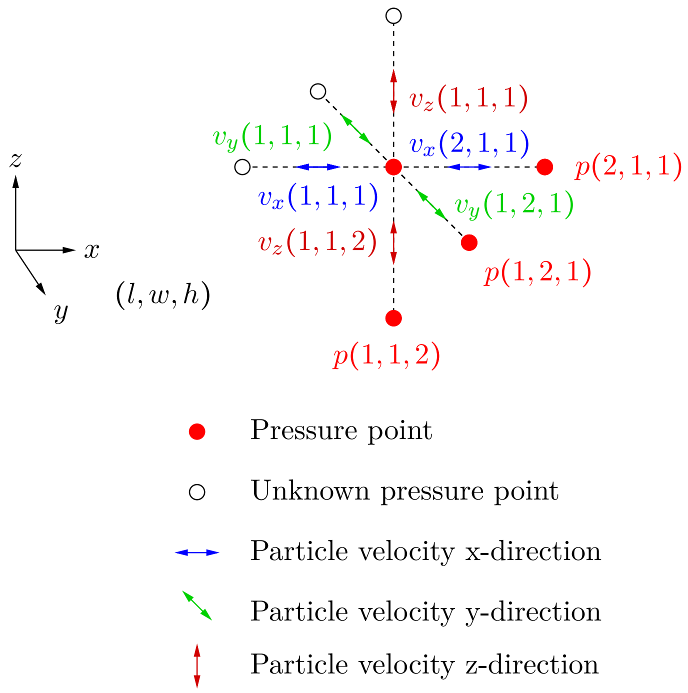

Figure 1.4: The figure visualized a boundary plan through particle velocity plan x in 1 dimension

All simulation is implemented in MATLAB. The x direction is implemented as matrix row direction and y direction is implemented as matrix column direction. The height h is implemented as the third dimension and the time is implemented as the fourth dimension. The particle velocity will be calculated as the first step and the pressure will be calculated as the second step. The time dimension is only as small as 2 pages, because it is only necessary to save at $[l-1]$ and at $[l]$. A page is one dimension in a matrix system, where the number before ´´page" describes the time in this context and not a two dimension page. There can be many dimensions for pages, where the implementation of the FDTD has two page dimensions. All time steps further away than $[l-1]$ are not used in the calculation and will be deleted for keeping the storage requirements as low as possible.

1.2.4 FDTD boundary

The particle velocity at the boundary in all direction has to be calculated as in Equation 1.10 and this is done as a step between the particle velocity matrices and the pressure matrices. This means, that for the particle velocity in x direction the first and last matrix row are calculated by the boundary formula and in y direction the first column and the last matrix column is calculated by the boundary formula.

1.2.5 FDTD plot

The aim of the section is to explain how the phase, gain and distance, resulting from **??**, are implemented in the FDTD simulation, and how the result of the simulation is compared with the polar plot from **??**.

To be able to simulate the result from **??**, three pressure points is used as the acoustical center. The pressure points are implemented as transparent point sources, where all the point sources is fed with a sinusoid of a single frequency where the phase and gain is adjusted to the specified frequency. The front speakers will always be feeded with a gain of one, which correspond to a pressure of ±1 Pa and zero phase. Therefore it is only the back speaker, which will be adjusted in phase and gain. The number of simulation run steps is the smallest room size divided by the grid size, such that the waves do not reflect from the wall.

A wave will expand in a circular form and since the simulation stops just before the it hits the first wall, the simulation result will be a circular shape. Some of the area is not simulated and is therefore without a valid pressure. Because of this, the simulation is made so large that the polar plot with a radius of 10 m can still be calculated, where the used simulation data is only a squared area inside the circular shape and the rest data is discarded.

The polar plot from the analytical model is based on the Root Mean Square (RMS) pressure, therefore the RMS pressure of the FDTD has to be calculated. The RMS of the FDTD simulation is calculated as following in Equation 1.22.

$$P_{\text{rms}}(i,j,k) = \sqrt{\frac{\left(p_{(\text{i,j,k})}^{[l=\delta t]}\right))^2 + \left(p_{(\text{i,j,k})}^{[l=2\delta t]}\right)^2 + \ldots + \left(p_{(\text{i,j,k})}^{[l=n\delta t]}\right)^2}{n}} \tag{1.22}$$

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