PROCEEDINGS of the
23rd International Congress on Acoustics
9 to 13 September 2019 in Aachen, Germany

Study of DIF boundary model with rectilinear FDTD scheme in voice booths

Vito ROMANELLI TRICANICO(1), Marcel BORIN(2), Carolina MONTEIRO(3), Rânnely ARAÚJO(4)

(1)Harmonia Acústica, Brazil, vito.romanelli@harmoniaacustica.com.br

(2)Harmonia Acústica, Brazil, marcel.borin@harmoniaacustica.com.br

(3)Harmonia Acústica, Brazil, carolina.monteiro@harmoniaacustica.com.br

(4)Harmonia Acústica, Brazil, ranny.araujo@harmoniaacustica.com.br

Abstract

Computational simulations are helpful when studying acoustical properties of rooms. This represents a challenge when very small rooms, such as office video/phone booths, are taken into account. The volume of these rooms generally does not exceed 15 m^3 and the first modes occur at higher frequencies if compared to larger rooms. Computational room acoustics considers two main techniques of modelling sound propagation: geometrical and wave-based. As geometrical modelling is not the most suitable approach for low frequencies since the phase performs an important part in the physics at this frequency range, in the present study, a rectilinear FDTD scheme routine was implemented to obtain the impulse responses for low frequency bands of a voice booth, using proper boundary conditions derived from digital impedance filters (DIF). In order to obtain broadband impulse responses, high frequency impulse responses were acquired using a commercial software based on geometrical methods. Both responses were later filtered with low-pass and high-pass filters based on Schroeder frequency, and then assembled. Finally, the results extracted from the impulse responses of simulations were compared with data obtained from field measurements.

Keywords: Sound, Computational, Boundary

1 INTRODUCTION

The need to perform computational simulations to obtain acoustic room parameters is evident even when there are pre and post *in situ* measurements in the room to be treated. They can predict room configurations without the need of moving to the field. In Brazil, rooms such as voice booths have been increasingly used in offices in order for the user to obtain a higher level of privacy, to make small meetings or calls. These rooms generally do not exceed 15 m^3 in volume, and Schroeder's frequency occurs at higher frequencies when compared to larger rooms, implying that booth has modal behavior in a broader band of frequencies. The modal frequencies end up reaching the speech bands, and there is still no consensus on how to measure room parameters in such small volume. In [1], a method is presented to measure the frequency response of this type of room, obtaining alike curves between several voice booths with different configurations and dimensions, performed by Harmonia Acústica [1]. Because of this behavior, this type of room is ideal for studying wave-based methods, such as the finite-difference time-domain method (FDTD) [2], finite volumes and finite elements. Ray-based methods are not the most suitable for this type of room because they suppress phenomena such as diffraction at low frequencies, not adapting the room's response, which can lead to large inaccuracies in the values generated by the simulation. The FDTD has its advantages by being easily implementable, impulse responses obtained are already in time-domain, and it has a DIY aproach, whereas other methods need a few other software available for creating meshes, and ABCs based on parameters other than the coefficient of absorption, such as impedance, often difficult to achieve for specific materials.

In 2008, Kowalczyk [3] presented boundary conditions for the FDTD method based on digital impedance filters

[1] www.harmoniaacustica.com.br

(DIF), which is obtained from the absorption coefficient of the boundary, taking into account important physical considerations, but presenting only numerical results. In 2017, Reuben [4] showed that the model has a good application for computational simulations, but also no tests have been done to attest the use of the program in comparison to *in situ* measurements. An advantage of this method however is that there is no need to do the simulation for each frequency band since the modelled boundary is independent of the frequency, decreasing the execution time intensively.

Although the model has shown convergent solutions for some simulated room configurations, it is desired to study the applicability of this boundary model, comparing results obtained from simulations and measurements. The intention is to find a point of similarity of parameters obtained with the simulation and with the measurements, presenting a mutual validation.

2 OBJECTIVES

The goals of this study are to compare measurements and simulation results of $1/3$ octave band frequency responses and T_{30} of the voice booth, in order to assess the applicability of the DIF boundary model in an rectilinear FDTD grid for very small rooms.

3 METHODOLOGY

The process for obtaining the impulse responses presented a certain level of complexity amidst the simplicities presented a priori. The method of measurements is also presented in this section.

3.1 The wave equation and FDTD method

After doing a discretization of the undamped homogeneous wave-equation [2, 5], and some algebra manipulation, one may get that the update equation for the sound pressure $p(x,y,z,t)$ in a rectinilear 3D FDTD grid is

$$p_{i,j,k}^{n+1} = \frac{1}{3} S_{i,j,k}^{n} - p_{i,j,k}^{n-1} , \tag{1}$$

where

$$S_{i,j,k}^{n} = p_{i+1,j,k}^{n} + p_{i-1,j,k}^{n} + p_{i,j+1,k}^{n} + p_{i,j-1,k}^{n} + p_{i,j,k+1}^{n} + p_{i,j,k-1}^{n} , \tag{2}$$

with i, j and k denoting spatial indices for x, y and z directions and n denoting the time-step index. The Courant number, a constraint given by $\lambda = c\frac{T}{X}$, where T is the time-step in seconds, X is grid spacing in meters and c is sound speed in air, was chosen to be $\lambda = \frac{1}{\sqrt{3}}$ to get to Equation (1) in its form. This value satisfies the condition to minimize the dispersion error with the rectilinear grid topology [3].

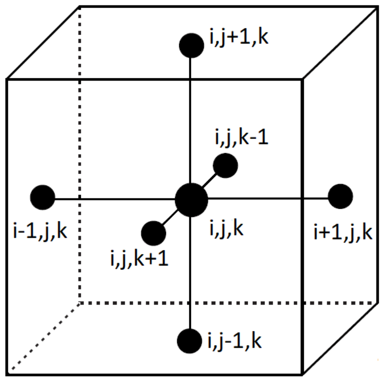

Figure 1. Rectinilear topology for the grid

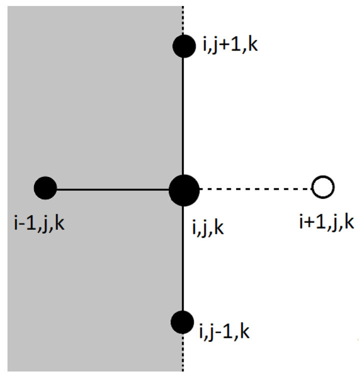

Figure 2. Ghost node for x_+ boundary

Different topologies would require Equation (1) to be more complicated. The rectilinear topology for the 3D FDTD grid can be described by interconnected nodes, presented in Figure 1.

3.2 Boundary model

The boundaries must be considered as locally reacting surfaces (LRS), i.e., the normal component of the particle velocity at the surface of the boundary depends on the sound pressure in front of the boundary element and not on pressure in front of neighbouring elements [3]. This holds for boundaries that do not propagate vibrations in the direction parallel to the boundary surface. For a boundary in the right, i.e., in the x_+ direction, it is necessary that the ghost node - the node that is out of simulation bounds - in position $i+1$ is eliminated in equation (1), as illustrated in Figure 2. The nodes of sound pressure in positions $i,j,k-1$ and $i,j,k+1$ are suppressed in the figure. For a pressure-centered approximation, Kowalczyk [3] showed that the equation for the right boundary is

$$p^n_{i+1,j,k} = \frac{\sqrt{3}}{\xi_\omega}(p^{n-1}_{i,j,k} - p^{n+1}_{i,j,k}) + p^n_{i-1,j,k}\,, \tag{3}$$

where ξ_ω is the specific acoustic impedance of the boundary. This approach may cause solution growth during execution depending on the form of the modelled room [6]. To reverse this problem, the velocity-centered approximation is used:

$$p^n_{i+1,j,k} = \frac{\sqrt{3}}{2\xi_\omega}(p^{n-1}_{i,j,k} - p^{n+1}_{i,j,k}) + p^n_{i,j,k}\,. \tag{4}$$

For normal incidence in the boundary (which is the case for locally reacting surfaces), ξ_ω may be written as

$$\xi_\omega(z) = \frac{1+R_0(z)}{1-R_0(z)}\,, \tag{5}$$

where $R_0(z)$ is the boundary reflection coefficient at normal incidence [3, 6], and $|R_0(z)| = \sqrt{1-\alpha}$, with α being the sound absorption coefficient of the boundary. Therefore, ξ_ω is expressed as an arbitrary magnitude IIR digital filter with transfer function given by

$$\xi_\omega(z) = \frac{b_0 + b_1 z^{-1} + b_2 z^{-2} + \ldots + b_N z^{-N}}{a_0 + a_1 z^{-1} + a_2 z^{-2} + \ldots + a_N z^{-N}}\,, \tag{6}$$

where each z^{-m} represents a delay of m samples in a multiplied discrete variable, by the time shifting property of the Z-transform.

Finally, substituting (4) in (1), and using (2) and (6), the final update equation for the right boundary is

$$p_{i,j,k}^{n+1} = \left[\frac{1}{3} \sum_{m=1}^{N} b_m (p_{i,j,k}^{n-m+1} + S_{i,j,k}^{n-m+1} - p_{i+1,j,k}^{n-m+1}) + \dots \right.$$
$$+ \sum_{m=1}^{N} p_{i,j,k}^{n-m} \left(\frac{\sqrt{3}}{6} a_m - b_m \right) + \dots$$
$$\left. - \sum_{m=1}^{N} p_{i,j,k}^{n-m+2} \left(\frac{\sqrt{3}}{6} a_m + b_m \right) \right] \div \left(\frac{\sqrt{3}}{6} a_1 + b_1 \right) . \quad (7)$$

To get to (7), the transformation $m \to m+1$ is needed so that the sums start at 1 because the chosen programming language has no zero-based arrays. For the designed digital filter coefficients happens that $a_m \to a_{m-1}$ and $b_m \to b_{m-1}$ in equation (6), e.g., the first coefficients in the sums, a_1 and b_1 will actually be a_0 and b_0 in (6), respectively.

The purpose of writing Equation (7) in three distinct parts is because they have different constraints over the sums, so that the *for* loops iterations have variables with time index greater or equal than 1. The constraints are given by

$$n \geqslant m \, , \quad (8a)$$
$$n \geqslant m+1 \, , \quad (8b)$$
$$n \geqslant m-1 \quad \wedge \quad m \geqslant 2 \, , \quad (8c)$$

for first, second and third sum in (7), respectivelly.

Equations for other boundaries are obtained analogously, but the subtracted term in the first sum would be the ghost node to be eliminated for that boundary, e.g., for a boundary in negative x direction (x_-), the first part of the update equation becomes

$$\frac{1}{3} \sum_{m=1}^{N} b_m (p_{i,j,k}^{n-m+1} + S_{i,j,k}^{n-m+1} - p_{i-1,j,k}^{n-m+1}) \, , \quad (9)$$

and if the boundary material is different, the digital filter coefficients values a_m and b_m would also be different.

The equation for a surface boundary in the form of (7) has a straightforward approach, but equations for boundaries such as edges and corners can get very complicated, so the applied technique was to use an arithmetic sum over the filter coefficients of each direction. As an example, for a (x_+, y_+, z_+) corner the first sum for the adopted update equation was

$$\frac{1}{3} \sum_{m=1}^{N} (b_{xm} + b_{ym} + b_{zm})[3p_{i,j,k}^{n-m+1} + S_{i,j,k}^{n-m+1} - (p_{i+1,j,k}^{n-m+1} + p_{i,j+1,k}^{n-m+1} + p_{i,j,k+1}^{n-m+1})] \, , \quad (10)$$

and for an (x_+, y_+) edge was

$$\frac{1}{3} \sum_{m=1}^{N} (b_{xm} + b_{ym})[2p_{i,j,k}^{n-m+1} + S_{i,j,k}^{n-m+1} - (p_{i+1,j,k}^{n-m+1} + p_{i,j+1,k}^{n-m+1})] \, . \quad (11)$$

3.3 Grid excitation and constraints

The chosen pulse injection for the grid was a simple unit impulse hard source [7] in a position above the table, although hard sources can cause some distortion in the results for lower frequencies. This kind of pulse is known to have a flat response and excite all frequencies in the grid.

In general, the highest valid frequency of the simulation must be half the sampling frequency, given by $f_N = \frac{1}{T}$. In the case of the rectilinear FDTD scheme, it needs to be even lower [3, 4]: $0.196 f_N$. Thus, in the routine was adopted that the Schroeder frequency of the room, $f_S \approx 500$ Hz, must satisfy that $f_S \leqslant 0.196 f_N$, so the grid spacing X must be chosen with care, since $X = \sqrt{3}cT$. Since $c = 345.1$ m/s and adopted grid spacing was $X = 0.055$ m, then $f_N \approx 10868$ Hz.

Figure 3. Measurements in the voice booth

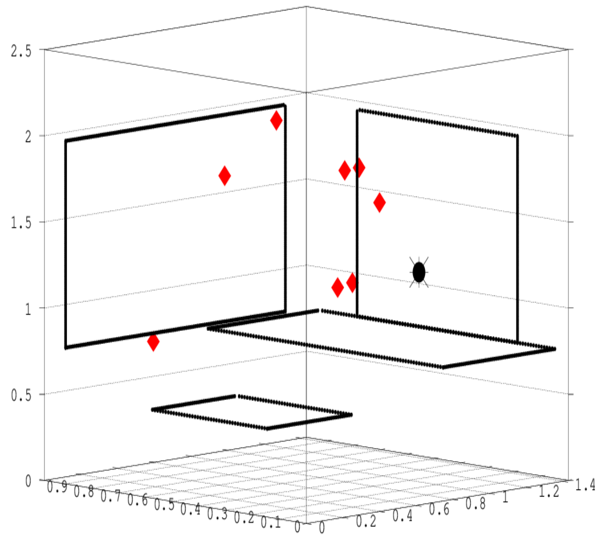

Figure 4. Graphical model of the voice booth (in meters) and the 7 chosen points

3.4 Chosen language and tools

In order to test the model of digital impedance filters for the voice booth, a routine was created using the Torch [2] framework, which has Lua [3] as a base programming language, using LuaJIT compiler. Lua was chosen because of its syntax simplicity and speed, and Torch has multidimensional tensors, making implementation a bit easier.

The boundary conditions were imposed with conditionals for the space. No visual tools or CAD programs were used to aid in the room modelling.

The arbitrary magnitude IIR digital filter coefficients can be obtained from a function with the same purpose in *Matlab®* [4] (function *fdesign.arbmag*) or *GNU/Octave* [5] (function *fir2* from the *signal* package). In order to get the correct coefficients, it is needed that the filter is modeled in function of the reflection coefficients [6] for each frequency, and the filter order adjusted for the required precision, fitting the resulting curve to the

[2] http://torch.ch/
[3] https://www.lua.org/
[4] https://www.mathworks.com/products/matlab.html
[5] https://www.gnu.org/software/octave/

reflection coefficient curve for the boundary, then using the resulting coefficients.

For both functions, there is a need to adjust the coefficients of the IIR filter modelled for the boundary. If the chosen function for modelling the filter is *fir2*, for example, there will only be numerator coefficients for R_0 since *fir2* returns an arbitrary magnitude FIR filter. So, if $R_0(z) = r_1 + r_2 z^{-1} + ... + r_N z^{N-1}$, from (5), results that $b_1 = 1 + r_1$, $a_1 = 1 - r_1$ and so on.

3.5 Field measurements

The measurements for the studied voice booth are described by Araújo, [8], as it is written below.

In order to evaluate the energy decay of the booth, the reverberation time (RT) was measured *in situ* for each one of the setups presented in [8]. These measurements followed the procedure from the standard ISO 3382-2:2012 [9].

The 1/3 octave band frequency response of the room was also measured for all setups using the sound power level of the pink noise from 50 Hz to 20 kHz characterized according to ISO 3745:2012 [10] and emitted from an omnidirectional sound source positioned in one of the room corners. The measurement followed the methodology for voice booths presented by Monteiro and Borin [1].

4 RESULTS AND DISCUSSION

The result of the FDTD routine execution is the sound pressure for every node in the modelled voice booth in setup G presentend in [8], for every time sample. The room impulse responses (RIR) of 7 points distributed in the modelled room were extracted from this field - check Figure 4. The RIRs for the same points were also obtained from a commercial software that uses geometrical acoustics.

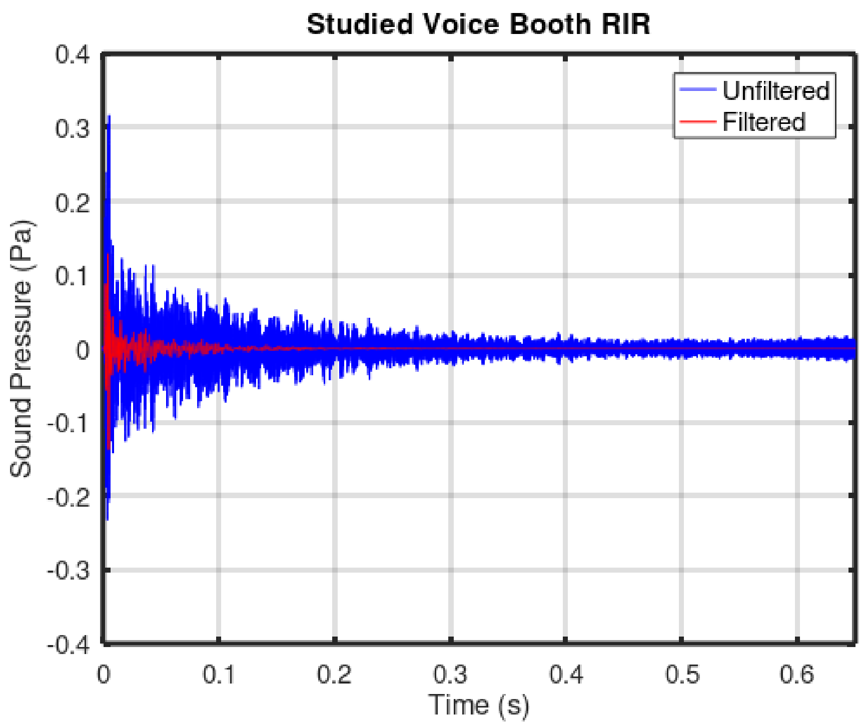

Figure 5. FDTD RIR for position [x y z] = [0.74 0.39 1.14] m, before and after filtering

After this process, RIRs obtained with FDTD and geometrical acoustics were filtered at 500Hz - a low-pass filter was used for FDTD and a high-pass for geometrical acoustics. Then, by doing a Fast Fourier Transform (FFT) in the filtered impulse responses, and transforming the values to $1/3$ octave bands, the frequency responses were assembled.

For assembling the frequency responses, a scaling was needed in order to adjust magnitudes for the lower and higher frequencies. This happens because the impulse response obtained with geometrical acoustics is normalized from -1 to 1, and the FDTD results are in Pascals. The scaling factor was chosen so that the 1/3 octave band frequency response of the room was acceptable.

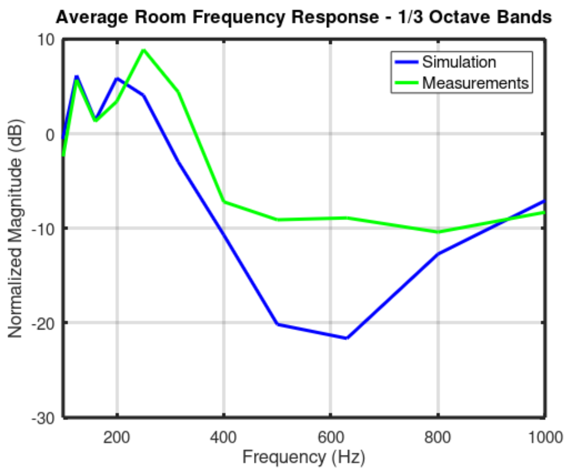

Figure 6. Comparison of frequency response of the voice booth

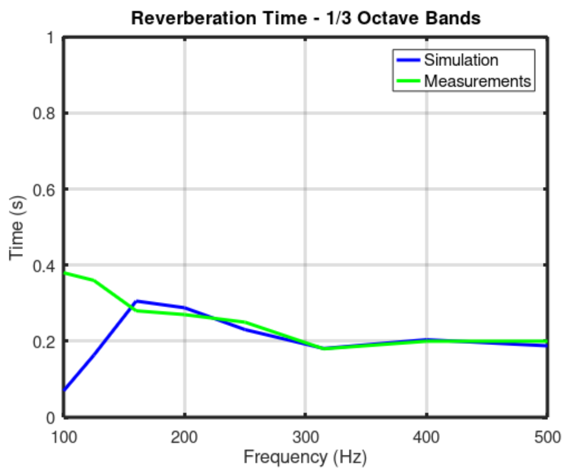

Figure 7. Comparison of reverberation time of the voice booth

Moreover, for the simulation values, a level of 135 dB was subtracted from the original 1/3 frequency response, which is the approximate sound power level of the source in the FDTD routine. As it can be noted, the response is fitted from 100 Hz to 160 Hz if compared to the measurements, and suffer some distortion 160 Hz onwards. However, up to Schroeder frequency, the curve for the simulation follows the general behavior of the curve obtained from measurements. Other causes for the distortions may be the equations chosen for edges and corners, or differently chosen absorption coefficients for the materials used in the voice booth, since not all the materials had this values registered.

The reverberation times were obtained for the low frequencies by filtering the RIRs obtained with the FDTD method with band-pass filters for each band. Thereafter, the results were obtained by calculating a backward cumulative integral [11] curve for the magnitude of the filtered RIRs, an then fitting a line to this curve for first -30dB decay, in order to get the T_{30}. This time, results extracted from simulation are very near the measurements, except for lower frequencies, as it can be seen in Figure 7.

5 CONCLUSIONS

As it was seen, the boundary model using digital impedance filters for the rectilinear FDTD grid showed acceptable results for the presented equations, despite differences in some frequency bands and in the equations for edges and corners. It showed a general similarity with the results obtained from measurements. Also, the method for measurements in the voice booth may or may not be the most suitable for this type of room, despite the similar results with the simulation, still needing more investigations [1]. The results for frequencies lower than 100 Hz were not studied because at the time of the simulations, there was no information of absorption coefficients at these bands for the materials used in the voice booth. Further investigations about the scaling factor

[12] need to be carried out. This factor was chosen in order to make the magnitude of the results obtained by simulations as close as possible to the ones presented by the measurements, because the RIRs obtained with the FDTD simulations are in pascals (Pa), and the ones obtained with geometrical acoustics are normalized from -1 to 1. The reverberation time values obtained from the simulated RIRs showed to be very near the values obtained by measurements up to the Schroeder frequency, which is the interest region of study for this method. Execution time of simulation was very slow in gerenal (approximately 2 day of calculations), and in the future it will be investigated the possibility of paralelizing the code with CUDA® [6] [4, 13] for shortening execution time.

REFERENCES

[1] C. Monteiro and M. Borin, "Acoustic quality evaluation of voice booths using 1/3rd octave band frequency response," *Inter-Noise*, no. October, 2018.

[2] N. J. Giordano and H. Nakanishi, *Computational Physics*. Pearson/Prentice Hall, 2006.

[3] K. Kowalczyk, *Boundary and medium modelling using compact finite difference schemes in simulations of room acoustics for audio and architectural design applications*. PhD thesis, 2008.

[4] M. R. Thomas, *Wayverb: A Graphical Tool for Hybrid Room Acoustics Simulation*. PhD thesis, 2017.

[5] C. A. B. Kinsler L E, Frey A R and S. J. V, *Fundamentals of Acoustics 4th edn (New York: Wiley)*. 2000.

[6] S. Oxnard, *Efficient Hybrid Virtual Room Acoustic Modelling*. PhD thesis, 2016.

[7] D. T. Murphy, A. Southern, and L. Savioja, "Source excitation strategies for obtaining impulse responses in finite difference time domain room acoustics simulation," *Applied Acoustics*, vol. 82, aug 2014.

[8] R. Araújo, "Room acoustic design of voice booths in open plan offices," *Inter-Noise*, no. June, 2019.

[9] "ISO 3382:2008 - Acoustics – Measurement of room acoustic parameters – Part 2: Reverberation time in ordinary rooms.," standard, International Organization for Standardization, 2008.

[10] "ISO 3745:2012 - Acoustics – Determination of sound power levels and sound energy levels of noise sources using sound pressure – Precision methods for anechoic rooms and hemianechoic rooms.," standard, International Organization for Standardization, 2012.

[11] E. Brandão, *Acústica de Salas: Projeto e Modelagem*. Edgard Blücher, 2016.

[12] Marc Aretz, René Nöthen, Michael Vorländer, Dirk Schröder, "Combined Broadband Impulse Responses Using FEM and Hybrid Ray-Based Methods," *EAA Symposium on Auralization*, no. March 2015, 2009.

[13] C. J. Webb and S. Bilbao, "Computing room acoustics with cuda - 3d fdtd schemes with boundary losses and viscosity," in *2011 IEEE International Conference on Acoustics, Speech and Signal Processing (ICASSP)*, May 2011.

[6] https://developer.nvidia.com/cuda-zone