Procedural 3D Audio for AR Applications

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Abstract

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THird paragraph of abstract

Four paragraphs is enough I guess

Acknowledgements

We would like to express our gratitude and appreciation to our supervisor for his support and guidance throughout this thesis work. Several discussion sessions and advice helped us take the most out of this project and make this study possible.

We would like to express special thanks also to the rest of our classmates who did their thesis at the same time under the same supervisor and offered us their advice. Furthermore, we would like to thank **To do** name of the recording guy (1) for providing us with the acoustics room to perform recordings. And last but not least to our family and friends whose support throughout this thesis was invaluable.

List of Figures

List of Tables

Contents

Abbreviations

ASW	Apparent Source Width.
AVIL	Audio Visual Immersion Lab.
BRIR	Binaural Room Impulse Response.
CS	Compressive Sensing.
DOA	Direction of Arrival.
ERB	Equivalent Rectangular Band.
HATS	Head And Torso Simulator.
HOA	Higher Order Ambisonics.
HRTF	Head-Related Transfer Function.
IACC	Inter-Aural Cross Coherence.
ILD	Inter-Aural Level Difference.
ITD	Inter-Aural Time Difference.
STFT	Short-Time Fourier Transform.
WFS	Wave Field Synthesis.

Nomenclature

$\mathbf{\Omega}_{LS}$ Vector containing directions of Loudspeakers in reproduction.

$\mathbf{\Omega}_L$ Grid of directions used for the CS algorithm.

$\mathbf{\Omega}_s$ Subvector of $\mathbf{\Omega}_L$ containing only the prominent directions after CS processing.

$\breve{\mathbf{H}}$ Combined transfer matrix for mixed-norm problem.

$\breve{\mathbf{p}}$ Combined measurement pressure vector for mixed-norm problem.

$\mathbf{x}$ Combined amplitude.

ℓ_p Norm-p.

$\mathbf{H}$ Transfer Matrix for plane waves impinging on rigid sphere.

$\mathbf{p}$ Measurement vector for the pressure on the spherical array.

$\mathbf{x}$ Amplitude vector for plane waves impinging on the sphere.

$\widetilde{\mathbf{p}}$ Pressure vector reconstructed from prominent plane waves.

B_n^m Ambisonics coefficients.

L Number of plane waves in a discrete grid of directions.

LS Number of Loudspeakers in reproduction.

N Truncation order for the spherical Harmonic Functions.

P_n^m The associated Legendre polynomials of the first kind.

Q Number of sampling points on the spherical microphone array.

R_0 Radius of reproduction area.

Y_n^m Spherical harmonic Functions.

Ω Angular Dependency on both azimuth and inclination angle.

λ Regularization factor for natural field HOA processing.

$\mathbf{B}_N$ Ambisonics coefficients vector truncated at order N.

$\mathbf{S}$ Loudspeaker signals resulting from HOA decoding.

$\mathbf{W}$ Vector containing radial functions W_n.

$\mathbf{Y}_N(\mathbf{\Omega}_L)$ Spherical harmonics vector truncated at order N for all measurement angles in vector $\mathbf{\Omega}_L$.

$\mathbf{p}'$ Residual pressure.

ε Noise parameter for Compressive Sensing Algorithm.

a Radius of microphone array.

"w/ Residual" Exploiting the residual pressure (full implementation of signal path in Figure **??**).

"w/o Residual" Residual pressure is neglected (only upper path in Figure **??**).

Chapter 1

Introduction

Immersion and all these stuff that makes our thing good. Why we are doing it and what do we want to give to the community?

Audio in interactive projects like video games and VR/AR applications, plays a significant role for user immersion and realism. Visual and acoustic experiences are interconnected and lacking one of them spoils the whole experience.

The most difficult task is to produce realistic virtual sounds inside the application, difficult to distinguish them from the real ones. This can be achieved not only by playing back a realistic sound, but also by taking care of the environment effects and the context. For example, striking a nail on a board when it still vibrates from the previous struct, produces a different sound that gets added to the previous one [4].

Why is our method better that others? (eg wavetable)? And why we think this is the future of the audio in video games?

Chapter 2

Theoretical Background

Short overview of the theory parts
This is a way to link to explanations Direction of Arrival (DOA)

THis is a todo: **To do** todo test (2)
THis is smth done:

2.1 State-Of-The-Art

Wildes and Richards (1988) defined the angle of internal friction (ϕ), a shape-invariant acoustical parameter that heuristically categorizes sounds into material categories, as

$$\tan(\phi) = \alpha/\pi f, \tag{2.1}$$

where $\alpha = \tau_e$ is the damping coefficient, with τ_e being the time for the vibration amplitude to decay to $1/e$ of its original value after the object is struck and f is the vibration frequency[7].

2.2 Modal Analysis

In this thesis we are using solid objects that are struck in different ways to produce sound. These ways could be falling on the floor or colliding with another object. The sounds produced can be impact, rolling or scratching sounds. When an object is struck, the forces applied cause deformations to it, emitting sound waves through the vibration of its outer surfaces [13].

Modal analysis studies the response of models under excitation. It uses the 3D model of an object to calculate its modal modes (vibration modes). There are multiple ways to do this, with the most accurate being FEM (Finite Element Method). The objective of FEM is to calculate the natural frequencies of a structure when it vibrates freely.

Another method for modal analysis is the “Exampled-guided”, where data get extracted using example recordings of the objects being struck. Using a suitable algorithm it is easy to extract features from the recordings such as the fundamental frequency and its harmonics and the frequency peaks of the signal.

2.2.1 Data Extraction

Modal analysis is performed before modal synthesis, to extract the necessary data. Modal synthesis is the sum of damped oscillators each corresponding to a modal frequency,

Symbol	Description	Derivation
A_n	Initial amplitude	Modal analysis
d_n	Damping	Material properties
f_n	Modal frequency	Modal analysis

Table 2.1: Data extracted in modal analysis.

as it will be discussed further below. The data needed for synthesis are shown in the table 2.1.

Since every different point being struck produces different deformations on the object, we need matrices of size N (N being the number of struck points of the object). More specifically, we need a vector **f** of size **N** corresponding to the modal frequencies of every point, a vector **d** of size **N** corresponding to the damping ratios and a matrix **A** of size **NxK**, where K is the number of modal frequencies calculated in one point, which corresponds to the amplitudes of each mode in every point of the object. All the above gives the modal model which can be symbolized as $\boldsymbol{M = \{f,\ d,\ A\}}$ [13].

2.3 Modal Synthesis

In the modal synthesis part, using the data extracted above, we synthesize the struck sound corresponding to the object. There are different ways to synthesize impact sounds, two of them being "Sinusoidal Additive Synthesis" and "Filter-based Modal Synthesis". The former uses exponential damping and the latter band-pass filters where the damping is the Q-factor of the filter.

2.3.1 Sinusoidal Additive Synthesis

At a struck point k when vibrating in mode n, the impulse response of the model is:

$$y_k = \sum_{n=1}^{N} A_{nk}\ e^{-d_n t}\ \cos(2\pi f_n t) \tag{2.2}$$

if $t > 0$ and $y_k = 0$ if $t <= 0$ [13].

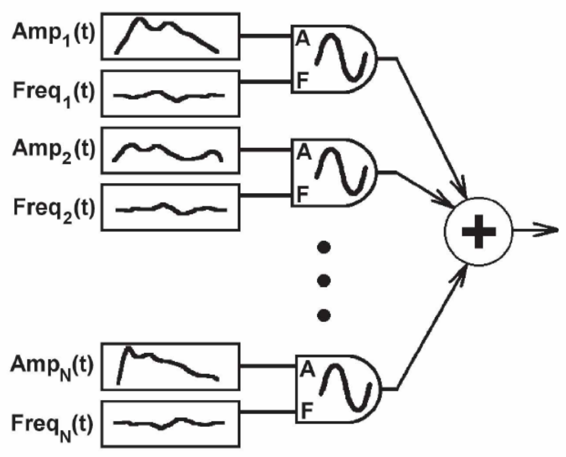


Figure 2.1: Sinusoidal Additive Synthesis Algorithm [4].

2.3.2 Filter-based Modal Synthesis

Band-pass Filters

At this point we will give some basic description of the band-pass filter since it is widely used in this thesis. Band-pass filters (BPFs) take a signal as input and give only a range of it as output, attenuating the rest of the frequencies. This range depends on the central frequency f_c. A filter of this kind is a result of a cascading of a low-pass and a high-pass filter circuit.

The passing range or "band" of frequencies is called **Bandwidth (BW)**. Defining as 0db the resonant peak, we can find the two cut-off frequencies ($f_{c_{\text{LOWER}}}$ and $f_{c_{\text{HIGHER}}}$) at -3dB. The range between them is the bandwidth (equation 2.3). In figure 2.2 we can see the frequency response of a BPF. [1].

$$BW = f_{c_{\text{HIGHER}}} - f_{c_{\text{LOWER}}} \tag{2.3}$$

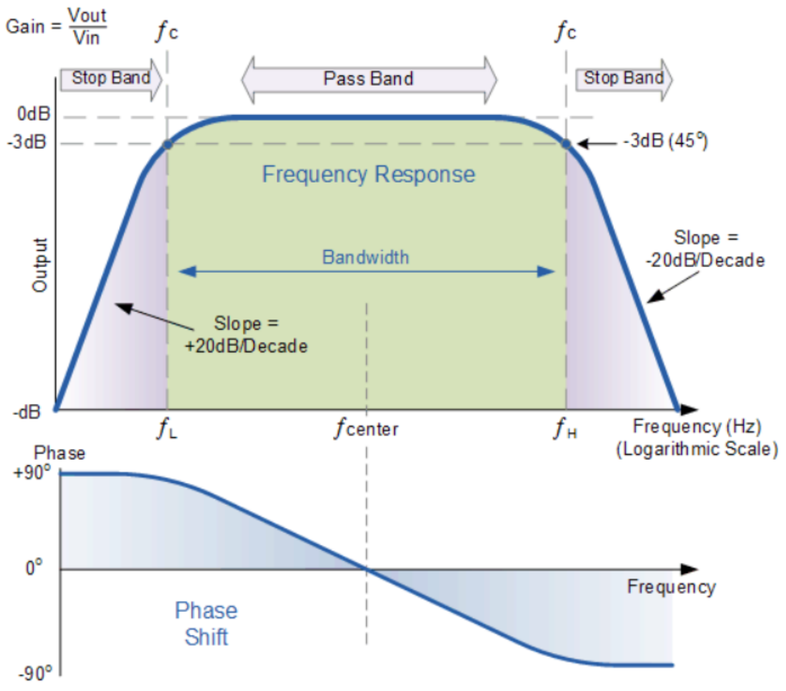


Figure 2.2: Frequency Response of a Band-pass Filter [1].

Synthesis

This method is also additive, since we are adding the outputs of a number of band-pass filters. To synthesize a sound using this method, we use as many filters as the modal frequencies. The filter takes as input an impulse, the center frequency which is the modal frequency and a **Quality factor (Q-factor)** which specifies the bandwidth of the filter. The Q-factor is calculated heuristically, depending on the material of the sound and is inversely proportional to the bandwidth ($Q = f_c/BW$), so the lower the Q-factor, the wider the bandwidth and vice-versa. Hence, more and less frequencies respectively will be included in the audible range. We call the above structure a *resonator*, which also includes a multiplication with the corresponding amplitude, taken from the A matrix.

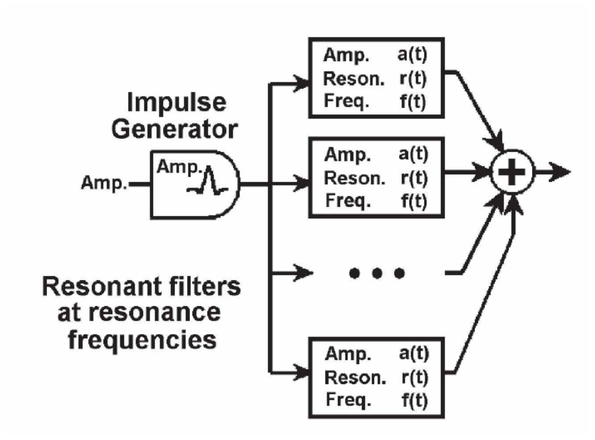


Figure 2.3: Filter-based Modal Synthesis Algorithm [4].

Chapter 3

Method

To do maybe name this chapter smth else?? (4)

A combination of the methods described in Chapter 2 is proposed in the present study.

3.1 Chuck language

Chuck language is a music programming language, made for "real-time sound synthesis and music creation" as mentioned in their website [6]. It's biggest advantage is the way it manipulates time. More specifically, the user specifies how long a sound will last, independent of other sounds that may play at the same time.

Modal features extraction code

We used the ChucK language at the starting point of our thesis to identify and extract the peaks of the recorded "wav" files. The algorithm used in this part of the thesis is made by Perry Cook for the course **"Physics-Based Sound Synthesis for Games and Interactive Systems"** held by *Perry Cook* and *Julius O. Smith* at **Kadenze Academy** [10].

First, we define *ten* to be the total number of peaks identified. After some trials, we found out that *five* peaks are already enough. However, the authors recommend to use twice the sufficient number, hence *ten* for our case. Afterwards, the algorithm having taken a recording as input, computes its histogram and identifies its peaks. The frequencies where peaks occur are the modal frequencies. Finally, the algorithm finds the maximum value of the signal on each peak, calculating the corresponding amplitude of each mode.

3.2 PureData

Pure Data (pd) is another music programming language. It is open source and the main difference from ChucK language is that pd is a visual or "patcher" programming language, using objects instead of code, linked together to form a sequence [9].

Re-synthesis patch

For reasons that will be explained below, we had to attach only one pd patch to every 3D object in the demo. Therefore, all synthesized sounds (impact, rolling and scratching) are being synthesized under one main pd patch.

The main part of the re-synthesis patch is the resonator. Using band-pass filters it gives the impulse response of an object struck at a specified point. It also clips the signal to give more brightness and harmonics. **To do** develop more (5)

3.3 Heavy Compiler

Heavy is a compiler that generates audio plugins from pd patches in interactive soudn and music applications [5]. In this thesis we used it to compile pd patches into Unity audio plugins. Heavy's interface is their website where users upload patches and then are able to download the corresponding plugins and put them into their applications. The plugins we used consist of DLL files and a C# (Unity code) script that allows communication of the plugins with the rest of the script and also enables the sound card to play sounds.

3.4 Unity®

3.5 Overview

CHAPTER 4

Measurements

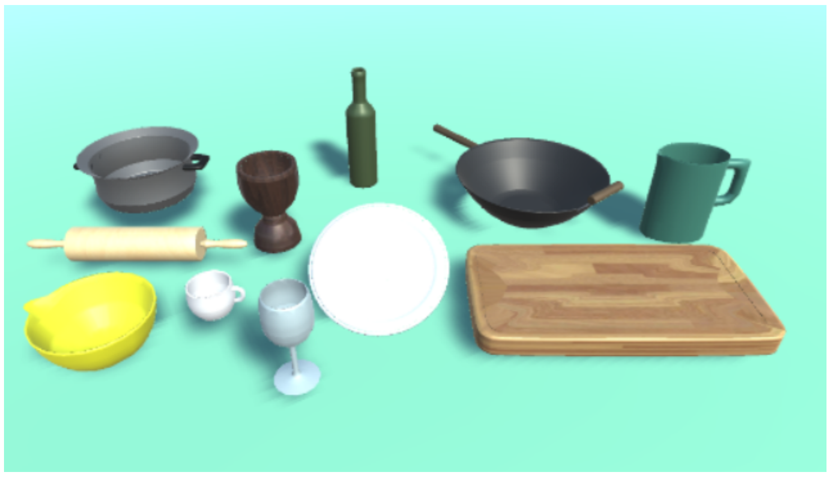
4.1 Recordings

As mentioned above, we used a sample recording of a wooden stick hitting every different area on the objects, to extract the necessary data. We separated the objects into areas depending on their shape and tried to keep the struck sound as uniform as possible inside the same area.

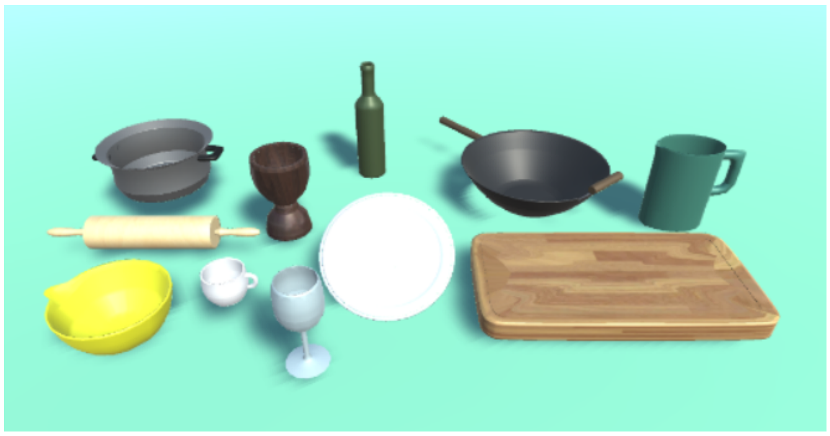
As the hitting object we used a drumstick shown in the figure 4.1. All recordings took place in the same room, using the same drumstick and the same hitting force. The only parameter changing every time are the point of the object being struck.

Figure 4.1: The drumstick used to struck objects during recordings.

We used eleven different objects of everyday life. The idea of choosing those objects came firstly from the need of both owning the object (to perform the recording) and also being able to model it for the demo (simple enough objects). Secondly, since we wanted to test the immersion of the synthesized sounds on users, we wanted to be sure that the sounds used are familiar to them. In figure 4.2 both real and modeled objects are pictured.



(a) Real objects.



(b) 3D models of the objects.

Figure 4.2: The eleven objects used in the thesis.

For the recordings we used a **To do** describe the device used for recordings (6). For the 3D models we used Maya Autodesk software [2] and exported them as FBX® files [3], a file format recognizable by Unity® software [12].

4.2 User tests

To examine the immersion of real-time produced and physics-based sounds on game players, we performed some *MUSHRA* tests to people. The stimuli of the test was both a recorded sound of a struck object and its corresponding synthesized sound using both sinusoidal and filter-based synthesis.

4.2.1 Stimuli

In the first phase of the test, participants where provided with a reference sound (the actual recording) and two testing sounds (the sinusoidal and filter-based synthesized sounds), and were asked to choose the one sounding closer to the reference. Afterwards, they heard only synthesized sounds and were asked to choose the one with the best sound quality. Lastly, we generated different stimulus giving the participants sounds from the same object but with different materials assigned to it and were asked to point out when material changes.

To do generate different stimulus sets, varying the size (7)

4.2.2 Procedure

Stimuli was presented to the participants through **To do** describe headphones (8), in a room with reduced external noise.

4.2.3 Participants

To do number of participants, age, gender, normal hearing, job (9)

Chapter 5

Implementation

To do Here we can put pictures and codes snippets (10)

Instead of having a huge amount of different pre-recorded sound files, which need a lot of space, we eliminate the problem to a 2D look up , where the algorithm matches the object first and then the exact point of the object that collided with some other object.

5.1 Impact Sounds

To do decribe the patch, describe the sound (starts low, goes to max ampl and then decays etc) (11)
To do Put spectrogram pictures of the sounds to describe them (12)

5.1.1 Sinusoidal Additive Synthesis

5.1.2 Filter-based Modal Synthesis

5.2 Rolling Sounds

5.3 Scratching Sounds

5.4 Assignment of Different Materials

Different materials can be assigned to the objects made for this thesis. The designer is able to choose between *plastic, wooden, ceramic, glass and metal* by adjusting a slider on the interface.

Metallic or glass sounds are more "ringy" than wooden or plastic ones that are more "thud". We achieve those sounds by changing the **Q-factor** of the **band-pass filters** used in the pd patch. Q-factor indicated the power loss in the filter. The higher the Q the less power is lost, so the resonator vibrates longer [14].

5.5 Changing the Size

In an application, the same object can appear in different sizes, so this thesis takes this into account. It is known that under the same excitation, the smaller the size of an object, the more high pitched sounds it will produce, because the sound waves travel a smaller distance.

Hence, we implemented a slider for the designer to choose the best sound that corresponds to the size of her object.

5.6 User Interface

Chapter 6

Results & Discussion

6.1 Results

6.2 Discussion

6.2.1 Bugs

- you have to press apply on prefabs
- the procedure of putting the freq and ampl data in
- it is object specific
- lack of acoustical richness that might characterize synthetic signals [7]

6.2.2 Which Synthesis Method Is Better?

6.2.3 Why our work can be used in VR/AR?

6.2.4 Did we manage to achieve what we wanted?

6.2.5 How can we improve our work?

- take into account the environment (reverberation etc)
- make objects destructible

Chapter 7

Conclusion

This is the conclusion
4-5 paragraph approx

Appendix A

Results of tests to users

Appendix B

User Guide to our product

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To do. . .

- ☐ 1 (p. v): name of the recording guy
- ☐ 2 (p. 3): todo test
- ☑ 3 (p. 3): this is done
- ☐ 4 (p. 7): maybe name this chapter smth else??
- ☐ 5 (p. 8): develop more
- ☐ 6 (p. 10): describe the device used for recordings
- ☐ 7 (p. 10): generate different stimulus sets, varying the size
- ☐ 8 (p. 11): describe headphones
- ☐ 9 (p. 11): number of participants, age, gender, normal hearing, job
- ☐ 10 (p. 13): Here we can put pictures and codes snippets
- ☐ 11 (p. 13): decribe the patch, describe the sound (starts low, goes to max ampl and then decays etc)
- ☐ 12 (p. 13): Put spectrogram pictures of the sounds to describe them