Long Project with Audiogaming

Additive Synthesis with Inverse Fourier Transform for Non-Stationary Signals

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Content

Part 1

Introduction

The company

AudioGaming
NATURAL BORN INTERACTIVE

- Localization: Toulouse, Paris
- Activity: Audio plug-in (VSTs and RTAS)
- Main customers: Film and Video Game Industry (Sony, Ubisoft)
- 10 employees

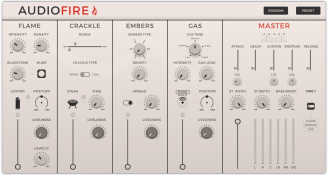

Figure: Audiofire: audio plug-in that recreates fire sound

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Introduction

Objective

- We are continuing the Audiogaming long project from 2015 (Emilie Abia, Lili Zheng, Quentin Biache)

Objective : Synthesizing sounds from their spectrum with a FFT^{-1}

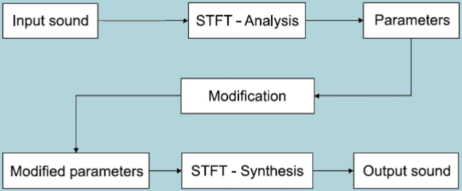


Figure: General approach for modifying a sound in the spectral domain

- We have to implement a new method of additive synthesis ⇒ computationally very fast

Introduction
Context of the Project

- 6 weeks only $\Rightarrow$ Focus on the synthesis method only.

Given codes in Python and Matlab from the 2015 project :

- Python : Analysis estimator of sinus parameters and sinus generation with those parameters (only stationary)
- Matlab : Some reasearch on the Non-stationary synthesis with the LUT of lobes

- We made our own Object Oriented Programmation tree structure in Python
- We remade all the codes to be coherent with the OOP tree structure

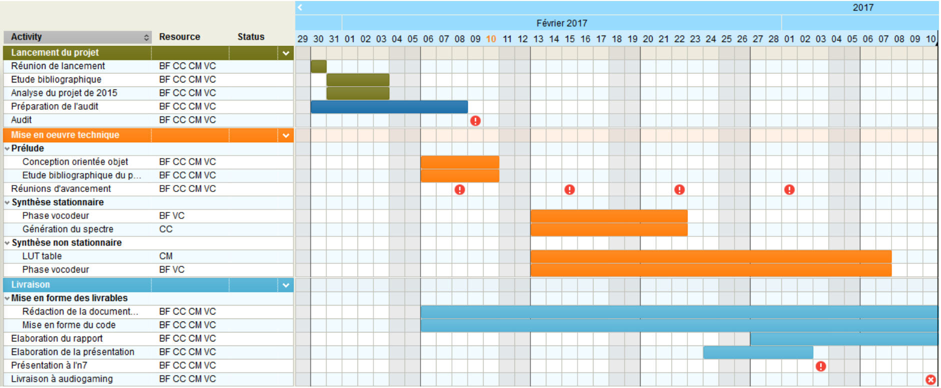
Introduction

Work Environment

Figure: *PyCharm* as Python IDE , *Slack* to communicate,*GitHub* to stock the codes and have a versionning, *Freedcamp* to plan the project events

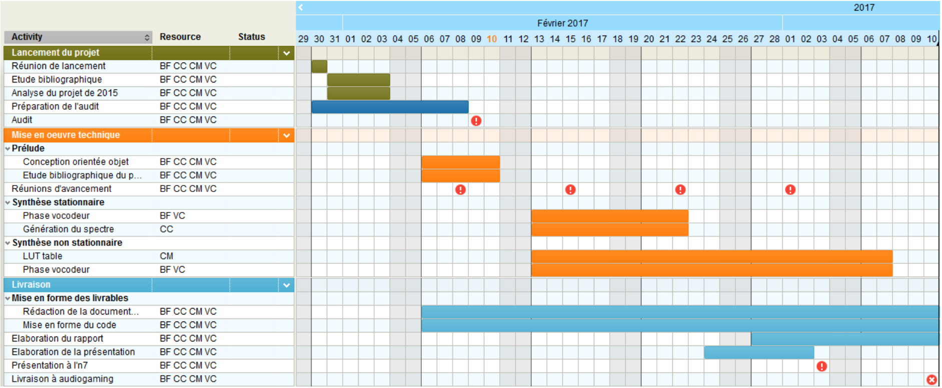
Introduction

Project Management : Gantt Chart (expected event)



Introduction

Project Management : Gantt Chart now



Part 2

Additive Synthesis
Time Domain

The sound signal is represented as a sum of N sinusoids:

$$x(t) = \sum_{n=1}^{N} a_n sin(2\pi f_n t + \phi_n)$$

- Very costly to implement
- Impossible to compute in real-time

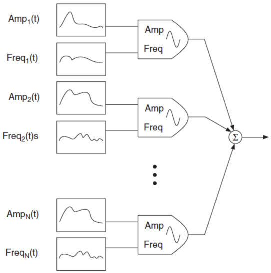


Figure: *The additive synthesis*

Method Overview : Windowing

Analysis

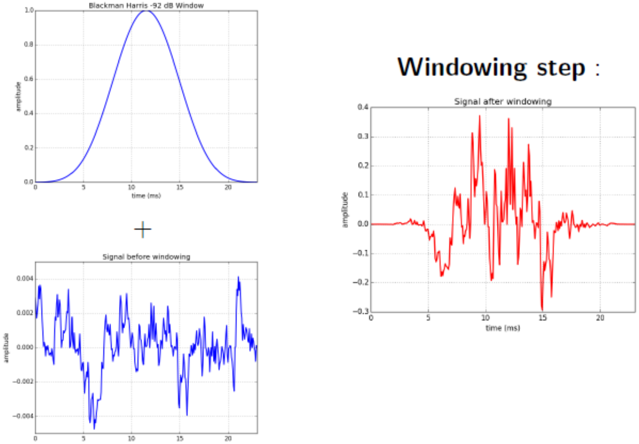


Figure: *Windowing step*

Method Overview : Peak detection in Frequency Domain

Analysis

Peak detection and extraction of parameters by STPT (particular Short Time Fourier Transform):

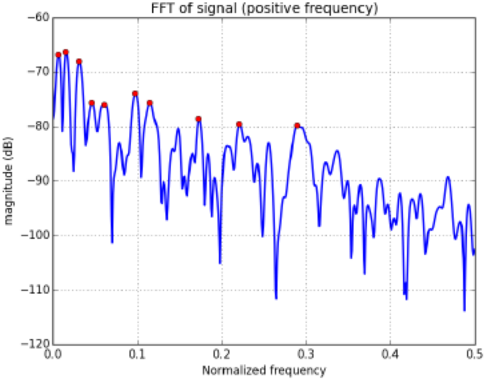


Figure: *Peak detection*

Method Overview : Result (FFT^{-1})

Synthesis

Additive synthesis with FFT^{-1} according to the parameters from the analysis:

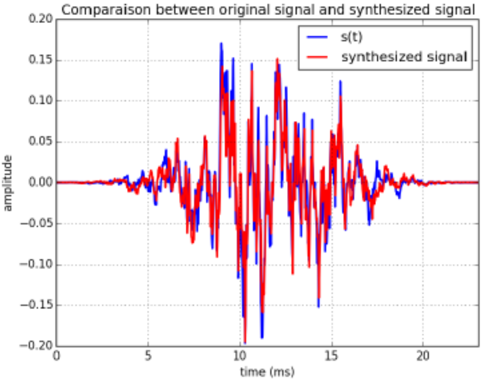


Figure: *Synthesized frame vs Original frame*

Part 3

Stationary Case

Stationary sinusoidal model

Mathematical model :

$$s(t) = a_0 exp[j(2\pi f_0 t + \phi_0)] \tag{1}$$

- 3 parameters: a_0 (amplitude), f_0 (frequency) and ϕ_0 (phase).
- Simplest model but useful for certain kinds of signals.
- Each spectral bin represents a stationary sinusoid.
- $\Rightarrow$ generate a synthetic spectrum with the desired parameters
  $\Rightarrow$ generate a main lobe derived from the Fourier transform of the normalized window w supposedly[1] used during analysis
  $\Rightarrow$ place it at the right position on the spectrum.

[1] Because no actual analysis happened

Stationary Case

Lobe generation

We generate the sinusoids in frequency domain :

- Window the signal to maximize the energy in the main lobe
- We only keep the main lobe for each sine (11 points)
- We assume that the parameters (amplitude, frequency, phase) are already given by the analysis
- We interpolate the relevant bins value if by any chance the wanted frequency $\hat{f}$ is not exactly on a bin, that is to say if $\hat{f} \notin \{\frac{2k\pi}{N}\}_{k=0...N-1}$

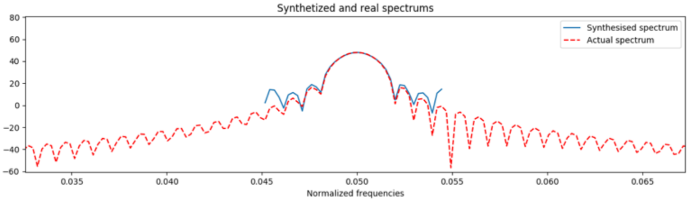


Figure: *Windowed sine lobe*

Stationary Case

Frames separation

The sound signal is a frame-by-frame signal:
The analysis hop size will be called R_a and the synthesis hop size R_s (moving step of the frame)

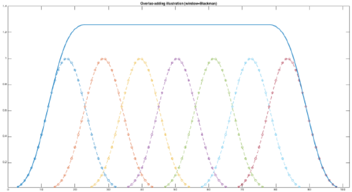


Figure: *Sum of small size Hanning windows*

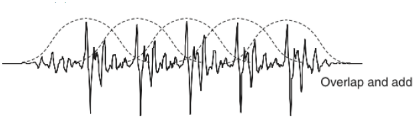


Figure: *Overlap and add*

Stationary Case

Phase Coherence

Phase coherence

The Phase coherence is not a problem in the Stationary case :

- We don't know the window effect on the phase : $f_{w,\hat{f}}(\phi) : \phi \mapsto \tilde{\phi}$
  $\Rightarrow$ We calculate its influence on the first frame and assume the same influence on the other frame.
- We then multiply the generated lobe by $\frac{A}{2}$ and set the lobe phase to $\tilde{\phi} + 2\pi\hat{f}R_a$
- In the purely stationary case, the expected phase shift is the theoretical phase shift :

$$\begin{cases} \tilde{\phi}_i = \tilde{\phi}_{i-1} + 2\pi\hat{f}R_a \\ \tilde{\phi}_0 = \tilde{\phi} \end{cases} \tag{2}$$

Quasi-Stationary Case

What is changing

Non-Stationary Case

Very different approach

Mathematical model :

$$s(t) = exp[(\lambda_0 + \mu_0 t) + j(\phi_0 + 2\pi f_0 t + \frac{\psi_0}{2} t^2)] \tag{3}$$

- 5 parameters:
  $(\lambda_0 + \mu_0 t)$ (overall amplitude)
  f_0 (frequency)
  ϕ_0 (phase)
  μ_0 (amplitude change rate (ACR))
  ψ_0 (frequency change rate (FCR))
- The analysis part give us all those parameters
- To manage the influence of the ACR and the FCR on the lobe $\Rightarrow$ Interpolation of Look-up table of already saved lobes with different (ACR,FCR).

Non-Stationary Case

Look up table

Non-Stationary Case

Phase Vocoder : Scaled-Phase Locking

- The idea is

Part 4

Stationary Case

Sine waves

Stationary Case

Triangular waves

The additive synthesis
Chirps

The idea is to try the method on some chirps signal. And then on real sounds, like instruments and voices.

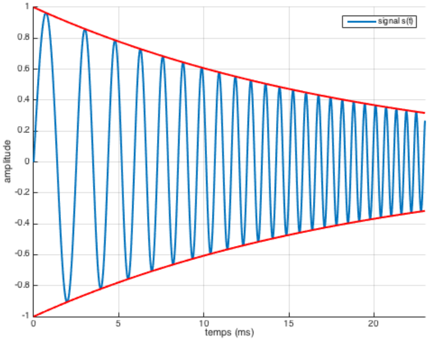


Figure: *Chirp signal to test*

Part 5

Conclusion

- 6 weeks only
- Research subject $\Rightarrow$ Can it really works ?
- Lots of trouble when we try to understand the phase coherence problem

References