Laboratorio di Informatica Musicale

SynthThesis

USER MANUAL

Version 1.0.1

Table of Contents

Introduction

SynthThesis is a virtual synthesizer based on the standard VST. Developed by Luca Mastrofini, a bachelor student at the University of Milan, with the help of the professors at the LIM (Laboratory of Computer Science and Music). SynthThesis is the outcome of a project that sums the three years at the University to show the conclusion of a chapter in the fields of both Computer Science and Music.
SynthThesis is developed in C++, with the framework JUCE. It provides an endless of possible configurations given by the different DSP blocks it has. Here is a list with a brief description.

Oscillators

SynthThesis houses two oscillators, which generate each a digital shaped waveforms. Each oscillator is configured to produce voices in stereo.

Filter

On the combined generated sound of the oscillators it is applied a Filter. This filter has the option to be Low Pass, Band Pass or High Pass. It is possible to set the cutoff frequency, resonance and apply ADSR and LFO modulation to the filter frequency.

Digital Volume Control

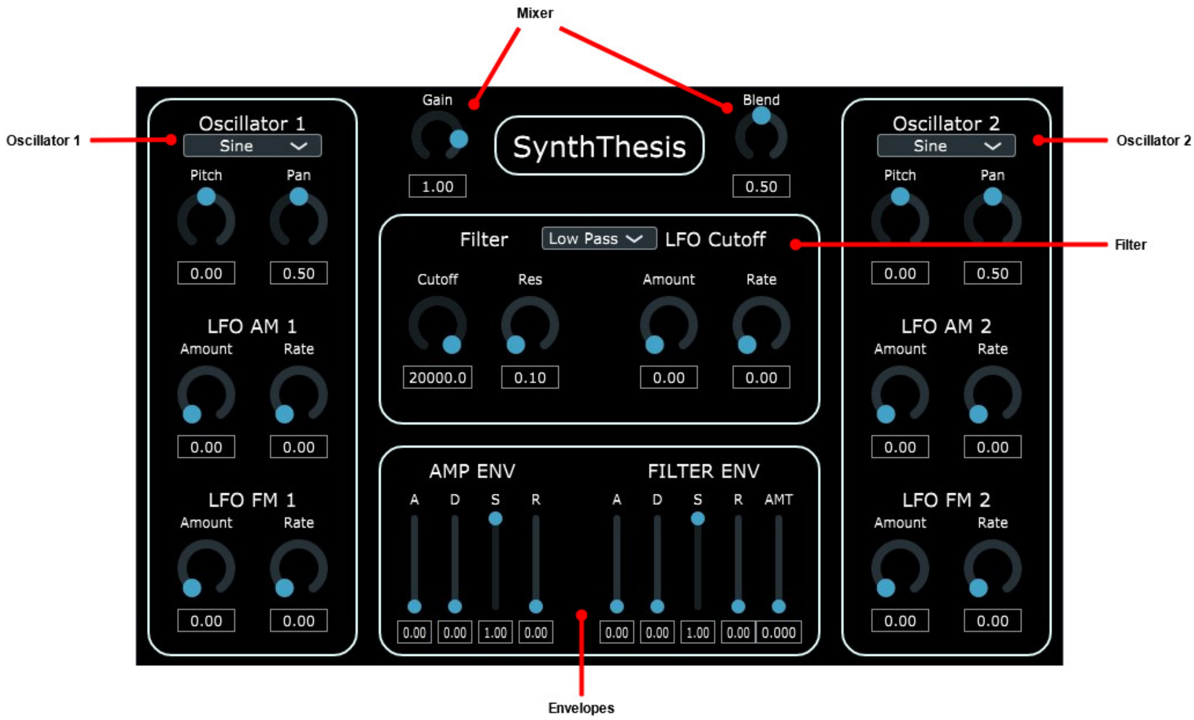
The volume control features the option to adjust the gain of each oscillator and the master gain and apply an ADSR envelope to the filtered sound.

Installation

The version 1.0.0 is only available for Windows as a 64bit architecture. To install the plugin copy the SynthThesis.vst file in the VST Plugin folder of your host. Here is a directory that usually is already set in your host preferences: `C:\ProgramFiles\CommonFiles\VST3`
Make sure after placing it in the correct folder, to go to your host Plugin preferences and scan the folder. After that it should appear on a list of your installed plugins. If the plugin doesn't work it is better to run it in bridged mode in the host plugin settings.

Overview

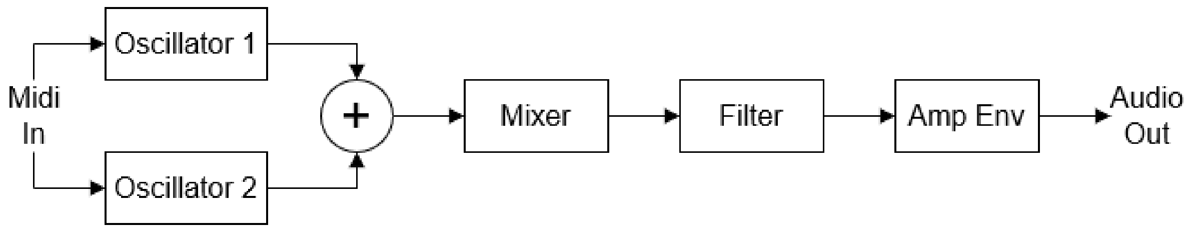
SynthThesis layout is divided in 4 main sections: Oscillator 1 and 2, Mixer, Filter and ADSR envelopes. Every section is fully dedicated to one DSP block. Every block is sur-



rounded by a cyan rounded rectangle to help with the visualization. The only exception is the mixer which is located on the sides of the title. In total there are 30 different changeable parameters.

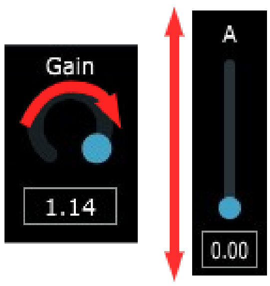
Audio Path

To understand how the audio path works, here is a scheme that explains how every DSP block receives and generates output and the order of how the blocks are displaced. Every arrow (expect the Midi In ones) represent stream of stereo sound data. The Midi note



frequency is plugged in the oscillators to generate two distinct sounds, they are added together and inserted in the mixer. The sound is then processed by the filter and in the end, an ADSR envelope is applied to the amplitude.

Controllers



There are three types of controllers in SynthThesis: Knobs, Sliders and Combo Box. To operate Knobs just move the cursor while keeping pressed with the left or right mouse button and dragging the blue dot in a circular motion, following approximately the below track. To tweak a Slider controller, just drag the blue dot in a vertical direction. Combo Boxes are to select a specific option. Just left click on the box and select the desired option by left clicking on it. Every controller (except combo boxes) can be set up to its default value by left or right clicking twice, also the current value is displayed in the white rectangular box below them.

Polyphony

The number of voices in SynthThesis is static and can't be modified. There are a total of 8 possible voices to play. Each voice generates the sound of two oscillators each where, in their default settings, are an octave apart. As more then 8 voices are played the subsequent voices will clear and then steal the first ones played.

Components Detailed

In this chapter we will take a closer look at the different sections of SynthThesis. Each section consist of several components, for each component we will explain their functionalities, limits, measurement unit and more.

Oscillator

As mentioned before, SynthThesis has two oscillators. The only difference is the frequency they play at: the oscillator 1 plays at the same frequency of the note, while the oscillator 2 plays one octave lower. The oscillators features the option to choose a waveform, set pitch and pan, control the FM and AM.

Waveform

There are 5 types of waveform to choose from: Sine, Square, Triangle, Saw-tooth and Noise. The square waveform is naive and not natural so can present aliasing in higher pitch frequencies. To change waveform just click on the menu right below the Title area.

Pitch

The Pitch parameter is to change the current frequency of the playing note. Its purpose is to create dissonance, it is used to change the scale or to create sweeping effects while automated. Pitch parameter is expressed in cents. The range goes from -1200 to 1200 cents. Each 100 cents is approximately one semitone, though 1200 cents are exactly one octave.

Pan

Pan controls the output of the oscillator to which channel it goes. At 0.5 the same amount is in both channels. At 0 it only plays on the left and on the value 1 it plays only on the right.

Amplitude Modulation

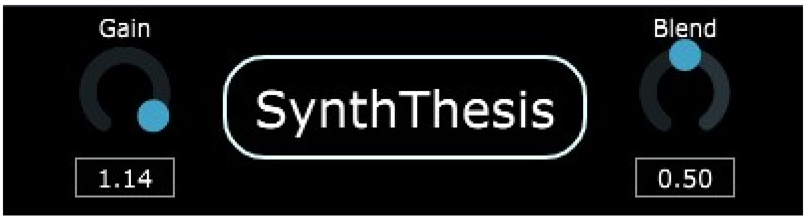
The LFO that modulates the amplitude of the oscillator is a sine wave. The Amount changes how much the LFO influences the actual amplitude and can go from 0 to 1. The rate is the frequency of the sine wave. It starts from 0 and has a maximum of 20Hz.

Frequency Modulation

The LFO that modulates the frequency oscillator is a sine wave. The Amount has the same functionalities of the one listed above, but this one is expressed in cents. At 1200 cent the frequency oscillates from -1 octave to +1 octave from the corresponding note played. Rate is exactly the same in the AM.

Mixer

The Mixer is collocated in the upper part of the GUI on the sides of the title "SynthThesis".



Gain

The Gain parameter applies gain to the final output. The values are from 0 to 1.2.

Blend

Blend is the control that gives priority to one of two the oscillators. By turning the knob all to the left the oscillator 2 will be muted, and vice versa. At 0.5 the output is equal from both oscillators.

Filter

After the two oscillators sounds are mixed together a filter is applied. In the Filter box we can control the parameters of the filter such as the type, the cutoff frequency and the Resonance and also the LFO on the cutoff frequency.

Filter Type

There are 3 types of filters to choose from. To change type, click on the box on the right of the title "Filter". Then choose the filter by left clicking again on the desired one.

- Low Pass Filter: Attenuates frequencies above the cutoff frequency.
- Band Pass Filter: Attenuates a range of frequencies around the cutoff frequency.
- High Pass Filter: Attenuates frequencies below the cutoff frequency.

Cutoff and Resonance

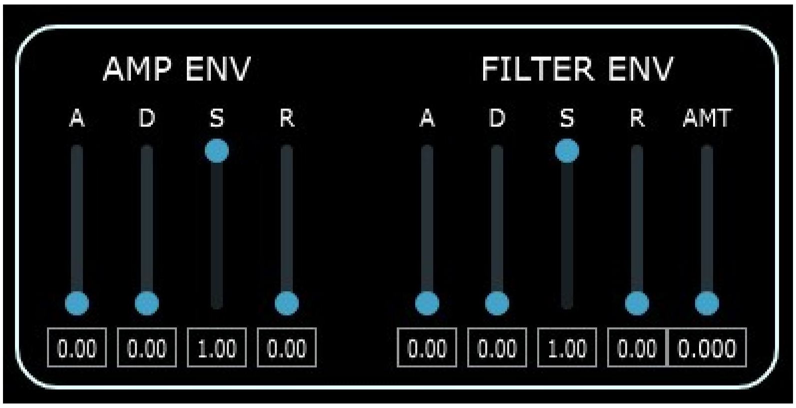
The Cutoff knob changes the cutoff frequency of the filter. Its range goes from 20.0Hz to 20000.0Hz and has a skewness factor of 0.25, giving more priority to the lower values then the higher ones. To avoid aliasing the frequency is capped to a maximum of 20000Hz even after the LFO and ADSR are applied. The resonance applies an amount of filter resonance at the cutoff frequency. The range goes from 0.10 to 4.00.

LFO Cutoff

The LFO on the cutoff frequency is implemented equal to the FM in the oscillator. The amount is expressed in cents and had a range of 1200 cents, while the rate can go up to 20Hz.

Envelopes

ADSR Envelopes modify a parameter value in time. Every ADSR envelope is composed by the attack time, decay time, sustain level and release time.



Amplitude Envelope

The Envelope is directly applied to each sample produced by the oscillators and filter. The attack (A), decay (D) and release (R) are expressed in seconds and have a range of 0.00 to 5.00, while the sustain (S) is a level that has a maximum values of 1.

Filter Envelope

The filter envelope modulates the cutoff frequency of the filter in time. It is exactly the same as the Amplitude Envelope except it features an extra slider: the amount slider (AMT). This parameter changes the influence of the ADSR on the cutoff frequency. If the AMT is 0 there is no envelope applied. The maximum value it reaches is 1. At 1 it adds a maximum of 20000.0Hz to the cutoff frequency.

Final Notes

Limitations of Responsibility

This software is developed by Luca Mastrofini. Its purpose is to demonstrate the acquired knowledge through the three years of computer science and music at the university of Milan, without commercials ends nor quality claims. Therefore we decline any responsibility of any malfunctioning that could cause the lost of data and other inconveniences.

Aknoledgemts and Contacts

This project is developed by Luca Mastrofini with the help of the professors Federico Avanzini and Giorgio Presti. To report bugs and for other comments please right at the email: lucam96@yahoo.it.