FSynth Pro

Audio Resynthesizer

User manual v1.0.2
March 2009

1. Contents

2. Description

FSynth Pro is a stereo Virtual Studio Technology (VST) plug in that allows real-time capturing and re-synthesis of audio events.

3. Demo limitations

The DEMO version of this plug in has the following limitations:

- Stereo setting is disabled
- The parameter display is disabled.

The full version does not have these limitations.

4. Installation

This plug in comes without installation program. The installation can be performed manually by the following two steps:

- Extract the file 'jb_fsynth_pro.dll' from the corresponding zip file, using an (un)zip program or using the build-in functionality from Microsoft Windows XP or Vista;
- Store the dll file in the directory where your host program stores all VST plugins. This directory depends on the host program. Please refer to the manual of your host program to determine the correct directory.

If you have used the demo version of this plug in (with the word 'demo' in the file name) and would like to install the full version, or if you have earlier beta versions, you are strongly advised to delete all earlier versions of this plug in before installing newer versions.

5. Concept of FSynth pro

FSynth Pro is an 'audio resynthesizer'. This means that new audio events are detected from an input signal, and used to re-synthesize a new signal with a similar timbre as the newly detected event. This concept is visualized in Figure 1. In an analysis stage, the onset of a new event is detected by increases in the input signal envelope (top panel). The timbre of the detected event is captured in a certain sampling interval.

In the subsequent synthesis step, a new sound is generated according to the timbre captured in the sampling interval. This new sound can be fully adjusted in terms of attack time, decay time, sustain time and level, and release time. Furthermore, additional parameters such as the stereo width and a filter envelope can be adjusted to shape the new sound event.

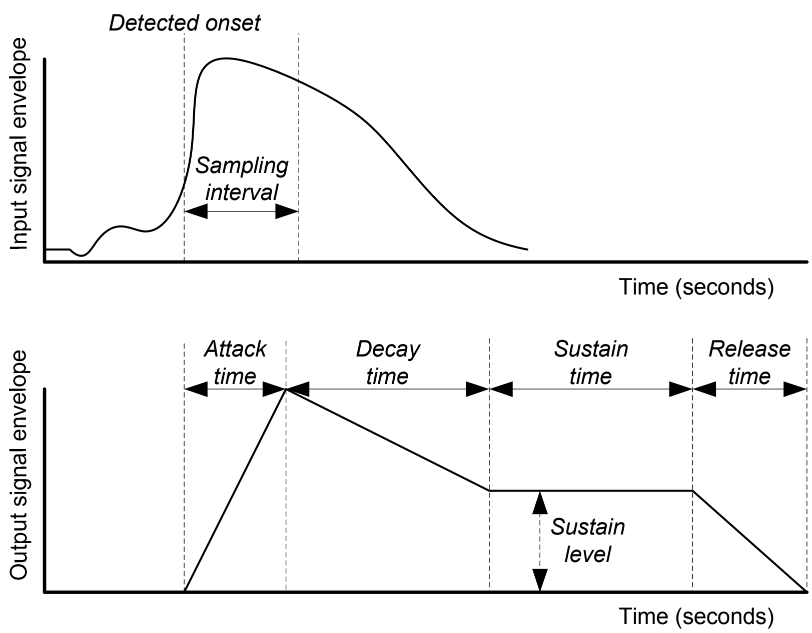

Figure 1 - Concept of FSynth Pro.

6. Usage

6.1 Graphical User Interface (GUI)

The Graphical User Interface (GUI) of FSynth Pro is shown in Figure 2. The GUI is split in several parts:

- A parameter display indicating current parameter values;
- Output trim slider;
- A metering section with meters for output level monitoring (with peak indication);
- A reverb section to add / modify reverb to the output;
- An envelope ADSR part for controlling the envelope attack, decay, sustain and release curves
- A filter ADSR part for controlling the filter attack, decay, sustain and release curves
- Generic settings to control the character and operation of the plug in.

Figure 2 - Graphical User Interface (GUI) of FSynth Pro.

6.2 Controlling GUI elements

GUI elements are controlled by simple mouse clicking and dragging. The following modes of operation can additionally be employed on sliders and rotary knobs:

- Control + left mouse click: reset GUI element to default value
- Shift + left mouse: enable micro control

Controller values will be displayed in the parameter display just below the plug in name on top of the GUI.

(a) Bypass

The '**Bypass**' know allows to bypass the audio processing. When bypass is enabled, the plug in will not modify the audio signals while the various meters will continue to operate.

(b) Mono

The '**Mono**' switch toggles between full polyphony and a single (monophonic) voice. The latter will cause 'voice stealing' effects if new sound events are detected which may have pleasant effects for e.g. drum loops.

(c) Analog

The '**Analog**' switch toggles between an analog sounding filter and a digital sounding filter.

(d) Reverb

The '**Reverb**' switch is used to switch on/off the internal reverberation. The parameters of the reverb (level, reverb time and room size) can be modified with the knobs '**rev level'**, '**rev time**' and '**rev size**'.

(e) Threshold

This knob determines the sensitivity for finding new events. If set to large values, FSynth will not easily find new events. Low values will easily trigger new events. Detection of a new event will be visualized by the corresponding LED.

(f) Filter freq

This knob determines the low-pass filter frequency of the internal filter.

(g) Stereo

This knob determines the stereo ‘width’ of the output.

(h) Dry/wet

The ‘**Dry/wet**’ knob allows to mix the input and output sounds in various amounts.

(i) Randomize

This knob determines the amount of randomness in the internal synthesizer. A value of zero indicates that the output will be fully deterministic, often resulting in a somewhat metallic sound. A larger value will increase the randomness in the synthesizer output.

(j) Resonance

This knob determines the amount of resonance of the low-pass filter.

(k) Filt env

This knob determines the effect of the filter envelope onto the filter cut-off frequency. For a value of 0 (knob is centered), the filter envelope will not adjust the filter cut-off frequency. A positive value will result in increases in the cut-off frequency with increasing filter envelope value. A negative value will decrease the cut-off frequency with increasing filter envelope.

(l) Envelope ADSR

This section modifies the attack, decay, sustain and release times of the output envelope, as well as the sustain time.

(m) Filter ADSR

This section modifies the attack, decay, sustain and release times of the filter envelope, as well as the sustain time. The filter ADSR allows to modify the filter cut-off frequency automatically.

(n) Output slider and VU meters

The output VU meters indicate the current output level (in dB) with a peak-hold function. The peak values are displayed just below the VU meters. Reset of the peak-hold values is performed by a mouse click on the peak value display.

Disclaimers

7. Specifications

Property	Supported values
Supported input/output formats	Stereo Mono (as double-mono input via host)
Plug in delay	2048 samples
Supported bit depths	32 bit float 64 bit float
Number of parameters	25
Supported sample rates	11025 - 96000 Hz Algorithm optimized for 44100 - 48000 Hz
VST version	2.4
Maximum polyphony	12

8. Known issues

- Changing the sample rate on the fly may cause a loss of current parameter values.

9. Change log

Version 1.0.2

- Fixed denormal problem related to reverb

Version 1.0.1

- Initial version.