HISSTools Freeze

Quick User Guide

HISSTools Freeze is plug-in baed on a phase vocoder design that allows you to create rich textures from an input sound, morphing between different input spectra, blurring time or stopping it altogether. The plug-in includes a dynamic morphing filter that can add movement to a frozen sound and three different modes for sampling the input. Whichever mode you are in, this is a plug-in made for explorations of the ambient!

You can use HISSTools Freeze to create a range of ambient textures, from strange reverb-like effects textures to epic morphing soundscapes!

Contents

Installation

To install all you need to do is to unzip and copy the relevant version of the plug-in to the folder which your host uses for plug-ins.

On Mac OSX these will be:

AudioUnit
/Library/Audio/Plug-ins/Components/ [or ~/Library/Audio/Plug-ins/Components/]

VST 2
/Library/Audio/Plug-ins/VST/ [or ~/Library/Audio/Plug-ins/VST/]

VST 3
/Library/Audio/Plug-ins/VST3/ [or ~/Library/Audio/Plug-ins/VST3/]

On Windows VST folder locations vary, so you may need to check for your host, but likely locations are:

VST 2
C:\Program Files\Steinberg\VstPlugins

VST 3
\Program Files\Common Files\VST3\ [Native plug-ins bitdepth]
\Program Files (x86)\Common Files\VST3\ [32bit plug-ins on 64bit Windows]
$APPFOLDER\VST3/

To uninstall simply delete the plug-in.

Usage Overview

The core of HISSTools Freeze is sampling of the input spectrum for resynthesis, which can be done in an automated or entirely manual fashion. This is settable using the **Mode** control. In *Regular* mode the input spectrum is sampled at a regular interval. In *Random* mode the average rate of sampling is still controllable, but samples will be taken in a irregular, randomised manner. In *Manual* mode, the input spectrum is only sampled each time the user clicks the **Freeze** button, which in other modes is used to pause the automated sampling process, and apparently 'freeze' time.

There are four key controls that affect the sampling and resynthesis process, which are the four dials in the upper part of the plug-in UI - **Blur**, **Fragment**, **Sample** and **Morph**. Before the spectrum is sampled it is average over a period of time controlled by the **Blur** control. On its own this creates a reverb-like effect, but in combination with longer **Morph** times it results in richer more organic textures, rather than the more synthetic sounds that can result when it is set to zero. After this averaging the **Fragment** control is used to set whether the whole spectrum is sampled together (0%) or whether a narrower band is sampled (with the rest of the spectrum left as it was before). As this control approaches 100% the bands become very narrow (around a quarter of an octave). In *Regular* and *Random* mode the **Sample** control sets the rate of automated sampling. The ***Morph*** control controls the length of the crossfade between input samples.

The four main controls can interact in complex ways, for instance setting the **Morph** time longer than the **Sample** time can result in even slower moving textures, due to the fact that the process is not given enough time to reach its target before a new input sample is taken. In another example, setting **Sample** very fast, with **Fragment** high can lead to multiple frequency bands changing at once due to the fact that FFT process is calculated at discrete points in time.

The **FFT Size** and **Overlap** controls set the timing and frame size for the spectral processing. Higher values for the **FFT Size** will lead to a higher frequency resolution, and more hi-fi sound, but lower settings can be used to create a more lo-fi robotic sound. The highest settings might require you to set your hosts buffer size higher in order to avoid audio drop-outs. The **Overlap** control has a less obvious effect on the sound, but higher settings may also increase the perceived quality in certain scenarios, as well as increasing CPU usage.

At the bottom left-hand side of the plug-in are the controls for the *Morphing Filter.* The morphing filter generates randomised filter shapes across a number of bands (set by the **Num Bands** control) and crossfades between them. The controls allow you to control the shape and timing of the filtering process. The **Strength** control sets the maximum extent of the filter bands, whilst the **Tilt** control allows you to bias the filter towards low (negative values) or high (positive values) frequencies. The timing of crossfades is controlled by two controls - the **Interval** control, which sets the minimum time between each filter shape, and the **Random** control, which sets a maximum random amount that is added to the value of the **Interval** control.

At the bottom right-hand side of the plug-in you will find an output section that allows you adjust the output width and gain.

The parameters are covered in a detail in the *Parameter List.*

Controls

- Mouse over a dial to show its value
- Adjust dial values by dragging with the mouse, or by double-clicking to type in a new value
- Adjust numeric panels by dragging with the mouse, or by single-clicking to type in a new value
- Adjust menus (which are indicated by a downward triangle) by clicking on the value to cycle through the values, or clicking on the triangle to display a drop-down menu.
- All controls can be reset to default value by single-clicking on them with the shift key held down
- To make fine adjustments to dials and numeric panels control whilst mousing depress shift ***after you have started dragging***
- The morphing filter controls are greyed out when the filter is disengaged (when **Strength** is set to 0dB)
- The **Sample** control is greyed out when in manual mode as it has no effect

Parameter Listing

Parameter	Value	Sets
FFT Size	512 /1024 /2048 / 4096 / 8192 / 16384 / 32768	FFT size used for processing Larger sizes increase frequency resolution and tend to sound richer Low values can sound interestingly lo-fi
Overlap	2 / 4 / 8	Number of overlapping frames Increasing this required more CPU but can sound more hi-fi
Mode	Regular / Random / Manual	Mode used to sample the input The first two of these are automatic In *Manual* mode the user must press **Freeze** to sample the input
Freeze	Off / On	Freeze mode In *Regular*/*Random* mode this parameter is used to pause automatic sampling In *Manual* mode this is the only way to take new samples of the input
Blur	0 to 2000 milliseconds	Time interval that is averaged before the input spectrum is sampled This control creates time blurring of the input separately to the sampling process
Fragment	0 to 100%	Amount that the spectrum is split/ fragmented when sampled As this is increased new samples will grab a increasing small frequency band
Sample	5 to 5000 milliseconds	Interval between samples of the input spectra (this is exact in *Regular* mode and the average in *Random* mode)
Morph	0 to 10000 milliseconds	Time taken to morph to a newly sampled input spectrum.
Filter Num Bands	1 to 60	Number of bands in the morphing filter
Filter Strength	0 to 24 dB	The strength of the morphing filter
Filter Tilt	-100 to 100%	Tilt of the filter towards low or high frequencies
Filter Interval	10 to 5000 milliseconds	The minimum time for each filter change
Filter Random	0 to 5000 milliseconds	The random time for each filter change
Width	-12 to 12 dB	Output width adjustment
Gain	-12 to 12 dB	Output gain

Minimum System Requirements

Mac
OS 10.11 or higher
32 bit or 64 bit VST 2.4, VST 3 or AudioUnit host
Minimum Intel i5 CPU recommended (AVX is required)

Windows
Windows 7 or later
32 bit or 64 bit VST 2.4 or VST 3 host
Minimum Intel i5 CPU recommended (AVX is required)

If you find a bug, please email A.Harker@hud.ac.uk

Enjoy using HISSTools Freeze!