Adoppler Enhanced 2

V 2.03

AZ Audio

1. Introduction

AdopplerEn2 is a powerful VST2 effect for windows VST Host. As you can notice, the effect have seriously changed since the first version, when it was only a modulated delay and a modulated panner designed to reproduce the Doppler effect on the audio source. In version 2, the modulation is not fixed and each audio processing function can be modulated by a lfo. The processing is not only made on a single stereo channel but on 8 parallel channels. There are also eight lfos that can be used by all the channels and can have various shapes. So if the original Doppler effect, which the plugin was originally designed for, still can be produced, a wide variety of new ones are also possible, such as chorus, flanger, panner, multi tapes delays, strange distorted filters/delay sweeps, combs or totally new and unrealistic send effects. Paradoxically, AdopplerEn2 is better and more interesting to use with this last category.

The whole interface is based over parallelism. The channels and the lfos use a shared editor which displays the values of the selected channel or lfo. Some hints and the values of edited controls are displayed at the bottom. Each Lfo or channel can be copied and paste to another channel or lfo, ensuring a fast and efficient workflow.

AdopplerEn2 is easy to use, however this manual will fully describe all the parameters and will expand certain considerations which was not possible to put directly as hint in the plugin. At the same time it will ensure you to have a perfect understanding of the effect and of its flow.

2. Input

Input gain: define the audio input gain, without affecting the feedback.

Buffer size: define the maximal buffer size, from 0 to 2 seconds. This setting can be used to get more accurate results when tweaking a channel delay time. Note that this parameter is not designed to be changed in real time (automation or midi control).

3. Channel bar, Lfo bar

Channel selector : define which channel you are editing.

Lfo selector : define which lfo is being edited. When changing a modulation source or when changing the channel in sight this selection is automatic.

C: Copy either the current channel parameters or the the current lfo parameters. The feedback value is skipped during this operation and it's always set to z when pasting a copied channel.

P: Paste previously copied parameters

4. Channel overview

The flow of a channel is always: delay>filter>(feedback)>amp>panning>out. Each channel uses the same recording buffer so they can be defined as some tape reader heads which is a bit different from the case where the buffer is not shared. The main consequence is that the feedback of a channel is read by all the tape reader heads. Each channel uses a total stereo processing chain.

Delay: define the tape reader delay time (distance between the recording head and the reader head). The value inspector is displaying this parameter as time and as ratio of a beat length. This can be used to design tempo synced presets, even if the effect is not designed for this. For example when changing the tempo, the delay time will have to be adjusted manually.

Delay modulation source: define which of the 8 lfos will modulate the delay time.

Delay modulation amount: define the amount of modulation applied to the delay time. The amount is computed by this way : delay time (max) – percentage of the delay time.

Pan: define the position of the channel output into the stereo field.

Pan modulation source: define which of the 8 lfos will modulate the stereo position.

Pan modulation amount: define the amount of modulation applied to the stereo position.

Filter type: define the filter kind of the channel. LP will only let the frequencies under the cutoff frequency pass. BP will only let the frequencies around the cutoff frequency pass. HP will only let the frequencies over the cutoff frequency pass. The filter is inactive when there is no filter type highlighted.

Cutoff frequency: define the cutoff frequency of the filter.

Cutoff modulation source: define which of the 8 lfos will modulate the cutoff frequency

Cutoff modulation amount: define the amount of modulation applied to the cutoff frequency The amount is computed by this way : cutoff frequency (max) – percentage of the cutoff frequency.

Resonance: define the gain at the cutoff frequency.

Amp: define the amplification of the channel.

Amp modulation source: define which of the 8 lfos will modulate channel amplification

Amp modulation amount: define the amount of modulation applied to the channel amplification The amount is computed by this way : amplification value (max) – percentage of the amplification.

Feedback: define the amount of reinjection inside the recording buffer. The feedback is bipolar and could cancel the feedback of another channel in some particular cases. As the feedback is copied before the amplification, a channel can provide feedback even if its output amplification is set to null. The sum of all the feedback channels is processed by a sinusoidal shaper which keeps audio in a acceptable digital range and can also, in some particular cases, result into a good sounding saturated feedback. However summing feedback must always be used with care.

Power: activate the channel.

5. Lfo overview

Each lfo is identical. They feature advanced resynchronisation possibilities. Their curves are computed each sample, even if it would not be always necessary, it maintains the overall quality of the effect.

Freq: define the frequency of the oscillator. If the sync switch is activated then the frequency will be snapped to the musical time, going from a period of 8 beats to a bit less than one beat quarter.

Sync: set the lfo frequency as dependent of the host tempo (musical time)

Shape: select the shape of the lfo: off, parabolic, triangular, saw down, saw up, square, random. When a lfo is not used it's better to set it to off, which will completely skip it to be processed and save a bit of computing power.

Smooth: smooth the shape of the lfo. This smoothing is useful with square,random and saw shapes as they often jump from 0 to 1 without transition. In this case the artefacts produced by the jumps over the target parameter will be attenuated. It can also be used if two synced lfos haven't the same frequency (which can produce important gaps in the curve) as the smoothing is not part of the lfo but an effect applied directly to the lfo output.

Resync: set which lfo will resynchronize the focused one. If the source differs from the target then the resynchronization will append when the source will reach its maximal value. If the target is the same as the sync source then it won't be resynchronized. Note that in all the cases a global and automatic resynchronization is always performed when the host transporter start playing (if the host informs the plug of this).

Resync offset: define the value of the lfo when it get resynchronized. This apply to all resynchronization types, including the automatic one on transport start. Between 0 and 49 the lfo will have a lower value than the resynchronization source, between 51 and 100 the lfo will have a greater value than the resynchronization source. At 50 the lfo will be perfectly synced to its resynchronization source.

6. Output

Output gain: define the gain of the output, this apply to the dry signal and the sum of all the processing channels.

Mix: define the ratio between dry and wet audio, going from fully dry to totally wet.

7. Presets

Presets can be changed using the VST host preset loading/saving system or with the effect internal system (it's using the common fxp system in both cases). Presets can be handled in the popup menu appearing when right clicking the interface.
The first menu item is used to save an fx program. AdopplerEn2 presets are stored in a predefined folder (common apps data\Az Audio\AdopplerEn2\).Presets must not be located in a

sub folder. The Next items are the available fx programs. Click to load. Note that depending on the vst host, the program name might not be updated even if the loading was successful.

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AZ Audio gives the warranty that AdopplerEn 2 is an original creation, that it doesn't use or contain stolen, illegal, unlicensed or malicious code.

Programming, manual and presets by Basile Burg (azertopia@free.fr)

Graphics by Scott Kane (s0nkite@hotmail.com)

Virtual Studio is a Technology by Steinberg media Gmbh

And...

Have fun with AdopplerEn2 !