



# VoiceFX

## VoiceFX for Alto Studio 4.0 - User Manual

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## Introduction

VoiceFX is Tsugi's main technology for vocal processing. It includes a large number of digital audio processing functions that can be used to create robot, alien or creature voices as well as other effects such as radio transmissions.

VoiceFX is available in different forms in Tsugi software, including as a tool in Alto Studio (starting with version 3.2). It is therefore accessible from the "Tools" menu or by using the Ctrl+V keyboard shortcut.



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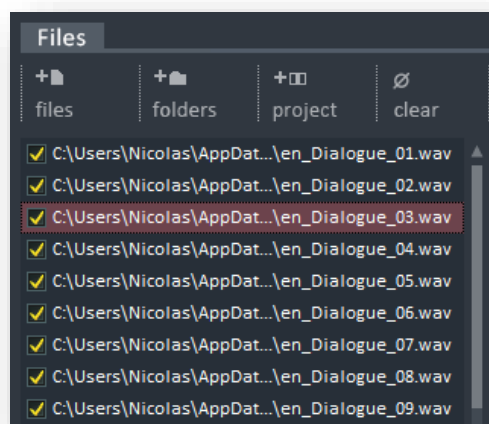
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## **Main Window**

The VoiceFX window is divided in 2 main areas. In the panel on the left, you can specify the files to process and determine how the results should be saved. Please note that VoiceFX only processes mono files. In the central area, you can edit the processing chain, by building a stack of effect racks.

### **Selecting the files to process**

In the “Files” section, you can select the files you want VoiceFX to process. Simply click on the “Files”, “Folders” or “Project” buttons at the top of the list. You can also remove all the files from the list by pressing “Clear”. Note that when adding folders, all the subfolders are also automatically added.



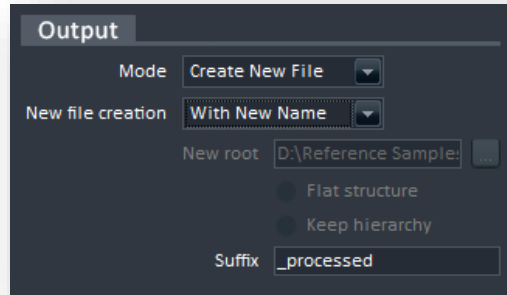
If an Alto project is currently open, once you pressed the “Project” button, you will be able to add all the files from one or several languages in this project. It is also possible to select only the dialogue lines spoken by a given character. Finally, you can add files by dragging them from the Windows Explorer onto the files list.

Once the files are in the list, you can exclude some of them from the processing by clicking on the checkbox located on the left of their path. Only the files whose checkboxes are marked will be processed. At the bottom of the list, a button allows you to quickly select / deselect all the files.

### **Specifying how to save the results**

The “Output” section lets you specify where you want to save the dialogue lines processed by VoiceFX. You can either overwrite the files you processed or create new ones. In this case, you can choose to save the new files at the same location but under a different name, or under a new root folder.





If you choose to save the processed files alongside the original ones, you will be able to enter a suffix. If you choose to save them under a new root, you can specify the path to that root. You can also select if you want to replicate the original hierarchy of subfolders under that root or if you want to save all the processed files at the same level, directly under the root.

### **Basic editing**

Here we describe how to create and edit a processing chain, by stacking racks. Although the user can add, move and remove racks as needed, there will always be an input rack at the top of the processing chain and an output rack at the bottom.

Adding a rack: drag a rack from the list on the right, and drop it where you want to insert it. A horizontal line will be displayed to show where it will be created. Double-clicking on a rack in the list will automatically add it at the end of the processing chain, just before the output rack. It is possible to add several instances of the same rack to the processing chain.

Removing a rack: click on the X icon of the rack to remove it from the processing chain. Alternatively, you can also click on a rack to select it, and then press the Delete key to remove it.

Moving a rack: it is possible to rearrange the order of the racks (except for the input/output racks) by dragging / dropping them in the stack.

Expanding / collapsing a rack: to make more space while still seeing the racks used in the processing chain, you can collapse a rack by clicking on the small triangle pointing down at the left of the rack. The triangle then points to the right and you can click on it again to expand the rack.

Enabling / disabling a rack: to enable / disable a rack, simply click on its power button located in the top-left corner. This is the equivalent of a bypass function. The input and output racks cannot be disabled.

Loading / saving: it is possible to save and reload a processing chain. The extension of the file is .vfx.

Presets: some presets were created by the Tsugi team to get you started. You can select them in the combo box at the top-left of the processing chain.



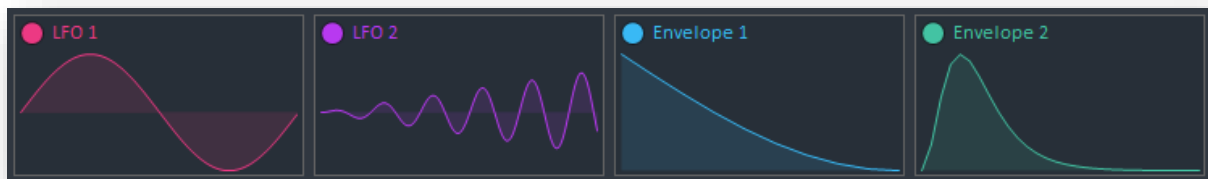
## Preview and Processing

There are basically two steps when using the VoiceFX tool. The first step consists in editing the processing chain and adjusting its parameters. To test that processing chain on a given audio file, it is possible to either click on that file from the list, or to drag and drop the file from the Windows Explorer. Then, simply press on the “Play” button.

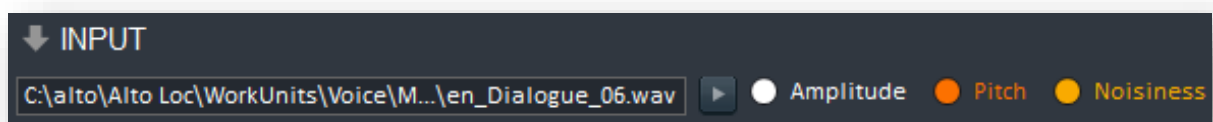
Once you are satisfied with the results, the second step is to batch-process the files from the list. This is done by clicking on the “Process” button. A progress bar will be displayed and all the files selected in the “Files” section will be processed and then saved according to your settings in the “Output” section.

## Modulations

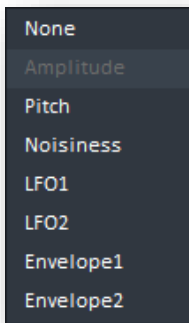
At the bottom of the processing chain are the modulators - 2 envelopes and 2 LFOs - that can be used to continuously modify some of the parameters of the racks in the processing chain. To edit an envelope or a LFO, simply click on its graph (see below to learn more about their parameters).



In addition, the audio file being processed can be analyzed and its amplitude, pitch and noisiness curves used to modulate the rack parameters.



Therefore, there are a total of 7 types of modulations. All modulation sources are represented by a different color. They can be assigned either by dragging the colored dots close to their name onto the slider of the parameter to modulate, or by right-clicking on a parameter and selecting the modulation source (or “none” to remove the current modulation).



Parameters whose sliders are plain (i.e. don't have a central dot) cannot be modulated.

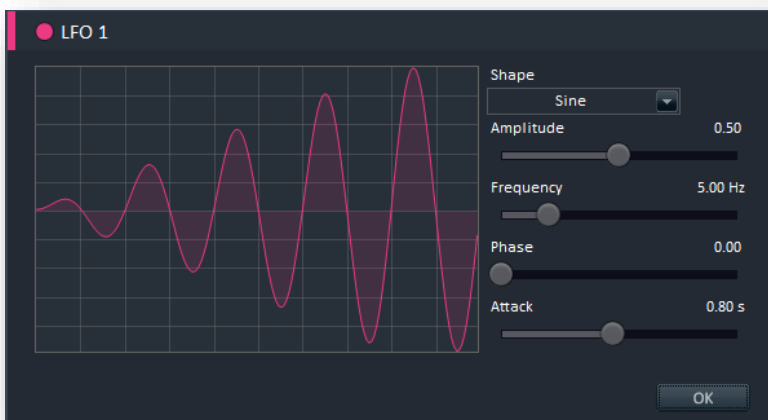
By default, the amount of modulation is set to 1.0 (i.e. the full range of a parameter). Therefore, when you assign a modulation source to a parameter, the full track of the slider changes to the color associated with the modulation source.



To adjust the modulation amount, simply click on the thumb of the slider and move the mouse vertically.

## LFOs

A LFO module generates a periodic, low frequency signal that can be used to control parameters. Its amplitude, frequency, waveform and attack can be set.



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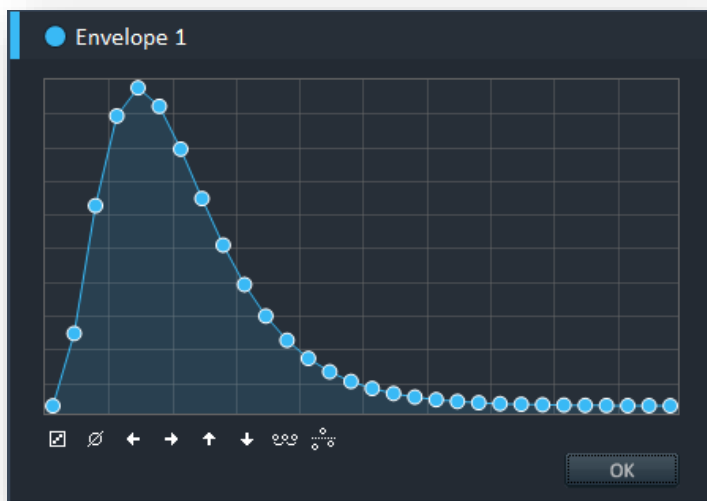
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## Parameters

- *Shape*: the waveform generated by the low frequency oscillator: pulse, random, sample and hold, saw up, saw down, triangle or sine.
- *Amplitude*: the amplitude of the control signal generated, between 0 and 1.
- *Frequency*: the frequency of the low frequency oscillator, from 0.01 Hz to 100 Hz.
- *Phase*: the phase of the signal generated, expressed in degrees (from 0 to 360).
- *Attack*: the attack of the signal generated i.e., the duration of a fade-in from 0 to full amplitude.

## Envelopes

An envelope generates a control signal from a multi-segment curve and can be used to modulate rack parameters.

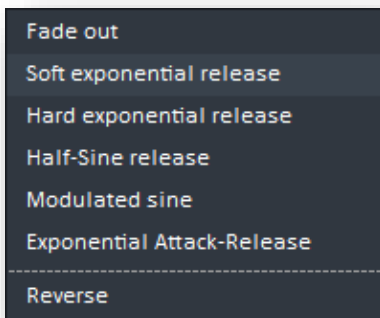


Under the envelope, the following buttons are present, from left to right:

- Randomize the envelope
- Clear the envelope
- Move the selected points left
- Move the selected points right
- Move the selected point up
- Move the selected points down
- Space the selected points evenly
- Set the selected points at the same level



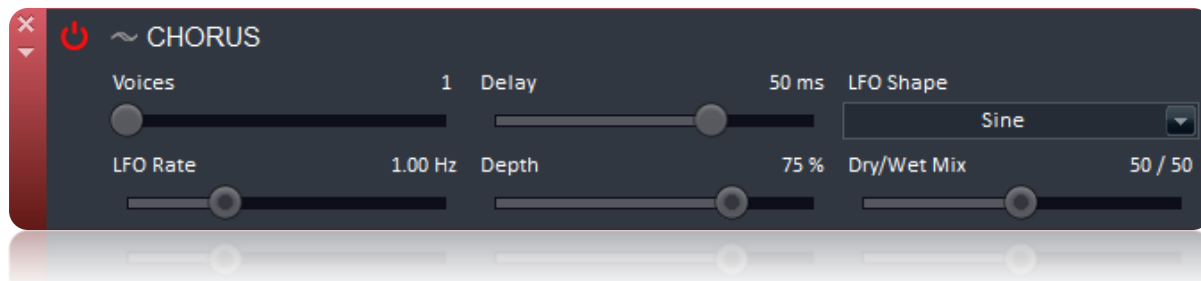
A context menu is also available. It offers a few presets as well as a function to reverse the envelope's curve.



## Racks reference

### Chorus

This is a typical chorus effect which combines several delayed voices.



#### Parameters

- *Voices*: number of voices (from 1 to 8) added to the original signal.
- *Delay*: delay time of the new voices in milliseconds.
- *LFO Shape*: shape of the LFO modulating the delay lines: sine or triangle.
- *LFO Rate*: modulating rate (0.1 Hz to 10 Hz).
- *Depth*: amount of modulation expressed as a percentage.
- *Wet/Dry mix*: percentage of wet signal.

### Clipper

This rack implements a hard clipper with a selectable threshold. Anything signal above the value of the threshold will be brought back to the value of the threshold.

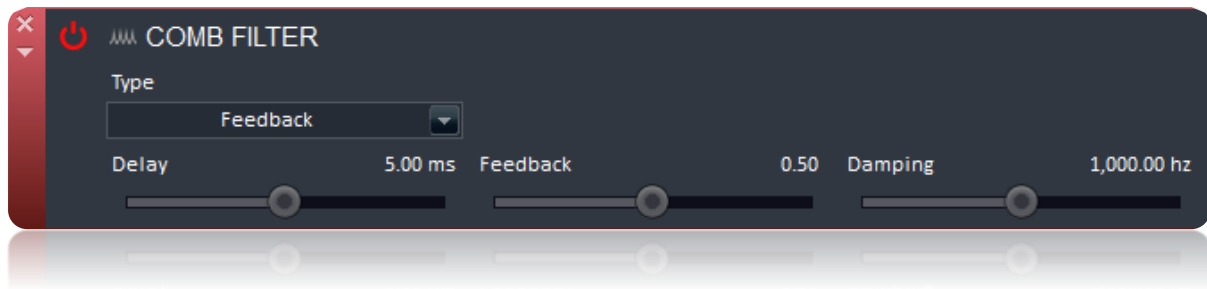


#### Parameters

- *Threshold*: the value of the threshold above which the signal is clipped (from 0 to 1).

### Comb Filter

This rack is a comb filter with feedback and feedforward modes. The feedback and the damping of the filter can be adjusted.

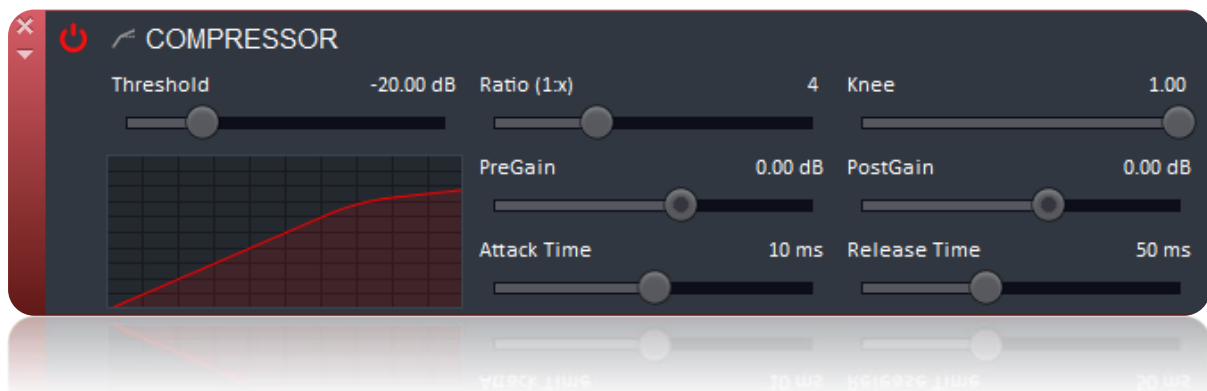


### Parameters

- *Type*: the type of comb filter, feedback or feedforward.
- *Delay*: the length of the delay line, from 0ms to 10ms.
- *Feedback*: the amount of feedback.
- *Damping*: the damping applied to the feedback signal, from 20Hz to 20000Hz.

### Compressor

This is a basic compressor limiter with threshold, knee, ratio, attack and release parameters. The compression curve is displayed.



### Parameters

- *Threshold*: level in dB at which the compressor starts operating.
- *Ratio*: compression ratio from 1 to 20.
- *Knee*: knee value between 0 to 1 (for a softer transition to compression).
- *PreGain*: gain applied before compression, in dB.
- *PostGain*: gain applied after compression, in dB.
- *Attack Time*: period during which the compressor is decreasing the gain to reach the level that is determined by the ratio. Between 0 and 200 ms.
- *Release Time*: period during which the compressor is increasing the gain to reach the level determined by the ratio. Between 10 and 3000 ms.

## Convolver

This rack performs a convolution between the signal being processed and an external signal.

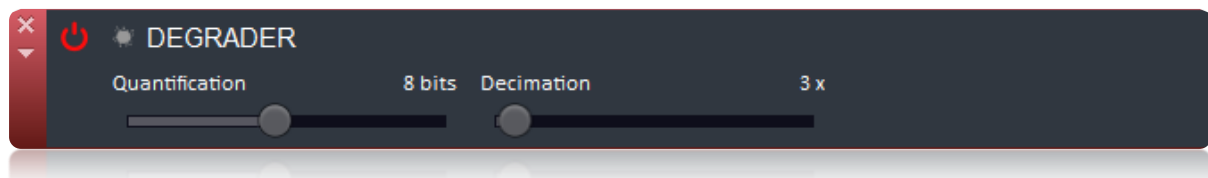


### Parameters

- *Sample*: an external audio file to be convolved with the input signal.
- *Gain*: the gain, expressed in dB, applied to the external signal before processing.
- *Quality*: the quality of the convolution, from 0 to 10. The higher the value, the higher the quality but also the higher the CPU cost.

## Degrader

This rack degrades the quality of the input signal by decimation and quantification, simulating both a lower sampling rate and a lower bit-depth.

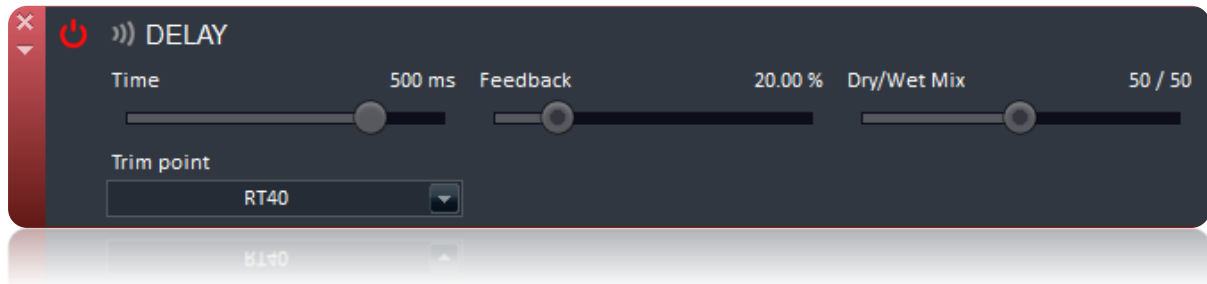


### Parameters

- *Quantification*: the number of bits on which the input signal is quantified.
- *Decimation*: the factor by which the input signal is downsampled.

## Delay

This rack is a typical Delay effect.

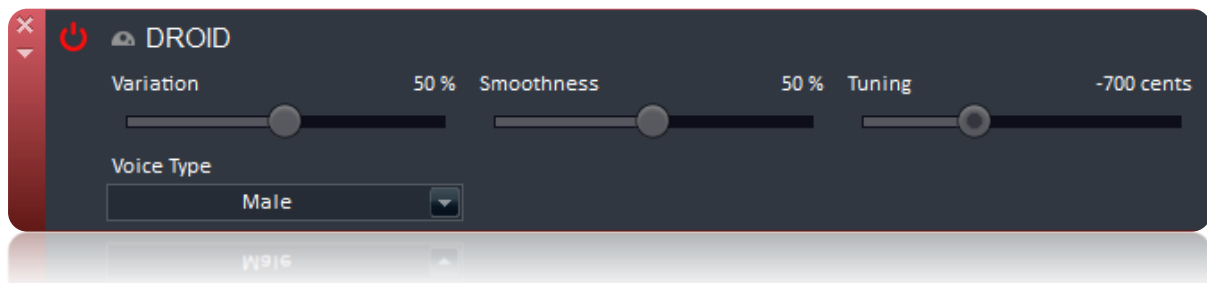


### Parameters

- *Time*: time of the delay from 0.01 to 2 seconds.
- *Feedback*: percentage of the processed signal reinjected in the processing loop.
- *Mix*: mix between the direct and processed signals (in this case, % of processed signal that makes up for the output signal).
- *Trim point*: how long the delay is allowed to decay before being cut off. RT20, RT40 and RT60 correspond to the times it takes for the level to reduce by 20, 40 and 60 dB respectively.

## Droid

This is a fun rack that generates R2D2-like bleeps from the dialogue that goes in. The resulting signal sounds like a droid but with features that mimic the original voice.



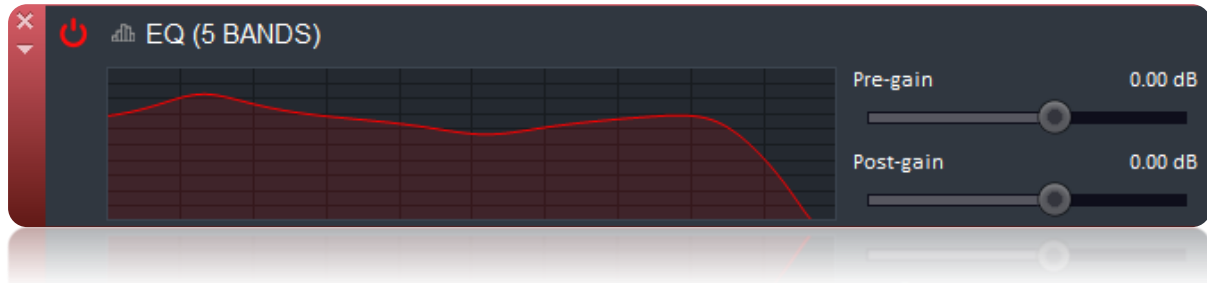
### Parameters

- *Variation*: the magnitude of the amplitude / pitch variations in the droid signal.
- *Smoothness*: determines how abrupt the transition between two sounds generated by the droid is.
- *Tuning*: amount of detune of the droid sounds compared to the original voice (expressed in cents from -1 to +1 octave).
- *Voice Type*: this parameter is used to specify the type of input voice, in order for the algorithm to perform a better analysis.



## EQ (5 bands)

This rack implements a 5-band graphic equalizer. For each band, it is possible to specify the gain, frequency, Q and type of the corresponding filter. Bands can also be disabled. Pre and post gains are available to control the level of the signal.



### Parameters

- *Pre-Gain*: gain applied to the incoming audio signal, before it is processed by the equalizer. Expressed in dB.
- *Post-Gain*: gain applied to the audio signal, after it is processed by the equalizer and sent to the output of the module. Expressed in dB.
- *EQ Curve*: the curve corresponding to settings of the equalizer. By clicking on the graph, a second window appears, in which it is possible to specify the behavior of the EQ.



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Click on a red dot to adjust the parameters of the corresponding band: gain in dB, center/cutoff frequency in Hz, Q factor, as well as the type of filter: bell, lowshelf, highshelf, lowpass or highpass.

They can be adjusted with the sliders and controls under the graph, or directly by moving the dot on the graph (for the gain and frequency). It is also possible to enable / disable a filter by clicking on the numbered dot under the graph.

Finally, right-clicking on the graph gives access to the context menu, allowing you to reset the curve or to call a preset.

## Filter

This rack offers a set of biquad filters of different types, with cutoff and resonance parameters. The frequency response curve is displayed.

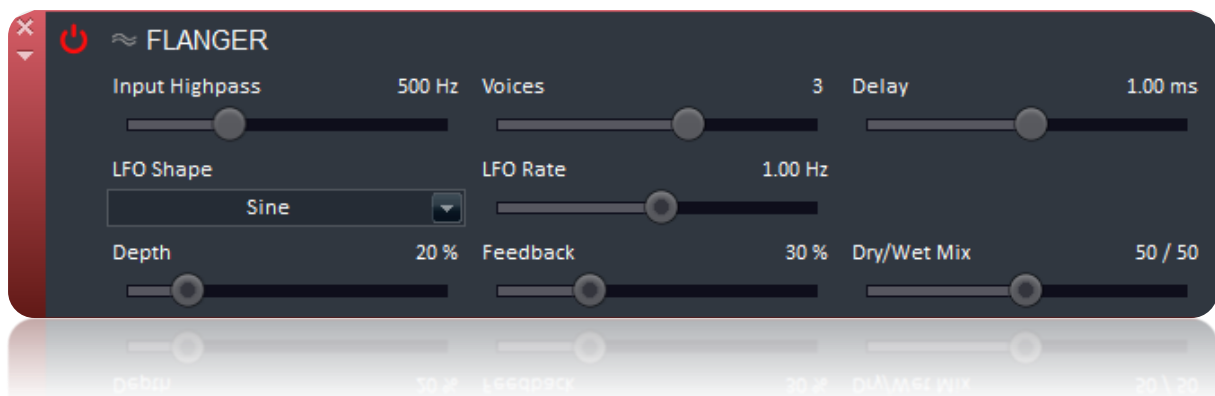


### Parameters

- *Type*: type of the filter (lowpass, highpass, or bandpass).
- *Cutoff*: cutoff or central frequency of the filter, from 10 Hz to 20000 Hz.
- *Resonance*: resonance (Q) of the filter, from 0.1 to 10.

## Flanger

This rack implements a flanger effect. It is similar to the chorus module but with a shorter delay, and an extra feedback parameter.



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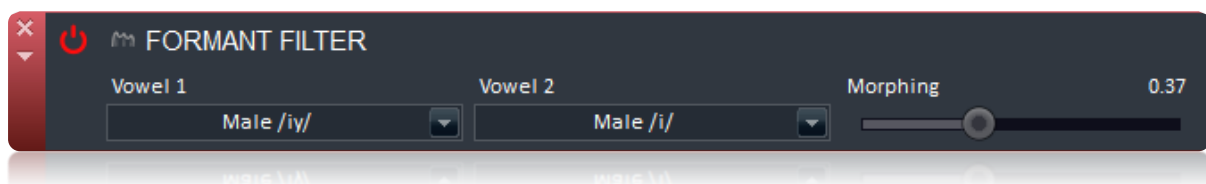
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### Parameters

- *Input Highpass*: cutoff frequency of a highpass filter applied to the input signal.
- *Voices*: number of voices added to the original signal.
- *Delay*: delay time of the new voices in seconds (0.1 to 10 ms).
- *LFO Shape*: shape of the LFO modulating the delay lines: sine or triangle.
- *LFO Rate*: modulating rate (0.1 Hz to 10 Hz).
- *Depth*: amount of modulation expressed as a percentage.
- *Feedback*: amount of processed signal reinjected in the processing loop, expressed as a percentage.
- *Dry/Wet mix*: percentage of processed signal in the output signal (in addition to the direct signal).

### Formant Filter

This rack implements a filter that can emphasize several frequencies (formants) corresponding to the sound of vowels. A morphing parameter allows the interpolation between two sets formants.



### Parameters

- *Vowel 1*: first set of formant frequencies corresponding to a vowel.
- *Vowel 2*: second set of formant frequencies corresponding to a vowel.
- *Morphing*: value between 0 and 1 which determines the final set of frequencies as being an interpolation between the 2 sets (if 0 we are using the frequencies of the first set, if 1 the frequencies of the second one).

### Frequency Shifter

This rack implements a typical frequency shifter.



### Parameters

- *Frequency Shift*: the amount of frequency shifting, from -1000 Hz to +1000 Hz.





## Gain

This is a simple gain rack that can be inserted at any point in the processing chain to increase or decrease the level of the signal.



## Parameters

- *Gain*: the gain to be applied to the input signal, expressed in dB.

## Input

This rack is where we select the sample(s) to be processed. It has a pre-gain and a tuning parameter, as well as a reverse playback option. When a sample is selected, its waveform is displayed.



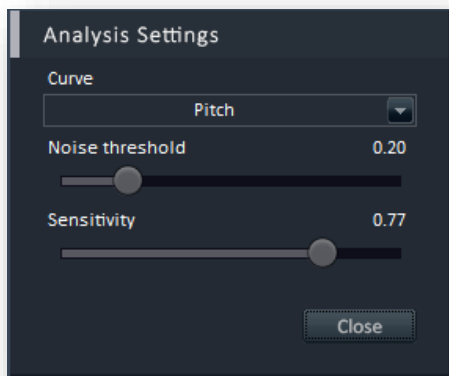
## Parameters

- *Sample*: the sample to be processed. When the actual processing happens, files will be sequentially selected from the file list on the left of the VoiceFX window. During preview, it is possible to select a single file from the list by clicking on it, or to drop an audio onto the graph area. Please note that only mono files can be processed.
- *Gain*: the gain to be applied to the audio file, expressed in dB.
- *Tuning*: the tuning applied to the audio file, expressed in cents (from -1 to +1 octave).
- *Reverse*: select this option to play the audio file in reverse, which is sometimes great as a basis for alien voices.

The amplitude, pitch and noisiness envelopes of the signal can be extracted and used as modulation sources for the parameters of the other racks.

You can press on the small cog icon to specify the settings - such as noise threshold and sensitivity - for each type of analysis.

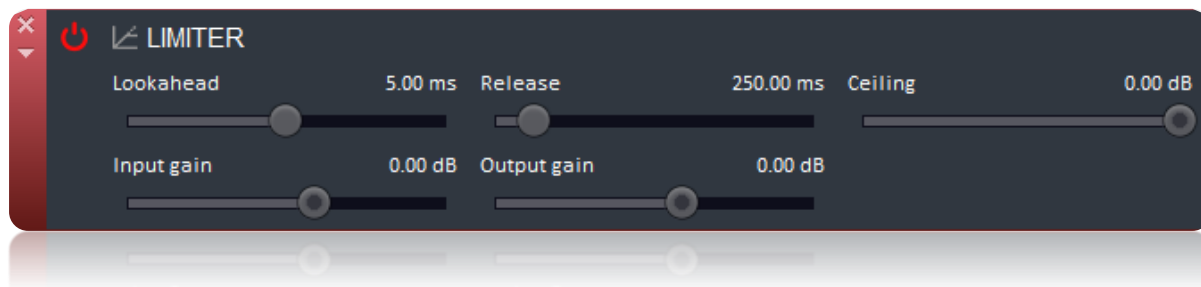




This rack is always at the top of the processing chain. It cannot be moved, delete or disabled.

### Limiter

This rack limits the amplitude of the input signal.



### Parameters

- *Lookahead*: the duration where the signal is analyzed in advance.
- *Release*: the period where the limiter is resuming the gain to the normal level.
- *Ceiling*: the maximum level passed to the output.
- *Input gain*: the gain applied to the input signal.
- *Output gain*: the gain applied to the output signal.

### Mixer

The mixer allows for an external sound source to be mixed with the original signal. For example, it can be used to add a background noise to simulate a radio effect.



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### Parameters

- *Sample*: the external sample to be mixed with the incoming signal.
- *Looping*: if this option is selected and the external sample is shorter than the signal to process, it will be looped.
- *Gain*: the gain to be applied to the external sample before the mixing, expressed in dB.
- *Tuning*: the tuning applied to the external sample, expressed in cents and between -1 and +1 octave.

### Output

The output is always the last rack on the stack (i.e., at the bottom) and it offers a final gain stage.

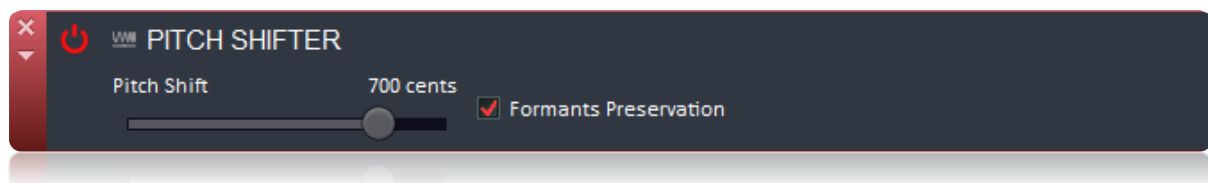


### Parameters

- *Gain*: the final gain stage of the processing chain, expressed in dB.

### Pitch Shifter

This rack changes the pitch of the voice while preserving the duration of the audio, with or without formants preservation.

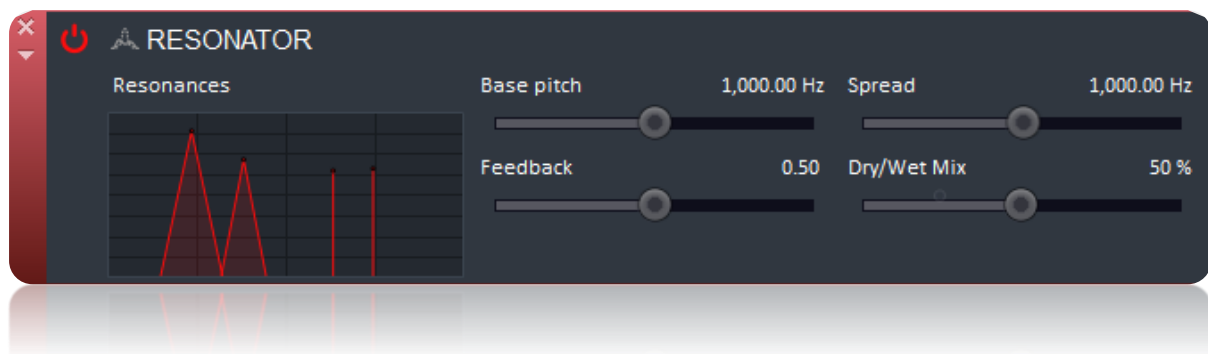


## Parameters

- *Pitch shift*: amount of pitch shifting to apply, from -1200 cents (-1 octave) to +1200 cents (+1 octave).
- *Formants Preservation*: if this option is selected, the frequencies of the formants are not shifted.

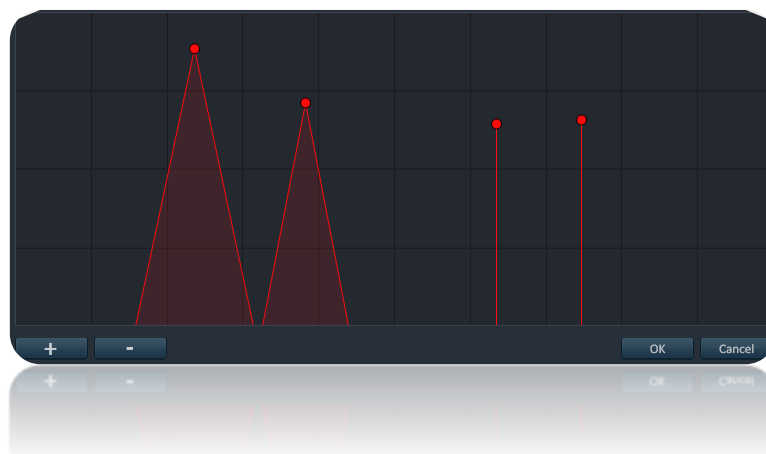
## Resonator

This rack adds resonances to a signal. Their amplitude, frequency, and bandwidth can be adjusted.



## Parameters

- *Resonances*: the location of the resonances we want to create in the spectrum. Click on the graph to open a secondary window to edit their positions, amplitudes, and bandwidths.



- *Base pitch*: the lowest pitch displayed by the graph (and at which a resonance can be created). From 20 Hz to 20000 Hz.
- *Spread*: determines the range between the lowest and highest pitches available to create a resonance. Resonances can be created between Base pitch and Base pitch + Spread. From 1Hz to 20000 Hz.



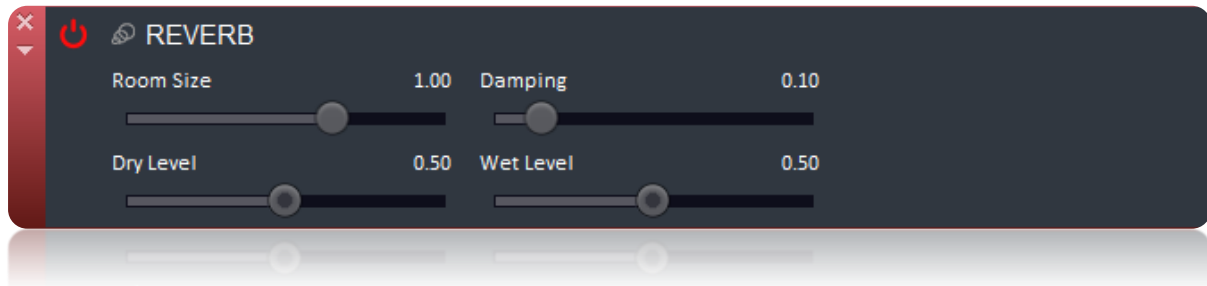
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- *Feedback*: the amount of feedback, from 0 to 0.99. Larger values created more pronounced resonances.
- *Dry/Wet Mix*: percentage of processed signal in the output signal.

### **Reverb**

This module implements a simple algorithmic reverberation effect. It is not intended to recreate vast reverberant spaces by using impulses and convolution (the convolver rack can be used to that effect).

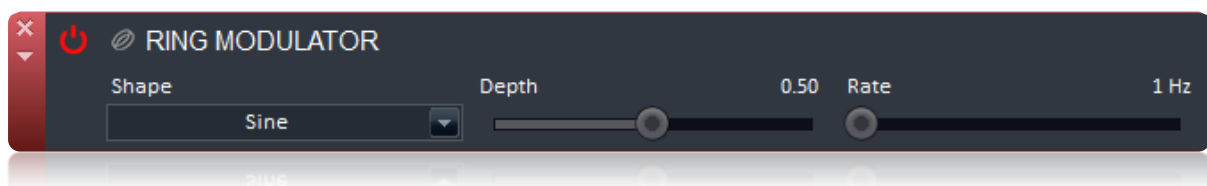


#### **Parameters**

- *Room Size*: this parameter simulates the size of the room. A bigger room will be more reverberant. The valid range goes from 0.1 to 2.
- *Damping*: determines the attenuation of the reverberation (0 to 10).
- *Dry Level*: amount of direct signal that is output (from 0 to 1).
- *Wet Level*: amount of reverberated signal that is output (from 0 to 1).

### **Ring Modulator**

The ring modulator multiplies the input signal by the output of an oscillator. It can be used advantageously to create robotic voices.



#### **Parameters**

- *Shape*: the shape of the signal generated by the internal oscillator: pulse, random, sample and hold, saw up, saw down, triangle or sine.
- *Depth*: the depth of the modulation (0 to 1).
- *Rate*: the frequency of the internal oscillator (from 0.01 Hz to 5000 Hz).

## Saturator

This is a basic waveshaper effect, used for example to create distortion.

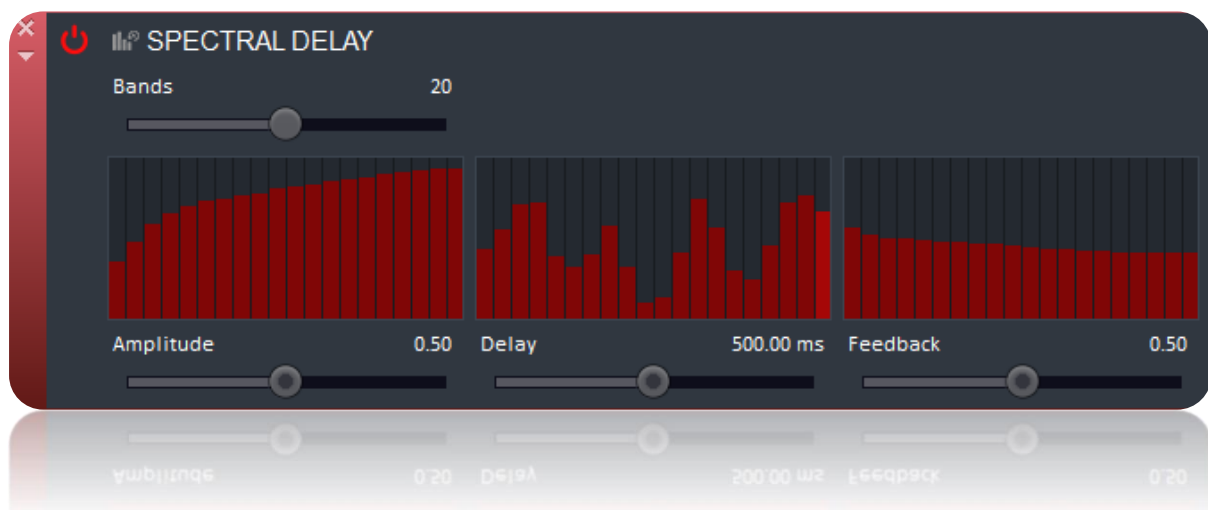


### Parameters

- *Type*: the type of curve used to create the saturation effect. Many choices are available: PowerA, PowerB, Tanh, Cubic, Cosine, ArcTan, Pre-amp.
- *Saturation*: amount of saturation applied, from 0 to 1.
- *Pre-gain*: the gain applied before the saturation.
- *Post-gain*: the gain applied after the saturation.
- *Block DC*: this option allows you to block any DC offset generated by the saturation effect if you want.

## Spectral Delay

This rack applies a delay effect independently to each frequency band of the input signal. For each band, the level, delay length, and feedback can be adjusted.



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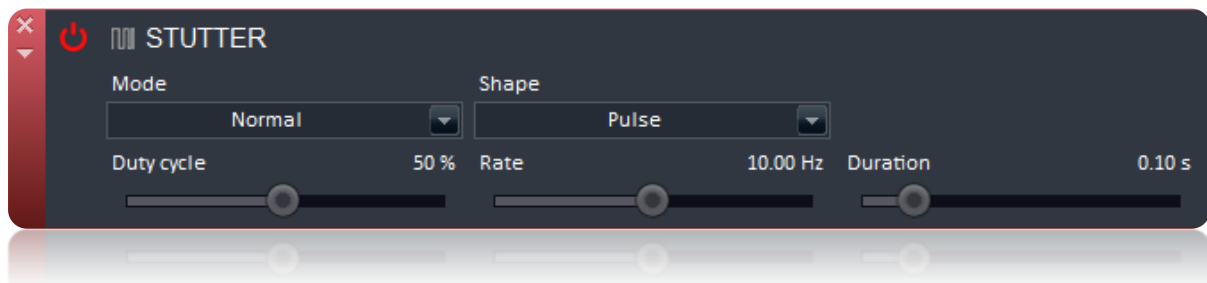
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## Parameters

- *Bands*: the number of bands in which the input signal is divided, from 8 to 32.
- *Band levels*: a bar graph (left) allowing for the easy input the level of each band. The maximum value corresponds to the value of the Amplitude parameter (see below).
- *Band delays*: a bar graph (center) allowing for the easy input of the delay duration of each band. The maximum value corresponds to the value of the Delay parameter (see below). The same presets than for the band levels are available.
- *Band feedbacks*: a bar graph (right) allowing for the easy input of the feedback applied to each band. The maximum value corresponds to the value of the Feedback parameter (see below). The same presets than for the band levels are available.
- *Amplitude*: the maximum amplitude of the bands, which is used as a reference by the band levels bar graph (from 0 to 1).
- *Delay*: the maximum duration of the delay in the bands, which is used as a reference by the band delays bar graph (from 0 to 1s).
- *Feedback*: the maximum feedback applied to the bands, which is used as a reference by the band feedback bar graph (from 0 to 0.99).

## Stutter

This rack can be used to create typical stuttering vocals and glitch effects, by periodically applying a gate function to the input signal.



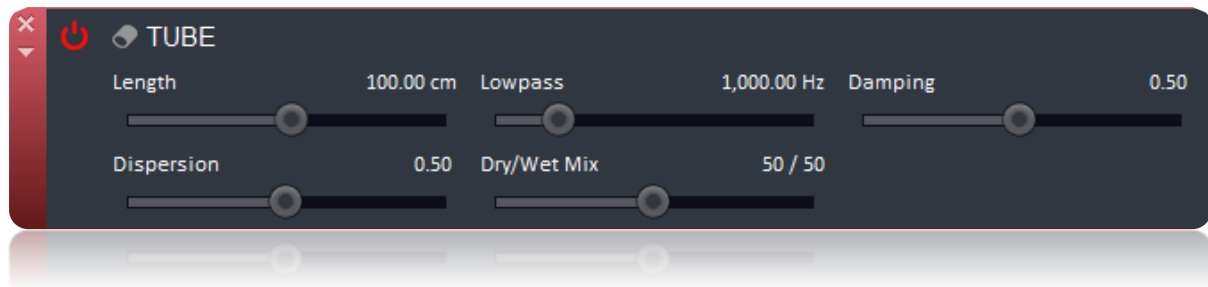
## Parameters

- *Mode*: stuttering mode, actually determining if the gate signal is used as such, or inverted.
- *Shape*: shape of the gate signal: Pulse (default), Saw Down, Saw Up, Sine, or Triangle.
- *Duty cycle*: the duty cycle of the gate signal (only used for Pulse and Triangle shapes). Set to 50% by default.
- *Rate*: the frequency at which the gate is applied, expressed in Hz (from 0.01 Hz to 1000 Hz, 10 Hz by default).
- *Duration*: the duration of the shape of the gate (from 10 ms to 10 s, 100 ms by default).



## Tube

This rack is a virtual tube whose acoustic properties can be adjusted by changing its length, dispersion, and absorption rate.

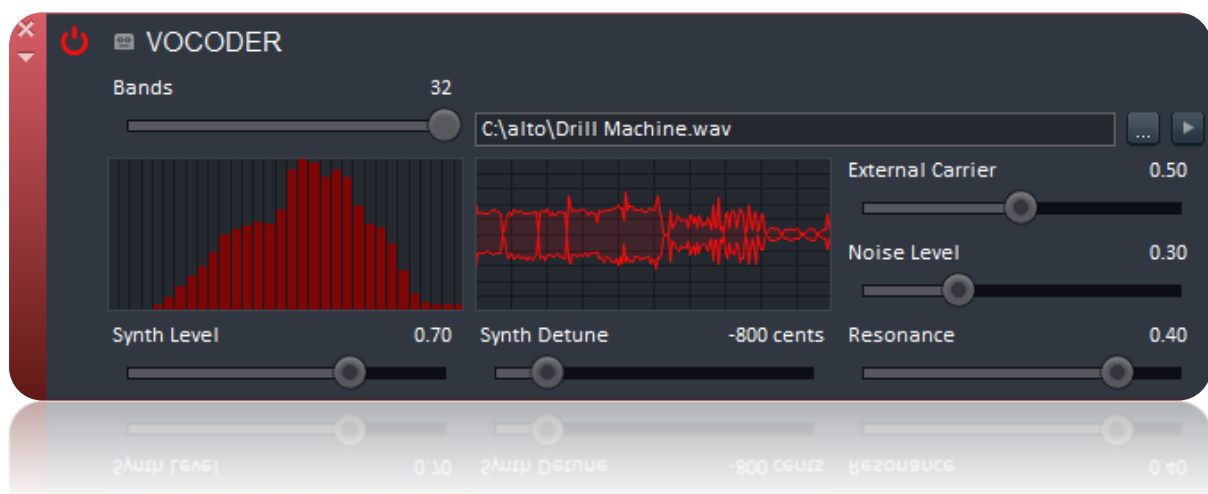


### Parameters

- *Length*: length of the tube (from 10 cm to 10 m, 1 m by default).
- *Lowpass*: frequency of the internal lowpass filter (from 0 Hz to 20000 Hz, 1000 Hz by default).
- *Damping*: damping rate of the tube (from 0.01 to 1, 0.5 by default).
- *Dispersion*: dispersion level of the tube (from 0 to 1, 0.5 by default).
- *Dry/Wet Mix*: percentage of processed signal in the output signal (in addition to the direct signal).

## Vocoder

This rack simulates a vocoder whose number of bands can be selected. The level of each band can be adjusted. The vocoder has an internal oscillator and a built-in noise generator that can be combined to create the carrier signal, as well as an input for an external carrier signal. This rack is especially useful to create robotic voices.





## Parameters

- *Bands*: number of bands in the vocoder.
- *Bands Levels*: level of participation of each band to the output. It is possible to edit the levels by drawing with the mouse.
- *Sample*: the external wave file that can be used as a carrier signal.
- *External Carrier Level*: level of the external carrier signal, if any (0 to 1).
- *Noise Level*: level of the internal noise generator (0 to 1).
- *Synth Level*: level of the internal oscillator (0 to 1).
- *Synth Detune*: tuning of the internal oscillator from -1200 cents (-1 octave) to +1200 cents (+1 octave).
- *Resonance*: this is the resonance of the bandpass filters used by the analysis and synthesis stages (0 to 1).

## **Annex 1: Keyboard Shortcuts**

Select first rack in chain	<b>Home</b>
Select last rack in chain	<b>End</b>
Select next rack	<b>Ctrl + Tab</b>
Delete selected rack	<b>Delete</b>

