SWAYED

User manual (v1.0.1)

Stryde Audio
April 10, 2026

CONTENTS

INTRODUCTION

Thank you for purchasing Swayed!

Swayed is modeled after a famous 1989 synthesizer that combined FM synthesis with sample based synthesis. To get an idea of its capabilities it's recommended to start by checking out the presets, and to tweak some of their settings to get a feel for how they affect the sound.

This manual contains a complete description of every knob and button in the user interface. If you're wondering what a certain parameter means you can find the answer in here.

INSTALLATION

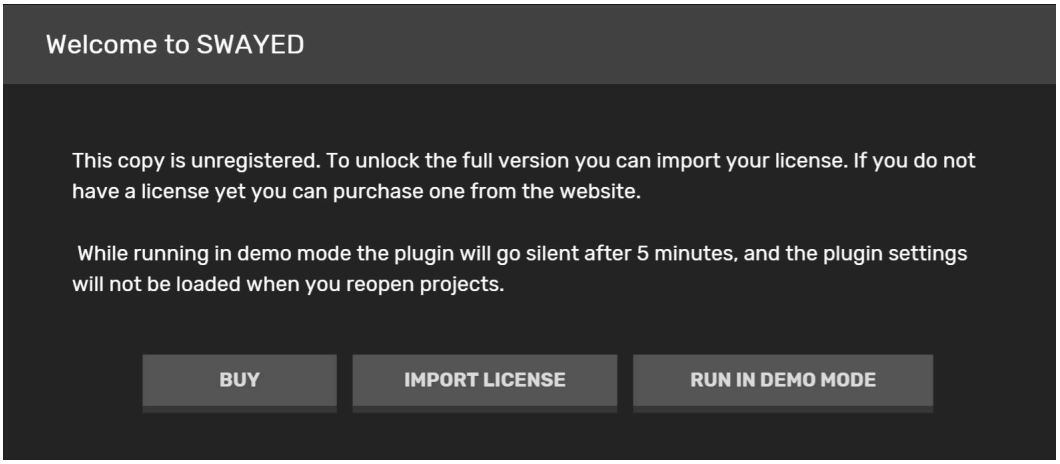
Installing the plugin

You can find the installer on the Stryde Audio website at **strydeaudio.com/swayed**. This installer is the same for the trial version and the full unrestricted version.

On Windows it can be installed as a VST3 plugin, and on macOS as a VST3 or Audio Unit plugin. There is no standalone version.

Launching the plugin

Launch the plugin by inserting it on a track in your DAW. You will be greeted by a window that allows you to start demo mode, or to active your license. If you choose to run in demo mode the plugin will go silent after 10 minutes, and plugin settings will not be saved and loaded when you reopen your project.



Activating your license

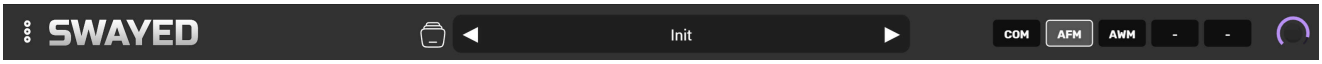
When you purchase a copy of Swayed you will receive a license key file via email. If you lose your license key file you can always download another one on the Stryde Audio website at **strydeaudio.com/my-licenses**.

After you launch the plugin click the **IMPORT LICENSE** button and select your license key file. You can now use Swayed without any restrictions. There is no limit to the number of activations.

OVERVIEW

This chapter gives a quick overview of the user interface.

Upper bar



The top of the user interface shows a toolbar with several controls.

Volume control

On the right we have the output volume control. This setting is preset independent and will persist when changing presets.

Screen navigation

Next to the output volume control is the screen selector. This shows up to 5 screens you can navigate to. The **COM** screen allows you to change settings that apply to the patch as a whole. Next to **COM** are up to 4 element screen buttons. Patches consist of up to 4 separate elements that are layered. These buttons can show **AFM** for a FM synthesis element, **AWM** for a sample synthesis element, or **DRUM** for a drum voice.

Right click on these buttons to solo/mute individual elements, and to copy/paste them.

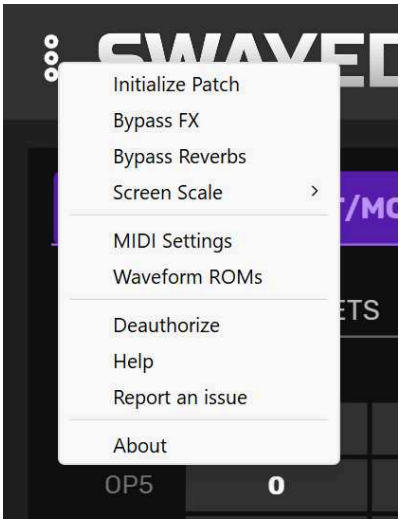
Preset selection

In the middle are the preset selection controls. The text block in the middle shows the name of the currently loaded preset. Clicking it will open a popup where you can select another preset from the currently filtered list of presets. Filtering will be discussed in the chapter on the preset manager.

The triangular shaped buttons allow you to select the previous and next preset from this filtered list.

The cabinet icon opens up the preset manager.

Menu



On the far left of the toolbar you will find a button with 3 dots. Clicking it will open a menu with several options.

- **`Initialize Patch`**: Resets to a new patch.
- **`Bypass FX`**: Sets the plugin in FX bypass mode. This setting is independent of presets and is not saved with a preset. Useful if you already know you want to use your own effects and want to browse through presets, hearing how they sound dry.
- **`Bypass Reverbs`**: Same as FX bypass mode, except it will only bypass reverb effects. The reverb effects are inspired by the ones in the original hardware, and are quite lofi. Useful if you want to use your own higher quality reverbs but keep the other type of effects.
- **`Screen Scale`**: Select a zoom scale for the screen.
- **`Waveform ROMs`**: Opens the screen that shows the currently loaded waveform.
- **`Authorize/deauthorize`**: Activate or deactivate your license.
- **`Manual`**: Opens this manual.
- **`Report an issue`**: Opens a web browser to a bug report form. If you encounter any issues with the plugin, you can provide information about the error here.
- **`About`**: Shows an info box with credits.

Lower bar

The bottom of the user interface shows another bar with some controls and information.

The voices indicator shows how many voices are currently active, as well as the maximum number of voices for this preset. When notes are pressed and all voices are active, the older voices will be stolen.

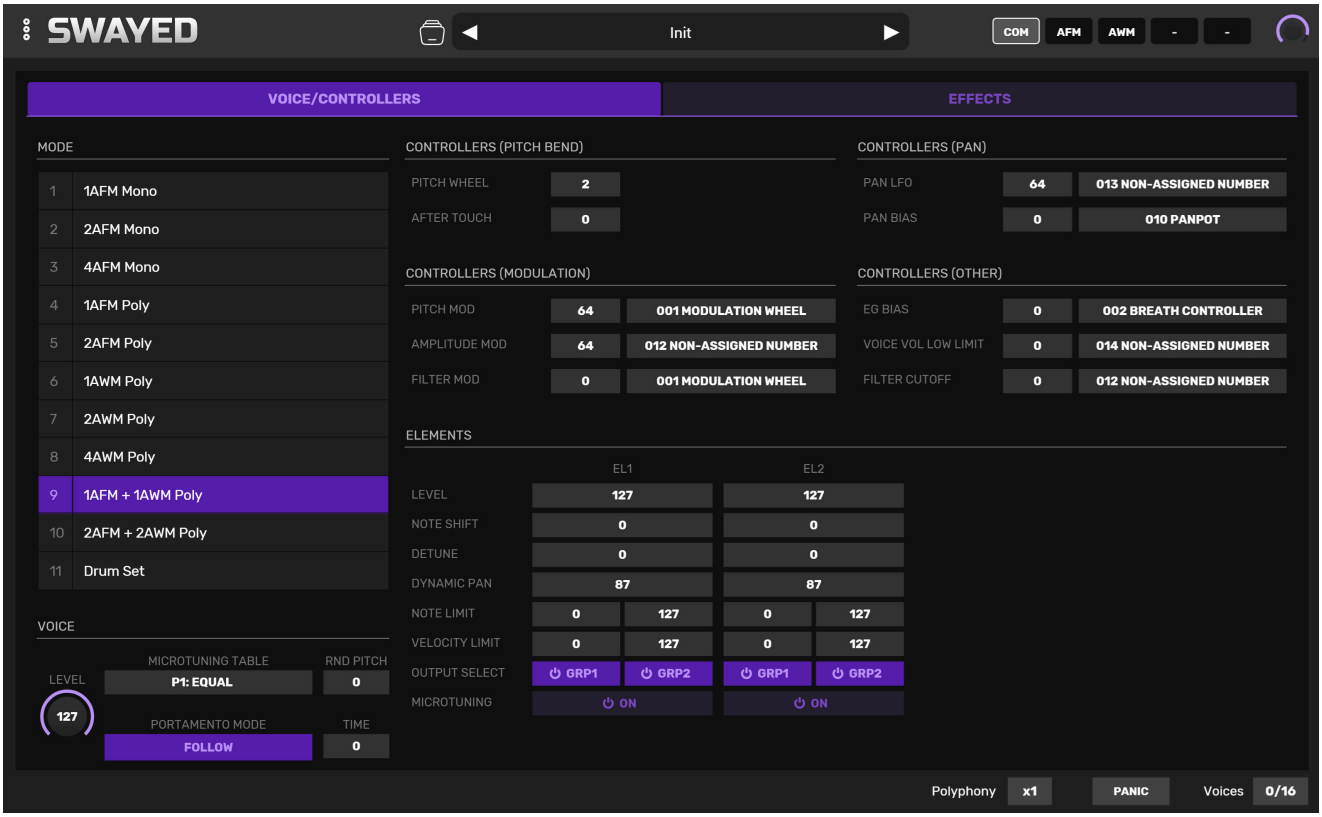
Pressing the **`PANIC`** button kills are currently playing voices. This can be useful when you have notes that are hanging because of a maximum release time.

The polyphony button allows you to change the maximum number of voices. In the original hardware the number of voices was determined by how many elements the preset contains. 1 element patches have 16 voices, 2 element patches have 8 or 16 voices (depending on the element type), and 4 element patches have 4 or 8 voices. This behaviour is replicated in Swayed. With the polyphony multiplier you can multiply this number to get extra polyphony. For example, if the max number of voices for your current preset is 16, then setting polyphony to x2 will increase that to 32.

You can resize the GUI to any size you like by click dragging the bottom right of the screen. Note for Logic users: you need to drag from the bottom right corner of the GUI, not from the corner of the actual plugin window.

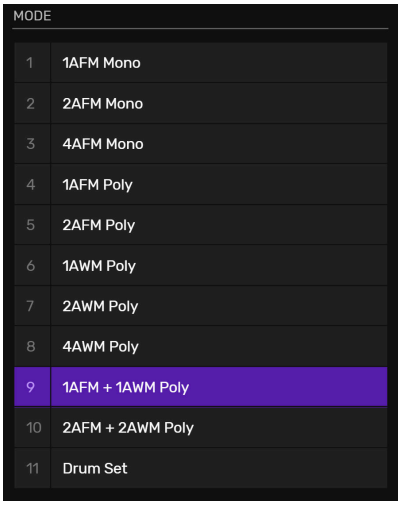
VOICE / CONTROLLERS SCREEN

This chapter explains the voice/controllers tab of the **COM** screen. To access this screen, click on the **COM** button in the upper toolbar, and then select the **VOICE/CONTROLLERS** tab.



Voice setting sections

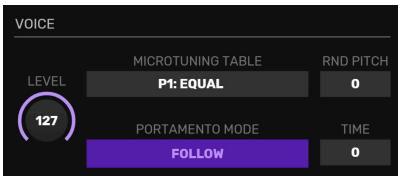
Mode



The left part of the screen shows the **MODE** section. There are 11 options. The voice mode determines how many elements (layers) there are, their type, and if the patch is monophonic or polyphonic. Click on one of the options to change the voice mode. Elements will be added or subtracted to your patch.

- **1AFM Mono**: 1 FM synthesis element, monophonic
- **2AFM Mono**: 2 FM synthesis elements, monophonic
- **4AFM Mono**: 4 FM synthesis elements, monophonic
- **1AFM Poly**: 1 FM synthesis element, polyphonic
- **2AFM Poly**: 2 FM synthesis elements, polyphonic
- **1AWM Poly**: 1 sample synthesis element, polyphonic
- **2AWM Poly**: 2 sample synthesis elements, polyphonic
- **4AWM Poly**: 4 sample synthesis elements, polyphonic
- **1AFM+1AWM Poly**: 2 elements: 1 FM and 1 sample, polyphonic
- **2AFM+2AWM Poly**: 4 elements: 2 FM and 2 sample, polyphonic
- **Drum Set**: special voice type for drum kits

Voice



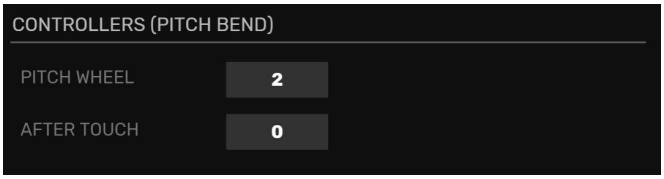
The bottom left part of the screen shows the **VOICE** section. This section contains the following controls:

- **LEVEL**: Sets the volume of the patch.
- **MICROTUNING TABLE**: Sets the microtuning preset.
- **RND PITCH**: Sets the level of random pitch drift on a note. The elements of a patch each have a different pitch deviation. This causes subtle detuning within the patch.
- **PORTAMENTO MODE**: There are 3 different portamento modes. In polyphonic mode the portamento is always set to Follow. In monophonic mode the portamento can be fingered or full time. Fingered = portamento is only applied when playing legato. Full time = portamento is applied to every note.
- **PORTAMENTO TIME**: The glide time between two notes connected by portamento.

Controller sections

The controller sections allow you to bind CC controllers to voice parameters. For each controller you can pick the CC number and the modulation depth.

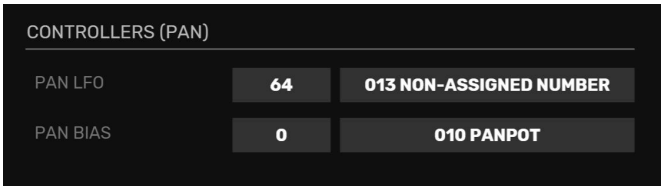
Controllers (Pitch Bend)



The top middle part of the screen shows the **CONTROLLERS (PITCH BEND)** section. This section contains the following controls:

- **PITCH WHEEL**: The range of the pitch bend wheel in semitones.
- **AFTER TOUCH**: The number of semitones that after touch will affect the pitch.

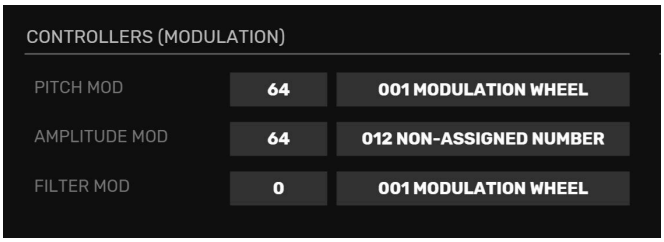
Controllers (Pan)



The top right part of the screen shows the **CONTROLLERS (PAN)** section. This section contains the following controls:

- **PAN LFO**: This setting determines how much the specified controller will add to the depth of the panning LFO.
- **PAN BIAS**: This setting determines how much the specified controller will affect the pan position.

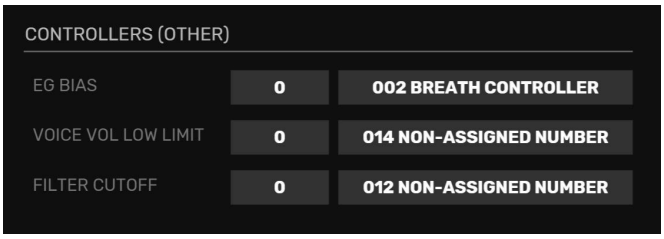
Controllers (Modulation)



The middle part of the screen shows the **CONTROLLERS (MODULATION)** section. This section contains the following controls:

- **PITCH MOD**: This setting determines how much the specified controller will add to the vibrato depth.
- **AMPLITUDE MOD**: This setting determines how much the specified controller will add to the tremolo depth.
- **FILTER MOD**: This setting determines how much the specified controller will add to the wah-wah depth.

Controllers (Other)

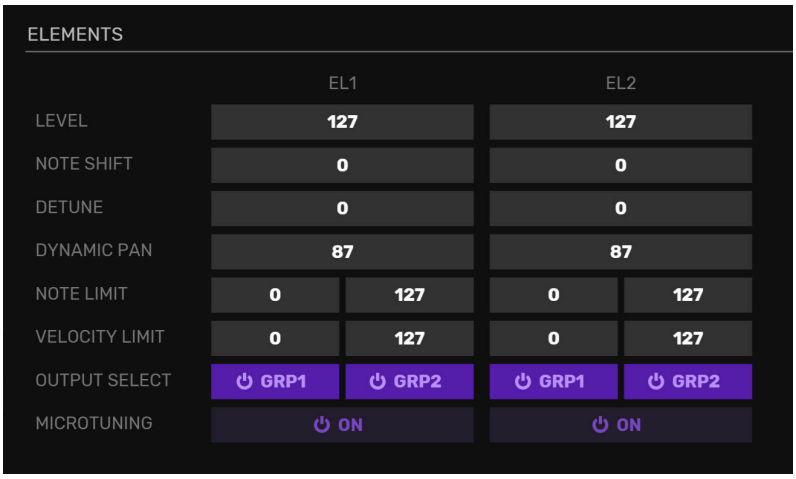


The middle right part of the screen shows the **CONTROLLERS (OTHER)** section. This section contains the following controls:

- **EG BIAS**: This setting determines how much the specified controller will add bias to the amplitude envelope level points.
- **VOICE VOL LOW LIMIT**: This setting determines how much the specified controller can lower the volume. If this is set to 0, the controller can make the voice completely silent. If set to, 100 the controller can reduce the volume somewhat. When set to 127 the controller will have no effect.

- **FILTER CUTOFF**: This settings determines how much the specified controller can increase the filter cutoff frequency of each element. If this is set to 0, the controller has no effect. If this is set to 127 the controller can open up the filters entirely.

Elements



The middle bottom part of the screen shows the **ELEMENTS** section. There is a column for every element in the voice. The columns contain settings that are also available on the element page. The following parameters can be adjusted here:

- **LEVEL**: The volume of the element.
- **NOTE SHIFT**: The transposition of the element (in semitones).
- **DETUNE**: Fine tuning of the element pitch.
- **DYNAMIC PAN**: The panning preset for the element.
- **NOTE LIMIT**: The range of notes that can play the element.
- **VELOCITY LIMIT**: The range of velocities that can play the element.
- **OUTPUT SELECT**: The active output groups for the element.
- **MICROTUNING**: Toggle that determines if the element responds to the microtuning preset for the voice.

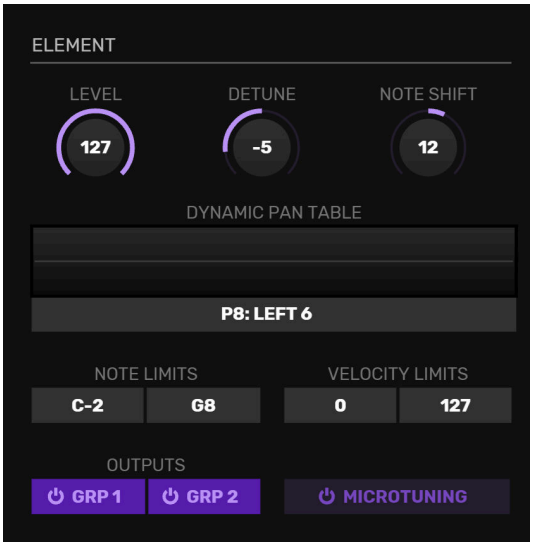
AFM SCREEN

This chapter explains the **AFM** screen. To access this screen, load a patch that contains AFM elements and select one of them in the upper toolbar.

Element/Modulations Tab

This tab shows the general element settings. This tab is almost equivalent to the one for AWM elements. This tab contains the following sections.

Element



This section contain the basic settings for the element, relating to volume, pitch, panning and note range.

We find the following controls:

- **LEVEL**: The volume of the element.
- **NOTE SHIFT**: The transposition of the element (in semitones).
- **DETUNE**: Fine tuning of the element pitch.
- **DYNAMIC PAN**: The panning preset for the element.
- **NOTE LIMIT**: The range of notes that can play the element.
- **VELOCITY LIMIT**: The range of velocities that can play the element.
- **OUTPUTS**: The active output groups for the element.
- **MICROTUNING**: Toggle that determines if the element responds to the microtuning preset for the voice.

Dynamic pan

The dynamic pan preset configures the stereo action of the element. It can combine the following elements to influence the pan position:

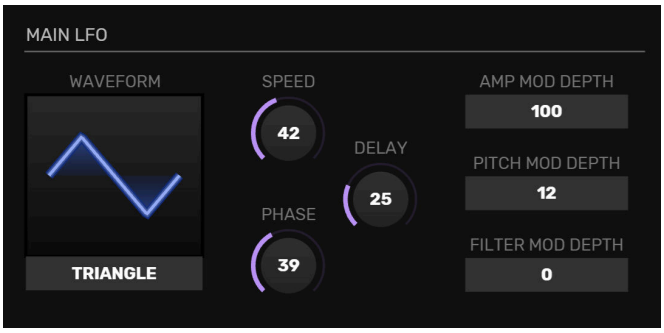
- An envelope
- A LFO
- Velocity
- Note number

There are 64 dynamic pan presets, and 2 internal slots. The 64 presets are fixed, but the internal slots are meant to be defined by the user. The current version of Swayed does not yet have an editor for panning presets, but if you load a SysEx bank that contains patches that refer to an internal panning preset, *and* if the panning preset is also stored in the SysEx file, then it will be properly loaded to its internal slot. If you save such a preset to your library, the panning information is stored along with it.

Outputs

Different output groups can have different effect processing applied. It can make sense to only send some elements to group 1 and others to group 2. For more information, see the chapter on effects.

Main LFO



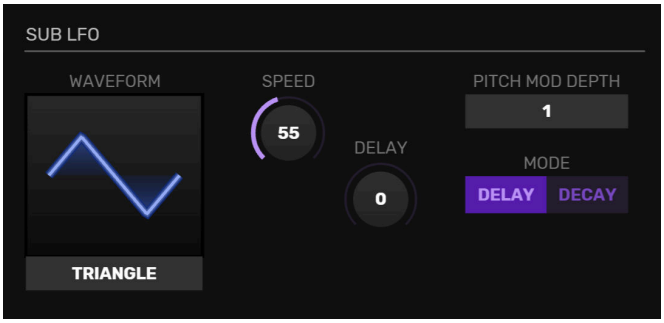
The main LFO generates a control signal that can be used to modulate an operator volume, operator pitch, and filter cutoff of the element.

The main LFO has the following settings:

- **WAVEFORM**: The waveform of the LFO (triangle/saw down/saw up/square/sine/sample & hold). The sample & hold waveform generates a random signal that changes at an interval determined by the speed setting.
- **SPEED**: The speed of the LFO.
- **DELAY**: The time delay before the LFO starts modulating.

- **PHASE**: The initial phase of the LFO. This determines at which point in the waveform the LFO starts when a note is played.
- **AMP MOD DEPTH**: Determines how much the LFO modulates the output level of the operators. The modulation depth for a specific operator also depends on its amplitude modulation sensitivity setting. It needs to be higher than 0 for there to be any modulation at all. Refer to the section of modulation targets for more information.
- **PITCH MOD DEPTH**: Determines how much the LFO modulates the pitch of the operators. The modulation depth for a specific operator also depends on its pitch modulation sensitivity setting. It needs to be higher than 0 for there to be any modulation at all. Refer to the section of modulation targets for more information.
- **FILTER MOD DEPTH**: Determines how much the LFO modulates the filter cutoff of the element. The modulation depth also depends on the filter modulation sensitivity setting. It needs to be higher than 0 for there to be any modulation at all. Refer to the section of modulation targets for more information.

Sub LFO



The sub LFO is an additional LFO that only affects the pitch of the entire element. It is unaffected by the pitch modulation sensitivity setting.

The sub LFO has the following settings:

- **WAVEFORM**: The waveform of the LFO (triangle/saw down/square/sample & hold). The sample & hold waveform generates a random signal that changes at an interval determined by the speed setting.
- **SPEED**: The speed of the LFO. The speed for the sample & hold waveform is higher than for the other waveforms.
- **MODE**: When set to the delay, the sub LFO will not start modulating until an amount of time has passed. The duration of this delay is set by the delay parameter. When set to decay, the sub LFO will start modulating immediately, but the modulation depth will decrease over time. The time it takes until the modulation depth reaches zero is set by the delay setting.
- **DELAY**: The delay or decay time, depending on the **MODE** setting.
- **PITCH MOD DEPTH**: The depth of the vibrato effect produced by this LFO.

Pitch Envelope

The pitch envelope can produce a change in pitch over time. The pitch offset and rate per segment can be modified by dragging the control points in the graph.

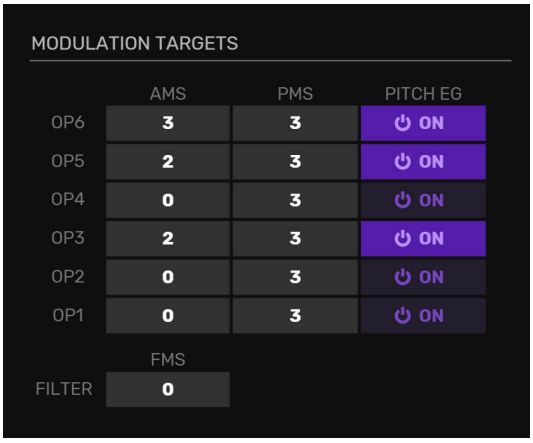
It's important to note that the graph does not accurately depict the pitch over time. The speed at which an envelope segment is traversed is determined by both the change in offset as well as the rate. Lower rates take exponentially more time, but are only linearly displaced horizontally, to make sure editing remains comfortable. In other words: doubling the horizontal distance between two control points does not mean it takes exactly two times as long to traverse that part of the envelope.

Double click a control point to reset the offset to 0. Right clicking the graph opens a popup menu with the option to reset the envelope, and options to copy or paste the envelope.

Other settings that affect the pitch envelope:

- **RATE SCALE**: Sets the keyboard rate scaling. Positive values mean higher notes will make the envelope go faster. Negative values mean higher notes will make the envelope go slower. When set to 0 the envelope speed is unaffected by the note.
- **RANGE**: The maximum range over which the pitch changes occurs (8 octaves, 2 octaves, 1, octave, 1/2 octave).
- **VEL SW**: Velocity switch. When turned on, higher velocities will cause the pitch change to be over a higher range.

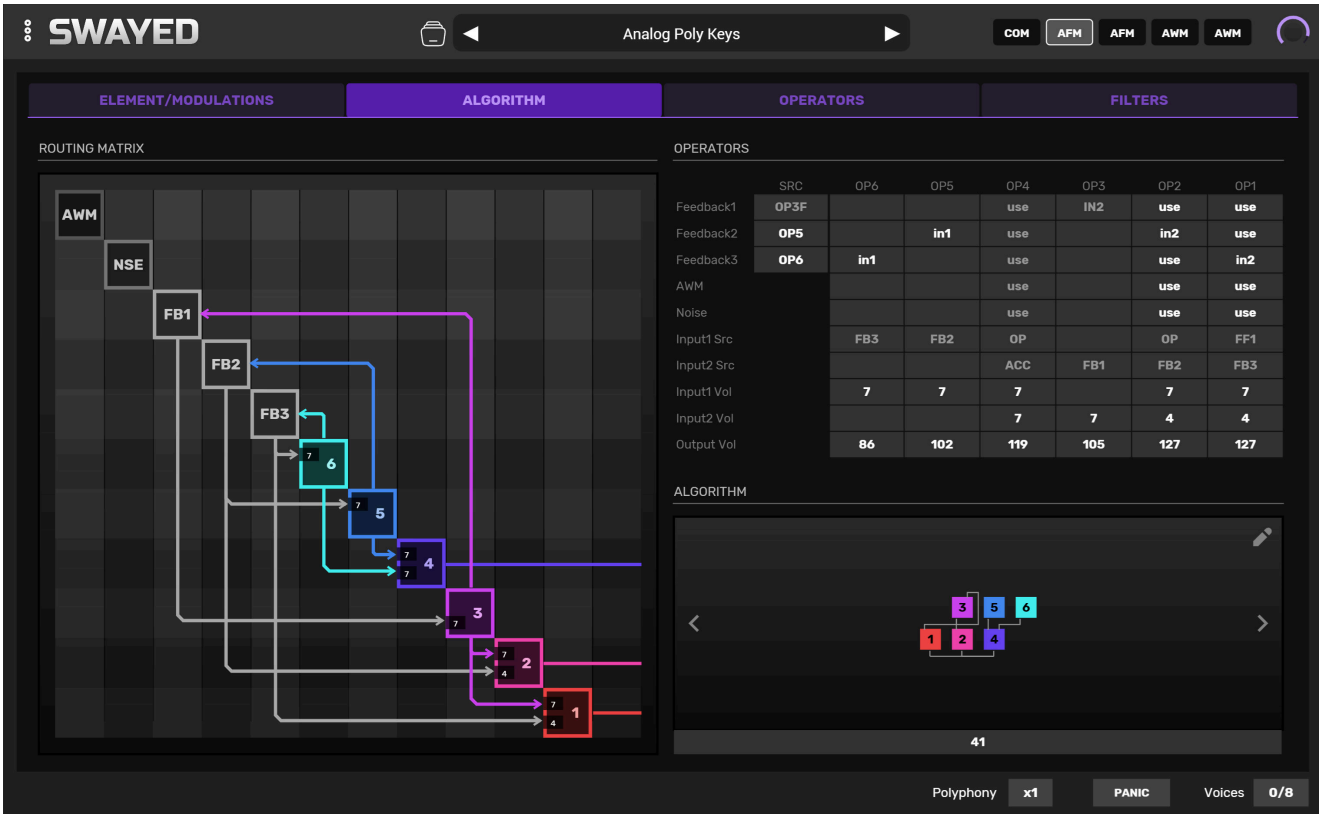
Modulation targets



This section contains settings that affect the depth of the modulations produced by the main LFO, the pitch EG and the controllers that are set up in the voice/controllers screen.

- **AMS**: Amplitude modulation sensitivity. This setting determines how much a specific operator is affected by amplitude modulation.
- **PMS**: Pitch modulation sensitivity. This setting determines how much a specific operator is affected by pitch modulation.
- **FMS**: Filter modulation sensitivity. This setting determines how much the filter cutoffs of this element are affected by filter modulation.
- **PITCH EG**: The pitch envelope can be switched on or off per operator.

Algorithm Tab



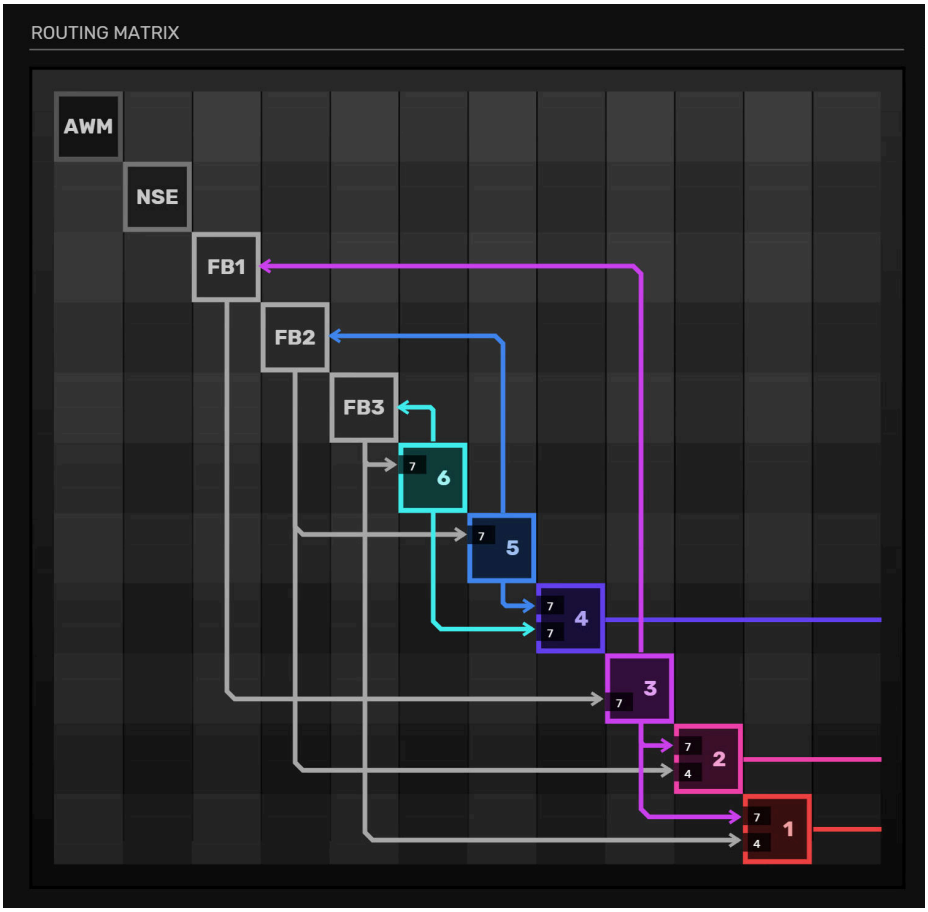
Algorithm

The algorithm control shows the **default** layout of the currently selected algorithm. It's important to note that adding additional feedback paths does not update the gray connection lines that are shown in this image.

Use the arrow buttons to select the previous or next algorithm. There are 45 algorithms in total. Alternatively you can draw the algorithm label at the bottom to quickly scroll through the algorithms.

If you tap the pencil icon in the top right coner you will enter freeform edit mode. This allows you to completely reconfigure the algorithm. This will be explained in the freeform section.

Routing matrix



The routing matrix shows the actual connections of modulators to carriers, including the adjustments made to the algorithm by setting feedback paths or by going into freeform algorithm edit mode. There is a box for each operator, a box for each feedback bus, a box for the **AWM** input, and a box for the noise input. This control is for visualization only, and cannot be used to actually change connections. Connections are changed through the operator table on the right of the screen.

The arrows that go downwards and to the right signify connections from modulators to carriers. The small numbers next to the arrow heads show the input volume for that connection. In the example you can see that operator 6 and 5 modulate operator 4. and the operator 3 modulates operator 2 and 1. Additionally, feedback bus 1 modulates operator 3, feedback bus 2 modulates operator 5 and operator 2, and feedback bus 3 modulates operator 6 and operator 1.

The arrows that go upwards and to the left signify connections from operators to the feedback bus. Such a connection means the operator is the source of the feedback path. In the example it's shown that operator 3 is the source foor feedback path 1,

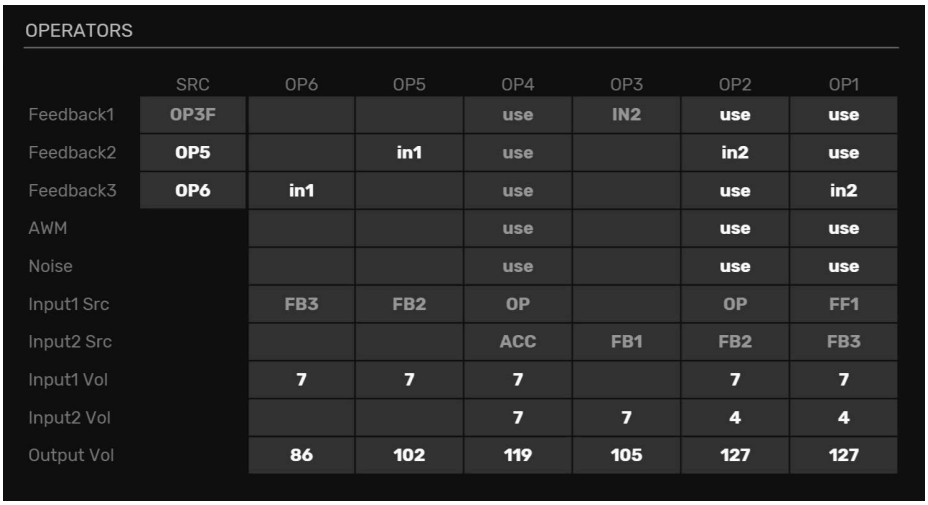
operator 5 is the source for feedback path 2, and operator 6 is the source for feedback path 3.

The lines the go from an operator to the right edge of the matrix signify output connections. The operators with these lines are carrier operators that are summed to the output bus.

If there is a connection from AWM to an operator, that means the output of an **AWM** (sample) element (if present) will modulate the operator. In a **1AFM & 1AWM** voice the only **AWM** element (element 2) is connected to the only **AFM** element (element 1) in this way. In a **2AFM & 2AWM** voice the first **AWM** element (element 3) is connected to the first **AFM** element (element 1), and the second AWM element (element 4) is connected to the second **AFM** element (element 2).

If there is a connection from **NSE** to an operator, that means a white noise signal will modulate the operator.

Operators



	SRC	OP6	OP5	OP4	OP3	OP2	OP1
Feedback1	OP3F			use	IN2	use	use
Feedback2	OP5		in1	use		in2	use
Feedback3	OP6	in1		use		use	in2
AWM				use		use	use
Noise				use		use	use
Input1 Src		FB3	FB2	OP		OP	FF1
Input2 Src				ACC	FB1	FB2	FB3
Input1 Vol		7	7	7		7	7
Input2 Vol				7	7	4	4
Output Vol		86	102	119	105	127	127

The operators table allows you to actually configure the algorithm. The structure of this table resembles the interface of the original hardware. The **Feedback/SRC** cells connect operators to feedback buses. For example, if **Feedback2/SRC** is set the **OP5**, it means feedback bus 2 sources its feedback from operator 5. If it says **OP5F** that means the setting is fixed by the algorithm and cannot be changed.

The **Feedback/OPn** cells allow you to route feedback paths into the operator. Select **in1** to route it to input1 and **in2** to route it to input 2. The **AWM/OPn** cells and **Noise/OPn** cells work similarly.

Each operator has a maximum of 2 input sources for modulation. The active input sources are shown in the **Input1Src/OPn** cells and **Input2Src/OPn** cells. These cells are readonly. The content is determined by the configuration of the selected algorithm and the customized feedback/AWM/noise paths. The possible values are:

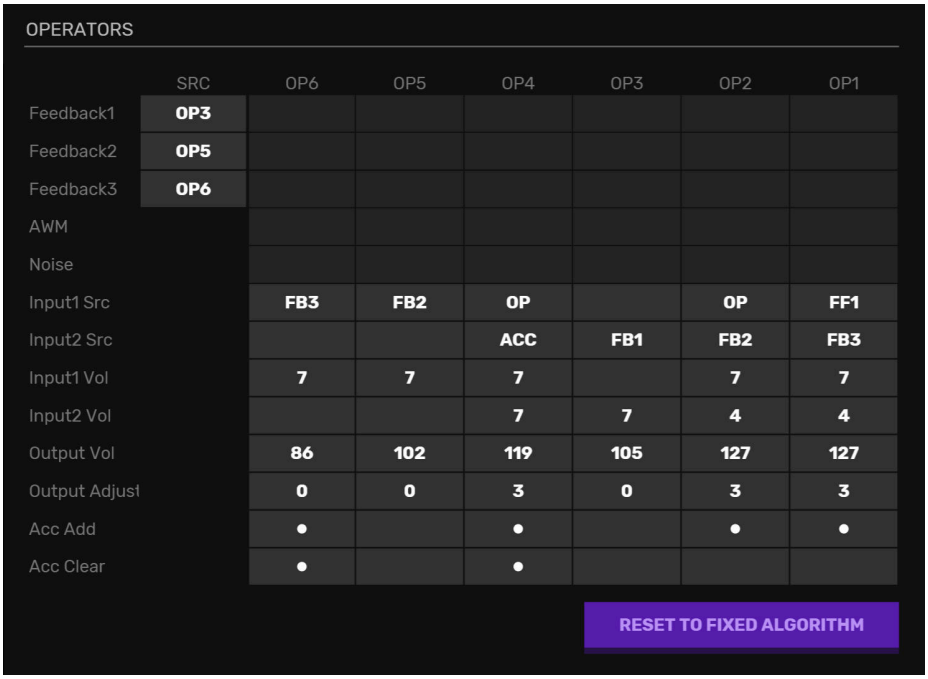
- **OP**: Receives the output of the previous operator.
- **ACC**: Receives the output of the accumulation buffer.
- **AWM**: Receives the output of the AWM element.
- **NSE**: Receives the output of the noise generator.
- **FB1/FB2/FB3**: Receives the output of a feedback bus.

- **FF1/FF2/FF3**: Receives the output of a feedforward bus. If operator 6 is the source for feedback path 3, then this output is already available in the current sample frame to operators 5 to 1, as a feedforward connection (before it actually becomes feedback). Some algorithms utilize this to have three or more operators modulating one carrier.

The strength of an input modulation can be tweaked by setting **Input1Vol/OPn** and **Input2Vol/OPn** cells. 0 means no modulation, 7 means maximum modulation.

Finally, the **OutputVol/OPn** cells set the output volume of the operators. For carrier operators it is a way to control the output volume, and for modulation operators it is a way to control the timbre. If a modulator operator is at a lower volume it will create less FM modulation on its carriers.

Freeform edit mode



Tap the pencil icon on the algorithm display to enter freeform edit mode. When doing so the operators tabel will change. Three new rows of cells appear, and the **Input1Src/OPn** and **Input2Src/OPn** cells are now directly editable. You use these cells to route in feedback paths, **AWM** or **NSE**, but you can also change them to the other values: **OP**, **ACC**, **FF1/FF2/FF3**. Consult the routing matrix to see how your changes affect the actual routing. Since you can now directly configure the inputs, it's no longer necessary to use the **Feedback/OPn**, **AWM/OPn** and **Noise/OPn** cells to route feedback. These cells are therefore emptied and disabled in freeform edit mode.

OutputAdjust/OPn cells allow you to further reduce the volume of operators. These are used by default to even out volumes between algorithms with different number of output operators. The algorithm with 6 outputs would otherwise sound much louder than the algorithm with 1 output, for example. Higher values mean a lower volume.

AccAdd/OPn cells indicate if an operator should write its value to the accumulator.

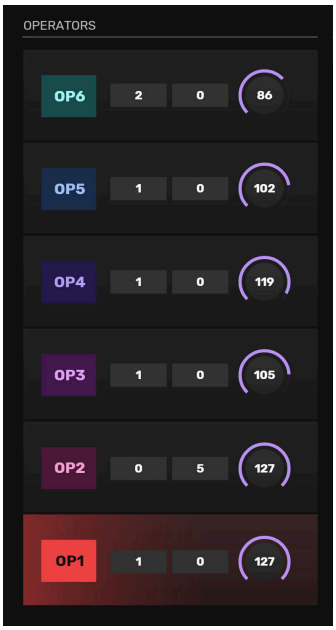
AccClear/OPn cells indicate if the accumulator should be cleared at that operator stage.

It's recommended to just experiment a bit with these cells and watch the routing matrix to see how it all fits together.

Operators Tab

The operators tab offers controls to adjust the settings of the operators, giving you control over their level, timbre, envelope, key scaling and modulation sensitivity. The following sections explain each part of the user interface in detail.

Operators



The operators section on the left allows you to select an operator for further editing with the other controls. The currently selected operator is highlighted. It also allows you to quickly adjust the operator level and ratio without having to select it first. The round knob sets the level, and the two number fields set the ratio. The left field represents the whole number and the right field is the fraction.

Envelope

The envelope section allows you to shape the level of the selected operator over time. The level and rate per segment can be modified by dragging the control points in the graph. The control point labeled **R** is the release point. The segments after that affect the operator level after releasing the note.

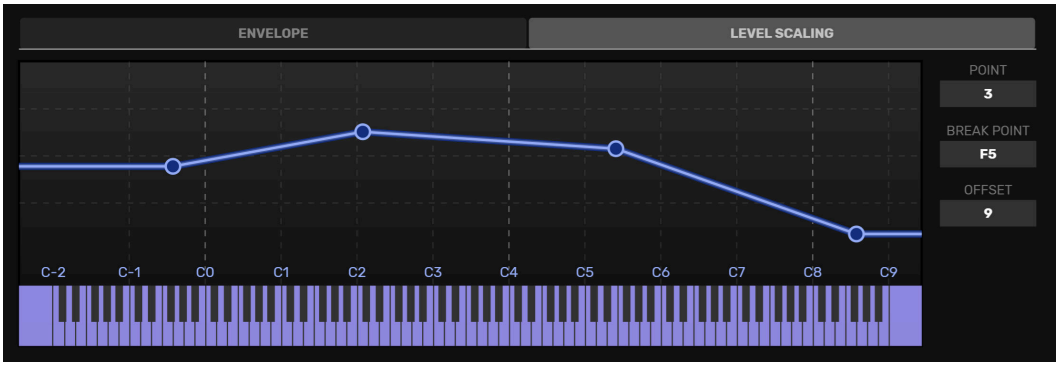
Just as with the pitch envelope, it's important to note that the graph does not accurately depict the level over time. Lower rate values are exponentially slower, but the horizontal rate offsets of control points are scaled linearly for easy editing.

Double click a control point to reset the offset to 0. Right clicking the graph opens a popup menu with the option to reset the envelope, and options to copy or paste the envelope.

Other settings that affect the operator envelope:

- **RATE SCALE**: Sets the keyboard rate scaling. Positive values mean higher notes will make the envelope go faster. Negative values mean higher notes will make the envelope go slower. When set to 0 the envelope speed is unaffected by the note.
- **LOOP**: Sets the loop start point. The envelope will loop from the release point back to this point. By default its set at the release point (point 3), indicating no looping but a sustain at the release level. The selected loop control point has an **L** label. The highest rate values can make the envelope loop at audio rate speeds, allowing for some interesting timbres.

Level Scaling



The level scaling section allows you to change the level of the operator across the note range. For example, if you want a sound to be less bright as you play higher up the keyboard, you can turn down the level for the higher keys on a modulating operator.

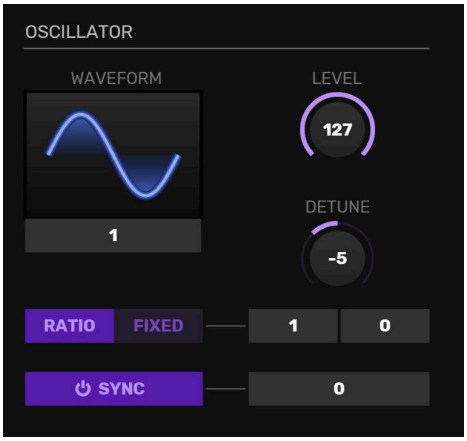
There are 4 break points to be placed. The horizontal location determines the note, the vertical location the adjustment in the level, which can be positive or negative.

This number is simply added to the operator level. The final calculated operator level can never exceed 127 or go below 0. The graph shows a line between two breakpoints that indicates what the offset will be at each note on the keyboard.

Double click a breakpoint to reset the adjust back to 0. Right clicking the graph opens a popup menu with the option to reset the level scaling, and options to copy or paste the level scaling.

When you play a note, a vertical bar will be briefly shown, indicating where in the contour the played note falls.

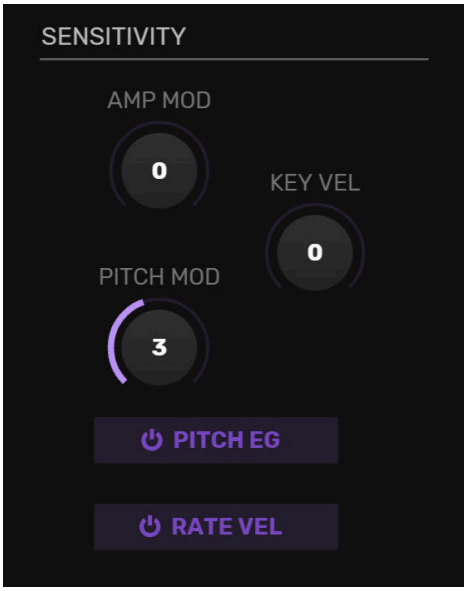
Oscillator



The oscillator section allows you to change the basic properties of an operator. It contains the following controls:

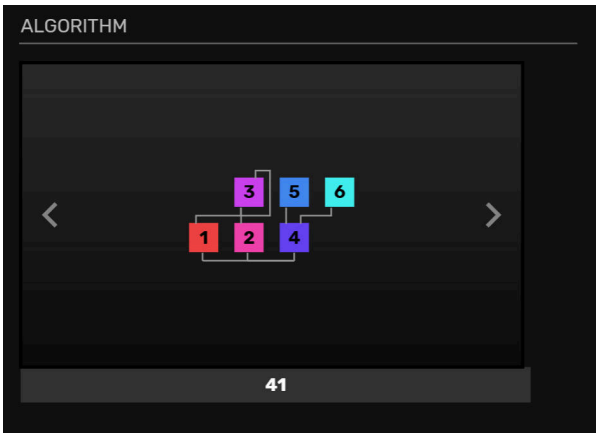
- **WAVEFORM**: The waveform of the operator. You can choose from 16 shapes.
- **LEVEL**: The output level of the operator.
- **DETUNE**: Finely tuned pitch adjustment.
- **RATIO**: When set to ratio, the operator will play at a pitch that is determined by the played note and the ratio. It takes the frequency of note and multiplies it with the ratio. The ratio itself can be set with the values next to it. There is a coarse and a fine control, combining to a ratio. So if it says 1 and 0 the ratio is 1.0. If it says 2 and 23 the ratio is 2.23.
- **FIXED**: When set to fixed, the operator will play at a fixed frequency, no matter what note you play. The frequency you can set ranges from 0 Hz to 9762 Hz. Drag the left half of the frequency label for coarse adjustments, and the right half for fine adjustments.
- **SYNC**: When phase sync if on the operator phase will be reset each time a new note starts. You can set this phase to reset by dragging the number next to it. 0 means a 0 degree offset, 127 (the maximum) means a 360 degree offset.

Sensitivity



- **AMP MODE**: Amplitude modulation sensitivity. This setting determines how much the operator is affected by amplitude modulation.
- **PITCH MOD**: Pitch modulation sensitivity. This setting determines how much the operator is affected by pitch modulation.
- **KEY VEL**: Key velocity sensitivity. This setting determines how much the level of the operator is scaled by the velocity.
- **PITCH EG**: If switched on, this operator is affected by the pitch envelope. Otherwise, the pitch envelope is ignored by this operator.
- **RATE VEL**: If switched on, the rate of the first segment of the operator envelope will be scaled by the key velocity. How much this rate is scaled depends on the key velocity sensitivity setting. At positive settings, stronger played notes will result in a faster attack. At negative settings, stronger played notes will result in a slower attack.

Algorithm

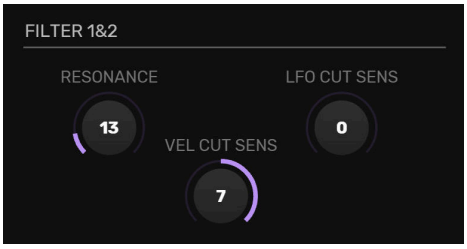


The algorithm control is the same as the one on the algorithms tab, with the exception that you cannot enter freeform edit mode from here. Use the arrow buttons to select the previous or next algorithm, or drag the number label at the bottom to quickly scroll through algorithms.

Filters Tab

This tab allows you to set up the filters for the element. There are two 12dB/oct filters available. The first of the two can be a low pass or high pass filter, the second one can only be a low pass filter.

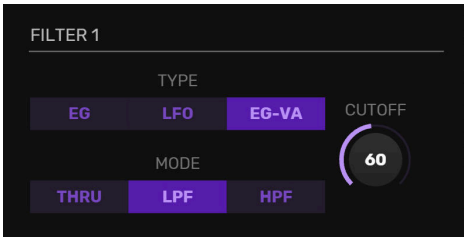
Filter 1&2



This section contains some controls that apply to both filters.

- **RESONANCE**: The resonance of the filters. Only the low pass filters can be resonant. If filter 1 is set to highpass mode this setting has no effect at all, not even on low pass filter 2, if it's active.
- **VEL CUT SENS**: This setting controls how much the key velocity will affect the cutoff frequency. For positive settings, higher velocities increase the cutoff. For negative settings, higher velocity decrease the cutoff.
- **LFO CUT SENS**: This determines how much the filter cutoff is affected by the filter modulation from the main LFO. It also determines how much the filter cutoff is affected by the contoller assigned to filter cutoff.

Filter 1



This section contains controls that only affect the first filter.

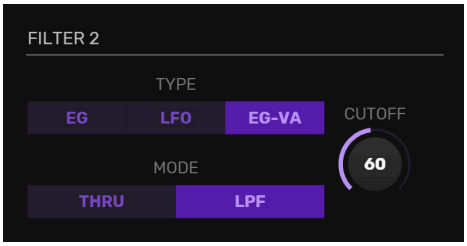
- **MODE**: Sets the filer mode. **THRU** means the filter is bypassed. **LPF** sets it to a low pass filter. **HPF** to a high pass filter.
- **CUTOFF**: Sets the cutoff value (0Hz to 22430 Hz for **LPF**, 0Hz to 11660Hz for **HPF**).
- **TYPE**: The filter control type.

When **TYPE** is set to **EG** the filter cutoff is determined by the envelope. Key velocity still has effect if its sensitivity is set to anything other than 0. The controllers assigned to the filter modulation depth and filter cutoff will have their value sampled at the note on event, but will not be updated during the note.

The **EG-VA** setting is mostly the same as the **EG** setting, except the key velocity will only affect the rate and target level of the first envelope segment (provided velocity cutoff sensivity is set to a value other than 0). For positive settings of the sensitivity, higher velocities will increase the rate and target level of the first segment. For negative settings the effect is reversed.

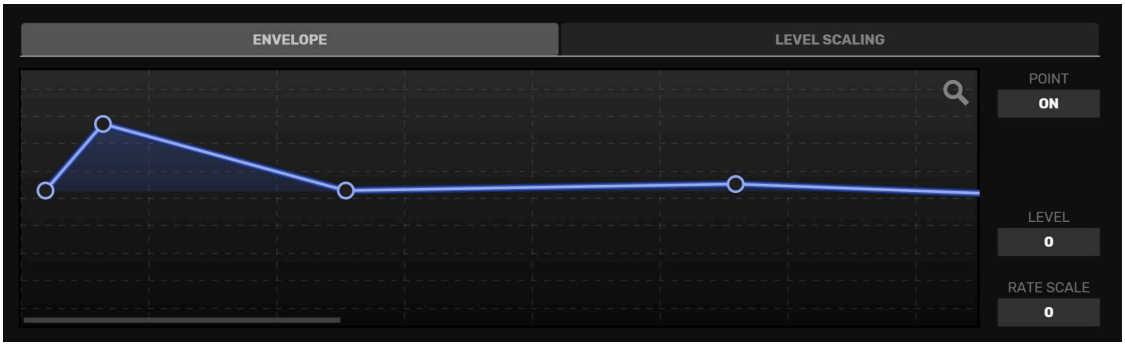
The **LFO** setting will make sure the filter cutoff is controlled by the main LFO modulation, as well as the controller assigned to filter modulation. Unlike the **EG** and **EG-VA** settings, the controllers are effective over the entire duration of the note. Key velocities will still affect the cutoff if the sensitivity is anything other than 0.

Filter 2



The settings for filter 2 are the same as the ones for filter 1, with the exception that the **HPF** mode is not available.

Filter 1/2 Envelope



The envelope section allows you to modify the filter cutoff over time. This requires the filter type to be set to **EG** or **EG-VA** mode. In **LFO** mode the envelope is ignored. Filter 1 and 2 each have their own envelope.

The envelope is bipolar, offsetting the selected filter cutoff from -127 to 127. This offset is labeled as -64 to 63 in the envelope. So if your filter cutoff is 100, and the offset at an envelope point is +3, then the actual cutoff of the filter is 106 at that point. The actual cutoff is always clamped between 0 and 127.

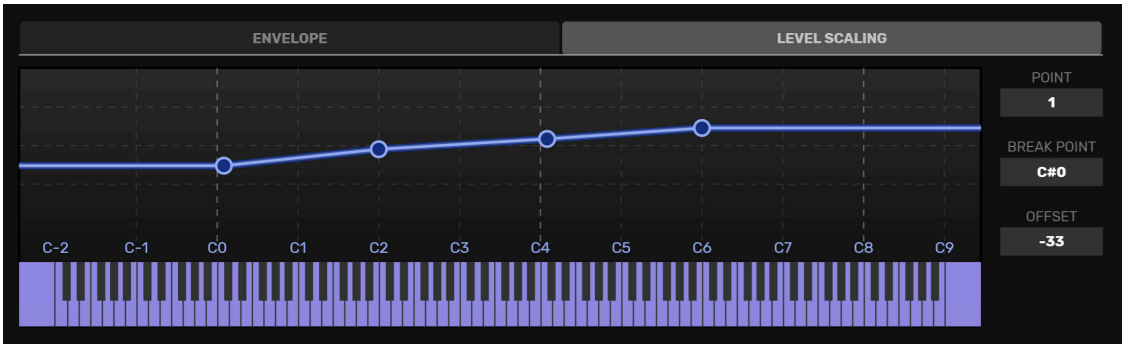
The cutoff offset and rate per segment can be modified by dragging the control points in the graph. The control point labeled R is the release point. The segments after that affect the filter cutoff after releasing the note.

Just as with the other envelopes, it's important to note that the graph does not accurately depict the level over time. Lower rate values are exponentially slower, but the horizontal rate offsets of control points are scaled linearly for easy editing.

Double click a control point to reset the offset to 0. Right clicking the graph opens a popup menu with the option to reset the envelope, and options to copy or paste the envelope.

The rate scale setting allows you to set up note based rate scaling. Positive values mean higher notes will make the envelope go faster. Negative values mean higher notes will make the envelope go slower. When set to 0 the envelope speed is unaffected by the note.

Filter 1/2 Level Scaling



The filter level scaling section allows you to change the cutoff offset of the filter across the note range. For example, if you want a sound to be less bright as you play higher up the keyboard, you can turn down the offset for the higher keys.

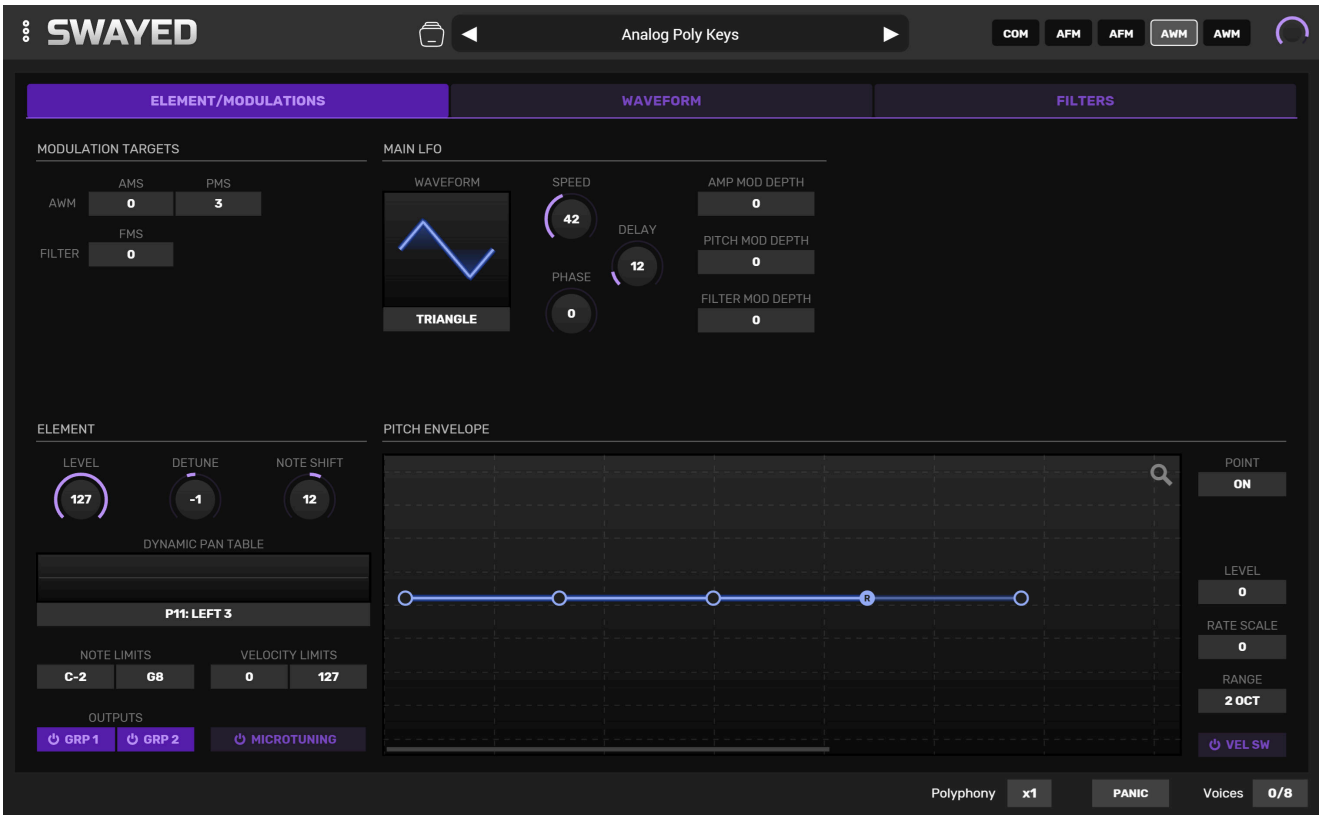
There are 4 break points to be placed. The horizontal location determines the note, the vertical location the adjustment in the cutoff, which can be positive or negative. It ranges from -127 to 127. The graph shows a line between two breakpoints that indicates what the offset will be at each note on the keyboard.

Double click a breakpoint to reset the adjust back to 0. Right clicking the graph opens a popup menu with the option to reset the level scaling, and options to copy or paste the level scaling.

AWM SCREEN

This chapter explains the **AWM** screen. To access this screen, load a patch that contains AWM elements and select one of them in the upper toolbar.

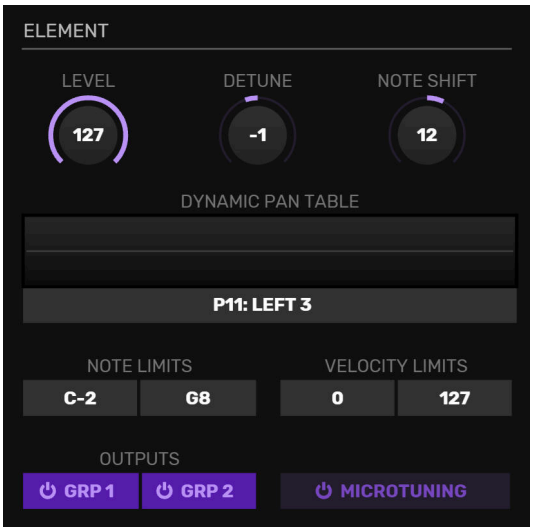
Element/Modulations Tab



This tab shows the general element settings. This tab is almost equivalent to the one for AFM elements. Unlike AFM elements, AWM elements do not have a sub LFO, and the modulation sensitivities target the entire element instead of an operator.

This tab contains the following sections.

Element



This section contain the basic settings for the element, relating to volume, pitch, panning and note range.

We find the following controls:

- **LEVEL**: The volume of the element.
- **NOTE SHIFT**: The transposition of the element (in semitones).
- **DETUNE**: Fine tuning of the element pitch.
- **DYNAMIC PAN**: The panning preset for the element.
- **NOTE LIMIT**: The range of notes that can play the element.
- **VELOCITY LIMIT**: The range of velocities that can play the element.
- **OUTPUTS**: The active output groups for the element.
- **MICROTUNING**: Toggle that determines if the element responds to the microtuning preset for the voice.

Dynamic pan

The dynamic pan preset configures the stereo action of the element. It can combine the following elements to influence the pan position:

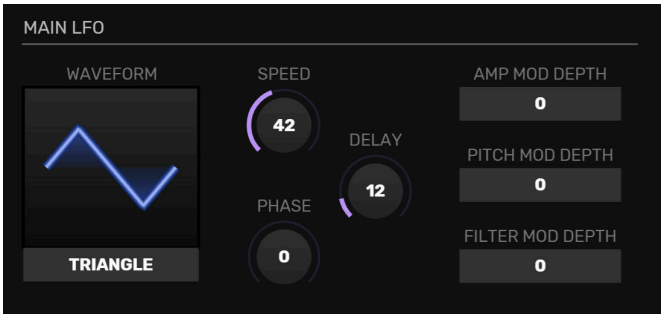
- An envelope
- A LFO
- Velocity
- Note number

There are 64 dynamic pan presets, and 2 internal slots. The 64 presets are fixed, but the internal slots are meant to be defined by the user. The current version of Swayed does not yet have an editor for panning presets, but if you load a SysEx bank that contains patches that refer to an internal panning preset, *and* if the panning preset is also stored in the SysEx file, then it will be properly loaded to its internal slot. If you save such a preset to your library, the panning information is stored along with it.

Outputs

Different output groups can have different effect processing applied. It can make sense to only send some elements to group 1 and others to group 2. For more information, see the chapter on effects.

Main LFO



The main LFO generates a control signal that can be used to modulate the volume, pitch and filter cutoff of the element.

The main LFO has the following settings:

- **WAVEFORM**: The waveform of the LFO (triangle/saw down/saw up/square/sine/sample & hold). The sample & hold waveform generates a random signal that changes at an interval determined by the speed setting.
- **SPEED**: The speed of the LFO.
- **DELAY**: The time delay before the LFO starts modulating.
- **PHASE**: The initial phase of the LFO. This determines at which point in the waveform the LFO starts when a note is played.
- **AMP MOD DEPTH**: Determines how much the LFO modulates the output level of element. The modulation depth depends on its amplitude modulation sensitivity setting. It needs to be higher than 0 for there to be any modulation at all. Refer to the section of modulation targets for more information.
- **PITCH MOD DEPTH**: Determines how much the LFO modulates the pitch of the element. The modulation depth also depends on its pitch modulation sensitivity setting. It needs to be higher than 0 for there to be any modulation at all. Refer to the section of modulation targets for more information.
- **FILTER MOD DEPTH**: Determines how much the LFO modulates the filter cutoff of the element. The modulation depth also depends on the filter modulation sensitivity setting. It needs to be higher than 0 for there to be any modulation at all. Refer to the section of modulation targets for more information.

Pitch Envelope

The pitch envelope can produce a change in pitch over time. The pitch offset and rate per segment can be modified by dragging the control points in the graph.

It's important to note that the graph does not accurately depict the pitch over time. The speed at which an envelope segment is traversed is determined by both the change in offset as well as the rate. Lower rates take exponentially more time, but are only linearly displaced horizontally, to make sure editing remains comfortable. In other words: doubling the horizontal distance between two control points does not mean it takes exactly two times as long to traverse that part of the envelope.

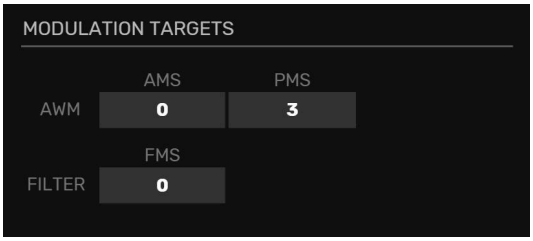
Double click a control point to reset the offset to 0. Right clicking the graph opens a popup menu with the option to reset the envelope, and options to copy or paste the envelope.

Other settings that affect the pitch envelope:

- **RATE SCALE**: Sets the keyboard rate scaling. Positive values mean higher notes will make the envelope go faster. Negative values mean higher notes will make the envelope go slower. When set to 0 the envelope speed is unaffected by the note.

- **RANGE**: The maximum range over which the pitch changes occurs (2 octaves, 1, octave, 1/2 octave).
- **VEL SW**: Velocity switch. When turned on, higher velocities will cause the pitch change to be over a higher range.

Modulation targets



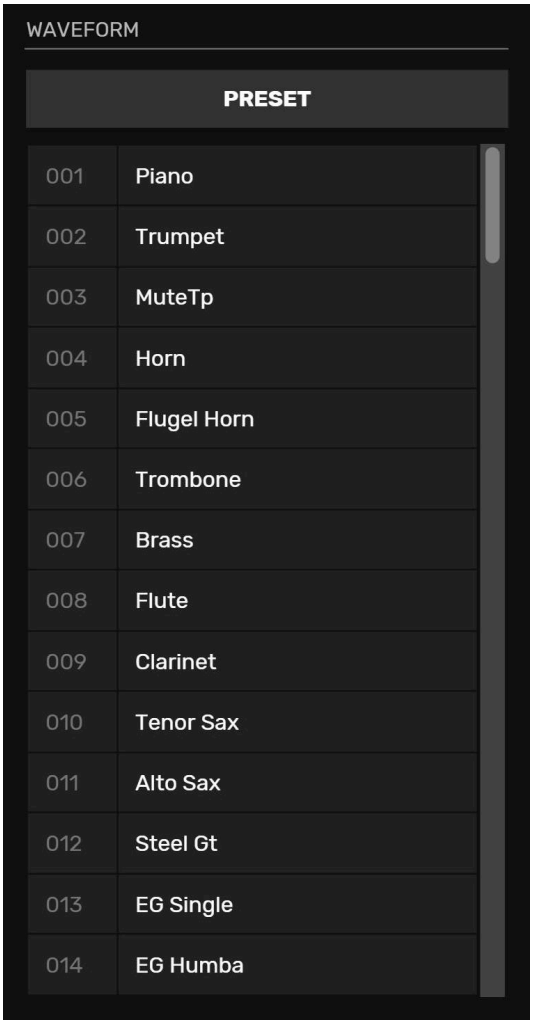
This section contains settings that affect the depth of the modulations produced by the main LFO, the pitch EG and the controllers that are set up in the voice/controllers screen.

- **AMS**: Amplitude modulation sensitivity. This setting determines how much this element is affected by amplitude modulation.
- **PMS**: Pitch modulation sensitivity. This setting determines how this element is affected by pitch modulation.
- **FMS**: Filter modulation sensitivity. This setting determines how much the filter cutoffs of this element are affected by filter modulation.

Waveform Tab

The waveform tab allows you to set up the sample that's used for playback.

Waveform



The waveform section contains the list with waveforms to choose from. First, in the top of the section there is a control to select a source. The source can be one of the following:

- **PRESET**: A collection of 118 waveforms, containing multisamples, sampled attacks, oscillator waveforms, attacks, non-pitched sounds, other effects, drum sounds.
- **CARD**: An expansion card from your library.
- **AFM**: Use the output of an AFM element as the source. Pitch modulations will have no effect in this case.

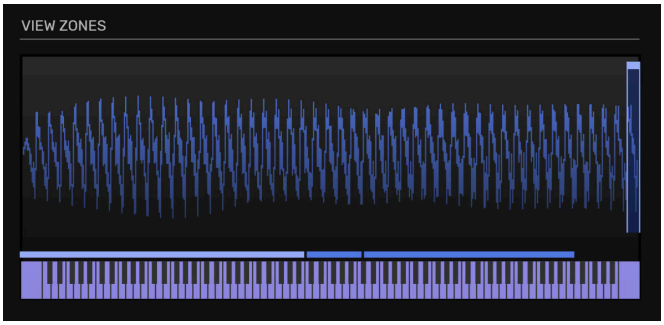
Expansion cards

Swayed ships with one expansion card of 81 additional waveforms. All waveform ROM files placed in the **Waveforms** folder will be picked up as cards that can be selected. To find this folder, click the 3 dots in the top left corner of the screen, go to the waveforms screen, and select the **OPEN IN EXPLORER** button (on Windows) or **OPEN IN FINDER** button (on Mac) from there.

A future update of Swayed will include a card editor, allowing you to import your own samples.

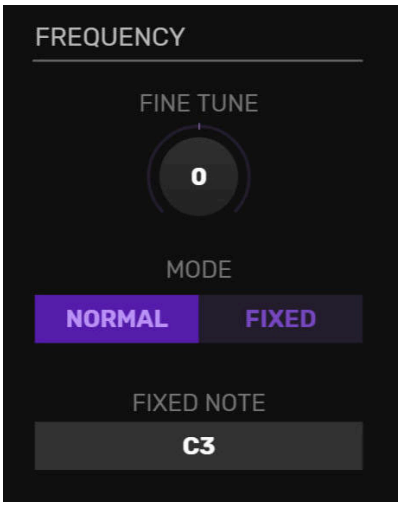
Waveform ROM files that are in the format of the original hardware will be picked up as well. You could create these files with the sy.factory software.

View zones



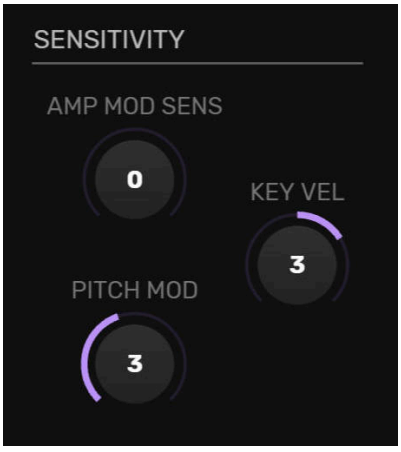
This display shows the current waveform, with its looping points. When it's a multi-sample it shows how the different zones are laid out on the keyboard. This control is only meant for visualization, and does not provide any control over parameters.

Frequency



This section contains control that affect the pitch of the played back sample. The **FINE TUNE** control allows you to finely detune the pitch. The **MODE** determines if the pitch is key tracked or a fixed frequency. When it's set the **NORMAL** the pitch follows the frequency of the incoming MIDI note (unless altered by the fine tune or detune/note shift settings). If it's set to **FIXED** the pitch will be determined by the note value set for **FIXED NOTE**.

Sensitivity



This section contains settings that affect the depth of the modulations produced by the main LFO and the controllers that are set up in the voice/controllers screen.

- **AMP MOD SENS**: Amplitude modulation sensitivity. This setting determines how much the element is affected by amplitude modulation.
- **PITCH MOD**: Pitch modulation sensitivity. This setting determines how much the element is affected by pitch modulation.
- **KEY VEL**: Key velocity sensitivity. This setting determines how much the level is scaled by the velocity. At positive settings, playing with more force results in a louder playback. At negative settings, playing with more force results in a softer playback.

Level envelope

The envelope section allows you to shape the level of element over time. The level and rate per segment can be modified by dragging the control points in the graph. The control point labeled **R** is the release point. The segment after that affects the level after releasing the note.

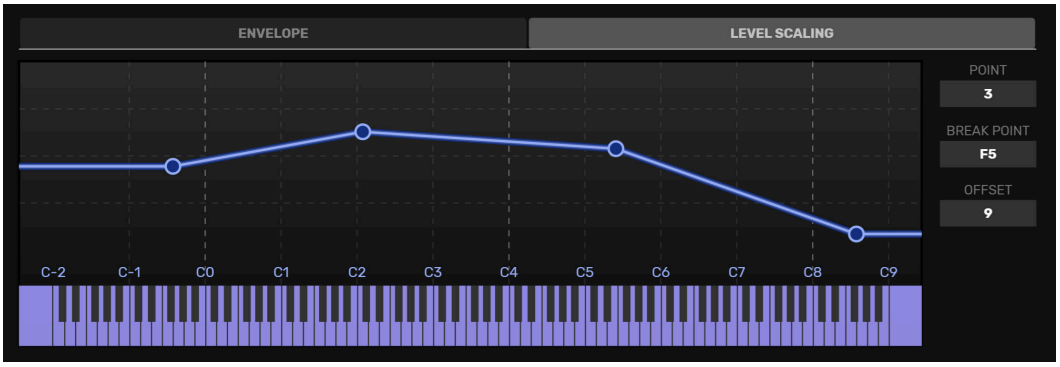
Just as with the other envelopes, it's important to note that the graph does not accurately depict the level over time. Lower rate values are exponentially slower, but the horizontal rate offsets of control points are scaled linearly for easy editing.

Double click a control point to reset the offset to 0. Right clicking the graph opens a popup menu with the option to reset the envelope, and options to copy or paste the envelope.

Other settings that affect the level envelope:

- **RATE SCALE**: Sets the keyboard rate scaling. Positive values mean higher notes will make the envelope go faster. Negative values mean higher notes will make the envelope go slower. When set to 0 the envelope speed is unaffected by the note.
- **HOLD**: If this is switched on, the envelope starts at full volume. The rate of the first segment changes to a "hold time" that specifies how long full volume lasts before moving on to the next segments of the envelope. If this is switched off, the envelope starts at silence and increases to full volume at the rate set by the first segment. It always goes to full volume during the first segment – the target level of the first segment cannot be changed.
- **VEL SW**: If this is switched on, the overall envelope speed will be affected by the note velocity. The strength of this speed adjustment is determined by the **KEY VEL** setting. If **KEY VEL** is positive, playing more strongly will result in a faster envelope. If it's negative, playing more strongly will decrease the overall speed.

Level scaling



The level scaling section allows you to change the level of the element across the note range. For example, if you want a sound to go softer as you play higher up the keyboard, you can turn down the level for the higher keys.

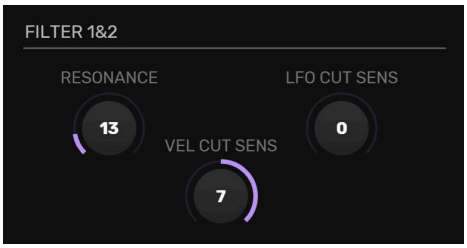
There are 4 break points to be placed. The horizontal location determines the note, the vertical location the adjustment in the level, which can be positive or negative. This number is simply added to the element level. The final calculated element level can never exceed 127 or go below 0. The graph shows a line between two breakpoints that indicates what the offset will be at each note on the keyboard.

Double click a breakpoint to reset the adjust back to 0. Right clicking the graph opens a popup menu with the option to reset the level scaling, and options to copy or paste the level scaling.

Filters Tab

This tab allows you to set up the filters for the element. There are two 12dB/oct filters available. The first of the two can be a low pass or high pass filter, the second one can only be a low pass filter.

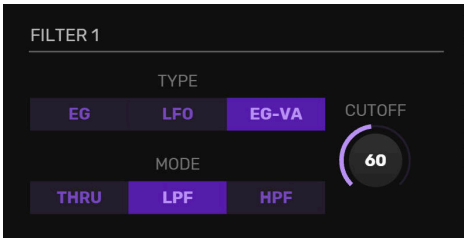
Filter 1&2



This section contains some controls that apply to both filters.

- **RESONANCE**: The resonance of the filters. Only the low pass filters can be resonant. If filter 1 is set to highpass mode this setting has no effect at all, not even on low pass filter 2, if it's active.
- **VEL CUT SENS**: This setting controls how much the key velocity will affect the cutoff frequency. For positive settings, higher velocities increase the cutoff. For negative settings, higher velocity decrease the cutoff.
- **LFO CUT SENS**: This determines how much the filter cutoff is affected by the filter modulation from the main LFO. It also determines how much the filter cutoff is affected by the contoller assigned to filter cutoff.

Filter 1



This section contains controls that only affect the first filter.

- **MODE**: Sets the filer mode. **THRU** means the filter is bypassed. **LPF** sets it to a low pass filter. **HPF** to a high pass filter.
- **CUTOFF**: Sets the cutoff value (0Hz to 22430 Hz for **LPF**, 0Hz to 11660Hz for **HPF**).
- **TYPE**: The filter control type.

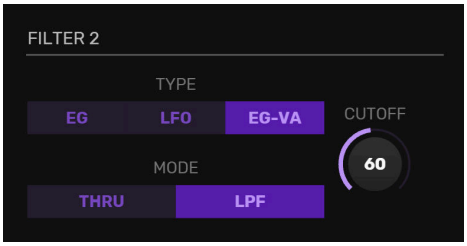
When **TYPE** is set to **EG** the filter cutoff is determined by the envelope. Key velocity still has effect if its sensitivity is set to anything other than 0. The controllers assigned to the filter modulation depth and filter cutoff will have their value sampled at the note on event, but will not be updated during the note.

The **EG-VA** setting is mostly the same as the **EG** setting, except the key velocity will affect the rate and target level of the first envelope segment (provided velocity cutoff sensivity is set to a value other than 0). For positive settings of the sensitivity, higher velocities will increase the rate and target level of the first segment. For negative settings the effect is reversed.

The **LFO** setting will make sure the filter cutoff is controlled by the main LFO modulation, as well as the controller assigned to filter modulation. Unlike the **EG** and **EG-VA**

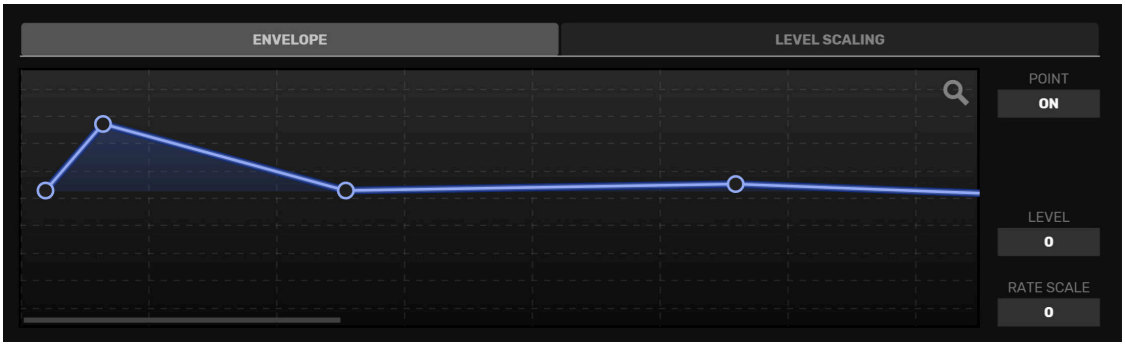
settings, the controllers are effective over the entire duration of the note. Key velocities will still affect the cutoff if the sensitivity is anything other than 0.

Filter 2



The settings for filter 2 are the same as the ones for filter 1, with the exception that the **HPF** mode is not available.

Filter 1/2 Envelope



The envelope section allows you to modify the filter cutoff over time. This requires the filter type to be set to **EG** or **EG-VA** mode. In **LFO** mode the envelope is ignored. Filter 1 and 2 each have their own envelope.

The envelope is bipolar, offsetting the selected filter cutoff from -127 to 127. This offset is labeled -64 to 63 in the envelope though. So if your filter cutoff is 100, and the offset at an envelope point is +3, then the actual cutoff of the filter is 106 at that point. The actual cutoff is always clamped between 0 and 127.

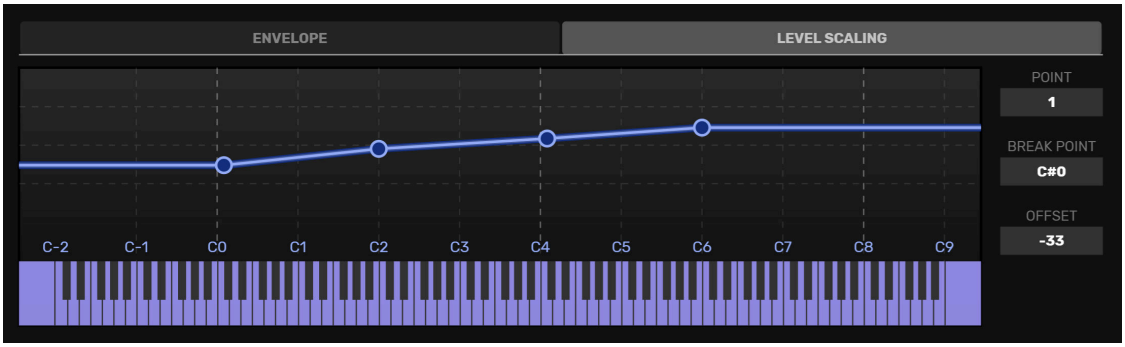
The cutoff offset and rate per segment can be modified by dragging the control points in the graph. The control point labeled R is the release point. The segments after that affect the filter cutoff after releasing the note.

Just as with the other envelopes, it's important to note that the graph does not accurately depict the level over time. Lower rate values are exponentially slower, but the horizontal rate offsets of control points are scaled linearly for easy editing.

Double click a control point to reset the offset to 0. Right clicking the graph opens a popup menu with the option to reset the envelope, and options to copy or paste the envelope.

The rate scale setting allows you to set up note based rate scaling. Positive values mean higher notes will make the envelope go faster. Negative values mean higher notes will make the envelope go slower. When set to 0 the envelope speed is unaffected by the note.

Filter 1/2 Level Scaling

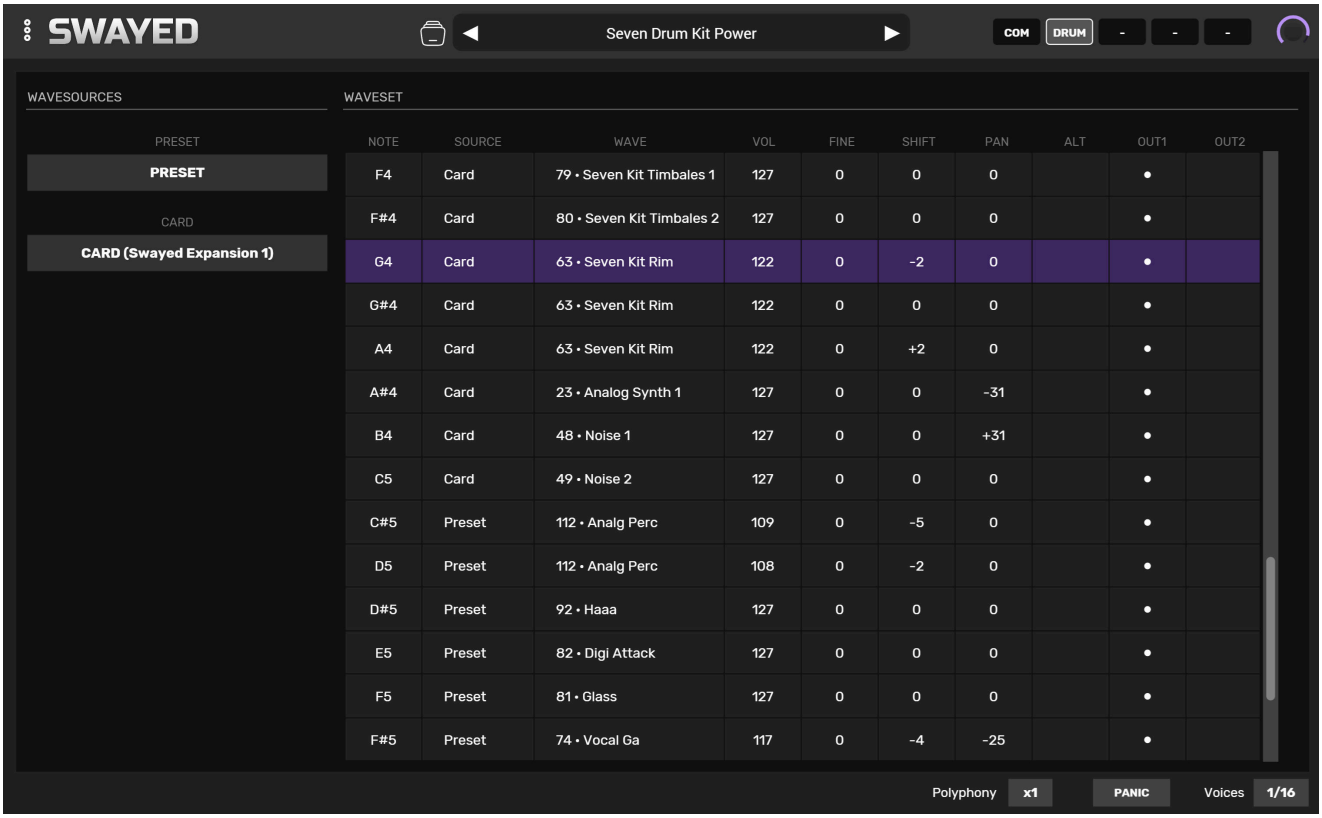


The filter level scaling section allows you to change the cutoff offset of the filter across the note range. For example, if you want a sound to be less bright as you play higher up the keyboard, you can turn down the offset for the higher keys.

There are 4 break points to be placed. The horizontal location determines the note, the vertical location the adjustment in the cutoff, which can be positive or negative. It ranges from -127 to 127. The graph shows a line between two breakpoints that indicates what the offset will be at each note on the keyboard.

Double click a breakpoint to reset the adjust back to 0. Right clicking the graph opens a popup menu with the option to reset the level scaling, and options to copy or paste the level scaling.

DRUM SCREEN



This chapter explains the **DRUM** screen. To access this screen, load a drum voice patch and select the **DRUM** element in the upper toolbar.

Drum voices only consist of 1 element. With a drum voice you can trigger different waveforms in a single patch. Each note from **C1** to **C6** has one waveform assigned. When a waveform is triggered, it's played back with a fixed envelope. This envelope varies from sample to sample, and cannot be changed.

The standard maximum polyphony is 16, but can be increased with the polyphony multiplier if needed.

The interface

Drum voices have a separate interface. It consists of two sections.

Wavesources section

This section allows you to select which waveform ROM to use for the **PRESET** and **CARD** slot.

Waveset

This is a list of zones that assigns one waveform to each MIDI note from **C1** to **C6**. You can select a waveform from the **PRESET** or **CARD** source. Each note can be configured with additional parameters. You can assign one waveform to multiple notes and configure them differently if you like. For example, if you want to be able to play toms at different pitches.

The following parameters can be set:

- **VOL**: The volume of the played back waveform.
- **FINE**: Fine tuning of the played back waveform.
- **SHIFT**: The tuning (in semitones) of the played back waveform.
- **PAN**: The panning of the played back waveform (from left to right, there are no dynamic pan tables involved here).
- **ALT**: A flag that allows you to put this zone in the “alt” group. Waveforms in the alt group cannot play simultaneously. If you trigger a note in the alt group, all currently playing notes in the alt group will stop. This can be useful for hihats, for example.
- **OUT1** and **OUT2**: The output groups to send the sound in this zone to.

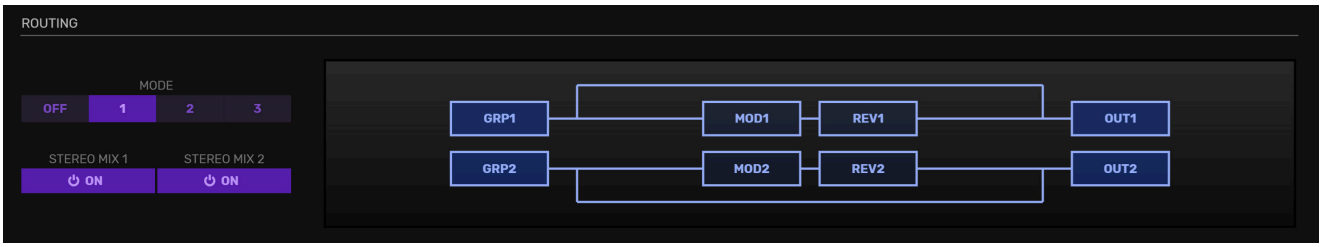
EFFECTS SCREEN

Swayed is equipped with a number of basic digital effects that can be added to enhance your patch. The group outputs 1 and 2 can be processed in parallel or in series, depending on the selected routing mode. To enter enter the effects screen, select **COM** and navigate to the **EFFECTS** tab.

The effects section can be bypassed entirely in the top left menu. This setting persists across patch changes, which makes it easy to browse presets while hearing how they sound dry.

The UI sections

Routing



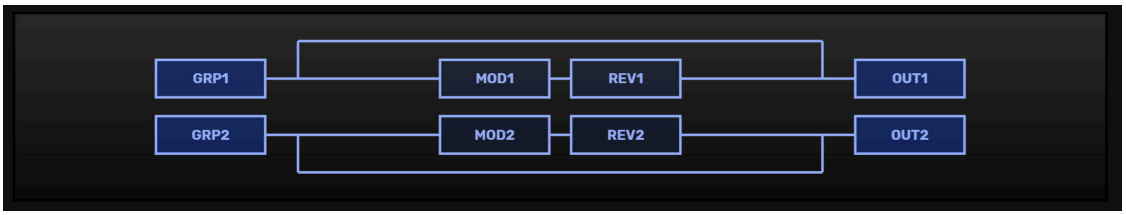
The routing section allows you to select how the effects are configured. There are 3 different modes, plus an **OFF** setting. Additionally, the stereo mix for group 1 and group 2 can be turned on or off. When a stereo mix for a group is on, the dry sound is mixed with the processed sound, effectively creating a 50:50 dry/wet mix.

Mode OFF

When the mode is set to **OFF**, the output of group 1 and 2 are sent directly to the output:

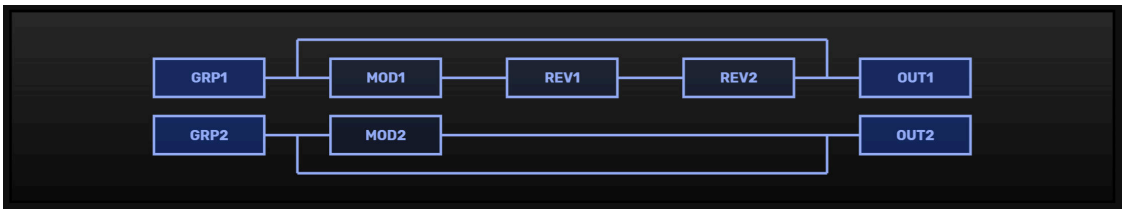
Mode 1

When the mode is set to **1**, group 1 is sent into modulation effect 1, followed by reverb effect 1. Group 2 is sent into modulation effect 2, followed by reverb effect 2. The output of the reverb effects is sent to the output.



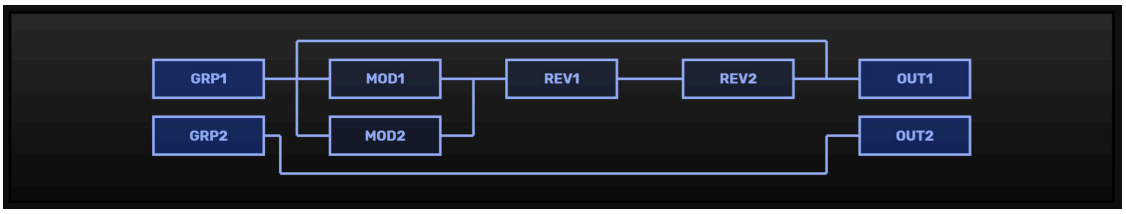
Mode 2

When the mode is set to **2**, group 1 is sent into modulation effect 1, followed by reverb effect 1 and then reverb effect 2.. Group 2 is sent into modulation effect 2, which is sent directly to the output without going into a reverb effect.

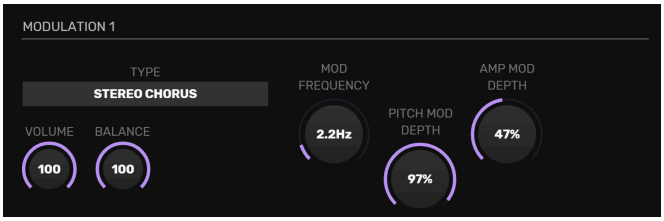


Mode 3

When the mode is set to **3**, the output of group 1 is sent in parallel to modulation effect 1 and modulation effect 2. Then the left channel of modulation effect 1 is used as the left channel input for reverb effect 1, and the right channel of modulation effect 2 is used as the right channel input for reverb effect 1. The output of reverb effect 1 is sent into reverb effect 2. The output of reverb effect 2 is sent to the output. Group 2 is directly sent to the output.



Effect sections



There are 4 different effect sections in the UI. There is one for modulation effect 1, one for modulation effect 2, one for reverb effect 1, and one for reverb effect 2. Each effect section contains the following controls.

- **TYPE**: The effect type. There are 4 different modulation effect types, and 40 reverb effect types.
- **VOLUME**: The output volume of the effect processor.
- **BALANCE**: A dry/wet control for the effect processor.
- **PARAMETERS**: Each effect has 3 or 4 parameters, depending on the effect type. Each effect type has its own specific parameters.

The effects

All modulation effects are stereo in, stereo out. Reverb effects 1 to 34 are mono in, stereo out. Reverb effects 35 to 40 are stereo in, stereo out. This means that if you use one of the mono in reverb effects, the dynamic panning info will be lost going into the effect processor. If **STEREO MIX** is turned on for the group, this dry signal path will keep the dynamic pan.

The “reverb” effects are not all actual reverbs, there are also delays, distortions and EQs.

Modulation effects:

- **STEREO CHORUS**
- **STEREO FLANGE**
- **SYMPHONIC**
- **TREMOLO**

Reverb effects:

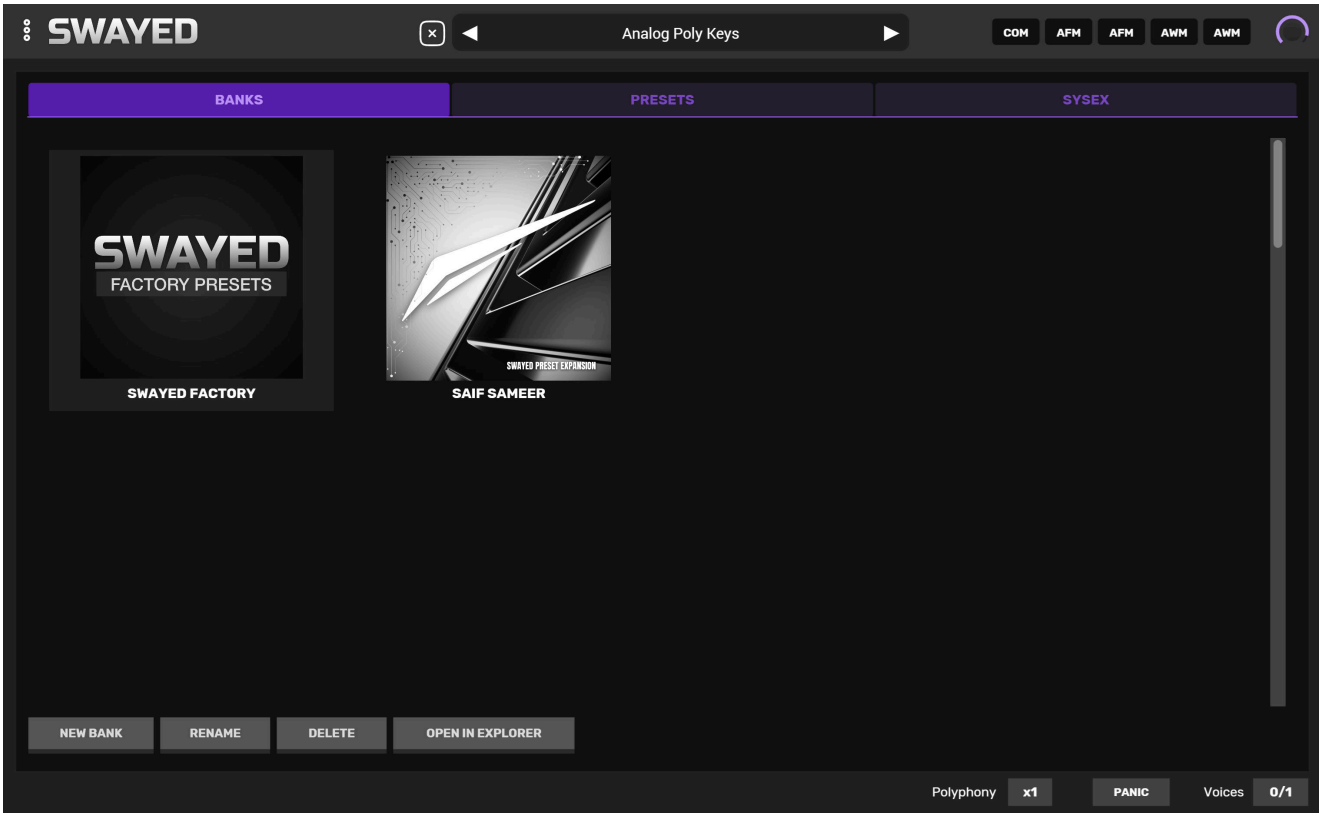
- **REVERB HALL**
- **REVERB ROOM**
- **REVERB PLATE**
- **REVERB CHURCH**
- **REVERB CLUB**
- **REVERB STAGE**
- **REVERB BATHROOM**
- **REVERB METAL**
- **SINGLE DELAY**
- **DELAY L,R**
- **STEREO ECHO**
- **DOUBLER 1**
- **DOUBLER 2**

. **PING-PONG ECHO**
. **PAN REFLECTION**
. **EARLY REFLECTION**
. **GATE REVERB**
. **REVERSE GATE**
. **FEEDBACK ER**
. **FEEDBACK GATE**
. **FEEDBACK REVERSE**
. **SINGLE DELAY & REVERB**
. **DELAY L/R & REVERB**
. **TUNNEL REVERB**
. **TONE CONTROL 1**
. **SINGLE DELAY & TONE CONTROL 1**
. **DELAY L/R & TONE CONTROL 1**
. **TONE CONTROL 2**
. **SINGLE DELAY & TONE CONTROL 2**
. **DELAY L/R & TONE CONTROL 2**
. **DISTORTION & REVERB**
. **DISTORTION & SINGLE DELAY**
. **DISTORTION & DELAY L/R**
. **DISTORTION**
. **INDEPENDENT DELAY**
. **INDEPENDENT TONE CONTROL**
. **INDEPENDENT DISTORTION**
. **INDEPENDENT REVERB**
. **INDEPENDENT DELAY & REVERB**
. **INDEPENDENT REVERB & DELAY**

PRESET MANAGER SCREEN

The preset manager screen allows you to select and organize your preset library. It contains three different tabs.

Banks Tab



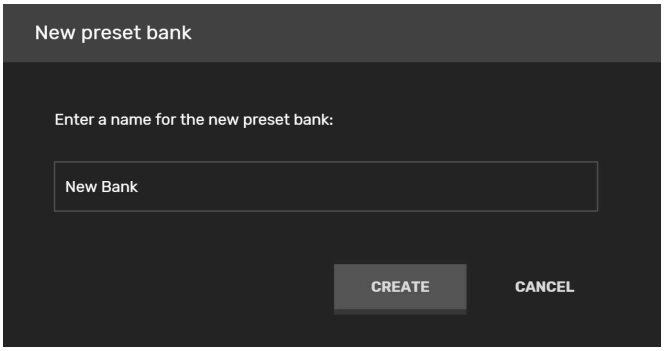
The banks tab shows the preset banks that are in your library, and provides control over these banks. Swayed ships with a factory bank of 128 presets, and an additional bank of 140 presets made by sound designer Saif Sameer.

Banks are just folders in the preset folder. You can place an **`icon.jpg`** or **`icon.svg`** in the folder to give it an icon that will show up in the banks tab.

Double click on a bank to open the presets tab, with that bank selected as a filter. This allows you to go through the presets of that bank.

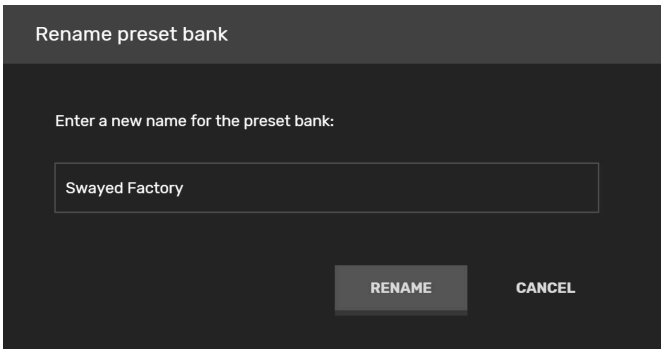
The bottom of the screen has a number of action buttons.

New bank



The **NEW BANK** button allows you to create a new empty preset bank. A dialog will pop up where can enter the name of the bank.

Rename



With the **RENAME** button you can change the name of the selected preset bank. A dialog will pop up where you can enter the new name of the bank.

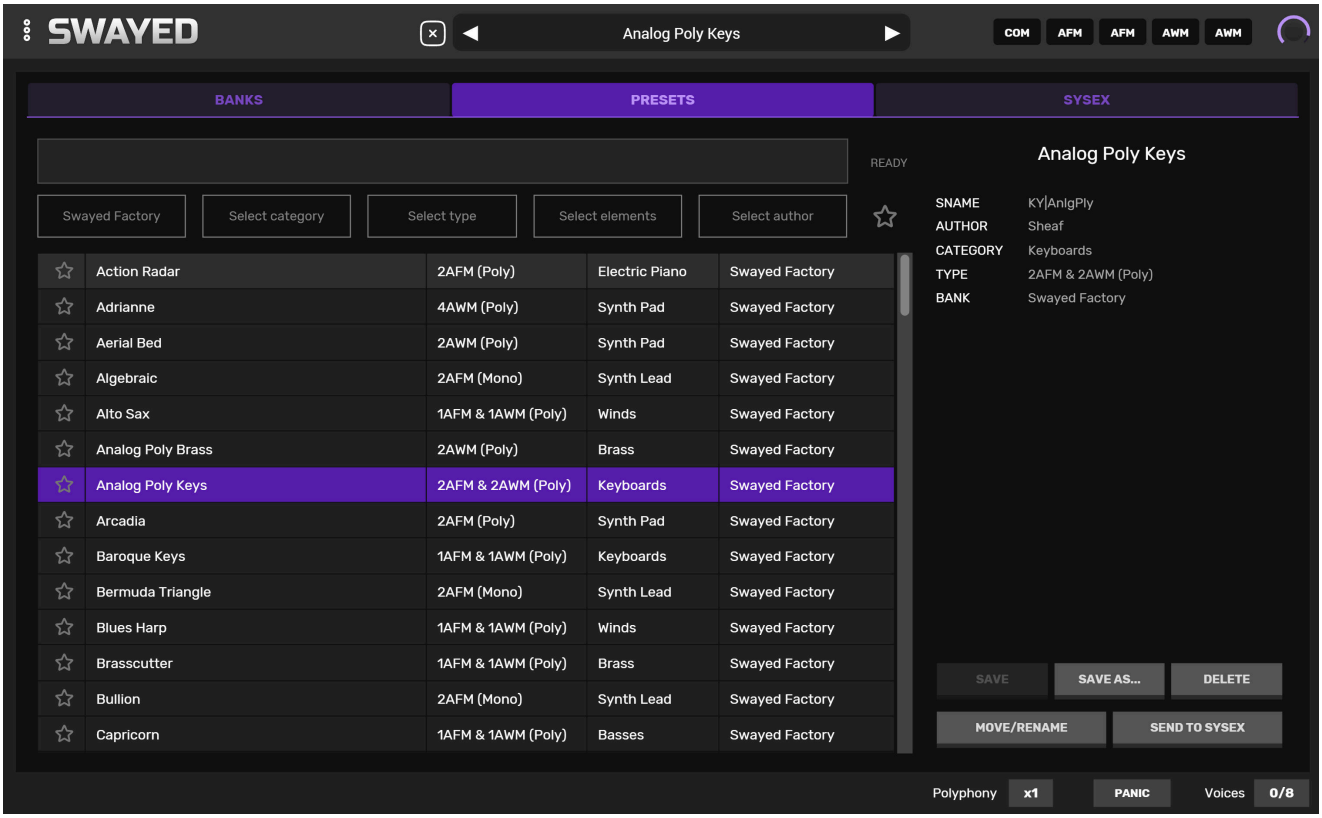
Delete

The **DELETE** button deletes the selected bank. A dialog will pop up to confirm the removal. If confirmed, the bank is gone and cannot be restored.

Open in Explorer/Finder

The **OPEN IN EXPLORER** button (on Windows) or **OPEN IN FINDER** button (on Mac) opens the preset folder in a OS File Explorer or Finder window. You can also categorize your banks there, by creating/renaming/deleting folders and copying or moving presets around.

Presets Tab



The presets tab contains the list of presets in your library. You can scroll through this list, filter the presets by various fields, select presets by clicking them, or mark them as favorite by clicking the star icon next to the name. You can use the up/down arrow keys on your keyboard to select the previous/next preset from the currently filtered list.

The preset list is loaded in the background. When the field next to the search bar shows **READY** this loading is done. Click it to reload the list from disk. The preset index is cached to disk for faster future loads.

The right side of the screen shows info about the currently loaded preset, as well as a set of buttons that allows you to perform various actions on that preset.

Filters

The filters provide a way to more efficiently search the preset list. The following filters are available:

- **BANK**: The preset bank.
- **CATEGORY**: The category / type of sound.
- **TYPE**: The voice type - Drum, AFM, AWM or a combination.
- **ELEMENTS**: The number of elements: 1, 2 or 4.
- **AUTHOR**: The soung designer.
- **FAVORITE**: Indicated by the star icon. When on, it only shows the presets you've marked as favorite.
- **SEARCH**: The text field at the top allows you to search by name.

The categories are the same categories that were used for a lot of presets made for the original hardware. Each category corresponds to a two letter code. You'll find

many presets in SysEx files use this encoding scheme for the name. These are the categories, and their codes:

- **SP** - Synth Pad
- **AP** - Acoustic Piano
- **BR** - Brass
- **EP** - Electric Piano
- **ME** - Musical Effect
- **WN** - Winds
- **ST** - Strings
- **BA** - Basses
- **SC** - Synth Comp
- **SL** - Synth Lead
- **PL** - Plucked
- **KY** - Keyboards
- **OR** - Organ
- **PC** - Percussion
- **CH** - Choir
- **SE** - Sound Effects
- **DR** - Drum Voices
- **SQ** - Sequencing
- **KS** - Keyboard Split
- **CT** - Controllers

Preset info

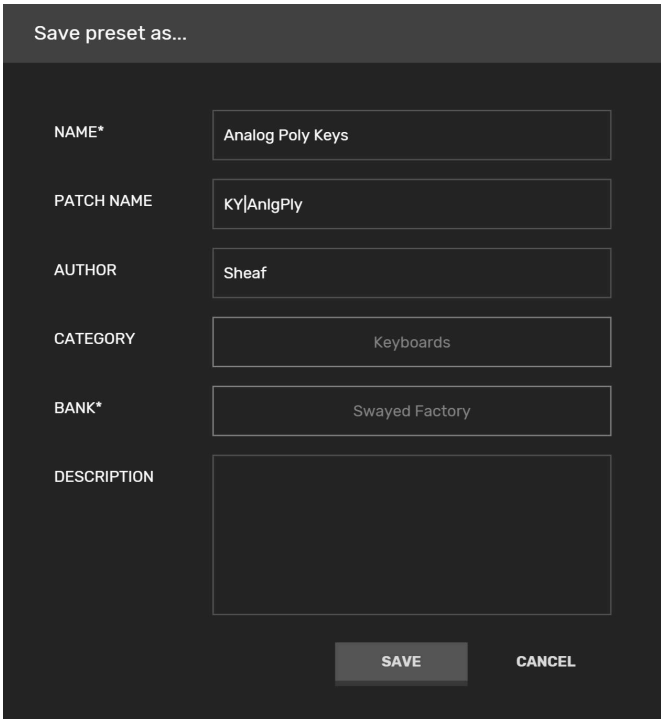
The preset info section on the right side of the screen shows the following information about the currently loaded preset:

- **NAME**: The full name of the preset.
- **SNAME**: The SysEx name. Name of 10 characters that's used for importing/exporting SysEx files. On the original hardware preset names could only be 10 characters.
- **CATEGORY**: The category / type of sound.
- **TYPE**: The voice type Drum, AFM, AWM or a combination.
- **BANK**: The preset bank containing this preset.

You can perform the following actions on the selected preset:

- **SAVE**: Saves the preset by overwriting it. This is disabled if there are no changes, or if the preset is loaded from a SysEx (that means it doesn't have a location as a preset in the library yet).
- **SAVE AS**: Saves the preset by writing it to a new location.
- **DELETE**: Deletes the preset.
- **MOVE/RENAME**: Renames the preset or moves it to a different bank.
- **SEND TO SYSEX**: Sends the preset to a location in the SysEx bank.

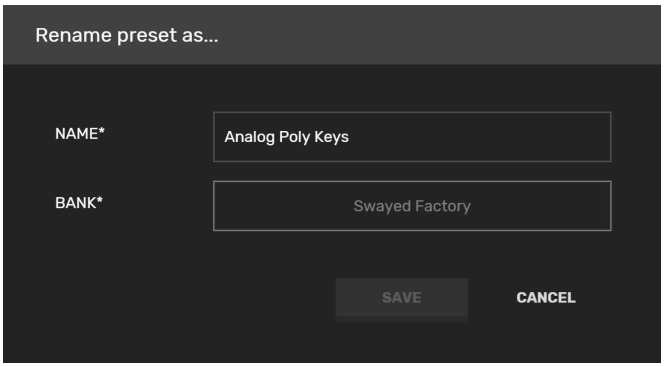
Save as



When choosing **SAVE AS** a dialog will popup that allows you to enter the preset information. It contains fields for the name, the SysEx patch name, the author, the category, the bank and a description.

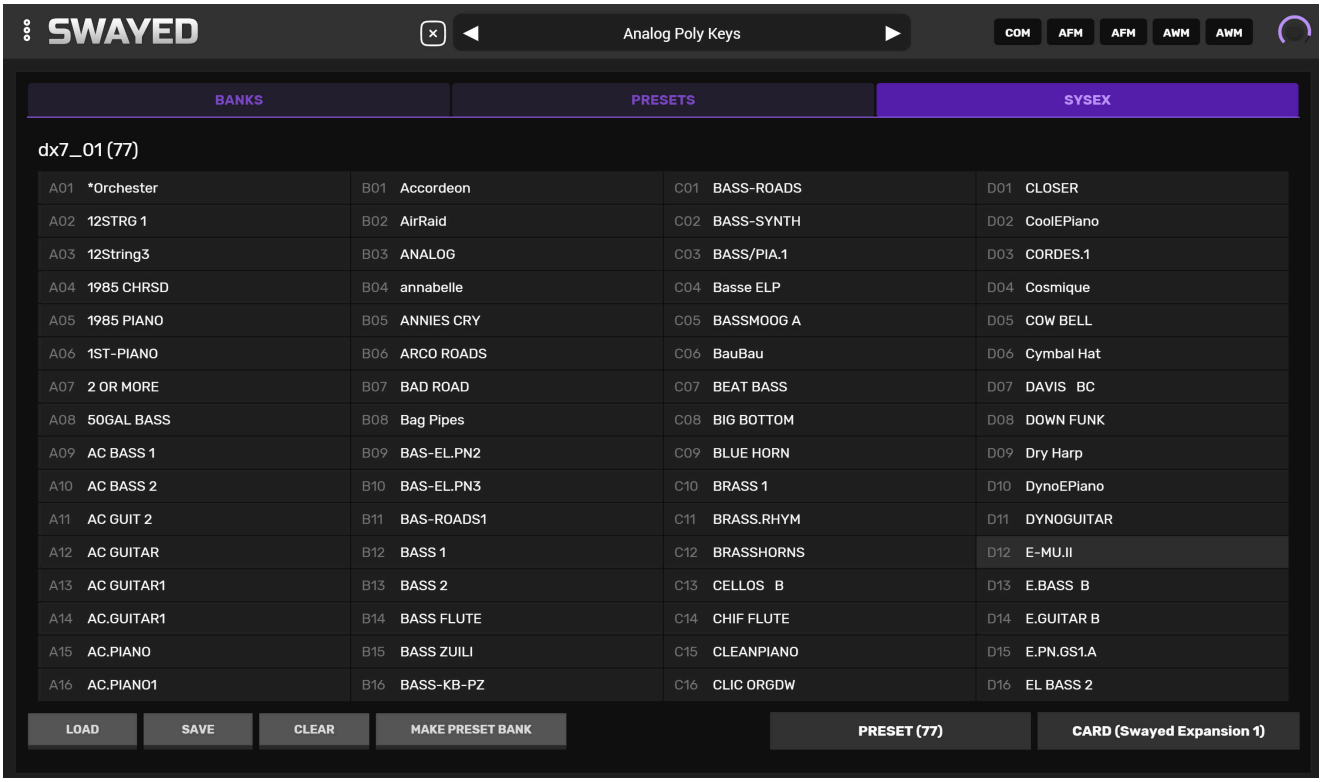
This will duplicate the preset if you choose a new name for it. There will still be a preset with the old name and preset bank location, unless the save was initiated from a preset loaded from a SysEx bank.

Move/rename



When choosing **MOVE/RENAME** a dialog will pop up that allows you to enter the new preset name and the bank to move the preset to. Unlike the save as option, this will not preserve the preset with the old name.

SysEx Tab



On the SysEx tab you can import and export SysEx files that follow the format of the original hardware. Each bank has exactly 64 patches. You can audition SysEx patches first without having to actually import them to your preset library. Just click on them in the table. You can use the up/down arrow keys on your keyboard to select the previous/next preset from the currently loaded bank.

If you like the entire bank you can choose to convert it to a preset bank to add to your library, or you could save individual patches as presets. To save an individual patch, click it to load it, then go to the preset tab and select **SAVE AS**.

Action buttons

The **LOAD** button will load a SysEx file from your harddrive. SysEx files from the 77 and 99 series original hardware models are supported.

The 99 model has more waveforms and different effects, but the waveforms had differeent numbers. Therefore, if you import a 99 model file, the common waveforms between the 77 and 99 model will be mapped, but for the patches that contain waveforms unique to the 99 these will default to a triangle waveform. Those patches will show a warning icon shown next to their name. The effects are translated, but obviously this cannot be a perfect translation since they are different. Hall reverbs will be mapped to hall reverbs of the same decay length, delays will be mapped to delays, et cetera.

The **SAVE** button exports the current SysEx bank to a file.

The **CLEAR** button clears the entire bank. As described in the previous section, you can send your presets to the SysEx bank to create your own bank. This bank could then be transferred to the original hardware using a program like MIDI-OX.

The **MAKE PRESET BANK** converts the SysEx bank to a preset bank, and adds it to your library.

Waveform sources

In the bottom right part of the screen you can select the waveform sources that will be used for **PRESET** and **CARD**. By default there only the factory waveforms for **PRESET** and the Swayed Expansion 1 waveforms for **CARD**. If you put additional ROM files in the designated folder, they will be selectable here.