USER MANUAL

What is Transfigure

The Header

Modulators
- Envelope Follower
- Trigger Unit
- Random Unit
- XY PAD
- Multi-Breakpoint Envelopes

Sound Processors
- Spectral Engine
- The Resynthesis Processor

Effect Units

The Band setup

Host integration

Installation
- Windows
- macOS

Uninstalling

Authorization

Contact

What is Transfigure?

Transfigure is your creative partner in sound. With just a few clicks, it turns any track into a living, breathing sonic landscape full of depth, motion, and emotion. Its clean, intuitive interface makes complex processes feel effortless. Whether you're refining details or creating entirely new textures, Transfigure gives you the freedom to shape sound your way. The central XY pad allows for simultaneous parameter control and expressive automation, perfect for performative sound design and evolving texture.

Key Features:

Experience the freedom to edit and shape sounds in real time, enhancing both workflow and creativity.

Transfigure masterfully blurs the line between effect and instrument, transforming every audio source into an artful expression.

Versatile Sound Effects: Choose from a wide range of effects, including reverb, delay, distortion, and spectral effects to add depth and character to your audio.

User-Friendly Interface: Designed with emphasis on usability, Transfigure's interface simplifies the audio processing journey, allowing you to focus on your creative vision without distractions.

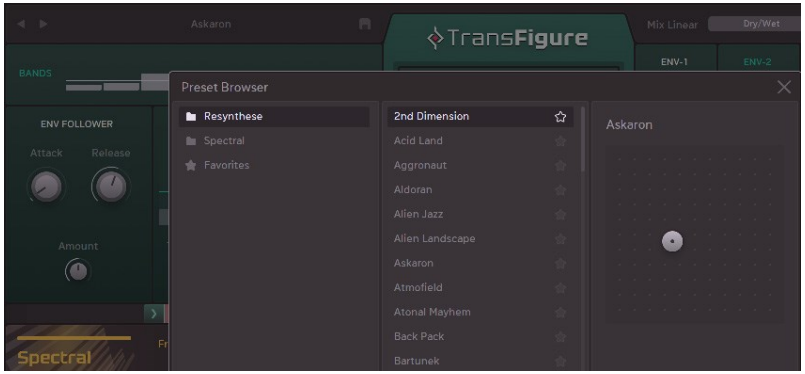
Presets: Get started quickly with a huge library of presets.

Transfigure is ideal for musicians, sound designers, and anyone looking to transform their audio content into something never heard before. Unlock new dimensions in sound with Transfigure, and take your audio projects to the next level!

The Header

Preset Browser

Manange your presets here.



Click the name of the current preset to open the preset library dialogue and step through the factory presets via the < > buttons.

Click the “Disk” button to save your own presets in the “User” folder.

The factory presets are stored in in your User documents folder:
/Users/[Your Username]/Documents/Sugar Bytes/Transfigure/Presets

Mark a preset as a favourite by clicking the star icon to quickly find it in its respective folder.

Master Mix

The Master Mix menu offers four blending modes that determine how the Mix control blends the incoming dry signal with the Transfigure output signal.

Linear: A linear mix – the center position provides 50% dry and 50% processed signal.

Equal: The crossfade is shaped according to the “Equal Power Law” which leads to some signal amplification at 50/50 mix.

Dry + Wet: The wet signal is added to the dry signal.

Wet Only: Only the effect signal is audible and the Mix parameter becomes a volume control.

Global Dry/Wet

Set the Dry/Wet Level (by clicking and dragging left/right), blending the incoming signal with the Transfigure output.

Undo/Redo

 Never fear a mistake. Undo and Redo recent actions with the curved arrows.

Global Tools

 The magic wand button opens up a drop-down menu.

Here, you can **delete all modulations** or **MIDI assignments** at once.

Swap the 2 generic effects.

Click **Random** in oder to perform a sound randomization of all parameters and effects.
Three **Randomize modes** are available via the Settings Dialog here.

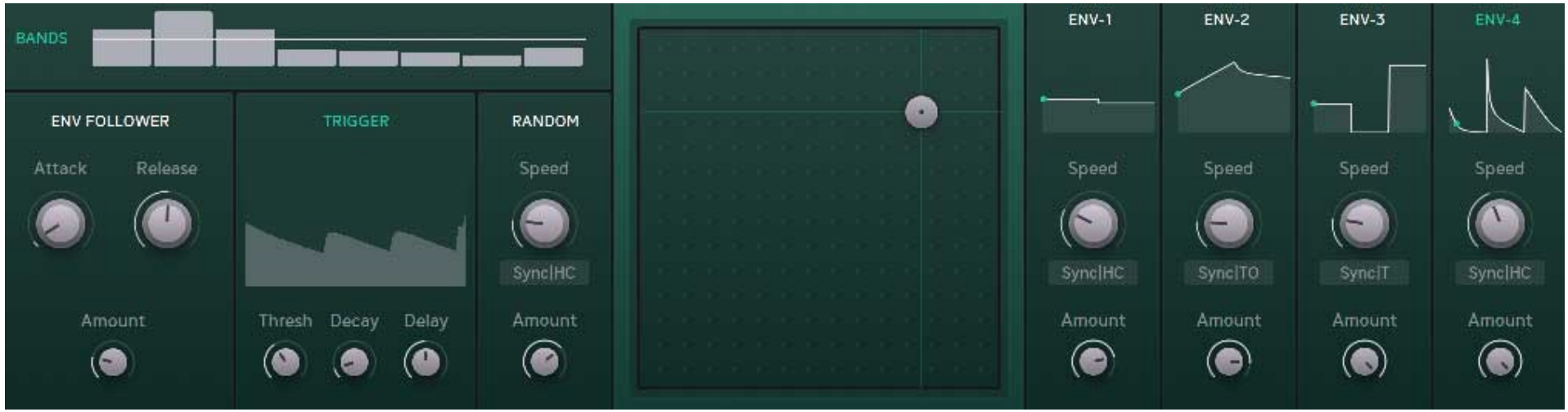
THE 3 SECTIONS

Modulators

Sound Processors

Effects

Modulators



There are basically 8 modulators in Transfigure.

The **Envelope Follower**, a **Random Processor**, the X and Y output of the **XY Pad** and the **4 Multibreakpoint Envelopes**.

To assign a modulator, just right-click a control and set the modulation amount.

Click the MIDI socket symbol to assign incoming MIDI CCs.

You can copy, paste, or reset modulator setups for a faster workflow.

Modulator controls can also be modulated, allowing you to create highly complex setups with ease.

Simply clear all modulation at once in the Global Tools (the magic wand button on top right).

Besides the **XY Pad**, every modulator has an **Amount control**.

Envelope Follower

On the left, you'll find the Envelope Follower alongside the Trigger Unit.

You can choose a specific **frequency band** 

or simply use the overall **volume level** (no band selected) to adjust the Envelope Follower.

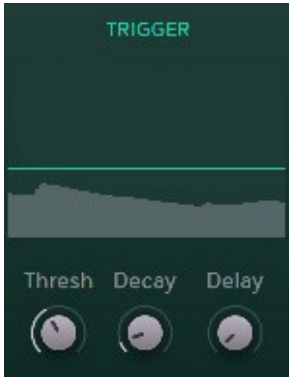
The Envelope Follower has the classic **Attack** and **Release** controls to define the basic behavior.

Note that a high attack value will prevent large modulation amounts; smaller values are often more suitable.

The **Amount control** lets you scale the output of the Envelope Follower.

Note that for the Envelope Follower, the amount goes up to 200%. This allows you to have a meaningful signal even if your input signal doesn't have much energy.

Trigger Unit



The trigger unit, beside the envelope follower, is only used in certain areas. It can be used to **trigger the freeze functionality** in the sound processor or to **(re)start the multi envelopes**.

The **Threshold** defines a certain level; if the audio signal goes above that level, a trigger is created.

The **Decay** will internally lower the threshold over time. This is useful if your input signal is generally too low to ever reach the threshold.

By having the decay lower this threshold, it ensures that at some point a trigger will happen. Once a trigger is created, the decay will start again.

The **Delay Control** will delay the creation of the trigger. This can be quite useful when the trigger is used to activate the freeze effect in the

Sound Processor. Often, you want the freeze to happen just a few milliseconds after the signal peaks, so the peak signal is held in place.

Random Unit

The Random unit creates random values; the rate can be controlled by the **Speed** control. Lots of different modes are here to explore:

In **Host Clock mode (HC)**, the timing of the trigger is related to your host clock and therefore the BPM of your project.

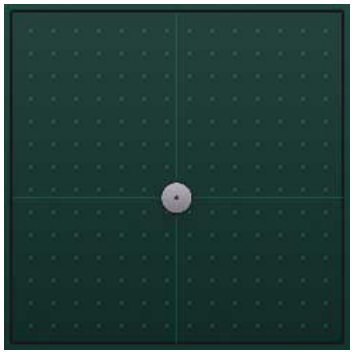
So the suboption **Sync** gives you plain synced options, while **SyncTP** includes less common triplet and dotted timings.

Seconds
Switches to milliseconds/seconds instead of BPM sync.

The **Host Clock Glide mode (HCG)** does the same as the Host Clock mode, with the exception that it not only generates random values but also includes an automatic transition from the previous value to the new one within the defined interval.

The **Trigger Glide option** generates a new random value whenever the trigger unit produces a trigger. It will move to the generated value over the defined time.

The **Trigger (no glide)** will just create a new random value whenever the Trigger Unit creates a trigger.



XY PAD

The XY Pad is where the fun starts.

Each preset includes an XY Pad assignment, so make sure to touch and move it after you load a preset.

Often the presets are set up in a way that in the left bottom corner you will have the pure unaffected audio signal.

Changing the XY Pad will then add or mix in the effects signal.
So in the top-right corner, you will often have the most intense or sophisticated setup.

The 4 Multi-Breakpoint Envelopes

Our envelopes are designed to deliver maximum functionality. They can create classic LFO-like shapes such as sine or saw, but can also be used as a step sequencer – and much more.

When you click on the envelope, the **Envelope Editor** will be opened, but first let's check the outside controls.

The **Speed control** defines the speed the envelope is read.

The **Amount control** will scale the output of the envelope.

As in the Random Unit, there are different modes to explore here.
The first four main modes are related to the speed.

In **Host Clock Mode (HC)**, the speed is related to the speed set in your DAW. The submodes let you further control which speeds are available.

Sync: Straight divisions of your BPM

Sync TP: Divisions of your BPM, but also triplet and dotted timings are included.

Sec: Time in seconds the envelope needs to run from start to end.

The main **Host Play mode (HP)** offers the same submodes as the Host Clock Mode (HC).

Here is the difference though:
In HC mode, you are also synced to the transport position of your DAW, even after changing the speed. While this ensures absolute correctness of the envelope position, it will lead to jumps when changing the speed while your DAW is running.

The HP Mode works differently. There will be an internal clock that starts when you start the transport in your DAW.
If you then change the speed, the internal clock will adjust, but there will be no jumps as in HC mode.

When in **Trigger Mode (T)**, the envelope will be started when a trigger is created.

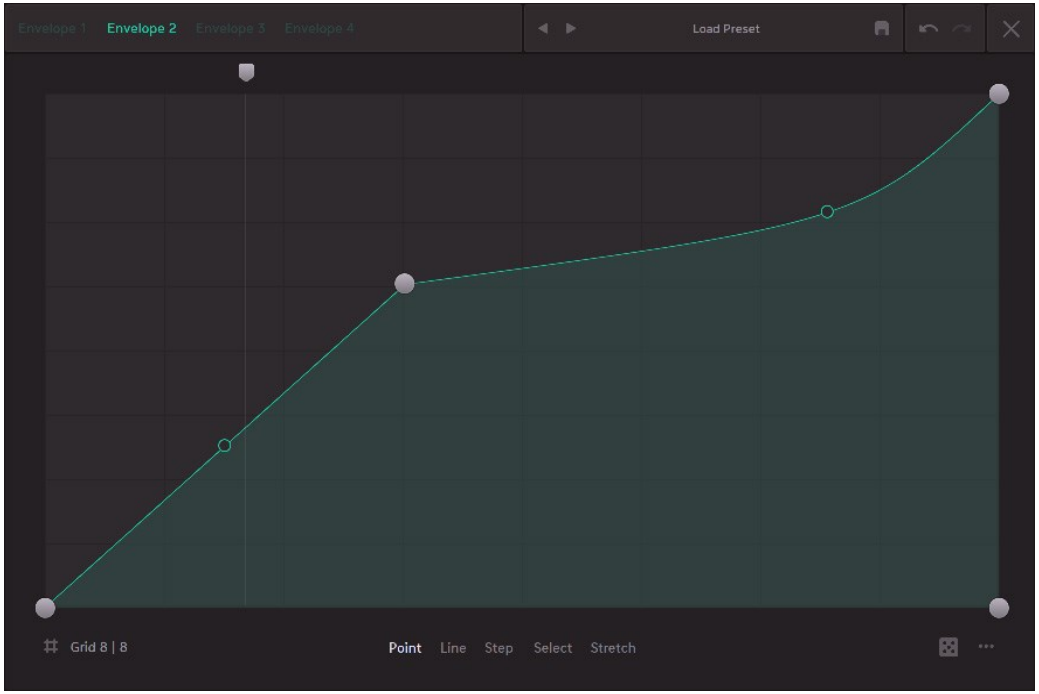
In **Trigger Oneshot mode (TO)**, the envelope will stop when the end is reached.

The **Envelope Follower mode** will transform the output of the envelope follower. You can use this to create some quite interesting setups. For example, you can create a hard trigger, so that only when the envelope follower is above 50%, the envelope will output 100%.

The **X/Y mode** lets you create an output related to the X/Y position of the XY Pad. This can be extremely powerful to bring the XY Pad to its full potential. For example, you could define that only if the X value of the pad is over 90%, the envelope will output some defined value. You will note that in our presets this is often used, for example, to trigger a freeze functionality if X or Y is above 50%.

In Random Mode the output of the Random Unit is used and transformed related to the envelope.

The Envelope Editor



This is a straightforward envelope editor.

In the top left area, you can quickly switch between the 4 available envelopes.

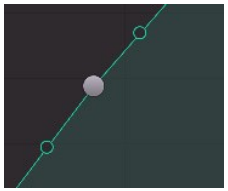
The Envelope Editor provides its own preset system.

To save a preset, simply click the disk icon.

Any changes you make can be undone or redone at any time.

To close the Envelope Editor, click the X button in the top-right corner.

The Envelope



The envelope is defined using anchor points and control points.

Clicking in an empty area creates a new anchor point.

You can move both anchor and control points by left-clicking and dragging them.

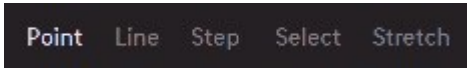
To delete an anchor point, right-click on it.

Double-clicking a control point will reset it.

Grid

If the grid is active, the main points can only be placed on the grid.
The grid can be set up in a way that X and/or Y distances are divided by 2 to 32.

The 5 Edit Modes



In **Point mode**, you will have the classic click behaviour already described.
In **Line mode**, you can't touch individual points; you will 'draw' a line if you just click and move around.
In **Step mode**, the envelope will behave like a step sequencer. The number of steps is defined by the grid X resolution.
In **Select mode**, you can select multiple points and then move them all together.
In **Stretch mode**, you can bend the envelope by clicking a point and moving in any direction.

*There are two **shortcuts** available: **Strg**: only stretches the Y dimension. **Alt**: only stretches the X dimension.*
Also note that you can make a selection in Select mode and then switch to Stretch mode to only stretch your selection.

 The **Random control** gives you random envelopes. The created curves will depend on the edit mode you are in.

Workflow functions:

 With **Copy and Paste**, you can copy and paste envelopes.

Normalise will look for the highest Y value and then stretch the whole envelope so that the highest Y value becomes 100%.

Flip will flip the whole envelope vertically or horizontally.

Move will move the envelope left or right.

THE 2 SOUND PROCESSORS

Spectral Engine

Resynthesis Processor

The Spectral Engine

The spectral engine works in the frequency domain, which means that in contrast to the time domain—where the signal is a continuous signal (wave)—the audio signal comes as energy levels for certain frequencies at a certain point in time.
This offers the possibility to manipulate our sound in a much more natural and intuitive way.

Low / Range:

A hard frequency filter — Low defines the lower border, and Range the upper one.

Resolution / FFT (64 – 4096):

Defines the spectral resolution – in other words, into how many frequency ranges the signal will be split.
In general, most algorithms work best at 1024 or 2048. Lower resolutions often result in a more distorted output, while higher resolutions create a smoother, more airy sound.

Freeze Rate:
The analized input will be frozen for a certain time.

In **Sync mode** the freeze interval is related to the Host BPM.

At **0%** no freezing is applied.

At **100%**, the spectral information will be used to freeze the signal permanently.

In **Trigger Mode**, the generated signal from the trigger unit will be used to freeze the signal for the set time.

In **(Re)trigger Mode**, it's possible to recapture the frequencies even if the set time has not yet passed.

Character:
This control defines the behaviour when a freeze happens. The explanation gets a bit technical, so you should test with your ears first and foremost. In general, Hold stands for a clean freezing approach, where the signal remains truly constant.

Hold A/B:
When freezing an audio signal in the frequency domain, there are basically two main components per frequency: the energy of the frequency and its phase. In Hold A/B mode, you can blend between two different phase recreation methods.
A mode uses the phase information that was present when the signal was frozen.
B mode uses a more mathematical approach to the phase.

Hold / Live:
Mix between the frozen phase and the phase information of the current input.
This brings back some character of the actual input signal.

Hold / Rnd:
Mix between the frozen phase and a randomly created phase.
This gets noisy, but in a spectral context.

Live / Rnd:
Mix between the input phase and a random phase.

Stereo:
Adds different offsets to the phase creation for left and right.
Adds a stereo difference to the sound.

LFO:
The phase for recreation will be modified by an LFO.

Unisono:
A frequency-dependent offset will be added to the phase. The resulting sound reminds of a unison control known from classic synthesizers.
Note the Band setup window for this section. It allows you to define not only which frequencies should be frozen, but also how much.
Obviously, the number of frequencies is dependent on the resolution.

Delay Time:
Adds a delay in the spectral domain.
The Delay Time control sets the overall maximum delay time.
Your signal will be unaffected if **Delay** or **Delay Amt** is set to 0%.

In order to edit the timing for each frequency, click on the Band setup. There are two options:

Always: The delay is constantly applied, as you would expect.
Freeze Link: The delay is only fed and advanced when a freeze happens.

Delay Amt: Controls how the delay signal will be added.

Add: Adds the delay signal.
Mix: Mixes between the delayed and the "original" signal.
Feedback Delay: In this mode some part of the delay signal will be feed back into the input.
Feedback Inverse: The delay signal will replace the original signal; you can add negative feedback.
The negative feedback will result in frequency cancellation depending on your signal. Expect to find some interesting sounds here.

Transpose:
Realtime-pitches your signal ±12 semitones.

Fine:
Realtime-pitches your signal ±100 cents.

Harmonize:
The Harmonize section will adjust the different frequency bands in way to create the selected scale.
Frequencies are transformed to fit the defined scale.

The **scale** is choosen in the menu under the Root control.
The **Root control** defines the rootnote the scale relates to.

With 0%, no harmonisation is applied, while higher values gradually apply more frequency bending.

Under the Harmonize parameter you can choose between scale and auto tune.

Scale Mode:
In scale mode, every frequency will be forced into the given scale.
Note that with lower resolutions, the result will become more and more distorted — so it's better to use higher resolutions here.
Unfortunately, you will lose some punch and transients when forcing the frequencies into the scale.

Auto Tune:
This will analyse the most dominant frequency and then adjust the overall pitch to bring it to the selected scale.
Root sets the note note.

Note that in Auto Tune mode, the Attack parameter is used to defined the speed at which a pitchchange happens.

Please also note that in Chromatic mode, it won't have any effect, since every note is allowed anyway.
If you select a scale like Major and Root C, for example, only notes from the C major scale will be created. So even if you change the transpose from the previous section, everything will still remain in C major – just higher or lower.

The Major/Minor Chord mode will only use three notes from the scale to create a clean chord.
Chrom Inverse will invert the chromatic scale internally, resulting in a very experimental sound.
The modes after Chrom Inverse are special in that they produce certain setups related to the root note.
So they are not scales in the classical sense, but more like pre-composed musical progressions.

Attack / Release:
A high attack value will gradually fade in frequency energy — you will definitely lose transients and punch here.
Release will decay frequency energy, resulting in a reverb-like sound.
Note the Band setup window for this section. It allows you to define not only which frequencies should be affected, but also how much.

Amount / Modifier:
This section offers different algorithms in the spectral domain.
In general, **Amount** brings in the effect, and **Modify** alters its behaviour.

Transient:
Transients will be isolated; background sounds will be silenced.

MP3:
Frequencies with low energy will be silenced. The Modifier will dynamically adjust the threshold.
So, if combined with Freeze, for example, and Modify is set to 100%, you will still have some dynamics because of the dynamic threshold.

Band Shift:
Allows you to lower certain frequencies via the Band setup. The modifier can shift the defined frequency setup.

Thin: The thin functionality looks at a small energy window just above the **noise floor**.
If the energy of a band lies within this window, it will be assigned a constant value.
The Modifier lowers the window, so that only certain frequencies will pass.

Especially with a higher Envelope Release, this can produce some glimmering spectral sounds.
Unfortunately, this only works well for higher FFT sizes; for lower sizes, it leads to noisier and more distorted sounds.

Spectral Noise:
The spectral noise mode looks at the energy of frequencies where the energy is lower than a threshold defined by the Modifier.

Only these low-energy signals will pass; the more dominant frequencies will be filtered out.
You could call this the negative image of your sound. Of course, the result will be very noisy, but still related to your original sound.

Noise Gate:
The noise gate adjusts the frequency energy that is below a threshold defined by the **Amount control**.
The **Noise Level** defines whether these frequencies get suppressed (< 50%) or amplified (> 50%).

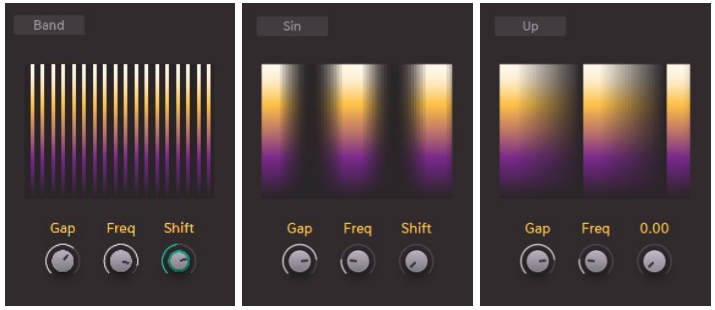
Stereo Enhancer:
Enhances the stereo image of your sound by amplifying the energy difference between the left and right channel per band.

Band Isolate:
The band isolate function only lets through the single frequency band with the most energy.
The Modifier widens the range around the strongest frequency band, thus letting more frequencies come through.

Binary:
There is only one certain energy level allowed, anything below will be zeroed, anything above will be set to the level.
The modifier adjusts the breakpoint level.

The Gap Engine

Here you can create gaps in the frequency spectrum. The gap modes define the character of the gaps.



In **Band** mode the gaps are absolute related to the settings.

In **Sin** mode a sine envelope is applied to smooth between gaps and signals.

In **Up/Down** mode a linear fade from/to the gap is applied.

The **Gap control** defines the width of the gap. With Gap at 0%, there will be no effect.

Frequency defines at which intervals the gaps will appear. A higher frequency results in more gaps across the spectrum.

Shift will shift the entire gap setup across the spectrum.

Reshape: Applies the original volume signature back to your signal.
This is especially useful if you've changed the overall volume significantly via Freeze or Release.

Stereo: Applies the setup from the Band setup below.
The idea is to define a specific frequency distribution for left and right.
You can simply randomize the Band setup to create some distribution of left and right frequencies, and then bring them in via the Stereo control.

Volume:
The overall volume of the processor.

In **Equal mode**, you get a blending functionality, while in **Wet mode,** you just add the spectralised signal to the original one.

Mix:
Mix between the original signal and the output of the spectral processor.

The frequencies are mapped to pitches (g♯-2 to g8)

The Resynthesis Processor

The Synthese processor uses spectral analysis to get an idea of the frequencies that are present.

It then uses up to 120 oscillators to (re)create something which, depending on the setup, is either close to or quite far from the original sound material.

The frequencies are mapped to pitches, but can be transformed and squeezed into scales.

The 4 different modes alter the frequency distribution of the analysed input signal:

In **normal mode** the analysed frequencies will directly be used for the synthesis engine.

In **inverse mode** the frequencies are completely mirrored. This means high frequencies in the input signal will result in low frequencies created via the synthesis engine.

Center will bend the input frequencies to the center of the currently defined range via **Low and Range**.

Border will stretch out the frequencies to the edges of the defined range.

Freeze Len / Freeze Rate (if set to Sync mode)
Defines when the results of the frequency analysis are used.

In **Sync** mode the freeze interval is related to the Host BPM. It can be set to **off** , so that the original frequencies are always used; **hold**, so that the frequencies that were present at the time will be used; or the result of the analysis is used in intervals defined by the control.

In **Trigger Mode**, the generated signal from the trigger unit will be used to define when the analysis result will be used.
The control then defines for how long the result will be used.
At 0% no freezing is applied. At 100%, the spectral information will be used to freeze the signal permanently.

In **(Re)trigger Mode**, it's possible to recapture the frequencies even if the set time has not yet passed.

Thresh:
Only frequencies with a certain energy level will get through.

Delay:
The pitch information will be delayed. The overall timing is defined via the control. The modes switch between **Always** and **Freeze Link**.
Always simply always applies the delay, while Freeze Link only advances the delay when a freeze is happening.

Delay Amt:
The Delay Amt control adjusts the amount of the delay signal or the feedback applied internally. There are **four modes:**
Add Delay: Adds the delayed signal
Mix Delay: Mixes between original and delayed signal
Feedback delay: In this mode some part of the delay signal will be fed back into the input.
Feedback inverse: A negative portion of the current delay output is fed back into the input resulting in very weird energy levels.

Delay is the first section that has a **Band setup**. You will find it in several sections and basically lets you define a per-frequency value.
Clicking on it will open the **Band editor**, handled here.

For the Delay you will define the relative delay time for each individual frequency. For a classical delay, you would set every band to 100%, but it gets much more interesting if you set up other distributions. For the delay time, a simple linear up/down will already give you some spectral delay feeling.

Transpose / Fine:
This will obviously transpose your sound up or down. But since this is applied to the analysis data, it will not affect the harmonic setup defined in the next section. Fine will slightly pitch your signal up or down.

The Harmonic section:

The Root control defines the root note of the scale you want to apply.

Please note that in Chromatic mode, it won't have any effect, since every note is allowed anyway.
If you select a scale like Major and Root C, for example, only notes from the C major scale will be created. So even if you change the transpose from the previous section, everything will still remain in C major – just higher or lower.

The Major/Minor Chord mode will only use three notes from the scale to create a clean chord.
Chrom Inverse will invert the chromatic scale internally, resulting in a very experimental sound.
The modes after Chrom Inverse are special in that they produce certain setups related to the root note.
So they are not scales in the classical sense, but more like pre-composed musical progressions.

The Voicing control does very different things according to its mode.

Noise: Adds randomness to the pitch. This helps to "get out" of the scale and get closer to the original signal.

Noise Static: Applies a static random pitch offset. It gives you a way to detune the synthesizer.

Unisono: A classical unison effect, by pitching voices slightly up and down.

Root Separation: This basically cleans up the lower part of the pitch spectrum by applying an octave scale instead of the chosen one. The control defines the position where the real scale is applied.

Single Note: In this mode, only the most dominant frequency in the original signal will be used for synthesis. The voice control adds two more voices and spreads them apart up to an octave above or below.

Blend Scale: This will blend from chromatic to the actual defined scale.

Attack: Here we control how fast the spectral analysis data is converted into an actual oscillator playing the pitch.

Release: When the frequency signal in the analysis drops, we can give it some decay time. You can create some reverb-ish, dramatic-like effects with a high release.

This section is important when you have the feeling that there are some crackles or unwanted noise.
We allow near-zero attack and release times to offer the full spectrum of possibilities.
In most cases, you should use at least a minimum attack and release (around 5%) to avoid abrupt changes in the audio signal.

This section has a **Band setup**. This means you can define for every pitch how much it should be affected by the attack/release control.

Pitch Mod
The Pitch Mode control adds variation to the frequency of the 120 oscillators in very different ways, depending on the selected mode.
Be careful here, as the pitch can quickly get very high since it is unaffected by the range.

Env Follow: Directly uses the output of the envelope follower and adds it to the frequency of the oscillator.

Sample Difference: Adds the difference of previous and current input sample value to the frequency.
This can lead to very pulsating, but also harsh sounds.

Sample Neg Difference: Same as sample difference, but the value is subtracted.

Sample Value: The current absolute input sample value is directly added to the frequency.

Sample Neg Value: The current absolute value of the input signal is subtracted from the current frequency.

Offset: A static negative offset is added to the frequency.

Lfo: The current sample value is used to drive a sine oscillator, whose output is then added to the frequency.

Lfo Neg: The envelope is used to create a sine LFO, whose output is then subtracted from the frequency.

Glide: This control works kind of like a counterpart to Pitch Mod. Especially the Sample Value and Difference modes can result in drastic frequency changes on a per-sample basis. With Glide, you can smooth out these fast changes.
There are a lot of interesting results if you just add a tiny bit of Glide to keep the drastic frequency changes in check.
Glide also applies to changes in Transpose, Fine, Root, and Voicing.
You will get a cinematic feeling, for example, when changing the Root of the Major Chord scale with maximum Glide on.

Low/Range

These controls will limit the pitch range the oscillator will work in. Low sets the lower border, while Range defines the range to the top.

The Oscillator settings:

While the sound parameters look very basic, they offer a wide range of sounds.

Wave: Seamlessly morph from sine to triangle to square to saw

PW: Add some pulse width

FM: Add some self-frequency modulation

Related to the Extra mode:

FM Variant: Change the shape of the internal FM carrier

Var1: The Frequency modulation happens with an oscillator which blends from sine to square.

Var2 : The Frequency modulation happens with an oscillator which blends from sine to saw up.

Bit: Add bitcrushing to the waveform

Reshape: Applies the original volume signature back to your signal.
This is especially useful if you changed the overall volume significantly via Freeze or Release.

Stereo: Shifts the oscillators of the different pitches to the left and right, adding some spatial width to your sound.

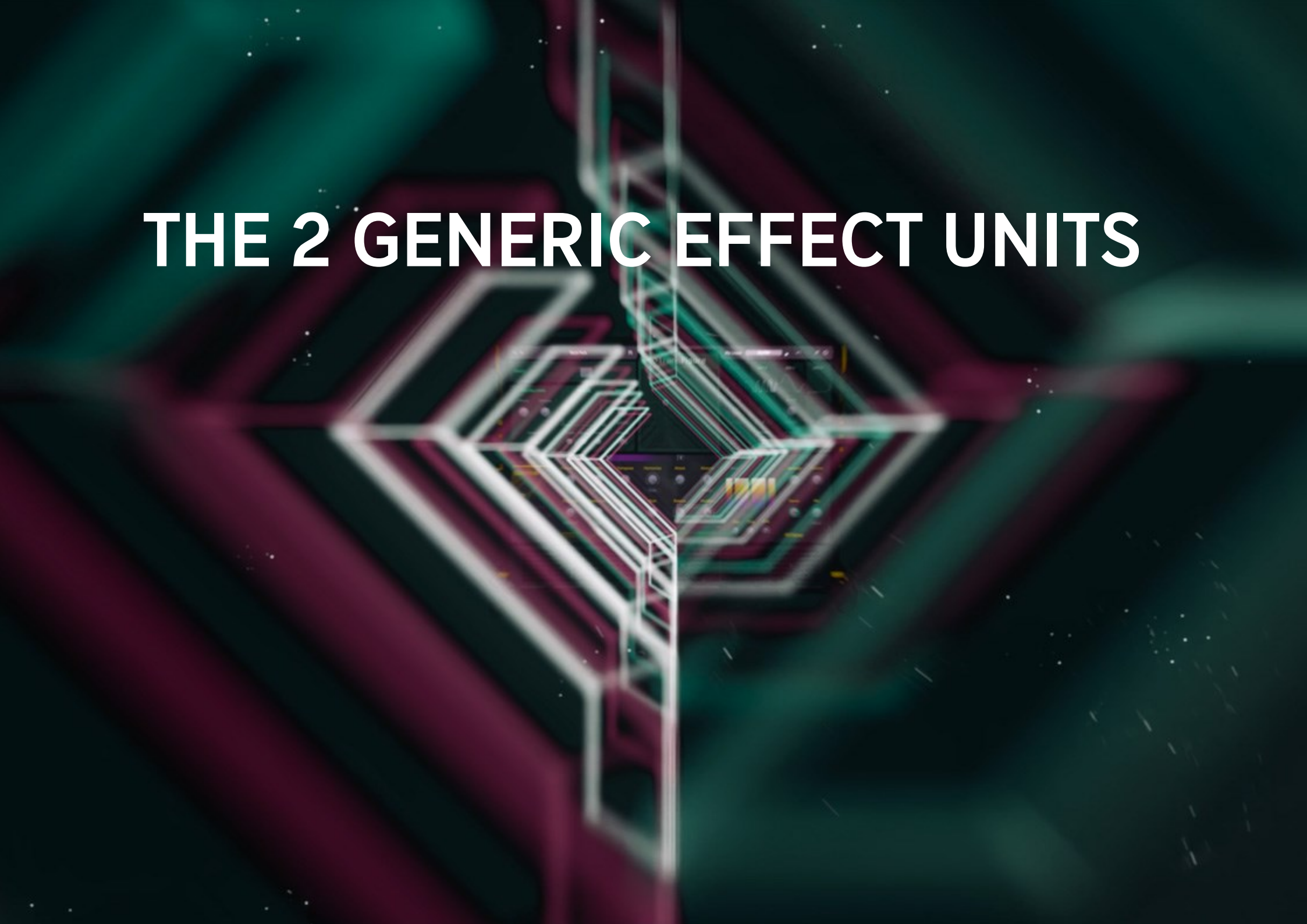
Volume: Just the overall volume of the processor. Don't hesitate to make a step sequencer assignment and fire up some stutter effect, for example.

Mix: Mix between the original and the output of the Synthese processor.

In **Equal mode**, you get a blending functionality, while in **Wet mode**, you just add the Synthese signal to the original one.

Note the Band setup. All our presets have a dedicated setup where high frequencies are gradually lowered. Otherwise, high frequencies would often be too loud in relation. But of course, you should tweak it to your liking.

THE 2 GENERIC EFFECT UNITS



Here you'll see the parameters for the individual effects for the two identical multieffect units.

In the Global Settings of TransFigure you can decide whether the effects are applied in front or behind the sound processor.

Choose from 6 different effects in the FX type dialog of FX1 and FX2.

All effects have a Mix control to adjust the Dry/Wet level.

Equal: The crossfade is shaped according to the "Equal Power Law", which leads to some signal amplification at 50/50 mix.

Wet: The wet signal is added to the dry signal.

Wet Only: Only the effect signal is audible and the Mix parameter becomes a volume control.

Delay

A typical delay with synced delay times.

Delay Left/Right
Adjusts delay time (synced to Host-BPM)

Sync Plain: Includes only straight timings (1/1...1/64).

Sync All: Includes dotted and trippled timings.

Sync 16: Synctimes in 1/16 steps

Milliseconds: Time in milliseconds.

Feedback
Determines how long the delayed signal will decay until it ends.

Pitch
The feedback signal will be pitch shifted.

Pan

Input: Control the pan of the input signal.

Feedback: Control the pan of the feedback signal.

Cross: Feeds signals from the opposite channel into the feedback path.

Ping Pong: Feeds a mono signal to the left channel, and a part of the delay's left channel to the right channel.

Cutoff
A filter to shape the frequency content of the signal.

Lowpass: Lets only frequencies below the set point pass.

Highpass: Filters out everything below the selected frequency.

Comb: Generates and adds a delayed version of the input signal.

Resonance
Resonance of the filter.

Input
The Input section of a delay determines how the original (dry) signal enters the delay processor.

Phaser

The good old 70s sound, deep and wide phasing at its best.

Freq
Controls the base frequency.

Feedback
Controls the feedback amount of the delays.

Normal: The Feedback parameter returns the signal back to the Phaser input, exaggerating the effect.

Inverse: Applies a 180 degree phase shift to the feedback signal.

Poles
The number of phaser poles used.

Lfo Amt
The amount of modulation applied to the base frequency. Different waveforms for the LFO are Sine, Triangular, Saw up/Saw down, Square and 4 Step. The Bode modes are quite special as they simulate and endless up/down movement of the LFO.

Lfo Spd
Defines the speed of the LFO.

Sync: Phasing speed, always in sync to your song tempo.

Free: LFO speed in milliseconds.

Stereo
Controls the stereo spread.

Filter

The Filter basically has 3 section.
A classic filter with Cutoff and Resonance, a Vowel filter with vowel control and Resonance.
And a distortion unit with Drive, Bitreduction and Samplerate reduction.
The classic filter offers a variety of filter types to choose from. Note that there is also an option to completely turn it off.

The Vowel filter can choose from various vowel combinations, and has also certain filter types to choose from.

Cutoff
Determines the filter frequency. 14 filter types, from classic HP/BP/LP to a strong COMB filter and a Vowel Mode are available.

Highpass: This filter can be used to reduce low frequency noise.

Bandpass: Allows frequencies within the cutoff range through, and rejects (attenuates) frequencies outside that range.

Lowpass: Frequencies below the cutoff are allowed through.

Comb: Creates and adds a delayed version of the input signal.

12dB LP: Low-pass filter with 12dB/octave slope.

12dB HP: High-pass filter with 12dB/octave slope.

12dB BR: Band-reject filter with 12dB/octave slope.

12dB BP: Band-pass filter with 12dB/octave slope.

24dB LP: Low-pass filter with 24dB/octave slope.

24dB HP: High-pass filter with 24dB/octave slope.

24dB BR: Band-reject filter with 24dB/octave slope.

24dB BP: Band-pass filter with 24dB/octave slope.

Moog 1 – 2: Special variants of a classcial Moog lowpass filter.

Off: No filter is applied.

Reso (Cutoff)
Controls the Filter resonance.

Vowel
Choose the two vowels between which the Vowel parameter crossfades.

Reso (Vowel)
Determines the filter resonance/sharpness of the vowel filter.

Highpass: This filter type can be used to reduce low frequency noise.

Lowpass: Frequencies below the cutoff frequency are allowed through.

Bandpass: Perfect for use with the vowel mode.

Comb: Creates and adds a delayed version of the input signal.

To ensure a rich vowel sound, dial in a high resonance setting. Bandpass and Comb modes deliver the best results in vowel mode.

Drive
Applies a classic overdrive effect.

Bit
Reduces the bit resolution of the signal, it offers different modes for slightly different flavours.

Normal: Classical bitcrusher.

High: Quiet signals are more affected then louder.

Low: Louder signals are more affected then quieter ones.

Clip: The audiosignal gets attenuated before bitcrushing is applied.

Progressive: Instead of bitcrushing the signal itself, the bitdepth of the difference between single samples is reduced.

SRate
Srate reduces the samplerate of the signal and also comes in different flavours.

Control the rate at which samples are captured and hold.

Normal: Classic sample and hold.

Dynamic: The sample rate reduction depends on the energy of your signal.

Absurd: The samples will fall to zero resulting in a very unnatural sound.

Smooth: The samples will be captured and then interpolated to the next sample.

Reverb (Pitch/Diffusion)

A reverb algorithm internally uses multiple delays to create a spacious sound.
These delays usally have certain timings to maximise diffusion. Here we offer the possibility to go a different route.

By selecting a scale under the **Pitch parameter** the delaytimes will be adjusted accordindlgy, resulting in an harmonic imprint on the reverbed signal.

This effect is maximised if diffusion is set to 0 and the pitch is transposed up by the octave menu under the **Diffuse parameter**.

In normal mode with Pitch 0% and diffusion 100% and no octave transposing you will have a classic reverb with no harmonization.

Note that this harmonic effect is especially notable when the reverb tail is frozen.

Size
The room size, for bigger or smaller reverbs.

Damp
Dampens the reflection of the reverb.

Width
Stereo imaging for a wider sound.

Freeze
Locks the current state of the reverb tail and holds it indefinitely.

Low Cut
Removes low-frequency content from the reverb signal.

High Cut
Attenuates the higher frequencies in the reverb signal.

Looper

The Looper captures and repeats sections of audio, allowing you to create rhythmic patterns, evolving textures, or layered soundscapes. It can be synced to the host tempo or run freely in seconds for creative flexibility.

Trigger
The Trigger parameter defines whether the looper is active or not.
When the value is above 50% the audio is captured and looped, if below 50% no effect is applied.

Size
Defines the length of the loop.

Sync: Synchronizes the loop length to the host BPM.

Sync TP (Sync Triplets): Extends sync options to include triplet-based timings.

Seconds: Switches to time-based mode, setting loop length in milliseconds/seconds instead of BPM sync.

Spd Strt
Determines the playback speed at the beginning of the loop.

Loop: Toggles continuous looping.

Continues: Lets the loop continue seamlessly without retriggering when re-engaged.

Spd End
Sets the playback speed at the end of the loop, enabling gradual pitch/time shifts.

Forward: Plays the loop in forward direction.

Pin Pong: Alternates playback direction (forward–backward) for more variation.

Slope
Changes the type of speed transition (Start to End) from linear to quadratic.

Decay
Reduces the level of repeated loops over time, making each repeat softer.

Swing
Adds rhythmic swing to the loop playback when synced, shifting certain beats for groove.

Volume
Adjusts the overall output level of the loop.

Capture

The capture effect, makes it very easy to continously create a looping and glitching soundscape.

Intervall

The intervall parameter defines the rate at which the audiosignal is captured.
The playoption (forward/ping pong) defines whether the repeated signal is played forward or in pingpong mode.

Repeat

The Repeat parameter defines how often the signal is repeated within the capture intervall.

Mesc: Here you can define for how many millisecond the sound is played before a repetition happens.

Tonal: The repetition is very short so that a tonal effect is created.

Divisor: Simply define how much repeats happen within the capture intervall.

Decay

Applies a volume envelope when lower than 100%.

Loop Soft: Creates an envelope which smoothly fades out every repetition.

Loop Hard: Uses a quick fade out for every repetition.

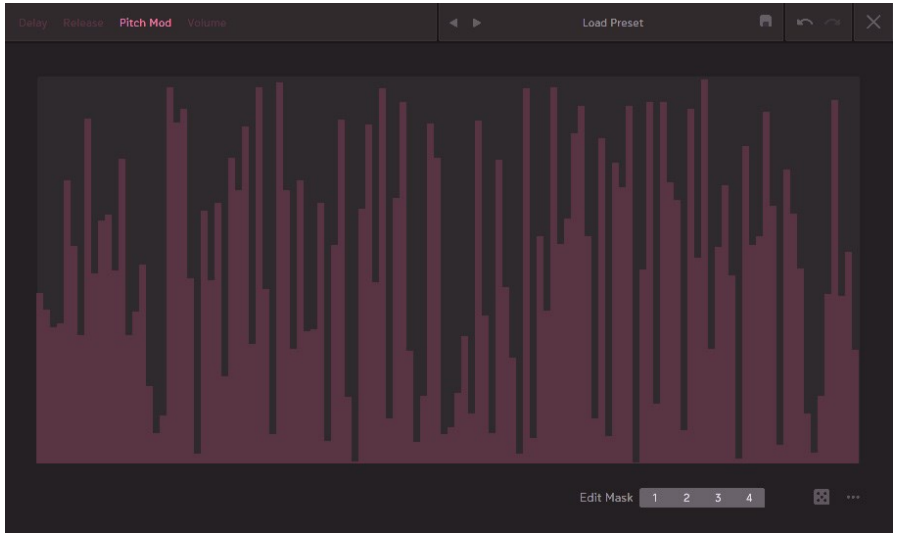
Intervall: Applies the envelope for the duration of the capture intervall instead of every repetition.

Pitch
Alters the pitch of the repetition playback.
In +-12 and +-24 the pitch will be constantly altered accordingly.
In move mode the pitch will be changed with every repetition.
In wrap mode the pitch will also be gradually changed, but when they reach 1/2 octave above/below, the pitch will be set to the opposite.
This resulty in constantly changing pitch.

Stereo
Will change the panning for every repetition.

Amp
Is a simple volume control, it allows you to apply your own envelopes via modulation.

The Band setup



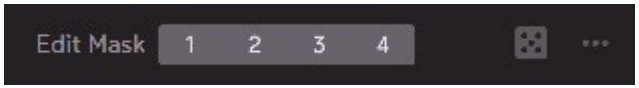
The Band setup editor lets you specifically design the behavior for each frequency.

There is a difference between the Band setup in Spectral and Resynthesis.
In Resynthesis, we have a static data size of 120, corresponding to the pitch range (G♯-2 to G8).
In Spectral, the data size is dynamic and depends on the selected FFT size (32 to 2048).
In general, the functionality always depends on the section you are in.
For example, in Resynthesis within the Delay section, you define the relative delay time for each pitch, while in the Volume section, you simply control the volume of each pitch.

In the top-left corner, you can browse through the different available sections. In the top-middle, you'll find the preset section.
On the right side, you have the options to undo/redo and to close the editor.

In the main data area, you can simply click and draw. There are also some powerful shortcut variations:
With Shift + Click, you can move the entire dataset. With Ctrl + Click, you can stretch the dataset — for example, transforming a linear upward dataset into a sloped curve.

Only in Spectral, when the number of bands exceeds 256 (FFT size 512 and above), you have the option to zoom into a specific range.
Activate the zoom option and select the data range you want to modify.
All modifications (preset load, random, drawing) will then only affect the selected range.



At the bottom, we have the Edit Mask.

This tool lets you exclude certain bands from being edited.
For example, if only 1 is active, you'll only edit bands 1, 5, 9, and so on.
If only 2 and 3 are active, you'll only edit bands 2, 3, 6, 7, 10, 11, and so on.
This allows you to create very interesting setups. For example, in the Delay Time section, we could activate 1 and 2, and then load the "Up" preset.

Next, we activate 3 and 4, and load the "Down" preset.
This results in a very interesting delay time setup, where alternating frequencies have different delay times.
There are many interesting setups to explore here.

In the bottom-right corner, we have the options to Randomize, Copy/Paste, and Smooth the data.

Note that the Band setup for the Stereo parameter in Spectral behaves slightly differently, as it defines the stereo position of each band.
Therefore, it has a bipolar setup: any value above 50% pushes the signal to the left, while anything below 50% pushes the band to the right.

SETTINGS

Signal Flow
Put the FX in front (FX First) or behind (FX Last) the sound processor.

Random
Three Randomize modes are available:
Gentle will only create some modulations while the **Extreme** setting will results in many modulation and a more chaotic setup in general.

UI Scale Factor (Zoom)
Adjust the GUI size to fit your personal preference.

HOST INTEGRATION

Transfigure works as a VST2 / VST3 / AU / AAX plugin.

Make sure that Transfigure is installed in the plug-ins folder used by your DAW.

Some hosts need to perform a plug-in rescan when a plugin is installed for the first time.

Please find all detailed instructions for your DAW below.

On **macOS**, the Transfigure installer will automatically install the VST2, VST3, and the AU flavour into:

VST2:
/Library/Audio/Plug-Ins/VST/**Transfigure.vst**

VST3:
/Library/Audio/Plug-Ins/VST3/**Transfigure.vst3**

AU:
/Library/Audio/Plugins/Components/**Transfigure.component**

If you're using Pro Tools, make sure that **AAX** is checked during the installation.
The Transfigure.aaxplugin file will be installed into: /Library/Application Support/Avid/Audio/Plug-Ins/**Transfigure.aaxplugin**

On **Windows** per default, the Transfigure installer will install the VST2 and VST3 plugin variant into:

VST2:
C:\Program Files\VSTPlugins**Transfigure.dll**

VST3:
C:\Program Files\Common Files\VST3**Transfigure.vst3**

To use Transfigure in Pro Tools make sure to check **AAX** during the installation. Per default, the Transfigure.aaxplugin file will be installed into: C:\Program Files\Common Files\Avid\Audio\Plug-Ins**Transfigure.aaxplugin**

Either add the related path to the plugin search folder in your DAW or copy the files from the default installation folder to the folder your DAW is looking for. Then make a rescan.

Installation

Download (requires login) the latest version here. Double click the installer and follow the instructions provided by the installation process.

The **Standalone** version and **Manual** will be installed into:

Windows: C:\Program Files\Sugar Bytes\Transfigure
macOS: /Applications/Sugar Bytes/Transfigure

Presets will be installed into

\Users\[Your Username]\Documents\Sugar Bytes\Transfigure

Do not move Transfigure presets after installation!

Default installation paths on **macOS**:

VST2
/Library/Audio/Plug-Ins/VST/Transfigure.vst

VST3
/Library/Audio/Plug-Ins/VST3/Transfigure.vst3

AU
/Library/Audio/Plug-Ins/Components/Transfigure.component

AAX
/Library/Application Support/Avid/Audio/Plug-Ins/Transfigure.aaxplugin

Default installation paths on **Windows**:

VST2
C:\Program Files\VSTPlugins\Transfigure.dll

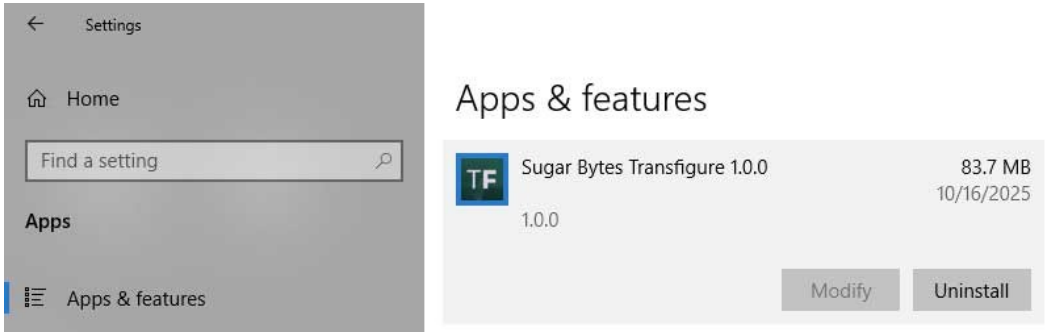
VST3
C:\Program Files\Common Files\VST3\Transfigure.vst3

AAX
C:\Program Files\Common Files\Avid\Audio\Plug-Ins\Transfigure.aaxplugin

Uninstallation

Windows

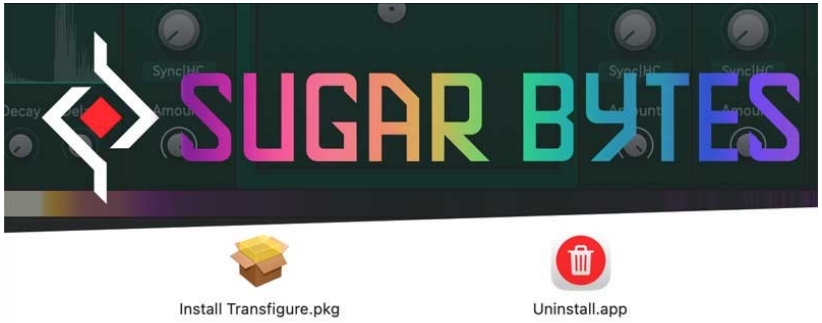
Uninstall Transfigure by using the Add/Remove Program feature in the Windows Control Panel



or via the unins000.exe in the application folder of Transfigure.

macOS

Open the downloaded Transfigure.dmg file, then invoke the uninstall.app. Select the plugin items you want to remove and confirm by clicking 'Uninstall'.



Authorization

The serial number is requested during installation. If the serial number is missing or incorrect, Transfigure won't work.
Check the top area of the Transfigure interface to verify whether your serial number is VALID. Performing a quick reinstall should fix any issues. One License covers both the macOS and Windows version and can be activated for two computers at the same time.
For use on more than two computers, please buy an additional license.

Contact

Sugar Bytes GmbH | Made of passion

Greifswalder Str. 29 | 10405 Berlin, Germany
phone:+493060920395

Sugar Bytes are:
Rico Bernhardt & Robert Fehse
Vincent Miethe
Nadine Purrmann