



introduction

Thank you for choosing **croquesolid**, the ultimate one-stop solution for quick, solid, and tasteful drum design! Whether you're a seasoned professional or just starting out, **croquesolid** offers a streamlined workflow for effortlessly shaping acoustic drum tracks. This versatile plugin is designed to clean, refine, and enhance your drums with precision and character, all within a single and intuitive interface.

For any inquiries, support, or feedback, please don't hesitate to contact us:

- **General inquiries:** info@touragedsp.com
- **Product support:** support@touragedsp.com

Alternatively, you can use our contact form on the website.

installation

croquesolid is available in VST3 and Audio Unit (AU) formats, ensuring seamless integration with all major digital audio workstations (DAWs) on both Windows and macOS platforms.

Windows Installation (VST3)

1. **Download:** Obtain the latest croquesolid installer from our official website.
2. **Run installer:** Double-click the downloaded installer and follow the on-screen instructions.
Note: In some cases, the Windows Defender Smart Screen might show up with a 'Windows protected your PC' message. If so, then please click *More info*. Under 'publisher', the installer certificate holder ('Sascha Eversmeier') will be shown, indicating that this is a genuine and signed **TOURAGE DSP** install package. You can now proceed installing safely by clicking 'Run anyway'.
3. **File locations:**
 - croquesolid VST3 plugin will be placed in the default VST3 directory
C:\Program Files\Common Files\VST
 - Additional plugin-specific configuration files located under
C:\ProgramData\TOURAGE DSP\croquesolid
 - Upon first launch of the plugin, these two folder will be created:
C:\%UserProfile%\Documents\TOURAGE DSP\croquesolid\UserPresets
C:\%UserProfile%\Documents\TOURAGE DSP\croquesolid\UserSamples
4. **Launch DAW:** Open your preferred DAW and perform a plugin rescan if necessary to detect croquesolid.

macOS Installation (VST3 and AudioUnit (AU))

1. **Download:** Obtain the latest croquesolid installer from our official website.
2. **Run the installer:** Open the downloaded installer package and follow the on-screen instructions.
3. **File locations:**
 - croquesolid VST3 plugin will be placed in the default VST3 directory
/Library/Audio/Plug-Ins/VST3
 - The AudioUnit version of the plugin will be placed in the default AU directory
/Library/Audio/Plug-Ins/Components
 - Additional plugin-specific configuration files located under

/Library/Audio/Application Support/TOURAGE DSP/croquesolid

- Upon first launch of the plugin, these two folder will be created:
 - ~/Documents/TOURAGE DSP/croquesolid/UserPresets
 - ~/Documents/TOURAGE DSP/croquesolid/UserSamples

4. Launch DAW: Open your DAW and rescan plugins if needed.

croquesolid sample pack

Part of your plugin license is a drum-sample pack we've created for users who need alternative sounds to enrich or replace their kicks, snares or toms.

The library can be downloaded separately from the user area:

- <https://touragedsp.moonbase.sh/download/croquesolid>

It comes with its own installer which copies the samples automatically inside the product installation folder. Alternatively, you can choose another folder or drive.

After copying the samples, the installer updates the **croquesolid** configuration with the chosen folder to make sure samples are found automatically.

At the time of writing this version's manual, the library is a high-quality collection of multi-layer samples of

- a handcrafted Italian-brand studio drum kit (translucent ash-wood shells),
- 20" kick, 14" snare, 12" and 14" tom,
- coated batter heads on snare & toms,
- clear kick beater head w. damping ring, resonant head w. 5" hole,
- each snare & tom recorded in stereo using matched pair of small-diaphragm studio mics,
- kick recorded in mono using a specialised kick mic.

plugin usage & workflow

preset selection

Clicking the preset list located in the top of the interface opens up a list of settings ready to be tried out on your drum signals. Those presets stay in their *factory* location, but can be used as 'blueprint' for your own settings. Saving as personal preset (via the disc icon) will place it into the user folder.

global modes

croquesolid is designed to provide an intuitive and efficient workflow for drum processing. Its modular structure allows you to tailor the processing chain to your specific needs.

Once the plugin is loaded on a track (or a bus) in your DAW, and no matter if you're starting fresh or using a preset, it's best to first check if the global **mode** is what you're after. It optimises processing for different drum elements:



- **kick mode:** Tailors the processing chain for kick drums.
- **snare | toms mode:** Optimises settings for snare and toms.
- **room | bus mode:** Adjusts for processing of room-microphone, overhead, or drum-bus tracks. Or any other signal, if you like :)

Choosing the right **mode** automatically configures optimal settings for detection sidechains, filters, and timing parameters.

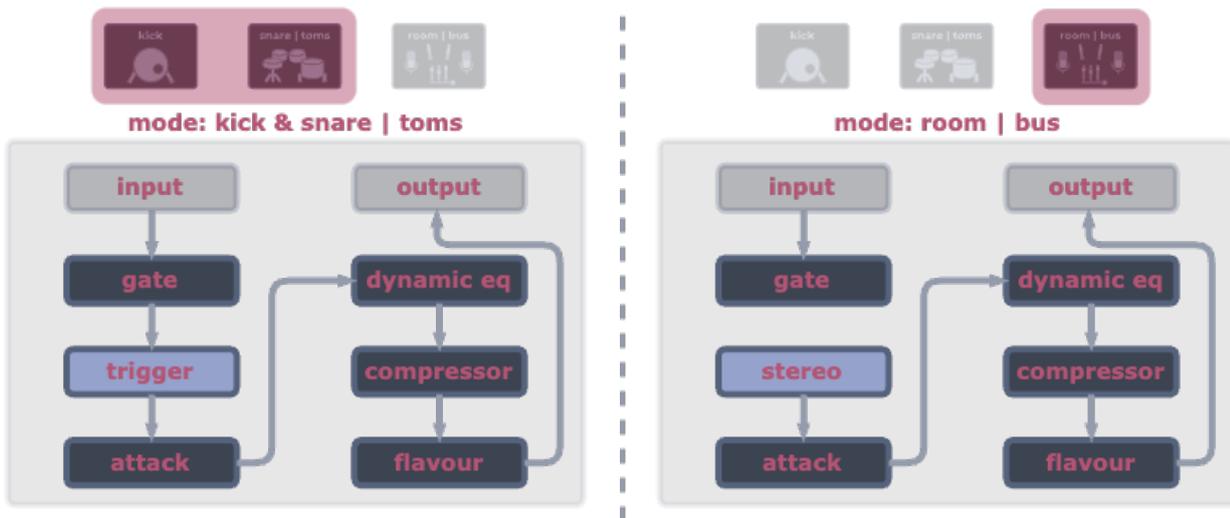
Please notice: two sub-sections in croquesolid are only available depending on the selected mode, sharing the same slot in the user interface, mutually exclusive (either-or):

1. **trigger:** Because triggering internal samples (or sending midi notes to other plugins) based on detected drum hits only makes sense on individual kick, snare or tom hits, the trigger section is not available in room/bus mode.
2. **stereo:** Typically, kick, snare or tom tracks in a multi-mic studio session are in mono. While it is always possible to internally process (and trigger) stereo signals, the **stereo** section is tailored for complex and multi-source signals. We chose to only make it available in room/bus mode, as to keep the interface clean and focused.

plugin signal flow

The signal flow within the plugin can be customised, with one limitation: mode-dependent interchange of the trigger and stereo section.

- The **trigger** section is available in **kick & snare | toms** mode.
- The **stereo** section is only available in **room | bus** mode.

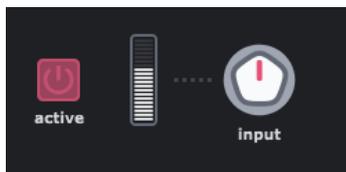


Changing the processing chain is simple: Hover over an empty area in a section's top bar. When clicked, the cursor changes to a hand icon, allowing you to drag the section (which is now blinking and appears detached, visually). Drop it over another section to swap their positions. You can repeat this as needed to reconfigure the signal path.

Changes in the processing order are saved with your presets, and likewise are also saved with your DAW project.

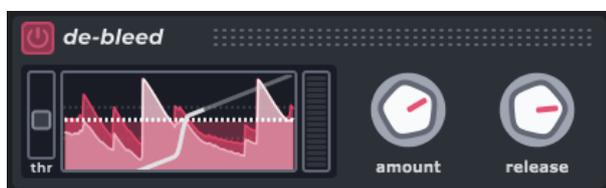
input section

This is where your signal enters **croquesolid** before any processing begins.



- **bypass:** Disables internal processing entirely, allowing for quick comparison.
- **input:** Adjusts the input level before processing, and check against the input meter. A little care about incoming levels goes a long way, as most sections expect a certain operational window (which might be large, but still).

de-bleed section (gate/expander/de-bleeder)



The **de-bleed** module reduces unwanted spill and background noise, like hi-hats or cymbals bleeding into snare or tom mics.

A waveform display shows both the original and processed signal for precise control,

along with an input-output curve to help fine-tune threshold and amount (reduction).

Typically, precise gating of acoustic drum tracks is a crucial balancing act, and many gates or expanders fall short in dissecting the audio with regard to what is *signal* and what is *noise*. In **croquesolid**, **de-bleed** is actually a combination of classic gating, soft-knee downward expansion and dynamic filtering. Depending on the global **mode**, the process adjusts for drum hits and sustain, and sets appropriate parameters automatically.

- **threshold:** Controls the level at which the gate opens. The threshold should be set to open on substantial drum hits, while still catching the musically relevant ghost notes, but still dampening unwanted background/ambient noise or signal bleed from other parts of the drum kit.
- **amount:** This knob sets a number of internal parameters that decide over the degree of gating (or more precisely: downward expansion). As can be observed on the operational curve, amount changes what traditional expanders call a ratio. Since gating in croquesolid also does dynamic filtering, the knob also sets some of it.
- **release:** Controls how the plugin shapes the transition from the initial hit to the sustain. A longer release allows more of the drum's body and resonance to come through but may also let in more background bleed. Shorter settings isolate the transient more tightly and suppress bleed more aggressively, affecting how clearly onsets are separated from surrounding signal content.

sample trigger section



Available in **Kick** and **Snare/Toms** modes, the trigger section allows you to blend or replace drums with onboard **multi-layered samples** for extra weight or texture. You can also import own samples for a customised sound.

For details on using custom WAVs, refer to the **multi-layer sample format** section at the end of this manual.

how triggering works

To trigger a sample accurately, croquesolid must differentiate between **drum hits** and unwanted background noise, such as bleed from other drums, ambient noise, or rumble. Depending on the recording, this process can be straightforward or challenging.

Tip: Pre-conditioning the signal by lightly gating it in the **gate** section can help improve detection accuracy.

setting up solid triggering

Triggering should be configured with both loud and soft hits in mind. Ensure the **input** level is correctly set so that the plugin receives a *healthy* signal, ideally within the upper half of the non-red segments on the meter.

Also, make sure the global **mode** is set correctly. This allows the trigger section to apply the optimal parameters for signal detection.

Once the input is set, configure triggering using the following controls:

- **threshold:** Defines the minimum level required for triggering. A good approach is to loop a section with soft hits (such as snare ghost notes), watch the waveform, and adjust the red **threshold** line to match the attack of the drum hits. Anything above this line will trigger a sample. Internally, **croquesolid** ensures that minor waveform fluctuations won't cause unintended retriggers or flams.
- **offset:** Typically, drum samples don't start at their exact attack transient. There can be a slight fade-in or a zero-crossing cut to prevent clicks. This is especially important when layering samples with the original sound. Adjusting **offset** shifts the original waveform back in time, aligning both layers audibly. The plugin automatically reports this delay in milliseconds to the host for *latency compensation*.
- **vel (velocity curve):** By default, all samples in the **Default Sample Pack** are recorded with maximum headroom and dynamic resolution, aiming at a 1:1 dynamic

response between the incoming audio and the triggered sample. The plugin automatically selects the appropriate WAV file based on MIDI velocity.

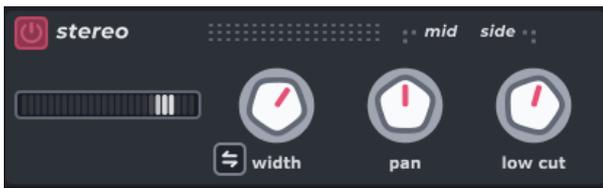
- Increasing the **vel** parameter compresses the dynamic range. Quiet hits become louder, and the sample selection is weighted more towards higher-velocity layers. This helps make performances sound tighter and more consistent, especially when the original playing is soft or uneven.
- Reducing **vel** expands the dynamic range, which is useful for preserving or enhancing detail in softer performances with subtle ghost notes or nuanced playing.
- **tun (tuning)**: Adjusts the sample *pitch* in semitones. Like with real drums, higher tuning shortens decay and shifts resonances upward, making the drum sound tighter and snappier.
Tip: Use the mouse wheel for fine-tuning in sub-semitone steps
- **decay**: Controls the sample's length as a percentage of its original duration.
 - **100%** retains the full sample
 - Lower values shorten the decay using an exponential curve, mimicking natural damping
- **mix**: Sets the balance between the original audio and the triggered sample.
 - **0% mix**: full sample replacement
 - **100% mix**: no sample, only incoming audio

additional features

- **MIDI in- & output**: The trigger section also accepts external MIDI note events. This way you can use **croquesolid** as a regular drum sample player, if required. The plugin also sends MIDI note events to external instruments within your DAW, allowing you to trigger third-party drum samplers instead of using the internal engine. Events are always send upon triggering.
 - The **MIDI note value** is determined by the selected **mode** and can be customised.
 - Check your DAW's manual for instructions on setting up external triggering and to confirm whether your host supports this feature. Compatibility may also depend on the plugin format.
- **trigger scan time**: For greater precision, the **scan time** of the trigger process can be adjusted.

Refer to the **Settings Menu** page for details on configuring MIDI output and scan time.

stereo width section



Available in Room/Bus mode, this section enables you to adjust the stereo image of stereo-miked sources. Internally, it features a mid-side (M-S) matrix, enabling you to widen or narrow the stereo field, correct the center position of the kick or snare, and

tweak the spectral balance of the perceived soundstage.

Note: This section is only effective when processing a signal with true stereo information.

-  **exchange:** Swaps the left and right channels of a stereo signal, making it easy to switch between drummer's and audience perspective.
- **width:** Adjusts the stereo field. Turning it left narrows the image; turning it right widens it. The correlation meter next to the knob shows the amount of "stereo-ness": according to standards (measured in degrees 0..180°), a mono signal (0°) is shown fully right on the meter. Stereo signals with "good" phase integrity, like X/Y-miked overheads, or anything not being spaced to far apart, will show readouts between center and probably 70-80% of the meter (towards the right). A meter readout of less than 50% (towards the left) indicates more than 90° phase difference, which is usually something to consider in a mix: Often, A/B mic setups have a large stereo field that might sound great using headphones, but could be problematic when listened on stereo speakers, and especially when mixed down to mono.
- **center:** Moves the perceived center of the stereo image. Useful for correcting mic placement issues on overheads, like the kick or snare not being dead-center.

Tip: temporarily increase **width** for extra precision; it will exaggerate the position, making **center** adjustments more drastic. When done, decrease **width** again.

- **low cut:** This control the corner frequency of a low-cut (high-pass) filter working on the side information of the stereo signal only. By moving the filter upward in the spectrum, bass information is removed from the sides. This can come in handy when the overall image is widened but some 'rumble' is coming from all around the kit. Keeping bass energy in the center is usually a key aspect in modern music production.

attack section



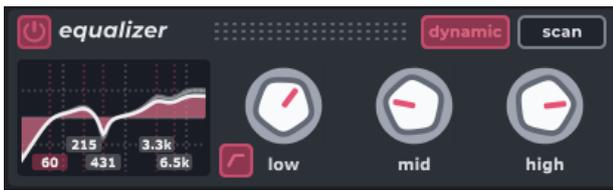
The **attack** section provides a simple yet powerful transient shaping tool, to add punch or soften the attack of your drum hits.

- **amount:** this knob is bipolar, with its center position set to 0 (no signal change). Turning clockwise enhances transient attack, adding extra punch - use cautiously on already spiky sources! Turning the knob to the left decreases the attack, making drum hits sound more softer, or "woolly".

Tip: use this section to bring the drum kit (or its parts) more forward in a mix, or to move it more to the back. Often this helps getting the balance right in an otherwise 'spectrally-ok' mix, by working on the *tightness* and *feel*. Since this section is located before the dynamic eq and compressor, quite a good amount of dynamic nuance is possible in this plugin!

- **length:** Controls the duration of the added or reduced transients. Increasing length on positive values turns the perception of attacks from snappy to punchy, but at the cost of affecting too much of the signal envelope. Same goes for reducing transients: small length settings help taming a too snappy drum hit, while larger ones would actually dig a hole in the envelope. As usual, taste and application are the key, but our general advise is to make it as short as possible, but without causing audible distortion artifacts.

equalizer section



This 3-band EQ (plus switchable low-cut) is tailored for dynamically enhancing key frequencies or attenuating unwanted or problematic ringing often associated with acoustic drums.

The equalizer has a few unique features:

- Each band's frequency spot is determined by scanning the input signal. In other words, the plugin "listens" briefly and locks onto the dominant spectral content, eliminating the need to sweep or guess frequencies. This makes tonal shaping fast and intuitive.
- The shape and bandwidth of each band are influenced by the global mode, optimised for punchy & modern drum sounds.

dynamic mode

Unlike static equalization, which applies fixed gain values, dynamic filtering takes the momentary signal energy into account when applying gain. This typically results in more natural, less "EQ-ed" sound, producing unobtrusive adjustments that preserve the musicality of the source.

spectral scan process

The section features automatic frequency scanning, identifying the three most dominant spots in the frequency spectrum (narrowed down by the chosen mode).

Clicking the **scan** button triggers capturing four consecutive drum hits, taking a spectral snapshot on each of them and averaging the results. This provides a representative analysis of the source's frequency content.

Tip: For complex material, position your DAW's play cursor over a region that best represents the sound you want to adjust. If you're focusing on a single hit, activate the scan while looping that specific section.

When a knob (**low**, **mid**, or **high**) is at the center position, no gain is applied to the corresponding frequency band.

- Turning the knob to the right applies gain to the selected frequency range. In **dynamic** mode, this means the band is expanded based on the signal envelope,

making that frequency "pop out" and become more pronounced. The maximum gain applied is determined by the knob's setting (in decibels).

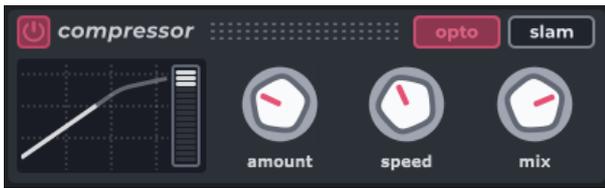
- Turning the knob to the left inverts the process. In **dynamic** mode, decreasing gain causes the frequency band to be dynamically compressed in response to its own signal energy. This is effective for momentarily damping resonances or controlling unwanted artifacts without affecting the overall tone.
- When **dynamic** is engaged, the gray region in the frequency display represents the potential gain adjustment based on the knob settings, while the red overlay in the foreground shows the actual dynamic gain being applied in real time.
- The automatic scan process has been designed to speed up your workflow, and to eliminate the need to know exact frequencies at where to set the EQ bands. But that doesn't mean it won't tell you: point inside the display with the mouse and it will momentarily show the values for **low**, **mid** & **high** frequencies as well as the **low-cut** (if enabled).
- The **low-cut** filter is 12dB/oct by default. It can be set to a steeper slope (24dB/oct) on the menu page.

frequency bands explained

Typically, drum instruments of any kind share the same properties:

- The fundamental frequency: Normally, this is where a drum is tuned to, on the batter head (or where the mic is pointing at). With cymbals, the fundamental is mostly determined by the diameter and material properties like alloy, thickness or curvature. Anyhow, this is the prominent pitch that is perceived when such an object gets hit. Some people call this thump, or thwack. Here we just call it **low**.
- An instrument's body, or its ringing: On most acoustic drums, there's usually something sticking out which is not the fundamental. Sometimes you need it, sometimes it's just bloody annoying. A snare drum's kettle resonance pattern, for instance: In some music styles, some ringing is absolutely essential, without this the drum would sound sterile or dead, whereas in other genres, or a soft tune or ballad, any ringing might spoil it all. In this section, **mid** is usually the point that has been tracked somewhere above the fundamental as a dominant spot, no matter if good or bad (you decide). In **room | bus** mode, the mid range can greatly help smoothing out overly harsh cymbals, as - once locked to a problematic frequency - the bandwidth is automatically chosen to be small and *just enough*.
- Upper-end harmonics, or treble click: beating a drum also generates high frequencies at the upper end of its frequency spectrum, like the *click* of a kick beater, or the *crack* of a snare, induced from the wires underneath. Cymbals are loved (and notorious) for creating an abundance of treble noise and *wash*. A lot happening in the treble region is quite broadband, which is what the plugin detects by making a statistical snapshot, and what is then taken as corner frequency for the high band.
 - In **kick & snare | toms** mode, the **high** band is a peaking filter plus a high shelf. This helps enhancing the *click* or *crack*, and gives some extra *air* to cut through in the mix with beauty.

compressor section



Inspired by renowned studio classics, the compressor offers two modes, **opto** and **slam**. Both include modelling old hardware control behaviour, but also some aspects of the gain staging like input/output transformer coupling and filtering.

As indicated above, the selected compressor is also dependent on the global mode, which includes overall texture, internal timings and detection filter settings.

technical note

The following section contains in-depth details. While not necessary for basic use, it may provide deeper insight into the plugin's behaviour.

- **opto**: A punchy, yet forgiving mode modelled on classic optical-compressor circuitry, great for balancing a dynamic performance while retaining the sonic integrity of the kit or individual instruments. It imparts the memory effect that the old vintage classics were popular for: prolonged gain reduction leads to increasingly longer recovery time. Conversely, those devices have quite a slow initial attack time, but strangely get faster and faster the more the gain-control element is working hard at attenuating the signal. This overall behaviour makes it really hard to make an optical compressor sound bad, and it's the glueing forgiveness we all love, and what we modelled in the opto mode.
- **slam**: A harder and more intrusive mode, achieving the typical in-your-face sound of vintage FET-style compressors incorporating that choking effect on fast transients. Back in the days, before there were (semiconductor) OP amps and highly linear VCA circuits for precise gain adjustment (and in the log domain, because: decibels), using a transistor (or precisely: FET, field-effect transistor) as the 'valve' to control gain was having to deal with a very small usable linear operational range (out of which circuits went crazy very easily). The only way to achieve stability in the regulation was to generate a closed-circuit *feedback* control loop. Instead of first looking at a signal and then decide what to do with it (like modern devices would be able to do), a feedback compressor is essentially gain-reducing at its input, thus feeding the detector a signal that it already has 'seen' and processed. The old way is essentially what we do in our house when it gets cold: we start to freeze, we turn up the heating, then realise that it becomes too warm, then turn it down a notch. Then repeat until we're comfortable. A compressor that does so is constantly gain-riding, and the effect is that it sounds quite pleasing and

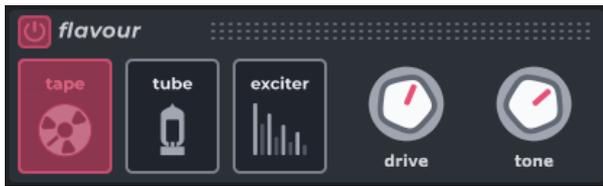
musical. But there's a catch: such a feed-back loop can't do more than roughly 3:1 or 4:1 compression ratio, meaning: not much. It therefore has to apply additional gain in the loop. This is where it gets tricky, and what made the old studio classics so special: adding additional gain in the control loop drastically increases the ratio, so that the unit can be used as limiter. The key issue is timing. Though not explicitly labelled on classic units, an adaptive attack and release system is in play, by looking at long-term fluctuations of the signal. If such a feedback compressor wouldn't use add-on gain very dynamically in its control loop, it would *choke* the audio on just about every slight transient.

We have modelled this behaviour to achieve a balance between aggressive transient control and musicality in **slam** mode.

compressor parameters

- **amount:** Adjusts how much gain reduction is applied to the signal. At the same time, it applies some gain makeup at the output, which helps in achieving a similar perceived volume
- **speed:** Tweaks the time constants and other internal parameters for the compression response. Generally, low speed settings lead to slow attack & recovery, while turning up the knob will make it faster in both aspects. As said, the compressor will always impart some program dependency, so this knob sets up the general *route*.
- **mix:** Blends the compressed and dry signal for parallel compression effects, or if you feel you should preserve some of the original transients and colour.

flavour section



Add warmth, saturation, and character with analogue-inspired processing. The **flavour** section offers three distinct processing modes, each with its own unique tonal characteristics.

All modes share two adjustable parameters, but their function depends on the mode choice:

tape

Emulates the saturation and glue of analogue tape.

- **drive:** Controls the input gain into the tape simulation. Lower values introduce subtle warmth, while higher settings enhance saturation, smooth transients, and create a soft, compressed feel. Increased drive also adds high-frequency smear, characteristic of tape recording.
- **tone:** Adjusts the amount of pre-emphasis on high frequencies before hitting the tape. Higher settings result in a brighter sound but can also introduce treble distortion at extreme values.

tube

Simulates a class-A tube preamp, adding rich harmonic content and warmth. Compared to the **tape flavour**, the tube section typically sounds more snappy, with more micro-dynamics.

- **drive:** Sets the gain of the tube stage. Low settings keep the signal mostly clean with subtle odd- and even-order harmonics, while introducing gentle soft-limiting of transients. Increasing drive intensifies harmonic saturation, adding grit, weight, and impact, especially on low-frequency signals.
- **mix:** Blends between the clean input signal and the processed tube signal, allowing for subtle enhancement or full saturation.

exciter

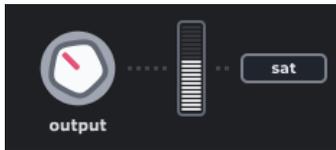
Enhances clarity, presence and low-end punch through dual-band harmonic saturation and phase-response processing.

- **low:** Enriches bass and lower-mid frequencies with harmonics, adding depth and weight.
- **high:** Saturates the upper-mid and treble range, making the sound brighter, clearer,

and more refined. Use with care, as excessive enhancement can lead to ear fatigue.

output section

Controls the final output signal level.

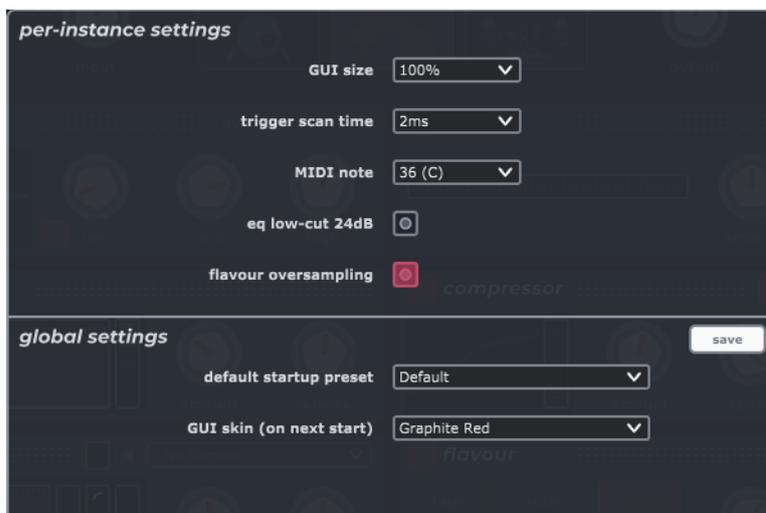


- **Output gain:** Adjusts the overall level after all processing.
- **Sat:** Engages soft saturation to prevent digital clipping by rounding off peaks near 0dBFS.

settings menu



Clicking on the 'burger icon' in the top-right corner of the interface opens the **settings** view in the upper half of the screen. Here, you can customize some things that are saved per-instance in your DAW project as well as set up the global behaviour across all instances.



per-instance settings

- **GUI size:** scales the user interface from 50% to 200% of the original size. This can be useful when working with multiple instances of the plugin on limited screen space, or if you're using a small laptop display and need better visibility.
- **trigger:**
 - **scan time:** This determines how much audio is buffered before detecting a drum hit. Instead of reacting to the first detected transient, the system analyzes an entire window, ensuring it captures the most accurate onset. While slightly increasing latency, this method improves detection accuracy and reduces false triggers caused by transient inconsistencies.

The added latency is automatically corrected by adjusting the processed audio and requesting latency compensation from the host.

- **MIDI note:** As explained in the **trigger** section, triggering internal samples also generates MIDI note events, allowing you to use an external sampler instead of or alongside the internal engine. Select the desired note value from this list, ranging from C2 (MIDI note 36) to B2 (47).
- **eq low-cut 24dB:** as mentioned in the equalizer chapter, engaging the low-cut filter normally blocks low frequencies with a 12dB-per-octave slope. This might be sufficient for many applications, but should you require a steeper cut-off, here's the tool.
- **oversample (flavour section):** The sub-modules of the **flavour** section all generate a good amount of harmonics, meaning they add distortion by principle. In the analogue domain we just wouldn't care, and enjoy the warmth and richness created. But in digital we're faced with the Nyquist frequency (half the project sample rate) as the boundary where there's no way further. Instead, spectral contents that want to go past are actually mirrored and thrown back into the spectrum inversely. Analogy: like wheels in old TV movies appearing to roll backward at certain speeds. This is called aliasing. It typically has no harmonic relationship to the original signal, and - to cut it short - sounds awful.
Enabling the **oversample** option makes sure that the simulated **tape**, **tube** or **exciter** runs at a sample rate high enough to keep aliasing at a minimum. The **flavour** section processes internally at 176.4kHz - 192kHz, depending on your project sample rate. This ensures added harmonics decay naturally without audible artifacts.

global settings

These settings apply to all plugin instances in your DAW (and equally to multiple DAWs in your system if you are using more than one):

- **default startup preset:** If you have a standard workflow and always wish to launch the plugin with a specific (custom) preset, select it here.
- **GUI skin (on next startup):** croquesolid comes with selectable skins. At the moment, these are:
 - **Graphite Red**, a high contrast skin. Vivid, fresh & friendly.
 - **Muted Blue**, a lower-contrast skin, aimed at late-night studio sessions in dim light, easy on the eyes and non-fatiguing.

Note: since changes to global settings aren't saved with a preset, don't forget to hit the save button in this section.

triggering custom samples

croquesolid allows you to add your own samples to the **trigger** section. To ensure they are recognised correctly, your files must meet the following requirements.

user sample location

- On Windows, your own samples should be placed inside this folder:
C:\%UserProfile%\Documents\TOURAGE DSP\croquesolid\UserSamples
- On macOS, the user sample location is:
~/Documents/TOURAGE DSP/croquesolid/UserSamples

one-shot samples

Single WAV files can be placed directly in the sample location, either in the root folder or inside dedicated subfolders. These are treated as one-shot samples and are mapped across the entire MIDI velocity range.

While they do not offer micro-dynamics or timbre variation, velocity-to-volume mapping is still applied for natural playback.

sample-set structure

A *sample set*, as with the **Default Sample Pack**, follows specific rules:

- All WAV files to one *sample sound* must be located in a common sub folder.
- A sample set can contain up to 127 velocity layers, each with up to 8 variations. Every layer and every variation corresponds to an individual WAV file.
- The sample set name is determined by the folder name.
- All files within a sample set must share a common substring in their filenames (detailed below).
- Each WAV file represents a velocity layer within the sound set. The more files included, the finer the dynamic resolution.
- The MIDI velocity range (0–127) is evenly divided based on the number of files, ensuring each file corresponds to its own velocity range.
- Each velocity layer can contain up to 8 variations to prevent repetitive playback (the 'machine-gun effect'). Variations are played in a rotating cycle when the same layer is triggered repeatedly at a similar velocity.

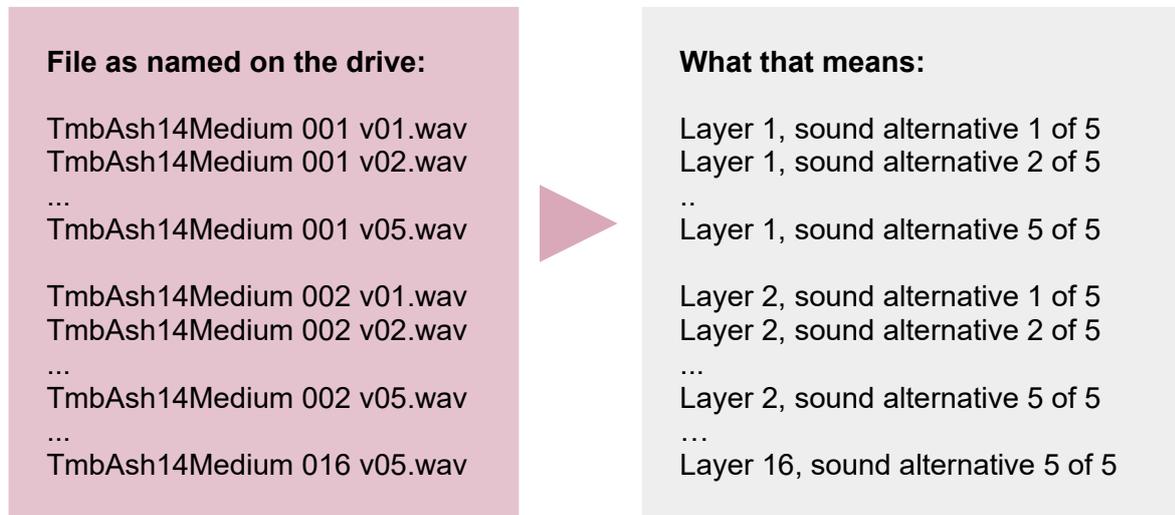
audio format

The **Default Sample Pack** is recorded at 44.1kHz / 24-bit. You can import files at any sample rate used in your DAW project, as long as all files within a set have the same format.

Note: Higher sample rates significantly increase memory usage.

file naming convention

To ensure **croquesolid** correctly assigns samples to velocity layers (and variations within those layers), files must follow a specific naming pattern. Let's look at an example:



A single filename is therefore composed like so:

SAMPLESETNAME_{001..128}_v{01..08}.wav

creating custom samples

There are plenty of resources available online about recording drums, microphone placement, and studio techniques. While everyone has their own approach, modern studio drum sounds follow a general consensus, making it easy to find inspiration.

However, when it comes to sampling acoustic drums, there's far less widely available information, and the process is often treated as a kind of 'black magic'. Many commercial sample libraries might give the impression that creating your own drum samples isn't worth it, suggesting that you need high-end gear, vintage drum kits, pristine studio acoustics, and extensive experience to do it right.

We strongly encourage you to experiment with recording your own drum samples, even if it's just a few hits for your own productions. If it's not perfect on the first try, there are still great benefits:

- Your sound, your identity. Custom samples are unique to you, becoming part of your signature as an artist or engineer.
- A backup for mixing. If you're already tracking drums, it's easy to take a few minutes to record extra hits for later use. These can save a mix if a performance isn't quite right or needs subtle reinforcement.
- A great learning experience. Sampling teaches you a lot about individual drum sounds, mic techniques, and the fine details of editing and consistency. The extra attention to detail will pay off in all aspects of production.

approaches to sampling a drum kit

When recording drum samples, there are two main approaches:

1. Full-kit studio setup
 - This method mirrors a typical studio recording session: miking up the full kit as if tracking a live performance. The advantage is that your samples will sound identical to the recorded kit, including natural snare buzz, sympathetic resonances, and the interaction between drums.
 - Ideal for mixdown replacement, where the samples blend seamlessly into the original performance.
 - Can include close mics only or room mics for a more spacious sound.
 - Requires proper studio space, multiple microphones, and more setup time.

2. Isolated kit piece sampling

- This approach records each drum separately, without interaction from other kit elements.
- Fewer mics, simpler setup, and less reliance on room acoustics.
- Works well in home studios and is easier to repeat consistently (distances and mic placements can be documented).
- When recording in stereo, you can position mics per drum while keeping their relative spacing similar to a full kit.

The Default Sample Pack follows this second approach, with a few tweaks:

- The kick drum was miked conventionally: a mono kick mic, placed about 1/3 inside the shell, angled toward the beater.
- Snares and toms were recorded in stereo using an A/B pair of small-diaphragm condenser mics (cardioid pattern).
- The mics were spaced 30cm (~1ft) apart, angled inward, and positioned above the drum head, placed equidistant between the center and rim to balance *impact* and *ringing*.
- The recording space was a dry control room (RT60 \approx 0.2s), so no additional ambience was captured.

sampling process

Recording multi-layer drum samples requires patience and consistency. If you're recording alone, choose a time when you're fresh, focused, and relaxed.

- Consistency is key. Striking a drum at consistent velocities, gradually increasing power step by step, is crucial.
- Our recordings typically contain 15–20 velocity layers per drum, from ghost notes to full power hits. More layers come with better dynamic resolution, but require more work, discipline and of course memory on disk and in RAM.
- Most snare and tom samples in the **Default Sample Pack** have five variations per velocity layer, helping to avoid the machine-gun effect. Typically, having more inter-layer variations is more important than the absolute number of layers.

Since recording more variations than needed provides flexibility, we recorded extra takes and later selected the best five during editing.

editing and exporting

For best results, samples must be cleanly cut, properly levelled, and consistently formatted before being imported into croquesolid.

The workflow may vary depending on the tools you use (external audio editor or DAW), but the key steps remain the same:

- Ensure each hit is accurately sliced, with no unwanted silence before the attack.
- Apply consistent gain staging across all layers to preserve dynamics.
- Keep recorded dynamic range intact, don't over-process.
- Avoid clipping (0dBFS), especially with integer WAV formats.
- Light high-pass filtering may be useful for snares and toms, depending on mic choice.
- If needed, use envelopes or fades to shape the decay of each sample.
- Export files in WAV format while maintaining a uniform bit depth and sample rate across the entire set.

While there's no single right way to approach this, finding an efficient workflow with your preferred tools will make the process much easier over time.

Most importantly: **have fun!**