Softube

INSTRUMENTS USER MANUAL

Supporting VST/VST3/AU/AAX

Rev. Oct 30, 2020

Softube User Manual

Disclaimer

Every effort has been made to ensure that the information in this manual is accurate. However, there are a chance that we have made mistakes, and we hope that you understand that we are only humans. Please let us know about the mistake, and we'll fix it in the mix (or in the next version of this manual).

Support

On the Softube website (`www.softube.com`) you will find answers to common questions (FAQ) and other topics that might interest you.

Support questions can be posted at `http://www.softube.com`, where we will help you as fast as we can!

Web: `www.softube.com`
E-mail: `info@softube.com`
Phone: +46 13 21 1623 (9 am – 5 pm CET)

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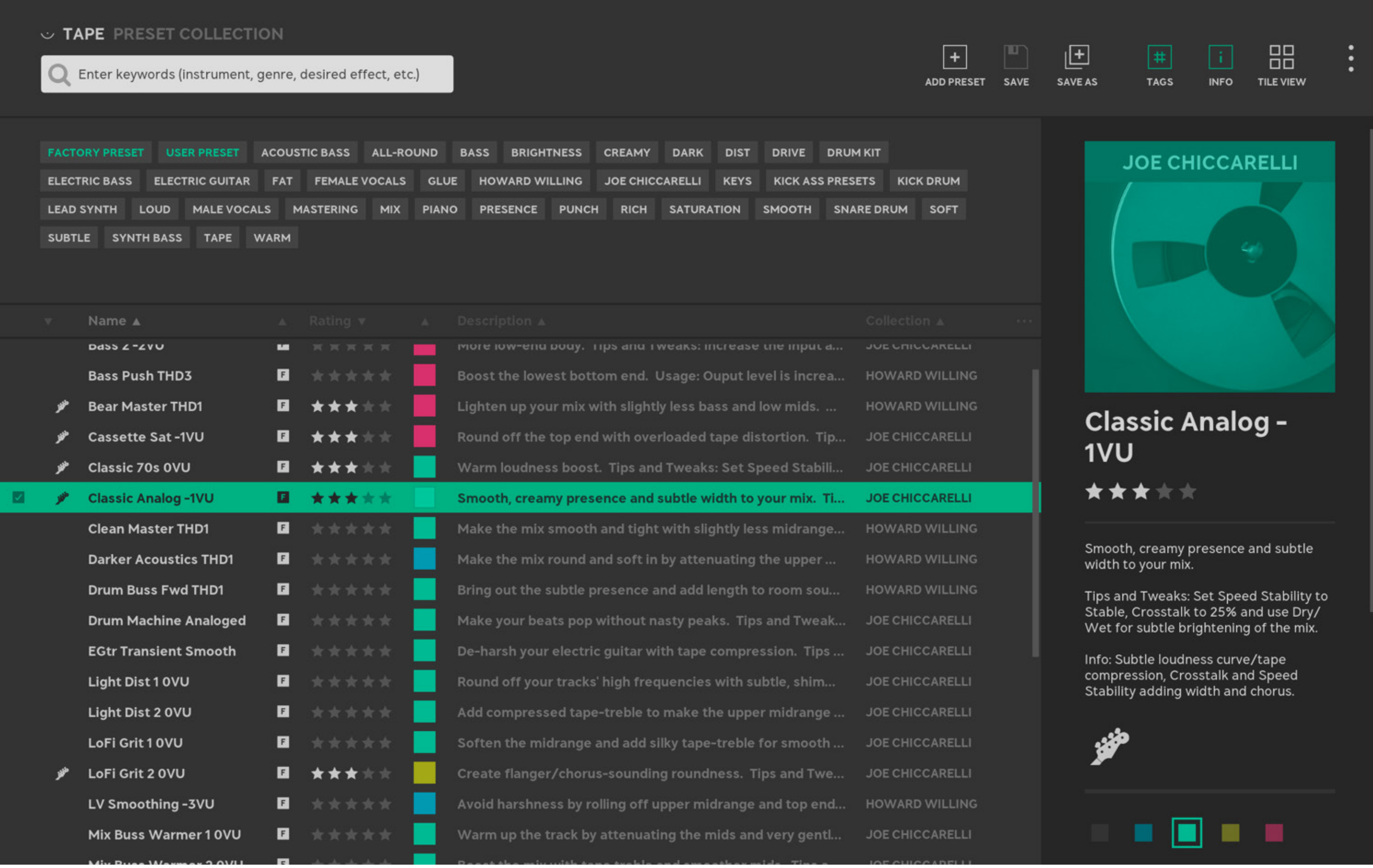


1 Preset Collection

The Preset Collection is a tool to organize your presets in logic, simple, advanced or mysterious (?) ways (it’s up to you!), or just a simple mechanism to save your favorite sounds and easily browse through artist's presets.

You can either use the simplified version in the menu bar of each Softube plug-in, or you can press the open window icon to open the full Preset Collection.

Browsing presets

Use the ◀▶ buttons in the menu row at the top of the plug-in to step through presets. Click ▼ to open a menu to select presets. By default, presets are sorted by "collection", usually by artist or theme.

Saving Presets

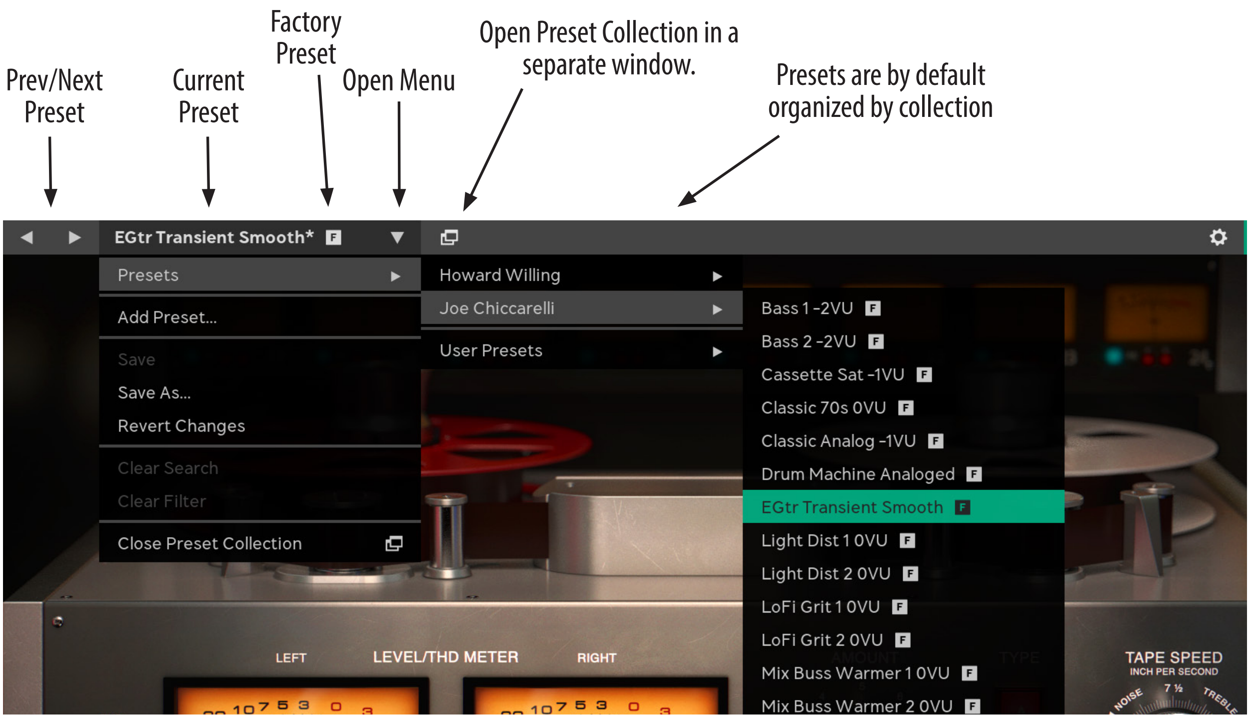
Press "ADD PRESET..." in the dropdown menu to save the current settings as a new preset. Type the name of the preset and press enter. If you've made changes to a current preset and wish to overwrite that preset, just press "SAVE". Press "SAVE AS..." if you want to save it with a new name.

Searching for Presets

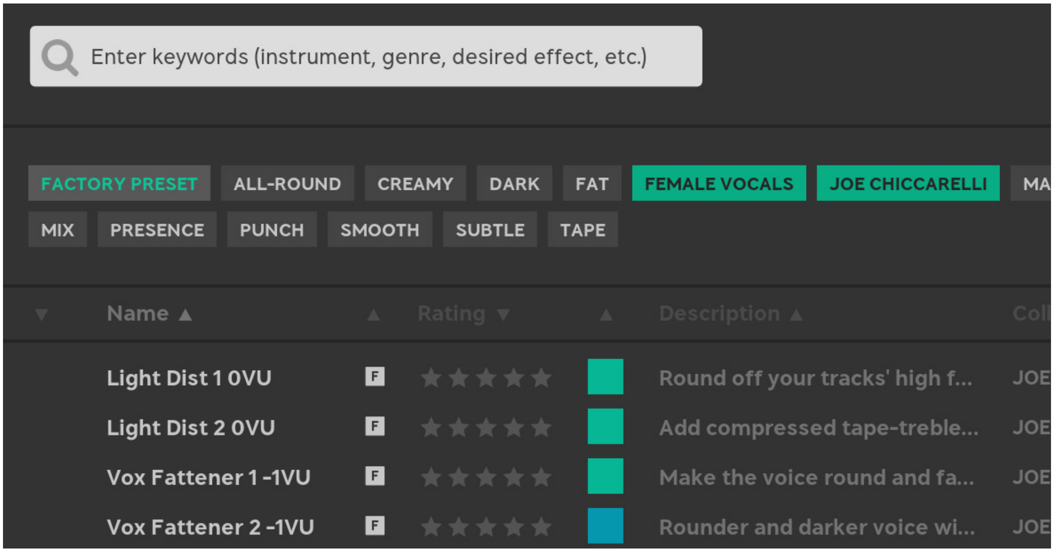
If you want to find a specific preset and have a lot of presets, there are many smart ways to search for preset, and all of them require that you open the Preset Collection by clicking the ⧉ button.

Search by tag

All presets have a number of tags associated with them, in general they describe the function ("DISTORTION", "EQ", "COMPRESSION"), the use case ("FEMALE VOCALS", "BASS") and the character ("CREAMY", "DARK"), etc. In plug-ins that use modules, such as MODULAR, the tags also include which modules are used by the preset (for example "SATURATION KNOB")



Browse presets directly from the menu bar at the top of the plug-in.



Two tags ("Female Vocals" and "Joe Chiccarelli") has been selected to show all Joe's presets suitable for female vocals.

Search box

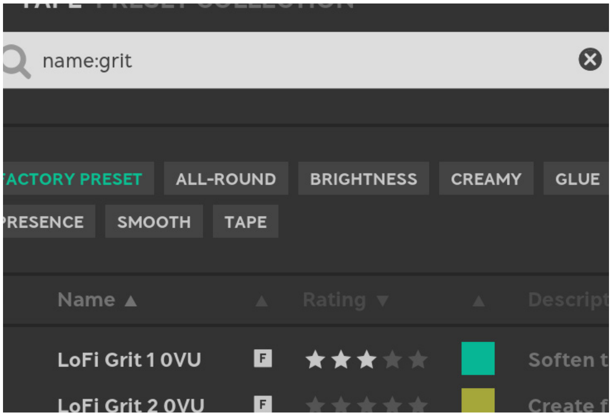
A search in the SEARCH BOX will search through all metadata (name, tags, description, etc). If you want to narrow your search, you can specify what you want to search for by using a qualifier, such as "name:" or "description:".

Possible search qualifiers are *name, desc, description, tag, tags*, and *collection*.

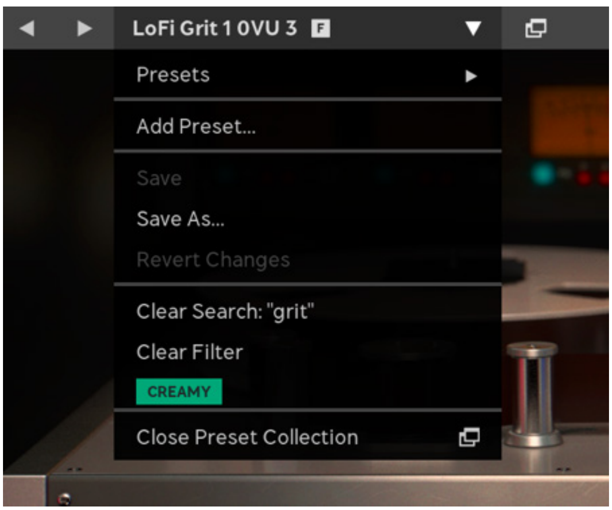
Search in the menu bar

When you have selected a subset of presets, for example by searching for "vocals" or using a tag, the search results are available directly from the plug-in's menu bar. That means that you can easily step through the presets using the ◀▶ buttons.

Open the menu to see the search criteria or clear the search.



Type "name:" before the search criteria to limit the search to the name of the preset.



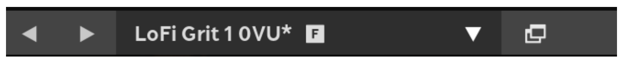
The menu shows the current search criteria, currently selected tags, and options to clear the filter and search criteria.

Plug-in Settings vs. "Metadata"

A Softube preset consists of two parts: the plug-in's settings and the description of it, what we call "*metadata*". Metadata is everything that's not the included in the plug-in's settings, for example preset name, description, color, rating, tags.

When you make a new preset by clicking "ADD NEW PRESET…" the only metadata that you save is the preset name. You need to open Preset Collection to add other metadata, such as a description and tags. If you instead use "SAVE AS…" when you save a preset, the metadata (for example tags) in the currently selected preset will be carried over to the new preset.

Whenever you change the settings in the plug-in, the preset name will be marked with an asterisk * to indicate the current settings are different from the current preset. It also indicates that the preset has changes that aren't saved. You need to click "SAVE" or "SAVE AS" to save those changes.



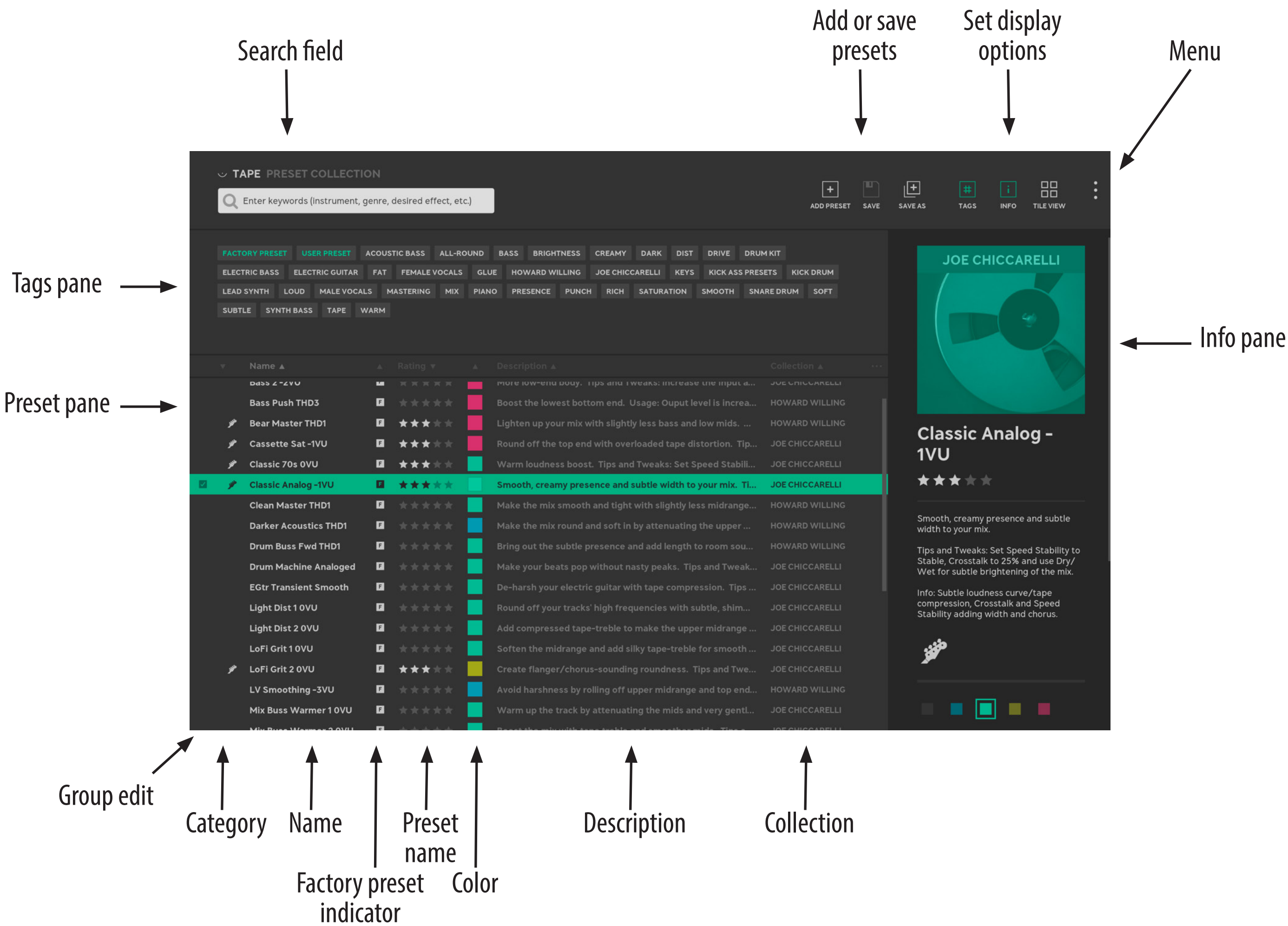
Asterisk (*) indicate that the settings of the plug-in is different from the saved preset.

When you make changes to the metadata, you don't have to save these. All metadata changes are saved immediately in the preset database.

**) An asterisk looks like this: **
**) ibid.*

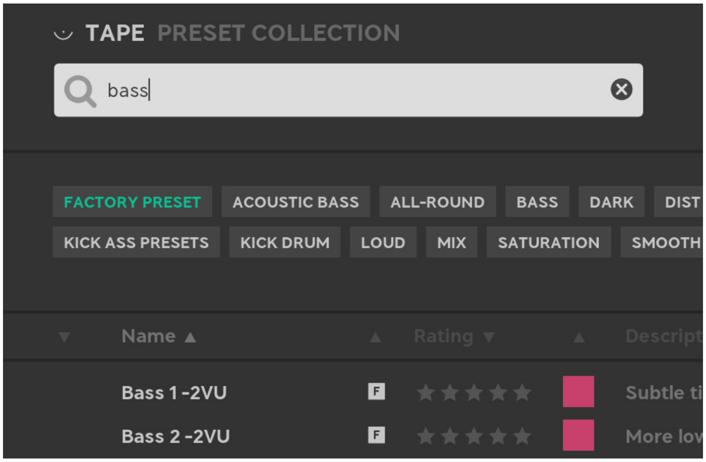
Preset Collection

Click ⧉ to open the Preset Collection. Here you can organize, colorize, tag, add icons and images and sort your presets.



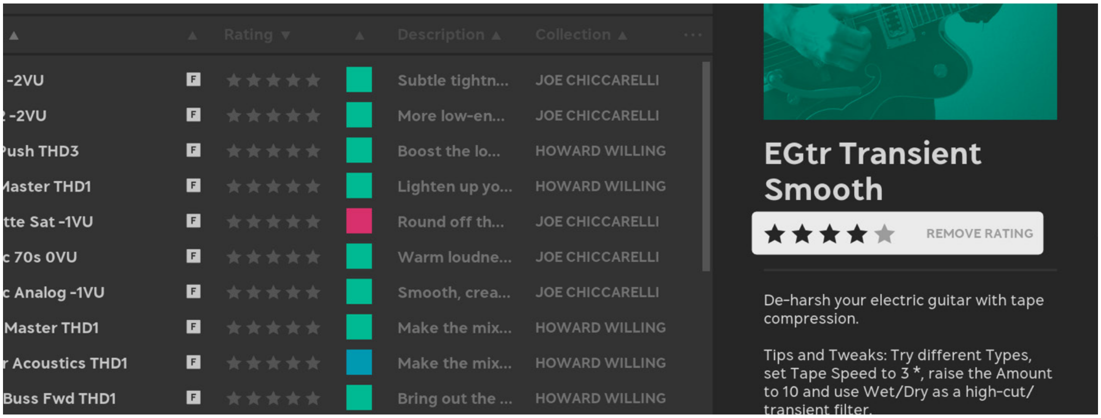
Workflow

For a plug-in with hundreds of presets, the easiest way to start is to type something in the search field, such as “bass”, or select an appropriate tag.



Type in the search field to find presets.

Use the keyboard up/down arrows to step through presets, and if you find something you like (or dislike), change the rating of that preset so that you can easily find it later.



Click on the stars in the Info pane to change the rating. It is also possible to right-click the preset and change rating from the context menu.

Philosophy

The main ideas behind Preset Collection are

1. A preset name doesn't give enough information about how to use a preset. Sometimes you need more info, for example what to listen for, how to tweak it, in which context etc.
2. Everyone wants to organize their presets in different ways.
3. Tags are a simple way to create "folder like" structures, but without being limited by placing the preset in a single folder. A tag can be a use case, a project name, or just about anything!

With the preset's description you'll be able to add info, for instance how many dBs of gain reduction you need for the drum bus to really glue together, and with names, tags, ratings, categories, and colors you can organize those presets any way you want. The tags become powerful if you want to organize presets after projects you're working on. Tag each preset you make/use with the project name, and you'll have an extra dimension to use when you browse presets.

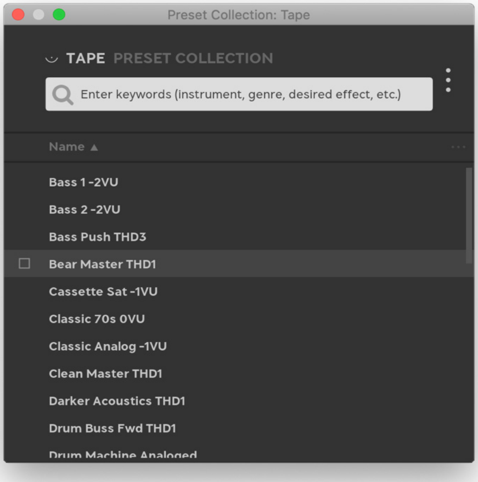
> Don't be afraid to add or change metadata in the factory presets. (And it's always possible to restore the original metadata later if you want to!)

Organizing presets

All factory presets come with tags and color, but you can remove those colors and tags and organize them in other ways. You can mark your favorite presets with a high rating, a specific tag ("kick ass presets!", "great for accordion"), a color, or a category, to make them easier to find later. You can select several presets at once to, for example, reset their color, and use your own color scheme. All fields, except COLLECTION, are possible to change.

Customizing the Preset Collection

You can decide yourself how much or how little info you want to show in the Preset Collection, and these settings are stored globally.



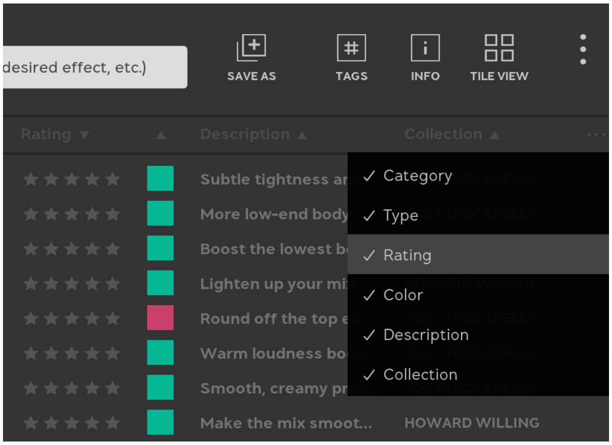
The minimal view of Preset Collection, only preset names and search field are visible.

Showing/hiding info panes

Click on [#] or [i] to show and hide the Tags and Info pane, respectively.

Showing/hiding columns

Click on the menu (•••) next to the columns header to turn on/off columns in the preset pane.



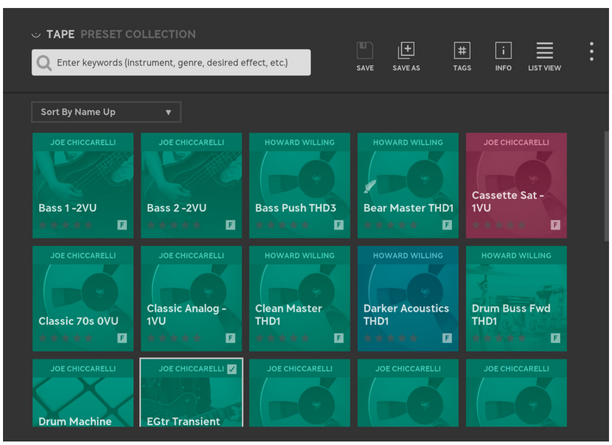
Select which columns are visible from the menu.

Sorting

Use the up/down arrows (▲▼) next the the column header to change preset sorting.

Tile view/List view

Click on Tile View to show presets as tiles, or List View to get back to the default mode.



The Tile View.

Resizing tags pane or window

You can easily resize the Preset Collection window or the size of the TAGS pane by clicking the edge and drag it.

User Interface

Search Row Searches in name, description, tags, etc. To specify a particular field to search in, use one of these qualifiers: *name, desc, description, tag, tags*, and *collection*, for example "*collection:chiccarelli*"

Add Preset Create a new, empty, preset from the current plug-in settings.

Save Preset Overwrite the selected preset with the current plug-in settings . Only possible with user presets. If the current preset is a factory preset, use "SAVE AS" instead.

Save As Save the current plug-in settings, together with the selected presets meta data (tags, description, etc) under a new name.

Tags Show/hide the TAGS pane.

Info Show/hide the INFO pane.

Tile view, List view Switch between a list of presets (LIST VIEW) or tiled images (TILE VIEW)

⋮ Menu Opens the menu with some additional options. See below for Menu options.

Menu Options

Add Preset Add a new preset.

Save Overwrite the selected preset with the current plug-in settings.

Save As Copy the selected preset to a new preset with the current plug-in's settings.

Show/Hide Tags Show/hide the TAGS pane.

Show/Hide Info Show/hide the INFO pane.

Show Tile View Show the TILE VIEW instead of LIST VIEW.

Import Preset(s) Import presets from file, for instance if you downloaded a ".softubebundle" file from www.softube.com

Export Selected User Preset(s) Exports the currently selected presets to a ".softubebundle" file, so that you can send them to a friend.

Enter Group Edit Mode Let's you select several presets at once, which is useful if you want to export a batch of presets, or change tags, description or other meta data in several presets at once.

Preferences Opens additional preferences, listed below.

Preferences

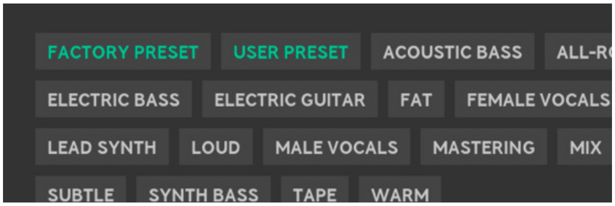
Warn When… Turn warnings on/off when for example deleting or overwriting presets.

Show Images In Info Panel Turn off to hide the image in the INFO PANEL.

Group presets by collection/group presets by category Change how presets are grouped in the presets menu in the plug-in menu bar.

Pro tip: select "Group Presets By Category" to get a flat list for all uncategorized presets in the menu bar.

Tags pane ⊞

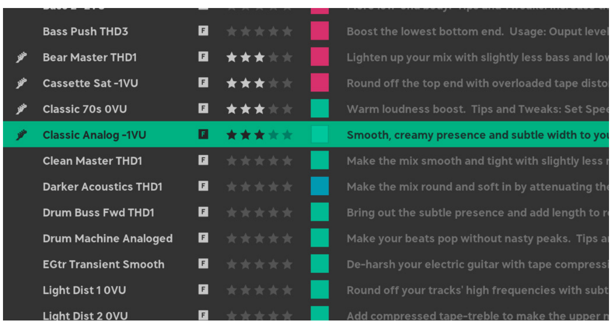


Use the Tags pane to filter presets by tag.

The TAGS PANE lets you filter presets by selecting one or several tags. Select for example the tag "MALE VOCALS" if you want to find all presets suitable for male vocals. You can select several tags at once.

You can add tags to a preset by clicking "ADD TAG ✐" in the INFO PANE, and subsequently remove tags by clicking on the ⊗ on the tag.

Presets pane



The Presets pane shows all presets in the current search or filter.

The PRESETS pane contains all presets in the current search. You can sort presets, select a preset, or edit a preset from the presets pane. The PRESETS pane can either be visualized as a list, or as a "TILE VIEW" with images for each preset.

> Tip: Double click on the preset name to change it.

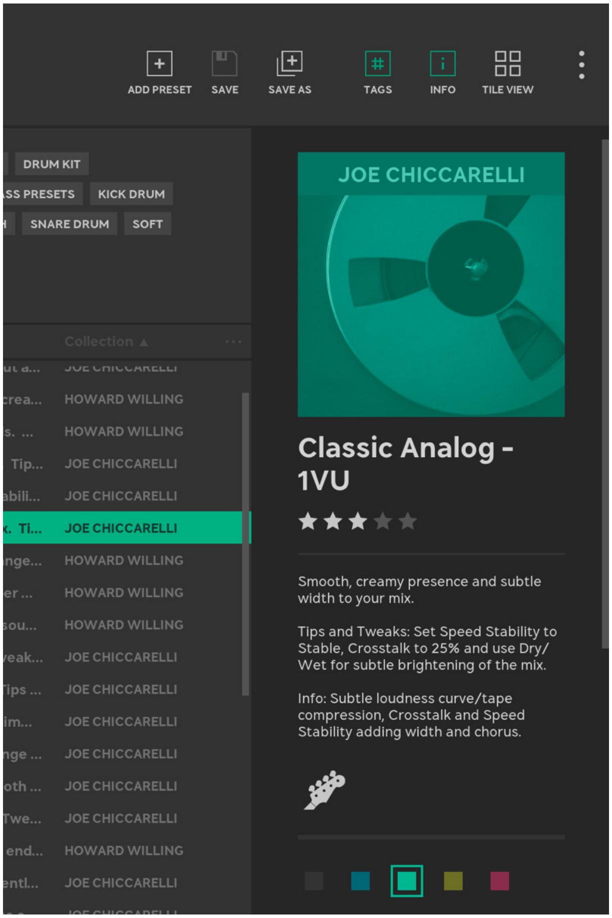
To sort presets in the LIST VIEW, click on ▲▼ for a column to sort down or up for that column. In the TILE VIEW, open the "SORT BY…" drop down to select sorting.

Context Menu (Right Click on a Preset)

Right-click on a preset to bring up a context menu with the following options:

Save	Save preset
Save As…	Save as a preset with a new name.
Revert Changes	Revert all changes made to the preset.
Rename…	Rename the current preset
Export…	Export the current preset
Delete	Delete the current preset
Restore Factory Metadata	If metadata edits, for example color, tags or ratings were made to a factory preset, you can revert them here.
Category	Sets the category of a preset. By default, no categories have been set.
Rating	Sets/changes the rating (0-5 stars) of a preset.
Color	Sets/changes the color of the preset.

Info pane



The Info pane contains all metadata for the currently selected preset and lets you edit these fields.

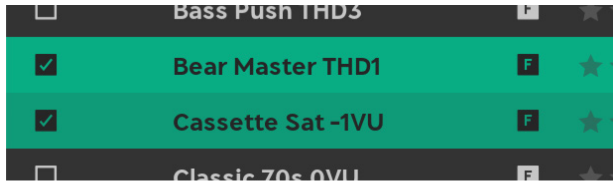
In the INFO pane you can edit the metadata of the selected preset. You can change image, name, rating, description, category, tags and color. Whenever it is possible to change a field, a pen icon ✎ will appear. All changes made to the description of the plug-in are saved immediately.

Change Image

The image needs to be in PNG format, and will be converted to black-and-white on import.

Group Edit Mode

It's easy to edit several presets at once, for instance if you want to change rating or add a tag to multiple presets. Enter the GROUP EDIT mode by selecting several presets in the check box on the left. Once in GROUP EDIT mode, you can use SHIFT + click to select a range of presets.



Select several presets by clicking the check box to the left.

In GROUP EDIT mode it is possible to edit several presets at the same time:

- Change rating, color, and category
- Append text to the name or description
- Add tags, or remove tags common for all selected presets

2 MIDI Mapping

All Softube instruments can be mapped to any midi-controller that can output MIDI controller change (sometimes just called MIDI CC) data. This can be done through the MIDI CC linking mode menu at the top right in the plugin.

When clicking on the MIDI symbol, this dropdown menu appears displaying all previously mapped parameters in your project/song (if any) and at the top the option for you to enter MIDI CC linking mode by clicking on the text that says “Click here to enter MIDI CC linking mode”. At the bottom, the “Load from file...” and “Save to file...” options – more about these later.

After entering MIDI CC linking mode by clicking on the text “Click here to enter MIDI CC linking mode”, the on-screen message will prompt you to “select parameter to link with MIDI CC” and this means that you can click on any parameter you would like to link to your controller.

After clicking on a parameter of choice you'll be prompted to move the slider or dial on your midicontroller to link this physical controller to the chosen parameter in the Softube instrument. If there were a mistake you can always click on “Exit MIDI CC linking mode” to abort this operation.

Now you have tweaked your MIDI controller and will see the knob you linked it to moving on screen while you're moving it – like magic! An on-screen message displays the parameter name link and to which MIDI CC message it is linked to. OK, something went wrong? Don't worry – you can unlink the controller again just as easily by clicking on the unlink "parameter name"-message.

Any number of parameters can be linked to the same MIDI controller knob although multiple knob cannot be linked to the same parameter.

Exit MIDI CC linking mode by clicking on "exiting MIDI CC linking mode" in the on-screen pop-up or by clicking on the MIDI symbol in the top left corner again. All linked parameters will be listed in the drop-down menu until you click elsewhere in the GUI.

The "Load from file..." and "Save to file..." options in the top menu are used for saving and loading previously mapped MIDI controller templates in the file format ".softubemidipreset". The saved midi map contains information about the instrument-type and will not be able to load within another Softube instrument.

MODULAR EXCLUSIVE

3 Heartbeat

SYNTHESIZED SOUNDS AND imaginary worlds have inspired musicians since the mid 1900s when Dr. Bob Moog invented the first ever voltage controlled synthesizer modules, and eventually launched the electronic synthesizer as a new instrument into the limelight of every day musicians. Around the same time, electronic organ-makers looked into ways of electronically reproducing drums and rhythmic sounds. In the 70s, the electronic drum machine made its way into the public mind and electronic drum production could soon be heard in everything from disco, electro and hip-hop to pop and rock.

This legacy of finding new and exiting electronic percussive sounds is something we want to convey in Softube Heartbeat—the joy and excitement of exploring new and interesting percussive worlds by looking back at history, but at the same time adding something new to the concept.

Introduction

Heartbeat is an innovative software drum synth with a familiar, yet unique, sound character. A world class effects section is included, as well as the innovative Auto Layer Machine which will take your beats to unexpected places. While Heartbeat draws inspiration from the best analog drum synths from the 1980s, it does not emulate any existing drum machine. The sound mostly originates from Softube's own modeled analog synthesis, which has been augmented with carefully selected waveforms.

The core of Heartbeat consists of the eight instruments. You will find two different bass drum instruments, which can be as punchy and deep as you want them, but are also perfectly capable of producing snappy and hard hitting woody textures. The two dedicated snare drum instruments have six parameters each which allows you to achieve anything from edgy rimshots, soft and whispery snare rolls to machine-like claps.

The two percussion instruments are identical and can be used to model anything from 80s style synthetic toms to cowbells and noise drops. And just like the other instruments, the hihat and cymbal channels offer flexible synthesis engines—tweak to your heart's desire! But the idea behind Heartbeat is to make it a one-stop shop for your beat creation, so we also added an effects section and the innovative Auto Layer Machine.

1. Drum channels
2. Master bus
3. Auto layer machine
4. Valley People Dyna-mite
5. Filter echo
6. TSAR-1D Reverb
7. Utility
8. Instruments
9. Mixer
10. Global

Heartbeat's Sections

The left half of Heartbeat's graphical interface is taken up by the eight **Drum Channels**. These all consist of (from top to bottom) the **Utility** section, the **Instrument** and the **Mixer Channel**. By default, the **Mixer Channels**' outputs are summed and sent to the included **Valley People Dyna-mite** compressor/limiter/gate (read more below), and then on to the **Master Channel** on the right side of the interface.

Below the **Valley People Dyna-mite** unit are the **Filter Echo** and the **TSAR-1D Reverb** effects. Each mixer channel has send knobs (labeled ECHO and REV respectively), that determine the level of sound that is being fed from the respective channels into the two send effects. The output of the send effects is then summed with the output of each **Drum Channel**, and fed into the **Master Bus**.

Above the **Valley People Dyna-mite** unit you will find the **Auto Layer Machine**. This is a device that can be used to layer sounds or trigger a chain of events, in order to create new sound textures or create automatic fill patterns in up to four steps. By pulling the **Chaos** slider to the right, an element of randomness is introduced—so Heartbeat has a mind of its own and might give you some unexpected results.

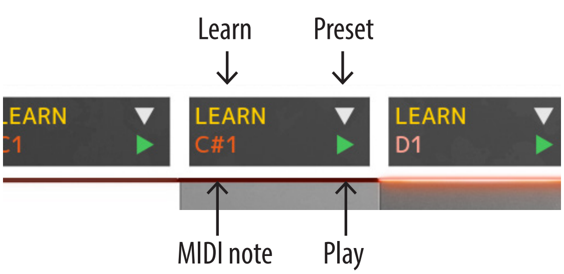
Getting Started

If you're familiar with working with software instruments, this section may be all you need to get started. Refer to the in-depth parts of this manual to learn the details.

After you have finished installing Heartbeat, open a new song in your DAW, and launch Heartbeat which you will find in the DAW's software instruments folder.

Setting up MIDI

Heartbeat is by default set up so the eight **Instruments** respond to the MIDI notes that are most commonly used for drum machines and drum software. The red text in the **Utility** section (the black square at the top of each **Drum Channel**) displays the MIDI note set up for each **Instrument**. If you would like to change it, click and hold the red text, and pull up or down. Or click **Learn** and strike the desired key/pad on your MIDI controller to assign this key/pad to the **Instrument**. Please note that the **Hihats** instrument receives input from two different MIDI channels, as it can be used for both closed and open hihat sounds. If you don't have a MIDI keyboard or pad controller available, you can use your mouse to click the green arrow in the **Utility** section, which will trigger the sound.



Presets

Clicking the white arrow will open a list of presets for that specific **Instrument** or effect. There are also presets available for entire **Heartbeat** kits (with settings for all eight instruments, effects, levels and master settings) via the usual preset function in your DAW.

Instrument and mixer

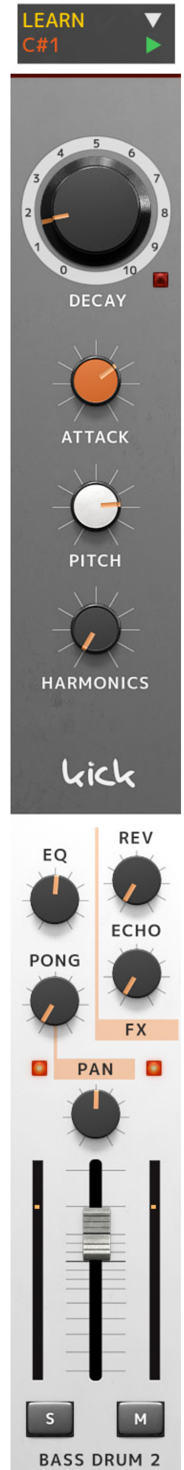
In the **Instruments** sections, you will see the settings for the instruments. These are all clearly labeled and adjusting them will yield apparent changes to the sound. Below these are the **Mixer Channels** with effects sends, a one-knob EQ (adapted for each **Instrument**) and an auto-pan function called **Ping/Pong**. The **Pan** knob and volume fader acts as you would expect, as do the **Solo** and **Mute** buttons.

Effects

The parameters of the three effects are clearly labeled. The **Pre** button to the right of **Filter Echo** and **TSAR-1D** inserts these effects before the **Valley People Dyna-mite**, which means the reverb and echo tails are also affected by **Dyna-mite's** processing.

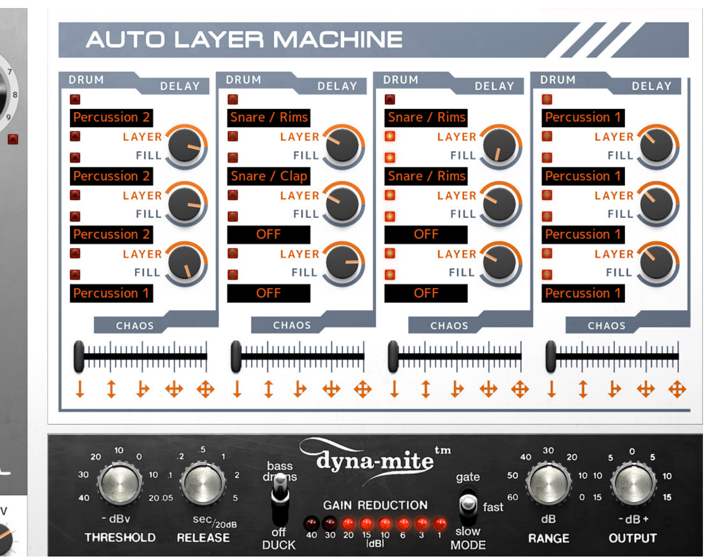
Master bus

The output from the **Instruments** and the effects are all summed in the **Master Bus**. **Mono Cut** collapses any stereo sounds below the selected cutoff frequency into mono, to ensure phase compatibility in the important lower frequencies. **Width** makes the entire stereo image wider or narrower.



Global parameters

At the bottom of the graphical user interface, you can determine Heartbeat's overall sensitivity to MIDI velocity, as well as separately determine how much velocity will affect **Pitch**, **Attack** and **Decay**. **Time Gate** can shorten all **Instrument** sounds independently of their velocity to create a stuttery, machine-like sound—very useful for creating variation to the sound by quickly adjusting a single knob.



Auto Layer Machine

Auto Layer Machine can be used to easily layer sounds from two or more **Instruments** for new textures, or to trigger flams or autofills. A quick way of learning what it does is to try the different factory presets and note the differences to their settings. Click the white arrow in the **Utility** section of one of the four **Auto Layer Machine** channels, and play the pattern by clicking the green arrow, or hitting the MIDI key assigned to that **Auto Layer Machine** channel (as indicated in red text in the **Utility** section). Refer to the detailed section for further information.

Keyboard shortcuts

All knobs in can be reverted back to its preset-settings by ALT-clicking on the parameter.

Fine-adjust any parameter in Heartbeat by CTRL-clicking (Windows) / CMD-clicking (Mac OS).

By clicking the **Setup** button below the Heartbeat logo, you can choose some basic settings for Heartbeat, such as turning off the tool tip pop-up windows.

Sound Architecture

See the image below for a description of the signal flow. The incoming MIDI signals can trigger either the instruments or the Auto Layer Machine. If an Auto Layer Machine channel is triggered, this in turn triggers the instruments.

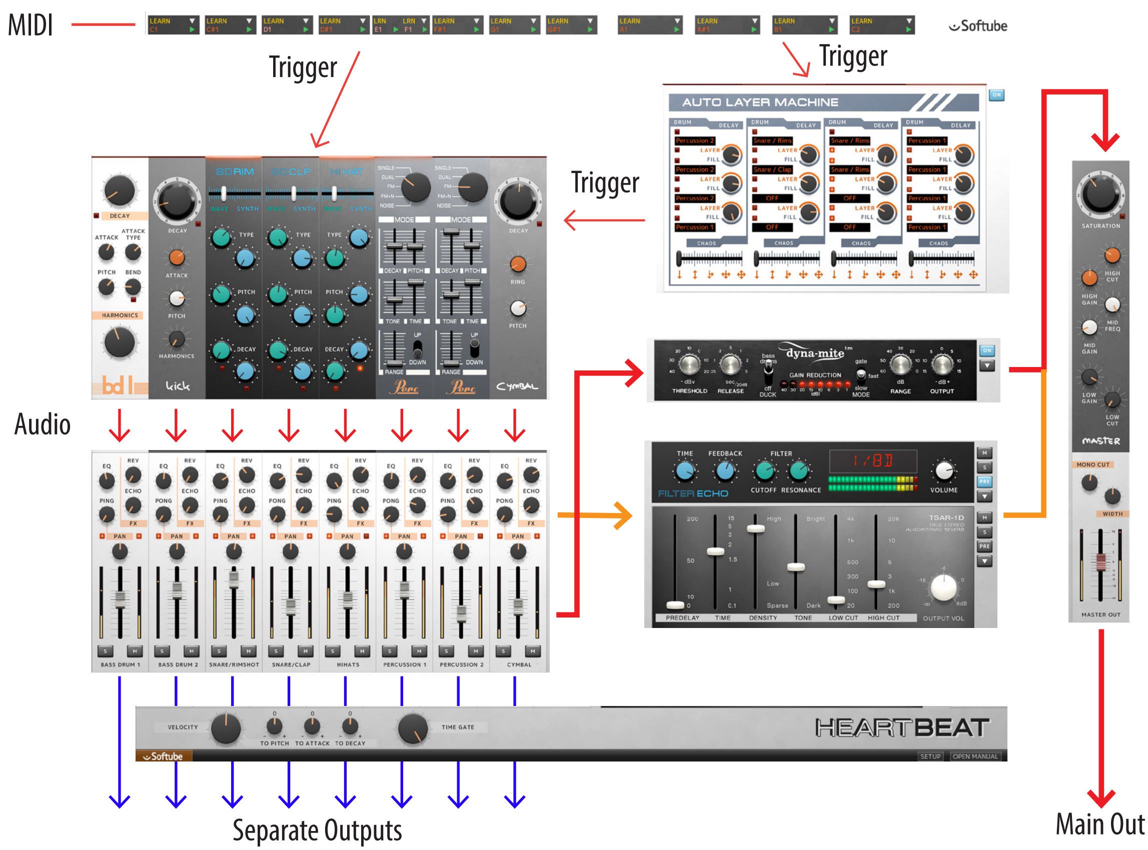
After the trigger, the instruments generate drum sounds that is routed to the corresponding mixer channels, and then routed through different paths:

1. The main signal (red) is sent through the mixer channel's volume fader and mixed with the other instruments, sent to the Valley People Dyna-mite, gets summed with the signal from the send effects, passes through the Master Bus and is eventually sent out via Heartbeat's Main Out to the DAW channel.
2. If the user chooses to, one signal is sent via the mixer channel's **Rev** send to the TSAR-1D Reverb and another is sent via the mixer channel's **Echo** send to the Filter Echo (orange).
3. One signal (blue) is sent *pre-fader* to the respective instruments' Separate Output, to be used as an isolated signal in your DAW, if this is supported by your DAW.

If the TSAR-1 Reverb or Filter Echo's **Pre** buttons are activated, the output from the effects are instead routed to the Dyna-mite, instead of directly to the Master Bus section

If **Duck** is set to BASS DRUMS, Bass Drums 1 and 2 are also routed to the sidechain of Dyna-mite, where it controls Dyna-mite's behavior.

For a more detailed overview, please see the chapter "Block Diagram".



Utility Section

The Utility section is the black field on top of each Drum Channel (and Auto Layer Machine channel).

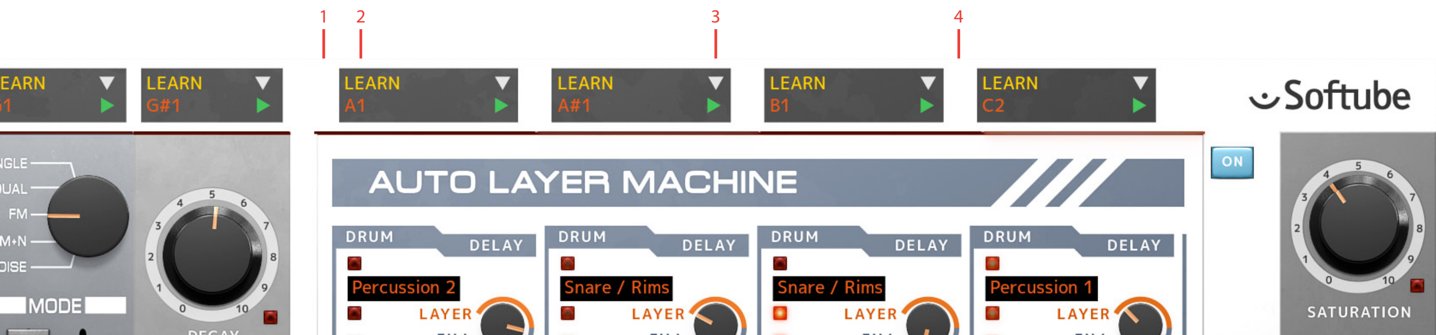
Learn The Learn function is a quick way to assign a key on your MIDI keyboard or pad controller so it triggers the corresponding Instrument, in case you would like to change it from the factory settings. Click Learn, which will start blinking to indicate that it is awaiting an incoming MIDI note. Press the MIDI key on your keyboard controller (or strike the pad on your MIDI pad controller) that you want to assign to the Instrument. The MIDI Note indicator (red text below the Learn button) will show the new MIDI note you assigned to the Instrument, and the Instrument will now respond to incoming MIDI data on that note number. Please note that the Hihats instrument has two Learn buttons, as it can be used for both closed and open hihat sounds.

MIDI Note The red MIDI Note indicator is located just below the Learn button, and tells you which MIDI note is assigned to the corresponding Instrument. You can change this by clicking, holding and dragging up/down the MIDI note number, as an alternative to using the Learn function explained above.

Channel Presets (white arrow) Click the white arrow to open the channel presets pop-up menu. This reveals a small selection of presets for each individual Instrument, intended as starting points for your own sound creation. Since both percussion channels use the same sound architecture, they also share the same channel presets. The same goes for the Auto Layer Machine channels. Only the Instrument parameters and the Equalizer (EQ) are affected by the channel presets. The effect sends (Rev, Echo), Ping/Pong, Pan, Volume, Mute and Solo are unaffected.

Play (green arrow) Clicking the green arrow will trigger the corresponding Instrument with maximum velocity. This function is handy when auditioning Instrument sounds without a MIDI keyboard or pad controller connected to the computer.

1. MIDI note
2. Learn
3. Channel preset
4. Play



The Instruments

The eight drum instruments occupy most of the upper left part of Heartbeat's graphical user interface. From left to right, you will find two different bass drum channels, two snare drum channels of which one is more suitable for typical snare sounds and the other leans towards clap sounds, a hihats channel (with both open and closed hihat sounds), two identical and very versatile synth percussion channels and finally a cymbal channel. Below, you will find a detailed description of each of these Instruments.

The equalizer (**EQ**) is an integral to the Heartbeat sound and should be thought of as part of the drum sound.

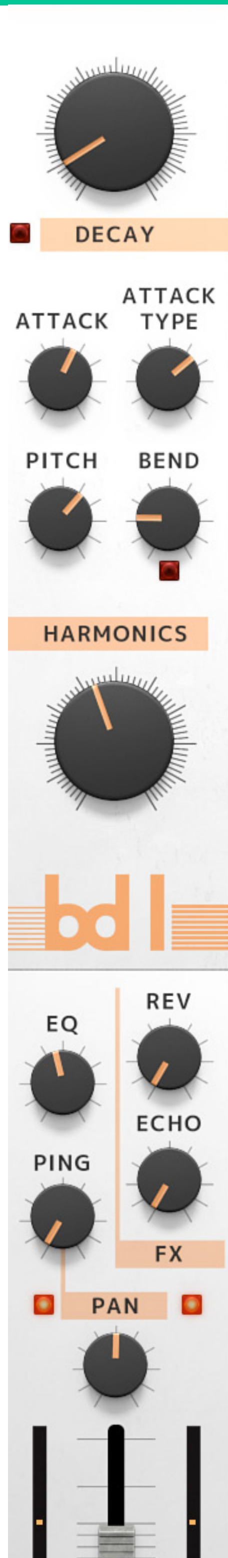
1. Bass drums
2. Snare drums
3. Hihat
4. Percussion
5. Cymbal

Bass Drum 1 “BD 1”

BASS DRUM 1 is highly flexible and was inspired by a well known Japanese drum machine from 1984 that went more or less unnoticed until the end of the 80s, when it became the core of the new house music scene of Chicago, Detroit and New York. Its sound stems from a modeled analog synthesized tone with a slight drop in pitch in its decay, augmented with a waveform attack transient.

Decay	Sets the duration of the bass drum sound. Turn counter-clockwise for short popping sounds and clockwise for longer ones.
Attack	Sets the level of the attack transient waveform.
Attack Type	Sets the character of the attack transient. Turn counter-clockwise for electronic style, harsher sounds, and clockwise for more “woody” and acoustic sounds.
Pitch	Sets the initial pitch of the modeled analog synthesis.
Bend	Turn clockwise for a fast pitch bend that goes up and then down again. Set fully counter-clockwise to bypass.

To create short percussion-like sounds , set the **Eq** counter-clockwise to remove the bottom end from the bass drum, .



rmonics	Adds harmonics/distortion to the synthesized tone. Can go from subtle overtones into harsh and bit-crunchy territory above the 12 o’clock position.
EQ	Boosts or cuts the low frequencies of the bass drum.

House music kick drum

Characteristic of the house music kick drum is its short and distinct snappy attack along with its moderate decay. By changing **Decay**, **Bend** and **Harmonic** you’ll get different and useful variations.

Decay: 25%
Attack: 50%
Attack Type: 0%
Pitch: pretty much what ever you like, but 0% will do.
Bend: 50%
Harmonics: anywhere between 0% to 20% will do.
EQ: 30%

Acoustic style kick drum

The acoustic bass drum is short and dry. Decrease the **Attack** volume if you want the impact to be a bit smoother.

Decay: 10%
Attack: 100%
Attack Type: 71%
Pitch: -75%
Bend: 84%
Harmonics: 0% (clean)
EQ: 0%

Bass Drum 2 "Kick"

The second bass drum is circuit modeled from a classic Japanese drum machine from the early 1980s. It has been heavily used in many genres, ranging from electro and hip-hop to techno and R&B.

Decay	Sets the duration of the bass drum sound. Turn counterclockwise for short popping sounds and clockwise for longer ones.
Attack	Adjusts the filter level of initial click transient. Turn counterclockwise for a darker and more subdued click character, and clockwise for a more edgy and apparent click.
Pitch	Sets the pitch of the bass drum sound.
Harmonics	Sets the amount of clipping distortion.
EQ	Boosts or cuts the low frequencies of the bass drum.

By using **Velocity To Pitch** in combination with this bass drum it is possible to create nice sounding deep baselines.

Electronic style booming kick drum

Bass Drum 2 is very suited for this type of booming electronic sounds with long Decay times.

Decay: 100%
Attack: 20%
Pitch: -32%
Harmonics: 0% (clean)
EQ: 35%

Techno style kick drum

Short and distinct kick that will cut through any mix.

Decay: 10%
Attack: 100%
Pitch: -45%
Harmonics: 0% (clean)
EQ: 0%

Hollow distorted kick drum

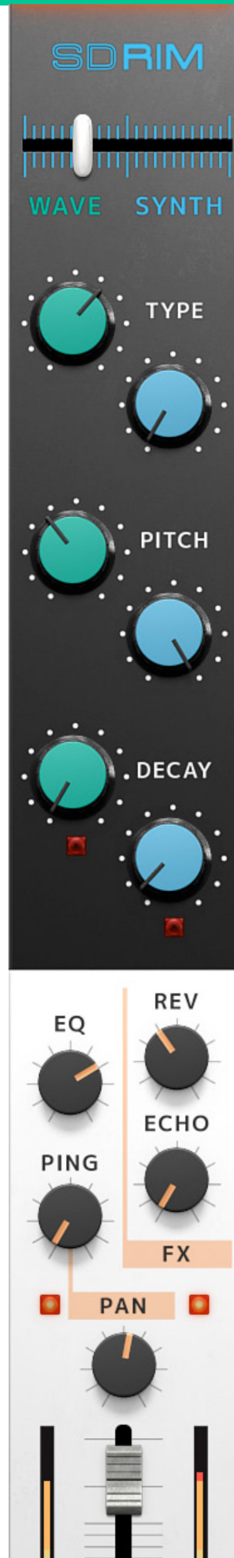
This setting makes the bass drum sound more like a synth bass.

Decay: 73%
Attack: 100%
Pitch: 100%
Harmonics: 100% (clean)
EQ: -100%

Snare/Rimshot "SD RIM"

The SNARE/RIMSHOT channel blends snare drum waveforms with modeled analog synthesis in a highly flexible manner. The balance between them is determined by the slider at the top, and the WAVE and SYNTH portions each have three knobs that adjust their respective sound character—green for WAVE and blue for SYNTH.

Wave/Synth	Sets the balance between the waveform and synthesized portion of the drum sound.
Type	Sets the character of the sound. WAVE (green knob) ranges from a rattling snare to a hard rimshot, SYNTH (blue knob) takes the sound from a pitched note to a noise sound.
Pitch	Sets the pitch for each sound, and the cutoff for the noise in the synthesized part.
Decay	Sets the duration of the WAVE and SYNTH parts respectively. Turn counterclockwise for short popping sounds and clockwise for longer ones.
EQ	Boosts or cuts the frequency range where most of the snare drum's tonal content is found. The neutral setting is at 12 o'clock, turn counterclockwise to cut this frequency range (emphasizing the snare/noise character) and clockwise to boost (emphasizing the tonal character).



House snare

A snare sound close to that of a very popular drum machine from the 80s.

Wave/Synth: -42%
Wave type: -34%
Synth type: 0% (TONE)
Wave pitch: 0%
Synth pitch: -27%
Wave decay: 100%
Synth decay: 21%
EQ: 39%

Acoustic Style Snare

A more acoustic sounding snare drum.

Wave/Synth: -13%
Wave type: -60%
Synth type: -57%
Wave pitch: -52%
Synth pitch: -55%
Wave decay: 77%
Synth decay: 72%
EQ: -21%

Create dynamic and interesting sounds by combining a wave portion with short decay with a synth portion with long decay, and vice versa.

Snare/Clap "SD CLP"

The second snare drum is the SNARE/CLAP and works similarly to the SNARE/RIMSHOT. However, both the WAVE and SYNTH portions of the SNARE/CLAP have a different sound character which lends itself more to clap style sounds.

> Create a double-clap sound by using a clap wave combined with a synthesized clap.

House Clap

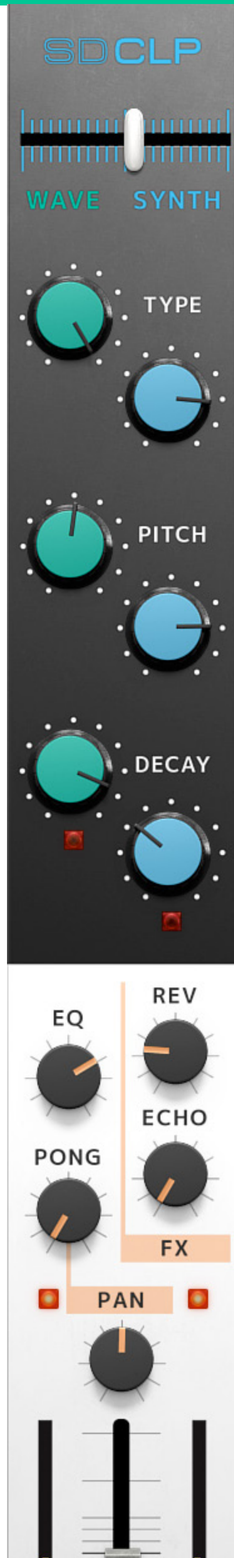
A clap sound close to that of a very popular drum machine from the 80s.

Wave/Synth: -80%
Wave type: 100% (CLAP)
Synth type: 100% (CLAP)
Wave pitch: -7%
Synth pitch: 80%
Wave decay: 100%
Synth decay: 57%
EQ: 71%

Drummachine Snare

An 80s style digital sounding snare drum.

Wave/Synth: -64%
Wave type: -35%
Synth type: 0% (SNARE)
Wave pitch: 0%
Synth pitch: -20%
Wave decay: 64%
Synth decay: 36%
EQ: 76%



Wave/Synth	Sets the balance between the waveform and synthesized portion of the drum sound.
Type	Sets the character of the sound. WAVE (green knob) ranges from different snare sounds to tight claps, while SYNTH (blue knob) takes the sound from a slightly noisy tonal character to a dark and sluggish clap sound.
Pitch	Sets the pitch for each sound, and the cutoff for the noise in the synthesized part.
Decay	Sets the duration of the WAVE and SYNTH parts respectively. Turn counterclockwise for short popping sounds and clockwise for longer ones. For the SYNTH portion of the sound, longer **Decay** times also decreases the tightness of the clap, which gives it an even more loose and sluggish character.
EQ	Boosts or cuts the frequency range where most of the snare drum's tonal content is found. The neutral setting is at 12 o'clock, turn counterclockwise to cut this frequency range (emphasizing the snare/noise character) and clockwise to boost (emphasizing the tonal character).

Hihats

The HIHATS instrument of Heartbeat can make both closed and open hi-hat sounds. One sound chokes the other. So if the open hi-hat is played, followed by the closed hi-hat the open hi-hat will be immediately cut off by the closed hi-hat. The HIHATS are laid out in the same way as the two snare drums (SNARE/RIMSHOT and SNARE/CLAP), with a crossfade slider at the top which sets the balance between hi-hat waveforms (WAVE), and sounds generated by modeled analog synthesis (SYNTH). As with the snare drums, the green knobs affect the WAVE portion and the blue knobs affect the SYNTH portion of the sound.

Wave/Synth	Sets the balance between the waveform and synthesized portion of the hi-hat sound.
Type	Sets the character of the sound. WAVE (green knob) ranges from classic drum machine-like hi-hat sounds to a more acoustic sounding character. SYNTH (blue knob) takes the sound from a cluster of high pass filtered pulse waveforms in the far counter-clockwise setting, to a filtered white noise when turned clockwise.
Pitch	Sets the pitch for the WAVE sound (green knob), and sweeps a low cut filter for the SYNTH portion (blue knob).
Decay	Sets the duration of the WAVE and SYNTH parts respectively. Turn counterclockwise for short popping sounds and clockwise for longer ones.
EQ	Boosts or cuts the frequency range where most of the hi-hat's tonal content is found. The neutral setting is at 12 o'clock, turn counterclockwise to cut this frequency range (emphasizing the noise character) and clockwise to boost (emphasizing the tonal character).

Combine the transient of the wave with a longer, noisy, decay of the synth part to get dirty nice hi-hat sounds.

Synthesized Hi-hat

A sound similar to that of a very popular early 80s drum machine.

Wave/Synth: 93%
Wave type: -45%
Synth type: 52%
Wave pitch: 57%
Synth pitch: 68%
Wave decay: 6%
Synth decay: 24%
EQ: 86%

Digital Hi-hat

Dry 80s drum machine style hihat.

Wave/Synth: 33%
Wave type: 3%
Synth type: 100% (NOISE)
Wave pitch: -4%
Synth pitch: 74%
Wave decay: 66%
Synth decay: 6%
EQ: -49%

Percussion 1 and 2 “Perc”

he two PERCUSSION instruments are identi-
al to each other, so the information given
ere covers them both. They are highly flexible
nstruments that draws a lot of inspiration from
 lesser known, but very powerful, Japanese synth
ercussion unit from the early 80s. The sound of
he PERCUSSION 1 and 2 are purely generated
rom modeled analog synthesis.

Mode	This dial selects one of the following five modes.

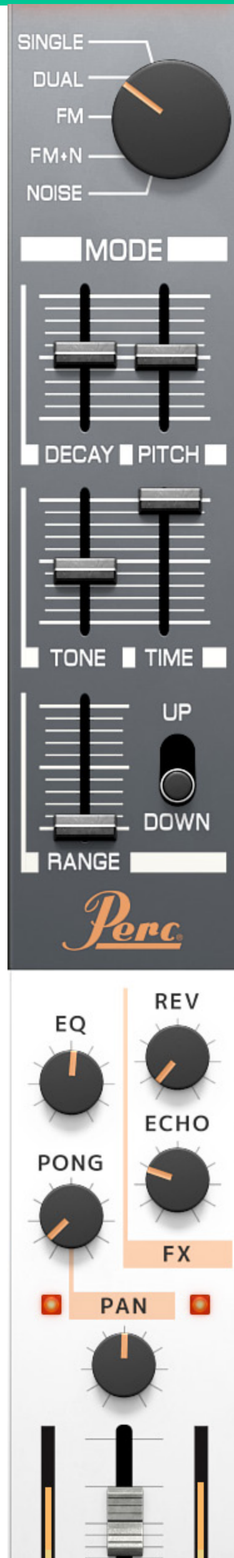
SINGLE: Employs a single triangle wave oscillator. This is great for disco style toms, additional bass drums and short harmonic snaps.

DUAL: Employs two triangle wave oscillators with a fixed pitch ratio between them. This is ideal for cowbell, agogo bell and marimba type of sounds.

FM: One oscillator is frequency modulated by the other with a fixed ratio. This is useful for disharmonic metal-like sounds.

FM+N: The same as above, but with added noise. Can be used to generate other-worldly metallic sounds.

NOISE: White noise. This is good for generating shakers, thunderous snares and special effects.



Decay	Sets the duration of the sound.
Pitch	Sets the initial pitch of the oscillators.
Tone	Sets the initial cutoff frequency of the low-pass filter.
Time	Sets the speed of the pitch bend and in noise mode the speed of the filter-sweep.
Range	Sets the amount of pitch bend or filter-sweep in noise-mode.
p/Down	Sets if the bend goes upwards or downwards (filter sweep in noise mode).
EQ	Boosts or cuts the frequency range where most of the drums tonal content is found. The neutral setting is at 12 o’clock, turn counterclockwise to cut this frequency range (emphasizing the noise character) and clockwise to boost (emphasizing the tonal character).

The percussion modules without the bend (**Range** at zero) in combination with the **Velocity To Pitch** parameter makes it possible to create small pseudo-melodies and baselines.

Cymbal

The sound of Heartbeat's Cymbal is purely generated by modeled analog synthesis. It draws inspiration from several early 80s Japanese drum machines. But the **Ring** parameter has been added for the ability to get a more bell-like high pitched ringing sound.

Decay Sets the duration of the sound.

Ring Sets the amount of "ring" character.

Pitch Adjusts the filtered mix of the harmonics within the cymbal sound.

EQ Boosts or cuts the frequency range where most of the cymbals tonal content is found. The neutral setting is at 12 o'clock, turn counterclockwise to cut this frequency range (emphasizing the noise character) and clockwise to boost (emphasizing the tonal character).

A high setting of **Ring** makes the cymbal sound more like a ride cymbal, while a low setting makes the cymbal more vintage drum box sounding.

Noise Hat

Using the cymbal as an extra noisy drum-machine like hihat.

Decay: 0%
Ring: 0%
Pitch: 100%
EQ: 100%

Noise Ride

A cymbal sound close to that of a very popular drum machine from the 80s.

Decay: 14%
Ring: 10%
Pitch: 40%
EQ: 48%

Short and Sweet

Short cymbal sound with some ringing.

Decay: 20%
Ring: 34%
Pitch: 53%
EQ: -47%

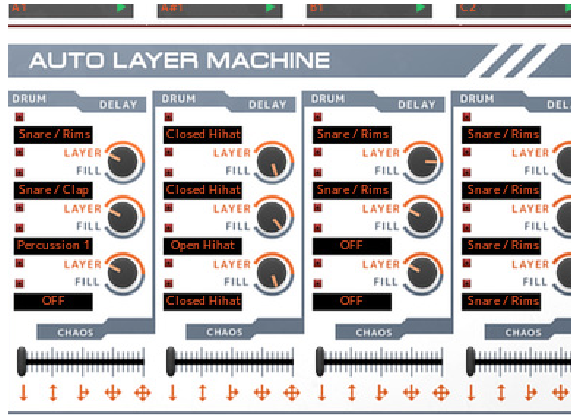
The Mixer

The Mixer section takes up most of the lower left part of the Heartbeat graphical user interface. The parameters are identical for all eight mixer channels, with the exception of the **EQ** (equalizer) which is tuned for each individual instruments, although the knobs look identical.

EQ	Adjusts the equalizer setting. It is tuned for each channel and optimized to work with the sweet spots of the individual instruments.
Rev	Sets the signal level being sent from each instrument to the TSAR-1D REVERB unit, and therefore how much reverb is added to the instrument. The **Rev** send is post-fader, meaning that the send level is also affected by the setting of the **Volume** fader. This keeps the proportion between the direct sound and the reverb intact even if the **Volume** fader is turned up or down.
Echo	Sets the signal level being sent from each instrument to the FILTER ECHO, and therefore how much delay is added to the instrument. The **Echo** send is post-fader, meaning that the send level is also affected by the setting of the **Volume** fader. This keeps the proportion between the direct sound and the reverb intact even if the **Volume** fader is turned up or down.
Ping/Pong	The automated panning function. Sets the amount of automatic panning for each drum hit.
Pan	The initial position of the instrument in the stereo panorama.
Volume fader	Sets the volume of the instrument.
Solo (S)	Activating SOLO for a mixer channel mutes all othter channels (unless they are also in SOLO mode).
Mute (M)	Activating MUTE for a mixer channel turns off the sound from this channel.

Auto Layer Machine

AUTO LAYER MACHINE takes up most of the upper right part of the graphical user interface. It can be used to easily layer sounds from two or more instruments for new textures, or to trigger flams or auto-fills—you could compare it to a basic MIDI sequencer. AUTO LAYER MACHINE consists of four channels, each with its individual MIDI note number assigned. The four channels are divided into a **Drum** and a **Delay** column. In the **Drum** column you will find four slots per channel. Hitting the assigned MIDI key will make the corresponding AUTO LAYER MACHINE channel generate a chain of events, moving from the top slot to the bottom one (in its default state). Each trigger will be slightly lower in velocity which is apparent when using the **Velocity** parameters.



Get started!

The easiest way to understand what the AUTO LAYER MACHINE does is by trying the settings in the factory preset you will have every time you launch a new instance of Heartbeat. Here, the four AUTO LAYER MACHINE channels are set up to perform different tasks.

Layering

Hit the MIDI key A1 to trigger the first AUTO LAYER MACHINE channel (or click its green arrow in the UTILITY section, the black field above the channel). You will hear that this triggers three of the instruments—SNARE/RIMSHOT, SNARE/CLAP and PERCUSSION 1—simultaneously. This creates a layered sound. You can also see the names of these three instruments in three of the slot windows of the first AUTO LAYER MACHINE channel, indicating that the slots have been assigned to these instruments.

Patterns and fills

If you instead trigger the second AUTO LAYER MACHINE channel, by hitting A#1, you will hear the closed and open HIHATS playing a short pattern with four hits. Again, you can see in the slots that they are assigned to the closed and open HIHATS. But unlike the first channel, they were not playing simultaneously—a delay was added for each step.

This is done with the knobs in the **Delay** column, to the right of each slot. The **Delay** knobs set how long it takes after a slot has been triggered until it passes on the trigger impulse to the slot below it. In the first AUTO LAYER MACHINE channel, you will see that the **Delay** knobs are all set to LAYER (fully counter clockwise), meaning that there is no delay from one slot to the next—the instruments are triggered simultaneously.

But in the second channel, they have other positions, which is what creates the delay between the instruments being triggered, and thus creates the small pattern you hear each time you trigger the channel. If you change the positions of the **Delay** knobs, you will hear the short pattern change accordingly.

Velocity

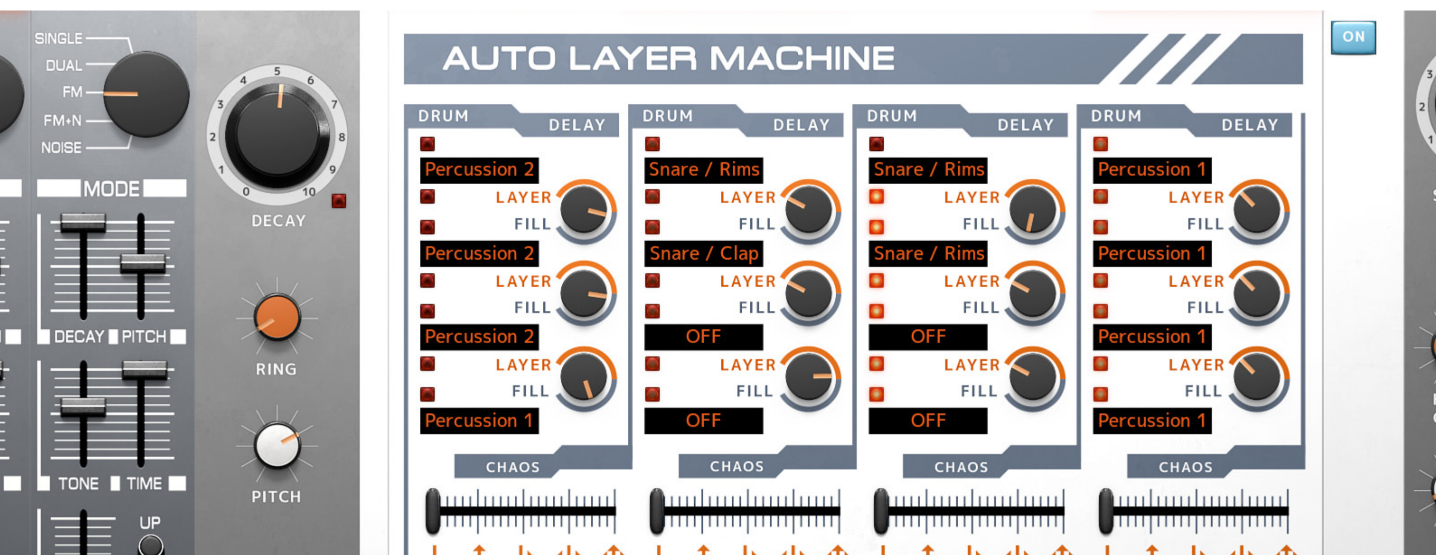
The Instrument in the first Auto Layer Machine slot will be triggered with the velocity of the incoming MIDI note. For each subsequent step in the Auto Layer Machine, the velocity will automatically drop by a predefined amount. This means that if the incoming MIDI note has a very low velocity to begin with, the subsequent steps might drop below 0 velocity, and thus not trigger the Instrument at all.

On Turns ON and OFF the Auto Layer Machine. You can save some CPU power by turning the Auto Layer Machine off when not in use.

Slot window The slot windows in the **Drum** column indicates and determines which of Heartbeat's instruments is triggered via the slot. Click or SHIFT-click to select next or previous instrument. You can bypass the slot entirely by selecting OFF. It is also possible to click and drag to scroll back and forth among the instruments.

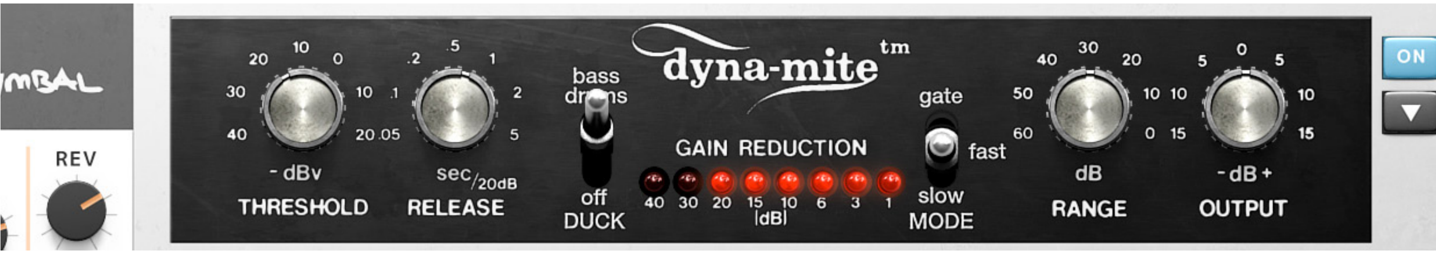
Delay Determines how long it takes after an instrument has been triggered until it passes on the trigger impulse to the slot below it. By setting it to ZERO (the knob indicator pointing at LAYER), there is no delay, so the two instruments are triggered at the same time and thus layered. By turning it clockwise, the following trigger will be more delayed. Use this to create flams or automatic fill patterns. When the indicator is by the orange part of the marking, the delay is expressed in milliseconds in the tooltip window that pops up. Turn it to the blue side to set the delay expressed as beat divisions of the DAW project's tempo.

Chaos In its default state with the slider set all the way to the left, the trigger impulses move from the top to the bottom as indicated by the orange arrow underneath the Chaos slider. Moving the slider a bit to the right will enable the Auto Layer Machine to reverse the direction of the triggers so that some triggers will randomly populate upwards instead of only downwards. Moving the slider even further will make the trigger impulses "spill" over to the adjacent Auto Layer Machine channels, as indicated by the arrows. In its far right position, you will have full chaos with triggers sent everywhere in a rather unpredictable manner. Even more so if you have all four Chaos sliders to the far right!



The Effects

Heartbeat includes three different effect units: The VALLEY PEOPLE DYNA-MITE compressor/limiter/gate, the TSAR-1D REVERB and the FILTER ECHO. These are shared by all the drums. The signal level sent from each drum to the TSAR-1D and the FILTER ECHO is set with the **Rev** and **Echo** knobs in each mixer section. VALLEY PEOPLE DYNA-MITE is inserted across the stereo sum of all the instruments, so as long as it's activated it affects all instruments (apart from the bass drums when **Duck** is set to BASS DRUMS).



Valley People Dyna-mite

The VALLEY PEOPLE DYNA-MITE built into Heartbeat is a specially adapted version of the Valley People Dyna-mite plug-in, separately available from Softube. The original analog Valley People Dyna-mite unit came out in the early 80s and was a very popular tool for gating/expanding, compressing/limiting and ducking—highly loved for its ability to compress sounds with fierce aggression, and to gate in an ultra-musical manner. The Heartbeat version features four operating modes:

Compression

A compressor is basically an automatic volume control, which turns down strong sounds but leaves the weaker sounds unaffected. This makes the dynamic range of the sound (the difference between strong and weak parts) smaller, which is why it's called compression. Using compression lightly can make the sound compact and coherent (often described as glueing the sounds together), while using it heavily can create an aggressive mash. With the **Mode** switch set to SLOW, the Dyna-mite will act as a compressor with a slow attack. This lets the initial transient of the sound through before Dyna-mite reacts and starts compressing, resulting in a punchy and snappy sound. The **Threshold** knob sets the threshold level. Any time a sound reaches above this level, Dyna-mite will start compressing—so a high threshold setting will only affect the strongest peaks, while a low threshold setting will affect most of the incoming sound, resulting in a very apparent compression. The **Release** knob determines the time it takes for the Dyna-mite to recover after it has compressed. This can be used to emphasize the rhythmic feel of the beat, making the Dyna-mite "breathe" in time with the music.

Limiting

A limiter is a very fast compressor that uncompromisingly slams down the sound any time it exceeds the threshold level. Its original use was to protect loudspeakers from sharp sound spikes that could potentially damage them, but it can also be used creatively for music mixing. Set Dyna-mite's **Mode** switch to FAST, and it will act like a limiter. Compared to the SLOW mode, you will note that the Dyna-mite now doesn't let the initial transient of the sound through. Instead, the sound hits a brick wall, creating an aggressive and pumping sound—even more so with a low threshold setting.

Gating

With the **Mode** switch set to GATE, Dyna-mite will shut off the sound completely if it drops below the threshold level (set by the **Threshold** knob), and open up as soon as the sound rises above the same level. This can be used to make sounds appear shorter (for example creating gated reverb effects), and to get rid of low-level sounds for a cleaner and more focused impression. If the Dyna-mite is set to gate out the weaker sounds entirely, the **Range** knob can be used to mix them back in, but at a lower level than they originally had. This is called *expansion*—you expand the dynamic range of the mix by making the weak sounds (the ones below the threshold level) weaker, and thus in comparison making the strong sounds (above the threshold level) stronger. So an expander is basically a compressor in reverse. This can be used to enhance and alter the dynamic feel of a drum beat.

Ducking

Ducking is the effect of one sound source controlling the output volume of another. In its Heartbeat version, Dyna-mite can be set to let the bass drum channels duck the others. By setting the **Duck** switch to BASS DRUMS and the **Mode** switch to either FAST or SLOW, every time a hit from one of Heartbeat's bass drums is strong enough to reach above the threshold level, the sound level of all the other instruments will be turned down by Dyna-mite. This creates a pumping and energetic effect that is prominently used in a lot of electronic dance music. The **Mode** switch can also be set to GATE while the **Duck** switch is set to BASS DRUMS. In this case, the gate opens up every time the bass drum hits, and shuts off the sound of the other instruments between the bass drum hits.

On	Turns Dyna-mite on or off.
Presets	Clicking the button with the white arrow below the On button brings up Dyna-mite's preset menu. It contains some examples of applications of Dyna-mite compressor which are good starting points for further tweaking.
Threshold	Sets the threshold level, above which the Dyna-mite starts to limit or compress (in FAST or SLOW modes), or lets the sound through (in GATE mode).
Release	Adjusts the time it takes to restore the original gain after gating/compressing.
Duck	Activates/deactivates bass drum ducking, which makes the bass drums affect Dyna-mite's processing of all the other instruments.
Mode	Sets the main mode of operation: GATE, FAST (limit) or SLOW (compress).
Range	Sets the maximum amount of gain reduction.
Output	Sets the output volume. If necessary, turn this up to compensate for the volume loss caused by compressing/limiting.

Filter Echo

Filter Echo is a gritty little delay effect with a resonant lowpass filter in its feedback loop. The filter can be set to near self-oscillation for that lo-fi sound.

Turns off the sound of the Filter Echo.

Solo (S) Solos the sound of the Filter Echo.

Pre Places the Filter Echo before Dyna-mite in the effects chain, which means Filter Echo's sound will also be affected by Dyna-mite's processing. When the Pre button is not activated, the Filter Echo's output will be post the Master Bus Saturation effect, but before the Master Bus equalizer.

Presets Clicking the button with the white arrow below the Pre button brings up Filter Echo's preset menu. It contains some examples of Filter Echo settings that are good starting points for your tweaking.

Time Determines the delay time, how long time passes between each delay "hit". In the left half of the knob's path, the range is from 1 to 1000 milliseconds. In the right side, the delay time can be set in divisions of the DAW tempo, ranging from 1/64 to 1/2 beat. The latter is useful for setting the delay to act in time with the song.

Feedback This set the amount of feedback, how many delay repeats there will be. It ranges from one repetition to roughly 10 repetitions at full feedback.

Cutoff Sets the cutoff frequency of the low-pass filter.

Resonance Sets the resonance of the low-pass filter.

Volume Sets the output volume.

TSAR-1D Reverb

The TSAR-1D is a version of Softube's acclaimed TSAR-1 Reverb, adapted for use with Heartbeat. For more information, please see page 188. TSAR-1 Reverb is available as a separate plug-in from Softube.

Mute (M) Turns off the TSAR-1D.

Solo (S) Solos the TSAR-1D reverb.

Pre Places the TSAR-1D before Dyna-mite in the effect chain, which means that the reverb will also be affected by Dyna-mite. When the **Pre** button is not activated, TSAR-1D's output will be after the master saturation, but before the master equalizer.

Presets Clicking the button with the white arrow below the Pre button brings up TSAR-1D's preset menu.

Predelay Determines the time between the dry signal and the reverb tail. Set to 0, there is no delay. Delayed settings are often used to achieve the impression of a large room, by making the reverb tail arrive later.

Time Sets the duration of the reverb sound, from short to long.

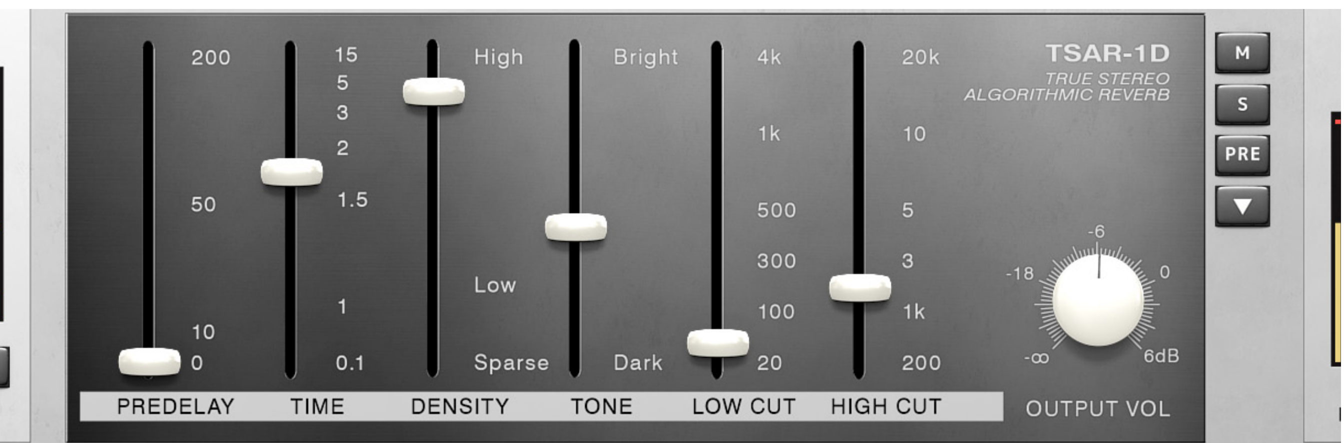
Density Adjusts is the thickness and smoothness of the reverb.

Tone Overall tone of the reverb signal.

Low Cut Applies a low cut filter to the reverb sound, taking away the lower frequencies that might make the sound cluttered and undefined.

High Cut Applies a high cut filter to the reverb sound, taking away the higher frequencies that might make the reverb sound take up too much space in the mix.

Output Vol Output volume of the reverb.



Master Bus

The Master Bus is the section on the far right of Heartbeat's graphical user interface. This affects the main output of Heartbeat, letting you add saturation, make EQ adjustments to the overall sound and more.

Saturation	Sets the amount of saturation applied to the entire drum mix, post the Dyna-mite compressor. It mimics the saturation that can be achieved by devices using electronic vacuum tubes, which results in a pleasing and thick saturation.
High Cut	Applies a high cut filter to Heartbeat's output, which cuts treble frequencies and makes the overall sound darker. This is very similar to the High Cut function of a DJ mixer.
High Gain	Boosts the treble frequencies.
Mid Freq	Sets the center frequency of the midrange equalizer filter.
Mid Gain	Boosts the midrange at the frequency determined by the Mid Freq knob.
Low Gain	Boosts low frequencies.
Low Cut	Applies a low cut filter, which cuts bass frequencies and makes the overall sound thinner. This is very similar to the Low Cut function of a DJ mixer.
Mono Cut	Determines a cutoff frequency below which everything is summed to mono. This is a great way to ensure that your mix will sound solid on any playback system, since it guarantees that you will have no phase issues in the low end of the frequency spectrum.
Width	At 12 o'clock, Width is disabled and all stereo settings work as expected. Turning it counterclockwise gradually makes the sound become more mono. Turning it clockwise makes the sound wider.
Master Out	Master fader which controls the volume of Heartbeat's main output.

Velocity

At the lower left of Heartbeat's graphical user interface, below the mixer section, are the global **Velocity** parameters. These knobs determine how responsive Heartbeat will be to the velocity of the incoming MIDI signals—i.e., how strong or soft the incoming MIDI note is. The **Velocity** settings are global for Heartbeat's instruments, meaning that they effect them all simultaneously.

Velocity Sets how strongly the velocity of the incoming MIDI note affects the volume of the instrument being triggered. Setting this to 0% (fully counterclockwise) will result in no volume difference of the sounds, regardless of the velocity. Conversely, when **Velocity** is set to 100% (fully clockwise), the Instruments will respond very dynamically to the velocity. So in this setting, higher velocities give louder sounds.

To Pitch Sets how the initial pitch of Heartbeat's instruments is affected by velocity. In its 12 o'clock position, velocity has no effect on pitch. Turning it counterclockwise will result in higher velocities giving the sounds a lower initial pitch. Conversely, turning it clockwise will result in higher velocities giving the sounds a higher initial pitch.

To Attack Sets the amplitude of the attack portion of the instrument's sounds. In its 12 o'clock position, velocity has no effect on attack levels. Turning it counterclockwise will result in higher velocities giving lower attack amplitude. Conversely, turning it clockwise will result in higher velocities giving the sounds a higher attack amplitude. This applies only to instruments that have the **Attack** parameter

To Decay Sets how the decay time of Heartbeat's instruments is affected by velocity. In its 12 o'clock position, velocity has no effect on decay time. Turning it counterclockwise will result in higher velocities giving the sounds shorter decay times. Conversely, turning it clockwise will result in higher velocities giving the sounds longer decay times.

Time Gate

Time Gate is a fun and useful function that cuts the decay short of all instruments globally. This creates a jerky and chopped up cool sound reminiscent of old 80s sample-based drum machines with very small memory. Since Time Gate is controlled by a single knob, it offers a quick way of altering the sound of the entire beat—this is not least useful for live applications.

Using Multiple Outputs

Heartbeat is designed to be a one-stop shop for drum sound creation, where the resulting sound package comes out of a single stereo output. But for added flexibility, it is also possible to send the individual instruments through separate outputs, and have them appear on individual mixer channels in your DAW. The separate outputs can be used in situations where you would like to add mix effects from your DAW to the individual instrument sounds of Heartbeat—for example if you have a particular reverb plug-in in your DAW that you would like to use for the snare drums, and only for the snare drums. Or if you want Heartbeat's bass drums to duck all the other sound sources in your song, such as synths and vocals.

When using separate outputs, the signal from the instrument is being split into two. One is sent the usual way through Heartbeat, via the volume fader and the effects to Heartbeat's Master Bus. The other one is sent to the direct output. This is tapped out of the DRUM CHANNEL mixer pre-fader. This means that the DRUM CHANNEL volume fader (as well as the **Solo** and **Mute** buttons) will *not* affect the signal being fed to the direct output. If you want an instrument sound to only be sent to the direct output, and not appear in your main Heartbeat mix at the same time, you can set the corresponding DRUM CHANNEL's volume fader to zero, or press its **Mute** button.

Sending effects to a separate output

If you want all the instruments on separate outputs, and also get the send effects (FILTER ECHO and TSAR-1D) as a separate stereo signal, in total 8 + 1 stereo pairs, you can achieve this using the **Solo** buttons for the FILTER ECHO and TSAR-1D—then you will only have the outputs of these sent to Heartbeat's MASTER BUS.

Using multiple outputs

Your DAW will automatically detect that Heartbeat has a total of nine stereo outputs—the MAIN OUTPUT, plus a stereo pair for each of the eight instruments. These outputs are named in accordance with the DRUM CHANNELS:

- BASS DRUM 1
- BASS DRUM 2
- SNARE RIMSHOT
- SNARE CLAP
- HIHATS
- PERCUSSION 1
- PERCUSSION 2
- CYMBAL

The different DAWs all have their own particular ways of handling instruments with multiple outputs, such as Heartbeat. Therefore, please refer to your DAW's manual to learn how to use Heartbeat's multiple outputs on your particular system.

Presets

Heartbeat features 50 different preset drum kits, ranging from classic drum machine sound-a-likes to more contemporary sounds of all kinds. They also contain settings for the effects as well as programmed Auto Layer Machine settings. Each preset name begins with a two-letter acronym of the name of the creator. They are:

KU	Kristofer Ulfves, Softube
CB	Christoffer Berg (Depeche Mode, The Knife, Hird)
DG	David Giese (Joxaren, Flogsta Danshall)
TB	Tomas Boden (Differnet, Liminals)

Setup window

Clicking the **Setup** tab in the bottom right corner of Heartbeat's graphical user interface will bring up the setup configuration menu. Some of the changes made here will only take effect after relaunching Heartbeat.

Always Use Small GUI	This toggles between bigger and smaller versions of the graphical user interface. We recommend that you check this box if you use Heartbeat on a small computer screen, such as a laptop screen.
Enable Tooltip	Toggles Tooltips on and off. These are the small pop-up windows that appear when hovering the mouse pointer over most of Heartbeat's controls.
Show Value Display	Toggles the value display in lower left corner of Heartbeat on and off.

Credits

Oscar Öberg – product lead and signal processing. **Kristofer Ulfves** – research, sound design, presets and user manual. **Niklas Odelholm** – graphic design and presets. **Patrik Holmström** – GUI programming. **Henrik Andersson Vogel** – user manual and marketing. **Paul Shyrinskykh** – quality assurance. **Arvid Rosén** – framework programming. **Ulf Ekelöf** – graphics rendering. **Torsten Gatu** – framework programming. **Mattias Danielsson** – technical support. **Johan F. Antoni** – help with initial concept. **Andreas Tilliander** – hardware reference. **Tomas Boden** – testing and presets. **Christoffer Berg** – testing and presets. **David Giese** – testing and presets. **Jakob Herrman** – sound reference. **Marcus Schmahl** – demo and feedback

Block Diagram

This is a simplified block diagram of the Heartbeat functionality and signal paths.

MIDI notes are received by the Auto Layer Machine and the Drum Channels. Audio from the Drum Channels are routed both to the separate outputs as well as to the Valley People Dyna-mite.

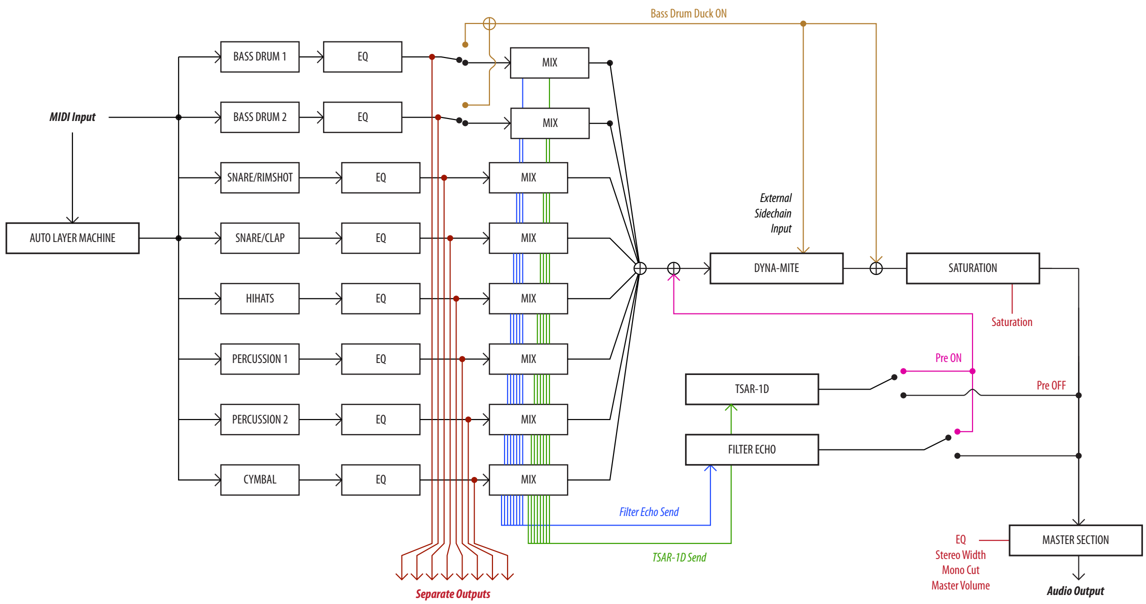
Bass Drum Ducking

If Dyna-mite's **Duck** switch is set to BASS DRUMS, the audio from the bass drums are being routed to the external sidechain of the Dyna-mite, as well as being mixed together with the output of the Dyna-mite.

Send Effect Pre/Post

If the send effects (TSAR-1 Reverb and Filter Echo) are set to PRE (**Pre** button is blue), the outputs from the effects are routed to the input of the compressor.

If the send effects' **Pre** button is off (**Pre** button is gray), the output from the effects are routed directly to the master section, but after the saturation circuit.



ALM/Filter Echo times chart

Name	Length	Swing value	
1/64	64th note		
1/32	32th note		
1/16	16th note		
1/16+	16th note	54% swing	
1/8T-	slightly short 8th note triplet	16th note with 62% swing	
1/8T	8th triplet	16th note with 66% swing	
1/16D	dotted 16th note	16th note with 75% swing	length of a 16th plus a 32th note
1/8	8th note		
1/8+	8th note	54% swing	
1/4T-	slightly short quarter note triplet	8th note with 62% swing	
1/4T	quarter note triplet	8th note with 66% swing	
1/8D	dotted 8th note	8th note with 75% swing	length of a 8th plus a 16th note
1/4	quarter note		
1/2T	half note triplet	4th note with 66% swing	
1/4D	dotted quarter note	4th note with 75% swing	length of a quarter plus a 8th note
1/2	half note		

4 Model 72 Synthesizer System

Softube Model 72 Synthesizer System is our new fully modeled vintage monophonic synthesizer based a true legend in music instrument history. The hardware was originally launched in 1970 and totally changed the market for how synthesizers were perceived, making them finally accessible to the average musician. It was compact, powerful and easy to use. Our golden model left the factory floor in September of 1972, hence the name. This synthesizer contains plenty of analogue quirks such as excessive distortion in filter and amp when pushed, as well as organic and rich tone generation from its three oscillators.

Getting Started

Model 72 Synthesizer System is really two plugins – a Model 72 Instrument, to be used on instrument channels controlled via MIDI - and a Model 72 FX, which is to be added to audio channels in order to being able to use the filter external input.

Model 72 makes use of two separate GUI sizes, one for small screens (without keyboard) and one for larger screens (with keyboard). You can force Model 72 to use the small GUI by clicking in that checkbox in the option menu and then restarting the plugin.

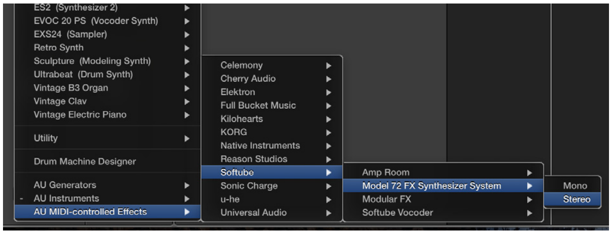
When using Model 72 as an instrument solely for creating sounds, add the Model 72 Synthesizer System plugin to an instrument or MIDI track in your DAW. Now, either use a connected MIDI keyboard controller or edit notes in your DAW to play your plugin synthesizer. Model 72 will respond to the full range of your keyboard and can also be configured to respond to Velocity.

When using Model 72 as an audio processing unit, add the Model 72 FX version to your audio or mix track. As you turn up the External input volume knob you'll see the overload lamp start to respond to the incoming audio but you still cannot hear anything – this is because, just like the original hardware instrument, Model 72 needs to be played in order to open up its filter and amp and let the sound out. So you'll need to route an a MIDI channel to your effect and play a note in order to hear the sound be processed through. How this is done differentiates between different DAWs. Consult your DAW user manual to see how your DAW is setup. Here's two examples of how this is done:

In Ableton live, add Model 72 FX to the channel with the audio-clip that you want to process, then add another, adjacent MIDI-channel to control the envelope. Route the MIDI-channel as pictured above in order to get your MIDI-information for the note-gates to Model 72 FX when you're playing.

In Logic – things are a bit different here. Here's what to do:

1. Create an audio track and import an audio file. Or optionally, route the output of the audio track away from the stereo bus (if you only want to hear the audio passing through Model 72 FX).
2. Create a software instrument track.Populate the software instrument track with Model 72 FX from the menu "AU MIDI controlled Effects"

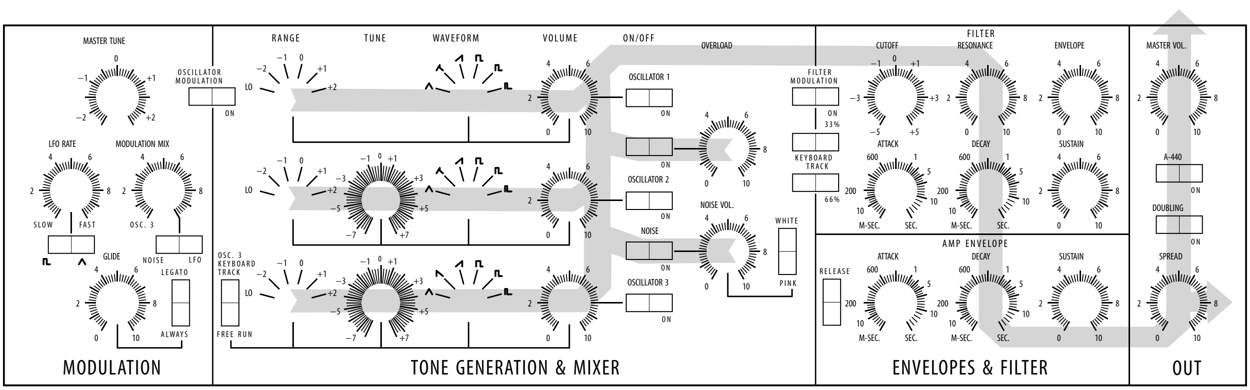


3. In the top-right corner of the instrument GUI, select "Side Chain", then "Audio".
4. Now, notes on the software instrument track will control the triggering of the envelopes of Model 72 FX that processes the sound from the audio-track.

You might already have noticed the difference between the Model 72 instrument plugin and the Model 72 FX version – Yes, the Model 72 instrument features a feedback path from the main output directly back into the external input, equivalent to connecting a cable from the hi gain output to the external input of the real hardware unit. This creates a roaring, raw and exciting feedback tone not found elsewhere.

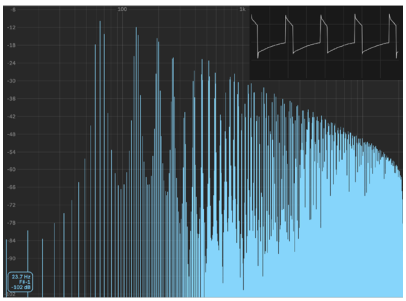
Sound architecture

The Model 72 is THE classic subtractive synthesizer. In fact, few might argue that the Model 72 hardware origins set the standard for how subsequently released subtractive synthesizers usually is setup. In a subtractive synthesizer you have at heart sound generation that creates harmonically rich content that then is filtered and amplified to form the final sound. The shaping of the sound is done with either momentous modifiers like Envelope generators or with periodically cycling modifiers like Low Frequency Oscillators (LFOs).

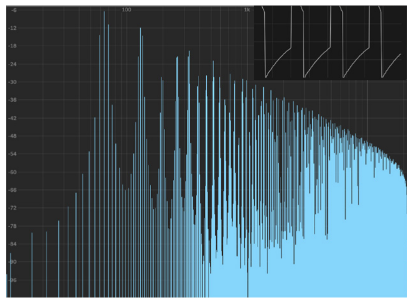


Tone generation & Mixer

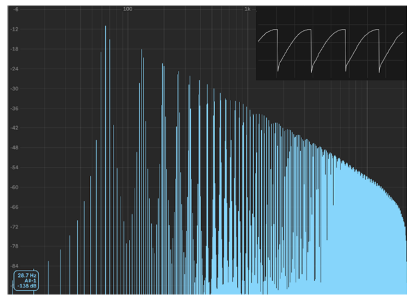
This is the heart of Model 72. The Tone generation section consists of three oscillators, top two of which are identical in terms of output waveforms and features. They are both linked to the incoming note-information and always tracks musically across 9 octaves. The third oscillator is a bit special since it can be unlinked from the incoming note-information and be used as a "free-running" modulation oscillator with extended tuning range on the tune knob. This third oscillator also features slightly different waveforms (an inverted sawtooth instead of shark-fin).



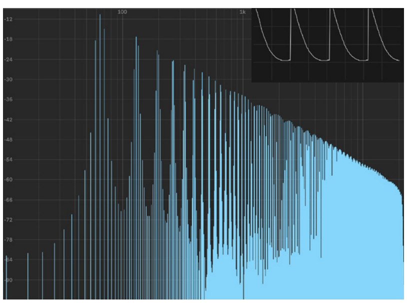
Narrow pulse



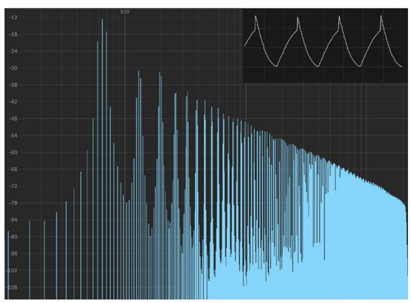
Pulse



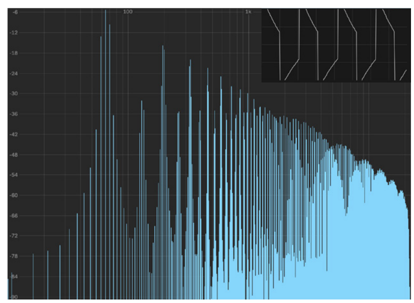
Ramp



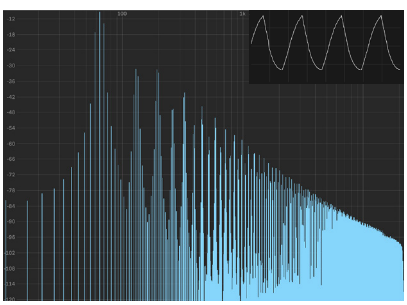
Sawtooth



Sharkfin

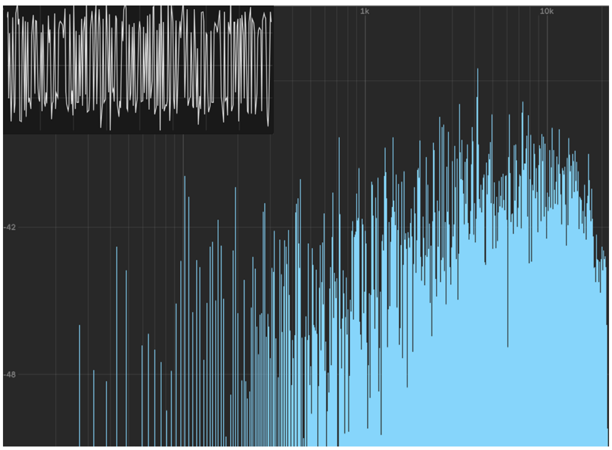


Square

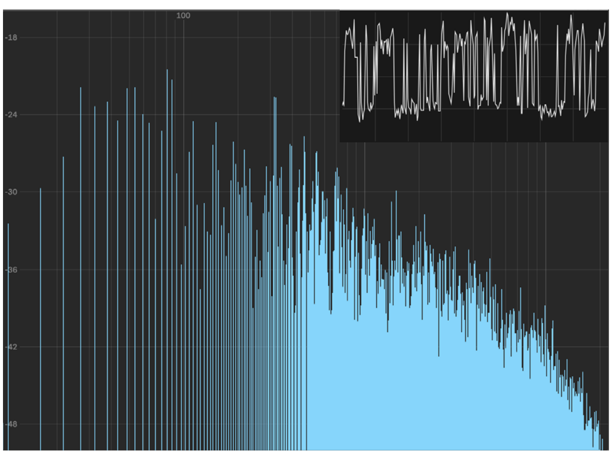


Triangle

The mixer-section consists of five switches for turning on and off the different sound sources mix into the filter and amp section of the Model 72. Next to these five switches are the volume knobs, setting the mix volume for each sound component. Apart from the three oscillators mentioned above there's also a feedback path (or external audio input in the FX version) and a white/pink noise generator.



White noise

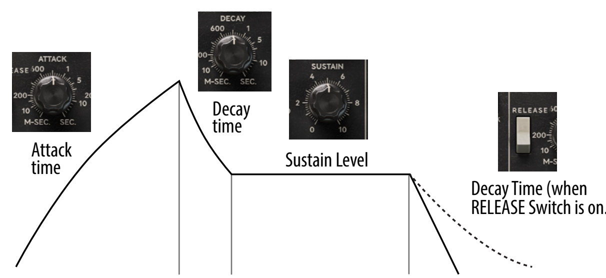


Pink noise

Envelopes & Filter

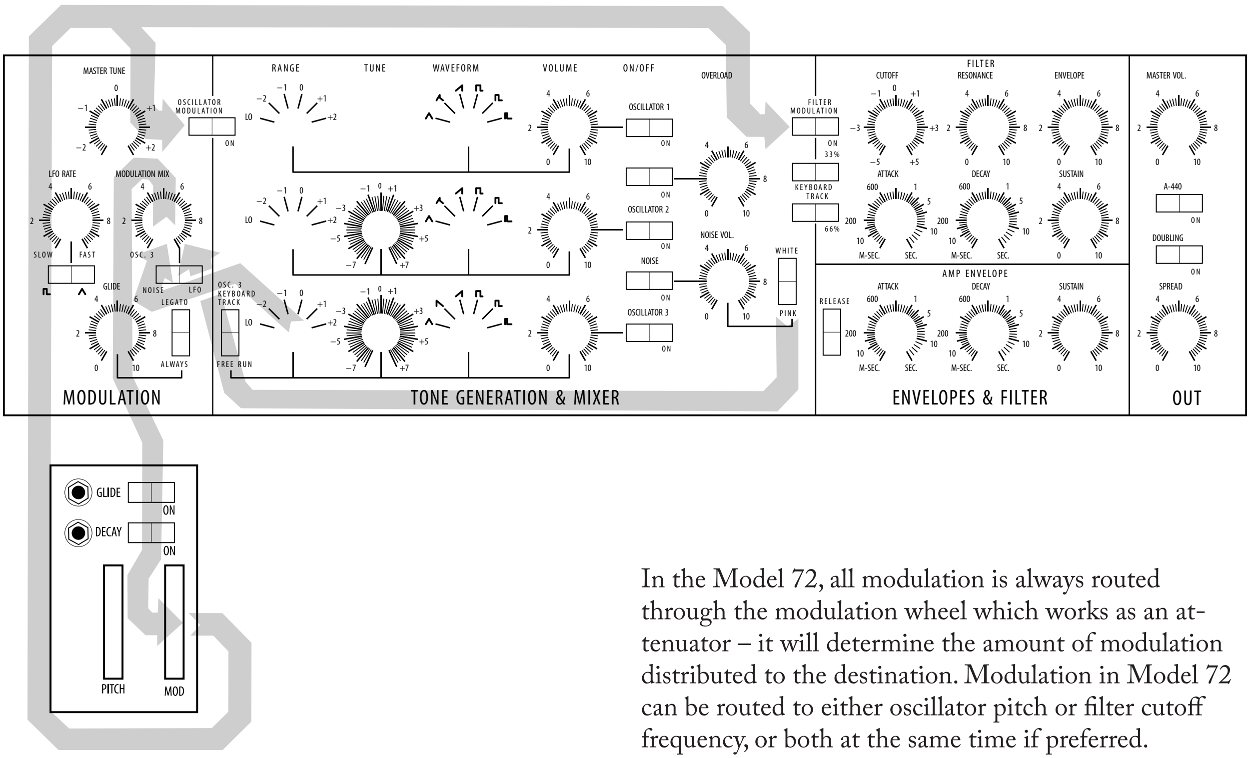
The next section to the right of the mixer is the Envelopes & Filter. The combined signals from the mixer enters the filter which is a voltage controlled 4 pole low-pass filter with resonance control (feedback). The filter cutoff frequency can be controlled via the cutoff knob but is also dynamically modified in real-time by the Model 72 different modulators. These are the dedicated envelope-generators in this section, but also the separate modulation section described below.

An Envelope generator is used to describe time-based changes in a sound. In the Model 72, the top envelope generator (marked by the attack, decay and sustain knobs) is used to shape the harmonic content of the sound. The bottom one, marked “Amp Envelope”, is used to shape the amplitude of the played sound. Both of these envelope generators are started when you play a note and will go through their individual phases in time set by the knobs.



Modulation

To the far left on the Model 72 panel we will find the modulation section. Here you'll find the master tune knob that sets the overall fine tuning of the Model 72 as well as the note glide rate and modulation mix. Apart from the classic options of using either the oscillator 3 output or the noise, we have also added a dedicated LFO (Low Frequency Oscillator) that can be used as an optional modulation source.



In the Model 72, all modulation is always routed through the modulation wheel which works as an attenuator – it will determine the amount of modulation distributed to the destination. Modulation in Model 72 can be routed to either oscillator pitch or filter cutoff frequency, or both at the same time if preferred.

The Out section

On the far right of the panel is the output section. It contains foremost the Master Volume knob that controls overall output volume. Just below the Master Volume knob is the A440 button which is the tuning guide tone on and off. As implied by its name it turns on a 440hz (A4) sine oscillator that is used for swift tuning purposes.

Below the A440 button is another switch: the Doubling feature and its associated knob, Spread. When enabled, this function will add the effect of having the notes played doubled by another Model 72 while the spread knob will set the how much this doubling will spread out in the stereo field.

The Keyboard

The big GUI of the Model 72 features a on-screen keyboard with the main purpose for fast and ease of use for sound design. Without reaching for your midi keyboard you'll have the possibility to trigger sounds off Model 72. The build in keyboard will play notes equal to a mid velocity (64) level but cannot be recorded as automation.

This on-screen keyboard does not reflect incoming MIDI and this is intentional and by design: We believe that this is representative of what the action of a real-world vintage synthesizer with add on MIDI retrofit looks like when controlled over MIDI.

The Expanded View

But wait – there's more! We have added another, "hidden" expansion panel that let you tweak some of the finer details of the oscillator behavior as well as some performance aspects of the Model 72. Click on the wooden board below the control panel (above the keyboard on the big GUI) and the slide-in expansion panel will pop up, revealing another 9 parameters for tweaking. The mouse-cursor will change from an arrow to a small pointing hand to indicate this click-area. Click just above or on the sides of the expansion panel to close it again. The state of this panel (open or closed) is remembered in your projects and saved presets.

Click here to open expanded view

Parameter description:

Here follows a more detailed description of each knob and switch in Model 72:

Master Tune This is the master tuning of the Model 72. It affects the pitch of all three oscillators as well as the keyboard tracking of the filter (added feature from hardware). Range is up and down two and a half semitones which is great for fine tuning against the included reference oscillator (see A440 switch). If greater adjustments or transposition is needed, the pitch-bend wheel can be used for this purpose (see below).

LFO rate This knob adjust the speed of Model 72's built in LFO (Low Frequency Oscillator) from a minimum speed of around 1/3Hz (knob fully counter clockwise) to maximum of around 100Hz (knob clockwise).

Modulation mix Adjust the balance between the Oscillator 3 and Noise (or the optional LFO when selected by the Noise/LFO switch mentioned below) on the modulation bus.

LFO waveform This switch sets the waveform of the LFO between Square and Triangle. The square waveform is great for automatic trills when modulating pitch, while the triangle wave is better for vibrato.

Noise / LFO This switch determines what the right modulation source in the modulation mix will be - Noise (as on the original hardware) or the optional LFO.

Glide This knob will set the glide time between each note in normal (always) or legato mode (see below). Glide is turned off when glide time is set to 0.

Legato This switch turns glide legato mode on and off (always). In normal mode (always) glide will always occur between two played notes. Key priority mode will whether the overlapping notes will glide or not (see separate description below in the section of the expanded view).

Oscillator Modulation This switch turns on or off the pitch-modulation to the oscillators. All modulation is routed via the modulation wheel and its effect will be different dependent on the type of modulation source (the mix of oscillator and noise or LFO), its modulation-rate and the amount.

Osc 1 Range This knob dial will set the octave reference point for oscillator 1. The "0" position is the normal range (sometimes called "8´ range") where an A3 midi-note will produce a A 440hz note. The "-1" and "-2" position transposes the oscillator one and two octaves below this respectively. Similarly, "+1" and "+2" position transposes the oscillator one and two octaves above the "0" position range. The "LO" position is special and places the oscillator five octaves below the "-2 range", well into subsonic level.

Osc 1 Waveform The waveform dial of oscillator 1 will enable you to choose which type of sonic character the oscillator will have. The different waveforms available are triangle, shark-fin (kind of a mixture between a triangle and sawtooth), sawtooth and then three different kinds of pulse waveforms ranging from square-wave (50% cycle) to two more narrow pulse-waves.

Osc 2 Range This knob has the same functionality for oscillator 2 as the Osc 1 range has for oscillator 1, see detailed description above.

Osc 2 Tune This knob is the individual tune knob of oscillator 2 and can be used to detune this second oscillator up or down seven semitones in relation to oscillator 1. Note that this knob add detuning in addition to Master tune and pitch-bend, which affects all oscillators.

Osc 2 Waveform This knob has the same functionality for oscillator 2 as the Osc 1 Waveform knob had for oscillator 1, see detailed description above.

Osc 3 Keyb Tracking This switch toggles Oscillator 3 between free-running mode (essentially a form of "unvoiced" mode) and keyboard tracking mode. When Oscillator 3 is in free-running mode it will not track the on-screen keyboard or incoming MIDI notes.

What would that be good for then, you might ask? Well, in the original hardware this was a way of using the oscillator 3 as a free-running modulator in "LO range mode" routed to the pitch of the other oscillators or the filter cutoff frequency. But of course you can also do this while in tracking mode as well, even when using in oscillator 3 in audible range! More of this later of course.

Good to know is that the range of the Osc 3 tune knob increases when free-running mode is enabled.

Osc 3 Range This knob has the same functionality for oscillator 3 as the Osc 1 range has for oscillator 1, see detailed description above.

Osc 3 Tune This knob has the same functionality for oscillator 3 as the Osc 2 tune knob has for oscillator 2, see above. A notable exception from this behavior is that the range of the Osc 3 tune knob increases when the free-running mode is enabled by the Osc 3 keyboard tracking switch turned off (see above).

Osc 3 Waveform This knob has the same functionality for oscillator 3 as the Osc 1 Waveform knob had for oscillator 1 and 2, but with one difference: instead of the shark-fin waveform of oscillator 1 and 2, oscillator 3 has a ramp waveform. This is great for usage as a modulation source and also to counter the falling sawtooth in oscillator 1 and 2 – together they create a very nice "soaring" sound when mixed.

Osc 1 Volume This is the mix volume knob for oscillator 1. It set the relative volume of the oscillator going into the filter.

Osc 1 Switch This is the oscillator 1 on/off switch. When on, it enables the audio signal from the oscillator to enter the mixer.

Feedback / External Input Volume This is the volume mix for feedback on Model 72 Instrument, or the external audio signal for Model 72 FX version.

When feedback is enabled (via the Feedback switch turned on) on the Model 72 Instrument version, this knob will control the volume of the feedback signal emulating a cable connected between hi gain output of the hardware and external audio input. Feedback is thus very dependent on mixer level of the oscillators, filter settings and amp envelope. When feedback is turned up past 4 or 5, feedback will start to self-oscillate and create an uncontrollable feedback tone reflecting also the behavior of the hardware.

On the Model 72 FX version this knob will control the volume of the external audio fed into the filter of the Model 72. The preamp overload lamp will indicate when the signal clips although this distortion might also be regarded as desirable.

Feedback / External Switch The switch adds the Feedback or external signal to the mixer (see further description above).

Osc 2 Volume Same functionality as the volume knob for oscillator 1 but for the oscillator 2 audio level into the mixer.

Osc 2 Switch This switch has the same functionality as the oscillator 1 on/off switch but for the oscillator 2. It enables or disables the audio from oscillator 2 reaching the mixer.

Noise Volume This knob will set the level of the noise going into the mixer.

Noise Colour This switch sets the which type of noise is being used (also for modulation), White or Pink. White noise in the Model 72 is more focused on high end frequencies, while Pink noise have more focus in the low frequency areas.

Noise Switch Enables or disables the audio from the noise generator to reach the mixer.

Osc 3 Volume Same functionality as the volume knob for oscillator 1 but for the oscillator 3 audio level into the mixer.

Osc 3 Switch This switch has the same functionality as the oscillator 1 on/off switch but for the oscillator 3. It enables or disables the audio from oscillator 3 reaching the mixer.

Filter Modulation This switch enables the modulation bus, the mixed and damped modulation signal, to reach and control the cutoff frequency of the filter. All modulation is routed via the modulation wheel and its effect on the filter will be different dependent on the type of modulation source (the mix of oscillator and noise or LFO), its modulation-rate and the amount. For example a medium rated LFO can be used to create a auto-wah effect being routed to the filter, while a audio-rate oscillator routed to the filter can create a distortion type of effect.

Cutoff Frequency This knob is the most important knob in the filter section. It controls the filter cutoff frequency which is the roll-off point of the lowpass filter in Model 72. The cutoff frequency also represents the reference point to which all other kinds of modulation (keyboard tracking, the envelope generator and the modulation from modulation bus), are added to.

Resonance This controls the internal feedback of the filter itself. Higher feedback creates a resonant peak around the cutoff frequency of the filter which at a certain point (resonance set to about 8 or 9) starts to self-oscillate. The self-oscillation itself creates a nice sine-oscillator that can be tuned and played via the keyboard tracking of the filter (see below).

Envelope Amount This knob will set the amount influence the filter envelope generator will have on the cutoff-frequency. The more envelope amount, the more the filter will open up with the attack, sustain and decay phases of the envelope. When adding midi velocity to envelope amount (via the expanded view), this knob will also mark the maximum value the envelope will influence the filter cutoff.

33% Switch This switch turns on 1/3 of the tracking of the keyboard to the filter cutoff. Keyboard-tracking means that higher notes will open up the lowpass filter to let the upper harmonics out. When combined with the 66% switch this will have the keyboard-tracking at 100%, meaning that the filter cutoff fully tracks the notes played (you can even play on the self-oscillating filter itself).

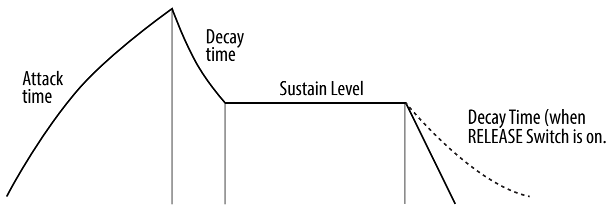
66% Switch This switch turns on the remaining 2/3 of the tracking of the keyboard to the filter cutoff. Effect on this switch by itself is twice as much tracking as the 33% switch but still not 100%. Combine this with the 33% switch to have the keyboard-tracking at 100%

Filter Attack The filter attack knob controls the time of initial "rising" phase of the filter envelope generator. It opens up the filter and ranges from very fast to around 10 seconds.

Filter Decay This knob controls the time of the "falling" decay phase of the filter envelope generator. It can be heard as a gradually dulling effect as the filter closes and ranges from very fast to around 50 seconds at maximum. It will also dictate the release time of the filter envelope when the release switch is enabled (see below).

Filter Sustain The filter sustain level is the "hold level" phase in the envelope generators of the Model 72. It defines the level at which the cutoff frequency is held until the note has ended (key is released). A little known fun fact is that the sustain level change rate is determined by the decay setting – this can be heard when changing the sustain level of the filter while holding down a note.

Release Switch This switch affects both the Filter Envelope generator and the Amp Envelope generator. It turns on the release phase of both Envelope generators which are controlled by each decay knob (Filter Decay and Amp Decay). When this knob is turned off, the release time for both envelopes will be set to zero (instant decay).



Amp Attack This knob controls the time of initial "rising" phase, the attack of the Amp envelope generator. It ranges from very fast, instantly hear sound set to minimum, to a smoother around 10 seconds long crescendo at maximum.

Amp Decay This knob controls the time of the "falling" decay phase of the amp envelope generator. Amp decay can be heard as a change in volume as the sound gradually diminish in amplitude down to the level set by the sustain knob. The amp decay times of Model 72 ranges from very fast (instant), to around 50 seconds at maximum. It will also dictate the release time of the amp envelope when the release switch is enabled (see above).

Amp Sustain The amp sustain level determines the "hold level" phase of the amp envelope and that means that it defines the volume level at which the sound will be held at after the initial attack and decay phases. The amp envelope generator will hold the amplitude at the same level until the note / key is released. The same rules for the rate of change while tweaking the amp sustain level applies as described in the section about the filter sustain knob (see above).

Master Volume The master volume knob determines the overall output level.

A440 Switch This is the reference tuning oscillator for Model 72. It plays an A at 440hz – great for tuning those complex FM modulated sounds! This switch turns it on and off.

Doubling Switch This switch turns on the doubling feature of Model 72. This emulates a doubling of the played baseline or lead-synth or whatever you're playing.

Doubling Spread This is knob determines the width of the doubling when enabled It goes from full mono at minimum, to fairly wide stereo spread at max. This knob will have no effect when the doubling switch is set to off.

Pitch bend The pitch bend wheel is unspringed just like the hardware original and can be used to transpose the whole keyboard. This in collaboration with the built-in tune oscillator (A440hz) comes in handy.

Mod Amount Modulation is an integral part of the Model 72 sound, so always remember that it is routed through the modulation wheel which sets the modulation amount. Both pitch and mod-wheels are parameters and thus saved as part of your project or preset.

The Expanded View Parameters

By clicking on the wooden board below the panel (or below the keyboard in the big GUI), the bottom expander panel with additional "pro-tweaking options" becomes accessible. They are:

Slop This parameter emulates the slow temperature drift of the individual oscillators on a vintage instrument over time. A low value is near ideal (no drift) and then go some 0.5 semi out of tune when reaching max, excessive drift.

Octave Trim This parameter is to emulate the accuracy of the global octave trim for the octave switches of the original hardware. This is a global multiplier for CV that is applied to the octave offset so that at zero, octaves track 100% correct, but at max octaves for all three oscillators stretch slightly more that 1 v/oct.

Stretch Trim This parameter is designed to emulate the CV/octave fine trim stretch for each oscillator. This is a per oscillator multiplier for CV applied to the CV tracking much a like how the octave trim works. Difference is that this stretch is a bit different per oscillator. A low value goes from ideal trim and then go some 0.1 - 0.2% out of tune when reaching maximum setting.

Velocity to Filter This parameter determines how MIDI note velocity will map to the filter cutoff frequency. A low value (minimum) on this parameter will have no effect and then scale up linear to at max, full velocity to cutoff scaling is obtained.

Velocity to Amp This parameter will determine how MIDI note velocity till map to the amplitude of the Model 72. Low value (minimum) will have no effect and then scale to a linear to maximum where full velocity will affect amp scaling so that MIDI velocity 0 will be totally silent.

Velocity to Filter Envelope This parameter determines how MIDI note velocity will map to filter envelope amount. A low value here will have no effect and then scale up linear to max where full velocity will affect filter envelope amount at 100% (which means what the Filter Envelope amount knob is currently set to).

Key Priority This parameter can be set to high, low or last and determine when which note have priority over another when multiple notes are played. It will also how legato portamento is applied.

Envelope Retrig This parameter determines when the envelope generators is re-triggered Normally - on the original hardware, the envelopes does not re-trig on when playing multiple notes at once (legato). Retrig is the option to have each note to restart the envelopes, regardless if one or more keys are playing at once.

Pitch bend Range This parameter will set pitch bend range up and down from minumum of a +-2 semitones to maximum of +-12 semitones.

Creating sounds

When creating sounds with Model 72 it is crucial to understand the fundamentally separate signalflows of the audio and modulating signals in the sound architecture. By understanding it is easier to new angles to a seemingly simple synthesizer. I want to share a few tips and tricks for your enjoyment:

pseudo PWM

Although Model 72 has different set pulse-waves of three different width, you might sometimes miss a feature more common in other subtractive synthesis: Pulse Width Modulation (PWM). This is a feature where the width of the pulse is changed by a external modulator at variable speed. Model 72 does not have PWM, but you can create a sound very similar to this by mixing the sawtooth waveform of oscillator 1 or 2 with the ramp waveform of oscillator 3 tuned very close in pitch. With very close pitch ratio you'll get a slow PWM like phasing, while detuned slighly more apart will get you a faster pulse modulating rate. One disadvantage of this method vs proper PWM is of course that the modulation rate is pitch-dependent and that modulation amount is hard to properly govern.

FM

By using Oscillator 3 as modulation source while in audio-rate, modulation applied to the pitch or the filter cutoff will create interesting harmonic or dis-harmonic overtones by the means of linear frequency modulation. To synthesize harmonic sounds, the modulating oscillator must have a harmonic relationship to the original carrier signal, in this case oscillator 2 and 3. As the amount of frequency modulation is increased (by the modulation wheel), the sound grows progressively complex. By modulating oscillator 1 and 2 with oscillator 3 set to frequencies that are non-integer multiples of the those frequencies, inharmonic bell-like and percussive spectra can be created. The disadvantage of using linear frequency modulation in the Model 72 is that the pitch the modulated oscillator is easily detuned out of range. This is when it is great to have the a440 reference oscillator close at hand to tune the sound back into normal usability with master tune and/or the pitch-bend wheel.

Filter FM distortion

Using oscillator 3 again as an audio-rate modulator, introducing modulation to the filter creates a kind of nice distorted sound.

Using the Filter as oscillator

The filter itself can act as an oscillator while in self-oscillation. Turn up the resonance beyond 8 and the filter itself can be used as a nice sounding sine-oscillator that, when keyboard-tracked (using the 1/3 and 2/3 switches) can be used for everything from haunting theremin-like lead sounds to subwoofer base sounds.

Chords

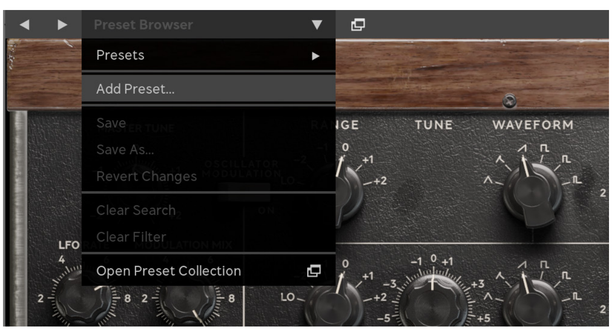
The fact that Model 72 has three oscillators lends it well to create monophonically controlled chord sounds. Set up a minor chord easily by clicking on the +7 label beside the oscillator 3 detune knob and +3 label beside the oscillator 2 detune knob.

Saving and loading sounds

Saving and recalling your own or other sounds is very simple in Model 72. In the top left corner of the plugin window is the preset display window and just to the left of that are the two buttons for next and previous preset quick browsing.

When clicking on these you will scroll back and forth in the list that is selected and filtered in the preset collection browser.

The preset collection browser window is quickly open by clicking on the window symbol to the right of the small preset name display. For details about Preset Collection see the separate chapter. When browsing the presets for Model 72, please note that there are separate presets for the Model 72 Instrument and the Model 72 FX version. While the presets for Model 72 Instrument are playoriented the presets for Model 72 FX are audio-processing oriented and won't probably sound at all before adding audio that is processed through it while playing it.



When you have created a really great sound that you want to save, simply click on the drop-down arrow just to the right of the preset display window and choose Add preset or Save as – you'll now be promted to give your preset a name (the same procedure can of course also be performed in the preset collection by itself) and press enter to save it.

Your preset has now been saved and can found among the user presets in Preset Collection. You can even share it with your friends by exporting it along with other presets as a separate ".softubepreset"-file. Details about this can be found in the Preset Collection chapter.

When just tweaking a sound you'll notice that the name will get an * addition after the name, indicating that this preset has been changed from its saved state. If you want to save those changes simply choose "save" in the preset dropdown menu.

Model 72 modules in Modular

When you purchase Model 72 Synthesizer System, you're also purchasing the possibility to use all the different components of Model 72 in our modular platform Softube Modular! There are seven different modules for you to use in Softube Modular: Tone Generator, Amp, Filter, Envelope, Noise/Glide, Doubling and Preamp. Here's a brief description of them:

Model 72 Tone Generator

These oscillators a lot hotter than the Doepfer VCOs included in Softube Modular. In order to get correct gain-staging, use a mixer or gain module to get the to perform well with the other modules in Modular. The depiction of the waveforms on the panel reflects the same error (flipped waveforms) as on the original hardware and includes all six waveforms used in Model 72. The FM jacks are AC coupled and thus are not suited for static CV like sequencer data.

Model 72 Amp

This amp does not only amplify - it distorts audio as well. The top CV jack is prepatched to 7.5v when no cable is inserted, this voltage is scaled through the Amp CV attentuator potentiometer. Additional CV (from for example an envelope) need to be inserted at CV in order for the Amp to amplify/pass audio.

Model 72 Filter

This classic fat lowpass filter selfoscillates when it goes past 8 and can track 1 v/octave fairly well when 33% and 66% switches are both engaged – they actually represent 1/3 and 2/3, but hey - the panel was not big enough for all decimals. Observe that the level knob is actually dampening for the input.

Model 72 Envelope

A gate above 0.66V in the gate jack activates the envelope. The decay knob also sets the release time when release switch is set to on (same functionality as in the Model 72 plugin). Release time is set to zero when the Release switch is turned off.

Model 72 Noise/Glide

This module consists of two parts – a white and pink noise generator, and a glide buffer. The white noise outputs far from ideal white noise but based upon performance of the original circuitry. Same goes for the Pink noise of course. The glide buffer is mainly designed for slewing control voltages but can of course be used to slew anything.

Model 72 Doubling

This essentially the same effect from the standalone Model 72 plugin but here this effect can be set gradually with the Doubling knob and the spread knob will expand the effect in stereo.

Model 72 preamp

This is the preamp for external audio from Model 72, it gives a very nice distortion when pushed.

Model 72 in Amp Room

Ever wanted some lush, warm and focused distortion or perhaps modulated sounds that you just can't seem to achieve anywhere? Then look no further! From the Model 72 Synthesizer System comes a great addition to the Amp Room platform: The Model 72 Envelope Filter module.

The Model 72 pedal is pretty straight-forward in its design, but here are the basics; the Amount knob controls how much of the envelope signal is used to open and close the filter and the Drive knob controls the input gain. The Cutoff knob opens and closes the filter, regulating the number of overtones that are let through. The Resonance knob changes the shape of the frequency response (the module starts to self-oscillate above 8) and the Smooth/Fast switch determines whether the envelope signal is smoothed out or not. For something like drums the Fast setting would be recommended, but for a guitar or bass Smooth would be the go-to setting.

Then again, experimenting is what this pedal is all about!

Credits

Kristofer Ulfves – Project lead, sound design, presets, user manual

Jacopo Lovatello – Tech lead, DSP models

Anton Eriksson - DSP models

Erik Sight – UI programming

Patrik Holmström - UI programming

Nis Wegmann – GUI design

Tord Jansson – programming and GUI engine customization

Maxus Widarsson – Deep testing, Qualification

Henrik Johansson – Testing, presets

Fredrik Mjelle – Testing, presets

Pontus Hagberg – Testing, presets

5 Monoment Bass

It is not always possible to achieve a modern bass sound with “just a synth”. While it is possible to create absolutely fabulous and punchy sounds with two oscillators, modulation and filters, a modern production usually requires more. And that “more” often consists of layers, movement, stereo imaging, and other effect processing that is normally done after recording the synth line.

But it doesn’t have to be that way. In **Monoment Bass** we threw out the “ordinary” oscillators and replaced them with a collection of well-crafted stereo samples; taken from unique, highquality synths, processed through the finest boutique gear, and sometimes with layer upon layer to create the sources that you need to craft your sound.

On top of that we created a synth workflow with filters, envelopes and modulation, all easily accessible and adjustable, for you to shape the sources into the tone you need right now.

And if the textures of the sources aren't enough, there's an effects section with ambience, spatialization, distortion, multi-band compressor, and equalization. All that you need to make your bass line take the desired position in the mix.

It's about sound quality and workflow. A modern bass sound in no time.

User interface

1. Source section
2. Filter and LFO section
3. Effects section, Tone section

The user interface consists of four parts:

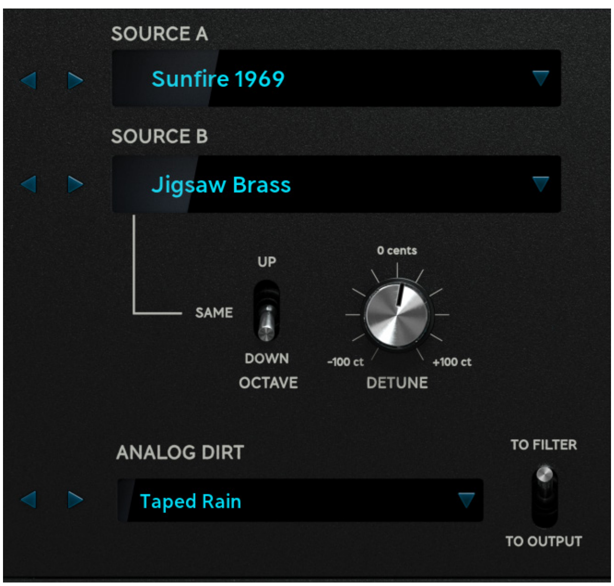
1. **Source section,** where you select different sound sources
2. **Filter and LFO section,** where you set the cut-off frequency of the filter, or use filter modulation to automatically change filter cut-off
3. **Tone,** where you balance your sources and set the punchiness of the sound.
4. **Effects,** the place for distortion, reverb, EQ, compression and spatialization.

On top of all this you also have a quick selector for presets that allows you to easily step through the presets in the Preset Collection, or if you've narrowed down the search by using tags or search words, step through the presets in the current search.

Source section

The Source section's main purpose is to select the sources that are the foundation of the bass sound. You do that by clicking on the title of the source and choose which source you want from the different categories. On the second source (SOURCE B), you can also select which octave it should play in, so that you can have SOURCE A and SOURCE B playing an octave apart for an even fatter sound, or you can slightly de-tune SOURCE B to get a nice chorus-y effect.

The last source does not come from pristine and high-quality synths, but from the different flaws they have. We call it ANALOG DIRT and these are a unique collection of noises that have been carefully sampled and adapted for Monoment Bass. These can be used as an almost regular noise source, or can be filtered together with the other sources and together create complex textures and richness. There's just something about good noise. It just makes everything feel more alive! It's like a pretty black-and-white photo. The noise makes it real, and without it, it would just look like something you snapped with your mobile. But the noise has to be *right*.



The level of noise, and also the amount of analog "goodness" is controlled by the **Aging** control in the TONE section.

To set the mix between SOURCE A and SOURCE B, use the **Source Mix** knob in the TONE section.

Parameters

Source A, Source B	Selects the sound for the first or second source. Click on the < > arrows or use the dropdown menu to navigate the different categories.
Octave	Sets the octave (-1, 0, +1) for the second source.
Detune	Detunes the second source. A bit of detuning makes everyone happier!
Analog dirt	Sets the sound of the analog dirt. Choose between noise sources and different attack sounds.
Dirt to filter	Engage if you want the ANALOG DIRT to be filtered by the same filter as SOURCE A and B. Otherwise it will go directly to the EFFECTS section. The amount of ANALOG DIRT is effectively controlled by the **Aging** knob in the TONE section.

What source-ry is this?

The sources are divided into different categories, to help finding the right source quickly. Below we describe each category, and also point you to our favorite sources.

Analog Clean

A collection of rather clean analog sources without any harshness and distortion. These sources are a good starting point for add distortion using the DRIVE section. By combining a clean and less clean source you can create an unique sonic richness.

Listen to: **"Monster Saw"**, a rather fat analog sound designed with the Schmidt Synthesizer, try out DRIVE with the MODERN setting.

Analog dark

A collection of analog sources that don't have a full frequency spectrum and sound rather dark. Each source use specific filtering of the synthesizers they were sampled from, which gives another character using the filter section in Monoment Bass.

Listen to: **"Voltage Overload"**, sound designed using the Schmidt Synthesizer, using Schmidt's unique filters to darken the sound.

Analog Punchy

A collection of analog sources with a punchy attack, mainly designed with a filter envelope. Each synths' envelope and filter have something special and we didn't just want to offer oscillator sounds from several synths, but also capture the different filter and envelope behavior.

Listen to: **"Unisono puncher"**, sound designed using the Schmidt Synthesizer, but here we used the filters and their envelopes to create a punchy source.

Analog Rich

A collection of full spectrum analog sources. Don't call it super saw, these are all analog sources with analog richness!

Listen to: **"Big Ship"**, this monster sound was designed with the Modal 002 synth, play it again and again with open filter and Aging above 60% to hear the rich and lively sound.

Digital Clean

A collection of rather clean digital sources without any harshness and distortion - but with early digital "low resolution" fun! These sources are a good starting point for adding additional distortion. By combining a clean and less clean source you will get unique sonic richness.

Listen to: **"Early Digital 1"**, this sound was designed with the Modal 002 synth, capturing sonic fingerprints similar to the famous PPGs. Try out DRIVE with the MODERN setting.

Digital Dark

A collection of analog sources that don't have a full frequency spectrum and sound rather dark. Each source use specific filtering of the synthesizers they were sampled from, which gives another character using the filter section in Monoment Bass.

Listen to: **"Winter Night"**, a really warm, but digital, sound designed with a rare Jomox Sunsyn, which great filters we had to use!

Digital Noisy

A collection of rather noisy digital sources of all kinds. The different kinds of harshness was designed with on-board features of the specific synths like distortion, *FM* or other modulations, as well as with external processing of the sources.

Listen to: **"Truck FM Radio"**, a sound designed using the Nonlinear Labs C15 synth - now you can hear that Monoment Bass is not a standard bass synth, but an instrument with really unique and inspiring sample sources!

Digital Punchy

A collection of digital sources with a punchy attack, mainly designed with a filter envelope. Each synths' envelope and filter have something special and we didn't just want to offer oscillator sounds from several synths, but also capture the different filter and envelope behavior.

Listen to: **"Synclavier FM1"**, did we say filter envelope before? This sound was designed using the famous Synclavier synth. It uses no filters, but has incredible sound shaping features, like harmonic envelopes and more.

Digital Rich

A collection of full spectrum digital sources. For these sources you have to try out FILTER section and filter envelopes to form the rich spectrum in the right way.

Listen to: **"Outburst"**, a monster of sound. Designed with the Nonlinear Labs C15. Make sure to use the filter here, unless you want the audience to panic from all fatness!

Organ

A small collection of sources that remind of organ sounds. These sounds are a fundamental part of timeless dance and house tracks, but they definitely fit many other genres as well.

Listen to: **"Hot Washed Organ"**, another sound designed with the Nonlinear Labs C15, not a typical house organ, but a really deep and noisy organ for your next pop, dance or indie track!

Processed

A small collection of sources that have been heavily processed with analog filters, FM modulation, secret stomp boxes and hardware modular devices. These sources enrich the Monoment source collection with sonic content, far beyond standard waveforms.

Listen to: **"Circuit Board Fire"**, the name isn't entirely accurate, but that's what it sounds like. Another sound designed with the Nonlinear Labs C15 and several secret processing tools on top.

Sub

A collection of sub bass sources, typically with a low harmonic content, which is useful for creating sub or 808 like basses. Be careful with your speakers! These sources are very useful when you need to add more bottom to any other source.

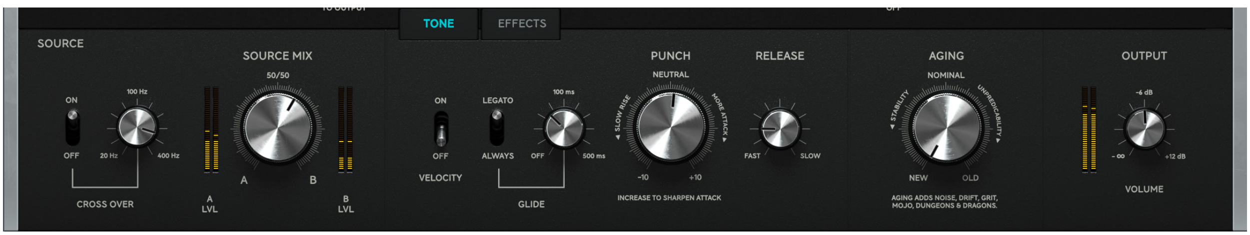
Listen to: **"Earth Shaker"**, the subtle richness and movement in the source makes even a single note bass line interesting. It was recorded with the Schmidt Synthesizer including filtering.

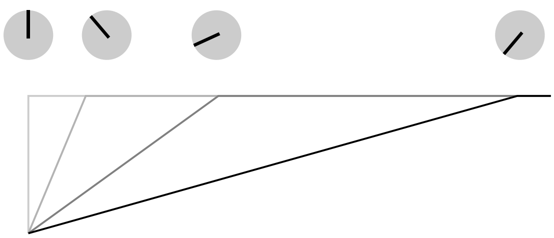
Tone section

The purpose of this section is to sculpt the tone: how you mix the sources, if it should have a sharp attack, or a slow fade-out. A lot of the "playability" of the sound can be adjusted here with just a few powerful controls.

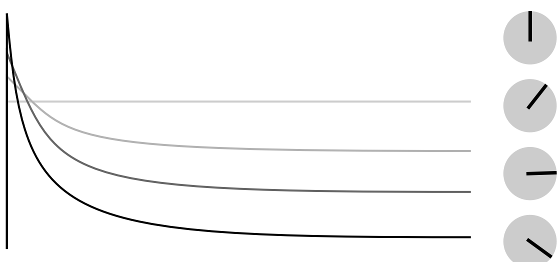
First step is to blend SOURCE A and B. In many cases it's enough with a single source, and you can set **Source Mix** all the way to A or B, but for modern production styles you often need to blend multiple sources. If you want to take the subs from SOURCE A, and the sizzle from SOURCE B, that's easy to do by engaging the **Crossover**. It separates SOURCE A and B at the **Crossover Frequency**.

Next step is to adjust the shape, the *envelope*, of the tone. That's easily done by adjusting the **Punch** control. Increase **Punch** and you'll get a snappier, punchier sound. Decrease it and you'll get a sound with a slow build-up. The **Release** sets the time it takes for the sound to fade-out.





Decreasing Punch from 12 o'clock down to nothing will make the onset of the note slower. This is useful for creating slow building sounds or to reduce the built-in attack of a source.



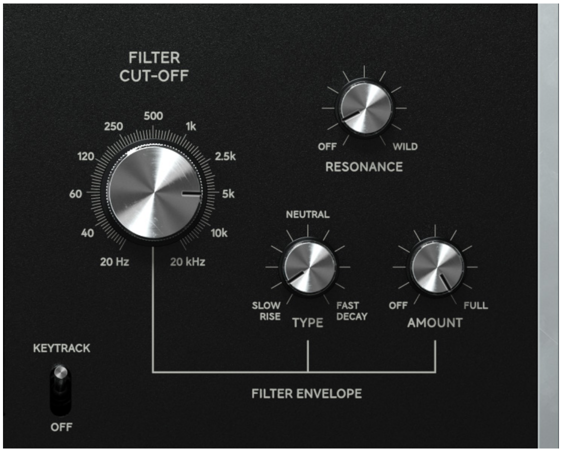
Increasing Punch from 12 o'clock will create a sharper attack with a lowered volume on the sustained note. Increase Punch to get a pluckier and sharper sound.

When designing Monoment Bass we wanted to make something that feels alive and real. Like a big beast that emits a lot of heat and creates earth-shattering lows. But sometimes you need something more static and a bit more predictable. That's what **Aging** does. More **Aging** gives you a more authentic, but also slightly more unpredictable sound with more dirt and grit. Don't want that? Fine, turn **Aging** all the way down.

Parameters

Crossover ON/OFF	Activates the crossover filter between SOURCE A and B.
Crossover frequency	Sets the crossover frequency. SOURCE A will get the low part, all frequencies below **Crossover Frequency**, and SOURCE B will get the frequencies above.
Source mix	Sets the blend between SOURCE A and B.
Source meters	Shows the volume level of SOURCE A and B after the crossover and mix.
Velocity ON/OFF	Engage to get a velocity sensitive instrument.
Glide type	Sets the type of glide or portamento. **Always on:** the note's pitch will always glide between the former and the current note, in the time that is set by **Glide Time**. **Legato:** the note will only glide when you play legato, ie. connected notes. If there is a space between to notes being played, the pitch will immediately jump to the latest played note. This mode is often called fingered glide, or auto glide.
Glide time	The amount of time it takes for the glide to reach the target note. At its minimum position (0.0), the glide functionality is turned OFF.
Punch	Sets the shape, the envelope, of the sound. At 12 o'clock it doesn't do anything. Increase it to get a sharper attack and more punch. Decrease it to get a softer attack with a slow build-up.
Release	Sets the time it takes for the note to fade out.
Aging	Sets the amount of unpredictability in Monoment Bass. More **Aging** means less stable pitch, more unpredictable attacks, more dirt and grit and more analog goodness.
Volume	Sets the final output volume.
Output meter	The level of the output signal.

Filter section



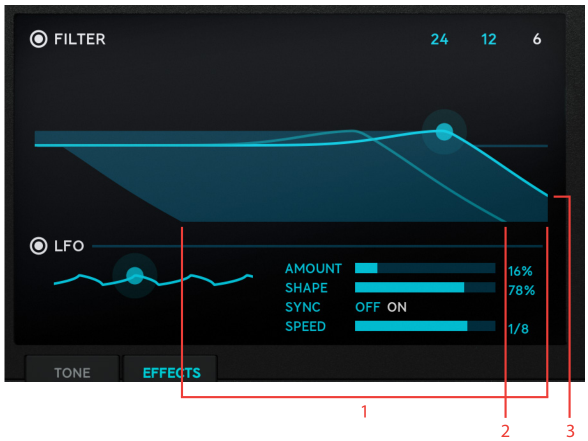
Even if a source, or a blend of sources, sound fantastic on its own, a synth bass sound is all about how you filter it, and how the filter changes when you hit a note. We want to make that as easy as possible and boiled it down to four main controls: the four knobs to the right.

Filter Cut-Off is the king. It rules them all. From subby bass to screeching high. That's what the **Cut-Off** do. **Resonance** is the trickster, that sets the amount of screechiness, and can be overdriven so that it distorts into a creative mayhem. (No, it's not your speakers breaking, it's how it should sound.) But the real heroes are **Filter Envelope Type** and **Amount**. They control how the filter changes when a note is being played! And that's how you get the classic synths sounds, like the "pluck" (the filter is closes when the note is played), or the "rise" (the filter is slowly opening then a note is played).

Filter Envelope Type goes from "slow rise" at its minimum position. The filter will slowly open up and reach the cut-off filter after a while. Turn it up to 11 o'clock and the filter will open up faster. At 12 o'clock it does nothing. Then when you increase it after that, the filter will close faster and faster, and at its maximum position the synth doesn't sound much more than a knock on your door. ("But hang on, I tried that, and IT DIDN'T DO NOTHING!")

Filter Envelope Amount controls how much of the **Type** effect you are getting ("Now you're telling me?") and with **Amount** on full, the filter sweep will go from nothing up to the cut-off frequency, or vice versa. Set it half-way, and the filter sweep will go from half-way between cutoff and nothing all the way up to the cut-off (or vice versa).

You can clearly see how the cut-off is set, how it is changing, and how much it can change in the filter graph.



1. Amount
2. Current frequency
3. Cut-off frequency

The combination of **Cut-off, Resonance, Type**, and **Amount** is by far the most powerful knobs in Monoment Bass. And we've spent a lot of time to make them as versatile as possible yet maintaining workflow and speed.

These filters distort. With too much resonance you'll get a very gritty (gritty-nice, not gritty-bad distortion!

Parameters

Filter cut-off Sets the cut-off frequency of the filter. Can also be adjusted by clicking and dragging the curve in the window.

Resonance Sets the amount of resonance (squealing) of the filter. With a lot of resonance, you can get the filter to distort in a very nice way! Can also be adjusted by clicking and dragging the curve in the window.

Filter envelope type Sets how the filter reacts to notes being played. Neutral at 12 o'clock, increase for a decaying sound, all the way to a short "plucky" sound. Decrease to get a rising filter sweep.

Filter envelope amount Sets how much the **Filter Envelope Type** should affect the filter cut-off. This is indicated by the shaded zone in the window.The current cut-off frequency (the combination of cut-off, filter envelopes and LFOs) can also be seen in the window.

In the window you can also set some additional filter parameters:

Filter ON/OFF Activates the filter.

24, 12, 6 Sets the type of filter:

24 dB/OCTAVE, classic analog synth filter with steep cut-off

12 dB/OCTAVE, typical 2-pole filter

6 dB/OCTAVE, a gentler filter used in a lot of modern productions. Technically, a 6 dB filter cannot be resonant, but many modern analog synths have combined the gentle 6 dB filter with a resonant circuit and achieving a filter like this.

> You will notice that heavy filters (like 24 dB/OCTAVE) will remove too much of the built-in character that the sources have, while 6 dB/OCTAVE lets the sources keep more of their character.

LFO (low frequency oscillator) is another way of automatically change the filter frequency, but instead of having it change when a note triggers, you can get it to rhythmically change in time with your music. In general, you want to set it to sync the tempo with your DAW (SYNC ON) and set the **LFO Speed** to 1/4TH notes.

LFO ON/OFF Activates the filter LFO.

LFO Rise (The dot on the LFO Curve) Sets the start point of the LFO, from triangle to saw-tooth.

LFO Amount Sets how much the LFO should affect the filter cut-off, similar to **Filter Envelope Amount.**

LFO Shape Sets the shape of the LFO, from shark-fin to triangle.

Sync ON/OFF	Activates tempo synced LFO.
LFO Speed/LFO DAW Speed	Sets the speed of the LFO, in seconds in the first case, in beats/bar in the latter case.

A note on filter modulation: synth enthusiasts will probably scratch their head about how the modulation amount on the filters work, so let's state it clearly: the modulation will always lower the cut-off frequency. Set the cut-off to the highest you need and adjust amount until the modulation goes low enough.

Effects section

The EFFECTS section consists of five different effects that you can use to further sculpt your bass sound. These effects have been carefully chosen and designed for bass sounds, and the control set has been thoroughly tweaked to give you as much control as possible with as few knobs as possible.

When dialing in distortion on a bass sound it's often best to listen to the bass in context of the whole mix. Bass distortion sounds very different in a context of other high frequency instruments.

Drive

If you need more dirt and grit, **Drive** is king. For a typical bass sound you often need one of two different types of distortion: either a low frequency roar (TRANSF.) that adds harmonics to the lowest frequencies so that they can be heard even on a small mobile speaker, or a full-band distortion to make the bass sound sit better in the mix (MODERN).

The algorithms for the DRIVE section come from the Harmonics Analog Saturation Processor plug-in, so if you like them, or want more options for the distortion, please check out the Harmonics plug-in.

Parameters

Drive ON/OFF	Activates the DRIVE section.
Type	Sets the type of distortion, TRANSF: a transformer based low-frequency distortion. Will only affect the lowest frequencies. MODERN: a full-band distortion
Drive	Sets the amount of distortion.

Ambience

Many have been taught that putting reverb on the bass is a classic mistake. That is often true, since the reverb smears out the bass in time, and you often need a very focused and tight bass. But what would creativity be if you can't break rules?

The ambience algorithms in Monoment Bass was designed for synth bass and are meant to enhance the stereo goodness and movement of the bass sounds. It can add amazing texture to your sound, and it is quite effective to use the SPATIALIZATION features in conjunction with the AMBIENCE to further tailor the sound.

Parameters

Ambience ON/OFF	Activates the AMBIENCE section.
Tone	Sets the tone of the AMBIENCE, with dark tones counter clockwise and brighter tones clockwise.
Dry/Wet	Sets the amount of AMBIENCE.
Type	Sets the type of AMBIENCE. NEUTRAL: A short all-round reverb, created with a modern hardware device combined with analog filtering. This ambience gives your basses some natural "air" and size. ELECTRIC: A short robotic reverb, created with a French reverb device from the 80s, combined with some secret offline processing of the recording. Use this ambience to create a very special sonic fingerprint of your basses. METAL: A unique metallic reverb, created with a French reverb device from the 80s. This ambience gives your bass sound some industrial metallic sonic flavor. RUSTY: A short unique reverb, created with a modern hardware device combined with offline pitch shifting and other vintage tools to give a rusty edge to your bass sound DIRTY: A dirty reverb, created with a famous modular reverb device and combined with some secret stomp boxes. Use this ambience to create a trashy little space around your bass sound.

EQ

Equalization, tone control, is an essential part of all sounds. The EQ in Monoment Bass is an extremely powerful two-knob algorithm that, of course, has been tailored for bass synths (do we need to say that again?).

You can see the two controls (**Bass** and **Tilt**) as having two different purposes. With the **Tilt** knob you adjust the overall sound, the balance between the high frequency harmonics and the sub-frequency content. The **Bass** control is what you use to specifically target the bass frequencies. In other words: **Tilt** is used to shape the character, and **Bass** is for making it sit in the mix.

Parameters

EQ ON/OFF	Activates the EQ section.
Bass	Counter clock-wise: cuts bass frequencies Clock-wise: reduces high frequency content With the semi-resonant nature of the cut-filter it's very easy to find sweet-spots where the low frequency rumble has been removed while the fundamental frequencies are being emphasized. This technique has been used in many famous low-frequency EQ circuits.
Tilt	Turn clock-wise for a brighter sound, turn counter clock-wise for a darker sound. This is the knob you reach for when you need to balance the bass sound in the mix.

Multiband

The knob to rule them all. Multiband makes everything fatter, clearer, better, more defined… it's hard to not use it on everything. We got our product manager Paul, who also designed the preset algorithms for **Weiss MM-1** and **Drawmer S73**, to design a one-knob multiband compressor especially for bass synth, and the result is the multiband you see in Monoment Bass. With just a bit of compression you'll get a warmer and fuller sound. Dial it up even more and you get something that is ready to cut on a record. Overdrive it and it gets a character on its own.

Parameters

Multiband ON/OFF	Activates the multiband compressor.
Amount	Dial in the amount of compression.
Meters	From left to right: the amount of low-frequency gain reduction, the amount of mid-frequency gain reduction, the amount of high-frequency gain reduction.

Spatialization

All sources and algorithms in Monoment Bass are stereo, and sometimes you want to enhance the stereo-ness of them, or maybe reduce it. That's what the SPATIALIZATION block does.

The **Sub Mono** knob forces the audio below the set frequency to become mono, which is great if you're cutting vinyl or need a firmer low end. The **High Freq Widener** will increase the stereo information in the audio. When used in tandem, the **Sub Mono** will still make all frequencies below in mono, regardless of what the **High Freq Widener** is set at.

Parameters

Spatialization ON/OFF	Activates the SPATIALIZATION effect.
Sub Mono	All audio below the set frequency will be in mono. Pull it all the way up to force the output to be completely in mono.
High Freq. Widener	Increase to get more stereo width.

The SPATIALIZATION section is a mid/side matrix with built-in EQ. **Sub Mono** enhances the "mid" part while **High Freq. Widener** enhances the "side" part.

Sources of sources

We spent a huge effort selecting and recording the sources for Monoment Bass, in order to reach a new level of quality and fun using this product.

All samples were recorded in 96 kHz in stereo via Merging Technologies Hapi AD converters and resampled to 44.1 kHz after all external processing.

All samples were designed with and sampled from real hardware synths. We didn't sample the standard Moog or Roland synths for basses, but those exclusive and rare products such as the following synths:

- Schmidt Synthesizer
- Modal 002
- Non Linear Labs C15
- Kawai K5000
- Jomox Sunsyn
- Yamaha DX5
- NED Synclavier

Synths from four decades were sampled via secret processing chains of hand-picked boutique pre-amps, EQs and compressors. We sampled every 3rd note over two and a half octave and sampled every note three times to capture typical minimal sonic variations, controllable via the **Aging** knob.

The ATTACK samples in the ANALOG DIRT section are based on real recordings of objects with a certain attack sound, sonically fitting to a bass for additional punch and dirt. The NOISE samples are based on real noise source recordings and processed sounds from field recordings. Those noises add controllable dirt to the basses.

Monoment For Modular

When you purchase Monoment Bass, you're also purchasing the possibility to use it as three separate blocks in our modular platform **Softube Modular**!

The three different modules for you to use in Softube Modular: **Monoment Source, Monoment Filter** and **Monoment FX**. Here's a brief description of them:

Monoment Source	This is the Source engine from Monoment Bass, available as stand-alone module. It has a built in amplitude envelope, waveform mixing, dirt level and aging. It can be played monophonically within modular over midi or monophonically via the gate and note jacks. Glide, Punch, Release and Aging parameters features external CV control via jacks.
Monoment Filter	This is the Filter section from Monoment Bass, available as stand-alone module. It has a built in filter envelope and LFO. The envelope is triggered through the external gate jack. All sequencer functions - Slew, Range, Amount and Swing parameters - features external CV control. The LFO can be automatically locked to the DAW tempo or set to free running mode.
Monoment FX	This is the Effect engine for Monoment Bass, available as stand-alone module. It has four modes that can be run one at a time – Drive, EQ, Multi comp and Spatialization. Up to three parameters in each mode has external CV control.

Credits

Tobias Menguser – Initial concept, multi sample sound recording. **Niklas Odelholm** – Sound, visual and product design, project management. **Björn Rödseth** – DSP programming. **Erik Sight** – Framework programming. **Filip Thunström** – GUI programming. **Jacopo Lovatello** – Filter design and modeling. **Paul Shyrinskykh** – Compressor sound design. **Johan Bremin** – Quality assurance. **Kristofer Ulfves** – Synth expert. **Ulf Ekelöf** – 3D graphics modeling. **Klaus Baetz** – programming.**Madison Mars, Vandalism, Cr2, WA Production, Black Octopus, Function Loops** – Factory presets.

6 Parallels

Parallels is all about sonic exploration. Softube Parallels is a plugin synthesizer featuring high quality pre-recorded multi-waveforms in the Source-section, custom made filters in the Shaper-sections and a special selection of different effects in the, you guessed it - Effects section (right hand side panel). Most of the parameters in the different sections can be modulated by one of the five modulator types (LFO, RND, EUC, ENV, SEQ) available in the Mod Pod section (left hand side panel).

We hope you will find Parallels inspirational and that you will have fun sound exploring its hidden depths!

Foreword by Johan Antoni, concept creator of Parallels

I have been using, selling and buying synthesizers as long as I can remember. My father had a music store and imported the famed EMS Synthi-synthesizers (among others) in the mid 70s. You could say that I grew up with synthesizers and that they occupied much of my time since I was a kid. Since 1996, I followed in my fathers footsteps and have been running (dare I say?) Stockholms premier synth shop, selling mainly vintage, quirky and boutique synths. Or any synth I find interesting, really. Lately, its been mostly expensive or rare modular systems.

A couple of years ago, after having played around with a Korg 800DV and an Oberheim TwoVoice in the store for a couple of lazy summer days with no customers, I realized that I sort of missed a less complicated and instead more direct way of doing more complex sounds. As you may or may not know, both of these synths mentioned are old 70s vintage analog synths with a kind easy-to-follow architecture, very versatile filters (even compared to todays standards), not that many parameters and they work by giving you two of everything. In short, they are very competent, user friendly, and they allow you to layer two sounds on top of each other in order to make complex sounding sounds very quickly and easy.

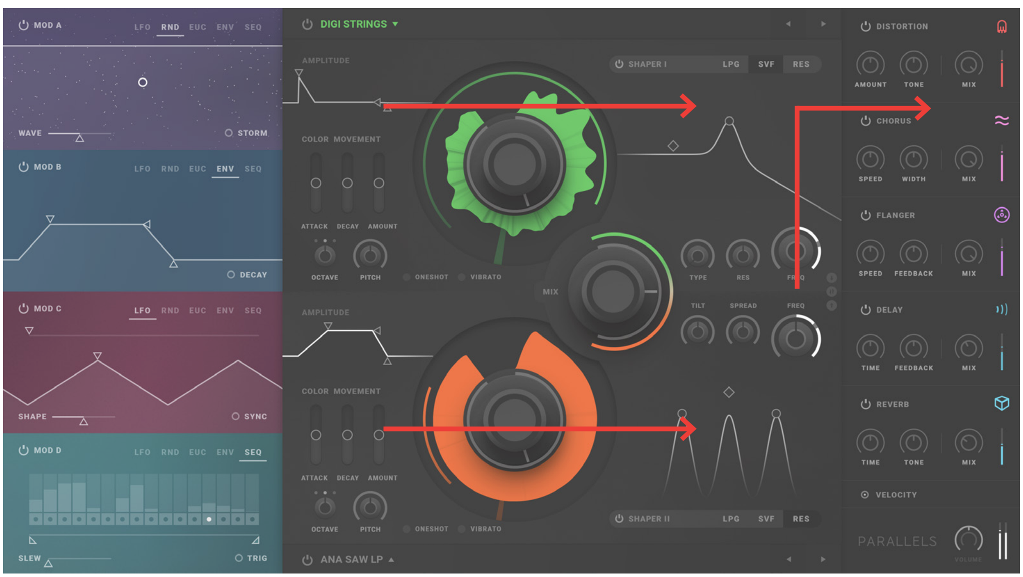
Combined with my idea of the ”source waveform” based synthesis (explained later in this manual) which allows you to switch quickly between different snapshots of synthesis, this became a concept I could not let go of. I wanted this as a software plug-in. After some very basic software idea tests I approached my friends at Softube and they loved the concept; why not start developing this right now?

For my part, what followed was producing the ”source waveforms”. This took place mostly after hours in the store using whatever I could think of (mostly vintage synth gear), some basic programming at home and a period visiting friends in Tokyo.

User interface

1. Mod pod
2. Source section
3. Shaper section
4. Effect section

Parallels signal flow chart



Introduction to Parallels's different Sections

Parallels is a very simple, but at the same time, complex 14 voice polyphonic synth. Polyphony is shared among two layers, upper and lower. Parallels's simplicity lies in that it has a very logical signal flow for the audio. However, for the complex part, Parallels also has a modulation resource, the Mod Pod, with multiple slots that can be customized and routed to a huge number of available destinations. Here follows a brief description of the different sections of Parallels:

Source section

Let's start looking at the Source section: placed in the middle left at the top and bottom, Parallels's Source section and is the heart of the instrument. Each Source section, the upper and lower parts, contains a waveform that is selected through the Wave dropdown menu or via the wave selection arrows. This waveform is displayed as wrapped around so called the Color knob which clearly indicates the waveform start-point and modulated loop. Each Source section also have a dedicated Color modulation envelope and a dedicated amp envelope.

Shaper

The Shaper section contains three different filtering modes – **Low Pass Gate (LPG)**, **State Variable Filter (SVF)** and **Resonator (RES)** - each one with three modulatable parameters.

Mod Pod

This Modulation Pod (alias Mod Pod) contains five different kinds of modulation that can be routed virtually everywhere in Parallels (with a few exceptions). It contains four Modulation Slots named Mod A to D which can contain any of the five different modulation types at ones. The Modulation types are a Low Frequency Oscillator (LFO), a digital Random Source (RND), a Euclidian sequencer (EUC), a polyphonic modulation envelope (ENV) and a modulation sequencer (SEQ).

Effects

The effects section contains five different effects in series that can be turned on or off/bypassed, each with control over two parameters and dry/wet portion. The effects are Distortion, Chorus, Flanger, Delay (unsynced/synced) and Reverb.

MIDI

Parallels is a virtual instrument that interfaces with your DAW through the MIDI protocol. Many DAWs load these types of "virtual instruments" on to so called Instrument tracks which often are set up to send MIDI information by default to the plugin instrument on channel 1. Parallels actually receives its MIDI information on all 16 channels (MIDI omni mode).

The Velocity option turns on and off Velocity to amplitude response. This mean that, when Velocity is set to on, soft strokes on your MIDI keyboard will create pretty quiet notes while harder strokes will create louder notes. When Velocity is turned off, any change in velocity will thus not be reflected in quiet and loud notes.

The keyboard range of Parallels covers a little more than five octaves (G0 – C#6). MIDI notes received outside this range will not be registered and thus not heard.

The pitch bend range is always + - 2 semitones.

Modulation wheel (MIDI CC#1) is always routed to vibrato amount, engaged when vibrato is turned on. The present value of modulation wheel is not saved in the patch.

Interface

Press CTRL/CMD to have fine-adjust mode activated (as in our other plugins) which also will prevent more than one step being changed at the same time in the sequencer.

Furthermore, for controls that change two parameter at the same time (such as for the shaper controls), it can be "locked" into just X or Y direction by pressing SHIFT when dragging. These two controls can also be combined (where applicable).

Getting started

Basics – what is this Source Waveforms, Shapers, Envelopes, Mod Pods etc???

Sound is a vibration that typically propagates as an audible wave of pressure, through a transmission medium such as a gas, liquid or solid. In digital audio (as in Parallels), sound is signal that has been recorded as or converted into digital form. This means that the sound wave of the audio signal is encoded as numerical samples in continuous sequence stored in a digital medium (for example a computer hard drive).

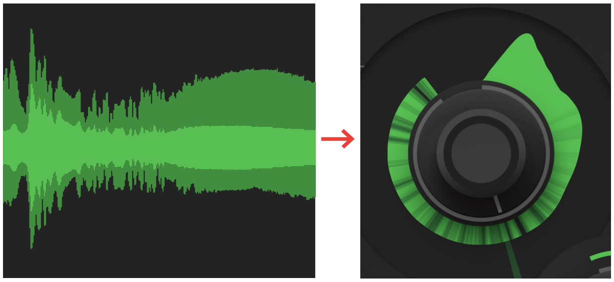
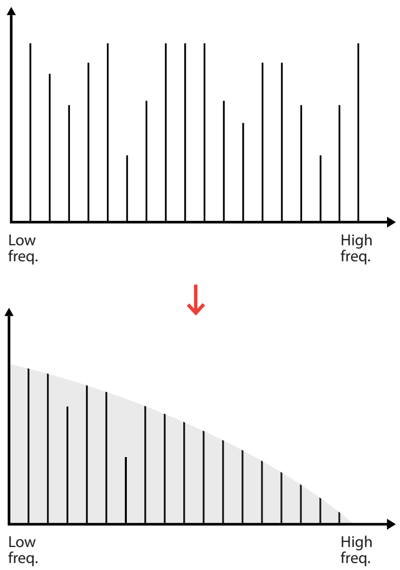


Figure wave to waveshape.

Parallels's Source section uses pre-recorded digitized audio that has been meticulously recorded in high fidelity and prepared to cover multiple octaves in order to ensure high quality and great musical performance. Each waveform has been selected to be musically inspiring to work with and also each represent a sonic transformation in the sonic spectral properties, but also sometimes in amplitude and pitch. Each waveform can use any startpoint and be swept back and forth freely by the dedicated Color envelope shape or any modulation.

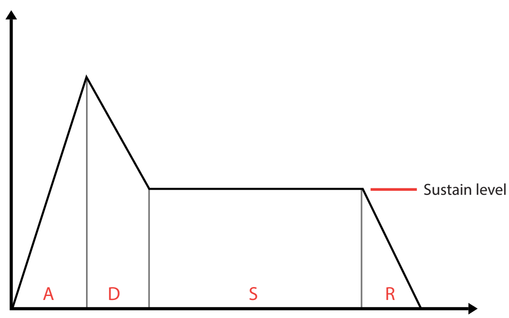
The Source waveforms in Parallels creates a sonically rich spectrum. This spectrum can be further shaped into containing the desired amount of overtones and amplitudes with the so called Shaper section. The Shaper section contains different kinds of filters (and distortion). Each filter affects the spectral outcome of the Source waveforms in different musical ways.



Example of lowpass filtering.

An envelope generator describes a triggered series of timed events, in instruments normally triggered by the note start. The envelope is used to represent the natural sound properties in a formalised model. There are different models and school of thougths when it comes to envelopes in synthesizers. Parallels uses several different ones (AD, ASR and ADSR) for flexibility.

The **Attack** portion is the rise time of the envelope, i e the time it takes for it to reach its peak. The **Decay** time is the time it takes for the envelope to fall from the peak down to the next level, often to be the **Sustain** level. The Sustain level is the level at which the note is statically suspended and sustained until the note is released (key is released). The last envelope phase is the **Release** phase, describing the time it takes from the note is released until it has all died out.



ADSR Envelope.

Getting Started: Let's get to know Parallels's Source section!

1. We will start off by turning off all modulation and all effects.

2. Continue on by turning off the lower Source section and the top Shaper. Now when you play Parallels, all you should hear is the raw waveform of the upper Source section.

3. Turn down the upper Source Movement Amount slider to zero. Now try to move the big Color knob clockwise and counterclockwise back and forth while playing. You will notice that the sound goes from a duller string sound to brighter one the further clockwise you will go. And at the very end the waveform is silent.

Now turn the Color knob back all the way counter-clockwise (Color 0%).

Now image that you could do the clockwise and counterclockwise movement back and forth motion you previously did with your mouse in an automated manner - Yes, this is precisely what the Color Movement Envelope is made for! The Movement Attack time is the timing of the clockwise movement, the Movement Decay time the timing of the counterclockwise movement and Movement Amount the striking distance (i e the multiplier) of the first two.

4. Try out the Color Movement Envelope by setting a fairly long Movement Attack (~1300ms) and Decay (~2500ms), with a fairly high Movement Amount (75%). Now play a long sustained note, listen and watch as the playhead of the Source waveforms follows your instructions and now sweeps forward clockwise at first, then turning counterclockwise and returning to the start-point set by the big Color knob.

5. Now, try out different settings where you experiment with different Color knob settings for different start-point of the sweep and different Movement Amount for different strike range. Also try different time ratios of the Movement Attack and Decay. You'll notice that when Movement Amount is too high in combination with a high value on the Color knob (initial startpoint) the Movement envelope can sometimes sweep outside the waveforms boundaries resulting in a silenced or partially note. This can however be used to our advantage (more of this later).

6. OK, let's try something else. Set the Movement amount to zero again and the Color knob at default position 50% (alt-click on the knob to set it back to initial value).

7. Now, adjust the amplitude-envelope at the upper left corner of the Source section, into a smoother curve. This is done by adjusting the attack time and sustain level to match the picture above. Play a note. You will notice that the note now is quieter in the beginning, gradually fading up to sustained level and staying there until the note is released.

8. Now, adjust the amplitude-envelope curve to match the picture above again: Attack time at zero, sustain level at 100% and release time at zero. Play a note again and you will notice the instant full level amplitude response of the waveform now. And when you release the note, the sound is silenced at once.

9. Now, using the techniques you learned above, try out and explore the other waveforms by browsing the different waveforms in the waveform menu or using the arrow buttons in the upper right corner of the Source section. You will notice the vast amount variation available already within Parallels's Source section itself, and we've only just begun!

Getting Started: Now let's turn our attention to Parallels's Shaper section.

1. Let's turn on the Shaper section again. SVF mode is the default Shaper mode for the upper Shaper section.

2. Now try clicking on the hotspot in the displayed SVF (state variable filter) curve - it's on the crest or peak of the curve. Try clicking and moving it around while playing. You'll notice that by changing this graph in horizontal direction changes the SVF cutoff frequency while changing the graph in vertical direction changes the SVF resonance. Note also that the cutoff frequency and resonance knobs changes with the curve and vise versa.

3. Click on the other hotspot in the filter diagram and try to move it. You'll notice that it is linked to the Type knob and locked to be modified only horizontally. When the Type parameter is changed from a low value to a high value, the SVF topology is seamlessly changed from a lowpass filter, on to a bandpass filter and finally a high pass filter. The displayed graph also reflects this behaviour.

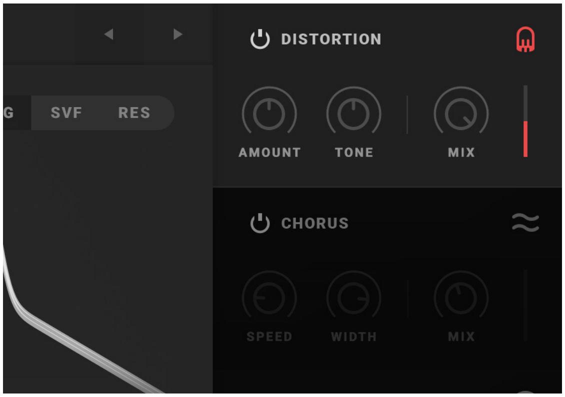
4. OK, Let's look briefly at the Shaper RES mode, the Resonator. Click on the RES tab on the Shaper mode selector on the top right, an interactive three peak filter curve appears. The Resonator shaper mode is great for creating formants or vocal sounds, but can also be used for crude phasing duties. By clicking on the middle peak hotspot and dragging your mouse or touchpad left and right, the RES Center frequency is changed. This is also reflected on the Freq knob in the lower right part of the Shaper window.

5. Let's click on the hotspots on one of the side-peaks in the filter curve. Clicking and dragging your mouse or touchpad reveals that moving the side-peaks hotspots vertically changes the RES Tilt (peak emphasis), while moving them horizontally changes the RES spread (the space in frequency between the different peaks). This change is, of course, also reflected on the adjacent knobs on the lower middle and left in the Shaper section.

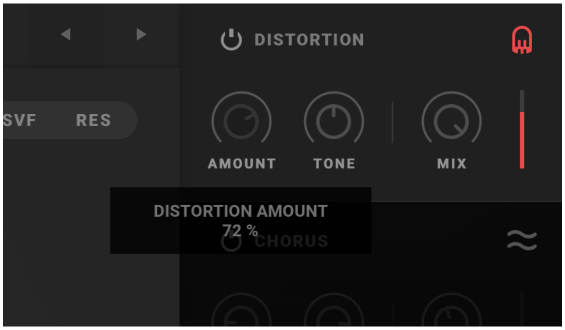
6. At last, let's look at the Shaper LPG mode. LPG stands for Low Pass Gate which is a special vactrol-controlled filter and VCA combination controlled via a so called vactrol, often used in so called West-Coast modular synthesis. Parallels's Shaper LPG mode draws some inspiration from the classic LPG to create a filter with a controlled "sloppy" cutoff response. Click on the LPG tab on the Shape mode selector on the top right to bring up the LPG filter curve.

7. Clicking on the different hotspots in the filter curve for the LPG reveals similar behaviour as the other two modes. The peak hotspot moved vertically controls the LPG resonance, while moving it horizontally, it will control the LPG cutoff frequency. The other hotspot on the left is locked to only vertical movement as it reflects the LPG Slew, i e the amount of "sloppyness" behaviour the Shaper LPG will display when its cutoff frequency is modulated (more of this later). Let's move on to the Effects section!

Getting Started: Getting to know Parallels's Effects section!



1. OK, let's start by clicking on the activate-button on the top effect in the Effects section (on the right in Parallels), distortion. When the color graphics light up (in the case of distortion, it is red), the effect is active.

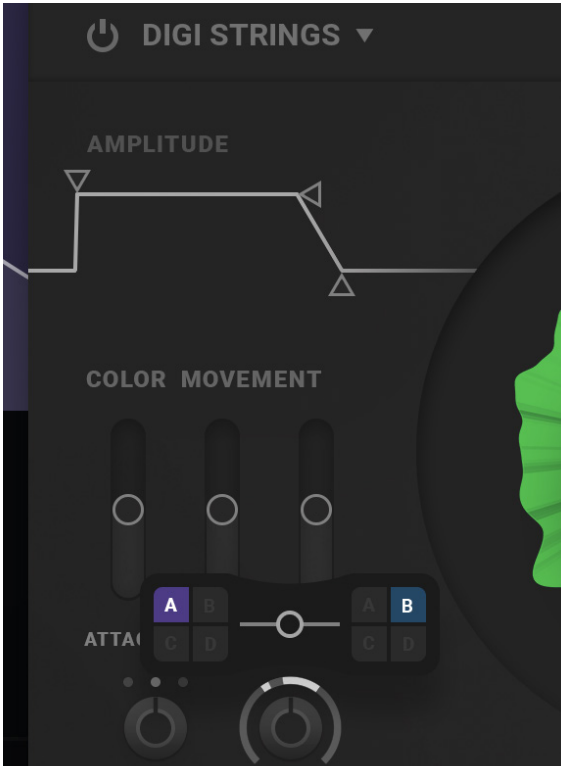


2. Each effect features two specially selected editable parameters and a dry/wet mix (the knob on the right). All the effects in Parallels is chained in series with a signal flow from top to bottom, which is good to know when working with effect.

Getting Started: Looking at Parallels's Modulation section

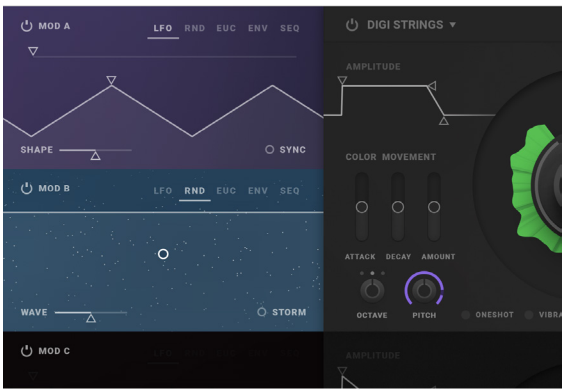
On the left panel we will find the Modulation Pod (or Mod Pod for short), containing all the four different modulation slots named A, B, C and D. Most of the parameters in the different sections can be modulated by one of the five modulator types (LFO, RND, EUC, ENV, SEQ) available in each slot of this Mod Pod section.

1. Let's get started by adding an LFO in Mod Pod slot A (the slot on top).



2. Now, let's add some modulation to the pitch of the upper Source. This is done by clicking on the outer ring of the pitch and dragging upwards. You will see the modulation ring start to fill with white color from the middle and out.

3. By adjusting the Modulation mix slider between the Modulation sources you will see that the color of the modulation will change accordingly. When the Modulation mix is turned all the way towards left where Mod Pod Slot A is selected as modulation source, the modulation color turns violet; the same color as the Mod Pod Slot A has. Play a note and you will hear the LFO affecting the pitch up and down, like a siren. Experiment by changing the speed, shape and rise/fall times in the LFO while playing.



4. Now, let's add another modulation source. Click on the Mod B Active button to active Mod Pod Slot B and select RND as Mod B Type as pictured.

5. Now, click on the Pitch knob outer ring again to bring up the modulation sources and mix controls for this destination. Slide the Modulation mix all the way to the right, towards mod B as modulation source, and you will see the modulation color turn blue; the same color as modulation slot B. Now, when playing a note you will hear a highly buzzing erratic sound due to the noise-source affecting the Source waveform pitch. Experiment by changing the different parameters in the RND source now selected in modulation slot B.

6. As we noticed before, when the modulation mix slider is set somewhere in the middle, in between the modulation source selected on the left and on the right, the modulation color is white; indication that there is a mixture of two modulation sources going on here.

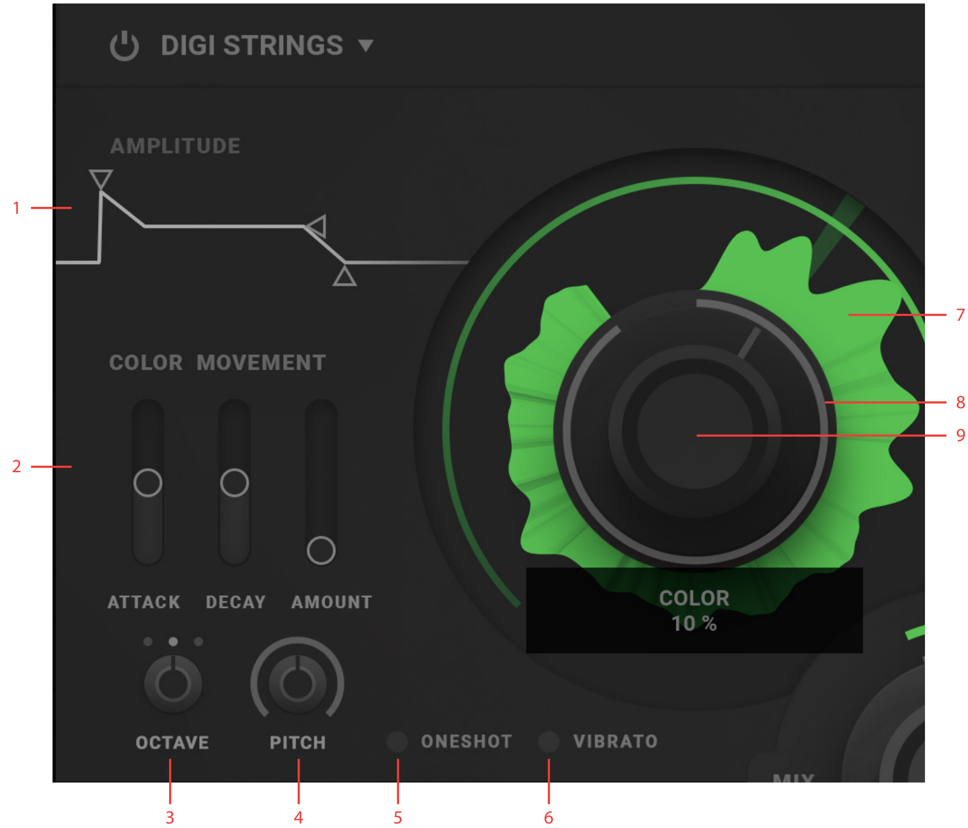
7. As pictured and described, you can route any modulator to any destination by clicking on the other ring of a parameter of choice. Then chose the modulation from two of the four modulation slots and the mix balance between them. Set the amount of modulation at the destination by clicking on the other ring of parameter again, and drag your mouse or mouse-pad upwards (the ring around the parameter will fill up).

Note that some parameters cannot be modulated - all non-knob-type interface will not accept modulation routing from the Mod Pod. This is true with the exception of the Octave knob that's not available for modulation (despite being a knob).

Parallels in detail

In these sections that follows, you'll find all detailed information about each and every part of Parallels.

The Source Section



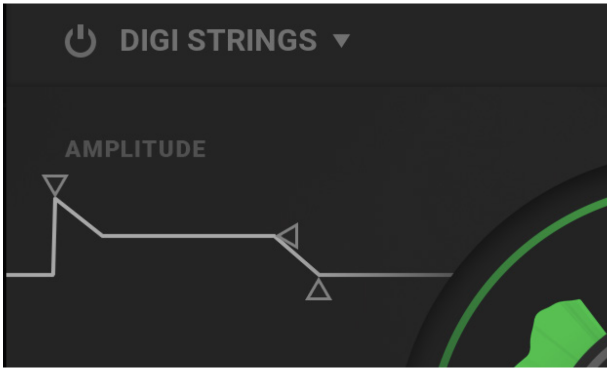
1. Amplitude envelope
2. Color movement envelope
3. Octave
4. Pitch
5. Oneshot
6. Vibrato
7. Waveform
8. Modulation
9. Color knob

Active on/off	This parameter activates and deactivates the Source with its adjacent Shaper section. When a section is deactivated, its graphics is dimmed indicating that it is no longer heard.
Wave menu	In this dropdown menu you can select from each of the different wave category.
Wave selection arrows	With the wave selection arrows you can rapidly change between waveforms without using the drop-down menu. Notice that a changed waveform will not be heard until a new note is pressed.

A complete Waveforms list is presented in "Appendix" on page 121.

The Amplitude envelope

The amplitude envelope controls the overall change of amplitude over time. It is a so called ADSR (Attack, Decay, Sustain, Release) envelope with simplified controls (i e one common control for Decay and Release times).



Amplitude envelope attack (1ms - 16sec) This horizontal slider controls the attack portion of the (upper) layer's amplitude envelope. Keep this short (low values) for instantly responsive and snappy sounds, while using higher values for extending the attack time for slower, fading in notes.

Amplitude envelope sustain (0-100%) This vertical slider controls the sustain level of the amplitude envelope for the upper layer. This is the sustained volume level that is held after the initial attack and decay times of the envelope. If the sustain level is at 100% the decay will not be pronounced, effectively turning the envelope into a ASR (Attack Sustain Release) envelope.

Amplitude envelope release (1ms-16sec) This horizontal slider controls the decay and release portion of (upper) layer's amplitude envelope. This dual function parameter controls both the initial decay phase appearing right after the attack phase, but simultaneously also controls the final decay (often also called release) that appears when a note is released. Setting this parameter low creates very fast, abruptly ending sounds, while using higher values creates lush sounds with long hanging notes.

The Color envelope

The Color envelope is a dedicated AD(Attack and Decay)-envelope that will perform a sweeping of the start and loop-position of the Source waveform, first in a clockwise motion during attack phase, and then in a counterclockwise motion during the decay phase.

Color (0-100%) This parameter control the waveform playback start- and loop-point. This is the offset point from which the playback of the waveform always will begin.

Color movement envelope attack (1ms-16sec) This parameter controls the attack time of the dedicated Color Movement AD (Attack Decay) envelope. The attack time is the timing of the rising portion (attack) of the envelope.

Color movement envelope decay (1ms-16sec) This parameter controls the decay time of the Color Movement envelope. The decay time is the timing of the falling portion (decay) of the envelope.

Color movement envelope amount (0-100%) This parameter controls the amount of color movement inflicted by the dedicated Color Movement envelope. Color movement is the modulation of the start- and playback-position of the Source waveform. The movement inflicted by the Color Movement envelope is added to the offset position set by the Color knob, as well as the modulation added from the Mod Pod (see further description below).

Octave (-1 / 0 / +1) This parameter let the user transpose the Source waveform up or down once octave. Default value is 0 (no transposition).

Pitch (-12 – 0 - +12 semitones) With this parameter, the user can detune the pitch of the Source waveform up or down from the default offset. Maximum values are + and – one octave. Modulate this parameter with the Mod Pod sequencers for creating pseudo-arpeggios (see further description below).

Oneshot (on/off)	This parameter turns on the Oneshot playback mode of the Source waveform. In Oneshot mode, sustained loop-functionality is turned off and the waveform is played from the where the current start-position (Color knob + the modulation determines where). The Color-envelope is also disabled in Oneshot mode.
Vibrato (on/off)	This parameter turn on the internal vibrato. Vibrato is a low-frequency pitch modulation that emulated the vibrato of a violinist or cellist. When this parameter is engaged, MIDI modulation amount (MIDI controller #1) will control the maximum amount of vibrato (default is 100%).

The Mix Balance functionality

Src Mix	This is the mix balance in volume between the upper and lower layer in Parallels. When this knob is turned fully counter-clockwise, only the upper layer will sound. And of course, if this knob is turned fully clockwise, only the lower layer will sound. This described behavior is also reflected in the graphics as the waveform's shown color intensity also will reflect the current state. When a waveform is faded to grey, it is not heard.

The Shaper Section

1. Shaper active on/off
2. Shaper model
3. Hotspot
4. Interactive filter diagram
5. Filter parameters 1-3

Active on/off This parameter activates and deactivates the Shaper section. When the Shaper is deactivated, its graphics is dimmed indicating that it is no longer heard and the sound from its adjacent Source section is bypassed through to the Effect section.

Type LPG/SVF/RES This parameter determines which tone shaping model the Shaper section will use. The three types are:

The LPG (Low Pass Gate) Is a lowpass type of filter inspired by West Coast synthesis, with a variable slew, cutoff and resonance control. While the cutoff and resonance controls works pretty much as on any like other lowpass filter with feedback control, the slew parameter controls the response of modulation applied to cutoff frequency, mimicking the "slop" of a vactrol control.

LPG frequency (20Hz - 20kHz) This is the cutoff frequency of the LPG. When modulated, the LPG Slew will determine the "sloppiness" its response (see below).

LPG resonance (0 - 100%) This is the amount of feedback in the LPG Shaper.

LPG slew (0-100%) This is the amount of "sloppiness" in the cutoff response. It ranges from 20 ms to 2000 ms, minimum to maximum response.

The SVF (State Variable Filter) Is an OTA-style filter with variable characteristics (lowpass, bandpass, highpas) and separate control over cutoff and the a wild resonance.

SVF frequency (16Hz - 17.6 kHz) This is the cutoff frequency of the SVF Shaper.

SVF resonance (0 - 100%) This is the amount of feedback in the SVF Shaper.

SVF type (lowpass - bandpass -highpass) This parameter set the SVF type used in the Shaper which ranges from lowpass mode with knob at counterclockwise, bandpass with knob at 12 o'clock, to highpass mode at fully clockwise.

The RES (Resonator) Is a three band variable resonator-bank with control over the center frequency, as well as the tilt and spread of the bands.

RES center freq (159Hz – 2518kHz)	This is the center resonator peak cutoff frequency of the RES Shaper mode.
RES tilt (0 – 100%)	This parameter set the RES tilt balance used in the Shaper which ranges from high freq boost with knob at counterclockwise to low freq boost at fully clockwise.
RES spread (0 - 100%)	This parameter set the spread between the three different bands, from roughly doubling in frequency for each band at fully counter-clockwise, to up to eight times the frequency at fully clockwise (x2 - x8).

The Shaper Copy and Swap functionality

The Shaper sections also contains some smartness - you can copy and swap setting between the upper and lower Shaper by clicking on one of the three different "arrow"-buttons (see further description below).

Copy shaper settings to lower.

Swap shaper settings between upper and lower.

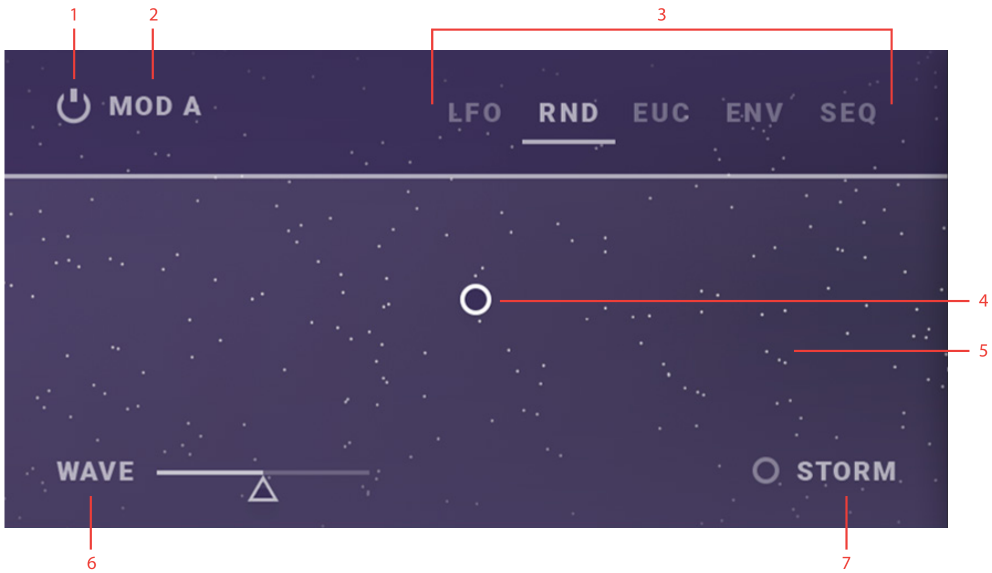
Copy shaper settings to upper.

Copy from (arrow on module you want to copy from)	Clicking on this arrow on the upper or lower Shaper instantly copies the settings for the Shaper type currently selected, selects the same Shaper type on the other Shaper.
Swap	Clicking on this "double arrow" symbol instantly swaps the settings of the upper and lower Shaper. Clicking the same cymbol again swaps back.

Note: No modulation information is copied or swapped with the functions described above. Because of this, settings copied from the upper Shaper to the lower Shaper or vise versa can sound quite different although the same Shaper type and settings are being used. This can of course be cured by manually selecting the same modulation sources, mix and amount.

The Mod Pod

The left section of Parallels contains the modulation slots A to D and is called the Modulation Pod or **Mod Pod** for short. Each slot in the Mod Pod contains a configurable modulation source. This means that any of the modulation slots can be assigned to do similar modulation tasks and to be assigned just about anywhere. All slots have their own individual modulation color that is reflected in the assigned modulation.



1. Modulation active on/off
2. Modulation slot A-D
3. Modulation type
4. Hotspot
5. Interactive modulation diagram
6. Modulation parameter 1
7. Modulation parameter 2

Active on/off This button turns the Modulation Slot on or off. When turned off, modulation is cut from all destinations using it and thus can radically change the sound.

Modulation type (LFO/RND/EUC /ENV/SEQ) With this type button you can choose which kind of modulation you want the modulation slot to use. Here follows the detailed description of each of these modulation types available in Parallels:

LFO (Low-frequency Oscillator)

An LFO is a low-frequency oscillator - a cyclic rise and fall type of wave oscillation that can be used to modulate your target parameter destination with as slow movement as 0.1Hz (i e 1/10 cycle each second, meaning that cycle time is 10 seconds).

LFO rise (0-100%) This parameter sets the how much of the LFO cycle that will be the rising portion and thus also the remainder that is to be the fall portion.

LFO shape (0-100%) This parameter set the shape on the rise and fall period of the LFO curve. It goes seamlessly from inverted exponential (0%) to linear (50%) to exponential at (100%).

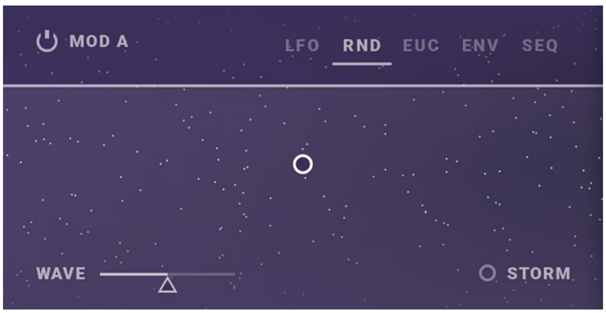
LFO DAW speed (4 bars/1 bar/half note/ fourth note/eight note/sixteenth note) When the LFO sync is turned on, each LFO cycle is quantized into division of a beat ranging from one cycle every 4th bar to one cycle each 16th note.

LFO speed (0.1 Hz – 50Hz) The LFO speed parameter sets the cycle time of the LFO, from a very slow 0.1Hz to a moderate 50Hz (just above audio-rate).

LFO sync (off/on) This parameter turn on and off the DAW tempo sync. When turned on, each LFO cycle is quantized into division of a beat (see LFO DAW speed).

RND (Random modulation generator)

The RND is a digital random modulation generator that outputs bursts of random pulses at a regular rate. This means that for each clock-cycle a pulse may of may not appear. The clocked rate in the RND is called “rain” to illustrate the light drops of rain at a low rate, to the heavy rain of a high rate. There's also a low- or highpass filtering of the generated noise as well a internal LFO controlled amplitude-shaping called “wave”. The “Storm” option makes the wave go bananas!



RND rain (0-100%) This parameter which is the movement in the Y-axis on RND panel, controls the internal speed of the clocked random source, from the light drops of rain at a low rate (0%), to the heavy rain of a high rate (100%) - white noise. The Y-axis movement can be locked by holding down shift and moving the mouse vertically for only adjustment of the RND rain (clocked speed).

RND noise (lowpass - flat - highpass) This parameter which is the movement in the X-axis on RND panel, controls the filtering of the generated noise, from the heavy lowpass filtered version at 0%, on to flat (no filtering) at 50% and on to heavy highpass filtered at 100%. The X-axis movement can be locked by holding down shift and moving the mouse horizontally for only adjustment of the RND noise (filtered noise).

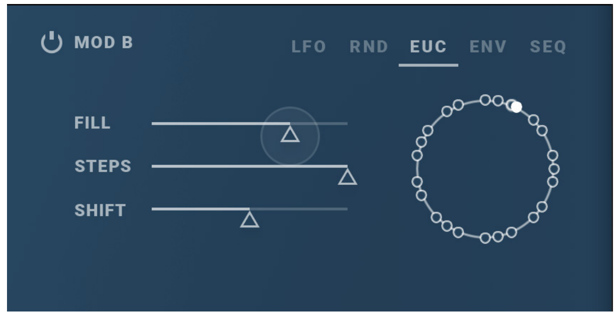
RND wave (0 – 100%) This parameter controls the amount of “breathing” scaling to the modulation amount. This is a cyclic slow amplitude modulation of the filtered random pulse output that can be turned completely off by setting the RND wave to 0%.

RND storm (on/off) This option completely turns off the wave modulation and instead amplitude modulates the RND output with another uncorrelated white noise source.

EUC (Euclidian sequence generator)

The Euclidian sequence generator (EUC) in Parallels can be used to create a number of evenly distributed gates (modulation maximum) over the course of one bar. The number of gates are filled from the first beat clockwise by a percentage and the resulting gates can be shifted around the beat for (nearly) endless array of possibilities. The EUC is always synced to DAW tempo and position, therefore its effect will not be heard when the DAW transport is stopped.

EUC fill (0-100%)	This parameter determines how many percentage of the available steps in a bar that will send gates.

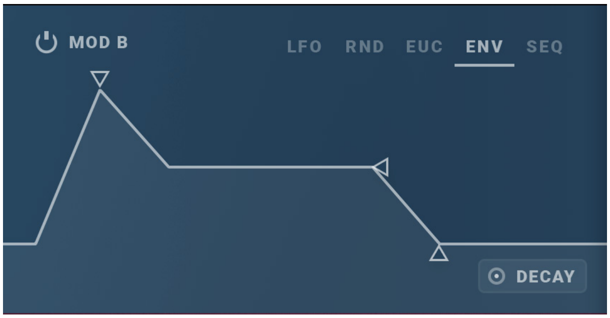


For example, in a 16 step sequence a 25% fill will result in 4 distributed gates.

EUC steps (1-32)	This parameter determines the number of evenly distributed gates (modulation maximum) over the course of one bar.
EUC shift (0-100%)	The EUC shift parameter is used to shift the steps around the perimeter by percentage.
EUC Retrig (off/on)	This parameter determine if the EUC cycle will retrigger on next 16th note.

ENV (modulation Envelope generator)

The modulation Envelope generator of Parallels is a ASR or ADSR generator (configured by the ENV Decay on/off button). It can be used to, for example, modulating the filters in the Shaper section, sweeping the waveforms loop start-points and altering the dry/wet level of the effects.



ENV Attack (5ms - 4000ms) This parameter set the attack time of the modulation envelope. It defines the rise time until the envelope has reached its peak. A lower attack value equals a faster rise time.

ENV Sustain (0-100%) The Sustain parameter defines the hold level of the modulation envelope. This is the level that the envelope will rise to from zero through the attack phase that lasts as long as defined by the ENV attack parameter (described above).

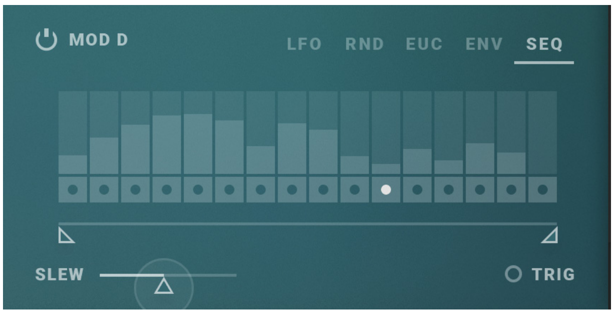
ENV Release (50ms - 8000ms) This parameter defines the final fall time of the modulation envelope. It takes from a key is released until the modulation envelope has fallen down to zero again. When the modulation envelope is set to ADSR mode via the ENV Decay option.

ENV Decay (off/on) When this option is turned on, the modulation envelope changes its curve to a ADSR envelope, adding the optional decay stage that will be proportional to, and controlled by the same parameter as the ENV Release (descriped above). In a ADSR envelope, the decay portion describes the fall time from the peak of the envelope, down to its sustain level.

Note: the modulation envelope is the only modulation parameter that is assigned per voice, that is affects each destination polyphonically (true for all destinations except for destinations with the effect section).

SEQ (modulation Sequencer)

The modulation Sequencer in Parallels is designed to provide slewed sequenced modulation or gating information that can be used to modulate different things: pitch, filter cutoff etc. It's speed is a 16th note division set by the DAW sync tempo and its alignment is to transport when DAW is running. The modulation sequencer has two modes: modulation and trig. Where modulation sequencing is stepped level set by each step, the trigged gates of the trig mode is either on or off, and each gate only lasts half of a 16th note (a 32th note). In modulation mode you can easily draw you modulation curves continuously across the 16 steps, and fine-adjustments can be done by pressing and holding CTRL (pc) or CMD (mac) to fine-adjust. This will also prevent more than one step being changed at the same time in the sequencer. Note that the draw functionality can be turned off in the setup menu.



SEQ first step (1-16) This parameter sets the first step of the modulation sequence. If this step occurs after the last step, then the sequence will go backwards. For example, if the first step is 12 and the last step is 8, then the sequence will start at step 12, then play steps 11 through 8 and loop back to step 12 on the next 16th note.

SEQ last step (1-16) This parameter sets the last step if the modulation sequence. If this step is positioned before the first step, the sequence will reverse order as described above.

SEQ slew (0-100%) The slew parameter defines the slew rate of the output modulation from the modulation sequencer. The higher slew value, the slower lag (or glide) is added to the output modulation.

SEQ Trig (off/on) This parameter controls the output mode of the sequencer. Modulation mode (SEQ Trig: off) sequencing is stepped level set by each step, and Trigger mode (SEQ Trig:on) is trigged gates.

SEQ step CV 1 – 16 (0-100%) This is the amount of modulation to be generated by this step.

SEQ step trig 1 – 16 (off/on) In trig-mode, this is the amount (either 0 or 100%) to be generated by the gates of each step. Each gate will last for half a 16th note.

Modulation Routing

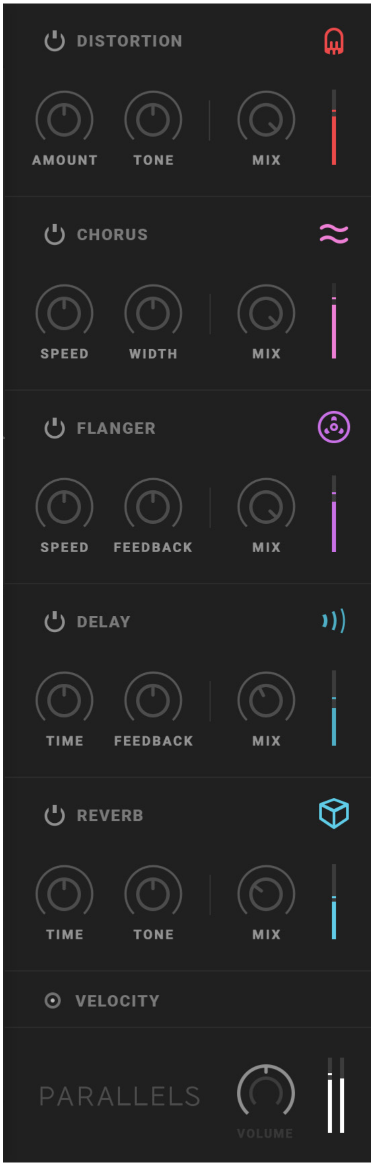
In Parallels, nearly all parameters in Parallels can be modulated from the modulation sources available in the Mod Pod. Click on the outer ring, modulation ring, on a knob to bring up the available modulation sources (A/B/C/D) and the mix balance between them. By clicking on the modulation ring while dragging your mouse/mousepad upwards you will add modulation amount to that destination. The color ring around the destination knob will light up with a color reflecting the modulation amount and mix of the sources chosen to modulate this knob. A 50/50 mix of two sources will result in a white color. There's is also a small indicator on the ring showing the presently applied sum of modulation.

Modulation routed in Parallels is only positive with one exception: Modulation of the to pitch is bipolar, which means that a Source waveform will stay in pitch when modulated with a LFO. This also means that when modulating pitch from the modulation sequencer at full range, nominal pitch will be at 50% on a sequencer step giving the sequencer a modulating pitch range of a total of two octaves.

Note: A few of the parameters cannot be modulated and those are primarily sliders: Color Attack, Color Decay, Color Amount and Octave. Also, oneshot and vibrato on/off options cannot be modulated from the Mod Pod. Change of modulation routings is only reflected in the next triggered note.

The Effect Section

The effects signal flow in Parallels are setup in series, from top to bottom. Each effect has its own dry/wet mix and bypass switch.



Distortion

This effect saturates the signal with a vacuum tube-like flavor.

Distortion Amount (0-100%)	The amount of distortion on the effected signal.
Distortion Tone (0-100%)	Dark or Bright distortion, you decide.
Distortion Mix (0-100%)	This is the dry/wet mix between the dry signal and the effected (distorted) signal. Blend to your preferred ratio.

Chorus

This effect emulates the behavior of a classic stereo-chorus, often found in polyphonic synthesizers of the late 1970s and early 80s.

Chorus Speed (0-100%)	The internal chorus speed.
Chorus Width (0-100%)	The stereo width and amount of the chorus.
Chorus Mix (0-100%)	This is the dry/wet mix between the dry signal and the effected (chorus) signal. Blend to your preferred ratio.

Flanger

A flanger is effect a time-dilation effect where parts of the signal is slightly delayed, feedback and mixed with a constant change of he delayed timed. The results is not so different from a phaser but can still be distinguished as it creates an unlimited series of equally spaced notches and peaks, both harsh and sweet sounding at the same time.

Flanger Speed (0-100%)	This is the flanger internal sweep speed (approximate range 0.5 – 5Hz).
Flanger Feedback (0-100%)	This parameter controls the feedback path of the flanger, but also the stereo width of the sweep; i e the more feedback, the wider flange.
Flanger Mix (0-100%)	This is the dry/wet mix between the dry signal and the effected (flanged) signal. Blend to your preferred ratio.

Delay

This delay is an echo effect quite similar to the Filter echo of Softube Heartbeat. It features both a set delay-speed on the first half on the speed knob. But also a variable delay-speed set as division of a beat synced to your DAW, on the second half on the same knob.

Time (1-900ms, 1/16, 1/32, 1/16, 1/16+, 1/8T-, 1/8T, 1/16D, 1/8, 1/8+, 1/4T-, 1/4T, 1/8D, 1/4, 1/2T, 1/4D, 1/2)	This parameter determines the delay time of the echo; i e how long time passes between each delay “hit”. On the left half of the Time knob, the range is from 1 to 1000 milliseconds. On the right side, the delay time can be set in divisions of the DAW tempo, ranging from 1/64th to 1/2 of a beat. The latter is useful for setting the delay to act in time with your song.
Feedback (0-100%)	Is set the amount of feedback, how many delay repeats there will be. It ranges from one repetition to roughly 10 repetitions at full feedback.
Delay Mix (0-100%)	This is the dry/wet mix between the dry signal and the effected (delayed) signal. Blend to your preferred ratio.

Reverb

The Reverb in Parallels is Softube custom reverb algorithm tailored to suit the needs of a synth sound. It creates everything from a short slap-back room to a vast synthesized textured space. Tweak it according to your needs.

Time (0-100%)	This parameter changes many things at once about the reverb. Low values makes the pre-delay more apparent, with less feedback and shorter times in the feedback loop. 20-40% creates a small space and with values higher than 50% a very large space is created by the reverb.
Tone (0-100%)	This parameter sets the filtering of the feedback networks in the reverb. While 50% is a flat filter topology, lower values below 50% will dampen higher frequencies in the reverb (dark reverb). Values above 50% will dampen lower frequencies in the reverb (brighter reverb).
Reverb Mix (0-100%)	This is the dry/wet mix between the dry signal and the effected (reverb) signal. Blend to your preferred ratio.

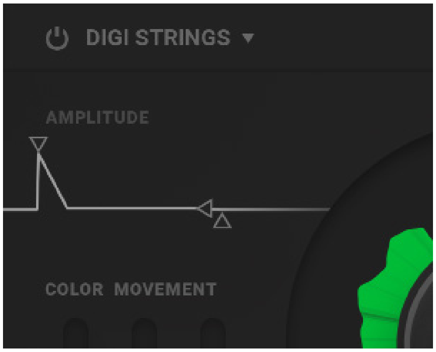
Setup

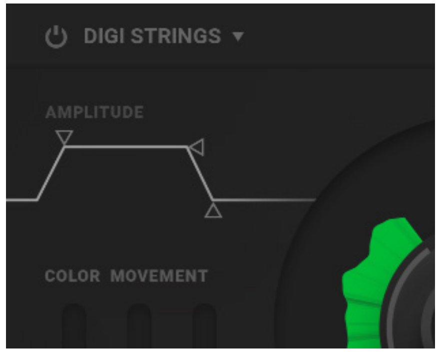
By clicking on the Setup window tab in the lower right corner of Parallelss widow, a screen of global options will be displayed. Many of the options requires shutting down and opening Parallels again (good to know). The setup options are as follows:

Warn when deleting presets from preset collection	Turns off or on the dialogue appearing when deleting user presets in the preset collection. Factory presets cannot be deleted, although they can be filtered out (not shown).
Warn when overwriting presets in preset collection	This option turns off or on the dialogue appearing when overwriting user presets in the preset collection.
Use Open GL graphics	Turns off or on Open GL graphics acceleration. This is an option that affects all Softube Plugins and if selected, a DAW restart is required before change of this option is active.
Color Blind Mode	This option enables or disables colors specially adapted for the colorblind. Close and open the Parallels GUI for the change to have effect.

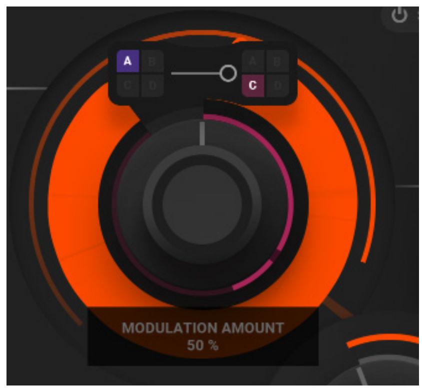
Show Colors in Menu	This option enables or disables colors in the wave menu reflecting the colors of the waveforms already before they are selected. Close and open GUI again for the change to have effect.
Always use smaller GUI	This option forces Parallels's rescaler to always open the plugin in a smaller window. Close and open the Parallels GUI for the change to have effect.
Show tooltips	Turns on or off the tooltips (screen text overlay for parameters). Close and open the Parallels GUI for the change to have effect.
Show value display	Turns on or off the value display for edited parameters in the lower left corner. Close and open the Parallels GUI for the change to have effect.

Tips and tricks

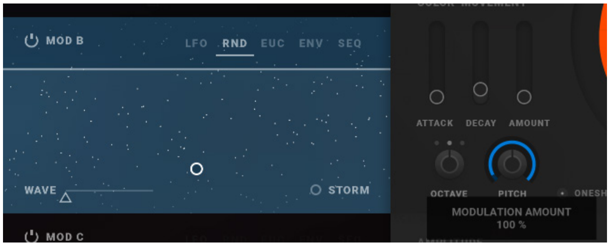




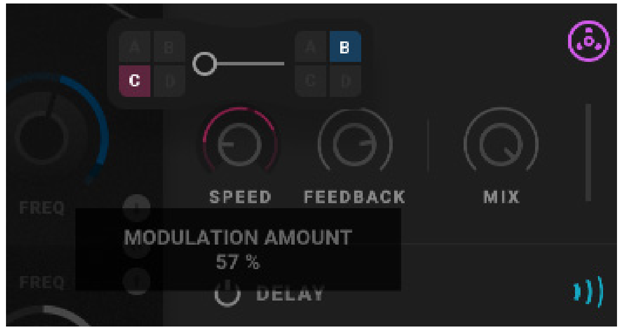
Use the Amp Envelope to change the dynamic character of any sound. Pictured above is the difference on the Amp Envelope between a plucked string sounds and slow, bowed stringed sound.



Most of the different Source waveforms have change of harmonic content when sweeping the Color knob clockwise. Make use of these harmonic changes either by using the dedicated Color Movement Envelope or by assigning a modulation slot to create a harmonic change while playing. And of course you can also do both at once. Pictured above is how a LFO in Mod Pod C is assigned to sweep the lowpass filter in the "Ana Saw LP" Source Waveform.

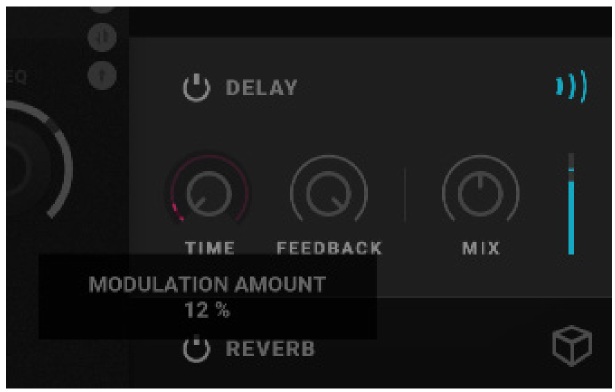


Use the RND to create random octave jumps with the RND Wave parameter set at 0. Use RND modulation on full 100% amount to make the RND pulses to create moderatly slow random octave jump.



You can modulate the internal LFO speed of the flanger with a modulation LFO by assigning a Mod Pod containing the LFO to the LFO speed knob of the Flanger (as pictured above).

Total polyphony is shared between upper and lower layer. Double your polyphony by turning off one layer if you do not need it.



Create your own unusual flange effect by assigning a modulation LFO to speed knob of the Delay effect (as pictured).

Create a tremolo by using a modulation LFO to modulate mix with one Source turned off.

Another way of creating a tremolo effect is by setting a late start in the Source Wave and assigning a modulation LFO to modulate playback.

Parallels For Modular

When you purchase Softube Parallels, you're also purchasing the possibility to use it as five separate blocks in our modular platform Softube Modular!

The four different modules for you to use in **Softube Modular: Parallels Source, Parallels Shape, Parallels Mod, Parallels Env** and **Parallels FX**. Here's a brief description of them:

Parallels Source This is the Source engine from one half of Parallels Lead, available as stand-alone module. It has a built in amp and color envelopes, vibrato and one shot mode. It can be played polyphonically within modular over midi and/or monophonically at the same time via the gate and note jacks. All envelopes stages, pitch parameters and color parameters features CV control via the external jacks.

Parallels Shape This is the Shaper section from Parallels, available as stand-alone module. It hasthree built in modes (LPG, SVF and RES). All controls are CV-controllable via external CV control.

Parallels Mod This is the Modulation section from Parallels. It features four different modulation modes that all can be externally controlled in different ways. The random source can be externally clock, as does also the euclidian and step sequencers. Sequencers have extensive CV control over first, last step and output slew. When not externally clocked the sequencers is automatically locked to the DAW tempo at all times.

Parallels Env This is the Modulation Env from Parallels, available as stand-alone module. It is externally triggered by a gate over 1.33v and features CV-control over each phase at any time. When Decay is activated it shifts from ASR to ADSR type.

Parallels FX This is the Effect engine for Parallels, available as stand-alone module. It has five modes that can be run one at a time – Distortion, Chorus, Flanger, Delay and Reverb. Three parameters in each mode has external CV control.

Credits

Johan Antoni – Initial concept, multi sample sound-recording and library, testing and presets. **Kristofer Ulfves** – Project lead, sound design, presets, user manual. **Björn Rödseth** – Tech lead, programming. **Kim Larsson** – DSP modeling, programming. **Erik Sight** – programming. **Patrik Holmström** - programming. **Jacopo Lovatello** – programming. **Filip Thunström** – programming, preset conversion. **Tord Jansson** – programming and GUI engine customization. **Oscar Öberg** – Mentoring, programming. **Arvid Rosén** – Mentoring, programming. **Manuel Colomb** – GUI design. **Joe Lawton** – Testing, presets. **Henrik Johansson** – Testing, presets. **Fredrik Mjelle** – Testing, presets. **Christoffer Berg** – Testing, presets. **Maxus Widarsson** – Deep testing, Qualification. **Johan Bremin** – Deep testing, Qualification. **Sven Bornemark** – Testing. **Fanny Hökars** – User Manual layout.

Appendix

Waveforms list

Cathegory	Index no	Name	Description clockwise harmonic change
Digital Synth	001	Digi Guitar	Synth-guitar sound
	002	Digi Noisy Seagulls	a slowly swelling synthetic sea-gull sound
	003	Digi Strings	A somewhat brassy synth-string sound
	004	Digi Meta 1	Digital wave that propagates from a soft to a harsh sound
	005	Digi Meta 2	Digital wave that evolves from a soft pad into a harsh piercing sound.
	006	Digi Meta 3	Digital chunky chord sound that evolves into a huge space.
	007	Digi Meta 4	Digital guitar into added harmonic content.
	008	Digi Meta 5	Digital plucked string sound extended into a held tone warped into distortion.
	009	Digi Meta 6	Digital percussive looped sound into warped distortion
	010	Digi Meta 7	Digital guitar into warped distortion.
	011	Digi Superwave	Static sawtooth into a detuned swarm of supersaw.
	012	Digi Table 1	Swept digital wavetable with different harmonic content.
	013	Digi Table 2	Another swept digital wavetable with different harmonic content.
	014	Digi Table 3	Swept digital wavetable from even to odd harmonics.
	015	Digi Table 4	Digital additive synthesizer style wavetable.
	016	Digi Table 5	Swept formant wavetable.
	017	Digi BP Amped	Digital distorted triangle wave through bandpass.
	018	Digi LP Decimated	Digital distorted lowpassed triangle wave.
	019	Digi LP Dist Overload	Another digital distorted lowpassed triangle wave.
	020	Digi Tri Meta 1	Digital choir into distortion.
	021	Digi Tri Meta 2	Digital female choir into distortion.
	022	Digi Tri Meta 3	Digital voice into distortion.
	023	Digi Wave	Digital wave with filter sweep.
Analog Synth	024	Ana Hoover	Classic hoover sound from the analogue classic.
	025	Ana Sweep	Filter LP sweep sound from an analogue poly classic.
	026	Ana Saw BP	Sawtooth bandpass-sweep from a vintage analogue synth.
	027	Ana Saw HP	Sawtooth highpass-sweep from a vintage analogue synth.
	028	Ana Saw LP	Sawtooth lowpass-sweep from a vintage analogue synth.
	029	Ana Squ BP	Squarewave bandpass-sweep from a vintage analogue synth.
	030	Ana Squ HP	Squarewave highpass-sweep from a vintage analogue synth.
	031	Ana Squ LP	Squarewave lowpass-sweep from a vintage analogue synth.
	032	Ana Tri BP	Trianglewave basspass-sweep from a vintage analogue synth.
	033	Ana Tri Hp	Trianglewave highpass-sweep from a vintage analogue synth.

	034	Ana Tri LP	Trianglewave lowpass-sweep from a vintage analogue synth.
	035	Vintage Saw	Classic vintage sawtooth sound through lowpass sweep.
	036	Vintage Saw Feedback	Saturated classic vintage sawtooth sound through lowpass sweep.
	037	Vintage Saw Res	Classic vintage sawtooth sound through resonant lowpass sweep.
	038	Vintage Squ	Classic vintage squarewave sound through lowpass sweep.
	039	Vintage Squ Res	Classic vintage squarewave sound through resonant lowpass sweep.
	040	Vintage Tri	Classic vintage trianglewave sound through lowpass sweep.
	041	Vintage Tri Feedback	Saturated vintage trianglewave sound through lowpass sweep.
	042	Vintage Saw LP Amped	Amped vintage sawtooth sound through resonant lowpass sweep.
Combo	043	Combo 1	Combination of waveforms with an increased harmonic complexity.
	044	Combo 2	Combination of waveforms with an increased harmonic complexity.
	045	Combo 3	Combination of waveforms with various pulse wave content.
	046	Combo 4	Combination of squarewaves with different tuning.
	047	Combo 5	Combination of waveforms morphing into each other.
Chords	048	Chords 1	A minor 3rd interal tuned up to a 5th
	049	Chords 2	Unison tuned up to a minor 3rd
	050	Chords 3	Minor 3rd with increased FM modulation.
	051	Chords 4	Minor 3rd stab with increased assymetric triggering.
	052	Chords 5	Minor 3rd stab with increased assymetric triggering.
Drones	053	Drone Combo 1	Atonal drone with increased FM amount.
	054	Drone Combo 2	Atonal feedback sound.
	055	Drone Combo 3	Atonal cyclic silent drone fade up.
FM	056	FM Mallet	Classic icy FM Mallet sound.
	057	FM 1	4 operator FM sweep with feedback.
	058	FM 2	Even broader 4 operator FM sweep with feedback.
	059	FM 3	4 operator FM sweep, different pitch relations
	060	FM 4	Dubstep style FM sweep
	061	FM 5	Harsh FM sweep
	062	FM 6	Broader and harsher FM sweep
Physical Mod	063	Phys Mod Hybrid	Woodwind pulse, inverse amplitude sweep
	064	Phys Mod 1	Hollow body wave
	065	Phys Mod 2	Bowed string wave
	066	Phys Mod 3	Modal string, inverse amplitude sweep
	067	Phys Mod 4	Rubber into odd harmonics sweep

	068	Phys Mod 5	Rubber into FM sweep
Stacked	069	Stack 1	Oscillator stack with pitch variations in the beginning and end
	070	Stack 2	Sine stack through waveshaper
	071	Stack 3	Bounce stack with increased rate
	072	Stack 4	Pulse stack
	073	Stack 5	Square to Triangle shape stack
	074	Stack 6	Triangle to pulse shape stack
	075	Stack 7	Sinusoid sub stack to FM transformation
	076	Unison 1	Unison squares with added pitch modulation
	077	Unison 2	Unison tilt modulated saw/triangle
	078	Unison 3	Unison pulse with added pitch modulation
	079	Unison 4	Unison square-sine with added pitch modulation
Distorted	080	Thrashy 1	8-bit style pulse trash wave
	081	Thrashy 2	8-bit style pulse trash wave
	082	Thrashy 3	Lo-fii FM with sync portion
Environment	083	Ebisu 1	Enviromental sounds from Ebisu area blended with sines
	084	Ebisu 2	Enviromental sounds from Ebisu area
	085	Ebisu 3	Gong from Ebisu with freezed looping
	086	Ebisu 4	Power station in Ebisu
Chaos	087	Chaos 1	Chaotic analogue synth sweep
	088	Chaos 2	Chaotic analogue synth sweep
	089	Chaos 3	Particle synth sweep
	090	Chaos 4	Sine noise decay
	091	Chaos Phys Mod 1	Chaotic physical modelling synth sweep
	092	Chaos Phys Mod 2	Particle string into square
	093	Chaos Phys Mod 3	Irregular string model
	094	Chaos Phys Mod 4	Particle string into pad
	095	Chaos Phys Mod 5	Atonal string, inverse amplitude sweep
	096	Vintage Chaos 1	Westcoast Chaos
	097	Vintage Chaos 2	Westcoast Chaos

7 Statement Lead

Statement Lead is the next product in the series that started with Monoment Bass. The Monoment Bass user will feel right at home with Statement Lead, although some parts are different:

First, this is obviously a lead instrument - it's polyphonic and covers 5 octaves. It also features quite different source waveforms from Monoment.

Statement also features a slightly different set of Softube effects: Drive, Reverb, Delay, a one-knob Multiband compressor and Spatialization.

Statement, like it's brother Monoment, is a no-brainer, always-sound-good, easy-to-use polyphonic lead machine that creates complex sounds but feels analog.

Due to its polyphonic nature, we have omitted the glide feature and instead added an auto-glide

feature. The idea is that you can trigger a glide/pitch shift at the onset of the note/chord, or at the end of it. You can select how you want to trigger it, for instance if Velocity is above a certain threshold, if Aftertouch reaches a certain level, or maybe every time a note is played or released.

Statement has the same simplified control set of the envelopes as Monoment - Punch/Release instead of a full ADSR, "Env Type" instead of a full ADSR on the filter, simplified controls in the effects section.

User Interface

The user interface consists of four parts:

1. Source section, where you select different sound sources
2. Filter and Sequence section, where you set the cut-off frequency of the filter, or use the dedicated filter sequencer to automatically change filter cut-off or amp.
3. Tone, where you balance your sources and set the punchiness of the sound.
4. Effects, the place for distortion, reverb, delay, compression and spatialization.

In Statement, we have again created a synth workflow with filters, envelopes and modulation, all easily accessible and adjustable, for you to shape the sources into the tone you need right now. And to compliment the textures of the sources, the effects section with reverb, delay, spatialization, distortion and multi-band compressor is there to help you; all that you need to make your awesome synthesizer sound take its desired position in the mix. Again, it's about sound quality and workflow. A modern synth-lead sound in no time.

Source section

The Source section is the heart of Statement. Each sound is build up by a mix of one or two waveforms. Adding to that is a analog dirt section with looped or transient material to enhance your sound. The two sources are always sent to the filter section while the dirt has the option to be added on the side if needed.

The Source Material for Statement

The extensive Source material for Statement uses meticulously recorded and processed material from rare and expensive synthesizers such as Schmidt Synthesizer, Synclavier, Oberheim Four Voice and Black Corporation Deckard's Dream. The soundmaterial in all consists of 90 waveforms in 11 different categories:

Analog Sync – Oscillator hard- and soft-sync type of waveforms.

Analog Clean – fairly basic analogue sound waveforms.

Analog Dirty – analogue waveforms with a bit more grit and distortion.

Digital Rich – digital rich waveforms with plenty of harmonic content

Digital Noisy – digital waveforms with a bit more grit and lo-fi sound

Analog Rich - analogue rich waveforms with plenty of harmonic content

Digital Dirty - digital waveforms with a bit more grit and distortion.

Digital Percussive - digital waveforms with a defined attack.

Digital Voiced - digital waveforms with different kinds of formant character.

Digital Clean - fairly basic FM and additive waveforms.

Analog Percussive - analogue waveforms with a defined attack.

Parameters

Source A This is where you select your waveform for Source A out of the 11 different Source categories. Click in the display to activate the drop-down menu for an overview or use the back and forward arrows to go through the different waveforms one by one.

Source B This works similar to Source A selection but for second source waveform.

Source B Octave This is the relative transposition of the Source B section. It can be set to be one octave below, at the same octave or one octave above the pitch of Source A.

Source B Detune The detuning of Source B against the pitch of Source A. It ranges from minus seven semitones to a plus seven semitones.

Analog Dirt This is where you will select your dirt layer sound out of 20 (15 looping and 5 transient attack) different waveforms. Remember to turn up the dirt level on the tone page in order to properly hear this feature.

Dirt to Filter / to Output This switch will determine if the Analog Dirt sound is mixed with the rest of the sources (A and B) and sent into the filter, or if it is sent directly dry to the output section.

The Filter and Sequencer section

To the right on the Source section, is the Filter-section with its built-in modulation sequencer. At center is the main filter display window that shows filter interactive filter characteristics as well as the modulation sequencer with its controls. To the right of the filter display are the controls that are directly related to the filter and its built-in envelope. The envelope determines how the filter changes when you hit a note. As we wanted to make that as easy as possible for you we boiled it down to four main controls: Filter cutoff, Resonance, Envelope Type and Envelope amount.

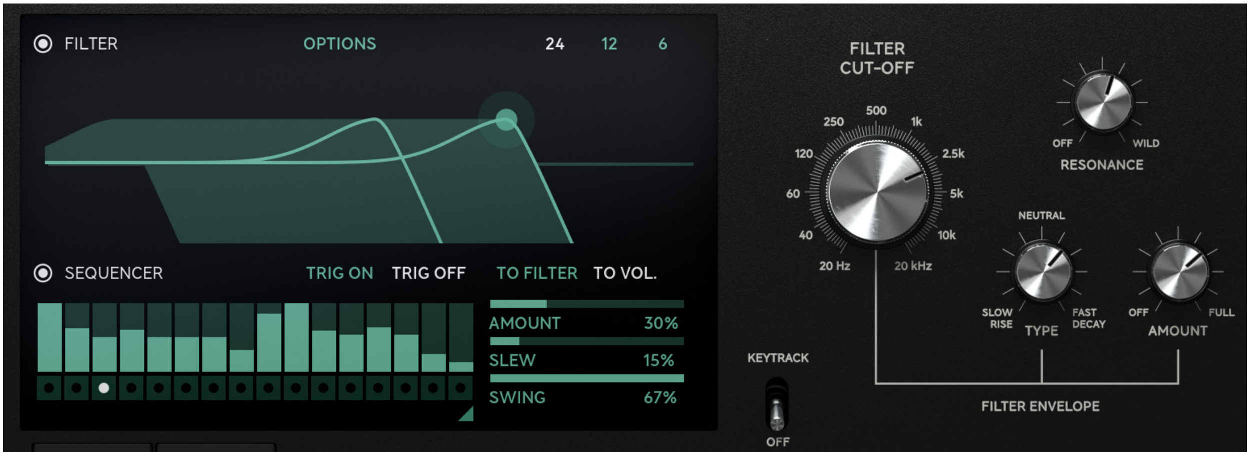
Cut Off This knob set the cutoff frequency of the filter in Statement. The cutoff frequencey determines the roll-off point of the lowpass filter after which high frequencies are diminished. The steepness of this roll-off is dependent on the Filter type (see below).

Resonance This knob controls the pronounced feedback around the cutoff frequency in the filter. The more resonance, the more "hollow" or "singey" the filter gets.

Env Type This knob defines the envelope type and timing. From a very slow attack with the knob set at fully counter clock-wise, to a very fast attack at the knob at fully clock-wise. And of course all the interpolating times in between. So if you, for example, would want a medium decaying filter envelope then set the envelope type knob at 2 o'clock.

Env Amount The envelope amount knob defines how much the envelope will affect your filtgers cutoff frequency. This is value is reflected in the filter graph as a "shadowed area" showing the end-destination level of the cutoff-frequency when applied this amount. While playing, you will also clearly see how it is changing in the filter graph.

Key Track This switch enables full keyboard tracking to the filter cutoff. This means that the filter tracks and opens the filter cutoff in relation to the notes that is played.



Filter Type In the top right corner of the filter graph window, you can change between the three different filter-types in Statement: 24 dB/octave, 12 dB/octave and 6 dB/octave – essentially three lowpass filter typer with different roll-off curves with 6 dB/octave being the most “gentle” and 24 dB/octave being the most “aggressive”.

Filter On Off This is a switch in the upper left corner of the filter graph window to swiftly bypass the filter for instantly listening to the Source waveforms without the coloration of the filter. When turned off, the Filter graph window is blacked out.

The Modulation Sequencer

The built-in modulation sequencer is another way of automatically change the filter frequency or output volume, either stepped or slewed. The sequencer is always running with its tempo synchronized to the DAW clock so you'll always can get a rhythmic change in time with your music.

Seq On/Off This small button at the upper left of the Sequencer window enables or disables the sequencer. When turned off, the sequencer window is blacked out.

Seq levels/trigs The 16 steps in the modulation sequencer can be set to be either levels or distinctive triggers (on/off). Just click on a level drag with your mouse to adjust it. In trig mode, click repeatedly on a step to toggle it on or off. Only one mode can be used at a time.

Seq Amount This will determine the effect the modulation sequencer will have on its destination (filter cutoff or output volume, see below). If the filter is turned off (filter graph is black) then the sequencer will not have any effect.

Seq Slew This parameter will sets the slewing of the output sequencer data applied to the filter or output volume. More slew equal smoother sound. Great when using the sequencer as more of an programmed, synced LFO.

Swing This parameter will change the timing of the even steps (2,4,6,8,10,12,14 and 16) to make it swing more towards triplets. Range is from 33% to 67% where 50% represents normal, straight timing.

Seq to filter/to Vol This double click area determines whether the sequencer controls to the cutoff frequency of the filter or the output volume. Experimentation with this one is fun – a sequence that modulates the filter makes a totally different sound when modulating the output volume!

Seq Trig On Off This switch toggles between normal sequencer level mode (Trig: off) and sequencer trigger mode (Trig: on).

Seq Range This sets the loop-length of the sequencer. It can be set as low as 1 (no loop) and as high as 16 (all 16 steps).

The Tone section

The Tone section is where you sculpt the tone: here you can mix the sources and choose whether you prefer an overall sharp attack or a slow fade-out for your sound. The main "playability" of the sound is adjusted here with just a few powerful controls. Here's an overview.

Next step is to adjust the shape, the envelope, of the tone. That's easily done by adjusting the Punch control. Increase Punch and you'll get a snappier, punchier sound. Decrease it and you'll get a sound with a slow build-up. The Release sets the time it takes for the sound to fade-out.

Parameters

Dirt Level This parameter will set the dirt volume level from non-existant to pretty subtle - but yet again, the dirt in Statement is meant to be a sonic spice not the source of a sound.

Source Mix This knob sets the balance between Source A and B before they enter the filter.

Mono Unison This three stage switch changes Statement between normal polyphonic mode and two different mono unison modes. Unison mode 2 is slighly more pronounced than M1. In unison performance is monophonic, meaning that one note at a time is heard. Last note played is prioritized.

Velocity This switch toggles Velocity sensitivity on or off. The level of sensitivity can be changed in the options menu (see further description below).

Punch This parameter sets the overall amplitude envelope of the sound in Statement. Increase Punch and you'll get a snappier, punchier sound. Decrease it and you'll get a sound with a slow build-up.

Release This parameter sets the time it takes for the body of the sound to fade-out. Note that long reverb and delay can prolong the sound further (see section below).

Auto-glide

The special auto-glide feature of Statement Lead means that you can start or end each note with an automatic pitch-bend up or down with a programmable time and range. There are also some conditional features such a threshold for setting when the bend should and should not occur. Let's go through them all:

Auto Glide Start This parameter determines whether the glide will occur in the start of the sound. When set to the velocity option, the threshold-level will determine at which velocity the auto-glide is triggered.

Auto Glide End This parameter determines whether the glide will occur at the end of the played sound. When set to the velocity option, the threshold-level will determine at which velocity the autobend is triggered. When set to the aftertouch option, any aftertouch above the threshold occuring while note is playing will trigger the auto-glide at the end of the sound (when the key is released).

Auto Glide Pitch Shift This determines where the glide will start or end. Range goes from minus two octaves to plus two octaves.

Auto Glide Threshold This threshold determines the trigger-point for the auto-glide when used in conjunction with velocity and/or aftertouch (see description above).

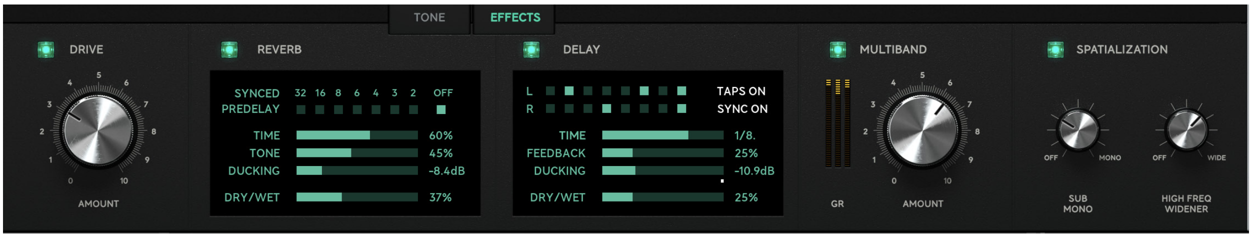
Auto Glide Speed This is the auto-glide speed rate. It goes from off, fast to fairly slow.

Aging Sets the amount of unpredictability in Statement. More Aging means less stable pitch, more unpredict-able attacks, more dirt and grit and more analog goodness.

Volume This parameter sets the overall output volume

The Effects section

The effects section consists of five different effects that you can use to further sculpt your lead sound. These effects have been carefully chosen and designed for polyphonic synthesizer-lead sounds, and the control set has been thoroughly tweaked to give you as much control as possible with as few knobs as possible.



Parameters

Drive On Off This switch enables or disables the Drive effect. This transformer drive is quite subtle but gives your sound the warm edge it needs.

Drive Amount Sets the amount of drive coloration.

Reverb On Off This switch enables or disables the Reverb effect simulation. This is an algorithmic reverb with a pre-delay.

Reverb Pre-delay This is the pre-delay that always is synced to the DAW tempo. The pre-delay is set to be a subdivision of the tempo and ranges from 32th notes at min to half notes ("2") through 16th, 8th, 8 triplets ("6"), quarter notes and quarter note triplets. When set to "off", the pre-delay effect is disabled.

Reverb Time This parameter set the reverb time from very short to very long (near infinite). The decay time of the reverb depends on the reverb tone setting – a neutral setting (50%) will generate the longest reverb times.

Reverb Tone This parameter is the equalization of the reverb signal. Low settings of this parameter brings out more of the low end while high settings brings out the high end.

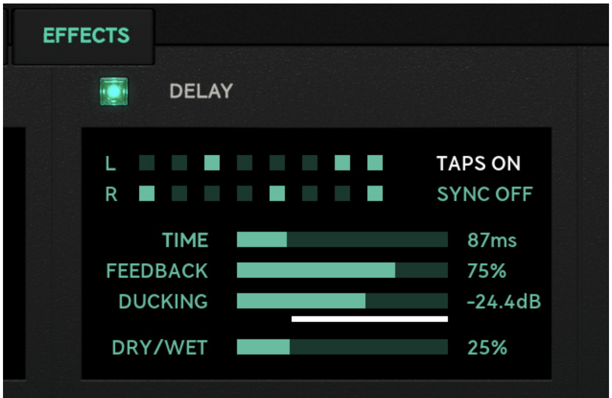
Reverb Ducking This feature enables the user to have create a sound where the decaying reverb from previously played notes is reduced in volume when a new note is played. This parameter will determine how much the reverb will "duck" against the new dry sound.

Reverb Dry Wet This parameter sets the dry/wet balance of the reverb.

Delay On Off This switch enables or disables the Delay effect. It creates musically interesting echos that can be defined time-based in milli-seconds or as divisions, synchronized of the DAW tempo.

Delay Taps On Off The delay effect can be used as a standard one tap delay with feedback (taps off) or as a multi-tap delay (taps on) where up to eight delay taps per channel, left and right, is defined (see below).

Delay Tap L 0 – L7, R0 - R7 These buttons turns the delay taps on and off. The taps farthest to the right (L7 and R7) is the longest and will represent the chosen time, while the ones to left of this are subdivisions down to 1/8 of the time at the very left end (L0 and R0). Delay taps are like playheads in a tape-recorder and will, when enabled, play back a perfect replica of the played sound delayed in time.



Delay Sync On Off This turn the DAW synchronization on or off.

Delay Time This parameter sets the Delay time. It ranges from very short (32ms) to long (2 seconds). How it is presented is dependent whether the Delay is synced to the DAW or not (see above). When sync is set to off, the delay time of the longest taps (L7 and R7) are set in milliseconds. When DAW sync is turned on, the delay time is defined as subdivisions of the DAW tempo.

Note that in "1/2-tempo" the odd taps will is affected by the swing parameter set in the modulation sequencer, this is deliberate and very musical.

Delay Feedback This sets the amount of feedbacked signal sent from the slowest delay taps (L7 and R7). The internal feedback is send through a low.

Delay Ducking This feature enables the user to have create a sound where the decaying delay signal from previously played notes is reduced in volume when a new note is played. This parameter will determine how much the delay will "duck" against the new dry sound.

Delay Dry Wet This parameter sets the dry/wet mix balance of the delay effect.

Comp On Off This switch enables or disables the one knob multiband compressor. Multiband makes everything fatter, clearer, better, more defined...use it on everything! With just a bit of compression on your synth-lead you'll get a warmer and fuller sound.

Comp Amount The sets the amount of compression. Overdo it and a character on its own gets through.

Spatialization On Off This turns the Spatialization feature on or off. While all sources and algorithms in Statement are stereo, you sometimes want to enhance the stereoness of them, or maybe reduce it. That's what the Spatialization effect does.

Sub Mono This monomaker sets all audio below the set frequency will be in mono. Pull it all the way up to force the overall output of Statement to be completely in mono.

High Freq Widener This stereo Widener let you enhance stereo differences and suppress mono behavior.

Options menu

Hidden in plain sight - top middle of the Filter graph window - is the options menu. In here you'll find global parameters not often changed. They are:

Pitch range This is the customizable pitch-bend range. This will determine how much the pitch of the Sources will bend up and down when a MIDI pitchbend message is received. You can configure the bend-range to be anything from zero to twelve semitones, quantized in semitone steps.

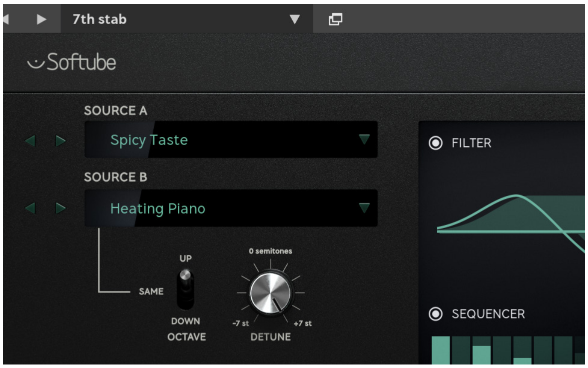
Mod Wheel Depth This will determine how much the internal modulation will affect pitch when applied through external MIDI modulation message (midi CC#1).

Mod Wheel Rate This sets the speed of the internal modulation oscillator, from a fairly slow pace at 0.3Hz to fairly quick pace at 30Hz.

Velocity Range This parameter set sensitivity to incoming MIDI velocity, where the minum value is totally oblivious to velocity information and at maximum setting it has full range (meaning that velocity received with a value of 0 will be totally silent).

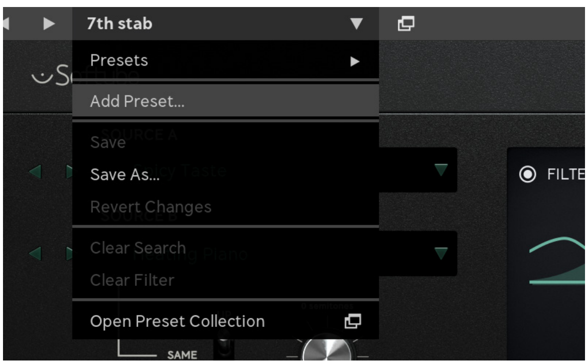
Saving and loading sounds

Saving and recalling your own or other sounds is very simple in Statement Lead. In the top left corner of the plugin window is the preset display window and just to the left of that are the two buttons for next and previous preset quick browsing.

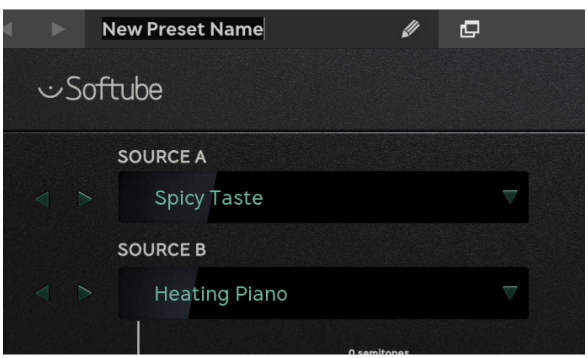


When clicking on these you will scroll back and forth in the list that is selected and filtered in the preset collection browser.

The preset collection browser window is quickly open by clicking on the window symbol to the right of the small preset name display. For details about Preset Collection see the separate chapter.



When you have created a really great sound that you want to save, simply click on the drop-down arrow just to the right of the preset display window and choose Add preset or Save as – you'll now be promted to give your preset a name (the same procedure can of course also be performed in the preset collection by itself) and press enter to save it.



Your preset has now been saved and can found among the user presets in Preset Collection. You can even share it with your friends by exporting it along with other presets as a separate ".softubepreset"-file. Details about this can be found in the Preset Collection chapter.

When just tweaking a sound you'll notice that the name will get an * addition after the name, indicating that this preset has been changed from its saved state. If you want to save those changes simply choose "save" in the preset dropdown menu.

Statement For Modular

When you purchase Statement Lead, you're also purchasing the possibility to use it as three separate blocks in our modular platform Softube Modular! The three different modules for you to use in Softube Modular: Statement Source, Statement Filter and Statement FX. Here's a brief description of them:

Statement Source	This is the Source engine from Statement Lead, available as stand-alone module. It has a built in amplitude envelope, waveform mixing, dirt level and aging. It can be played polyphonically within modular over midi and/or monophonically at the same time via the gate and note jacks. Dirt Level, Punch, Release and Aging parameters features CV control.
Statement Filter	This is the Filter section from Statement Lead, available as stand-alone module. It has a built in filter envelope and sequencer. The envelope is triggered through the external gate jack. All sequencer functions - Slew, Range, Amount and Swing parameters - features external CV control. The Sequencer is automatically locked to the DAW tempo at all times.
Statement FX	This is the Effect engine for Statement Lead, available as stand-alone module. It has five modes that can be run one at a time – Drive, Reverb, Delay, Multi comp and Spatialization. Up to three parameters in each mode has external CV control.

Credits

Tobias Menguser – Initial concept, multi sample sound recording, presets

Niklas Odelholm – Sound, visual and product design

Kristofer Ulfves – Project lead, presets, user manual.

Erik Sigth – Framework programming

Patrik Holmström – GUI programming

Filip Thunström – GUI programming

Jacopo Lovatello – DSP programming

Alexander Näs – DSP programming

Björn Rödseth – Programming, Mentoring

Arvid Rosén – Filter optimization

Maxus Widarsson – Quality assurance

Ulf Ekelöf – 3D graphics modeling

Klaus Baetz – programming

Tord Jansson – programming and GUI engine customization.

Henrik Johansson – Testing, presets.

Fredrik Mjelle – Testing, presets.

Patrick Detampel – presets

Erik Putrycz - presets

8 SOFTUBE AB'S END USER LICENSE AGREEMENT ("EULA").FOR PLUG-INS AND OTHER SOFTWARE PRODUCTS (ver 2019-06)

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However, upon the earliest of your (i) ACCEPTANCE OF THIS EULA, (ii) DOWNLOAD AND/OR FOLLOWING THE INSTALLATION OF THE PRODUCT AND/OR (iii) USE OF THE PRODUCT, you agree to be bound by the terms and conditions of this EULA together with the applicable conditions of the Terms.

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Brian Paul
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