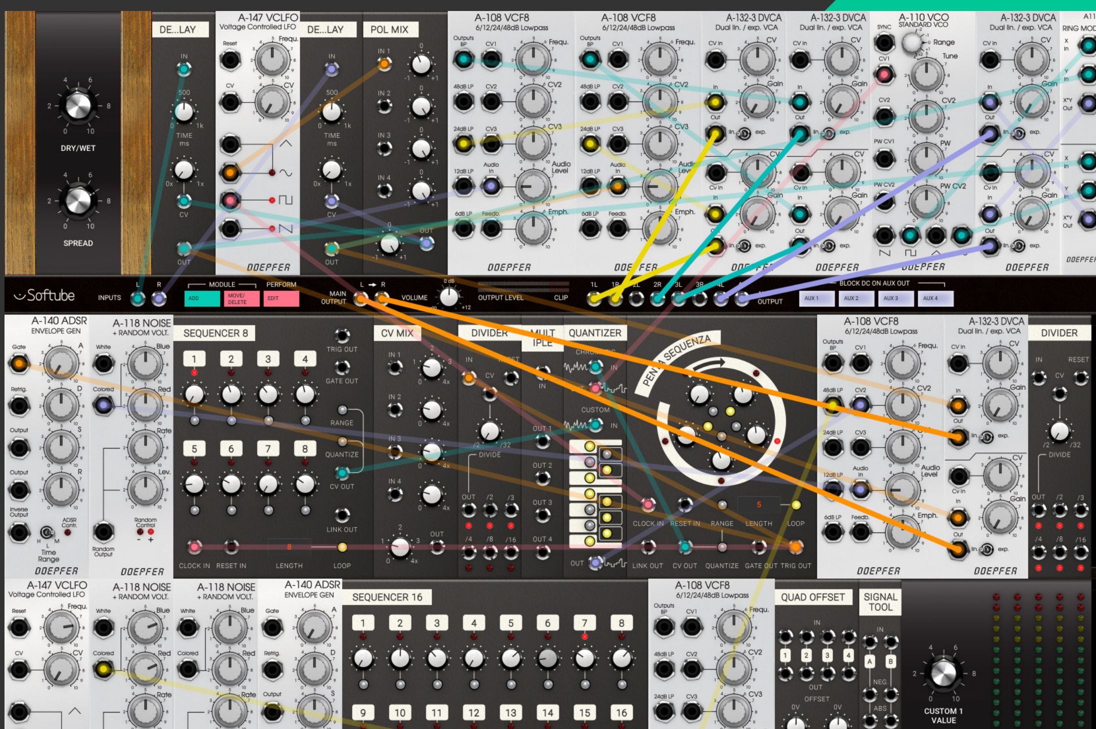
Softube

MODULAR

USER MANUAL

Supporting VST/VST3/AU/AAX Native

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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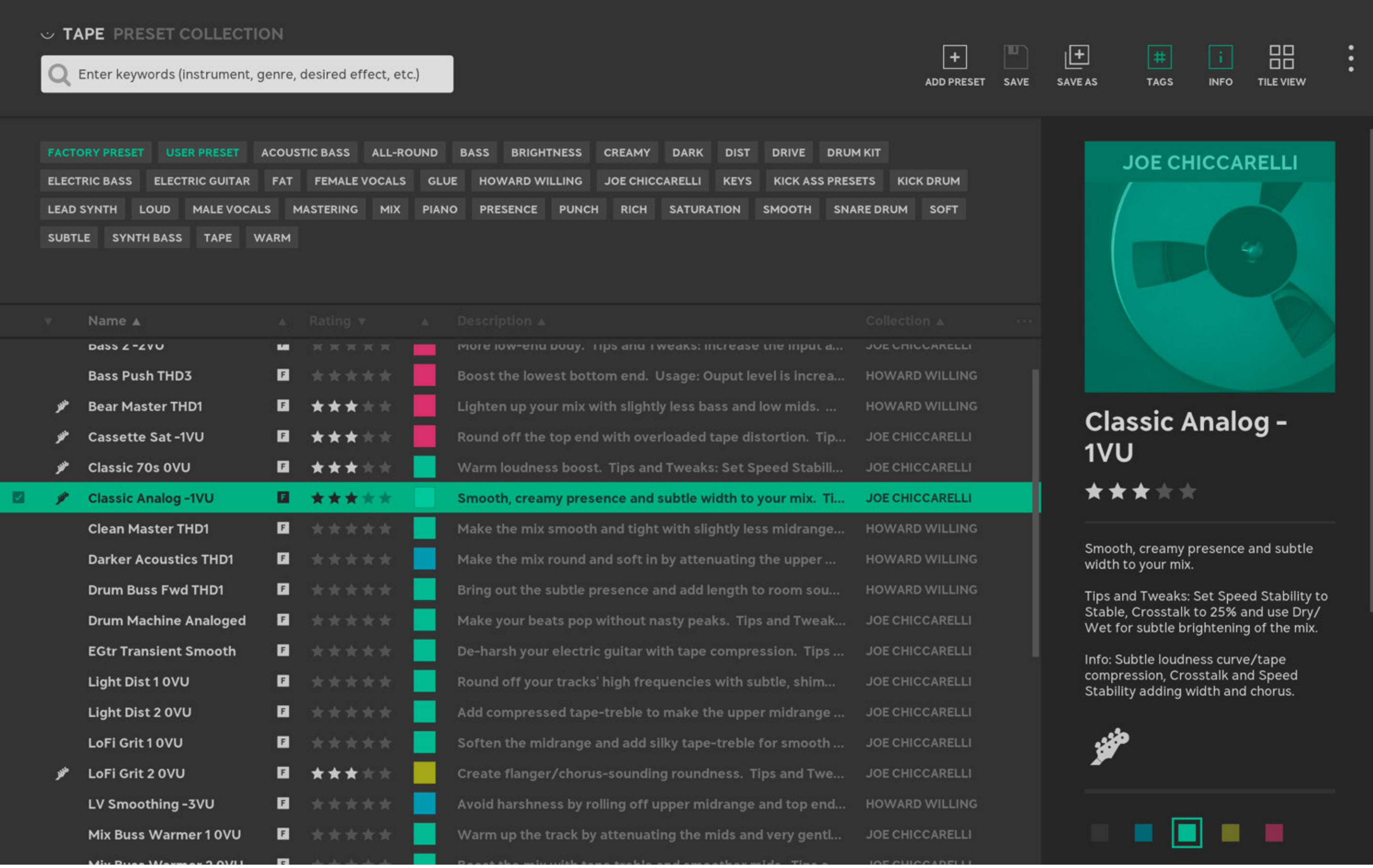


1 Preset Collection

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

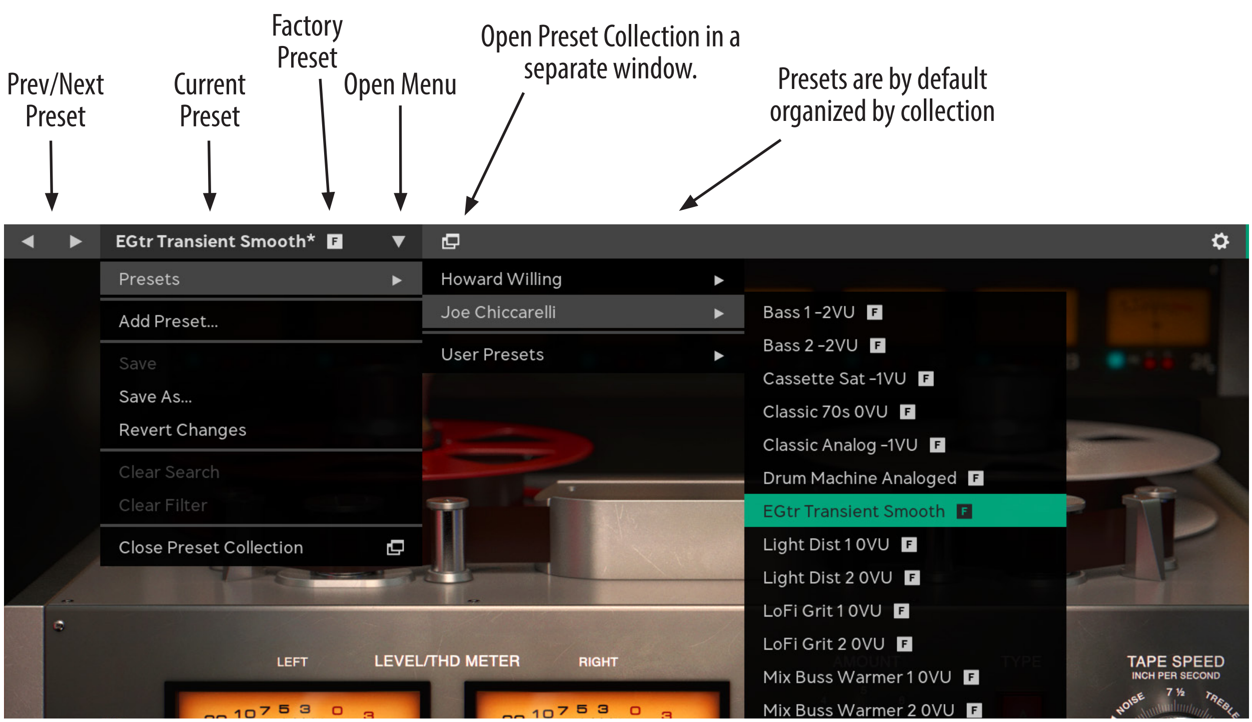
Press "ADD PRESET..." in the dropdown menu to save the current settings as a new preset. Type the name of the preset and press enter. If you've made changes to a current preset and wish to overwrite that preset, just press "SAVE". Press "SAVE AS..." if you want to save it with a new name.

Searching for Presets

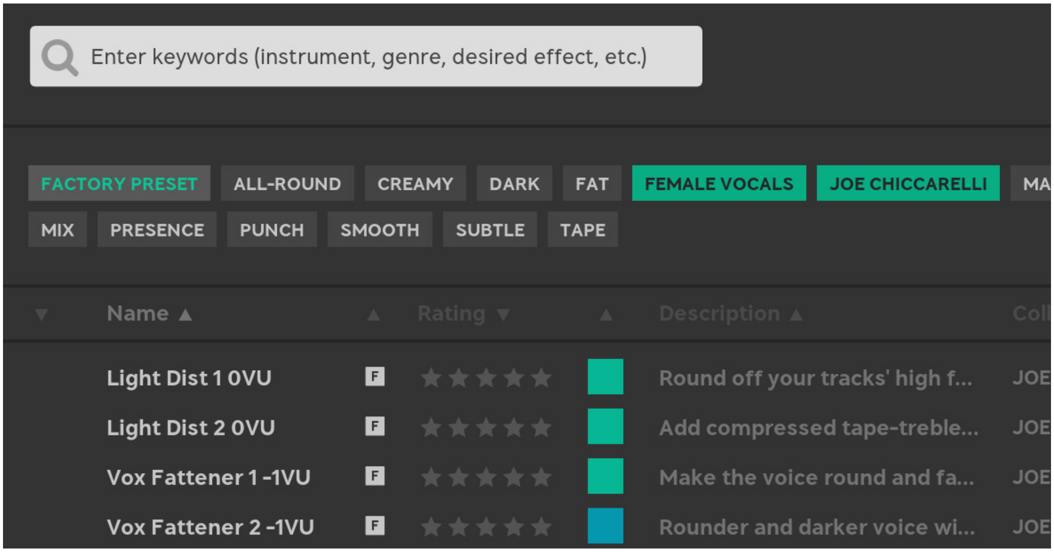
If you want to find a specific preset and have a lot of presets, there are many smart ways to search for preset, and all of them require that you open the Preset Collection by clicking the ⧉ button.

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

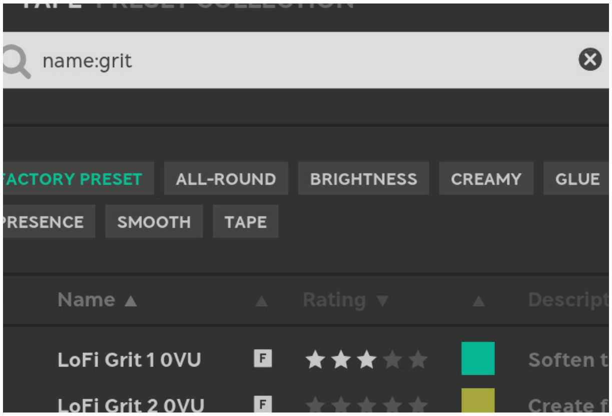
A search in the SEARCH BOX will search through all metadata (name, tags, description, etc). If you want to narrow your search, you can specify what you want to search for by using a qualifier, such as "name:" or "description:".

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

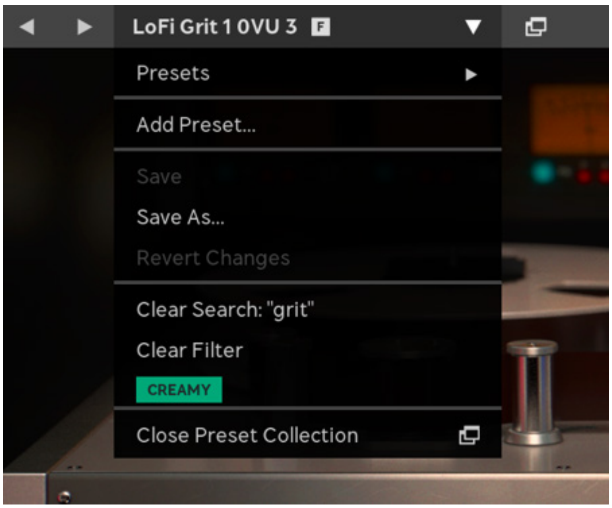
Search in the menu bar

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.



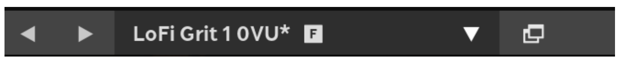
The menu shows the current search criteria, currently selected tags, and options to clear the filter and search criteria.

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.



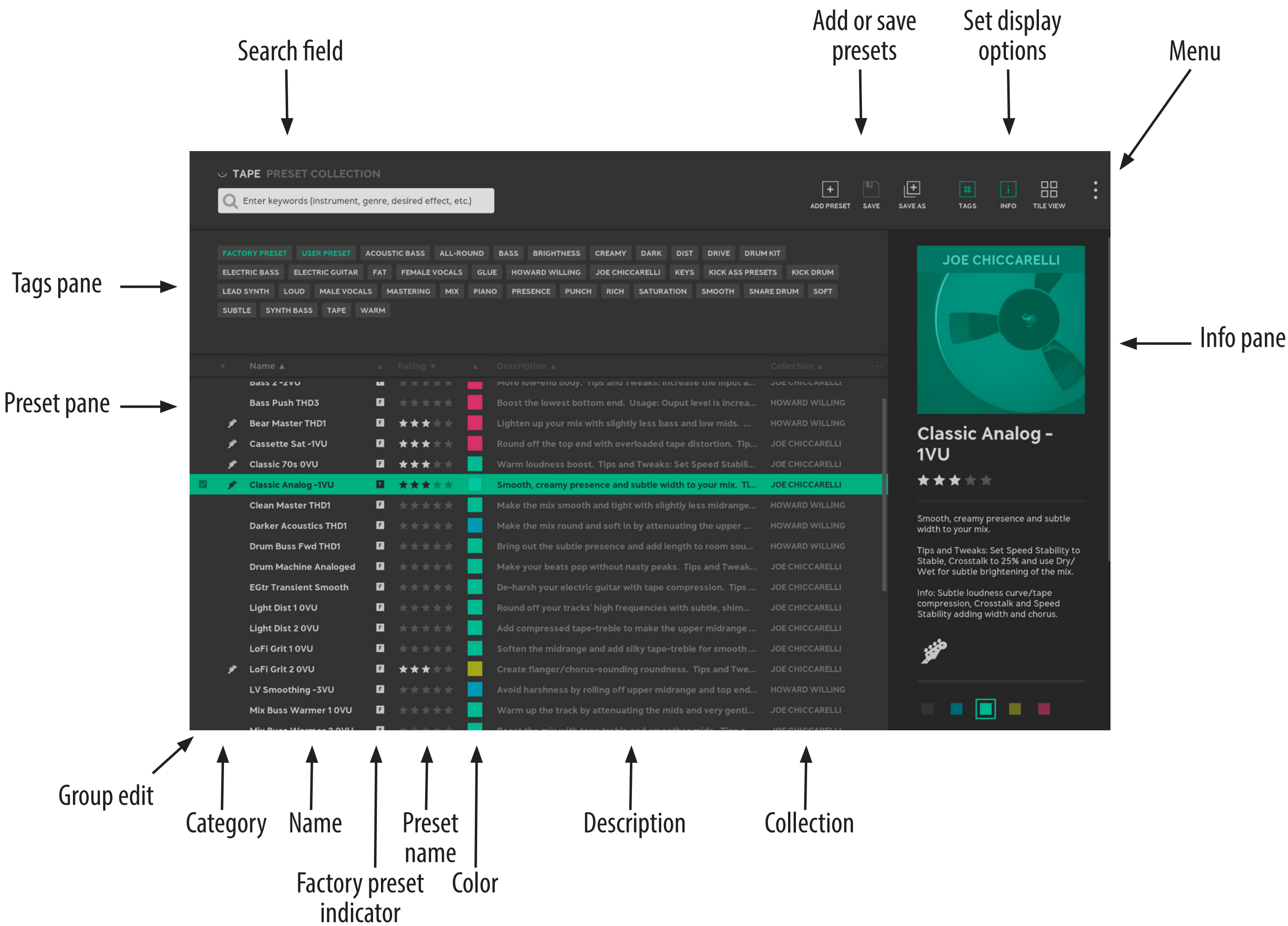
Asterisk (*) indicate that the settings of the plug-in is different from the saved preset.

> 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.*

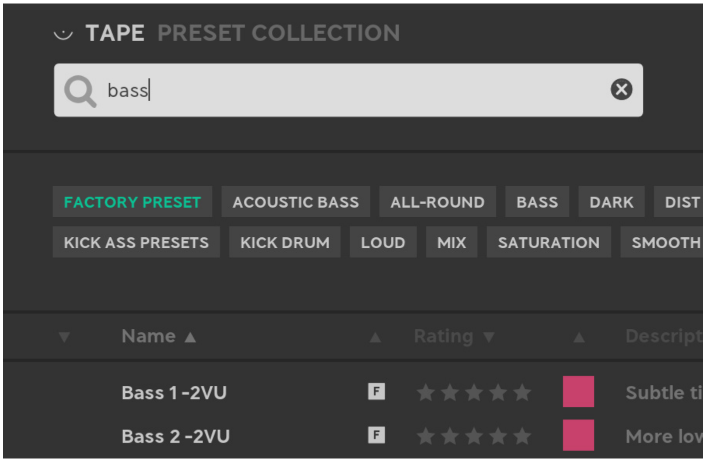
Preset Collection

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



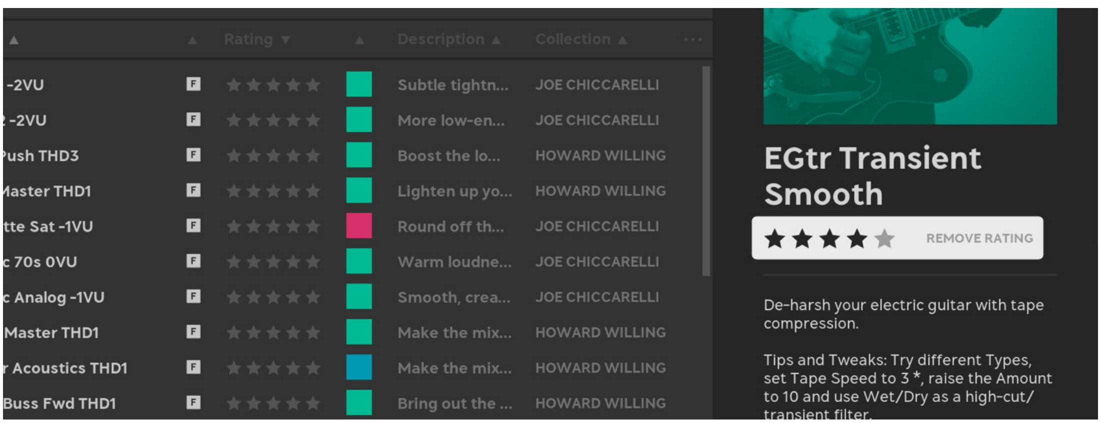
Workflow

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.

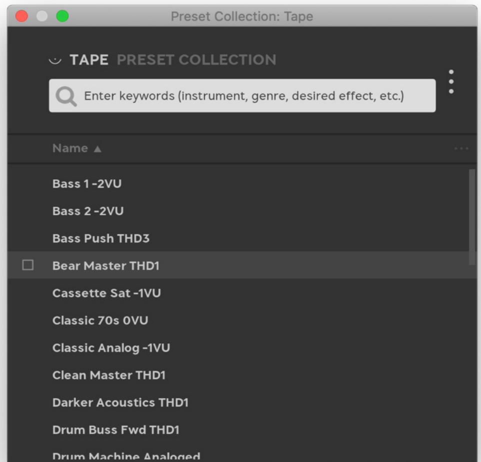
> Don't be afraid to add or change metadata in the factory presets. (And it's always possible to restore the original metadata later if you want to!)

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.



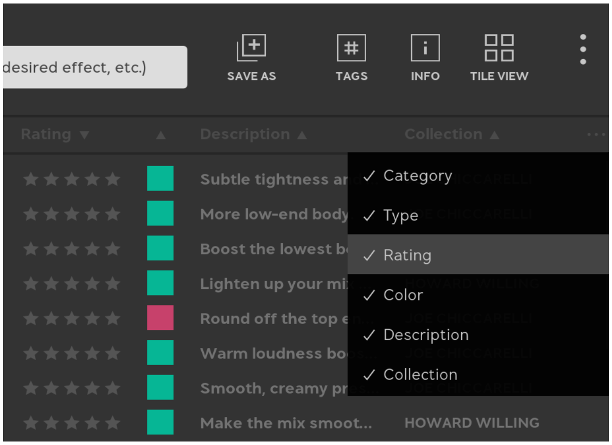
The minimal view of Preset Collection, only preset names and search field are visible.

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.



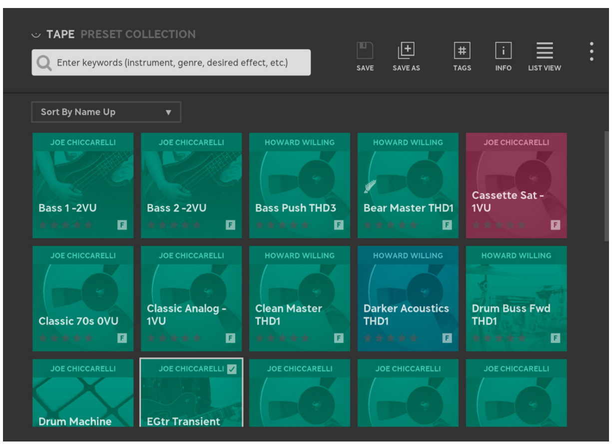
Select which columns are visible from the menu.

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

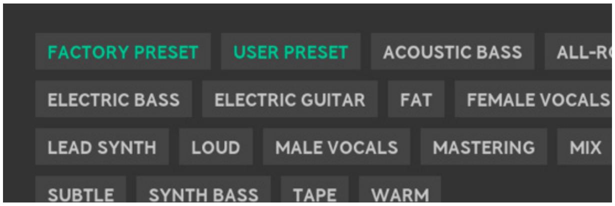
Warn When... Turn warnings on/off when for example deleting or overwriting presets.

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 ⊞

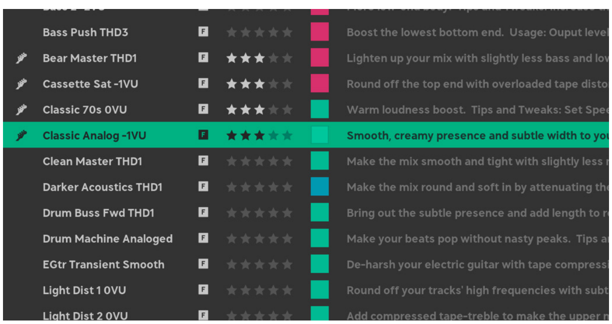


Use the Tags pane to filter presets by tag.

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.

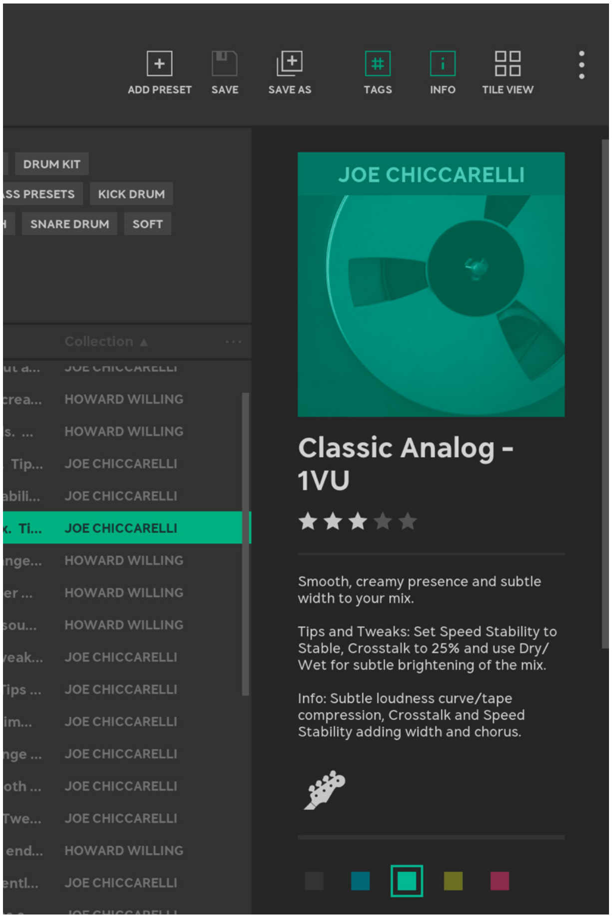
To sort presets in the LIST VIEW, click on ▲▼ for a column to sort down or up for that column. In the TILE VIEW, open the "SORT BY..." drop down to select sorting.

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.

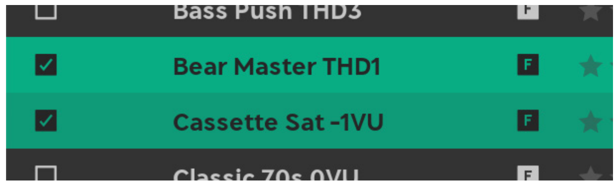
In the INFO pane you can edit the metadata of the selected preset. You can change image, name, rating, description, category, tags and color. Whenever it is possible to change a field, a pen icon will appear. All changes made to the description of the plug-in are saved immediately.

Change Image

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

Group Edit Mode

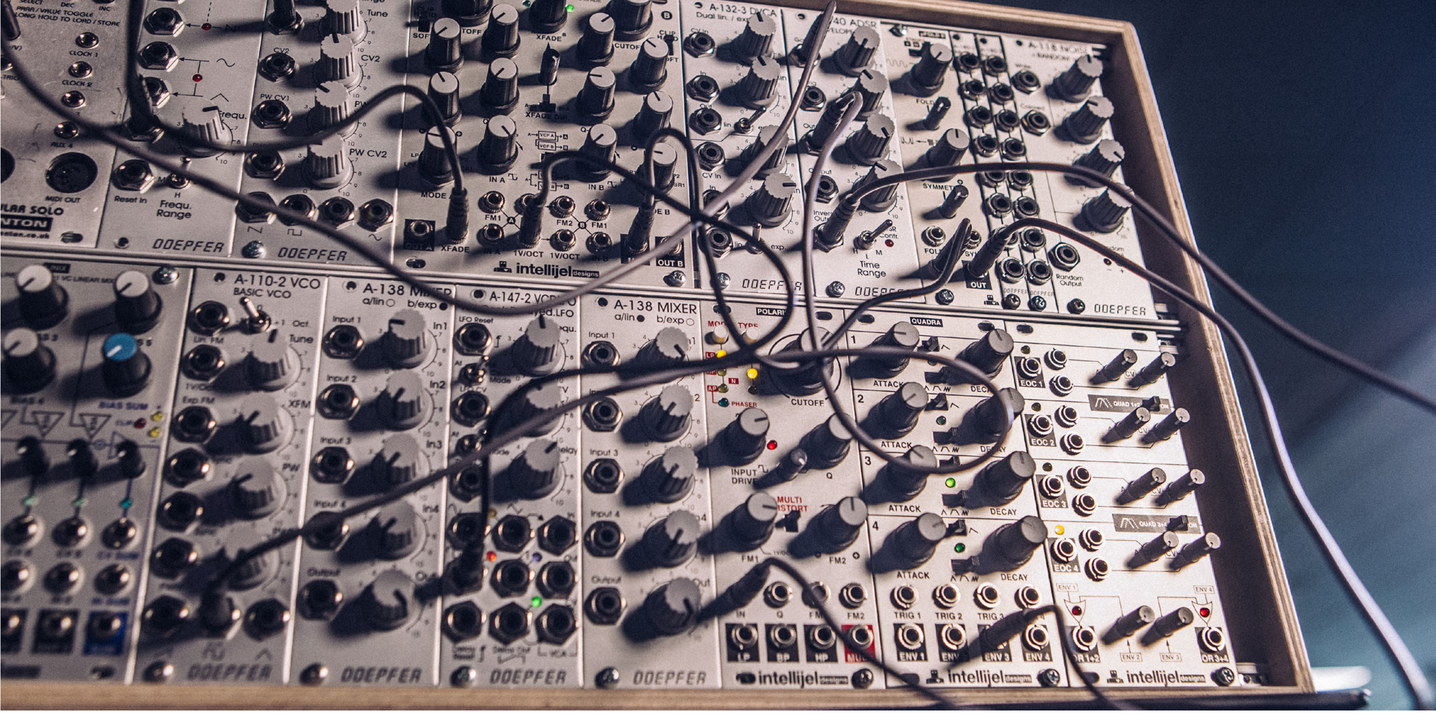
It's easy to edit several presets at once, for instance if you want to change rating or add a tag to multiple presets. Enter the GROUP EDIT mode by selecting several presets in the check box on the left. Once in GROUP EDIT mode, you can use SHIFT + click to select a range of presets.



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



1 Modular

BEFORE THERE WERE MINIMOOGS, Odysseys or any of the Japanese compact performance synthesizers, there was the modular synthesizer. The modular synthesizer consisted of the core building blocks—modules—for sound sculpting; tone generators (also known as oscillators), fixed filters and modifiers. These were separate units which could be interconnected, usually using patch cords. The fact that the individual modules could be patched in a number of configurations opened to a wide range of sonic options.

Doepfer And The Eurorack Synth

In the mid 1990s, there was a modular synth revival when German synth designer **Dieter Doepfer** launched his range of modular synthesizers. Doepfer's system was called A-100, and the modules were housed in Eurorack casings, a rack format previously used extensively in the telecommunications industry. Soon, a large number of synth vendors started coming out with their own modules for the Eurorack standard. Today, there are over 4000 Eurorack modules available from over 150 different vendors.

Softube Modular

Softube Modular is a virtual Eurorack modular synthesizer, which is used as a plug-in in any major DAW. Modular was developed in close collaboration with Dieter Doepfer, and six Doepfer modules are included in the basic Modular package, along with more than 20 of Softube's own utility modules. More modules can be purchased separately to expand the options from the basic package.

Modular can be used just like an actual Eurorack modular synth, by adding different modules and combining them using virtual patch cords. A Eurorack modular synth is by definition monophonic and so is also the Softube Modular, although a certain degree of polyphony can be achieved. More of this later. The analog synth modules have been modeled, component by component, by Softube's engineering team. Therefore, Modular is sonically impossible to distinguish from its analog counterparts. Furthermore, just like a real life Eurorack synth, any audio signal can be fed into the modules. To facilitate using Modular this way, we have created a separate plug-in called Modular FX.

It goes without saying that the experience of connecting actual patch cords and tweaking physical synth modules can never be replicated in the software domain. But the fact that Modular resides in the computer opens to a few possibilities that are not available with physical Eurorack synthesizers. First of all, patches (complete setups of modules, their interconnections and settings) can be easily saved, instantly recalled and shared with other Modular users. Second, a large number of professionally made presets are available. That makes Modular instantly usable also to inexperienced users—and offers a great way to learn more about modular synths. Furthermore, Modular offers Performance modules, which makes it easier to highlight and adjust a select number of parameters on the fly, even with complicated patches. This makes Modular ideal to use for live performances or for inspiring music production in the studio. Plus—if you have access to physical synth modules, you can actually interconnect these with Softube Modular via an audio interface.

Quickstart Guide

If you're familiar with working with software (and hardware) synthesizers, this section may be all you need to get started. Refer to the in-depth sections of this manual to learn the details.

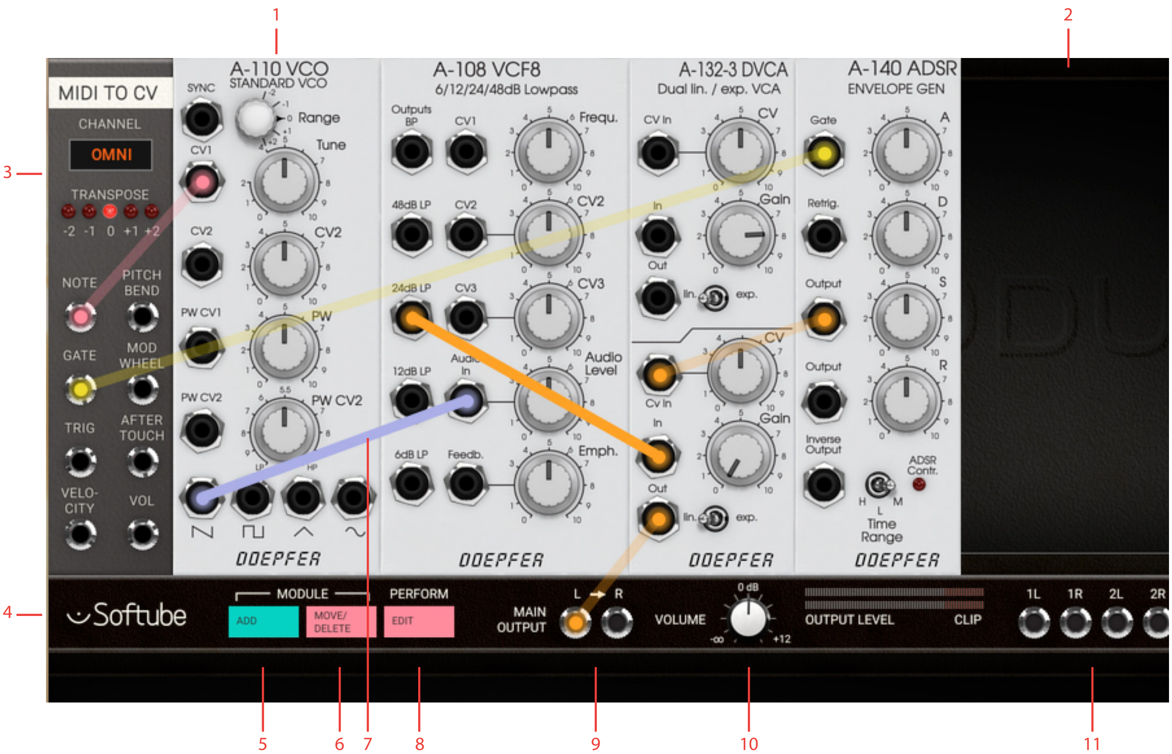
When launching Modular, you will find that its default state is an empty patch, and as no modules have been added to the rack yet, you find yourself presented a choice of modules in the MODULE SELECT VIEW. By selecting modules you can start to create your patch. You can also open a preset patch from the extensive factory library in your DAW. This is slightly different depending on which DAW and plug-in format you're using (please refer to your DAW's User Manual). If you open a preset patch, you will note that the patch cords appear when you move the mouse pointer over them, and otherwise fade away.

> To add multiple modules without exiting the Module Select view, press Shift while clicking to select.

To connect two modules, click and hold a jack. To facilitate patching, the inputs to which the patch cord can be connected will be marked with a color, green (INPUTS) and red (OUTPUTS). An output can only be connected to an input and vice versa. Simply drag and drop the cord on the desired jack. Multiple patch cords can be routed from an output (red) to several destination inputs (green), but an input jack can only accept one patch cord at the time. If you add more modules than fits your screen, you can scroll to see them all by clicking and dragging anywhere, except over a knob or jack. Scrolling can also be performed by smart gestures or scroll wheel if your computer supports this. Any patch can contain up to 100 modules, depending on the size of the modules. Just like a physical Eurorack synth, to hear your creation, you will need to finish your patch by connecting it to Modular's outputs. These are located in the CENTER ROW. Similarly, in order for your patch to be controllable via MIDI, you need to add the **MIDI to CV** module to interface with your DAW. Getting MIDI clocks from your DAW is achieved through the **DAW Sync** module.

User Interface

These are the main sections and features of Modular.



1. This is a Softube emulation of a Doepfer A-110 Oscillator module.
2. This is the empty rack space where new modules are placed.
3. To make Modular respond to incoming MIDI signals from your DAW or a MIDI controller, you need a MIDI To CV module in your patch.
4. Center Row. Here, you have access to Modular´s main functions, as well as inputs and outputs.
5. Click to enter Module Select View, where you select which modules to use in your patch.
6. Enter and exit Move and Delete mode.
7. This patch cord connects one of the oscillator´s outputs to the input of the Doepfer Filter module.
8. Enter Performance knobs linking mode.
9. Main Output. Both these need to be connected to achieve a stereo image to your DAW. Left output is normalized to the right, so connecting only L will result in a mono signal output on both channels.
10. Output volume and meter.
11. Aux Outputs.

Overview

The Softube Modular plugin consists of an empty Eurorack that can be filled with Eurorack modules, your building blocks, for building your patch. The CENTER ROW is situated between the first and second row of your rack area. The CENTER ROW consists of the buttons for **adding modules, moving around or deleting** them, as well as the button for **editing your Performance modules**.

Also situated on the CENTER ROW is the INPUT (only on Modular FX) and OUTPUT jacks for audio. Here's also the main **Output Volume** knob and its **Output Level** meter which indicates if the signal is clipping. The Modular Eurorack workspace can be scrolled up and down by clicking and dragging on the background or on the module panels in the patch view. Using the scroll wheel on your mouse or smart gestures (two finger scrolling) is also supported.

When scrolling, the CENTER ROW will scroll along, but stay at the top of the screen. Controls within modules (knobs, slider, switches) can be moved by clicking and dragging the mouse cursor over the parameter. A parameter can be reset to its pre-set value by clicking while holding ALT key. Fine adjustments can be done by holding ⌘ (Mac) or CTRL (Win), while clicking and dragging on a parameter.

Handling Modules

The modules are the building blocks from which you create your modular synth patch, just like in a real life Eurorack modular. When you launch Modular, you will see the two top rows of rack space (empty by default), divided by the horizontal CENTER ROW. The rack space is where the modules will go as you add them. You can scroll down to see additional empty rows of rack space.

On the far left side of the CENTER ROW, two buttons are labeled **Module**, one to **Add** and the other to **Move/Delete**. Clicking **Add** opens the MODULE SELECT VIEW. Here, you will see all your available modules, sorted by brand and type. Placing your mouse pointer over a module opens a brief description. The details of each module is covered in `"Modules In Detail" on page 35`. Clicking a module adds it to the rack. You can hold SHIFT and click to add several modules at once to the rack. Exit MODULE SELECT VIEW window by clicking the pink "X" icon in the upper right corner.

Move/Delete View

Having added modules to your rack, you can click the **Move/Delete** button in the CENTER ROW to move or delete them. While in MOVE/DELETE VIEW, the **Move/Delete** button is lit and the added modules are darkened with an added "X" mark in each upper right corner. Clicking on this "X" will delete the module.

By clicking at a module, it becomes highlighted, indicating that it is now selected for moving. Click on an empty space in the rack to where you want to move the selected module or move the module by clicking and dragging the module to the desired location. While being moved, the module will turn opaque when you drop it to a new location. If you drop it while it's semi-transparent it will be positioned back to its previous position.

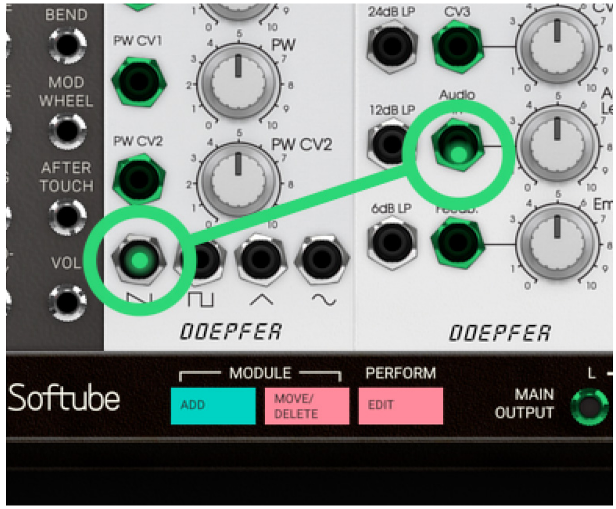
Exit MOVE/DELETE VIEW by clicking on **Move/ Delete** button again. The button is unlit as you're back in normal PATCH VIEW.

Patching

When you have added modules to the rack, you need to manually connect them using patch cords. To add a connection, click and hold an output jack on a module. A circle appears, and if you move the mouse (still holding the mouse button down), a patch cord is drawn from the output.

Any input jacks that can be used as a destination for your current connection will turn green. If you are instead patching from an input, all the available destinations will of course be outputs, marked red. Drag the patch cord to one of these and let go—the connection has been made. When you move away the mouse pointer, you will see the patch cord fades away. The two connected jacks will be marked with the same color as the patch cord, and if you move the mouse pointer back near any of the patched jacks, the cord will reappear.

To alter a connection, click on a jack and move the cord to another jack. Since multiple connections can be patched from the same output jack, patchcords can only be removed from the "input-jack side". To remove a connection, drag the cord on the input jack side to somewhere where there is no other jack and let go. Multiple cables can be stacked from the same output but only one can connect to the same input. This can be useful for example when using a single envelope generator output to control multiple other modules.



Patch from an **output** jack, and all **inputs** will get highlighted with a **green** color.

The Output Section

Any patch must end with a connection to the **Main Output** jacks on the CENTER ROW, otherwise, no sound will come out of Modular. The LEFT output is normalled to the RIGHT, which means that if your patch is only connected to the LEFT output, the sound will be in mono. To create a stereo patch, both LEFT and Right outputs must be used. The **Volume** knob lets you adjust the output volume to reasonable levels, as indicated by the **Output Level** meter. Avoid running this into the red segments, labeled CLIP.

The CENTER ROW also has four stereo pairs of auxiliary outputs, labeled **Aux Outputs**. These can be used to send separate outputs to your DAW.

> This can be useful for further processing in your DAW, or to route signals from Modular to a hardware Eurorack system.

The **Block DC On Aux Out** buttons is a feature used to make the Aux outputs sending ordinary AC (audio voltage) in order to interface with your hardware Eurorack system. However, setting the **Block DC** to OFF should be used with utter care. A large DC offset directly to your speakers can damage your hearing and/or speakers.

While patching from an **input**, all **outputs** will get highlighted with a **red** color.

Understanding Control Voltage

A physical Eurorack modular synth communicates via CV (control voltage), which Modular emulates. CV is used to control parameters such as pitch, cutoff frequency, pulse width and more. An interesting aspect of Eurorack modular synths is that there is no real difference between audio and controller (CV) signals. These can be mixed or used to interact with each other, for example using the output of an oscillator to modulate the cutoff frequency of a filter, so called "Filter FM".

Events in a Modular synthesizer are often triggered by a GATE or TRIGGER pulse. A GATE signal is a pulse that is extended high until released, typically the output of a MIDI TO CV module where the GATE signal corresponds to MIDI note on (GATE rising edge) and MIDI note off (GATE falling edge) messages.

Certain modules requires a short pulse, rather than a GATE. This is called a TRIGGER pulse. Examples of modules that work best with TRIGGER pulses are for example the Heartbeat drum modules (using the MIDI TO TRIGGER module or BEAT SEQUENCER) and SEQUENCER modules.

Interfacing With DAWs And MIDI Controllers

The fact that Modular uses CV also means that it interfaces to DAWs and MIDI controllers in a less integrated—but much more flexible—manner compared to most software instruments. Any connection between Modular and a DAW or MIDI controller requires at least one of the modules in the DAW AND MIDI INTERFACING category in MODULE SELECT VIEW.

In order to use MIDI signals from a DAW or a physical MIDI controller to play notes or control parameters, you need to add one of the MIDI to CV conversion modules to your patch. These modules convert the incoming MIDI data to virtual voltage, which the Modular's modules can react to. You will then need to patch the outputs of these modules to the modules you would like to be affected by the DAW or MIDI controller.

Similarly, to make any of Modular's modules synchronize to the DAW tempo, you need to add the DAW Sync module, and patch that to the sequencer or LFO module you would like to have synchronized. See the detailed description of the DAW AND MIDI INTERFACING modules in the `"Modules In Detail" on page 35` and go through the `"A Basic Modular Synth Patch" on page 26` to see an example of this in use.

Performance Modules

Both in the studio and during a live performance, a big part of the attraction to using synthesizers is the ability to alter sounds on the fly by adjusting parameters as the music plays. But even with complex patches using a large number of modules, there is usually a limited number of parameters you are interested in tweaking as you go. These parameters can, however, be spread out across a number of different modules that may be placed far from each other in the rack, making live adjustments impractical. To facilitate sound tweaking on the fly, we have included the user definable PERFORMANCE modules. These have controls (knobs, switches and/or sliders) that you can freely assign to any parameter within your patch. This lets you gather the parameters you would like to play with in a single spot. The parameters become easy to find and easy to assign to a physical MIDI controller.

For more info, please see "Performance Modules" on page 31.

Sequencing

Four different sequencer modules are included in the Modular basic package. You will find them under the SEQUENCER category in MODULE SELECT VIEW. These can be used for analog style step sequencing, and can be synced to your DAW tempo using the DAW SYNC module.

For more info, see "The Sequencer Modules" on page 63.

Modular FX

The Softube Modular can also be used as an audio effect. So instead of basing a patch on an oscillator, you can base it on any audio signal–a guitar, a drum loop or a vocal recording. The only difference is that, as an audio effect, the CENTER ROW will also contain an INPUT stereo pair. This is where the audio enters the Modular FX. Patch and process your incoming audio through any of the Softube Modular effects in order to build your own effects, or just load some of the included Modular FX presets.

Basic Terminology

CV Control Voltage. A signal that controls other modules.

VCO Voltage Controlled Oscillator. A module that generates waveforms (makes sound).

VCF Voltage Controlled Filter. The basic approach to synthesizers is to use a harmonically rich audio signal, such as a sawtooth waveform, and use a filter (VCF) to remove "unnecessary" harmonics. This is called *subtractive synthesis*.

ADSR Attack/Decay/Sustain/Release. Converts on/off information to a nice envelope with variable attack, decay, sustain and release.

VCA Voltage Controlled Amplifier. This is a CV controlled volume knob, usually used to apply an ADSR envelope to control the volume of a signal.

LFO/VCLFO Low Frequency Oscillator, usually Voltage Controlled. Basically the same as a VCO, but for very low frequencies. Use to create vibratos, pulse-width modulations, etc.

MIDI Mapping

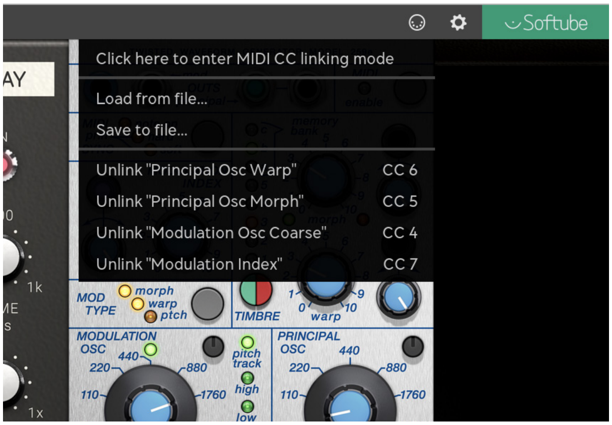
Softube Modular can be mapped to any midi-controller that can output MIDI controller change (sometimes just called MIDI CC) data. This can be done through the MIDI CC linking mode menu at the top right in the plugin.

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 Modular. 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". A word of caution - since the Modular parameter set is dynamic this functionality can be confusing when loading a previously saved map within a ".softubemidipreset"-file on to a preset structure that it was not intended to control.

The saved midi map contains information about the instrument-type (Modular in this case) and will not be able to load within another Softube instrument (for example Model 72).

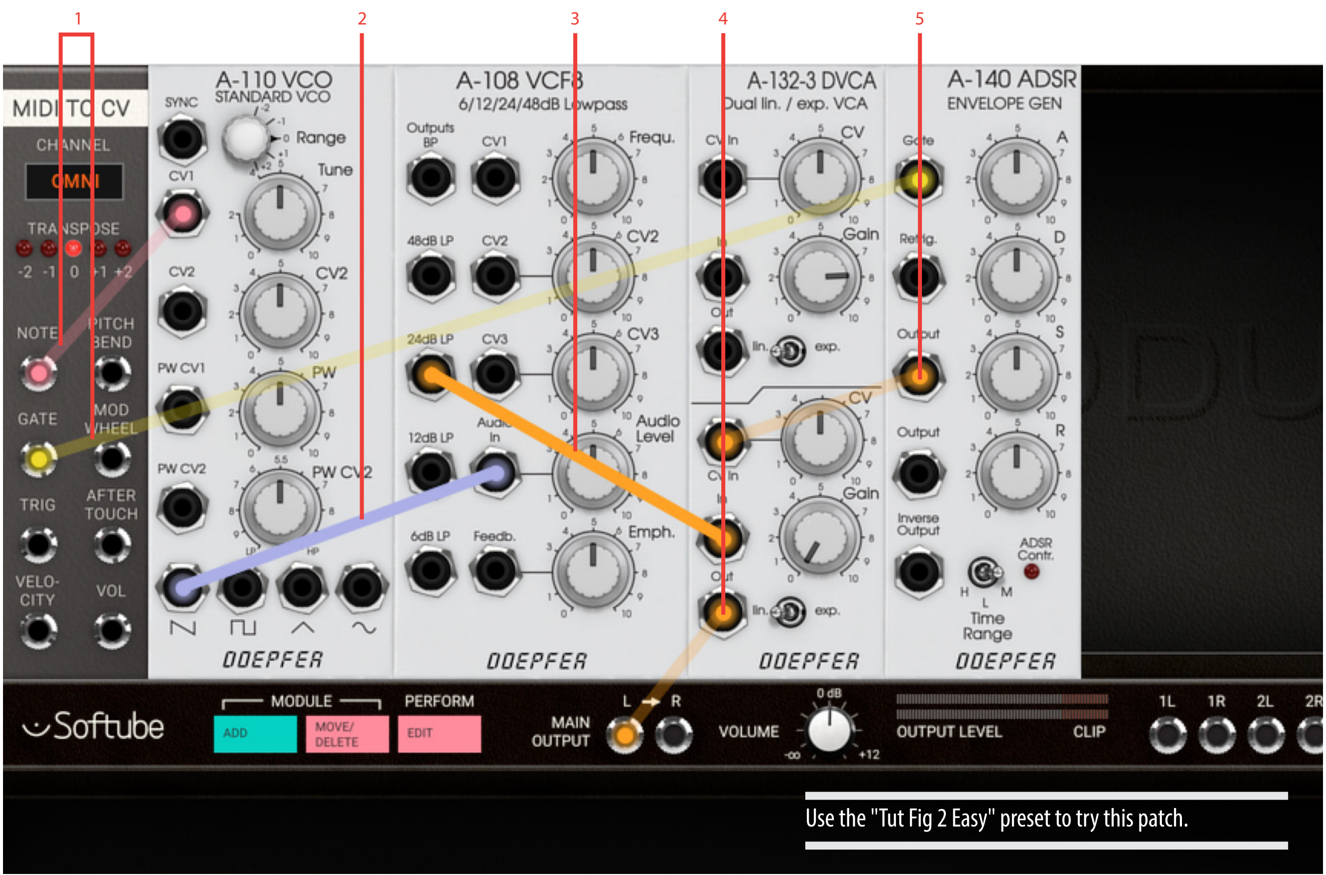
A Basic Modular Synth Patch

This is the smallest patch that uses all of the classic synth parts.

1. The MIDI TO CV sends note information to the tone generating oscillator (VCO), and on/off (**Gate**) information to the envelope generator (ADSR) .
2. The sawtooth output from the VCO is sent to the low pass filter (VCF).
3. The low pass filtered signal is sent to the amplifier (VCA) where its volume will be controlled by the ADSR via the **CV In**.
4. The ADSR converts the on/off information into an envelope with variable **Attack, Decay, Sustain** and **Release**, which is used to control the amplifier (VCA).
5. On the VCA, turn down **Gain** so it is 100% controlled by the CV signal. Finally, the output from the VCA is sent to the **Main Output**.

When sending a MIDI note from your DAW (for example by pressing a key on your MIDI controller keyboard) a note with a slow onset will be heard.

Raise the cutoff frequency (**Frequ.**) on the VCF module fully clockwise to brighten the tone. This can be automated and controlled by the ADSR envelope—connect one of the outputs of the ADSR to the **CV2** jack of the VCF, lower the cutoff frequency again and a familiar filter sweep will be heard each time you strike a note.



Use the "Tut Fig 2 Easy" preset to try this patch.

Basic Sequencer Patch

This is another simple example on how to set up a eight step sequencer to be synced by the DAW to control the simple patch build above. The DAW SYNC module outputs a clock, in this case 16th notes, that is patched into the **Clock In** of the SEQUENCER 8 module.

In order to get the sequencer to start from the beginning when the DAW playback is restarted, the **Reset** pulse jack of the **DAW sync** module is connected to the **Reset In** jack of the **Sequencer 8**. Now, when the DAW is started, the Sequencer will slave to its tempo and output CV, gate and trig information. By connecting the **CV** out to the **CV1** in on the DOEPFER A-110 VCO module, the SEQUENCER 8 steps can control the pitch of the oscillator. When controlling the pitch of an oscillator it is recommended to engage the **Quantize** button on the sequencer module, since this will quantize the outgoing voltage to a chromatic scale. **Gate Out** or **Trig Out** can be used to trigger an envelope generator, such as the A-140 ADSR, used in the example above. The difference between the **Gate Out** and the **Trig Out** is that the **Gate Out's** pulse lengths are longer and that several programmed gates after each other, creates a legato note on the **Gate Out**. In the example above you can see and hear this on step 6 in the sequencer.

Use the "Tut Fig 3 Sequencer" preset to try this patch.

Use the "Tut Fig 4 Beat Sequencer" preset to try this patch.

Beat Sequencer Patch

Using the BEAT SEQUENCER with the DAW SYNC module is more or less the same procedure as the previous example, but with the big difference that this sequencer module doesn't send out any pitch information, only triggers. Another difference is that the BEAT SEQUENCER has four tracks of triggers, each with its own output. These trigger outputs can be used to trigger any kind of event: an envelope (like the A-140 used in the example), another sequencer, a drum module, a clock divider, a sample and hold module et cetera.

In this patch the DOEPFER A-140 ADSR output is used to control not only the volume of the sound (by connecting it to the A-132-2 DVCA's **CV** input), but also to control the A-108 VCF's filter cutoff frequency (by connecting it to **CV2** in), as well as to control the pitch sweep of the A-110 VCO (by connecting it to **CV2** in on the A-110). This creates the synth-tom sound which is triggered from the BEAT SEQUENCER.

Linking Multiple Sequencers Patch

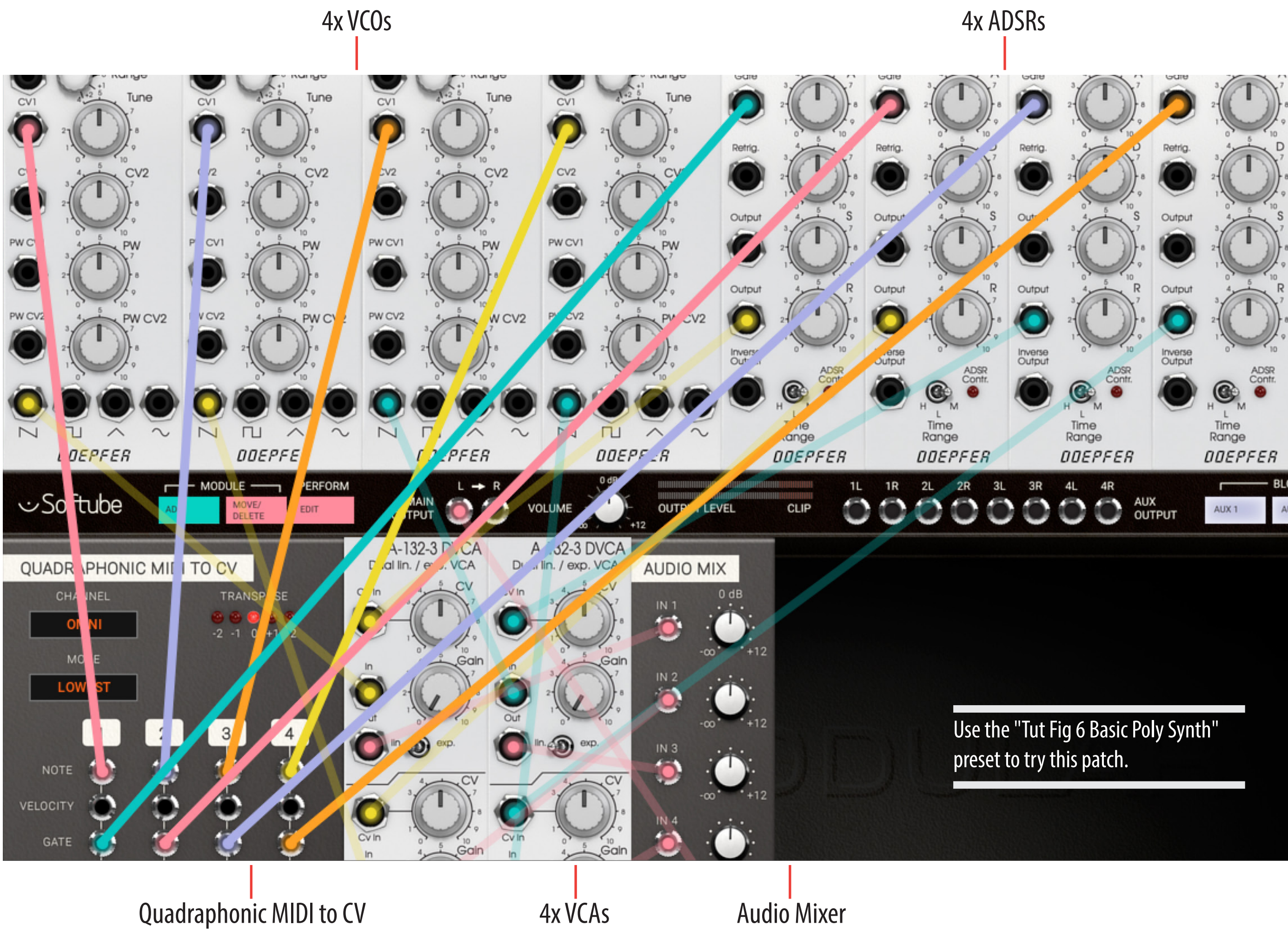
This example patch uses two linked 16 step sequencers in order to achieve a 32 step sequence. Both 16 step sequencers are clocked from the same clock source (16th notes from DAW SYNC). But they get their reset clocks from the linked out pulse which is transmitted after the last step when the sequencer is not in loop mode. The CV and gates from the two sequencers are mixed together to control the simple synth setup (VCO, VCA, ADSR) at the bottom. The logic tool is used for the logic function OR on the DAW SYNC's **Reset** pulse and the **Link Out** pulse from the second sequencer. This patch has some flaws but it is a good starting point for further patching.

Use the "Tut Fig 5 Linked Sequencer" preset to try this patch.

Basic Polyphonic Synthesizer Patch

It's possible to build four voice polyphonic patches with Softube Modular using the QUADRAPHONIC MIDI TO CV module. But just like with a world Eurorack system, this takes four times more modules if each voice needs its own oscillator, filter, etc.

This example demonstrates how a polyphonic patch is set up. Each voice in this really simple patch consists of a VCO going into a VCA that is controlled by an ADSR envelope. Each voice gets its own note CV from the **Quadraphonic MIDI to CV** module and in the same manner, each voice also get its own gate information sent to its dedicated ADSR envelope. The ADSR output is connected to the **CV In** on the VCA controlling the overall volume of that voice. The VCA of each coive is then summed in the **Audio Mixer** before being sent to **Main Output**. Experiment with the QUADRAPHONIC MIDI TO CV voice modes to see the difference between rotating assign and lowest assign modes.



Performance Modules

The PERFORMANCE MODULES offer a nice way of getting easy access to a selection of parameters from all modules in your patch in one convenient location. The modules can be used to create what could be called a macro-area containing all the parameters you want easy accessible for your performance. A Performance knob, switch or slider can be linked to any parameter within your patch, making it possible for you to create a custom interface to your patch.

1. Click Perform Edit to enter Performance Edit Mode. Available Performance Knobs turn green.

2. Click on a green Performance Knob to select it. Assignable knobs will turn green, and the Performance Knob turns yellow.

Performance Linking In Detail

The knobs, switches and sliders on the PERFORMANCE MODULES can be edited to control parameters within your patch, all from one convenient place. Click on the **Perform Edit** button on the CENTER ROW to enter PERFORMANCE EDIT MODE, which is indicated by the activated **Perform Edit** button. Patching is disabled in PERFORMANCE EDIT MODE.

3. Click on a green knob to assign it to the Performance Knob.

In PERFORMANCE EDIT MODE the knobs on the PERFORMANCE MODULES in your patch that are unassigned will become green tinted, and the text label underneath will also say "CUSTOM VALUE".

By clicking on one the green performance knobs, the knob color turns yellow indicating that you have selected this particular knob for being assigned to a parameter within your patch.

4. When a knob has been assigned to a Performance Knob it can only be controlled from the Performance Knob, which is indicated by a red color.

All controls in your patch that are available as destinations, will now be indicated by turning green. Click on the control (knob, switch, slider) within your patch you want as your linked destination. This knob will turn transparent red to indicate that it is linked to a performance knob. The slaved parameter will get disabled as long as it is slaved to a performance module.

The Performance knob name can be edited by

5. Hover over the Performance Knob to change to a custom name, or set the ranges of the assignable knob.

hovering your mouse pointer over it until a window appears. In this window, you can write your custom name for the knob's function and also set custom ranges using the **Min** and **Max** knobs.

To unlink a performance knob, double click it and the parameter slaved to it will lose its red color and be available for normal use again.

To exit PERFORMANCE EDIT MODE, click the **Perform Edit** button again and its appearance is reverted back to normal as you're back in PATCH VIEW.

LEDs on Performance modules

LED indicators on Performance modules and its corresponding input jack is for visual reference and can be used for displaying an audio level, gate indication or clock rate. Blinkenlights!

The Preset Library

Modular comes with more than 200 preset patches. Use your DAW's preset browser to browse through the patches. They are organized as follows:

Bass	Different kinds of bass synth sounds.
Brass	Trumpets, horns (OK, we also included some flutes here).
EFX	Effect sounds.
Lead	Different kinds of lead synth sounds.
Pads	Mono and poly chord sounds.
Perc	Different kinds of percussion sounds.
Sequenced	Sequenced and pseudo-sequenced sounds.
Strings	Lush and analog, acoustic and murky.
Tutorial patches	Good starting points, described in the tutorial examples above.
Heartbeat	Sounds that make use of the Heartbeat modules.
Expansion	Sounds that requires add-on modules, for example Intellijel modules.

Artist	Presets created by artists.
Modular FX	Presets made for the insert version of Modular, to be used with an external audio source. These are only available in Modular FX.

Setup Window

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

Always use smaller GUI	This forces the GUI to remain small even if higher resolution is supported by your computer screen. Uncheck this if you want to use a larger GUI.
Use medium GUI on large screens	This forces the GUI to remain medium sized even if higher resolution is supported by your computer screen. Also uncheck this box if you want to use the largest GUI.
Show value display	Toggles the value display in the lower left corner of Softube Modular on and off.

About CV and Gate Standards

The mid 1960s was an exciting time with huge leaps forward in technology for synthesizing sound. It has since then been widely debated who really invented the voltage control for synthesizers. Nevertheless, voltage control was a significant invention that enabled all synthesizer modules to communicate to each other and it was **Bob Moog** that standardized CV. 1 Volt/Octave for pitch was the first standard to prevail although many other manufacturers for a long time also used other standards such as Hz/Volt (which respond to each octave by a doubling of voltage).

For triggering of electronic switches and envelope generators within his system, Bob Moog used the so called *s-trig* (short trig) standard where others used the *V-trig* (voltage trig) standard. The latter eventually became more popular and is today the standard when we talk about CV (control voltage for pitch) and Gate (for triggering sequences, envelopes etc).

The Eurorack standard, more or less set by **Dieter Doepfer** with his A100 system, use the linear CV and V-trig standard. There are three types of signals within a Eurorack system:

Audio Signals	Ranges from -5 to +5v
Control Voltages	Typically range within 0 to +8v, as for example the output of an ADSR
Trigger Voltages	Range 0 to 5v, trigger occurring at leading edge

All these three signals can be patched without restrictions within the system, which means that audio can be used to trigger logic modules designed for use with trigger voltages or as modulators connected to inputs designed for control voltage. The system will not differentiate between the types which makes the Eurorack system extra intuitive, open for experiments and happy accidents.

Modules In Detail

In this section each module of the Softube Modular is described in detail, with added suggestions on use. In *Description Legend* below we define how each module is described and what the different sections described refers to.

Description Legend

Names in bold
Refers to knobs, switches, dials, input/output jacks, etc.

NAMES IN CAPS
Refers to a parameter setting, or the name of a module or function.

[descriptive text]
Text within brackets is a description of a symbol on the panel, for example that of an waveform or other graphical element.

Parameters

Refers to knobs, dials, switches on module and are described from left to right, from top to bottom on module.

Inputs

Refers to the input jacks of the module used for input of audio or CV. They are described from left to right, from top to bottom on the module.

Outputs

Refers to the output jack of the module used for outputting audio or CV. They are described from left to right, from top to bottom on the module.

In Use

Detailed description of the module's functions as well as some tips and tricks.

Doepfer

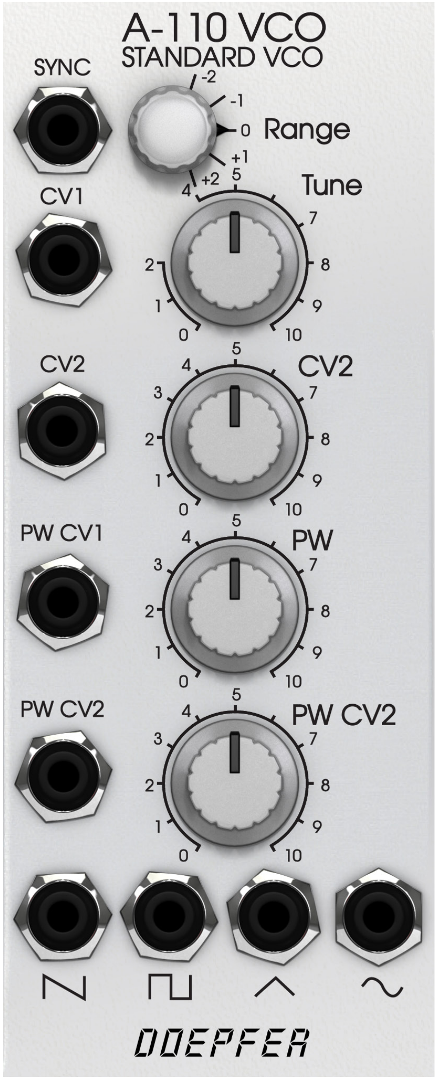
This category contains the modules from legendary Eurorack manufacturer Doepfer. These include a VCO, VCF, VCA, ADSR Envelope, LFO and a Noise source—all you need for building your own basic subtractive synthesizer patch.

Doepfer A-110 VCO (Voltage Controlled Oscillator)

An oscillator is a tone generator, the most basic building block of a synthesizer. The Doepfer A-110 can generate four different waveforms, each with its distinctive own overtone spectrum.

Parameters

Range	Octave offset ranging five octaves
Tune	Fine tune up and down one whole note, preset setting 5 is tuned to 440 Hz when playing an A3 from a MIDI keyboard.
CV2	Sets the amount of CV modulation of the incoming signal at CV2. Knob turned fully clockwise is full 1v/octave range.
PW	This knob sets duty cycle of the pulse wave. The preset setting 5 is 50%, also known as a square wave.
PW CV2	This knob controls the amount of pulse width modulation by incoming CV at PW CV2 jack.



Inputs

SYNC Incoming signal on this jack resets the oscillator, often referred to as hard sync on other synthesizers.

CV1 CV control of pitch, full range 1v/octave. Connect your note CV from MIDI to CV converter module to this jack.

CV2 CV modulation input, connected with CV2 knob (see above).

PW CV1 CV control of pulse width, full range.

PW CV2 CV control of pulse width, connected with PW CV2 knob (see above).

Outputs

[saw] This jack outputs a saw wave, a harsh and clear sound containing both even and odd harmonics.

[pulse] This jack outputs a variable pulse wave, a more hollow sound than the saw wave which contains mostly odd harmonics when the pulse width is 50% (PW knob set to 5).

[triangle] This jack outputs a triangle wave, also a very hollow sound suitable for synthesizing flutes, vibes and organ sounds.

[sine] This jack outputs a sine wave which is a waveform that only contains the fundamental, none of the overtones. This is thus not suitable for subtractive synthesis (shaping the sound with a filter).

In Use

Connect the **Note** CV out from your MIDI TO CV converter to **CV1** on the A110 for accurate full tracking of the A110. By connecting **Note** CV into the **CV2** and using the **CV2** knob, microtonal changes smaller than 1 V/Octave tracking can be obtained. Connect **Sync** jack to the square output of another oscillator to achieve hard sync. This means that A110 is reset by the other oscillator which can be heard when raising the pitch of the A110 above the other one's—the harmonic spectrum of the outputted signal of the A110 is now changed. Control the A110 pitch by envelope, LFO or similar to achieve moving "hard sync sweeps".

The Softube Modular Doepfer A110 module models the real hardware with all its flaws and quirks. This means, for instance, that the triangle and sine outputs are not "mathematically clean" but has the additional harmonics just like the original circuits they are modeled upon. This gives the module character and a nice musicality, but if you prefer a "clean" sine waveform we advise you to use the UTILITY SINE OSCILLATOR instead.

Doepfer A-108 VCF8 (Voltage Controlled Filter)

Voltage Controlled Filter based on the well-known transistor ladder filter (the "Moog" ladder) but with a unique external feedback path.

Parameters

Frequ	Cut off frequency of the filter.
CV2	CV2 amount, scales the incoming signal on the **CV2** jack.
CV3	CV3 amount, scales the incoming signal on the **CV3** jack.
Audio Level	Incoming volume of the Audio input jack.
Emph	Emphasis, this controls the feedback amount of the filter. Often also called resonance or Q.

Inputs

CV1	CV control of cutoff frequency, full range 1v/octave. Connect your note CV from MIDI TO CV converter module to this jack if you want your filter to fully track your keyboard.
CV2	CV control of cutoff frequency. This jack is tied to the **CV2** knob that scales the amount of the signal from this input.

CV3 CV control of cutoff frequency. This jack is tied to the **CV3** knob that scales the amount of the signal from this input.

You can have several CVs controlling the cutoff frequency, the different CV inputs are added together within the module.

Audio In Insert the signal you want to filter here. This jack is tied to the **Audio Level** knob that sets the volume of the incoming signal.

Feedb This jack breaks up the internal feedback path from the **48 dB** output back via the **Emphasis** knob. This enables different modules to be inserted into the feedback loop. For example, inserting a VCA enables voltage controlled resonance. This also makes it possible to feed back other filter outputs than the 48dB LP to obtain a different resonance behavior.

Outputs

BP This is the bandpass output of the filter, which means that it only passes audio centered around the set cutoff frequency. Emphasis sets the width of the pass band.

48dB LP This is the 48dB per octave low pass filter output, the filter's steepest rolloff.

24dB LP This is the 24dB per octave low pass filter output, often associated with the "fat" American synthesizer sound.

12dB LP This is the 12dB per octave low pass filter output.

6dB LP This is the 6dB per octave low pass filter output, the filter's least steep rolloff which gives it a very gentle impact on filtered sound.

In Use

The Doepfer A-108's audio input is very sensitive, so distortion is possible even with normal levels. For example, distortion appears about from position 5 with the A-110 oscillator. In self-resonating mode, the filter can be used as a sine-oscillator source, tracking 1V/octave on the **CV1** input jack.

Doepfer A-132-3 Dual Linear/ Exponential VCA

The A-132-3 is composed of two identical Voltage Controlled Amplifiers where each has a manual gain control and a Control Voltage input with attenuator. The character of the Control Voltage response can be set to either linear or exponential.

Parameters

CV	Control Voltage input Amount. This knob scales the incoming CV controlling the VCA.
Gain	This knob sets the Gain offset of the VCA. It can be thought of as the "minimum" level or "idle" level setup of the VCA.
Lin/Exp	This switch sets the scale response to input Control Voltage at CV in.

Inputs

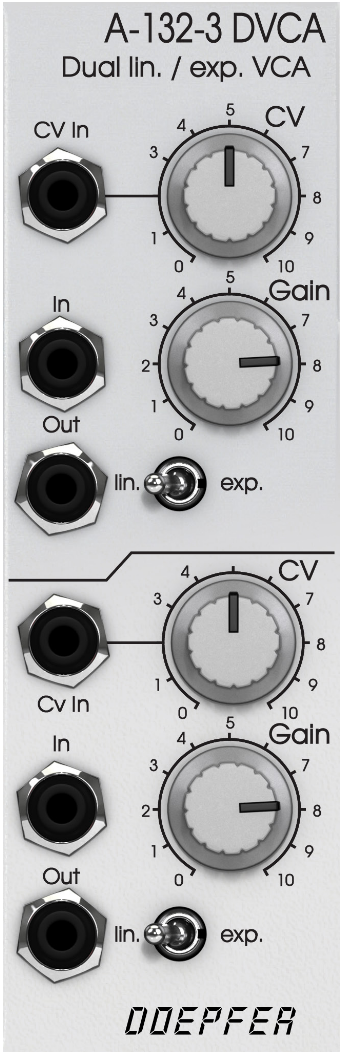
CV in	An incoming signal controls the amplification of the VCA.
In	The input signal which amplitude is to be controlled.

Outputs

Out	Output of the VCA, the resulting signal.

In Use

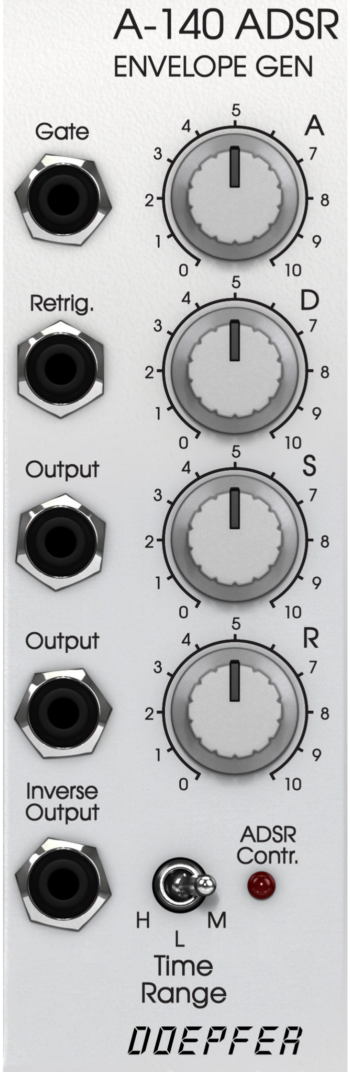
The VCA can be used to shape and scale both audio and control signals. The most common example would be to let an ADSR envelope control the output volume of a synthesized sound in the classic subtractive configuration VCO-VCF-VCA. Other examples include letting the VCA scale the vibrato amount of an LFO to pitch where the CV input is controlled by the Modulation wheel CV from a MIDI to CV module. The VCA can also be used to create tremolo effects when controlled from a LFO or used with the A-108 VCF8 to enable it to have Voltage Controlled feedback.



Doepfer A140 ADSR Envelope Generator

An ADSR envelope generator is a gate-triggered time-controlled series of events outputting timed changes in voltage output. The different phases represents that of an acoustic sound described in its **Attack**, **Decay**, **Sustain** and **Release** phases.

The **Attack** phase is the swelling phase, from zero to the maximum value of the envelope. The **Decay** phase is the subsequent falling time period until the **Sustain** level is met. The **Sustain** level determines the level as long as the MIDI key (or gate voltage) is held. The **Release** phase is the ending time period occurring when the key (or gate voltage) is released and determines the time it takes until the envelop have reached zero again; the end point.



Parameters

A (Attack)	This knob sets the *Attack* time of the envelope.
D (Decay)	This knob sets the *Decay* time of the envelope.
S (Sustain)	This knob sets the SUSTAIN level of the envelope.
R (Release)	This knob sets the RELEASE time of the envelope.
H L M (Time Range)	This 3-position switch lets you select the overall speed of the ADSR. H: HIGH, very slow. Up to several minutes. M: MEDIUM, standard setting. L: LOW, very short time periods down to less than 100 µsec.

Inputs

Gate	This input triggers the ADSR envelope from a gate signal.
Retrig	This input retriggers the envelope from a trigger signal.

Outputs

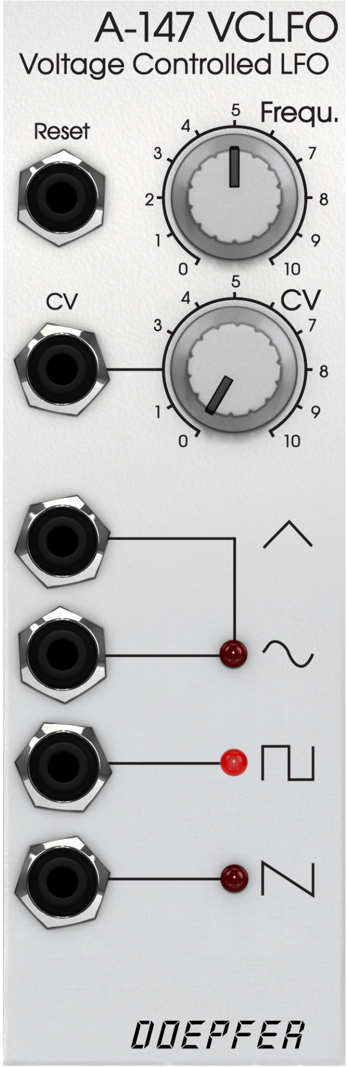
Output	This is the output voltage from the envelope.
Output	This is a duplicate of the output above.
Inverse Output	This is an inverted version of the **Output**.

In Use

The Doepfer A-140 ADSR is a module that is triggered from a gate generated from for example a key pressing (using the MIDI TO CV module) or a sequencer. **Attack** and **Decay** sets the time of the first two stages before arriving at the set **Sustain** level. **Release** is the fall time from **Sustain** level when gate has been released. The ADSR envelope can be used for controlling the VCA, VCF or VCO. Also use the ADSR is to sweep the pitch of a VCO synced to another VCO for sync-sweeps.

Doepfer A-147 Voltage Controlled LFO

A LFO is an oscillator especially adapted to be used as a modulation source running at a slow rate. An LFO is often used to create trills, vibrato, tremolo or control of slowly evolving soundscapes.



Parameters

Frequ	This knob sets the initial speed of the LFO and ranges from 0.01 Hz to 50 Hz.
CV	This knob sets the Control Voltage amount that the signal on the CV input will affect the LFO speed.

Inputs

Reset	This input jack resets the LFO on a rising edge signal such as a Gate signal.
CV	Input of Control Voltage affecting LFO speed.

Outputs

[triangle]	Triangle wave output, suitable for vibrato use (modulating pitch).
[sine]	Sine wave output, also suitable for vibrato and tremolo.
[square]	Square wave output, suitable for trills (modulating pitch).

[sawtooth]	Sawtooth wave output, suitable for creating a beating kind of motion in your sound.

In Use

The most obvious use for the Doepfer A-147 is of course as a modulation source for creating vibrato (modulating pitch) or tremolo (modulating amplitude). By connecting the A-147 **Sine** or **Triangle** output to a CV in of an oscillator a vibrato can be obtained.

Connect the mod wheel or after touch output of the MIDI TO CV module to the **CV** input to control the amount from your MIDI keyboard.

Using the A-147 to slowly sweep the cutoff frequency of your filter is another great way of creating timeless and classic synth sounds.

Another fun thing to do is to use multiple A-147 LFO resetting each other to create complex and "pseudo-sequenced patterns".

Doepfer A-118 Noise + Random Voltage

Module A-118 is a noise and random voltage generator. It produces three types of signal: white noise, colored noise, and random voltage. White and colored noise can be used as audio sources, and the random voltage is a useful source of voltage control, especially for its low frequency content.

Parameters

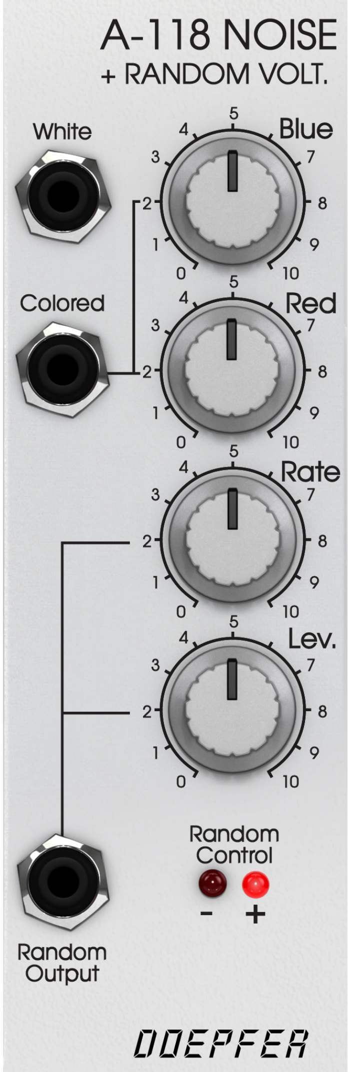
Blue	This knob boosts the high end of the colored noise output.
Red	This knob boosts the low end of the colored noise output.
Rate	This knob controls the slew rate of the random output. Lower values gives less slew and thus more vivid changes in the random output.
Lev	This knob controls the output level of the random output.

Outputs

White	This is the white noise output.
Colored	This is the colored (equalized) noise output.
Random Output	This is the random noise output, it consists merely of low end heavily filtered noise.

In Use

The Doepfer A-118 is a dirty little noise module emulating the amplified noise of a transistor with additional outputs for colored noise for that eerie wind sound. Use **Random Output** to breathe life into boring patches and use **Blue** noise in combination with a clocked sample and hold module for that classic droid sounds.



Doepfer A-114 Ring Modulator

The Doepfer A-114 Ring Modulator contains two separate ring modulators. Each ring modulator outputs the product (Multiplication X*Y) of the signals at inputs X and Y. It behaves similar to a VCA but responds to both positive and negative voltages (4-quadrant multiplication). Ring modulator could therefore be described as a version of amplitude modulation (AM). Where ordinary amplitude modulation outputs the original carrier frequency fC as well as the two side bands (fC - fM, fC + fM), ring modulation cancels out the carrier frequencies, and just lets the side-bands pass to the output.

The emulation of the A-114 Ring Modulator featured in Softube Modular reflects the original hardware slight variation of trimming between the top and bottom channels. The top channel does not fully suppress the X component out of the X*Y output as the bottom channel are. The distortion models also differ slightly between the two to mirror the behavior of the hardware measured.

Inputs

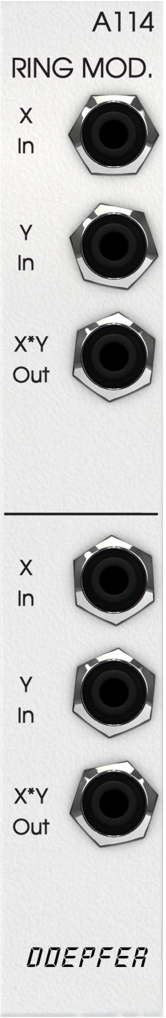
X in 1	input for audio signal X, top channel.
Y in 1	input for audio signal Y, top channel.
X in 2	input for audio signal X, bottom channel.
Y in 2	input for audio signal Y, bottom channel.

Outputs

X*Y out 1	audio output for the top channel.
X*Y out 2	audio output for the bottom channel.

In Use

A ring modulator is great for the production of bell-like sounds, alien voices, or just to produce new timbres out of exciting components.

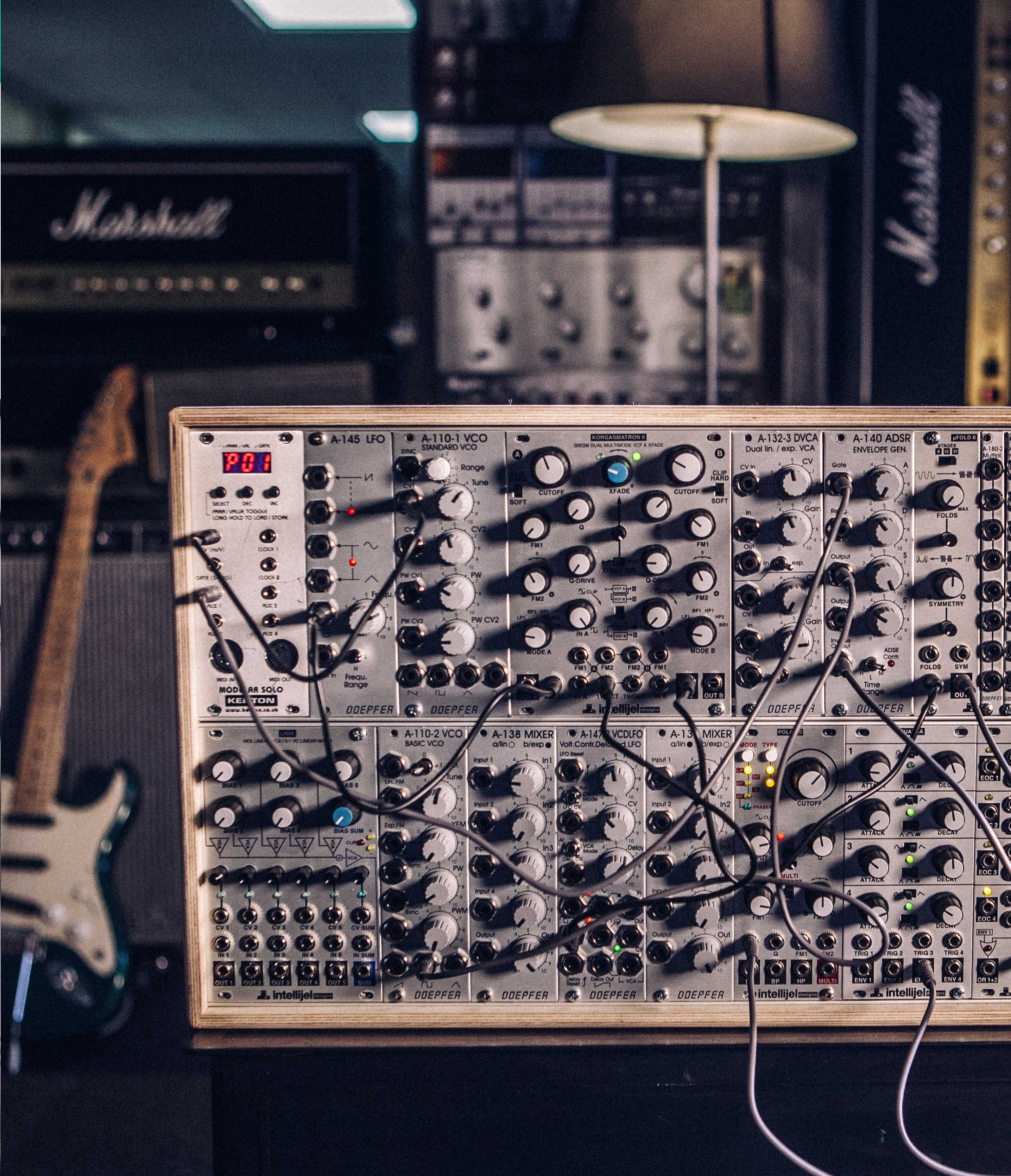


Intellijel

Eurorack brand Intellijel, run by **Danjel van Tijn** and associates, is based in Vancouver, British Columbia in Canada. These branded add-on modules from Intellijel are available for purchase at `http://softube.com/buy`.

The Intellijel modules are RUBICON THROUGH ZERO OSCILLATOR, KORGASMATRON II and the µFOLD II. These three modules all make a fine addition to the more traditional subtractive basic system of Softube Modular, as well as a step towards more West coast thinking in terms of synthesis.

For more information regarding the Intellijel modules, please see "29 Intellijel Rubicon, Korgasmatron II & uFold II" on page 200.



Marshall
P01
MODULAR SOLO
KENTON
A-145 LFO
A-110-1 VCO
STANDARD VCO
KORGASMATRON II
A-132-3 DVCA
Dual lin. / exp. VCA
A-140 ADSR
ENVELOPE GEN.
A-110-2 VCO
BASIC VCO
A-138 MIXER
DOEPFER
intellijel

ROLI Seaboard RISE Module

Interface the innovative keyboard controller **ROLI Seaboard RISE** with Modular and get access to five dimensions of touch.

Configuration

Please make sure that ROLI Seaboard RISE is set to its default settings. To load these, start the application ROLI DASHBOARD FOR RISE. Press the MENU button on the top right, and then select RESET in popup menu that appears.

The RISE's **Power/Mode** button on the lower left side should now be WHITE, to indicate the RISE is in MIDI mode. If your Seaboard is not in MIDI mode (the button has a different color), press the button repeatedly until it turns white.

Set up your DAW for working with the RISE as described in detail on ROLI's web site: `https://support.roli.com/article-category/rise-setting-up-with-other-software`

This is particularly important to obtain the correct behavior in Ableton Live as described here:

`https://support.roli.com/article/using-the-seaboard-with-ableton-live`

Parameters

Glide to Note	Turns on and off the RISE's ability to glide between notes. Turning this parameter OFF does not only disable the gliding zones beneath and above the RISE keybed, but also disables the vibrato (**Glide** expression) achieved by striking a note and then wiggling your finger sideways on the playing surface. Turning this parameter OFF is advisable for situations where you want the RISE to hit precise chromatic pitch values and don't want to glide between notes (or want to use the glide to control any other parameter).

Mode POLY ROTATE: Cyclic voice allocation, a new voice is assigned with each received MIDI note from RISE.

POLY LOWEST: Only one voice channel is triggered as long as only one note is played at the time. If one voice is playing (key is being held, gate is high), the next note added will assign voice channel two and so on.

MONO: In this mode, only channel 1 is triggered by playing RISE.

Indicators

Voice LED 1-4, All Indicates which voice is currently being triggered by your RISE. All is lit if any voice is active.

Touch Faders 1-3 Indicates the value set on the Touch Faders 1-3. These faders are active in RISE's MIDI mode (the **Power/Mode** button on the lower left on RISE is white) and sends out CV ranging from 0 to 5V via the jacks CV1-3 (see more on the CV 1-3 outputs below).

XY Touchpad Indicates the value set by the XY Touchpad on your RISE. These are connected to the X and Y outputs which send out a corresponding control voltage, ranging from 0 to 5V (see more on the X and Y outputs below).

Outputs

CV 1-3 These jacks send out CV, 0 to 5V, controlled by the **Touch Faders 1-3** on your RISE.

Pedal This jack sends out CV, 0 to 5V, controlled by an expression or sustain pedal connected to the RISE pedal jack.

X and Y These jacks sends out CV, 0 to 5V, controlled via the **XY Touchpad** on your RISE.

Voice Outputs 1-4 These are the polyphonic outputs for CV and Gate information. Voice assign set by the **Mode** parameter will affect the way these output jacks are assigned.

Note This is the note CV output (1V per octave) for each of the four voices. Connect this output to the main tracking input on your oscillator (such as CV1 on DOEPFER A-110). When the **Glide to Note** function is activated, this output is not only affected by the played MIDI key but also the RISE channel's glide.

Gate These jacks output a logical high signal (5V CV) when receiving a MIDI note on message (a key is struck on your RISE). At MIDI note off (key is released) it instantly falls back to zero (0V CV). Use this output to connect to a **Gate** input on an ADSR envelope module.

Strike This jack outputs 0 to 5V CV reflecting the velocity value (0–127) received from the RISE. The harder you strike the RISE keys, the higher CV output you will get here.

Press This jack outputs 0 to 5V CV when receiving pressure from the RISE. This means that Modular can receive polyphonic pressure via the RISE module.

Glide The **Glide** jack outputs positive or negative CV (1V per octave), corresponding to the horizontal upwards or downwards bend from the note played. The **Glide** jack is useful as a modulation source especially when the **Glide to Note** function is set to OFF.

Slide The **Slide** jack outputs CV matching the vertical slide movement performed on your RISE keys.

Lift This jack outputs CV matching the speed of release on a released key on the RISE. The faster you release a played note, the higher the CV output here will be.

All Gate The **All Gate** jack output sums the individual channel gate outputs to a single mono gate output. A logical high signal (5V CV) is sent when receiving a MIDI note on message (a key is struck on your RISE).

All Press	This jack outputs the average pressure CV from all the channels. This means that the more fingers used and the more pressure applied on the RISE's playing surface, the higher the CV output will be.
All Glide	The **All Glide** jack outputs positive or negative CV (1V per octave), corresponding to the average horizontal upwards or downwards bend from the notes played.
All Slide	The **All Slide** jack outputs CV corresponding to the average vertical slide movement performed on your RISE keys. One useful usage of the **All Slide** output is to control the filter cutoff in a paraphonic patch (a patch with multiple oscillators but only one filter). This makes it possible to control filter sweeps by moving all of your fingers vertically on the RISE.

In Use

When interfacing with the Seaboard RISE in Modular through Modular's RISE module, it's good practice to use SLEW LIMITER modules to soften the sudden shifts in CV out, which can otherwise result in audible clicks and undesirable behavior. We recommend that you study the RISE preset patches included among Modular's presets for inspiration.

The expressiveness of the RISE makes it possible to make really organic sounding patches using for example pressure, strike, glide, slide and lift. The **XY Touchpad** is really useful when controlling panning and volume at the same time. The **Touch Faders 1-3** are useful for example when creating drawbars on an organ or controlling volumes on different channels.

DAW and MIDI Interfacing

These modules are mainly for interfacing the Softube Modular with your DAW. It features DAW SYNC (MIDI to clock converter), MIDI TO CV (monophonic MIDI to note and gate converter), QUADRAPHONIC MIDI TO CV (four channel, polyphonic MIDI to note and gate converter) and MIDI TO TRIGGER (four MIDI notes to trigger outputs, for triggering of percussive sounds from individual MIDI notes).

DAW Sync

This module is an efficient way of getting clock pulses in sync with your DAW. Connect the 16 notes output to click in on your sequencer module and the reset pulse out to reset in for the most common application of this module.

Outputs

1/1	This jack outputs a short pulse on the first beat of each measure.
1/4	This jack outputs a short pulse on each beat (quarter notes).
1/8	This jack outputs a short pulse on each eighth note.
1/16	This jack outputs a short pulse on each sixteenth note.
1/32	This jack outputs a short pulse on each thirty second note.
Reset	Sends out a trigger pulse at the beginning of a bar in the DAW's sequencer.
Run	This jack sends out a high (+5v) signal, when DAW is running.

In Use

Typical use for the DAW SYNC module is to supply clocks for the sequencer modules. Connect for instance 1/16 output to clock in on the sequencer module and the **Reset** out jack on DAW SYNC to **Reset** in on the sequencer.

The **Run** jack can be used together the LOGIC TOOL module with for example a gate from a sequencer before sending to an envelope GATE IN. In this way, the ADSR gate input is sure not to be held high when DAW is stopped.

MIDI to CV

Your average workhorse MIDI to CV converter! This module converts incoming MIDI notes to monophonic **CV** and **Gate** outputs. There is also a trigger out for resetting oscillators with each keystroke if that's to your liking. Outputs for the most common performance MIDI controllers such as pitchbend, mod wheel, volume and aftertouch are also provided.

Parameters

MIDI channel This sets the MIDI to CV receiving channel. OMNI is for receiving on all channels, while numbers 1 to 16 reflects specific MIDI channels.

Transpose Transpose incoming MIDI notes, plus and minus one or two octaves.

Note Priority Set the note pitch priority when playing several MIDI notes at once. LOW PRIORITY means lower MIDI notes are selected over higher, while HIGH PRIORITY has the opposite behavior. LAST note priority simply means that the last received MIDI note is selected for output note pitch. Note that this only applies for note CV, not velocity CV, since velocity always has the last note priority.

Outputs

Note 1 volt per octave note CV output. Use this output to connect to your main tracking input on your oscillator (**CV1** on A-110 for example).

Pitch Bend This jack outputs +/- 1V CV (+/- 1 octave) when receiving MIDI pitch bend.

Gate This jack outputs a logical high signal (5V CV) when receiving a MIDI note on message (a key is stroke on your keyboard) from DAW. At MIDI note off (key is released) it instantly falls back to zero (0V CV).

Use this output to connect to a **Gate** input on an ADSR envelope module. When receiving several MIDI note on messages, the gate will remain high (at output 5V CV) until last note is released with a MIDI note off message (your key on your keyboard is released).

Mod Wheel This jack outputs plus 1V CV when receiving MIDI modulation (CC #01).

Trig This jack outputs a short logical high pulse (5V CV) when receiving a new MIDI note on message from the DAW.

Aftertouch	This jack outputs 0 to 5V CV when receiving a MIDI aftertouch message (channel pressure) from DAW.
Velocity	This jack outputs 0 to 5V CV reflecting MIDI velocity value (0–127). The harder you play on your keyboard, the higher CV output you will get here.
Vol	This jack outputs plus 0–5V CV when receiving MIDI volume (MIDI CC#07).

In Use

Note and **Gate** outputs can be used to hook up your basic monophonic synthesizer patch, as described in the first tutorial example (see above). **Trig Out** can be used to reset oscillators and LFOs, or retriggering of ADSR envelopes such as the A-140. Velocity and aftertouch outputs can be used to control filter cutoff modulation or amplitude.

Quadraphonic MIDI to CV

This is a four channel MIDI note to CV and gate converter for creating those four voice polyphonic patches that would cost a fortune in real life.

Parameters

Channel This sets the MIDI to CV receiving channel. OMNI is for receiving on all channels, while numbers 1 to 16 reflect specific MIDI channels.

Transpose Transpose incoming MIDI notes, plus and minus one or two octaves.

Mode This sets voice allocation mode. ROTATE is cyclic voice allocation, so that a new voice is assigned with every MIDI note received from DAW. LOWEST is voice allocation that assigns only voice channel one as long as only one note is played at a time. If one voice is playing (key is still being held, gate is high), next note added will assign voice channel two and so on.

Outputs

Note 1 volt per octave note CV output. Use this output to connect to your main tracking input on your oscillator (**CV1** on A-110 for example).

Velocity This jack outputs 0 to 5V CV reflecting MIDI velocity value (0–127). The harder you play on your keyboard, the higher CV output you will get here.

Gate These jacks output a logical high signal (5V CV) when receiving a MIDI note on message (a key is stroke on your keyboard) from the DAW. At MIDI note off (key is released) it instantly falls back to zero (0V CV). Use this output to connect to a **Gate** input on an ADSR envelope module.

Volume This jack outputs plus 1V CV when receiving MIDI volume (MIDI CC #07).

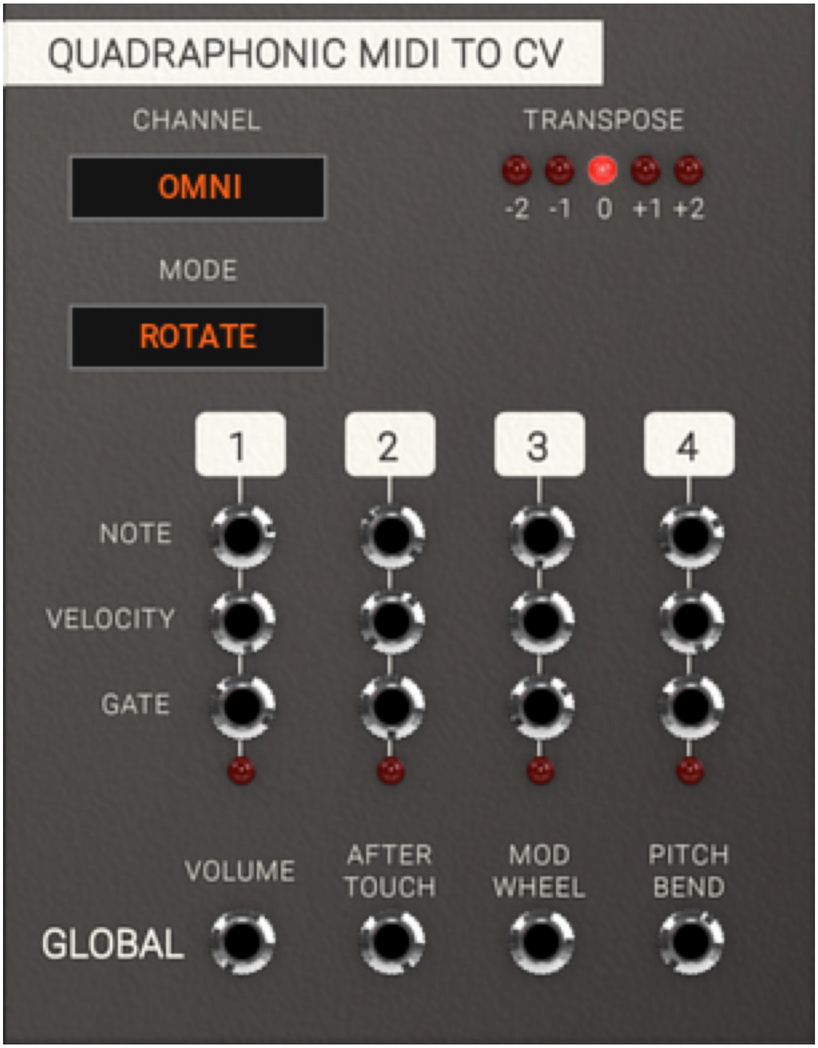
Aftertouch This jack outputs 0 to 5V CV when receiving MIDI aftertouch message (channel pressure) from DAW.

Mod Wheel This jack outputs plus 1V CV when receiving MIDI modulation (CC #01).

Pitch Bend This jack outputs +/- 1V CV (+/- 1 octave) when receiving MIDI pitch bend.

In Use

Connect the **Gate** outputs to four ADSR modules controlling VCAs and the **CV** outputs to four VCOs and you have the basics for a four voice synthesizer (see tutorial example `"Basic Polyphonic Synthesizer Patch" on page 30`. The QUADRAPHONIC MIDI TO CV can also be used together with the signal tools to create a simple four tone arpeggiator (see for example the preset *Seq Melodic Arp Mono*).



MIDI to Trigger

A four channel MIDI to trigger module, primarily made to trigger percussive sounds off specific MIDI notes. It emits short trigger pulses when a MIDI note is received on the set MIDI channel.

Parameters

Channel	This sets the MIDI to CV receiving channel. OMNI is for receiving on all channels, while numbers 1 to 16 reflect specific MIDI channels.
Note	This is the specific MIDI note which the MIDI TO TRIGGER will receive on this trigger channel. The **Activity LEDs** will blink when a MIDI note message is received on the corresponding channel.

Outputs

Trigger	This jack outputs a short logical high pulse (5V CV) when receiving a MIDI note on matching the one set on this channel.
Velocity	This jack outputs 0 to 5V CV reflecting MIDI velocity value (0–127). The harder you play on your keyboard, the higher CV output you will get here.

In Use

Set desired receiving MIDI channel and MIDI notes by clicking and dragging. **Trigger Out** emits short trigger pulses and **Velocity** outputs last known velocity CV, which is held until next MIDI note is received on the channel.

Effect

This category currently only contains the DELAY module, but will be definitely be expanded upon in the future.

Delay

This module delays an incoming signal up to 1000 ms (one second).

Parameters

Time	This knob sets the time between 1 ms to 1000 ms (one second).
CV Amount	This knob sets the CV Amount affecting the delayed time of the inputted signal.

Inputs

In	Use this input for the audio or CV you want delayed.
CV	This is input is used for control voltage of delayed time. Closely linked to CV Amount.

Outputs

Out	Output of delayed signal.

In Use

Use the DELAY module to delay any audio or CV signal. Connect it to a mixer and feedback it through a filter to create a classic tape style effect.

Short delays swept in time from a LFO can also be used to create flangers, chorus and such effects. Simple but powerful module.

Mixer Modules

These are mixers suitable for summing audio (exponential response) and CV (linear response) as well as a polarizing mixer suitable for subtractive mixing, but of course all mixers can be used for either type of signal. All mixer modules have the same controls.

Audio Mixer

Mixer module with an exponential response, making it suitable for mixing audio signals. This mixer uses a dB scale for its knobs, with a maximum gain of 12 dB.

CV Mixer

A no-nonsense CV mixer with a linear response, making it suitable for mixing CV signals. This mixer scales the signal up to 4 times.

Polarizing Mixer

A mixer module with a gain -1 to +1.

Parameters

Volume 1-4	These volume knobs set the amplification of input signal. from -∞ to 12 dB, from 0 to 4x, and from -1 to +1 in the different mixers.
Output Volume	Sets the main output mix amplification. Same range as the input **Volume** knobs.

Inputs

Input Jacks 1–4	The four input jacks for the signals to be summed

Outputs

Output	Mixed and amplified output of the four input signals.

Audio Mixer In Use

The AUDIO MIXER is of course preferably used for mixing audio signals, however nothing prevents you from using it for mixing of any kind of signals since there is no DC block on the input or output.

CV Mixer In Use

The CV MIXER is of course preferably used for mixing CV signals, however nothing prevents you from using it for mixing of any kind of signals.

Polarizing Mixer In Use

Use the POLARIZING MIXER for adding and subtracting signals. The POLARIZING MIXER is great to for example mix the outputs of the A-108 to create new and interesting filter characteristics. It is also useful for mixing control voltages or just about any kind of signals.

The Sequencer Modules

These are simple 16, 8 and 5 step note sequencers, as well as the BEAT SEQUENCER, a four channel trigger sequencer suitable for triggering percussive elements. All are driven by clock inputs that can be taken from the DAW SYNC module or whatever kind of pulse output.

Beat Sequencer

Classic x0x style sequencer for programming percussive sounds (for example with the HEARTBEAT modules, purchased separately). The sequencer can be looped or not. In LOOP OFF mode the **Link Out** sends a trig after last step which can be used to link together several sequencers. SWING sets the delay time of every other pulse input.

Parameters

Gate/Trig Buttons 1–16 These buttons determine whether or not a trig is outputted on the corresponding output jack.

Indicator LEDs 1–16 These LEDs indicate which steps are active.

Length This parameter sets the final step played in the sequence. Change the value by clicking on it to add one step at the time, or click and drag to set whatever value between 2 and 16 you want.

Loop This button determines whether the sequence will loop back after the last step or not. An unlit **Loop** button means no loop back, and that the **Link Out** jack sends a pulse after the last step has been reached and one more click has been received at the **Clock In** jack.

Swing This knob sets the delay time of each other pulse input. **Swing** is also dependent on reset pulse in, which can be used for interesting and surprising effects.

Inputs

Clock In By sending a short pulse or gate into this input jack, the sequencer will progress one step. The most common application of this jack is to connect it to **1/16 Output** of the DAW SYNC module.

Reset In By sending a short pulse or gate into this input jack, the sequencer will instantly reset to the first step. The most common use is to connect it to the **Reset Out** or **1/1 Output** of the DAW SYNC module.

Outputs

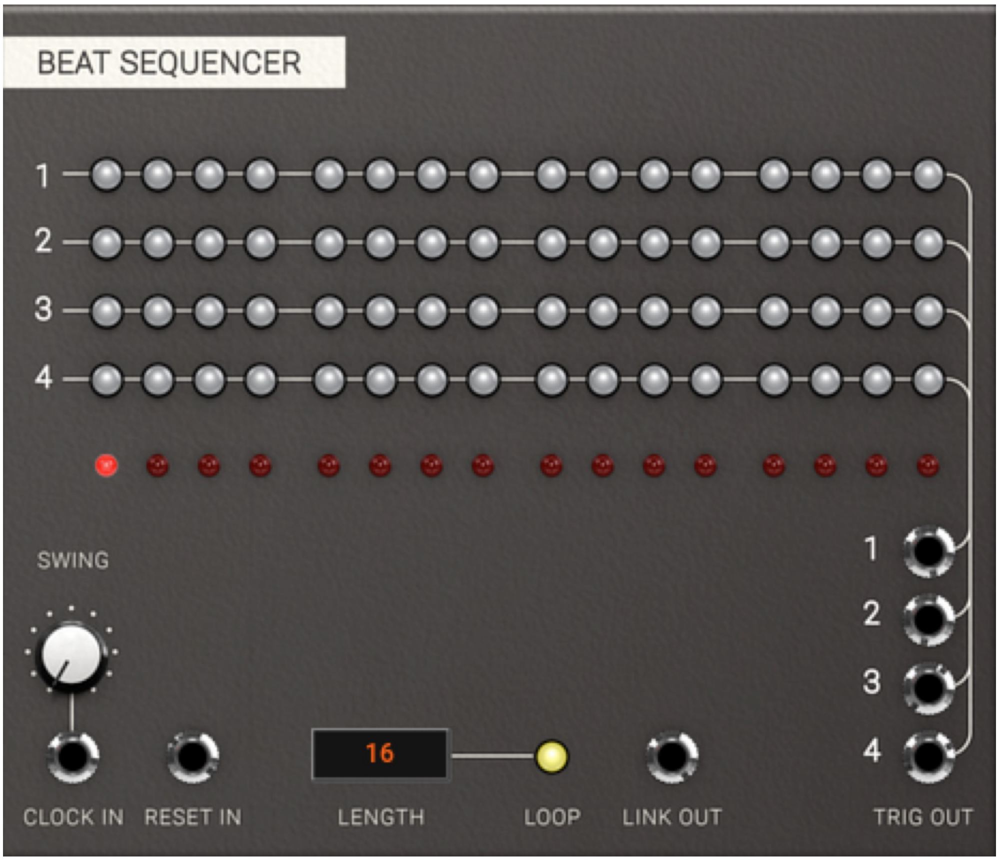
Trig Out This output jack sends short pulses for each activated step. This type of triggers is useful for triggering percussive elements, using for example the HEARTBEAT modules.

Link Out This jack outputs a short pulse after the last step has been reached when the module isn't in LOOPED mode and the sequencer receives another clock in. Use it to reset the next sequencer you want to link in order to expand your sequence beyond 16 steps.

In Use

Patch **Trig Outs** to **Trig Ins** on HEARTBEAT modules or to **Gate In** on the ADSR envelope you want to trigger.

To create accents, use one channel to control an envelope module and a VCA and run your sounds through that to control their dynamics. **Swing** means that every second clock pulse gets delayed, so if you for example use the 16th notes output from DAW sync to feed the clock input on the sequencer, only the steps with equal numbers (2,4, 6 etc) will be affected. This becomes apparent when programming pulses on all 16 steps.



Sequencer 16, Sequencer 8 and Penta Sequenza

Classic step sequencers with **CV Out** and **Gate Out**. Multiple gates after each other extends the gate output over several steps. A separate trig channel is provided as well as chromatically quantized CV scaling. Sequences can be looped or not.

Parameters

Indicator LEDs 1–16	These LEDs indicate which steps are active..
CV Knobs 1–16	These knobs set the amount of output CV at the **CV Out** jack when the corresponding step is active. CV AMOUNT is shown in volts down in the value display in the lower left side of the plug-in.
Gate/Trig Buttons 1–16	These buttons determine whether a gate and a trig is outputted on the **Gate Out** and **Trig Out** jacks when the corresponding step is active. Several consecutive gates creates legato notes, but separate triggers.
Length	This parameter sets the length of the sequence. Change the value by clicking on it to add one step at a time, or click and drag to set the value between 2 and 16.

Loop This button determines whether the sequence will loop after last step or not. If **Loop** is OFF, **Link Out** will send a pulse after a completed sequence.

Range This button switches the **CV Out** between limited range (1 octave) and full range (5 octaves). When activated, full clockwise position of the CV knobs will correspond to a one octave jump.

Quantize When activated, the **CV Out** jack will be quantized to the nearest semi tone. Use this function when using the sequencer to program note information.

Inputs

Clock In By sending a short pulse or gate into **Clock In**, the sequencer will progress one step. The most common application is to connect the **1/16** output of the DAW SYNC module to the **Clock In**.

Reset In Reset the sequencer step by sending a pulse to this jack. Most common application is to connect the **Reset Out** or **1/1** output of the DAW SYNC module to the **Reset In**.

Outputs

Trig Out Sends short pulses for each activated step that the sequencer passes when clocked. These kind of triggers are useful for triggering percussive elements, such as the HEARTBEAT modules.

CV Out Sends the programmed CV for each step. **Range** button and **Quantize** button affects the output CV information (see above).

Gate Out Sends a gate for each activated step that the sequencer passes when clocked. Several consecutive gates creates a combined long gate (legato).

Link Out This jack outputs a short pulse after the last step has been reached when the module isn't in LOOPED mode and the sequencer receives another clock in. Use it to reset the next sequencer you want to link in order to expand your sequence beyond 16 steps.

In Use

When linking more than one sequencer, use LOOP OFF mode, where the **Link Out** sends a trig after last step, which can be used to link together several sequencers. Use the **Link Out** trigger to reset the next sequencer, also in LOOP OFF mode, clocked from same clock source (see "Linking Multiple Sequencers Patch" on page 29). The sequencer can be used not only for note information, but also as a programmable modulation source.

The PENTA SEQUENZA is ideal for use as a quirky modulation sequencer. Using an odd number of stages for running a more traditional sequencer like the 16 step sequencer in parallel can create some really interesting polyrhythmic effects.

The Utility Modules

The Utility Modules is a collection of simple useful modules, such as a simple SINE OSCILLATOR, an ENVELOPE FOLLOWER, MULTIPLE, OFFSET, SLEW LIMITER, CLOCK DIVIDER, LOGIC and SIGNAL TOOL, as well as SAMPLE & HOLD and CV-controllable switches.

Sine Oscillator

This is a simple and mathematically correct sine oscillator with perfect tracking. It was originally created for testing purposes in the early stages of the development of Softube Modular.

Parameters

Pitch	This knobs sets pitch offset ranging from 0.1 Hz up to 10 kHz. The 12 o'clock position is a C at 64 Hz (this position is easily reached by ALT clicking on the pitch knob).

Inputs

CV in	This is a normal 1 V/octave tracking input jack. Connect note CV out from the MIDI TO CV module for musical use of the sine oscillator.
Sync	This is a reset input jack for syncing purposes. Similar to hard sync on other oscillators.

Outputs

Out	This is the main output jack.

In Use

The sine oscillator is great as a space efficient LFO or as FM operator in use with for example the INTELLIJEL RUBICON.

Envelope Follower

This module can be used to generate control voltages from audio sources, but also as a simple ASR envelope.

Parameters

Attack	This knob controls how quickly the envelope follower will react to a rising edge amplitude change.
Release	This knob controls how fast the envelope follower will react to a falling edge amplitude change.

Inputs

In	This is the input jack. Connect it to audio or gate signal from your MIDI to CV module if you're using it as a regular envelope.

Outputs

Out	This is the output jack. Resulting CV from the envelope follower is output here.

In Use

Use the envelope follower to, for example, create an auto-wah by letting audio controlling a low pass filter. The envelope follower can also be used as a simple ASR (Attack, Sustain, Release) envelope if it's triggered with a gate signal.

Multiple

This module multiples the incoming signal to four identical outputs.

Inputs

In	The input signal.

Outputs

Out 1 - 4	Four identical outputs for distribution of the input signal to four different sources.

In Use

The MULTIPLE can be seen as quite unnecessary in the Softube Modular since all outputs can be routed to any number of inputs, but it can be used for esthetic reasons, just to keep your patch tidy. It is also true to how many real life Eurorack modular systems look like when not using "stackables" (stackable cables).

Offset

The OFFSET module can be used both as a static CV source, but of course also to offset the input CV.

Parameters

Offset	This knobs sets offset between -10V through 0V to a maximum of 10V.

Inputs

In	Input signal to be offset goes here.

Outputs

Out	This is the sum of the input signal and the set offset. If a signal is not inserted at input, output will directly correspond to set offset.

In Use

Offsets can be used to set level threshold, for instance with the A>B function in signal tool. Offsets can also be used to control several CV controllable parameters at the same time and linked to a performance knob for "morph-type" operations.

Quad Offset

More or less the same as OFFSET, but with four channels contained within the same module to save space.

Parameters

Offset 1 - 4	These knobs set the offsets of the four channels between -10V through 0V to a maximum of 10V.

Inputs

In 1 - 4	Input signals to be offset goes here.

Outputs

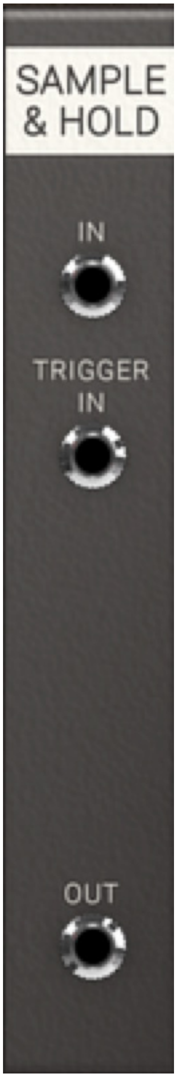
Out 1 - 4	These jacks output the sums of the input signals and the set offsets. If signals are not inserted at input, the outputs will directly correspond to set offset.

In Use

More or less the same areas of use as the single offset module.

Sample and Hold

The SAMPLE AND HOLD module listens to an incoming signal and samples the control voltage at a time set by the incoming trig pulse at the trigger input.



Inputs

In	Input signal to be sampled from. Can be any signal imaginable.
Trigger In	Pulse input. Input signal at the **In** jack is sampled at rising edge of a pulse.

Outputs

Out	Outputs the sampled and held CV signal. CV is held until next trigger pulse is presented at the trigger input.

In Use

The classic application of sample and hold is to create something called a "stepped random control voltage". By sampling white noise with a steady stream of pulse (preferably from a LFO) a "stepped random control voltage" is achieved, creating a sound famililar from robots and droids of the sci-fi flicks of the 1970ies. Another application is to trigger it at high frequency to create a bitcrusher (look at the preset "*Bitcrusher*" in the Modular FX for examples of this).

Slew Limiter

A slew limiter is a device that smoothes an incoming signal, limiting the maximum rate of change of the output voltage per unit of time. It is often called lag generator. It creates a *glide* or *portamento* when applied to a note CV signal.

Parameters

Time	This knob sets the slew time from 0 to 10 seconds.

Inputs

In	Input CV signal here is smoothed out.

Outputs

Out	CV output of the SLEW LIMITER.

In Use

Can be used on **Note CV** to create portamento or glide effects. Random bursts of white noise can be filtered into slowly evolving random CV changes.

Divider

This module outputs a clock pulse for every *x* pulses it receives. Use for example the **/3** output to trigger an output pulse for each third incoming pulse.

Parameters

Divide	Sets the division of the clock on the **Out** output.

Inputs

In	Clock pulse input.
Reset	Reset pulse output and all internal counters and clocks.
CV	CV control the clock division of the **Out** jack.

Outputs

Out	This is the programmable division output. It outputs pulses divided between 2 and 32 times and are affected by the **Divide** offset knob and the **CV** division input.
/2	Outputs a pulse for every 2nd input pulse.
/3	Outputs a pulse for every 3rd input pulse
/4	Outputs a pulse for every 4th input pulse
/8	Outputs a pulse for every 8th input pulse
/16	Outputs a pulse for every 16th input pulse

In Use

Clock division means that clock pulses arriving at the input are counted and output pulse will be generated for example every 16th pulse at the /16 output. The division module is most commonly used as clock divider, but can of course also be used as a suboscillator if connected with a pulse output of an oscillator.

Logic Tool

This module is used to perform simple logical operations on DC signals where signals over 2.5V equals a logical high state (1) and below 2.5V equals a logical low state (0). The module has two inputs with inverting logical outputs. The two inputs are compared in logical function truth tables and each logical function has its own output.

Inputs

In A	Signal input of channel A: the first operand.
In B	Signal Input of channel B: the second operand.

Outputs

Inv A	Inverted logical output of channel A. Output is high for inputs below 2.5 V, as indicated by the LED.
Inv B	Inverted logical output of channel B. Output is high for inputs below 2.5 V, as indicated by the LED.
AND	Output jack for logical AND function. This output is high when **both** channel A and channel B inputs are high.
NAND	Inverted function of the AND output jack.
OR	Output jack for logical OR function. This output is high when either channel A or channel B, or both inputs are high.
XOR	Output jack for logical XOR (exclusive OR) function. This output is high only when either channel A or channel B input are high.
NOR	Inverted function of the OR output jack.

In Use

To make a sequencer stop mid sequence, you can connect the DAW SYNCS **Run** signal to **A**, and **Gate Out** from the sequencer to **B**, and use the **AND** output to control an audio source. This means that both the **Gate Out** and **Run** must be active for **AND** output to become active.

Signal Tool

This module acts as a toolbox for continuous CV signals and does not differentiate between logic high and low like the LOGIC TOOL (see above).

Inputs

In A	Signal input of channel A, the first operand.
In B	Signal input of channel B, the second operand.

Outputs

Neg. A	Negative output of channel A. An input of for example 2V will give an output of -2V.
Neg. B	Negative output of channel B.
ABS A	Absolute output of channel A. An input of 2V and a signal of -2V will give same output of 2V.
ABS B	Absolute output of channel B.
Max	Outputs the maximum of channel A and B. If channel A has 2.5V and channel B has 2V **Max** will output 2.5V.
Min	Outputs the minimum of channel A and B. If channel A has 2.5V and channel B has 2V **Min** will output 2V.
A+B	Outputs the sum of channel A and B.
A-B	Outputs the difference of channel A and B.
A>B	Outputs logical high (5V) when A level exceeds the level of B.
Clip	Clips channel A at channel B's input level.

In Use

The signal tool has many uses, for example, it can be used to compare levels on incoming audio on Modular FX to create faux compressor type patches.

Switch 1 to 4

This CV controllable switch can be used to direct a CV source or audio signal to any of four different destinations.

Parameters

Select	Select which of the four outputs that will be connected to the input.
LEDs	Indicate which of the four outputs is active.

Inputs

In	Input jack of the switch.
CV	CV input for controlling the connected output.

Outputs

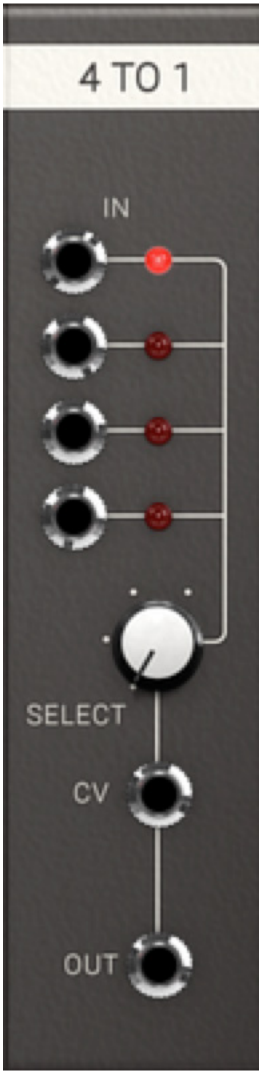
Out 1 - 4	Output jacks.

In Use

The switch 1 to 4 can have a lot of uses such as creating interesting modulation routes or pseudo-sequenced noises.

Switch 4 to 1

CV controllable switch that can be used to select one of the four connected CV or audio sources and send it to a set destination.



Parameters

LEDs	Indicate which of the four inputs is active.
Select	This knob is the offset for which of the four inputs will be connected to the output.

Inputs

In 1-4	Input jacks of the switch.
CV	CV input for controlling the connected input to the output.

Outputs

Out	The output jacks of the Switch 4 to 1 module.

In Use

SWITCH 4 TO 1 can be used to select modulation routes or change distributed clock divisions for your sequencer. On some of the presets for the Modular FX, the SWITCH 4 TO 1 module is used to turn on and off an effect, guitar-pedal style.

Dual Pan

Two channel, CV controllable panning module that can be used to place a mono signal in a stereo mix, or to use as an A/B switch. The panning curve uses a constant power pan law with a +3 dB boost at the edges.

Parameters

[pan indicator LEDs]	Indicates the panning balance for the left and right outputs of channel 1 and channel 2.
Pan knob	Sets the **Pan** offset on its respective channel. The overall pan amount is determined by the **CV** input and the **Pan** knob.

Inputs

In	Audio input jack of channel 1 and channel 2.
CV	CV input for external control of the pan of channel 1 and channel 2.

Outputs

L, R	The outputs of channel 1 and 2.
1+2	These are the combined outputs of channel 1 and 2. L outputs from channel 1 and 2 are mixed together in output L 1+2, and R outputs from both channels are mixed together in output R 1+2.

Dual Pan in use

The DUAL PAN module can be used to manually or automatically pan a mono signal in a stereo mix. A typical example would be two percussive sounds that the user wants to place individually in the stereo image. The summed output, **1+2**, can then be used to add the two stereo panned signals together.

You can also use the DUAL PAN as an A/B switch to send a CV or audio signal to different inputs.

Dual X-Fade

Two channel, CV controllable crossfade module that can be used to mix two mono signals with a CV controllable balance input. Crossfade curve response is linear with a -6 dB dip in the middle.

Parameters

[crossfade indicator LEDs]	Indicates the crossfade balance for the LEFT and RIGHT inputs of channel 1 and channel 2
X-fade knob	Sets the crossfade offset on its respective channel. The overall crossfade amount is determined by the CV input and the **X-fade** knob.

Inputs

L, R	Input jacks for the two signals to which the crossfade is applied.
CV	CV input for external control of the crossfade function.

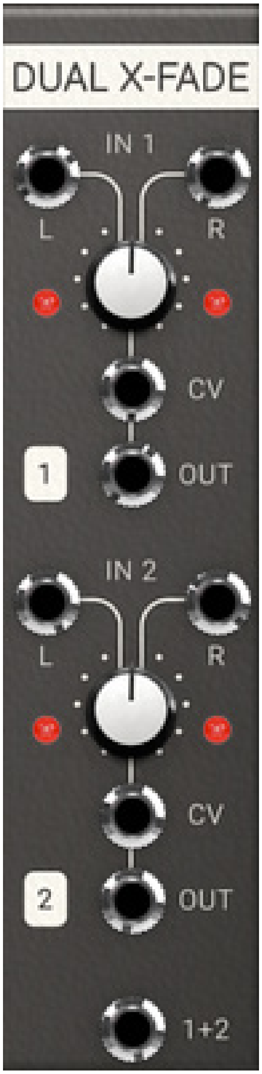
Outputs

Out	The outputs of channel 1 and 2.
1+2	This is the combined output of channel 1 and 2.

Dual X-Fade in use

The DUAL X-FADE module can be used for mixing different waveforms together, switching between sources, or modulating sources.

A typical use case would be an automated fade between two signals. Cross-fades can for be used as a pseudo-low pass filter when fading between outputs from the same oscillator, for example a sine or triangle on input 1, and saw or square on input 2. Then use an envelope or LFO to "sweep the filter" by patching the CV input.



Utility Quantizer

The Utility Quantizer module looks at input CV and outputs a quantized CV based on either a chromatic scale (top section) or a user definable scale (set on the bottom section). Chromatic scale means that each octave of 1V each is divided into 12 equal parts that corresponds to a chromatic 12 tone scale on a standard black and white keyboard.

Parameters

C to B buttons	In the bottom section of the Quantizer the user definable scale is found. Each button active (on) represent a note part of your user definable scale.

Inputs

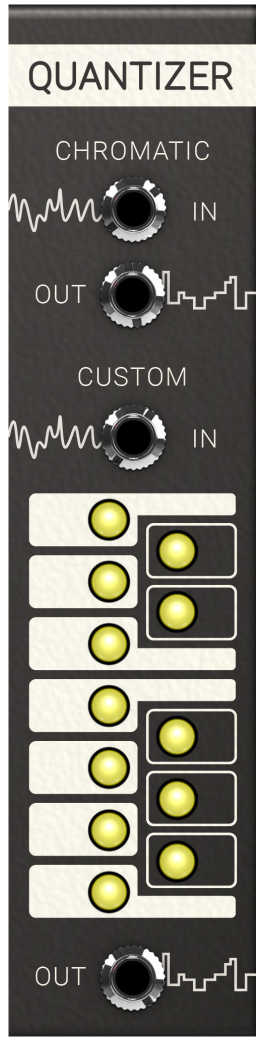
Chromatic CV in	This is the CV input that you want quantized to a chromatic 12 note scale.
User Definable CV in	CV input for the user definable scale part. Input any control voltage here that you want to be quantized to your custom scale.

Outputs

Chromatic CV out	This is the CV output of the 12 note chromatically quantized voltage.
User Definable CV out	CV output for the user definable CV scale. The output CV here is quantized to the notes active via the C to B buttons.

In Use

The Utility Quantizer is great fun! Use it for example to extract musical material out of the random output of the A-118 Noise module. Or apply it to any kind of sequencer in order to stay in the right (customized) key when you transpose the sequence.



Heartbeat Modules

Heartbeat consists of seven drum modules, and a dedicated two-channel one knob equalizer. These are the drum modules from Softube Heartbeat, and if you own and have installed Softube Heartbeat the modules are available right from the Modular.

The drum modules are the same as in Heartbeat, with the addition of input and output jacks. If you need more information on each individual module, please see the Heartbeat chapter in this user's guide.

Heartbeat Bass Drum 1

The Bass Drum 1 channel from Heartbeat—the drum synth with a soul! It's very versatile and can sound anything from analog boomy to acoustic dry. All the parameters that are velocity controllable in Heartbeat can be CV controlled in the Modular.

Heartbeat Bass Drum 2

Circuit modeled from a classic Japanese drum machine from the early 80s. The parameters that arevelocity controllable in Heartbeat can be CV controlled in Modular.

Heartbeat Snare/Rimshot

A mixture of carefully selected waveforms and synthesis. It is designed to do snare drum and rim shot sounds. All parameters that are velocity controllable in Heartbeat can be CV controlled in Modular.

Heartbeat Snare/Clap

It is a mixture of carefully selected waveforms and synthesis. It is designed to do snare drum and clap sounds. All parameters that are velocity controllable in Heartbeat can be CV controlled in Modular.

Heartbeat Hihat

A mixture of carefully selected waveforms and synthesis. It does hihats, both analogue sounding and acoustic. All parameters that are velocity controllable in Heartbeat can be CV controlled in Modular.

Heartbeat Percussion

The percussion channels from Heartbeat.

Heartbeat Cymbal

Purely generated by modeled analog synthesis and 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. All parameters that are velocity controllable in Heartbeat can be CV controlled in Modular. Please refer to the Heartbeat User manual for tweaking examples.

Original Heartbeat instrument plug-in.

Heartbeat EQ

Each mixer channel in SOFTUBE HEARTBEAT has a one knob equalizer, with tailor made EQ curves for each drum sound. The close integration between these equalizers and its drum is a large part of how to shape the sound from the drum, but instead of adding the EQ to each drum module we decided to make it a module on its own so that you can use it wherever you feel it works.

Parameters

Drum E	Boosts or cuts the frequency range where the tonal content of the instrument selected by mode switch is found. The neutral setting is at 12 o'clock, turn counter-clockwise to cut this frequency range and clockwise to boost.
[mode]	Selects which frequency bands that the EQ affects. For example, BD boosts or cuts the low frequencies of the BASSDRUM and so on. OFF is bypass mode.

Inputs/Outputs

In	Audio input.
Out	Audio output.

In Use

Use the Heartbeat EQ it to equalize your Heartbeat drum sounds or why not add some bass to your favorite synth sound? Each setting (BD, SD RIM, SD C, HH, CYM, PERC) has its own sweet spot. OFF is bypass.

MIDI Step Sequencer

The MIDI Step Sequencer is a sequencer containing 64 programmable steps. Each step can contain a note on pitch or rest, tie and accent. The inspiration comes from the legendary Roland SH-101 and TB-303 step sequencers, but with some twists of our own.

Getting started

1. Let's program and play back a simple sequence! Let's start off by adding and connecting the following modules as pictured above.

2. The MIDI Step Sequencer by default starts in record mode. This means that any MIDI notes played into the MIDI Step Sequencer will be recording into its memory and it will progress it one step for each note. Let's try inserting an C4 note by playing it on a MIDI keyboard connected to your DAW sending MIDI on the same channel as Modular is situated.

3. Dang! You played a C4 and hopefully also heard this reflected by the simple VCOVCA-envelope patch. You can also see that the step sequencer automatically progressed to position 2.

4. Let's add a rest on position 2, meaning that this step will be silent. Click on the Rest button to switch rest "on" and the sequencer will progress to position 3.

5. OK, now let's add a longer note linking together positions 3 and 4. Now let's first add a another C4 on position 3 by playing a note again.

6. Dang! Now, when on step 4 (the position indicator showing 4), click on the tie button which lits up and a T is lit in the lower part of the Note display. Then play another C4 to extend the C4 note from step position 3.

7. Now let's listen back. Click on the play mode button and the MIDI step sequencer will now be ready to step through the programmed steps when clock-information from the DAW sync module is fed to its clock input. Start your DAW and you'll hear and see your sequence of the two C4 notes being played back.

Now experiment with the other parameters – try inserting tie, accents and rest into your programmed sequence by clicking on record again, and scroll back and forth in your sequence by clicking on the position window (you can use both click or click+drag). Accent is a trigger out pulse for a second accent envelope or another external event (more details on accent below).

Parameters

Glide This is the amount of glide time added to the output CV when to steps are tied together with the tie function (see further description below). This creates a bend between notes of different pitch.

Pos This is the indicator and parameter for current position / step. In record mode this parameter can be used with click and drag in order to set current position to write to. Also, clicking on the display will progress it one step at a time. In play mode this parameter will only display played back position and is not clickable.

Channel This parameter set the MIDI at which the MIDI Step Sequencer receives its MIDI note information. Default channel is omni which is MIDI notes received on all channels simultaneously.

Direction This parameter tell the sequencer in which order the steps will be played back:

forward – Sequencer playback start from position one and progresses forwards until the last position (corresponds to Length) is reached, or until a reset pulse is received at the Reset in jack.

backward – Sequencer playback start from last position (corresponds to Length) and progresses backwards until the first position is reached. When a reset pulse is received at the Reset in jack, the sequencer will revert to last position.

pingpong – Sequencer playback start from position one and progresses forwards until the last position (corresponds to Length) is reached. When last position is reached another clock-pulse at the Clock in jack will progress the sequencer backwards until first position and then start over. When a reset pulse is received at the Reset in jack, the sequencer will start from first position.

random – Sequencer playback position is a random step within the loop length. When a reset pulse is received at the Reset in jack another random position is recalled.

Play / Rec These buttons changes between Playback mode and Record mode. In Playback mode, the MIDI Step Sequencer is waiting for clock triggers to enter the clock in jack in order to progress forward. In Record mode, the MIDI Step Sequencer is expecting the user to enter MIDI note information on the selected MIDI channel. In this mode, step position can also be scrubbed back and forth by clicking or clicking+dragging on the position display.

Tie When this button is activated, the previous step and current step will be tied together – this is called legato in musical language. When two steps are tied together the glide buffer is activated making it possible to "glide" from one note to another (also known as "glissando"). Tie is indicated written in the MIDI Step Sequencer memory by appearing as a "T" shown in the display on the steps it is written when being played back.

Accent When this button is activated, a short trigger pulse is sent out of the Accent out jack. This pulse can for example be used to trigger an envelope for extra punch in your basspatch. Accent is indicated written in the MIDI Step Sequencer memory by appearing as a "A" shown in the display on the steps it is written when played back.

Rest When this button is turned from off to on, a rest is entered into memory and the MIDI Step Sequencer will step forward to the next position (a rest is written into memory). Note that this is only for when the Rest button is turned from off to on, not the other way around. A rest entered into the memory of the MIDI Step Sequencer means that this step has a gate output value set to zero (in effect will not trigger any envelopes). When played back, the Rest is indicated written in the MIDI Step memory by appearing as a "R" shown in the display.

Note that Tie, Accent and Rest information previously inserted into your sequence step in discarded when entering a new note value into the same step of the sequence via MIDI.

Indicators

Pos display In this display window the currently played back position.

Note display This display shows information on the played back note related to the current position. Note is represented by a letter (C-B) and with an octave the suffix. A lit up "#" sign indicates a sharp note and a "-" sign is used in addition to show the two lowest octaves (C-2 to C0). The "T","A" and "R" indicators are lit to indicate an added Tie, Accent or Rest in a step position.

Inputs

Clock in A pulse received at this jack will progress the MIDI Step Sequencer one step forward when in play mode.

Reset in A pulse received at this jack will reset the MIDI Step Sequencer to the start position when in play mode. Note that start position will be different depending on which play mode (forward, backward, pingpong or random) is currently being used.

Outputs

CV out This jack outputs the note CV information on the currently selected step. CV notes values are discrete with the exception of notes linked together with ties where glide is applied.

Gate out This jack output a gate when a note is playing. If notes on adjacent steps are linked together with ties, the gate will remain high until the next step not containing a tie.

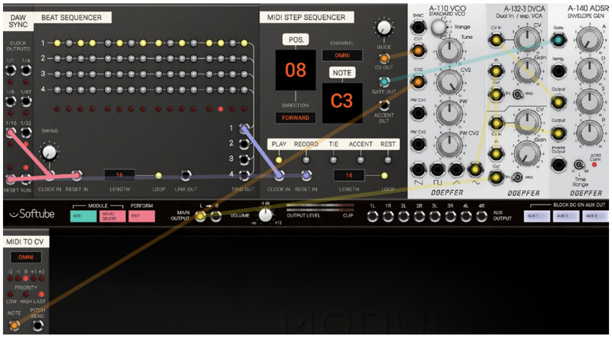
Accent out This jack will output a trigger when an accent is activated on a step.

The MIDI Step Sequencer module in use

The MIDI Step sequencer is of course great to use a no-frills programmed step-sequencer in the vein of the Roland-sequencers that inspired it, but here's a few more interesting applications to use it:

1. Pseudo arpeggiator

Use the MIDI Step Sequencer as a "programmed" arpeggiator, in this case a sequence that is being triggered by the timing information from the beat sequencer and transposed from the MIDI-to-CV module.



2. Polyphonic offset

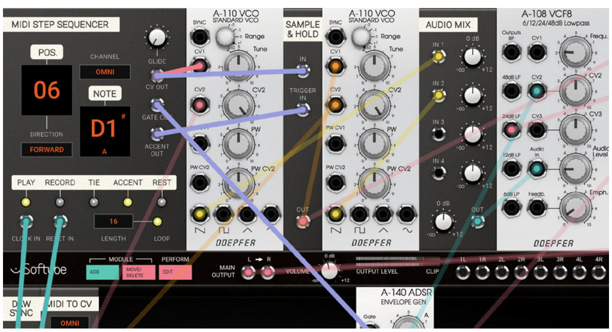
Use a MIDI Step Sequencer as programmed polyphonic offset to your incoming notes per voice in a polyphonic patch. This can be observed in detail in the preset called "Quad Ring Seq" (required Mutable Instruments Rings):

3. Quantized Random Memory

Insert a programmed scale of your choice into the MIDI Step Sequencer and use that as a random source of pitch CV to accompany your melody line. Fun and creative!

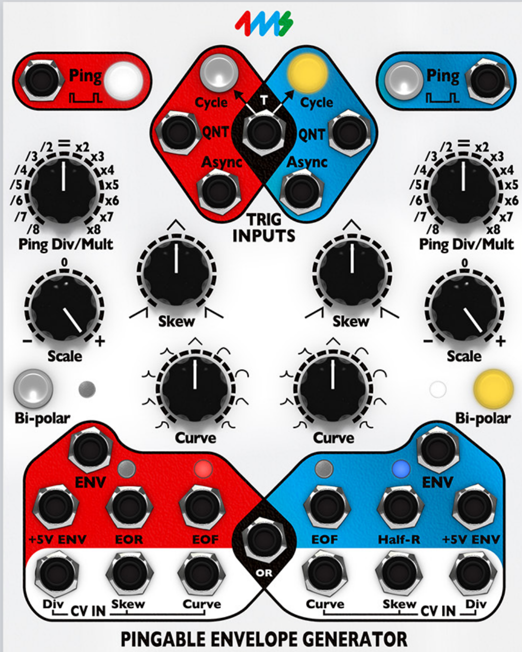
4. Accented Duophony

Use the MIDI Step Sequencer's accent functionality to trigger a second oscillator's sample and hold and separate envelope. In this way a neat musical “cat-and-mouse-chase” is created between the oscillators where one mirrors the other every now and then. Great for creating sequences that alternates between unison and harmonies. An example is found in the preset called “MIDI Step Duo”.



Credits

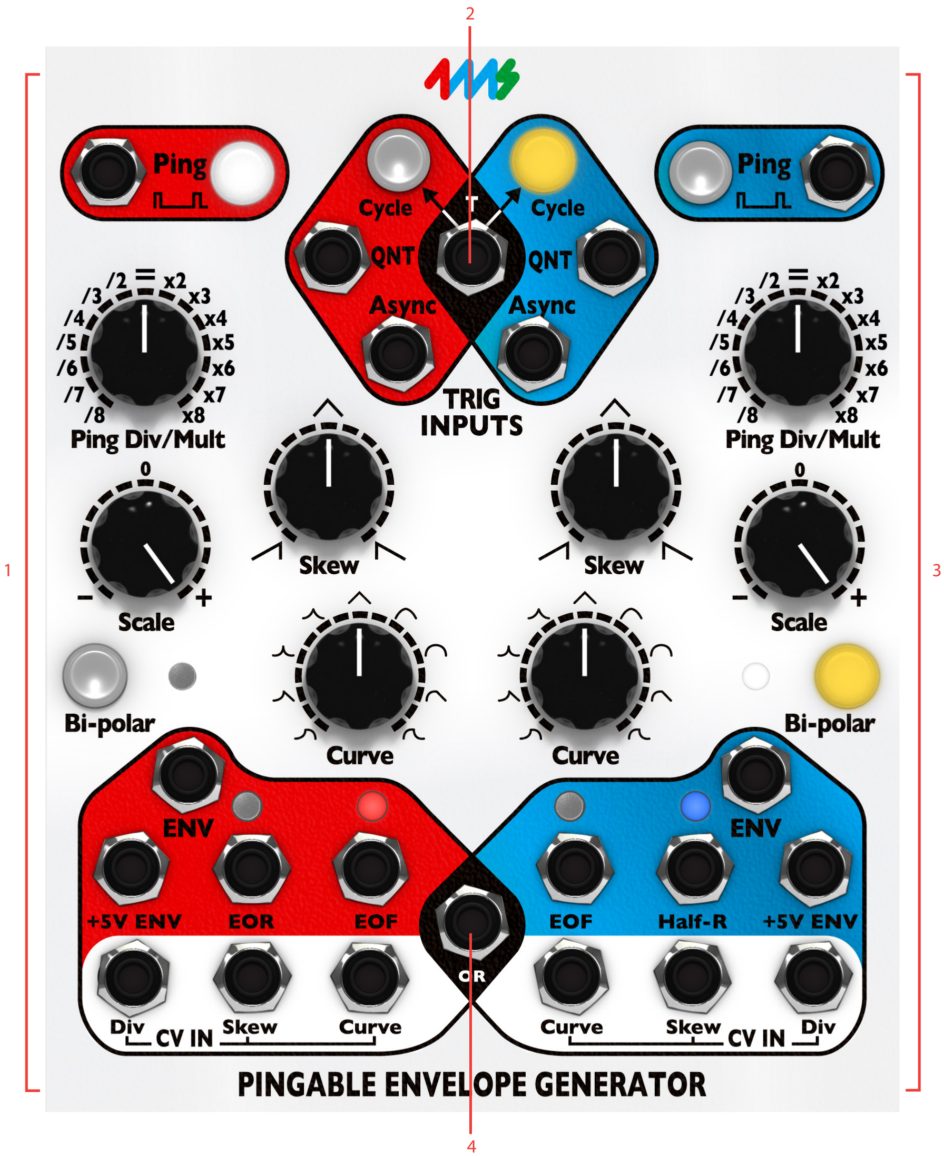
Oscar Öberg – concept, modeling, sound design. **Kristofer Ulfves** – concept, marketing, presets, user's guide. **Björn Rödseth, Kim Larsson** – modeling. **Arvid Rosén** – modeling and validation. **Patrik Holmström** – graphics programming. **Niklas Odelholm** – graphic design, programming, model validation. **Paul Shyrinskykh** – quality assurance. **Torsten Gatu** – programming. **Henrik Andersson Vogel** – marketing, user's guide. **Danjel van Tijn** – hardware design and model validation. **Dieter Doepfer** – hardware design and model validation. **Chris Assall** – model validation. **Kabuki** – presets.



2 4ms Pingable Envelope Generator (PEG)

The 4ms Pingable Envelope Generator (PEG) is a dual envelope generator whose envelope lengths are set by the time between external clock pulses or "pings". The dual envelopes are of simple two-stage, rise-and-fall design, and can be used in cycle mode as LFOs. When gated it can also serve as a simple rise-sustain-fall (or ASR) envelope, commonly used in monophonic synths of the 70s.

Overview



1. Red channel
2. Toggle for both channels
3. Blue channel
4. Combined output

Description

PEG consists of two identical envelope channels, Red and Blue. They can be used independently or via the mixed (OR) output.

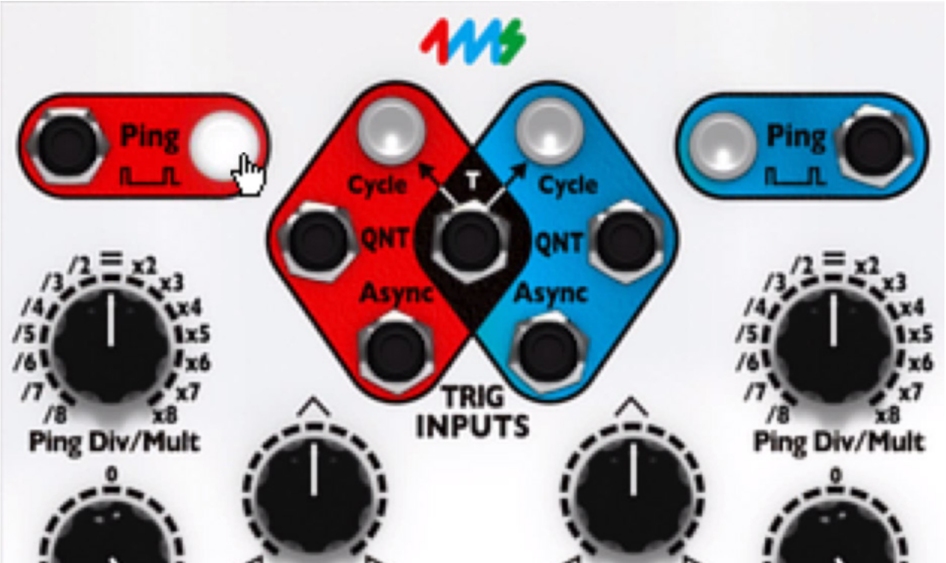
In PEG, CV control over envelope shape, skew and ping (clock) division/multiplication per envelope channel (Red/Blue) is possible. PEG features a multitude of different triggering and cycling options (AD, AR, quantization, cycle, cycle toggle) as well as different level stage trigger outputs (EOR, EOF and Half-R).

Each of the envelope channels also features a Tap tempo button for manual pinging.

The firmware features used in the PEG firmware version 4.3 are all emulated by Softube except the 2 second tap clear feature and System Mode (factory presets are being used).

Getting Started With PEG

1. Set the tempo of an envelope channel by repeatedly clicking on one of the ping buttons at the top of the PEG module. "Pinging" an envelope channel will set the internal tempo for that envelope and determine its overall length. An envelope channel can also be "pinged" by patching an external clock into its Ping input jack.



2. Use the pinged envelope channel as a rise-sustain-fall (same as an ordinary ASR) envelope by patching a gate into the Async jack. Now, when a gate appears on the Async jack, the rise-portion of the envelope will play through and the envelope will then stay at its maximum level (i e sustain level) until the gate is released. When the gate is released, the fall-portion of the envelope will be initiated and played through.

3. PEG does not normally produce sound on its own, so patch the PEG Envelope CV output to the whatever module you want to modulate. This could be the pitch of a oscillator, the cutoff frequency of a filter or a voltage controlled amplifier (VCA), just to mention the most common use-cases.

4. The output envelope amount can be adjusted by using the Scale knob. It can also be used to invert the output envelope CV.

5. Experiment with different envelope lengths by "pinging" the envelope at different tempos. You can also change the speed by turning the Ping Div/Mult knob to set the envelope to be up to 8 times faster or slower then pinged tempo. Changing the Curve or Skew of the envelope will not affect the overall rise and fall time, only the balance between the two.

6. Use the cycle function by clicking on the Cycle button at the top if you want the envelope to keep on repeating independently. This is great when you want to use PEG as an LFO.

Parameters

Red Tap Button

This is the manual tap tempo button for the Red envelope channel. Manually "ping" the red envelope channel to a certain tempo by repeatedly clicking on this button. The button will flash in sync with tapped tempo.

Note: Pinged PEG tempos are saved in a Modular patch, but the phase difference between the two envelope channels (or other PEG modules) will not be saved. This is to avoid other (in our opinion worse) side effects of saving the phase and this can make patches using cycle sound different when loaded than compared to when they were saved.

Red Cycle Button

This turns Red envelope cycle on and off. When cycle is enabled the envelope will keep on triggering on every ping.

Blue Cycle Button

This turns Blue envelope cycle on and off. When cycle is enabled the envelope will keep on triggering on every ping.

Blue Tap Button

The manual tap tempo button for the Blue envelope channel. "Ping" the red envelope channel manually in a certain tempo by clicking on this button. The button will flash in sync with tapped tempo. If more than two clicks are received, the average time between the three latest clicks will set the pace of the envelope (unless the third click occurs more than 50% different than the timing period of the first two clicks).

Red Ping Div/Mult Knob

The Div/Mult knob sets the offset division or multiplication of the pinged tempo of the Envelope. If this knob is set to "=" (preset value), no division or multiplication takes place. Division makes the output envelope times longer (up to 8 times longer) and Multiplication makes the output envelope times shorter (down to a 1/8 of pinged timed).

Blue Ping Div/Mult Knob
The Division / Multiplication offset for the Blue envelope generator (see description above).

Red Skew Knob
The Red Skew knob sets the proportion between the rise and fall times of the Red envelope generator. All the way counter-clockwise the rise portion is the dominant (longer) part. With the Skew knob set at 12 o'clock both portions are equal and with the Skew knob set all the way clockwise the fall portion becomes the dominant (longer) part.

Red Skew Knob
The Blue Skew knob sets the proportion between the rise and fall times of the Blue envelope generator in a similar manner as the Red Skew knob (see description above).

Red Scale Knob
This knob sets the envelope amount and polarity output at the ENV (red) output jack. With the Red Scale knob set all the way counter-clockwise, the ENV (red) output will output a fully inverted (negative) envelope shape (-10V), while the knob set at 12 o'clock (0), will produce no envelope output at the ENV (red) output jack. Fully clockwise (preset position) will produce a full positive envelope output (+10V) at the ENV (red) output. The +5V ENV (red) output jack is not affected by the Red Scale knob and will always output a fully positive envelope shape.

Blue Scale Knob
The Blue Scale knob has a similar behavior as the Red Scale knob but will of course affect the ENV (blue) output jack instead (see description of the Red Scale knob above). The +5V ENV (blue) output jack is not affected by the Blue Scale knob and will always output a fully positive envelope shape.

Red Bi-Polar Button
When engaged, this button centers the Red envelope around 0V with a approximate span of -5V to +5V.

Red Curve Selection
This knob selects the desired envelope-curve for the red envelope from a plethora of 17 different exponential, linear and logarithmic waveforms. As you turn the knob clockwise the first 4 curves are asymmetrical with exponential attacks and different decays, while the next 4 curves are symmetrical, ranging from exponential to logarithmic in 4 interpolated steps. Middle curve is linear (triangle wave). Curves 10 to 13 are symmetrical, ranging from linear to logarithmic, and the last 4 curves are asymmetrical with logarithmic attacks and varying decays (logarithmic to exponential).

Blue Curve Selection
This knob selects the desired envelope curve for the blue envelope with a similar function (see description above).

Blue Bi-Polar Button
This button centers the Blue envelope around 0V with an approximate span of -5V to +5V.

Tip: Use envelope Cycle mode if you want a PEG channel to work as an LFO.

Indicators

Red Ping LED
This LED button indicates the pinged tempo of the Red Envelope with tempo alterations set by the Red Div/Mult knob.

Red Cycle button LED
This LED button indicates whether or not the Red Envelope is in cycle mode or not (see further description above).

Blue Cycle button LED
Indicates whether or not the Blue Envelope is in cycle mode or not (see further description above).

Blue Ping LED
Indicates the pinged tempo of the Blue Envelope with tempo alterations set by the Blue Div/Mult knob.

Red Bi-Polar button LED
This LED button indicates if Bi-Polar mode engaged for the Red Envelope or not.

Red Envelope LED
This LED lamp reflects the output of the +5V ENV (red) jack and thus does not reflect amount or polarity of the ENV (red) output.

Blue Envelope LED
This LED lamp reflects the output of the +5V ENV (blue) jack and thus does not reflect amount or polarity of the ENV (blue) output.

Blue Bi-Polar button LED
indicates if Bi-Polar mode is engaged for the Blue Envelope or not.

Red EOR LED
When the red envelope finishes the rise portion of its cycle, the EOR (red) output goes high (+5V) and this LED is lit.

Red EOF LED
When the red envelope finishes the fall portion of its cycle, the EOF (red) output goes high (+5V) and this LED is lit.

Blue EOF LED
When the blue envelope finishes the fall portion of its cycle, the EOF (blue) output goes high (+5V) and this LED is lit.

Blue Half-R LED
The Blue Half-R LED is lit when the blue envelope has gone through half (50%) of its rise portion.

Inputs

Ping (Red) Input
Short pulses sent into this input "ping" the red envelope channel and set the internal tempo for it. Faster pulses sent here means shorter envelope, and pulses further apart will effectively result in a slower envelope. Tempo is derived from the time between the two latest pulses.

Ping (Blue) Input
Works similar to the ping (red) input, but obviously affects only the blue envelope.

QNT (Red) Input
This is the quantized input for the red envelope. Quantized means that any pulse that appears here will not trigger the envelope until the next "ping" is indicated on the red envelope channel LED. If gated (this input held high, at +5V) for a period longer than two pings in succession, the envelope will repeat (similar effect to cycle mode).

T (Toggle) Input
When a gate is applied (this input held high, at +5V)

to this input, both envelope channels' Cycle buttons will toggle state, on to off and vice versa. When the gate is released they will revert to their previous state. The "T" jack is useful for toggling between the two channels: set one channel in Cycle mode and the other channel to non-cycling while using the output from the OR jack (see the outputs section below). The "T" jack is also useful for turning both channels on/off at the same time.

QNT (Blue) Input
Blue envelope channel quantized input. Works similar to the QNT (red) input described above but for the blue envelope channel.

Async (Red) Input
When a trigger is received at this jack, the red envelope will trigger or re-trigger immediately. Holding a high gate (+5V) on this jack causes a sustain period followed by a fall period when the gate goes low (ASR envelope).

Async (Blue) Input
Work similar to the Async (red) input jack.

Div (Red) CV Input
A CV signal patched to this input will add to the value set by the Red Ping Div/Mult knob. Range is -5V to +5V.

Skew (Red) CV Input
CV control over the Red envelope Skew. This will add to the value set by the Red Skew knob. Range is -5V to +5V.

Curve (Red) CV Input
CV control over the Red envelope Curve. This will add to the value set by the Red Curve knob. Range is -5V to +5V.

Div (Blue) CV Input
A CV signal patched to this input will add to the value set by the Blue Ping Div/Mult knob. Range is -5V to +5V.

Skew (Blue) CV Input
CV control over the Blue envelope Skew. This will add to the value set by the Blue Skew knob. Range is -5V to +5V.

Curve (Blue) CV Input
CV control over the Blue envelope Curve. This will add to the value set by the Red Curve knob. Range is -5V to +5V.

Outputs

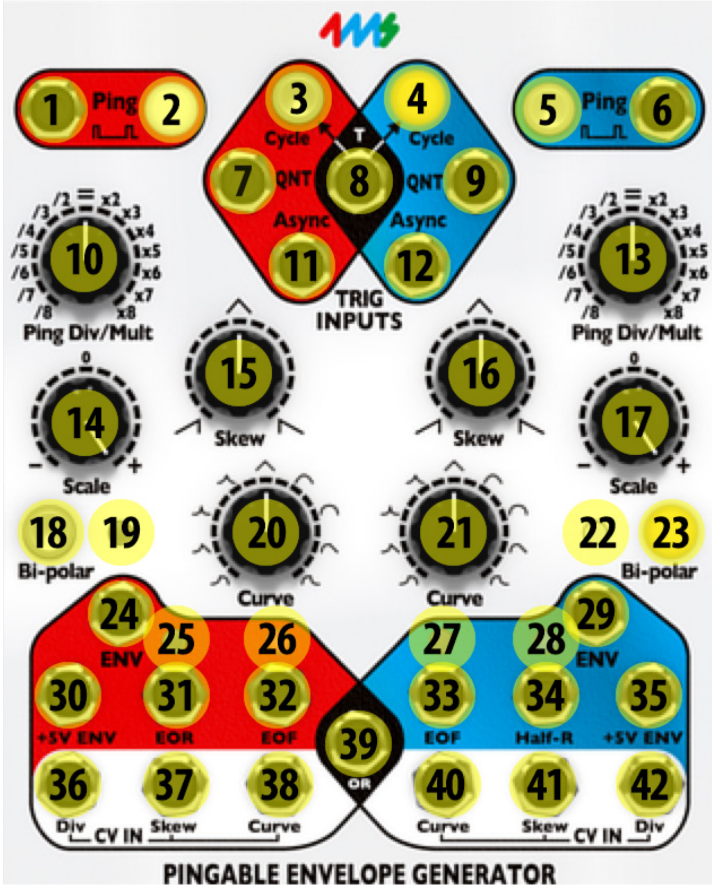
ENV (Red) Output
This is the main envelope CV output of the red envelope channel. The CV amount output here is controlled by the Red Scale knob and Red Bi-Polar button. Range is -10V to +10V.

ENV (Blue) Output
This is the main envelope CV output of the blue envelope channel. The CV amount output here is controlled by the Blue Scale knob and Blue Bi-Polar button. Range is -10V to +10V.

+5V ENV (Red) Output
This is the unregulated envelope CV output of the red envelope channel. Range is always 0V to +5V.

EOR (Red) Output
This jack output +5V at the end of the rise portion of the envelope when the fall portion is initiated. Note that this jack does not output +5V during sustain when gating the Async input jack.

EOF (Red) Output
This jack outputs +5V at the end of the fall portion of the red envelope or when the rise-portion is in progress.



Parameters, Indicators, Inputs and Outputs

1. Ping (Red) Input
2. Red Tap Button
3. Red Cycle button LED
4. Blue Cycle button LED
5. Blue Tap Button
6. Ping (Blue) Input
7. QNT (Red) Input
8. T (Toggle) Input
9. QNT (Blue) Input
10. Red Ping Div/Mult Knob
11. Async (Red) Input
12. Async (Blue) input
13. Blue Ping Div/Mult Knob
14. Red Scale Knob
15. Red Skew Knob
16. Blue Skew Knob
17. Blue Scale Knob
18. Red Bi-Polar Button
19. Red Bi-Polar Button LED
20. Red Curve Selection
21. Blue Curve Selection
22. Blue Bi-Polar Button LED
23. Blue Bi-Polar Button
24. ENV (Red) Output
25. Red EOR LED
26. Red EOF LED
27. Blue EOF LED
28. Blue Half-R LED
29. ENV (Blue) Output
30. +5V ENV (Red) Output
31. EOR (Red) Output
32. EOF (Red) Output
33. EOF (Blue) Output
34. Half-R (Blue) Output
35. +5V ENV (Blue) Output
36. Div (Red) CV Input
37. Skew (Red) CV Input
38. Curve (Red) CV Input
39. OR (Black) Output
40. Curve (Blue) CV Input
41. Skew (Blue) CV Input
42. Div (Blue) CV Input

EOF (Blue) Output
This jack outputs +5V at the end of the fall portion of the blue envelope or when the rise-portion is in progress.

Half-R (Blue) Output
This jack outputs +5V when half of the rise portion of the blue envelope has been played through. Note that this jack also output +5V during sustain when gating the Async input jack.

+5V Env (Blue) Output
This is the unregulated envelope CV output of the blue envelope channel. Range is always 0V to +5V.

OR (Black) Output
The OR jack will output the highest voltage value from either the red or blue ENV jack at any given moment. One way to use this is to think of the OR jack as a mix out, and use the Scale knobs as level knobs and the Bi-Polar buttons to bring down the relative level of a channel (kind of like a mute button).

The PEG Module In Use

The 4ms Pingable Envelope Generator is a very versatile module with many fun uses. Here are some suggestions:

1. CV controllable length of the PEG cycle - Use a CV-controllable LFO or VCO to ping the tempo of PEG. By varying the pitch of the driver the length of the PEG envelope can then be easily be altered, for example via an additional LFO or sequencer.

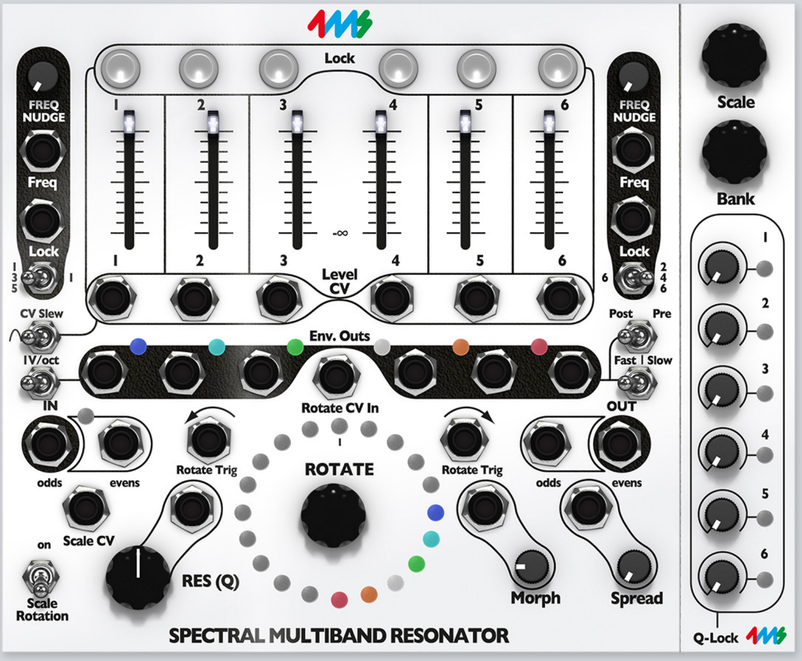
2. Ratcheting envelopes - Clock one envelope module from the DAW sync clock module with a steady pace, for example quarter notes. Use another faster input to clock the QNT input while an LFO is sweeping the Div input. The pace of the faster input, for example 32nd notes, will set the upper limit to the ratcheting speed. If you desire a more controlled approach to ratcheting, using a CV to turn on and off the cycle function is the way to go.

3. DAW synced LFOs - Use the DAW sync module to ping a cycling PEG at a regular pace, for example by using the 16th notes output. The reset jack or whole note jack should also be connected to the Async input to ensure PEG to start at the same phase each time (or each bar when using the whole note output). Using the Tri (preset) shape will be the most common use case when using PEG as an LFO, but of course you can use any shape to create your own customized LFO.

4. PEG Percussion - PEG is a great module to use for percussive sounds. Use the Skew and Curve knobs to shape the pitch bend of the tuned percussion and the Scale knob to set pitch bend amount. Experiment with different curve types for different sounds.

Credits

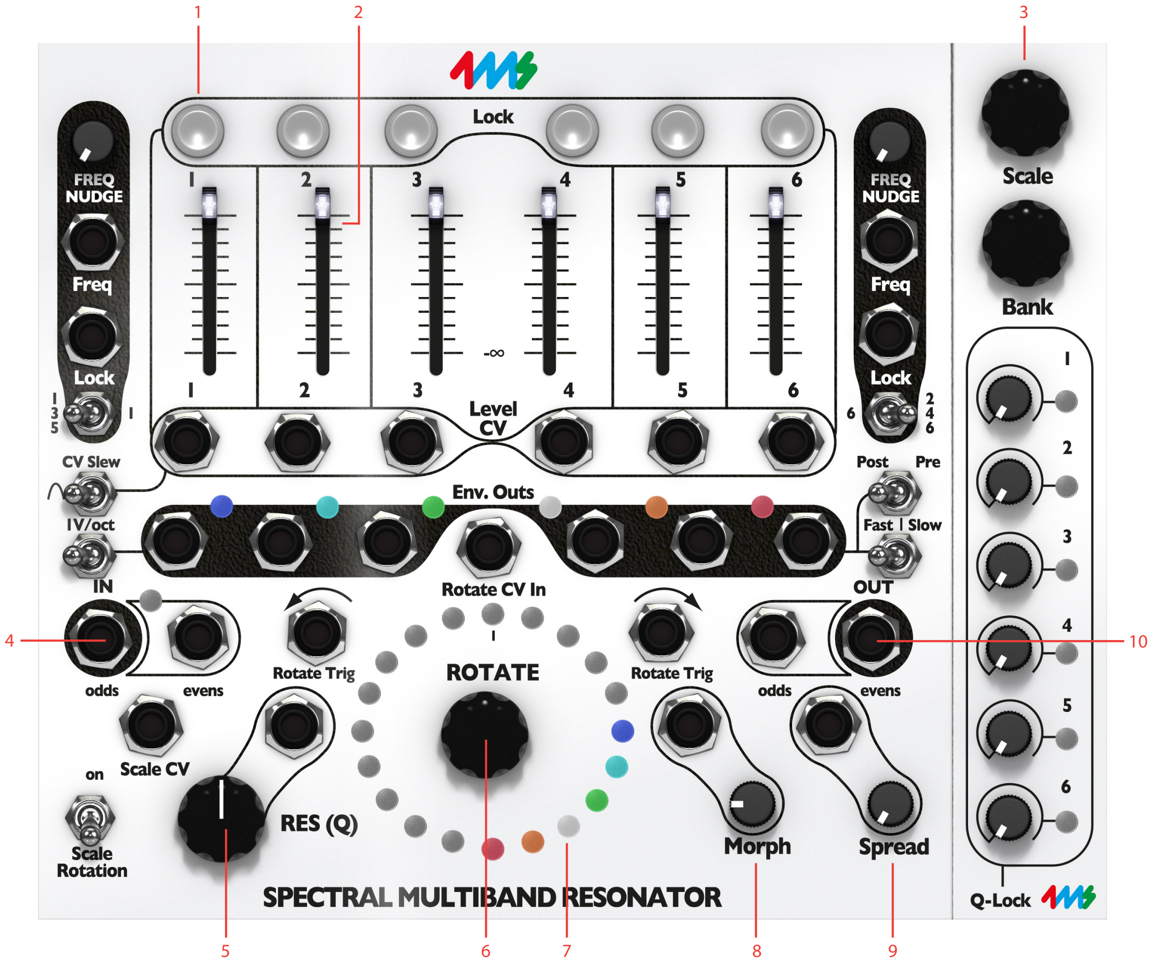
Oscar Öberg – modeling, project management, validation. **Kristofer Ulfves** – presets, validation, user manual. **Igor Miná** – user manual layout, hardware photos. **Niklas Odelholm** – GUI graphics.



3 4ms Spectral Multiband Resonator (SMR)

THE SPECTRAL MULTIBAND RESONATOR FROM **4ms** is an innovative multiband resonant filter which can process audio like a classic filter bank, ring like a marimba when plucked/struck, vocode, re-mix tracks, harmonize, output spectral data, quantize audio to scales and much more.

The SMR consists of 6 channels of resonant bandpass filters, all individually locked to a 20 notes scale displayed in the centre of the unit via the LED ring. All 6 filters can easily be swept around their present note scale manually or via CV. Notes scales cover everything from western scales, alternative tunings to more esoteric *Gamelan* and *microtonal* scales. Each filter's output volume can be controlled via CV and each



1. Filter band Lock buttons
2. Filter band Volume sliders
3. Expander section
4. Inputs
5. Resonance
6. Rotate
7. LED ring
8. Morph
9. Spread
10. Outputs

channel also has an envelope follower that can be set to output different kinds of modulation and pitch CV.

The official firmware v4 of the original 4ms hardware was mainly used when modeling the SMR behavior, the exception being the added optional 1 V/oct switch functionality of firmware v5 for the envelope follower outputs (see detailed description below) and the lack of user definable scales. The experimental non-tracking filter type is also not featured in this version.

User Interface

The interface might look daunting at a first glance, but is not that difficult. The most important dials, buttons and jacks are explained in the overview below.

Color Schemes

The SMR consist of 6 bandpass filters. Each filter-band is identified by a number and a color, which is shown on the LED ring. Each filter can be set to almost any frequency, except that it has to be within one of the available scales. More on that below.

Filter-band 1 is always deep blue, band 2 cyan, band 3 green, band 4 white, band 5 orange and band 6 red. This color reference is also shown on each filter-band's Env Out LED indicator.

The LED ring has a second function: It switches function temporarily when operating the scale and bank knobs, giving information about the scales' relation to the different filter-channels.

Description of Banks and Scales

The SMR is all about scale rotation. The LED ring is an important indicator to watch in order to understand the state of each of the different filter channels in relation to scale rotation. Rotation can be either clockwise or counter-clockwise.

When rotation occurs, all six bands will move in the same direction and the six lights on the light ring will fade to the next available free spot. If a band is locked it will not rotate, and that band will instead be indicated by a blinking light corresponding to that band in the LED ring. If a band is rotated such that it is intent to land on top of a locked band or on another band moving in the opposite direction (which occurs when spread is increased), it will then move past the locked or moving band to the next available free spot on the other side.

Rotation is closely linked to **Morph**, which sets the pace at which rotation can happen. No matter how much rotation you tell the SMR to do, it will only go as fast as the Morph setting allows. Rotation is also queued, so if you quickly turn the **Rotate** encoder ten notches to the right and have a very slow Morph setting, the SMR will start rotating all the channels one step at a time until it's rotated a total of ten times. Each of the ten rotations will happen at the Morph speed. Reversing direction will clear the queue, so if you want the SMR to stop rotating, you can turn the **Rotate** encoder one click in the other direction. The SMR will forget the remaining rotations in the queue and just rotate once in the new direction.

Getting started with SMR

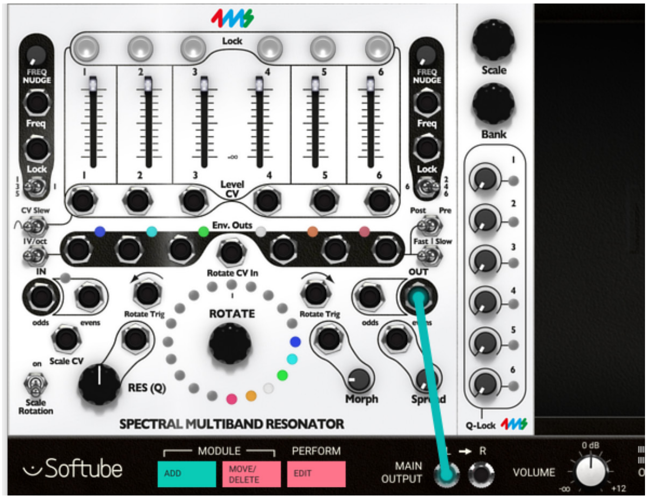
Here are a couple of guides to quickly get you started with the SMR.

Tutorial 1: Droning

Preset patch "**Exp SMR tutorial 1**"

The easiest SMR patch is making droning chords. Only one cable is needed! Set up the SMR as follows:

1. Add the SMR module to your empty rack by clicking on the **Add** button in modular.
2. Connect a cable from the **Evens** output jack (marked black on panel) to you main output.
3. Turn up the **Res (Q)** knob to 100%.
4. At the top of the SMR are the six **Volume** sliders. Each one controls the volume of a filter/resonator channel. Push down all six sliders, then slide each one up, one at a time. You should hear a pitch fade up in volume. Each slider controls a different frequency.
5. As you adjust the slider, watch the LED ring. Each slider is associated with a color. The color gets brighter on the LED ring as the slider moves up and down. Each of the 20 spots on the LED ring is associated with a pitch (a frequency, or a musical note).
6. Now turn the **Rotate** encoder clockwise by clicking and dragging upwards. Hear how the pitches shift up and the lights rotate one step clockwise. Click and drag the **Rotate** encoder downwards so it rotates counter clockwise. Hear how the pitches shift down and see how the lights rotate counter-clockwise.
7. Keep on spinning the **Rotate** encoder so that the lights move past 12:00 (due north, marked by a tick mark on the light ring). Now notice how the pitches go to the top and then start at the bottom. By rotating and pushing different sliders up and down, you can make different chords.



Connect the output of the SMR to the Main Output of Modular.

8. Play with the **Morph** knob as you continue to turn the **Rotate** encoder knob. **Morph** will set the speed at which rotation happens. With **Morph** at 100%, when you turn the **Rotate** encoder, the SMR will slowly fade from one position to the next. With **Morph** at 0%, the SMR will instantly jump from one position to the next (this can sound "clicky").
9. Now try playing with the **Spread** knob, which controls the number of empty positions in between each channel in the scale. With **Spread** at 0%, the channels occupy adjacent spots on the LED ring. As you turn **Spread** knob up slowly, the channels will jump from having one position between them, then two positions, then three and so on. As the spacing increases, the channels will get pushed around and eventually the highest channel will wrap around the lowest.
10. Try out playing with the **Q (Res)** knob. The SMR has a digital noise source normalized to the audio IN jacks. With the **Q (Res)** knob set to 0%, you should hear a filtered version of this noise. Adjust the slid-

Trigger the SMR with the Beat Sequencer and use a high Q value to create a rhythmic "ping".

ers, use the **Rotate** encoder knob and **Spread** knob to bring in different bands of noise. Notice that, as you increase the Q value by turning the **Q (Res)** knob, the output slowly changes from filtered noise towards pure sine waves. This is the effect of a tight bandwidth (Q) or resonance where only very select frequencies from the noise source are allowed to pass through to the output. If you have a favorite noise module (or any complex sound source), try running it into the input jack(s).

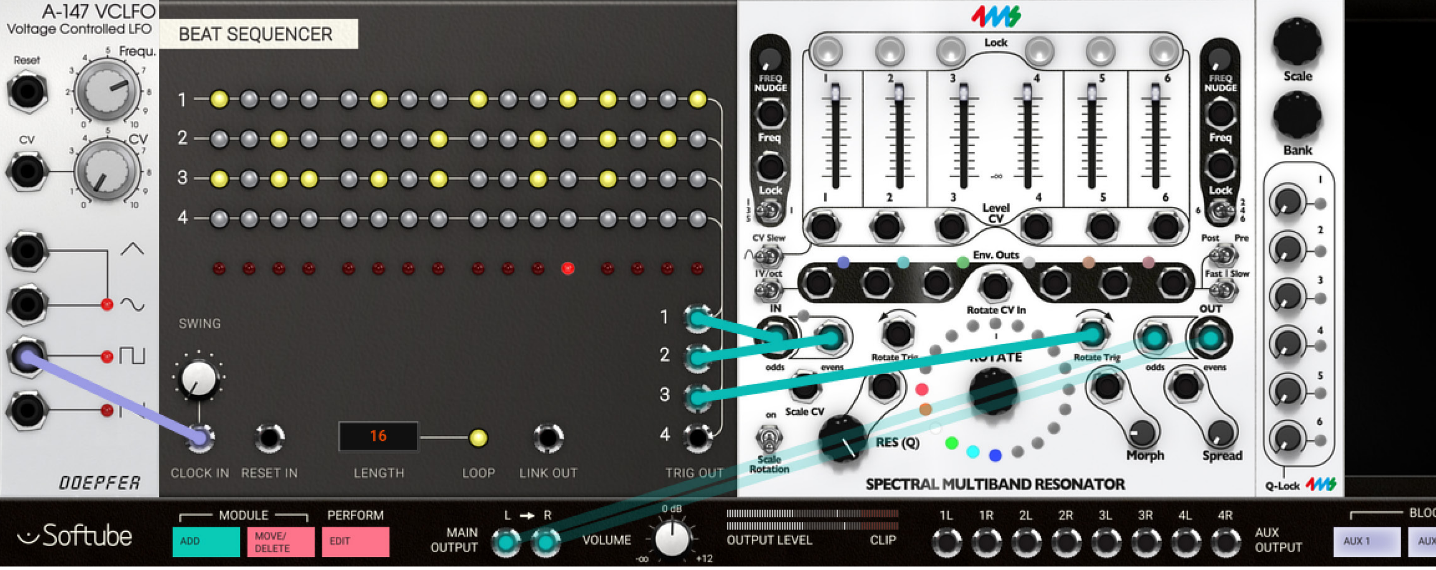
11. Finally, try running a 1V/oct melody line from a sequencer into one or both **Freq** jacks. Flip the **135 | 1** and **246 | 6** switches to select which channels are tracked, and which stay steady.

Tutorial 2: "Pinging" the Filter to Create Percussive Sounds

Preset patch "**Exp SMR tutorial 2**"

This is a great example on how to create resonant percussive sounds. You need a trigger or gate source from a MIDI TO CV or SEQUENCER module.

1. Patch the **Evens** output jack to main output left or, for stereo output, also patch the white **Odds** output jack to the main output right.
2. Patch your trigger sources into the **Odds** and **Evens** in jacks. Two sources with different rhythmic content is ideal, however if you only have one trigger source, just patch into the **Odds** in jack. In the preset example we have used the BEAT SEQUENCER driven from a LFO.
3. To start, have the triggers firing about once or twice per second. The odds IN jack goes to channels 1, 3, and 5. The evens IN jack goes to channels 2, 4, and 6. So each trigger source can strike a chord of three notes.
4. Listen to how the channels resonate when struck with a trigger or gate, they should sound like a gong or marimba. Adjust the **Q (Res)** knob to change how "ringy" the sound is: low Q is like a wood block, high Q is like a large bell. The amplitude of the triggers also effects the sound, try attenuating or boosting the triggers before they reach the SMR. Notice that the SMR will be struck on both the rising and falling edge of a gate, so two sounds will be heard for every gate. Triggers will cause only one sound since the rising and falling edges are very close together. Adjust the **Rotate** encoder and **Spread** knob to change the pitches of the notes, and play with the sliders to adjust the level of each note.
5. Input another trigger source into the **Rotate Trig** clockwise or **Rotate Trig** clockwise jacks. If you don't have another trigger source, you could input an LFO or envelope (0 to +5V) into **Rotate CV In** jack and the rotation will track the waveshape of the



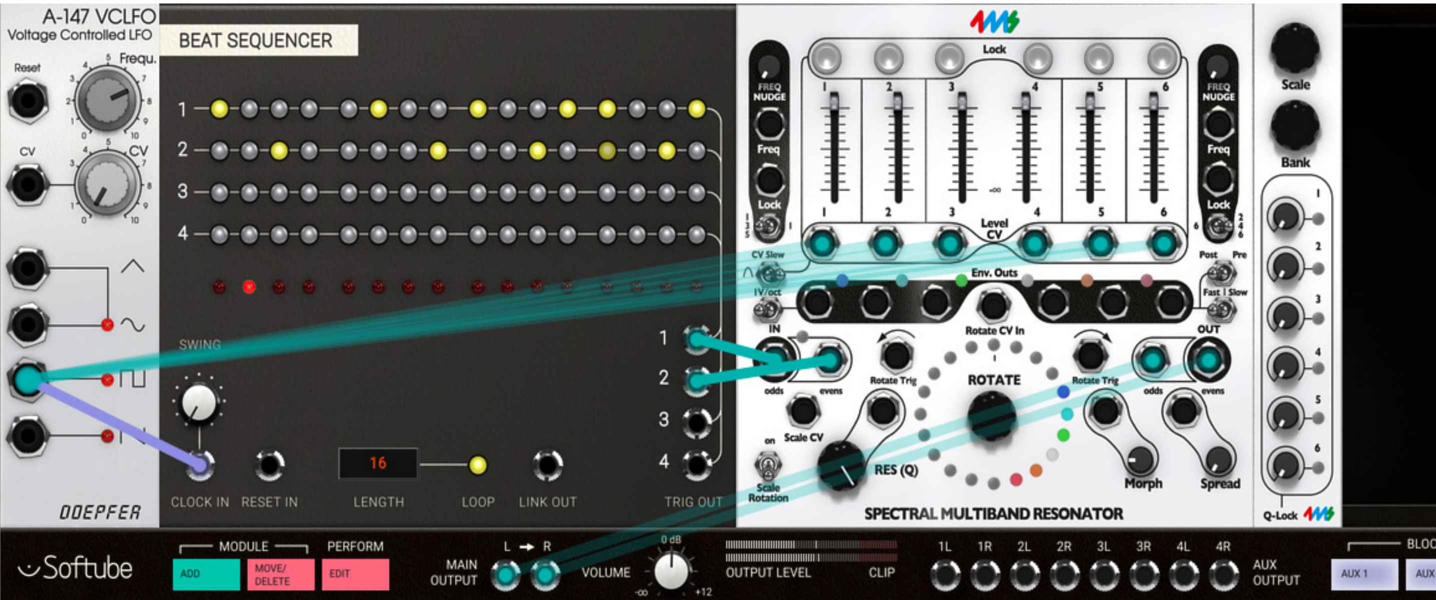
Play around with Rotate and Spread to change the pitches.

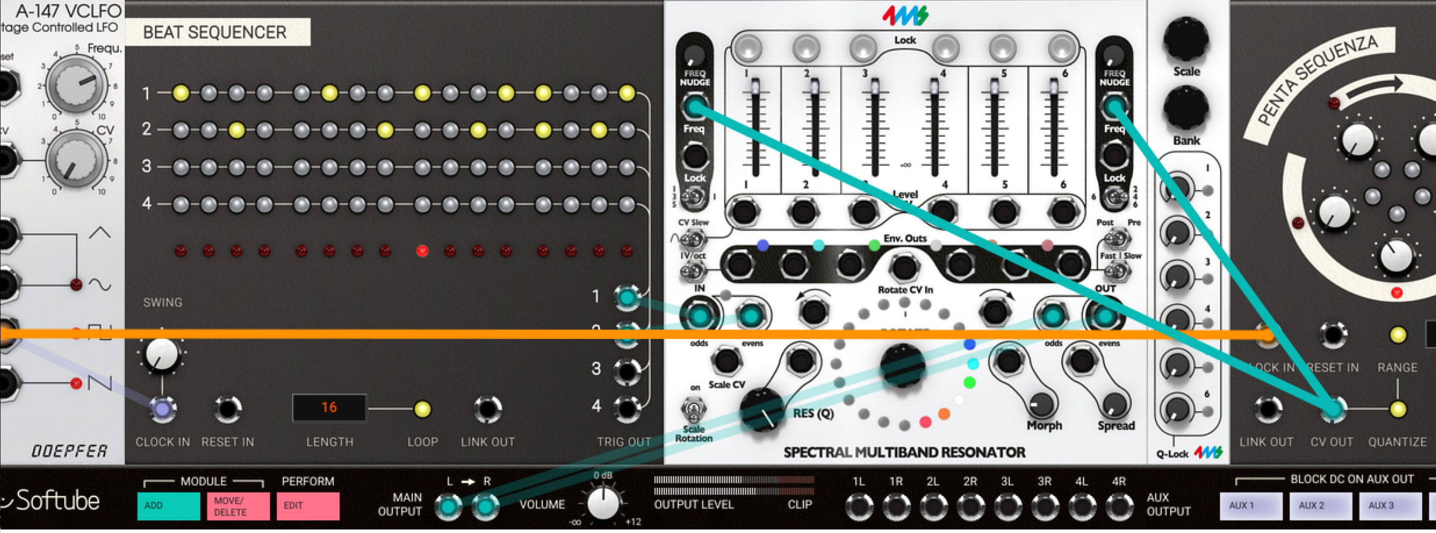
incoming CV.

Play with slowing down the rotation from the external triggers by increasing the **Morph** knob value. Try setting the rotation trigger source super fast and then turning **Morph** knob up and down to adjust the speed of the motion. **Morph** is a powerful way to limit musical movement (and it can be CV controlled).

6. Try out clicking on the **Scale** knob in the expander section, and hold down the mouse button. This will display the currently selected scale and bank by one or many blinking LEDs in the upper half, and a solidly lit lower half of the LED ring. Notice how the LEDs change: the bottom six LEDs are one color and there should be one LED that's slowly blinking colors in the top half of the ring. This is the currently selected scale. Click and turn the **Scale** encoder and the blinking LED will move: each position in the top half of the ring represents a different scale (there are eleven for every bank). Listen to how each scale is different. If you like, patch CV into the **Scale CV** input jack to control the scale selection with another

Use an LFO to set the channel volumes, adjust the CV Slew to get smooth out the CV waveforms.





Use an external sequencer to have a melody line track the pitches.

module.

7. Run an LFO square wave CV into the **Level CV** jacks. Now you can try flipping the **CV Slew** switch to the left to smooth out the click that occurs when a gate snaps up or down. Setting **Morph** higher when you flip the **Slew** switch will cause more slewing to happen, which means very fast CV in the **Level CV** jacks will be rolled off.

8. Try running a fast decaying envelope into one or both **Freq** jacks. Time it to the trigger sources to get membrane drum sounds. Or try a 1V/oct melody line from a sequencer to track the pitches. Try flipping the **135|1** and **246|6** switches to select which channels are modulated/tracked, and which stay steady.

Tutorial 3: Basic Remixing

Preset patch for Modular FX "**Exp SMR Tutorial 3**"

Remixing audio tracks often involves highlighting certain instrumentation and getting rid of others. This can be done by boosting and cutting frequency bands with the SMR and this how it is done:

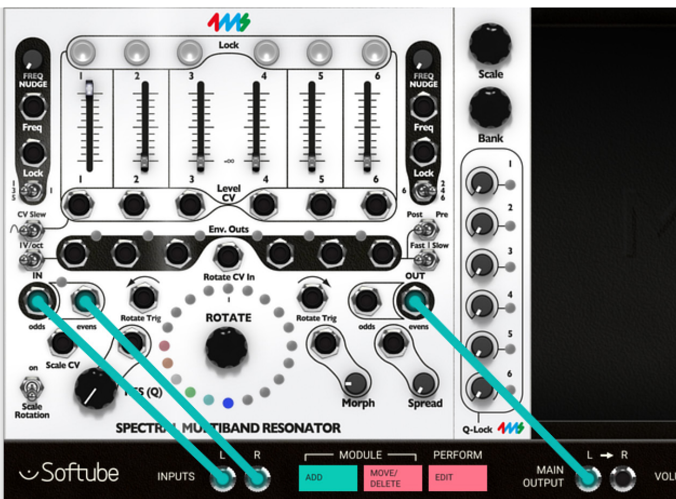
1. Input a audio signal into the **In** jacks (use the **Odds** input jack for mono, or both jacks for stereo).

2. Take the output from the **Out** jacks to your mixer (use the **Evens** jack for mono, or both jacks for stereo).

3. Turn **Res (Q)** knob all the way down.

4. Click on the **Bank** encoder and the bottom six LEDs should change color, each color represents a different bank. Keep holding down the mouse button and keep changing by dragging until the you've selected the deep blue bank. Any bank will work, but the deep blue bank is nice for re-mixing because the scales are set to common graphic EQ frequencies.

5. Slide channel one's **Volume** slider up and the rest of the sliders down. Turn the **Rotate** encoder to rotate



Basic remixing. Connect the audio from the DAW to the SMR's inputs.

channel 1 up and down the frequencies while you listen to the audio. Keep rotating until you find the frequency band you want to boost. Adjust the **Res (Q)** knob to narrow the band and add resonance if you wish. You may also want to adjust the left **Freq Nudge** knob to fine tune the frequency.

6. When you find the frequency setting you want, tap the channel's **Lock** button above the **Volume** slider. This will lock the rotation as well as the **Freq Nudge** setting. If you also want to lock the Resonance setting, adjust the top **Q-lock** knob on the expander section to the right hand side. The **Lock** button above the **Volume** slider will be lit if the channel is locked and on the expander section, the Q-lock LED will be lit if **Q-lock** is active (**Q-lock** knob is above zero). If you don't lock the Q, then the channel's resonance will be controlled by the **Res (Q)** knob and jack along with all the other channels with unlocked Q.
7. Repeat the process (steps 5-7) for channels 2 to 6. Note that for channels 2, 4, and 6 the **Freq Nudge** knob on the right is used, while for channels 1, 3, and 5, the **Freq Nudge** knob on the left is used. Make sure the **135 | 1** switch is set to 135, and the **246 | 6** switch is set to 246 (otherwise the respective **Freq Nudge** knobs will only control channels 1 and 6).
8. After setting all the channels to a useful frequency, adjust the mix of the six sliders. Play with adjusting the Q of each channel of the expander panel. You can range from essentially a re-EQ'ed track to harmonizing, resonant bowl sounds. Since each channel has a different Q value, both sounds can coexist.
9. To take the next step, consider running LFOs, envelopes, or clock gates into the **Level CV** jacks. Consider rotating some channels while keeping others still. Or run an LFO or envelope into the **Rotate CV** input. Play with **Morph** to make the rotation slower or faster. Another idea is to run a slow LFO into the **Freq** input, to create a filter sweep effect.

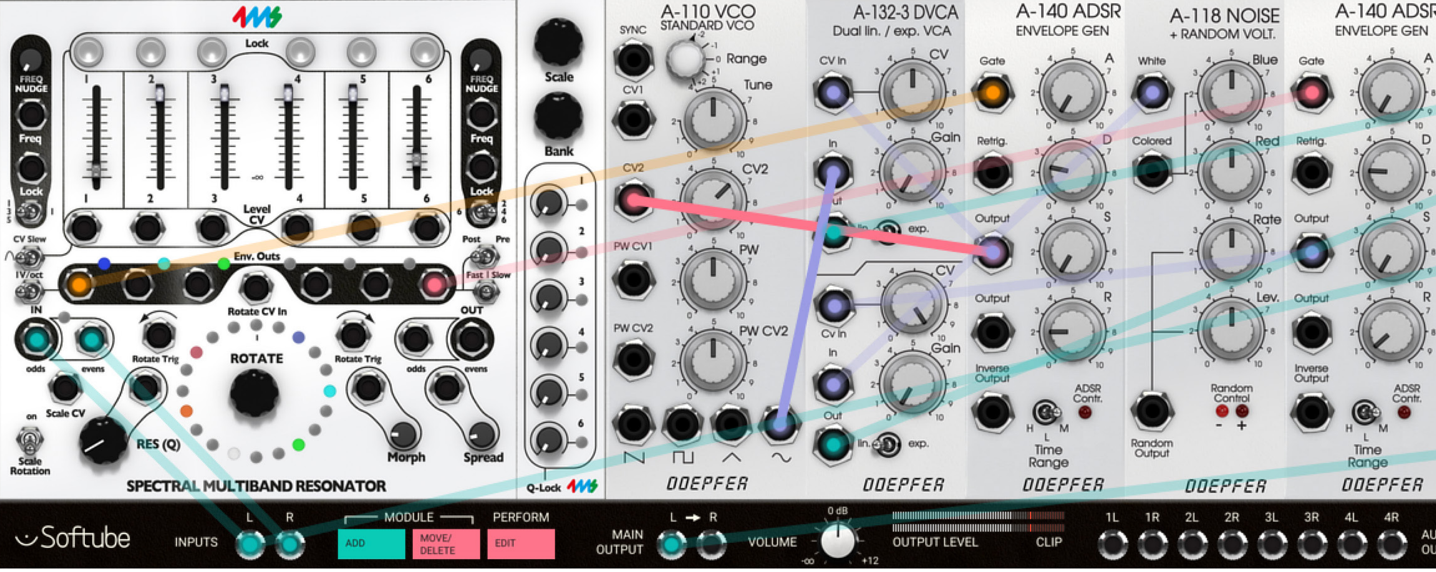
Keep this patch going, and continue on to the next patch idea...

Tutorial 4: Advanced Re-Mix Techniques

Preset patch for Modular FX "**Exp SMR tutorial 4**"

This patch idea is a powerful way to sync your modular system with an external audio source, whether it's a prerecorded track or a feed from other instruments being played live. The basic idea is to convert different frequency bands into trigger outputs, with the rhythm of the triggers matching the rhythm of the original audio's instrumentation.

1. Start with step 1 through 4 of the RE-MIX patch on the previous example: Patch audio inputs and outputs, turn down **Res Q** and select the EQ bank scale (deep blue).
2. Set the **Select** switch Odd (**135/1**) to 135, and the Select switch Even (**6/246**), to 246.
3. Set the Envelope follower out switch (**Fast/Trig/Slow**) to the center position (triggers), and set the Trigger detect switch (**Post/Pre**) to POST. The **Scale Rotation** switch should be turned off (down position).
4. Let's say you have a dance track with a kick, snare, hi-hat, and some melody lines happening. You can use the SMR to convert each kick drum hit into a trigger output, which you can use to trigger your own kick drum (or clock your master clock, or trigger anything on the modular). Start by patching the **Env Out** jack for channel 1 into something that makes sound when triggered.
5. Turn down all the sliders and bring up the slider for channel 1 to around 50%. Listen to the SMR's



Advanced remixing. Let the SMR trigger an external sound source, a Doepfer VCO in this case.

output as well as the externally triggered module. You might want to monitor the input audio, too. Use a audio mixer and mix in your dry signal from MODULAR's inputs.

6. There are four parameters to play with to get beat-synced trigger outputs: **Rotation, Freq Nudge, Q,** and filter channel volume level. Turn **Rotate** so that channel 1 is triggering the external module whenever the kick drum hits. Probably for a typical kick drum, channel 1 should be on the 2nd, 3rd, or 4th note (it depends on the pitch of the kick you're trying to tune into). Set **Freq Nudge** to dial in a frequency in between two rotation spots. Turn **Q** up to narrow the band. Around 50% is a good place to start for syncing to drum sounds.

7. Set the **Level** slider to set the threshold at which the trigger fires. If you set it too high, the ENV OUT light will stay on, and it it's too low it'll stay off. If the slider seems to have no effect, double-check that the trigger detect switch (**Post/Pre**) is set to POST.

8. When you have found a good spot, use the lock function above the filter channel level slider to lock that frequency. Also experiment with different Q values for this particular channel on the expander module.

9. ADVANCED: Use an external CV source (try an LFO or a sequenced envelope) into the **Level CV** jack with the slider all the way up. This way you will have an automated mute/unmute of that filter channel.

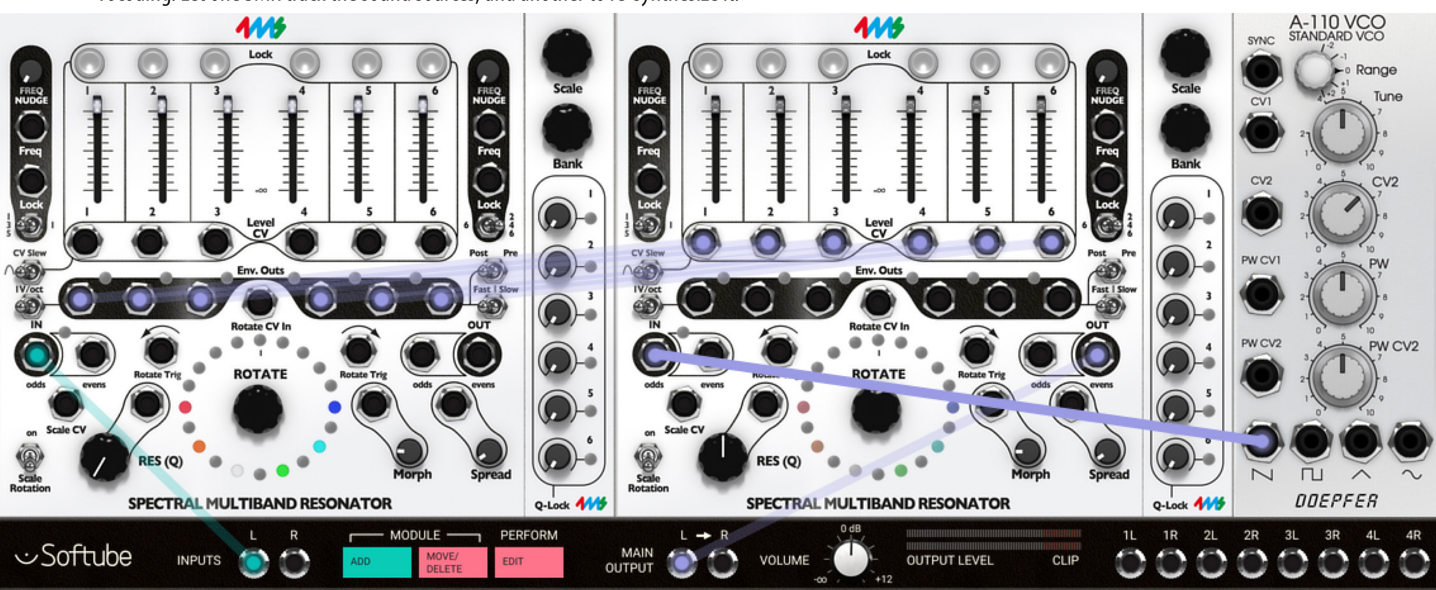
Tutorial 5: Vocoding and other spectral transfers

Preset patch for Modular FX "**Exp SMR Tutorial 5**"

Using two SMR's, you can create spectral transfer effects. One type of spectral transfer is known as *vocoding*, which is classically used with human speech and a sawtooth oscillator.

1. Patch a human speech audio signal into the first SMR's **Odds** input jack. Use a vocal sample, or a direct lined microphone from your audio interface. You are not limited to just using human speech, but it's a great way to start playing with this technique.
2. Patch the carrier signal into the second SMR's odds input. Use a sawtooth VCO, or perhaps an FM'ed VCO. You could also use another sound source or a complex sound by patching one signal in via the left jack and another via the right jack.
3. Patch the second SMR's **Evens Out** to the MODULAR **Main Out** so you can listen. You may also wish to patch the first SMR's **Evens Out** to a mixer and for monitoring the modulating signal.
4. Flip Trigger detect switch (**Post/Pre**) to POST on the first SMR.
5. Patch all six **Env Outs** jacks on the first SMR to the six **Level CV** input jacks on the second SMR. If this is your first time, keep them in the same order (1->1, 2->2...)
6. Play with **Rotation, Spread, Scale** and **Banks** of both SMRs. You might want to start in the blue bank (graphic EQ) and with the channels positioned as shown in the diagrams above.
7. Play with the **Res (Q)** knob of both SMRs. To fine tune the patch, follow the technique outlined in the RE-MIX patches, setting the note, scale, bank, and resonance of each channel on both SMRs.

Vocoding. Let one SMR track the sound sources, and another to re-synthesize it.




RES (Q)
(SCALE)
SPECTRAL MULTIBAND RESONATOR

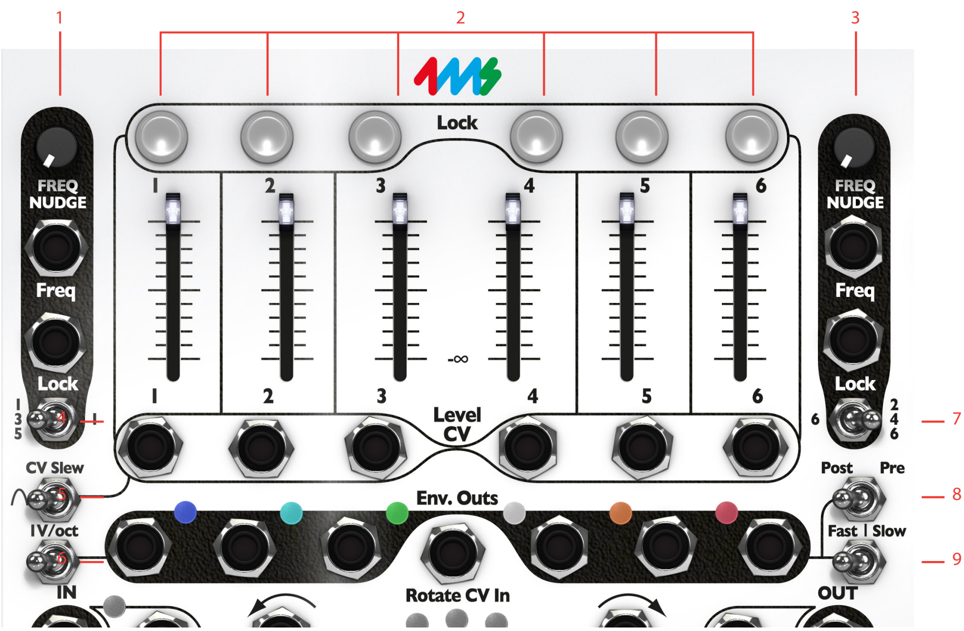
Parameters

Freq Nudge (odd bands) This knob detunes the odd bands: 1, 3 and 5, or just band 1, dependent on how the **135/1** switch is set. (See description below.) De-tunes up to 5 semitones in extreme clockwise position.

Lock Buttons 1 – 6 These toggle buttons are the frequency lock buttons for the different filter channels. When an unlit button is clicked once, it will lit up white (active). Once clicked again, it will turn unlit, grey, again (not active). When a channel is frequency locked it will not be affected by any change by the **Rotate** encoder knob, scale or bank encoder, nor will it be affected by any input frequency related CV (**Scale CV, Rotate CV, Spread, Nudge**) as long as it is activated. Notice that the **Lock** button state is the offset value for the **Lock Input CV** (see description below).

Freq Nudge (even channels) This knob de-tunes the even bands: 2, 4 and 6, or just band 6 , dependent on how the **6/246** switch is set. (See description below.) De-tunes up to 5 semi tones in extreme clockwise position.

Volume sliders 1- 6 These are the volume sliders for the different filter channels in the SMR, which control the volume out of that particular filter band for the odd and even outputs. These sliders will also affect the triggering levels for the envelope follower



1. Freq nudge (odd bands)
2. Lock buttons 1-6 and volume sliders 1-6
3. Freq nudge (even bands)
4. 135/1
5. CV Slew
6. 1V/oct
7. 6/246
8. Post/Pre
9. Fast/Trig/Slow

outputs when the **Trigger Detect Switch** is set to post (see further description below).

135/1 (Select Switch Odd) This switch determines whether or not all odd filter channels will be affected by the **Freq Nudge** (odd channels) and external CV through the **Freq (Odd)** input jack. When this switch is in "135" position, all odd channels are affected, but if switch is in "1" position only filter band 1 (blue), is affected.

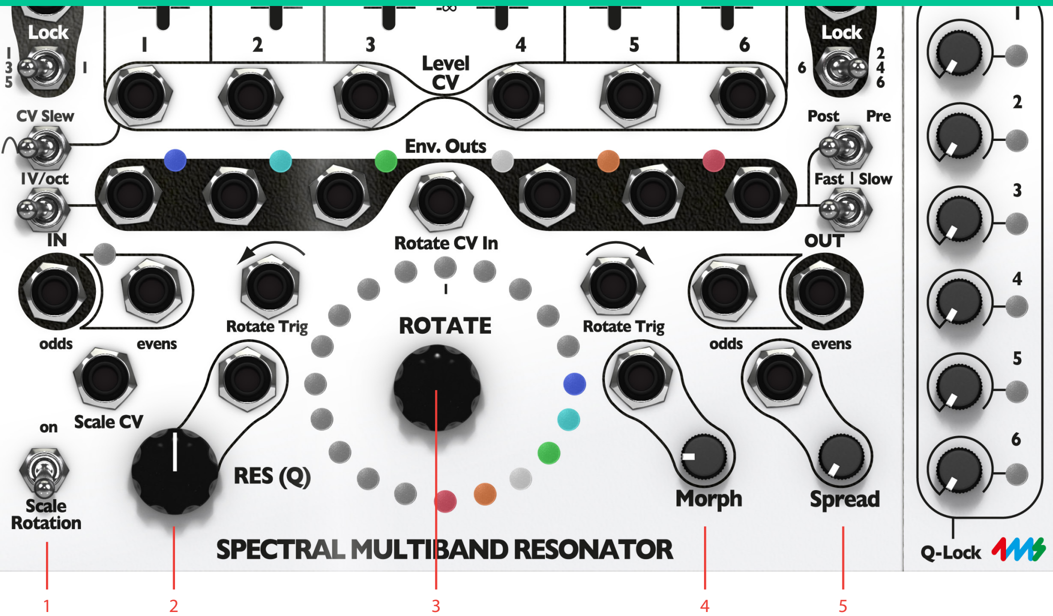
6/246 (Select Switch Even) This switch determines whether or not all even filter channels will be affected by the **Freq Nudge** (even channels) and external CV through the **Freq (Even)** input jack. When this switch is in "6" position, only filter band 6 (red), is affected, but if switch is in "246" position, all even bands are affected.

CV Slew The **CV Slew** switch can be set to soften the CV patched into the **Level CV 1-6** jacks. This is handy to avoid crackles and abrupt volume changes. **Slew** is engaged when the switch is in the left position. The slew rate is affected by the **Morph** knob position when the **Slew** switch is switched on in the left position.

Post/Pre (Trigger detect switch) The trigger detect switch, on the panel named **Post/Pre** sets whether envelope follower detection will occur before or after level set by the **Volume Sliders 1-6**. When set to "POST" position, the output of the envelope followers, **Env Outs 1-6**, will depend on the signal level set by the **Volume Slider 1-6** and **Level CV 1-6** inputs. When set to "PRE" position, envelope follower outputs will only reflect signal at the main audio inputs **In Odds** and **In Evens**.

1V/Oct (Env. Out) This switch changes the behavior of the envelope follower outputs, **Env Outs 1-6**; from envelope follower/trig functionality in "OFF" position, to quantized note CV reflecting the different channels in the "1 V/OCT" position. This functionality of the original hardware firmware version 5 is very handy when using the SMR as a quantizer or chords-generator.

Fast/Trig/Slow (Env. Follower Out) The envelope follower out switch, marked FAST/TRIG/SLOW on the SMR panel, sets the character and timing factor on the **Env Out 1-6** CV outputs. In FAST and SLOW mode the envelope follower will output a continuous CV reflecting the audio level of that particular band. FAST response time means that CV level will react quicker than in SLOW mode. When this switch is set to MID position, the **Env Out 1-6** will either output nothing or a high pulse when a



1. Scale rotation
2. Resonance
3. Rotate
4. Morph
5. Spread

certain level (-40dB) is exceeded and will remain so until signal has passed below this level again.

Scale Rotation switch When this switch is set to "ON" position, each revelation of the **Rotate** encoder, by the **Rotate Trig** (clockwise) or the **Rotate Trig** (counter-clockwise) inputs will continue on the next available scale within the bank, instead of starting over at the same scale. Each scale revelation starts at the 12 o'clock position.

Res (Q) This knob controls the overall resonance offset of the filters, from no resonance to a clear oscillation sine tone suitable for instrumentation. This knob setting will be added to the CV of the **Res Q** CV input. A filter band that has an active **Q-Lock** on will not be affected by this knob until its lock has been released.

Rotate The **Rotate** encoder lets you rotate the filter channels of the SMR around the presently selected 20 step scale. Filter bands rotated past 12 o'clock will be starting at the beginning of the scale again, or in case the **Scale Rotation** switch is set to "ON", when passing 12 o'clock it will go to the next scale in the bank. The response of the encoder is affected by the **Morph** knob (see description below).

Morph This knob sets the speed of the cross fade between the different steps in the selected rotated scale and affects all changes within that scale (the **Rotate** encoder, **Rotate CV**, **Rotate Trig Clockwise** and **Counter-clockwise**, **Spread**). This means that when **Morph** is set fully counter-clockwise, changes to rotated scale will be very abrupt and instantaneous. When **Morph** knob is turned clockwise, the rate will change to slower and slower. When a change to the rotate encoder or any related rotate scale change is faster than the change rate of the **Morph**, there's a "memory" build into the **Morph** which will remember and play out the stored changes. This very apparent when for example sending triggers into the **Rotate Trig** jack at a pace that exceeds that of the **Morph** and then subsequently changing the **Morph** rate to a faster pace, the stored rotate steps will be rapidly played out. The stored **Morph** memory will however never be larger than 127 steps.

Spread The **Spread** knob set the spacing in steps between each filter band in the selected scale. When this knob is set fully counter-clockwise, the filter spacing will be 1 step, meaning that all bands will appear besides one another in the currently selected scale. When turned clockwise, the spacing between each step will increase successively by 1 step at a time, up until the channels are 14 steps apart. Evidently, this means that they already after 3 steps spacing will be spread beyond the 20 note scale and will "spill over" into the next scale upwards and downwards in the bank (if scale rotation switch is turned on). The **Spread** knob will work as an offset when simultaneously controlling **Spread** via the **Spread CV** input jack.

Expander Section

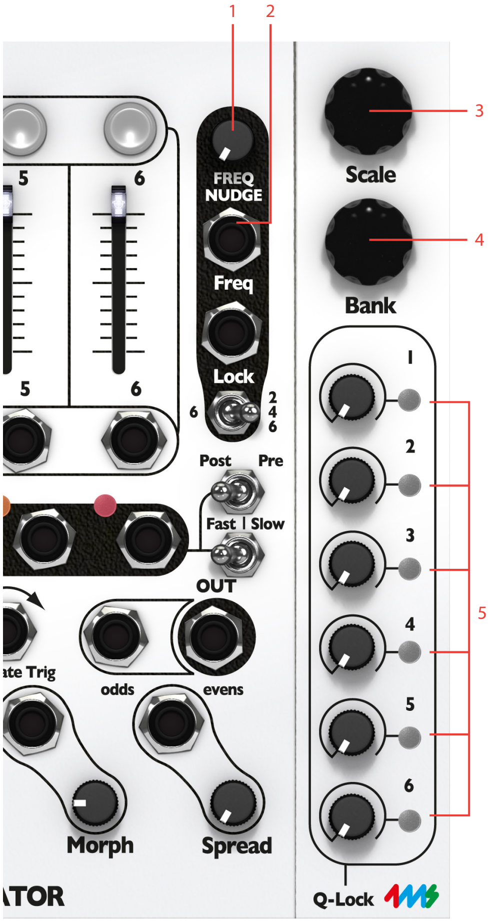
The expander section adds easier control over parameters that in the hardware are somewhat hidden.

Scale This encoder shifts all the filter channels (not presently locked and down the 11 available scales within the presently selected bank). The selected scale will be seen on the upper part of the encoder ring when clicking on and dragging the mouse cursor up and down. The colors on each of the different scale LEDs represents the channels assigned to that particular scale. If more than one band uses the same scale, that scale led will toggle between the different colors of the channels.

Bank This encoder shifts all bands available (not locked) between the different scale banks. More information on which bank colors represent which scale banks below.

Q-locks 1 – 6 The **Q-locks** are individual control over each filter channel's Q value. When a **Q-lock** is raised above 0% that band will not be affected by the **Res (Q)** knob or the **Res (Q) CV** input as long as it is locked to the value set on this expander section.

Indicators



1. Lock button LEDs
2. Volume slider LEDs
3. Scale
4. Bank
5. Q-Locks 1-6 and Q-Lock LEDs

Lock buttons LEDs	Shows if a band is locked or not. When a band is locked it will not be affected by scale rotation movement of any kind (encoder or CV induced).
Volume Slider LEDs	The intensity of these white LEDs shows the volume of the audio of the particular filter channel. A channel clipping will indicate this by flashing white light.
Env Out LEDs	These LEDs reflect the output CV of the envelope follower on the **Env Out 1-6** jacks. They will pulsate when the envelope follower is in fast, slow or trigger mode. Note that they will still reflect the envelope follower output CV even though the 1V/OCT function is active.
Input Clip LED	This indicator is lit when there's clipping on the SMR input.
The LED Ring	This is the most important indicator on the SMR, which normally shows the 20 scale steps in the currently selected scale. The different colors each represent the filter channels 1-6. When the scale or bank encoder is activated by the user, this LED ring will also briefly show the currently selected scale and bank (until the user releases the mouse button).

Expander Section

Q-locks 1 – 6 LEDs The Q-locks LEDs indicate that a **Q-lock** for that particular channel is active, thus that particular channel is not affected by the main **Res (Q)** knob or **Res (Q) CV** input jack.

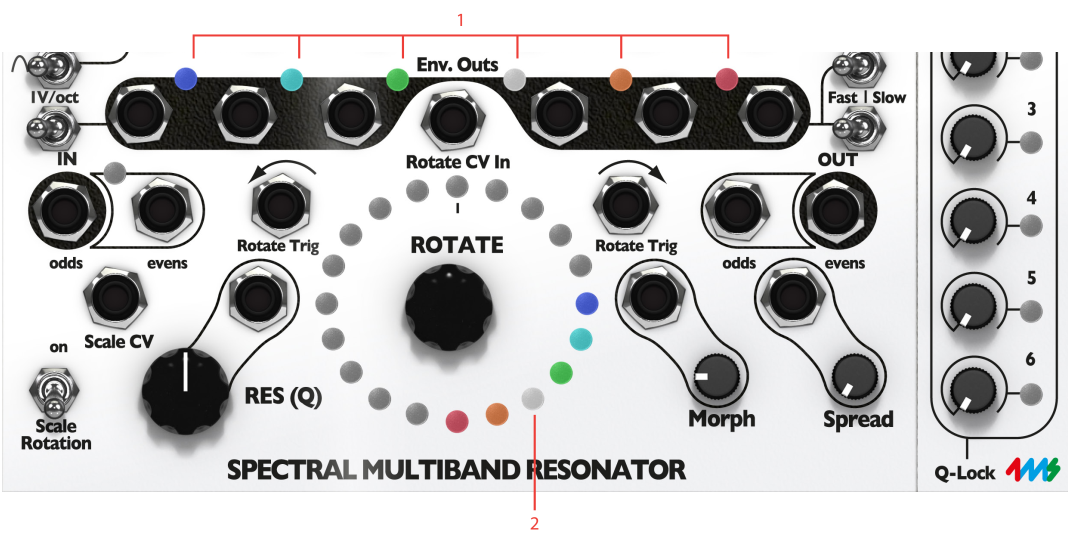
Inputs

Freq (Odd) This is the main 1 V/octave CV input for the ODD filter channels of the SMR. When the **135/1** switch (Select switch ODD) is set to "135" all odd channels will be affected by the CV input, but when the switch is set to "1", only the first band will be affected.

Freq (Even) This is the main 1 V/octave CV input for the EVEN filter channels of the SMR. When the **6/246** switch (Select switch EVEN) is set to "246" all even channels will be affected by the CV input, but when the switch is set to "6", only the last band, number six, will be affected.

Lock (Odd) This is the lock toggle input for the odd channels. It will trigger and toggle the lock function when a CV below 2.5V goes above that threshold. This function will affect the band currently selected via the **135/1** switch (select switch ODD). If a channel already has its lock button active, a pulse at this input will toggle it off.

Lock (Even) This is the lock toggle input for the even channels. It will trigger and toggle the lock function on a rising edge trigger (CV below 2.5V goes above that threshold). This function will affect the band currently se-



1. Env. Out LEDs
2. The LED ring

lected via the **6/246** switch (select switch EVEN). If a channel already has its lock button active, a pulse at this input will toggle it off.

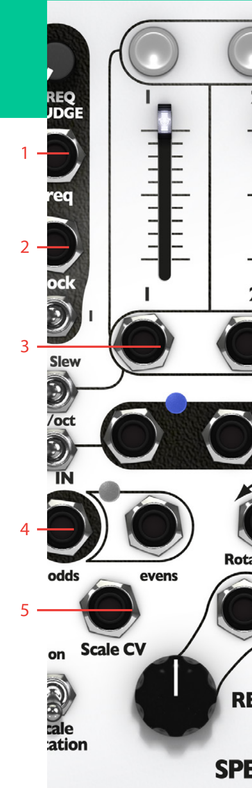
Level CV 1- 6 These inputs are the CV inputs for controlling each band's level. When a CV signal is input here, the **Volume Sliders 1-6** will act as attenuators of the incoming CV, meaning that you will still be able to control the maximum volume of each band from the panel.

In Odds This is the common input for all odd filter channels. It is pre-patched to the **In Evens** jack when that jack is not in use. This means that the input signal goes through all filters, odds and evens, if no signal is patched into the **In Evens** jack.

In Evens This is the common input for all even filter channels. When a signal is inserted here it will go through all even channels (2, 4 and 6). When no cable is inserted here it is pre-patched from the **In Odds** jack (see above).

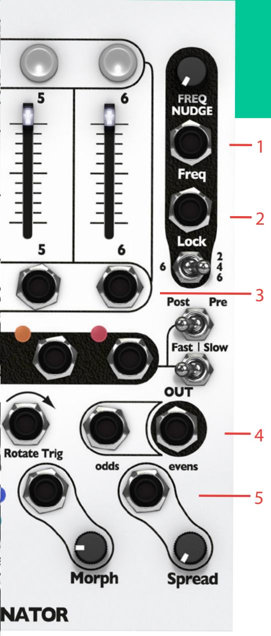
Rotate Trig (Counter-clockwise) When a rising edge trigger (CV below 2.5V goes above that threshold) appears on this input, all channels not locked will shift one step back counter-clockwise. This is equivalent to turning the **Rotate** encoder one click to the left.

1. Freq (odd)
2. Lock (odd)
3. Level CV 1-6
4. In odds, In evens
5. Scale CV, Res (Q) CV



Rotate Trig (Clockwise) When a rising edge trigger (CV below 2.5V goes above that threshold) appears on this input, all channels not locked will shift one step forward clockwise. This is equivalent to turning the **Rotate** encoder one click to the right.

Scale CV Input CV on this jack will select one of the 11 available scales within the currently selected bank. Click on the Scale knob in the expander section to have a visual check on the what scale that is currently selected.



1. Freq (even)
2. Lock (even)
3. Level CV 1-6
4. Rotate Trig
5. Morph CV, Spread CV

Res (Q) CV This is the jack for external CV control of the resonance of all filter channels not currently locked by the **Q-locks** on the expander section.

Morph CV This jack is the external CV control of the morph parameter that can be offset via the **Morph** knob on the panel.

Spread CV The **Spread CV** jack is the external CV input for the spread parameter that can be offset via the **Spread** knob on the panel.

Outputs

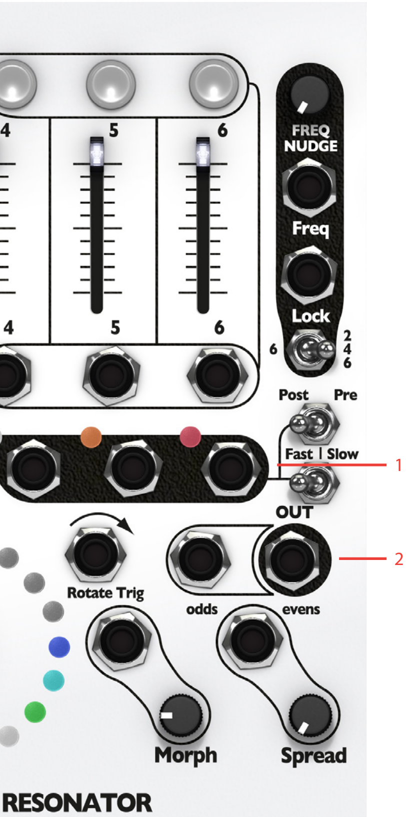
Env Outs 1 – 6 These outputs, **Env Out 1-6**, are the envelope follower outputs for each band. The outputs here are always control voltages (CV), not audio. When the **1V/oct** switch is set to "OFF", either envelope follower (FAST or SLOW) control voltage or triggers (TRIG mode) will appear on these jacks.

The trigger length on these outputs will depend on the set filter band level (**Level CV** input and **Volume Slider**) when the **Post/Pre** switch is set to "POST". When the **1V/oct** switch is set to "1V/OCT", each of these output jacks will output the quantized CV of each band's current cutoff frequency.

Out Odds This is the output jack of all odd filter channels. It is pre-patched to the **Out Evens** jack if no cable is patched into the jack (see description below).

Out Evens This is the main output jack of the SMR. If no cable is patched into the **Out Odds** jack, this is the output of all filter channels in the SMR.

Outputs



1. Env Outs 1-6
2. Out odds, Out evens

Scale Bank Colors

When turning the **Bank** encoder in the expander section, part of the LED ring will change color to indicate which scale bank that will be active when the mouse button is released.

1. Western scale (white)

Just intonation. Each scale contains a root note and one interval, repeated over 10 octaves. Each scale uses a different interval (m2 to M7) and spans 27.5Hz to 15kHz

2. Indian pentatonic (green)

Scales 1-8 have five notes per octave, spanning four octaves. Each is one octave above the previous. Scales 9-11 contain selected notes.

3. Alpha scale, selected notes (lavender).

Each scale contains intervals from Wendy Carlos's micro- tonal scale, spanning roughly from 20Hz to 20kHz

4. Alpha scale, selected notes (cyan).

More variations. All intervals are multiples of 78.0 cents (perfect fifth split into nine equal parts), so there are no octaves, yet rich harmonies.

5. Gamma scale, selected notes (orange)

Each scale contains intervals selected from Wendy Carlos's non- octave repeating Gamma scale, spanning roughly 20Hz to 20kHz.

6. 17-note per octave (yellow)

The first scale starts at 13.75Hz and each note is equally tempered to a 17-note octave. Each scale is one octave above the previous.

7. Chromatic (purple)

The first scale starts at E2 (82.4Hz) and each note is one semi-tone above the previous. Each scale spans an octave and a half and is six semi-tones above the previous.

8. Diatonic 1/2 (magenta)

Each scale is a diatonic scale spanning almost three octaves. The first two scales start on A, the next two start on A#, then B, C, C#, D. In each pair, the second scale is two octaves above the first.

9. Diatonic 2/2 (rose)

Continuation of previous bank, starting at D#, E, F, F#, G, G#

10. Western dual-interval (light green).

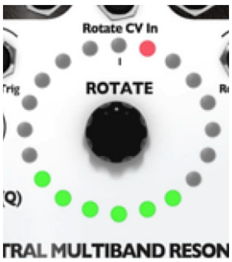
Each scale starts on G1 (49Hz), followed by two intervals. The triad is repeated across seven octaves, an octave higher each time. The pair of intervals varies per scale: M2/P5, M3/flat5, m3/M5, M4/P5, m3/#5, P4/P5, M3/M6, m3/flat6, M3/#5, m3/m7, P5/M7

11. Mesopotamian (Red)

Various ancient scales from Mesopotamia. Scales 1, 3, 5, 7, 9, and 11 have six notes starting on A2 (55Hz), and then repeats over 3 octaves. The remaining scales are copies of the previous scale, but three octaves higher.

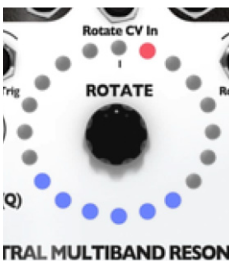
12. Shrutis, Indian scales (Yellow-green).

Uses 20 notes from a 22-note octave, so each scale spans one octave and is one octave higher than the previous scale.



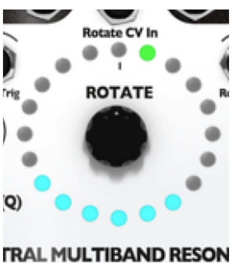
13. Graphic EQ (blue)

These are common frequencies in Eqs. The first scale is 20Hz, 40, 60, 80, 100, 150, 250, 350, 500, 630, 800, 1k, 1.3k, 1.6k, 2k, 2.6k,3.5k,5k,8k,10k. Each scale is the same as the previous scale, shifted up about 3%



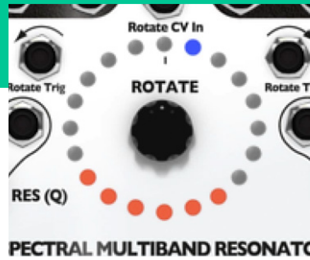
14. Gamelan (azure)

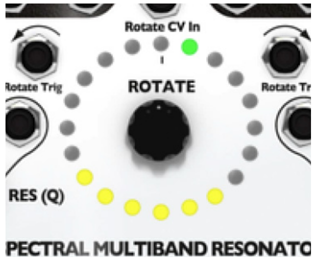
Scales 1 and 2 are the Gamelan Pelog (5-note) scale repeated over 4 octaves. Scales 3, 4, and 5 are Slendro (7-note) scales spanning 3 octaves each. Scales 6-11 are a selection of five notes from the Slendro scale.

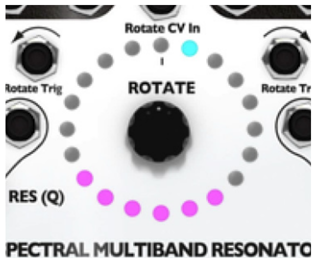


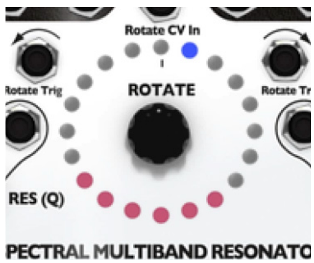
15. Bohlen-Pierce (Lime)

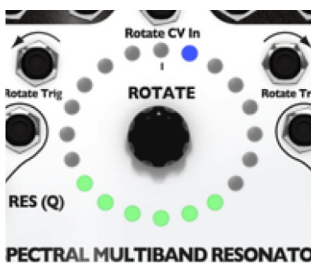
Instead of octaves, the BP scale is based on the tritave (triple the root frequency), with thirteen notes per tritave. Each scale starts on C1 and contains four selected notes per tritave, spanning five tritaves







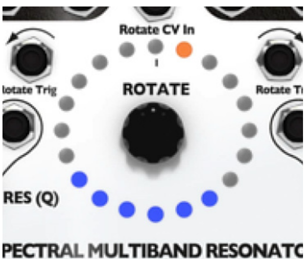


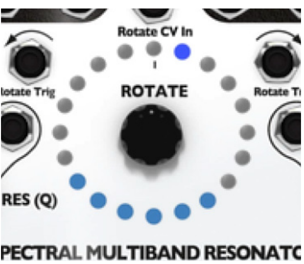


In Use

As described in the tutorial examples and shown in the preset patches, the SMR is a very flexible module that can be used for a diversity of different tasks. For example:

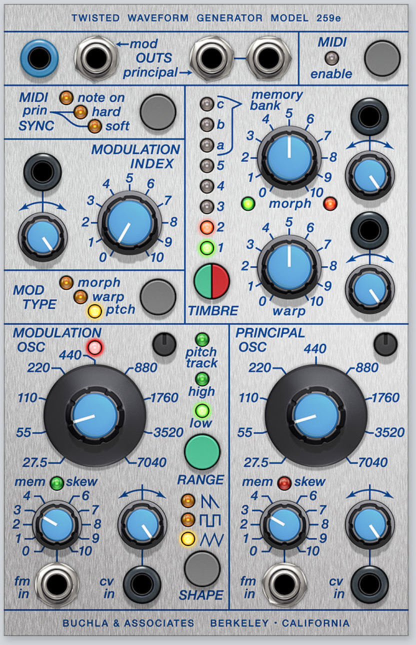
- as oscillator bank for additive synthesis (marimbas, gongs, woodblocks)
- pulse triggering odd and even for different rhythms
- as vocoder
- as phaser
- as CV source/arpeggiator/sequencer
- as stereo-width thingy
- as multiband EQ
- as remix tool for extracting triggers from recorded material
- as quantizer
- as crazy ass effect
- as basis for multiband compressor patches
- as synth filter





Tip

Manually max the **Spread** knob and bring it down to ZERO in order to get your channels in order again, in case were they might have been set in different order via previous locks. The same technique can be used to bring all channels to the same scale by maxing and then setting the **Scale** knob to zero.



4 Buchla 259e Twisted Waveform Generator

Donald Buchla, an inventor based in Berkeley, USA, started building and designing electronic instruments in the early 1960s. It all started when he was commissioned by avant-garde composer **Morton Subotnick** to build a modular instrument for composing and performing live electronic music. The result was called the **Buchla Box**—Don Buchla's first modular synthesizer. This evolved into Don's first commercially available system, the Buchla 100 system, which was followed in the 70s by the legendary **Buchla 200 system**.

In the early 2000s, Don decided to reinvent and reimagine the 200-system, and the new Buchla 200e system was born. The 200e system is an interesting hybrid between analog and digital technology, and the Buchla 259e module is a perfect example.

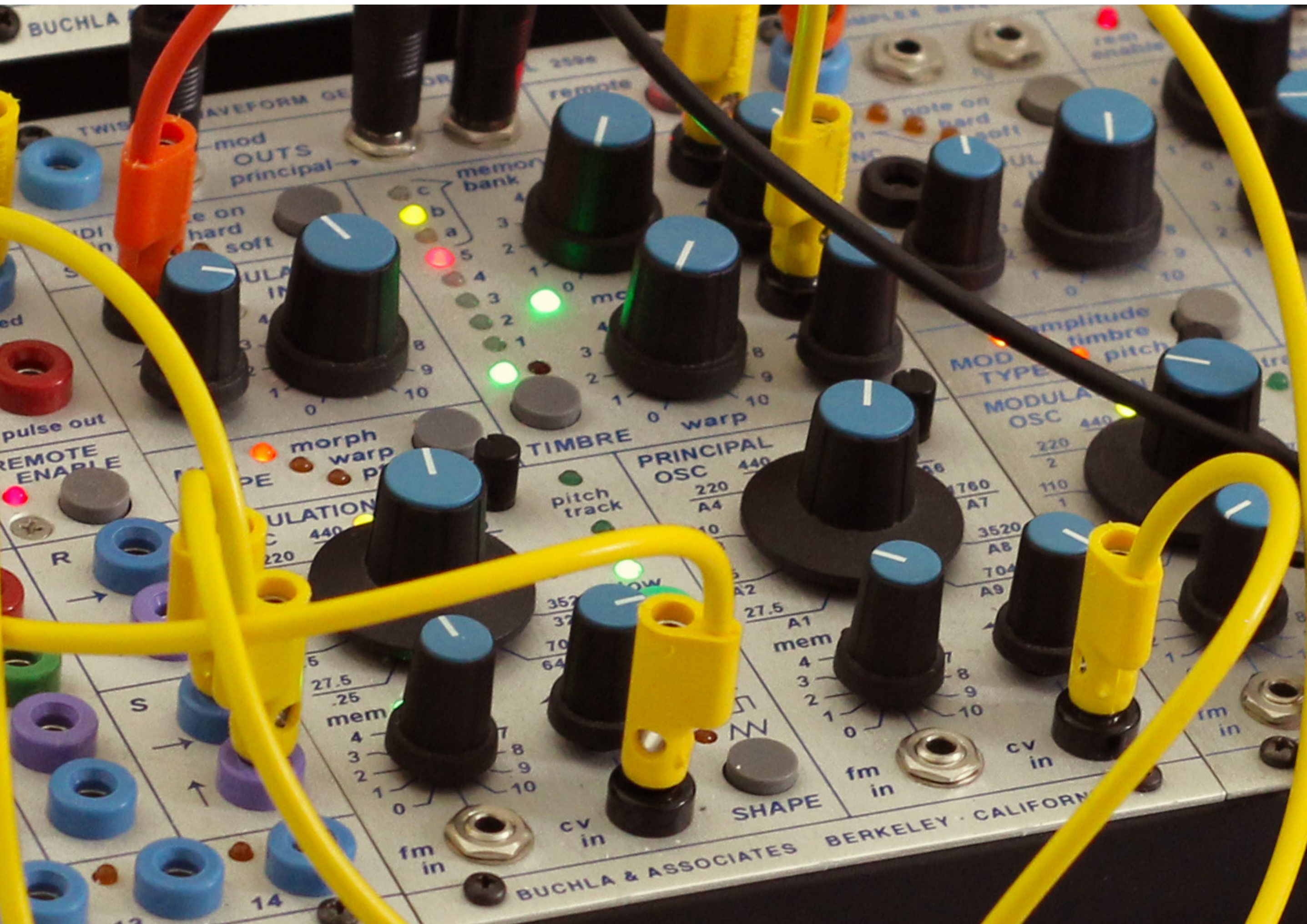
The **259e** is in one sense a modern day digital implementation of the original **Buchla 259 Complex Wave Oscillator**, but in many other ways it's a different beast altogether. The 259e's crazy memory scanning functionality and its cold and very digital sounding waveshaping capabilities, with plenty of characteristic aliasing distortion, makes it a prime candidate for sonic experimentation beyond the ordinary.

Overview

Softube's **Buchla 259e Twisted Waveform Generator** plug-in for **Modular** features the digital waveshaping, aliasing noise and fold over frequencies of the original hardware. The plug-in captures all the self-modifying, screeching, snarling responses from the original's downright odd inner workings—which added "twisted" to its name.

The 259e consists of two separate oscillators. One is the Modulation oscillator, which primarily serves to modulate the second one, called the Principal oscilla-

The Buchla 259e hardware



tor. The Modulation oscillator can be switched between High and Low frequency ranges, where High generates notes in the audible region and Low is mostly for when the oscillator is used as an LFO which modulates the Principal oscillator. The Modulation oscillator can be switched between three classic waveform shapes, and three different modulation targets can be selected, one by one or in any combination.

The sine wave generated by the Principal oscillator is simultaneously applied to two of the eight available waveshape tables. Step through the tables (1-5, a-c) by pushing the green and red buttons. Green and red LEDs indicate which tables are currently selected.

Tables a, b and c are actually not tables in the classical sense—they are simply portions of the 259e operating program, full of unpredictable noise and frequent silences. When these tables are selected, the FM controls are re-assigned to table scanning functions and the FM inputs become table modulators. In this state, the Mem Skew LEDs above the FM controls light up.

A morph voltage pans between the two tables and a warp voltage varies the amplitude of the sinusoidal (driving) waveform. Both these functions can be modulated by the modulation oscillator.

Buchla Philosophy

The Buchla systems vary the harmonic content in their patches by using a waveshaper on a rudimentary sound source such as a sine oscillator, rather than shaping of harmonic content through filtering, which is the more commonly used method. Buchla's type of synthesis is often referred to as the West Coast style.

Where a VCF removes harmonic content from a signal, a waveshaper warps the waveform by applying either a fixed or variable mathematical function. This real time process alters the shape of the waveform, adding harmonics and resulting in a more abrasive sound compared to the typical East Coast style characteristics (classic subtractive synthesis).

In a Buchla synthesizer system, audio and modulation signal paths are separated by their different jack and cable formats. Audio jacks are silver and AC coupled, while modulation jacks are colored "banana" jacks which are DC coupled. The form factor of the jacks and the different cables make it easy to see which signal goes where and creates a natural signal flow in the Buchla environment.

Softube has reimagined the 259e in a Eurorack environment, adapting levels and connectivity to work as any Eurorack module. This means that the 259e module featured in Modular, unlike its hardware counterpart, will have no limitations when it comes to connectivity—you can connect any type of output to any kind of input, whether it's audio or modulation. But for the sake of familiarity, we kept the look and color of the hardware audio and modulation jacks. Softube has also adapted the signal levels to harmonize better with the other modules in the Modular system.

The following adaptations have been made when converting the Buchla 259e to Softube Modular:

* Audio and CV can be freely patched to each other.

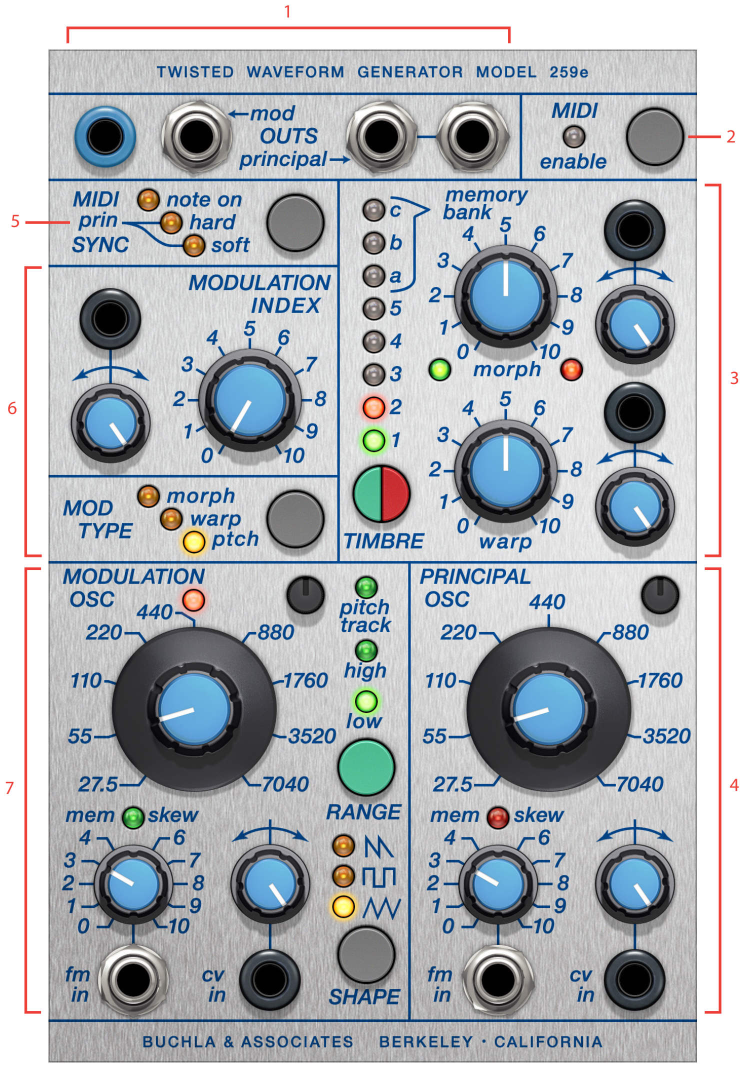
* CV voltages rescaled by around 0.9 times, so that setting the frequency CV attenuators to max gives a 1V/oct response.

* Audio voltages have been rescaled by around four times, to match Eurorack levels better.

* Remote Enable replaced by MIDI Enable. Press once to go to MIDI learn mode (blink), play a MIDI note and it locks to that channel. Press again to turn MIDI off.

* the AC coupling of the FM jacks was disabled when in Slew mode in order to make them more useful when in use with regular DC CV modulation such as LFOs, envelopes and sequencers.

Overview



1. Main outputs
2. Midi enable
3. Principal oscillator Timbre section
4. Principal Oscillator
5. Modulation oscillator Sync behaviour
6. Modulation conversion between Modulation oscillator and Principal oscillator
7. Modulation oscillator

Detailed Description

MIDI Enable Button

When clicked once in its preset state, the LED will start flashing, an indication for that the 259e is awaiting a MIDI note on message. When the MIDI note is received, the 259e automatically sense will sense on which MIDI channel and lock onto that channel (LED is lit). By clicking once more, MIDI is reset to off status and LED will go unlit. Received MIDI range is from A-1 to A7 (8 octaves), same range as coarse tune. MIDI pitchbend is received on the same MIDI channel and responds to +-1 octave.

MIDI Prin Sync Button (Modulation Osc Sync)

This button toggles between the different reset behaviors of the Modulation Oscillator:

- **Preset mode** - no synchronization is set (all LEDs are unlit).
- **Soft sync mode** - uses a phase-lock loop to achieve synchronization.
- **Hard sync mode** - truncate synchronization to the Principal oscillator. This means that the Principal oscillator will hard reset the Modulation oscillator.
- **MIDI note sync** - synchronisation by incoming MIDI note messages on the channel set by the MIDI enable function (see description above). This means that this function will not be operational when MIDI enable is set to off.

Morph

The **Morph** knob is the offset used to fade between the green and red wave-shape tables of the Principal oscillator. When this knob is fully counter-clockwise and no modulation is applied to morph from neither the Modulation oscillator or the **Principal Osc Morph CV** jack, the green LED just beneath the knob will be lit indicating that presently, only the green wave-shape is output on the Principal outputs. When turning the morph knob gradually clockwise, more and more of the red wave-shape will be faded into the Principal outputs, and at fully clockwise, only red is output. The **Morph** knob works as an offset, to which modulation is added, either via the **Principal Osc Morph CV** input or via the **Modulation Index** routing.

Principal Osc Morph CV

This is the bi-polar amount knob to scale the incoming CV at the **Principal Osc Morph CV** jack. At 12 o'clock position, amount is 0 and then increasing clockwise up to 1 times scaling at the 5 o'clock position. Turning the knob counter-clockwise from 12 o'clock to 7 o'clock position sets amount to fully -1 times (CV has inverted impact on morph). Modulating morph CV is always relative to the position of the **Morph** knob which works as an offset.

Modulation Index CV

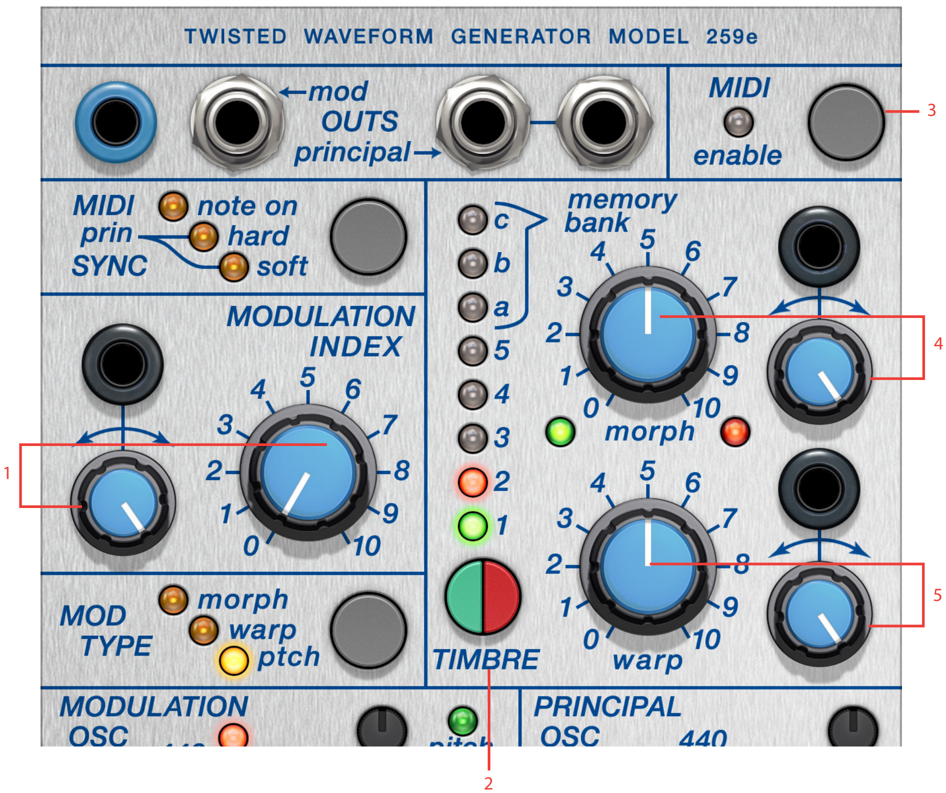
This is the bi-polar amount knob to scale the incoming CV at the **Modulation Index CV** jack. At 12 o'clock position, amount is 0 and then increasing clockwise up to 1 times scaling at the 5 o'clock position. Turning the knob counter-clockwise from 12 o'clock to 7 o'clock position sets amount to fully -1 times (CV has inverted impact on the Modulation Index). Modulation CV is always relative to the position of the Modulation Index knob (offset). See further description on Modulation Index below. The signal patched into the **Modulation Index CV** input will affect the amount of modulation selected via Mod Type and offset via the Modulation Index, positively or negatively. For example FM amount between Modulation oscillator and Principal Oscillator can thus be affected by an external ADSR envelope.

Modulation Index

The **Modulation Index** knob sets the amount of the selected modulation between the Modulation oscillator and the Principal oscillator. This is done via Mod Type button (see description below). The **Modulation Index** knob works as an amount offset, which is affected by the signal at Modulation Index CV input. This signal can be set to affect the Modulation Index positively or negatively.

Timbre Bank Buttons

These green and red buttons control the different wave-shape tables of the Principal oscillator. There are different wave-shape tables for the green and red sides, although some of them are the same. When clicking ov n one of the green and red buttons, the corresponding colored LED will move one step up the wave-shape selector ladder. Both sides, green and red, will share the same indicator which makes the LED lit up orange when green and red indication appears on the same step. When clicked through until step C, one more click will cause the that side to cycle back to step 1. Green and red **Timbre** banks are the same for position 1, A, B and C, but differ slightly in harmonics for position 2,3,4 and 5 on the green and red side. Positions 1 through 5 are different regular wave-shapes, while position A-C reflect the fun and quirky memory bank mode. In this mode, the wave-shapes are based on scanning through the digital code memory of the Buchla 259e with the **Mem Skew** function (see further description below).



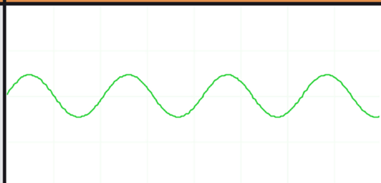
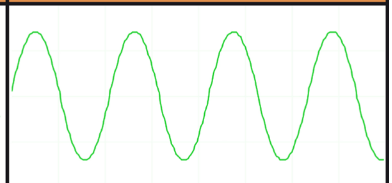
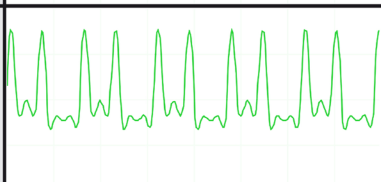
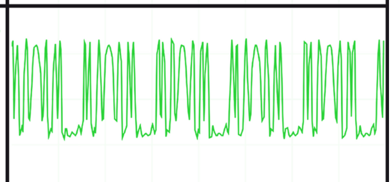
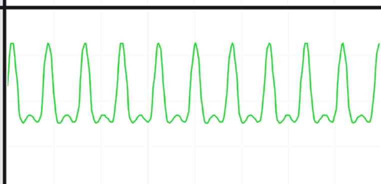
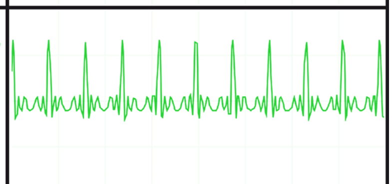
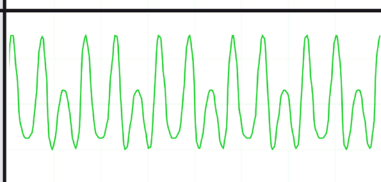
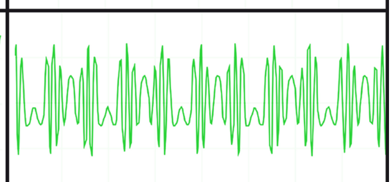
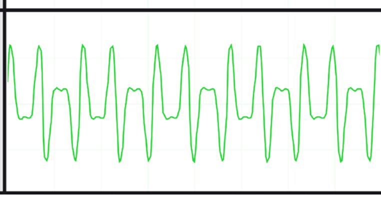
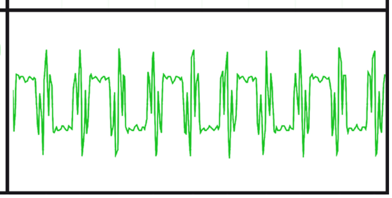
1. Modulation Index Offset and Attenuator
2. Green and red Timbre selection
3. Midi enable button
4. Morph knobs
5. Warp knobs

1. Modulation oscillator coarse and Fine tune
2. Green and red FM Amount (or Mem Skew)
3. Pricipal Coarse and Fine tune

Principal Osc Warp

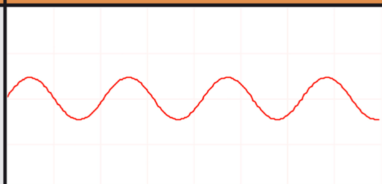
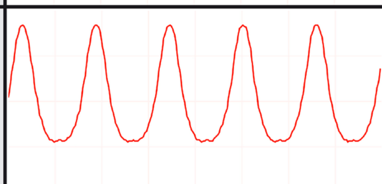
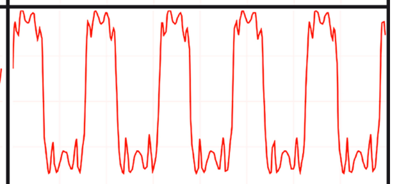
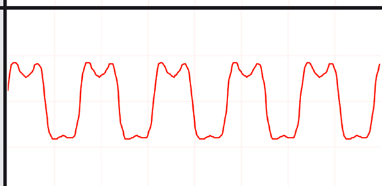
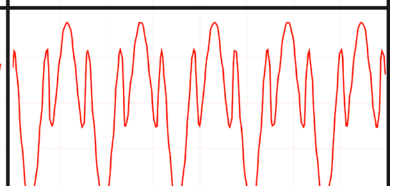
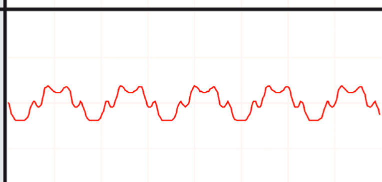
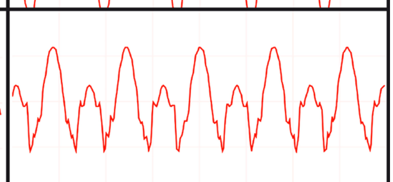
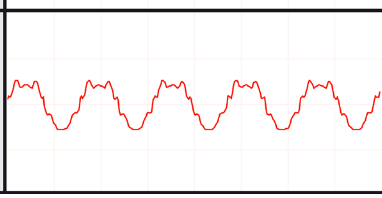
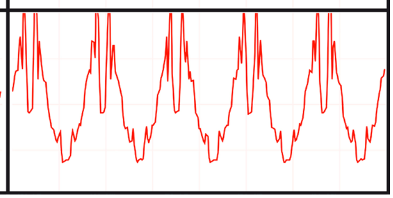
The **Osc Warp** knob controls the folding of each of the timbres (except timbre nr 1, sine) working as an offset for the modulation applied to the Principal **Oscillator Warp CV** input. Increased warp changes the output harmonics of the Principal Oscillator. When the in **Mem Skew** mode, the **Principal Osc Warp knob** will set the range of Memory being scanned via the **Mem Skew Index** and CV input.

Buchla 259e Principal Oscillator Timbres:
Green Timbres

Timbre	Warp = 0	Warp = 10
Bank 1		
Bank 2		
Bank 3		
Bank 4		
Bank 5		

Waveforms captured at 440hz

Buchla 259e Principal Oscillator Timbres: Red Timbres

Timbre	Warp = 0	Warp = 10
Bank 1		
Bank 2		
Bank 3		
Bank 4		
Bank 5		

Waveforms captured at 440hz

Principal Osc Warp CV

This bi-polar amount knob to scales the incoming CV at the **Principal Osc Warp CV** jack. At 12 o'clock position, amount is 0 and then increasing clockwise up to 1 times scaling at the 5 o'clock position. Turning the knob counter-clockwise from 12 o'clock to 7 o'clock position sets amount to fully -1 times (CV will have inverted impact on the **Principal Osc Warp)**. **Principal Osc Warp CV** is always relative to the position of the main **Principal Osc Warp knob** (offset). The signal patched into the Principal Osc Warp CV input will affect the Warp amount, positively or negatively.

Mod Type button

This button determines which destination, or combination of destinations, the Modulation Oscillator will modulate through the Modulation Index. Multiple presses on this button switches between the none (all three LEDs - morph, warp and ptch, are unlit), one, two or all three of the available Modulation destinations: **Morph, Warp** and **Ptch** (Pitch). There are in all 7 different combinations available

Modulation Osc Fine

This knob is for fine-tuning the pitch of the Modulation Oscillator. The range is roughly up and down two semitones.

Principal Osc Fine

This knob is for fine-tuning the pitch of the Principal Oscillator. The range is roughly up and down one and a half semitone.

Modulation Osc Coarse

The **Modulation Osc Coarse** knob sets the coarse tuning for the Modulation oscillator. When **Modulation Osc Range** is set to Low the range of this knob goes from 0.25Hz to 64Hz. When set to High range is 27,5Hz to 7040Hz. In Pitch Track mode, the knob function as offset to pitch set by the Principal Oscillator frequency.

Principal Osc Coarse

The **Principal Osc Coarse** knob will set the coarse tuning of the Principal oscillator ranging from 27,5Hz up to 7040Hz. In MIDI mode this knob will not always reflect the pitch heard because, in MIDI mode, pitch is determined by the last MIDI note played. And when you then tweak the Coarse-knob, it will work as an offset from the MIDI note. In practise this means that if you play an A4 in MIDI mode, the following time you adjust the **Principal Osc Coarse** knob, the pitch of the Principal Oscillator will start at 440Hz regardless of where the knob points, and work in linear fashion towards the lo and hi extreme endpoints of the knob. A new MIDI note will force the frequency back to that note value. This means that if a MIDI sequence is played, the **Principal Osc Coarse** knob can be used to temporarily tweak the frequency of the active notes, while it will always jump back in tune for every received MIDI note.

Modulation Osc Range button

- **Pitch track** - when this LED is lit, the Modulation Oscillator pitch is tracking that of the Principal Oscillator, but offset by the **Modulation Osc Coarse** knob. In this mode, the LED above the **Modulation Osc Coarse** knob indicates if the pitch-relation to the Principal Oscillator is below (red light), above (green light), or spot on (LED is unlit) when the **Modulation Osc Coarse** knob is set to 12 o'clock.
- **High** - the Modulation Oscillator operates in the audio-region with the **Coarse** knob ranging from 27 to 7040 Hz. In this mode, the LED indicator above the **Coarse** knob remains unlit
- **Low** - the Modulation Oscillator operates in the sub-audio range with the **Coarse** knob ranging from .25 Hz to 64 Hz. The LED indicator above the **Coarse** knob in this mode reflects the cycles of the Modulation Oscillator (red being negative part and green being positive part).

Modulation Osc FM
This knob sets the amount of frequency modulation to affect the Modulation Oscillator when patching in an external signal into the FM in input jack. When the Principal Oscillator's green timbre table is in **Mem Skew** mode (using bank A,B or C), this knob's functionality is re-assigned to table-scanning. This is indicated by the **Mem Skew** green LED indicator being lit above this knob. In Mem Skew mode, the **Modulation Osc FM** knob will act as an offset for the Principal Oscillator green table scanning while signals patched into the FM in input jack will affect the scanning (without any attenuation).

Modulation Osc CV
This knob will attenuate modulation signals patched into the CV in input jack situated below this knob. Knob position 12 to 5 o'clock will attenuate positively clockwise (0 to 1 v/octave), while knob position 12 to 7 o'clock will attenuate negatively anticlockwise (0 to -1v/octave).

Principal Osc FM
This is the amount of frequency modulation to affect the Principal Oscillator when patching in an external signal into its' FM in input jack. When the Principal Oscillator's red timbre table is in **Mem Skew** mode (using bank A,B or C), this knob's function is re-assigned to table-scanning. This is clearly indicated by the **Mem Skew** red LED indicator being lit above this knob. In **Mem Skew** mode, the Principal Osc FM knob will act as an offset for the Principal Oscillator red table scanning while signals patched into the Principal Oscillators FM in input jack also will affect the scanning (without any attenuation).

Principal Osc CV
This knob will attenuate modulation signals patched into the **Principal Oscillator CV** in input jack situated below. Knob position 12 to 5 o'clock will attenuate positively clockwise (0 to 1 v/octave), while knob position 12 to 7 o'clock will attenuate negatively anticlockwise (0 to -1v/octave).

Indicators

MIDI enable LED
This LED starts blinking when clicking on the MIDI mode button, indicating that the 259e is waiting for a MIDI note on message to determine which MIDI channel to listen to. When having received a MIDI note, this LED will stop blinking and remain lit until MIDI mode button is clicked again to switch 259e to manual mode.

MIDI prin sync LEDs
This LED indicates sync status of the Modulation Oscillator. For further detailed information on the different indicated sync modes, see description above (MIDI prin sync button).

Timbre Bank LEDs
These LEDs tells you which timbre banks, green and red that are selected for the Principal Oscillator.

Morph LEDs
These LEDs indicate the balance of the timbre banks heard on the Principal Oscillator output, directly reflecting the sum of the **Morph** knob and **Principal Osc Morph CV** input.

Mod Type LEDs
These LEDs indicate which modulation destinations in the Principal Oscillator that currently are selected to be connected to the Modulation Oscillator. Selected destinations can be any combination of one, two or all three destinations. None is also an option (no LEDs lit).

Modulation Osc LED
Situated just above the **Modulation Osc Coarse** tune knob is the Modulation Osc LED reflecting the internal function of the oscillator. When the Modulation Oscillator is set to Low Mode it will reflect the oscillator rate with red being the negative portion and green being the positive portion of the waveform.

Mem Skew LED Green

This LED is lit when the Principal Oscillator green timbre bank is using one of the memory banks A, B or C. It indicates the temporarily changed functionality of the **Modulation Osc FM** knob and CV input (for further detailed information, see above description about the **Modulation Osc FM** knob).

Modulation Range LEDs

These LEDs reflect Modulation Oscillator operational mode: **Low, High** or **Pitch Track**. See further detailed description above on the **Modulation Oscillator Range** button.

Mem Skew LED Red

This LED is lit when the Principal Oscillator red timbre bank is using one of the memory banks A, B or C. It indicates the temporarily changed functionality of the **Principal Osc FM** knob and CV input (for further detailed information, see above description about the **Principal Osc FM** knob).

Inputs

Principal Osc Morph CV input

This is the CV input for the blend of the red and green timbre banks on the Principal Oscillator.

Modulation Index CV input

CV input for scaling the Modulation Index between the Modulation Oscillator and Principal Oscillator.

Principal Osc Warp CV input

This jack is the input for externally controlling the **Warp** function of the Principal Oscillator from a CV modulator (LFO, envelope, etc).

Modulation Osc FM input

Audio input for frequency modulation of the Modulation Oscillator or, in **Mem Skew** mode, external CV control of the green timbre banks memory scanning. This jack is AC coupled in ordinary FM mode which means that it will work better with audio signals (this is how the original hardware works). In **Mem Skew** mode this jack will change to DC coupled mode to better work with modulation signals such as LFOs, envelopes and sequencers.

Modulation Osc CV input

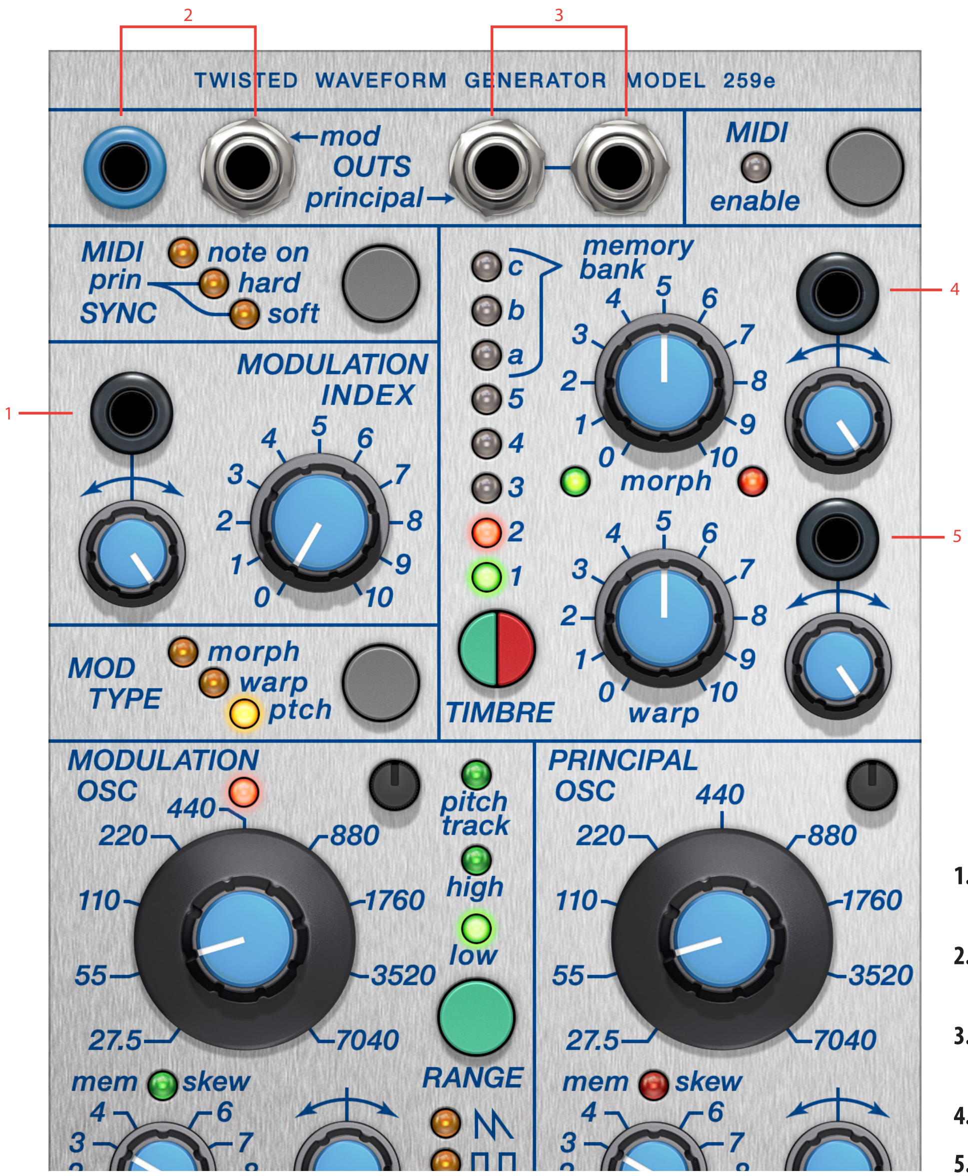
This jack is external CV control of the Modulation Oscillator pitch, tracking 1V/octave when the **Modulation Osc CV** knob is set to max (preset value).

Principal Osc FM input

Audio input for frequency modulation of the Principal Oscillator or, in **Mem Skew** mode, external CV control of the red timbre bank's memory scanning. This jack is AC coupled in ordinary FM mode which means that it will work better with audio signals (this is how the original hardware works). In Mem Skew mode this jack this jack will change to DC coupled mode to better work with modulation signals such as LFOs, envelopes and sequencers.

Principal Osc CV Input

This jack is external CV control of the Principal Oscillator pitch, tracking 1V/octave when the Modulation Osc CV knob is set to max (preset value.



1. Modulation attenuation CV in
2. Modulation oscillator CV and Audio output
3. Principal oscillator outputs
4. Morph CV in
5. Warp CV in

1. Modulation oscillator FM and CV in
2. Principal audio FM and CV in

Outputs

Modulation Osc CV Out
This is the DC coupled output of the Modulation Oscillator spanning approximately 0 to 8.6V. This output is very suitable when using the Modulation Oscillator as an LFO to modulate other module.

Modulation Osc Audio Out
This is the AC coupled output of the Modulation Oscillator. Use it when you're patching the Modulation Oscillator as an ordinary oscillator to external filters and such.

Principal Osc Audio Dual Out
These two identical AC coupled outputs are the main outputs of the Buchla 259e where the resulting waveform of the Principal Oscillator is heard.

The Buchla 259e Module In Use

Due to its complexity, patching the 259e can be pretty challenging but also great fun.

Here are some interesting patch-scenarios for you to try out:

Building Filter-Like Effects With 259e
Morphing between a timbre bank with little harmonics and a timbre bank with lot of harmonics can give the you the illusion of a filtered sweep. Try for example using green timbre bank 1 and red timbre bank 3 for a sine to square like sweep. Experiment with different warp amounts to give the red timbre part different harmonic content.

Synced Modulation
The fact that the Modulation Oscillator can be synced to incoming MIDI note messages and slaved to the Principal Oscillators pitch makes it interesting for audio rate modulation of pitch, Morph and Warp. Try for instance to have the Modulation oscillator to rapidly Morph the Principal Oscillator banks while in pitch track mode.

Use a envelope generator gated from a MIDI to CV module to sweep the Modulation Index and this will create a chorusing motion in the harmonics of the Principal Oscillator output.

Modulation Oscillator As External LFO

The Modulation oscillator can of course also be used as a ordinary LFO that can control external modules such as another 259e via its modulation (blue jack) and audio (silver jack) outputs. Using the Modulator Oscillator as LFO you can take advantage of the MIDI note on reset if you for instance would like a tremolo effect that will sound exactly the same for each note.

Using Aliasing As A Desirable Effect

The principal oscillator on the 259e has a pretty gritty, nice sound full of gorgeous aliasing in the upper regions. This can be used as a great source of metallic sounding effects, hihats, gongs etcetera

Using The Modulation Oscillator As Main Oscillator With Hard/Soft Sync

The Modulation Oscillator can of course also be used as main audio source. Especially the hard- and soft-sync features add new and exciting harmonic possibilities not offered by the Principal Oscillator. The Buchla 259e sync options do not sound like the other oscillators and have highly musical qualities.

Skew In Practise

The 259e's timbre banks A,B and C enable a special Memory Scan mode where parts of the 259e internal program memory is played back as audio through the Principal Oscillator outputs. In this mode, Warp will set the scope of swept memory and the FM in knob (Mem Skew) will set the offset (i e reference point in memory), where incoming CV 0 to approximately 5.88V will sweep portion of the memory around the set offset. The 259e for Modular has been modified to be used with DC signals which makes it possible to for example very precisely sequence Mem Skew positions with a CV sequencer.

Drones

The dual timbre banks of the Principal Oscillator coupled with the built-in modulation of the Modulation Oscillator makes the 259e ideal for creating long, evolving and exciting drone sounds. Just add a couple of slow LFOs sweeping Morph, Warp, Modulation Index as well as Mem Skew inputs and you'll have a killer drone in no time. Just add reverb.

Creating thundering bass sounds

By mixing the audio outputs of the Modulation Oscillator and the Principal Oscillator and slightly detuning them, the core basis for great bass sounds can easily be created. Use a lowpass filter like the Doepfer A-108 VCF8 or the Doepfer A-101-2 LPG to further filter out high frequencies and get a pronounced bottom end.

All in All: The Buchla 259e is a very creative and musical module that will bring you many hours of pleasant music making. Have fun!

Credits

Oscar Öberg – modeling, project management, validation. **Kristofer Ulfves** – presets, validation, user manual. **Igor Miná** – user manual layout, hardware photos. **Todd Barton** - testing, presets. **Patrick Detampel** – testing, feedback. **Henrik Andersson Vogel** – marketing. **Bitplant** - graphics.

5 Buchla 296e

Donald Buchla, an inventor based in San Francisco, was in the late 1960s commissioned by Avant Garde composer Morton Subotnick to build a modular instrument for composing and performing live electronic music, the Buchla Box - Don's first modular synthesizer. This evolved into Don's first commercially available system, the Buchla 100 system, followed in the 70s by the Buchla 200 system.

The Buchla 296e Spectral Processor of the Buchla 200e system was released in 2010. The 296e has its roots in the classic Buchla 296 Programmable Spectral Processor, part of the original 200-system built in the 1970s and made famous by artists such as Suzanne Ciani and Morton Subotnick. Where the original 296 had more or less the same functions, the 296e utilizes digital technology for entering and presenting information through the touch LED-display.

At the heart of the Buchla 296e are 16 well defined band-pass filters coupled with 16 VCAs and 16 envelope followers. Each band's audio level is under full control from the A-B morph, CV inputs or the Spectrum Xfer (spectral transfer) functionality. There's is also a freeze function for freezing the output level of each band in order to to sampe the spectral "footprint" into one of the A, B Levels as well as into the Decay page memories.

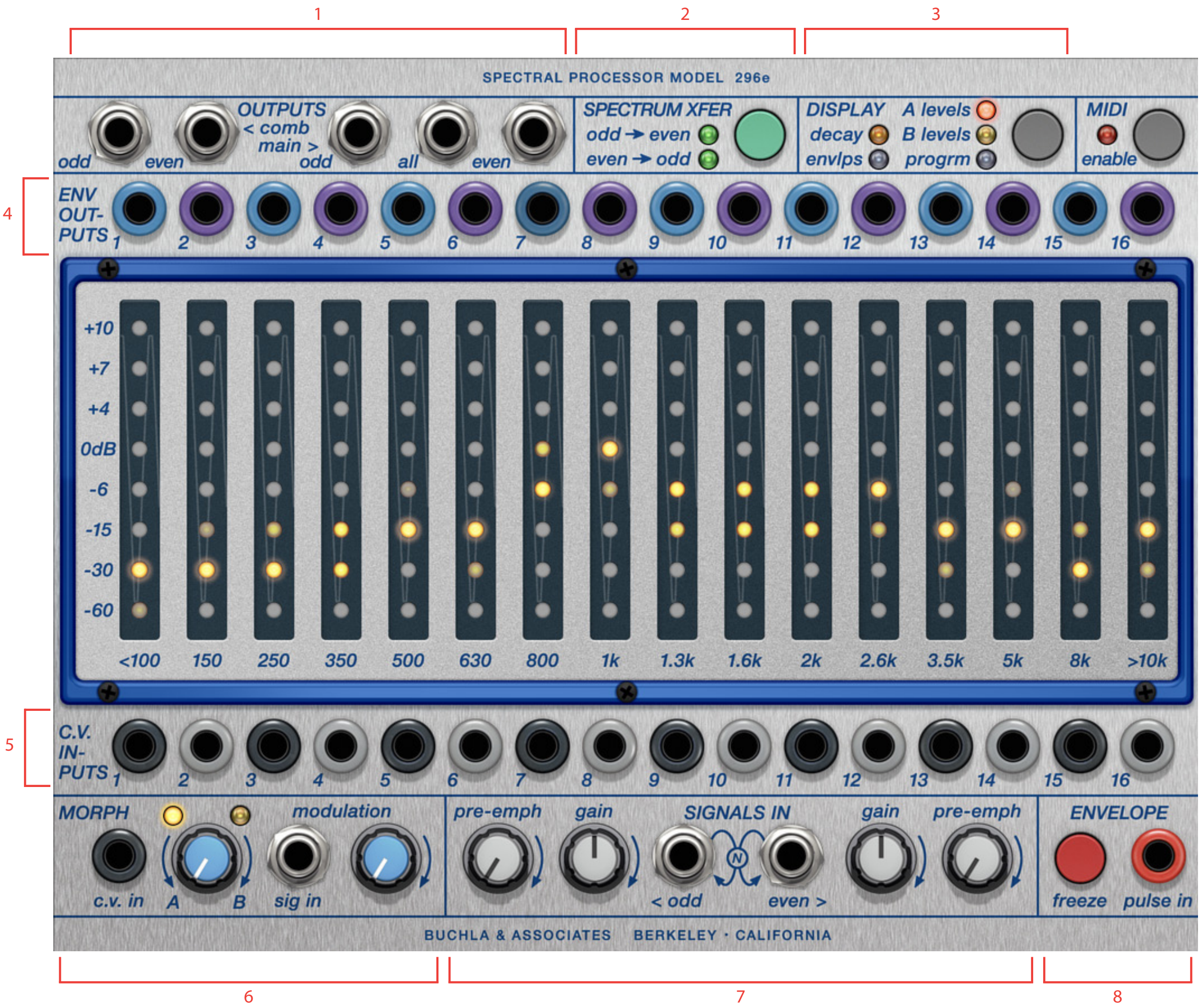
This freeze toggle button feature differs slightly from the momentary switch of the hardware original. There is also a new MIDI mode that replaces the remote button of the hardware that lets you "play" the bands of the 296e from a MIDI keyboard. These differences from the original hardware are described in detail below.

In the "In Use" section at the end of this user manual we have also listed some potential fun use-cases for the Buchla 296e Spectral Processor. Hope you will enjoy the read and have a happy experience using Buchla 296e for Softube Modular.

Buchla philosophy

In a Buchla electronic music system, audio and modulation paths are separated by their different jack and cable formats: Audio jacks are silver while modulation jacks are colored "banana"-jacks which are DC coupled. The form factor of the jacks and the different cables makes it easy to see which signal goes where and creates a natural signal flow in the Buchla environment. Softube has reimagined the Buchla 296e Spectral Processor in the Softube Modular virtual eurorack environment, adapting levels and connectivity for it to work as any other module. This means that the 296e module featured in Modular will have no limitations when it comes to connectivity – you can connect any type of output to any kind of input, whether it be audio or modulation, but for familiarity, we kept the look and color of the hardware audio and modulation jacks.

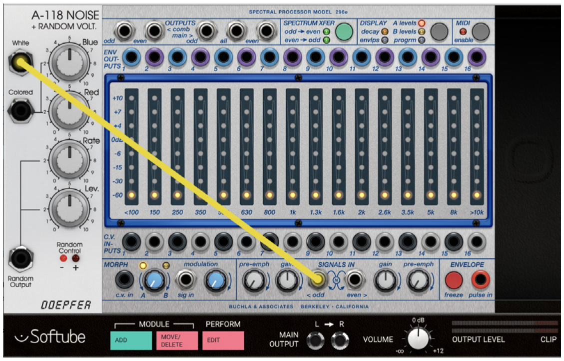
User interface



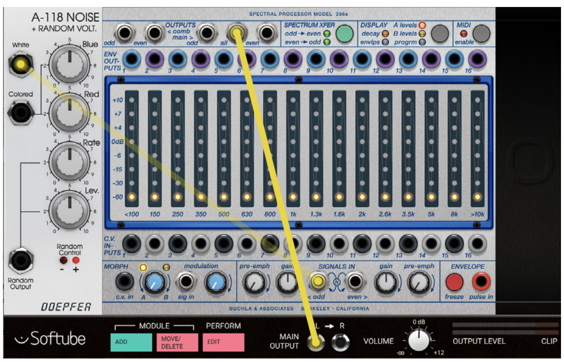
1. Audio outputs
2. Spectrum transfer modes
3. Display modes
4. Envelope CV outputs
5. VCA CV inputs
6. Morph section
7. Audio signal inputs
8. Freeze section

Getting started with Buchla 296e

The Buchla 296e Spectral Processor is a quite complex module but there's an easy way of getting started. Here's how:



1. Connect a noise-source to one of the signal in jacks at the bottom of the 296e. Make sure that Gain is turned up at least half way.

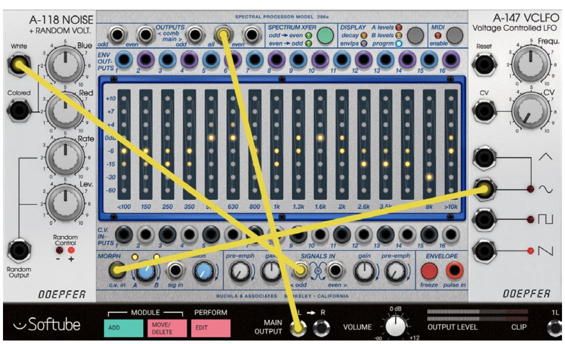


2. Connect the All output at the top of the 296e to Modular's main outputs.

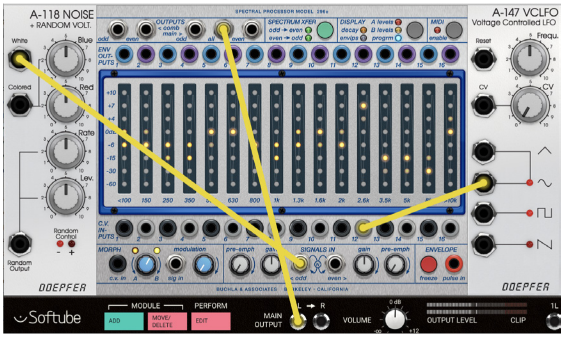
3. Select Display mode A by repeatedly clicking on the Display Levels button until the A Levels LED is lit (orange). Now you can use your mouse-pointer to “draw” a curve across the 16 bands of the 296e. You'll notice that you'll be hearing sound from the corresponding band when its level is turned up.

4. Click on the Display Levels button once more and the B Levels LED lights up (yellow). Again, use the mouse-pointer to "draw" a curve across the 16 bands, but this time there won't be any audible difference. This is because we're still listening to the levels set by the A Levels page, indicated by fact that the Morph A-B knob is set fully counter-clockwise (Morph A).

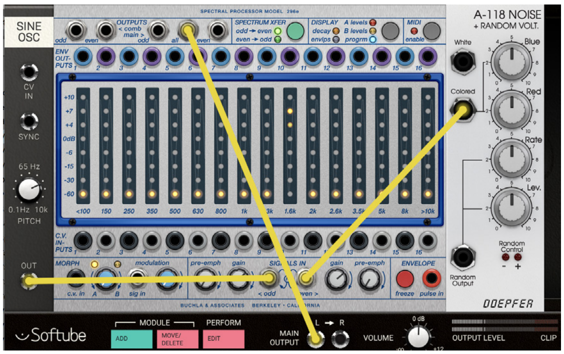
5. Now, click on the Display Levels button once more and the "progrm" option is selected (blue LED lit). Now the LED touch-display will show the resulting output levels. Use the morph A-B knob to "morph" between the A levels and B levels, meaning that output levels adjust seamlessly between the amplitude levels set at the A Levels and B Levels page.



6. Now let's control this motion remotely by connecting a modulator to one of the two modulation inputs. For this purpose I'm using a Doepfer A-147 LFO and connecting it to the Morph CV input. Now, watch as the morph between the A Levels and B Levels are automated. Note that you'll still hear the resulting "programmed output" at the All out audio output although another Display mode is selected and reflected in the touch-display.



7. OK, let's repatch the LFO to control the VCA of a single band instead. By patching the LFO signal into the 2.6k (the 12th band) input, the LFO will add to the signals controlled by the curves drawn on the A and B levels page.



8. Now, let's explore the Spectrum Xfer functionality of the 296e. Xfer meaning transfer and spectrum transfer makes it possible to cross-couple the envelope outputs of the odd (1, 3, 5, 7, 9, 11, 13, 15) to the VCA inputs of the even bands (2, 4, 6, 8, 10, 12, 14, 16) and vice versa.

Let's erase everything in the patch and start again by adding a Sine oscillator and inserting that into the odd input and turning up the gain slightly past 12 o'clock. Now let's add a A-118 noise source and connect it to the even input, turning up the gain also slightly past 12 o'clock. So far you will hear both sources (the sine-oscillator and noise) from the All output of the 296e.

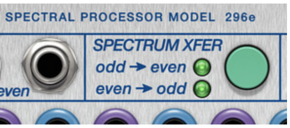
Now, when clicking on the Spectrum Xfer button and Odd>Even LED lights up, you will only hear the harmonic spectrum transfer from the odd bands applied to the even bands. And since the sine-oscillator does not contain any additional harmonics besides the fundamental, the pitch of the sine-oscillator now basically decides which of the even bands that is amplified. Try changing the pitch of the sine-oscillator and you'll see (and hear) the spectrum transfer follow.

Note that, while in Spectrum Xfer mode, morph or externally controlled CV is ignored (also described in further detail below).

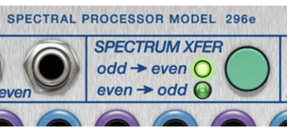
9. As a final lesson in this "getting started"-segment, let's look closer at the envelope follower outputs. Let's use a white noise source to feed the 296e again. To the right in this patch I'm setting up four sine oscillators with different tunings, each through their own VCA. The VCA themselves are driven from the envelope outputs of bands 4, 7, 10, and 13. The resulting sound is a kind of a "tuned particle noise". Now, by using the display envelope mode (as pictured) you can "draw" the decay curve differently for the different bands (band 4, 7, 10, and 13 used here).

Parameters

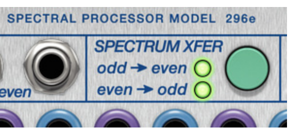
Spectrum Xfer mode button This button switches between the four different spectral transfer modes in the 296e. In normal mode (off), both of the Spectrum Xfer mode LEDs (odd>even and even>odd) are turned off meaning that no internal coupling is being made between the odd and even bands.



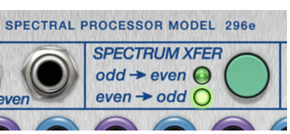
Clicking on the Xfer mode button once will light the odd>even LED and will internally couple all odd bands envelope outputs with all even bands CV inputs. In this mode, only the Even output will output audio.



Clicking on the Spectrum Xfer mode button again and both the odd>even and even>odd LED will lit up, indicating internal cross-modulation between all odd and even bands.



Clicking on the Spectrum Xfer mode button yet again will light up only even>odd LED indicating a odd to even internal coupling. In this mode, only the Odd output will output audio.



Clicking on Spectrum Xfer a fourth time, will turn off the spectral transfer altogether and the 296e is back in normal operation mode again.

Note: In all of the active spectral transfer modes, all external signals connected to the CV inputs are neglected, morph is not active and the envelope output signals will reflect the bands internal routing.

Display mode button Click this button to switch between the different display modes for the touch-display. Some modes are for input of data ("A Levels", "B Levels" and "Decay") while some modes only will show the results ("Progrm" and "Envelps"):

A levels In this mode, the touch-display is used for programing the odd and even levels for the Morph A. Morph can then be used to fade between the levels set in the A and B levels display pages (for further description see below).

B levels In this mode, the touch-display is used for programing the odd and even levels for the Morph B (for further description of Morph see below).

Progrm This mode uses the touch-display merely as a meter display for showing the resulting current output for the bands.

This means that the Progrm page in "normal mode" (no spectrum xfer mode activated) will show the levels applied to the VCAs for each band. This can be the blend of the A and B levels via the morph knob, as well as added external modulation via each band's CV input.

In Spectrum Xfer mode, the Progrm page will show the levels of the VCAs, but in this mode they're coming from the envelopes that are internally coupled while all other modulation (Morph blend, CV inputs) are neglected. See also further comment above in the Spectrum Xfer mode description.

Decay This mode uses the touch-display to set the envelope-follower decay time for each band. The response is exponential which means that the first bottom five LEDs on the touch display will have a very subtle change in decay, where the top three LEDs will have quite drastic changes. This is intentional and works the same way as the original hardware. Maximum decay time around 40 seconds.

Envlps This mode uses the touch-display to display the envelope-follower outputs for each band.

Note: You can also go between the different display modes by clicking on each mode's text. In this way it is possible to for example jump directly from A levels to Decay and vice versa.

MIDI enable This button enables MIDI mode. When MIDI mode is enabled, the 296e first listens to the incoming MIDI (blinking LED), locking on to any channel where a MIDI note on message is found (LED becomes solid). MIDI notes C0-D#1 plays the 16 bands with MIDI velocity controlling the amplitude of each band. There's also other repeating keyranges control at C2-D#3, C4-D#5, C6-D#7.

Note that in MIDI mode applied Morph is disabled. Also note that Spectrum Xfer modes will override MIDI mode input, meaning that MIDI note information will be ignored in Spectrum Xfer mode.

Morph Knob A-B This knob morphs between the A Levels and B Levels. The resulting output levels are displayed in Progrm mode (see description above).

Sig in Modulation This knob sets the amount of modulation applied from the signal at the “sign in” input.

Pre-Emphasis Odd This sets the amount of high frequency pre-emphasis applied to the Odd signal.

Gain Odd This is the gain amount of the Odd signal. It determines the loudness level of the signal input at the Odd input. Note that although the signal input at the Odd input is pre-patched to the Even input, the Gain knobs are not and thus still need to be set separately.

Gain Even This is the gain amount of the Even signal. It determines the loudness level of the signal input at the Even input.

Pre-Emphasis Even This set the amount of high frequency pre-emphasis applied to the Even signal.

Freeze This button is kind of special. It freezes the current state of all of the 296e levels applied to each band's output VCAs until clicked again; a toggle button functionality. This behavior is equal to holding down the freeze button on a real world hardware 296e. In addition to this, when toggling from freeze mode to normal mode in Level A, B, or Decay display mode, the spectrum of the current input signals is sampled to the memory of that particular page. Toggle button active is indicated by blinking LEDs on the touch display.

Note: When the 296e is in freeze state via the Freeze toggle button, no other buttons or touch-display entry will be accepted until freeze toggle button is turned off again (LEDs stop blinking). However, this does not apply to externally modulated freeze states (via the Pulse in jack) where the only parameter temporarily off limits to the user, is editing the touch-display while the 296e is sampling the audio spectrum into memory (during Pulse in jack at high, +5V).

Indicators

Spectrum Xfer mode LEDs When any of the Spectrum transfer ("Xfer") mode LED is lit, this indicates that envelope outputs of the Odd bands are coupled (pre-patched) to the even bands and vise versa. When both "Odd>Even" and "Even>Odd" LEDs are lit at the same time, this indicates full cross-modulation (not easily possible via external cables). When both Spectrum Xfer mode LEDs are turned off, "normal" mode (no pre-patching) is activated.

Display mode LEDs Indicates current state of the touch-display (A levels, B levels, Progrm, Decay, or Envlps).

MIDI mode LED This LED indicates MIDI mode activated (LED is lit), MIDI learn (LED is blinking while the 296e is waiting for a MIDI note on message on a MIDI channel) or MIDI mode off (LED is unlit).

Program Level1-16 A levels, B levels – these are two differently programmed sets of output levels of the 296e that can be setup to be morphed between. Morph means that the two sets of output levels are continuously changed into the other at the turn of the Morph knob or by applying external modulation to the Morph CV input. The resulting (Morph) curve is displayed in Program mode ("progrm" selected). When Decay mode is selected, the touch-display will show the envelope decay times for each band and in Envelope mode ("envlps" selected), the resulting output envelopes CV outputs for each band is displayed. If the Freeze function is manually latched by the toggle button, frozen bands will be indicated by blinking LEDs on the touch display.

Envelope outputs 1-16 Envelope display mode (Envlps) shows the output state of the envelope-followers at each band on the touch-display (but with no touch in this mode as it is purely for visual feedback).

Morph LEDs The Morph LEDs indicate the transition between the A levels and B levels, from Morph A LED fully lit while Morph B LED is unlit to gradually the opposite.

Inputs

CV input 1-16 These are the CV inputs for controlling the attenuated output of 296e's 16 bands. The CV signal applied here is summed to that of the drawn A and B level curves.

Morph CV in This is the CV input for external control over the 296e Morph functionality. Any CV input here is added to the offset set by the knob.

Audio Mod in (sig in) This is the audio input for external modulation of the 296e Morph function. Any signal input here will do.

Odd Signal in This is one of the two audio inputs for the 296e, the input for the odd bands. Both of the inputs are normalized to each other which means that if no cable is inserted in the other input jack, a signal input to either one will be fed to both. For example, if a signal is inserted at the Odd Signal in jack without any cable present at the Even signal in jack, the same signal will be fed to all bands - odd and even - of the 296e. Note that both sides still have their own dedicated gain and pre-emphasis gain controls though.

Even Signal in This is the other of the two audio inputs for the 296e, the input for the even bands. The same description as for the Odd Signal in jack applies here (see above).

Pulse in This CV input is the remote control of the Freeze functionality. As soon as a logical high (CV above +2.5V) appears on this input, the Freeze function is engaged which is indicated by flashing LED display levels.

Outputs

Odd Comb out This is the comb filter outputs for all odd bands before internal attenuation.

Even Comb out This is the comb filter outputs for all even bands before internal attenuation.

Odd out This is the attenuated output for all odd bands.

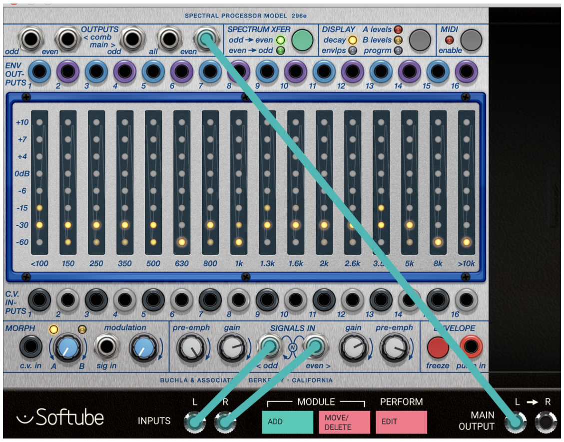
All out This is the attenuated output for all bands, odd and even. This is to be considered the most commonly used output.

Even out This is the attenuated output for all even bands.

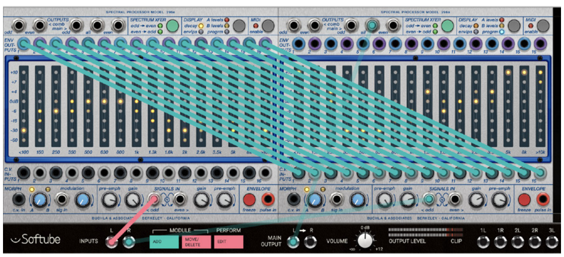
Envelope out bands 1-16 These are the CV outputs from each band's envelope follower. These levels are affected by inputs signals, internal or external linking or attenuation and the decay times set in the Decay display mode.

Note: The visualisation of the CV outputs at each Envelope out band is available in Envelope display mode ("envlps" selected).

The Buchla 296e module in use

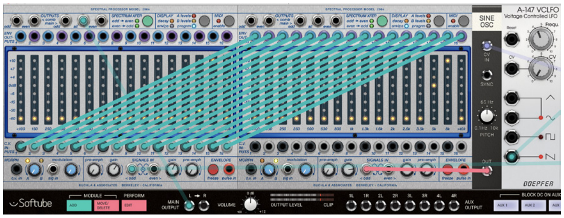


In this patch we have connected a rhythmic source like a drum loop to the left inputs of Modular FX and a pad-like loop to the right inputs. By using the Spectrum Xfer mode odd>even and only listening to the even outputs, a crude version of a vocoder is created (8 band vocoder). Try adjusting the Decay times at the Decay page for different flavors.

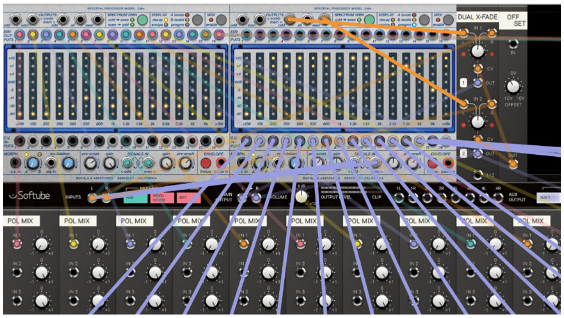


We also use multiple instances of the buchla 296e together to create a "fuller" vocoder sound than described above. The left 296e is fed by the rhythmic part coming from the left input and the right 296e is processing the audio coming in through the right input. The key thing here is the envelope outputs of the left 296e that are routed to the CV inputs of the bands of right 296e.

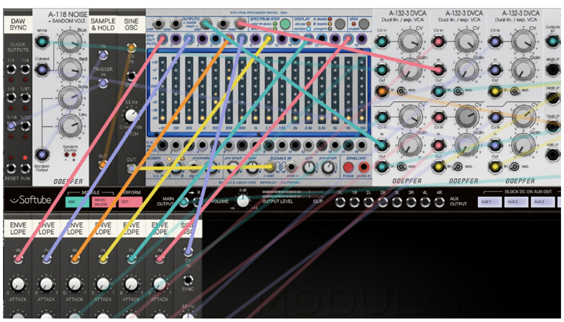
The 296e's envelopes can of course be used to extract beat information out of an audio-track fed into Modular FX. Here's an example of how you could set up the 296e together with some Heartbeat-modules to produce a drum-replacer.



In the original 296 from the Buchla 200-system there was a function called programmed control that let you use the 296 as a bandpass focusing filter. The patch pictured above is something in that direction with an analyser part (right) that tracks the sineocillator and a carrier part that filters the A-110 VCO.



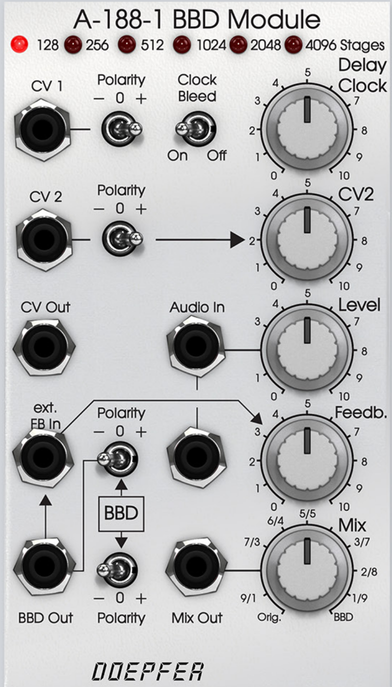
This is an interesting concept - using one 296e as analyser (left) and then using the inverted envelope signal for each band to reduce gain off the second 296e with all levels at full. This gives us control over the ducking decay for each band as well as a parametric EQ at the same time. The Dual X-fade and offset setup at the right is for blending some original signal back in.



You can also take advantage of the 296e's analysing capabilities and use the envelope as a kind of "audio-driven sequencer". In this patch we use the pitch of the top left sine oscillator to drive the 296e, but of course any type of audio could be used.

Credits

Jacopo Lovatello – programming, modeling. **Kristofer Ulfves** – product owner, testing, preset, user manual. **Kim Larsson** – mentoring. **Björn Rödseth** – mentoring. **Arvid Rosén** – mentoring. **Bitplant** – GUI graphics. **Fanny Hökars** – user manual layout.

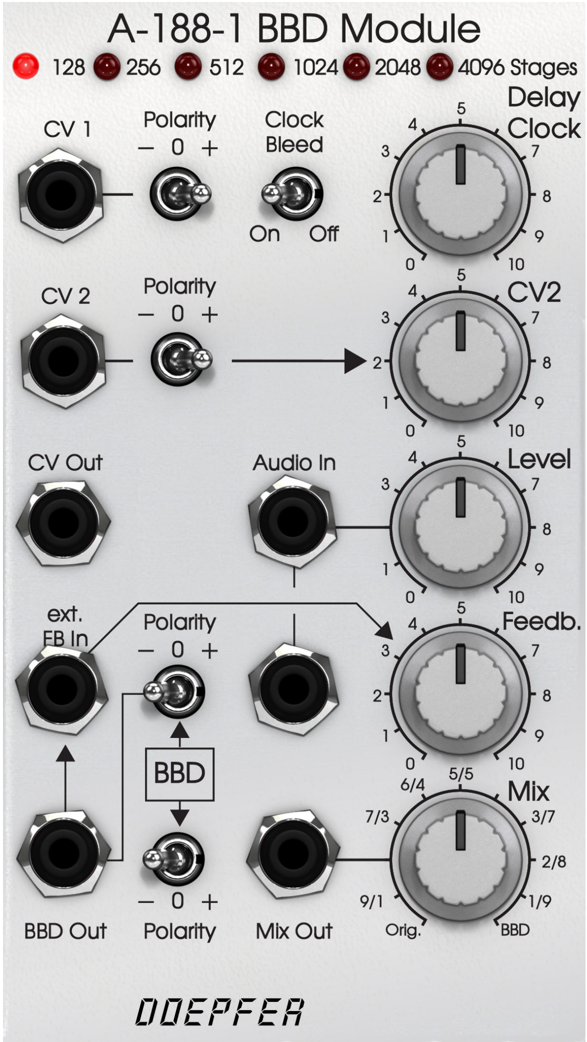


7 Doepfer A-188-1 BBD module

The Doepfer A-188-1 BBD module for Softube Modular is an emulation of a so called Bucket Brigade Device, a kind of early "digital" memory used in the 70s and 80s in all kinds of delay-based effects such as choruses, flangers, ensemble-effects and of course "analog" echo-units.

A Bucket Brigade Device can be described as a long row of level-memories that pass their position at a high speed providing a rudimentary form of digital memory. The speed at which these memories are passed through the module is determined by an internal clock, normally run at very high speed. The length of the memory held is then determined by the number of stages in the BBD-chip used in the module. The BBD chips have a very specific working area that only guarantees correct behavior (no degradation of the sound) above a certain clock-frequency. This is why gradual degradation of the delayed signal occurs when the clocking-frequency of a BBD chip is lowered.

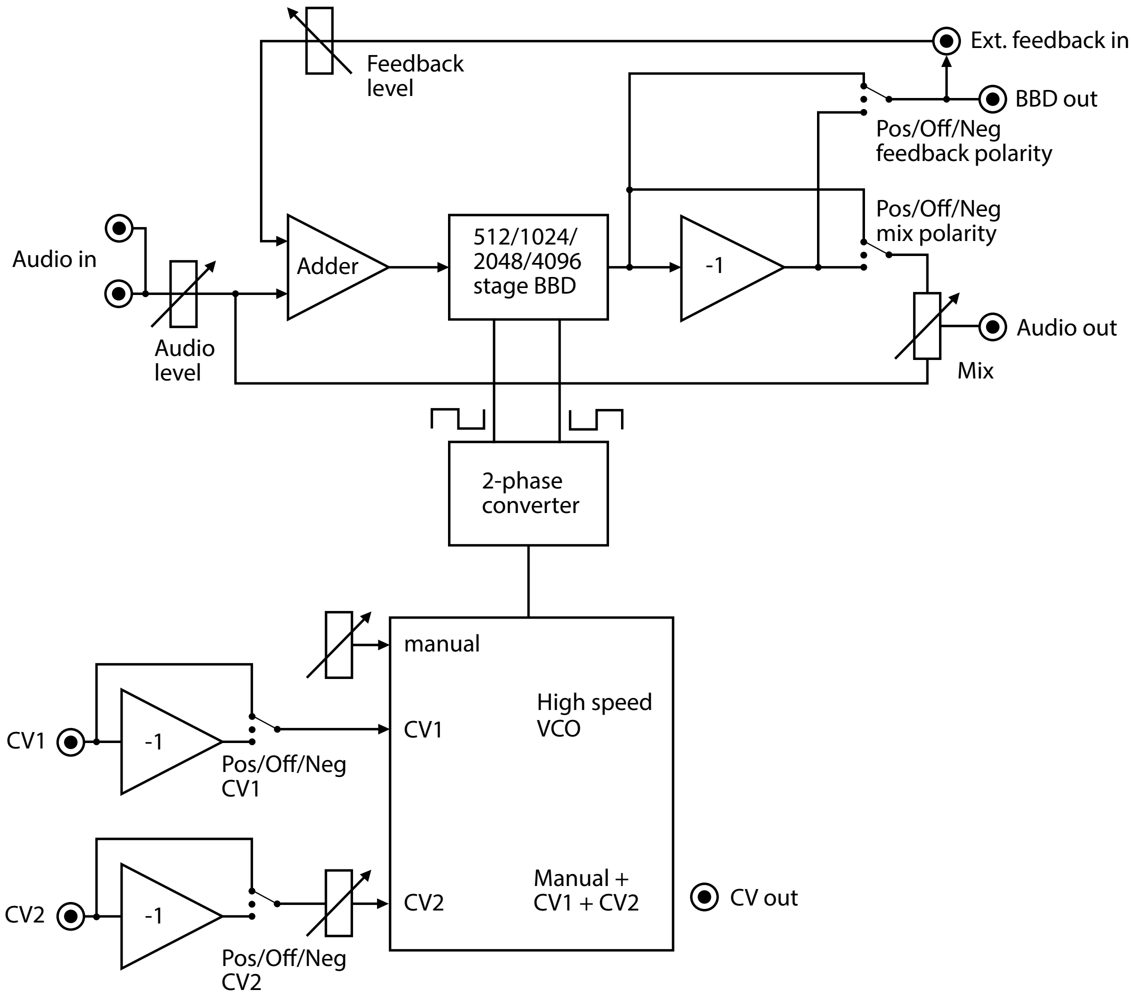
The Doepfer A-188-1 hardware module is normally shipped with one specific BBD chip in order to fit a specific purpose. Some modules ship with a chip with 128 stages, some with 256 stages and so on. The Doepfer A-188-1 hardware we have used came with 1024 stages using a modern BL 3207 BBD chip inside, and was thus the basis of the model used in this Softube emulation. This chip is a bit noisy and the module has apparent clock-bleed through when lowering the clock-frequency, so we have therefore made this noise and clock-bleed optional via the new Clock-bleed switch. Another change from the hardware version is the absence of the clock out and external clock input, simply because it was not possible to emulate and support the extremely high clock-frequencies in the MHz area within Modular.



Overview

The Doepfer A-188-1 BBD module has been designed with experimentation in mind. This is why the signal going through the BBD chip is not filtered or processed in any way (as is often the case in dedicated units such as choruses). Thus the output signal is deliberately rough sounding and full of aliasing artifacts when input is pushed too high or internal clock-frequency is lowered. At full feedback the module can self-oscillate or "scream" in a distinctive (pleasing) manner.

The internal clock can be CV-controlled via the CV1 or CV2 input jacks (the latter also scaled via the CV2 attenuator knob). Built into the A-188-1 BBD is also a feedback path with positive or negative feedback controlled via the feedback attenuator knob. This path can be broken and fed via external modules in order to gain control of this aspect as well (see further examples below).


Feedback
level
Ext. feedback in
BBD out
Pos/Off/Neg
feedback polarity
Pos/Off/Neg
mix polarity
Audio in
Audio
level
Adder
512/1024/
2048/4096
stage BBD
-1
Audio out
Mix
2-phase
converter
manual
High speed
VCO
CV1
-1
Pos/Off/Neg
CV1
CV1
CV2
-1
Pos/Off/Neg
CV2
CV2
Manual +
CV1 + CV2
CV out

Parameters

Stages buttons 128/256/512/1024/2048/ 4096: these buttons determine how many stages of BBD chip is emulated with the A-188-1 BBD module. Fewer stages tends to work well for short delay-based effects such as flangers and choruses, while more stages work better for longer echoes and reverb-like effects.

Polarity switch CV1 This switch sets the effect incoming control voltage in the CV1 input will have on the internal delay clock speed. When the polarity switch is set to "-" incoming CV signal through the CV1 jack will be inverted, while when polarity switch is at "+" CV is positive. "0" position turns incoming CV off on the CV1 input.

Polarity switch CV1 This switch sets the effect incoming control voltage in the CV1 input will have on the internal delay clock speed. When the polarity switch is set to "-" incoming CV signal through the CV1 jack will be inverted, while when polarity switch is at "+" CV is positive. "0" position will turns incoming CV off on the CV1 input.

Clock Bleed switch This switch turns on and off the authentic clock-bleed and noise that is part of the original Doepfer A-188-1 hardware. We thought it would be nice to have the possibility to get rid of these sound artifacts so we added this switch in the place where the original hardware has its Clock out and external clock input jacks (see above).

Delay Clock knob This knob sets the internal clock speed for the Doepfer A-188-1 and thus also the amount of delay introduced as the signal is stored and carried through the bucket brigade stages. This knob set at high values or fully clock-wise will cause short delays while preserving the fidelity of the original signal. With lower values, longer delays will be produced but also less fidelity in the resulting delayed sound. You will also notice hissing noise and apparent clock-bleed when lowering the delay knob while the clock-bleed switch is in the "on" position.

Polarity switch CV2 This switch works similar to the Polarity switch for CV1: It sets the effect incoming control voltage in the CV2 input will have on the internal delay clock speed. When the polarity switch is set to "-" incoming CV signal through the CV2 jack will be inverted, while when polarity switch is at "+" CV is positive. "0" position turns incoming CV off on the CV2 input. Notice that the control voltage input through CV2 input is scaled via the CV2 attenuator knob described below.

CV2 knob The CV2 attenuator knob scales the incoming control voltage on the CV2 input jack. When turned fully clockwise, the full input control signal is affecting the delay clock speed.

Audio input Level knob This knob set the level of the audio input. This knob can also be used to increase the audio level beyond the point where the BBD starts to distort which might be desirable in some cases.

Feedback Polarity switch The Feedback Polarity switch will determine whether the wet BBD signal fed back to the input is polarity flipped or not. Negative ("-") value will result in a different sounding feedback than a Positive ("+") one. When the Feedback Polarity switch is set to "0", no feedback is active in the module.

Feedback knob This is the attenuation of the wet BBD signal sent through the feedback path. When set at fully clockwise, attenuation is 0% which means that feedback will go well into self-oscillation.

Mix out Polarity switch The Mix out Polarity switch will determine whether the wet BBD signal sent to the Mix Out jack is polarity flipped or not. Negative ("-") value will result in a different sounding feedback than a Positive ("+") one. When the Mix out Polarity switch is set to "0", no wet BBD component will be heard in the mix output.

Mix knob This knob sets the mix between the clean signal and the wet, delayed, signal for the Mix Out jack.

Indicators

Stages buttons indicators The stages LED buttons light up to indicate which number of stages is selected.

Inputs

CV1 This is the first control voltage input for modulating the delay time from an external modulation source.

CV2 This is the second control voltage input for modulating the delay time from an external modulation source.

Audio in These are the dual audio inputs (one to the left of the Audio input Level knob and one to the left of the Feedback knob). Inputting audio into both will create a 50-50 mix of the two inputs before the Audio Level attenuator knob.

Ext FB in	External Feedback input. This is the input for an external feedback loop where you for example could insert a VCA to CV control feedback (see user example below).

Outputs

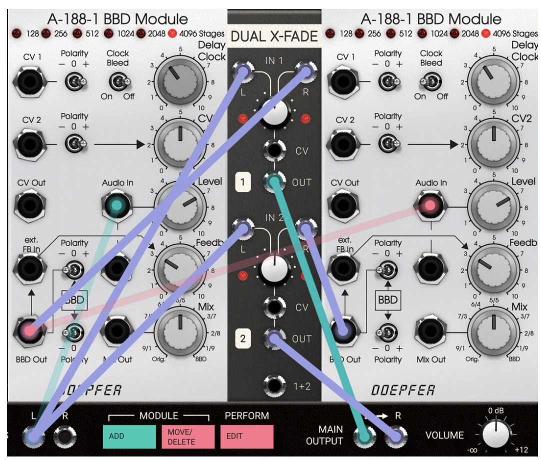
CV out	This is the tracking CV output for the delay clock frequency. This is great for example to create external anti-aliasing filters with CV controlled cutoff that track the delay time.
BBD out	This is the fully wet (effect) output of the BBD module. Great as the output entry point for an external feedback loop.
Mix out	This is the main output for the mixed signal between the dry (input) and wet signal of the Doepfer A-188-1 BBD module.

The Doepfer A-188-1 BBD module in use

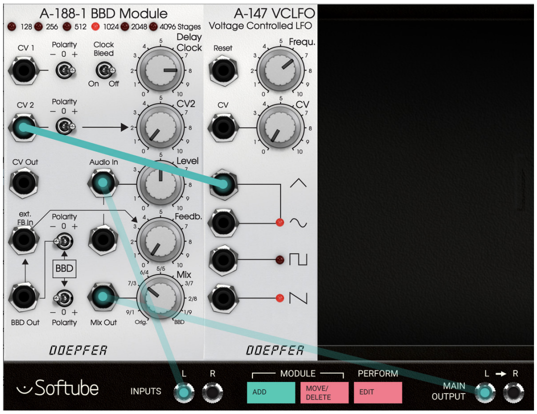
The Doepfer A-188-1 BBD can be used for creating all kinds of time-based effects. Everything from obvious echos and reverb-like effects to choruses, flangers and ensemble effects. But the Doepfer A-188-1 can also be used as crude distortion effect or feedback oscillator for Karplus-Strong synthesis.

The most obvious effect the Doepfer A-188-1 BBD module is used for is creating a simple echo effect. This is done by using the 4096 mode and lowering the delay rate to a figure somewhere between 3 and 6. Remove the clock-bleed if you like by flipping off the clock-bleed switch or by external low pass filtering.

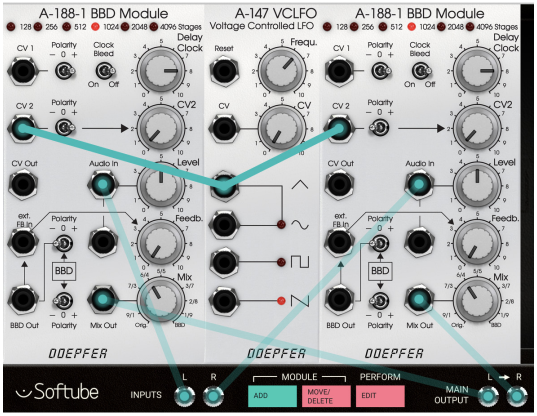
A pingpong echo can also be achieved by using two BBD modules and an external mixer for the mixing of the dry and delayed signal.



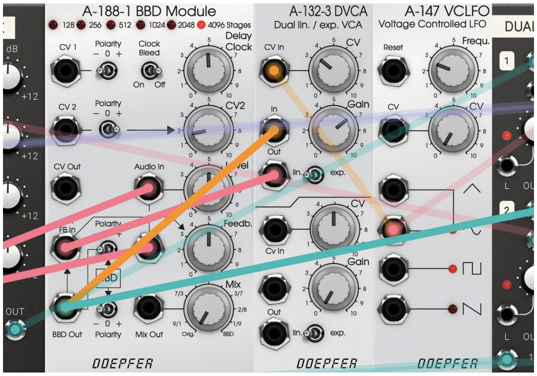
Building a simple chorus using the Doepfer A-188-1 BBD only requires an external LFO to create some movement in the short delay line.



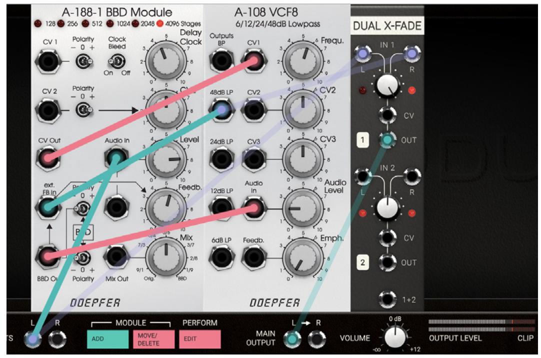
A Stereo chorus is just an extension of the patch described above. Just add another BBD-module and feed that from the same LFO, but with inverse phase. This is done by flipping the CV2 Polarity switch to "-".



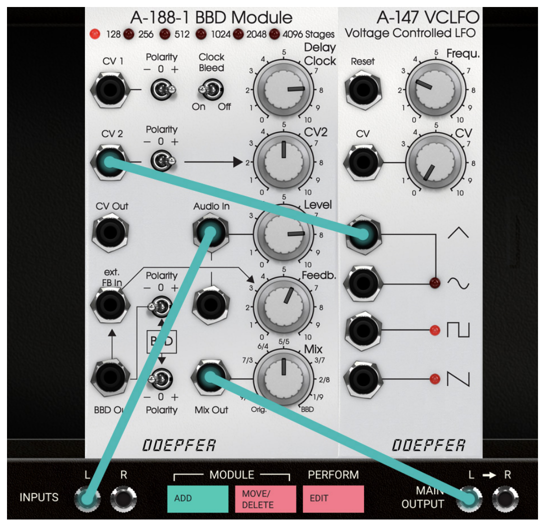
The BBD Out and ext. FB in jacks can be used to attach an external feedback loop to the BBD module. In this example an external Doepfer A-132-3 VCA module is used together with a Doepfer A-147 LFO to periodically control the feedback of the short echo created by the BBD module.



In this example we're using an external tracking low pass filter CV in the feedback path in order to remove clock bleed and get a "muddier" and "dubby" echo sound. Notice that the filter cutoff frequency now perfectly tracks the delay length, so that short echoes are brighter while longer echoes are duller with less high end.

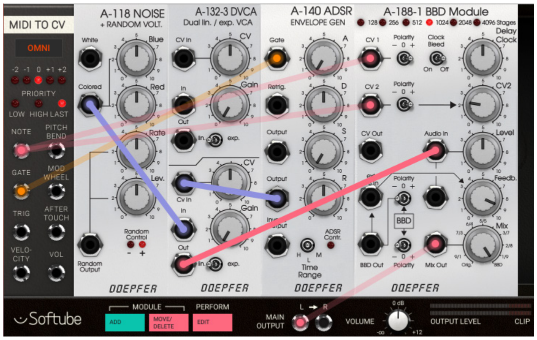


Creating a flanger patch with the Doepfer A-188-1 BBD module is very easy. All we need are to use short delay times (using 128 stages here), and to sweep that delay time up and down from an LFO. Experiment with modulation depth by adjusting the CV2 knob and feedback by adjusting the feedback knob.



The Doepfer A-188-1 BBD module has a very distinctive distortion that can be utilised for striking lo-fi effects. Use very few stages (128 or 256) and change the delay length for into even more lo-fi, bit-crunchy territory. In this example we also use an external mixer to push the input volume even harder..

The Doepfer A-188-1 BBD module can be used for Karplus-Strong synthesis. Pictured here is a patch where a short burst of noise is used to trigger the BBD module acting as a resonating string. The CV inputs are adjusted to enable this patch to track somewhere around an octave, adjustment for broader ranges could be tricky though.



Credits

Jacopo Lovatello – programming, modeling. **Arvid Rosén** – modelling, mentoring. **Kristofer Ulfves** - product owner, testing, preset, user manual. **Niklas Odelholm** – GUI graphics. **Fanny Hökars** – user manual layout. **Igor Miná** - hardware photos.

8 Doepfer A-101-2 Vactrol LPG

THE DOEPFER A-101-2 VACTROL LPG IS a vactrol based combination of a 12 dB Low Pass filter (LP) and VCA. This combination is often called *Low Pass Gate* (LPG), a term coined by synth legend **Don Buchla** for the **Buchla 292** module of his classic 200-system range. The original Buchla module is still highly sought after and has been the inspiration for a number newer modules over the years, the **Doepfer A-101-2** being one of them.

The **Doepfer A-101-2 Vactrol LPG** features vactrol control of its LP and VCA parts, which gives it that desirable woody, organic response revered by many. In the LPG mode the sound becomes more dull as the loudness decreases, a characteristic of all acoustic instruments; the harder you strike, pluck, or bow, the richer the overtone structure. Consequently, the Doepfer A-101-2 Vactrol LPG a very good module for imitating the tonal response of tuned percussion. By exciting the vactrols with short bursts of noise or impulse spikes through the CV inputs, the low pass gate response can be used to produce sounds resembling hand drums, steel drums or marimba.

Several different A-101-2 units were used during measuring process, due to the individuality of each unit. This is the origin of the Vact switch which we added for users of Modular to be able to alter between three different sets of vactrols for different flavors and responses.

The classic "LPG", low pass gate, mode is called L+V/EXT on the hardware and user interface.

User Interface

F/A Cutoff for filter, amp offset for VCA or both at the same time (L+V/EXT mode). Function of this parameter is dependent on position of **Function** switch and **G1** and **G2** inputs.

CV2 Attenuator for incoming **CV2** input. At fully clockwise position, incoming CV at **CV2** input will have full range, in effect the same as **CV1** input.

Lev Attenuator for incoming audio. When set above 5, normal oscillator or modulator level signals (such as the A-110 VCO) will distort.

Res This knob controls the resonance of the low pass filter. When set above 8, the filter will go into self oscillation when the **Function** switch is set to LP mode (see below).

Function This switch determines the operation mode of the Doepfer A-101-2. When set to LP mode (switch set to the left) only the low pass filter part will be active. In L+V/EXT mode both the filter and the VCA will be active, meaning that audio is fed first through filter then VCA. EXT means that the **G1** and the **G2** inputs both are active (see below). In VCA mode (switch set to the right), only the VCA part of the Doepfer A-101-2 will be active and thus no low pass filtering will take place.

Vact The switch selects between three different types of vactrol responses. F is for fast response times, M is medium and S is for slow response.

Indicators

LED The red LED at the top right reflects the accumulated CV level into the LPG. The LED itself is set in series with the vactrols in the Doepfer A-101-2 Vactrol LPG hardware circuit, thus accurately showing the CV response to the **F/A** knob and the CV inputs.

Inputs

CV in 1 Full range CV input for controlling the cutoff for filter, amp offset for VCA or both at the same time (L+V/EXT mode).

CV in 2 CV input for controlling the cutoff for filter, amp offset for VCA or both at the same time (L+V/EXT mode). The effect of the CV input here is dependent on the **CV2** attenuator knob.

Audio In Main audio input. Connected with the input level knob (**Lev**) that controls the incoming audio level (see above).

G1 External control input for the Doepfer A-101-2 Vactrol LPG operating mode. When the **Function** switch is set to L+V/EXT position, a logic high or low will select operating mode as according to the chart on the panel (see description below).

G2 The other control input for external control of the Doepfer A-101-2 Vactrol LPG operating mode. When the **Function** switch is set to L+V/EXT position, a logic high or low will select operating mode according to the chart on the panel: **G1** set high (above approx. 0.55 V) and **G2** set low (below approx. 0.55 V) will set the Doepfer A-101-2 Vactrol LPG in LP mode. **G1** set low and **G2** set high will result in VCA mode, and finally both **G1** and **G2** set low will set to the LPG mode L+V/EXT (default).

Note: Both inputs high (G1 and G2) is a forbidden state, which will bypass the signal through the module all together. Also worth noting is that when the Function switch is set to LP or VCA position, the G1 and G2 inputs set high will repel the effect (G2 active in LP mode and G1 active in VCA mode). These are all undocumented features of the original hardware and will produce unpredictable results.

Outputs

Audio Out	This is the output of the Doepfer A-101-2 Vactrol LPG, and its output is determined by how the **Function** switch is set. In LP mode only low pass filter is output here, in L+V / EXT mode the signal put through both low pass filter and then the VCA. And when the **Function** switch is set to VCA, only that the signal passed through the VCA will be heard.

In Use

The Doepfer A-101-2 Vactrol LPG has a very organic response due to its vactrol control, which can be used to closely mimic the characteristics of an acoustic instrument. The vactrol non-linearities makes the low pass gate a very good module for imitating the tonal response of tuned percussion. By feeding in short bursts of noise or impulse spikes through the CV inputs, the low pass gate can used to produce sounds resembling hand drums, steel drums or marimba. It can of course also be used as a pretty straight forward low pass filter or just VCA.

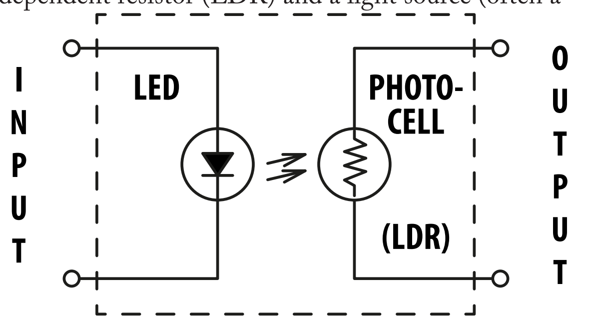
The Doepfer A-101-2 Vactrol LPG also passes and processes DC voltage, which makes it possible to pass a DC offset through and ping the LPG through the CV inputs to get the vactrol response CV on the output.

The Doepfer A-101-2 Vactrol LPG has an input that is quite sensitive in LP mode, so you can use this to distort your signals easily by cranking up the input level.

Use the **G1** and **G2** inputs with the switch in its middle position (L+V/EXT) to sequence the output mode (low pass, LPG or VCA mode) using for example sequencer gates or high outputs from the logic and signal modules.

What the Heck are Vactrols?

Vactrols are a common kind of optical isolator often used in circuitry that requires different circuits to be electronically isolated from each other but still able to interact. A vactrol consists of a combination of a light dependent resistor (LDR) and a light source (often a



LED, light emitting diode) both integrated in a small light-proof case.

The vactrol works by a very simple principle: When input current is applied to the LED element the output LDR turns on and, as the only connection in between the two is light, the output produced is largely dependent on the LDR response. An increase in the LED brightness will cause a decrease in the LDR resistance. This effect can be used in circuits that require variable resistors to obtain the desired function (for example

VCFs, VCAs, VC phasers, VC trigger delays, VC slew limiters, VC Envelope Generators, VC LFOs, VCOs and many more).

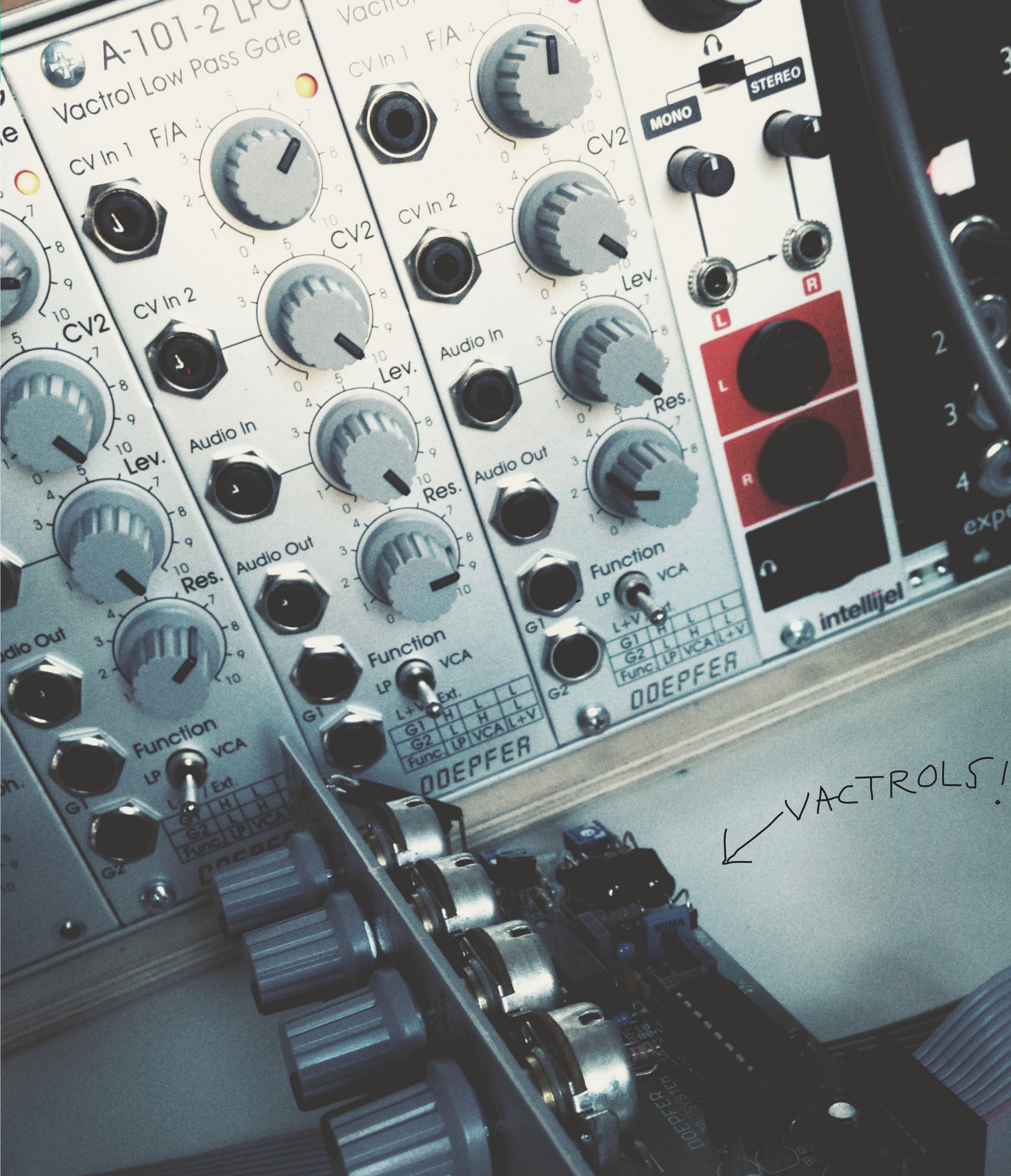
However, mechanical and electrical tolerances will cause each vactrol to behave a bit different from another (for example due to differences in alignment and distance between LED and LDR). This is why identical circuits featuring vactrols will behave or sound a little bit different from each other. Also, another aspect of the vactrol response is the LDR element; that will not respond immediately to the illumination change induced by the LED but will have a certain "sluggishness" built in. The LDR response curve is largely logarithmic and thus it may take a few seconds until the LDR reaches its "dark resistance" (the maximum resistance without illumination). In the upper brightness range it will also respond much faster: up to 50 Hz and more, which corresponds to a response time in the 10 ms range. The essentials of vactrols' behavior will depend very much upon the LDR type and the light-sensitive material used.

As a consequence of the things described above, vactrol based compressor circuits (often called opto compressors) have historically often been considered particularly natural sounding; while a modern VCA compressor can respond almost theoretically perfect with a constant ratio and predictable attack/release curves, many older optical designs have strange attack and release characteristics. In optical compressors, this is originally due to the relatively slow response of a vactrol compared with a transistor-VCA based circuit. This is perhaps the reason why the traditional opto compressors have so much perceived character and "warmth".

The Doepfer LPG response in Modular, coupled with a VCA or another LPG in VCA mode, can yield some really nice sounding results which can be heard among the presets for Modular FX.

Credits

Oscar Öberg – modeling, project management.
Kristofer Ulfves – presets, validation, user manual.
Henrik Andersson Vogel – marketing. **Niklas Odelholm** – graphics, validation.



A-101-2 LPG
Vactrol Low Pass Gate
CV In 1
F/A
CV2
CV In 2
Lev.
Audio In
Res.
Audio Out
Function
LP
VCA
G1
G2
Func
DOEPFER
MONO
STEREO
L
R
intellijel
VACTROLS!

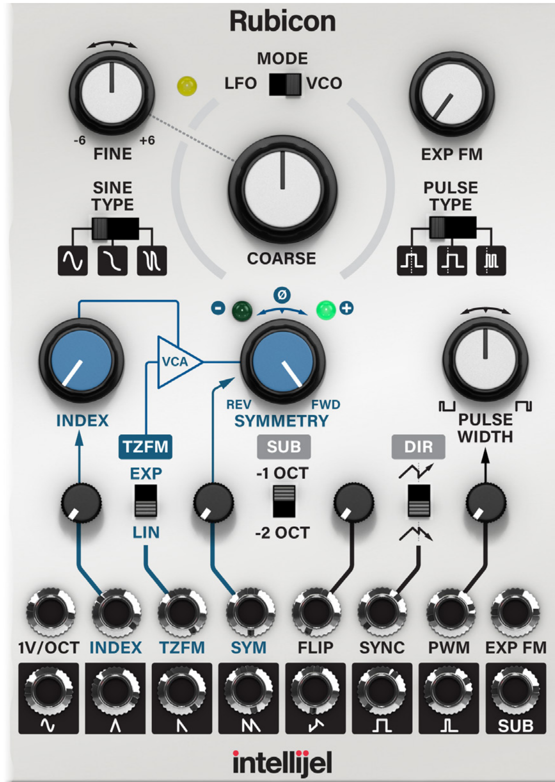
9 Intellijel Rubicon, Korgasmatron II & uFold II

Eurorack brand Intellijel, run by **Danjel van Tijn** and associates, is based in Vancouver, British Columbia in Canada. These branded add-on modules from Intellijel are for purchase at `http://softube.com/buy`.

The Intellijel modules are RUBICON THROUGH ZERO OSCILLATOR, KORGASMATRON II and the µFOLD II. These three modules all make a fine addition to the more traditional subtractive basic system of Softube Modular, as well as a step towards more west-coast thinking in terms of synthesis.

Intellijel Rubicon

The Rubicon is David Dixon's masterpiece bringing a new approach on thru-zero frequency modulation into the Eurorack world. Rubicon exclusive tone and extreme tracking (0.1% over 8 octaves) is ideal for experimenting with FM synthesis.



Parameters

Fine	This is the fine tuning knob, ranging up and down six semitones.
Exp FM	This knob sets the amount of exponential frequency modulation to be applied by the signal coming from **Exp FM** input jack (see below).
Sine Type	This switch changes the shape of the sine output jack between sine, sigmoid and double sigmoid.
Coarse	This knob is coarse tune that ranges from a few Hz to above hearing range.
Pulse Type	This switch determines the shape and phase of the **Pulse** output jack. It switches between centered, edge aligned and edge aligned, double frequency.
Index	This is the offset gain of the signal connected to the through zero **FM (TZFM)** jack.
Symmetry	This sets the symmetry offset of the output waveforms. Normal forward

position is preset to the right, center position is 0% which means waveforms are slowed down to a stop. Left position is completely reversed. Note that **Symmetry** affects 1V/Oct tracking! For the best tracking keep the **Symmetry** knob in full clockwise position.

Pulse This knob set the pulse width offset for the pulse output.

[index CV amount] This knob sets the index CV amount of the incoming signal at the index jack affecting the amplitude of the through zero frequency modulating signal.

TZFM Exp/Lin Switch which determines the response of the through zero frequency modulation, exponential or linear.

[symmetry CV amount] This knob sets the symmetry CV amount of the incoming signal at the symmetry jack.

SUB This switch determines whether the sub output will be one or two octaves below Rubicons main pitch.

Soft Sync CV This attenuator is to adjust the incoming signal at the soft sync input jack (flip).

Dir Reset direction. This switch selects whether the hard sync input will reset the VCO to a rising or falling direction.

PWM CV This knob controls the amount of pulse width modulation on pulse waveform by incoming CV at PWM jack.

Inputs

1V/oct CV control of pitch, full range 1V/octave. Connect your note CV from MIDI TO CV converter module to this jack in order to get your Rubicon to track pitch.

Index The incoming signal at the index jack affects the amplitude of the through zero frequency modulating signal.

TZFM The signal input here, through zero frequency modulates the Rubicons waveforms. Connected with **TZFM Exp/Lin** switch described above.

Sym The CV signal input modulates the symmetry setting, and is equivalent to turning the symmetry knob back and forth.

Flip This is the soft sync input. The triangle core of the Rubicon reverses on the incoming sync pulse and the attenuator sets the probability that a pulse will cause reversing. This jack requires edge-y waveforms like square or sawtooth (sawtooth is best). There is a mini attenuator on this input (see above).

Sync	Hard sync jack input. Use the reset switch (Dir, described above) to select if the waves will reset on rising or falling edge.
PWM	CV input controlling the pulse width of the pulse output.
Exp FM	Exponential CV input of the Rubicon. Fully clockwise is 1v/octave tracking.

Outputs

[sine]	Sinewave output, but also of the sigmoid and double sigmoid if the Sine type switch is used.
[triangle]	Triangle wave output.
[saw wave]	Saw wave output.
[double saw wave]	Saw wave output one octave above Rubicon fundamental frequency.
[zig-zag]	Output of the zigzag wave, a cross between upwards and downward sawtooth.
[square]	Square wave output. This output is unaffected by PWM.
[pulse]	Pulse wave output.
[sub]	Sub oscillator output. This output is unaffected by PWM.

In Use

The Intellijel Rubicon can flip all waveforms backwards using the symmetry function, all except the zig-zag, strangely enough. Really cool waveforms can be achieved by sweeping symmetry while Rubicon is being hard synced by another oscillator. Also very ear-pleasing is using the **TZFM** input by another oscillator in linear mode to create classic "icy" FM timbres.

Intellijel Korgasmatron II

The Korgasmatron II takes its name from the classic Japanese MS series from the early 80s. While the Korgasmatron II is still as gritty as the original hardware, it is also more flexible than its predecessor. The dual filter channels A and B can be set as 2-pole high pass and low pass in series, just like the original, but can also be set to work in parallel and in a whole variety of other modes.

Parameters

Korgasmatron has two identical channels, A and B, with the same basic functionality in both channels. The parameters below controls the patching and blend between channel A and B.

Xfade	This knob sets the cross-fade offset between channel A and channel B.
Xfade CV Amount	This knob controls the CV amount from the **XFade CV** input.
Xfade Dir.	This switch sets the polarity of the **XFade CV Amount** input. When set in the left position, polarity will be positive (cross fade will go from channel A to channel B). When set in the right position, polarity will be negative (the cross fade will go from channel B to channel A).
[filter configuration]	This switch sets the Korgasmatron II's filter configuration of channel A and B. Set to the upper position for a serial filter configuration (one filter after the other), or the lower position for a parallel configuration. The serial

path is disconnected if INPUT B is used in serial mode

The parameters below are identical in both channel A and B.

Clip Type This switch toggles the channel's clipping type between HARD (upper position) and SOFT (lower position).

Cutoff This is the cutoff frequency

Q This knob controls the Q value (feedback) of the filter. Often called emphasis or resonance.

Q-Drive Adjusts headroom in feedback loop, in effect adjust the volume of the resonance when self-oscillating.

FM1 This knob controls the CV amount from the **FM1 CV** input to **Cutoff** frequency.

FM2 This bipolar knob controls the CV amount from the **FM2 CV** input to **Cutoff** frequency. Center position is zero amount, while turning the knob clockwise from center will yield positive modulation and counterclockwise will yield negative modulation.

Mode This switch sets the filter topology and filter type.

The different modes are described on the next page.

In This is the input volume of the filter.

Inputs

FM1 This is the CV input jack for the **Cutoff** frequency. The **FM1 CV** input jack is connected to **FM1 CV Amount** knob (described above).

FM2 CV input jack for the **Cutoff** frequency. The **FM2 CV** input jack is connected to **FM2 CV Amount** knob (described above).

Xfade CV input jack for cross fade sweep

In Audio input jack.

1V/Oct 1V/octave CV-control for the cutoff frequency. Connect your note CV from MIDI TO CV converter module to this jack in order to get your Korgasmatron II to track playable pitch for approximately two octaves (see in use below) .

Outputs

Out Output jack.

Mix Output from both channels, determined by the XFade configuration.

Filter Modes

LP2 2-pole low pass (12dB/octave slope)

LP1 1-pole low pass (6dB/octave slope)

BP1 Band pass, cuts away low and high frequencies and leaves a narrow area around cutoff frequency. The narrowness is set by the Q value.

HP1 1-pole high pass (6db/octave slope)

HP1 2-pole high pass (12db/octave slope)

BR1 Band reject, also called notch. Cuts away frequencies in a narrow area around cutoff and leaves low and high frequencies.

In Use

The output jacks on the Intellijel Korgasmatron II is marked with by a square to make it easier to find them. Output A and B are marked by black squares and the Mix output is marked by a blue square.

Korgasmatron II is an excellent filter for distortion combined with colorful filtering duties. 1V/Octave tracking makes it possible to use the Korgasmatron II as a dual sine-oscillator when self oscillating. However, just like the hardware, tracking stretches more or less 1 and half octave which limits its scope as oscillator. The Korgasmatron II is tracking best when Q knob is set fully clockwise

The highly flexible **Xfade** in combination with different filter modes makes it a easy to create moving and interesting filtering.

Intellijel µFold II

Collaborated design with David Dixon, modeled after the some of the most beloved timbre shaping modules in classic synth modules; both in terms of how the were designed electronically as well as how they sound. The µFold II is one big sweet-spot for shaping your oscillator or feedback loop for filtering.

Parameters

Stages	This switch selects the number of stages of folding used, 2, 4 or 6.
Folds	This knob controls the folding offset (initial level).
[Folds CV amount]	This knob controls the folding amount scaled from **Fold CV** input.
Symmetry	This knob sets the through-zero symmetry offset.
[Symmetry CV amount]	This knob controls the symmetry amount scaled from **Sym CV** input.

Inputs

Fold	CV input for controlling folding, closely linked with the **Fold CV Amount** knob (see above).
Sym	CV input for controlling **Symmetry**, closely linked with the **Symmetry CV Amount** knob (see above).
In	This is the audio or CV input for the µFold II. Insert whatever signal you want to fold here.

Outputs

Out	This is the output of the µFold II.

In Use

Connect your oscillator to the µFold II input jack and out comes a folded signal. The µFold II also works nice to fold low-frequency signals such as LFOs. A neat trick is to connect input to output to create a white noise source with full range.

Credits

Oscar Öberg – concept, modeling, sound design. **Kristofer Ulfves** – concept, marketing, presets, user's guide. **Björn Rödseth, Kim Larsson** – modeling. **Arvid Rosén** – modeling and validation. **Patrik Holmström** – graphics programming. **Niklas Odelholm** – graphic design, programming, model validation. **Paul Shyrinskykh** – quality assurance. **Torsten Gatu** – programming. **Henrik Andersson Vogel** – marketing. **Danjel van Tijn** – hardware design and model validation. **David Dixon** – hardware design.

10 Mutable Instruments Braids

The Mutable instruments Braids eurorack module was released 2013 and has since then become one of the most desirable and popular modules among the eurorack crowd. It is a voltage controlled digital oscillator featuring 45 different oscillator algorithms in a wide variety of synthesis paradigms. Hidden beneath the surface is a plethora of useful functions such as built in quantization scales, envelope, VCA and bit reduction – Braids is truly the Swiss-army knife of oscillators!

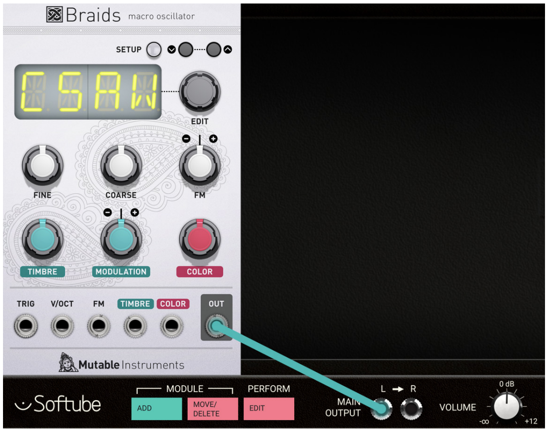
Overview

1. Setup button
2. Down button
3. Up button
4. Edit dial
5. FM modulation knob
6. Timbre modulation knob
7. Color knob
8. Display
9. Coarse tune knob
10. Fine tune knob
11. Timbre knob
12. Trig input
13. V/Oct CV input
14. FM input
15. Timbre CV input
16. Color CV input
17. Output

Getting started with Braids

Braids is a very deep and versatile module, but here's some steps on how to get started and getting to know your Braids module in Softube Modular:

1. Select your Braids module from the module select view (brought up via the "Add" button on the center bar). When your Braids module is selected and placed in the virtual rack, connect Braids out jack to main out in order to hear what it sounds like. You will hear a buzzy sawtooth sound, the CSAW algorithm.



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2. Now add MIDI control over pitch by adding a MIDI-to-CV module, found among the DAW and MIDI interfacing modules in the module select view. When this module is selected and added to your rack, connect the MIDI note output on the MIDI-to-CV module to the V/Oct CV input jack. Now, when your DAW plays MIDI notes, you will hear Braids module pitch tracking the notes played.

3. Braids features a multitude of different oscillator algorithms. Change the currently used algorithm by turning the edit knob (click on and drag your mouse up or down). Changing algorithms can also be achieved by clicking on the up or down button located on the top right on the Braids module.

4. Some algorithms in Braids need to be excited, pushed or plucked into oscillation in order to be heard. Change algorithm to no 31, “Bell”. You’ll notice that this algorithm is dead silent. Connect the trig output of the MIDI-to-CV module to the trig input of Braids. Now, when playing a MIDI note in your DAW, an inharmonic Bell sound is heard. You can change some properties of the bell sound by tweaking the timbre and color knobs.

5. Add a modulation source to perform some automatic tweaking of a parameter, for example by connecting an A-147 LFO to the timbre CV input of Braids. The modulation amount applied to timbre can now be set via the modulation “attenuverter” knob. Attenuverter means that no modulation is applied when this knob is at 12 o’clock, full positive modulation when it is fully clockwise, and of course full negative modulation when it is fully counter-clockwise.

6. Braids features a setup menu for deeper configuration. This mode is entered by clicking on the setup button. While in setup mode, this button remains lit to indicate that editing is being done within setup. Here you’ll find settings for samplerate, bit-depth, quantisation, the internal vca and envelope. These features are all described in detail in the section below called “The Setup menu”.

Braids oscillator algorithms overview

Mutable Instruments Braids features a formidable smorgasbord of different oscillator algorithms. There are 45 different oscillator algorithms, all listed below. The list below features a short description of each of the algorithms, what they are and what function the timbre and color knobs have.

Classic analog waveforms

Wave	Description	Timbre	Color
	CS-80 imperfect saw	Notch width	Notch polarity
	Variable waveshape	Waveshape	Distortion/filter
	Classic Sawtooth/Square	Pulse width	Saw <-> Square
	Sine/triangle into wavefolder	Wavefolder amount	Sine <-> Triangle

Digital synthesis

Wave	Description	Timbre	Color
	2 detuned harmonic combs	Smoothness	Detune
	2 square VCOs with hardsync	VCO frequency ratio	VCO balance
	2 saw VCOs with hardsync	VCO frequency ratio	VCO balance
	Triple saw wave	Osc 2 detune	Osc 3 detune
	Triple square wave	Osc 2 detune	Osc 3 detune
	Triple triangle wave	Osc 2 detune	Osc 3 detune
	Triple sine wave	Osc 2 detune	Osc 3 detune
	3 ring-modulated sine waves	2/1 frequency ratio	3/1 frequency ratio
	Swarm of 7 sawtooth waves	Detune	High-pass filter
	Comb filtered sawtooth	Delay time	Negative/positive feedback
	Low-fi circuit bent sounds	Sample reduction	Bit toggling

Vocal synthesis and formants

Wave	Description	Timbre	Color
ZLPF	Direct synthesis of low-pass filtered waveform	Cutoff frequency	Waveshape
ZPKF	Direct synthesis of peaking filtered waveform	Cutoff frequency	Waveshape
ZBPF	Direct synthesis of band-pass filtered waveform	Cutoff frequency	Waveshape
ZHPF	Direct synthesis of high-pass filtered waveform	Cutoff frequency	Waveshape
VOSM	Sawtooth with 2 formants	Formant frequency 1	Formant frequency 2
VOWL	Speaking-toy vowel synthesis	A, e, i, u, o	Gender
VFOF	Vowel synthesis	A, e, i, u, o	Gender
HARM	A mixture of sine harmonics	The center frequency	Distribution of the amplitude of each harmonies
FM	2-operator frequency modulation synthesis	Modulation index	Frequency ratio
FBFM	2-operator frequency modulation synthesis with feedback	Modulation index	Frequency ratio
WTFM	Chaotic 2-operator frequency modulation synthesis	Modulation index	Frequency ratio

Physical simulations

Wave	Description	Timbre	Color
PLUK	Plucked string	Decay	Plucking position
BOWD	Bowed string	Friction	Bowing position
BLOW	Blown reed simulation	Air pressure	Instrument Geometry
FLUT	Flute simulation	Air pressure	Instrument Geometry
BELL	Bell simulation	Decay	Inharmonic content
DRUM	Metallic drum simulation	Decay	Inharmonic content
KICK	808 style kick-drum model	Decay	Click amount
CYMB	808 style cymbal noise spectrum	Filter	Squares <-> Noise
SNAR	808 style snare-drum model	Tone	Snappy

Wavetables

Wave	Description	Timbre	Color
WTBL	21 wavetables	Wavetable position	Wavetable selection (quantized)
WMAP	16 x 16 waves	X position	Y position
WLIN	Linear wavetable scanning	Wavetable position	Interpolation quality
WTx4	Quad wavetable scanning	Wavetable position	Chord type

Noise

Wave	Description	Timbre	Color
NOIS	Tuned noise (2-pole filter)	Filter Resonance	Response LP - HP
TWNQ	Noise sent to 2 resonators	Resonance	Resonators frequency ratio
CLKN	Clocked digital noise	Cycle length	Quantization
CLOU	Sinusoidal granular synthesis	Grain density	Frequency dispersion
PRTC	Droplets granular synthesis	Grain density	Frequency dispersion
QPSK	Modem noises	Bit-rate	Modulated data

Parameters

Setup This button toggles between the Wave selection and Setup mode in Braids. When in Setup menu, this button is used to edit a value of a certain menu parameter (see further description below). You can always exit from the setup menu swiftly by clicking on the display.

Up arrow This button is equal to turning the edit dial one step upwards in a menu or increasing a value by one when editing a setup option.

Down arrow This button is equal to turning the edit dial one step downwards in a menu or increasing a value by one when editing a setup option.

Edit dial This dial scrolls upwards and downwards and is used to select between different Braids' different algorithms and menus.

Fine This is the fine tuning of Braids. Turning this knob tunes Braids one semitone up and down in pitch from its knob center position.

Coarse This knob controls the coarse tuning of Braids. It has a full range frequency that span over 9 octaves for most of the oscillator algorithms.

FM This knob is a so called attenuverter that controls the positive or negative frequency modulation of the currently selected oscillator algorithm. At counter-clockwise frequency modulation is at full minus 100%, at 12 o'clock 0% and at fully clockwise, it is at 100%.

Timbre The Timbre knob performs different functions for each oscillator algorithm. The value set is an offset to the Timbre modulation entered via the Timbre CV in jack which is being attenuated via the Modulation knob (see description below).

Modulation This knob is the amount of positive or negative modulation applied to the timbre parameter via the Timbre CV in jack.

Color The Color knob also performs different functions for each oscillator algorithm. Its value is also an offset to the control signals entered via the Color CV in jack (see further description below).

The setup menu

The Setup menu on Braids can be entered at any time by clicking on the Setup button when in Wave selection mode. The display on Braids will then show the text "WAVE" on screen, indicating that we have now entered the first page of the setup menu. While in the menu, the Setup button LED remains lit **green** while browsing between the different setup options, and changes to lit **red** when setup is pressed again to edit/change a parameter.

Selecting a setup option.

While editing a setup option, the setup button LED is lit **red**.

In the setup menu, the edit Dial and/or the up and down arrows are used to scroll between the different setup menu pages displaying the different setup parameters. The setup button is then used for entering and exiting edit of a certain setup parameter.

Exiting the setup menu is done by clicking the setup button again when on the "WAVE" screen (or on the "v 1.8" firmware version screen). But remember that **you can always also swiftly exit from the setup menu by clicking on the display** - this will send you directly to the currently used waveform page.

Listed below are all setup parameters:

WAVE This is the "transition" page of the setup menu. If the setup button is clicked when on this page, setup menu is exited out to the Wave selection mode.

META This parameter changes the behavior of the FM CV input if engaged. When META is set to "on", the FM CV input jack can be used to change between the different oscillator synthesis models.

BITS This parameter sets the bit-depth of the data sent to the (emulated) DAC.

RATE Selects the refresh rate of the (emulated) Braids DAC.

TSRC This parameter selects a trigger source. "EXT." uses the gate/trigger input jack while "AUTO" option makes Braids detect changes in the V/Oct CV input larger than a semitone and generates trigger on each of these.

TDLY Trigger Delay. This parameter applies a delay between the moment the trigger is received and the moment the note is "struck" on the physical models.

|\ATT This is the attack time of the internal AD envelope generator.

|\DEC This is the decay time of the internal AD envelope generator.

|\FM This is the amount of FM modulation induced by the internal AD envelope generator when triggered.

|\TIM This is the amount of Timbre modulation induced by the internal AD envelope generator when triggered.

|\COL This is the amount of Color modulation induced by the internal AD envelope generator when triggered.

|\VCA This toggles the internal VCA on and off (default value). When set to "on", Braids will remain silent until the internal AD envelope generator is triggered as it will then control the VCA level.

RANG This parameter determines the range of the Coarse knob. The option "EXT. " adjust the range of this knob to +/- 4 octaves around the note received on the V/Oct CV input (if this option is selected a very low frequency will be heard if nothing is connected to V/Oct CV input). The option "FREE" (default value) adjust the range of the coarse knob to +/- 4 octaves centered around C3 (261.5 Hz). The optional "XTND" mode provides a larger frequency range but disables accurate V/Oct scaling as a side effect. The last option "440" locks the oscillator frequency to 440Hz exactly – helpful when tuning another VCO.

OCTV Octave selection. Transpose the oscillator up or down.

QNTZ Quantization of incoming cv on the V/Oct CV input jack. The frequency can be quantized to semitones, to any one of the other available scales or disabled (default value).

ROOT This setup parameter selects the root note for the selected quantizer scale.

FLAT When engaged, this parameter applies a detuning in the lower and higher frequencies to recreate the tuning imperfections of VCOs.

DRFT This parameter recreates the drifting of a badly designed VCO.

SIGN When engaged, this parameter applies grunge, glitches and imperfections to the output signal.

v 1.8 This is the last item in the setup menu and it shows the firmware version being emulated in Braids for Modular. Clicking on setup here also exits to the Wave selection menu.

Inputs

Trig in jack The trig in jack on Braids has three functions: Braids physical models need to be "excited" by an impulse on this input to give birth to a sound. Other models will treat the trigger as a reset signal, bringing the phase of the oscillator algorithm to 0. This envelope can also be used to trigger the internal AD envelope applied to the parameters of your choice in order to create sound animation without the need of an external envelope.

V/Oct CV in jack This is the 1V/Oct frequency input. Use this input to control the pitch of Braids from the MIDI-to-CV module or from a CV sequencer.

FM in This is a frequency modulation CV input. The Scale and polarity of the signal input here is controlled via the FM attenuverter knob.

Timbre This is the CV modulation input of the Timbre parameter of each oscillator model in Braids. The CV modulation that is input via the Timbre CV in jack is being attenuated via the Modulation attenuverter knob to add a positive or negative modulation to the offset value set by the Timbre knob on the panel.

Color This is the CV modulation input of the Color parameter of each oscillator model in Braids. The CV modulation that is input via the Color CV in jack is being added to the offset set via the Color knob on the panel.

Outputs

Out This is the main output jack for audio output. Connect this to main output of modular to hear the results of your tweaks.

MI Braids module in use

There are many fun uses for a module as versatile as Mutable Instruments Braids, but here's some examples:

Standalone Synth

By activating the VCA function among the setup functions in Braids, the internal VCA will linked to the internal envelope. This example shows how you to link up your Braids module after VCA is set to "on" and TSCR to "Exti".

Polyphonic Synth voice

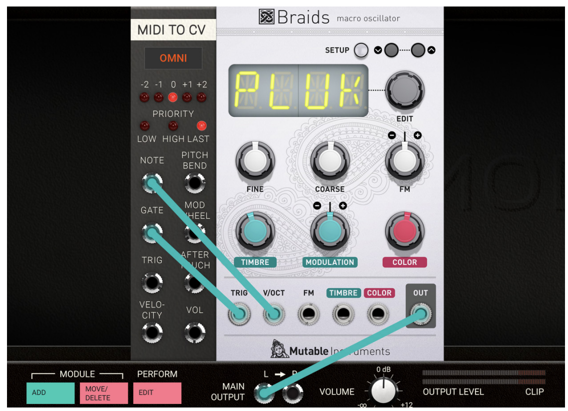
A similar technique that is described above can of course be used to create polyphonic voices (see example pictured above). Only downside to using the internal envelope of Braids is that it does not contain a sustain/hold level and thus are not suitable for long sustained chords.

Wavetables

The oscillator algorithms "WTBL", "WMAP", "WLIN" and "WTx4" are all different takes on wavetable synthesis. This technique, used in some of the most iconic early digital synths, plays back parts of pre-recorded and processed single cycle waveforms. The timbre knob and CV input modulation set currently played back waveform. In the example pictured an LFO is sweeping the wavetable of a chord set in the "WTx4" oscillator mode.

Acoustic modelling

Using some the acoustic modelling oscillator algorithms such as "PLUK" can produce some really exciting results, especially in combination with some of the special microtonal quantisation modes such as "JAPA" or "GAME" that can be selected in the setup menu.



Drums

Braids has some very nice analog drum models for bassdrum, snare and cymbal-noise built in. In the example pictured the algorithms "KICK" and "SNAR" are triggered from the trigger-sequencer. A combination of the "CYMB" algorithm, a Doepfer A-132-3 DVCA and two envelopes are used to create open and closed hihat.

Wave sequencing

So called wave sequencing patches can also be creating with Braids by switching on the META function in the setup menu. This enables the user to create sequences where the CV for each step will control the algorithms currently used. In the example pictured, the META functionality is used to create a (monophonic) drum track where the CV knobs for each triggered step will determine which kind of drum will be played back as well as the pitch of that drum.

Credits

Eric Hampusgård – programming, modeling. **Émilie Gillet** – original code, feedback. **Kristofer Ulfves** – project management, presets, validation, user manual. **Oscar Öberg** – programming, mentoring. **Arvid Rosén** – mentoring. **Fanny Hökars** – user manual layout. **Bitplant** – GUI graphics.

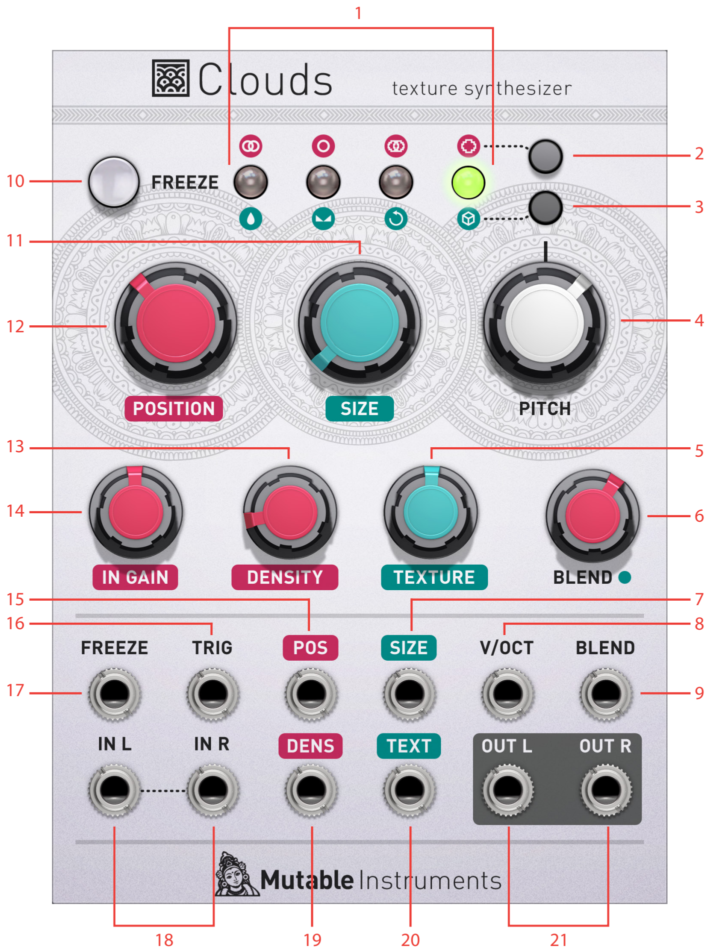
11 Mutable Instruments Clouds

Mutable Instruments Clouds is a granular audio processor. It creates textures and soundscapes by combining multiple overlapping, delayed, transposed and enveloped segments of sound taken from an audio recording buffer. The unique focus of Clouds is on real-time processing and manipulation of streaming audio, rather than playback of pre-recorded samples. This means that Clouds continuously records incoming audio through its inputs into the audio buffer, which can be up to 8 seconds long (by using the reduced audio quality setting). This limitation of the original hardware has been kept in the Softube Modular version in order to ensure a one-to-one hardware-software compatibility and similar user experience. The Softube Modular version of Clouds is based on firmware revision 1.31.

Using this continuously recorded audio, Clouds synthesizes a sonic texture by playing back short, overlapping segments of audio – also known as grains – extracted from it.

One could describe Clouds as a vessel with an audiostream running through it, featuring a tap that drips audio droplets out of the vessel for us to listen to. The tap sets the pace at which the audio droplets are dripping out this vessel. The pace is however also controlled by the size of each drop.

Overview



1. LED meters
2. Audio quality selection button
3. Blend mode selection button
4. Pitch knob
5. Texture knob
6. Blend knob
7. Size CV input
8. V/Oct CV input
9. Blend CV input
10. Freeze push button
11. Size knob
12. Position knob
13. Density knob
14. Input gain knob
15. Position input
16. Trig input
17. Freeze gate input
18. Left and right audio inputs
19. Density CV input
20. Texture CV input
21. Left and right outputs

Freezing, fragmentation and dissolution of the instantly unexpected is the way of Clouds, rather than the careful planning of what might very well not come. Have fun!

Description

Getting started with Clouds:

1. A great way of understanding how Clouds works is to run a well known and easily recognisable piece of audio through it. Try for instance using a drum/instrument loop on an audio track in your DAW. Use Modular FX loaded as an audio insert effect on your track and route audio through Clouds in your virtual rack. You should now hear an eerie, ghost-like version of your audio loop playing through Modular FX and Clouds when you hit play in your DAW.

2. Try adjusting the Density knob of Clouds slowly from fully clockwise, via the 12 o'clock position, to fully counterclockwise while your loop is playing. Notice that the sound seems to "break up" and become silent around the 12 o'clock position, and then resume on the far side as you keep turning the knob. This is because Density controls the speed at which Clouds is playing back sample "grains" from the audio-stream that is being fed through the module.

7 o'clock to 12 o'clock positions on the Density knob plays back the audio at a regular pace which decreases the further towards the 12 o'clock position we get.

On the far side, 12 o'clock to 5 o'clock positions, the audio is played back at an irregular pace, starting slow around 12 o'clock and getting increasingly more rapid as we turn the Density knob towards the end-position, 5 o'clock.

3. Now try adjusting the Texture knob from the end position, fully clockwise, slowly over to fully counterclockwise. Notice how the sound character changes. This is because of the different volume envelope applied on each grain. While the envelope at fully clockwise (Hann window) is smooth, the envelopes at 12 o'clock (triangle) and at the fully counterclockwise position (square) sound more harsh. This is extra obvious when the Density knob is set to one of the slower rates (regular or irregular) around 12 o'clock.

4. Now try clicking on the Freeze button. It will light up green and the audio-stream through Clouds will remain frozen until the Freeze button is clicked again (or CV turns it off via the Freeze CV In jack; more on this later).

When Clouds is in its Freeze state, try "audio-scrubbing" through the frozen audio buffer with the Position knob. Find a start of a drumbeat or any other percussive sound in your frozen audio.

5. Now, reduce the pace at which the sampled "grains" are played back again by setting the Density knob to around 10 o'clock. You should now hear a steady pulse of audio grains played back. Try adjusting the Size knob. You'll notice that Size not only affects the length of the played back grain, but also the pace at which it is played back. Try also adjusting the Pitch shift knob. This shifts each grain up or down 2 octaves from the original pitch (at the 12 o'clock position).

6. OK. Next let's look at the Blend modes. Unfreeze Clouds by clicking on the Freeze button again and you will hear your DAW-loop playing through Modular FX and Clouds again. By clicking on the lower button to the right of the LED row you will cycle which Blend mode is active. Select the Blend mode furthest to the left with the drop symbol; the Dry/Wet mode. You'll notice that the Blend knob on the lower right on the Clouds module will change color to white and knob position is set to fully clockwise. This means that the Blend VC Dry/Wet parameter is currently set to fully wet mode. Tweak the Blend knob gradually to fully counter-clockwise to only hear the dry signal. Notice that the LED above the currently selected Blend mode changes color from unlit to green and then over yellow, to red at the Blend knob set to a fully clockwise position. The color will then always be reflected whenever tweaking the Blend knob regardless of which mode that is currently selected. In this way, you can always keep track of the four different Blend parameters current value when tweaking one. At VC Dry/Wet set to 12 o'clock you will hear both dry and wet signals at the same time and notice that there's a slight delay between the two. This is normal and the same behavior as in the original hardware. With the Pos knob you can set where in the audio buffer Clouds will playback of sampled grains, effectively creating a delay between recorded and played back audio. If a CV source is inserted to the Blend CV input, it will add to and control the currently selected Blend Mode parameter.

7. Let's move on to the next Blend mode, the VC Stereo Spread, indicated below the second LED from the left by a stereo image symbol. By repeatedly clicking on the Blend mode selection button until the second LED from the left is lit green, the VC Stereo Spread mode is selected. This is also indicated by the color of the Blend knob (now red) and the new position of the knob to reflect the current value of VC Stereo Spread. This parameter determines the stereo spread of each played back sampled grain, i.e. the stereo width of Clouds. Set the Blend knob all the way counterclockwise for fully mono output on the left and right outputs, or all the way clockwise for extreme stereo spread of each grain.
8. The third Blend mode, indicated below the third LED from the left by a curling arrow symbol, is VC Feedback. This parameter determines how much of the the output signal will be fed back into the input audio stream. Be careful with applying too much feedback as it is easy to get loud output from Clouds when the feedback starts building. When in Freeze mode this parameter changes to a kind of additional reverb control (for more information see detailed description below).
9. The last Blend mode is VC Reverberation. It is indicated below the fourth LED from the left by a box symbol and set how much internal reverb is applied to Clouds output.

Tip: Use performance module linking to gain easy access to all four Blend parameters at the same time.

Note: Since Clouds is essentially a digital module, it induces some latency to its processed signals. Time critical elements processed through Modular FX and Clouds will appear "out of sync" if not compensated for beforehand.

Parameters

Freeze Push-Button	When this button is engaged (is lit), Clouds "freezes" the audio buffer held in memory and the current grain will continuously loop until the freeze button is dis-engaged again by pressing it a second time. During freeze all Clouds' controls will work as normal with all functions now applied to the grain "loop". Only two functions will be different: Input Gain will not affect the currently playing grain "loop" since it only controls incoming audio level. Also Feedback function will be given a different function since feedback can no longer occur in the recording buffer (because it is frozen). Instead, the output signal is routed through delays and all-pass filters to get the feedback build-up to occur in this extra recording space, giving the sound a very reverb-like nature.
Audio Quality Selection Button	This button toggles between the four different sound-qualities in Clouds: 16 bit stereo, 16 bit mono, 8 bit stereo and 8 bit mono. The current mode will be displayed briefly on the 4 LED bar by a red indication below the currently selected mode symbol (displayed above the 4 LED bar). Clicking on the rose symbols above the LED indicator bar will also swiftly change between each of the blend modes. Note that Clouds' 8-bit

has a lovely flavor of 8-bit μ-law companding which make it sounds like a cassette with less hiss, but more distortion.

Blend Mode Selection Button This button determines the current state of the Blend knob and input jack. Clicking on this button enables the user to toggle between (from left to right) VC Dry/wet mode, VC Stereo Spread, VC Feedback and VC Reverberation mode. The current mode will be displayed briefly on the 4 LED bar by a green indication above the currently selected mode symbol (displayed below the 4 LED bar). Clicking on the green symbols beneath the LED indicator bar will also swiftly change between each of the blend modes.

Blend Mode Selection Button

VC Dry/Wet Balance Mode When VC Dry/wet balance mode is selected, the Blend knob (and the input CV at the Blend input CV jack) sets the balance between the dry and wet signal, 'dry' being the unaffected portion of the sound fed through Clouds, and 'wet' being the affected signal.

Blend Dry/Wet

VC Stereo Spread When VC Stereo Spread mode is selected, the Blend knob (and the input CV at the Blend input CV jack) will determine how each grain is distributed in stereo. In VC Stereo Mode, turning the Blend knob clockwise will "widen" the stereo output of the Clouds.

VC Feedback As VC Feedback mode is selected, the Blend knob (and CV input at the Blend jack) will set the internal feedback of Clouds. Depending on the settings of other parameters of Clouds such as Size, Density and Pitch, this feedback will be more or less harsh. Be careful though – it is quite easy to get Clouds feedback to self-oscillate which causes internal distortion. When the Freeze function is engaged, the

VC Feedback can no longer occur in the recording buffer (it is frozen) and thus feedback changes function to a kind of additional reverb control.

VC Reverberation In VC Reverberation mode the Blend knob (and the input CV at the Blend input CV jack) will control how much signal is fed into Clouds' built-in reverb. The Blend knob set fully clockwise in VC Reverberation mode will generate a very "wet" and long reverb tail.

Position Knob With the position knob, you will select from which part of the buffer the grains are taken, i e setting the starting-position from which the sample-grain are being played back in Clouds. If you change this knob while Clouds is processing sound that goes through, you will hear the played back audio affected in different ways. In Freeze mode, turning this knob will scrub the grain sample buffer back and forth while audio is played back.

Size Knob This knob sets the size of the played back grains. With this knob turned all the way counterclockwise, grains will be very tiny and sound metallic depending on what settings the density and texture knobs are set to. Turning the Size knob clockwise while sound is played back can result in cool "time-stretched" effects. Of course a motion similar to this can be automated through the size CV inputs.

Pitch Knob With the Pitch knob each played back grain can be played back at a different pitch and, by the nature of Clouds, the pitch-shifted signal will still retain its tempo and timing. Slightly pitch-shifting each grain is also extra apparent when using feedback. The fully counterclockwise position gives a minus 24 semi-notes (two octaves) down transposition, while fully clockwise transposes each grain 24 semi-notes (two octaves) up.

Input Gain Knob This knob sets the attenuation of the signal at Audio in jacks. Fully counterclockwise, signal will be completely silent. Watch the LED meters to monitor the input signal so that it's not distorting while being fed through Clouds.

Grain Density Knob This control sets Clouds internal playback density and behavior of the played back grains. If this knob is at 12 o'clock, nothing will be heard at all since the playback density of grains will be set to none. Twist the knob slowly fully counterclockwise to get more dense, regularly played back grains with a constant, high rate. Or twist the knob slowly fully clockwise to increase density but with irregular playback, which sounds different and smoother than constant rate.

Texture Knob This knob sets each audio grains volume envelope. In the fully counterclockwise position, the envelope curve is very step and will give a harsh sound to Clouds, especially when the Density knob is set high (fully clockwise or counterclockwise). At twelve o'clock, the Texture knob gives each grain a more sloped and softer triangular shape and at the fully clockwise position a Hann window shape – a raised cosine distribution is used. The latter gives Clouds a blurry, pleasant sounding texture when played back.

Blend Mode Knob This knob controls one at a time, the parameters determined by Blend mode button selection: VC Dry/wet balance, VC Stereo Spread, VC Feedback or VC Reverberation (see full description above at Blend Mode selection).

Indicators

Freeze Button LED When the Freeze Button LED is lit, Clouds does not stream audio through it, but merely plays back what's currently in the audio-buffer.

Meters (4 multicolor LED bar) This bar of four multi-colored LEDs have different display functions. By default, it displays input level by lighting one LED after another from left to right. Adjust your input level by turning the In Gain knob until two or three LEDs remains more or less constantly lit (on sounds with an even sound level).

When the Audio Quality Selection button is clicked the LED meter bar will briefly show which audio quality is selected by a red LED indication. Red LED meaning (from left to right): 16 bit stereo, 16 bit mono, 8 bit stereo and 8 bit mono.

When the Blend Mode Selection button is clicked in order to change Blend mode, the LED meter bar will briefly show which mode is selected at the moment by a green LED indication.

When the Blend knob is moved, the LED at the position of the selected parameter will change color from none to green, over via yellow to red to indicate increased parameter value. The current values of the other Blend values will also be reflected by their LED colors at the same time.

Inputs

Freeze Gate Input Jack
When a high (+5V DC signal) appears on the Freeze Gate input jack, Clouds will "freeze" the audio buffer and hold it in memory until the input jack goes low again. If the freeze button is engaged at the same time, it's current state will override the input CV of this jack.

Trig Input Jack
When a trigger appears at the Trig input jack Clouds will trigger and play a grain, regardless of the density setting. This means that it is possible to trigger a short snippet of a "sampled" sound that is currently in the buffer while using another CV to sweep through the position in the buffer (see below).

Position CV Input Jack
CV input into this jack mirrors the function of the Position knob. This means that you sweep the playback position of the audio buffer memory in real-time while audio is streaming through, or when in "freeze mode".

Size CV Input Jack
This jack controls the played back size of the sample grains and mirrors that of the Size knob. Very cool time-stretch effects can be achieved by CV modulating this input, changing the parameter during playback

V/Oct CV Input Jack
CV input jack to control the pitch transposition at 1 Volt per octave. This jack adds to the function of the Pitch knob, which works as an offset in relation.

Blend CV Input Jack (Controls The Currently Selected Blend Parameter)
This jack controls the currently selected Blend parameter: VC Dry/wet mode, VC Stereo Spread, VC Feedback or VC Reverberation mode.

Audio In Left
Left input jack for Stereo audio input.

Audio In Right
Right input jack for Stereo audio input. This jack is normalized to Left input if no cable is inserted here (i.e. mono in).

Density CV Input Jack
CV input jack to mirror the Density knob control.

Texture CV Input Jack
CV input jack that mirrors the Texture knob control.

Outputs

Audio Out Left	Left output jack for Stereo audio output.
Audio Out Right	Right output jack for Stereo audio output.

The Clouds module in use

The Clouds module can be used in many interesting ways. Here's some suggestions:

Sampler

One of the coolest uses of Clouds in Modular is to use it as a kind of crude sampler. Use Modular FX to record your prefered sound into the sample buffer, freeze, set density to 50%(12 o'clock) and use the trig input to trigger individual grains. Use the Pos knob to scrub the buffer back and forth, and find the part of the Sample you want to play back. By adjusting the size from fully counterclockwise (only one sample grain played back) to full clockwise, the whole sample buffer can be played back from the start position set by the pos knob.

AM – RM kind of effect

By setting the Density knob at roughly 9 o'clock and Texture knob fully counterclockwise, the signal through Clouds take on a gritty tone. Now adjust the pitch knob to some odd ratio for clangorous rich ring-mod type sounds.

Delay

Using Clouds with full regular Density and Texture at fairly square (both knobs at fully counterclockwise), it can be used for some fairly straightforward delay-duties. Adjusting Size changes delay time and adjusting Blend feedback - you guessed it - delay feedback. Set the balance between the dry signal and the delayed signal with Blend Dry/Wet.

Pitch-shifter

This is a bit trickier. Since Clouds is not specifically made for pitch-shifting, different harmonic material requires slightly different Texture and grain Size settings for optimal pitch shifting duties. This is also depending on the ratio required - it is easier to achieve perfect one octave up or down pitch shifting with Clouds than for example a clean +7 semitones pitch shift. Experimentation is the key here.

Time-stretch

By modulating the size CV input with an lo frequency rising sawtooth wave with roughly the same speed as the played back audio through Clouds, a time-stretch kind of effect can be achieved.

Robotize

By automatically trigger freeze Clouds audio buffer, a cool "robotizer" effect can be constructed. This works exceptionally well on beats and rhythm material.

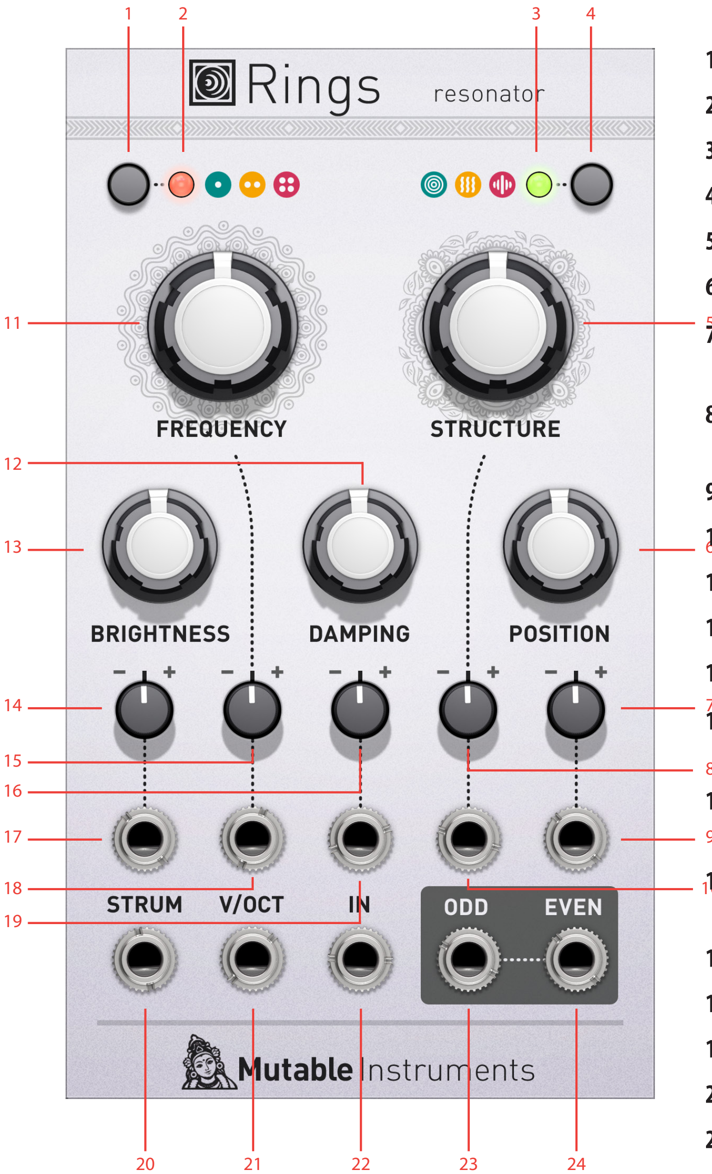
Credits

Eric Hampusgård – programming, modeling. **Émilie Gillet** – original code, feedback. **Kristofer Ulfves** – project management, presets, validation, user manual. **Oscar Öberg** – mentoring. **Arvid Rosén** – programming, mentoring. **Igor Miná** – user manual layout, hardware photos. **Bitplant** – GUI graphics. **Tomas Bodén** – presets. **Robin Rimbaud** – presets.

12 Mutable Instruments Rings

The Mutable instruments Rings is a eurorack modular module that was originally released during winter 2015/2016. Rings brings physical modeling synthesis to Eurorack from a more modular angle than, for instance, the models featured in Braids and other modules from Mutable Instruments. Instead of trying to be a complete instrument, Rings focuses on the key ingredient, the resonator, ready to be excited by envelope clicks, trigger pulses, granular noise or any other audio source. The Softube Modular version of Mutable Instruments Rings is based on firmware revision 1.1 with the all features described in the original hardware's user manual.

User interface



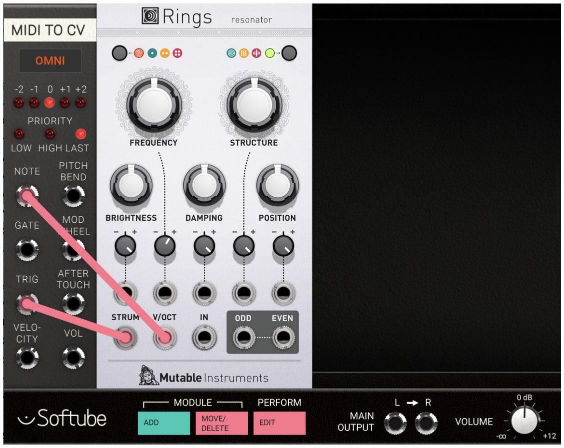
1. Polyphony select button
2. Polyphony mode LED
3. Resonator mode LED
4. Resonator select button
5. Structure knob
6. Position knob
7. Position CV Modulation knob
8. Structure CV Modulation knob
9. Position CV input
10. Structure CV input
11. Frequency knob
12. Damping knob
13. Brightness knob
14. Brightness CV Modulation knob
15. Frequency CV Modulation knob
16. Damping CV Modulation knob
17. Brightness CV input
18. Frequency CV input
19. Damping CV input
20. Strum input
21. V/Oct input
22. In
23. Odd output
24. Even output

Rings is a resonator, the essential ingredient at the core of several physical modeling techniques. It transforms an external, un-pitched excitation audio signal (such as a click, a burst of noise, or whatever is fed into it) into a full-bodied pitched sound. You could describe Rings as the bar, the tube or the bunch of strings you'd cause to vibrate with an external signal in order to create your sound. However, Rings has some smartness already built into it that makes it very easy to use (more of this later).

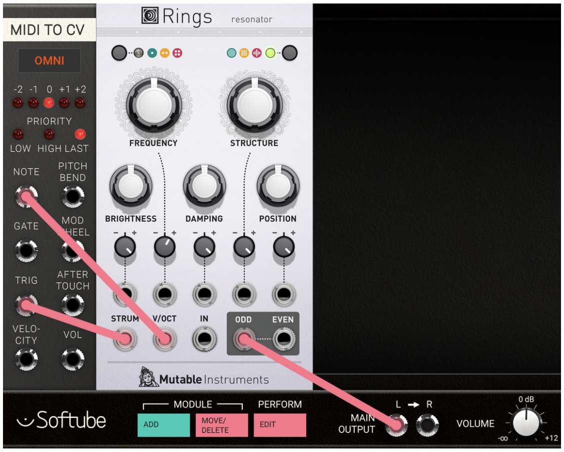
Getting started with Rings

1. The easiest way to use rings is to connect it as a regular oscillator (well, almost). Add a MIDI-to-CV module and Rings to your rack.

2. Connect the Note CV output of the MIDI-to-CV module to the V/Oct CV input on Rings.



3. Now connect the Trig output of the MIDI-to-CV module to the Strum input on Rings.



4. And now connect one of Rings' outputs (Odd or Even) to Modular's left main output. Now, the MIDI notes from your DAW should be able to play notes on Rings. Experiment by selecting between the three different internal models by clicking on the upper right button. The upper right LED color will change from green to yellow, and to red, as you change models and sound of Rings.

5. You can now experiment by adding a connection between the Velocity output of the MIDI-to-CV module to Rings Brightness CV input. Now, playing harder on your MIDI keyboard (or sending MIDI note information with less velocity) will make Rings sound brighter, while playing softer makes Rings sound will a duller/ softer sound.

6. By connecting the remaining Rings output (odd or even) to Modular's main right output, you will be able to clearly hear the dynamic panning effect between Rings' voices in polyphonic mode (top left LED mode red or yellow) and pitch spread in monophonic mode (top left LED mode green).

Interface

As mentioned earlier, Rings has some smartness already built into it that makes it easier to use without having to connect every input. The general design philosophy for Rings is that whenever you don't patch an input, the module tries to infer it from the other inputs. For example, the Strum trigger input indicates string changes, but if you don't provide it, the module will look at sudden changes in the
V/Oct CV input to decide if a new string is touched or not. If there's nothing in V/Oct CV input, Rings will detect transients on the In input. Similarly, if nothing is provided in the In audio input, the module will makes its own excitation signal by waiting for triggers in the Strum input, or, if there's nothing in this input, by detecting sudden changes in the
V/Oct input.

So, to sum it up: Ideally, **Rings** would need three input signals:

1. A trigger signal in the **STRUM** input, to indicate when the currently playing note should fade away, and when a new note is starting.
2. A CV signal in the **V/OCT** input, to control the note frequency.
3. An audio signal in the **IN** input, which will hit, strike or caress the resonator. But because it might not always be possible to get these three signals from your system, **Rings** makes the following assumptions:

- If no patch cable is inserted in the **IN** audio input, the module will synthesize its own excitation signal whenever a note is strummed. This excitation signal is either a low-pass altered pulse, or a burst of noise depending on the resonator type.
- If no patch cable is inserted in the **STRUM** audio in- put, the module will determine that a new string should be strummed either by detecting note changes on the **V/OCT** input. Or by detecting sharp transients of the **IN** audio signal when nothing is patched in the **V/OCT** input.

Rings Resonator models

The resonator select button, found at the top right on Rings, selects between the three different types of resonator models available. These are indicated by different LED colors on the top right LED and are:

1. Modal resonator (top right LED lit green)

Modal synthesis works by simulating the phenomena of resonance at play in vibrating structures. For instance, a string or plate will absorb certain frequencies while it will "ring" at some other frequencies, called the modes. Various materials or structures are characterized by different relationships between the frequencies of their modes, which Rings recreates. When we pluck a string, strike a drum or blow in a tube, the short burst of energy of the blow/impact contains many frequencies. Some of these frequencies will fall outside of the modes and are absorbed, while some of them will excite the modes and produces a stable, pitched sound. Each mode corresponds to a harmonic or partial in the spectrum of the sound, and is modeled by a band-pass filter. The Q factor of the filter will determine the sustain of the oscillations for the corresponding partials.

2. Sympathetic strings (top right LED lit orange)

Some string instruments, such as the sitar or sarod, make use of strings that are not directly struck/plucked by the musician. These strings are responding to vibration of the other strings, adding extra overtones or undertones to it. Rings simulates this phenomenon with a bunch of virtual strings (made with comb filters), allowing the addition of extra tones to an incoming audio signal. The tuning ratio between these strings can be altered.

3. Modulated/inharmonic string (top right LED lit red)

This last method is based on the extended Karplus-Strong method: the excitation signal is sent to a comb filter with an absorption filter, simulating the multiple reflection of a wave propagating on a string and being absorbed at its ends. However, to bring more variety to the sound, Rings adds three extra ingredients to this classic: a delay-compensated all-pole absorption filter creating more drastic plucking effects, delay time modulation emulating the sound of instruments with a curved bridge (like the sitar or tanpura), and all-pass filters in the delay loop, shifting the position of the partials and recreating the tension of piano string or completely bonkers inharmonic timbres.

Parameters

Polyphony select button	This mode select button switches between Rings monophonic mode (LED lit green), duophonic mode (LED lit orange) and quadraphonic mode (LED lit red). Polyphonic means that a note that has been struck can will still be sounding when the next note is struck.
Resonator select button	This button selects between the three different resonator models available within Rings (see detailed description above).
Frequency knob	This knob controls the overall tune of Rings. This knob is unquantized when nothing is patched to the V/Oct CV input. However, when cable is connected to the V/Oct input, this knob will work as a quantized pitch offset instead.
Structure knob	This knob controls the selected model's inner harmonic structure. Exactly how this is done differs slightly between the three different models: For the modal resonator (right LED lit red) the structure knob controls the frequency relationship between the partials and thus the perceived structure (plate, bar, string). When using the sympathetic string resonator mode, this knob will alter between set frequency ratios between the strings. And finally, when using the modulated/inharmonic strings resonator, this knob controls amount of modulation and detuning between the partials.
Brightness knob	This knob controls the upper harmonics response of the excited signal within Rings. Low values on this knob simulate materials like wood and nylon, while high values simulate materials like glass or steel.
Damping knob	The damping knob controls the decay time of the sound with a range between less than a 100ms set at the lowest value, up to about 10s, set at the highest value.

Position knob This knob controls the excitation position, meaning the point at which the string/surface where the excitation is applied. Applying the excitation right in the middle of the surface will cause, by summing, the even harmonics to cancel each other. This can be clearly heard when position knob is at 12 o'clock at Modal mode (right LED lit green) and in monophonic mode, no even output will be heard at all, as it is cancelled out.

Brightness CV Modulation knob This is the attenuverter for the Brightness CV input. This means that this knob controls the amount of positive or negative modulation applied to the brightness parameter via the Brightness CV in jack.

Frequency CV Modulation knob This is the attenuverter for the Frequency CV input. This means that this knob controls the amount of positive or negative modulation applied to the frequency parameter via the Frequency CV in jack. Note that the Frequency CV in jack is normalized to 3V when not connected, and thus will this knob work as a offset or fine-tune as long as no signal is connected to this jack (see also description below).

Damping CV Modulation knob This is the attenuverter for the Damping CV input. This means that this knob controls the amount of positive or negative modulation applied to the damping parameter via the Damping CV in jack.

Structure CV Modulation knob The attenuverter for the Structure CV input. This means that this knob controls the amount of positive or negative modulation applied to the structure parameter via the Structure CV in jack.

Position CV Modulation knob This is the attenuverter for the Position CV input. This means that this knob controls the amount of positive or negative modulation applied to the position parameter via the Position CV in jack.

Inputs

Brightness CV input This jack is for remote controlling of brightness within Rings. Control signals (CV) input here will affect the brightness parameter as described above.

Frequency CV input Signals input here affects the tuning of Rings. Note that the Frequency CV in jack is normalized to 3V when not connected.

Damping CV input This jack is for remote controlling of damping within Rings. Control signals (CV) input here will affect the damping parameter.

Structure CV input Remote control of structure. Control signals (CV) input here will affect the structure parameter.

Position CV input Signals input at this jack controls of the strike position within Rings. Control signals (CV) input here are added to the Position parameter.

Strum input This is the trigger input for polyphonic operation. A trigger received by Rings on this jack will assign a new voice and thus making "overlapping" voice-operation a reality. Try this out by setting the module in 4-voice mode (left LED lit red), turn damping fully CW to get super long releases, and just play your notes on the V/OCT input - they will overlap without cutting each other. A new trigger will cut the least recently played note and will start a new note – just like voice-stealing on poly-synths. When no cable is patched into this jack it is normalized to the V/Oct and In jack.

V/Oct input CV input jack for chromatic scaling (1V per Octave) for Rings. Connect your the Note CV from the MIDI-to-CV module here.

In This is the main audio signal input for excitation signal in Rings. If no signal is connected, this jack is normalized to an internal noise/burst generator that reacts to changes in the V/Oct input.

Outputs

Odd Audio output for odd harmonics. This jack is normalized to the Even jack if it is not connected, meaning that its audio will be mixed and output through there.

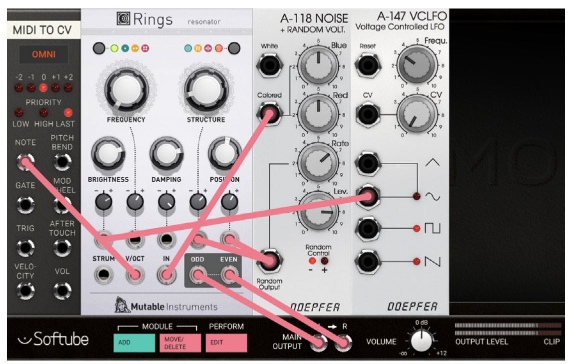
Even Audio output for Even harmonics. This jack is normalized to the Odd jack if it is not connected, meaning that its audio will be mixed and output through there.

The Rings module in use

Mutable instruments Rings is a very versatile and characteristic sounding module. Here's some uses:

An obvious use for Rings is to utilize the sympathetic string model (yellow right LED mode) in order to create credible plucked string sounds. Experiment with different brightness, damping, position and structure settings for different string sounds. Try also mapping velocity to brightness for additional expressive control over MIDI. An interesting exercise is also to replace the internal excitation signal with an external source for a different sound - the easiest way to try it in this patch is to add an additional patch cable from the trig jack out of the MIDI-to-CV module to the In audio input on Rings.

Using a random noise source or another continuous signal for excitation is a great way of creating rich moving drone-sounds. In this example we're using the modulated/inharmonic string model (red right LED mode) to create a deep drone.

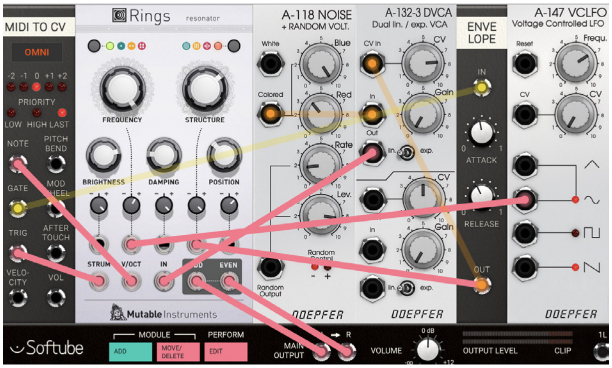


Rings is great also for processing external sound! Try feeding any percussive, harmonic or non-harmonic sound into Rings using the external inputs of Modular FX to add flavor. Try for instance using the sympathetic string resonance model to create Sitar-esque drones for your guitar. Or why not adding cool modal resonating drones triggered from your drums?

Try using an LFO to slowly modulate the position of the modal model (top right LED green) for a cool phasing type of clock sound.

Make use of the polyphonic feature of Rings to create a guitar strumming emulation. In this patch we use an AD-envelope generator patched through a quantizer further on into Rings to create automatically strummed arpeggios in a selected scale. The left CV mix module limits the range of the envelope into the quantizer, while the right CV mix module adds together the arpeggio CV and note CV from the MIDI-to-CV module for transposition.

Create flute-like sounds by feeding enveloped white noise through Rings in the Modulated/inharmonic string model (top right LED lit red) with the resonance frequency tunes fairly high.



Credits

Eric Hampusgård – programming, modeling. **Émilie Gillet** – original code, feedback. **Kristofer Ulfves** – project management, presets, validation, user manual. **Oscar Öberg** – programming, mentoring. **Arvid Rosén** – mentoring. **Fanny Hökars** – user manual layout. **Igor Miná** - hardware photos. **Bitplant** – GUI graphics.

13 Saturation Knob For Modular

Saturation Knob is a modeled output distortion that can be used anywhere you need some grit. Use it to fatten up bass lines, add some harmonics and shimmer to vocals, or simply destroy your drum loop.

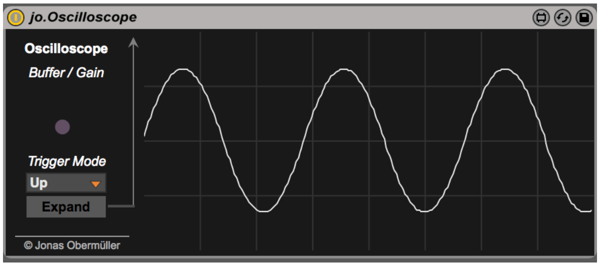
The three modes, Keep High, Neutral and Keep Low, give you three kinds of distortion characters. If you want to squash a drum loop, but at the same time keep the bass drum fairly intact, you can for example use the Keep Low mode. Vice versa you can use the Keep High mode to keep the hi-hats intact but distort the lower end of the bass-drum and snare.

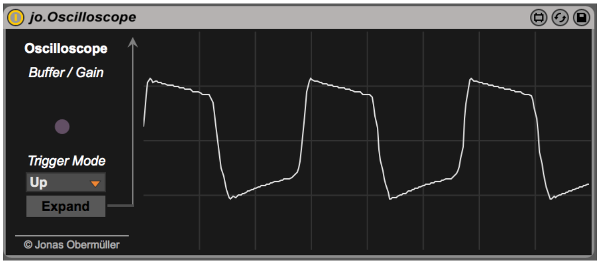
Overview

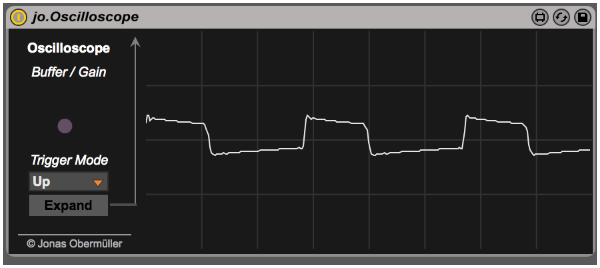
1. Saturation knob
2. Saturation type
3. Saturation CV input
4. Left and right inputs
5. Left and right outputs

Parameters

Saturation knob
This knob is THE actual saturation knob from which the plugin got its name. It controls the saturation amount, clipping the waveform and ear pleasingly squishing as saturation is increased. When set fully anti-clockwise no saturation takes place and turning the knob obviously increase saturation.







Saturation Type
This switch decides the balance of the saturation:

Keep Low – Saturation is applied on the high end of the signal fed through.

Neutral – All frequencies are equally distorted.

Keep High – Saturation is applied on the low end of the signal fed through.

Inputs

Saturation CV input
This input is external CV control of the Saturation amount. CV input here will be added to the amount set by the saturation knob.

Left input
Left of the stereo inputs, insert your signal to distort here. This input is normalized to both stereo outputs if no signal is inserted into the right input

Right input
Right of the stereo inputs, insert your signal to distort here.

Outputs

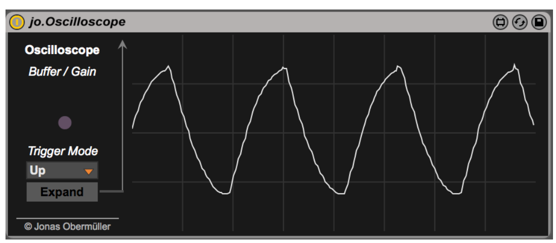
Left input
Left of stereo outputs, this output is a mono mix of both stereo inputs distorted through the Saturation Knob module if no other signal is inserted into the right output.

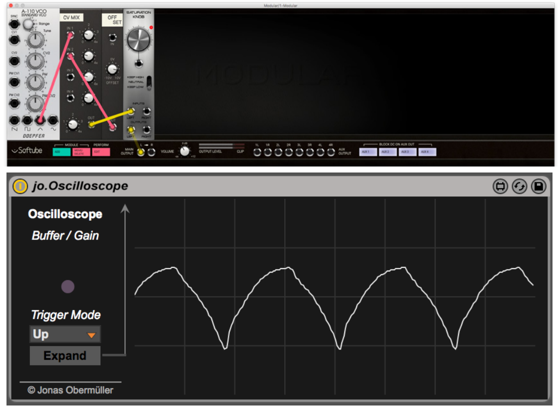
Right input
Right of the stereo outputs.

The Saturation Knob module in use

Using the Saturation Knob module in the feedback line of the A-108 VCF or in a delayed feedback line to create gritty cool echoes is highly recommended.

The Saturation knob mode can also be used as a crude wave-shaper for waves that lack a rich harmonic register such as sine and triangle waveforms. Use offsets and/or CV-mixers to control the asymmetry of the clipping.





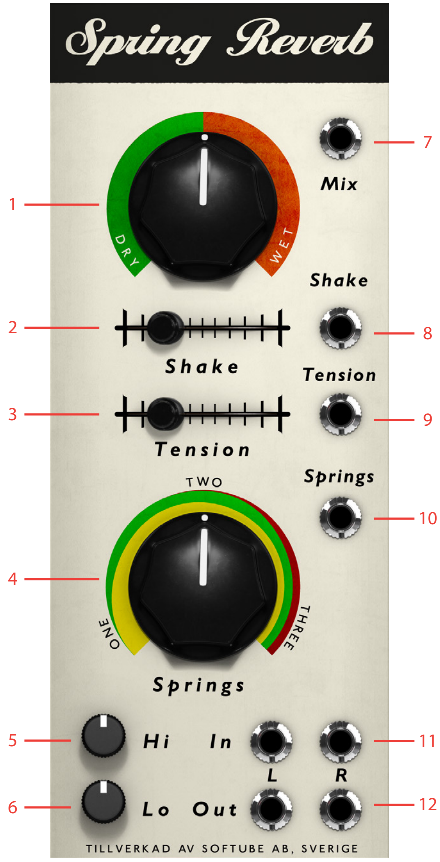
Use case 2 (continued).

The saturation becomes more pronounced if the input signal contains a lot of uneven harmonics. Try for instance to patch two sine-oscillators through an audio mixer and then into the Saturation Knob module, and then set their relative pitches to an uneven frequency relationship (for example 65.4hz and 177.6hz, a ratio of 2.71). Now, when the saturation is increased, inharmonic sidebands appear as a very harsh pronounced distortion in the audible signal. When changing the second oscillator frequency so that the relative pitch relationship is even (for example 65.4hz and 98,1hz, a ratio of 1.5) the sidebands created by the distortion have a harmonic relationship and sound much more pleasing. This is the same principle on which “power chord” theory with electric guitars is built.

14 Spring Reverb For Modular

Back in the 1930s, Laurens Hammond had started developing and selling his now famous Hammond electro-mechanical Organ. While he had bought a small church organ, having it installed in his living room, he soon found out that it didn't sound as impressive without the reverberation of the church hall. To cure this, he borrowed technology from Bell Labs who had developed a device to emulate the artificial delay in long-distance calls using wire and springs. Hammonds new tweaked version of Bell Labs device was the first real artificial reverberation unit and it was huge. Eventually, in the 1960s, the spring reverb units had shrunk enough to fit into guitar amplifiers and so they also were incorporated in the early synthesizers such as the EMS VCS3 (1969), ARP 2600 (1971) and Buchla Music Easel (1973).

Overview



1. Dry/Wet mix knob
2. Shake slider
3. Tension
4. Springs
5. Treble (Hi)
6. Bass (Lo)
7. Mix CV in
8. Shake in
9. Tension CV in
10. Springs CV in
11. Left and right inputs
12. Left and right outputs

Description

The Spring Reverb module for Modular is based on Softube's stand-alone native plugin and shares a lot of the same features. An addition to the Modular version is the CV control of all parameters. For more detailed information about Spring Reverb, see also its separate chapter.

Parameters

Dry/Wet Mix Knob This knob will set the Dry vs Wet ration for your Spring reverb. For a fully wet signal, turn this knob fully clockwise.

Shake Slider This slider emulates the spring tank being shaken or spring being struck resulting in a thunderous pleasant sound, just like a real spring tank would. When the slider is moved to a position other than halfway between far left and right, the reverb will go "sproing". When the mouse button is released while adjusting this slider, it will automatically revert back to its middle position. Think of it as the slider having a spring-motion (pun intended).

Tension By adjusting this slider the sound of the strings will change from slow and smooth (left) to quick and harsh (right). By setting this parameter differently, many sounds from different reverb tanks can be achieved just by changing the tension and the number of springs (see below). As a general rule, cheaper reverb tanks usually feature high tension and two springs, while a more expensive reverb tanks might feature three springs and a lower tension.

Springs This knob controls the number of springs audible in the Spring-reverb. Fully counter-clockwise, only one spring will make sound, while at 12 o'clock two springs are audible, and finally at fully clockwise the sound of all three springs are mixed to the output.

Treble (Hi) This knob changes the high end equalization of the Spring reverb module output. Tweaking the treble knob clockwise will boost the top-end frequencies above 5 kHz of the wet portion, and tweaking the same knob fully anticlockwise will cut off some of the same top frequencies.

Bass (Lo) This knob changes the low end equalization of the Spring reverb module output. Tweaking the bass knob clockwise will boost the low-end frequencies below 70Hz of the wet portion, while tweaking the same knob fully anticlockwise will cut off some of the same sub frequencies.

Inputs

Mix CV In This CV input controls the Dry/Wet ratio for your Spring reverb and works with the Dry/Wet Mix knob as offset.

Shake In CV input jack that activates the Shake function of the Spring Reverb when it exceeds 0v. Lower CV voltage fluctuations will result in minor "spring strokes", while great leaps (gates etc) will create thundering spring noise-effects. This jack works in tandem with the Shake slider as offset.

Note: automating the shake slider from a performance module can slightly alter its functionality since the control will not have the spring function of the module's own control.

Tension CV In This CV input jack controls the tension of the springs. Swiftly modulating this input with a sine or triangle wave can create desirable chorus-like sounds.

Springs CV In CV input jack with control over the output volume of the three springs. Low CV will output only one spring, while an increasing CV amount (up to 5V) will add the two other springs in the mix output.

Left In This is the left audio input of the Spring Reverb.

Right In This is the right audio input of the Spring Reverb. It is normalized to the Left audio input when it is not patch, which means that only connecting the Left input via a patch cord will enter mono audio through the Spring Reverb.

Ouputs

Left Out The Left audio output of the Spring Reverb module.

Right Out The Right audio output of the Spring Reverb module.

The Spring Reverb module in use

The Spring Reverb module can be used in a variety of ways, some of them not normally associated with spring reverbs.

1. Use fast modulation of the tension input jack in order to get chorus-like sounds out of seemingly ordinary waveforms. Use this great trick to get animation and blurriness into your patches.

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2. Using a the Spring Reverb in a controlled feedback loop can also create interesting sonic textures. Spring Reverb can be a bit sensitive, especially in higher frequencies; a good practise is to have an envelope-follower controlled filter or VCA in the feedback path along with the Spring Reverb. And by using the envelope follower to simultaneously control the tension of the Spring Reverb in the feedback path,

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3. Another great way to customize your Spring Reverb sound is to use one or many bandpass filters as means of equalization of the output signal.

4. CV control of the Dry/Wet mix sequencer from a synchronised source such as a sequencer can also create cool jerky "frozen" beats reminiscent of 90s Big Beat fame.

5. The shake trigger input can also be used creatively by varying the amplitude of the input CV or pulse you're using, for example via velocity.

Credits

Peter Möller – mathematical modeling. **Torkel Svensson** – mechanical analysis. **Oscar Öberg** – modeling and implementation. **Niklas Odelholm** – framework programming and graphic design. **Kristofer Ulfves** – project management, presets, validation, user manual. **Igor Miná** – user manual layout. **Ulf Ekelöf** – 3D rendering and graphics.

MODULAR READY

2 TSAR-1 and TSAR-1R Reverb for Modular

The TSAR-1 reverb is not an emulation. It's not a stock design. It's not a snapshot of a space. The TSAR-1 is a powerful, modern reverb algorithm. It's alive and vibrant, it's gentle and dreamy, and above all – it's natural and believable. The TSAR-1 is the better-sounding alternative to the established, traditional reverb products. At a fraction of the price.

For a full description of TSAR-1 and TSAR-1R, please see its main chapter on page 188.

TSAR-1 User Interface

Both TSAR-1 and TSAR-1R are based on the same reverb algorithm, but uses different controls. TSAR-1 gives the user more power to control the details, while TSAR-1R is easy to dial in, and very easy to get a good natural sound from.

Parameters

ER Type SMALL, MEDIUM, LARGE

The Early Reflections (ER) Type give the user a sense of the room dimensions.

ER Mix 0-100%

The Early Reflections (ER) mix parameter sets the relationship between the early reflections and reverb tail.

Diffusion LOW, MEDIUM, HIGH

This parameter set the smoothness of the sound, but also a sound with high density also takes more space in the mix.

Modulation RANDOM, SLOW, FAST

The modulation parameter sets modulation of the internal feed-back loop. For most reverbs Random sounds most natural, but you can set it to Slow or Fast for a more chorus-y effect.

Pre-Delay 0-1000ms

This parameter sets amount of delay between the early reflections and reverb tail.

Reverb Time 0.15 – 15s

This knob sets the time it takes for the volume of the reverb tail to drop 60 dB. In a real world analogy, the reverb time would be how much the walls of a room reflect the sound. For instance, a big room with much acoustic treatment has shorter reverb time than a stone-wall church of the same size.

Density 0-100%

The density parameter sets whether you want a small space with thicker, smoother reverb with more reflections (high density) or a larger, thinner sounding one (low density). Low density reverbs are very handy if you need a long reverb that doesn't take up too much energy in your mix.

Reverb Tone -100 (Dark), through 0 (Neutral) to 100 (Bright)

This knob will adjust the tonality/color of TSAR-1's reverb tail.

High Cut 0.2 – 20kHz

This knob adjusts the cutoff point for the high frequencies for both the tail and early reflections.

Output Volume -inf – 6dB

Sets the output volume of the TSAR-1 module (including dry signal).

Reverb Mix 0-100% (dry to wet)

This knob sets offset mix between the direct signal and the reverb signal (including early reflections).

The TSAR-1 module in Modular builds on the same principles as the TSAR-1 plugin. For more detailed information on the inner workings of the TSAR-1 algorithm, look at the TSAR-1 chapter in the Softube Plug-Ins Manual.

Inputs

Volume CV This input jack let you externally control the overall output Volume of the TSAR-1 module in Modular (including the Dry signal).

Tone CV The Tone CV input jack externally controls the tonality/color of TSAR-1 Module's reverb tail.

High Cut CV The High Cut CV input controls the cutoff frequency of for the tail and early reflections.

In L This is the left input jack of the TSAR-1 module, it is normalized to the right input if no cable is inserted there meaning that a mono signal easily can be spread across the whole reverb.

In R This is the right input jack of the TSAR-1 module.

Reverb Mix CV	This CV jack is for external control of the Reverb mix. Incoming CV at this jack is added to the Reverb Mix offset set by the Reverb Mix knob.

Outputs

Out L	Main output of the TSAR-1, left channel.
Out R	Main output of the TSAR-1, right channel.

Feeding TSAR-1 on to itself can be lots of fun, but it can create unpleasant feedback, be careful not to ruin your ears!

TSAR-1 In Use

Add your TSAR-1 module anywhere in your patch where you want to insert reverb, most commonly would probably be right before patching to main output. Use the **Dry/Wet** knob to adjust the level of reverb in your patch. But you could of course also use the TSAR-1 module as a completely "WET" source and mix it through an external mixer module such as the Audio mixer or X-fade mix modules.

The presets supplied in Modular and Modular FX shows some different uses of TSAR-1 in Modular.

TSAR-1R User Interface

The TSAR-1R is very easy to dial in. Just select the type of room you want (from STUDIO to LARGE HALL) and decide if you want a BRIGHT, NEUTRAL or DARK sounding reverb. That's it!

TSAR-1R Reverb is the little sibling of the more adjustable TSAR-1 Reverb. They share exactly the same high end reverb algorithm, but TSAR-1R offers extreme ease of use for a fast and intuitive workflow.

Parameters

Predelay	0-200ms
	The Predelay has the same function as the TSAR-1 Module, but is limited to 200 ms maximum delay time (which is more than enough for natural sounding reverbs).
Reverb Time	0.2 – 15s
	In the TSAR-1R module the Reverb Time parameter adjusts an internal pre-delay, the early reflections, diffusion, density and decay time – all at once. All these parameters have been meticulously fine-tuned to give you a natural sounding result.
Color	DARK, NEUTRAL, BRIGHT
	This parameter set the overall tonal character of the TSAR-1R module. BRIGHT is useful for strings and vocals, or when you want to give the perception of a highly reflective room, NEUTRAL for normal halls or studios, and DARK for smaller spaces with a lot of acoustic damping.
Reverb Mix	0-100%
	This knob sets offset mix between the direct signal and the reverb signal.

Output Volume	-inf – 6dB Sets the output volume of the TSAR-1R module (including the dry signal).

Inputs

Reverb Mix CV	This CV input jack is for external control of the Reverb mix. Incoming CV at this jack is added to the Reverb Mix offset set by the Reverb Mix knob.
Output Volume CV	This input jack let you externally control the overall output Volume of the TSAR-1R module (including the Dry signal).
In L	This is the left input jack of the TSAR-1R module, it is normalized to the right input if no cable is inserted there.
In R	This is the right input jack of the TSAR-1R module.

Outputs

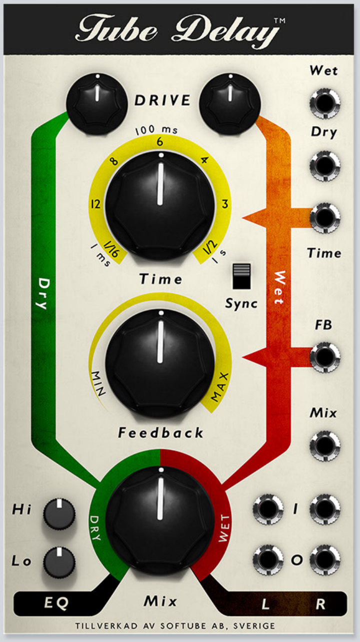
Out L	Main output of the TSAR-1R module, left channel.
Out R	Main output of the TSAR-1R module, right channel.

Feeding TSAR-1 on to itself can be lots of fun, but it can create unpleasant feedback, be careful not to ruin your ears!

TSAR-1R In Use

Add your TSAR-1R module anywhere in your patch where you want to insert reverb, most commonly would probably be right before patching to main output. Use the **Dry/Wet** knob to adjust the level of reverb in your patch. But you could of course also use the TSAR-1R module as a completely "WET" source and mix it through an external mixer module such as the Audio mixer or X-fade mix modules.

The presets supplied in Modular and Modular FX shows some different uses of TSAR-1R in Modular.

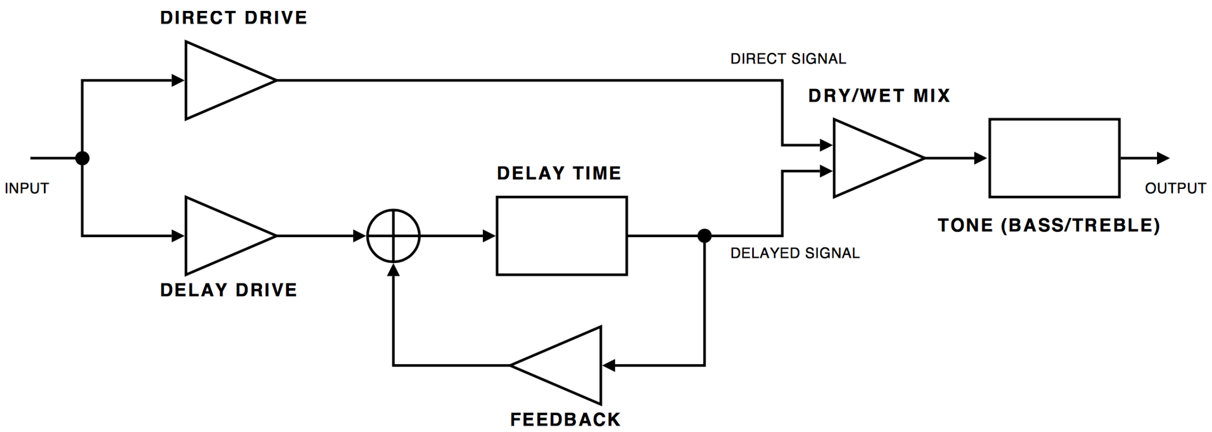


15 Tube Delay For Modular

The Tube Delay module for Modular is based on Softube's stand-alone native plugin and shares a lot of the same features. Tube Delay features a dry and a wet path, both with built in tube saturation. The wet path features the delay line which has an analogue tape echo behavior and an additional tube saturation in its feedback path. Added on the Modular module are, of course, CV control capabilities for most of the parameters. Note that the maximum delay time is always one second, which can make Tube Delay unsuitable for certain synced echoes. For more detailed information about Tube Delay, see also its separate chapter.

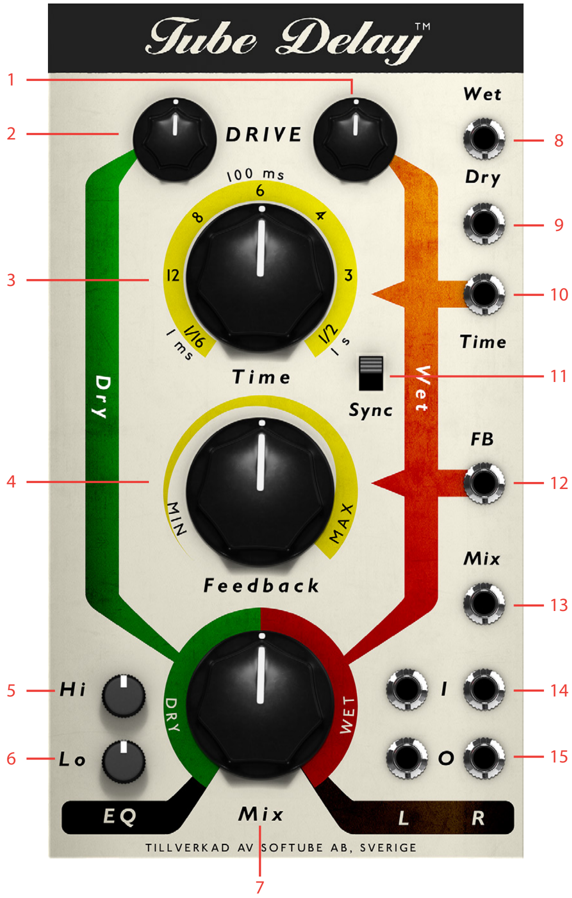
Block Diagram

Below, the a block diagram depicting the Tube Delay is shown. As you can see, the dry signal (direct signal) isn't very dry at all. It is affected by both the tone stack and the tube circuits in the Direct Drive knob.



For simplicity, the Tone control has been depicted as the last stage after the Mix knob. In reality, the Tone control is embedded within all tube stages, and will thus affect all distortion.

Overview



1. Wet drive
2. Direct drive
3. Time knob
4. Feedback knob
5. Treble (Hi)
6. Bass (Lo)
7. Mix knob
8. Wet drive CV input
9. Dry drive CV input
10. Shake in
11. Sync
12. Feedback CV input
13. Mix CV input
14. Left and right inputs
15. Left and right outputs

Parameters

Direct Drive This knob controls the amount of tube drive in the dry signal path.

Delay Drive This knob controls the amount of tube drive in the wet signal path.

Time This knob controls the length of the delay in milliseconds, or, when the Sync switch is set to on, in divisions of a beat. The maximum length of delay after the original signal in the Tube Delay module is one second. In millisecond mode (the sync switch set to off), Delay Time will adjust the time from 1 to 1000 ms. The first half of the control goes from 1 to 100 ms, the second half goes from 101 to 1000 ms. This gives tweaking in the 1 to 100 ms range a greater resolution.

Sync This Tempo sync switch locks the delay time of the Tube Delay to the tempo setting of the host application. The Delay Time knob sets the length of the delay in these fractions of a measure: 1/16, 1/12, 1/8, 1/6, 1/4, 1/3 and 1/2. To get to these values directly, simply click the numbers around the knob.
It's also possible to get values in between the fractions by adjusting the knob.

Feedback This knob will affect the amount of the wet signal that is sent back into the wet path. With this knob set around 80% and above (dependent on EQ setting) the delay will go into self-oscillation. Increase or decrease of Treble and Bass will affect this as well.

Mix This controls the mix of Dry and Wet signal in the output.

Treble This knob controls the high end of the built-in equalization in the Tube Delay module. It affects both the Dry and Wet section.

Bass This knob controls the low end of the built-in equalization in the Tube Delay module. It affects both the Dry and Wet section.

Inputs

Wet Drive CV Input A modulation CV (from example a LFO, envelope or sequencer) input here will control the tube-drive of the Wet (i e the delayed) portion of the Tube Delay.

Dry Drive CV Input A modulation CV (from example a LFO, envelope or sequencer) input here will control the tube-drive of the Dry portion of the Tube Delay.

Time CV Input This is a CV modulation input for the delay time. Its range is +-5v.

FB CV Input This is CV control input for the feedback amount.

Mix CV Input This is CV control input for the Dry/Wet Mix amount.

Left Audio Input This is the left audio input of the Tube Delay.

Right Audio Input This is the right audio input of the Tube Delay. The left channel is normalized to the right audio input if no signal is input at the right audio input. This results in mono audio.

Ouputs

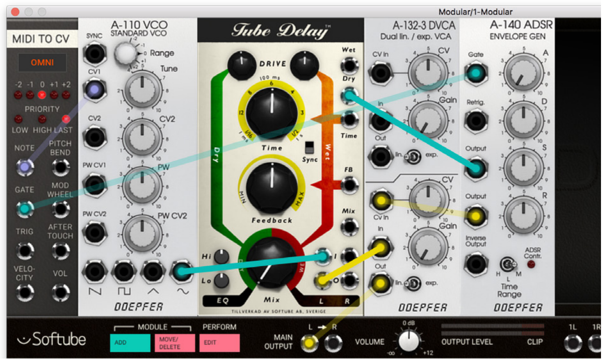
Left Audio Output This is the left audio channel of the output channels of Tube Delay.

Right Audio Output This is the right audio channel of the output channels of Tube Delay.

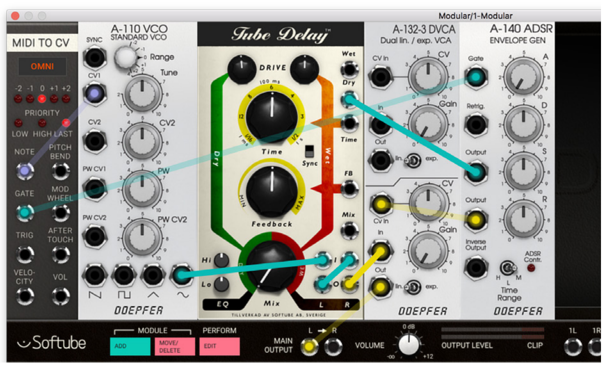
The Tube Delay module in use

The Tube Delay module in Modular is more than just a delay: because the dry signal path is really a model of a tube preamp, you can also use the Tube Delay module in fully "dry" mode to color or distort audio passing through it like you might use a "real" tube preamp.

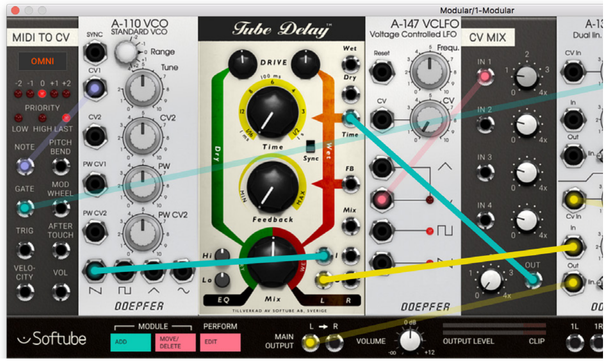
1. Use the tube drive of the module for waveshaping of oscillators where the different characters of the dry and wet section can be further enhanced by CV controlling the mix at the same time.



2. Chain the inputs and outputs in a criss-cross pattern to get twice the effect of the drive feature.



3. Subtle and fast modulation of the time parameter, creating variations on a short delay, can successfully be used to create chorus-like effects.



Credits

Torsten Gatu – original sound design and framework **Oscar Öberg** – modeling and framework **Niklas Odelholm** – graphical design and framework **Arvid Rosén** – framework **Peter Möller** – framework **Ulf Ekelöf** – 3D rendering **Kristofer Ulfves** – User Manual, testing and presets **Igor Miná** – user manual layout, photo **Henrik Midtgaard** – original concept.

16 Vermona Random Rhythm

Vermona is a brand of the HDB company, based in Erlbach, Saxony in Germany and have a long tradition as manufacturers of music gear going back to the early years as VEB Klingenthaler Harmonikawerke – a manufacturer of accordions.

The current company is descended directly from the nationally owned enterprise Vermona in the former GDR which has been building electronic music instruments and accessory for decades.

Vermona Random Rhythm is a random-based dual-channel trigger sequencer created with performance in mind. It creates rhythmic patterns out the probability appearance of quarters, eighths, sixteenths and triplet notes within a bar. Each probability trigger output is controlled via four corresponding sliders for each of the two rhythm channels while a fifth sequence output combines all the triggers.

Overview

Vermona Random Rhythm consist of two rhythm sections, 1 and 2. Four sliders on each channel correspond to the note-values quarters, eighths, sixteenths and triplet notes and influence the probability that they will appear at each step in the generated pattern.

Each of the two rhythm-sections can have their own master-clock and work in realtime or dice mode.

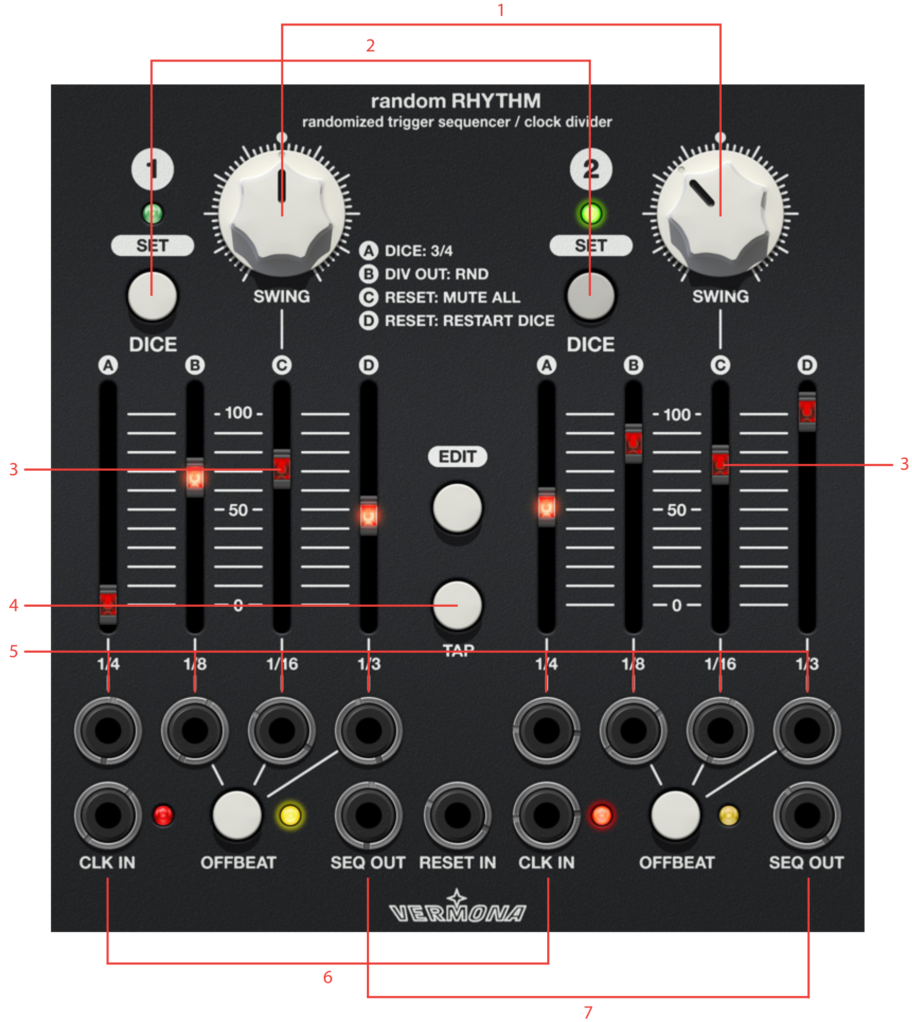
Realtime means, that the module is continuously generating new random values for each note value whereas in dice-mode it generates them for a complete 3/4- or 4/4-bar (this is the default mode on the module). In both modes, the probability sliders will still influence the resulting sequence that appears on the sequence out jack.

Beside the sequence output of each rhythm-section there are individual outs for the quarters, eighths, sixteenths and triplets. You can use them with or without random. The latter makes randomRHYTHM a precise tool to generate or multiply clocks.

Vermona Random Rhythm can create its own or work with external clocks (per rhythmsection if you like).

Additional fun is guaranteed by the flexible RESET input that lets you mute the outputs or restart your diced bar. As with all parameters, the functionality can be set individually per rhythm-section.

randomRHYTHM is a creativeness booster. It is easy to create complex rhythms or straight four-to-the-floor beats. Its foolproof user interface lets you play it without thinking about randomness and probabilities, although that's the fundamental concept of the module. You don't need to be a stochastic genius! It doesn't matter if you use it as sole rhythm-base or as valuable addition to other sequencers. randomRHYTHM gives you a different view to rhythms and the way to generate them.



1. Swing
2. Dice button(s)
3. Probability sliders
4. Tap tempo
5. Individual outputs
6. External clock input(s)
7. Sequence output

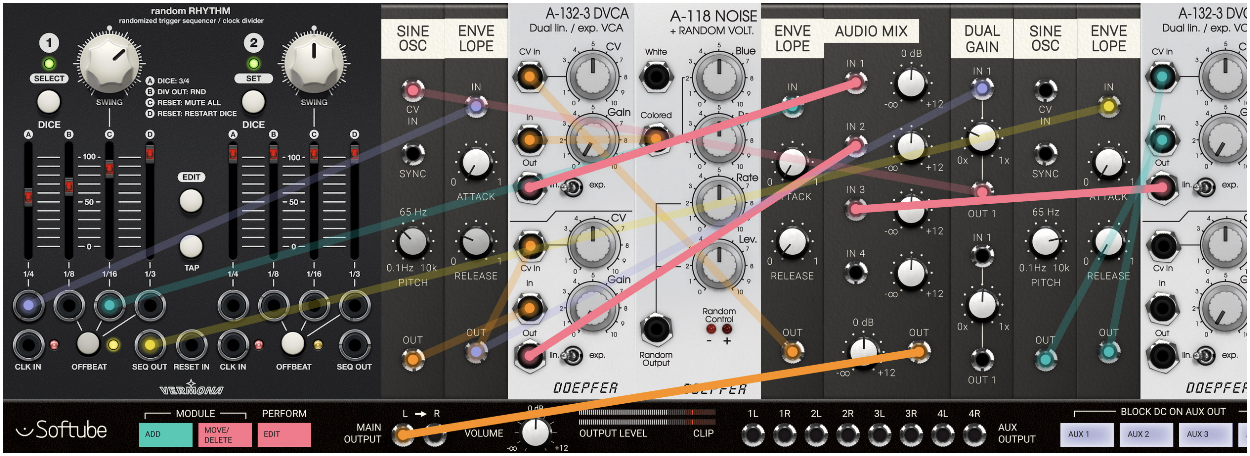
Getting started

1. Add Vermona Random Rhythm to your Modular virtual rack along with a sineoscillator, an utility envelope and a Doepfer A-132-2 DVCA.

2. Connect the output of the quarter note (¼) jack output on channel to the envelope input jack in order to trigger the envelope, and then setup the envelope to control the volume of the sine-oscillator through the VCA as pictured.

3. Try out experimenting with different levels of probability in your played back quarter note pattern by changing the quarter note slider position the corresponding channel.

4. Add a noise-generator and route that trough the second half of the A-132-3 DVCA controlled by a second envelope. Route the output of the sixteenths note jack to this second envelope as pictured.



5. By changing the probability slider of the sixteenths notes you'll notice that you're also filtering out different parts of the sixteen notes pattern. By reseting the slider to the previous position you can always come back to the trigger pattern generated for this particular output.

6. By clicking on the Dice button on top of the corresponding channel, a new pattern for all four note-values - quarter, eights, sixteenths and eigth note triplets - is being generated.

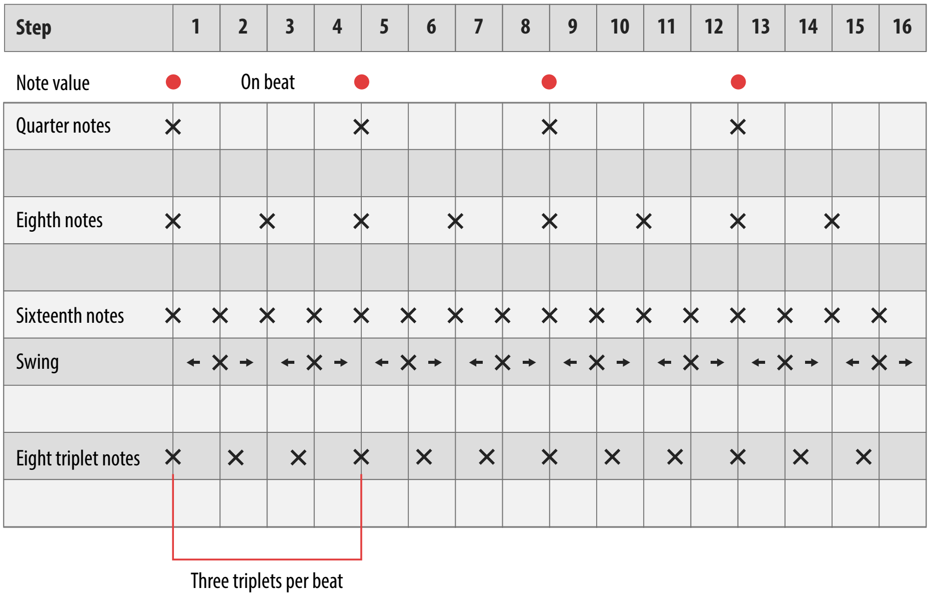
7. Try turning the Swing knob on top of the corresponding channel to hear the sixteenths notes being shifted in time to create swing (sometimes also refereed to as shuffle) in the pattern.

8. By clicking on the off-beat button on the corresponding you will also notice that sixteenths notes that occur at the same time as the quarter notes (on beat) is removed out of the sixteenth note pattern.

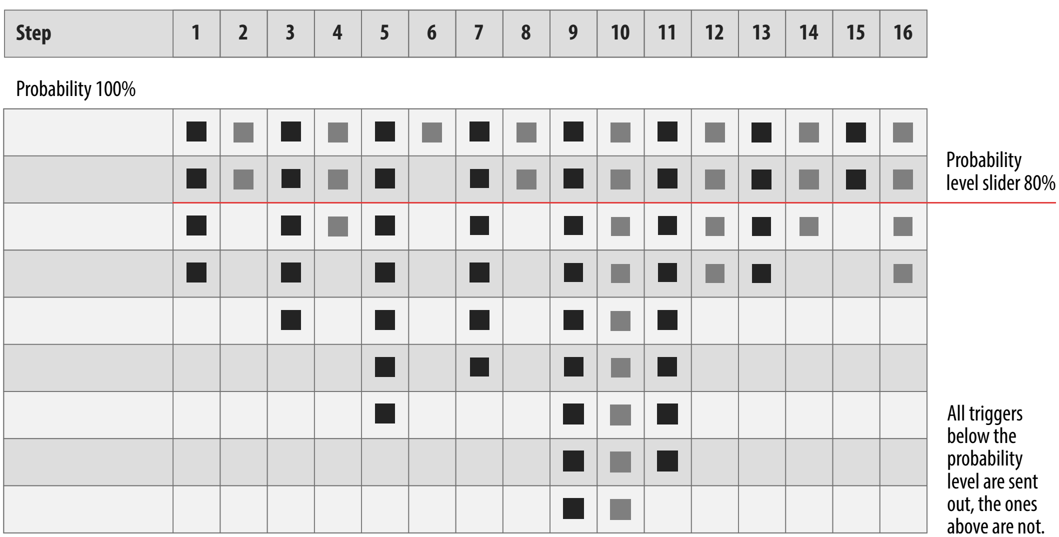
9. Add yet another oscillator, envelope, Doepfer A-132-2 DVCA and a mixer to create a third percussive element to this pattern. Patch Seq Out of the corresponding pattern to the envelope in order to hear the combination of all four note probabilities. Notice that this output is not affected by the off-beat button.

On the next page there's a visual view of the generated note-patterns of the Vermona Random Rhythm.

Overview Generated pattern triggers



Probability Example pattern Sixteenth notes



Parameters

Ch1 Realtime This parameter turns real-time mode on or off. Realtime is activated (green LED is unlit) by clicking on the channel number on top of each channel. Realtime is the when the module is continuously generating new random values for each note value, whereas in dice-mode it generates them for a complete 3/4- or 4/4-bar (this is the default mode).

Ch1 Swing Shifts the 16th notes several clock-cycles ahead or behind in timing to create "swing" in your generated rhythmic pattern.

Ch2 Realtime Same as Ch1 Realtime but for channel 2.

Ch2 Swing Same as Ch1 Realtime but for channel 2.

Ch1 Dice Clicking this momentary button, Dice, will generate a new probability pattern for all of the individual pulse and combined (Seq out) outputs on channel 1. This button can also be used to de-activate Real-time mode on channel 1 (see description above).

Ch2 Dice Clicking this momentary button, Dice, will generate a new probability pattern for all of the individual pulse and combined (Seq out) outputs on channel 2. This button can also be used to de-activate Real-time mode on channel 2 (see description above).

Ch1 A Level Determines the probability that a trigger should occur for all quarter notes (1/4) for Channel 1. A high value will output a 4/4 ("4 on the floor") beat suitable for triggering a kick-drum. Lower value will determine the chance in percentage that a trigger will occur on each step.

Ch1 B Level Determines the probability that a trigger should occur for all eigths notes (1/8) for Channel 1. A high value will output a regular eighths note pulse while lower values determines the possibility that a trigger will occur on each step.

Ch1 C Level Determines the probability that a trigger should occur for all sixteenths notes (1/16) for Channel 1. High value will output a regular sixteenths note pulse while lower values determines the possibility that a trigger will occur on each step.

Ch1 D Level Determines the probability that a trigger should occur for all eights notes triplets (indicated 1/3 on panel) for Channel 1. High value will output a regular eighth note triplet pulse while lower values determines the possibility that a trigger will occur on each step.

Ch2 A-D Level Description remain the same as for Ch1 A-B Level but applied on channel 2.

Edit This button switches Vermona Random Rhythm into edit mode, see full description in separate section below.

Tap This is the tap-tempo button. Clicking on this button three or more consecutive times will extract the tempo and set internal clock of Random Rhythm to this tempo. This button has no effect if both clock-inputs are in use.

Ch1 Offbeat This toggle button will invoke the Offbeat mode for the channel 1 eighth, sixteenths and eighth triplet outputs. This means that these outputs will no longer send out triggers on the beat (at the same time as the quarter notes). The Offbeat triggers are indicated by the LEDs on top of the probability sliders.

Ch2 Offbeat Same functionality as for channel 1.

Indicators

Ch1 Realtime This green LED situated right below the channel 1 symbol on the left but above the "SELECT" label. This LED is unlit when in Realtime mode and lit when in Dice mode (default).

Ch2 Realtime This green LED situated right below the channel 2 symbol on the right but above the "SET" label. The functionality is the same as for channel 1.

Ch1 A-D Level The LED indicators on top of each probability slider show the off-beat triggers. The visual sum always represents the triggers output at the sequence out jack regardless of the Div Out: RND edit parameter (see description below).

Ch2 A-D Level Same functionality as channel 1.

Ch1 Clk in This indicates the tempo that this channel is running, whether it was set by internal clock (tap tempo) or external clock inserted at the Clk in jack.

Ch1 Offbeat When this indicator is lit, the 8ths, 16ths and triplets on the beat (quarters) are excluded from its associated jack.

Ch2 Clk in This indicates the tempo that this channel is running, whether it was set by internal clock (tap tempo) or external clock inserted at the Clk in jack.

Ch2 Offbeat Same functionality as Ch1 Offbeat.

Inputs

channel 1, clk in	This clock input for channel 1. Expected signal to be input here are quarter notes equalling same behavior as tap-tempo button.
reset in	Dependent on this setting in the edit menu, this input will mute the outputs, restart your dice feed or restart your pattern (default behavior is restart). Note that this can be set to for the two channels to react differently independent of each other.
channel 2, clk in	Same functionality as Ch1 clk in, but separate clock for channel 2.

Note that the channel clock inputs are both normalized to each other, meaning that any clock inserted into one channel jack will automatically also drive the other channel unless this channel has a clock of its own (tap-tempo or external).

Outputs

Channel 1, 1/4	Trigger out for channel 1 quarter notes, its behavior can be set in edit mode.
Channel 1, 1/8	Trigger out for channel 1 eighths notes, its behavior can be set in edit mode.
Channel 1, 1/16	Trigger out for channel 1 sixteenths notes, its behavior can be set in edit mode.
Channel 1, 1/3	Trigger out for channel 1 eighth triplet notes, its behavior can be set in edit mode.
Channel 2, 1/4	Same functionality as similar jack at Ch1, for channel 2.
Channel 2, 1/8	Same functionality as similar jack at Ch1, for channel 2.
Channel 2, 1/16	Same functionality as similar jack at Ch1, for channel 2.
Channel 2, 1/3	Same functionality as similar jack at Ch1, for channel 2.
Channel 1, seq	Combined trigger out sequenced probability output for channel 1.
Channel 2, seq	Combined trigger out sequenced probability output for channel 2.

EDIT mode

There are four additional settings that can be adjusted for each section. These settings will be saved as part of the saved project or preset. For the EDIT parameter, the DICE buttons of rhythm-section 1 and 2 have different functions: SELECT (DICE channel 1) and SET (DICE channel 2) as market on the panel.

To change the EDIT parameters press the EDIT button and the green LED above the SELECT button will start flashing. In addition, the red LEDs of the channel probability sliders will no longer appear the same anymore (no flashing for indication of sequence triggers). Instead, now the LEDs of the active EDIT parameter are light up permanently.

By repeatedly pressing SELECT (the left DICE button), the edit functions A, B, C and D for rhythm-sections 1 and 2 are selected one after another. This is indicated by the LED of the corresponding slider flashes. The user can change the edit parameters from off to on (or vise versa), by clicking of the right DICE button (SET) and the corresponding parameter enabled. To exit edit-mode, simply press the EDIT button once more. The red LEDs of the probability sliders visualize the switching in two ways: The parameter that has been selected using SELECT has a flashing LED. If a parameter has been enabled the corresponding LED of the slider flashes with real short stops. If a parameter has been disabled, the corresponding LED of the slider flashes with longer interruptions. For the other seven sliders, which are currently not being selected, the red LEDs are lit for enabled parameters and off for disabled parameters.

The EDIT parameters

A - DICE 3/4

The factory-setting of Random Rhythm is a 4/4-beat-resolution, with dice-mode enabled. However, it is possible to switch to a 3/4-beat. By combining both rhythm-sections with 4/4- and 3/4-beats, the generated patterns are continuously shifted against each other. This can also be interesting when used in combination with external sequencers.

Parameter active - LED on = 3/4-time

Parameter inactive - LED off = 4/4-time (factory default)

When hitting DICE, random Rhythm always generates random-values for a 4/4-beat, even when EDIT parameter A - DICE 3/4 is activated.

B - DIV OUT: RND

The random-function can be switched off for the individual outputs. In this case, continuous trigger-patterns equaling a clock-signal, are being generated and sent to the corresponding outputs 1/4, 1/8, 1/16 and 1/3. Here, the sliders do not carry out any function. However, the OFFBEAT button d can still be used to specify whether all trigger impulses of the corresponding output will be send or just the impulses that follow the basic concept of the divided pattern.

Parameter active - LED on = random (factory default)

Parameter inactive - LED off = continuous clock-signal

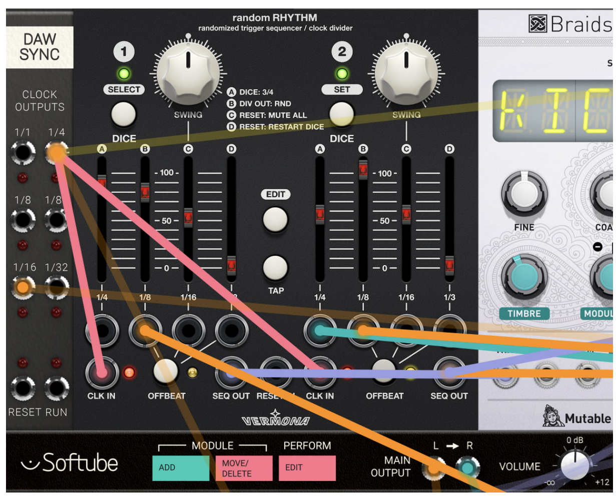
The RESET Functions C and D each specify the operating mode of the input RESET IN. Only one of the two functions can be enabled per rhythm-section, since these functions are mutually exclusive. Without any function being activated, the input RESET IN h takes no influence on the corresponding rhythm-section.

C - RESET: MUTE ALL With this function being enabled, the RESET IN input h allows to mute all trigger-outputs (1/4, 1/8, 1/16, 1/3 and SEQ OUT). Whenever a permanent, positive voltage above 2 volts is applied to input RESET IN, no trigger-impulse will be send out to the corresponding outputs. Once the voltage drops below 2 volts, the outputs will be released again.

D - RESET: RESTART DICE This function is only active in dice-mode. With this function being enabled, a trigger-impulse (positive slope) can be used to reset the active pattern to its start. The operating mode for RESET IN h may differ for the rhythm-sections 1 and 2. This leads to interesting rhythmical effects. For example: connect a square-wave-LFO to the RESET IN input h with RESET DICE activated for rhythm-section 1 and MUTE ALL activated for rhythm-section 2. At the LFO's rising slope, rhythm-section 1 is being restarted, while rhythm-section 2 is being muted for the duration of the positive half-cycle of the square-wave. With the slope descending, the trigger-impulses for section 2 are sent again.

Synchronizing Vermona Random Rhythm to DAW tempo

Vermona Random Rhythm was designed to mainly act as a master-clock in your eurorack system, but can be synchronized to DAW tempo by feeding it quarter pulses. Here a suggested setup for synchronizing it in the Softube Modular environment running in a DAW:

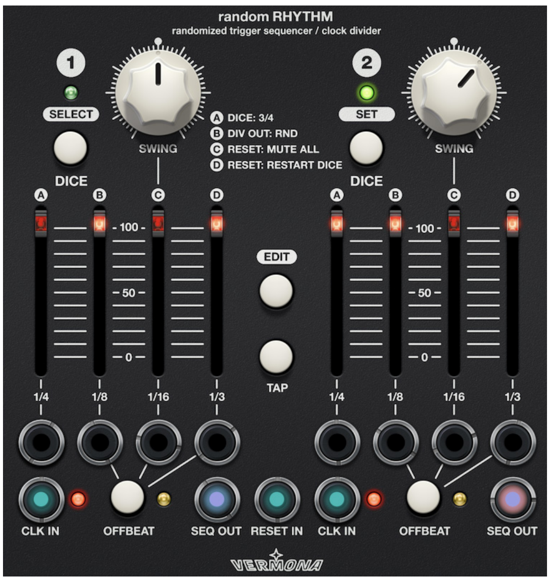


Vermona Random Rhythm will treat the quarter pulses the same way as manual tap-tempo entered through the tap-tempo button meaning that it will only stabilize tempo and start on the third beat. This is normal and the way the original hardware module is designed.

Vermona Random Rhythm module in use:

The Vermona Random Rhythm can be used in a variety of different ways, but here's some suggested uses:

1. A great way producing evocative rhythm patterns is to have one channel set to 4/4 time and the other to 3/4 time. This is done in edit mode as shown below (edit parameter A on channel 1 turned off and turned on on channel 2).



2. By setting the Div Out: RND parameter to "off", this can be used to have one of the the two rhythmic channels to serve as a master clock where the individual outputs are set to continuously send out their values. This is done by setting the edit parameter B (Div Out: RND) on one of your channels.

3. Use one channel to clock the other – this is a fun and creative way to generate new and surprising rhythmic patterns.

Credits

Eric Hampusgård – programming, modeling.

Thomas Haller – original concept Vermona, feedback.

Swen Strobel – original hardware Vermona.

Marcus Faust – original software Vermona, feedback.

Kristofer Ulfves – project management, presets, validation, user manual.

Oscar Öberg – programming, mentoring.

Arvid Rosén – mentoring.

Bitplant – GUI graphics.

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Softube may however terminate the relevant Product license with immediate effect if you breach the provisions of this EULA or the Terms and/or fail to make the agreed payments under the Terms. In case of a violation of our and our licensors', if any, intellectual property rights to the Product or other material breach of this EULA or the Terms, such termination may follow immediately upon detection of such violation/breach. Upon termination of the relevant Product license, you must delete the Product from your equipment and any copies thereof.

7. Agreed limitations in your right to use the Product

You may use the Product for only as many simultaneous users, servers or other limiting parameters as are set out in the Product (by means of technical limitations), applicable Product manual or Products instructions, as set out in the Terms, the EULA or otherwise.

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You may neither alter nor modify the Product and may not analyze, reverse engineer, decompile or disassemble the Product or any part of the Product, incorporate the Product into any other application Product, or print out the Product, except and only to the extent that such activity is expressly permitted by applicable law, notwithstanding this limitation.

8. Limited warranty for the Product

Warranty for media upon which the Product is delivered, if applicable
In addition to what is set out in the Terms, we warrant the media on which the Product is recorded (to the extent the Product is delivered upon such media, otherwise not) to be free from defects in design, material and workmanship under normal use for a period of ninety (90) days from the date of purchase (as evidenced by a copy of the receipt).

Unless otherwise set forth in this EULA together with the Terms and/or following from mandatory and applicable consumer legislation, Softube's entire liability and your exclusive remedy in case of warranty breaches is an obligation to replace he media not meeting the Softube's limited warranty and which is returned to Softube or an authorized representative of Softube with a copy of the receipt. Softube will have no responsibility to replace any media damaged by accident, abuse or misapplication.

Other defects in relation to the Product
Without prejudice to the foregoing and such statutory rights available for you as a consumer (or an individual with corresponding rights) under applicable law, our responsibility and liability with regard to possible defects in the Products shall be limited to faults arising during the above ninety (90) day period and entailing that the Products do not conform to the relevant specification when used on advised equipment and in accordance with the documentation and other instructions, gener-

ally or specifically issued/published from time to time by us. Softube may in its own discretion decide how such possible defects shall be handled, i.e. through rectification, circumvention, issue of a Product update/upgrade, replacement of the Product or through any other appropriate means. We shall not have any liability to remedy defects in the Product where such defect results from any default, non-compliance or non-agreed use of the Product by you. Subject to the foregoing, all conditions, warranties, terms and undertakings, express or implied, statutory or otherwise, in respect of the Products are hereby excluded. Softube's entire liability in case of a defect during the warranty period, which we are unable to rectify, shall be (in our reasonable discretion) either

(a) your right to return the Product (and deletion of the relevant Product license key as per our instructions) and our following repayment of Product price paid by you;

or

(b) our rectification or replacement of the Product (as to ensure that it meets the applicable specification) that does not meet our limited warranty.

This limited warranty is not valid if the alleged defect of the Product has resulted from accident, abuse, or misuse. A possible replacement Product will be warranted for the remainder of the original warranty period or thirty (30) days from your receipt thereof, whichever is longer. Softube disclaims all other warranties, either express or implied, including but not limited to implied warranties of merchantability and fitness for a particular purpose, with respect to the Product and the accompanying materials.

9. "AS IS" and LIMITATION OF LIABILITY

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You however acknowledge and agree that Softube's licensors, subject to the existence of relevant provisions in the agreement and/or understanding between us and relevant third party licensor, if any, are third-party beneficiaries of this EULA with respect to the provisions of this EULA that create rights of or obligations to such third party licensor and/or Softube, as the case may be and that, upon your acceptance of the terms and conditions of this EULA, also such third party licensor will have the right to enforce the provisions of this EULA against you in its capacity as a third party beneficiary to this EULA.

11. Version(s) of the EULA

Both technology development as well as our in-licensing of third party technology and intellectual property rights, if any, may and will result in Softube amending and adjusting the EULA from time to time. The EULA governing your purchase of the product shall however remain the version that originally applied in connection with the original purchase of the Product. However, if and to the extent you purchase a new version and/or additional features to a Product, and in connection with such purchase accept the applicability of a new EULA, that new version of the EULA shall apply.

12. Questions, comments or remarks in relation to the Products, this EULA or the Terms

If and to the extent you have questions, comments or remarks in relation to us, the Products or the EULA/the Terms, please contact us, as to enable us to answer your questions, solve the Product issue and to improve our services to you as an end-user. The easiest way of getting into contact with us is via e-mail support@softube.com or such other communication means as is set out on our website.

13. Applicable law and dispute resolution

If you are a consumer (or an individual with corresponding rights) under applicable and statutory law, we will submit us to the choice of law and competent venues for dispute resolution, as set forth under such applicable statutory rules. Otherwise (i.e. primarily for other end-users than consumers), with regard to applicable law, venue, and procedure for dispute resolution, the corresponding provisions of the Terms shall apply for possible disputes between us and you.

ACKNOWLEDGEMENTS AND LICENSES

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