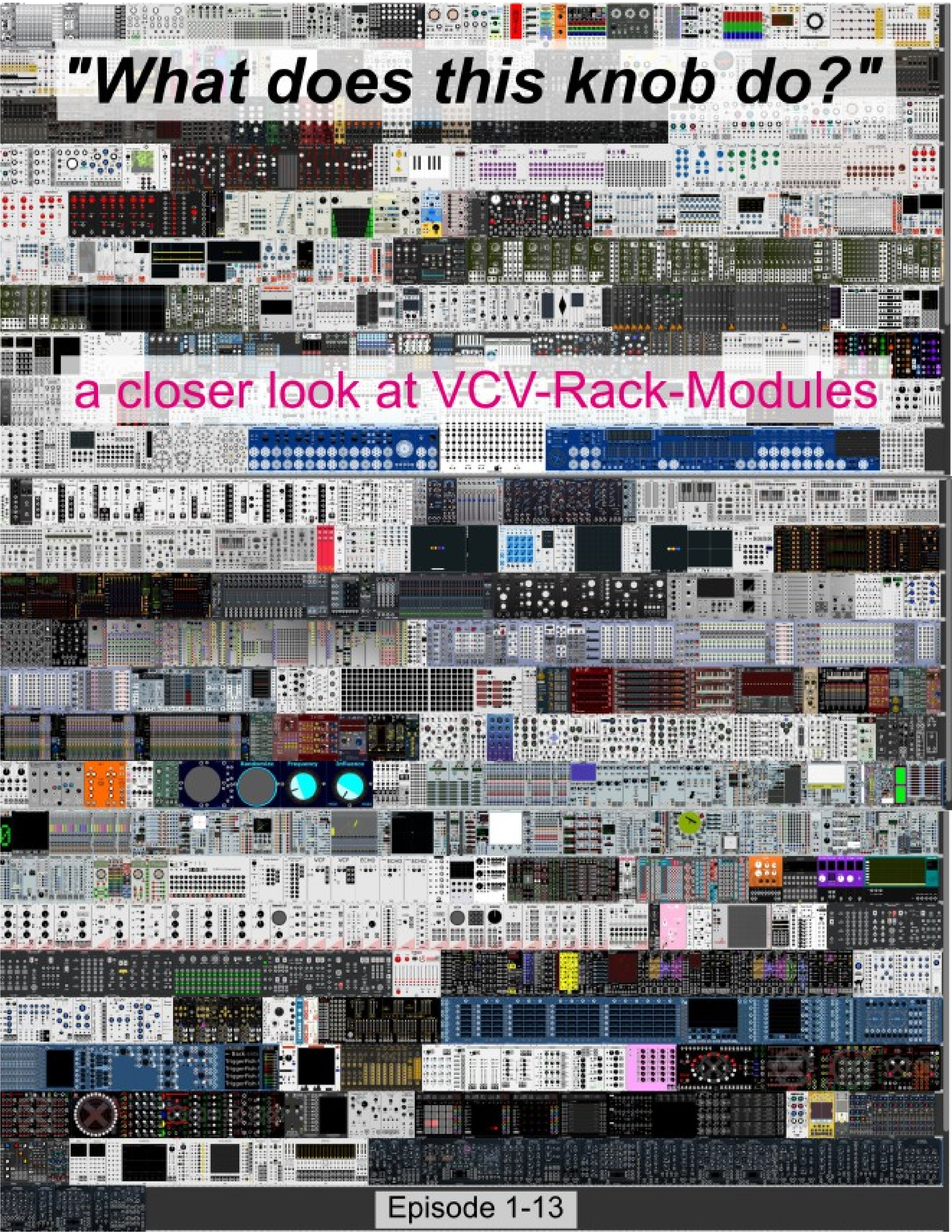
Waves unlimited

"What does this knob do?"

a closer look at VCV-Rack-Modules



Episode 1-13

Editorial

Hello world and welcome to this summary of the video episodes 1-13 of "what does this knob do?". You can read this as it is, but I recommend watching the respective videos, because I often refer to things only visible there.

It was clear to me from the beginning that I would start a mammoth project with it. Introducing all modules is already a very high demand and takes time. In so far 13 episodes I have presented all (then) available VCO, VCF and VCA. In the meantime many modules have been added or revised, so I am working on updates as well as episode 14 on Envelope Generators. And not all descriptions may fit anymore.

These are already a lot of modules and until I use one next time, I may have forgotten my experience with it again. Fortunately, I have my written records and do not have to search for the modules in my videos. And it came to mind that apart from me nobody has this possibility. That's why I decided to prepare my scripts and make them available as a book in pdf format.

In this book you will find all the episodes, the modules presented in it and the times in the video so you can listen to the sound samples as well. In the index, the modules are listed both by module name and by name of the developer and also by category. Where some modules fit into several categories and therefore may appear several times.

Some modules may look different in your Rack, that's because I had to change the colors of many modules to be able to work with. This is not foremost a matter of taste, but of my viewing habits. I don't share this custom skins, unless a developers gives me direct permission.

I hope this book is as useful as the videos and it inspires you to try the featured modules for yourself. Because even if I let my subjective opinion flow, it is just my opinion and you may discover quite different qualities.

Have fun with VCV-Rack and many creative moments wishes you

the1andonlydrno

Episode 1: Introduction and my setup

Hello world, it's really done!

It was not as easy as I thought to make this video. But with a lot of trial & error, looking for the right program and after the missing parcel from amazon was finally replenished, it is now actually time. Here comes my 1st episode of "wous mächdn da Gnobf" or for non-franconians "what does this knob do?".
I try to speak high german, but now and then the Franconian will surely break through in me, If you then do not understand something, then there are still the desired english subtitles. And by the way, this is not a word by word translation of everything I say in the video. It is how it was supposed to be, without stuttering and ähms. I thought, that's the better way. So enjoy.

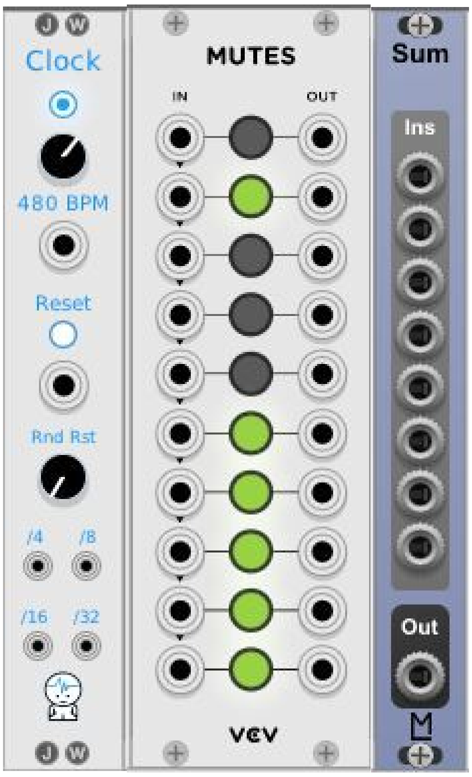
What is this about? In the broadest sense it's about music, especially about synthesizers. Whoever gets off here, I recommend to googling for the term and to deal with the basics. For everyone else, it's all about the modular system VCV-Rack. I can not say how grateful I am to Andrew Belt and the many module developers that there is a free modular system with end-less free modules, which is constantly evolving thanks to open source. And you can even save your patches and make them available to others. I would never have hoped for such a thing in my wildest dreams. THANK YOU THANK YOU THANK YOU!!!

Of course, the multitude of modules also brings with it some questions. In principle, of course, anyone who deals with the matter, for example, knows what a VCO is. However, if we take a closer look at the different oscillators, almost all of them have some peculiarities that are not immediately obvious. And since the most exciting thing about module systems is the alienation of modules, so to use them for things for which they are not really designed, it is extremely important to know what the individual components of a module do. I've been sit-ting in front of it many times wondering "wous mächdn da Gnobf", so I've started to create a VCV-Rack work environment that allows me to compare and scan modules. My subjective impressions will be as simple as possible and I hope you can understand them. I also work wherever I can with measurement methods and graphical representation, because it makes many things easier to understand.

In this first episode, I introduce you this work environment (setup), or the modules used by

me here before. I always try to keep this constellation, even the basic wiring. Over time, you will see all the modules again, because I will introduce the modules according to their function in groups, starting with oscillators or VCO's. Of course, I do not want to miss the point that there are many great tutorials on a variety of modules that have helped me a lot. Of course, I can not enumerate all of them here, but I just have to mention a few, but I do not do it in any particular order. Nigel Sixsmith (the Art of Sound - Talking Rackheads), Omri Cohen (Improvised Patch Tutorial and others), Andrew Mercer (Buckydurddle), Leonardo Laguna Ruiz (Vult Modules), Greg Brouelette (Modular Curiosity), Hagen Ueberfuhr (theklirrfactor) and of course all rackheads in the VCV-Rack official user group.
You can find links to this under the video, also to modules that you can not download via the Plugin Manager. Basic information on installing and using VCV-Rack can be found on the official website. Do you have questions, wishes, suggestions? I am glad about every comment.

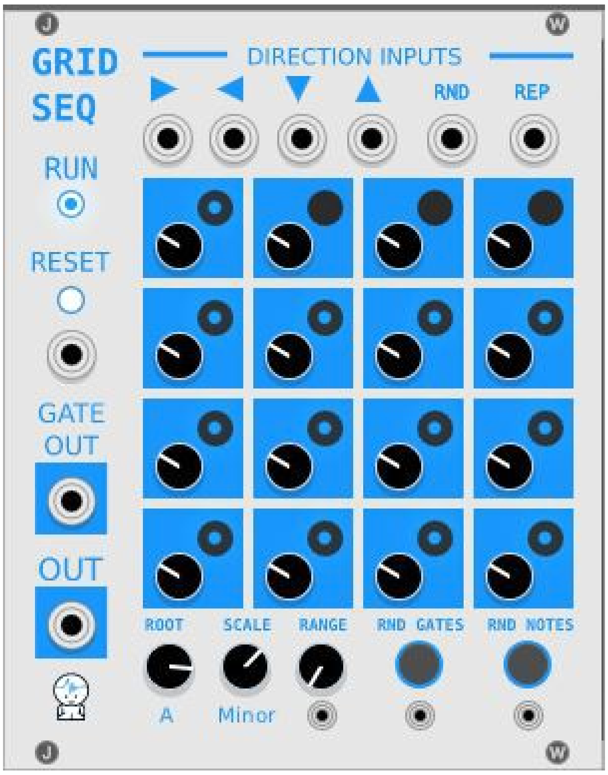
My setup: I do not have a Midi keyboard or a corresponding controller connected, instead my signal comes from a sequencer. Wherever possible, I used modules from the VCV Fundamental range. My other selection criteria were functionality, simplicity and space requirements. Some modules are also out of sight because I will rarely access them. But let's start from left to right.



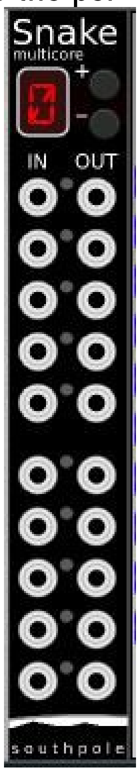
Here we first have the **Simple Clock** from **JW-Modules**. It not only offers an extremely large range (15-3840 bpm), but also different divisions of the original beat number (by 4, 8, 16, 32), so I am flexible enough for a variety of tasks. Currently my speed is 480 bpm and I go with 120 bpm in the sequencer.

However, the various clock signals go first in **Mutes** from **VCV Fundamental** and then in **Sum8** from **ML-Modules**, so I can mute the unneeded clock pulses.

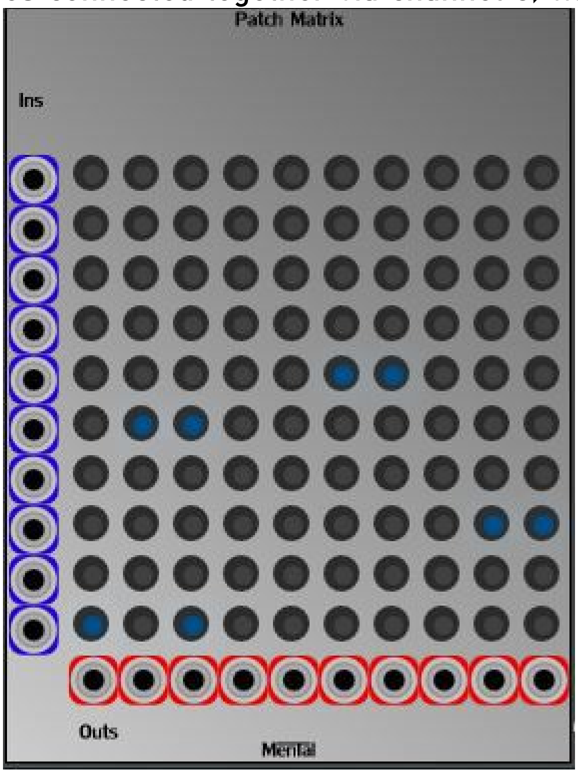
My chosen speed goes into the **Gridseq** of **JW-Modules** and here it is triggered from left to right. Otherwise I use the internal quantizer with pitch a and the minor tone scale. In order to hear the volume envelope, only the 1st quarter of the bar is triggered, the other 3 are not active. In order to have no "ghost notes", I also checked "ignore gate for **V/Oct** out" (in the context menu, click with the right mouse button on module). Here I have also selected the mode "continous". I do not use all the other functions of this sequencer at the moment and therefore discuss them with sequencers later.



From here I use **Tidy Up Cables** from **LogInstruments** to get an overview of my wiring. In my opinion, an extremely useful accessory to not lose the perspective in the cable jungle.



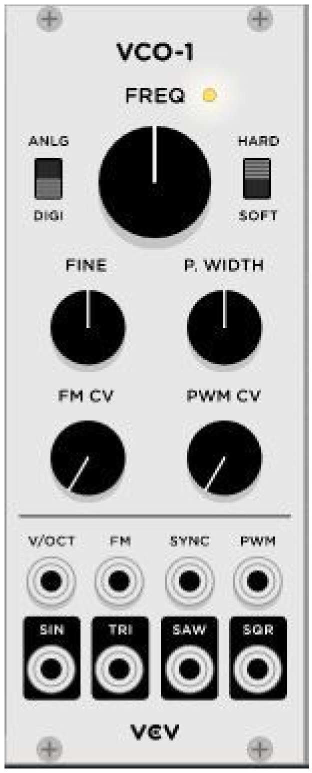
Just like **Snake** from **Southpole**, which is very helpful when connecting cables across the screen. As you can see, I have 6 of these multicores connected together via channel 0, with both inputs and outputs connected to the **Patch Matrix** of **Mental** and various rows. Two more **Snakes** provide the mixer with effects.

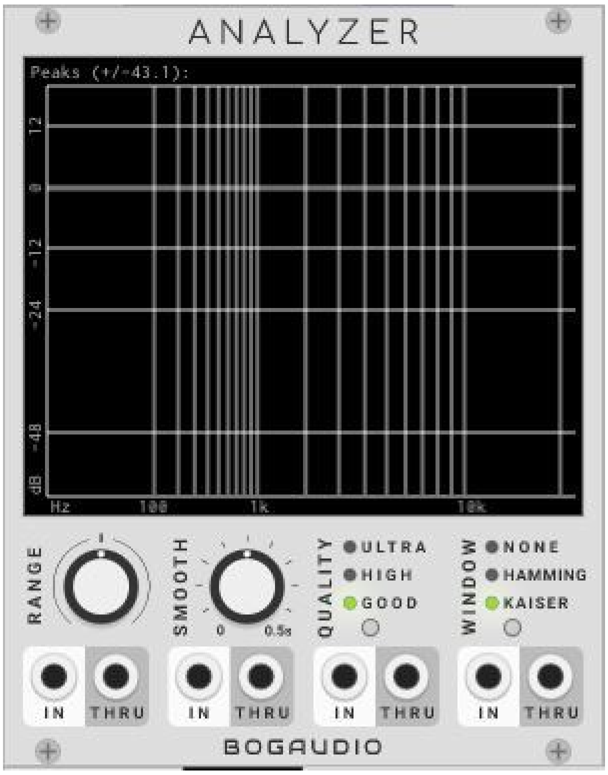


Here **Delayplus Stereo Fx** and **Reverb Stereo Fx** from **AS**, which are placed 3 rows below the mixer and through the use of **Snakes** the necessary long wiring unnecessary. With such a tidy screen, you can easily work well and effectively.

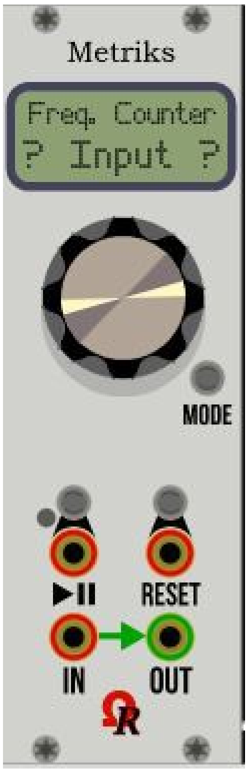
The already mentioned **Patch Matrix** from **Mental** also helps to keep order and offers the possibility to flexibly interconnect the inputs and outputs of modules without having to pull cables.

In my setup I have the **VCO-1** from **VCV Fundamental** as inputs with a **sine** signal on 2 paths. To see how different modules or components of modules change the signal, I once have the most original possible signal of the VCO, which can be visualized in the **Patch Matrix** with the **Analyzer** from **Bogaudio**,





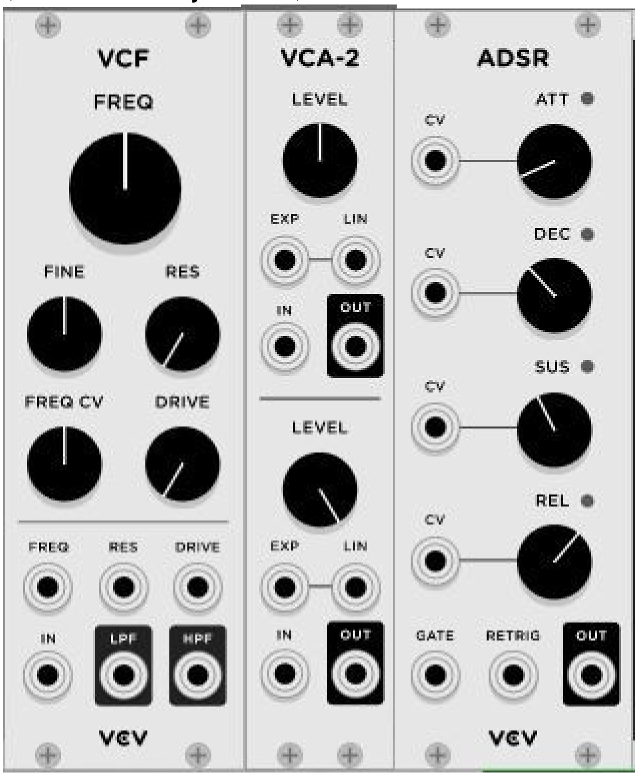
Metriks from **Ohmer Modules**,



the **Full Scope** from **JW-Modules**

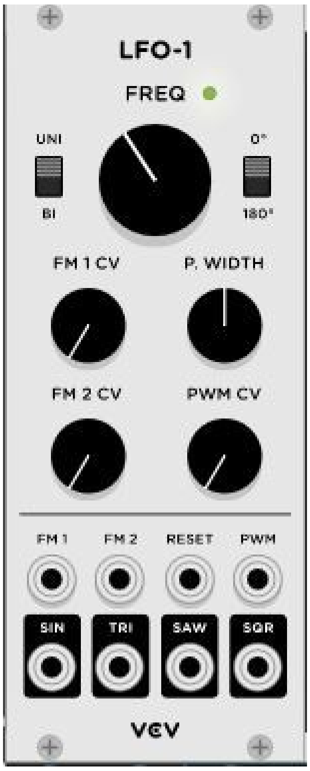
and the **Spectrum Analyzer** from **LogInstruments**.

I also send the signal via **VCF** and **VCA** (modulated by **ADSR**) from **VCV Fundamental** to



hear the volume envelope again. In principle, it is also possible to connect the signal directly to the audio output so that you only hear the oscillator. However, the signal coming from the VCA is returned to the **Patch Matrix** and connected to the audio output.

The path of the 2nd signal (2nd line) differs. Of course, we also want to hear this 2nd signal, so in the **Patch Matrix** we connect **Snake** 0 to another instance of **Snake** 0. From this further instance the signal can now be fed into any module, e.g. as **V/Oct** at an oscillator. The signal coming from the module is in turn passed through this instance of **Snake** on 2 channels back into the **Patch Matrix** and connected there to the visualization modules. The basic structure of the visualization is the same, whereby I use the 2nd signal input of the modules to compare the signals directly. A second instance of **Metriks** also uses this second signal to measure frequency changes. In addition, this second signal is sent over the path VCF / VCA to make the envelope audible. This signal can also be

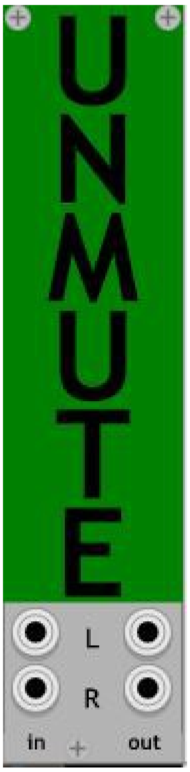


sent directly to the audio out. The original signal is on the left, the 2nd signal on the right channel.

The visualization or measurement always takes place before the signal has been sent via VCF / VCA in order to obtain a picture that is as unchanged as possible. Optionally, the **Patch Matrix** can be used to make cutoff frequency and / or resonance modulation

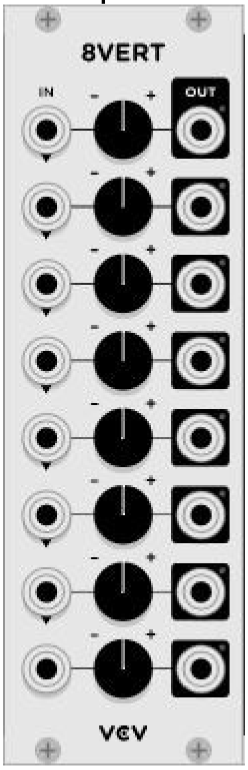
with a **Fundamental LFO-1**. Also the already mentioned effects can be switched on via the aux control of the mixer.

Speaking of mixers, here I use the mixer **9ch, 3groups, 4 aux** from **mscHack**, which is now available in 3 sizes, which I think is great, because these mixers are built like "real" mixers. More about it in the corresponding episode.



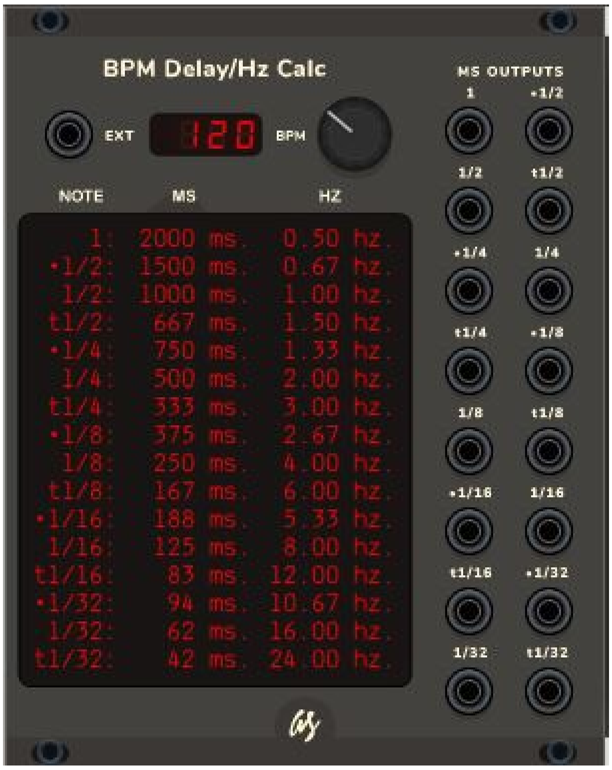
I have not mentioned a few modules, such as **Big Mute Button** by **Alikins**. It does what it should, heavenly silence and that is sometimes veeery important. That's why I placed this module in front of the mixer, because although this of course has a mute function, this is just pressed faster and can thus save lives.

I use **Specific Value** of **Alikins** to create different values, which can be added in the patch matrix.



With the **8vert** of **VCV Fundamental** I increase or decrease the signal strength as needed.

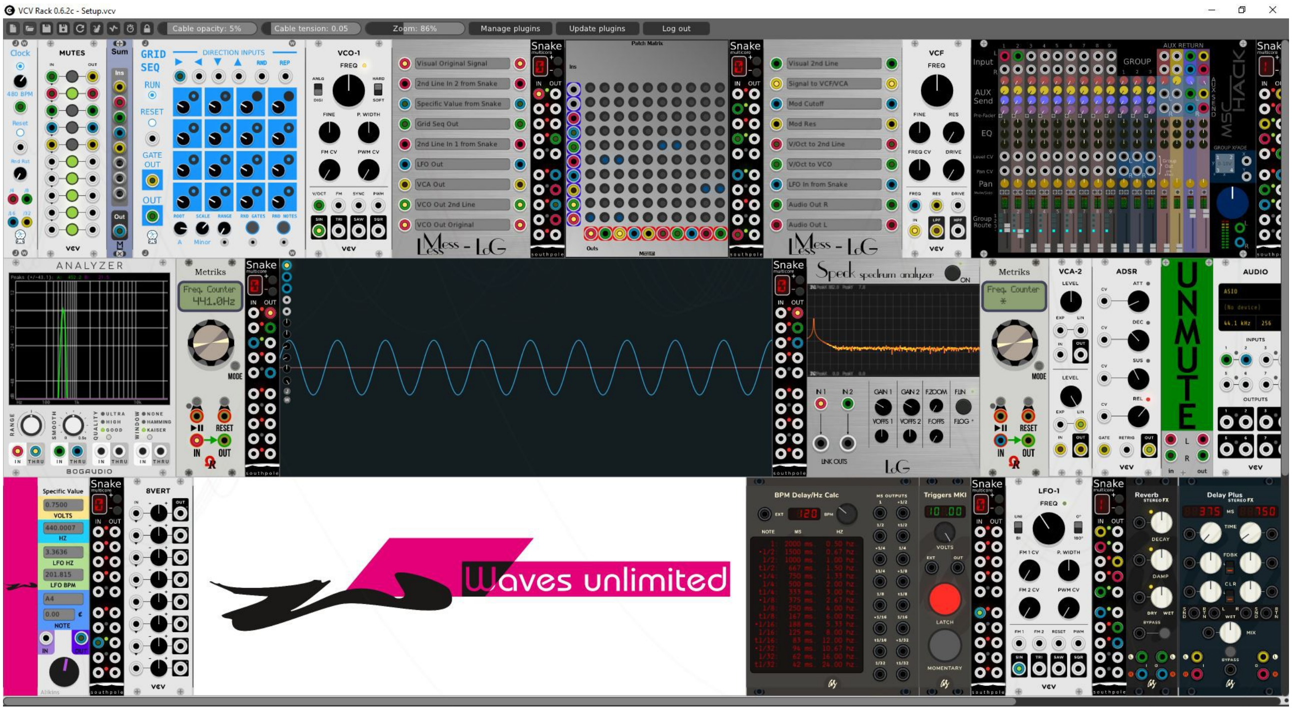
With the **bpm to delay / Hz calculator** from **AS**, I determine the milliseconds that must be set in a delay effect to get specific results.



In order to generate permanent and momentary trigger pulses in different volt-strengths I use the **Trigger MK1** of **AS**.

So, that's my basic setup, which will come up again and again in my videos. As I said, I will discuss each of these modules again, as well as any other freely available module. I will also discuss the commercial modules that I own. But I will certainly not buy every available module, I simply lack the necessary change. But there are so many modules and I do not know how long it will take me to get through it. In the meantime, there will always be new modules, new versions and probably also VCV-Rack 1.0. In any case, I'm really looking forward to exploring the modules and sharing my thoughts with you. Maybe you can do something with it, but maybe you also think of other uses or I missed something. Whatever, let me know.

One thing is for sure, VCV-Rack thrives on the rackheads and their ideas. This is the musical globalization of which musicians like Peter Gabriel have always dreamed, "One World - One Voice" but with a lot of noises and unlimited waves!



Episode 2: XFX Wave from Blamsoft

Hello world and welcome to "what does this knob do?".

The second episode is about oscillators, respectively about one oscillator, namely the **XFX Wave** by **Blamsoft**. This oscillator is a commercial module (in the meantime it's free), which means you can install it through VCV-Rack's Plugin Manager, but you must purchase it first.

However, Andrew Best from **Blamsoft** kindly provided me with this module, so it also gets its own episode. Many, many thanks Andrew, I am very happy that I have the opportunity to try out this oscillator and to share my impressions.

I've had it for a couple of days now and I'm really excited about it. But just form your own opinion.

The **XFX Wave** is a **wavetable oscillator**. It has **70 different wavetables** (meanwhile controllable per CV), which are waveforms that circle through one after the other. **Each wavetable consists of 256 individually selectable waveforms**. And the whole thing can of course be changed and modulated in a variety of ways. Who thinks now, a module with such a variety is certainly very complicated to use, I can take that fear away. The manual is effectively 7 pages and is very easy to understand, if only available in english, but that should not be a problem. But even the oscillator itself is very clear and in my opinion very easy to understand.

But let's take a closer look at the module. At first glance, I notice the displays because I'm a big fan of waveform visualizations. So once a real plus point with me.

The waveform shown is not so extraordinary. This standard **sine** wave I compare here in **Full Scope** with my reference oscillator, the VCO1. We can see that the waveform is identical, with the red curve being the **XFX Wave** and slightly out of phase to better see the two curves in comparison.
This **sine** wave is the first waveform of the **wavetable "basic waves"** and we know that each **wavetable** consists of 256 waveforms. By turning "pos" I move through a **wavetable** and can choose one of the 256 waveforms.

The transitions of the waveforms are fluid, so you can "**morph**" from one to the other. The display shows very nicely how the **sine** wave first becomes a **triangle**, then a **sawtooth** and finally a **square** wave with different **pulse widths**.

The name "**basic waves**" for this **wavetable** is no coincidence. A jack and another rotary knob on the "pos" knob are a clear indication that morphing can be controlled and automated via control voltage with a modulator / controller.

I have it here for example connected to the **LFO 1** of **Fundamental** and if I turn up the control corresponding to the influence of the modulation, then movement comes into the **wavetable** and the more I turn up, the stronger the modulation.

With the **pos** controller, I determine which section of the **wavetable** is automatically processed. The wider it opens, the narrower the bandwidth of waveforms. You can see and hear that especially well at the end of the spectrum. If the slider is fully on the left, the entire **wavetable** will be traversed.

And as I said, this is just the range of **basic waveforms**. If one selects **another wavetable**, e.g. Digital1, resulting in the same modulation, very interesting sound spectrum. And as mentioned, there are 70 **wavetable**s, which I do not want to introduce here individually. If interested, you can watch a video on the website of **Blamsoft**.

But back to our **sine**, which I now mute once again. Let's see what other buttons there are. There is a **noise generator** on the one hand. As you might expect, this adds some **noise** to the sound.

The result is a signal consisting of the waveform displayed in the upper display plus **noise**. When fully open, I only get the **noise**. This is not visible in the built-in display, but the **Full Scope** shows the change in the signal very well.

The next 3 buttons belong together and they have to do with another peculiarity of the oscil-lator, namely, it can be operated up to **eight voices in unison**, that is not just an oscillator, but in extreme cases, there are eight oscillators that oscillate with each other.

I select the number of oscillators with "**density**". This oscillation can be absolutely synchron-ous, which only by a change in volume or by the fact that the signal is "denser" shows, or - and now it's exciting - with "**detune**" I can easily detune them and then get a movement that gets more and more extreme the farther the knob is turned on. With the control "**spread**" I distribute the unison **voices** over the entire stereo spectrum, the farther the knob is opened, the bigger the stereo effect.

And of course it will be even more interesting if we select another **wavetable** again. Eight os-cillators synchronized and slightly to strongly detuned and distributed over the entire stereo range, sound remarkably.

But let's get back to our sinus curve. In the lower part we have a second bigger display named "**harmonics**". Behind it hides a further form of synthesis, namely the **additive synthesis**.

Classic **additive synthesis** adds harmonic frequencies to a **sinuswave**, creating new wave-forms and new sound structures. We also know this principle of organs, which are changed by drawbars in the sound.

In the **XFX Wave** it is possible to **add frequencies to each waveform**, each **wavetable**. In the display, a frequency curve can even be "**painted**" and thus represents a possibility for real-time control of the sound.

Where above the center line, the addition of frequencies is displayed and below a phase shift of frequencies, the result of which is the extinction of individual frequencies.

And back to the **sinuswave**. The buttons in the middle section are self-explanatory, "**octave**" transposes the signal up to 3 octaves higher or lower, "**semi**" each 11 semitones higher or lower, and "**fine**" moves higher or lower within 0.5 semitones. Of course, "**level**" is the overall volume.
Below the level knob is the **frequency modulation** control knob, and next to it the associated socket, which can be connected to an external source, e.g. here again with an LFO. This is nothing special, but should be mentioned anyway.

However, it is particularly in the field of mods, i. e. **modifiers**. In initialized mode, all 3 are off, as you can see here. A click with the left mouse button in one of the associated display then reveals what is behind it, namely **25 different ways to modulate the sound**.

Among them are old acquaintances like **frequency modulation**, **phasedistortion** or **hard sync**, but also exotics like **mirror sync**. Again, I do not present every single option and refer back to the videos on the website of **Blamsoft**.

It is possible to determine how strongly the modulation is to affect the signal by means of a rotary control and here too there is a CV input with which the **modulation strength** and the **polarity** of the modulation can be regulated and automated by an external controller. Again I use the LFO as an example.

With the switch "**post**" you can define, whether the signal should be modulated before or after the **additive synthesis**. Post means the modulation of the added end signal.

And as mentioned, there are three identical mods, so I can select **three different sources for modulation** and also automate this again.

By combining all these possibilities, you can create infinitely many and very complex waveforms and sound images.

Granted, using other oscillators, modulators, waveshapers, wavefolders, etc., it's also possible to create appropriate waveforms and sounds, but not so easy, not so comfortable, not as compact as with this module.

Here I really have everything in one and it just sounds absolutely great.

Yes, I'm excited about this oscillator, I have to say that honestly. But I also always find **wavetables** very exciting.
Of course, it always depends on which music you make. For me, the **XFX Wave** is definitely **one of my absolute favorites**.

Thanks again, Andrew, for putting it at my disposal.

If you want to create extraordinary and versatile sounds and in a very simple way, I can highly recommend this oscillator.

Episode 3: Oscillators which correspond to Tides from Mutable Instruments

Hello world and welcome to "what does this knob do?"

As you can see, this is the 3rd episode and I already posted the 4th, why this? While I was working on my third episode, Francesco Mulassano of **Soundmit** asked in the official VCV facebook group, who is interested in testing a module that officially does not exist yet and makes a video about it. Guess who couldn't resist! **Soundmit** is a sponsor of the VCV facebook group and now also a sponsor of a new oszillator developed by **Autodafe**. If you haven't seen it yet, it is here in my youtube channel.

The third episode continues with oscillators.

I've been thinking for a long time in which order I should discuss the modules and I've decided to go plain alphabetical, so I think everyone can live with it. But each rule also has exceptions, so I take modules that match other modules as soon as the corresponding modules are reviewed. As you might guess, these are especially the modules originally designed by **Mutable Instruments** as hardware for eurorack. Their code is open source and therefore there are some versions of different developers for VCV-Rack.
Exactly with such a case, I will now start with **Cycles** of **Aepelzens Parasites**, which corresponds to both the **Tidal Modulator** of **Audible Instruments** as well as **Splash** of **Southpole** and **Southpole Parasites**. The original module is **Tides** by **Mutable Instruments**, which also comes with a variety of firmware in the hardware version. **Tides** is - like all **Mutable Instruments** modules - very versatile and not really unique to any function. **Tides** is an oscillator, but also an envelope generator and an LFO. Depending on what I use it for, the handling is of course

different. The firmware makes **Tides** a completely different module and these versions are also provided by the VCV-Rack plugins more or less. For this reason, it makes sense to **compare the modules** with each other, but I want to limit myself to the use as an oscillator in this episode, I discuss the other functions in the associated episodes.

Even in terms of optics, it is noticeable that both **Cycles** and **Tidal Modulator** are strongly oriented towards the design of the original module, while **Splash** has a slimmer appearance with smaller buttons. Since space in VCV-Rack is not necessarily the problem right now, and on the other hand, even the smallest buttons can grow with the zoom function, it is only a matter of taste which version you prefer.

Two of the modules are the same as the original module, with the added option of the alternative firmware **Sheep**, which is also the name used by **Audible Instruments**, while **Southpole** calls it **Lambs**.

The two versions of **Parasites** contain in addition to the original the variants **Two Bumps** and **Two Drunks**. **Cycles** even provides the optional quantizer and is thus very close to the parasite's firmware.

But let's take a look at the four modules with the original version. I have initialized all modules, so all modules have exactly the same setting and receive the same signals. Therefore, the sound should be the same at every module. We will hear and see that this is not necessarily so. Unfortunately, I do not have the hardware version as a reference, but for me it does not matter which module sounds more authentic. As so often it is a matter of taste which sound you like better.

But let's take a closer look at the modules. I would like to start with the inputs and outputs. Each module has **4 outputs**, which are outlined in gray or black and therefore can be quickly identified. For the moment I use the output **BI**, which of course stands for bipolar. This means that the signal goes into both, the positive and the negative range. Specifically, these are between -5 and +5 volts, we see this in scope in the blue curve. The lowest peak is -5, the top +5 volts. **UNI** stands for unipolar, so the signal is only in positive range, here from 0-8 volts. Very nice to see here in the red curve.

In addition, however, the signals of both outputs have a completely different waveform and also serve to **convert a bipolar into a unipolar signal**. To demonstrate this, I connected the **LFO 1**, which is currently on **UNI**, to the level input of the oscillator. We can see in the display the clearly unipolar blue original signal of the LFO and the resulting red curve, which is clearly bipolar. If I switch the LFO to bipolar and use the output in the oscillator, the result is unipolar. Needless to say that sounds completely different depending on which output is used, because of course it is also possible to use this signal to further modulate or combine the outputs or split or or, the possibilities are limited only by the imagination limits.

And then there are still 2 outputs, **high** is a unipolar signal, which reaches its highest point at the end of the attack phase and remains there until the cycle starts again or the curve is retriggered. If one uses this signal for modulation; here I have fed it into the **V/Oct** input; the signal is only modulated during the sustain phase, as shown in the red curve. The signal of the **low** output is also unipolar and reaches its highest point at the end of the decay / release phase and remains there until a new trigger pulse triggers the wave again. In the region of the control signal, a kind of gap arises in the modulated signal, that means, it changes cyclically both the frequency and the amplitude. Of course, all outputs can be used simultaneously and combined.

By contrast, the **inputs** are almost self-explanatory. **Trig** is for a trigger pulse that triggers the signal. In any case, a signal must be present if the module is to act as an envelope generator. **Clock** is intended for synchronization with an external clock. A trigger signal on the **Freeze** input keeps the signal at the value it had when it received the trigger until it is triggered again. **FM** is the usual CV input for **frequency modulation**, the influence of which can then be adjusted via the corresponding rotary control. The **level** input I have already briefly described above, but in principle all types of modulation signals / control voltages can of course be fed.

Finally, the three sliders **Shape**, **Slope** and **Smoothness** each have a CV input to connect an external modulation source. What these regulators all do, I'll come to that soon. Before that it is important to **deal first with the buttons**.

With the top one selects the mode, **3 modes** are available, attack / decay envelope (green led) attack / release envelope (red led) and cycle mode (led off or yellow). For the two envelopes a trigger pulse must be connected, otherwise nothing will happen. For operation as an oscillator, however, only the cycle mode is relevant.
To see what the individual controllers do, I initialize all the modules once again. We all see the same triangular wave. With the **frequency control** I can control the pitch within +/- 4 octaves. The transition is fluent and the input **V/Oct** can also be used as a CV input, which is why it is probably also called the module cycles freq. For all modules, the lowest possible frequency is 1 Hz, with the upper limit for the **Southpole** modules at 255 Hz and for the other two at 262 Hz. Thus, the **Southpole** modules do not quite reach the c4. Whether that bothers, everyone has to decide for themselves.

Let's go back to the middle position, as with most modules, this works with a right-click on the button. To test the **frequency modulation**, I connected the initialized LFO 1 to all modules, so they all receive the same signal. In the frequency **Analyzer** we see that this time the two **parasites** show a nearly same behavior and the two with the **original** firmware. The parasite modules are more stable in both low and high frequencies, and it is also a matter of personal preference if that is relevant.

I disable the **FM** again, so we can see better what happens to the signal. Also with the **Shape** controller the difference between **parasites** and **original** is not to be overlooked and also not to be ignored. The frequency spectrum also shows more high frequencies in the **parasites**. The same applies to the intermediate steps, only the **triangle** is the same for all. **Slope** control works the same for all modules. Its main task is in envelope mode, where it is responsible for the balance between rising and falling. More about that in episode 14. For the sound of the oscillator, the **Slope** provides a kind of "division" of the signal, with the knob left, the upper part of the signal left and the lower right, but this is inverted, which is not heard. If we look at the frequency spectrum, the signal is symmetrical. **Smoothness** is in the initialized setting in the middle position, a sure indication that the signal can be changed in both directions. In the neutral position we see a **triangle** wave. If we want to smooth it out, we turn the knob to the left and slowly the **triangle** becomes a **sine** and then almost a flatline. On the other hand, the signal becomes rougher, it gets rough edges, the curve is folded, the farther the knob is turned on, the more folds are added and thus more overtones. Generally speaking, **Smoothness** is removed or added. It seems that the **Smoothness** controller works the same way for all modules.

Let's see how the identified differences in the interaction of the individual components are noticeable. All knobs are all the way to the right side, it's hard to see in the scope, but there seems to be a difference between the **original** and the **parasites**, which is to be expected because the shape control adds more high frequencies. Also with the shape slider to the left we have the same result and as expected, all modules sound the same when the knob is in the middle. Also in the **Analyzer** we can see that the frequency response is the same for all settings as soon as the shape slider is in the middle position. And of course it gets really exciting when modulation comes into play.

Let's get to the alternative firmware **Sheep** or **Lambs**. While owners of the hardware have to decide which version they want to install or, of course, buy the module several times, the users of VCV-Rack have it much easier. A right mouse click on the module and a checkmark on the desired option is all it takes to make the **Tidal Modulator** of **Audible Instruments** or splash of **Southpole** a **wavetable** oscillator.
Both modules now also have this term in the header. The two versions of **parasites** do not have this option, but there are **Two Bumps** and **Two Drunks** modes, but more on that later.

If the module is used as **wavetable** oscillator, some elements of the control surface change. **BI** and **UNI** remain the same, the **LOW** output becomes **SUB** and here a **square** wave is output, the frequency of which is output 1 octave below the original signal set on the freq knob. At the **HIGH** output, a **1-bit** version of the original signal can be tapped, which of course has a beautiful lowfi aspect. For the further consideration of the functions of the controllers, I let all 4 outputs run in parallel and initialize the module again.

Here we see from top to bottom, the **BI** signal, the **UNI** signal, the **SUB** signal and the **1-bit** signal. With the top button you select the sound bank. Green led is called **additive harmony** (what this means we'll see soon), led off stands for **pulse width modulated waves** and red led for waveforms of the **WMAP** table of **Mutable Instruments'** oscillator **Braids**, in VCV-Rack the **Macro Oscillator** of **Audible Instruments** and **CornrowsX** of **Southpole**. With the button below, you can select the frequency band between low (green), mid (off) and high (red), with the fine tuning again via the freq button.

The **FM** button also has the same function, just like **Smoothness**. But things are different with **Shape** and **Slope**, which are now called **Row** and **Column**. The name actually says what these buttons do, it's about rows and columns. We know this name from tables and something else is not a **wavetable**. Rows are parallel to the x-axis and columns are parallel to the y-axis, so each waveform within the **wavetable** has exact coordinates that can be adjusted with these knobs. Of course, that does not mean that you really know the coordinates to select a particular waveform, because the transition in both directions is fluent. In principle one could say with the **wavetable PWM**, with row one selects a family and with column how far the selected family member in the kinship is apart. If we take a sinusoidal curve, we get that if both row and column are at the left end and the led's are off. Then row changes the signal e.g. to a pulse wave (other family), while column retains the Fundamental curve but adds additional frequencies and thereby "distorts" the signal. In the umpteenth generation, the original curve is still to be guessed, but actually the signal has become a **sawtooth**. In the case of the pulse wave, this means different **pulse widths**, so to speak a **PWM**.

When I select the bank **WMAP**, at the beginning (coordinate 0/0) is a **sine** wave. Column duplicates these and increases the frequency, but the **sine** wave is still clearly visible. Exciting here are above all the intermediate steps in which different degrees of intensity, that means, **sine** waves with the same frequencies but different amplitudes arise. On the one hand, the signal seems to be "thinned out", on the other hand, however, clearly more overtones can be heard, which give the whole a phased or metallic character. When turning the row regulator, it is noticeable that the individual rows have a stronger relationship here, after a few steps I already achieve the same result here as with the **PWM** at the end of the column. But this signal is also reproduced by the column control. And here we even hear chords in the intermediate areas!

We come to the green led and thus to the already mentioned **additive harmonics**. Here I have a quasi **sine** wave, if the row controller is in the center position and the column control on the left stop. If I turn now on the column controller high frequencies are added. With the row control I determine the number of overtones, starting with a very dull signal on the left, up to a quasi **sawtooth** on the right. Here, too, interesting, especially metallic and percussive tones can be represented.

Is there a difference between **Lambs** and **Sheep**? Except for the fact with the frequency controller, which I already mentioned at the beginning, not really. Both the graphical evaluation and the acoustic impression can not determine. All in all, **Lambs / Sheep** is an outstandingly sounding and extremely versatile oscillator, which, of course, is only expressed through external modulation via LFO or other sources.

Finally we come to the two **parasites** variants. **Two Bumps** enables **additive synthesis** by

adding **sine** waves of different frequencies and amplitudes. The outputs are configured as in **Lambs / Sheep**, except for uni, whose signal mixes the harmonics randomly and thus introduces an interesting random element into the sound.

A trigger on the **Freeze** input triggers a new mix, but in any case the **UNI** output must be used. A trigger on the **Clock** input causes a new distribution of the harmonics, but only for the moment when the trigger arrives. Then the curve is as before.

A trigger on the **TRIG** input causes a reset of the current curve, so it restarts with every trigger pulse. For a gate signal this is perceptible as a shift in phase.

The two switches also have another function in this mode. The top chooses the **kind of harmonics**. When the red led is illuminated, only octaves are added, green odd harmonics, and led off, all harmonics are included.

To hear and see this, we first need to press the **Smoothness** knob, because if it is in the middle position **Shape** and **Slope** have no function. To get an optically visible signal, I turn freq,

Shape and **Slope** to the left stop and move **Smoothness** to the left until I almost have a sinusoid. The signal is barely audible, but good to see. If I move shape or slope slowly to the right, whole tones, mainly octaves, are added. With the smooth control, I choose how the timbre is affected by the center frequencies selected with **Shape** and **Slope**. So I can select 2 dominant mid frequencies, whose amplitude is thus increased, the second frequency is always less emphasized than the first. In the initialized state, I have a c5, according to **Hot Tuna** of **Nysthi**, this does not change if I change the nature of the harmonics, by pressing the appropriate button, even if it sounds like that.

If I turn red and move the freq slider to the left stop, it becomes a c1, as one might expect. I move **Shape** and **Slope** also to the left stop nothing happens, no surprise. However, if I move **Smooth** towards the left stroke, less and less the adjacent frequencies of the main frequencies set with **Shape** and **Slope** are emphasized, one could also say that these are filtered out or subtracted. The curve changes until it becomes a pure **sine**, in which only the main frequency sounds, which is acoustically barely perceptible at c1 and finally the amplitude of this frequency is lowered so that only a flat line remains. So we have no signal, so bringing an emphasis through the controls shape and slope does nothing. Amplified silence remains silence. That's the sound of silence ;-)

The situation is different with the **sine** wave, as we saw earlier. If we move the shape slider towards the center we hear very well the intervals and also **Hot Tuna** shows us this, but if we reach the next octave, shows **Hot Tuna** again a c1 and we hear it that way, because so the 1st overtone of the fundamental frequency or the 2nd harmonic was added. The same is true if more octaves (i. e. the other harmonics) are added. Unlike drawbar organs, I have here but also the intermediate frequencies and can also emphasize these to get a different sound.

In short, with **Shape** and **Slope**, I select my 2 emphasized main frequencies, and with **Smooth**, I control how much the adjacent frequencies are affected. On the left of the middle position, the main frequencies and their neighbors are emphasized. To the right of the middle position, first the main frequencies and then the neighbors are faded out, so that only a few frequen-cies are heard at the right stop. The basic function of the controls, **Shape**, **Splope** and **Smooth**, of course, remain the same, but the other two modes offer completely different sound spectra.

Let's take odd harmonics mode and leave the smooth slider in the same position as before. If the **Shape** and **Slope** are on the left stop, then we have a pure **sine** wave again, I move the **Shape** slider to the right, I select again frequencies to emphasize, in the octaves I have a pure **sine** wave again or if in both **Shape** and **Slope** an octave is selected, the frequencies cancel each other out and I have a flatline. But between the octaves it becomes very interesting. The waveform becomes chaotic and contains all sorts of frequencies and timbres. If I move the **Smooth** controller further to the right, I no longer have a pure **sine** wave, so there are also different frequencies in the octaves, so that there are always partial cancellations, which results in interesting sound spectra.

If no led is lit, we have the entire spectrum of harmonics. If we start from the pure **sine** wave again, then we get natural tones during the rotation of the **Shape** controller ala "also sprach Zarathustra", whereby these are more strongly separated with the **Shape** controller, the transition is more fluent with the **Slope** controller. By combining the two controllers, very clean intervals can be displayed. If we change the output signal with **Smooth**, this consists of numerous different frequencies, as clearly seen on the spectrum. Very nice here you can see the two selected frequencies and the influence of the **Smooth** controller. If both frequencies are still strongly separated at the beginning, this loses more and more as the bandwidth of the affected neighboring frequencies increases. Until the middle position no separation is present. If we turn the knob further to the left, you can clearly see the hole that is forming where the frequencies are fading out more and more.

One last option is still open, namely the second button. This adjusts the quality, that means, the accuracy of the **sine** waves. Red led is the highest quality, so as you can see here is a beautiful round curve. In contrast, the green led with the lowest accuracy, which is noticeable in spikes in the curve. If the led does not light, we have medium quality, so only slight kinks in the curve. Of course, all three variants also bring differences in sound, the more angular the curve, the more it sounds like a **triangle** wave, that means, sharper or colder.

But are there any differences between the two modules? The most striking thing is certainly the different tuning of the two oscillators. So the frequency in the middle position of the freq controller is cycle at 523 Hz c5, at two bumps it is 509 Hz and thus between b4 and c5. However, this seems to be a known problem according to github, which resulted in converting from VCV-Rack 0.5 to 0.6.

Now let's look at the version "**Two Drunks**". At **Southpole Parasites**, this is also the name of

the module and the individual controls describe how it works. With **Cycles** one must again remember, which function the regulators have in this version. **Two Drunks** is also called dual random walk simulator, roughly speaking, **2 probability generators**.

The two generators are each **randomly clocked with an independent clock signal**. Channel 1 represents a trembling or fluttering whose intensity can be adjusted. Channel 2 simulates a random coin toss, whereby the probability of the outcome can be influenced. The low and high outputs output these clocks, respectively. They are **both derived from the same master clock**, with the **frequency knob** controlling the frequency of this master clock. Each step of the clock generates a voltage at one of the **unipolar** (channel 1) or **bipolar** (channel 2) outputs. The course between the steps can be selected with the **Shape / Interpolation** knob. The transition between the waveforms is fluid, so that not only classic shapes, such as **square** wave, **triangle** or **sine** are possible, but many mixed forms.

If I move the controller from left to right, you can see clearly that the signals of both outputs change in opposite directions, I have the **square** wave at channel 1 at the left stop, at channel 2 right. The reason for this is that both channels share the same parameters. In the middle position, the shares are the same, that means 50/50. The ratio changes by turning the knob to the left or right. This applies to all buttons.

The **Smoothness** slider sets the **maximum size of a step**, but within these parameters, step sizes are random.

The **Slope** knob adjusts the **behavior of the two clocks**. For **channel 1** it acts as a **random delay** of the main clock, the knob is on the far right, this equates to zero delay, and channel 1 follows the main clock. Turning the knob to the left increases the random delay and the clock slows down and begins to flutter.

For **channel 2**, it controls the **likelihood** that a tick of the main clock produces a tick of channel 2. Completely left is the probability 1 and the signal follows exactly the main clock. The farther it is turned to the right, the more ticks are deleted. In the middle position the probability is 1/2; rightmost, the probability is very small (but not zero) and almost all ticks are left out.

As usual, the **FM** knob is an attenuator for the CV input of the same name, but it also has a second function, which refers to the two random clocks. For **channel 1**, the **random distribution** of on / off states is set, for **channel 2** the **pulse width of the clock signals**.

The module has three modes, which are set with the mode switch. When the led is off, it runs **cyclically**. The main clock keeps ticking and stops only when a gate is received in the freeze input. When the led is green, the module is in **trigger mode**. A new cycle is started only when a trigger is received in the trig input. Also, this trigger is triggered at the clock outputs of channel 1/2 delayed / random. If the led is red, it is in **gate mode**. The master clock is cyclically switched as long as the input value at the trig input is high.

The **range** button adjusts the frequency range of the oscillator as usual: low (green led), medium (led off), audio rate (red led).

Two Drunks is also a very interesting **noise generator** if you set the range to audio frequency and boost the frequency to maximum.

Phew, these are the VCV modules that correspond to **Tides** from **Mutable Instruments** and all of its variants. I think you realized that these are great and versatile modules and of course I only scratched the surface here. And not to forget that there are other uses, such as LFO or envelope generator, which I have not yet received here. For this reason, these modules will return in other videos. I hope I made you want to play and experiment with these fantastic modules. Thank you for your interest in my videos and off course, many thanks to Andrew Belt and all the developers without whom all this wouldn't be possible.

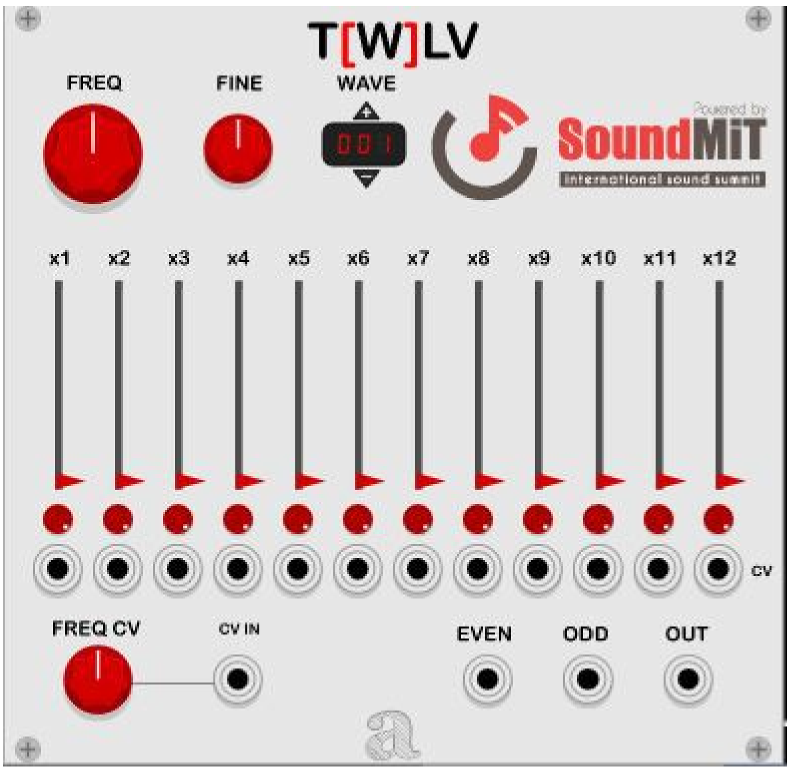
Episode 4: T [W] LV Harmonic Oscillator from Soundmit/Autodafe

Hello world and welcome to "what does this knob do?"

Yes, I decided to do my videos from now on in german and english, because it's much easier to record a **voice** than doing subtitles. So I hope you understand my franconian english

While I'm still working on my third episode, which is getting a bit longer and more detailed, Francesco Mulassano of **Soundmit** asked in the official VCV facebook group, who is interested in testing a module that officially does not exist yet and makes a video about it. Sure, that I have reported there immediately. **Soundmit** is a sponsor of the VCV facebook group and now also a sponsor of a new module developed by **Autodafe**. Below the video you will find links to the various pages of **Soundmit** and **Autodafe**.

The module is the **T [W] LV Harmonic Oscillator**. A VCO with **12 selectable harmonics**, each a multiple of the fundamental frequency, the level of each harmonic can be adjusted and even controlled by CV. The oscillator provides the fundamental waveforms, with the rectangle represented with different **pulse widths**.



Let's take a closer look. Here I have my usual test setup, supplemented by the **Multiscope** from **trowaSoft**, because I would like to present the 3 outputs of the module like in comparison.

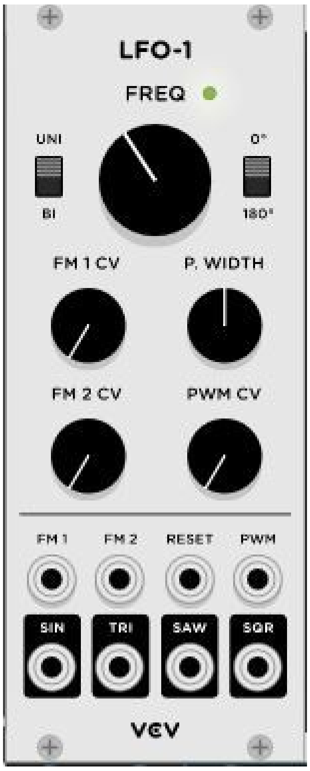
I also use **Nysthi**'s **Surveillance** to adjust the output level of the oscillator to my reference signal, because this is (for a good reason) much lower, as you can see in the red curve here. So now we have the same level on both signals.

Once the module is initialized, you will not hear anything, because there is no signal yet. As you can see, however, I get the same **sine** wave as with my reference signal when the fundamental frequency is fully added. I can now add each single harmonic alone and see exactly the multiples of the fundamental frequency, but they all have the same amplitude. I set the fundamental frequency via the **frequency controller** in the range of 26 to 2600 hertz. It becomes interesting, when you **add the harmonics** to the fundamental frequency. On the one hand, we can clearly see that the amplitude is increasing, on the other hand we see here in the **Multiscope** that different signals are outputed at the three **outputs out**, **even** and **odd**. As expected, all frequencies are at the output out, the even at the output even and the odd at the output odd. Here's very nice to see that at the odd output currently only the fundamental tone is applied. If I add an odd harmonic, this signal also changes, and of course, the signal on the out output, while the even signal remains the same. In addition, but also increases the amplitude and thus the volume of the signal. If I add more harmonics, the amplitude will continue to increase and the signals at the corresponding outputs will also change. In the **Analyzer**s you can see the added frequencies very well. By changing the amount of harmonics, I get completely different sound patterns, exciting again here is the randomize function of VCV-Rack.

For further consideration, I initialize the module again and activate only the fundamental frequency. As already mentioned, not only do we have **sine** waves available, as is customary in **additive synthesis**, but **all basic waveforms**. Clicking on the plus of the wave display brings us to the next waveform and the minus leads us back. Personally, I find that a bit awkward. Here I would have liked a CV controllable rotary switch to be able to morph through the waveforms, but that does not mean that there is a lack of control options.

At first glance, it looks like a **V/Oct** input is missing and so I have asked, the input signal goes always in the **frequency CV in**. In the center position, the output signal is not affected by the input signal. If you turn the control all the way to the right, the input signal can pass fully, if you turn the control to the left, the input signal is inverted. In addition, the pitch changes as well. An exciting idea, which certainly needs some getting used to.

Let's come to the absolute highlight of this oscillator. **Each of the 12 harmonics has its own**



CV input, so I can control each one with a control voltage, wow! To clarify what this means, I have selected a **square** wave. I influence this with my **LFO 1** from **Fundamental.** Significantly, the increase in the level is audible. With the sliders, I set the minimum amplitude with the knob the strength of the modulation. If I add more harmonics, they will be modulated as soon as I have a corresponding signal. So right now this is just the keynote. Now I send the same control voltage to the overtones. It can be clearly seen in the **Analyzer** that the frequencies remain the same but the amplitudes are modulated. Which of course changes the sound. And of course I can also use different modulation sources. Here, for example, an LFO on a **Sssh ...** Sample & Hold by **Southpole**.

Each harmonic can practically have a different modulation source, even sequencers with different timing are possible. And of course we can combine that with an input signal, e.g. a sequence. As a result, very lively sound structures are possible, which almost come close to **wavetable** synthesis. If you could still modulate the waveform by CV ... But who knows, maybe there will be a version 2.

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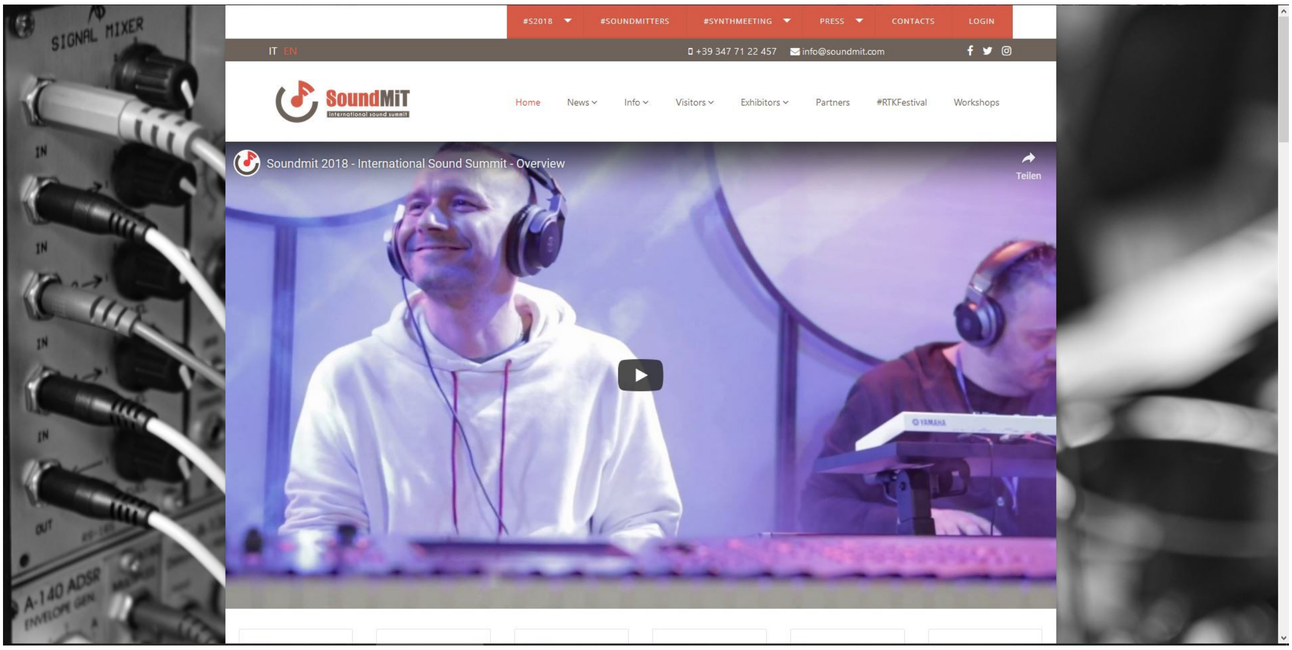
But even the missing of this possibility does not diminish the extremely positive overall impression. I really like the sound and the possibilities of changing the sound, handling and design. It's fun to play with this oscillator. I hope he is available soon and I can only recommend him.



Episode 5: Oscillators – Developers with A (with exceptions)

Hello world and welcome to "what does this knob do".

After a long gap between episodes 2 and 3, and the 4th episode coming before the 3rd was finished, I'm happy to give you already the 5th episode. We are still on the subject of oscillators, and since I go alphabetical, todays subject are the remaining modules or developers with A. But there are also exceptions today, so we have also **21kHz** and also a module of **Southpole**, which corresponds to an module of **Audible Instruments**.

The **21kHz Palm Loop** is also the module I want to start with. This module is still pretty new and it is a solid oscillator with different waveforms and a few special features. Initialized it is tuned to c4 and can then be adjusted in the range of +/- 4 octaves. With **course** I have the possibility of tuning in +/- 7 semitone steps and with **fine** in +/- 1 semitone steps. So far nothing surprising. But that changes when we look at the **waveforms**, while the shapes are standard. In the top row we have **sawtooth** and **sine**, in the bottom row **square** wave, **triangle** and again **sine**, but these three are **tuned one octave lower**. Also interesting are the three inputs, **reset** triggers the wave at each impulse, and **ex** and **lin** offer different possibilities of **frequency modulation** from gentle to crass. This can definitely start a lot.

I continue with another newer module, the **Slic** of **Animated Circuits**. It's part of the **Animated Circuits** free welcome package and a simpler version of the commercial module (meanwhile free) **Cosmic** with a single oscillator and 2 waveforms. **Slic** is therefore also the abbreviation of slice of cosmic. This module is a **phasedisortion** oscillator that generates its waveforms by modulating a **sine** wave at different speeds and distorting the original wave.
The fundamental frequency of the oscillator is 261.63Hz (c4). The frequency can be modulated by a signal at the

FM input. The intensity of the modulation is adjusted with the bipolar **FM** control. Two waveforms are available, **sawtooth** and **reso2** (from the available waveforms of **Cosmic**). The **strength** of the phase distortion is set with the **big blue button**. When turning from full left to right, the waveform changes from a **sine** wave to a **sawtooth** or the reso2 wave with a type (resonant) filter sweep effect. The **phasedistortion** can be controlled by CV and the **modulation** is adjusted with the **small blue button**. This button is an attenuator that inverts the input signal when it is completely left. Even with this slimmed-down version you can create very interesting and lively sounds, so it's definitely worth taking a look at the commercial modules of **Animated Circuits**.

Sometimes you do not need an extremely versatile oscillator, just a reliable and simple module. This is definitely true of many of the oscillators presented today, including the two from **AS**. Although **Tiny Sawish** sounds like a simple **sawtooth**, but it is anything but that. The output signal has more of something from a pointy cap and this can be splitted with the mod controller, creating a resonance that reminds of smacking and that can be controlled per CV. In contrast, the **Tiny Sine** is a bit of boring, a **sine** wave signal, which can be regulated in pitch, is all he offers. But like I said, sometimes you don't need more. And besides, connect the **Tiny Sawish** to the **Tiny Sine** and you have wonderful **FM** sounds. I really like the two tinys.

The **Autinn** modules are also quiet simple, but still offer interesting waveforms. **Oxcart** and **Saw** I see as different sides of the same coin, both are **modified saw teeth**. **Oxcart** sounds sharper, as you can see in the graphics and **Saw** rounder. The signals are nearly inverted but not exact. **Sjip** is the expert for **round waves**, if a sinus is too **square**, you should try this module. **Square**, I can not call with the best of intentions as a **square** wave, perhaps as a rectangle, which was heavily beaten, so it looks more like a pointy cap. Nevertheless, this is an interesting waveform. Despite rather spartan equipment, **Jette** is a very versatile sound generator. Plugged in the initialized state, it shows a quasi **square** wave and sounds like that. Using the large white switch, you select different waveforms, a clean **triangle** and a distorted **sawtooth**.The knob is responsible, as with the other modules, for adjusting the pitch. With the **sliders** one **fades out harmonics** of the signal and thus alters the character of the sound, the signal becomes thinner and thinner and finally a flatline. If I turn down everything, except the root, I have a **sine** wave with every waveform. If I add harmonics again, you can modulate the most different waves. Also with this module the randomize function always brings amazing results.

We've already heard that there are numerous modules from **Mutable Instruments** for VCV-Rack. One of these modules is **Braids**, which is available in VCV-Rack in two versions, as **Macro Oscillator** from **Audible Instruments** and as **CornrowsX** from **Southpole**. Incidentally, the successor of **Braids**, who calls himself **Plaits**, is already available for VCV-Rack, here he

is called **Macro Oscillator 2** by **Audible Instruments** and is part of the preview package, which can be purchased for $ 20 (meanwhile free).
But let's just stay with **Braids**. As usual from **Mutable Instruments**, we have here again a very versatile module, which wants to be properly researched. **Omri Cohen** has numerous videos in his **youtube channel**, which I highly recommend. You should also download the **cheatsheet** from braids illustrated, because you just can not remember anything. Depending on which program / which sound you select, the controls timbre and color have different characteris-tics.

Completely unspectacular are the output **out** and the input **V/Oct**. A trigger signal at the **trig** input resets the oscillator phase. Physical or percussive models like pluk or kick need this trigger signal to play the sound properly. The trigger input can also be used to trigger the in-ternal AD envelope.

Control voltage (CV) can also be used to control **frequency modulation FM** and the **timbre** and **color** buttons. With **CornrowsX**, the length of a trigger delay can also be adjusted with **tdly**. But before we explore the differences between the two versions, let's first look at the similarities. In the initialized state we can see in the display the synthesis model CSAW, an emulation of a **sawtooth** of the **Bladerunner** synthesizer Yamaha CS-80. Also the small un-cleanness in the waveform was also emulated to reproduce the sound character as authen-tically as possible. Here you can really see the love of detail.

I do not go through all the synthesis models here, these are explained very well in the men-tioned **cheatsheet**. Dividing the models into groups, we have **classic analog waveforms**, **digi-tal synthesis**, **physical simulations**, **percussion**, **wavetable**, and **noise**. Use the edit or shape slider to select the desired model. The abbreviations in the display are self-explanatory af-ter some time, but a look into the **cheatsheet** is also always helpful.

With **fine** and **course** you can adjust the pitch, at **fine** +/- 1 semitone, at **course** +/- 2 octaves. **CornrowsX** also has an **extra octave slider** that does not have the smooth transition of the course knob. If a signal is set on the **FM** input, the **frequency modulation** can be set with the **FM** knob. A very interesting, but well hidden feature has to be activated by a right click on the module and a tick at **meta**, then I can **switch the synthesis model**, i. e. the sound, with the **FM** button via the CV signal on the **FM** input. The farther the **FM** knob is moved from the center position, the more models are traversed. A very exciting **variant of wavetable synthesis**. In this mode, however, it is no longer possible to manually select the synthesis model via the slider.

Also with right click one can select further options. **Drft** simulates a non-tuning stable oscillator, **sign** applies waveform errors to the output signal. **CornrowsX** also offers the **flat** option, which applies detuning in the lower and higher frequencies to represent some of the tuning errors of VCOs. Through all three options, the modules sound **more analogous**, which is indeed often desired. Again, we can feel the attention to detail by the developers.

As already mentioned, the **timbre** and **color** controls are different for each synthesis model. In principle, however, it can be said that **timbre** determines the main development and movement of the timbre and **color** controls a second dimension of the sound. In between is the modulation attenuator, which controls the amount and polarity of the modulation applied by the timbre CV to the timbre parameter. Incidentally, in the context menu we can also opt for operation as a **low CPU**, currently an important criterion in VCV-Rack.

Let's go to the other options of **Braids**, which only **CornrowsX** offers. First, we have here a **built-in envelope generator**, which also does not have to be operated as in the original menu, but has its own controller. Great! The same applies to the **built-in quantizer**, both for the **root** and for the **tone scales** are knobs available and the range of scales is very versatile with 48 possibilities. With **rang** we determine the area which the course regulator affects, particularly interesting here is the mode free and the possibility to operate the oscillator as LFO.

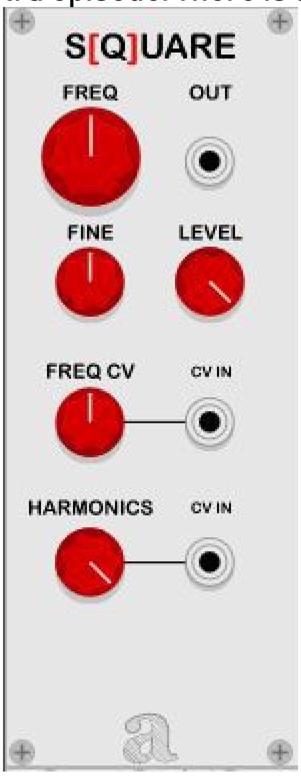
All three knobs **timbre**, **modulation**, and **color** are equipped with an attenuator that controls the **amount of modulation from the internal AD envelope generator** to these parameters. The amplitude can also be influenced, with its value, between 0 and 15, being set via the inner VCA, which in turn can be accessed via the context menu. If all these settings are zero, the **trig** input will operate as **sync / reset** input.

Finally, the **bit rate** can be controlled from 2 to 16 bits, and with **rate** the sample rate of the signal to generate beautiful lo-fi sounds. But also in the context menu hide a few more options. **Auto** tracks changes in the **V/Oct** frequency input greater than a semitone and generates a new trigger on each of these triggers. This allows e.g. control by a note sequencer that does not provide a gate signal.

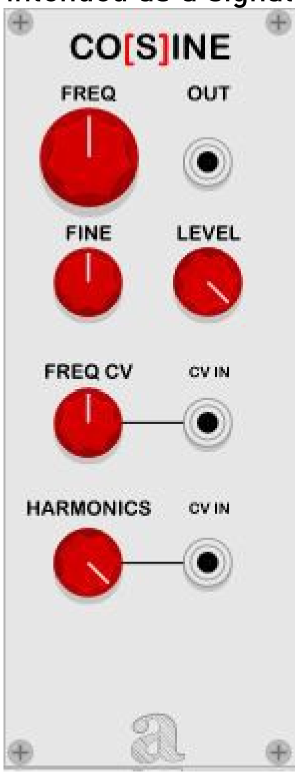
One last option lurks in the context menu of **CornrowsX** and this is called **paques**. I have found no documentation about it, so I can only assess its function subjectively. It is a noisy, folded pulse wave whose cycle reminds me of a morse signal. I have not found a way to change the display number 49, but all buttons work normally. Maybe one of you knows more about it and shares this knowledge with us here.

No matter which of the two versions one chooses, **Braids** is a great module with a lot of potential (see also the mentioned videos by **Omri Cohen**).

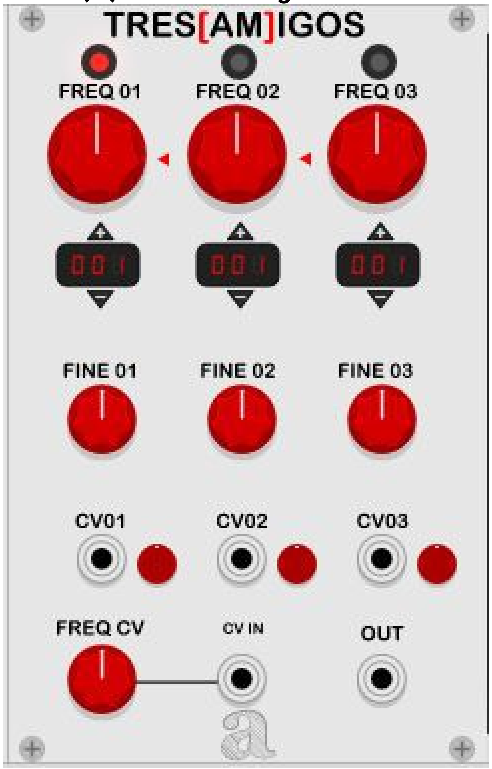
The conclusion of today's episode are three oscillators from **Autodafe**, actually it should be four, but the **T (W) LV**, **Autodafe** has created for **Soundmit**, I have already presented in an extra episode. There is also an **FM** module and a snare module on the website of **Autodafe**. The freely available drum modules I introduce elsewhere. Now for **S(Q)UARE**. As the name suggests, this is a **pulse wave oscillator**. But it is not a "normal" pulse wave, but a mix between pulse and **sawtooth** and there are also some small convolutions in the signal, which makes it very much alive. The frequency range is +/- 2 octaves and can be adjusted via the corresponding control, or via the fine control +/- 1.5 semitones. As with the **T (W) LV,** the CV input is intended as a signal input. With the **harmonics control** we can reduce the number of overtones and smoothen the signal. I can also control this via CV, which can generate bell-like sounds.



The structure of **CO(S)INE** is exactly like **S(Q)UARE**, but the Fundamental waveform is of course completely different. Unfortunately, anyone who has expected only a phase-shifted **sine** by name will have to disappoint. The signal produced here consists of **sine** waves with different amplitudes and frequencies, the result sounds **like a bandpass filter**. All controls and inputs work as in **S(Q)UARE**, although here too the **modulation of the harmonics via CV** is very interesting.



TRES(A)MIGOS delights us in the initialized mode with a classic **sine** and the already men-



tioned frequency settings. Again, the input signal goes through the CV in and also affects the root. But since we have three friends, three oscillators, we have everything but freq CV and CV in three times. Above each freq button is a switch that activates the respective oscillator and below a display with + and - to select the waveform. Again, we know this from the **T (W) LV** , so here too, I would want a stepless transition of the waveforms and the ability to control them via CV.

At this point, however, it is over with the similarities and **TRES(A)MIGOS** show what is in them. Namely, we do not have three identical individually controllable oscillators, but **3 oscillators that influence each other**. If oscillator 1 is disabled, the others will not work either. If I activate oscillator 1, I can activate oscillator 2 alone or together with oscillator 3, but not 1 and 3. The signals of the oscillators add up, which already allows many interesting sounds.

But of course there is more, because the oscillators are connected to each other, oscillator 2 modulates oscillator 1 and oscillator 3 modulates its amigo 2, which modulates the 1 again. So here we have a **complex amplitude modulation**. And not only do we have **sine** waves that modulate, but **8 waveforms per oscillator**, and that's not all, each oscillator has a controllable CV input. Easy to use, many options, I think, I will spend more time with the 3 amigos.

And that was it for the fifth episode. I hope you enjoyed it and you want to play and experiment with VCV-Rack and its great modules. I am pleased about your feedback. And not to forget, many thanks to Andrew Belt and all the module developers, without whom all this would not be possible. As always you will find the most important links here under the video. That's it from me, take care

Episode 6: more Oscillators – Developers with B

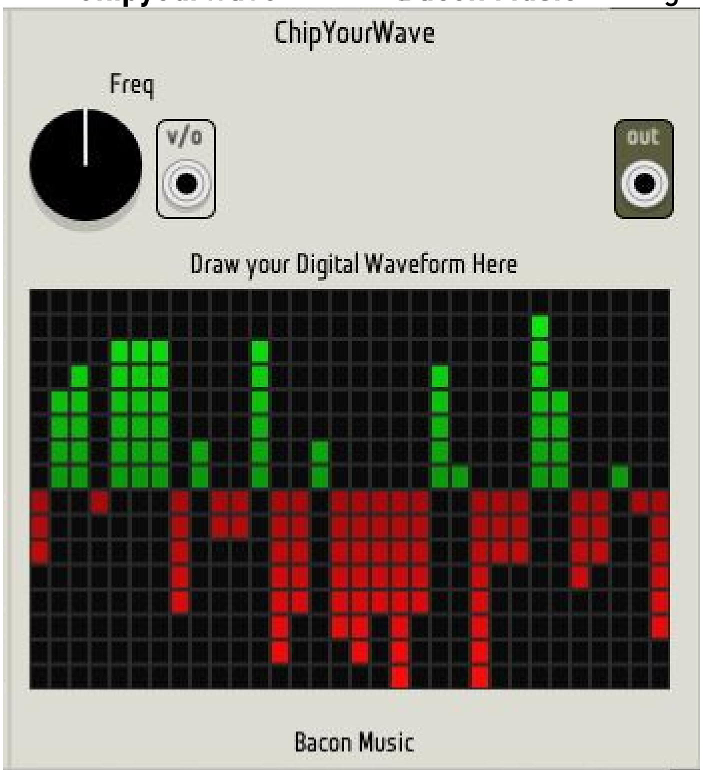
Hello world and welcome to "what does this knob do?".

Let's see if I can finally achieve my goal of releasing a new episode every week. Every week numerous new releases, every week a new very cool patch contest and in between further actions, keep me really busy. And actually, I also want to make music. And then there is also something like the "real" life. But the intent is and I work to realize it. Today with the 6th episode and more oscillators. The letter B brings back some very interesting modules.

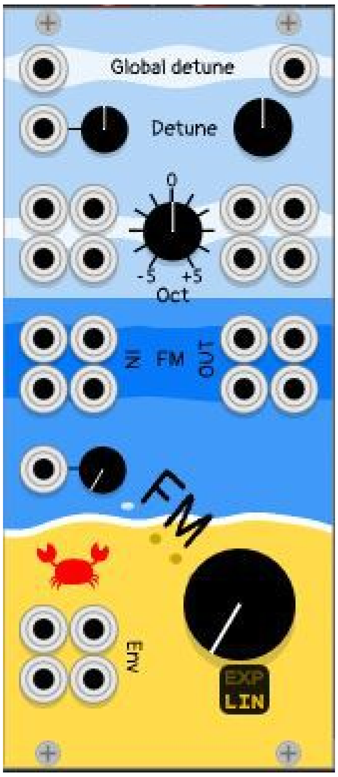
Let's jump right in the middle with **Bacon Music**. Even the name makes you hungry and the modules are also real delicacies. **Karplus Strong Poly** is, as the name implies, a module with

Karplus Strong synthesis. On this interesting form of synthesis alone I could make a whole episode and on the internet you will also find many contributions, here only so much, it is a synthesis form, which is mainly used for the **simulation of stringed instruments**. And just as a string does not sound when it is not vibrated, this oscillator also produces no sound when there is no signal at the trigger input. The input signal goes into the **freq** input and can be tuned in the range of 1 Hz to 3160 Hz. Due to the extreme overtones of the different waveforms, the pitch determination is sometimes quite difficult. In the middle position a c3 is displayed anyway. The waveforms are selected using the **packet** slider, **Random** (a sort of folded **sine** wave whose amplitude and convolution are highest at the moment of the trigger and then decreasing, it changes phase and offset randomly, simulates string vibration, and sounds quite realistic), with the filter a you can make several strings out of one string. **Square** is more like a lo-fi **triangle** wave in the initialized state, but becomes a rectangle when knob a is turned all the way to the right, completely to the left the **triangle**'s amplitude increases and it becomes cleaner. **Saw** starts as a mixture of **sawtooth** and **sine** wave, turning into a cyclically phase-changing **sawtooth** on the far right and a quasi-inverted signal on the left, but with a higher amplitude. **Noisysaw** adds even more **noise** to the already slightly rusty saw signal, but otherwise the waveforms are identical. The additional **noise** again gives the impression of a vibrating string. **Sin** is a beautiful round sinus, which is quieter and slightly resonant right and left louder and drier. **Sinchirp** is a lo-fi **sine** with resonance, here you can clearly see that the knob a is turned to the right more high frequencies and turned to the left more deep, so it should probably be a **band-stop filter**, since in the middle position both frequency ranges reduced are. Currently, only this one filter type is implemented, but 2 more seem to be provided and you can probably select this with the currently missing rotary knob. Both packet and filter A can be controlled by CV. With the also CV controllable attenua-tor control regulates the release time, the further the slider is turned to the right, the shorter it is. Although the oscillator is **not finished yet**, you can now conjure up very interesting sounds with it. From guitar to piano and harpsichord, to completely synthetic sounds and bells, everything is possible. It is important to know, however, that a change of sound does not happen immediately but fluently, which also allows great effects. To **try out** what is pos-sible, I recommend again the randomize function.

Also **Chipyourwave** is from **Bacon Music** and again we have an exceptional oscillator. Paul Walker, the developer of these modules obviously has a soft spot for chip sounds. The theme is here, **paint your own waveform**. And indeed what we paint in the display is the waveform shown in the scope. Unfortunately, the randomize function only works on the freq slider and you want to practice painting, but definitely a cool idea and the result always sounds like chip wave.

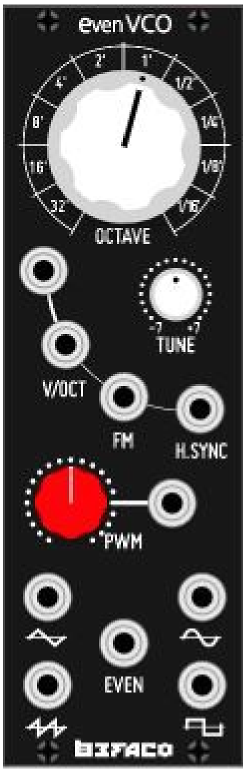


Our next candidate is exotic in every way. It is the **Bargkass FM Operator**, which does not exist in the Plugin Manager and which **has to be compiled** by itself. Who does not know how to do that, I recommend the corresponding videos by **Leonardo Laguna Ruiz**. As always, you will find the link under the video. I have no idea if it's's done yet, but he's definitely a fun playground right now. Of course, the first thing that strikes the eye is the look, that may not be for everyone, but in case of doubt, just change the svg file with inkscape. I like it that way and somehow the design fits in too. But what is it and what does it? **FM Operator** actually



says so. It's about **frequency modulation**. He has **4 inputs** and **4 outputs**. In principle, I could therefore send in 4 different signals and distribute them to 4 sources. But of course it is also possible to pick up an input signal in the associated output and **loop it in again** to switch the inputs in series or in parallel. And with trick 17 it is also possible to **create a feedback loop**. Of course, the output signals can also be used for CV control of the envelopes and the **FM**, also for pitch control or for the detune. Or you control these parameters in a classic way with external controllers, such as LFO. This oscillator can be stinking boring, if you like, but it gives the creative a variety of sounds that you would never have expected from the exterior alone. **A true hidden champion**.

Let's get to a module that does not really need an introduction, the **Even VCO** from **Befaco**. All VCV modules from **Befaco** are freely available and they are also available as hardware for the eurorack. The **Even VCO** can certainly be described as a classic among the VCV modules, it was also one of the first modules. At first glance, it looks pretty unspectacular. Its range goes from c0 to c9 and is set via the large **octave** knob. The fine tuning is done via the **tune control** and depends on the octave. Between c0 and c7 the accuracy is 1 cent, over c8 it is 4 cents and 13 cents above c9. The **pitch** can also be controlled by CV, but there is no setting option. **V/Oct** is the usual input and also **FM** and **hardsync** can be controlled by CV, but also there is no internal adjustment possibility. So far, the **Even VCO** is more spartan. The **pulse width** of the **square** wave can be controlled by CV and there is also the **PWM** controller. Other waveforms are **triangle**, **sawtooth**, **sine**, and **even**. The latter waveform is the peculiarity and the namesake of the oscillator, a waveform in

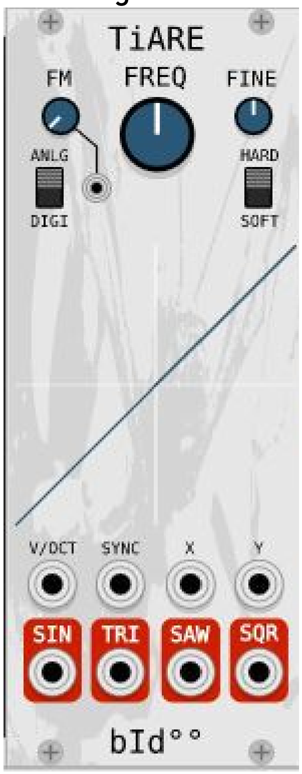


which the **even overtones are emphasized**. This sounds rounder than a pulse wave, but sharper than a **triangle** wave. But what makes the **Even VCO** a classic is simply its sound. It sounds **warm and analog**, in the oscilloscope we also see that the waveforms are not perfect, just as it should be with an **analog** oscillator. As mentioned in the last episode, sometimes you do not need an extremely versatile oscillator, but a reliable and good sounding module, like the **Even VCO**.

While I'm on this episode, a new oscillator has just been released by a developer I discussed

in the last episode. The module is the **Tachyon Entangler** from **21kHz**. The **Tachyon Entangler** is an edge-smoothed chaotic sync oscillator and basically consists of two **Palm-Loop** oscillators (see last episode) with chaos that can be cross-synchronized for likelihood. By default, oscillator A is the master.

The controls **octave**, **coarse** and **fine** are master pitch controls for both oscillators and work like the palm loop. The **ratio B** knob changes the frequency ratio of oscillator B with respect to the master pitch. If I turn this knob, you can hear the classic sync sound. Here I would have liked a CV control. Which does not mean that it would lack it. Each oscillator, like the palm loop, has **exponential** and **linear FM** inputs and associated attenuators. In addition, both have **chaos** and **sync** controls. The **chaos** button introduces a **random element** into the oscillation, making the signal noisy. The **sync** knob influences the probability of one oscillator being synchronized with the other. Full counterclockwise is no synchronization and fully clockwise is hard sync; intermediate settings create disturbing and stuttering effects. The chaos and sync settings also have modulation inputs and associated attenuators.

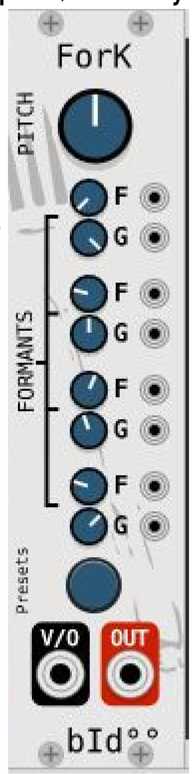


Each oscillator also has a **V/Oct** input and a **reset** input, which reboots the waveform each time a trigger is received. Finally, each oscillator has two **outputs**, **saw** and **square**. As with palm loop, the **square** output is tuned **one octave lower**. With the **square** signal, you can use the **ratio** knob to create harmonious levels, if you could control it via CV ... The **Tachyon Entangler** has opened certainly a lot of potential, but it certainly takes some time to develop that.

The next 3 **Bidoo** modules are also anything but standard. Although **TiARE** and **clACos** are based on the **VCO-1** from **Fundamental**, they are far from pure copies. Common features are the pitch of **9 octaves**, the

fine tuning in 2 semitone steps, the CV controllable **frequency modulation** and the ability to switch between **analog** (as a warmer) and **digital waveform** and hard / soft **sync** (softer). The external synchronization is possible via the corresponding input and of course there is also a **V/Oct** input. Like the **VCO-1**, the waveforms are **sine**, **triangle**, **sawtooth**, and **rectangle**. The special feature of the **Bidoo** modules are the **diagonal lines**. Because these are not just a design feature. Both oscillators are **phasedistortion** oscillators, wherein in the starting position, the phase distortion is 0 and thus the signal is linear, that is unchanged to the **VCO-1**. The blue line passes through a coordinate system with the zero point of both axes in the middle. So we have 4 sectors on each line, and I can **click and move that line**, keeping the start and end points the same and creating a kink in the line at the touch point. In the oscilloscope you can see very nicely how the wave changes and this can be automated or controlled via the CV inputs x and y. As a result, complex, changing waveforms are possible and thus extraordinary sounds.

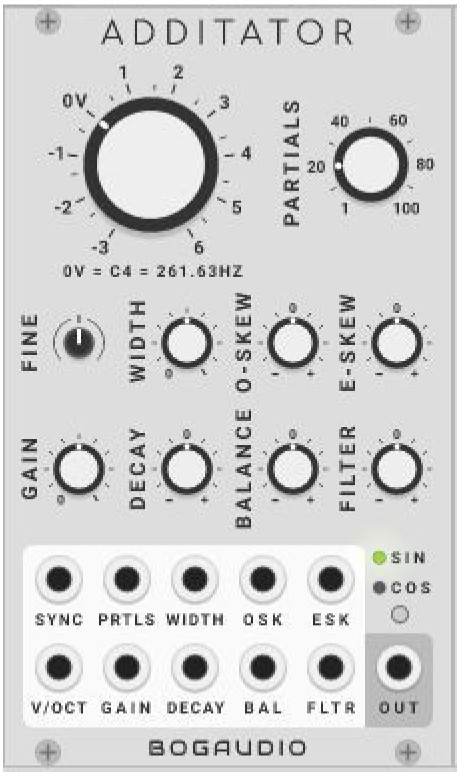
clACos goes one step further and offers **4 oscillators in one module**. Each oscillator has its own waveform selector, which can even be controlled via CV. Unfortunately, there are no single outputs, but only a mixed signal from all oscillators, but that is also important, because the oscillator sounds very well-behaved in itself and therefore requires a good filter and a comprehensive modulation, here is for example **Caudal** from **Vult**, as he provides 12 random tensions.



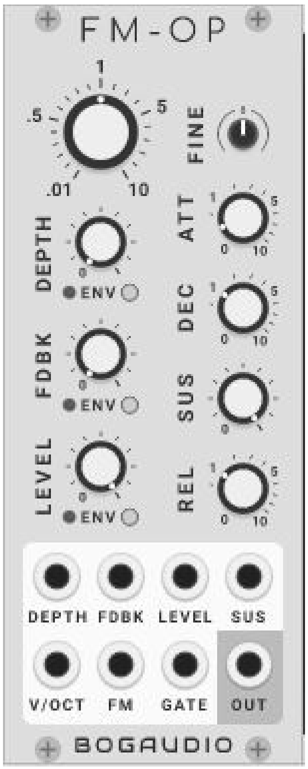
With the formant oscillator **ForK** VCV-Rack learns to **speak french**, at least according to developer Sebastien Bouffier, over the presets button typical french vowels can be accessed. Otherwise, the pitch can be adjusted via **pitch** and the **formants** in 4 frequency bands, each with frequency f and gain g. This can also be controlled by CV. Formidable!

The conclusion of today's episode is made by 4 oscillators from **Bogaudio**. **VCO** is a standard VCO similar to **VCO-1** from **Fundamental**. His **range** is from c1 to c10 (which one does not hear anymore). With the **fine** knob, you can detune +/- one semitone. There are 4 waveforms available: **rectangle**, **sawtooth**, **triangle**, and **sine**, which can be used simultaneously. The **pulse width** of the **square** wave can be controlled by CV. Likewise, the **frequency modulation**,

where you can decide between **linear** (light) and **exponential** (strong) modulation. For external synchronization there is also a **sync** input and of course a **V/Oct** input for the input signal. **XCO** could be considered the **big brother** designate. It has everything **VCO** has, in addition it has a CV input for the modulation depth of the **frequency modulation** and offers the possibility to influence the waveforms individually and output both as a single and as a mix signal. With a controller (or by CV), the **proportion** of waveforms on the mixed signal and their **phase** can be controlled within the signal. In addition, **each waveform has an individual modulation type**, the **square** wave is the **pulse width**, the **sawtooth** a **saturation**, the triangular wave a **Sample & Hold** and the **sine** wave **FM feedback** and these modulations are influenced by CV. Great!. All of these options greatly enhance the "normal" **VCO**, and that's probably what the name **XCO** symbolizes.



The **Additator** is, as the name implies, an **additive oscillator**. The output signal is the sum of **up to 100 individual sine / cosine waves**. The **number**, **frequencies** and **amplitudes** of the partials can be controlled both manually and via CV. With **partials** you determine the number of partials, with 1 you have only one **sine** wave and the further the knob is turned to the right, the more **sine** waves are added. **Width** specifies the distance of the partials in the frequency. In the middle position, each subsequent partial is one octave higher than the previous one. **O-skew** adjusts the spacing of odd-numbered partials up or down relative to width. **E-skew** adjusts the distance of even partials up or down relative to width. **Gain** controls the output level of the signal. **Decay** regulates the decay of the individual partials. In the middle position, the amplitudes decrease proportionally with increasing frequency. **Balance** cuts the amplitudes of the odd or even partials. **Filter** manipulates the sub-amplitudes to simulate low-pass or high-pass filter effects. And as I said, everything can be controlled by CV. Not to mention that in addition to **sine** waves also **cosine** waves can be added, resulting in a completely different sound. The basic operation remains the same.



One to go and that is the **FM-OP**. Of course, this is an **FM** oscillator, so **frequency modulation** ala **Yamaha DX7**, where several **FM-OP** must be connected to each other. Each individual acts as an operator of an **FM** algorithm. There are a few videos about this **FM** synthesis and what it has to do with the algorithms, only so much at this point, the arrangement of the operators in series or parallel, and the strength of the influence of operators on each other determine, the algorithm and, ultimately, the sound. The **depth** and CV controls can be used to influence the input of an external **FM** signal on the **FM** input and the internal feedback of an operator. As well as the strength of the **level** knob adjustable output signal. The oscillator has a built-in **ADSR** envelope generator with which **depth**, **feedback** and **level** can be controlled. For this purpose, a signal must be present at the gate input and the associated button be activated under the respective controller, so that the green light is on. The **sustain** parameter can also be influenced by CV. Anyone who has ever created an **FM** sound knows how difficult it is to get the desired result. Even the smallest changes to the controller can have huge effects, because each operator influences at least one other. And it is not different here, it is even more difficult, because you have to create the algorithm yourself and see no absolute numbers of controls, only approximate knob positions. At least you can now save presets, that's ever very helpful. Whether one prefers this free architecture of **FM** synthesis, or **Dexter** of **Valley**, which provides ready-made algorithms and still has many additional options that go far beyond **FM** synthesis, is certainly a matter of taste. In any case, I think it's good that there are also such modules in VCV-Rack that require experiments and demand a lot from the user.

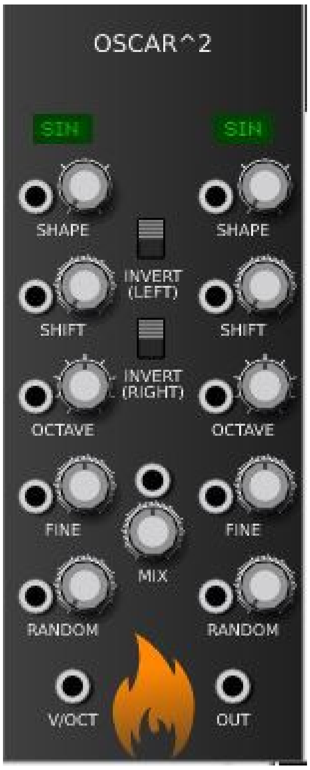
And that's for the sixth episode. I think there were some interesting modules and also some that are only accessible in practice. And that's what my videos are all about, and I want to initiate the process of getting closer to the modules. Therefore, this will always be just a rough overview so you can decide if you are interested in a module or not. Tips and tricks for individual modules can be found on youtube. Just enter the module name or VCV-Rack. As always, you will find links to the most important web pages and also to the developers of the modules introduced this time.

I look forward to your feedback, no matter if wishes, suggestions or criticism. And not to forget, many thanks to Andrew Belt and all the module developers, without whom all this would not be possible. That's it from me, let it be good

Episode 7: Oscillators – Developers C-L

Hello world and welcome to "what does this knob do?".

Now it's making huge strides, as there are not that many developers with C-L offering oscillators. Episode 8 will be the letters M-N, episode 9 is completely S and episode 10 T-V (which Leonardo will probably be very happy ;-)). It will not be long until all oscillators are introduced. The next are the filters. A short note in advance. Some modules will look different for you than for me. That's because I've taken the liberty of changing the background of some modules so I can work better with them. However, I usually left the rest of the design. Without the developer's permission, I will not give away these skins. In principle, however, anyone with inkscape can modify the svg file. However, I definitely recommend making backups of the originals and the new skins. But now let's talk about oscillators.



It starts with **Oscar^2** by **CharredDessert**. **Oscar^2** is called **advanced waveform oscillator**. It offers **two oscillators** combined with adjustable parameters. It has a single **V/Oct** input and an output for the resulting waveform. Waveforms can be selected via the **shape** knob or CV.

Sine, **triangle**, **sawtooth** and **square wave** are available. Both oscillators **can be** individually **inverted**, which also allows different sounds. With **shift**, the initial position of the waveforms can be moved from 0 to 10, with 0 not being moved at all and 10 being shifted 100% to the right. With **octave**, you set the octave between c0 and c8 and with **fine** you can tune again by +/- one octave. **Random** regulates the possibility of skipping a waveform in one cycle. Instead of outputting a waveform, 0 is output. The output signal is a combination of the two waveforms, mixed by the mix input, which can be controlled by CV like any other knob. Who follows my videos, knows that I am a fan of CV inputs, but I miss 2, namely **FM** and **PWM**, which usually has every oscillator. And if I'm already complaining, single outputs for the two oscillators would also be great. I do not want to say that **Oscar^2** is a bad oscillator. But we know that the better is the enemy of the good.

In github, **dBiz** does not reveal much about its modules, but the oscillators confidently say

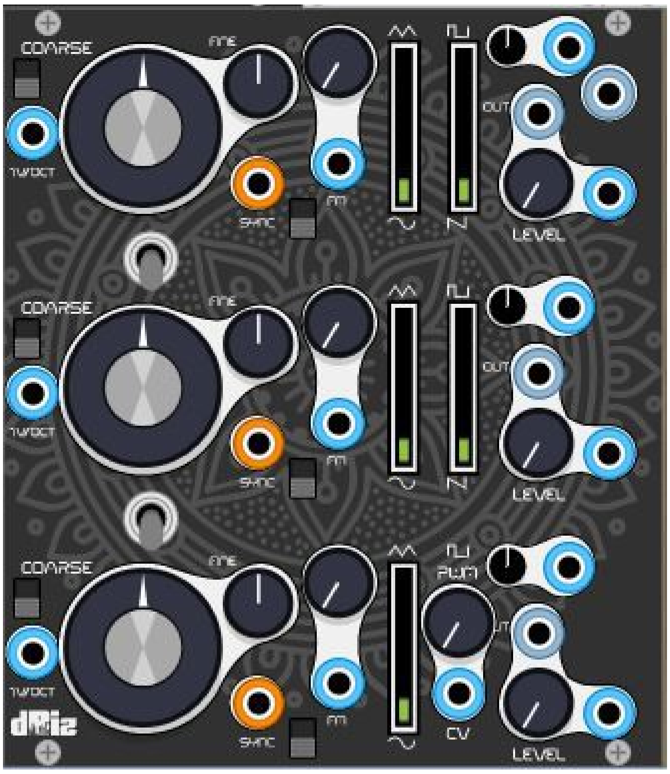
"they do not need introductions". But when you see the picture, you can not really imagine it, and actually there is a lot to say about **dBiz**'s oscillators. The **DAOSC** has a few elements of the **VCO-1** from **Fundamental** (which I like to take as a reference), but also a few special features. Above all, it consists of **2 identical individually controllable oscillators**. With **coarse** you can tune it in the range of +/- 4 octaves and with **fine** +/- 3 semitones. It has **3 outputs**, one for each oscillator and a total output, although the mixing ratio of the two oscillators can not be controlled. But otherwise everything can be adjusted. With **contrast** one fades over between **sine** wave and **square** wave. If different waveforms are set in both oscillators, **sawtooth**-like and triangular-like waveforms also appear at the **sum** output. Further deformation of the waveform by adding and hiding frequencies can be done with **dist**. This will make the previously bevelled **square** wave the perfect rectangle when the knob is on the far right and will get an edge on the other side. **FM** is the usual attenuator for an external signal at the **FM** input. The next two controls add harmonics or overtones to the sound, distorting the waveform as well. **Odds** are odd harmonics, **pair** (french for even) even harmonics. And all these controls have one CV input and they are available for each oscillator.

DVCO is similar but different and it's even harder to open up. The pitch is the same as the **DAOSC** and is set here via **Hz** and **fine**. However, we also have the option here of the oscillator or the oscillators, because there are again **2 in one**, to operate as an **LFO**. In addition, both oscillators can also be **synchronized** via a slider. **FM** also works like in his colleague. What we also know from **VCO-1** is the ability to switch between **analog** and **digital waveform** and hard / soft **sync**. **V/Oct**, **sync** input and **PWM** input are also old acquaintances. The rotary encoders have **PWM**, i. e. pulse width modulation **and PW**, which can generally be used to set the pulse width of the modulation.

This works for **any waveform containing a pulse**, including mixed forms. Wave is used to infinitely variable select the waveform. The waveforms are tapped at **6 outputs**. Each oscillator provides, in addition to the **normal** one, a **mirrored** version of the waveform, marked with a minus. In addition, oscillator b is in the **analog mode inverted** to oscillator a, so we have 4 different waveforms simultaneously available and in addition a **sum** signal from the 4 individual signals and we still have the output **RM**, which should be an **average** of 4 individual signals. Anyway, another signal. To better identify the fundamental waveforms, I switch the waveform to **digital**. The **basic waveforms** are sinusoidal if both wave controls are on the left stop. Here you can see the reflection, but the inverted **sine** remains **sine**. If I move a wave controller slowly to the right, first a **triangle**, then a **sawtooth** and finally a very narrow pulse wave, whose width can be adjusted with the pw-controller. Clearly you can see in the changes, the reflection of the waveform. In **digital** mode, both oscillators are identical, inverted in **analog** mode. But that's not all, **DVCO** still has 2 inputs, **carrier** and **modulator**. We know these terms, e.g. from **vocoders**, so I can introduce 2 signals here and use them as a carrier signal and modulator, the signal from the **DVCO** is then only at the individual outputs, the introduced signal at the sum outputs. Everything **very exciting** and certainly not self-explanatoxry.

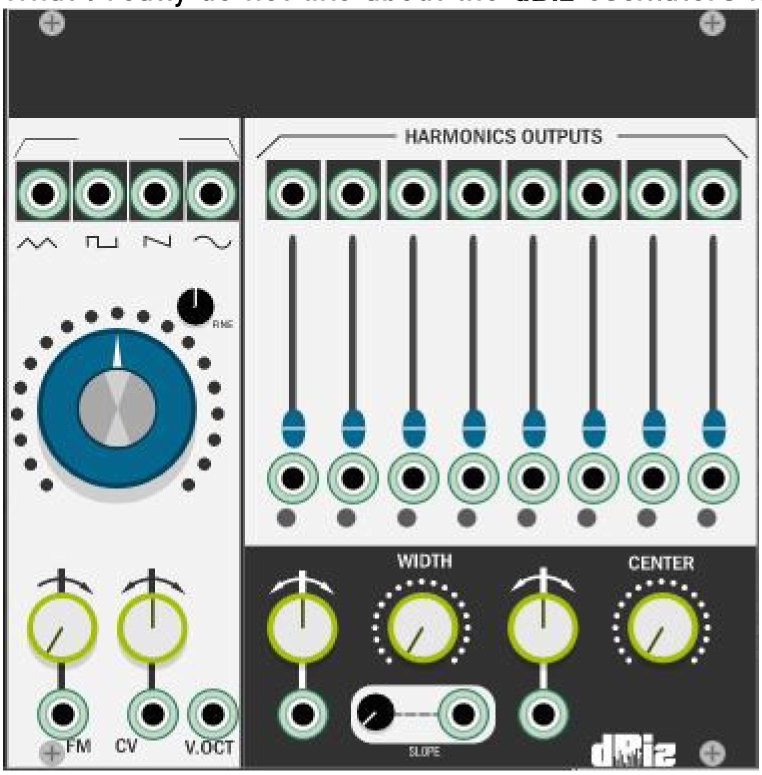
SuHa is one of the new modules in the new design and actually self-explanatory. **SuHa** stands for **subharmonic oscillator** and that's exactly what it's all about. In principle, we also have **2 oscillators here**, but each one **also** has **two sub oscillators**, which are connected to it **in tune**, that means I detune the main oscillator, the subs are still in tune to it. As usual, the main oscillator has +/- 4 octaves and is infinitely variable. The sub oscillators start at the very left with the same tone as the main oscillator and can then be tuned down to 1 Hz. For the entire module, there is a **master volume** with CV input and each oscillator and sub-oscillator has its **own volume control**, as well as its **own V/Oct** input and a separate **output**. So you can send in **6 different signals** and pick up each at its own output. But we can also build a **sum** signal and thereby create **complex waveforms**. A simple principle, but with some potential.

The **Triple Oscillator** is exactly what you expect, namely **3 oscillators in one**. The three oscillators each have their **own output**, but there is also a **sum output**. Besides this they are almost identical. At inputs there are as usual **V/Oct**, **sync** and **FM**. In addition, you can adjust the



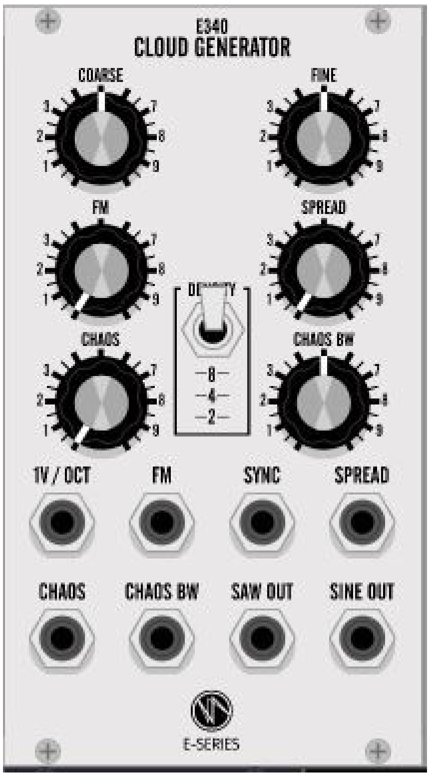
volume for each oscillator by CV or manually and the **mixing ratio** of the two waveform controls. There are the usual waveforms available, **triangle**, **sine**, **sawtooth** and **square**, where **oscillator 3** has no **sawtooth**, but **pulse width modulation**, which is naturally controlled by CV. Exceptional is the choice of waveform. Here for, there is a **slider** and on the other hand a **knob**, which can also be controlled by CV. If this knob is on the leftmost, only the left fader is active and you can choose between **triangle** and **sine**. If it is on the far right, only the right fader is active and thus rectangle and **sawtooth**. However, since both the faders and the knob have flowing transitions, almost **countless hybrids** are possible and then the automation by CV. One last option is still connected to the 2 small gray switches. Hereby I can choose, if each oscillator should receive its **own input signal** or if all of them are processing the same signal. This option also allows **chords**. Also a very interesting oscillator.

What I really do not like about the **dBiz** oscillators is their CPU hunger and **Verbo** really

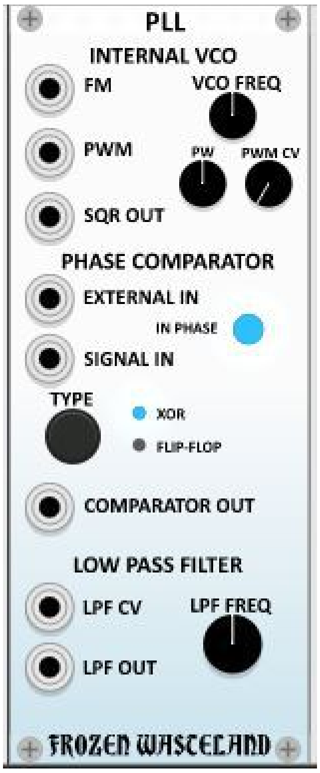


shoots the bird, as you can see on the red bar. My computer is certainly not the worst, but these modules force him over and over again and I have to take other oscillators. I would be thrilled if these modules were more economical. But so much only besides, we come to **Verbo**, (Which is a recreation of the **Verbos Harmonic Oscillator**. I didn't know that, when I made the video and therefore it is not mentioned,) It reminds me of the **T(W)LV** of **Autodafe / Soundmit**,

which I already introduced. It is about the **addition of harmonics**, so **sine** waves with different frequency and also here each one can be controlled separately. But that ends the similarity. In principle, **Verbo** consists of 2 parts, which is also visually presented.
On the left is a **standard oscillator** with the 4 **basic waveforms triangle**, **rectangle**, **sawtooth** and **sine** wave. The **pitch** is adjusted via the large knob with +/- 4 octaves and with **fine** in +/- 3 semitones. **FM** and pitch can be controlled by CV and there is a **V/Oct** input. So nothing special. The **sine output** is also the output for the **sum of the harmonics** and thus for the **additive** part, whereby **each partial tone can also be output separately**. Between outputs and inputs are **sliders**, with which the amplitude of each partial tone can be determined or one controls it by CV at the inputs. The blue lights underneath are related to the 5 other knobs. With these we can **define** harmonics to be emphasized. **Center** determines the frequency to be emphasized, **width** regulates in which range around the frequency the harmonics are emphasized, the two controls on the left and right of it **regulate**, if and how the edge areas are emphasized and **slope**, how fluent or choppy the transition between emphasized harmonics and the other ones is. The latter three again have CV inputs. In any case, an interesting oscillator, especially if you combine the two parts and forms a sum signal from it. If he did not need so much CPU ...



The next oscillator is again a classic and was also one of the first modules. It is the **E-Series Cloud Generator**, the hardware version for the euro-rack is the **Synthesis Technology E340 Cloud Generator**. In the extreme case, **16 oscillators** are used for one sound if both waveforms saw and **sine** are used simultaneously. Per selector you can choose per output for 2, 4 or 8 oscillators. But you know that infinitely many oscillators that are tuned exactly the same, are absolutely boring. Therefore, one usually uses the fine tuning to generate beats. Although we also have the possibility to **tune** or **detune** in the range of +/- 4 octaves resp. +/- 1 semitone, this does not help if there is not another oscillator in the game. But we have heard, this module includes several oscillators. And here comes the **spread** button into play. In **Bogaudio**'s **Frequency Analyzer** we can see very well what **spread** does. It increases the gap between the oscillators. With 2 **sine** waves this is not that breathtaking, but if we take 4 or even eight, things look different. And listen to the **sawtooth**, here is even the smallest movement enough, to produce a **fat sound** with a wonderful hovering. Here you really have to be careful, too much is really too much here. In no time we have a **supersaw**. And so that this detuning does not remain static there is the **chaos control**, which puts the oscillators in random oscillation, the farther the knob is turned up, the stronger the oscillation. Their speed can be adjusted with the attenuverter **chaos bw**. Of course **chaos**, **chaos bw** and **spread** can also be controlled by CV, just like **FM** and **sync**. Again an oscillator that can not do much, but what he can do is making him great. And the sound is beyond doubt.



The next module is something very special. **Frozen Wasteland's Phased Locked Loop** is inspired by **Doepfer's A-196** module and it is also recommended to read the description on **Doepfer's Website**. I have to admit that I did not understand the function of this module either, but in principle the module consists of **3 units**, the **internal VCO**, the **phase comparator** and the **low pass filter**. The **phase comparator** compares the signal of the **internal VCO** with the frequency of an **external signal** (**Frozen Wasteland** recommends a kind of **square** wave at the input signal in). Thus, it generates a **digital** signal indicating whether the frequency or phase difference of the two compared signals is positive, zero or negative. This signal is **smoothed** with the **low-pass filter**.

Let's take a look at it in practice and especially in the oscilloscope. As recommended, I introduce a **square** wave into the **PLL** input signal and go out to the output **sqr out**. The signal is shown here, a standard **square** wave at pitch c4. The frequency range of the **PLL** VCO can be difficult to measure with the on-board resources of VCV-Rack, but according to **Doepfer** it should be between 2 Hz and 100 kHz. Without external signal we have a simple oscillator with +/- 4 octaves, which produces only **square** waves, which can be **regulated** in the pulse width and that also by CV. But even here we have a difference to the usual oscillators, because here both **pulse width, and frequency is changed**. So the **PWM** changes the **pitch**, in addition to the usual thinning out of the signal. The usual CV input for the **frequency modulation** is available, but not adjustable.

As soon as an external signal is connected to **signal in**, the **phase comparator** starts to work. He compares the external signal, as seen here this red **square** wave, with the signal of the **Internal VCO**, here these blue one, which is **square** wave, which is constantly changing in

frequency and pulse width. Because of the **pulsing**, the difference between the two signals is sometimes **positive**, sometimes **negative**, and sometimes **zero**. The resulting signal is the yellow **square** wave, which has clear dropouts and also a different frequency, as we also hear very clearly. At the moment, the **phase comparator** is at **xor**, either or, so the blue dot, that indicates when the two signals are in phase, blinks constantly. So there is only a very short phase equality at the moment when the signals overlap. If we switch to the module **flip-flop**, the **phase comparator** once compares the positive range of the signals and once the negative range. As a result, there is a longer phase equality and the signal has less dropouts. A small change in frequency brings the signals almost in line and we get a nearly flawless signal. The further we lower the frequency, the greater the match, as indicated by the blue light. In **xor** mode, the coincidence increases with decreasing frequency, but it is still much lower, so you can clearly hear the different frequencies and **FM**-like sounds arise. The signal at the end of the comparator can be tapped and used further. But it is also internally in the **low-pass filter**, whose frequency can also be controlled manually by CV. In addition, this also has its own output, with which its signal can be used again for the control of other modules. Moving the frequency knob from left to right clearly shows how the output signal increases in motion. In the middle position one obtains a **sine** wave at low frequencies at the VCO controller, which fluctuates however strongly. At mid frequencies, the output of the filter is the most chaotic and gets smoother as the frequency gets higher. If the VCO frequency knob is in the middle and I turn the filter frequency knob further to the right, the signal becomes more and more round and shows about 2 o'clock a kind of smoothed **sawtooth**, further turned the signal is more chaotic and splits. If the VCO frequency regulator is on the left stop, you can use the frequency control filter to generate all standard waveforms. In conjunction with the PW and **PWM** controls it can now also generate typical **PWM** effects, but also quite crazy versions of it. In the **flip-flop** mode the knobs work similarly, but you can tap interesting envelopes on the filter output or interesting triggers and gate effects when the VCO knob is turned off.
Instead of the internal VCO, one can also use an **external signal**, which is introduced at the **external in** input. In principle, the controller works the same, of course, but here it depends very much on which signals are used, the more different the signals, the more extreme is the result. The **PLL** is definitely a module to deal with, but it also rewards with its **extraordinary sounds**.

The next two oscillators are often used as a reference or model for other oscillators, **VCO-1** and **VCO-2** from **Fundamental**, **VCV-Rack's** proprietary oscillators. But reference also

means that other oscillators have to compete with them because standard does not mean bad. I have already mentioned at some reviews, which makes this standard. **Pitch** of +/- 4 octaves, **fine** tuning of +/- 3 semitone steps. The 4 standard waveforms **sine**, **triangle**, **sawtooth** and **square** can be used simultaneously. The **square** wave can be changed in the **pulse width** and this can also be **controlled by CV**, as well as the **frequency modulation**. There is an input for external **synchronization** and you can switch between **hard** and **soft-sync** and you can choose between **analog** and **digital waveform**.

The **VCO-2**, however, differs in a few things. It has no 4 individual outputs for the waveforms, but only **one output** and the waveform can be **continuously adjusted** or controlled via the wave knob or CV. Nevertheless, all standard waveforms are possible, but also **mixed forms**. Two good and reliable oscillators and sometimes it does not need it anymore.

The next two oscillators **VCO-F1** and **VCO-F2** are **almost equal** to the **Fundamentals**. But, like all **Gratrix** modules, each **output** is available **6x** and each **input** is **7x**. So you can save on multi

or merge modules. That too can sometimes be useful.

And here comes **Woldemar** and he is not a noseless black magician, but an interesting oscillator from **Lindenberg Research**. He calls himself **voltage controlled analog oscillator** and so it is already clear, that he has no switch between **analog** and **digital** waveform and also a sync function we are looking for in vain. Its range goes from c0 - c8 and this is set via the **octave control** and the **coarse** button. And the **octave controller** has a **low** position and this allows **Woldemar** to be used as an **LFO**. As soon as this mode is activated, a blue light illuminates and an led display with the current frequency appears. The controller **coarse** can be set between 0 and 25,000 Hz. Woldemar has **2 V/Oct inputs**, whose signals are added, is only one occupied, of course, we have only 1 signal. **FM** and **pulse width** of the **square** wave and the **triangular** wave (and thus of course the sum signal) can be controlled by CV. The pulse width also manually. We have the **basic waveforms sawtooth**, **square**, **sine** and **triangle** available and also **white noise**. Each **waveform** has its own **output** and except for the **noise**, all the waveforms go into a **sum** output mix, with the mix **ratio** set via an attenuator for each waveform. In the middle position, no signal is output, turned clockwise increases the proportion of the waveform in the mix signal and when it is turned **counterclockwise**, the proportion of the **inverted** signal increases. Even the basic sound of **Woldemar** is beautifully **analogue**, but the **variety of sounds** through the mix is **breathtaking**. Too bad that not even the **noise** can be mixed, you can indeed loop it into the **V/Oct** input, but unfortunately not regulate and that affects all the waveforms. Nevertheless, it is a very versatile oscillator and of course LFO.

Finally, I have now an oscillator of which I do not know what to think of it. We are talking about **LogInstruments Crystal**. He has **three inputs**, with probably a gate signal must be present in the upper left, in the middle more or less **V/Oct** and at the right input a sort of modulation signal. At least I found this out by trying, because the github page of **LogInstruments** says only "easy-to-break signal generator with unpredictable outcomes. Up to 32 oscillators employing an alias-free signal generation technique, it has three input CVs, detune control and laser gem. Explore it and handle with care." Yeah, I can confirm that with the **unpredictable** results. **Detune** detunes up to 32 oscillators, the number of which is set via **voice**s. This can be seen in the **laser gem** in the number of facets, with full number of **voice**s you can see a **diamond**, but the result is getting used to. I had difficulty with more than 6 oscillators to produce a melodic sound in the long run. **Vol** is of course the volume, with **rst**, the wave is reset, which could be quite useful if there would be a CV input. **Lock** ensures that the **pitch** does not change. The three outputs actually provide three different signals, which can be seen in the frequency spectrum. **Without input** signal, it has a **beautiful filigree** sound, which I would like to have with an input signal, but so far **I have not found a way** to accomplish that. So he leaves me with a very ambiguous impression. **Too bad, the idea certainly has potential or I'm just too stupid to understand the principle.** Maybe you've had other experiences and I'm actually on the wrong track. I would be happy to hear from you.

And that's how I reached the final word. I think also this time there were exciting modules waiting to be discovered by you. As always, you will find links to the most important web pages and also to the developers of the modules introduced this time. I look forward to your feedback, no matter if wishes, suggestions or criticism. And not to forget, many thanks to Andrew Belt and all the module developers, without whom all this would not be possible.

That's it from me, take care

Episode 8: Oscillators – Developers M-N

Hello world and welcome to "what does this knob do?".

As promised, I'll introduce you today to oscillators from developers of M-N. I also make up for a module that was somehow not displayed as an oscillator in the module browser. I will start with this one, too.

I introduced **Chipyourwave** from **Bacon Music**, but **Chipwaves** slipped through. Developer Paul Walker also shows his enthusiasm for chip sound with this oscillator, he simulates the sound generation of the NES. **Triangle** and **pulse** wave are provided. The **triangle** wave is **stepped** and therefore lo-fi, and the **width of the pulse wave** can be set in **4 steps** with the **duty cycle** control. The frequency range reaches into the **LFO** range, I measured 11. 6 Hz and with the **triangle** wave up to f6 and the pulse wave up to f8 and then there is another **V/Oct** input and that's it.

One of the most diligent and innovative module developers in VCV-Rack is **mscHack**. I have to admit, at first the design of the modules kept me from trying them out. But behind the sim - ple, almost child-friendly façade, some real treasures are hidden. First of all, we have the **OSC 3**. At first glance he seems to be missing a lot, but he has other things that aren't actually

in an oscillator. In principle, it consists of **3 identical oscillators**. Everyone has the usual **V/Oct** input, a **trigger** input, which must have a signal in any case, otherwise nothing will happen. In addition, both filter **cutoff**, **resonance** and **volume** can be controlled by CV. The **filter** is already the first special feature, because this is usually an extra module. Not with the **OSC 3** and the filter can be used as **lowpass**, **highpass**, **bandpass** and **notch** filters or off. And as already mentioned **cutoff frequency** and **resonance** are controllable. Each oscillator offers the four

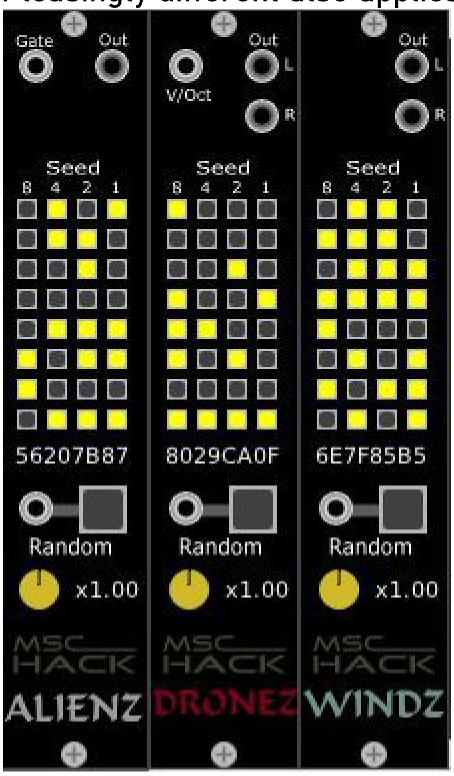
basic waveforms sine, **triangle**, **square** and **sawtooth** plus **white noise**. Only **one** waveform **at a time** can be output per oscillator. A simple **envelope generator** with **attack**, **decay** and **release** is also integrated to trigger the envelope, that's why the trigger pulse is required. The sound of each oscillator consists of **up to 7 identical part waveforms**. With **n-waves** you set their number, with **detune** they are detuned against each other, which we know from the **E-Series Cloud Generator**, there it is the task of the **spread** button. Here, **spread** is responsible for delimiting the partials to each other. Far left is a clear demarcation, turning the knob further to the right, the transitions are always softer or more fluent. With **each waveform**, however, this effect is perceivable **differently**. A very interesting kind of sound shaping. Each oscillator has **2 outputs**, since these are labeled with left and right, that means probably that it can be operated in stereo, which indeed makes sense because of the spread. Unfortunately, there are **no master functions** for the whole module, i. e. **V/Oct** global, mastervolume and sum output. But of course everything can be improvised, that's why it does not matter so much. More serious, however, is the **lack** of opportunity **to tune** the oscillator. The pitch must therefore be determined by the input signal or you generally use it for **bass passages**. Sonically, he is almost predestined for this. An interesting oscillator with some possibilities.

With the **Wave Morph Oscillator**, we can once again **paint our own waveform** and thereby

generate virtually any waveform, provided that 17 steps of coordinates are sufficient. Where coordinates are not quite correct, because the values of the x-axis are fixed and only the y-axis can be shifted in the range of +/- 6 volts, whereby always the instantaneous value of a point can be seen. Nevertheless, a lot can be started with it. With **draw** you can activate the draw mode and with **rubberband** how much the neighboring dots are influenced by the drawing. For a quick overview, I'd recommend clicking on the **rand** button, which wave morph will provide right away. This all looks like business development or temperature scale, but is a **complex randomly generated waveform**. If that's too coincidental, you will also find a few **presets** and here old acquaintances like a **square** wave, a **sine**, a **cosine**, a kind of **mixture** of **sawtooth** and **sine**wave, a pure **sawtooth**, a **phase-**

shifted sawtooth and a **pure signal** with + each / - 6 volts and 0 volts. By clicking on **invert**, any waveform, even self-generated or random, can be inverted. With the button **copy** you can copy a waveform into one of the **3 Wave Morph Oscillators**. This one selects over 1, 2, 3 and the output signal always corresponds to the volts of the input signal at the morph input. If I switch the oscillator to **solo**, only the tuned oscillator with the corresponding waveform will sound. In this mode, with a CV signal at the input **change**, also the choice of the oscillator can be automated. Both **trigger** and **gate** signals are possible. So we never get a mixed signal from all 3 oscillators. The 3 oscillators alone can also be activated by -10 volts for oscillator 1, 0 volts for oscillator 2 and +10 volts for oscillator 3 at the input **morph**. If I enter - 5 volts, I get the same mixture of oscillator 1 and 2, if I enter 2 volts, the output signal consists of 80% oscillator 2 and 20% oscillator 3. Of course, finer subdivisions are also possible and can therefore be morphed through the waveforms with a changing input signal, e. g. by LFO or envelope. From light beating to total chaos, everything is possible, depending on which waveforms I choose and which modulator. The output volume is set with **level** and this can be controlled by CV. We also find the same **filter** here as in the **OSC 3**, so I do not go into it here, but with this one I can edit the waveforms very nicely. I recommend using an **oscilloscope** when working with the wave morph to see what you are doing and see all the changes. Again, the **lack of pitch adjustment** is noticeable. But this oscillator is **pleasingly different**.

Pleasingly different also applies to the next 3 modules of **mscHack**. They look similar and work similar, so I trade them together. We are talking about **Alienz**, **Dronez** and **Windz**. I have no idea if they are intended for specific sound design, I see here rather the **random sound generation** but in this they are quite big. If not everything deceives me, there are 4,294,967,296 ways to combine the 32 seeds, because everyone can have the state on and off, i. e. 2 to the power of 32, so in principle you could do targeted sound design, but for that it would be necessary to try all combinations, because each combination, even if only of 2 seeds, has a different result. Their **exact code** can be read and remembered for reuse, or you can simply **save the preset** and reload it if necessary. But I found out that the seeds of the fifth row are all



variations of the pitch a, but with different modulations or overtones. The internal modulation, which is permanently linked to each seed, is a **filter modulation** ala **wha wha** and its **speed** is set with the yellow control between 0.1 and 8, so both positive and negative modulations are possible and in the middle position at 1 there is no modulation. The **random** button in the middle of the modules is very prominent and as I said, here I can see how they are used. Especially since this can also be **triggered**. **Dronez** also has a **V/Oct** input and this should also be used, then the drone sounds really come into their own. Because that's what **Dronez** is for, and that's what he's doing great for.

Windz has no **V/Oct** input, it simply generates different **types of noise** and combines them into wind **noise** and here too the speed of modulation can be controlled and the random generation of the combinations can be triggered. **Dronez** and **Windz** are **stereo** designed.

Alienz is only **mono** and has a **gate** input instead of **V/Oct**. Here you can of course introduce a **modulation source**, but also **external signals** such as another oscillator, perhaps even **Dronez**. **Alienz**'s seeds are all **special effects** and would have fit wonderfully on board the Enterprise. And just in the **combination** of the modules are extremely **wacky** things possible. These 3 oscillators are certainly not intended for everyday use, although I can often imagine **Dronez**, but they are absolute **specialists** when it comes to special effects. You can't usually do them that easily.

The next oscillator fits again very well into the category **basic**. It's the **Rogue** from **MSM**. At

this point I have to highlight the **manual**, which is really **fantastic** and contains all previous modules of **MSM** and perfectly answers the question "what does this knob do? Nevertheless I introduce the modules of **MSM** of course, equal rights for all.

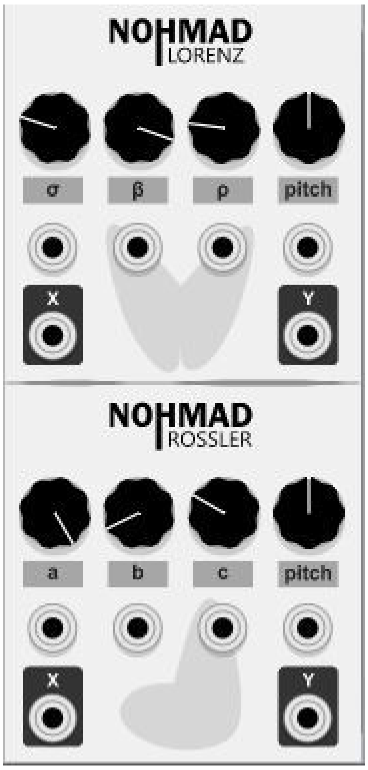
Ok, the **Rogue** has some features of the **VCO-1** from **Fundamental**, which is always my reference. It has the same **range**, from +/- 4 octaves and **fine** tuning +/- 3 semitone steps. It can be switched from **analog** to **digital** waveform and for further consideration I now also use the **digital** waveform. Here we see that the **digital** waveform is cleaner but not perfect and therefore still has a slight **analog** character. It offers the possibility of **frequency modulation**, but even has 2 modes, **linear** and **exponential** with **extra inputs** and controllers. It has a **V/Oct** input and a **reset** socket with which the waveform can be reset and restarted via **trigger**. In principle, a **gate** signal is also possible here, but the reset is only triggered if the

threshold is exceeded. At waveforms **ramp** (an inverted **sawtooth** wave), **sine**, **triangle**, **sawtooth** and **square** are available, whereby the pulse width of the **square** wave can be controlled manually and by CV. In addition, there are also 2 waveforms that correspond to a **rectified sine**. One represents the **full** waveform and is output at **fwrs** and the other only **half** of the waveform and this is applied at **hwrs**. And as if that wasn't enough, the oscillator can also be operated as an **LFO**. In any case an interesting variant of the **VCO-1**.

The next oscillator could be described as **Rogue**'s big brother. Basically the **OSCiX** consists of 2 **Rogues**, but without the rectified **sine** waves. But it has the waveform **fold**, but more about that later. Both oscillators can be **synchronized** with each other by switch in **both directions**, i. e. oscillator a with oscillator b and vice versa and even both, which creates a **synchronization loop** and synchronization can even be controlled by CV. Both oscillators have their own controls for **linear** and **exponential frequency modulation** and these can even be operated without CV, which allows really great sounds, but of course there is also an input jack for CV control. The **FM index control** selects the frequencies to be influenced for the **FM of both oscillators** and this can also be controlled via CV. Oscillator A generates the waveforms **sine**, **triangle** and **sawtooth**, oscillator B **sine**, **square** and **fold**, whereby the pulse width of the **square** wave can be determined with the **PW** controller and controlled by CV. Let's get to **fold**. This output remains mute first if it is occupied. With **fold index** you generate a waveform and determine its amplitude, this can also be done via CV. This waveform already contains a first convolution, further convolution is done with the **fold control**. The more folds, the more chaotic the signal becomes and sometimes sounds metallic. The **fold CV input** with the corresponding control is assigned to the fold control, i. e. the number of folds can also be dynamically controlled. The **folding** can take place from bottom to top and vice versa. The **degree** of folding from which direction is set using the **up** and **down** knobs, CV inputs are also

available for this. Finally, it is also possible to insert an **external signal** and fold it. All in all, the **OSCiX** is extremely versatile and easier to use than you would expect from its possibilities. All that remains is that all **MSM** modules have both a light and a dark design.

Let's get to stranger things, that's all I can call **Strange Attractors** from **Nohmad**. First I had



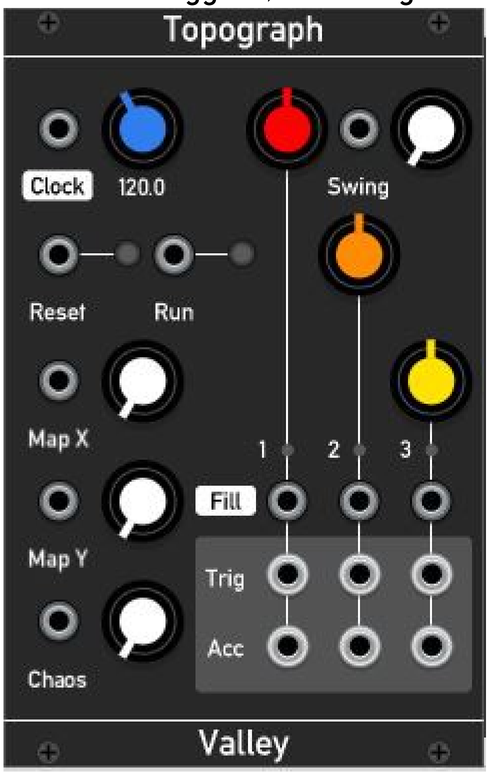
to get to know what the **Lorenz** and **Rössler** attractors were and honestly I didn't understand much about them. It is generally a stretch-fold operation. These can be calculated mathematically with corresponding variables and can therefore also be displayed. In **Lissajous** mode of **JW-Modules Full Scope** you can see them very nicely. Each attractor has an x and a y coordinate and these are output at the corresponding **outputs**. An input signal can be introduced into **pitch** and the pitch can also be adjusted, at least to some extent. Because the other controllers determine the variables of the respective equation. **Lorenz** has the variables **sigma**, **beta** and **rho**, **Rössler a**, **b** and **c**. If at **Lorenz** all controls are at the left stop and the pitch control is in the middle position, we have a kind of **sine** to which frequencies are randomly added and removed again. If everything doesn't fool me, **sigma** adds low frequencies by stretching the waveform, **beta** adds medium frequencies and folds them and **rho** adds high frequencies. The further the knobs are turned up, the more equal frequencies are generated and overlap, partly they are added, partly they cancel each other out. The result is a **bipolar** signal with randomly changing amplitudes and frequencies, which can become very chaotic, but also generates regular waveforms by selectively adding and removing frequencies.

Rössler is a little easier to grasp. If all controllers are on the left, we have an even **sine** wave with the frequency 234. 83 Hz. Regulator **a** compresses the shaft, which slightly increases the frequency and additionally every second amplitude. Interestingly, the signal at output y is mirror-inverted to output x. Controller **b** stretches the signal again, making it more regular again, but the frequency decreases again. Knob **c** distorts the signal by adding overtones, increasing and decreasing individual amplitudes more or less regularly. The further the slider is turned to the right, the more chaotic the waveform is, but it repeats itself again and again and basically always remains a **sine** wave. That's my **subjective** view of the **Strange Attractors**. I think you should just try it out here and it certainly doesn't hurt to see the corresponding **Lissajous** graphic. **Strange Attractors** is certainly not a bread & butter oscillator, but it has its appeal just as surely.

Also the **Quadratic Iterator** from **Nonlinear Instruments** is anything but commonplace. As

the name suggests. Roughly speaking, **iteration** is a mathematical procedure in which the result of a calculation serves as the basis for the next calculation and this is repeated until one has an almost valid result. We are here in the basics of **chaos theory** and with this procedure one wants to predict chaotic processes. This is certainly not the task of this module, but the term above the big button tells us that it's all about **chaos**. This is also the only parameter that can be changed. In this module 2 iterators work and the result of their calculations is output at the outputs **x** and **y**. The chaos button defines the parameter **r** in the programmed equation **xn+1 = r -xn - (1 - xn)**. I don't know what value the developer has set for **xn**, but usually it is between 0 and 1. mathematical calculations have shown that chaos starts at a value of **r>3. 5**, so the controller starts with that. The more it is turned up, the more chaotic the signal becomes. When the offset controller is closed, **r** is the same for both iterators. If you move it to the left, the 2nd iterator gets a different **r**. The strength of this deviation is determined with the offset control. And so that the chaos does not remain static, the value for **r** can also be controlled by CV. The intensity of this modulation is determined with the **CV-c** controller. What do we do with it now? Together with an envelope generator and a good filter you can do a lot with it from melodic to percussive. But he's a real **challenge**.

Let's see if the **Nysthi** modules are also such a challenge, at least they have many controls. And I have to admit, that scared me off at the beginning and Antonio Tuzzi is just so busy to release at least one new module every week that you can hardly keep up. All right, challenge accepted. Let's start with the **Deepnote Generator**. He first welcomes us by clicking on start with a swelling chord, which I have heard somewhere before. Ah yes, **THX**! After that, he's mute. Next to the start button is an input, let's see what happens if you introduce a **V/Oct** signal here. Nothing, obviously this is a **gate/trigger input**. After hours of experimenting I asked Antonio Tuzzi, the head behind **Nysthi**, who explained that **51 VCO's** and **150 envelopes** are running internally and he is not to be used as normal VCO, only as basic material for sampling. Well, that takes me further. Ok, there was a module of **Nysthi** among the oscillators, which processes samples. But the **Quadsimpler** is only a sample player, but its big brother the **Complexsimpler** can also sample itself. Since sampling is not our topic today, I did it in the background and already loaded the file into the **Quadsimpler**. We'll get to that in a moment. First of all I'm still at the **Deepnote Generator** and so I'm not giving up using it elsewhere. I let myself be carried away to the saying that **you can create drums with every module**, so why not with this one as well. The **THX** sound has a rather long **attack** time and also the **duration** is not exactly short and therefore we have adjustment possibilities, exactly like for **4 frequency ranges**. As usual with **Nysthi**, click on the number you want to change and move the **mouse** up or down. With several **Deepnotes** and different triggers, like e. g. the **Topograph** of **Valley** you can do quite a bit and I just decided that I will do it in the **very cool patch contest 10**. So if you want to know what I'm doing with the **Deepnote Generator**, you should check out my video.



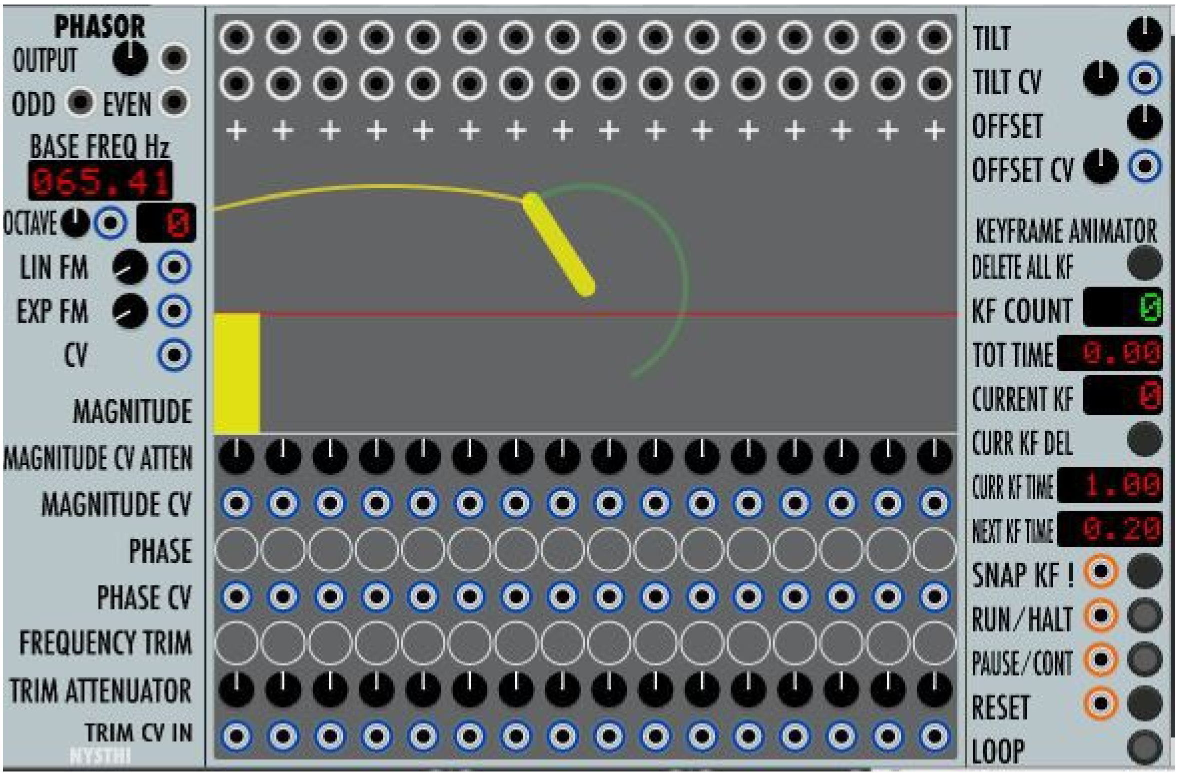
Here I continue with the **Quadsimpler**, which I already fed with the sound of the **Deepnote**. As already mentioned, the **Quadsimpler** is a **sample player** and, as you can see and as the name suggests, you can play **4 samples at the same time**, which is remarkable in itself. All 4 units are identically constructed. First of all we have the **white field**, here of course the sample should be **inserted** and this can be done simply by **right-clicking** and selecting the slot to be equipped. This will open a window to select the **wav** or **aiff** file. It couldn't be simpler. Once the sample is loaded, you can see its waveform in the **display** and **start** it by clicking trig. Immediately a **blue bar** starts to race back and forth and does not stop until you click either **gate** or **one-shot**. In **gate** mode, it waits for a gate signal at the **trigger** input and plays the sample as long as the gate signal continues. Pressing or holding the trigger button repeatedly also has the same effect. In **one-shot** mode, the sample is played once as soon as a **trigger** signal is received or the button is pressed. Holding the button is useless and with a gate signal, the sample is triggered only once as soon as a voltage is applied, just 1v is enough. As soon as a trigger signal arrives at the **stop** input, the sample is stopped. It is therefore possible to play back the sample for a desired time using trigger signals. This trigger signal can also be triggered internally via the **eoc** end of cycle output. When the cycle of one sample ends, the other is activated when the **eoc** of one unit is connected to the trigger in another. This allows the samples of the four units to be played back automatically one after the other.

Start CV is assigned to the start slider, which sets the start point within the sample. The CV input can also be moved dynamically and automatically, e. g. with an LFO or an envelope. The **green bar** always indicates where the starting point is. The **FM** controller only works when a control voltage is applied to the corresponding input and controls the sampling speed of the sample. These can be set manually with the **speed** control, whereby in the initialized position the original speed of the sample is applied and with the clockwise direction the speed increases and decreases counter-clockwise. In the center position the speed is 0 and to the left of the center position the sample is played backwards. With **pan** you can of course adjust the panorama and each unit has its own output. But of course there is also a common **stereo** output. And if you want to add more samples, e. g. more **Quadsimplers** to the stereo signal, you can use the **chain mixer control** with its 2 inputs. The **Quadsimpler** impresses by its **simplicity**, it is a simple sample player and doesn't want to be any more. Therefore, there is probably no way to define the start and end point within the sample. This can be improvised via a trigger pulse at the stop input. Or you can use the big brother **Complexsimpler**, which can do that.

In the **LFO Multiphase**, I first thought it accidentally slipped into the oscillators, then I saw the x10 switch in the frequency range. As LFO it goes from 0 - 10 Hz, x10 to 100 Hz and that is g2 and with an extremely fast trigger like the 3840 bpm of **JW-Modules Clock** we get to 640 Hz, because with **tap trig** you can set the frequency depending on the speed of the trigger, either by clicking on tap as fast as you want it to be, or by introducing a **trigger** signal. The same applies to the **sync** input. Usually it is used to synchronize 2 oscillators, so I can insert the signal of another oscillator and its frequency controls the modulation of the module, but you can also simply click on the **tap sync** field at the desired speed. The **LFO Multiphase** generates the 4

Fundamental waveforms **sine**, **triangle**, **sawtooth** and **square**, whereby with the corresponding controller it is possible to change continuously from one to the other. This can also be controlled by CV, but here you can not only adjust the modulation **intensity**, but also whether the **transition** between the waveforms should be smooth or choppy, with the switch **discrete/morph**. With **uni/bi** you switch between unipolar and bipolar signal. The fine-tuning is actually very **fine** here, because it is not even a half step up and down. **FM** in together with the corresponding controller is not only the usual **frequency modulation**, but also the **V/Oct** input of the module when the controller is turned up completely, as we already know from the **Autodafe** modules. The **PW/PWM** knobs and the corresponding input socket determine the pulse width of the waveforms and act on **all waveforms**. This separation between fixed pulse width and pulse width modulation makes it possible to create very interesting and lively sounds, especially through the various available **phases**. Eight fixed and one freely definable phase are available. While the shifted phase without pulse width modulation is mainly optically detectable or, of course, if you use several phases at the same time, adding the **PWM** results in different waveforms and sounds. The **angle** of the free phase can be adjusted manually or controlled by CV. The **amount** of modulation can be set using the **free phi mod** slider. So I have to say, this module has positively surprised me. The sound and the variety of sounds is much bigger than I thought and I can imagine it very well in the area of bass sounds. But also experimental sounds in higher pitches can certainly be realized by the tap functions.

Speaking of experimental, the **Phasor Harmonic VCO** scared me when I first saw it. Series after series of inputs, outputs and knobs and then the graphics in the display, even the candy colors are of no use. Fortunately, he looks more dangerous than he is. Above all, we can simply **hide** individual elements, which I will do for now. **Phasor** has 2 **different context menus**, depending on where you right-click. In the **side** areas the usual menu appears, which all VCV-Rack modules have. But if you click in the **middle**, many options appear which can be set or which determine the behaviour of **Phasor**. I deactivate show **phasor**, show **path** and show **wave** one after the other by removing the check mark with a left click and the graphic is hidden. Alternatively, you can simply switch on the **low power mode**. Now click on full reset and we have a simple **sine** wave with the frequency 65. 41 Hz, which is also shown in the **basefreq Hz** display. This corresponds to a c2 and therefore **Phasor** is tuned 2 octaves lower than standard oscillators from the beginning. By changing the value in the display, simply click on the number to be changed and then move it up or down with the **mouse**, I get



value ranges from 16. 35 Hz to 261. 63 Hz, i. e. C0 - c4 and with the **octave** control I can still detune +/- 2 octaves. So we get up to c6 and down to the absolute **LFO** range and there are dedicated outputs for that, but more about that later. The octave can also be detuned by CV. For **frequency modulation** there are 2 options **linear** and **exponential** and both have their own controller and input and can therefore be used **simultaneously**. CV is the usual **V/Oct** input here.

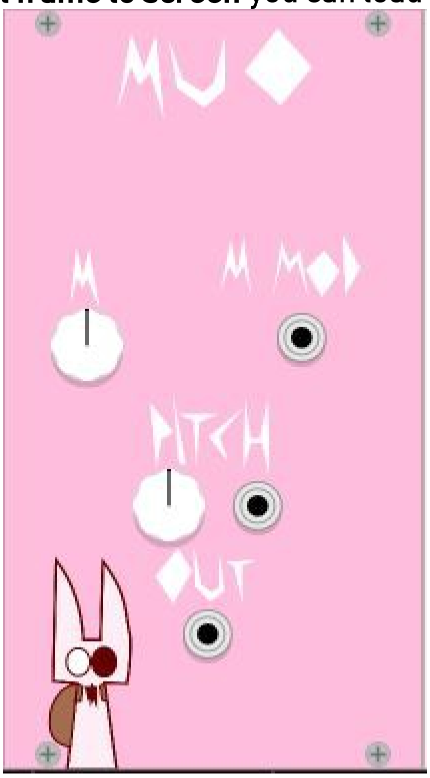
Phasor has **16 oscillators**, each with 2 **outputs**, one for the **normal** signal and one for the **LFO** signal. The latter does not work in low power mode, so it makes sense to hide the graphic elements individually. Both output signals can also be used **simultaneously** (because they actually each have an oscillator) and by left-clicking on the **+** this is closed **-** and thus the frequency is **inverted**. The sign is called **frequency sign** and can also be set randomly, but more about this later. The **red line** in the display represents 1v and is important for the setting magnitude, in principle the **volume** of the selected frequency. This can be between - and + 2 and can easily be changed by left-clicking and then moving up or down. This can also be controlled by CV. The **magnitude CV atten** slider is used to adjust the **amount** of modulation, which can be positive or negative. With **phase** the phase of the overtone can be set between 0 and 360 degrees or controlled dynamically via the phase CV input. With **frequency trim** the frequency of a harmonics can be shifted by +/- 20% to achieve **inharmonic** effects. This can also be controlled by CV and the strength is adjustable via the **trim** attenuator. Here, too, a negative signal is possible, so that the attenuator should actually be called an attenuverter. Smartass mode off! Before we turn to the right side of the module, let's have a look at the

context menu, because that's really something.
While **initialize**, as usual, resets the entire module, **full reset** resets only the center area. The next settings only make sense if more than one oscillator is working, so I click on **randomize all magnitudes**.
Again, **Phasor** differs from other modules because it has different modes for **random** selections. In the standard context menu, a random setting can of course also be generated for the entire module. In the basic setting, the harmonics are tuned normally, i. e. the 1st harmonic comes next to the Fundamental, so at c2 this is c3, which corresponds to a doubling of the frequency. The 2nd harmonic is then the basic frequency x3 and so on. With **Phasor**, however, I can increase the **interval** between the frequencies to **square**, i. e. fundamental x4 (22) corresponds to the 1st harmonic and Fundamental x9 (32) to the 2nd and so on. But that's not all, **cubic** also goes up to 3 (x8; x27. . .) and **square root** (x1. 41; x1. 73. . .). This alone can generate an enormous amount of different sounds. We already had the low power mode, with **frequency trimmers as FM inputs** you can realize a separate **frequency modulation** for each oscillator or each harmonic. For the next setting option, I activate the 3 elements of the graphic again. **Zoom out** reduces the graphic, **zoom in** enlarges it and with **zoom reset** I have the initial value again.
We know from the **additive synthesis** that any waveform can be generated with **sine** waves of different frequencies and amplitudes. **Phasor** takes advantage of this fact and therefore offers **presets** for the waveforms **sawtooth**, **square** and **triangle**, which can be selected with reset saw, **square** and **triangle**. What **random kfs** is all about, we'll come to that later. As already mentioned, various elements of **Phasor** can be created individually at **random**. **Magnitude**, **phase**, **frequency trimmer** and **frequency sign**. Each of these elements can also be reset individually to **zero** or, in the case of frequency sign, to the starting position. If you right-click in the field directly over a **magnitude** attenuator, you have the option to set this magnitude to full, half or zero and to change the corresponding sign.

If you need to go to the bathroom, have a drink or a smoke, now's a good time. Which we come to the final spurt.

To see what **tilt** does, I first choose one of the fundamental waveforms, the **sawtooth**. Controller in the middle position always means that positive and negative values can be set. Turned clockwise, high frequencies are raised, the higher the frequency, the greater the effect. Turned counterclockwise, the same happens at low frequencies. This can be automated

with **tilt CV** and a control voltage at the corresponding input. If the **tilt control** is in the center position, you can use **tilt CV** to switch between the emphasized frequencies.
Offset shifts all harmonics up or down, i. e. increases or decreases the **amplitudes**, which also changes the **ratio** of the partials within the overall sound. This is why strong offset val-ues are also noticeable by a change in sound. This can also be controlled by CV, where only values are added, i. e. the offset is either positive or negative, but not alternately.
And as promised, I'll tell you what random **kfs** is all about. Let's take a look at the **keyframe animator**, which is a very powerful tool. Initially it shows a green zero in the **kf count** display, because it does not yet have a **keyframe**. In the context menu you can **add** 1/10 or even 100 **random keyframes**. Now I decide for 10, the green 0 has turned into a green 10 and a click on **run/halt** starts it immediately. The **keyframes** pass through what you see on the display and then the oscillator **stops**. A **keyframe** is a snapshot of a sound, as it can be seen in the dis-play. However, some parameters of them can also be set. **Tot time** is the total time of all **keyframes**. **Current keyframe** shows which **keyframe** is currently playing. I can also select one here, otherwise you can simply see the changing numbers here. With **delete all kf** all **keyframes** are deleted and the green number is reset to 0. With **curr kf del**, only the dis-played **kf** is deleted. Each **kf** can have a different duration and this is defined in the field **curr kf time** for the current **kf** or **next kf time** for the following **kf**. This works as always with a click on the number to be changed and move up or down. **Snap kf** creates a new empty **kf**, which can be filled with the current screen content, for example. In the context menu there is the option **save screen to current frame** for this, with **load current frame to screen** you can load a previously created **cf** into the display. **Run/halt** starts and stops the **keyframes**. **Pause/cont** stops the playback and restarts it at the stopped location. **Reset** resets the counter back to **kf** 1 and with **loop** the series of **kf** is repeated again and again. Except for loop, all of these buttons also have a **trigger** input to automate their functions. And I'm actually done with that module monster. Try it yourself, it is certainly not the easiest module, but **worth every effort**.



Finally, I now have an oscillator that could well be described as the opposite of **Phasor**. But that doesn't mean anything about the quality. We are talking about the **Messed Up Oscillator** from **Rodentmodules**. You could call him

minimalist and in any case he immediately catches the eye with his comic design. But he's not a beeping rodent, he can roar quite well. Unlike many other modules, it is tuned to a (440 Hz) and its range ranges from a1 to a7, I can see its strength in the **bass** range, which is why I also choose the low pitch, which also makes it easier to see the waveform. This is a **mixture** of **sawtooth** and **sine** and can be modulated with the controller **m**. This causes the waveform to be folded and slightly distorted. And this can be controlled via CV. Next to the pitch control is the **V/Oct** input and that's it. A simple oscillator with an **interesting basic sound**, which certainly has its fields of application.

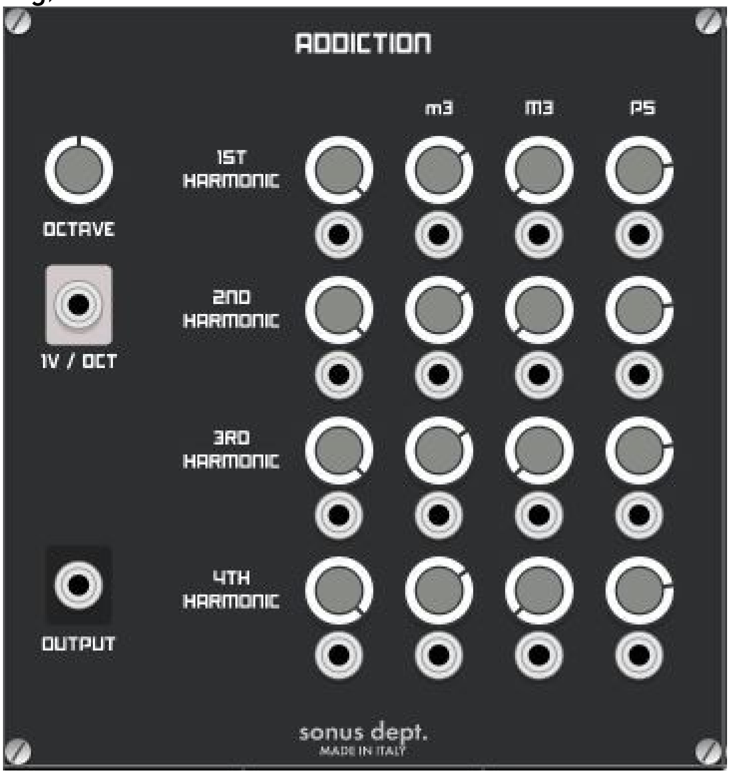
And that was episode 8, which is probably not so easy to digest today. My recommendation, let it work and then grab the modules yourself and try them out for yourself. As always you will find links to the most important websites and also to the developers of the modules presented this time.

I am looking forward to your feedback, no matter if wishes, suggestions or criticism. And not to forget, many thanks to Andrew Belt and all the module developers without whom this would not be possible. That's it from me. Have a good time.

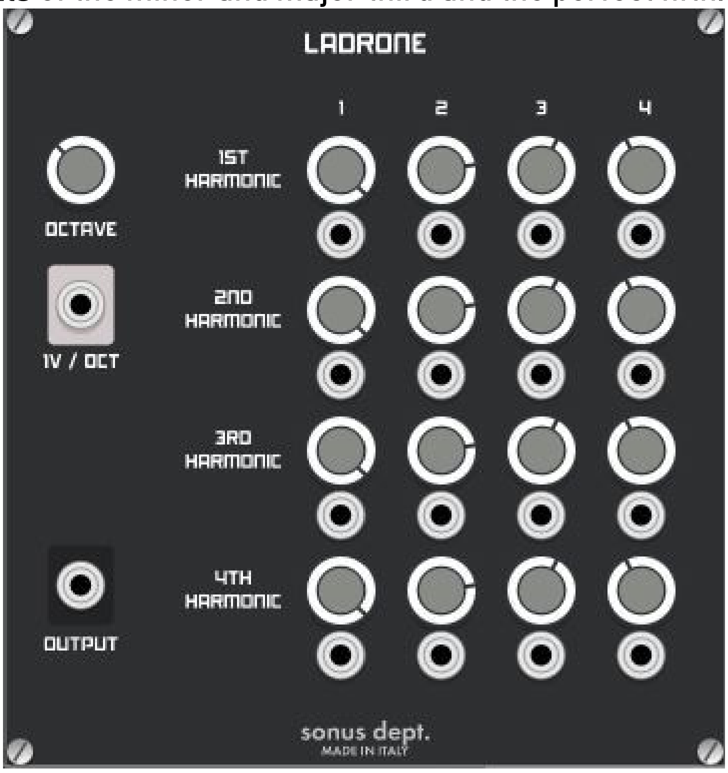
Episode 9: Oscillators - Developers with S

Hello world and welcome to episode 9 of "what does this knob do?".

We are approaching the end of the series on oscillators for VCV-Rack. Then I actually presented all currently available oscillators, if I have them. Some of the paid modules are missing, I will submit them as soon as I have them. And of course new modules will be added over and over again and of course I will introduce them as well. Now, however, I continue with developers with the letter S.

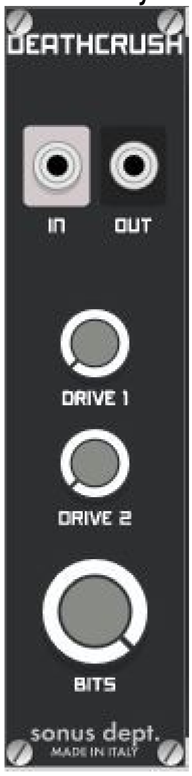


Sonus Modular has 5 candidates in the race: **Addiction** is an **additive sine** wave oscillator, or actually **16 oscillators**. Its frequency spectrum ranges from 16. 3 Hz to 4009 Hz. With the **octave** control it can be tuned in the range of +/- 3 octaves. It has one **V/Oct** input and only one output. It has **4 identical rows**, each consisting of a **harmonic** and the **intervals** of the minor and major third and the perfect fifth. Each of these oscillators has a **volume** control and this can also be controlled via CV. The basic tones are c3, c4, g4 and c5. By increasing or decreasing single oscillators, you can build different **chords** or **intervals**.



Ladrone is absolutely identical in appearance. Its frequency range is also identical. Here we have a **harmonic** and the **1st**, **2nd** and **3th** overtone, which corresponds to the 2x, 3x and 4x fundamental frequency and is therefore called this way. Basic tones are c2, c3, g3 and c4. The Fundamental tone is in any case a pure **sine** wave, the harmonics contain a kind of **phase modulation**. Here, too, the sound is generated by increasing and decreasing the harmonics. Does not sound so much like a chord due to the phase modulation and the other intervals. And it should sound more like **drone** and it does so especially in low registers and it is a perfect addition to **Addiction**.

Chainsaw is not satisfied with one oscillator either. Its range goes from c1 to c7, but can be detuned by +/1 octave for each of the **three oscillators**. Especially the **fine-tuning** is important if you have three identical oscillators. This makes everything possible, from a slight sweep to a **supersaw**. Very interesting is also the **shape control**, which makes the waveform and the sound rounder when turned to the left and sharper when turned to the right. Oscillator 2 and 3 can also be **synchronized** with oscillator 1 at the push of a button. Fortunately it has a CV in and so this can be controlled by an external trigger and can be permanent with a fixed voltage, e. g. by the **MK 1 Trigger** of **AS**. The knobs **shape** and **tune** also have CV inputs. But for an oscillator that calls itself **Chainsaw**, it is a bit too nice for me and needs some distortion, as for example his colleague **Deathcrush**.

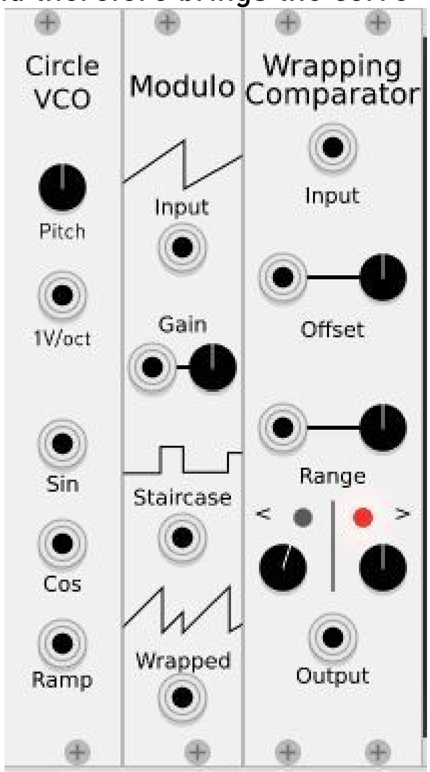


Oktagon is actually an **LFO** with **8 different phases**, which all have the same frequency, because there is only one frequency control, but a phase shifted by 45%, whereby the waveforms are not congruent and start at different times, as you can see here very nicely at the flashing lights of the cables. A slider changes the mode to **oscillator** and here we have a **sine** wave. The shifted phase only becomes noticeable when several phases are used simultaneously, because the frequencies of the phases then influence each other. The tonal range extends from c2 to c6 and can also be controlled via CV. This socket also serves as **V/Oct** input.

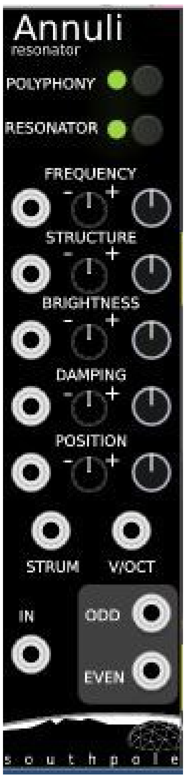
Osculum is also called **unusal oscillator**. In any case the possibilities for soundshaping are

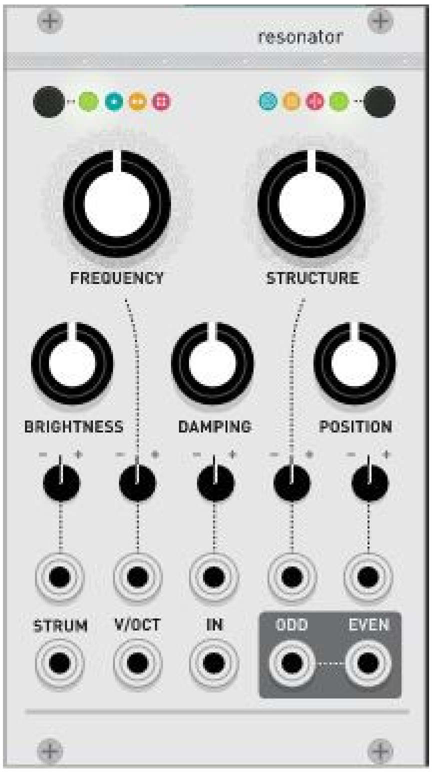
unusual, except for an octave control, which allows a tuning between c1 and c7, nothing can be influenced. It has **four outputs**, labeled wave 1 to 4 and these waveforms are all different. From 1-4 the number of harmonics increases and with it the timbre becomes brighter. In my opinion, this oscillator almost screams for the possibility of **mixing** the waveforms in a flexible ratio and then outputting them as an overall signal. This can be achieved e. g. with an attenuator, here the **8vert** of **Fundamental** and a sum or merge module, here the **Sum MK II** of **ML Modules** which can also reverse the polarity. And the sound becomes more interesting. But that's the nice thing about modular systems, you just need a starting point to shape the sound further with other modules. **Sonus Modular** goes its own way with its oscillators. They provide **interesting base material** which needs to be formed further.

The **Circle VCO** from **s-ol** also wants to be formed further and therefore brings the corresponding material with it. Because in itself it doesn't offer much, its frequency range goes from 11. 6 Hz to 6300 Hz and it produces a **sine** wave, a **cosine** wave, thus a **sine** shifted by 45 degrees and a **sawtooth**, here called **ramp**. Then there's a **V/Oct** input and that's it. But that's where his two colleagues come in. **Modulo folds** the input signal, whereby **staircase** is a coarser convolution and **wrapped** allows fine nuances. With **gain** you can adjust the intensity of the **convolution** and this can be controlled by CV.



If we also bring in the **Wrapping Comparator**, the sound becomes even more versatile. **Offset** is used to set the phase and this can also be controlled via CV. **Range** refers to the range in which the two underlying controllers become active. These modulate the range or compare its signal with the input signal and output this value at the output. No matter which input signal comes from the **Circle VCO**, the result is always a **square** wave, but with **different pulse widths** and these can be controlled or shaped by CV at different points. And the **Circle VCO** has its name, because of its representation in **Lissajous** mode of an oscilloscope, here **Full Scope** of **JW-Modules**. With these 3 modules you can also **create** wonderful **graphics**. The whole thing is called **oscilloscope music** and you can find a link under the video. https://www. youtube. com/watch?v=dnxmfnkgxiy I find the whole thing **very exciting** and will deal with it even more closely. But also the sound possibilities are quit good.





Annuli from **Southpole** once again corresponds to a module from **Mutible Instruments** for the euro-rack, namely **Rings**. And there is also another version for VCV-Rack, namely **Resonator** from **Audible Instruments**. Interestingly, this was not displayed as an oscillator, but since it corresponds to **Annuli**, I introduce it here. Let's start with the entrances. Each of the 5 knobs has its own CV input and an associated attenuator. Besides the usual **V/Oct** input for a CV signal, there is a **strum** input for a trigger signal and the **in** input for an audio signal. Ideally, a signal should be present at all three. The strum signal only works in **polyphonic** mode. Here the sounding **voice freezes** for a moment, it fades away, a new signal starts on the next **voice**. This creates the effect of **strings** or other resonance bodies struck shortly after each other. The **signal in** excites the resonator to sound.

The module has 2 outputs, whereby in **polyphonic** mode the **even** harmonics are output on one and the **odd** harmonics on the other. I just mentioned the polyphony mode. The module can be operated in three different modes. **Green** led corresponds to **one voice**, **yellow** led to **two voices** and **red** led to **four voices**. For these modes there are also three variants of resonators. **Green** corresponds to a **modal resonator**, which simulates some frequencies being raised and others being damped, as when striking one string or another resonance body. **Yellow** corresponds to **sympathetic strings**, which simulates the resonance of unattached strings or resonating bodies, e. g. harmonics.

Red led corresponds to **modulated** or **inharmonic strings**. This is a classic **Karplus Strong** synthesis, that is, the excitation signal is sent to a **comb filter** with an absorption filter that simulates the **multiple reflection** of a wave that results from the vibration of a string and that is absorbed at its ends. However, this is extended by a delay-compensated allpole absorption filter for more drastic **plucking** effects, delay-time modulation modeled after the sound of instruments with a **curved bridge** (such as sitar or tanpura), and all-pass filters in the delay loop that shift the position of the partials and simulate the **tension** of a piano string. And inharmonic tones.

The basic frequency is set in the range of 5 octaves via **frequency**. The function of **structure** depends on the choice of resonator. In the modal resonator it determines the frequency ratio between the partial tones, in sympathetic strings mode this parameter controls the frequency ratios between all strings, if modulated or inharmonic strings is active, this parameter controls the amount of modulation and detuning of the partials.

Brightness adjusts the level of higher harmonics in the signal by the simultaneous effect of a low-pass filter on the exciter signal and the attenuation filter on the rest of the potentiometer. Low Values simulate materials such as wood or nylon. High values simulate materials such as glass or steel. **Damping** controls the cooldown of the sound.

Position controls to which point of the string / face the excitation is applied. Applying the excitation in the middle of the surface will cause the straight harmonics to cancel each other out, resulting in a "hollow" sound reminiscent of a **square** wave. This reminds a bit of **PWM** or comb filter effects.

As always, there are a few special features hidden in the **context menu**. On the one hand we have the mode **FM voice**, which allows typical **FM** sounds a la **DX 7**. **Quantized sympathetic strings** is a variant of the sympathetic strings, in which only tones that match the Fundamental tone are heard. **Reverb strings** is a subtype of modulated/inharmonic strings with a high reverb content.

And then there's **Disastrous Peace Mode**. This was originally an **easter egg** from **Mutable Instruments**, which was not so easy to activate. In the VCV-Rack versions, however, it can be easily activated in the context menu and we hear an organ sound. **Annuli** also changes its name and the name of the controls, which makes it much easier to work with this mode. **Odd**, **even** and **V/Oct** are the same. **In** becomes **fx in** and an effect or modulation signal can be inserted here. **Strum** becomes **env trig** and the internal envelope is triggered again at each trigger pulse. The **polyphony** button becomes **chordsize**, but it seems to have no effect. Instead of the resonator versions we only have different effect types, **formant filters**, **chorus** and **reverb**, whose power is controlled by **fx amount**. **Frequency** becomes **root note**, which defines the root note of the chord. Using **structure/chord type** we set the distribution of the tones within the chord. **Registrations** simulates different registers of an organ and thereby changes the timbre, i. e. also what **brightness** did before. **Decay/attack** is responsible for the **decay** and **swelling** of the sound and finally **fx amount** for the influence of the selected effect on the sound. **Disastrous Peace** is a nice encore, but actually it doesn't fit with the rest. **Annuli** or **Resonator** is a module for all sounds created by a form of vibration, be it strings, fur or solid surfaces. And it's just great at that. For me **one of the best** modules in VCV-Rack.

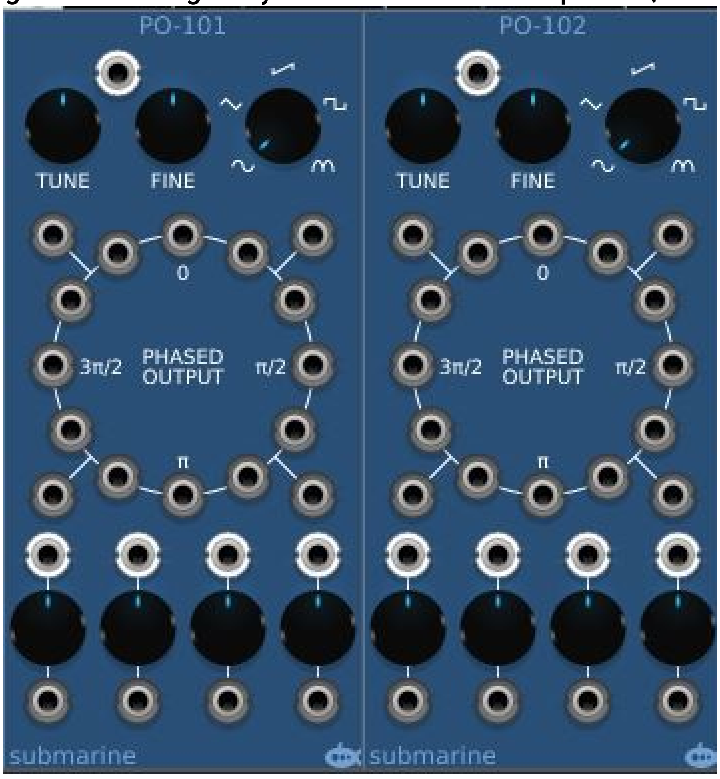
Squinky Labs is also a developer of extraordinary modules. The **Functional VCO** is not so special in principle. It is simply a variant of the **VCO-1** from **Fundamental**, which needs much less CPU and therefore a good and important module. Everything else is absolutely identical, so I give myself the further introduction and refer to my corresponding video.

The **Chebyshev Waveshaper** (this was about the 1st version, now we have **Cheby II**) is not yet officially available as I make this video. Therefore many thanks to Bruce Frazer, who made it available to me in advance for this episode. **Chebyshev** polynomials enable the **digital** generation of waveforms with any overtone structure at very low computing power. In this module the first ten **Chebyshev** polynomials are used to generate the first ten **harmonics** of the harmonic series. The oscillator has a frequency range of 16. 4 Hz to 8820 Hz, with the **finetune** control you can even reach 14700 Hz. Initialized it shows c5, so that it is already 2 octaves above most other modules. The **preset** button allows you to switch between three settings, one adding all harmonics and one setting all but the Fundamental to zero. The third setting is always the last one you used. The **proportion** of the individual polynomials is determined by the small black knobs and can also be controlled by CV. This alone allows for very lively sounds, but as always, it only becomes really exciting with modulation. The **mod** controller is assigned to the corresponding socket and enables **exponential frequency modulation**, the **LFM** controller including socket **linear frequency modulation**. The **gain** control is responsible for the waveshaping and has 2 modes **fold**

and **clip**.

If the led of this slider is off or green there is no difference between the two modes. If you turn the gain control up until the led is red, you can hear and see very clear differences. **Clip** simply cuts off the signal while **fold** is folding the frequencies. The **even** and **odd** controls are used to set the proportion of even and odd harmonics. **Slope** controls the fading out of high harmonics. Both **slope** and **gain** can be controlled via CV, and all harmonics can be controlled via the **eg** socket.

An external signal can be inserted into the **ext in** input, which then runs through the waveshaper instead of the internal VCO. Especially with modulation **Chebyshev** shows what he is capable of, even in initialized mode a modulation with **Caudal** from **Vult** is enough to teach **Chebyshev** how to speak. If **even** and **odd** could now also be regulated by CV, that would be the greatest thing. Maybe there will be an update (this and more is in **Cheby II**). But even so, the sound is **convincing** all along the line and is extremely versatile.



At first glance, the oscillators from **Submarine** are frightening, but on closer inspection they are not complicated at all. The **PO-101** has a frequency **range** of 11. 6 Hz to 6300 Hz and can be detuned by +/- 3 half steps with **fine**. It offers **sine**, **triangle**, **sawtooth**, **square** and **rectified sine** waveforms. It has one **V/Oct** input and **20 outputs**. 16 of these have **solid phases** and four **adjustable phases**, which can also be controlled by CV. The solid phases are on the one hand at a distance of 15 degrees and on the other hand there is another phase after every 22. 5 degrees. As with all oscillators with several phases, this is of course only heard when several phases are used **simultaneously**. Especially because of the four free phases and the possibility to control them with a controller, interesting sounds are possible.

The **PO-102** is absolutely identical, except for its frequency spectrum, which is why it is primarily an **LFO**.

We already experienced the **PO-204** in the **very cool patch contest #1**. Its frequency spectrum ranges from 1. 4 Hz to 6300 Hz, which is why it can be used both as **VCO** and **LFO**. It can also be detuned by +/- 3 half steps with **fine**. It has only one oscillator but **4 outputs** at which different signals can be tapped. Every signal can have a different waveform, **sine**, **sawtooth**, **rectified sine**, **triangle** and **square** and these can be traversed continuously and this can be controlled by CV. Each of the 4 **voice**s can have a different **phase** and this is also infinitely adjustable via CV. In addition, each signal can be **multiplied** up to 16x, which increases the frequency. This can also be controlled by CV, but here the steps are clearly audible. Also here I miss a summing output, for which I can regulate the ratio of the individual **voice**s, but also here one can make do with an attenuator and summing module.

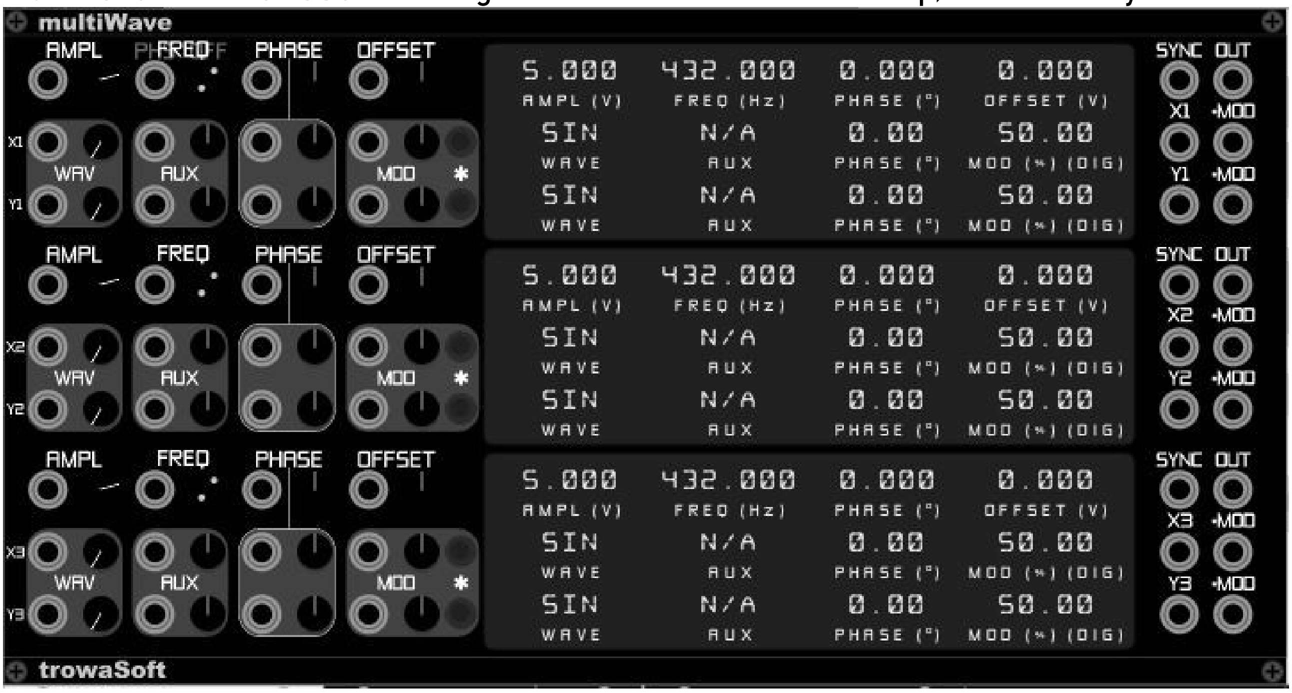
And that was episode 9, again with some very interesting modules. Next week as announced, the last episode with oscillators for the time being, then we'll continue with filters. As always you will find links to the most important websites and also to the developers of the modules presented this time.

I am looking forward to your feedback, no matter if wishes, suggestions or criticism. And not to forget, many thanks to Andrew Belt and all the module developers without whom this would not be possible. That's it from me. Have a good time.

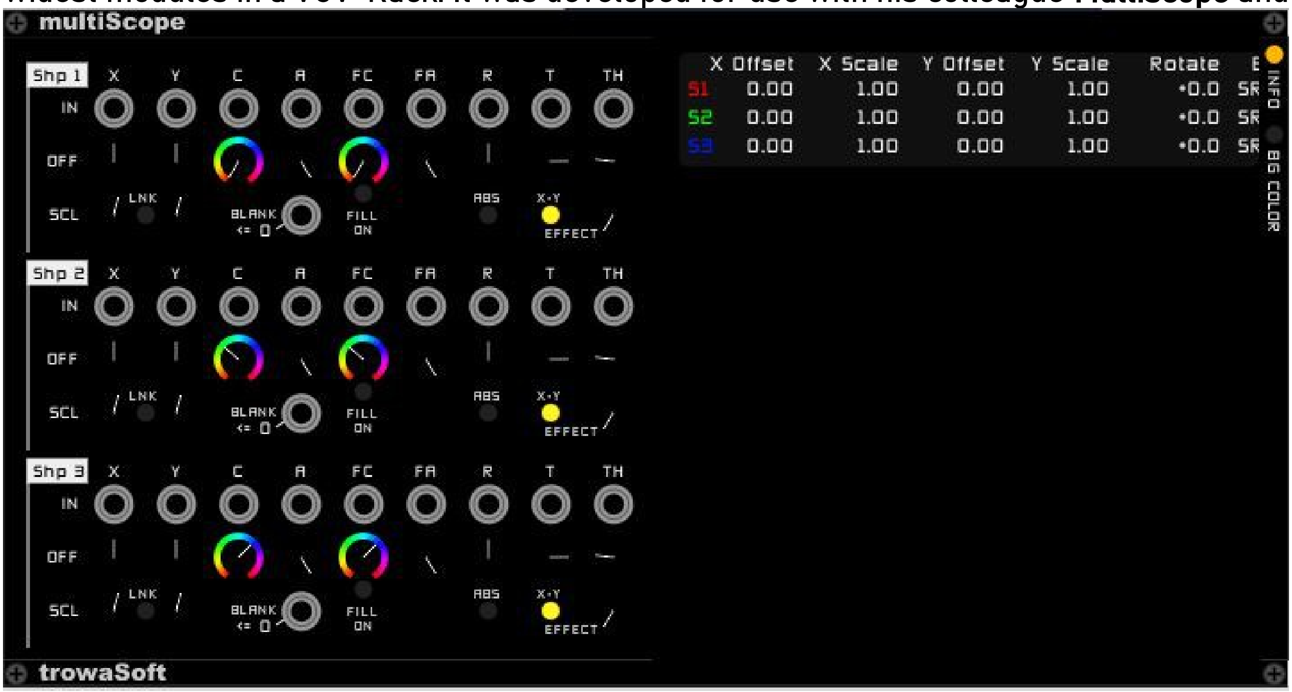
Episode 10: Oscillators - Developers T-V

This episode was originally supposed to be the last one with oscillators for VCV-Rack. In the meantime, however, a few oscillators have dropped in, which fill another episode. Neverthe-less, we will now continue with developers of T-V.

Multiwave from **trowaSoft** is a huge module. With a width of 48 hp, it is definitely one of the



widest modules in a VCV-Rack. It was developed for use with his colleague **Multiscope** and

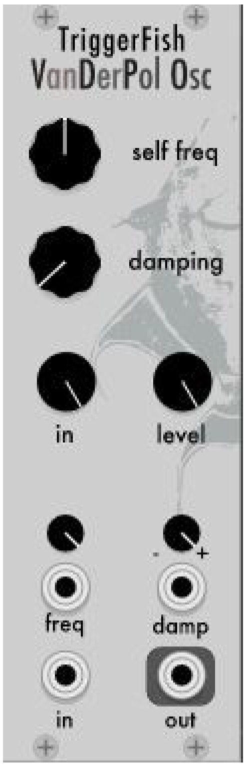


also has the same dimensions. The **Multiwave** uses **3 oscillators**, each of which has 2 sepa-rate signal paths. However, both the oscillators and their signal paths are identical, so in principle we have a **six-voice digital oscillator**. The oscillators are numbered from 1 to 3 and the corresponding **voice**s are always x and y.

Let's start with the inputs, which are all on the left side except for the sync socket. All inputs are intended for control voltage, so CV and each one is assigned to a knob. **Sync** makes an exception here as well, because this is not controllable. **Ampl** stands for amplitude, in principle the volume of an oscillator, but the control range is from -10 to +10 volts, so the volume maximum is reached at both + and -10 volts. **Freq** stands for frequency and is therefore intended for **frequency modulation**, but also as **V/Oct** input. The frequency range is from 0 Hz to 20,000 Hz and can be adjusted via an endless controller. If the shift key on the computer keyboard is pressed simultaneously, a coarse detuning is possible, with the ctrl key a finer one. **Phase** shifts the waveform of an oscillator, or a **voice** along the x-axis, so that phase-effects can be generated even with one oscillator. **Offset** works similarly, but shifts the signal along the y-axis. Both have a value range of +/- 10 volts. Each **voice** can have a different waveform, these are adjusted with **wav** or via CV. There are **sine**, **triangle**, **sawtooth** and **square** waveforms. The **aux** controller only works if a signal is present in the corresponding CV in. Then it **inverts** the **sawtooth wave** or **influences** the **pulse width** of the **square** wave. With the **CV** input signal you have a pulse width modulation. With **mod** an **amplitude modulation** is activated, also here the value goes from -10 to +10 so that a negative modulation is also possible. A CV signal can also be connected here. The small button next to the knob determines the type of modulation, but this only works when a CV signal is present. The default setting is **digital amplitude modulation**, if the white dot lights up, we have a **ring modulation** in which the amplitude is not modulated evenly.

The **display** in the middle not only shows the currently set values, but also serves for **direct input of values**. This works for all values **except wav** and **aux**.

Beside the already mentioned **sync input** there is also a **sync output**, with which the oscillators can be synchronized with each other and also a synchronization loop can be generated. Each **voice** has its own output for the **unmodulated** signal and for the **modulated** signal. Each designated as **x** and **x-mod** or, **y** and **y-mod**. According to the description, not everything is implemented yet, which explains e. g. why the aux control does not work with every waveform. I hope it will also get a sum output for all **voice**s, because with that you could get a lot more out of this oscillator. Of course this can be improvised again, but here it takes some effort. In principle, I would also like to see a more compact design, although I think the display and the values displayed are pretty cool.



The **VDPO** of **TriggerFish Elements** has just arrived in the Plugin Manager. The abbreviation stands for **Van der Pol oscillator**. According to wikipedia, this is "an oscillatory system with non-linear damping and self-excitation. For small amplitudes, the damping is negative (the amplitude is increased); above a certain threshold value of the amplitude, the damping becomes positive, the system stabilizes and changes into a limit cycle".

The developer specifies the whole thing on his github side: "while it can self oscillate, best results are obtained by feeding it another oscillator at the input and playing with the self-freq, damping and input level to go from harmonic to inharmonic and chaotic."

In practice, the **self freq** controller controls the natural frequency of the oscillator. For this purpose an input socket including attenuator is also available, so that **frequency modulation** via CV is also possible. The frequency range is from 2. 9 Hz (with fully opened damping) to 4410 Hz (damping closed).

Damping controls the non-linear damping of the oscillator. This influences both the tuning and the harmonic and inharmonic content of the output waveform. The further the damping is turned on, the lower the frequency becomes. A slight damping is also available for the initialized setting. If you want to switch off the damping completely, the corresponding control must be at the left stop. In this position we have a **sine** wave, which slowly becomes a kind of **smoothed square** wave by turning the controller clockwise. There is also a CV socket and a controller for damping. However, this is an attenuverter that outputs modulation in the middle position 0 and negative values rotated to the left and positive values rotated to the right. The **level** slider determines the volume of the module and the in slider, how much the input signal is added at the input in the self-oscillation.

Because of the **complex real-time calculation**, this oscillator needs a lot of CPU and you certainly have to invest a lot of time to get to know its possibilities.

And here comes **Dexter** from **Valley** and I have to admit he's still a little scary although he won me a **very cool patch contest**. And I'm sure I'm just going to scratch the surface here, so I'll refer you to the excellent tutorials by Andrew Mercer and Omri Cohen. As always, you can find the links under the video. **Dexter** also has a great manual.

Dexter's not easy to put in a pigeonhole. In principle it is of course an **FM** oscillator with **4 operators**. But it also offers **wavetables**, **phase distortion** and **numerous sync options**, not to mention the **modulation** possibilities via CV. **Almost everything is also controllable via CV.**

Okay, we're approaching this monster carefully through the **master controls**. All settings in this area affect (more or less) the entire module. The most important thing in an **FM** module is the interconnection of the **operators**. Each operator can be regarded as an oscillator and their interconnection or mutual influence is called an **algorithm**. In **Dexter**, an algorithm describes how each operator modulates and synchronizes another operator and to which output its signal is sent. If you use a dedicated **FM** oscillator, such as the already introduced **FM-OP** from **Bogaudio**, you can assemble these algorithms yourself with any number of operators. The best known **FM** synthesizer **Yamaha DX7** had 6 operators and 32 algorithms. **Dexter** offers **23 algorithms** with its 4 operators, where 12 of these algorithms are common for both **voice**s A and B and the remaining 11 algorithms are assigned either A or B, which allows independent tones from each **voice**. The algorithms also determine the **synchronization** source for each operator, where operators are synchronized with their **parent** operator by default. This can be changed to **"neighbor"** in the **context menu**, which synchronizes an operator with its right neighbor in the module. (i. e. 1 -> 2, 2 -> 3 and 3 -> 4.) the algorithms are selected with the **algo**. button and are shown below as a **schema**. The operators are represented as **square**s of 1-4, lines show how they are connected and the letters a and b show which operator goes to which output. **Operator 4** is always shown in **orange**, which symbolizes that it can **modulate itself**. The strength of this feedback can be adjusted by the large orange control **fb** or controlled by CV, whereby the small orange control **fb** is responsible for this. I have to emphasize the **color design** of **Dexter** at this point, because this ensures that you can quickly find your way around despite all the complexity. The user **manual** also contains a numbered **diagram** of all implemented **algorithms**, which simplifies the selection of a specific algorithm enormously. Unfortunately, no numbers are assigned to the algorithms on the module.

Below the graph you can see 2 buttons, with **LFO** all operators of **voice** a are set to function as LFO and with **reset phase**, the phase of all operators is reset, so that they start their cycle again at the same time. Finally, each **voice** has its own **V/Oct** input. If only one of the inputs is occupied, an input signal can of course only be processed here. The other **voice** behaves like any oscillator without an input signal and generates the selected keynote.

This keynote can be set separately for both **voice**s. Both **voice**s have initialized the Fundamental c4 and can be detuned with **oct**. in steps of +/- 3 octaves. The current octave is also displayed by number, great. With **coarse**, an octave can be steplessly detuned by +/1 and again by +/- 10 cent with **fine**.

Voice A can also create **chords** with up to **6 notes** plus two **unison** modes with **5 or 7 notes**. The chords are selected with the **"chord"** slider and the chord name is also displayed. The chord notes can be inverted several times with the **"invert"** slider. By default, if the chords are larger than 4 notes, notes can be inverted by one octave or completely inverted so that they are behind the last note of the chord. The **notes** in the chord can be detuned if some **voice**s (sharper) and others (flatter) are tuned. The **detune** control goes quite deep, which means that some chords are completely transformed. If both outputs, Al and Ar, are used for the output signal of **voice** A, the result is stereo and the notes of the chord are distributed between the two. To prevent unwanted distortion, the output signal becomes quieter and quieter, depending on the number of notes in the chord. All chord parameters can also be controlled via CV and this can also be set with the associated attenuverters. The **manual** also contains a table of the possible **chords** and their **structure**. We will see the function of the **shape** slider in detail in the operators, here a global setting is made.

The **bright** slider adds high frequencies clockwise and reduces the number of frequencies when turned counterclockwise. In the middle position, this controller is neutral.

I already discussed the **fb** controller earlier. These functions can also be controlled via CV and these inputs also have an attenuverter.

In the area of master controls we also have the **outputs**. By default, A is the main output, but this can be changed via the context menu. As already mentioned, **voice A** has a right and left output for a **stereo** signal, as always the left output is also configured as a mono output. **Voice B** also has a **mono** output and **each operator** has its own output. So everything you could wish for, because each operator can also be controlled individually in volume.

Okay, take a quick breather before we proceed with the **operators**.

These are absolutely identical in design. Operator 4 can also modulate itself, but the setting for this is not made in the operator itself, but via the orange knobs.

The most important settings are not visible at first glance. To **access** it you have to **click** on the button next to the **gear symbol**. This will light up red to indicate that the **wavetable menu** is now active. The **wavetable button** also lights red. The **wavetable opal** is set by default. **Dexter** provides **35 different wavetables** and a description can be found in the **manual**. The **wavetable** is selected using the **blue table knob** under which the name of the currently set **wavetable** can also be read. The two **drop-down** fields under **sync mode** and **shape mode** refer to the **wavetable**. In principle, a **wavetable** is a series of several waveforms. Usually, related waveforms are used or at least waveforms for which you have a common term, such as basic, which contains the standard waveforms **sine**, **triangle**, **sawtooth** and **square** plus pulse width modulation. The waveforms are read one after the other at a set **speed** and used by the operator as the basic sound. You can **manually** select a waveform within a **wavetable** using the **large purple wave slider** (in the normal operator menu) or using a CV signal at the wave inputs and the small purple knobs.

Sync mode is responsible for this **speed**. **15** different synchronization **options** are available; the classic **hard sync** is preset, in which the reading speed corresponds exactly to the set frequency. There is also a table in the operating instructions on the **modes available** for this purpose.

Reading is done with the so-called **read phasor** and in principle this is done continuously and can be displayed as a **ramp**, i. e. a very flat rising **sawtooth**. However, **shape mode** changes the nature of this ramp. The default setting here is **bend**, which bends the ramp, so that the first half of the **wavetable** is read faster (due to gradient) than the second half (due to slope). By the way, I find it quite helpful to visualize the whole thing in order to be able to better explain the behavior of the **read phasor**. With the **large red shape slider** in the normal operator menu, the amount of deformation can be adjusted. In **bend** mode, this means how much the ramp **deflects** and thus the ratio between the reading **speed** of the first and second half of the ramp. **12 different shape modes** are available and I recommend to have a look at the **manual** which explains all 12.

In the **wavetable menu** we also see the buttons **mod 1&2** and **mod 3&4**, which are assigned to the 4 ochre colored **free attenuverters** mod 1-4. With one click one selects the two modulators to be processed, which is immediately indicated by a red light. A parameter to be modulated can be selected next to each mod by means of a drop-down field. **13 different parame-**

ters are possible. Most of them have no dedicated CV inputs.
Let's go back to the normal **operator menu**. The **mul** control multiplies the basic frequency set in the master controls section. Its control range is from x 1/8 (actually 1/8 of the fundamental frequency) to 24 times the fundamental frequency. The result always remains harmonious. If you want an inharmonic sound image to create metallic sounds, for example, this is possible with the coarse and fine controls, which work exactly as in the master controls section.
I already mentioned the controls **wave** and **shape** and with **level** the volume of the operator is determined or controlled by CV.
The **pre fade** button ensures that the output signal of the individual output of the respective oscillator is independent from the level button.
If the mode **post shpe** is active, the **frequency modulation** takes place only after processing the **read phasor** with the shape control. By default, the **read phasor** reads the **FM** modulated **wavetable**.
With **sync** the operators are synchronized with each other, whereby the **weak** button causes that the synchronization does not take place over the entire cycle of the signal, but only in the first quarter. And the **LFO button** turns the operator into an LFO.
The lower half of an operator is reserved for CV inputs and the associated attenuverters. As you can clearly see from the colors, **pitch**, **wave**, **shape** and **level** can even be **modulated** by several controllers **simultaneously**.
Take a deep breath. **Dexter**'s overwhelming. No matter if sound diversity or modulation possibilities, **Dexter** is unique and in my opinion **plays in its own league**. I can't even begin to imagine how to come up with such an idea and how to implement it. The only thing I would wish is that it needs less CPU, but I have no idea if this is even possible with this variety of possibilities. I take my hat off to Dale Johnson for this masterpiece.

And that brings us to **Vult**, hello Leonardo! If you look into the oscillator/VCO section of the Plugin Manager, you will notice that **Vult** is also listed with some filters. Representative for all **Vult** filters I have here **Freak** from the **Vult-Compacts** series, which contains all **Vult** filters. In principle, it is quite possible to oscillate filters with resonance themselves, so that they produce a sound even without an input signal. But I wouldn't call them oscillators because they simply

lack the implementation **V/Oct**. This may not always be relevant and you can also convert a **V/Oct** signal to another voltage type, e. g. with **Rescale** from **AS**, but I have not found a way to reproduce a simple sequencer melody so that it sounds exactly like a "real" oscillator. So I will discuss the filters as such and not with the oscillators. Because of course **Vult** also has extraordinary oscillators.

I would like to start with **Vessek**, which is both **freely available** and included in the premium series. However, the two variants do not differ from each other, whereas this is the case with other modules. **Vessek** consists of **2 identical oscillators**, where oscillator A can modulate oscillator B by **frequency modulation FM** and **amplitude modulation AM**. But more about this later. The **tune** switch defines how the tune control should behave and has three modes. **Fine** detunes by +/- 1 semitone, **coarse** detunes steplessly by +/- 1 octave, **semi** detunes also by +/- 1 octave, but in semitone steps. **Vessek** is tuned to c1 by default and can be tuned with the **oct** control in the range of +/- 3 octaves. Thus the direction is clear - **Vessek** is primarily predestined for low pitches. With **master tune** you can tune the whole module in the range of +/- 1 octave. Thus **Vessek** goes deep into the cellar and can of course also be used as an **LFO**.

The **PW** controller controls the pulse width of **all waveforms**, which means that the wave doubles with a **sawtooth** and an asymmetry with a **triangle**. But you have to go into the third octave to make the waveforms look like that. Of course, this should not be a criticism, **Vessek** calls himself an **analog oscillator**, so the waveforms are not perfect either and that's a good thing. **Pulse**, **sawtooth** and **triangle** are available. Both oscillators can be set **indepen-dently** and the output signal always consists of a **mixture** of the two. The mixing ratio is ad-justed with the **mix control**, with oscillator A at 100% on the far left and oscillator B at the far right. In the middle position both are involved in the output signal with 50% each. **FM** is used to set how much the output signal of oscillator A should modulate the frequency of oscillator B. **AM** is the same, only for the amplitude of oscillator B. **Sync** allows a stepless synchronization of the signal from oscillator B to oscillator A. If the controller is far left, we have no synchronization, it is far right, we have a perfect synchronization, a **hard sync**. The **shaper** slider is used to add high frequencies to the mixed signal, resulting in distortion. **Offset** adds a positive or negative voltage to the wave before it enters the **shaper**, affecting the frequency response, changing the **ratio** between high and low frequencies.

The **fade** knob, including the two sockets placed on it, is assigned to the **gate** input and only functions if a signal is applied to it. Then **fade** controls the **decay** time of an internal envelope, which is output at the output connected to it. An **external signal** can also be fed in at input **ext**. This can be a modulation signal or any control voltage. The signal output here can be used for both internal and external modulation. The **glide** parameter also has an output that can be used for further modulation. The intensity of the glide effect, i. e. the combination of tones, ranges from 0, i. e. complete separation, to 100% blurring of the boundaries. It can be used to adjust the gliding to the next tone and there are 2 modes that can be set using the slider. **Always** means that the effect is applied to each note and **skipgate** means that no effect is applied when the **V/Oct** input signal coincides with the gate input signal.

The great feature of **Vessek** and some other **Vult**-modules is the **modulation section**. Here we have **6 inputs** for control voltages of any modulators and one attenuverter each to adjust the level of modulation. The modulation target can be **any parameter** of **Vessek**. To select it, simply **click** on one of the **letters a-f** and then move the slider you want to be controlled by CV. Then all that remains is to set the attenuverter corresponding to the letter, with both positive and negative modulation values possible, and the desired parameter is controlled by control voltage. Simple and ingenious. And the best thing about **Vessek**, of course, is the sound, which I would describe as **bold analog.** This is not a reserved celebrating gentleman, but a real hooligan with dirt under his fingernails and ruffled hair. But also with an irresistible grin, so you have to like him right away.

Actually I should finish this episode here now, because what should come now and not completely go down? And that's why **Noxious** just got here. Optically, **Noxious** is very similar to **Vessek**, but **Noxious** is a **digital oscillator** and generates its sounds by **FM synthesis** and **phase modulation**. And it's part of the paid **Vult-Premium Package**. (but I can highly recommend its acquisition) and it is available in both **monophonic** and **polyphonic**. I present here the polyphonic version, since this is identical to the monophonic version except for the additional **voice**s.

The parameters of the module are optically combined as **areas**. **Osc** are all settings concerning the main oscillator, **FM** is the modulator area. **Chaos** generates random events, **phase** is responsible for phase modulation, **mix** adds additional sound sources and **glide** adds a portamento. The **poly** section is of course only available in the polyphonic version. And of course not to forget the ingenious **modulation** section with the freely configurable CV inputs and attenuverters.

Noxious is also a specialist in low frequencies and therefore its frequency range is identical to **Vessek**. The tuning range is also identical.

Wave controls the **even** and **odd** harmonics of the main oscillator. On the far left it generates a **square**-wave signal, on the far right a **saw** shaft. **Harm** defines the number of harmonics used by the main oscillator. On the left side only the base wave is output and thus a **sine** wave, on the right side the full waveform. This control has a similar effect to a low pass filter. A slider can be used to switch between audio oscillator and **LFO** operation.

The **tune** switch in the **FM area** determines how the frequency of the modulator is defined. There are 2 modes for this, **ratio** selects only frequencies that are a multiple of the main oscillator frequency. The control range is from 1/2 to 8 times the main frequency. **Free** selects any frequency between 1/2 and 8 times the main frequency.

Freq defines the frequency of the modulator. The behavior is influenced by the parameter **tune** (see above). **FM** sets the level of **frequency modulation** applied to the main oscillator. **Wave** defines the wave type used by the modulator. **Saw** produces a more edgy result that is interesting for aggressive sounds. **Sine** produces a softer modulation. Ext. **FM** is connected

to the **FM** input jack and controls the influence of the external signal on modulation.
The **chaos section** is used to give the sound random variations. To **trigger** modulation, a gate signal must be present, otherwise these controls do not affect the sound at all.
Spread defines how much the parameters can move after a gate signal. At the maximum level, the parameters can move from minimum to maximum. **Settle** defines the time that the parameters remain outside the current value. A low setting can help to create percussive sounds by performing a very fast transition. Larger values offer longer transitions. When the maximum is set, the parameters are not reset to the original value, but are reset to a new random state by a new gate.
All parameters in the **phase section** influence the phase of the main oscillator. **Overtone** folds the phase to add overtones, distorting the signal. **Body** transforms the phase exponentially, with both positive and negative values possible, resulting in a thinner or richer sound. This makes the sound thinner the further the knob is from the center position.
Boost changes the amplitude of the phase, turned to the left it increases, turned to the right it decreases, resulting in a fuller sound on one side and a thinner sound on the other. **Feed** adds positive feedback to the phase. The sound is reminiscent of an open resonance control on a filter.
In the **mix section**, we can use sub to add a suboscillator to emphasize the low frequencies. **Noise** adds white **noise**.
The **glide section** is the same as **Vessek**, but has no dedicated socket for an external signal. Also the **range** of freely configurable CV inputs is like in **Vessek** and of course you can also specify glide as destination.
Like in **Vessek**, we have a **V/Oct** input, a gate input and a master output for the final signal of the entire module. At **Noxious**, the already mentioned **FM** input is added. The monophonic version would be covered, everything else only refers to the **polyphonic** version.
In the **poly section** we have **three identical voices**, all branched off from the main oscillator. Each **voice** has its own separate output, but with the level control the sound can be mixed through the main output. However, this does not affect the volume of the signal at the output socket of the **voice**.
The additional **voice**s are only active when their output or CV inputs are connected. Each **voice** can be **tuned** individually according to the setting on the main oscillator slider.
Each **voice** has its own **V/Oct** input, so different input signals can be used **simultaneously**. And since **each voice** also has its **own gate input**, several gate signals can be used simultaneously, so that each **voice** can also react to the chaos engine setting at a different time.

And how harmful is **Noxious**? To remain by the **analog**y, **Noxious** is dangerous and insidious. He admits that he is dangerous, but while we are preparing for the mace's blow, he has already cut our hamstrings with a sharp stiletto and made us helpless. Okay, that's where the role-player in me has gone through, but I think you know what I mean. **Pure Distortion Oscillator** shines under its name and already tells us what to prepare for. The sound of **Noxious** (especially in the polyphonic version), does not kill you - because we have prepared ourselves for this – but it shows so many and subtle finesse in the interaction of the individual components that I am helpless and do not know how to describe it. Maybe just like this: "I've been trying for hours to generate a sound that's just garbage and I didn't succeed!" I don't know if **Noxious is** the best-sounding oscillator I know, but it never gets unpleasant and is therefore so **dangerous**. The addiction therapist in me just called me up and said, "you talk like an addict!" I think he's right.

And that was episode 10, which was extremely subjective and lurid. But sometimes that has to be and if I can't honestly give my opinion, I don't need to make my videos at all. Maybe you think I should stop or maybe you have ideas and suggestions or just want to praise me for my awesome videos. No matter what, let's hear from you. Next week I'll continue with filters and in between I'm working on an update with the newly added oscillators. This video will appear outside the series entitled". . . Update 1". As always you will find links to the most important websites and also to the developers of the modules presented this time.
And not to forget, many thanks to Andrew Belt and all the module developers without whom this would not be possible. That's it from me. Have a good time.

Episode 11: Filters Part 1

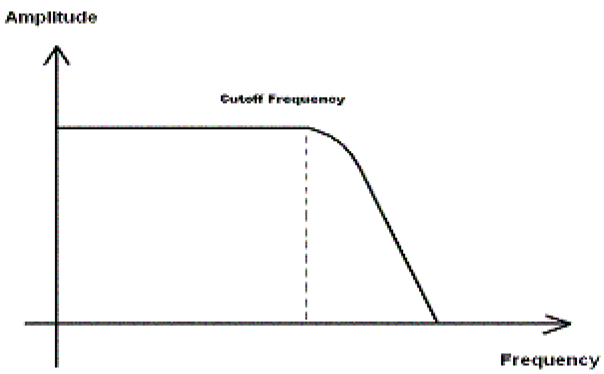
Hello world and welcome to episode 11 of "what does this knob do?".

With 1 week delay comes the first episode about filters in VCV-Rack. I could hardly speak last week, so unfortunately I could not produce a video. But this week there is a double episode with 25 filters. As usual, I go alphabetically by the name of the developer. I had to change my setup to be able to test the filters better. I'm sending a simple sequencer tune from **Synthkit's 4-Step Sequencer** to the **VCO-1** from **Fundamental**. Its signal goes directly to the VCA input of **Slap** from **Vult** and to the filter input and from there back to the VCA. In this way I can compare the unfiltered signal with the processed one and see what the filters do with it. Of course, I also look at the filter modulation and take on the one hand the **LFO-1** from **Fundamental** and on the other for random modulations **Caudal** from **Vult**. As before, I use to visualize **Full Scope** from **JW-Modules**, the **Bogaudio Analyzer**, and the **LogInstruments Spectrum Analyzer**.

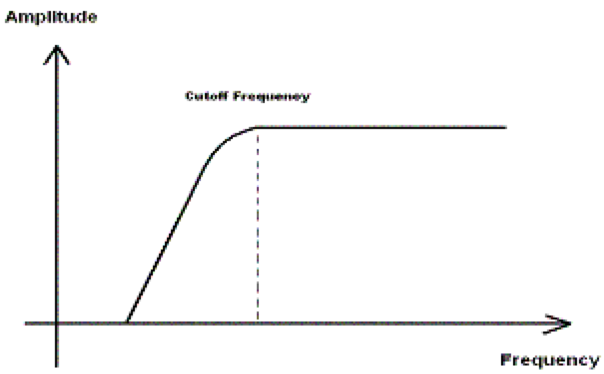
First of all a few basic things about filters. In the following I will use the term **VCF** (volt controlled filter) again and again when I speak of the module and abbreviate the different filter types as **lp** = lowpass filter / **hp** = highpass filter / **bp** = bandpass filter / **nf** = notch filter / **cf** = comb filter …

The VCF is an **essential part** of the **subtractive synthesis**, as it is mainly used in **analog** synthesizers. The VCF removes certain frequency ranges from the signal of the oscillators (VCO) and thus determines the timbre of the sound. This filtering out of frequencies has also given **subtractive synthesis** its name. The influence of a VCF is on overtone rich waveforms, such as **sawtooth**, stronger. The **sine** wave as a pure fundamental waveform has no overtones and thus nothing can be filtered out of this wave.

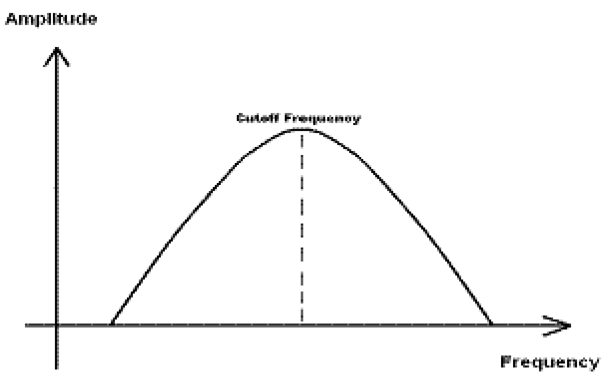
Subtractive Synthesis



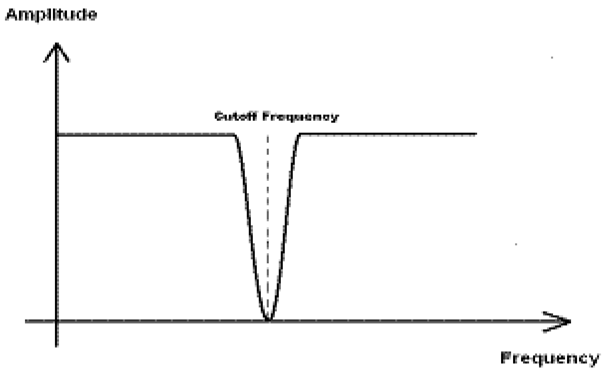
The **lp** is the most common filter form in synthesizers. This filter mode clears the frequencies **above** the set **cutoff frequency**. By eliminating the high frequencies, the sound sounds duller and deeper, while gaining more of warmth and fullness. The application of the low-pass filter is good for everything that should sound **warm** or **fat** - like bass or wide area sounds.



An **hp** is the technical counterpart to the **lp** and thus **erases** the **low parts** of a sound. This makes the sound **thinner** and **sharper**. The **hp** is essential for all types of **cymbal** sounds, such as hi-hat, crash, ride, etc.

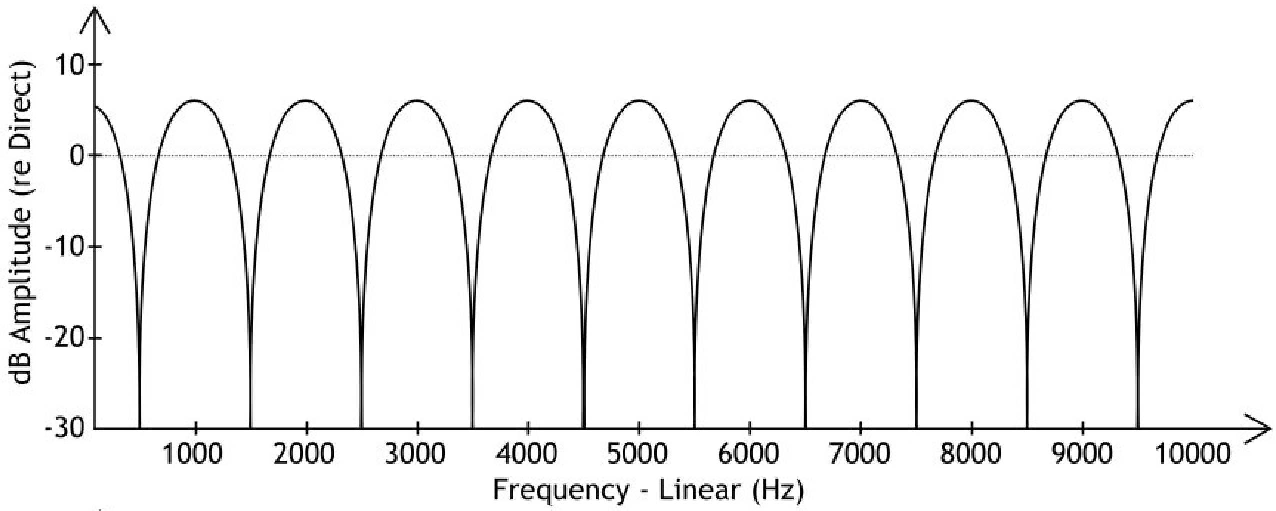


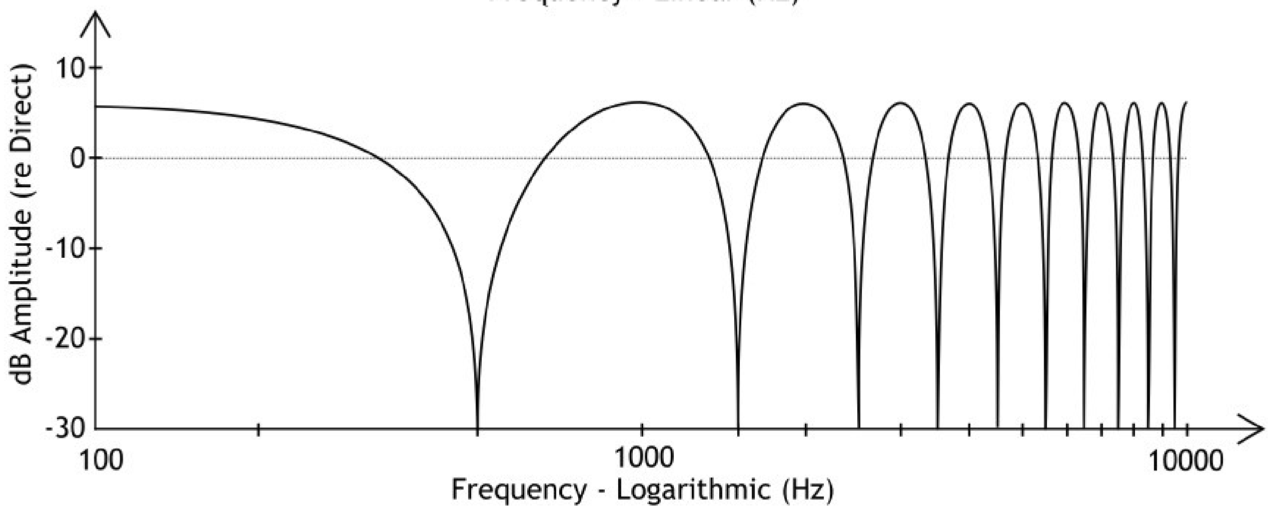
Technically, the **bp** can be realized by connecting one **lp** and one **hp** in **series**. In doing so, the **bp** passes the frequency range passed by both filters. If the cutoff frequency of both filters can be set separately, then very steep **bp** can be achieved. **Bandpasses** with **high slopes** are called **formant** filters. A formant area is a filtered center band that can be used to very well form **vowels**. If drastic sound changes or even vocal sounds are needed then the **bandpass** is the best choice.

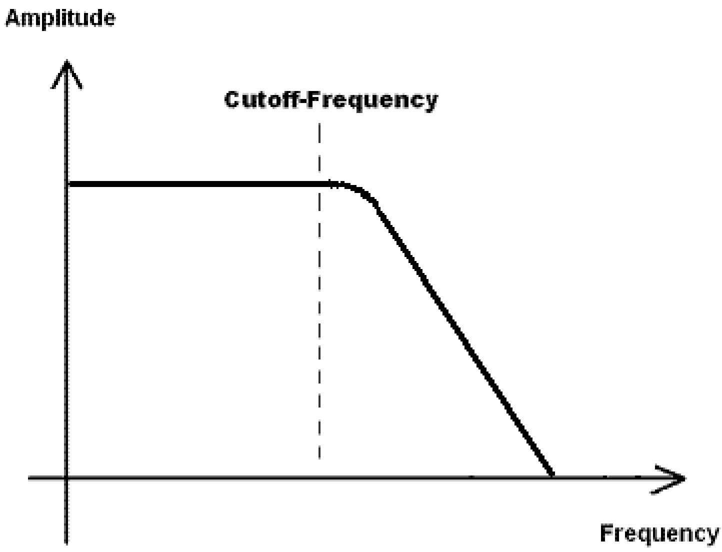


The technical counterpart of a **bp** is the **nf**. An other designation is **band-stop filter**. The **nf** is the **parallel** connection of an **lp** with an **hp**. Through this circuit appears a frequency gap - the so-called stopband - in the audio signal. The **nf** is an ideal tool to **remove** certain interference **frequencies** from an audio signal. These can be for example **humming**, but also **sibilants** when singing.

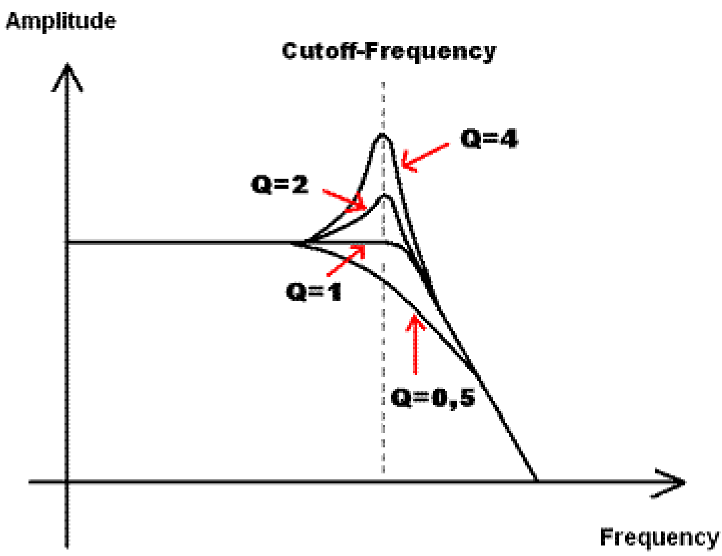
A **cf** is a filter that filters out **signals of groups** of a certain frequency. Unlike lp and hp, it filters several frequencies at the same frequency spacing or otherwise by a formula determined frequency spacing (e.g., logarithmic). The amplitude response (level over frequency) of the frequency response of the **cf** has a comb-like appearance, whence the name comes from.



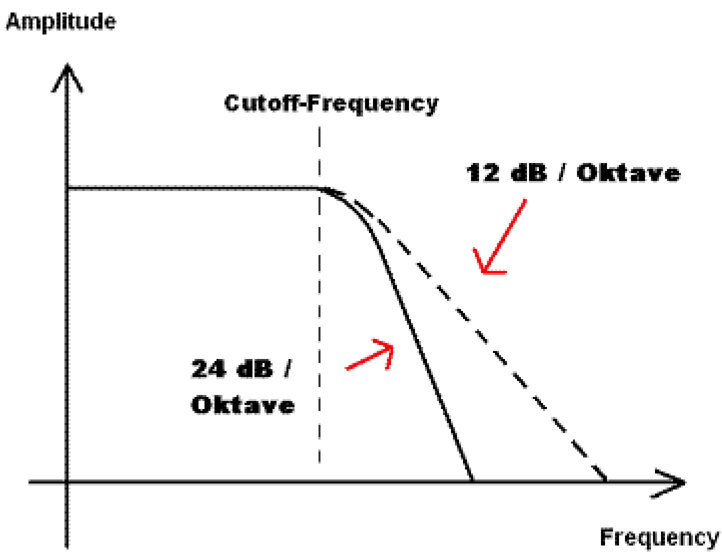




The most important parameter of a VCF is the **cutoff frequency**, which determines the frequency limit from which to filter. A slow full turn on and off of this regulator is called a **filter sweep**. In doing so, different tones are passed through.



The 2nd parameter present in most VCF is the **resonance**, sometimes referred to as **q**. The resonance knob increases the range around the cutoff frequency. At maximum resonance, most filters produce a **sine** tone with the pitch of the adjusted cutoff frequency, which is called **self-oscillation**.



One term that is repeatedly encountered in VCF is **slope**. A filter does not cut off frequency ranges abruptly, but they become weaker as they move away from the cutoff frequency. This attenuation has a logarithmic curve and is measured in decibels per octave (**db/octave**). The higher the **slope**, the higher the filter effect. The **slope** can also be increased by successively switching a plurality of similar filters. In this case, the edge steepnesses of the individual filters

are added together.
For example, the series connection of three low pass filters with respective edge slopes of 6 **db/oct** gives a low pass with the slope of 18 **db/oct**. Another description of the **slope** uses the term **pole**. **A 1-pole filter** corresponds to a filter with a slope of **6 db/oct** - accordingly, a **4-pole filter** stands for a filter with a slope of **24 db/oct**. The sound of the legendary minimoog is decisively shaped by such a sound. The VCF modules for VCV-Rack usually emulate clas-sic hardware VCF, which I will go into the individual modules, if I have this information.

It starts with 4 modules from **AS**, which are part of the paid **drums & filter** package, which can be purchased in the Plugin Manager. For this price you get **24 modules**, in addition to the **4 filters** still **20 drum modules**. So a real bargain.

But let's take a closer look at the filters. The **VC Filter** is in both **mono** and **stereo** available, but otherwise the two are not different from each other. The frequency range is specified be-tween 40 Hz and 12 mHz and it can be switched between operation as **low-pass** filter and **high-pass** filter. In addition to the usual parameters **frequency** and **resonance**, it has a mix attenuator, with which you can adjust the **ratio** between **unaltered** and **filtered** signal. This vastly upgrades the otherwise quite simple filter. As usual with filters, all parameters are controllable by CV. The filter packs well and has a characteristic of **12 db/oct** changes can be heard immediately, but it never sounds uncomfortable and the resonance does not resonate. Nothing exciting, but nothing bad. Just **rock solid**, like many modules from **AS**.

If you want to emphasize or weaken certain frequencies, a good way is to use a **Fixed Filter Bank**. A filter bank consists of several parallel filters of different cutoff frequency, which allow - similar to the **additive synthesis** - the targeted manipulation of certain frequency ranges. Depending on the application, the deepest filter may be a lowpass and the highest a highpass, but in general all filters are bandpass filters. The module of the same name from **AS** has **12 frequency ranges** with associated attenuators and in addition the

possibility to steeplessly add a **lowpass** or **highpass** filter and this also via CV. With this makes it is possible to achieve very interesting effects that are almost in the range of a formant filter. The characteristic of the two controllable filters I estimate at **6 db/oct** The regulators work very precise and effective, as it was not to be expected at **AS**.

And sometimes even a **Fixed Filter Bank** is too much and you just need a **parametric equalizer**. The **Paraeq** has a frequency range between 100 Hz and 12 kHz. The desired frequency is selected with the corresponding knob and either lowered or raised with the corresponding attenuator. The amount of modulation is determined by the level knob. The **Paraeq** works clean and effective, but for my taste CV input jacks are missed.

In summary, I can only repeat myself, **rock solid modules** that are nothing special, but do what they should do.

Autinn Bass is a bass synthesizer but without an oscillator and therefore primarily a filter. The signal from the oscillator is introduced via the **osc** input. But before no **gate** signal is applied to the corresponding input, nothing is heard. The gate signal triggers one of the internal envelopes. With the **white button** one chooses between **gate** or **trigger** mode. When **gate** is activated, the note ends with the end of the gate signal. If **trigger** is activated, the note will be played until the next note. That sounds like sustain, but it is not. Single notes can be accentuated. For this, a gate signal must be introduced at the **accent** input, where the **accent** can be either at the **beginning** or at the **end** of a note. The **amount** of accentuation is adjusted with the **accent** knob. As mentioned earlier, there are **2 internal envelopes**, one for the **volume** and one for a **filter sweep**. **Env mod** refers to the filter sweep or its influence on the signal. If the attenuverter is at the very left stop, there is no filtersweep, rightmost it is maximum. Its vanishing is set with the **decay** knob. Here, too, we have extremely short decay on the left and maximum on the right. The **cutoff** control works very subtly without signal at the accent input, with signal you can clearly hear the signal change. The real filter sweep effect only really comes into its own when the **resonance** control is also turned up a bit. The two **light bands** symbolize the envelopes. The first light is for the **attack** time, the 2nd is not occupied according to the description, the 3rd is for the **decay** and the 4th indicates when the envelope is not active. The center placed indicates when a note is accentuated. The lower light bar shows the volume envelope, which we can not change and the upper one the filter sweep envelope. The **light** next to **osc** indicates when the volume of the injected oscillator is too high and then turns **red**. On the right side there are the CV inputs and the associated attenuators. The characteristic of the module seems **6 db/oct** to be. The question that arises to me is, does it need an extra module for filter sweeps? Because that really seems to be the only function.

Let's see what his sister **Flora** can do. According to the description, this filter has a slope of **24 db/oct** and is based on the **classic analog transistor ladder filter**. Since you still hear nothing after the connection of inputs and outputs, of course, the first handle is after the **cutoff** controller. This opens the filter slowly and rounds the wave already slightly. **Resonance** works very subtly when the **cutoff** knob is fully turned on, but transforms the wave into a **sine** wave while retaining the basic shape. Turning the **cutoff** control slowly back on, the **resonance** is much clearer. If you turn the **drive** knob all the way down, you fade out the input signal and you only have the **self-oscillation** of the **resonance**. If you turn the **drive** control back on, you will notice how the input signal becomes more and more dominant in the output signal. All 3 controls can be controlled by CV and also have an attenuator for the degree of modulation. The **cutoff CV input** is also designed for a **V/Oct** signal, so you can feed an **external oscillator** or a modulation signal and thus additionally modulate the output signal. If I'm here for example feeding the output of **Bass**, I get very interesting sound images. I think you should use these two filters **together** and they will really show what they are capable of.

Bidoo describes its filter **pErCO** as "simple **lp**, **bp**, **hp** filter". It has the usual parameters **frequency** and **resonance**, although there is a CV input jack for both, but the degree of modulation can only be adjusted for the frequency. Outputs are available for **lp**, **bp** and **hp**, the characteristics of the **lp** should also be **24 db/oct**, with the **bp** and the **hp** probably **6 db/oct**. The **resonance** is **not** enough for **self-oscillation**. This filter is good and does not break anything.

lIMbO is called **4th order stereo lp ladder filter**. This is equivalent to **4-pole filters** or **24 db/oct**. In principle it is the **lp** range of **pErCO** with a few extras. In the initialized state you can see already in the **Analyzer** that the two signals match. This changes as we turn up **resonance**. Then **lIMbO** becomes much quieter, which can be compensated with the gain control. By default, the module is **linear**, and the gain control and the associated CV input jack affect the output volume. However, with the switch under the gain button, the VCF also sets to **non-linear**, which applies a **hyperbolic tangent function** to the signal, and **gain** adjusts the strength of this function. The result is a more **extreme resonance** that can distort the signal and **almost** approach the range of **self-oscillation**. Especially with external modulation by CV very interesting filter sweeps are possible here. As a result, he is no longer as nice as **pErCO** and also well suited for heavier sounds.

The **Blamsoft XFX F-35** has no less than **35 filter modes**. It is chargeable, but can be purchased via the Plugin Manager (now it is free!). It would be beyond the scope of this video to

go into any detail, so I refer you to the **Blamsoft website**, which **lists** all the filters. The link can be found under the video as always. Most filters have different characteristics, such as **2 Sallen-Key-Filters** (e.g. used in the **Korg MS20**), **8 State-Variable-Filters** (e.g. **Oberheim**), **16 Transistor-Ladder-Filters** (e.g. **Moog**), **1 Diodes-Ladder Filter** (e.g. **EMS**), but also exotics like **3 Dual Resonance Filters**, **2 Comb Filters**, **Frequency Modulation**, **Ring Modulation** and **Bitcrusher**. The controls are the same for all modes. **Drive** increases the **volume** but also ensures **saturation** of the signal. **Frequency** and **resonance** are the usual parameters, with the frequency range from 20 Hz to 20 kHz. All three parameters are **controllable by CV** and also the **strength** of the modulation is adjustable. Whereby both a **positive** and a **negative modulation** is possible. The **mix** knob determines the **ratio** between the input signal and the filtered signal, which is basically a dry / wet control and this can also be controlled by CV. And even the choice of the filter mode can be controlled not only by clicking on the display, but also by CV. At least now I am thrilled by this filter. He can not do everything, but extremely much and he can do things that other filters can not. I can only think of a term **"the swiss army knife of the filters in VCV-Rack"**.

Carbon from **CharredDessert** is called an emulation of a **Moog** resonance filter, so we can start with a **4-pole filter**. The range of the **cutoff** frequency can be between 20 and 6000 Hz and for this there is also a display which always shows the set value. The usual two parameters, **frequency** and **resonance**, can both be CV modulated, but the modulation level can not be adjusted. The **resonance** goes into the **self-oscillation** and can be really uncomfortable, here I would want but then an attenuverter, but of course you can do it externally. Nothing works without an input signal, so the module can not be used as an oscillator. But he is again a small fine module with good sound.

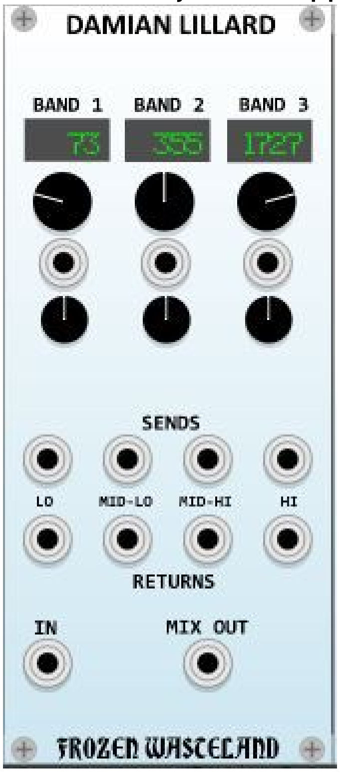
Apparently, **dBiz** does not want to tell us anything about its **Multimode Filter**, as he did with the oscillators. It has a lot to offer. I think he is a **4-pole filter** or and actually **2 identically constructed filters**. Each of these filters has its **own volume control** and you can also set their mixing **ratio** and via the **fade** input also via CV. This is especially interesting if the two sides have **different filter types**. Because that's the essence of a **multimode** filter. Here you have the choice between **lp**, **bp** and **hp**. The **frequency** knob works precisely but subtly. It has 2 CV inputs and associated attenuators, called **FM1** and **FM2**. Of course that means it can be modulated simultaneously by one envelope and another controller. The resonance is called **q** here and does **not** go to the **self-oscillation**, where it gets a bit more bite with the parameter **drive**. Also **q** and **drive** are controllable by CV. Each filter has its own input and output, that is, both can be used **separately**, but there is also a **mix** output and here is the actually much too good filter at once interesting. The combination of all parameters and

different types of filters will still not make a predator, but there are many different timbres and timbres possible, which always sound musical and that's also a lot of value.

Dr. Res of **Friedrichs Audio** is one of his **free** modules and unfortunately has no description, so I have to speculate here too and go back to the **lp** of a **4-pole filter**. However, this is also a **multimode** filter with the types **lp**, **hp** (estimated **12 db/oct**), **bp** (estimated **6 db/oct**) and **notch**. The **cutoff** frequency regulator works cleanly and effectively, and of course this can also be controlled by CV. What gives the filter its name are the two controllers **dr** for drive and **res** for resonance. **Drive** is the volume control of the module and the **resonance** goes into the **self-oscillation** and that also without input signal. Of course, these two can also be modulated by CV. As you might expect from this name, **Dr. Res** is an expert in **resonances**. Especially with modulation they can really shine.

The **Evil Turtle Productions' RezzoEQ** is part of a pay-what-you-want package available only through the developer's website. If I am not mistaken, we are dealing here with a **1-pole** filter. However, the developer calls it **"spring resonance filter"**. This system uses a very fast spring-rate algorithm, controlled by the input signal and the frequencies, acting either as a **low-pass filter or** as a **high-pass filter**. **Frequency** is the usual cutoff frequency control, which of course can be modulated by CV. **Energy** controls the amplitude as well as the frequency and thus makes the signal softer and quieter or in combination with the **damping**, the resonance shriller. **Damping** is available as an attenuator, turned to the right increases the attenuation and the signal becomes dull and quieter. Turned to the left, the high frequencies increase more and more, until a strong **resonance** sets in, which can be modulated even further with **energy** and thus sometimes sounds like a **distorted guitar**. **E-curve** changes the envelope of the energy parameter from short pulse to a gate that lasts until the next pulse comes. Both filter types have the same structure, but of course differ in sound. Really strong resonances are obtained, of course, if you use both filters together, which I think is also provided, after all, there is only one output. **Self-oscillation** and feedback are daily bread for this VCF and with modulation you can really make it sing. Too bad that only the frequency and the damping can be controlled by CV and single outputs, I would also like. Nevertheless, the **RezzoEQ** is an **exceptional** filter

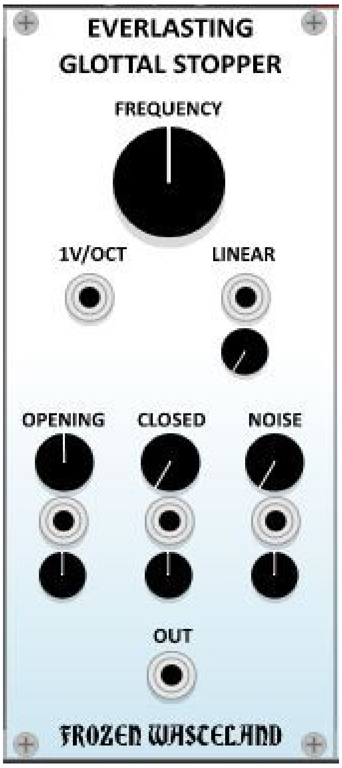
that certainly has its application.



Since we are just on the extraordinary, **Damian Lillard** of **Frozen Wasteland** is anything but a normal filter. Therefore, we can save the consideration of its characteristics. It has **three frequency bands**, which can be infinitely adjusted between 15 and 8400 Hz and this also by CV.

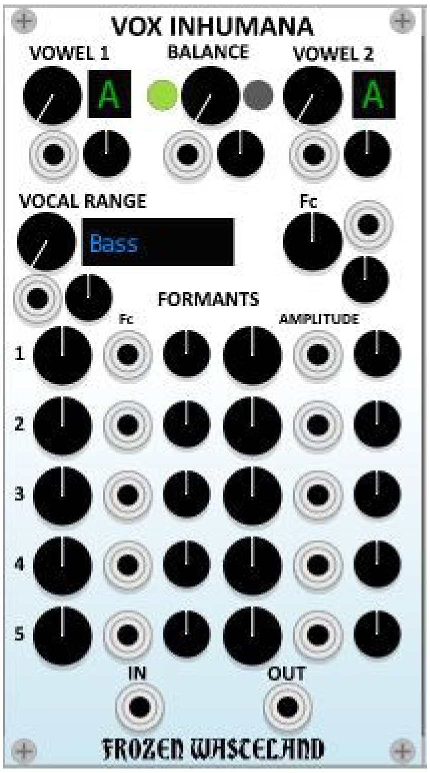
But what makes it so special are the **sends** and **returns**, which are divided into different frequency ranges. This means that each frequency range can be assigned its **own effect**. Two identical stereo delays and a modulation of the frequency bands with **Caudal** from **Vult** already produce a complete ambience piece. This inconspicuous-looking filter is actually a **playground for creativity** and much more than just a filter.

And if that were not exotic enough, the **Everlasting Glottal Stopper** puts one more on top.



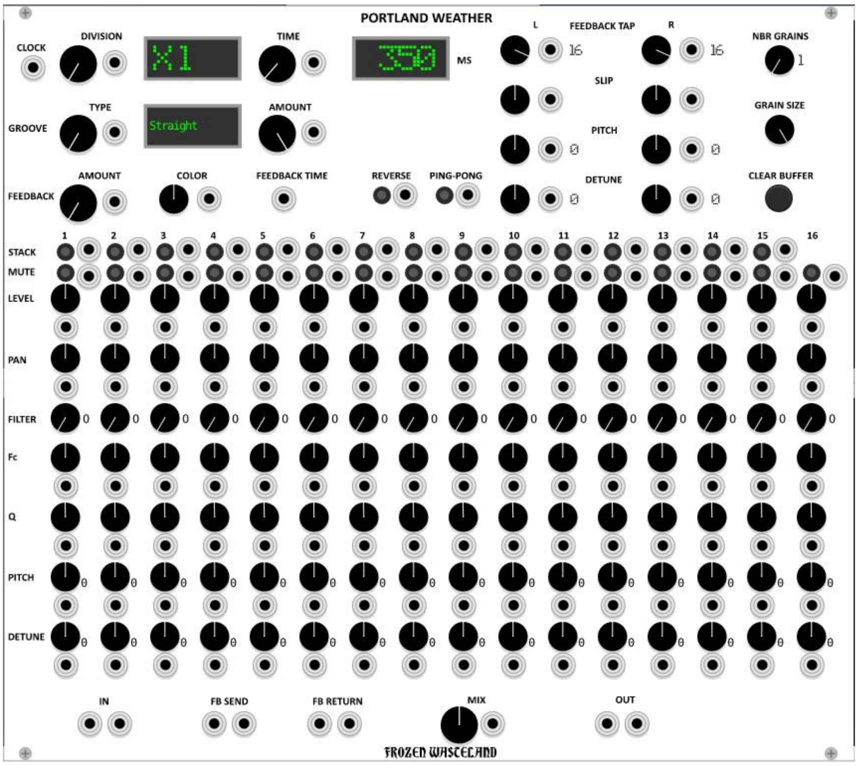
Glottal stop is a **phonetic term** and describes a "consonant formed by the sudden, un**voice**d release of a vocal fold closure". For this, a Mr. **Rosenberg** has made a calculation and defined a **glottal pulse**. This complex calculation forms the basis of this module, so here we have not only the usual frequency button, but also three parameters that come from phonetics. Roughly speaking, a sound is created by an interplay of breathing and articulation in the **throat**, **mouth** and **nasal cavity**. **Noise** emulates the breathing, **opening** and **closed** the opening and closing of the articulation organs. All three parameters can also be controlled by CV. Apart from a **V/Oct** input, through which the module can also be used as an oscillator, there is also an input for a **linear frequency modulation** with associated attenuator. This can also be used as an **input for an audio signal**. It is remarkable what is possible with these few parameters. However, in the description of the **Everlasting Glottal Stopper**, it says that it fits well with the **Vox Inhumana** module, so let's take a look at that next.

This is called **formant** synthesizer, so it's about **vowels**, these are formed by the interaction of formants. The first two formants are important for the understanding of the vowel, and the

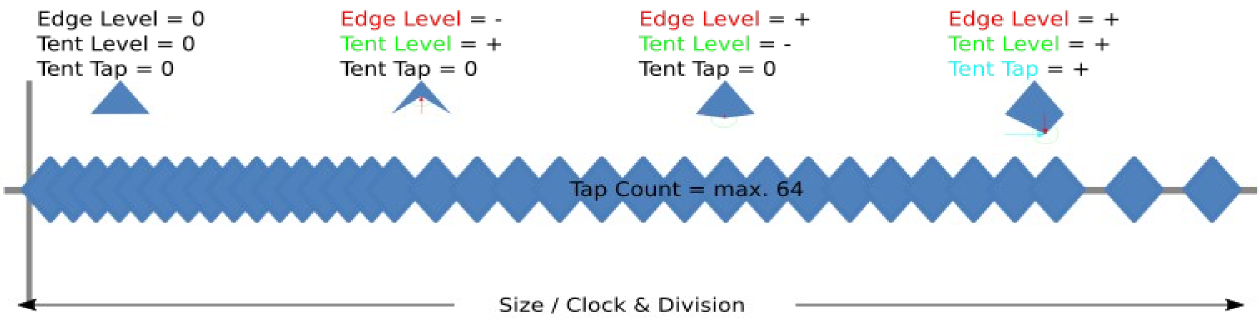


others are rather individual differences, such as gender, age, etc. Each vowel in **Vox Inhumana** consists of **5 formants**, i. e. 5 fixed frequencies and associated amplitudes. The letter A in pitch soprano e.g. from the frequencies 800 Hz with 0 db / 1150 Hz with -6 db / 2900 Hz with -32 db / 3900 Hz with -20 db and 4950 Hz with -50 db. In addition, each frequency has a bandwidth within which the vowel is still recognized. The basis for this is a table, which can be found on the github page of **Frozen Wasteland**. **Vox Inhumana** allows you to change the frequency of each formant by +/- 50% and the amplitude up to twice. And both can be controlled by CV and the modulation strength can each be set with an attenuator. Two vowels can sound at the same time, these can either be selected fixedly or via CV, whereby the **direction** and **speed** of scanning through the vowels can also be set by means of an attenuator. Both vowels can be individually sounded or mixed, depending on how the **balance** is set and also for this there is a CV input including attenuator. With the parameter **fc** you change the frequency of **all formants** at the same time and this can also be done via CV, of course, the intensity of the modulation is also adjustable here. Finally, you can still select the general vocal range with **vocal range**. The possibilities here are **bass**, **tenor**, **counter-tenor**, **alto** and **soprano** and even that can be controlled by CV and an attenuator sets the direction and range. Due to the many modifiable parameters and the modulation options **Vox Inhumana** can almost speak, but only in vowels and here then of course the **Everlasting Glottal Stopper** comes into play, because this produces **consonants**. Do we now have a complete speech synthesizer together? In principle, yes, but you will probably have to spend a lot of time to get the desired result. At any rate, while experimenting, I had some ideas and what could be said better about a module than it **inspires**.

And because it's just so nice now comes an extraordinary filter from **Frozen Wasteland**, **Hair Pick**. According to the description, this corresponds to the **combfilter section of the Intellijel/Cylonix Rainmaker**. Together with **Portland**



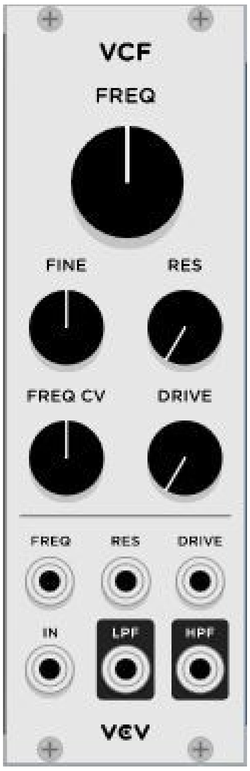
Weather, we have a complete **Rainmaker Clone** in VCV-Rack. But back to **Hair Pick**. All controllable parameters can also be controlled by CV. It has 2 inputs into which different signals can be fed and each one has its own output. If only the left input is occupied, the same signal can be tapped in both outputs. In principle, even a third signal can be introduced via the **V/Oct** input. However, this also affects other parameters, because each increase of the signal fed here by 1 volt halves the size of the comb filter. To understand what that means you have to deal with the **basics of a comb filter** again and I have created (based on this module or the rainmakers) a diagram for clarification.



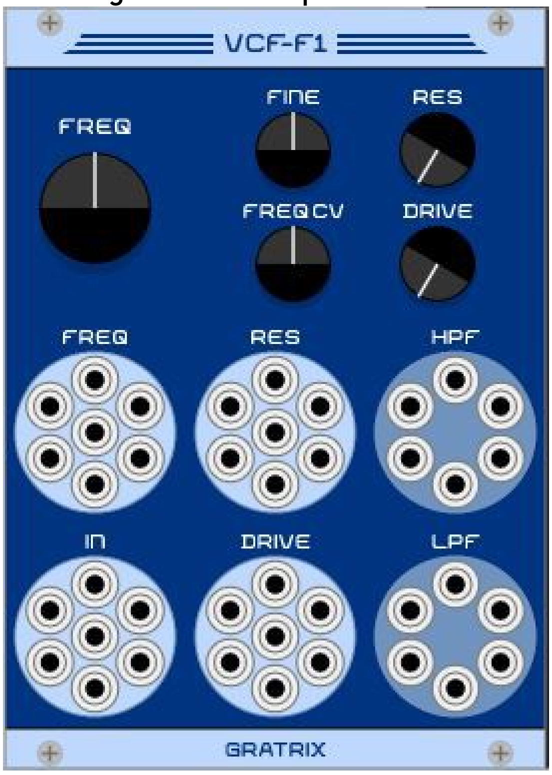
We can see here that the **size of the comb** can be defined in 2 ways, via the **size parameter** or via a signal at the **clock input**, which is divided by fixed factors with the division parameter. The smallest value is 1, so no division and the largest 256. I go for example. with a clock sig - nal with 480 bpm (equivalent to 2 Hz) in the clock input, divide this by division by 2 is the comb filter size 1 Hz. 1 volt in the input **V/Oct** halves the comb filter size, 2 volts halves this again and this works in factor 2 high x so on, where x corresponds to the volt strength. The smaller the size of the comb, the more of the original signal is left, with increasing negative volt values the original signal becomes less and less. And even if this is not intended, here is of course any signal possible, including LFO, envelope or whatever you use to create tension.

In addition to the comb filter size of course, the **distance** of the individual teeth of the comb is relevant and whether they are evenly **distributed**. Here we come to the **patterns**, of which there are **16**, where the numbers 9-16 are only variations of the first 8 with a random factor. The names of the patterns are shown in the display and graphically displayed in the **manual of Rainmaker**, which you should read through. Of course you can find the corresponding link under the video.
The spikes of the crest are called **tap**, and so the **tap count** parameter refers to the number of spikes within the defined range. Unfortunately, here too, as with the size knob, the set number is not displayed. Between 1 and 64 can be the number of taps.
To better understand the parameters **edge level**, **tent level** and **tent tap**, I refer to the diagram shown. If you look at the **tips** of a comb, they can look different. For example, completely **flat**, then all hair or frequencies below the surface are affected. **Edge level** defines whether the tip of the spike points **down** or **up**, down the frequencies in the area of the top are more affected, upwards stronger on the edges. **Tent level** defines how **pointed** the spike is. The sharper the point, the greater the difference between the influence of edge frequencies and center frequencies. **Tent tap shifts the tip** along the x axis, making the point asymmetrical. As a result, depending on the setting of the other two controls, a single frequency or the frequencies to the right or left of the tip are influenced more or less. That sounds more complicated than it really is, as the graphics will hopefully clarify.
Feedback type regulates **resonance** and **reverberation**. The parameter offers 4 settings. **Guitar** filters the feedback of the filter based on a simple exponential function, similar to the classic **Karplus-Strong** synthesis.
Sitar is basically the same, but with a nonlinear component in the filter, which slightly modulates the comb size by a rectified version of the filter input signal. This models the effect of the curved bridge of a sitar, with the length of the string slightly changing depending on the displacement of the string. The effect of nonlinearity is noticeable only at higher feedback levels and most pronounced when the comb size is changed simultaneously.
Clarinet has the same feedback filtering as the guitar, but here is a non-linear component in the feedback path. This creates odd harmonics, giving the sound a "hollow" character, similar to a **square** wave. In the **raw** type, the tap # 64 output is returned directly to the input without filtering. **Feedback amount** defines the strength of the feedback signal.
Of course, **Hair Pick** is anything but a off the peg product. It can be used as a simple **comb filter**, but the variety of parameters and modulation options make it a very **complex** instrument for sound design. What I would like to have is the possibility to create my own patterns, but

that is not possible with the original rainmaker, so it would be something completely new. But you may wish it …



The modules from **Fundamental** are indeed the basic equipment of VCV-Rack and thus naturally provide a certain standard. The **Fundamental VCF** is a **4-pole ladder filter**, so a true classic. It offers the variants **lp** and **hp** and these can also be used simultaneously. The **frequency** knob works clean and nuanced. With the **fine** knob, the **cutoff** frequency can be adjust-ed even more, especially with the **resonance** knob, so interesting sounds are possible. The **resonance** controller goes to **self-oscillation**. The signal is saturated with the **drive** control, with much resonance the signal can be slightly distorted. **Drive**, **resonance** and **frequency** are also controllable by CV for the frequency input there is also an attenuverter, which rotates turned to the right, that the modulation signal should open the filter and turn to the left. And that was about it. A simple, no-frills VCF that works clean.



Once again, **Gratrix** supplements the **Fundamental** modules with **additional inputs** and **outputs**. The **VCF-F1** has **7 signal inputs, frequency** inputs, **resonance** inputs and **drive** inputs, and **6 outputs** each for **lp** and **hp**. Everything else is absolutely identical to the **Fundamental** module.

Even simpler as the **Fundamental** filter is the **24db Lowpass Filter** from **Lindenberg Research**. There are only 2 adjustable parameters **frequency** and **resonance** and both are controllable by CV and the in-tensity of the modulation can be adjusted with an attenuator. Despite the spartan features, all classic filter effects can be realized and it sounds **beautifully analogous**.

His sister **Valerie** makes me nostalgic. After all, my first synthesizer was a **Korg MS20**. **Valerie** is, according to the description, a **12 db/octave low-pass filter** modeled after **Korg's** famous **1978 MS20** filter. All three parameters can be controlled by CV and each have their own attenuator. This means that depending on how the large knobs are set, a modulation on both sides may be possible and at **Lindenberg Research** you can even **see the modulation** in the form of a small arrow on the parameter knob. As soon as a signal is applied to the CV in, it appears and moves with the modulation when the corresponding attenuverter is rotated. Absolutely awesome! Of course we have **frequency** at parameters, where the range is from 0 to 20 kHz, i. e. a larger frequency range than in the original **Korg** filter. **Peak** here is the **resonance** that goes to **self-oscillation** and **gain** saturation, but that quickly becomes distortion. There are 2 modes for this, where mode A is similar to a non-linear tube distortion and mode B is a nonlinear fold-back distortion. The sound is always **warm**, ranging from soft tones to shrill shrieks. A great filter that's easy to use.

Alma is the sister of **Valerie**, but has very different qualities. Since there is ever from **3db/oct** to **48 db/oct** infinitely adjustable edge steepness. For the other 3 parameters there is again one CV input with associated attenuator. Again, arrows indicate the modulation. The **cutoff** frequency range is 20 Hz to 20 kHz. **Resonance** goes to **self-oscillation**, and **overload** adds saturation, which goes as far as extreme foldback distortion. This also indicates a blue light. Especially with the **slope** parameter, which sets the slope, **Alma** is something very special. The strawberry on the icing would still be, although that could be regulated by CV. As I said, you may wish it ...

Laika is the youngest of the three filter sisters from **Lindenberg Research** and she is also modeled on a classic, the **18 db diode ladder filter** used in the **Roland TB-303** in the **EMS-VS3**. Unlike this **Laika** can be used not only as **lp** but also as **hp** and also both filters can be use

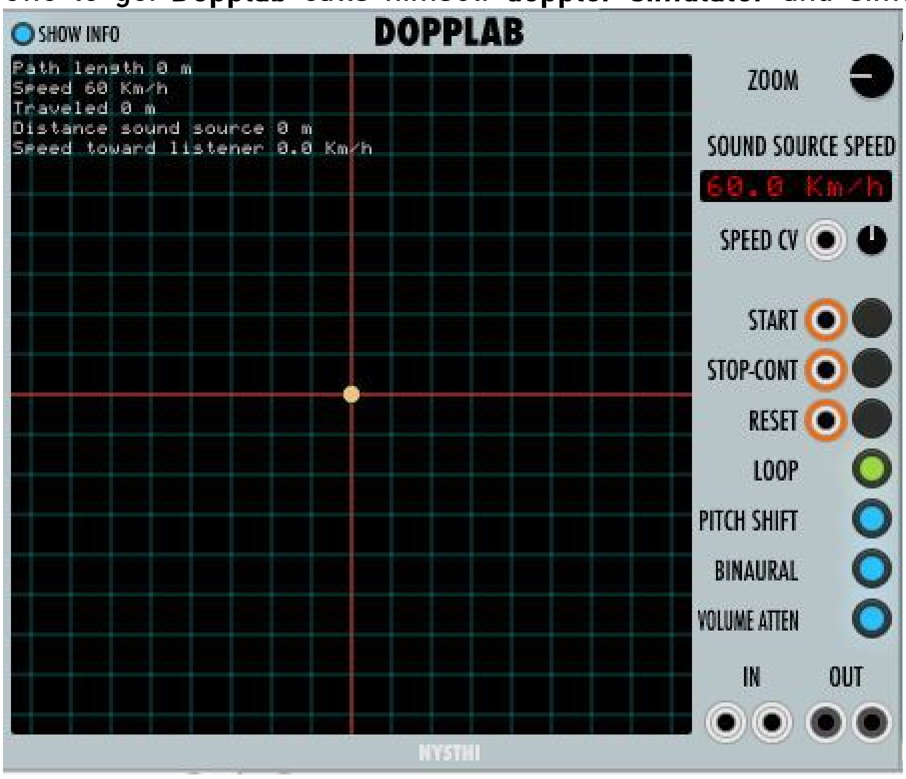
simultaneously. Again, all 3 parameters have their own CV input with associated attenuverter. Here the resonance is called **response**, but works as usual. With **saturate**, the signal is saturated or distorted again. With a bit of modulation, we have our own R2D2. A versatile filter, especially if you use both filters at the same time, which sounds good not only with acid.

TVCF by **MSM Modules** is based on a legendary module, the **Blippoo Box** by Rob Hordijk. He uses the **twin-peak principle** of two inverse-parallel low-pass filters, but goes beyond that other ways and therefore sounds different. With **2 filters**, it makes sense to use 2 different audio input signals. Where in principle also a third can be introduced via the input **V/Oct**, but it can also be any other voltage source here. Besides these 3 inputs, the module has 4 CV inputs, one for the controller **dual**, which determines the mixing ratio of the two filters. One for the parameter **ping**, which here is the **resonance** of both filters and of course also **self-oscillating** and one each the attenuverter **CV A** and **CV B**, which are assigned to the parameters **peak A** and **peak B**, where peak here stands for the **frequency**. In addition, with both filters, the filter type can be **steplessly** selected between **lp** and **bp**. Of course, I would have liked to have CV inputs. The more different the two filters are set, the more extraordinary the result sounds and even if a **V/Oct** signal is used and everything is modulated by CV, this filter does things that one would not expect from a filter. He looks so harmless but is definitely a wolf inside. Have fun taming.

The conclusion of today's double episode makes **Nysthi**. Of the meanwhile 72 modules (by now it's more than 100!), three are also listed under filters. Whereby **4DCBlock** immediately falls out, which filters out **dc voltage** from a signal and has no audible influence on an audio signal.

Dica33 is referred to on the github side of the developer only as a "very unstable filter", we don't learn more about it. By comparison, I estimate the characteristic to **12 db/oct**. And it's probably a **diode ladder filter**, working differently in the low frequency range. It has 2 parameters that can be controlled by CV, but no assigned attenuverter. **Cutoff** is the usual frequency regulator and **resonance** is called the **"risorstore"** and goes as far as **self-oscillation**. By the way, high values in both controllers are "rewarded" with a **smiley**. This filter is definitely missing the attenuverter, for a uniform modulation without dropouts is best suited a unipolar signal between 1 and 10 volts. So I think **Tidal Modulator** from **Audible Instruments** or the corresponding modules from **Aepelzens Modules** or **Southpole**, which I have all introduced in **episode 3**, are very well suited. But with almost certain probability **Nysthi** itself has a module, which represents a definite added value for this basically good-sounding filter.

One to go! **Dopplab** calls himself **doppler simulator** and simulates the property of the



Doppler Effect. This is a natural effect that arises when moving a sound source. In principle, the wavelength of a signal changes depending on the position of the receiver. This changes both the pitch and the volume of a signal and thus the perceived frequency. The further away the signal source is from the receiver, the deeper and quieter the sound is. As the signal source moves toward the receiver, the sound becomes higher and louder and also contains more harmonics. **Dopplab** defines the point in the **middle** of the display as a **receiver** and by **left-clicking** in the coordinate system points can be set, which together form straight lines, because we know that the shortest distance between 2 points is a straight line. On these routes, the **sound source moves**

relative to the receiver, due to the ever-changing distance between the two creates the **Doppler Effect**. The available parameters are **zoom**, which defines the size of the coordinate system. 1 box always corresponds to 10 meters, the zoom factor goes from ¼ to x4. The **speed** of the sound source can be set by clicking on the corresponding display and moving the mouse up or down between 0.1 and 999 km / h. In addition, a modulation signal can be mixed with the speed CV input and the associated attenuator. The **movement** of the sound source is started with **start** and stopped with **stop**. **Reset** resets the sound source to the starting point. All three functions can be controlled by **gate** or **trigger** and can thus be automated. When **loop** is activated, the sound source keeps moving and at the end jumps back to the beginning and so on. **Pitch shift** switches the pitch change on or off, **binaural** defines the slightly different perception of both ears. For the filter effect this should be activated. **Volume atten** switches the volume change on or off. This is also useful for a permanent filter effect. 2 input signals can be processed simultaneously and there are also 2 outputs, both of which should be occupied, for a clearer filter effect. If input 2 is not assigned, the same signal is output at both outputs. At input 2 can not only an audio signal, but also an **LFO**, etc. An **unusual filter**, Kraftwerk would certainly have had already for the production of Autobahn, because this drive-by effect can easily be produced with **Dopplab** . But due to the free definition of the routes also very crazy filter rides are possible. Exciting and interesting and certainly much more than a filter.

And hereby we are at the end of episode 11, which was a double episode. According to the current state, there are still 20 missing filters, which I pack in episode 12. Then it goes with VCAs. Probably in the meantime also the update 1 will appear with the new oscillators. As always, you will find links to the most important web pages and also to the developers of the modules introduced this time.
And not to forget, many thanks to Andrew Belt and all the module developers, without whom all this would not be possible. That's it from me, let it be good

Episode 12: Filters Part 2

Hello world and welcome to episode 12 of „what does this knob do?“

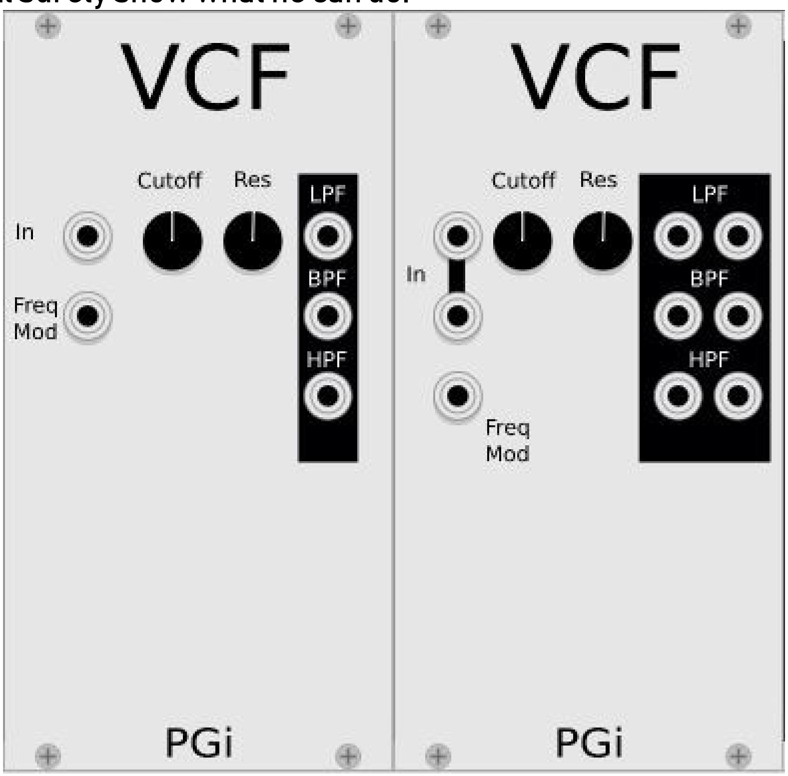
In this episode I show the other filters in VCV-Rack and with that I am already through and looking at VCAs next. And as with the oscillators applies here, new releases or new acquisi-tions of mine, I summarize in updates. Since the last episode more filters have been added, so I'll start with them right away.

Disfunctionalconvolvzilla (now only called Convolvzilla and a a lot of the issues are re-

worked) of **Nysthi** is described as a **convolver**. It can load and play all kinds of **wav** or **aiff** files, but actually it is meant for **impulse responses**. There are so many, but especially microphones, amplifiers and rooms, so reverb. I did not find anything for filters on the internet, but I did not search too hard. Finally, I found my it in my **VST plugins**. There are quite a few programs that emulate filters and so there is a great chance that impulse respons-es are included. The **Nysthi** module, with its almost unpronounceable name, not only **plays** the files, but **mixes** them with an input signal and outputs that mixed signal again. **2 different signals** can be processed **simultaneously**, each input signal also having an associated output, i. e. without input signal, the corresponding output remains mute. You can control both input and output levels and there is a stepless **dry/wet** slide. To hear something, active must be activated. The **original** signal can be heard when **bypass** is activated, this can also be controlled by triggers on the corresponding input socket. **Reset** resets the signal to the beginning, because the result can sometimes swing too high. For active and reset I would have liked associated CV inputs. The output signal can be reduced by -6 db using the button. A file can be **loaded** in two ways, either by **drag & drop** into the large display, or via the **context menu**, which is opened by right-clicking on the large display. Below that we can specify the **block size**. The smaller the value, the cleaner the signal, but the greater the load on the CPU. The two options, **impulse normal-**

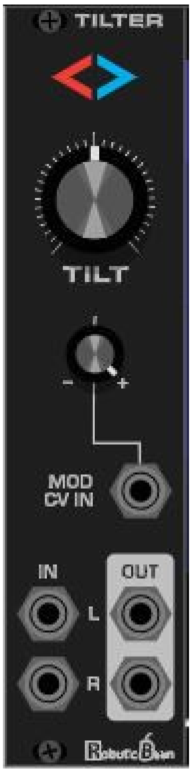
ized and **remove zero tails from impulse**, act directly on the loaded sample and ensure that no sudden peak levels occur or cut the sample off when the amplitude drops to zero. **Available impulses** shows all previously used impulses. I have no idea how many can be stored here, but for me there are already over 30 and these are listed alphabetically. Someone really thought it through! The last option is **invert screen**, which turns the background black and the sample white. Incidentally, the name of the loaded sam ple can also be seen in the yellow banner. Depending on what is loaded for a sample, this module is also a filter, although there is no way to edit its parameters (now we have early reflections and pre-delay). Nevertheless, with Disfunctionalconvolvzilla you can create extremely **extraordinary** sounds, if you have the right impulse response.

Even the micro-module **Slew** is not really a filter, but still has properties that go in this direction. In principle, the module is made up of **2 identical units**. Both have **1 input** and **output** and a **CV input socket**, with which the button **bypass** can be triggered. The actual modulation curve is set with the parameter **shape** and here you can hear and see quite a sound change, but here I would also have liked a CV input. However, this is always available for the parameters **attack** and **release**. These represent an own envelope for the modulation waveform and with small values in these parameters the modulation signal can be further shaped, at least it becomes duller. The main application of this module should not be a filter, because the possibilities are too limited. But in another episode with comparable modules, he will surely show what he can do.



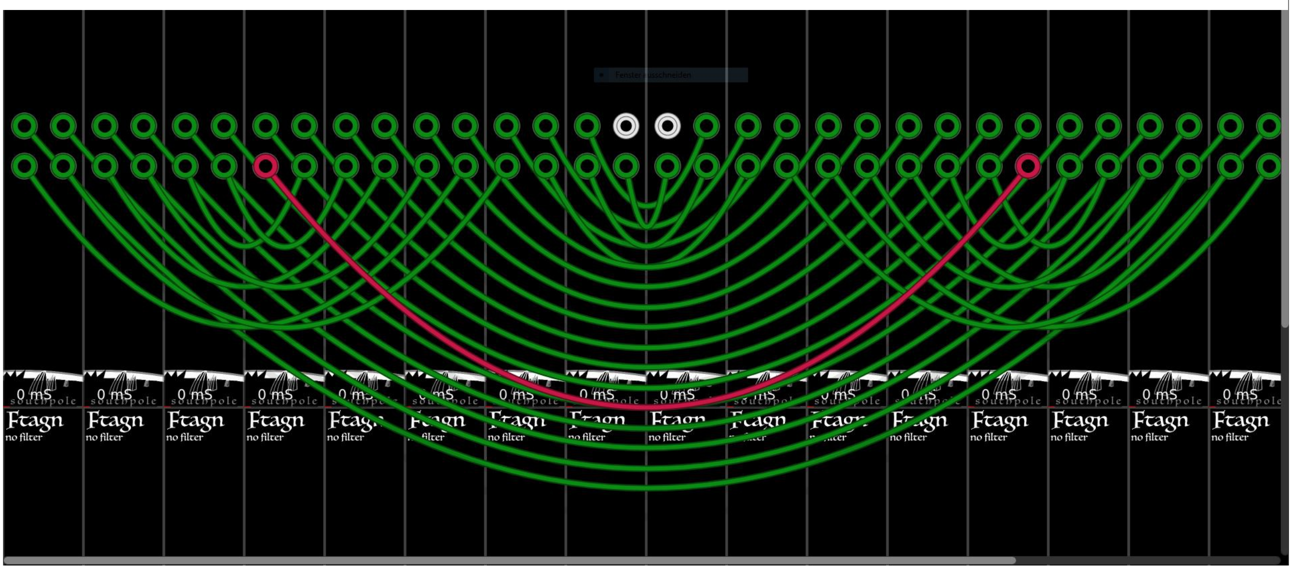
I almost overlooked the two VCFs from **PG Instruments** because they are not listed under filters. You can find them only if you enter the search term VCF. **PG VCF** and **PG Stereo VCF** actually only differ by the stereo option. And only if both inputs are assigned, you get a signal from both outputs. The modules are simple **1-pole filters** and offer at parameters only **cutoff** and **resonance**. Resonance does **not** go to **self-oscillation**. **Cutoff** can also be controlled by CV, but the intensity of the modulation must be controlled externally. The modules offer the three filter types **lp**, **bp** and **hp**, which can also be used simultaneously. The regulators are clean and effective. 2 simple filters that do what they should do. The format, however, I find unfavorable, you do not even make micro modules out of it, but the width of the VCF of **Fundamental** should be feasible.

The **Robotic Bean Tilter** is a paid module and can be purchased through the Plugin Manager.

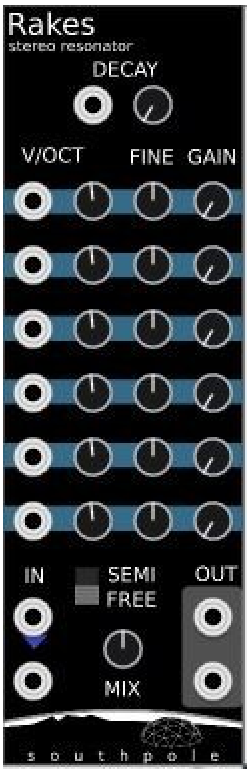


It is designed in **stereo**, if no signal is present at the 2nd input, the same signal can be tapped at both outputs. The module is both **low-pass** and **high-pass filter**, with a slope of **12 db/oct** each. The filter type is selected via the **tilt** control. In the middle position, the filter is neutral, **lp** turned to the left and **hp** turned to the right. Otherwise, this is the normal **frequency** controller and this can also be controlled by CV. With the associated attenuverter, the strength of the modulation can be adjusted and so also a filter sweep between the filter types are generated. Personally, I am not really convinced by this concept. I would have liked it better if the **tilt** parameter always sums both filters to 100%. So mid 50/50, 9 o'clock 25:75, 3 o'clock 75:25 and of course corresponding intermediate values. In addition still single exits for each filter and CV controllable resonance. Too bad, there are some free modules that have more on it.

One of my absolute favorite modules in VCV-Rack is the **ftagn** no filter from **Southpole**. The no stands for non-euclidean, that is, its **overwhelming sound quality** and **indescribable versatility** only come into its own in a **non-euclidean space** that has been painted with the **color out of space**. If you want to know how it sounds, I recommend the **Music of Erich Zann**.

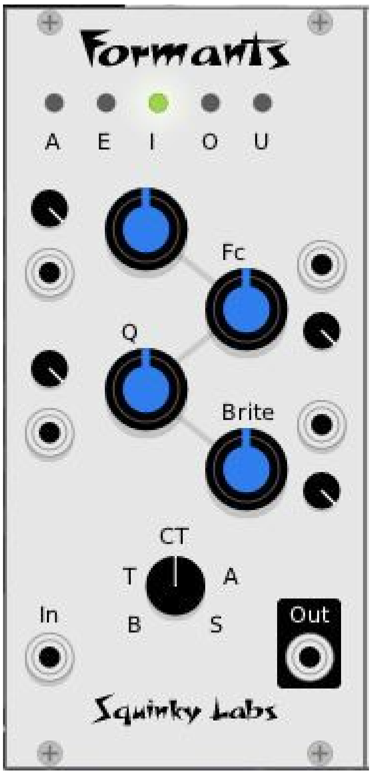


Essentially down-to-earth are **Etagère** or its stereo version **Deux Etagère**. According to the description, these were inspired by the euro-rack module **Shelves** by **Mutable Instruments**, but do not reproduce all the characteristics. The modules also have small print **"EQ filter"**, so we are not dealing with a pure equalizer, but with a module that combines both. The term **"variable filter bank"** comes to mind.. As with a **Fixed Filter Bank**, you have frequency ranges, but here you can change the frequency within the range and use the **gain** control to decrease all frequencies when turning it from the center to the left or raising it to the right. At frequency ranges it offers **highs**, **high mids**, **low mids** and **lows**. Each frequency range has the two CV controllable parameters **frequency** and **gain**. In the range of the **treble**, the gain control acts on all frequencies above the set frequency, in the range of the **bass** below the set frequency. The two middle bands are designed parametrically, i. e. with the **frequency** control one selects a frequency and with **q** a range around this frequency, which is lowered or raised with **gain**. That sounds partly like resonance. In addition, **Etagère** has its own output for each frequency band and one CV input for frequency and total gain. Especially with the latter, you have to be very careful, otherwise you have quickly a tinnitus or again R2D2. At **Deux Etagère** these CV inputs and also the single outputs are missing. It is **stereo** and the left channel is also usable as a mono. At the right output, only one signal can be tapped, even if the right input is assigned. If you are looking for a **Fixed Filter Bank** or an **equalizer**, you should definitely take a look at these modules because they offer all that and more.



Southpole's Rakes is also called stereo resonator and was influenced by **Ableton's resonator effect**. I do not know this, so I look at the module completely unbiased. We are dealing here with one or rather **six comb filters**, all of which have the same structure. Each has its own **V/Oct** input, so we can edit 6 different signals at the same time and use sliders to determine whether they are **quantized** in semitone steps or not quantized. In addition, the module has 2 further inputs, which are mixed in the output signal with the signals from the comb filters. If no signals are present at the individual inputs, only the mono or the stereo signals are processed by the comb filters. Here, of course, 2 completely different signals can be present. The 1st knob of a filter is the general **frequency**. With **fine** it can be slightly detuned to generate floating effects or strum effects, and **gain** sets the ratio of a filter in the overall signal. **Decay** controls the decay, which results in e.g. emulate the behavior of strings or other resonant bodies and that also by CV. With **Rakes** wonderful string sounds can be created, as with the classical **Karplus Strong** synthesis and partly it sounds like **Hair Pick** by **Frozen Wasteland**, which is also a combfilter. But in principle a lot more is possible, as the **randomize** function clearly shows. And if all inputs are occupied, there are so blatant things possible that VCV-Rack crashes me, because unfortunately the module is also a CPU eater. But it's **awesome**.

The next two filters from **Squinky Labs** are again something special. **Formants** says already



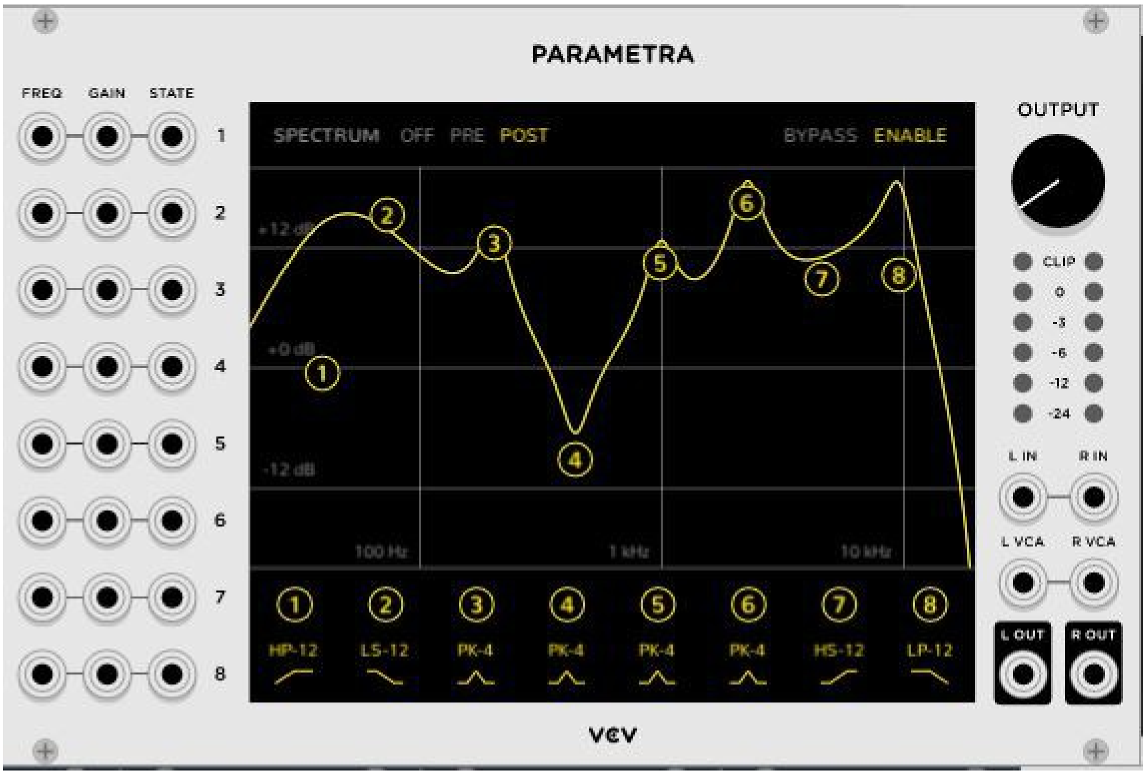
what he is, a **bandpass** filter with high slopes. Remember, **lp** and **hp** in **series** filters both bass and treble, leaving a narrow center band. We also had **formants** at **Frozen Wasteland**'s **Vox Inhumana**, so here's also **vocals**. We can already see this in the upper part of the module, together with a light bar that shows which vowel is currently playing, with the transition between the vowels flowing. With the directly below blue button you can switch between the vowels or CV. All blue controls have an associated CV input jack and an attenuverter. If this is on the far right, you have full modulation, it is in the middle, no modulation and on the left also full modulation, but with inverted signal. **Fc** stands for frequency control and here the frequency of the filter is adjusted. Here again, **q** is the resonance that does **not** go to **self-oscillation** and **brite** boosts the high formant's level when turned to the right and lowers it when turned to the left. Finally, the **pitch** is adjusted with the big black knob, the classical pitches **bass**, **tenor**, **counter tenor**, **alto** and **soprano** are possible here. And **Caudal** makes him sing already in the initialized state.

Growler is also called **vocal animator**. Here you can not select vowels, instead it consists of **4 bandpass filters**, each with its own formanttypical frequency. These run in parallel and their output signals are added together. **4 LFOs** with different triangular waves modulate these filters. This creates an effect that sounds like vowels again. The **speed** of the LFOs can be influenced via the LFO control or by an external signal, whereby the intensity can again be

adjusted by means of attenuverters, which behave as with formants. The **toggle** switch in the LFO section affects the **synchronization** of each LFO. At the bottom, the synchronization is greatest, slightly different in the middle, completely different at the top. The lights indicate the speed of the first three LFOs and the adjacent output jacks allow this LFO signal to be used in other modules. The controller **fc** sets again the frequency of the filter, **q** the resonance and **depth** the influence of the LFO on the filter. All three are controllable by CV and also have associated attenuverters that behave as usual. Of course, the 3 LFO outputs can also be connected to the CV input jacks, which modulates the module itself. With the toggle between in and out you can boost the bass a bit. For this the controller must point upwards.

The two **Squinky Labs** modules are similar but different. In any case, both are perfect for vocal sound. Formants is a specialist who feels very comfortable in his niche. Growler is also at home in this niche, but he also dares to go outside. Both sound great and need - as usual with **Squinky Labs** - very little CPU.

Parametra is one of the paid modules that **VCV** offers itself and with which both the development of VCV-Rack and their free modules are funded. So if you want to continue with VCV-Rack, you should take a look at these excellent modules and buy them. Although **Parametra** calls itself a **parametric equalizer**, it is much more than that. But the title of the manual also says **parametric filterdesigner** and that hits the nail on the head. Although it offers **17 different filters**, from **low-pass 48 db/oct** to **notch** filters, you can use up to **eight different filters at once** and individually set and control their **cutoff** frequency range and **gain** via CV. And you can also turn on and off each of the eight filter types by **trigger**. I'm really excited about the handling via the **display**. On the one hand, the **frequency spectrum** of the signal or the signals can be displayed, because the module has two inputs and two outputs. The options for the display are input signal, output signal or off. With bypass, the module can be deactivated and put back into operation with enable. In the initialized state, the display shows a straight line on which the numbers 1-8 are distributed at regular intervals. We can still see that the line has just 0 db and the range of the amplitude exceeds +/- 12 db, the exact value is not to be



learned from the manual, as well as the frequency range, here we only see that it well below 100 Hz starts and goes beyond 10 kHz. But this is not necessarily important for the handling. Much more important are in the lower half of the display again the 8 numbers that are of course are related to those on the line and each symbolize a filter. Among the numbers we also see the **selected filter types**, if you would like to select another one, you just have to **click** with the **left mouse** button on the filter **symbol** and then click on the filter you want and it is already selected and shows with a small graphic, what he does to the signal. To activate a filter, just click on the corresponding number above the filter symbol. This lights up, as well as the corresponding point on the line. The one on the line can be **moved**, even in the deactivated state. Namely, the cutoff frequency is set horizontally and the amplitude is set vertically. Simple and ingenious, you can **hear** what the filter does and then **see** it right away. And since this can also be controlled by CV, of course, all possible filter runs are possible, but of course, here also an envelope signal can be introduced. In the graph you can see not only the setting of the filter, but also its modulation. Speaking of modulation, in the inputs **VCA l & r**, a VCA signal can be introduced to regulate the output volume by envelope or similar. And we have here **eight different filters** available, all of which can work **simultaneously**, even in the same frequency range and which can all be modulated externally. This allows extremely **complex** filter effects to be realized. You can also use it as a simple parametric equalizer, but if you work hard with it, it offers an almost **inexhaustible reservoir of possibilities**. Although it costs $ 30 but is definitely a good investment.

The term filter is fixed in VCV-Rack with a name - **Vult**. And not only because there are no fewer than 11 modules listed under filters, but above all because of the sound. Here I introduce the commercial modules included in the **Vult Compacts** and **Vult Premium** packages. However, 7 of the modules are also available in the **Vult-free** package, with **Lateralus** and **Tangents** in the premium version having additional options, which I will discuss later.

The first is **Freak** from the **Vult Compacts** series, which contains all of **Vult**'s filters, but not

all the settings, inputs and outputs. I will therefore compare **Freak** with the original module every time. Adjustable parameters in **Freak** are **resonance**, **drive** and **cutoff** and all can be controlled by CV and there is also an attenuverter. Cutoff can even be modulated by **3 different sources** at the same time. The module is designed in **stereo**, whereby with only one input signal at both outputs the same signal can be tapped. But, not only that. By means of the switch **mod route**, the two inputs A and B of the **cutoff** frequency can be assigned differently to the outputs, which is also evident in the display. The default setting is **combined**, where the signals from A and B are added and passed to both outputs. The setting **individual** sends the signal from A to the left and from B to the right output. With **inverse**, the left output receives the signal from A minus the signal from B and the right output from B - A. This alone makes very interesting variations of the sound possible. All knobs clearly show how they work. The parameters from 0 to full and the attenuverters to the left negative and to the right positive modulation. Between the inputs and outputs is still a small switch for **low power**, which also makes sense, because **Freak** already needs quite a bit of CPU. However, the sound character also changes slightly, so you can see this switch as an added value in terms of sound variety. In the display we see not only the routing of the modulations A and B, but also the currently set filter model and its mode. With the switches underneath you select these. The first row is for the **model** and the second for the **mode**. Use the up arrow to select the next setting and the down arrow to select the previous one. In this way all options of the filters can be set.

Let's get to the **filter models**. By default, the filter **Vortex** is selected in **lp** mode. This is also called **russian filter**, since it is based on the famous russian **Polivoks**. The filter is a **lowpass** and **bandpass** that can **self oscillate**. We see in the **Analyzer** that **Freak** and **Vortex** are iden-

tical. If I turn **Freak** into the low power mode, you can clearly see that its slope is increasing. This is difficult to estimate, in the frequency response it looks like the slope will change from the beginning **6 db/oct** to **24 db/oct**. The original had a slope of **12 db/oct**. **Vortex** also has a low power mode, so that the signals are the same again. Sonically, the low power mode is slightly higher in altitude. The parameters of **Vortex** are exactly the same as those of **Freak**, but with **Vortex** both filters can be used at the same time, which definitely influences the special sound of the **Polivoks**. Describing the sound is always difficult, it ranges from warm bass to dazzling heights and **resonance** and **drive** unleash the predator in it. Personally, I usually prefer **Vortex** and send the signals from **lp** and **bp** to an attenuverter to **mix** their ratio. It would be nice if this would be possible in the module, but it also works like this. **Vortex** sounds great even in the **Freak**, but the single module topped that still tremendous.

The next module in **Freak** is **Unstabile**, but since this is a version of **Stabile**, let's take a look at it first. **Stabile** is a **state variable filter**, as it e.g. was used by **Oberheim** in **OB-X** and **SEM**. Especially the latter plays a big role in the filter design of **Stabile**. Because apart from the filter types **2-pol lp**, **bp** and **hp**, which are common for a state variable filter, it also offers a **SEM** output, at which a mixture of **lp** and **hp** is output, which is set with the parameter of the same name. This is designed as an attenuverter and fades to the left turned off the highs and turned right the low frequencies. This parameter is limited to the **SEM** output. **Cutoff** and **resonance** work as usual. **Stabile** sounds in **Freak** by the additional **drive** parameter more agressive, but also here only one filter type can be set and the option **SEM** is not available. As a result, I would even speak of different modules or **variants**. However, the single module is certainly more versatile again, especially if all 4 outputs are mixed via an attenuverter. **Stabile** sounds warm and is quite restrained.

Unstabile has integrated the **drive** parameter itself and also introduces **nonlinearities** and can also **oscillate itself**. Otherwise, the same applies as for **Stabile**. **Unstabile** gives the soft sound of **Stabile** a proper portion of dirt and that suits it well. Here I tend once again to single module, of course, never forget the initially mentioned specifics of **Freak**. **Stabile** and **Unstabile** are both unchanged in the **Vult free** package.

Lateralus is a **diode ladder filter**, this principle is known e.g. from the **EMS V3** or **Rolands TB 303**. The controllable parameters are again the same as with **Freak**. In the **premium** version there are **2 modes** that can be selected via the slider. **Df** is a gentler version of the sound, **th** more agressive. In the freely available version there is no choice. The special thing about **Lateralus** is the **selectable slope, 6, 12, 18 and 24 db/oct** are possible here, 18 db was incidentally used in the **Roland TB 303**. In fact, the individual module also makes it possible to use **all slopes at the same time**; different effects or modulations can be obtained. **Freak** also offers both modes and all slopes, but again the output is limited to **one filter type**. The big question is here, do I want to use different slopes at the same time. If not, in this case, the benefits of **Freak** outweigh.

Tangents is also available in a **freely** available version as well as part of the **premium** package and here too the **premium** version offers **additional modes**. **Tangents** is a **2-pole steiner parker filter**, so it uses the filter design of the **Synthacon**, which has been revived by the **Arturia Minibrute**. The parameters of the single module also correspond to those of **Freak**. The modes are **yu**, which corresponds to the original **Synthacon** filter, **ms** is a gentler version and **xx** is referred to in the description of **Tangents** only as "pure madness", so an extreme variant. Unlike the other filters, there are not multiple outputs, but **multiple inputs**. This makes it possible to introduce different signals with different filters to edit and output this mixed in the output. Here I recommend to switch an attenuverter before the inputs to change the ratio of the input signals. But **Freak** does not have that option again. Although it offers all modes and also the filter types, but also not the possibility of the single mod-

ule. That's why I prefer this again.

Ferox is included as a single module only in the **Vult premium** range. He calls himself **cmos filter** and since I have found nothing on the internet, except that it probably goes to semiconductor elements, I quote here once the **Vult** website "**ferox** is based on the ingenious **cmos** filter design. The **cmos** filter uses an **digital** inverter chip to replace operational amplifiers. The result filter is full of character, distorted and aggressive. **Ferox** can transform the simpliest waveforms into completely new sound. In some cases it will make you doubt that it is a filter at all.. "end quote. Apart from the usual three parameters, **Ferox** still has the parameter **bite**, which is not included in **Freak** and which provides an additional "bite". **Ferox** provides the filter types **lp**, **bp**, **hp** and **notch** and all can be used **simultaneously** in the single module. **Freak** does not offer that possibility again and I could get over it, but without bite parameters, **Ferox** is just a shadow of itself. The strength of **Ferox** is aggressive and distorted and requires the right bite. So again clearly the trend to the single module.

The next module in **Freak** is **Nurage**, but since this is more than a filter, I move it a bit and first turn to **Rescomb**. The name already tells, it is a **comb filter with resonance**. And he has com-

pletely different parameters than **Freak**. Therefore, I am also of the opinion that it is about different modules or **variants**. For **Rescomb** the frequency is determined by a signal at the **V/Oct** input, this can be a fixed voltage or a changing one. With **tune**, the frequency can be detuned. **Freak** uses the **cutoff** control as usual and we remember that this can be influenced by 3 different modulation sources at the same time. The parameter **comb** is an attenuverter and determines the comb filter size depending on the input signal and the signal at the **V/Oct** input. Turned to the left, the comb is smaller, larger to the right. In **Freak**, you can choose between **comb+** and **comb-**, and this affects the function of the cutoff knob. That is, the combsize is either larger when the filter is further turned on or when it is closed. The **feedback** parameter in **Rescomb** is synonymous with **resonance** and works accordingly. In **Freak**, the sound can be additionally distorted with the **drive** control, which can also be used to create other sounds. Both variants are interesting and have their own strengths. But for me, the version in **Freak** is just ahead by the drive and the modulation options.

The most recent filter in **Freak** is **Nurage**, which is just the **filter section of the module**, a mix of a **Buchla** filter and a **Korg MS 20** filter, hence called a **Borg** filter. It is influenced by the usual 3 parameters and offers only this one mode. However, the two CV inputs A and B also make a **low-pass gate** effect possible by introducing gate signals here. The **Nurage** filter in **Freak** is a good sounding **analog** filter that lives up to its big role models.

Nurage is part of the paid **Vult premium** package and much more than just a filter. He calls himself **dual low pass gate/filter**. In principle, a **low-pass gate** is a **mixture** of **low-pass filter** and **VCA**. Accordingly, the parameters are like other filters and also fulfill the same tasks, but there are other options.

Cutoff is therefore not only responsible for the filter cutoff, but also for the opening of the **gate**, for which he is assigned 2 CV inputs, including attenuverter. Fully turned on, the gate is also open at maximum. However, using only one gate signal will trigger both gates. If you use **2 gate signals**, you can switch the two signals **independently** of each other by means of the **gate switch**, which is standard on link. With the switch **vactrol** you activate a mode which simulates a **vactrol circuit**, or better said its characteristic to react only delayed at fast gate signals. This also causes a **delay** or a **reverberation** in the signal. Finally, one can activate it by means of **VCA switch**, so that the modulation acts on the amplitude and not on the filter. Because of all these possibilities and the great sound, which significantly enhances rhythmic parts, **Nurage** as a single module is much more versatile than the implementation in **Freak**.

Nurage is not the only **lowpass gate** of **Vult**, there is also the free module **Julste**. This is modulated according to the original **Buchla lowpass gate**. Here, too, a **gate** signal defines at what time and at which frequency a signal may pass. **Range** determines the opening of the gate, on the left the opening is minimal, right is maximum. With the **offset** parameter you set a minimum value for the gate turned to the left and turned to the right a maximum value. **Resonance** is the usual parameter. **Sharpness** defines the influence of **vactrol** on the signal. At the far left the gate corresponds to the inserted gate signal, at the far right it corresponds more to an envelope curve and consecutive signals are separated more sharply. All parameters can of course be controlled by CV and also have an associated attenuverter. With a switch you can select between VCA and lowpass. In **VCA mode**, only the amplitude of the input signal is modulated, and here only the parameter sharpness works. **Julste** works much more reserved than **Nurage**, but is still very good for rhythmic structures. I think they complement each other wonderfully because everyone can do something the other can not.

The episode concludes with **Tohe**, another module of the **Vult free** series. **Tone control** already says that we are dealing here not with a filter but an **equalizer**. It is designed in **stereo**, if only one signal is introduced, this can be tapped at both outputs. The frequency is set via the **tone control**. Turned to the left, high frequencies fade out, to the right low frequencies. This can also be controlled by CV and there is also an associated attenuverter. The **model** switch determines the behavior of the tone controller using an equalizer model. **Base** is a standard control with a very strong influence on the signal, **diez** corresponds to a 10-band equalizer, which allows a finer adjustment and **puff**, a dispersion model for very gentle and subtle frequency changes. The three modes and the ability to control CV make **Tohe** much more versatile than a simple equalizer.

All of the **Vult** modules shown here leave an **excellent impression** on me, although the design also appeals to me very much. Of course, more important is the sound and it's just great and actually each module sounds different, so it makes perfect sense to have it all available. Again and again, the question arises as to which of the paid packages from **Vult** you should get. The answer is simple, both. **Freak** is absolutely versatile, but he is just more the all-rounder and the individual modules rather the specialists. Nevertheless, **Freak** can do things that the specialists can not and vice versa. The premium package is now complete with **Nurage** and thus contains **8 exclusive modules**, **Lateralus** and **Tangents** in slimmed-down form are also in the free package and **Noxious** are included as a monophonic and polyphonic VCO. **Vult** compacts currently only consists of **Freak** and **Incubus** (by now there are 4

modules), but there will be more modules here.

But in order to not make one-sided advertising for **Vult** now, here's again the hint, **VCV-Rack and many free modules are funded by paid modules and these are all extremely cheap and definitely worth their money!**

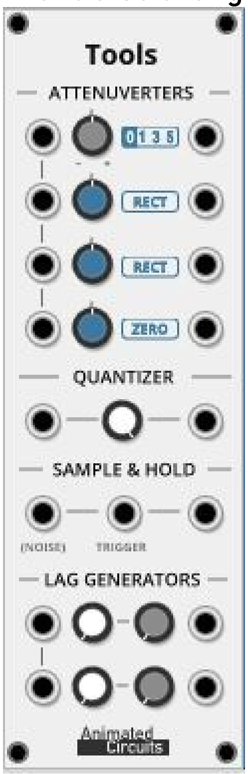
And hereby we are at the end of episode 12, so I introduced almost all currently available filters. As soon as enough new modules have been added or the bugs have been removed in the case of current modules that I have not yet presented, there will also be an update here. It continues next week with VCAs. As always, you will find links to the most important web pages and also to the developers of the modules introduced this time.
And not to forget, many thanks to Andrew Belt and all the module developers, without whom all this would not be possible. That's it from me, take care

Episode 13: VCA

Hello world and welcome to episode 13 of „what does this knob do?"

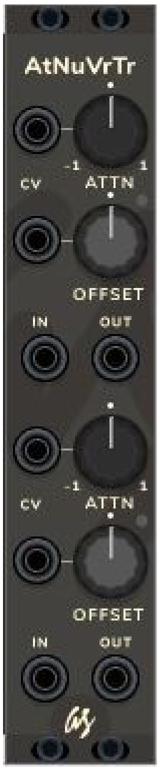
As announced, I'm going to deal with **VCAs** this time, but I'll also take a look at **attenuators/attenuverters**, because the tasks are similar and some modules appear in both sections. A few of the modules also have other areas of application, for these I limit myself to the functions as VCA or attenuator and look at the other functions in another episode that deals specifically with these. I will introduce all **52 modules** in this episode, because I assume that this is very fast because VCAs often only have input, output, volume control and maybe a CV input socket for linear or exponential modulation. In classical **subtractive synthesis** you don't need any more. VCO for sound generation, VCF for sound design, VCA to influence the volume of the sound and an envelope generator to control a temporal change of the sound and also the volume, because some sounds should start immediately and others have a settling time and the same applies to the reverberation. In this context one usually only thinks of the **envelope generator** or **ADSR**, but if it has nothing to control, it does nothing at all. So the VCA is very important, you could call it the **heart of the system**, because without it, there is no sound. And yet it is often treated very negligently, so we can be very curious to see how the various developers deal with it.

I have also changed my setup a little bit, or added something. Up to the **VCO-1** of **Fundamental** everything remained the same. Its signal now goes into **Fundamental's VCF** and from there directly to the mixer and audio output on the one hand and into a VCA or attenuator on the other, whose envelope is formed by **Fundamental's ADSR**. In the **Bogaudio Analyzer**, the **LogInstruments** spectrum **Analyzer** and the **JW-Modules Full Scope**, I juxtapose the two signals so that we can see what the VCA does with the input signal.



Tools from **Animated Circuits** is part of the welcome package and was therefore one of the freely available modules of the developer. In the meantime, however, all its modules are free of charge and can be downloaded from its website. **Attenuverter** is printed big on the module, but it's much more than that, but I won't discuss the other features today. All knobs in the attenuverter section are neutral in the middle position and rotated to the right increase the amount of the respective parameter in the output signal, while rotated to the left inverts the signal. In principle, each of the 4 attenuators has its own input and output. So everyone can receive a different signal, but as long as there is at least one input signal, this can be tapped at all outputs. The top row adds a **fixed value** to the input signal, 0, 1, 3 and 5 volts can be selected, whereby these values also become negative when turned to the left. The selected value is always output in the middle position. **Lines 2 and 3 are identical** and here the signal can be **rectified.** To do this, click on the **rect** field, which then inverts its color and has white writing on a blue background. If this field is not clicked, the knob works normally as an attenuator. The same applies to the **4th row**, where the signal cannot fall below 0 volt when the **zero** field is activated, all values below it are normalized to 0 volt. If the knob is turned to the left, the effect is the same, but the curve is inverted. **Tools** is a very interesting module, especially if the different areas are connected in series, but it drastically reduces its range of application because not a single parameter can be controlled by CV. That's too bad.

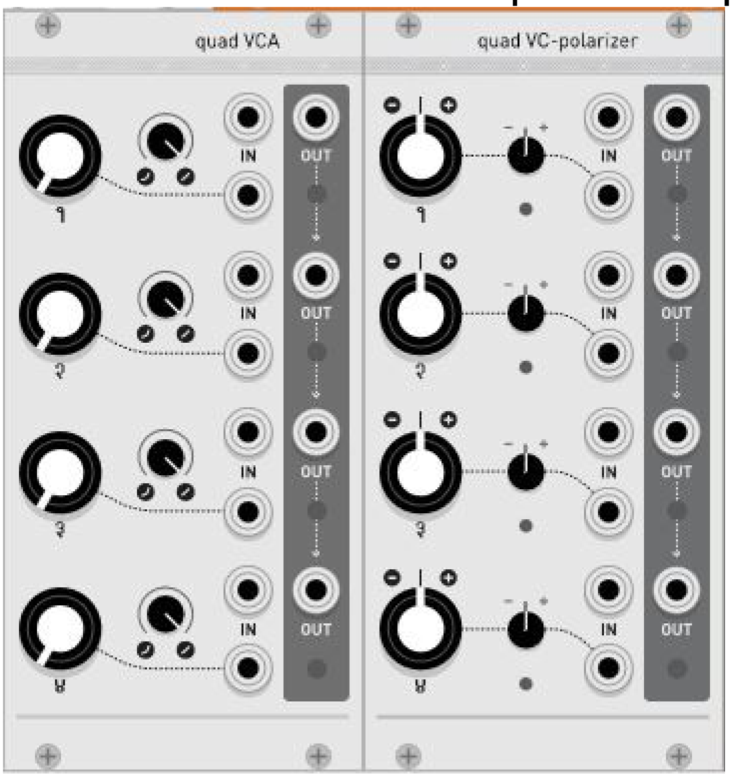
VCA and **QuadVCA/Mixer** from **AS** modules are much more classic. Both are identical in handling, whereby the **QuadVCA** still has the additional mixer option. **VCA** consists of **2 independent** areas, **QuadVCA** of **4**, which can also be output as a sum signal at the mix output if the other outputs are not occupied. The sliders attenuate the input signal. In the middle position the amplitude is halved, at the bottom 0 and at the top 100% of the input signal. So, like most VCAs, we are not dealing with an amplifier, but with an attenuator. Each section has an input, an output and a CV input, which is mostly used for an envelope signal, but of course other modulation signals are also possible, like LFO or velocity. The **lin exp** switch is used to define the type of amplitude change. **Lin** means linear and therefore an even increase or decrease. **Exp** means exponential, with which the change begins gently and then increases more strongly. This can be seen well in the scope when using an **ADSR**. Linear is a gentle change of the signal, exponential a sudden and jerky change. In **QuadVCA** this switch is marked with symbols. Red lights below the inputs also show which signals are summed in the mix output. 2 simple and straightforward VCA that do what they are supposed to do.



Also in the department attenuverter **AS** is represented with the module **AtNuVrTr**. Also this module consists of **2 independent identical areas**. In addition to input and output, there is also a CV in socket for each of the two parameters. If both knobs are in the middle position, no input signal is output or, if an **ADSR** is used to modulate the **attn** parameter, the same signal as with the VCA with 100%. If you turn **attn** to the left the signal is inverted, to the far right it corresponds to the input signal. With modulated **attn**, a slight inverted modulation is noticeable on the left and a doubling of the amplitude on the right. The **offset** parameter adds 0-10 volts to the signal when it is turned to the right and negative values when it is turned to the left. Thanks to the CV inputs **AtNuVrTr** can also be used as VCA, but only with linear modulation, but the offset parameter is definitely an added value.

As always, the **AS** modules leave a very positive impression on me and deserve the rating **"simple and good"** again.

The **Audible Instruments** modules **quad VCA** and **quad VC-Polarizer** look very similar but



function differently. Both consist of **4 identical independent units**, each with one input and one output socket, as well as a CV input socket with corresponding attenuator and a rotary knob for the parameter. Both are, like all modules from **Audible Instruments**, modulated after euro-rack modules from **Mutable Instruments**. **Veils** was the inspiration for the **quad VCA**. The large white parameter button is responsible for the signal strength and starts with 0 on the left and maximum on the right. But with a modulation signal in the CV input, the amplitude of the input signal is already reached in the middle position, and a doubling is at the far right. **Linear** and **exponential** can be infinitely adjusted with the small black knob. On exponential, the amplitude change is much clearer and here the value of the input signal is already reached at about 8 o'clock. With a balanced ratio at about 9 o'clock. The output sockets are connected to each other, if you use only the lowest

one, there is a **mixed** signal from all input sockets. What I definitely like about this module is the possibility of infinitely variable regulation between linear and exponential. That's a good idea. **Balaclava** from **Southpole** corresponds 100% to this module, but is much more compact.

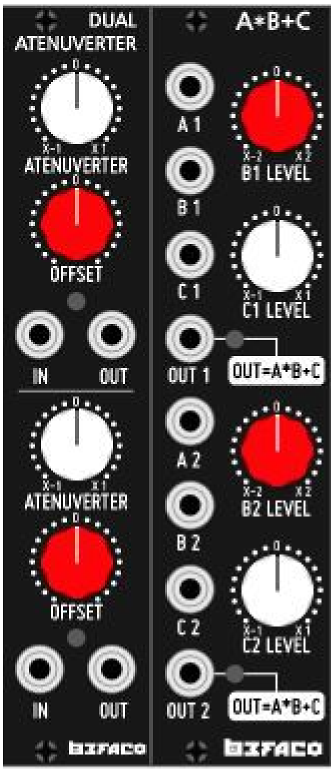
The **quad VCA-polarizer** corresponds to **Blinds**. The parameter knob does not pass a signal in the center position, fades it in up to 100% of the input signal when rotated to the right, and inverts it in the same ratio when rotated to the left. If this is on the

right and a modulation signal is present at the CV input, the amplitude is **increased** up to twice as much when the corresponding attenuator is turned to the right. If you turn it to the left, you get a slight modulation and the amplitude **decreases**, finally reaches 0 and is inverted. If the knob is at the left stop, we have the full **inverted** signal with full modulation. The same applies vice versa if the parameter controller is at the far left. The output sockets are also **interconnected** here. The led's under the modulation controllers light up green when the signal is positive and red when the signal is negative or inverted. Very interesting results can also be obtained by feeding an additional audio signal into the CV input socket, from a rectified waveform to a ring modulator. This module is also very interesting and offers a lot more than the normal VCA functions. Also here there is a more compact module from **Southpole**, called **Bandana**.

The **Keyframer/Mixer** is the VCV-Rack version of **Frames** and of course much more than a VCA. In principle we also have **4 VCA** available, all of which have an input and output socket and a corresponding controller. In this use there are no CV inputs and the signal strength can be 0 - 100% of the input signal. Each signal can be tapped at the output of the corresponding input. If the all input is used instead of the individual inputs, the signal can be picked up at all outputs and sent to different receivers in different strengths, for example. If the mix output is used instead of the individual outputs, a mix of the input signals is applied, whereby the mixing ratio of the signals can be adjusted. The most interesting thing is, of course, when you use the **key frames**, then a CV and envelope control is also possible. The principle is actually quite simple and we have already seen it similar with the **Phasor Harmonic VCO** from **Nysthi**. You take **snapshots** and then you play them one after the other, like in a movie. The snapshots or key frames here are the positions of the 4 white controls 1-4. To **create a key frame**, click on the round gray area **add**, whereupon the large led lights up green and the large frame button also lights up. Now set the **4 white knobs** as desired and turn the **frame button** until the green led goes out. Now you can create the next key frame. Each frame gets its own color. You can **switch** between the individual key frames by turning the **frame knob**, or by modulating the input frame and here the envelope comes into play. The intensity of this modulation is adjusted with the corresponding attenuator, whereby both positive and negative modulation are possible. Single key frames can be deleted by clicking on the gray round **del** button. The **+10v offset** slider is used to output a fixed 10v voltage to all inputs without an input signal. A trigger pulse is output at the **Fr. Step** output each time a new key frame is reached. All current key frames can be deleted at once in the **context menu**, which, as always, can be called up by right-clicking. The key framer can also be switched to **poly LFO** here, but I won't discuss this until another episode. In addition, the **interpolation** and the **response** curve for each of the 4 channels can also be defined here. The interpolation is the transition between the key frames, here there are 6 different possibilities, step (no transition), linear (even), accelerating (slow at first, then faster), decelerating (faster at first, then slower), smooth departure/arrival (slow start and end) and bouncing (jumpy transition). Response curve determines how the modulation works, here the usual options are linear and exponential. Even this short overview has clearly shown that this module is much more than a VCA. Therefore I can only recommend, as with all modules of **Audible Instruments**, to deal with it intensively. **The effort is worth it!**

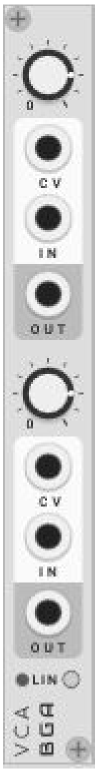
Autinn's **Amp** has the usual trio jacks and a large white knob for the signal, left the value is 0, in the middle equal to the input signal and right double the amplitude, so a real **amplifier**. With CV input the behavior of this control changes and to have a real envelope signal you have to turn it far to the left, preferably completely. And even then it jumps from 100% input signal to double value. And the modulation type is not adjustable, it seems to be an **exponential** modulation. To get a **linear** modulation here, you can add another module of **Autinn**, **Mera**. If it is switched before, the input signal of the modulation can be reduced to 0 and the modulation is linear. For this option the input and output of the **pre** section are to be used, with the white control the strength of the output signal is determined, whereby already in the middle position twice the input value is reached. The further this knob is turned to the right, the greater the amplitude. If you use the post section, you have the amplitude of the input signal on the far right and to the left this goes through the ceiling and to the far left each signal becomes a **square** wave. Both modules are real **amplifiers**, but I would have liked **Mera** to have a CV in as well. If you use them together, the versatility increases enormously.

The **Dual Atenuverter** from **Befaco** corresponds to **AtNuVrTr** from **AS**, but has no CV input.



A*B+C does not have it either, but each of the two units of this module has 3 inputs, 2 of which are adjustable. If an input signal is present at input A, this is multiplied by signal B and added to signal C. If no signal is present at one of these inputs, the value for b is 5v and for c is 10v. With the value of **attenuverter B** the signal of B is multiplied, whereby the range goes from -2 to +2. The same applies to C, where the range is -1 to +1. Output 2 is also the sum of both units. The controller paths are always exponential. If you insert an audio signal into input A and a modulation signal into input B, you can use **Control B Level** to determine the strength of this modulation. This can be positive as well as negative again and with the control and a signal at C an offset can be defined to cut off high or low frequencies again. Due to its unique functionality, this module has a lot of potential which goes **far beyond** the use as a VCA. But here, too, the combination of both modules is a real added value.

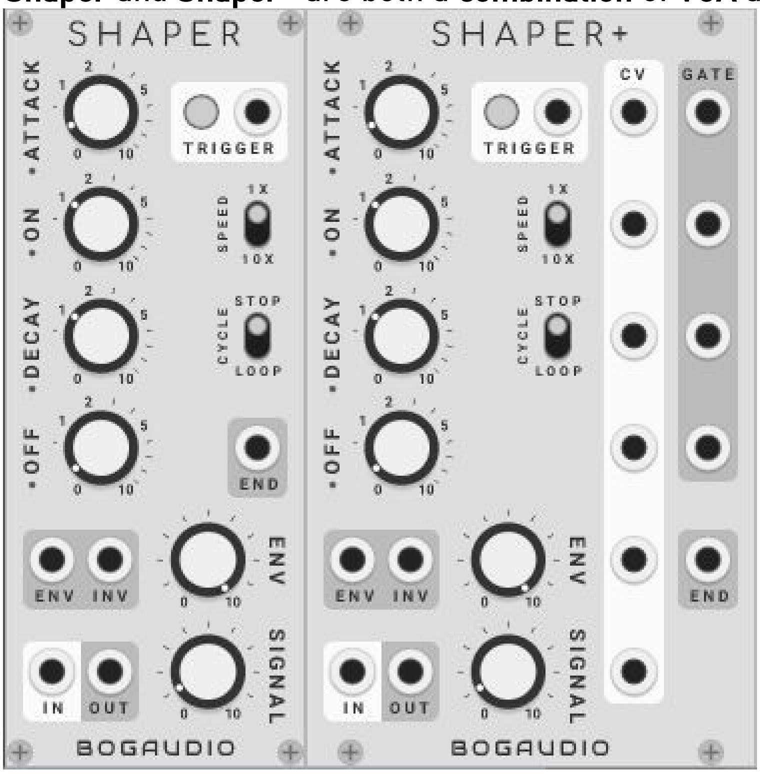
The modules of **Bogaudio** look (in the video) different, because I had to adapt their design a bit to work with them well. And I also made the colors to match a bog. **Bogaudio** currently has 50 modules in VCV-Rack and is therefore one of the most diligent developers. Some of these modules are small in size but large in performance. This also includes **VCAmp**, which has the usual sockets and can add up to 12 db to the input signal, i. e. 4 times the input value reached at 0 db on the scale. I also like the **slider** very much, whose led goes from green and yellow to red at maximum **amplification**. The modulation signal at the CV in is processed **exponentially**. Here I would have wished for another selector switch.



His colleague **VCA**, who is a standard VCA through and through, has this, with the three well-known sockets and a control to **attenuate** the signal, because here once again the maximum is at 100% of the input signal and is reached at the right stop. And as already mentioned, you also have the choice between **linear** and **exponential**.

Offset belongs back to the **attenuator** area, where the volume of the input signal is adjusted with the scale parameter. The amplitude is cut off at +/- 12db, as usual in VCV-Rack. If this is not desired, it is possible to switch off the **soft clipping** via the context menu. This makes the signal very loud again. The signal is inverted to the left of the middle position. This parameter has a CV input and corresponds to a linear modulation for modulation with an envelope. The offset control can also be controlled via CV and is the usual **offset** parameter, whereby an offset without frequency suppression is also possible here by switching off the clipping.

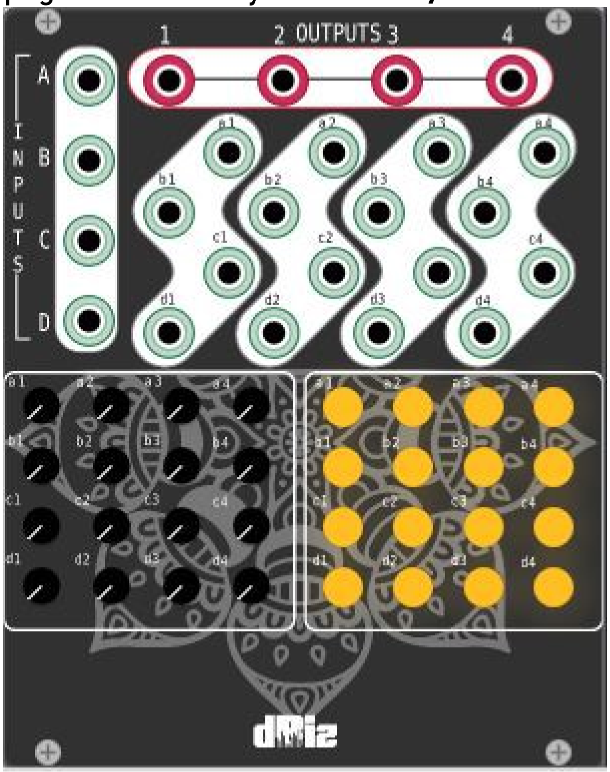
Shaper and **Shaper+** are both a **combination** of **VCA** and **envelope generator**. **Shaper+** also has a **CV input for each controller** and a **gate output for each section of the envelope**. Otherwise the two modules are **identical** and emulate the envelope unit of the **EMS V3**. The input signal at input in is amplified with the parameter signal. At position 0 there is no output signal, at 1 we already have 100%, at 2 200% and so on. As usual with an envelope generator, a **trigger** is required to start the envelope. There is an input and a button for this. If this is clicked, the envelope is triggered once when the cycle switch is set to **stop** and always repeated when it is set to **loop**. The speed between decaying and re triggering the envelope can be increased tenfold



with the **speed** switch, i. e. it takes 10x as long until the envelope is triggered again. The parameters of the envelope curve differ somewhat from the usual envelope generators and all have a control range of 0-10 seconds. **Attack** is the usual settling time, **on** defines how long the tone is held until the decay phase, and **off** defines how long it takes for the tone to completely vanish. The envelope can be tapped at **output env** and the inverted envelope at **inv**. The amount of the envelope signal is determined with the **env knob**. At the end of the envelope a trigger pulse is output at output **end**. In my opinion **Shaper+** is the more useful module, because the parameters of the envelope can be controlled by CV. The gate outputs also make sense. Instead of a speed switch with 2 options I would have liked to have a stepless knob. But still a great and versatile module.

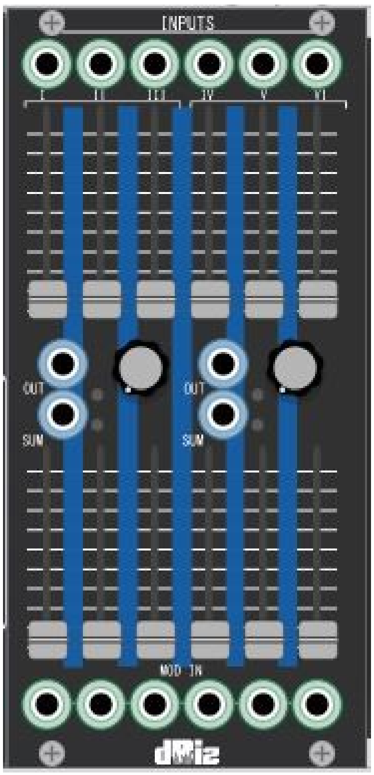
The **Bogaudio** modules are easy to use and very practice-oriented. It is fun to work with them and they do what they should do and more.

The **dBiz** modules once again look quite complex and hardly have a description on the github page. **VCA4** is only called an **in/out matrix mixer**, which is true but doesn't really help. Yes, it



has 4 inputs and 4 outputs each and to get 1 signal from input a in output 1, the black button a1 must be turned up, where 0 is on the far left, 100% in the middle and 200% on the far right and the field a1 must be yellow. That much is clear and therefore the module works as a VCA. But now it's interesting, I insert an envelope signal into **input a1**, then this modulates my input signal from **a** and I can tap the modulated signal at **1**, because **output 1** is assigned to **a1** to **d1**. But if I insert another modulation signal, here an LFO, into **input b1**, nothing happens. First a signal must also be present at **input b**, then I can set a sum signal at **output 1** from **a+b** whose ratio is defined with the parameters **a1** and **b1** and which is modulated with the corresponding modulation signal at **a1** or **b1**. And by deactivating the corresponding yellow button, the respective signal is also deactivated. It's really easy once you get the hang of it. Input and output result in a **letter number combination** and both input, rotary knob and yellow button must be activated with the same combination. Almost correct, the inputs with the combination do not necessarily have to be chosen. If they are not, the same signal can be tapped at all outputs. This means that the **active, summed and modulated input signals are output at the outputs.** And of course this doesn't only work with audio signals in the input, maybe you just take **modulation signals** and send out the output as such or you make a mixture of control voltages or also a wild mixture of all possible signals. I think this module is a **monster** in terms of its possibilities and we will certainly see it elsewhere and explore it further.

The **VCA530** is of course modeled on the **Roland VCA 530** and has all its parameters except



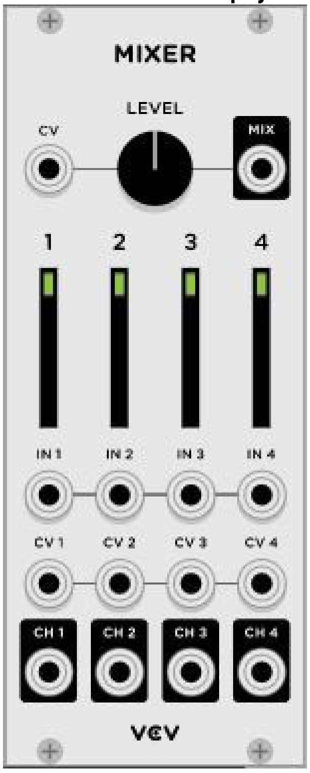
the switch between linear and exponential. In principle it consists of **2 identical VCA**s, which in turn each have **3 identical channels**. So you can connect **six sources** and adjust the volume of each one from 0 to about 3 times the input signal. Each VCA also has **3 mod in** inputs, which can also be controlled individually. The large round gain control adjusts the input volume of a VCA and the led's indicate overloads. As always, you should avoid red. A **sum** signal of the **3 inputs** and mod inputs is output at the **out outputs**. A **sum** signal of **all** inputs and mod inputs is available at the **sum outputs**. What I definitely miss here, but the original doesn't have that either, are single outputs. But clearly, this module sets other priorities, it sets standards regarding **modulation** possibilities in any case.

The VCAs from **dBiz** are both quite demanding in handling, but also offer a lot for it. I can only recommend trying them for yourself.

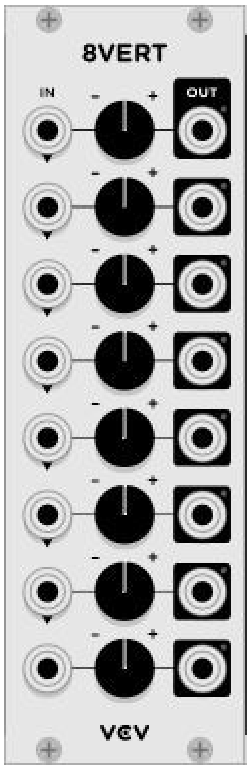
Since a VCA is something very fundamental, **Fundamental** of course also offers a lot here. The **VCA-1** is a simple VCA with the usual 3 sockets and a switch between **linear** and **exponential**. Once again, it is merely a signal **attenuator** that goes from 0 to 100%. The volume is adjusted by clicking on the **vu** display + **mouse** movement. Very nice you can see the difference between the two types of modulation when using an external modulation signal.

The **VCA-2** consists of **2 identical units**. In contrast to the **VCA-1,** the volume here is set classically via a rotary knob and **exponentially** and **linear** each have their own CV input socket and can also be used **simultaneously**. This enables interesting modulation effects.

The module simply called **Mixer** contains **four identical VCAs**. The modulation type is fixed



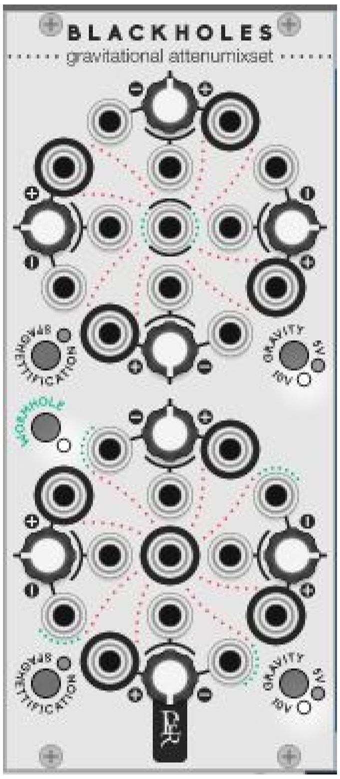
linear. In addition, this module offers a **mix** output, through which the **sum** of the individual VCAs can be output, but also when the level control is fully turned up, a doubling of the amplitude is possible. This can also be controlled by CV and an envelope signal introduced here seems to be processed exponentially.



8vert is a classic **attenuverter**. In the middle position the output value is 0, rotated to the right the input signal is output up to 100% and rotated to the left it is inverted. So we have **no** signal **amplification** here again. However, a special feature of this module is that an input signal is output at all outputs below it, if no further input sockets are occupied below it. It is therefore possible to **insert** a signal at the **top** input and **output** it at the **eight outputs** with different intensity and polarity. I really like this module, but I would like CV input jacks for the controls.

As always, the **Fundamentals** are **solid and reliable** modules that you can't go wrong with and are easy to use, especially for beginners.

And as always, where **Fundamentals** are not far away, **Gratrix** provides the standard modules with **additional** inputs and outputs. This is also the case with the **VCA-F1**. In principle, it is a **VCA** from the **VCA-2** of **Fundamental**, that is, it has separate inputs for **linear** and **exponential**. And because it is a **Gratrix** module, it has **7** of them each, just like the input. And it has **6 outputs**, whereby only the corresponding input can be tapped at each output. However, the middle **7th input** is output at **all** outputs. The **Gratrix** module also has 2 **mix** outputs, where on the one hand the sum of all inputs is output and on the other hand the signal of the middle input is amplified up to approx. 3 times, whereby the value 0 is again completely left. The modulation inputs also only affect the assigned input, except for the middle input, which again affects all of them. These additional features make the **Gratrix VCA-F1** more than just a copy and more versatile than its predecessor.

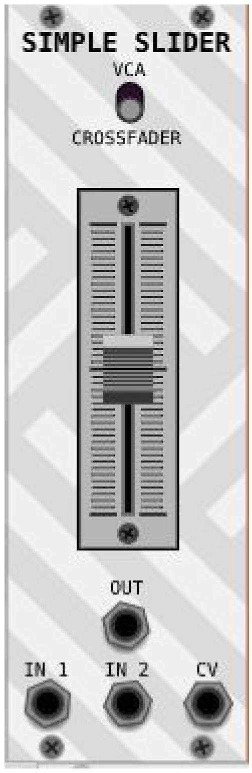


After the excellent videos by Omri Cohen, I thought about introducing **Blackholes** by **Geodesics** to the world. Isn't that like carrying owls to Athens? Maybe, but this episode wouldn't be complete without this great module. All right, let's go. **Gravitational Attumixset** is under the name, which is a nice term, but says nothing about the module. If you look at the description you see that the module consists of **8 VCA**s or **two 4-channel mixers**. Regardless of the great design, the module is actually a simple VCA. We have an input and an output and a CV in with a corresponding attenuator, which can **invert** the signal as it should. And with **spaghettification** there is also a switch between **linear** and **exponential**. And there are four of these units in each of the two areas. Nothing special so far. The output jack in the middle of a black hole outputs the sum signal of **all grouped VCA**s. That sounds interesting. A control voltage can be tapped at the corresponding outputs without an input signal. With the **gravity** switch you can configure the modulation input, +/- 10v for envelope and gate, +/- 5v for LFO and VCO. With the switch **wormhole** you activate the wormhole effect, which sends the sum signal of the first black hole to the second black hole and splits it among the inputs, so that a further signal modulation can take place here and this again can be separated and output in sum. It all sounds very complicated, but on closer inspection it is simple and ingenious. And of course, as with any VCA, you can feed in multiple audio signals and/or multiple modulation signals. Also here I can only recommend to **experiment**, because probably there is much more to do than the developers had imagined.

The **Amplitude VC Mixer** from **Hora** is one of the developer's **free** modules, which can be found in the Plugin Manager under **VCO-VCF-VCA free**, but can only be downloaded from the developer's website. The first thing that catches the eye here is the unusual design, which may seem a bit confusing at first. Here we have again **2 identical VCA** in one module. Input and output are as before, the envelope signal is put into the **strength** input and is amplified with the corresponding attenuator to the right up to 100% and inverted to the left. **Accent** defines the type of modulation, this is similar to **linear** and **exponential**, but will never be as smooth as linear. To the right, the accent is longest, to the left, short and sharp. This can also be controlled by CV. In the **led quadrant** you can see the **phase** and **amplitude** of the input signal vertically and the phase and amplitude of the parameter strength horizontally. Finally, both VCAs are output together in the **mix** output. For this an **external signal** can be mixed for which there is a separate input plus attenuator.

This module is most interesting for me if I tap the sum of both VCAs and the external signal at the mix output and both VCAs have different modulation sources. Suddenly he does things a normal VCA doesn't.

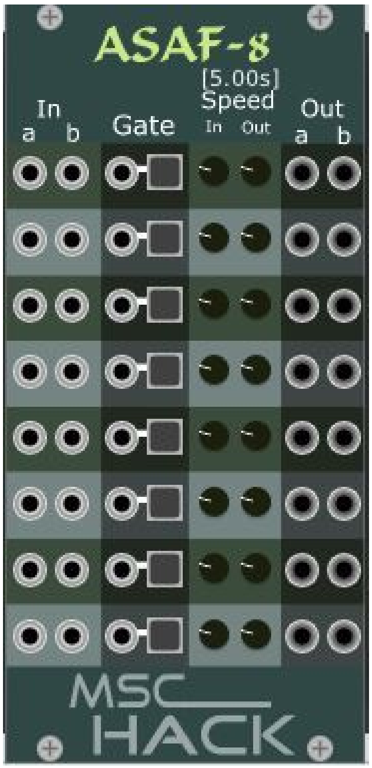
The **Pan VCA** from **MSM** modules also consists of **2 independent identical units**. At least in the switch position **norm**. Here we have an attenuator for each side, which starts at 0 on the left and does not reach twice the amplitude on the right. Each side has a separate input, output and CV input and a switch for linear or exponential modulation. There is also an output for the **sum** signal and the **inverted** sum signal. The CV input **p** and the parameter button **pan** only work in pan mode, which must be activated via the corresponding switch. These change the proportion of the two VCAs in the sum signal, whereby one of the signals never occurs alone in the sum signal. Also here it is most interesting to work with the sum of one side linear and the other side exponentially modulated and of course the panorama.



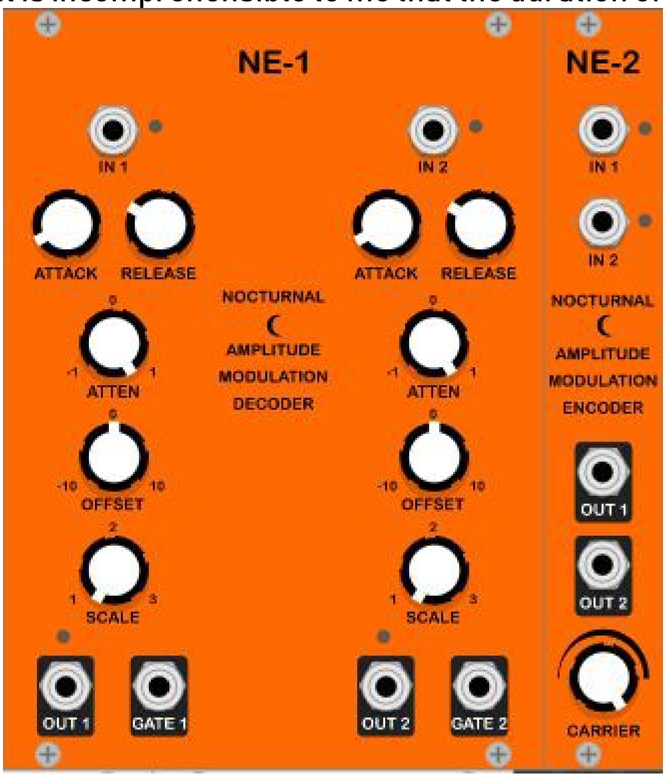
The **Simple Slider** also has **2 modes**. In **VCA mode** you can adjust the volume between 0 and 100% of the input signal or via CV. Control by envelope is not possible, but by LFO. In **crossfader mode** you can fade between 2 input signals. Here, too, control with LFO is possible, but not via envelope. Even though you can use this module as a VCA, I see the main task as a **crossfader**, because you can fade all kinds of signals into each other and control it via **LFO**.

The **MSM** modules are once again **unusual, but useful**. Maybe not for daily use, but every now and then they make sense.

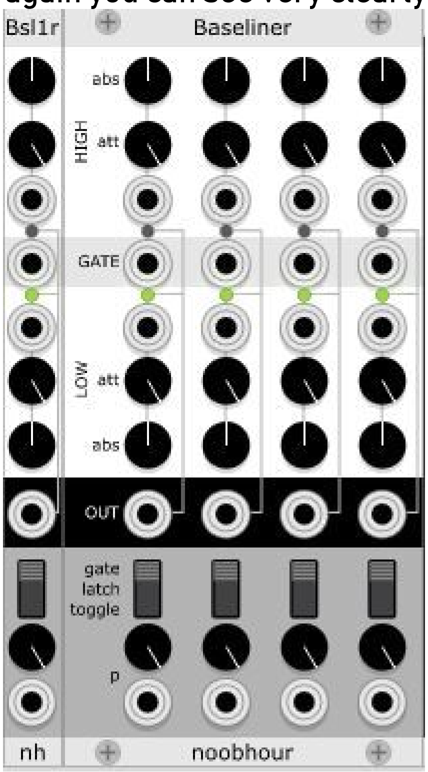
The **ASAF-8** autofader from **mscHack** is listed under **attenuator** and its task is to fade a signal in and out. This can be a **stereo** signal as well as a **mono** signal and in principle it does not have to be an audio signal. The module has **8 identical independent units**. Each unit has 2 inputs and 2 assigned outputs. The gate input expects an on/off signal, i. e. either a bipolar **gate** signal or a **trigger** pulse. An envelope only works when the original uni-polar signal has been converted. If the input receives a voltage (one volt is sufficient here), the adjacent **square** lights up green and the **fade-in** is started, the signal is then held as long as the voltage is present. If the light goes out, the voltage is gone and the **fade-out** is started. The **speed** in and out controls are used to set the duration of the two events. For this I would have wished for a CV input. By the way, when these controllers are actuated, the currently set time of the parameter is displayed in seconds. A small but useful detail. The **ASAF-8** is definitely a tool to **automate** a patch and it does that very well. Therefore, it is incomprehensible to me that the duration of the fading in and out cannot be controlled by CV. But maybe there will be an update that can do that.



The two modules **NE-1** and **NE-2** of **Nocturnal Encoder** are listed under **VCA** and **attenuator** respectively, but have very special areas of application. **NE-2** turns an **audio signal into a CV signal** and **NE-1** turns a **CV signal into an audio signal**. Both are intended for communication with the **eurorack** world outside of VCV-Rack. But what happens to an audio signal that has been

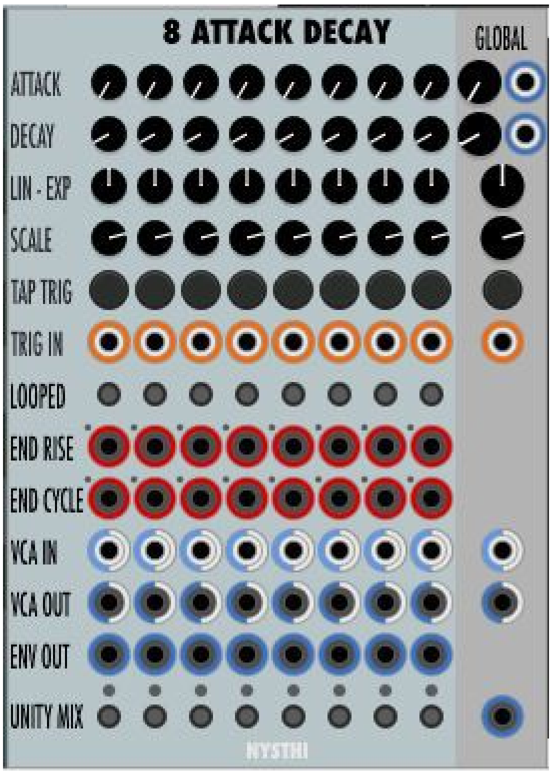


converted to CV and then back to audio? Is then the output signal identical with the input signal? Far from it and that makes it interesting for other applications as well. Obviously, we're on the **extreme margins** of VCA and attenuator here, and since we're usually talking about audio signals, I'll just take a closer look at that here. The audio signal is inserted into one of the **in** inputs of **NE-2** and output at the corresponding output. **Carrier** defines the strength of the output signal from 0 to 100%. The output signal is introduced into **NE-1**, which consists of **2 identical independent units**. In general it can be said that the **CV signal is analyzed and a new audio signal is generated from its envelope**. So basically an **envelope follower**. And this is exactly what all parameters in **NE-1** are designed for. **Attack** and **release** control the general speed of the envelope follower, with the values on the left being the lowest. The parameter **atten** defines the strength of the signal, in the middle position the value is 0, the rightmost full signal, the leftmost full inverted signal. With **offset** the signal is shifted by +/- 10 volt, thus **a uni-polar signal can become bipolar**. With **scale** the signal can be **amplified**, on the far left 100% (no amplification) is on the far right 300%. If you insert an audio signal into **NE-1**, it can be processed with the parameters and **scale** functions as an audio **amplifier**. If this signal is now reintroduced into the **NE-2**, it produces a **CV signal matching the envelope**, thus reversing the principle of the envelope follower. If a CV signal is inserted into the **NE-2**, **carrier** again defines the output strength of the signal and with the parameters of **NE-1** this can also be further shaped. However, the **offset** must then be **negative**. **Scale** does not define the volume of the signal, but the strength of the modulation. Once again you can see very clearly that it can be very interesting to use modules differently than the developer intended. **Have fun experimenting**.



Baseliner and **Bsl1r** from noobhour are listed under attenuator, but fit better under switch, so I will introduce them there first.

And of course **VCAs** and **Attenuators** are included in the more than 100 **Nysthi** modules. **8 Attack Decay** is of course first and foremost an **envelope generator** and therefore we will see it again in the corresponding episode, but it is also a full VCA or better said **8 VCA** and **1 special**. Each VCA has an input, an output and an output for the envelope. In addition, one input for a trigger pulse and one output each for a trigger generated at

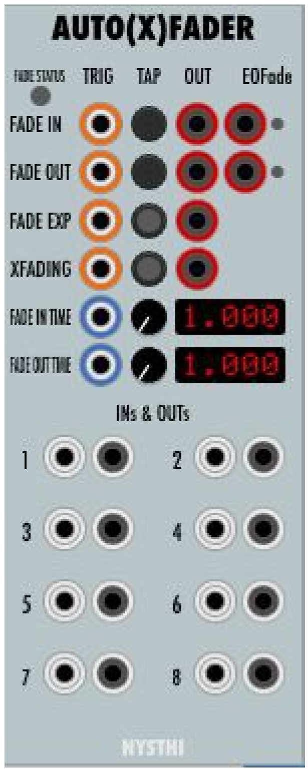


the end of the rise of the envelope and at the end of the cycle. Without a gate signal, nothing happens at first, as is usual with envelope generators. Here you can bypass this by activating the **looped** button. If you click on **tap trig** now, the trigger pulse is always repeated. If **looped** is **deactivated**, you can generate a unique trigger, a **one shot**. But of course you can also send a trigger pulse to the **trig in** input. The other parameters are **attack** and **decay** of the envelope, an attenuverter for **linear** and **exponential** modulation that is neutral in the middle and linear to the left and exponential to the right. **Scale** determines the signal gain, which is double on the far right and **inverts** the envelope on the far left. And then there's the **global** section, which caused me some headaches, because the three lower sliders don't really belong to it. Here we have one knob each for **attack** and **decay**, which affects **all 8 units** and can be controlled via CV. The modulation can also be set globally here, but can still be fine-tuned in each unit. The same applies to the **scale** parameter, which allows more nuanced but also stronger settings. And a **trigger** pulse here also applies to all units and supplements the pulses in the units. And then the **three other jacks**. The **VCA in** jack does not add its signal to the individual VCAs, but this signal can be influenced by all parameters of all units selected with the **unity**

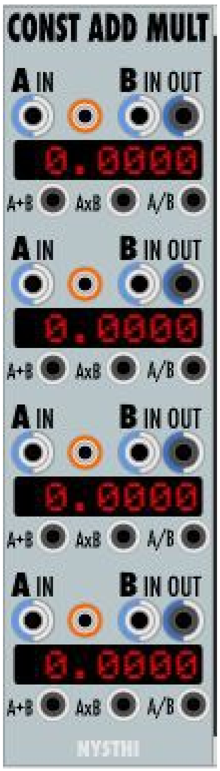
mix button. This makes **very complex envelopes** possible, of course, and I will discuss this in another episode. The **VCA out** output is therefore not the sum of the individual VCA signals, but for the signal processed with all envelopes. However, these envelopes are added and output as a sum at the **unity mix output** and can thus be used to modulate other modules. Especially these 3 jacks lift 8 attack decay far beyond a normal VCA or envelope generator, but you first have to understand how they work.

Sou means **source of uncertainty** and is a term **Buchla** brought into the world of synthesizers. **Nysthi** has several modules that have this term in their name and provide various random functions. **Sou Utils** consists of three different areas, each of which consists of **2 identical independent units**. **Scaler - offsetter amplifies** the signal with the parameter scale, whereby the same value with inverted signal is reached on the right 200% and on the left. We have no signal in the middle position. **Offset** adds +/- 10 volts, and thus a **uni**-polar signal can be made **bipolar** again and **vice versa**. A standard VCA volume effect is possible with an envelope in the CV input scale. The **offset** parameter can also be controlled via CV. The other two sections **octave folder** and **analog shift registers** are described in an other episode.

One module that I always use when I need a signal **amplifier** is **Surveillance**. The **range** switch is used to define the general behavior of the module or the global CV in parameter. If 0-10 volts is selected, the large black button functions as an attenuator from 0 volts left to 10 volts right. In the mode -5 to +5 volt, in the middle position the 0 volt and right +5 volt and left -5 is the inverted signal. Each of the 10 **ctrl** knobs is also designed as an **attenuator** and **multiplies** the signal by 5 or 10, depending on how the range is set. To the left, the signal is also **inverted** again, which is especially interesting for the 0-10 volt mode, because this also achieves negative values. And of course a different voltage can be set at each **ctrl** output. And in the **global CV in**, of course, any kind of signal can be introduced. Now it would be great if you could control all parameters via CV as well, e. g. to have a separate envelope for each parameter or velocity or LFO...

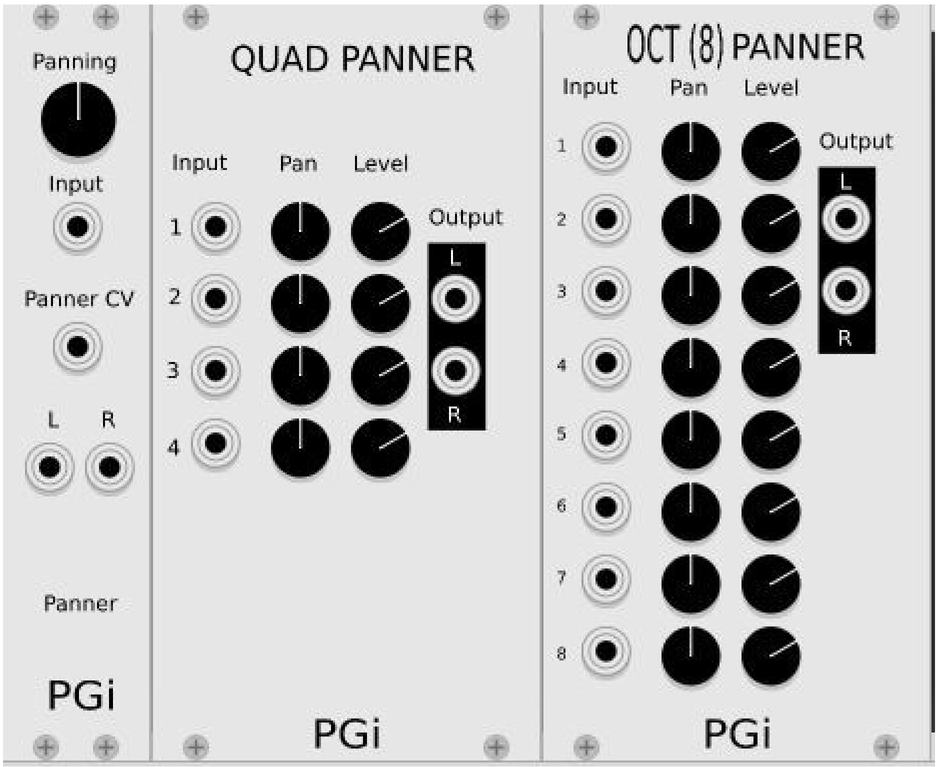


The **Auto(X)Fader** can also act as a modulated VCA in **fader mode**. In this case the parameters **fade in time** and **fade out time** take over the function of the envelope, and this can also be controlled via CV. A **trigger** signal in the input **fade in trig** triggers this or for a **one-shot** also a click on the field **fade in tap**. The outputs of the **out column** output the trigger signal unchanged. The rows **fade in** and **fade out** also each have an output **eofade**, at which a trigger signal is generated when the end of the respective fade is reached, i. e. either fully faded in or again at 0. For a **loop** it is necessary to connect the fade in **eofade** with the **fade out trig in**. Thus the end of the fade in triggers the beginning of the fade out. The **fade exp** button switches to **exponential** modulation. This can also be triggered by a trigger. The **xfading** button switches the module to **cross-fade mode**, which allows fading between the inputs and outputs of the left side and those of the right side. So you can generate up to **8 fades** or up to **4 cross-fades** with this module.



Const Add Mult adds, **multiplies** or **divides** the input signal by a fixed or variable voltage. The input signal is introduced in **A in**, a modulation signal or another CV source can be introduced in input **B in** and output unchanged at **B out**. Without a signal in **B in**, the voltage can also be adjusted by **clicking** in the number **display** and moving the **mouse** up and down. The further you click in the field on the left, the finer the setting can be. If both inputs are not used, the voltage set here is output at outputs B and A+B. The outputs are unique, whereby in the case of A/B a random value is output if B=0 and 0 if A=0.

The modules of **Nysthi** have one thing in common, they **cannot be defined to one task**. Anyway, I keep coming up with new things when I work with them. So also here is to try out and experiment absolutely announced.



The **Panner** modules from **PG-Instruments** can also be used as **attenuators**. Of course the **panorama** plays an important role here. 100% of the input signal is therefore only reached on the far left and far right. If the panorama is balanced, the amplitude is lower. With the level control of the **quad Panner** and the **oct Panner** this is adjusted. In principle, these two modules have the same structure, they consist of **4 or 8 identical units**, which have an input, a **panorama** control and a **level** control. All units are output together at one **stereo** output. The **Panner** has a slightly different structure. It lacks the level control, but it has a **CV input for panning**. Both an LFO and an envelope can be used to generate interesting amplitude changes. With the envelope, the panning slider also affects the separation between the successive envelopes. On the far left is a strong separation, on the far right it is rather fluid. It's also exciting to insert a CV signal into the input, but I'm not going to go into that until **Panner**s are reviewed in another episode.

I must admit, with some modules I let myself be deterred by their appearance. This is also the case with the modules of **PvC**, whereby I am not concerned here with questions of taste, but simply and touchingly with things such as contrast and clarity. That's why I took the liberty to adapt these modules to my viewing habits. Unfortunately, the github site of the developer tells very little about the modules, so I have to make it up again.

CoSuOf has a lot to offer that goes **far beyond** the scope of this episode. That's why I quickly **fade out the lower part** of the module. The easiest way to handle this module is to use it as a simple **attenuator**. To do this, a signal is introduced at the **pos** input and its volume adjusted with **level**. As usual, left 0 and right 100% of the input signal. With **offset** this can be shifted by +/- 10 volt. Both the upper and lower limits are cut off, thus naturally eliminating high or low frequencies. In **output sum** this signal can be tapped, in **output mus** the **inverted** signal. If you use the input **neg** everything is reversed. If the offset control is in the right half, i. e. positive, there is always a signal at the sum output that is trimmed in height, the direction of which is determined by selecting the **pos** or **neg** input. At the same time a signal cut in the low frequencies is present at the **mus** output and this can also be **inverted**. If the offset control is in the left, i. e. negative, range, **mus** outputs the treble-reduced signal and **sum** the bass-reduced signal. The further out the offset control is, the more frequencies are cut off, and the amplitude is also influenced by the level control. If the same input signal is applied to both inputs, the level controls must not be set to the same level, otherwise the signals cancel each other out. Of course, different signals can also be applied to both inputs. In such a case, the **sum** signal or the **inverted sum** signal is output at the outputs. Of course this can be influenced by **level** and **offset**. Sounds complicated? It is, and that was only half the module. The rest comes in the episode about gates.

Vamps, on the other hand, is really boring. It has 2 inputs, whereby both a stereo signal and 2 completely different signals can be introduced here. In addition, with a mono signal you get an output at both outputs of the left input. Otherwise only one input signal at the corresponding output. The modulation can be **linear** and **exponential**, and both have separate input jacks, allowing **simultaneous** modulation with both options. The output volume is set between 0 and 100% with the level control. And that's it. A simple **stereo VCA**.

The modules of **PvC** definitely **deserve more attention** and I will certainly look at them more closely now that they are more in line with my viewing habits.

Mix from **Qwelk** was also a real challenge. It consists of **2 quasi identical units**, which are

however for good reason once red and once blue deposited. They all have the property that both inputs, i. e. either **L & R** or **M & S**, need an input signal. If the **L & R** inputs are used with the same signal, this can only be tapped via the mid output. The parameter **AMP** is used to determine the maximum amplitude of the signal between 0 left and 200% right, but the attenuator **MID** must also be turned up for this, whose control range also goes from 0 left to 100% of the value of amp on the right side. In the blue module the inputs are **M & S** and the output **L**, the parameters are identical, but here **AMP** only goes up to 100%. If different signals are introduced into **L & R** or **M & S**, they add up in the **MID** or **L** output and subtract in the **SIDE** or **R** output. The **SIDE** or **R** parameter determines the strength of the subtraction. Also here the values go from 0 left to 100% right. Whereby always the right signal is subtracted from the left one, one exchanges these, one receives thereby another result. But of course the addition remains the same. If one of the input signals is a modulation signal, the amplitude of the signal is formed as described above and the modulation signal causes an offset. For **MID** and **L** this is **positive**, for **SIDE** and **R negative**. However, this does not influence the **amplitude**. If the input signals are interchanged, a positive inverted offset of the signal can be tapped at the outputs **SIDE** and **R**. The offset of the input signal is set to the value of the input signal. Of course, all this also works if none of the input signals is an audio signal. The parameters **MID & SIDE** and **L & R** each have a CV input. Here a modulation signal can be introduced to change the amplitude, e. g. an envelope signal. The strength of the modulation is adjusted via the corresponding knob. And that was the easiest handling of the module.

But my first thought was, **why are the exits of the red one in the blue area and vice versa?** Turning the knobs, very quickly showed that they were **not connected** to each other. Maybe they're supposed to be connected first. The first attempt to use either inputs **L** or **R** as input for an external signal was not really productive. The result at input **M** was exciting, though. My input signal is connected to this input, **SIDE** is connected to **S**, **L** to **L** and **R** to **R**. At output **MID** the usual signal is present, but at **SIDE** an inverted version, the peaks of which can be rounded with control **L**, i. e. a **wave shaping**. If I pick up the signal at output **L**, I also receive my original signal, but I can also smooth this out by turning control **L** to the left or deform it further together with control **R**. I get an inverted version if the input signal is inserted in **S** and **MID** is connected to **M**. And I think there's a lot more to discover. So again a VCA or attenuator that **blows all boundaries**.

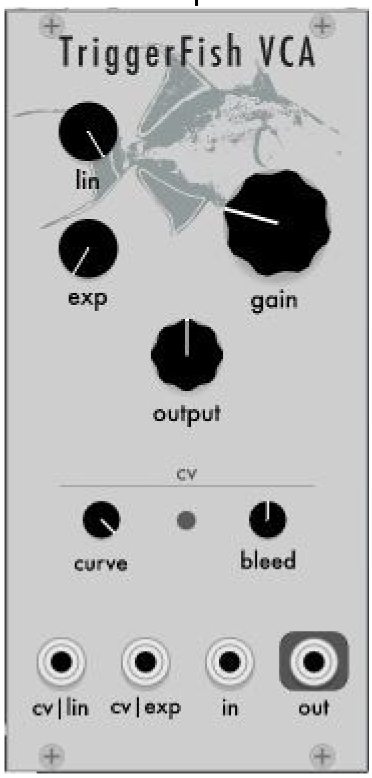
Besides the two modules **Balaclava** and **Bandana** already mentioned at the beginning, there

is also the module **AUX** in the VCA area of **Southpole**. Of course, this module is primarily intended for blending in effect modules, but can also be used as a VCA. The module has **2 aux ways**, whereby the **returns** of one are connected to the **sends** of the other, hence the name **effect loop**. If only one signal is used at input **ret l**, the same signal can be picked up at both sends. Its intensity is adjusted with the **feedback** knob. In the middle position we have no signal, right and left is identical. If you use only 1 signal at the input **ret r**, you have no signal in the middle position again, but turned to the left the signal comes only from the left send out and turned to the right from the right. If different signals are present at both inputs, it is possible to **fade** between them, whereby again no signal is output in the middle. All other controllers have no function here. But the module also has **two more inputs and outputs**, here simply called **l & r**. A signal at input **l** is output at both outputs. If 2 signals are present, each is output at the assigned output. The **mute** button mutes the outputs. In this mode it is possible to insert a modulation signal into one of the **ret** inputs. A modulation signal in the **left ret** input causes an offset in both signals, a modulation signal in the **right ret** input causes only an offset in the right signal. If different modulation signals are used in both inputs, the left one affects both signals and the right one only the right one. In this mode, the amount of the offset is set via the slider under ret. All other controls have no function, the **mute** button mutes the input signal and **bypass** switches the modulation. Of course, different signals can also be picked up at all send outputs in this module. The control under send determines the amplitude and the **feedback** control underneath determines the intensity of the offset. In the middle position the offset is 0 and turned to the right it is positive and turned to the left it is negative. If you use the other aux channel, it's the other feedback control. **AUX** is also a very complex module when used as a VCA. We will take a closer look at its qualities as an effect blending path in the effects section.

Falls is called **Attenumixer** and it can also be used differently. Easiest as a **6x attenuverter**, since it consists of **6 identical independent units**. Each has an input and output and an attenuverter, which does not let through any signal in the middle, to the right up to 100% and to the left the inverted signal. So first of all, nothing intoxicating. If no input signal is present, +/- 1 volt can be tapped at the output, i. e. in principle an offset. And the blue arrows already indicate it, if you **leave outputs free,** the signal of all above empty outputs can be tapped at the **next output**. This makes it possible to insert a **different signal into all inputs** and to pick up a **sum signal** at the lowest output, whereby the mixing ratio of the signals can be adjusted. It is also possible to form smaller sums, e. g. by occupying the third and sixth output, so that the signals of 1 and 2 can be picked up at the 3rd output and of 4 and 5 at the 6th output. I use this module e. g. to **generate very complex waves** from fundamental waveforms. By the way, a CV signal introduced into the input causes an offset, the intensity of which can be adjusted with the knob. With the range switch, the output signal can be increased tenfold, which can sometimes be very interesting for CV signals.

Southpole once again shows how you can achieve great things with small modules.

The **TriggerFish Elements VCA** is more or less standard again. It has one input and one output and one CV in each for **linear** and **exponential**. This means that both modulation types can be used **simultaneously** again. Since everyone also has their own attenuator, their ratio can also be adjusted as desired. Left as always is 0 and right 100%. **Gain** controls the volume of the input signal and **output** controls the output signal and adds saturation to it. The **curve** parameter acts on the **exponential** modulation and makes the transitions smoother when it is left and more defined when it is right. The **bleed** parameter works very subtly and adds some of the CV in signal to the output signal. Here again there is no effect at the far left and it becomes bigger the further the knob is turned to the right. An interesting VCA especially due to the possibility to mix the modulation.

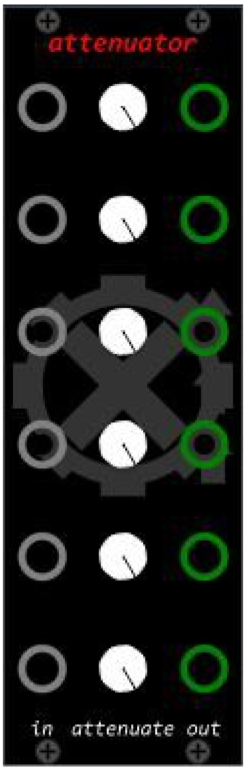


Among my most used modules are **Slap** and **Spank** by **Vult**. Both are **envelope generators with built-in VCA**, the latter especially for percussive sounds like drums. The handling is very simple. The VCA has one input and one output and to trigger the envelope a **gate** signal is needed and this envelope can be tapped at the output out to modulate further modules. The **soft/hard** switch determines whether the gate is to be closed quickly or slowly. **Soft** is slow, which means that the signal is not cut off immediately. In the envelope area we find the usual 4 parameters in **Slap**, **attack**, **decay**, **sustain** and release and all 4 can be controlled via CV and have one attenuverter each, which allows no modulation in the middle position and a positive modulation when turned to the right and a negative modulation when turned to the left. **Spank** has only the three parameters **attack**, **hold** and **decay**, which are more useful for **percussive envelopes**. However, these can also be modulated accordingly. The switch for the gate function has another option here, **loop**. This allows you to specify that the gate does not have to be re triggered each time, but **repeats** itself, making **rebounds** and similar effects possible.

Knobs consists of **2 identical independent units**. Each unit has a large knob whose function is determined by the switch next to it. In **uni** mode it outputs 0 volts on the far left and 10 volts on the far right, in **bi** mode it outputs -5 volts on the far left and +5 volts on the far right and in **mix** mode it outputs 0 volts on the far left and 100% of the signals of the corresponding small buttons on the far right. These 4 are all equal attenuverters, so in the middle 0 and left negative and right positive. This means that the signal is inverted on the left. If the big button is not set to mix, it causes an **offset** of the signals at the 4 inputs. Up to 4 different signals can be **mixed** per unit and the individual signals can also be **inverted**.

Flux is also a very versatile module and we will therefore meet it again elsewhere. Basically

it consists of **4 independent identical units**. To use it as an **attenuator**, the **mode** switch must be set to **off**, the other two functions do not interest us today. The **range** parameter ranges from 0 to 200% on the far left and with **offset** a value of +/-5 volt can be added to this signal. The switch **follower** defines whether the **rate** parameter should be **linear** or **exponential**. In the third mode **env**, the **rate** parameter follows the input signal as closely as possible. In and out are the usual sockets and **clk** is for an external clock signal, which is not relevant today either. Much more important is the well-known **Vult modulation display**. By clicking on one of the 6 letters and moving the desired button, it is selected and can be controlled via CV and the modulation strength and its phase, i. e. positive or negative, can be adjusted with the corresponding attenuator. So if I want to control the amplitude with an envelope, I connect the output of the **ADSR** to one of the CV input sockets and select the range parameter and I already have a volume envelope. Or I let the envelope modulate the rate parameter and get a kind of filter sweep or morph effect. And of course it can be combined with each other and we have 4 units and of course the other modes.



Also with these modules **Vult** shows once again impressively what is possible in VCV-Rack and that **versatility does not have to be complicated**.

The conclusion of today's episode is made by **TheXor's Attenuator** module and actually it's not anymore. An input, an output and a button that goes from left 0 to right 100%. And that in **6 units**. Pretty simple and and sometimes that's all you want.

And this brings us to the end of episode 13 and I've introduced all the modules listed under VCA or attenuator. For me, this episode was a real surprise, because I hadn't expected such a great variety on this topic. But that also showed me that there is still a lot of unused potential slumbering

here. I hope you have also become curious and just try something different. Next week, I'm looking at envelope generators. As always you will find links to the most important websites and also to the developers of the modules presented this time.
And not to forget, many thanks to Andrew Belt and all the module developers without whom all this would not be possible. That's it from me, let it go well with you

Afterword

And this is also the end of this summary. I am sure, you can find a lot of mistakes on the subject and in the language. Google translate gives sometimes very strange translations, but when you see my german script, you find a lot of colloquial language. And I use a lot of allegory and metaphors and google translate makes a word-for-word translation.

This book is a non-commercial work and may be printed or otherwise distributed as a whole or in parts, but only free of charge.
The pictures of the modules are made from screenshots, feel free to use them. The graphics are from somewhere in the web, don't ask me from which sites. I could have drawn them by myself, but sometimes I'm a lazy bastard.

That's also one of the reasons, **I made a few things clickable**. You can click on the number at the bottom of the pages to jump to the “Table of Contents”, where you also can click on a lot of things. You can click on “Editorial”, the terms “Episode *” and the name of the module to jump right away to that respective page.
If you click on the title of an episode, it should open the video in your browser and if you click on a timestamp, that should open the video at exactly that position. I hope that works for everyone.

I take in no way any responsibility in any damage to any software, hardware or persons, animals, plants and other living things from this planet or an other in the past, present or future.

May your prefered protector be with you!

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Tables on github: https://github.com/the1andonlydrno

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Links

The most of the modules can be downloaded free via the Plugin Manager https://vcvrack.com/plugins.html.

The official forum: https://community.vcvrack.com/

The official VCV-Rack facebook group: https://www.facebook.com/groups/vcvrack/

21kHz	https://github.com/21kHz/21kHz-rack-plugins
Aepelzens Parasites	https://github.com/Aepelzen/AepelzensParasites
Animated Circuits	https://www.animated-circuits.com/
AS Modules	https://github.com/AScustomWorks/AS
Audible Instruments	https://github.com/VCVRack/AudibleInstruments
Autinn	https://github.com/NikolaiVChr/Autinn/
Autodafe	https://github.com/antoniograzioli/Autodafe
Bacon Music	https://github.com/baconpaul/BaconPlugs
Bargkass	https://github.com/korfuri/
Befaco	https://github.com/VCVRack/Befaco
Bidoo	https://github.com/sebastien-bouffier/Bidoo
Blamsoft	http://blamsoft.com/
Bogaudio	https://github.com/bogaudio/BogaudioModules
CharredDessert	https://github.com/JerrySievert/CharredDesert
dBiz	https://github.com/dBiz/dBiz
E-Series	https://github.com/VCVRack/ESeries
Evil Turtle Productions	https://www.evilturtle.nl/store/plugins/vcvpack.html
Friedrichs Audio	https://gumroad.com/friedrichsaudio
Frozen Wasteland	https://github.com/almostEric/FrozenWasteland
Fundamental	https://github.com/VCVRack/Fundamental
Geodesics	https://github.com/MarcBoule/Geodesics
Gratrix	https://github.com/gratrix/vcv-gratrix
Hora	http://www.hora-music.com/
Lindenberg Research	https://github.com/lindenbergresearch/LRTRack
LogInstruments	https://github.com/leopard86/LOGinstruments
mscHack	https://github.com/mschack/VCV-Rack-Plugins
MSM	https://phal-anx.github.io/

Nocturnal Encoder	https://github.com/djpeterso23662/NocturnalEncoder
Nohmad	https://github.com/joelrobichaud/Nohmad
Nonlinear Instruments	https://github.com/NonLinearInstruments/NLNRI_VCVRackPlugins
Nysthi	https://github.com/nysthi/nysthi
PGi	https://github.com/imekon/PG-Instruments
PvC	https://github.com/phdsg/PvC
Qwelk	https://github.com/raincheque/qwelk
Robotic Bean	http://roboticbean.com/
Rodentmodules	https://github.com/RODENTCAT/RODENTMODULES
s-ol	https://github.com/s-ol/vcvmods
Sonus Modular	https://gitlab.com/sonusdept/sonusmodular
Soundmit/Autodafe	https://www.soundmit.com/en/vcvtwlv
Southpole	https://github.com/gbrandt1/southpole-vcvrack
Southpole Parasites	https://github.com/gbrandt1/southpole-vcvrack
Squinky Labs	https://github.com/squinkylabs/SquinkyVCV
Submarine	https://github.com/david-c14/SubmarineFree/
TheXor	https://github.com/The-XOR/RackPlugins
TriggerFish Elements	https://github.com/JTriggerFish/TriggerFish-VCV.git
trowaSoft	https://github.com/j4s0n-c/trowaSoft-VCV
Valley	https://github.com/ValleyAudio/ValleyRackFree/
VCV	https://vcvrack.com/Parametra.html
Vult	https://modlfo.github.io/VultModules