Voltage Modular: Cherry Audio modules

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Table of Contents

1 Additive Oscillator

The Additive Oscillator lets users combine the first eight partials in the harmonic series to create composite tones with a choice of sine, triangle, or variable-width pulse waves. Inspired by 70's "West Coast" synthesizer modules, it includes separate outputs for each partial, and allows the partial levels to be modulated in a number of creative and intuitive ways. First we'll explain the function of each control, then we'll explore the Additive Oscillator in use.

Keyboard CV - Standard 1V/octave CV input for pitch control.

Range - Sets the octave range in standard footage increments. The Lo setting is intended for modulation purposes and will generally be below audible range. (unless you're a whale and can hear 1Hz)

Wave Select Buttons - These select sine, triangle, or variable-width pulse wave. Only one can be selected at a time.

Pulse Width Control and Display - This knob allows manual control of pulse width. Center position will generate a 50% square wave; rotating left or right results in narrower pulse-widths. The waveform display shows the current width of the pulse wave.

Freq Mod - Applying CV's here modulates the base pitch of the oscillator; in other words, all partials are affected equally. This can be used for basic vibrato or siren effects with sub-audio modulation speeds, or wilder FM cross-mod when using audio-range signals.

PW Mod - Short for pulse-width, this mod input allows real-time modulation of the pulse wave. It won't have any audible effect if the sine or triangle wave is selected.

Top Row CV Jacks and Attenuator Knobs - These CV inputs allow level modulation of each partial. The small CV Amt knobs are CV attenuators. Incoming CV's are combined with slider settings.

Partial Level Meters - These give a visual display of partial volume levels.

Partial Level Sliders - Sets the levels of each partial in the harmonic series. 1 is the the fundamental frequency, 2 is the second harmonic, etc.

Partial Individual Out Jacks - Separate audio outs for separate processing of individual partials. Note that these are affected by the partial slider volume settings.

Center - Defines the center frequency of the slider "peak."

Init Amt - Sets the initial amplitude of the the slider "peak."

Center CV Jack and Attenuator Knob - CV input for external modulation of Center frequency.

Focus - Sets the width of the slider "peak."

Focus CV Jack and Attenuator Knob - CV input for external modulation of "peak" width.

Please see the "Modulation Time C'mon" section below for more information about Center, Focus, and slider peak.

Mix Out - Audio output of the mix of all partials.

1.1 Basic Use

Let's start by setting up a basic patch consisting of an Additive Oscillator and a voltage-controlled amplifier, as shown below:

Pitch from the IO Panel goes to *Keyb CV* input, *Mix Out* is routed to amplifier *Input*, and gate from the IO Panel goes to the amplifier's *CV In*. Amplifier audio out is routed to IO Panel *Main Outs to host*.

If you ignore the modulation options, the Additive Oscillator is super easy to use, and operation is just like using a standard analog oscillator - in fact, it's effectively the same as opening eight individual oscillators and tuning each to partials one through eight of the harmonic series (for example, 100Hz, 200Hz, 300Hz, 400Hz, 500Hz, 600Hz, 700Hz, and 800Hz).

With the above patch set up, go ahead and play the Additive Oscillator. Adjusting the fader levels affects the volume of each harmonic, sort of like a vintage tonewheel organ on steroids. Notice the blue level meter to the left of each slider - this may seem superfluous now because the meter is displaying the static level of the adjacent fader, but their usefulness will become clear once we begin exploring the Additive Oscillator's modulation possibilities.

1.2 Modulation Time, C'mon

If you're a seasoned synthesist, all of the controls at the left of the panel should be familiar. Things start getting nutty when the Additive Oscillator's modulation controls are used. Let's begin with the CV level mod jacks and wee knobs at the top of the panel:

1.3 Partial Level Mod Controls and CV Attenuators

The CV input jacks above each partial slider allow CV control of each individual partial level. The small knob beneath each CV jack is an attenuator for setting the amount of CV voltage. Note that partial level CV mod works *additively* with the current fader level setting. Of course this easy to see by looking at the blue level meter, which always shows the actual output level.

1.4 Center and Focus Mod Controls and CV Attenuators

Things get pretty awesome here! The Center and Focus controls essentially let you create a "peak" of faders and move the "peak" up and down across the faders either manually or via CV. It's much easier to demonstrate than it is to explain, so let's set the controls as shown below:

Notice that all sliders are set at zero, but the meters show a peak - partials three and six are quiet, while four and five are louder. Adjusting the Center *Init Amt* knob adjusts the level of the peak. Turning the *Center* knob sweeps the peak up or down across the partials. Remember that, like the partial level CV mod at the top of the panel, Center and Init Amt control settings combine with the existing fader settings and level CV mod.

(This is why we zeroed the slider settings for this demo, but of course these can set any way you like.)

You may have noticed that sweeping the *Center* control sounds really cool, which is why there's a *Center CV Mod* in jack and attenuator knob. This can be modulated by anything that outputs a voltage, but an LFO at a slow rate is a good place to start.

The *Focus* knob adjusts the peak width, from one partial wide to all partials at full blast. To its right is a *Focus CV Mod* in jack and attenuator knob, allowing CV modulation of *Focus* width.

1.5 Advanced Additive Oscillator Madness

- Try routing the two sections of a Mini LFO to *Center CV Mod* and *Focus CV Mod*. This simple setup will go a long way, especially if you use the triangle wave on one of the CV ins and the square on the other.
- Embellish the above routing by routing another LFO to the *PW Mod CV* in and selecting the pulse wave on the Additive Oscillator.
- If you've purchased the Misfit Audio Drum Modules package, the Drum Trigger Sequencer is a wickedly fun modulation companion - it's eight individual channels enable each harmonic partial to have its own complex rhythmic sequence. BTW, after experimenting with this, we added a couple of milliseconds of lag into the level CV ins to prevent clicking with hard on/off gate voltages.
- We've saved the wackiest for last. The Additive Oscillator's partial separate outputs can be routed *back into any of its mod inputs* for all manner of audio-range frequency modulation madness. Try routing these to the *Freq Mod* and *PW Mod* inputs as well as individual partial CV ins.

2 AirStep Oscillator

AirStep is a digital oscillator that allows sampled waveforms to play in stepped, sequential patterns of up to 64 steps in length. Step durations can be controlled via internal or external clock source, and as you might expect, AirStep features a number of CV mod inputs as well as dual trigger outs, allowing external augmentation of patterns.

2.1 How It Works

AirStep includes close to 300 onboard wave samples. One of these waves is selected for each step of a pattern. Duration, crossfade percentage, level, and tuning can be specified for each step, and pattern length can be up to 64 steps. If "normal" *Internal* or *External* clock mode is chosen, AirStep plays through the sequence beginning at step one when a gate voltage is received. If the wave crossfade settings are at 0%, waves will abruptly transition from one to the next, creating a rhythmic feel. If longer crossfade settings are used, waves smoother transition, creating gently evolving textures.

If *Manual* clock mode is active, the entire pattern can be smoothly or abruptly swept with the *Initial Position* knob or via CV for wildly evolving tonal and melodic "swept wavetable" style effects. (AirStep is not a wavetable oscillator - in fact, its stepped waveform patterns do a whole lot more than wavetable synthesis, but we'll further explain this later.)

2.2 AirStep Pattern Control Section

This is the giant section in the middle of module, and it's where sample waves and patterns are configured as well as how they'll play back.

Step- Patterns can be from 1-64 steps in length. The number of steps in the current pattern is displayed at the left of the screen. This isn't an adjustable parameter, per se, but steps may be added or removed using the *-Remove Step* and *+Add Step* buttons at the bottom of the window.

Waveform- Clicking waveform name area opens a pop-up menu where waves can be selected. These are subdivided into folders. Most of the waves will loop, allowing endless playback. Waves found in the *One Shot* and *One Shot Reverse* folders play for a finite time then stop.

Duration- Defines how long a wave will play with lower numbers playing shorter and higher numbers playing longer.

xFade- Any value over zero adds a crossfade from the current wave to the next, expressed in percentage. If crossfade is set to maximum, the crossfade begins at exactly the halfway point of the current step (50% setting), and ends at exactly the halfway point of the following step; the exact point in time is dependent upon each step's duration.

Level- Sets the volume of the current step.

Semi- Sets the basic range for the current step in semitones. Tuning range is up or down 36 semitones.

Fine- Fine pitch setting for detuning. Range is just over a fifth, up or down.

Trigger A/Trigger B- Enabling these boxes toggles two independent trigger CV at the very start of the step, sent to the *Trig Out A and Trig Out B* jacks, respectively. These have plenty of creative uses; the simplest would be triggering drum sounds. More applications open up by combining the trigger outs with a Cherry Audio Trig To Gate module.

Pro Tip: Using Trigger Outs For Clocking

> The trigger outs were initially conceived as "one-shot" trigger sources for drum sounds, sequencer start/stop, etc., but enabling the trigger for **all** steps of a pattern

effectively creates an external clock out. It isn't even necessary to enable the trigger box for each step - just click the *Global* button for *Trg A* or *Trg B* and check the box to enable it for all pattern steps. This makes syncing other modules to AirStep really easy. Patching this trigger out to a Clock Divider module and/or a Trigger to Gate opens up many creative possibilities. Furthermore, if you're using a module that needs a reset signal for accurate timing, enabling the other trigger out on pattern step 1 only works well.

Remove Step/+Add Step- Clicking these adds or removes steps pattern steps.

Inserting, Adding, and Deleting Steps Between Existing Steps- This isn't a visible button or control, but right-clicking on a step number at the left side opens a pop-up allowing insertion of steps before or after the step number you've clicked on, or deletion of the step. Inserted steps will have the same settings as the currently clicked step.

Page 1/2/3/4- A pattern may be up to 64 steps in length, but AirStep's display can only display 16 steps at a time. If a patterns exceed 16 steps, the *Page 2* button will become active and allow viewing of all steps. Additional *Page* buttons become active as more steps are added. The *Step* numbers in the left column and all parameters change to reflect the currently displayed steps. So much typing to explain something totally obvious, am I right?!?

2.3 Global buttons (Top Row)

The *Global* section contains a duplicate of the *Waveform*, *Duration*, *xFade*, *Level*, *Semi*, *Fine*, *Trg A*, and *Trg B* step controls, as well as a toggle button atop each parameter. If one or more of the buttons is engaged, the parameter settings in this row will override the individual step settings.

The idea behind global pattern controls is that the setting of some parameters are often the same for every step in a pattern (most commonly *Duration* and *xFade*). In this case, global controls save you from having to set a parameter to the same value for every step of the pattern. Because each *Global* button functions independently, it's easy to choose which parameters the global setting applies to.

Individual step settings will dim when a global button is engaged for a parameter, but step settings are retained (in case the G*lobal* button is bypassed).

2.4 Position, Clock, and Speed (Bottom Row)

The very bottom row of controls define how AirStep's patterns behave. We'll start in the middle, because we like to keep you on your toes!

CLOCK MODE

The *Clock Mode* setting determines the timing of pattern playback. The three modes are as follows:

- **Internal (INT)**- Patterns are locked to AirStep's internal clock. This is the default setting.
- **External (EXT)**- When an external sync signal is plugged into the *Ext Syn In* jack, patterns lock to incoming sync signals.
- **Manual (MAN)**- Manual sync mode disconnects pattern clocking altogether, allowing manual control of wave position. All *Duration* step controls are disabled (including *Global*).

Initial Position and Position Mod CV input and attenuator- This is only active when *Manual* clock mode is selected. It defines the current static wave position and is visibly represented by the red horizontal display line.

The knob range for each wave is divided across the knob throw according to the number of waves in the current pattern. For example, if the current pattern contained two waves, the first half of the knob range would represent the first wave, and the second half of the knob range would represent the second wave. If the current pattern contained four waves, the knob range would divide over four sections, and so on, up to AirStep's 64-step maximum.

Since *xFade* settings represent a percentage (i.e. not a time value), step *xFade* settings apply to wave transitions in manual clock mode as with internal and external modes.

If you're using one-shot samples in manual clock mode, you'll find that samples initiate playing one time just as the wave position "arrives." It's a little tricky, but suffice to say, our programmers did a bang-up job of making this work really well - it's one of those things that just behaves how you'd want it to.

Initial position can be CV-controlled using the *Position Mod* CV input and attenuator. CV modulation works in conjunction with the current setting of the *Initial Position* knob.

Initial Position/Manual Clock Mode Pro Tip

"Shutting off" clock and using *Manual* mode is useful for setting pitches one step at a time when creating melodic patterns. You can take this idea further by enabling the *Global* button for *xFade* and setting it to zero; this guarantees that each wave plays discretely (i.e., not partially blended with the preceding or following wave). Once pitches are set, the *Global xFade* button can be toggled off.

Loop- Enabling loop causes the pattern to return to step 1 following the last pattern step. The pattern will end at the last step with Loop disabled, but will continue to play the last wave step (as an analog-style oscillator would) if the waveform is looped. If the last step's waveform is a "one-shot" (i.e., non-looping), the wave plays and sound stops at the final pattern step.

Direction (DIR)- The Direction button toggles through three step playback orders: forward (1-2-3-4, etc.), reverse (4-3-2-1, etc.), and forward and reverse (1-2-3-4-3-2-1).

Speed and Speed Mod input and attenuator- The *Speed* works in conjunction with the pattern window *Duration* settings and acts as an overall multiplier or divider, affecting the entire pattern playback rate. It's used to speed up or slow down the entire pattern without tweaking the *Duration* settings for each step.

The *Speed* knob is voltage controllable via the bipolar CV jack and attenuator to its right.

2.5 Inputs, Outputs, and Controls

This is everything else on left and right side of AirStep that we haven't covered thus far.

Pitch CV jack- Accepts a CV input for pitch. Typically this would come from the *Pitch* jack in the IO Panel *CV Sources* section, or from a sequencer pitch CV out.

Gate CV jack- Accepts a standard +5V gate voltage input, typically from the IO Panel *CV Sources* section, or from a sequencer gate output. The *Gate* input is used to initiate pattern playback when *Internal* or *External* clock mode is selected. If *Manual* clock mode is selected, the *Gate* jack is effectively disconnected.

Ext Sync In jack- Used in conjunction with *Clock Mode/Ext*, this accepts a 96 PPQN sync signal to allow external clocking of patterns. **It's not a clock input** like you'd see an analog step sequencer (one clock= one sequencer step). It can be used with sync signals from either the IO panel *Transport/Sync Out* jack, or a Sync Generator module if you're using the standalone version of Voltage Modular. The sync signal can also be multed to a Sync Divider module if you're like to lock up to modules that expect to see standard "slow" clock signals.

With the *Speed* knob zeroed (center position), a pattern step duration setting of 24 is equal to a quarter-note. (48 = half-note, 16 = eighth-note, etc. We trust you to do the rest of the math.) When the clock mode is set to *Manual*, the *Speed* knob is quantized to integer values. This makes it easy to multiply or divide the overall pattern rate to rhythmic playback values that make musical sense.

Frequency Mod CV input and attenuator- Allows negative or positive pitch modulation. It's applies to all waves equally and has range of five octaves, up or down. That should be plenty!

Ext In A/B jacks- The *Ext In* jacks let you use an externally patched signal at any step. To use an *Ext In* jack, plug a source into the desired jack, click a waveform name in the pattern area and select *External Input A* or *B* at the top of the pop-up menu list.

At the risk of pointing out the obvious, make sure the external audio source is making noise when the pattern step plays. This isn't an issue if the source is an oscillator or something else that sounds continuously, but it can be if the sound source is a drum/percussion hit or other quick sample. In this case, one of the trigger outs could be used to fire off the sound at the appropriate time. See how we thought of everything?

Trig Out A/B jacks- These are the trigger output jacks corresponding to the *Trg A* and *Trg B* buttons described in the *AirStep Pattern Control Section* above.

Out- AirStep's audio output jack.

3 AirVector Envelope

AirVector is a quad envelope for audio sources that dynamically mixes up to four input sources. The volume of all four sources is controlled in four quadrants by a virtual joystick, and joystick movements can be programmed with a flexible and easy-to-use vector envelope. It's inspired by the vector synthesis concepts used in the classic Prophet VS, Korg Wavestation, and Yamaha SY22/TG33 synthesizers.

It's closely related to the Cherry Audio AirWave vector oscillator module, but unlike AirWave, it contains no onboard sound generation. Its primary intention is to dynamically mix audio waves to create composite sounds, but it can be used with any audio source material, from simple waves to entire songs.

With four individual outputs, it can even be used for quad-surround audio mixing.

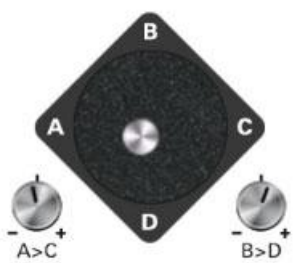
AirVector is functionally identical to AirWave, less the audio waves and associated controls. It's a bit smaller (18 HP versus 24 HP), and because it only accepts external audio inputs, there are no pop-up menus to configure the input source.

3.1 Inputs, Outputs, and Controls

We'll jump around the panel a bit, going from the basic stuff to the more intense vector madness parameters.

Gate CV jack- Accepts a standard +5V gate voltage input, typically from the IO Panel *CV Sources* section, or from a sequencer's gate output. The gate input is used to initiate AirVector's vector envelope; its functionality varies depending on the vector envelope's button settings (more on this later).

3.1.1 VECTOR JOYSTICK



Vector Joystick- The silver-knob-floating-over-a-diamond is a joystick controller that allows real-time control of the volume balance of up to four input sources. When the vector envelope is enabled, it also moves to show the current vector envelope position.

Like all knobs and controls in Voltage Modular, onscreen joystick movements can be recorded in a DAW as MIDI CC data. This is especially cool if you have a hardware instrument with a joystick controller (*see A>C and B>D joystick position knobs section below for setup info*), but the same can be accomplished by simply moving the onscreen vector joystick control.

A>C and B>D joystick position knobs / A>C and B>D Mod Input Jacks and Attenuators- A joystick describes control positions on two axes: X representing horizontal, and Y representing vertical. The *A>C* and *B>D* knobs are tied to the joystick X and Y axes.

Note that the MIDI controllers cannot be assigned to the joystick itself, so the *A>C* and *B>D* knobs are useful for assigning controllers within Voltage Modular, or hardware controllers such as joysticks, sliders/knobs/etc.

The *A>C Mod* and *B>D Mod* CV input jacks and bipolar attenuators are located to the left and right of the vector joystick diamond. These allow CV control of the *A>C* and *B>D* knobs, respectively. This can get a little nutty if you're trying to control both the X and Y axes simultaneously, but remember that AirWave can be configured for two waves at opposite sides with simple crossfading between them. If you're more ambitious, you can also use both mod inputs at once for more random results.

3.1.2 VECTOR ENVELOPE

The Vector Envelope enables precise automation of the vector mix joystick position. Unlike some implementations, it's designed to be easy to understand and setup is super quick. Please take a moment to carefully read the *How It Works* section, we promise, it's really simple to wrap your head around!

3.1.3 How It Works

AirWave's vector envelope has four steps, with each step's settings displayed in a horizontal row. Each step contains two settings defining the joystick position for that step. When the vector envelope starts playing, it moves through the steps. Instead of abruptly jumping from step to step, AirWave smoothly transitions from one position to the next, at a rate defined by the step's *Time* setting (as well as the overall *Speed* knob setting).

3.1.4 Vector Envelope Window Parameters

The step parameters are displayed across the top of the window and are the same for each of the four steps. All settings in the Vector Envelope window can be changed by either clicking and dragging a number, or by double-clicking and typing an exact value. Let's go over its parameters:

- **A>C**- X-axis (horizontal) position of the joystick for that step. Setting this to 0 corresponds to center position, i.e., a 50/50 mix of A and C waves. Negative settings move the joystick left toward wave A, positive settings move the joystick right toward wave C.
- **B>D**- Just like the A>C setting above, but defining the Y-axis (vertical) position of the joystick for that step. Setting this to 0 corresponds to center position, i.e., a 50/50 mix of B and D waves. Negative settings move the joystick up toward wave B, positive settings move the joystick down toward wave D.
- **Time**- Defines the length of the step, i.e. how long it takes for the joystick to arrive at the next step position. Lower value settings are faster, higher value settings are slower.
- **Snap**- Clicking the *Snap* button immediately sets that step's A>C and B>D position parameters to the current joystick position. This makes setting the A>C and B>D positions very easy. **The *Snap* buttons only work when the *Env Active* button is disabled.**

3.1.5 Vector Envelope Controls

Env Active- Enables and disables the vector envelope. Disabling the vector envelope is useful for "manually" playing the vector joystick, or if you're setting vector envelope window parameters using the *Snap* buttons discussed above.

When the *Env Active* button is toggled on, you'll notice that the vector envelope begins moving through its steps, even if no gate signal is present. It's designed this way to enable ever-shifting drones using the internal wave samples or the external audio inputs. (If the vector envelope were to only move when a gate voltage was present, the envelope would cease to move during an envelope generator's release phase.)

Retrig- When *Retrig* is enabled, the vector envelope restarts at step one every time a new gate signal is received, meaning every note will have the same vector envelope. Disabling *Retrig* causes the vector envelope to play continuously and ignore newly received gates. This is useful for continuously evolving sounds while playing a melody (particularly if the *Loop* button is engaged for non-stop vector movement).

If you're using the IO Panel *CV Sources Gate* output, vector envelope restart will be affected by the IO Panel *Single* and *Multi* trigger setting as with standard envelopes if *Retrig* is enabled. If *Single* trigger mode is selected in the IO Panel and the keyboard is played legato holding a previous key while playing new keys), the vector envelope will *not* restart when new keys are played. If *Multi* trigger mode is selected in the IO Panel, the vector envelope always restarts from step one when a new key is played, regardless of existing held notes.

Play Order- The *Play Order* button opens a pop-up menu that defines the playback order of the vector envelope steps. The default setting is 1–>4; this means the vector envelope plays step 1, step 2, step 3, then step 4. If the *Loop* button is engaged, it returns to step 1 and continues. The –> arrow indicates forward playback.

Selections with <–> bidirectional arrows indicate playback in both directions. For example 2<–>4 would play step 2, step 3, step 4, step 3, then step 2. If the *Loop* button is engaged, it would play step 2, step 3, step 4, step 3, then step 2, (loop here) then step 3, step 4, etc. Note that it won't repeat the same step twice when the loop occurs. We thought of that!

Loop- Enabling the *Loop* button causes the vector envelope to return back to the first step (depending on the *Play Order* setting) and continue playing.

Speed and Speed mod input and attenuator- The *Speed* works in conjunction with the vector envelope window *Time* settings and acts as an overall multiplier or divider, affecting the entire vector envelope playback rate. It allows you to globally speed up or slow down the entire vector envelope without tweaking the individual *Time* parameters.

The *Speed* knob is voltage controllable via the bipolar CV jack and attenuator directly beneath.

3.1.6 A/B/C/D CONTROLS

All of the following parameters, inputs, and outputs are available individually for each of the four waves.

A/B/C/D Level knobs- These allow a static boost or cut in volume for each waveform. Although all of the built-in samples are normalized for maximum volume, some sound louder or quieter than others; the level knobs make it easy to equalize each wave's initial (i.e., pre vector mix) volume. The *Level* knobs also affect signals plugged into the A/B/C/D *Ext In* jacks.

A/B/C/D In jacks- Patch the desired audio sources into these. Any number or combination of these can be used. For example, *A In* and *C In* only could be used to for vector control of two audio sources.

A/B/C/D Out jacks- Separate outs for each waves. Each wave's volume will be affected by the current vector position. These are useful for creating stereo mixes (by routing waves to a stereo mixer and using pan controls) or for processing of individual waves.

Mix Out jack- Mono mix of the current vector mix of all four sources.

4 AirWave Vector Oscillator

AirWave is a unique digital oscillator that dynamically mixes up to four sampled waveforms or external inputs. The volume of all four sources is controlled in four quadrants by a virtual joystick, and joystick movements can be programmed with a flexible and easy-to-use vector envelope. It's inspired by the vector synthesis concepts used in the classic Prophet VS, Korg Wavestation, and Yamaha SY22/TG33 synthesizers.

Vector synthesis makes it easy to create elaborate, constantly-shifting soundscapes. In addition to its very complete vector envelope implementation, AirWave includes a sample rate control for intense, aliased digital mangling, and plenty of CV control for real-time modulation.

4.1 Inputs, Outputs, and Controls

We'll jump around the panel a bit, going from the basic stuff to the more intense vector madness parameters.

Pitch CV jack- Accepts a CV input for pitch. Typically this would come from the *Pitch* jack in the IO Panel *CV Sources* section, or from a sequencer pitch CV out. *Pitch CV* globally affects all four digital waveforms. It has no effect on signals patched to the external input jacks.

Gate CV jack- Accepts a standard +5V gate voltage input, typically from the IO Panel *CV Sources* section, or from a sequencer's gate output. The gate input is used to initiate AirWave's vector envelope; its functionality varies depending on the vector envelope's button settings (more on this later).

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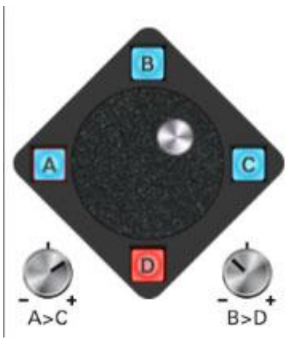

Vector Joystick- The silver-knob-floating-over-a-diamond is a joystick controller that allows real-time control of the volume balance of up to four waves (or external audio inputs). When the vector envelope is enabled, it also moves to show the current vector envelope position.

Like all knobs and controls in Voltage Modular, onscreen joystick movements can be recorded in a DAW as MIDI CC data. This is especially cool if you have a hardware instrument with a joystick controller (*see A>C and B>D joystick position knobs section below for setup info*), but the same can be accomplished by simply moving the onscreen vector joystick control.

Wave Select A, B, C, D buttons- Clicking the buttons at each corner of the joystick diamond opens a pop-up menu with 203 sampled waveforms. Click on a wave to select it.

If no wave is currently selected, the button will appear gray; the button turns blue when a wave is selected. If *External Input* is chosen, audio is sourced from the corresponding A, B, C, or D *Ext In* jack and the button turns red.

The built-in wave samples have categories and file names that indicate their general tone color and origin, such as "brass" or "perc," but bear in mind that these are relatively short samples (because you might have 37 modules open and be running Voltage on your grandpa's Windows 386 with 512K of RAM) that aren't intended to replace a 547 gigabyte symphony orchestra string library. In case it wasn't obvious, the included samples are intended to provide a whole bunch of weird tonalities to aid the in creation of odd and unique, evolving soundscapes.

AirWave can't load user samples, but if you'd like to use your own samples, we suggest using one or more Cherry Audio *Sampler I* or *Sampler II* modules, routing their audio into the *Ext In* jacks, and selecting *External Input* with the corresponding Wave Select buttons.

And if you really gotta have those fancy-pants realistic samples, you can always use Cherry Audio's *Plug-In Host* or *Mini Plug-In* host modules to load **any** virtual instrument within Voltage Modular, then run its output into AirWave's *Ext In* jacks.

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4.1.3 How It Works

AirWave's vector envelope has four steps, with each step's settings displayed in a horizontal row. Each step contains two settings defining the joystick position for that step. When the vector envelope starts playing, it moves through the steps. Instead of abruptly jumping from step to step, AirWave smoothly transitions from one position to the next, at a rate defined by the step's *Time* setting (as well as the overall *Speed* knob setting).

4.1.4 Vector Envelope Window Parameters

The step parameters are displayed across the top of the window and are the same for each of the four steps. All settings in the Vector Envelope window can be changed by either clicking and dragging a number, or by double-clicking and typing an exact value. Let's go over its parameters:

- **A>C**- X-axis (horizontal) position of the joystick for that step. Setting this to 0 corresponds to center position, i.e., a 50/50 mix of A and C waves. Negative settings move the joystick left toward wave A, positive settings move the joystick right toward wave C.
- **B>D**- Just like the A>C setting above, but defining the Y-axis (vertical) position of the joystick for that step. Setting this to 0 corresponds to center position, i.e., a 50/50 mix of B and D waves. Negative settings move the joystick up toward wave B, positive settings move the joystick down toward wave D.
- **Time**- Defines the length of the step, i.e. how long it takes for the joystick to arrive at the next step position. Lower value settings are faster, higher value settings are slower.
- **Snap**- Clicking the *Snap* button immediately sets that step's A>C and B>D position parameters to the current joystick position. This makes setting the A>C and B>D positions very easy. **The *Snap* buttons only work when the *Env Active* button is disabled.**

4.1.5 Vector Envelope Controls

Env Active- Enables and disables the vector envelope. Disabling the vector envelope is useful for "manually" playing the vector joystick, or if you're setting vector envelope window parameters using the *Snap* buttons discussed above.

When the *Env Active* button is toggled on, you'll notice that the vector envelope begins moving through its steps, even if no gate signal is present. It's designed this way to enable ever-shifting drones using the internal wave samples or the external audio inputs. (If the vector envelope were to only move when a gate voltage was present, the envelope would cease to move during an envelope generator's release phase.)

Retrig- When *Retrig* is enabled, the vector envelope restarts at step one every time a new gate signal is received, meaning every note will have the same vector envelope. Disabling *Retrig* causes the vector envelope to play continuously and ignore newly received gates. This is useful for continuously evolving sounds while playing a melody (particularly if the *Loop* button is engaged for non-stop vector movement).

If you're using the IO Panel *CV Sources Gate* output, vector envelope restart will be affected by the IO Panel *Single* and *Multi* trigger setting as with standard envelopes if *Retrig* is enabled. If *Single* trigger mode is selected in the IO Panel and the keyboard is played legato holding a previous key while playing new keys), the vector envelope will *not* restart when new keys are played. If *Multi* trigger mode is selected in the IO Panel, the vector envelope always restarts from step one when a new key is played, regardless of existing held notes.

Play Order- The *Play Order* button opens a pop-up menu that defines the playback order of the vector envelope steps. The default setting is 1–>4; this means the vector envelope plays step 1, step 2, step 3, then step 4. If the *Loop* button is engaged, it returns to step 1 and continues. The –> arrow indicates forward playback.

Selections with <–> bidirectional arrows indicate playback in both directions. For example 2<–>4 would play step 2, step 3, step 4, step 3, then step 2. If the *Loop* button is engaged, it would play step 2, step 3, step 4, step 3, then step 2, (loop here) then step 3, step 4, etc. Note that it won't repeat the same step twice when the loop occurs. We thought of that!

Loop- Enabling the *Loop* button causes the vector envelope to return back to the first step (depending on the *Play Order* setting) and continue playing.

Speed and Speed mod input and attenuator- The *Speed* works in conjunction with the vector envelope window *Time* settings and acts as an overall multiplier or divider, affecting the entire vector envelope playback rate. It allows you to globally speed up or slow down the entire vector envelope without tweaking the individual *Time* parameters.

The *Speed* knob is voltage controllable via the bipolar CV jack and attenuator directly beneath.

4.1.6 WAVE A/B/C/D CONTROLS

All of the following parameters, inputs, and outputs are available individually for each of the four waves.

Freq Mod input and attenuators- This single CV jack and attenuators affect the pitch each of AirWave's four waves and have a huge range of five octaves up and down. At extreme settings this translates to clicks and grumbles on the low side, and alias-y mess on the high side. Perfect for that Einstürzende Merzbow tribute band you've been dreaming up!

Kidding aside, besides basic vibrato duties, the *Freq Mod* inputs are handy for applying a step sequencer to play pitches with AirWave's waves in any combination. Expanding on this idea, disabling the *Key Trk* buttons means waves won't transpose, so the sequence can remain in the same key for one or more of the AirWave waves, while melodies are played on others.

Tune Semi knob and digital readout- Sets the basic pitch range for each wave in semitone steps. Tuning range is up or down 36 semitones. The digital number display shows the current setting.

Fine Tune knob and digital readout- Fine pitch setting for detuning. These can be used for subtle doubling effects, or to dial in a parallel harmony. Their range is just over a fifth, up or down.

Key Trk- Enables or disables pitch tracking when a CV patched to the *Pitch CV* jack. This is useful if you'd like the pitch for one or more waves to remain at a constant pitch while playing melodies on other waves. It's essentially equivalent to unplugging the *Pitch CV* input on a per-wave basis. All four *Key Trk* buttons are enabled by default.

Level knob- These allow a static boost or cut in volume for each waveform. Although all of the built-in samples are normalized for maximum volume, some sound louder or quieter than others; the level knobs make it easy to equalize each wave's initial (i.e., pre vector mix) volume. The *Level* knobs also affect signals plugged into the A/B/C/D *Ext In* jacks.

Ext In jacks- The *Ext In* jacks let you replace the built-in sample playback for any of the waves with a signal patched into them. To use an *Ext In* jack, plug a source into the desired jack, click the corresponding A,B,C, or D button in the vector joystick diamond, and select *External Input* at the top of the pop-up menu list. The Wave Select button turns red to indicate that it's in external input mode.

A/B/C/D Out jacks- Separate outs for each waves. Each wave's volume will be affected by the current vector position. These are useful for creating stereo mixes (by routing waves to a stereo mixer and using pan controls) or for processing of individual waves.

4.1.7 OTHER CONTROLS

Sample Rate and Sampler Rate Mod input and attenuator- By default, Airwave's digital wave samples play at 48 kHz sample rate, resulting in an audible bandwidth up to 24 kHz. When playing audio at reduced sample rates, standard hifi audio practice is to apply steep filtering above the highest audible frequency (i.e. half the sample rate) in order to reduce raunchy digital aliasing and artifact crud. But this audio crud actually sounds super cool, so there are no anti-aliasing filters in AirWave.The *Sample Rate* knob lets you reduce the sample playback rate all the down to a ridiculous 100 Hz, you can really go hog-wild. The evolving nature of Airwave's output plays particularly well with reduced sample rates.

And because twirling the *Sample Rate* knob sounds so neat, we've added a CV input and bipolar attenuator so an LFO, envelope, or other mod source do the twirling for you.

Mix Out jack- Mono mix of the current vector mix position of all four waves.

5 Amplifier

The Cherry Audio Amplifier module is a voltage-controlled amplifier, usable with audio or control signals. It's operation is relatively simple, but it remains one of the most important modules in the synthesis "tool box."

The idea of a voltage-controlled amplifier (VCA) is that an audio or control signal is patched to its input, then its amplitude can be externally controlled via the *CV In* jack. This is useful for turning audio or control signals on or off, applying envelope volume curves to sounds, regulating the amount of modulation signals applied to audio signals, and more. Think of it as a voltage-controlled gate, with a variable amount of gate opening.

5.1 Inputs, Outputs, and Controls

CV In jack- Control signal inputs such as gates, envelope generators, and mod sources (such as low-frequency oscillators) are patched in here. The most common control signal would be an envelope generator (for shaping the amplitude curve of notes), but any control signal can be patched here, including gates, LFO's, sequencers, noise generators, sample and holds, etc. The voltage level applied corresponds to the input signal's amplitude, with 0V = no signal passed and 5V = full amplitude passed.

CV Amount- Sets the amplitude of the control signal received at the *CV In* jack. This is a bipolar control with the middle position representing zero. Negative CV control increases as it's dialed to the left; positive CV control increases as it's dialed to the right.

Input jack- Use this jack to patch in audio or control signals to be affected by the *CV In* jack.

Gain- Adds up to 5 volts of gain. This works *in addition* to incoming *CV In* jack voltages. It's also useful for manually "opening" the amplifier.

Lin/Expo- These select the "curve" of the amplifier's response as the input CV rises from 0 to 5V. *Lin* or linear response curve is equally proportional across the voltage input range, where as an *Expo* or exponential curve is closer to how the human ear perceives volume. With that in mind, you'll likely want to use the *Lin* setting for modulation or control voltage situations, and use the *Expo* setting when an envelope generator is used to control an audio signal with the amplifier. Or just use whatever sounds best, we won't tell.

Output jacks- The *Output* jack carries the CV-modified version of the input signal. The *Inv Out* jack is an inverted version of the output signal. Be careful not to use both at the same level, because they can cancel the output entirely.

6 Analog Shift Register

The Cherry Audio Analog Shift Register is an eight-stage analog-style shift register that can be triggered via an internal or external clock source.

The concept behind an analog shift register (ASR) is similar to a sample and hold module which repetitively "samples" an input signal and outputs its voltage until triggered again. In fact the first output of the Analog Shift Register is **exactly** the same as the Sample and Hold module. What makes the ASR different is that every time a new sample is taken, the previous sample is "shifted" sequentially to the next output.

Typically the outputs are used to control the pitch of individual oscillators to create a canonic melody or pattern where the leading oscillator voice is "followed" by multiple subsequent voices.

6.1 Inputs, Outputs and Controls

Input jack + normalled white noise generator- This is the input jack for the audio or control signal to be sampled. If nothing is plugged in, it receives white noise from the internal white noise generator. Plugging a jack in overrides the normalled noise source.

Ext Trigger jack- A 5V pulse or gate received at this jack will externally trigger the module.

Internal Clock Out jack- If internal clock is selected, this allows to be used to trigger additional destinations, and allows the Analog Shift Register to function as a master clock source.

Trigger Source- The buttons *Int* and *Ext* select between the internal and external trigger source.

Rate- Controls the rate of the internal trigger source from 0.02 Hz - 50 Hz.

1-8 output jacks- These are the jacks where the sampled voltages will be output. Each sampled CV will initially be available at the first output and shifted sequentially to the next output with each following trigger. Note that voltages from the eighth output are not shifted back to the first output.

7 Arpeggiator

The Cherry Audio Arpeggiator is a classic "vintage-style" arpeggiator. In case you're wondering, an arpeggiator is basically a step sequencer that takes a chord as its input, and plays each note of the chord individually in an ascending or descending pattern over one or more octaves.

This module uses its polyphonic MIDI input jack to receive chords from a keyboard or DAW and convert them into a monophonic series of notes which are output as CV/gate signals. The rate at which the pattern is played can be set on the module or synced to an external clock source.

7.1 Inputs, Outputs and Controls

MIDI In jack- This MIDI input jack receives polyphonic pitch and note on/off messages from a MIDI controller or host DAW. Typically this will be connected to the *MIDI From Host* output on the I/O panel.

Rate- Sets the rate of the arpeggiator when it is not synced to an external clock source.

Gate Time- Sets the length of the 5V gate signal from 1 - 500ms for each step of the arpeggio pattern. The gate signal will be output at the *Gate Out* jack.

Clk In jack- This input jack can be used to sync the arpeggiator to an external clock source such as a sequencer or your DAW host. Typically the clock output from a module such as the Sync Divider is sent to this input jack but any signal can be used. The pattern will advance any time the input signal transitions from below 2.5V to 2.5V or higher.

To sync the Arpeggiator to your DAW host, connect the *Sync Out* and *Play* jacks from the Transport section of the I/O panel to a Sync Divider module's *Sync In* and *Reset* jacks respectively. Then connect the *Clock Out* jack from the Sync Divider to the *Clk In* jack of the Arpeggiator and engage its *Ext Clk* button.

> **Pro Tip**: To create a "swing" or "shuffle" feel, set the Sync Divider to 8th notes and send its clock output into a Delay module. Set the delay to 100% wet and 0% feedback and patch its output into the same *Clk In* jack of the Arpeggiator. The timing of the delayed signal can be adjusted to create a swung 16th note between the 8th notes.

Ext Clk- Engaging this button overrides the module's internal clock and allows the signal sent to the *Clk In* jack to externally control the rate of the arpeggiator.

Reset- This jack is used to force the module's internal clock to restart immediately when a signal of 2.5V or higher is received. Note that this will restart the clock, but not the arpeggiator pattern. The pattern is only reset once all keys in the chord are released and a new MIDI note or chord is played.

Pattern- These buttons select the order in which the notes of the chord will be played. *Up* plays the notes in order from lowest to highest, *Dwn* from highest to lowest, *Up&Dwn* will play the notes from lowest to highest then back to lowest again (the highest and lowest note will be played twice in a row) and *Rnd* will randomly cycle through the notes.

Hold- While engaged the arpeggiator will continue to run without having to continuously hold down keys. This allows you to play a series of chords without the arpeggiator stopping as you release keys between chords. Be aware that it will not stop until you disengage the button again. Mapping this to a sustain pedal or button on a MIDI controller could be useful for conveniently toggling this on and off.

Oct Range- Selects how many octaves the pattern will be played at before repeating.

CV Out jack- This is the output for the arpeggiated pitch CV. Typically this will be patched to the *Keyb CV* input of an oscillator to make the oscillator's pitch step through the notes of the chord being played but can also be used to control a filter's cutoff frequency or anything else with a CV input.

Gate Out jack- Outputs 5V gate signals for each step of the arpeggio pattern. Usually this will be patched to the gate input of an envelope generator whose output is patched to the CV input of an amplifier (VCA).

Clock Out jack- Outputs the clock signal of the arpeggiator. This is particularly useful for syncing other modules to the arpeggiator when using its internal clock.

8 Attenuverter

The Cherry Audio Attenuverter features three independent modules for attenuating and/or inverting audio or control signals. Attenuators are used to reduce the level of signals while inverters “flip” the polarity of a signal making positive voltage negative and negative voltage positive. While that doesn’t sound like too much fun, it is an extremely useful and invaluable tool within any modular system.

Signals within Voltage Modular start out as full-amplitude signals that often need to be turned down. An LFO, for example, can be used to create vibrato by subtly modulating an oscillator’s frequency. If the LFO signal is not attenuated first, the result will sound more like a sci-fi laser than vibrato!

Attenuverters are also handy when multiple CVs patched to a single input. Reducing their levels individually before the CV input is an effective way to "dial in" the perfect amount of modulation from each signal. The CV or mod amount knob on the module can then be used as a master modulation amount that attenuates all of the CVs at once while keeping their relative levels intact.

8.1 Inputs, Outputs and Controls

-∞ to Unity- This knob scales the amplitude of the incoming audio or control signal from 0% to 100%.

In jack- Input jack for the signal that will be attenuated and/or inverted.

Inv (invert) - Engaging this button will invert the polarity of the signal. All positive voltages from the input signal will be negative in the output signal and all negative voltages will be positive. Be careful not to mix a signal and its inverted signal together at the same amplitude or they will completely cancel each other out!

Out jack- Output jack for the attenuated and/or inverted signal.

9 Bandpass Filter

The Cherry Audio Bandpass Filter is a simple audio filter which allows frequency content at and around a specified frequency to pass through the filter while attenuating the signal above and below it. The width of the frequency band can be adjusted from quite wide to extremely narrow making this a tremendously versatile bandpass filter.

9.1 Inputs, Outputs and Controls

Input jack- Patch audio signals here.

Freq (Frequency)- Audio content at and around this frequency will be allowed to pass through the filter.

Width- Adjusts the width of the frequency band that is allowed to pass through the filter.

Gain- This is a post-filter gain control for adjusting the output volume of the filter.

Output jack- Outputs the filtered audio signal.

10 Bend Limiter

The Bend Limiter module is designed to easily configure incoming pitchbend messages received from a MIDI keyboard or DAW to “bend” the pitch of an oscillator up and down in different amounts from zero to 60 semitones (five octaves).

To set up the typical pitchbend behavior, connect the *Bend* output from the CV Outs section on the I/O panel to the *Bend CV In* jack, and connect the *Out* jack to an oscillator’s *Keyb CV* input (usually this will be in addition to a CV input from a keyboard or sequencer).

Although intended to scale the positive and negative voltages received from a pitchbend wheel, any signal can be altered. Try running an LFO through it. The amplitude of the positive and negative portions of its waveform can be scaled allowing the depth of modulation to be adjusted in each direction independently!

10.1 Inputs, Outputs and Controls

Bend CV In jack- For typical pitchbend wheel behavior, connect this input to the *Bend* output in the CV Outs section of the I/O Panel.

Steps (Up)- Sets the number of semitones a pitchbend wheel will raise the pitch of an oscillator when patched to its *Keyb CV* input.

Steps (Down)- Sets the number of semitones a pitchbend wheel will lower the pitch of an oscillator when patched to its *Keyb CV* input.

Out jack and LEDs- For typical pitchbend wheel behavior, connect this output to the *Pitch CV* input of an oscillator. The green and red LEDs give visual feedback of when the output voltage is positive and negative respectively.

11 Binary

The Binary module is a dual module that continuously tests an incoming audio or control signal to determine if its voltage is greater than zero (> 0) or less than zero (< 0) and outputs a selected voltage whenever the condition is true. The output will always be a binary signal, meaning it is either on (outputting the selected voltage) or off. It basically creates a series of gate signals which can be used to turn things on and off, trigger envelopes, step through sequencers etc.

One of the coolest things about modular synthesis and Voltage Modular is the ability to create unique relationships between modules so that they "react" to one another. The Binary module can "automatically" trigger an event or action based on other signals already present within a patch. It could be used, for example, to send only higher notes of an arpeggiator to a reverb send or change a drum sequencer's pattern every time an LFO passes zero.

11.1 Inputs, Outputs and Controls

> 0 and < 0 In jacks- Input jack for the audio or control signal whose voltage will be tested. The top module tests whether or not the signal is above zero and the bottom module tests whether or not it is below zero.

> **Pro Tip**: A DC Source module can be used to offset the input signal's voltage making it possible to test if a signal is above or below voltages other than zero.

-5V to +5V and LEDs- This knob selects the voltage that will be output any time the tested condition is true. The green LED shows that the condition is true while the red LED indicates that it is false.

Out jack- Output jack for the binary signal created. If both modules are testing the same input signal, the outputs will alternate outputting voltage and can be combined to create a square wave made up of both selected voltages.

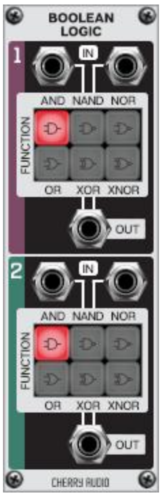
12 Blank Panel 6HP Black/White

The Blank Panel 6HP black and white modules are handy blank panels with "scribble strips" for patch notes. Simply double click anywhere within the module and begin typing. If more text is entered than there is space for, a scroll bar will appear on the right side of the module.

HP is short for "Horizontal Pitch" which is the standardized unit of measurement for the width of Eurorack modules.

Finally, Voltage Modular's Blank Panels are the most accurately modeled, warmest and best sounding blank panels ever created.

13 Boolean Logic



The Boolean Logic module is a dual module that combines two incoming gate signals using the common boolean functions AND, NAND, NOR, OR, XOR, and XNOR.

Each function creates a different gate-signal output based on the state of the two input signals. Using only one input will give the same result as combining it with a second input that is always "off." A signal is considered to be "on" when its voltage is 2.5V or higher and "off" when lower than 2.5V.

The result of each function is based on the rules that define it:

- AND only outputs a gate signal when both inputs are "on."
- OR outputs a gate signal whenever either one of the two inputs is "on."
- XOR (exclusive OR) outputs a gate when only one, but not both, of the inputs is "on."

The N (not) version of each function will create the same gate signal only inverted.

13.1 Let's look at a couple examples of how this might be used:

In the example below, the AND function is used to combine the gate out signal from a running sequencer and the gate signal from a keyboard so that the sequencer's gate pattern will only be output while a key is held down.

In this example, a gate signal from the CV Outs section of the I/O Panel is patched to one of the inputs of both modules. The top module function is set to OR and the bottom module is set to XNOR. This setup will trigger the sequencer to start when a key is pressed and stop as soon as all keys are released.

13.2 Inputs, Outputs, and Controls

In jacks- These are input jacks for the two gate signals that will be combined.

Functions- There are six possible boolean functions that can be used to combine the two inputs:

- **AND (and)**- While both input signals are "on" a gate signal will be sent to the output.
- **NAND (not and)**- While both input signals are "off" a gate signal will be sent to the output.
- **NOR (not or)**- While neither input signal is "on" a gate signal will be sent to the output.
- **OR (or)**- While either or both input signals are "on" a gate signal will be sent to the output.
- **XOR (exclusive or)**- While either, but not both, input signals are "on" a gate signal will be sent to the output.
- **NXOR (not exclusive or)**- While neither or both input signals are "on" a gate signal will be sent to the output.

Out jack- Outputs a 5V gate signal whenever the selected function tests true.

14 Chorus

The Cherry Audio Chorus module is a great sounding and flexible stereo chorus effect featuring CV control of delay time, feedback level, and wet/dry mix. Chorus is created by mixing an audio signal with one or more slightly delayed and pitch-modulated “copies” of itself and is often used to make a sound seem bigger, richer, and wider.

14.1 Inputs, Outputs and Controls

L(M) and R Input jacks- These are the mono or stereo audio input jacks. When using a mono input signal, patching it to the L(Mono) jack will feed the signal to both sides of the stereo effect.

Speed- Sets the speed of the pitch modulation from 0.01Hz to 8.0Hz.

Depth- Adjusts the depth of the pitch modulation.

Delay Time- Sets the amount of time, from 4ms to 50ms, that the “copies” are delayed.

Delay Time CV Amt jack and attenuator- CV input for externally controlling the delay time.

Feedback- Increases the number of “copies,” or layers, of the input signal by “feeding” the effected signal “back” to the effect input. This is the same principle as a delay pedal with repeating echoes only with shorter delay times.

Feedback CV Amt jack and attenuator- CV input for externally controlling the feedback amount.

Mix (Dry/Wet)- This knob adjusts the mix between the input signal (Dry) and the effected signal (Wet) that will be sent to the outputs.

Mix CV Amt jack and attenuator- CV input for externally controlling the dry/wet mix.

L and R Output jacks- These are the module's stereo output jacks.

14.2 Not Just Your Regular Chorus

It's good to know that the Chorus module can go WAY beyond typical pedestrian chorus duties, particularly when the *Feedback* is cranked past 90%. Excessive feedback settings and super slow mod speeds result in all manner of comb-filtered "robot" sounds that are particularly effective on drums.

15 Clock Divider

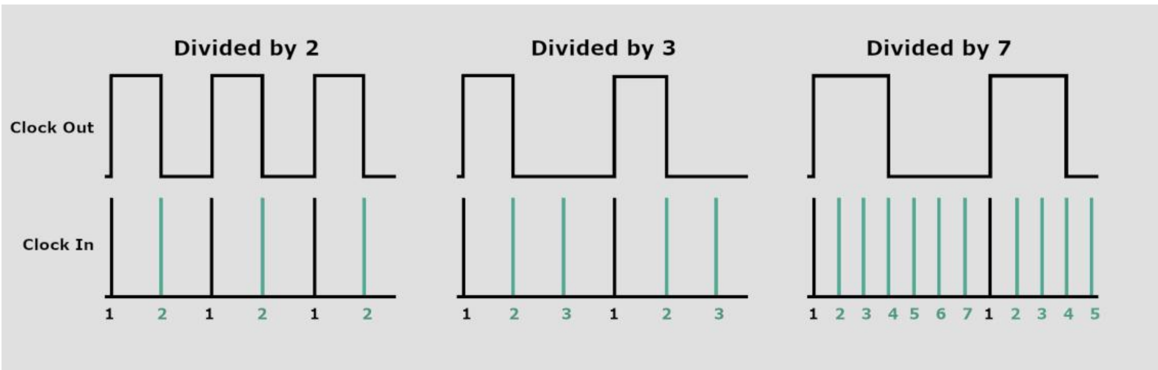
The Clock Divider is a dual module that slows incoming clock pulses by a factor of two to sixteen. The modules can run independently from one another or be linked so that the clock input and reset jacks from the first module are shared by both.

So what does it mean to divide a clock signal?

When a clock signal is divided, the timing of its pulses are not changed. Instead, the Clock Divider creates a new pulse-wave signal that represents only a fraction of the pulses received at the input.

Let's look at a sixteenth-note clock signal as an example. Dividing by a factor of two means that one pulse is output for every two pulses received resulting in an eighth note clock signal.

Dividing by a factor of three means that one pulse is output for every three received which is the equivalent of dotted-eighth notes. Dividing by a factor of seven doesn't result in a common note duration, but the same principle is used. The divider only outputs one pulse for every seven pulses which by the way can create some really cool poly-rhythms!



It is worth noting that the input signal doesn't need to be a designated clock signal. Any voltage transition from below 2.5V to 2.5V or higher will be interpreted as a pulse meaning you can use an LFO, the gate signal from a MIDI controller, or even an audio signal!

15.1 Inputs, Outputs and Controls

In jack- Input jack for the clock signal that will be divided. When the modules are linked the input signal from module one is internally routed to module two.

Reset jack- A 5V pulse or gate signal received at this jack will force the divided clock signal to restart on the next pulse received at the input jack. When the modules are linked the reset jack from module one will reset module two as well.

Divide By- This is the factor by which the clock signal will be divided. It can be set from one to sixteen by clicking on the up and down arrows to the left of the number display. A *Divide By* factor of one means the clock input will be unchanged (because duh, anything divided by one is itself!) but is a convenient way to temporarily bypass any clock division. A factor of two means one pulse is output for every two pulses received at the input. A factor of nine means one pulse is output for every nine pulses. You get the idea.

Out jack- Outputs the divided clock signal.

Input Link- When engaged, the input and reset jacks from module one are sent internally to module two. If a cable is patched to either of module two's inputs, the signal from the cable will "override" the link.

16 Compress

The Cherry Audio Compress module is a simple compressor effect with gain and reduction metering for controlling and/or shaping the dynamics of an audio signal. Compressors come in many styles and are used for various applications. This module has static attack and release times that are good for general dynamic control and/or creating punchier percussive sounds.

For anyone unfamiliar, compressors lower the output level of a signal once its input level passes a threshold. This helps reduce the dynamic range of an audio signal. This module uses a simple approach to compression with only two controls. First set the amount of compression using the *Peak Reduction* control, then use the *Gain* knob to compensate for any decrease in perceivable volume.

16.1 Inputs, Outputs, and Controls

Input jack- Patch audio signals here.

Peak Reduction- As this knob is turned up from 0% to 100%, the amount of compression, or gain reduction, is increased. This is the equivalent of reducing the threshold on many compressors. The inverted VU meter shows the amount of gain reduction.

Gain- This is a post-compression output-level control (often labeled "Make-Up Gain") that can be turned up to compensate for the decrease in volume caused by the compressor. The VU meter shows the level of the output signal.

17 Console Mixer

The Cherry Audio Console Mixer module is a powerhouse 12-input audio mixer featuring two effects sends with stereo returns and a three-band EQ (with semi-parametric mids), solo, and mute on each channel. This is the perfect “mission command” module for any live performance setting as well as a great utility for getting all of your sounds to gel just right.

17.1 Input Channel Section

Each of the twelve channels has identical controls, so let’s just look at channel one from top to bottom.

Input jack- This is the input jack for the audio signal which will be controlled by this channel of the mixer.

Send 1 and 2- These are post-eq and post-fader effect sends. Turning a knob up will send the audio signal to its respective *Aux Send* jack which can be patched to the input of an effects device such as a reverb or delay.

High Shelf Cut/Boost- This is a typical high-shelf EQ control and is used to darken or brighten the tone of the signal.

Parametric Mid Frequency- Selects the target frequency for the *Parametric Mid Cut/Boost*.

Parametric Mid Cut/Boost- This is a "bell-shaped" mids EQ control that decreases or increases the amplitude of mid-range frequencies targeted by the *Parametric Mid Frequency* knob.

Low Shelf Cut/Boost- This a typical low-shelf EQ control for decreasing or increasing the amount of low end in the audio signal.

Pan- Typical pan control for routing the channel's audio signal between the left and right master outputs of the mixer.

Level Fader- Controls the amplitude of the channel's audio signal from -infinity to +6.0 dB.

Solo- Typical solo button for isolating the channel's audio signal. This is a non-exclusive solo button meaning that more than one channel can be soloed at the same time.

Mute- Typical mute button for silencing the channel's audio signal.

17.2 Aux Sends and Effect Returns Section

Aux 1 and Aux 2 Send jacks- These outputs can be used to send audio from each channel of the mixer (via the *Send 1* and *Send 2* knobs) to two separate effects units.

Return L and R jacks- Patch the outputs of your effects units to these input jacks to "return" the effects to the mixer. The returned effects will be added to the mixer's master output. When using a mono effect, the *L Return* jack can be used as a mono return.

Pan- Pan control for adjusting the stereo placement of the returned signal.

Level- Adjusts the amplitude of the return signal.

17.3 Master Output Section

Master Pan- Pan control for adjusting the balance between the left and right master outputs.

Master Fader- Master level fader for adjusting the amplitude of the master outputs.

Mono/Stereo- These buttons select between monitoring the mixer's output in mono or stereo. When *Mono* is selected, the sum of the left and right channels are sent to both outputs. This can be a good way to check the phase correlation between the left and right channels. When two similar audio signals are out of phase, they can partially or completely cancel each other out. If a signal suddenly looses body or disappears when switched to mono, you may want to try inverting the polarity of one of the channels.

L Channel Inv- Inverts the polarity of the left output signal.

R Channel Inv- Inverts the polarity of the right output signal.

L and R Output jacks- Main stereo output of the mixer. Typically patched to the *Main Outs L* and *R* jacks in the I/O panel.

18 Crossfade

The Cherry Audio Crossfade module is a CV controllable two channel mixer, usable with both audio and control signals. This module smoothly mixes between two input signals with a single knob and/or the *CV Mod* input.

Remember that **any** two signals can be mixed. You could, for instance, mix two LFOs together to create a more complex shape, combine two different CV sequences, or even modulate between a sync signal and an oscillator if you were so inclined! Or maybe you'd rather use it like a DJ crossfader and mix between two drum beats.

18.1 Inputs, Outputs, and Controls

Input jacks 1 and 2- These are the input jacks for the two signals that will be mixed together.

CV Mod input jack- CV input for externally modulating the balance between the two input signals.

CV Mod Amount- Scales the amplitude of the CV signal received at the *CV Mod* input jack.

Initial Balance- Sets the initial balance of the two input signals before any CV modulation. It can also be used to manually mix between the two signals. Assign this knob to a midi controller's slider to create a DJ-style crossfader.

19 CV To MIDI

The CV To MIDI module converts incoming CVs for pitch, gate, velocity, pitch bend, mod wheel and aftertouch into MIDI data that can be used to control external hardware such as synthesizers or drum machines. Incoming pitch CVs can be transposed as much as three octaves up or down in semitone increments and the MIDI channel on which the data will be transmitted is assignable.

To send pitch and gate CVs from an Eight-Step Sequencer module to an external MIDI synthesizer, connect the sequencer's *Output* and *Gate Out* jacks to the CV To MIDI module's *Pitch* and *Gate* input jacks respectively. Then connect the *MIDI Out* jack of the CV To MIDI module to the *MIDI In* jack of a MIDI Out module and click its *Select MIDI Device* button to choose which external output to use.

Make sure your synth is set to receive MIDI on the same channel as the CV To MIDI module and you should be all set! (Note that if the MIDI Out module is set to a MIDI channel other than "All", the data will be output on that channel overriding the CV To MIDI module's channel setting.)

19.1 Inputs, Outputs and Controls

Pitch- Input jack for receiving pitch CVs that will be converted to MIDI note number messages.

Gate- Input jack for receiving gate CVs that will be converted to MIDI note on/off messages.

Vel- Input jack for receiving CVs that will be converted to MIDI velocity messages. The CV that is present at this input jack when a gate signal is received will be assigned as the velocity value of the MIDI note.

Bend- Input jack for receiving CVs that will be converted to MIDI pitch bend messages. Voltage from -5V to 5V will be mapped across to the pitch bend range.

Mod Whl- Input jack for receiving CVs that will be converted to MIDI mod wheel messages. Voltage from 0V to 5V will be converted to MIDI CC #1 values 0 - 127.

After Touch- Input jack for receiving CVs that will be converted to MIDI aftertouch (channel pressure) messages. Voltage from 0V to 5V will be converted to MIDI channel pressure values 0 - 127.

Transpose- Shifts the pitch of the MIDI notes up or down in semitone increments. The MIDI can be transposed as much as three octaves in either direction.

MIDI Channel- Selects the MIDI channel on which the converted MIDI messages will be output.

MIDI Out- Connect this MIDI jack to the input of a MIDI Out module to send MIDI to external software or hardware.

20 CV To MIDI CC Converter

The CV To MIDI CC Converter module converts up to four incoming CV signals to assignable MIDI CC (continuous controller) messages that can be used to control external hardware such as a synthesizer or drum machine. The four MIDI CCs are output on the selected MIDI channel and can be sent to external hardware using a MIDI Out module.

20.1 Inputs, Outputs and Controls

CV In jacks- Input jacks for the CV signals that will be converted to MIDI CC messages. Voltage from 0V to 5V will be converted to CC values 0 - 127.

MIDI CC#- selects which MIDI CC (continuous controller) number the CV signals will be converted to. MIDI CCs are used to control parameters on hardware equipment. A synthesizer, for example, may have a filter cutoff knob which can be externally controlled via MIDI CC# 102. Any CV signal from within Voltage Modular, be it an LFO, envelope, or sequence, can be converted to MIDI CC# 102 and used to control the synth’s filter cutoff frequency. Refer to your hardware’s user manual or MIDI implementation chart to determine which CC#s control its parameters.

MIDI Channel- Selects which MIDI channel (1- 16) the CC messages will be output on. Make sure your external hardware is set up to receive MIDI on the same channel as you designate here.

MIDI Out- Outputs all four CC messages on a single MIDI channel for controlling external hardware. Connect this to the input of a MIDI Out module and click its *Select MIDI Device* button to choose which external MIDI output to use.

21 DC Source

The Cherry Audio DC Source module is a two-channel DC voltage source. It outputs a constant voltage between -5V and 5V specified by the *DC Amount* knob.

This simple module can be especially useful for offsetting control voltages. The DC source can be mixed with any other signal to add or subtract voltage depending on its polarity. When mixed with a +/-5V LFO, for example, a 3V DC signal will shift the center of modulation from 0V to 3V resulting in an LFO ranging from -2V to 8V.

21.1 Outputs and Controls

DC Amount- Sets the DC voltage that will be output.

Output jack- Outputs the DC voltage.

22 DCO-60

The DCO-60 module is a fully self-contained polyphonic synth that models every aspect of the beloved classic Juno-106 synthesizer, from its self-oscillating filter to its dreamy chorus effect. In addition, it's loaded with patch points and mod inputs that immensely expand its flexibility and allow individual sections to be freely combined with other Voltage modules.

22.1 Patch Cord Normalling

22.1.1 IO Panel Normalling

DCO-106 takes advantage of Voltage Modular 2.0's patch cord normalling feature. This allows modules to be coded to communicate directly with IO Panel connections without the use of patch cords via "invisible" patch cords. The idea is to speed up setup by automatically configuring often-used cables and routings.

In the case of DCO-106, this means the IO Panel *Poly Pitch* and *Poly Gate* outputs are normalled to DCO-60's DCO *Poly Pitch* and ENV *Poly Gate* jacks, respectively. In practice, this means that the only cables that need patching to play a new instance of DCO-60 are one or both of the Master output jacks (which get patched to one or more of the IO Panel *Out To Host* jacks).

If a patch cord from another module is plugged into the DCO *Poly Pitch* or ENV *Poly Gate* jacks, the normalled, "invisible" cable is overridden, thus eliminating the IO panel connection. If you'd like to maintain the connection from the IO Panel, simply patch a cable from the IO Panel output as usual, in addition to the connection from a module as shown below.

The LFO section of DCO-106 can optionally take advantage of a normalled IO Panel connection as well (from the *Mod Wheel* output); we'll discuss this in the LFO section.

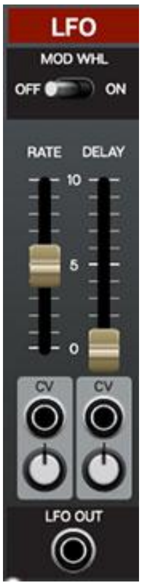
22.1.2 Internal Normalling

DCO-106 also makes use of a number of internally normalled connections. Looking at the DCO-106 panel, you'll notice that its sections have the appearance of multiple modules, with inputs at the top and outputs at the bottom. If these sections were standard Voltage modules, you'd need to patch each of them together appropriately to hear sound. (i.e. oscillator output to filter input, filter input to amplifier, etc.) In the DCO-106 module, all of these sections use normalled connections "under the hood," making it function like a conventional hard-wired standalone synthesizer. But as described above, normalled connections can be overridden by simply plugging cables into the jacks.

For example, to add a second poly oscillator, you would patch a cable from the IO Panel Poly Pitch out to a Poly Oscillator, then patch its output to DCO-106's HPF or VCF input. Because plugging into a normalled jack overrides the existing normalled connection (in this case the DCO output), a cable needs to be patched from one of the DCO outputs to the filter input, as shown in the image above. For finer control of oscillator balance, the external oscillator and DCO-106 oscillator could be routed through a poly mixer module prior to the filter input.

One important thing to keep in mind is that normalled input jacks are overridden when a cable is plugged into them, but normalled output jacks are not. Because of this, in the example above, it isn't necessary to patch a cable from the IO Panel Poly Pitch out to DCO-106's DCO Poly Pitch In jack to maintain the IO Panel to DCO connection. In other words, normalled outputs behave as if they're multed to more than one destination.

22.2 LFO Section



The low-frequency oscillator (LFO) is creates a sub-audio range triangle wave intended for modulation purposes.

Mod Wheel Switch- This is a handy, but potentially confusing feature, so don't skim this section. If the switch is in the Off position, the LFO is always "on." Its effect is immediately audible by moving the LFO sliders in the DCO and VCF sections. Moving the switch to the On position activates a normalled connection from the IO Panel CV Sources Mod Wheel jack, and enables control of LFO depth with an external keyboard controller mod wheel. If the mod wheel is all the way down, LFO depth is zero. This makes setting up a mod wheel to add vibrato or wah effects very easy.

Rate- Sets the LFO rate from 0.1 to 30 Hz. The rate can be CV controlled with the CV jack and attenuator beneath the slider.

Delay- Moving this slider up gradually delays the onset of LFO depth. The delay time can be set from 0 to 5 seconds. Because DCO-60 includes separate LFO's for each polyphonic voice, onset delay is independent for each note, which is a nice effect when playing melodies or arpeggiated note passages. The delay time can be CV controlled with the CV jack and attenuator beneath the slider. LFO Delay time can be CV controlled with the CV jack and attenuator beneath the slider. The bipolar attenuator knob's center position is zero; turning it to the right adds positive mod, and turning it to left adds negative mod.

When the Mod Whl switch is in the on position, the Delay slider "grays out" and is disabled (because you wouldn't want the onset of LFO mod delayed when controlling mod amount with a mod wheel).

LFO Out jack- The original Juno synths included hard-wired routings along with attenuation sliders to the DCO and VCF sections. This is replicated in DCO-60, but the LFO section may be used as a mod source for any of DCO-60's mod inputs by patching the LFO Out jack to any mod input (even external modules).

22.3 DCO Section

DCO-60 accurately models the unique master clock and divider architecture used in the original Juno synth oscillators.

Poly Pitch In CV jack- Accepts a CV input for pitch. This is normalled from the IO Panel Poly Sources Poly Pitch jack, so in most situations, no connection will be necessary. This input is useful if you'd like to patch in a sequencer or other pitch control source. And like all jacks in Voltage 2.0, it's actually a mult input with unlimited connections, so multiple inputs can be patched in.

Range- Sets the basic pitch range of the oscillator, displayed in traditional organ footage.

Wave Select buttons- These allow selection of sawtooth, pulse, or both waves simultaneously.

Pulse Width and LFO/Manual Switch-

- **Man position**- The Pulse Width slider sets the width or "duty-cycle" of the pulse wave. Setting the slider to zero outputs a perfect square wave, i.e. 50% duty-cycle. Moving the slider upward narrows its pulse width as well as the thickness of sound until it almost disappears.
- **LFO position**- Pulse width is modulated by the onboard LFO (often abbreviated to PWM). The modulation depth is set by the Pulse Width slider, and the rate is defined by the LFO section Rate slider.

In either switch position, the Pulse Width slider can be CV controlled with the CV jack and attenuator beneath the slider.

LFO slider- Adds frequency modulation from the hard-wired low-frequency oscillator section immediately to the left of the DCO. The LFO slider amount can be CV controlled with the CV jack and attenuator beneath the slider.

Sub slider- Adds a square wave one octave beneath the current range selection, useful for adding girth to oscillator tones. The Sub slider level can be CV controlled with the CV jack and attenuator beneath the slider.

Noise slider- Sets the level of the white noise generator. Both oscillator wave buttons can be disabled if white noise only is desired. The white noise level can be CV controlled with the CV jack and attenuator beneath the slider.

Waveform Individual and Mix Poly Output Jacks- Individual outs jacks for pulse, saw, sub oscillator, noise, and DCO mix output. Sub oscillator and noise outputs are not affected by their corresponding sliders and CV control in the individual outputs, but they will be affected in the Mix Out jack.

22.4 HPF Section

A highpass filter removes low frequencies as its cutoff frequency setting is increased, resulting in a thinning out of sound. The highpass filter section in the original Juno synths is a simple, non voltage-controlled affair, and acts more like a thickness control than a full-fledged filter. DCO-60 adds a CV input and attenuator, enabling voltage control.

In jack- The HPF poly *In* jack is normalled from the DCO Mix Out, so in most situations, no connection will be necessary. This input is useful if you'd like to patch sounds in from additional modules. Remember that connections will override the normalled signal from the the DCO; if you'd like to augment the standard DCO with additional sound sources, simply patch a cable from the desired DCO output to the HPF *In* jack along with any other poly audio sources.

Freq- Sets the frequency where low-frequency attenuation begins. The CV jack and attenuator beneath the slider allow voltage control of its setting.

Out jack- Output of the highpass section. This is normalled to the lowpass filter (VCF) input, so in most cases, it won't be needed.

22.5 VCF Section

The VCF section is a 24 db/oct lowpass filter that removes high frequencies as its cutoff frequency setting is decreased from max, resulting in a dulling of sound. DCO-60 features a super-accurate emulation and adds a CV input and attenuator for extensive voltage control.

In jack- The poly *In* jack is normalled from the HFP Out, so in most situations, no connection will be necessary. This input is useful if you'd like to patch sounds in from additional modules.

Freq- Sets the frequency where high-frequency attenuation begins. The CV jack and attenuator beneath the slider allow voltage control of its setting.

Res- Short for “resonance,” this emphasizes sound energy at and around the current cutoff frequency by adding feedback from the filter's output back to its input. At lower settings, this can be used to create mild resonances such as those heard in acoustic instruments. DCO-60’s implementation is fully self-oscillating - at extreme settings, it can be used as a sine wave generator, but be careful because high resonance settings can result in loud, screamy, dog-terrifying (and speaker blowing) occurrences. The CV jack and attenuator beneath the slider enable voltage control of its amount.

Env Slider and Mod Invert Switch- The *Env* slider sets the amount of envelope modulation applied from the envelope generator. With the invert switch in the up position, envelope affects the cutoff frequency positively. In the down position, envelope mod is inverted for “reverse” effects. The *Env* slider amount can be CV controlled with the CV jack and attenuator beneath the slider.

LFO- Adds cutoff frequency modulation from the onboard low-frequency oscillator section. The *LFO* slider amount can be CV controlled with the CV jack and attenuator beneath the slider.

KYBD- This is short for “keyboard” and causes the cutoff frequency to increase as ascending notes are played on a keyboard. The idea behind this is, because actual note frequencies rise as higher pitches are played, the *Kybd* slider adds a rising CV to the cutoff frequency in order to maintain the brightness of notes as higher pitches are played. The *Kybd* slider amount can be CV controlled with the CV jack and attenuator beneath the slider.

22.6 VCA Section

VCA is short for “voltage-controlled amplifier,” and for all intents and purposes, you can think of it as an automated, polyphonic volume knob.

In jack- The poly *In* jack is normalled from the VCF *Out*, so in most situations, no connection will be necessary. This input is useful if you’d like to patch sounds in from additional modules.

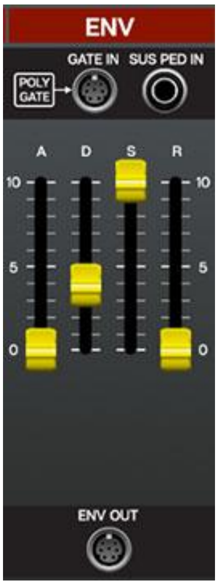
Level- In the original Juno synths, its main purpose was to act as a master volume control, in order to balance volume levels between presets. Voltage Modular allows volume adjustment in a number of different ways, so its importance isn’t as significant here, but we’ve replicated it as with the original. Typically this gets set to the center zero setting, and volume can be added or subtracted by moving it up or down. The *Level* slider amount can be CV controlled with the CV jack and attenuator beneath the slider, which can be useful for tremolo effects or wild amplitude modulation effects if the mod source is an audio-rate oscillator.

Env/Gate Switch- The original Juno synths included a single envelope generator that had to perform double-duty, applying to the lowpass filter as well as the VCA sections. In order to add some flexibility, the *Env/Gate* switch effectively disconnects the envelope generator from the VCA.

If the *Env/Gate* switch is in the *Env* (up) position, the envelope generator affects amplitude (VCA) and filter cutoff (if the the VCF Env slider is up) simultaneously. If the switch is in the down *Gate* (down) position, the envelope generator still affects filter cutoff, but note amplitude will instantly turn on and off, like an organ.

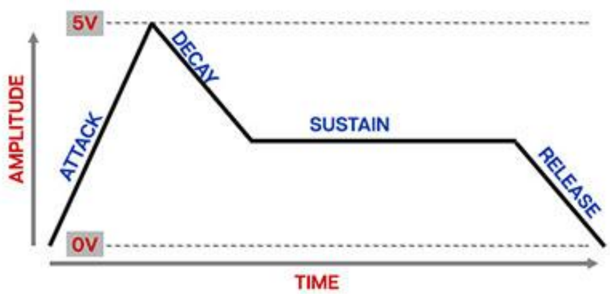
VCA Out- Output of the VCA section. This is normalled to the Chorus section input, so in most cases, it won't be needed.

22.7 ENV Section



The envelope generator section is a standard ADSR envelope generator used to shape amplitude curves and/or filter CV (or other stuff, if the Env Out jack is sued).

If you're not familiar with the operation of envelope generators, here's an overview:



When a gate voltage is sent either from the normalled IO Panel Poly Gate jack or from another source into the Gate In jack, the envelope generator outputs a dynamically changing voltage, according to the settings of its four stages. The Attack stage defines how long it takes for the output voltage to rise from 0 to 5 volts.

Once the attack stage reaches 5V, it moves to the Decay phase, which defines how long it takes to fall from 5V to the setting of the Sustain phase. Unlike the Attack, Decay, and Release phases, which define times, Sustain simply sets the held voltage level following the Attack and Decay phases - this usually equates to the envelope output level while holding down a key on a keyboard controller. Finally, the Release knob defines the the length of time it takes for the voltage to fall back to 0V when the gate input voltage is removed, i.e. when the key is released (hence the clever name).

"A" (Attack) slider- Defines the length of time for voltage to rise from 0V to 5V when the gate voltage is applied.

"D" (Decay) slider- Defines the length of time for voltage to fall from the attack stage 5V peak to Sustain stage setting.

"S" (Sustain) slider- Sets the held voltage level following attack and decay phases.

"R" (Release) slider- Defines the length of time for voltage to fall from sustain level to 0V when a key is released.

Env Out- Output of the VCA section. This is normalled to the VCA input and VCF *Env* slider, so in most cases, it won't be needed.

22.8 Chorus Section

DCO-60's Chorus section beautifully replicates the renowned, lush Juno-style stereo chorus.

Off, I, II buttons- These set the amount of chorus effect. Off is uh… off. The *I* and *II* positions are very similar, but the mod speed is slightly faster in the II position. Activating *I* and *II* at the same time results in a much faster, mono chorus sound.

22.9 Master Section

Volume- Sets DCO-106 master volume.

Tuning- Sets master tuning up or down by a half-step.

23 Delay

The Cherry Audio Delay module is a mono delay effect with up to two seconds of delay time, a high cut control, and normal and inverted outputs. While this module is primary used as an audio effect, it can also be used on controls signals.

23.1 Inputs, Outputs, and Controls

Input jack- Patch audio or CV signals here.

Time/ms- LED display for the delay time set using the *Time* knob.

Time- Sets the delay time from 0 to 2048ms.

Feedback- Controls the amount that the delayed signal is "fed back" to the input. Turning this up creates more repeats, or echoes, of the delayed signal. Be aware though, that at very high settings, the feedback loop will repeat indefinitely and continue to get louder and louder!

High Cut- This controls how much high-frequency content, or treble, is present in the delayed signal. When turned all the way to the right to *Bright*, the delayed signal will sound almost identical to the input signal, much like a digital delay pedal. As the knob is turned to the left towards *Dark*, the high frequencies of the delayed signal are reduced, or "rolled off," creating a darker tone more similar to old tape-echo machines.

Mix (Dry/Wet)- This knob adjusts the mix between the input signal (*Dry*) and the delayed signal (*Wet*) that will be sent to the outputs.

Output Inv jack- *Out +* is the output of the *Dry/Wet* mix. *Output -* is the Dry/Wet mix with its polarity inverted. This is most useful when using short delay times and can sometimes result in a more "hollow" or flanger-like sound when mixed with the original non-inverted signal. Note that this jack inverts the mix of both the dry and wet signals. If both outputs of this module are combined, they will completely cancel each other out resulting in silence! Therefore, it is usually a good idea to use the delay as a parallel effect or aux send with the *Mix* set to 100% when using the inverted output.

24 Digital Reverb

The Cherry Audio Digital Reverb module is a deep-toned reverb audio effect with variable room size, damping, and mono, mono-to-stereo or true-stereo operation. This module is capable of replicating the sound of a large range of room styles from small ambiences to long dark caverns.

24.1 Inputs, Outputs and Controls

L(M) and R Input jacks- These are the mono or stereo audio input jacks. When using a mono input signal, patching it to the L(Mono) jack will feed the signal to both sides of the stereo effect.

Room Size- Adjusts the reverb's decay length and character to simulate the size and frequency response of small and large rooms.

Damping- Adjusts the rate at which high frequencies in the reverb signal dissipate. This is used to simulate the characteristics of different rooms. A room that is full of people and/or has soft walls, for example, will soak up high frequencies quicker than an empty room with cement walls. Turning this knob to the left allows the high frequencies to last longer simulating brighter rooms, while turning the knob to the right will dampen the high frequencies more quickly to simulate a darker room.

Mix (Dry/Wet)- This knob adjusts the mix between the input signal (Dry) and the effected signal (Wet) that will be sent to the outputs.

L(M) and R Output jacks- These are the module's stereo output jacks. When using a mono input signal and wish to keep the reverb mono as well, use only the *L(Mono) Output* jack.

25 Diode

The Cherry Audio Diode module is a two-channel audio or CV polarity splitter. It takes any input signal and outputs its positive and negative voltages individually.

25.1 Inputs and Outputs

Input jack- Input the signal that you want to split here.

Positive Output jack- Outputs only the positive voltage received at the input jack. Any negative voltage is "clipped" off and outputs 0V.

Negative Output jack- Outputs only the negative voltage received at the input jack. Any positive voltage is "clipped" off and outputs 0V.

26 Distortion

The Cherry Audio Distortion module is an aggressive distortion effect unit with voltage control of distortion amount and audio level compensation. This is a great all-purpose distortion unit for adding some bite to a drum loop, attitude to a bass line, harmonics to a sub etc.

Note that although distortion is typically used as an audio effect, it can also be used to alter the shape or curve of LFOs and envelopes.

26.1 Inputs, Outputs, and Controls

Input jack- Patch audio signals here.

CV Mod jack- CV input for externally controlling the *Dist Amount.*

Dist (Distortion) CV Amount- Bipolar attenuator for CV signals received at the *CV Mod* jack. Negative CV control increases as the knob is dialed to the left; positive CV control increases as the knob is dialed to the right.

Dist (Distortion) Amount- Adjusts the amount of distortion imparted on the input signal. Low values can add subtle harmonics or saturation while high values can become quite aggressive.

Output Level- Adjusts the output level of the distorted signal. This is typically used to compensate for the raise in volume caused by adding distortion.

Output jack- Outputs the distorted audio signal.

27 Drum Highpass Lowpass

The Drum Highpass Lowpass module is a simple combination filter with steep slopes and convenient sliders for carving low and high frequencies from drum sounds or any other audio signal.

27.1 Inputs, Outputs and Controls

Input jack- Input jack for the audio signal that will be filtered.

Highpass- This slider is used to remove low-end frequencies from the input signal by "rolling off" audio content lower than the slider’s frequency. This value can be edited manually by double clicking on the slider.

Lowpass- This slider is used to remove high-end frequencies from the input signal by "rolling off" audio content higher than the slider’s frequency. This value can be edited manually by double clicking on the slider.

Output jack- Outputs the filtered input signal.

28 Drum Oscillator

The Cherry Audio Drum Oscillator module is a simple three-waveform oscillator made especially for creating vintage analog-style drum sounds. The pitch of the oscillator can be tuned from 30 Hz - 1200 Hz and can be modulated via its *CV Mod* input.

Analog drum sounds are often made by modulating the pitch of an oscillator with a short envelope to mimic the "smack" of a drum. To create a simple kick sound, first set the pitch knob to a frequency around 60 Hz. Then patch the sine-wave output to an Amplifier module's *Input* jack. Next, add a Percussion EG (envelope generator) module and patch its *Env Out* jack to the *CV In* jack of the amplifier. Now connect the *Trig* jack from the CV Outs section of the I/O panel to the *Trig In* jack of the Perc EG module. You should now hear a short tone when a key is pressed. Adjust the decay settings until you like the length of the drum. Now add a second Perc EG module that will be used to create a pitch envelope. Patch the *Env Out* jack to the *CV Mod* input of the Drum Oscillator and set the CV mod amount to about 75%. Finally, connect the *Trig In* jack to the same *Trig* output on the I/O panel.

Although the title seems pretty specific, the applications for this oscillator go far beyond creating drum sounds! Try using one of the waveform outputs as a ring modulation input or use several instances as “operators” in an FM synthesis patch.

28.1 Inputs, Outputs and Controls

CV Mod jack and attenuator- CV Input for externally controlling the pitch of the oscillator. When the CV Mod amount is at 100% the oscillator’s pitch will be mapped across a keyboard at 1V/Oct like a typical Keyb CV input.

Pitch- Sets the frequency of the oscillator from 30 Hz - 1200 Hz.

In order to play this oscillator in tune with a traditional oscillator, the pitch needs to be set to a frequency that is equal to one of the octaves of the note C.

C1 = 32.70 Hz

C2 = 65.41 Hz

C3 = 130.81 Hz

C4 = 261.63 Hz

C5 = 523.25 Hz

Waveform Outputs- Three individual outputs for sine, triangle, and square waveform oscillators. These can be used simultaneously in any combination and are all effected by the *Pitch* knob and *CV Mod* input.

29 Dual VU Meter

The Cherry Audio Dual VU Meter is an analog-style stereo VU meter for monitoring audio and CV levels. Analog VU Meters don't respond quickly enough to show every peak and transient of a signal and are therefore show the average volume, or "loudness," of a signal.

29.1 Inputs, Outputs, and Controls

Peak LED- This LED lights up to indicate when the signal is at or above 0VU.

VU Meter- The needle moves to show the current VU level of the signal. The louder the signal is, the further right the needle moves.

Text Label- Double-click to create a text label for each meter.

In jack- Patch audio or CV signals here to meter their loudness.

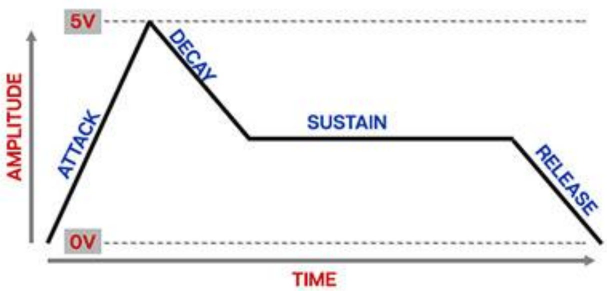
Meter Color- Pick between *Amb* (Amber) or Blk (Black) meters.

30 EG Station

EG Station features eight fully independent ADSR envelope generators. Though it's intended for use with the Cherry Audio *FM Station* oscillator module, it can be used with any modules in Voltage. It packs a lot of envelopes into a relatively compact footprint and features velocity CV inputs allowing control of envelope intensity.

30.1 How ADSR Envelope Generators Work

If you're not familiar with the operation of standard envelope generators, here's how they work: when a gate voltage is sent to one of the *Gate* jacks, the envelope generator outputs a voltage that changes dynamically according to its four stage settings.



The *Attack* stage defines how long it takes for the output voltage to rise from 0 to 5 volts. Once the attack stage reaches 5V, it moves to the *Decay* phase, which defines how long it takes to fall from 5V to the setting of the *Sustain* phase.

Unlike the *Attack*, *Decay*, and *Release* phases, each of which define a time, *Sustain* sets the held voltage level following the *Attack* and *Decay* phases - this usually equates to the envelope output level while holding down a key on a keyboard controller. Finally, the *Release* knob defines the the length of time it takes for the voltage to fall back to 0V when the gate input voltage is removed (typically when you let go of a key on a keyboard controller).

30.2 Inputs, Outputs, and Controls

30.2.1 Control Bars and CV Indicator Bars

EG Station's main controls are a bit of a departure from other Cherry Audio modules. Instead of standard knobs or sliders, its main controls are colored illuminated bars. These work just like the slider controls in other modules. To change values, grab the control bar at its top edge and move it or click anywhere in the control travel to jump to the setting (this is helpful when the control is set to minimum; you won't have to precisely click the tiny visible region of the bar). Hovering the mouse on a control bar causes it to light up. The controls bars behave the same as any standard Voltage slider control- right-clicking them allows all standard operations including Perform and MIDI controller assignments.

The control bars are color-coded to match the *FM Station* module's four operator sections, but otherwise each of the envelopes are functionally identical.

Gate jacks (normalled)- This is where you'll patch gate voltages to initiate an envelope generator cycle. The connection from the IO Panel *Gate* CV out is semi-normalled to all eight of EG Station's *Gate* inputs - in other words, even though there are no visible cables, it's automatically connected "under the hood." We think you'll find this makes using EG Station really fast and easy - it also dramatically reduces cable clutter.

If a cable or a cable bus is connected to a *Gate* input jack, the IO Panel connection is overridden for that jack (i.e., disconnected). This is useful if you don't want to use the IO Panel *Gate* jack output; when using a sequencer, for example.

- **Can I use a "trigger" to trigger an envelope generator?** It would seem logical, but the answer is, "sometimes, but generally, no." First let's clarify the difference between a gate signal and a trigger signal:
  - A **gate** is a *constant* voltage. If you're playing a keyboard, it remains high (i.e. +5V) as long as the key is held down.
  - A **trigger** is a *rapid spike* of +5V. It's useful for a number of things (like turning stuff on and off, or triggering "one-shot" drum sounds or modules).

With this in mind, EG Station needs to see a constant gate voltage to move through the *Attack* and *Decay* phases and hold during the *Sustain* phase. Removing the gate voltage following the *Sustain* phase tells it to move to the *Release* stage. Conversely, using a trigger signal will cause the envelope generator to *immediately* jump to the *Release* phase.

Velocity jacks (normalled)- This CV in jack allows the overall voltage output of the envelope to be controller via CV. Like the *Gate* jacks, all eight *Vel* CV inputs are semi-normalled to the IO Panel *Vel* CV outputs, so they don't need to be patched to function. The IO panel connections can be individually overridden by plugging poly cables into them. Velocity CV intensity is controlled by the *Vel* control bar.

Envelope Outs - The envelope voltage outputs have an envelope icon beneath them. Their output voltage ranges from 0V to +5V.

"A" (Attack) control bar- Defines the length of time for voltage to rise from 0V to 5V when the gate voltage is applied.

"D" (Decay) control bar- Defines the length of time for voltage to fall from the *Attack* stage 5V peak to *Sustain* stage setting.

"S" (Sustain) control bar- Sets the held voltage level following *Attack* and *Decay* phases.

"R" (Release) control bar- Defines the length of time for voltage to fall from *Sustain* level to 0V when gate is released.

Velocity control bar- This an attenuator controlling the intensity of CV from the *Vel* CV jack. Moving the bar up from middle position adds positive CV input.

This *Velocity* bar also displays an unfilled rectangle that displays incoming velocity CV in real-time.

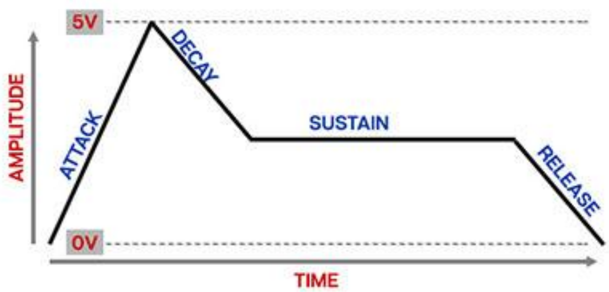
The *A*, *D*, *S*, and *R* control bars illuminate to indicate the currently active envelope stage. The *A*, *D*, *S*, *R,* and *Velocity* control bars also illuminate when the mouse is hovering over them.

31 EG-20

The EG-20 module replicates the dual envelope generators featured in the classic Korg MS-20 monosynth. In addition to their particular envelope curves and times, these include a couple of unique features.

If you're not familiar with the operation of envelope generators, here's an overview of a standard ADSR-style envelope generator:

When a gate voltage is sent to the *Gate In* jack, the envelope generator outputs a voltage that changes dynamically according to the settings of its four stages.



The *Attack* stage defines how long it takes for the output voltage to rise from 0 to 5 volts. Once the attack stage reaches 5V, it moves to the *Decay* phase, which defines how long it takes to fall from 5V to the setting of the *Sustain* phase.

Unlike the *Attack*, *Decay*, and *Release* phases, each of which define a time, *Sustain* simply sets the held voltage level following the *Attack* and *Decay* phases - this usually equates to the envelope output level while holding down a key on a keyboard controller. Finally, the *Release* knob defines the the length of time it takes for the voltage to fall back to 0V when the gate input voltage is removed (typically when you let go of a key on a keyboard controller).

Now that you're an ADSR envelope expert, let's go over the specifics of the EG-20 envelopes.

31.1 Inputs, Outputs, and Controls

EG-20 is a dual envelope generator; each envelope generator operates completely independently. Not only do they have different controls, their timing constants are different as well. **We'll refer to the left side envelope as EG 1 and the right side one as EG 2.**

EG1 / EG 2 Gate In jacks- This is where you'll patch gate voltages to initiate the envelope generator cycle. Most often this will come from the IO Panel *Gate* output. The standard gate voltage for Voltage Modular (and most hardware analog synths) is +5V, but EG-20 responds to gate voltages as low as +2.5V.

- **Can I use a "trigger" to trigger an envelope generator?** It would seem logical, but the answer is, "sometimes, but generally, no." First let's clarify the difference between a gate signal and a trigger signal:

Most standard envelope generators need to see a constant gate voltage to move through the *Attack* and *Decay* phases and hold during the *Sustain* phase. Removing the gate voltage following the *Sustain* phase tells it to move to the *Release* stage. With all that in mind, using a trigger signal will typically cause the envelope generator to *immediately* jump to the *Release* phase. However, EG-20's EG 2 can move through multiple stages using by using the *Hold Time* knob - more on this below.

EG 1 / Delay Time- Delays the onset of the EG1's attack phase by up to about 10.5 seconds when gate is high.

EG 1 / Attack Time - Defines the length of time for voltage to rise from 0V to 5V when the gate voltage is applied. EG1 max attack time is 30 seconds (!).

EG 1 / Release Time- Defines the length of time for voltage to fall from 5V to 0V when gate/key is released.

EG 1 / Sustain switch- When turned on, following the attack stage, this holds the output voltage at 5V until the key is released.

EG 2 / Hold Time- Turning this up holds the gate high when a key is momentarily struck for up to 22 seconds. This allows EG 2 to run through its stages without the need to hold a key for the duration, and also allows very brief gates to be used (i.e. trigger signals).

EG 2 / Attack Time - Defines the length of time for voltage to rise from 0V to 5V when the gate voltage is applied. EG1 max attack time is 14 seconds.

EG 2 / Decay Time- Defines the length of time for voltage to fall from 5V to sustain level

EG 2 / Sustain Level- Sets the held voltage level of held notes following the decay stage.

EG 2 / Release Time- Defines the length of time for voltage to fall from sustain level to 0V when gate is released.

Env Out (top) / Inv Out- These are the envelope voltage outputs. The *Out* voltage ranges from 0V to +5V. The *Inv Out* jack is a bit unusual - instead of outputting voltage from 0 to -5V, its output ranges from +5V to 0V. This mimics the functionality of the original MS-20 envelopes.

32 Eight Step Sequencer

The Cherry Audio 8-Step Sequencer is a fast and easy-to-use sequencer that can used not only for playing melodic sequences, but also as a modulation source for filters, amps, and more.

If you're not familiar with using step sequencers, the step sequencer concept is the forerunner of modern MIDI DAW software; the basic idea is that each step outputs a pitch and gate CV, making it act as a sort of "player piano" for melodies or CV control signals.

Eight steps (i.e. notes) may not seem like much, but step sequencers can be used for a variety of applications and are highly useful for pattern-based music and modulation.

32.1 Inputs, Outputs, and Controls

32.1.1 Transport Section

The top area of the module is analogous to standard tape deck-style transport controls.

Stop button and CV jack- Stops sequencer from running. The *Stop* button can be activated via CV using the jack below the button with any trigger or gate CV greater than +2.5 volts.

Start and CV jack- Starts sequencer running. The *Start* button can be activated via CV using the jack below the button with any trigger or gate CV greater than +2.5 volts.

Step- Advances current position to the next step. This is useful for setting pitches for each stage when the sequencer is stopped. The advance button also works when the sequencer is in play mode. Note that we didn't include a CV jack for step advance- the *Ext Clk* jack does exactly this.

Play Trig- The *Play Trigger* jack outputs a 5V trigger spike any time play mode is initiated (from the *Start* button or via CV control). This can be useful for starting ganged multiple sequencers and other functions.

Reset- This input jack is **really** important for locking sequencer timing to a DAW project or other sequencers. It force-resets the sequencer to the very beginning of step 1 the instant it receives a gate or trigger voltage.

If you're syncing to a DAW, route the IO Panel *Play* jack to the *Reset* input; if you're slaving to another sequencer, route the master sequencer's *Play Trig* out to the slave's *Reset* input.

And since we're getting hot 'n' heavy with all this clock talk, a quick primer on clock and sync signals would be er, timely:

- **Clocks: Not just a sappy song by Coldplay**- It's very important to clearly understand how clock signals work with step sequencers, so don't skim this section! One clock tick (i.e., a single voltage spike) equates to one sequencer step. If the clock signal was 120 clicks-per-minute, the sequencer would step 120 times a minute.
- **Can I just use the *Sync Out* jack on the IO Panel to sync the sequencer to a DAW? It says *Sync* right on it!**- You can, but not directly, so don't plug the IO Panel *Sync Out* jack into the sequencer *Ext Clock* (unless you like **really** fast music) . *Sync* and *Clock* signals are different - keep on reading...
- **What's the difference between *Clock* and *Sync* signals?**- Clock signals are explained above. To summarize, clock signals are the what-you-hear-is-what-you-get of timing signals: one click = one sequencer step advance. Sync signals are also a series of clicks, but they run MUCH faster and are intended to be subdivided down to musical note values via a Sync Divider module. They're typically expressed in pulses per-quarter-note, usually abbreviated to the catchy acronym, "PPQN." Voltage Modular's IO Panel *Sync Out* jack uses the common rate of 96 PPQN, with the actual speed of the pulses varying dependent upon the DAW host project tempo. The IO Panel *Sync Out* jack, along with a Sync Divider module, is the key to precisely syncing Voltage Modular sequencers to a DAW project. (If you're running Voltage Modular in standalone mode, the Sync Generator module can be used in place of the IO Panel *Sync Out* jack.)
- **I'm just reading, why are you making up all these fictional questions, weirdo?**- Hey, we're just trying to help. No need to get snippy.

Ext Clk button and CV jack- Clicking the *Ext* toggle button disconnects the 8 Step Sequencer's internal clock and accepts clock signals from *Ext Clk* jack. Note that the 8 Step Sequencer isn't too fussy about external clock sources; pretty much anything that creates rapid (or not-so-rapid) pulses can be used, including LFO's, oscillators, or even the gate or trigger output of another sequencer. Along these lines, note that external clock pulses don't have to consistently repeat either; any pattern of pulses can drive the sequencer *Ext Clk* input.

Num Of Steps and numeral display- These up/down buttons set the total number of sequencer steps from 2 - 8 steps. This defaults to 8 steps and can be altered with the sequencer in stop or play mode.

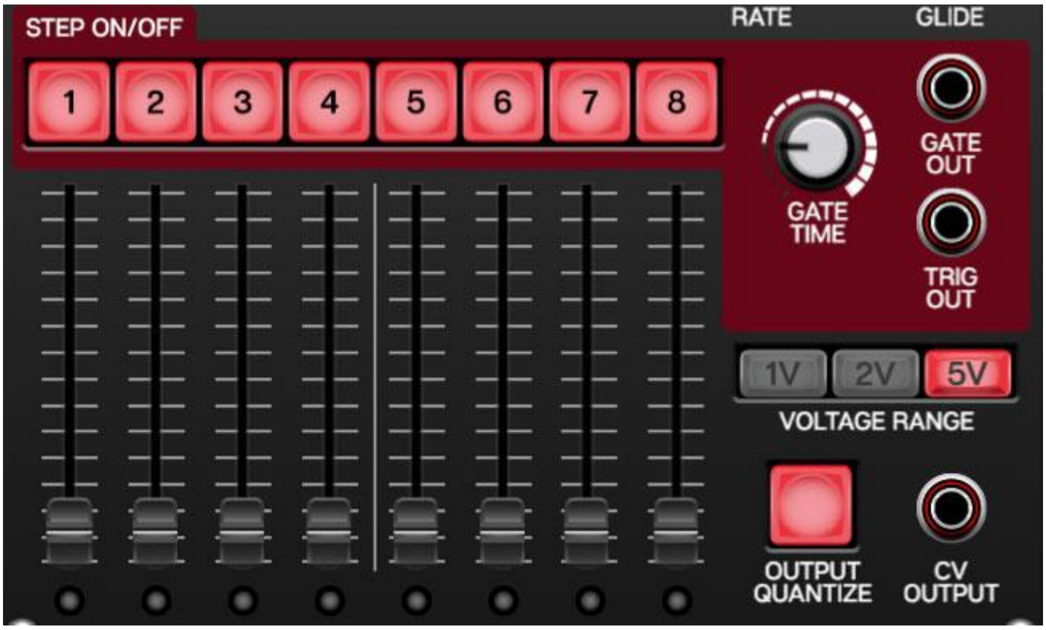
Rate and LED indicator- Sets the speed of the 8 Step Sequencer's internal clock from around 4 - 450 bpm. The LED indicator flashes with each "click" or step advance. The 8 Step Sequencer's *Rate* pop-up tooltip is calibrated to display tempos based on sixteenth-notes. For example, setting the *Rate* knob to 120 bpm plays 480 notes a minute (we did this because you'll likely want to play fast tempos such as this, and it's sort of kooky to set the knob to "480 bpm" just to get sixteenth-notes). External clock signals can used if faster or slower speeds are needed.

CV Offset- This input jacks lets you add or subtract overall voltage from the sequencer's output. Most commonly this would be used to transpose the key of a sequence during playback (from a keyboard CV, or another synced sequencer running at a slower rate), but it can also be used for more esoteric applications, such as routing an LFO to continuously vary the pitch of the entire sequence.

Glide- The *Glide* control causes notes to slide from one pitch to the next, as opposed to discretely jumping from one pitch to the next. Higher settings create a slower glides. Glide speed is **not** affected by the overall sequence rate; in other words, glide times between notes remain constant regardless of tempo.

32.1.2 Step and CV Slider Section

The bottom area sliders and buttons define how each sequencer step behaves. This is where the magic happens!



Stop On/Off buttons 1-8- Clicking these toggles the gate and trig output for each step. Keep in mind that disabling a step doesn't skip over it, it just creates a "rest" at this step, which is useful for creating rhythmic musical lines (as opposed to the ever popular never-ending-barrage-of-sixteenth-notes so popular in the modular synth community).

CV Slider- These sliders set the voltage sent to the *Output* jack for each step. The slider tooltip displays express their values either in MIDI note-style (e.g., "C3", "F#2", etc.) or in decimal value, dependent on the *Output Quantize* button setting (see *Output Quantize* button section below). Their range is defined by the *Voltage Range* switches (also explained below; reading is fun, right?).

Step Indicator LED- These guys light to display the current sequencer step. They're especially useful for setting pitches (in conjunction with the *Step* button) when the sequencer is stopped. Most importantly, they look really cool whizzing by at 100 mph when sequences are running.

Gate Time- Sets the length of the 5V gate signal from 1 - 500ms for enabled steps at the *Gate Out* jack. The *Gate Time* setting is not affected by the overall sequencer rate. It also has no effect on the *Trig Out* jack signals (because a trigger signal is always a rapid pulse).

Gate Out jack- This jack outputs 5V gate signals for active steps.

Trigger Out jack- This jack outputs 5V trigger signals for active steps.

Voltage Range- Selects the ranges of voltage for sliders.

1V = 0 to +1V (one-octave range)

2V = 0 to +2V (two-octave range)

5V = 0 to +5V (five-octave range)

Since Voltage Modular's pitch conforms to the 1V /octave standard, this means a 1V range equates to a range of one octave, a 2V range equates to two octaves, and a 5V range equates to five octaves. The 8 Step Sequencer's pop-up tooltip displays will change to reflect range button selection.

Output Quantize- Enabling *Output Quantize* forces fader values to snap to 1/2 step note increments. Without this, it would be difficult to set note values to play in tune (check out any 70s Kraftwerk record to hear the sound of wonkily tuned step sequencers). Disabling *Output Quantize* turns off pitch "snap" and allows any value to be set - this is useful when the sequencer is being used to modulate non-pitched destinations, such as filter cutoff or amplitude. The pop-up tooltip displays will show note or decimal values dependent on *Output Quantize* button position.

Output jack- Outputs the slider CV for the current step.

33 Eight To One Switch

The Cherry Audio Eight To One Switch module routes eight audio or control input signals to a single output jack. An input signal is only passed to the output when its respective “step” is active. The inputs can be stepped through sequentially with a manual or CV trigger, or targeted individually via discrete control voltages.

Switches are used to re-route signals without having to unplug or re-patch any cables. As an example, the Eight To One Switch could be used to send different modulation sources to a single destination or switch between different oscillator waveforms. The fun starts when you begin experimenting with different ways to step through the inputs!

33.1 Inputs, Outputs, and Controls

1-8 input jacks and LEDs- Input jacks for up to eight signals that will be routed to the output jack whenever their respective step is active. The small red LEDs give visual feedback of the active step.

Steps- Sets the number of steps that can be activated. When stepping through the inputs sequentially with either the manual or *Step Trigger* CV input, this sets the number of the last step before it will cycle back to step one.

Step Trigger jack- A 5V pulse or gate received at this jack will trigger the steps sequentially.

Step CV jack- CV input jack for switching between steps in any order. The control voltage range of 0V - 5V is evenly divided between the number of steps making it possible to target specific steps with discrete voltages.

Here are a couple examples of how the voltage is divided:

- If *Steps* is set to two, the 5V range is divided between the two steps. Step one is selected with voltage from 0V - 2.49V and step two is selected with 2.5V - 5V.
- If *Steps* is set to eight, the 5V range will be divided equally between the eight steps. Five divided by eight is 0.625 so, step one = 0V - 0.62V, step two = 0.63V - 1.24V, step three = 1.25V - 1.87V and so on.

If you don’t happen to make music with a calculator next to you, we recommend just playing around until you find the step you’re looking for!

Reset jack- A 5V pulse or gate received at this jack will immediately force the module back to step one. Note that resetting the module will be unnoticeable when using the *Step CV* input because the voltage received at its jack is constantly updating the active step.

Manual Step- Click this button to manually advance to the next sequential step.

Out jack- Outputs the active step's input signal.

34 Envelope Follower

The Cherry Audio Envelope Follower converts the amplitude of an incoming audio signal into a control voltage (CV) output. The module has VU meters to monitor the input and output signals as well as an adjustable input gain and envelope release time.

This is a great tool for creating dynamic CV signals that can be used to modulate just about anything in Voltage Modular. A drum loop, for example, could be used to modulate the cutoff frequency of a filter, the pitch or pulse-width of an oscillator, or the rate of a Super LFO!

34.1 Input, Output, and Controls

In jack- Input jack for the audio signal that will be converted to a CV output.

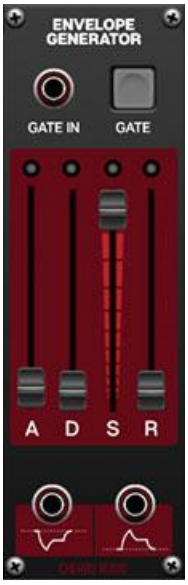
Gain- Scales the amplitude of the input signal from 0% to 200%.

Input Level- This VU meter displays the amplitude of the incoming audio signal after being scaled by the *Gain* knob.

Env Out- This VU meter displays the CV output signal.

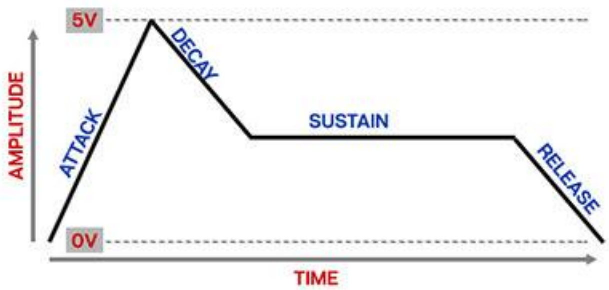
Release- Adjusts the amount of time it takes for the CV output to decrease in voltage as the input signal's amplitude decreases. When tracking percussive audio signals this will be very similar to a traditional envelope's release stage. However when tracking less dynamic input sources, the release time acts more as a smoothing value for the CV output.

35 Envelope Generator



The Cherry Audio Envelope Generator module is a standard "ADSR"-style envelope generator most often used to shape amplitude or filter curves. If you're not familiar with the operation of envelope generators, here's an overview:

When a gate voltage is sent to the *Gate In* jack (or the *Gate* button is held), the envelope generator outputs a voltage that changes dynamically according to the settings of its four stages.



The *Attack* stage defines how long it takes for the output voltage to rise from 0 to 5 volts. Once the attack stage reaches 5V, it moves to the *Decay* phase, which defines how long it takes to fall from 5V to the setting of the *Sustain* phase.

Unlike the *Attack*, *Decay*, and *Release* phases, each of which define a time, *Sustain* simply sets the held voltage level following the *Attack* and *Decay* phases - this usually equates to the envelope output level while holding down a key on a keyboard controller. Finally, the *Release* knob defines the the length of time it takes for the voltage to fall back to 0V when the gate input voltage is removed (typically when you let go of a key on a keyboard controller).

Now that you're an ADSR envelope expert, let's go over the Cherry Audio Envelope Generator module.

35.1 Inputs, Outputs, and Controls

Gate Button- Manually initiates the envelope generator cycle for as long as it's held. The same as sending a gate voltage to the *Gate In* jack.

Gate In jack- This is where you'll patch gate voltages to initiate the envelope generator cycle. Most often this will come from the IO Panel *Gate* output. The standard gate voltage for Voltage Modular (and most hardware analog synths) is +5V, but the Envelope Generator module will function with gate voltages as low as +2.5V.

- **Can I use a "trigger" to trigger an envelope generator?** It would seem logical, but the answer is, "sometimes, but generally, no." First let's clarify the difference between a gate signal and a trigger signal:
    - A **gate** is a *constant* voltage. If you're playing a keyboard, it remains high (i.e. +5V) as long as the key is held down.
    - A **trigger** is a *rapid spike* of +5V. It's useful for a number of things (like turning stuff on and off, or triggering "one-shot" drum sounds or modules).

Getting back to the Cherry Audio Envelope Generator module, like most standard envelope generators, it needs to see a constant gate voltage to move through the *Attack* and *Decay* phases and hold during the *Sustain* phase. Removing the gate voltage following the *Sustain* phase tells it to move to the *Release* stage. With all that in mind, using a trigger signal will cause the envelope generator to *immediately* jump to the *Release* phase (which might be useful in certain situations).

Some envelope generator modules can be used with a trigger signal if they have a "free-run" mode (for example, the Cherry Audio Perc EG module is *always* is in free-run mode and accepts gate or trigger signals). However, the standard Cherry Audio Envelope Generator module is designed to generally use gate signals.

"A" (Attack) slider- Defines the length of time for voltage to rise from 0V to 5V when the gate voltage is applied.

"D" (Decay) slider- Defines the length of time for voltage to fall from the *Attack* stage 5V peak to *Sustain* level setting.

"S" (Sustain) slider- Sets the held voltage level following *Attack* and *Decay* phases.

"R" (Release) slider- Defines the length of time for voltage to fall from *Sustain* level to 0V when gate is released.

LED stage indicators- In case it wasn't obvious, these guys illuminate to show the currently active envelope stage, and besides, the more blinking lights, the better!

Env Out / Env Out Inv- These are the envelope voltage outputs. The *Env Out* voltage ranges from 0V to +5V, whereas the *Env Out Inv* jack is an inverted version, with output ranging from 0V to -5V.

36 ESP-20 Processor

The Cherry Audio ESP-20 Processor recreates the slightly bizarre and endlessly useful External Signal Processor section of the classic MS-20 synthesizer. ESP-20 analyzes incoming audio and generates matching voltages for controlling oscillator pitch and envelope amplitude, letting you "play" Voltage Modular by singing, playing guitar, or any other monophonic signal source. ESP-20's frequency converter has a much wider range than the original for increased functionality, while still retaining the inherent quirkiness.

36.1 Inputs, Outputs and Controls

In jack- Plug audio signals in here.

Signal Level, Overload LED- Attenuates or amplifies incoming signal levels (up to +6 db) for optimum tracking performance; experiment with signal input level if tracking isn't working as desired. The overload *(OL)* LED illuminates if the input signal is too loud.

Amp output jack- The *Out* jack beneath the *Amp* icon outputs the the post-*Signal Level* audio.

High Cut Freq- This is a 12db per/octave, non-resonant lowpass filter and forms half of the *Band Pass Filter* block. Its intended use is to remove extraneous high frequencies in order to aid in pitch tracking.

Low Cut Freq- This is a 12db per/octave, non-resonant highpass filter and forms the other half of the *Band Pass Filter* block. Its intended use is to remove extraneous low frequencies in order to aid in pitch tracking.

Band Pass output jack- Though the aforementioned lowpass and highpass filters aren't necessarily intended for modifying audio signals, you're free to use them this way - the Band Pass output jack outputs the post filter(s) signal.

Threshold- This sets the level at which a 5V gate (constant) or trigger (instantaneous) voltage is output at the *Gate Out* and *Trig Out* jacks, respectively. ***Threshold* has no effect on the *Env Out* jack voltage**.

CV Offset- Adds or subtracts up to 5V to the *Freq>Volt CV Out* jack. Typically this would be used as a tuning or transpose control when patched to an oscillator CV input. It's generally advisable to begin with *CV Offset* at the *0V* position (remember that right-clicking a knob and selecting *Return To Default Value* is a quick and accurate shortcut).

Freq>Volt CV Out jack- Outputs the pitch CV derived from the incoming signal, as well as any offset voltage added with the *CV Offset* control. This generally will be patched to an oscillator's *Keyboard CV* or *Pitch CV* input.

Env Out- Outputs a continuous, dynamic voltage corresponding to the ESP-20 input signal level. The *Signal Level* knob will help to dial this in.

For more consistent CV dynamics, patching the input signal through a compressor module prior ESP-20's input can be helpful. (A third-party VST/AU compressor or limiter plug-in can optionally be used in conjunction with a Plug-In Host module.)

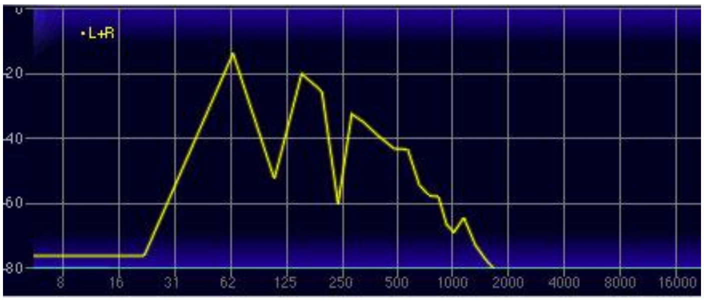
Gate Out and LED indicator- Outputs a constant 5V gate signal, dependent on the *Threshold* knob setting. The LED next to it lights to indicate when a 5V gate voltage is present.

Trig Out and LED indicator- Outputs a very brief 5V trigger signal (1ms in length), dependent on the *Threshold* knob setting. The LED next to it flashes to indicate when a 5V trigger voltage is present.

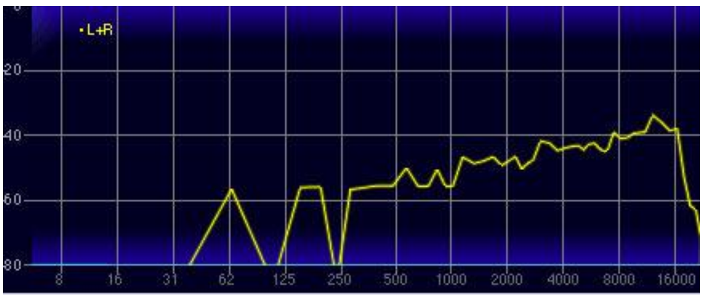
The *Trig Out* jack may output constant triggers if the *Threshold* knob setting is right on the edge of triggering. We're pretty sure this is called "hysteresis" - if this happens, a little tweak to the *Threshold* knob setting will fix it.

37 Filter

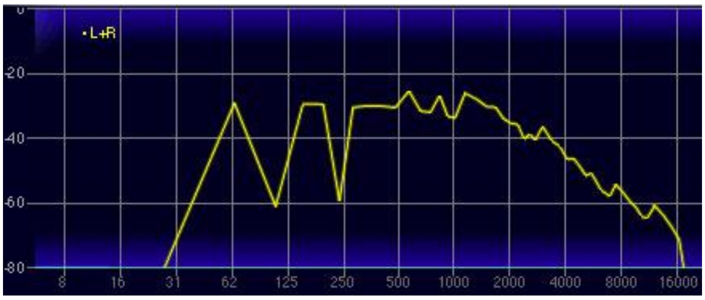
The Cherry Audio Filter module is a full-featured classic analog synthesis filter featuring lowpass, bandpass, and highpass outputs, 12- and 24-db per octave slopes, and two modulation inputs.



If you're not familiar with how filters work, a lowpass filter allows frequencies *below* the cutoff frequency setting to pass through, but blocks frequencies *above* the cutoff frequency. The frequency plot above shows the effect of a lowpass filter with its cutoff set at 412 Hz on a sawtooth wave. (The vertical axis represents amplitude and the horizontal axis represents frequency. "Axis: Bold As Love" represents a sweet Jimi Hendrix record from '67.) You can see how the high-frequency content trails off as it gets higher.



This plot shows the same oscillator signal and cutoff frequency setting using the highpass mode. This is the opposite of lowpass mode: high-frequency content remains, but low frequencies are removed as the cutoff frequency increases.



The plot above shows the same oscillator signal and cutoff frequency setting using the bandpass mode. Bandpass mode combines *both* lowpass and highpass modes, leaving sound only "in the middle." The cutoff frequency lies roughly halfway between the falloff on each side. By the way, this is pretty much how the tone controls on your stereo work!

Now that you know how filters work, let's look into Voltage Modular's "standard" filter:

37.1 Inputs, Outputs, and Controls

Audio In jack- Patch audio signals in here.

1V/Oct- This is a cutoff frequency modulation input intended to be used with keyboard CV inputs. It allows the cutoff frequency to follow or "track" notes played so that the relative brightness of notes follows note pitch.

Freq Mod CV 1 and 2 and attenuators- CV mod inputs affecting cutoff frequency. Each includes a bipolar attenuator knob. These are bipolar control with the middle position representing zero. Negative CV control increases as they are dialed to the left; positive CV control increases as they are dialed to the right.

Mod Type buttons- The *Mod Type* buttons allow linear or exponential modulation selection for each mod input. We'll give a couple of examples to clarify how they work:

- **Exponential**- For a given mod input voltage, the mod amount increases as frequency increases. For example, if the base cutoff frequency is 1000 Hz, and a bipolar wave is applied to the mod CV input, the cutoff frequency rises to 2000 Hz, and falls to 500 Hz. Because audio frequencies are inherently exponential in nature, the resulting cutoff frequency rises and falls exactly one octave.
- (This is why expo mod is generally used for oscillator mod - so that vibrato will rise and fall an equal amount above and below the pitch center.)
- **Linear**- For a given mod input voltage, the mod amount stays the same as frequency increases (hence the "linear" name). For example, if the base cutoff frequency is 1000 Hz, and a bipolar wave is applied to the mod CV input, the cutoff frequency rises to 1500 Hz, and fall to 500Hz. In other words, the frequency rises and falls by the same number of Hz in either direction.

  If the intent of the modulation is a tremolo or filter "wah wah" effect, linear mod is likely the wrong choice, because the audible effect will appear to be greater in one direction than the other - in this case, expo mod would be the best choice.

So when would we use linear mod? The advantage of linear mod is that it stays constant regardless of the base frequency, which makes it useful for audio-range modulation (i.e. mod frequencies 20 Hz and faster) when the mod source is to alter tone color. (As opposed to adding vibrato/wah-wah/tremelo, etc.) In use, you'll find that expo mod allows notes and scales to play in tune, whereas expo mod in the audio range allow neat ring mod-style sound effects, but doesn't usually allow properly pitched half-step scales.

Cutoff- Sets the frequency where attenuation begins. Attenuation will be above or below this frequency (or both) depending on which output is currently used. Also something I frequently hear at the bar, as in "you're cut off, pal!"

Resonance CV Mod and attenuator- CV mod input for filter resonance (see next section). This is a bipolar control with the middle position representing zero. Negative CV control decreases as the knob is dialed to the left; positive CV control increases as the knob is dialed to the right.

Resonance- Emphasizes sound energy at and around the current cutoff frequency by adding feedback from the filter's output back to its input. At lower settings, this can be used to create mild resonances such as those heard in acoustic instruments. At more extreme settings, resonance can create a pure sine wave at its own frequency (variable via the *Cutoff* knob). Be careful with this knob as it can get loud at extreme settings.

Slope- The nature of how a filter works is such that its affect on frequencies "falls off" above or below the cutoff frequency. Slope adjusts the "steepness" of this slope. A 12db per/octave filter has a shallower slope, giving it a clearer and brighter character, whereas a 24db per/octave filter's steeper slope gives it a tighter and darker tone (as well as more pronounced character with the resonance knob turned up).

Lowpass, Bandpass, and Highpass Output Jacks-

These are output jacks for lowpass, bandpass, and highpass modes, respectively. The icons visually represent the effect each has on incoming signals if the signal were to be viewed in a spectrum analyzer (check out the fancy diagrams in the intro). These can be used simultaneously, in any combination. Combining the outputs with a mixer can result in interesting curves.

38 FM Station

FM Station is a digital oscillator inspired by the Yamaha DX/TX-series synths. Like the classic Yamaha synths of the 80s, it uses frequency-modulation (FM) synthesis to create tones. In their day, they were giant step forward for and delivered an entirely new palette of sounds. Unfortunately, the original Yamaha synths gained a reputation for being difficult to program and as a result, most users simply used the factory presets.

Though FM synthesis is inherently less intuitive than standard subtractive analog synthesis, most of the difficulty in programming the original Yamaha synths had more to do with the minimal user interface than the actual synthesis method itself. In order to cut costs, almost all synth manufacturers of the era were switching from knob-per-function interfaces (think Minimoog/Prophet-5) to interfaces where a single slider or up/down buttons in conjunction with a small alphanumeric LCD was used for sound programming. This made creating sounds tedious at best, and given their large number of parameters, next to impossible on the Yamaha FM synths.

With this in mind, we designed FM Station to be as easy to understand and program as possible - all of its main parameters are indicated by colored bars, some of which change colors to clarify their current function. In plain English, FM Station is really fun and easy to use. We had a blast developing it and playing it - we hope you enjoy it as much as we do!

One more note: You'll likely need a fairly large number of envelope generators to optimally use FM Station. We contemplated building envelopes directly into the module, but this would've necessitated either hiding them behind tabs or pop-ups, or making a really large module, neither of which sits well with our "make all controls visible and grabbable at once" design philosophy. Instead, we created the *EG Station* module. Intended to sit directly beneath FM station, its eight ADSR envelope generators should be plenty to cover any EG mod needs when using FM Station. That said, *EG Station* can of course be used in conjunction with any other Voltage modules.

38.1 FM Synthesis Theory Nerdfest

Here we'll cover the basics of how FM synthesis works. You don't need to read this section to use FM Station, but it will greatly help your understanding, and we promise, no math equations, because as Butthead once said, "If I wanted to do math I'd go to thgool, huh-huh."

The underlying concept of FM synthesis is much like using the wavering output of a sine wave low-frequency oscillator (LFO) to modulate the pitch of an audio oscillator for vibrato. The LFO's rate is below the threshold of hearing (i.e. less than 20Hz), so you're able to clearly hear its up and down effect on the audio oscillator. In the patch below, the oscillator on the left is set to *Lo* range, and its sine wave output is routed to the second oscillator's *Freq Mod* input, creating a basic sine wave with vibrato.

The patch shown below is exactly the same, but the modulating oscillator's *Range* control has been changed to the *32*' setting, aka audio-rate modulation. In other words, its effect on the audio oscillator is no longer heard as vibrato- instead it changes the tonal color of the basic sine wave by creating additional frequencies known as "sidebands." The character of these sidebands can be altered considerably by manipulating the second oscillator's frequency modulation depth as well as changing the modulation oscillator's frequency. This modulating-one-oscillator-with-another-oscillator in various combinations is the basis of FM synthesis.

You may have also noticed the *Modulator* and *Carrier* dymo labels in the example patch. In FM synthesis, *carrier* refers to the oscillators used as audio sources; *modulator* refers to oscillators used to modulate the frequency of carrier oscillators - their audio is not directly heard.

38.1.1 Mod Type (aka, "I always wondered what that button was for.")

When setting up standard mod routings (i.e. vibrato, alternating pitches, etc.) a lot of us have likely clicked the *Mod Type* button both ways, not heard any significant effect on sound and moved along. But **in the case of FM synthesis, linear mod is of paramount importance for creating musical, tonal sounds.**

- **Exponential Mod**- For a given mod input voltage, the mod amount increases as frequency increases. For example, if the base frequency is 1000 Hz, and a bipolar wave is applied to the mod CV input, the cutoff frequency rises to 2000 Hz, and falls to 500 Hz. Because audio frequencies are inherently exponential in nature, the resulting frequency rises and falls exactly one octave. This is desirable for most standard sub-audio frequency modulation, i.e. vibrato, sirens, etc., because the audio rises and falls an equal amount above and below the median pitch.

- **Linear Mod**- For a given mod input voltage, the **mod amount stays the same as frequency increases** (hence the "linear" name). For example, if the base cutoff frequency is 1000 Hz, and a bipolar wave is applied to the mod CV input, the frequency rises to 1500 Hz, and fall to 500Hz. In other words, the frequency rises and falls by the same number of Hz in either direction.
-
- In the case of our vibrato or siren effect example, linear mod works poorly, because pitch will appear lopsided - for example, it would go down an octave but only rise a fifth.

To state it plainly, expo mod causes notes to go way out of tune as the mod amount is increase, whereas linear mod stays in tune and remains musical. (To be fair, audio-rate expo mod is great for creating laser sounds.) Try clicking the *Mod Type* switch in the demo patch above while adjusting *Freq Mod depth* settings and you'll instantly hear the difference.

By the way, besides being super basic, our two Voltage oscillators example doesn't really make for the best FM synth emulation, because the standard Voltage oscillator has some randomness built into its tuning and waves. This is desirable for accurately emulating an analog synthesizer, but not ideal for FM synthesis where it's best to have absolutely perfect waveforms and tuning. This is why you almost never see hardware analog FM synth oscillators - if everything isn't calibrated exactly perfect, it won't sound right or play in tune across a keyboard.

38.1.2 Algorithms and Operators

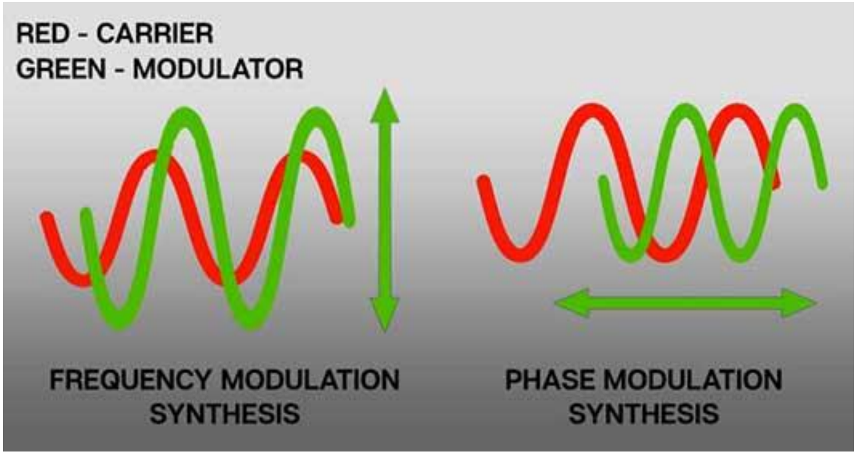
The preceding example patch illustrated the most basic FM synthesis configuration: one oscillator modulating another, aka one modulator modifying one carrier. Lots of great sounds can be realized this way, but it's roughly akin to an analog synth with a single oscillator. In order to expand the FM synthesis tonal palette, multiple modulator and carriers are configured in various pre-patched combinations to create more complex tones. These pre-patched combinations are known as **algorithms**. The carrier and modulator oscillators contained therein are generically referred to as "operators," and often abbreviated to "op."

The original Yamaha DX and TX synths were either "four-op" or "six-op." In other words, they included a total of four or six oscillators which could be configured as various combinations of carriers or modulators, depending on the currently selected algorithm. Like FM Station, the algorithms were graphically displayed on the top panel of the instrument for reference (these are actually selection buttons in FM Station, but we'll talk about that later on).

38.1.3 Frequency Modulation vs. Phase Modulation

The story goes that Yamaha licensed the use of the FM synthesis technology developed by John Chowning at Stanford University (FYI, the patent expired in the 90s). But oddly, Yamaha DX and TX synths don't use FM synthesis at all - they actually use the closely related *phase modulatio*n synthesis instead. What's the difference, why would they do this, and what does it mean to you? First, your bad-at-math help doc writer will attempt to explain the difference.

When an audio signal is frequency modulated, it can be thought of moving the wave up and down. Phase modulation can be thought of as rapidly moving the audio signal wave from side to side, thus changing the *phase* relationship between the two waves. Because both waves are constantly moving up and down, the effect is roughly the same on the resulting output wave.



There is one interesting quirk of phase modulation that's good be aware of - because the *Freq/Ratio* CV inputs affect phase, pitch changes are only audible *when the mod source is moving.* In other words, a constant DC voltage patched to a *Freq/Ratio* CV input (such as a keyboard pitch CV) won't affect pitch. Similarly, modulation waves with "straight" angles (such as sawtooth or triangle waves) will essentially sound the same as square wave mod, because their rate of change remains the same through their cycle, as opposed to a curve that's constantly changing - this is why all of FM Station's waveforms are curved.

A good analogy to understand phase mod is the doppler effect. When a car drives by honking its horn, the pitch of the horn changes as the car travels by, but if the car stops moving at any distance, the pitch of the horn remains constant. If you've ever messed with the time knob on an analog delay, they behave in the same way - pitch change is only audible while adjusting the delay time, and "catches up" as soon as you stop moving the knob. This is because a delay is performing exactly the same function - it's changing the phase (i.e. distance) of two copies of the same audio material.

Note that this only applies to the *Freq/Ratio* CV inputs; the *Fine* control bar and CV inputs are "standard" pitch CV ins (i.e. not phase mod) as is the master *Freq Mod* CV in at the far left of the panel.

Why did Yamaha choose phase modulation over frequency modulation? As it turns out, it takes less computational horsepower to perform phase modulation synthesis. This was an important consideration in light of the relatively limited computer horsepower available in the 80s. The other reason is that FM synthesis makes use of feedback loops for further tone colors, and with pure FM, these loops cause oscillators to go badly out of tune - with PM, tuning isn't an issue (we'll discuss these feedback loops later on).

We don't know why Yamaha continued to refer to DX/TX synthesis as FM as opposed to PM. Perhaps it was to capitalize on the prestige of the aforementioned John Chowing/Stanford research, or maybe they just thought FM had a better ring to it. Regardless, we'll continue to refer to it as "FM synthesis," even though it technically isn't.

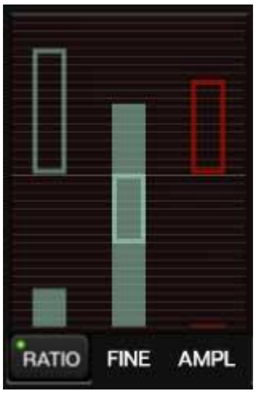
38.2 FM Station Basic Architecture

As discussed in the previous section, an "operator" is FM synth-speak for a single oscillator. FM synths generally include "pre-wired" combinations of oscillators, configured as modulators and carriers, known as algorithms. FM Station features four oscillators, making it a "four-op" FM oscillator. It includes eight selectable algorithms. Its operators each include eight selectable waveforms, and the fourth operator includes an adjustable feedback loop.

Because FM Station includes separate outputs and modulation inputs for all four operators, it can be used "a la carte," that is, if algorithm 8 is selected, all operators and mod routings can be manually patched for unlimited custom algorithm and mod routing, including multiple instances of FM Station for 8-op, 12-op, stereo or quad output routing, etc. etc.... it's Voltage Modular, what'd you expect?!?

38.3 Inputs, Outputs, and Controls

38.3.1 Control Bars and CV Indicator Bars



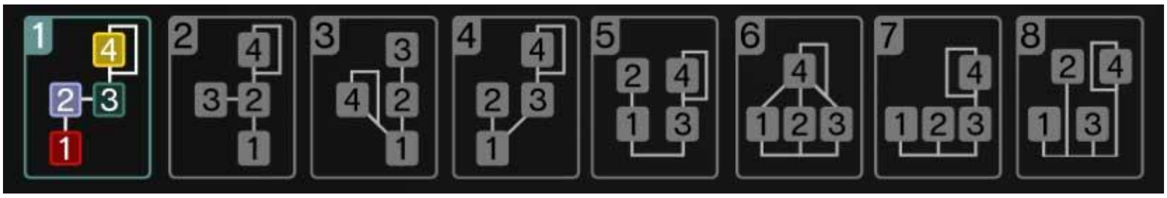
FM Station's main controls are a bit of a departure from other Cherry Audio modules. Instead of standard knobs or sliders, its main controls are colored illuminated bars. These work just like the slider controls in other modules. To change values, grab the control bar at its top edge and move it or click anywhere in the control travel to jump to the setting (this is helpful when the control is set to minimum; you won't have to precisely click the tiny visible region of the bar). Hovering the mouse on a control bar causes it to light up. The controls bars behave the same as any standard Voltage slider control- right-clicking them allows all standard operations including *Perform* and MIDI controller assignments.

You'll also notice the unfilled rectangular CV input indicators bars. These act as meters and display incoming mod CV levels in real-time. Besides being great eye candy, they're useful for quickly determining which carrier operators are making sound while under CV control.

Pitch CV jack- Accepts a CV input for pitch. Typically this would come from the *Pitch* jack in the IO Panel *CV Out* section, or from a sequencer pitch CV out.

Range- Sets the basic pitch of the oscillator, displayed in traditional organ footage. Unlike most other oscillators in Voltage Modular, there are only three range settings. This is because the *Freq/Ratio* controls offer a huge range of initial pitch settings, and the *Range* buttons are mainly intended for quick transposition.

Frequency Mod input jack and bipolar attenuator- Used for globally (i.e. all oscillators at once) modulating the base frequency. It's useful for adding vibrato with an LFO, siren noises, envelope-controlled pitch sweeps, etc. The input voltage is increased by turning from center position to the right; the input voltage is inverted and increased as the knob is turned to the left.



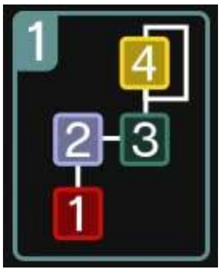
Algorithm selector buttons- The eight large diagram buttons across the top configure FM Station's operators into pre-patched routings known as "algorithms." The colored number boxes within correspond to each of the four operators; the operators in the bottom row of the diagrams represent carriers (i.e. audio sources), and operators above represent modulators (mod sources for the audio oscillators). In algorithms 5-8, all operators hooked to the horizontal line are carriers (i.e. audio sources).

Carrier oscillators are routed to FM Station's *Master Volume* and *Mix* output jack; modulator oscillators are routed only to carrier *Freq/Ratio* CV inputs. All four operators are **always** routed to the *Op1, Op2, Op3,* and *Op4* individual outputs along right side of the panel. The *Amplitude* control bars affect output levels in the *Mix* out as well as individual outs.

Algorithms are selected by clicking on the diagrams. The numbered operator boxes and outline frame will be colored for the currently active algorithm. A couple of things happen to help clarify which operators are currently configured as carriers and modulators in the currently selected algorithm. The text in the large colored buttons beneath the algorithm boxes changes to display *CAR* or *MOD* to indicate the status of each operator...

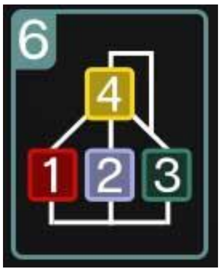
... and the *Ampl* control bars turn red for carrier operators, and green for modulator operators:

In following section, we'll go over a few of the algorithms; once you're gained an understanding of these, the others should be easy to grasp.



In algorithm 1, operator 1 is the only carrier. Operator 4 modulates operator 3, and operator 2 modulates operator 1. Although only operator 2 is directly modulating operator 1, the cumulative modulation of operator 4, then 3 can result in operator 2 having a highly complex modulation wave. That said, this algorithm is a good choice for simple patches where operator 2 modulates operator 1 (set operator 3 and 4 *Ampl* sliders to zero).

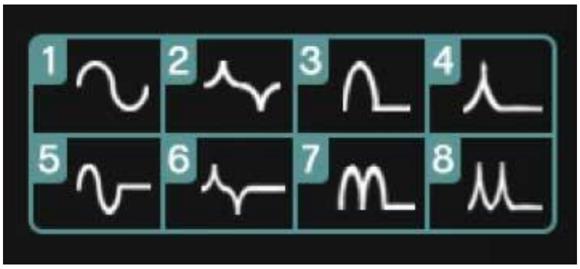
This simple modulation path is essentially the same as the two-oscillator demo patch at the beginning of this guide and is a good starting point for understanding FM synthesis.



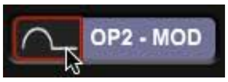
Algorithm 6 configures operators 1, 2, and 3 as audio sources, and operator 4 as a the sole mod source (with feedback loop). Algorithm 6 also resembles the gearshift pattern of my uncle's '74 Triumph Stag, which had some wicked electrical issues.

We wanted to specifically discuss algorithm 8, because it's the most interesting algorithm for the serious FM tweaker. When used with the *Mix* output, it's useful for additive synthesis/creating stacked chords, etc. But when used in conjunction with the individual outputs and the freq/ratio CV input jacks, it's the key to using FM Station "a la carte." Because it contains no pre-wired modulator routings, it allows users to roll their own custom algorithm routings.

It's also useful when routing separate outputs to stereo panned mixers and effects, as well as combining multiple instances of FM Station for doubling or tripling the number of available operators for wildly complex FM patches.



Waveforms selector buttons and panel diagram- The original DX/TX FM synths only used sine waves, but later models added extra waveforms. It should go without saying that a small number of waveforms on an FM synth isn't nearly as much of a limitation as with an analog or sample-playback synth; the original DX7 created a tremendous range of tones with sine waves only, but additional waveforms certainly expand the tonal palette, especially in a four-op configuration (as opposed to a more complex six-op configuration).



The panel display at the top right is simply a graphic displaying all eight waveforms (it's not a control of any kind). Waveforms are selected individually for each operator by clicking the black waveform selector button. These cycle forward with each click.

Freq/Ratio control bar, CV input and attenuator- This really a fancy name for pitch. These are tuned to the standard harmonic series of even overtones. They go from 0.25 of the base pitch all the way up to the 32nd overtone. It's unlikely you'll use the higher values as basic oscillator tones, but they higher values are intended for use as modulator sources, and they're the key to getting the signature FM bell-type sounds.

Unlike standard oscillators, the freq/ratio CV mod input isn't used for vibrato; remember that this is actually a phase modulation input, so it won't behave as expected if you patch a low-frequency oscillator or DC voltage source into it (such as a keyboard pitch CV). The main thing you'll use it for is "manually" routing the output of a modulator operator, but remember this is only necessary if the algorithm you've chosen doesn't already contian a pre-wired mod routing path.

Fine freq control bar, CV input and attenuator- This acts as a detune control with a range of up or down one octave, and default position at center. The CV input allows control via CV. This is a bipolar control with the middle position representing zero. Negative CV control decreases as the knob is dialed to the left; positive CV control increases as the knob is dialed to the right.

Amplitude control bar, CV input and attenuator- Sets the overall volume of the operator. If the operator is currently configured as a carrier, this affects the audio output level in the *Mix* and individual outputs. If the operator is currently a modulator, this affects the amount of modulation signal traveling to the operator it's routed to - this is the most significant sound varying parameter in FM synthesis.

The CV input allows control via CV. This is a bipolar control with the middle position representing zero. Negative CV control decreases as the knob is dialed to the left; positive CV control increases as the knob is dialed to the right.

Patching an envelope generator to a carrier operator is how you'll shape the amplitude curve of the sound, exactly as you would use an envelope generator in conjunction with a VCA in analog synthesis. Conversely, patching an envelope or other mod source to a modulator operator is the primary way to achieve dynamically shifting timbre.

Pay attention to the initial setting of the *Amplitude* control bar. If the operator is a carrier, you'll most likely want to set it to zero and patch an envelope generator to "turn the sound on and off" as you play the keyboard. Otherwise, setting the bar position above zero will result in a constant drone. You could run the *Mix* out to a VCA (*Amplifier*) module (controlled by an envelope generator), but there's really no need to, as each operator essentially contains its own built-in VCA.

Feedback control bar, CV input and attenuator (Operator 4 only)- FM synthesis makes use of feedback, present in operator 4 in FM Station. This routes the output of operator 4 back to its own *Freq/Ratio* input, with the amount set by the control bar. The resulting audio output greatly varies depending upon the currently selected waveform and amount. If the selected algorithm is using operator 4 as a modulator, this means the resulting modulation wave becomes considerably more complex than the original "unadulterated" wave.

One common FM synthesis trick is to use a sine-wave operator with feedback on its own, and turn the feedback control up about halfway, resulting in a fairly accurate sawtooth wave (hook up an *Oscilloscope* module to check this out). Sonically interesting results occur with different waveforms as well.

The CV input allows feedback control via CV. This is a bipolar control with the middle position representing zero. Negative CV control decreases as the knob is dialed to the left; positive CV control increases as the knob is dialed to the right.

39 Formula

This module uses the MXParser library written by Mariusz Gromada:

40 Glide

The Glide module is used to slide smoothly from one input control voltage to another. This is typically used to slide between pitch CVs creating portamento as one note glides to the next. The module features an adjustable speed, linear or constant curve, and a CV jack for enabling and disabling the glide in real time.

40.1 Inputs, Outputs and Controls

Input jack- Input jack for the control voltages you wish to glide between. Typically this will receive pitch CVs from a keyboard or sequencer.

Amount- Adjusts the speed of the glide from 0ms to 5000ms (5 seconds). When the curve is set to *Con*, this is the amount of time it will take to slide from one CV to the next. When the curve is set to *Lin*, the slide time will be different depending on the distance between CVs so this is actually adjusting the speed of the glide rather than the time it will take to complete each transition.

Response Linear/Constant- Selects between two options for transitioning from one CV to the next. When *Lin* (linear) is selected, the rate of the glide will remain the same regardless of how far apart the CVs are. Therefore gliding between voltages near one another will take less time than voltages that are farther apart. When *Con* (constant) is selected, the amount of time it takes to glide between voltages will be the same regardless of how far apart the voltages are.

External Engage jack- Allows the glide module to be enabled and disabled in real time using control voltages. Voltages 2.5V or higher will enable the glide while voltages less than 2.5V will disable it.

Output jack- Outputs a CV signal that slides from one voltage to the next. Typically this will be connected to the Keyb CV input of an oscillator to create portamento.

41 Hex Phaser

Besides being a stupendously good name for a comic-book villain, Hex Phaser takes the standard phaser effect and adds features out the proverbial wazoo. Though it vaguely resembles a classic 70s phaser effect, its unique feature set allows endless colors and tonal variations. These include individually switchable stages and feedback insert point, as well as extensive modulation, taking it waaay beyond the standard "two-knob special" stomp box.

41.1 A Little Background

In order to get the most from phaser, let's talk a bit about how phaser effects work. Phasers make use of a special type of filter called an "all-pass" filter. Unlike low-, high-, or band-pass filters, an all-pass filters allows all frequencies to pass through. So what good is that? Though it passes all frequencies, it alters the phase relationship of part of the audio spectrum. By itself, you wouldn't hear any audible difference, but when the signal is inverted and combined with the dry signal, phase cancellation occurs in regions of the audio spectrum. Varying the center frequency of the all-pass filter (i.e. where the phase cancellation occurs) results in the familiar swooshing sound we know as "phasing."

If a single all-pass filter is used, the effect is rather subtle. Engineers found that stacking multiple all-pass filter stages created a deeper, more pronounced swooshing effect. This is why phaser effects are sometimes referred to as four-stage, six-stage, etc. (Most of the classic stomp box phaser effects use four stages.) Another way phasing effects are intensified is by feeding back an inverted version of the phased part of the signal (i.e. not the unaffected dry signal) back into one of the stages.

In a typical hardware unit, the number of stages and feedback routing are fixed and unalterable. Hex Phaser allows all six of its stages to be enabled or disabled in any combination, and also lets you select the stage where the feedback routing occurs. This results in a lot of different sounds, especially in combination with the *Feedback* and *Q Width* controls.

41.2 Inputs, Outputs and Controls

We'll be jumping around a little bit in order to make the most sense of the control layout.

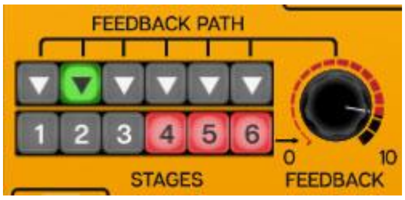
In L(M) and In R jacks- These are the mono or stereo audio inputs. Though it only has one set of controls, Hex Phaser is actually a true-stereo effect "under the hood." Unlike a typical stereo effect (e.g., reverb), it's desirable to have both channels share settings and have their modulation cycles related to each other (see the *Stereo Offset* control section for more on this).

- *Mono in/mono out* **-** Use In L (M) and Out L (M). The *Stereo Offset* control won't have any useful effect (it's setting won't matter).
- *Stereo in/mono out* **-** Use In L (M) and both outputs. The dry signal remains mono; the phased signal modulation is offset according to the *Stereo Offset* control setting. Hex Phaser is a great way to stereo-ize mono signals.
- *Mono in/mono out* - Use In L (M) and Out L (M). The *Stereo Offset* control won't have any useful effect (it's setting won't matter).

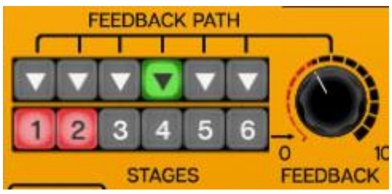
Stages- Hex Phaser features six all-pass filter stages (hence the name). Each of these can be enabled or disabled using the numbered buttons. As a general rule, more stages = lusher phase, but there are plenty of interesting tones to be had using very few stages. Note that the center frequency for each stage is slightly different - this was done because it actually sounds a little "larger" on the whole when multiple stages are used, and has the side benefit of sounding slightly different when individual stages are used.

If all stages are disabled, the unaffected dry signal is fed through - the equivalent of a bypass button.

Feedback, Feedback Path buttons and Feedback CV mod input/attenuator- The feedback knob adjusts the amount of inverted effect-only signal that gets mixed with the existing signal. This creates a more colorful and intense effect. The *Feedback Path* buttons let you specify which phaser stage receives the feedback signal, resulting in different tonalities.



Keep in mind that the signal always travels through the stages from left to right, so there are some non-tragic caveats to be aware of. In the example below, since stages 2 and 3 are disabled, *Feedback Path* "skips" stages 2 and 3 and is effectively the same as setting *Feedback Path* to stage 4.



In the screenshot to the left, the feedback signal is inserted *after* the enabled phaser stages, so it wouldn't result in any change in the phase sound (just an overall volume increase, and potentially loud, not-cool-sounding feedback mess). If the feedback path is routed after any enabled stages, the feedback path is automatically disabled. It won't void your warranty or anything like that, but it's something to be aware of.

The *Feedback CV* mod input and attenuator allow negative or positive CV control of feedback. The center setting correlates to no modulation.

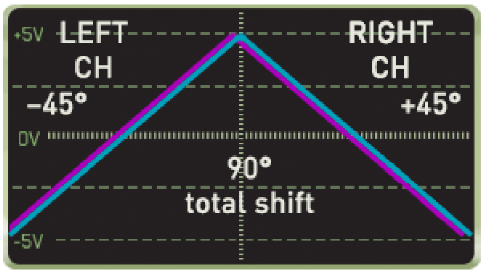
Rate and Rate CV mod input/attenuator- Sets the phaser speed modulation from 0.01 Hz to 7.5 Hz. You'll note that the knob is calibrated so that slower speeds occupy the first half of the knob travel. The *Rate CV* mod input and attenuator allow negative or positive CV control of rate. The center setting correlates to no modulation.

Depth and Depth CV mod input/attenuator- Adjusts the depth of the phase modulation, i.e. how far the all-pass filters sweep back and forth. The *Depth CV* mod input and attenuator allow negative or positive CV control of phase depth. The center setting correlates to no modulation.

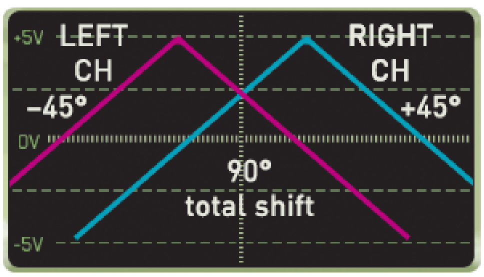
Q Width and Q CV mod input/attenuator- Sets the nominal frequency spread size or "bandwidth" of the all-pass filters. This control will really noticeably alter the overall tone and character - be sure to try it when experimenting with stage and feedback settings. The *Q CV* mod input and attenuator allow negative or positive CV control of Q width. The center setting correlates to no modulation.

Stereo Offset and Offset CV mod input/attenuator- Though it only has one set of controls, Hex Phaser is actually a true-stereo effect "under the hood," with two triangle-wave LFO's modulating two sets of all-pass filters. The LFO's always run at the same rate, and are phase-locked to each other

(When we say "phase-locked," we're not referring to the phaser effect itself, but rather the relationship of the two modulation waves. Sorry for the same-y and potentially confusing verbiage.)



If the *Stereo Offset* control is set to 0°, both modulation LFO's are exactly in phase, i.e. exactly the same, as shown the diagram below (we actually shifted the two waves a wee bit so they'd be visible, but you get the idea). This translates to the exact same phasing mod for both the left and right channels, resulting in a mono effect.



If the *Stereo Offset* knob is set to +90°, the left channel mod wave moves -45° to the left, and the right channel mod wave moves 45° to the right (for a total of 90° of phase offset as shown to the left).

Setting the *Stereo Offset* knob to negative values moves the mod waves in opposite directions. This "simultaneous movement" (as opposed to only shifting one channel) results in more interesting sound animation, especially when using CV control.

The *Offset CV* mod input jack and attenuator allow negative or positive CV control of stereo offset. The center setting correlates to no modulation.

Out L (M) and Out R jacks- These are the module's stereo output jacks. If the *Stereo Offset* control is set to 0°, the signal will be identical on both jacks. At any setting other than 0°, the outputs will be shifted apart dependent on the setting of the *Stereo Offset* control (and any mod applied to it).

Ext Mod/LFO Bypass button, knob and bypass jack- This bypasses Hex Phaser's internal LFOs and allows an external LFO, or any other mod source to modulate the allpass filter frequencies. Clicking the button disables the internal LFOs, and the attenuator allows negative or positive modulation. This is particularly useful if you'd like to use a tempo-synced LFO or a stepped sequencer.

Disabling the internal LFOs with the button also allows Hex Phaser to be used as a fixed filter, which can yield a lot of useful sounds. The *Rate*, *Depth*, and *Stereo Offse*t knobs won't do anything, but the *Stage* buttons and *Feedback* controls will still work. If you really want to get crazy with fixed settings, try connecting a DC Module to the *Ext Mod/LFO Bypass* jack for manual control of the all-pass filter frequencies with its DC voltage knob (make sure to set the CV attenuator to a non-zero value).

42 Invert

The Invert module is a dual utility module that flips the polarity of an audio or control signal. All positive voltages from the input signal will be negative in the output signal and all negative voltages will be positive.

Be careful not to mix a signal and its inverted signal together at the same amplitude or they will completely cancel each other out!

43 Ladder Filter

The Cherry Audio Ladder Filter is a CV-controllable, 24dB/oct, low-pass filter based on the famous transistor ladder filter patented by Bob Moog in 1966, and made famous in the Moog Modular and Minimoog synthesizers in the 1960s and 1970s. In addition to the classic Cutoff and Resonance (or "Emphasis") controls, we've added a Saturation knob as well for overdriving the tone even further! This filter is the definition of raw, raunchy, and powerful!

43.1 Inputs, Outputs and Controls

Input jack- Patch audio signals in here.

Freq Mod jacks and attenuators (1 and 2)- These are CV input jacks and attenuators for externally controlling the filter's *Cutoff Frequency.* Both jacks can be used simultaneously. The sum of both jack's voltage will control the cutoff.

Cutoff Frequency- Sets the cutoff frequency of the filter from 0Hz to 24,000Hz. Since this is a low-pass filter, all frequencies lower than this value will be allowed to pass through the filter while frequencies higher than the cutoff will be attenuated at a rate of 24db per/octave.

Emph (Emphasis) Mod jack and attenuator- CV input jack and attenuator for externally controlling the emphasis (resonance) of the filter.

Emphasis- Turning this knob up emphasizes sound energy at and around the cutoff frequency by adding feedback from the filter's output back to its input. On other filters this is often called Resonance. With higher settings, any modulations or knob twisting of the cutoff frequency becomes more pronounced.

Sat Mod jack and attenuator- CV input jack and attenuator for externally controlling the saturation amount.

Saturation- Adds distortion to the signal. This is where things begin to get raw and raunchy!

Output jack- Outputs the filtered audio signal.

44 LFO

The Cherry Audio "standard" LFO, or low-frequency oscillator, is a multi waveform, analog-style LFO. It generates six different control voltage (CV) waveforms and is capable of audio-rate modulation.

If you are unfamiliar with using LFO's, a simple application would be to slightly modulate an oscillator's frequency to create vibrato (pitch modulation), or to modulate a VCA's amplitude to create a tremolo effect (amplitude modulation). Modulating the cutoff frequency of a filter can create a dubstep-style wobble, or if modulated very slowly, long sweeping tonal shifts.

Many of the Voltage Module modules have dedicated CV inputs, sometimes labeled as "mod" inputs, all of which can be modulated with the LFO.

But remember, just because an input isn't labeled as CV or mod, that doesn't mean you can't route an LFO to it! Patch away and see what happens- unexpected results are what make modular synthesis so much fun.

44.1 Inputs, Outputs, and Controls

Reset jack- Any trigger or gate CV greater than +2.5 volts received at this input jack will force-reset the LFO's cycle. Connect the *Trig* or *Gate* jack in the IO Panel *CV Out* section to this input jack to reset the LFO each time a new note is played, or connect the *Trig Out* from a sequencer to keep the LFO in sync with the sequence.

Range- The *L* (low) and *H* (high) buttons alter the frequency range of the LFO. When *L* is selected, the LFO is a typical low frequency, or sub-audio rate, oscillator ranging from 0.02 Hz - 10 Hz. When *H* is selected, the LFO's frequency range is from 5 Hz - 400 Hz, making audio rate modulation possible.

Frequency- Sets the frequency, or rate, of the LFO. Frequency values are represented in Hz, or cycles per second. In other words, 1 Hz means it takes one second to complete a full cycle of the waveform. Therefore, 2 Hz = 0.5s, 4 Hz = 0.25s etc. Don't worry though, you don't need to know the math because the LED beside the knob flashes to display the LFO's current frequency.

Polarity- Sets the polarity of the LFO output waveforms. In the *Bi* position ("bipolar," meaning above and below 0V), the LFO will output signals ranging from -5V to 5V. In the + position, the output signals will range from 0V to 5V, and in the - position, the signals will range from -5V to 0V. Note that in the + and - positions, the amplitude of the LFO has been halved, and the center point of modulation has been shifted to 2.5V or -2.5V, respectively.

This can be desirable in many situations, but be cautious of using these modes with pitch modulation; if you were to create vibrato using the + mode, for example, your patch may sound great on its own, but the oscillators will actually be slightly sharp, and out of tune with the rest of the world!

Pulse Width- This knob adjusts the width, or "duty-cycle," of the pulse wave output. At 50%, a symmetrical square wave is produced, meaning the positive and negative portions of the cycle are equal lengths. As the knob is turned clockwise, the positive portion of the LFO cycle is increased; as it is turned counter-clockwise, the positive portion of the cycle is decreased. Be aware that the knob goes all the way to 0% and 100%- at either of these extremes, the modulation will be static.

Waveform Output jacks- These are the output jacks for the LFO signals. Each jack outputs a different waveform and can be used simultaneously in any combination. Shape options are square, random (i.e. sample and hold), sine, ramp up, ramp down and triangle.

45 Limiter

The Cherry Audio Limiter module is an extreme "brickwall" style limiter for audio signals. Limiters can be used subtly to "catch" the loudest peaks of a signal to keep it from clipping or quite aggressively to "smash" a signal's dynamics. They are famously known for their use in the mastering process to increase the overall level of a song while also making it sound punchier and "larger than life."

45.1 Inputs, Outputs, and Controls

Input jack- Patch audio signals here.

Reduction- Increases the amount of gain reduction. On many limiters, this knob is referred to as "Gain" or "Input Gain." Turning the knob up increases the level of the input signal being sent to the limiter's level detector. The louder the signal is, the more "Reduction" is needed to keep the signal from exceeding 0dB.

VU Meter (Red./Out)- This VU meter can be set to display the limiter's amount of gain reduction (*Red.*) or its output level (*Out*). When *Red.* is selected, the meter's needle will remain still at 0VU until the input signal exceeds 0db at which point it moves left to display how much the signal is being reduced to keep it from crossing 0dB. When *Out* is selected, the meter behaves as a typical VU meter displaying the level of the output signal.

Makeup Gain- This increases the volume of the output signal to "make up" for the amount of gain reduction imparted by the limiter.

Output jack- Outputs the limited audio signal.

46 Lyrinx Filter

The Lyrinx Filter precisely replicates the unique Voltage-Controlled Formant Filter of the super rare Synton Syrinx synthesizer. Consisting of two "peak" (bandpass) and one lowpass filter with multiple routing configurations, its excels at creating unique vocal-like timbres and includes extensive CV capabilities. This lets you easily create funky "talking" synth sounds. Lyrinx is super easy to use and capable of a wide array of timbres.

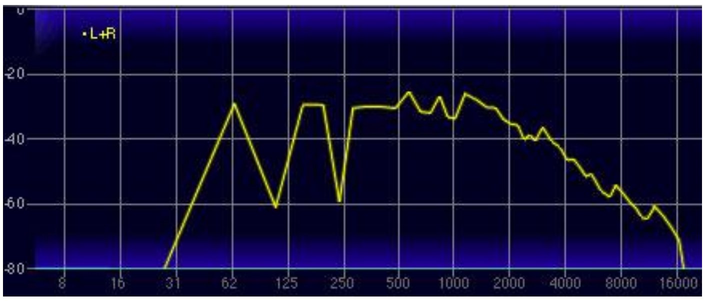
As mentioned, Lyrinx includes two "peak" (bandpass) filters and one lowpass filter. The two peak filters and one lowpass filter can be arranged in a few different ways dependent on the *VCF Routing* selector knob setting, but generally speaking, the signal flows from left to right going to the peak filters first, then to the lowpass filter.

46.1 How Bandpass and Lowpass Filters Work

For those unfamiliar with how different filter types work, let's review how bandpass and lowpass filters affect audio signals:

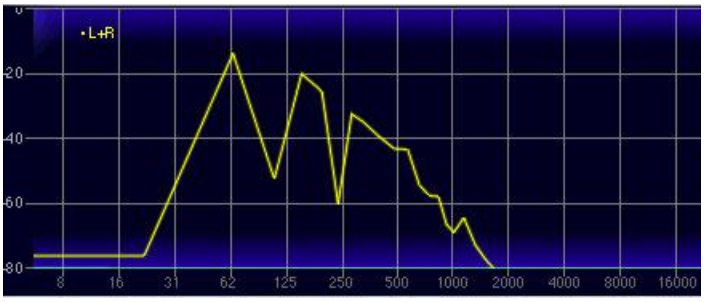
The original Syton Syrinx filter features two "peak" filters; as mentioned these are the same as bandpass filters. A bandpass filter is actually a combination of a lowpass filter *and* a highpass filter, configured in series (signal goes through one, then the other).

The lowpass section blocks frequencies *below* the cutoff frequency setting, but allows frequencies *above* the cutoff frequency to pass through. The highpass section does the opposite: it blocks frequencies *above* the cutoff frequency setting, but allows frequencies *below* the cutoff frequency to pass through.



The result is that only a "slice" of frequencies below and above the cutoff frequency are allowed to pass through. However, this slice has to be relatively wide, otherwise very little sound energy would remain audible.

The plot above shows how a bandpass filter affects an audio signal. (The vertical axis represents amplitude and the horizontal axis represents frequency.)



A lowpass filter allows frequencies *below* the cutoff frequency setting to pass through, but blocks frequencies *above* the cutoff frequency. The frequency plot to the left shows the effect of a lowpass filter on a saw wave. Notice how the high-frequency content trails off at the top of the audio spectrum.

46.2 Inputs, Outputs and Controls

The Lyrinx Filter contains a semi-terrifying quantity of controls, but it's actually pretty easy to use. Let's go over their functions.

In 1 and In 2 jacks- Lyrinx Filter features two individual inputs that allow a great deal of flexibility dependent upon the setting of the *VCF Routing* knob. In the first two positions, both inputs signals are mixed equally to one mono signal, so hey, free mixer. In the third and fourth positions, the two inputs are separately routed through the peak and lowpass filters (more on this in *VCF Routing* control section).

Keyb Track CV jack- This is a CV input, intended for use with the IO Panel *Pitch CV* output jack (though any CV source can be plugged into it). It allows cutoff frequencies to increase as higher notes are played on a keyboard or sequencer. The idea is to allow the harmonics of a sound to remain constant as higher notes are played. The *Keyb Track* input works in conjunction with the *Kbd Trk* knobs, which act as CV amount attenuators/boosters.

Peak 1&2 Out- Individual output jack for both peak filters.

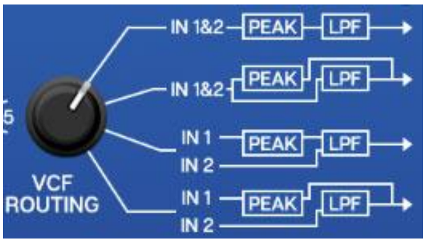
LPF Out- Individual output jack for lowpass filter. Depending on which input is used and the position of the *VCF Routing* selector, this may or may not be affected by the peak filters (sometimes making it the same signal as the *Mix Out* jack).

Mix Out- Output signal of all filter sections mixed together, in accordance with the *VCF Routing* signal flow diagram.

Kbd Trk (Oct/V) Pk 1&2 and LPF- These act as attenuators or boosters in conjunction with the the *Keyb Trk* CV input jack. *Pk 1&2* affects the cutoff frequency for both peak filters, while *LPF* affects the lowpass filter's cutoff frequency.

At a setting of 1 (i.e. 100%), this allows a sound's harmonics to remain constant (i.e., not become duller) as higher notes are played. The Synton Syrinx is unique in that its keyboard tracking controls go up to "2," equivalent to a setting of 200%. This allows far more contrast in filter brightness from low to to high notes than typical synth filters.

VCF Routing



- This selector switch allows four different routings through the bandpass and lowpass filters for different tonal colors. The front panel legend makes the signal flow easy to visualize.

- ***Position 1***- *In 1* and *In 2* are equally mixed, then routed through peak filters and lowpass filter in series. *In 1* or *In 2* can be used if a single signal is being processed.
- ***Position 2***- *In 1* and *In 2* are equally mixed, then the signal is multed and separately routed through the peak filters and the lowpass filter. *In 1* or *In 2* can be used if a single signal is being processed.
- ***Position 3****- In 1 is routed through the peak filters and the lowpass filter in series. In 2 is routed through the lowpass filter only.*
- ***Position 4****- In 1 is routed through the peak filters only. In 2 is routed through the lowpass filter only.*

Frequency*- Sets the frequency where attenuation begins. Attenuation occurs above and below the cutoff frequency for the peak filters, and only above the cutoff frequency for the LPF (lowpass filter).*

Peak 1 CV, Peak 2 CV, LPF CV mod inputs/attenuators- *Adjusts the depth of the cutoff frequency modulation, i.e. how much the filters sweep back and forth when a CV signal is applied. The attenuators allow negative or positive CV control of cutoff frequency. The center setting correlates to no modulation.*

Resonance Pk1, Pk2, and LPF*- Emphasizes sound energy at and around the current cutoff frequency by adding feedback from the filter's output back to its input. At lower settings, this can be used to create mild resonances such as those heard in acoustic instruments. At more extreme settings, resonance can create a pure sine wave at its own frequency. Be careful with the Resonance sliders as they can get loud at extreme settings.*

Note that subtle (or not so subtle) use of the individual Resonance controls is the key to creating unique "talking" filter effects (along with cutoff frequency CV mod).

Pk1 Res CV, Pk2 Res CV, LPF Res CV mod inputs/attenuators- *Adjusts the depth of resonance modulation. The attenuators allow negative or positive CV control of resonance level. The center setting correlates to no modulation.*

47 Mega Saw

Mega Saw delivers the immense "hoover" sound of multiple detuned sawtooth waves with up to 32 simultaneous saws plus CV-controllable detune amount. It's perfect for massive EDM lead lines, dramatic trailers, and more. We'll go over its functions, section by section.

47.1 Inputs, Outputs and Controls

47.1.1 Keyboard CV

Keyb CV jack- Accepts a CV input for pitch. Typically this would come from the PITCH jack in the IO Panel CV OUT section, or from a sequencer pitch CV out.

47.1.2 Saws

Number Of Saws and number display- This sets the total number of sawtooth waves you'll hear. The display shows the current number of saws (in case you don't feel like counting all those little lines around the knob).

Detune- Sets the average pitch detune amount for all sawtooth waves - higher values "spread" the detuning between the saws for a larger sound.

47.1.3 Detune Mod

Detune Mod attenuator and input jack- CV input for modulating the amount of detuning. Center setting is zero, and negative or positive modulation can be applied by turning the knob left or right.

Keyb CV jack- Accepts a CV input for pitch. Typically this would come from the IO Panel PITCH jack in the CV OUT section, or from a sequencer pitch CV out.

47.1.4 Freq

Range- Sets the basic pitch of the oscillator, displayed in traditional organ footage.

Fine- Fine-tune control for pitch.

47.1.5 Freq Mod

Freq Mod attenuator and input jack- CV input for modulating overall pitch. Center setting is zero, and negative or positive modulation can be applied by turning the knob left or right. Useful for adding vibrato with an LFO, siren noises, envelope-controlled pitch sweeps, etc.

47.1.6 Output

Output Jack- Where all those saw waves come out!

48 Micro Burst

The Cherry Audio Micro Burst module generates a “burst” of up to 32 CV triggers for each individual trigger it receives. Both the number of triggers per burst and burst rate are CV controllable.

This module can be used to replicate a number of acoustic sounds such as snare flams, hi-hat rolls and guitar strumming but is just as comfortable creating a cacophony of bleeps and bloops from another world! Try running a clock signal or LFO through the Micro Burst and use its output as the external trigger source for a Sample and Hold module. Modulating a resonant filter’s cutoff or an oscillator’s frequency can create some pretty crazy results!

48.1 Inputs, Outputs and Controls

Trigger jack- A 5V pulse or gate received at this jack will trigger a “burst” of triggers.

Num of Bursts- Sets the number of triggers, from 1 to 32, that will be output in each burst. The number can be externally controlled via its *Mod* CV input jack and attenuator.

Rate- Adjusts the speed at which the burst triggers are output. The time between triggers in the burst can be adjusted from 10ms to 100ms and can be externally controlled via its *Mod* CV input jack and attenuator.

Output jack- A burst of 5V pulses are output from this jack each time a single pulse is received at the *Trigger* input.

49 MIDI CC Converter

The MIDI CC Converter module converts eight selectable MIDI CC (continuous controller) messages from external MIDI devices to individual CV signals for use within Voltage Modular. This allows a hardware device's knobs and buttons to be used as CV sources.

49.1 Inputs, Outputs and Controls

MIDI Channel- Selects which MIDI channel the module will receive MIDI CC messages on. Set this to the same channel that your external device or devices are transmitting MIDI on. When set to *All,* CC messages from all 16 MIDI channels will be received allowing the module to convert CCs from multiple devices on different channels.

MIDI In jack- Receives MIDI messages from external MIDI devices. Typically this will be connected to the *From Host* MIDI output on the I/O panel or the *MIDI Out* jack of a MIDI In module.

MIDI CC#- Selects which MIDI CC (0 - 127) will be converted to a CV signal. Clicking on the small arrow to the right of the number display opens a pop-up menu of the standardized CC message assignments to pick from. Keep in mind that every piece of hardware will transmit different CC messages so it is usually easiest to refer to the MIDI implementation chart of your device or use the MIDI Learn function to find the number you are looking for.

> To find the CC# using the MIDI Learn function, right-click on any knob and select *MIDI Learn*, turn the knob you want to use, then look in the MIDI tab to see which CC# it transmitted. Once you know the number, you can remove the MIDI assignment by right-clicking on it in the MIDI menu and selecting *Unlearn*.

Many MIDI controllers allow you to assign any CC# to any knob or button so don't worry too much about what the list says. If a knob on your controller transmits CC# 5, it doesn't mean you have to use it to control portamento time.

CV Out jacks- Each jack outputs a CV signal for its respective MIDI CC#. MIDI CC values from 0 - 127 are converted to voltage between 0V and 5V. If no messages are received by the selected MIDI CC, no voltage will be output.

50 MIDI Channel Filter

The MIDI Channel Filter module allows easy selective filtering of any combination of channelized MIDI data and includes "All On" and "All Off" buttons to quickly enable or disable all MIDI channels.

This module can be used in various ways including muting/unmuting MIDI channels on the fly, splitting data to multiple MIDI destinations, or rerouting MIDI channels. You could, for example, use the MIDI Channel Filter to isolate a single channel of MIDI from a signal and reroute its data to a different channel and/or destination using a MIDI Output module.

50.1 Inputs, Outputs, and Controls

MIDI In jack- Patch MIDI signals here to selectively filter which channels are passed to the output jack. The small LED lights red when MIDI is being received.

Active Channels (1-16)- Click on these buttons to select which MIDI channels are passed to the output jack. Channels are active when their corresponding button is lit green. MIDI data received on active channels will be passed to the output while data received on unselected channels will be filtered out.

All Chnls (Off/On)- These *Off* and *On* buttons can be used to quickly select or deselect all 16 MIDI channels with a single click.

Midi Out jack- Outputs all MIDI data received on active channels. The small LED lights red when MIDI is being output.

51 MIDI Clock Divider

The MIDI Clock Divider allows Voltage Modular to be synced to external devices such as drum machines, synths, and sequencers which are capable of sending MIDI clock. This module divides a 24-pulse-per-quarter-note MIDI clock signal into slower, musically relevant note-values, and outputs a voltage-based clock signal that can be used to advance sequencers, switches, etc. inside of Voltage Modular. Clock divisions can be set from 1/32-notes to 4 bars including triplet and dotted values.

51.1 Inputs, Outputs, and Controls

MIDI Clk In jack- This is the MIDI input jack that will receive MIDI clock from an external device. Typically this will be patched to the *From Host* MIDI jack on the I/O Panel or the *MIDI Out* jack of a MIDI In module.

Reset jack- A 5V pulse received at this jack will immediately reset the clock divider. Note that most devices that send MIDI clock also send "Start" messages when the external device’s Play button is pressed which will automatically reset the MIDI Clock Divider. Remember to reset any sequencers, switches, etc. that are being triggered by the clock divider as well so that everything starts at the same time.

Note Value- Selects the note-value of the clock output pulses from 1/32-notes to 4 bars.

Any of the selected note-values can be changed to a triplet or dotted note-value by clicking the corresponding buttons which light up green when engaged.

For anyone unfamiliar with rhythmic note-values, a triplet clock will pulse three times for every two regular pulses of the same note-value, while a dotted-note clock will pulse twice for every three regular note-value pulses.

Clock Out jack- Outputs 5V clock pulses for syncing other modules in Voltage Modular. Often this will be patched to the external clock input of a sequencer but can be used for any number of things including advancing switches, resetting LFOs, and triggering sample and hold modules.

52 MIDI Drum Trigger

The Cherry Audio MIDI Drum Trigger module converts incoming MIDI notes to eight individual gate outputs and features easy-to-use *Learn* buttons for quickly mapping external devices. MIDI notes sent from the pads or keys of an external controller or drum machine can be mapped to individual drum modules, samplers etc. inside of Voltage Modular to create the modular drum-machine of your dreams!

52.1 Inputs, Outputs and Controls

MIDI Channel- Selects which MIDI channel the module will respond to. Set this to the same channel as your controller or external MIDI device. When set to "All," MIDI notes from all sixteen channels will be received allowing devices on different channels to trigger the module simultaneously.

MIDI In jack- Input jack for receiving MIDI notes from an external controller or MIDI device. This is typically patched to the *From Host* MIDI jack in the I/O panel or the *MIDI Out* jack of a MIDI Input module.

52.1.1 Triggers 1 - 8

Learn / MIDI Note- Displays the MIDI note-number assigned to each gate output. This can be reassigned by clicking the *Learn* button (the button will turn red), and playing the desired note on the device patched to the *MIDI In* jack.

Gate Out jack- Outputs a 5V gate signal while the respective MIDI note is being played. This will typically be used to play a drum sound via the trigger input of a drum module or gate input of a sampler but can also be used, for example, to start and stop sequencers or step through switch modules.

53 MIDI Input

The Cherry Audio MIDI Input module receives MIDI messages sent from an external MIDI device and converts them to CV signals for use within Voltage Modular. Using this module in addition to the CV Outs section of the I/O panel makes it possible to route MIDI data from multiple external MIDI devices to different parts of a patch.

An external sequencer could be used to play notes in one part of your patch while a midi keyboard controller is simultaneously used to play a different part of the patch.

53.1 Inputs, Outputs and Controls

Select MIDI Device- Click this button to select which external interface or port will be used for MIDI input.

Device Status- These two LEDs give visual feedback of the state of the selected external MIDI device. The *Active* LED will light up when the MIDI device is connected and working properly. The *Error* LED will light up if there is a problem with the MIDI device such as its connection being lost.

MIDI Out jack- MIDI output for passing the MIDI data received by the selected input device to other MIDI modules within Voltage Modular such as the Arpeggiator or Poly Octave Oscillator. The small LED next to this jack lights up when MIDI is being sent from the output.

Transpose- Shifts the pitch of the CV signals that are output from the *Pitch* jack up or down as much as three octaves in semitone increments.

Trig (Single/Multi)- Defines how gate and trigger voltages behave when a key is struck while another key is held. In *Single* mode, a new gate and trigger voltage will not be sent until all previously held keys are released. In *Multi* mode, new gate and trigger voltages are sent any time a new key is played. (Because the gate voltage is already "high," it will very briefly dip to zero volts when a new key is struck in order to let the module know to retrigger.)

Pitch- MIDI note number data is converted to a pitch CV signal and output from this jack. Typically this will be patched to the Keyb CV jack of an oscillator.

Gate- MIDI note on/off messages are converted to gate CV signals and output from this jack. This is often patched to the Gate In jack of an envelope generator to control a sound's amplitude and/or filter settings.

Vel- MIDI note velocity (how hard a key is pressed) values 0 - 127 are converted to CVs between 0V and 5V and output at this jack.

Bend- MIDI pitch bend messages are converted to CVs ranging from -5V (all the way down) to 5V (all the way up) and output at this jack.

Mod Whl- Mod wheel (MIDI CC# 1) values 0 - 127 are converted to CVs from 0V - 5V and output at this jack.

After Touch- MIDI After touch (channel pressure) values 0 - 127 are converted to CVs from 0V - 5V and output at this jack.

Sus- Sustain (MIDI CC# 64) on/off messages are converted to a CV gate signal and output from this jack.

54 MIDI Multiple

The MIDI Multiple module takes one MIDI input and "copies" it to four additional MIDI outputs so that one MIDI cable can be routed to multiple destinations. Since every jack in Voltage Modular can have up to six cables connected to it, it’s possible to merge up to six MIDI inputs and send the data to as many as 24 destinations!

The MIDI Multiple module can also be used to help organize or quickly re-route signals. The *From Host* MIDI jack, for example, could be patched to a MIDI Multiple which sends data to all of the MIDI modules in a patch. That way if you decide to change the MIDI input source, only one cable needs to be moved to re-route all of the MIDI in the patch.

54.1 Inputs and Outputs

Input jack- This is the MIDI input jack. All signals received at this jack will be “copied” to the four MIDI output jacks. Up to six cables can be connected to merge MIDI data.

Output jacks- Each of these four MIDI jacks will output the same MIDI data that is received at the MIDI input jack.

55 MIDI Output

The Cherry Audio MIDI Output module allows modules with MIDI out jacks, such as the CV To MIDI and CV To MIDI CC Converter, to control external MIDI hardware devices. This means that any MIDI capable sound module, synthesizer, drum machine or effects unit can be part of your modular setup! Create sequences, arpeggios and complex modulations inside of Voltage Modular and send them via MIDI to all of your favorite hardware.

The MIDI Output module can be used to merge up to six MIDI inputs. If more than six inputs are needed, multiple instances can be set to the same external device and their MIDI data will be merged.

In the image below, four channels of MIDI are being sent to the *MIDI In* jack. The MIDI Output module will merge the data so they can be simultaneously output by one MIDI cable. This allows four external devices to be individually controlled while “daisy-chained” together.

55.1 Inputs, Outputs and Controls

MIDI In jack- Modules with MIDI out jacks can be patched to this input to send their MIDI data to external MIDI devices. This jack can accept up to six MIDI inputs at once which will be merged and sent to the selected external MIDI device. The small LED next to the jack lights up when MIDI is being received.

MIDI Channel- Selects the MIDI channel on which the data received at the *MIDI In* jack will be output. When set from one to sixteen, MIDI data from all channels will be merged and output on the selected channel. When set to *All,* each channel of MIDI will be output on the same channel that it was received.

Transpose- Shifts the pitch of the output MIDI notes up or down as much as three octaves in semitone increments.

Select MIDI Device- Click this button to select which external interface or port will be used to output MIDI data.

Device Status- These two LEDs give visual feedback of the state of the selected external MIDI device. The *Active* LED will light up when the selected MIDI device is connected and working properly. The *Error* LED will light up if there is a problem with the MIDI device such as its connection being lost.

56 MIDI Trigger 3

The MIDI Trigger 3 module converts MIDI notes C3, D3, and E3 to trigger outputs for basic drum module setups. If you want a quick and simple way to convert three MIDI notes to trigger outputs this it! If three notes isn't enough, or you want to use different MIDI notes, check out the MIDI Drum Trigger module.

56.1 Inputs and Outputs

MIDI In jack- Input jack for receiving MIDI notes from an external controller or MIDI device. This is typically patched to the *From Host* MIDI jack in the I/O panel or the *MIDI Out* jack of a MIDI Input module. Remember that this module ONLY responds to MIDI notes C3, D3, and E3.

Trig Out jacks- Outputs a 5V pulse, or trigger, each time its respective MIDI note is received at the *MIDI In* jack. These will typically be used to trigger drum modules via their *Trig In* jacks but can also be used to trigger "one-shot" samples, envelopes or LFOs.

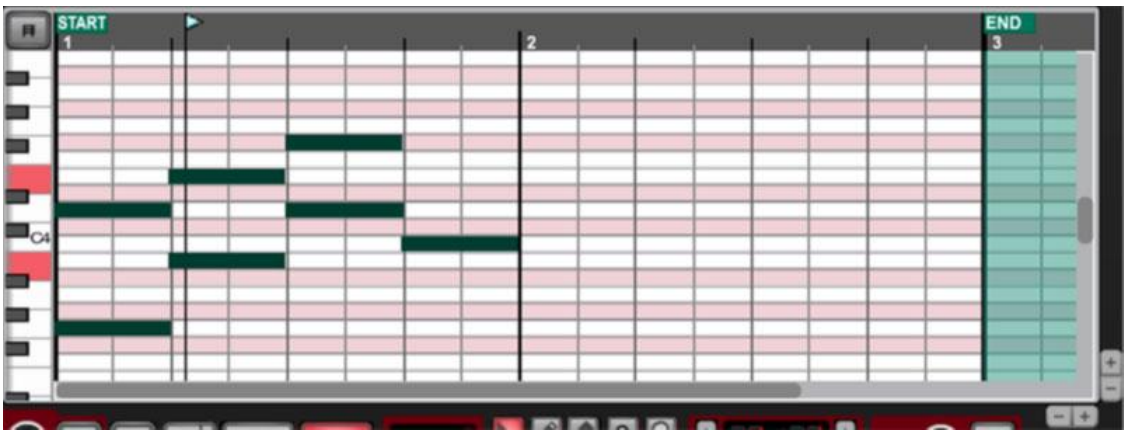
57 Mighty Piano Roll

Mighty Piano Roll is a full-function piano roll-type sequencer, similar to piano roll editors commonly seen in DAW editors. Not only does it allow graphic drawing and editing of note sequences, it's fully polyphonic, and supports independent MIDI channel assignment for each note.

In this guide, we'll interchangeably refer to Mighty Piano Roll as "MPR" (so don't confuse it with that homely Mazda car-van thing.)

Many of Mighty Piano Roll's functions have QWERTY keyboard shortcuts - these are indicated in [brackets] next to the function name.

57.1 Sequence Grid Window



This is where all notes are displayed, drawn, recorded, sliced, diced, etc. The sequence grid displays pitch vertically from low to high and covers the entire MIDI note range. White grid squares indicate natural notes (i.e. white keys) and pink squares indicate accidental notes (i.e. black notes). "The Pink Squares" may also be a band we saw open for Quarterflash and the The Go-Go's at Madame Wong's in 1981 (but maybe not).

Time is displayed horizontally with vertical lines indicating time divisions. The number of these lines/note value varies dependent upon current zoom level, but they have no effect upon playback.

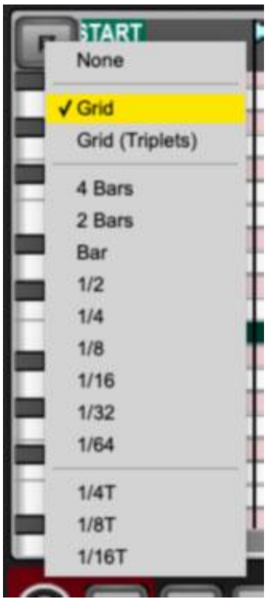
Start/End flags- At the top of the grid, these green flags indicate the start and end points of the current sequence. You'll generally want the *Start* flag at the very beginning of the sequence, but moving it can be useful if you'd like to change the start point of an existing sequence. Since only the area between the flags is active, any grid region to the right of the *End* flag is green to show that it's inactive.

If *Loops Sequence* is disabled, playback or recording will stop when the playback line reaches the *End* flag.

If a note is manually drawn in past the *End* flag, the *End* flag location will move forward.

Playback line/flag- The vertical line with a green triangle at the top indicates the current playback position. The playback line can be moved to any location by clicking in the dark gray area above the grid; it will snap to the closest current snap button value (located above the keyboard).

Snap button- The button directly above the keyboard opens the snap value menu. Note durations and locations are rounded to this setting. The icon on the button changes to show the current setting.



-/+ zoom buttons- Use these buttons located at the bottom right of the grid to enlarge or en-small (yep, made that up) the sequence grid view vertically or horizontally.

Scroll bars- These move the view range up and down or side to side. With short sequences (one or two bars) you probably won't need to use them. The *Grid* setting snaps notes to the currently visible vertical lines, which vary dependent on the zoom setting. *Grid (Triplets)* does the same, but rounded to the closest triplet value (in case you're creating a wicked waltz or one of those mid-tempo 90s dentist-office jams).

Grid notes- The colored rectangles on the grid represent notes, with note length corresponding to their width.

57.1.1 Grid Note Right-Click Menu

Delete- Erases selected note. Multiple notes can be deleted simultaneously if more than one is highlighted.

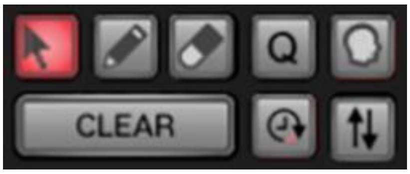
Select All- Highlights all grid notes.

Velocity- Sets MIDI velocity. Velocities can be set for multiple notes if more than one is highlighted. Notes drawn into the grid with the pencil tool will default to MIDI velocity 100.

MIDI Channel- Sets note MIDI channel. MIDI channel can be set for multiple notes if more than one is highlighted. Notes drawn into the grid with the pencil tool will default to MIDI channel 1. The MIDI channel of a note is indicated by its color as follows:

MIDI CHANNEL	COLOR	MIDI CHANNEL	COLOR
1	light green	9	orange
2	dark blue	10	red
3	medium blue	11	salmon
4	light blue	12	light purple
5	turquoise	13	gray purple
6	med green	14	violet
7	lime green	15	olive
8	yellow	16	blue purple

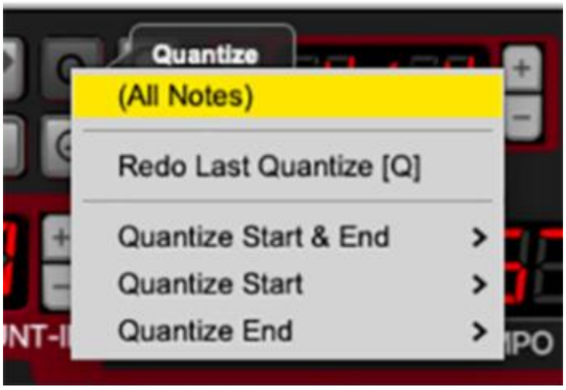
57.2 Tool Buttons



Select Tool (arrow) [Z]- The arrow tool is used to move notes by clicking and dragging, or to change note duration by clicking and dragging their left or right edge. Note durations will snap to the value currently selected with the *Snap* button.

Pencil Tool [X]- The pencil tool is used to create new notes by clicking on the grid. The duration of newly created notes is set with the *Note Duration* pop-up menu to the left of the tool select buttons.

Eraser Tool [C]- Click on notes with the eraser to... erase them. To erase multiple notes, click and drag over them. Remember that Voltage's undo buttons (the twirly arrows next to the patch preset box up top) can be used if you make a mistake.



Q / Quantize and pop-up menu- The quantize tool offers a number of powerful quantizing options. Unlike the previous tools it doesn't toggle on and off; it's used by first highlighting or more notes in the grid, then clicking and selected one of its quantize submenu options.

The submenu options are as follows:

- **(All Notes) / (Selected Notes)**- This indicates which notes will be affected by quantize operations. If no notes are currently selected, it will display *All Notes* and the selected operation will affect all notes in the grid. If one or more notes are currently highlighted, it will display *Selected Notes* to indicate that only the currently highlighted notes will be affected by quantize operations.
- **Redo Last Quantize [Q]**- In the menus beneath, you'll specify the desired quantization parameters (beginning of note, end of note, and quantization note value). Since you'll likely want to use the same type of quantization, *Redo Last Quantize* saves you from having to repeatedly delve two sub-menus down to quantize.
- **Quantize Start & End**- Rounds the start and end of note(s) to the chosen note value.
- **Quantize Start**- Rounds the start of note(s) to the chosen note value.
- **Quantize End**- Rounds the end of note(s) to the chosen note value.

Humanize (head icon) and pop-up menu-

The humanize tool is effectively the opposite of the quantize tool - it adds randomness to the timing of a note or notes.

Since most of us are pretty darn good at playing imperfectly, humanize is typically applied to perfectly quantized note passages; that is, notes that are sitting exactly on grid lines.

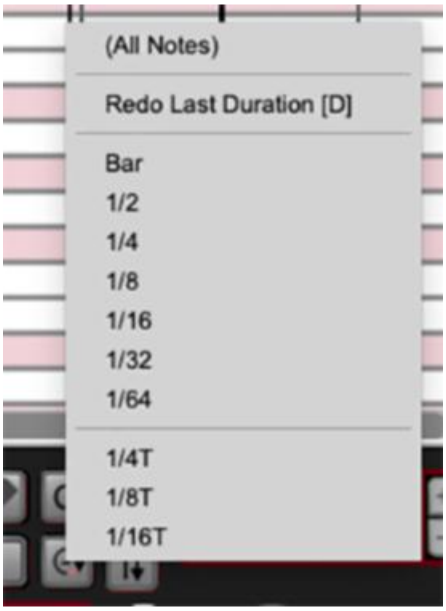
Its functionality is almost identical to the quantization options explained above, but instead of note values, it includes options for *Strong* (a lot of timing variance), *Moderate* (medium timing variance), and *Minimal* (very little timing variance). Its submenu options are as follows:

- **(All Notes) / (Selected Notes)**- This indicates which notes will be affected by humanizing operations. If no notes are currently selected, it will display *All Notes* and the selected operation will affect all notes in the grid. If one or more notes are currently highlighted, it will display *Selected Notes* to indicate that only the currently highlighted notes will be affected by quantize operations.
- **Redo Last Humanize [H]**- In the menus beneath, you'll specify the desired humanization parameters. Since you'll likely want to repeatedly use the same type of humanization, *Redo Last Humanize* saves you from having to repeatedly delve two sub-menus down to humanize.
- **Humanize Start & End**- Randomizes the start and end of note(s) to the chosen humanization amount.
- **Humanize Start**- Randomizes the start of note(s) to the chosen humanization amount.
- **Humanize End**- Randomizes the end of note(s) to the chosen humanization amount.

Clear- Deletes the currently selected note(s). **If no notes are selected, it will erase all notes on the grid**. Again, remember that Voltage's undo buttons (the twirly arrows next to the patch preset box up top) can be used if you blow it and erase everything.

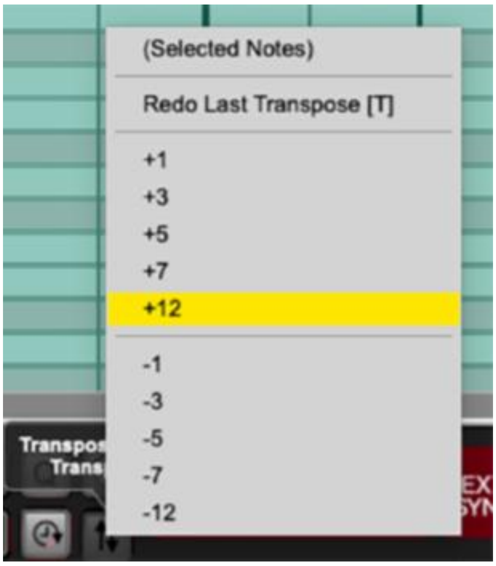
Set Duration (twirly arrow with clock with a dolphin shooting rainbows from its blowhole icon, ok, maybe only the first part)-

Allows a note duration to be set for one or more notes simultaneously. The pop-up menu format for this guy is similar to the quantize and humanization buttons.



- **(All Notes) / (Selected Notes)**- This indicates which notes will be affected by setting the duration. If no notes are currently selected, it will display *All Notes* and the selected operation will affect all notes in the grid. If one or more notes are currently highlighted, it will display *Selected Notes* to indicate that only the currently highlighted notes will be affected.
- **Redo Last Duration [D]**- In the menus beneath, you'll specify the desired duration parameters. Since you'll likely want to repeatedly use the same type of humanization, *Redo Last Humanize* saves you from having to repeatedly delve two sub-menus down to humanize.
- **Duration values**- Select these to set note lengths.

Transpose (up/down arrows icon)-



Transposes a note or multiple notes by moving them vertically in the grid. Again, the pop-up menu format is similar to the preceding controls.

- **(All Notes) / (Selected Notes)**- This indicates which notes will be transposed. If no notes are currently selected, it will display *All Notes* and the selected operation will affect all notes in the grid. If one or more notes are currently highlighted, it will display *Selected Notes* to indicate that only the currently highlighted notes will be affected.

- **Redo Last Transpose [T]**- In the menus beneath, you'll select the transposition interval. Since you may want to repeatedly transpose by the same interval, *Redo Last Transpose* saves you from having to repeatedly select a value. The *T* key on the QWERTY keyboard functions as a shortcut for *Redo Last Transpose*.
- **Transpose Intervals**- Choosing one of these values allows transposition up or down by any value up to an octave.

57.3 MIDI Jacks / Panic Button

MPR uses MIDI jacks for all note I/O.

MIDI In- Use this to input notes. Typically this will be patched from the IO Panel *MIDI/From Host* jack.

MIDI Out- Use this to route note playback to modules. It also acts as a MIDI Thru for incoming data.

Panic Button- The upside-down button with an exclamation point sends a MIDI all notes off message in case anything gets stuck.

57.4 Transport Buttons and CV Controls

Play Gate- Outputs a constant 5V gate signal when MPR is in play mode. It's unaffected by note-ons and note-offs.

Play/Stop- Starts and stops pattern playback.

Reset- Sets the playback line back to the beginning of the pattern grid.

Record/Record Quantization- The record button starts the transport and immediately begins recording notes from the MIDI input jack at the current location of the playback line. When pressed again, Might Piano Roll exits recording mode, but will continue to play. To stop the playback, whack ye olde *Stop* button.

The skinny right-side portion of the button sets record quantization - this automatically quantizes incoming notes to the selected value. The recording quantization button glows red if a note value other than *None* is currently selected.

Step Record- Allows notes to be manually input. Step recording always beings at beat 1 on the grid. Note duration are length of each step are specified via the *Note Duration* and *Step Length* pop menus, respectively. Note size will be the smaller of either note duration or step length. Single-note melodies or chords may be entered in step record mode.

Loops Sequence- Engaging this causes the pattern to repeat from the *Start* flag when the play line reaches the *End* flag. *Loop Sequence* is on by default.

Note Duration- Sets the length for note entered via Step Record mode, or drawn into the grid with the pencil tool.

Step Length- Sets the amount of time (in note value) that the play line jumps by when notes are entered using Step Record mode, or when controlling the playback bar using the *For*, *Rev*, and *Rand* Step CV trigger inputs. (See the next section for info on this)

57.4.1 Step CV Triggers

The five grouped jacks at the bottom allow control of playback in various ways via trigger CV's. Trigger CV's are typically 5V, but these jacks will respond to any voltage > 2.5V.

Play (Stop)- Toggles the play and stop modes.

Reset- Resets the playback line to the beginning of the pattern grid.

For (Forward)- Plays from current playback bar position for the length currently set in the *Step Length* pop-up menu to the right of the jacks.

Rev (Reverse)- The playback bar will jump back by the length currently set in the *Step Length* pop-up, then play for the length currently set in the *Step Length* pop-up. Sort of a "jump back and repeat."

Rand (Random)- The playback bar will jump back to a random location and play for the length currently set in the *Step Length* pop-up. The destination of the playback bar is always be a division of the current *Step Length* value. For example, if the S*tep Length* value is set to quarter-notes, the playback always lands on a quarter note. The playback bar will always land on even values regardless of its initial location.

57.4.2 Rec Gate

Rec Gate- Enables recording when a gate voltage is present. If the transport is stopped, the gate CV will start it; when the gate CV is removed, the transport continues to play.

Count-In- If set to a value other than *0,* when MPR receives a gate signal, it waits the number of bars before initiating recording. You'll most likely want to make sure the play pointer is on an even bar line, or else recording can begin on a funky bar line.

57.4.3 Time Signature, Tempo, Sync Out, Bar and Beat Outs

Sync Out- Outputs a 96 PPQN sync signal when MPR is in play mode. Remember that sync signals are different than gate and clock signals - if you'd like to synchronize something at a steady musical note value, use a Sync Divider module to convert the sync signal to a musical value. (If the timing is a little shifted, patch the *Play Gate* CV out to the Sync Divider *Reset* input.)

Time Signature- The grid time signature numerator and denominator can be set using the +/- buttons on each side of the display. The pattern grid line arrangement will reflect the current setting.

Tempo- Current tempo, expressed in beats per minute. The rate is adjustable with the big red knob.

Ext Sync jack and button- Allows control of tempo via an external sync source, such as the IO Panel Transport *Sync Out* jack or a Sync Generator module when enabled. As mentioned, remember that this input needs a 96 PPQN sync signal, not a clock signal.

Bar and Beat out jacks- These send a 5V trigger pulse at the beginning of each bar and beat respectively; their LED's indicate when a CV is sent. They're handy for turning things on and off or triggering percussion sounds or effects. If you're patching these to the gate input of an envelope generator, convert them to a gate signal (i.e. extend their duration) with a Trig To Gate module.

57.5 Interfacing Mighty Piano Roll With Other Voltage Modules

Mighty Piano Roll uses MIDI in and out jacks. MIDI I/O was chosen mainly because it allows support of up to 16 independent MIDI channels - any note in MPR can be assigned to any MIDI channel. This allows a great deal of flexibility with just one instance of MPR.

- The Poly Octave Oscillator is the easiest Voltage voice module to use with MPR: it's polyphonic, it has a MIDI jack for note input, and built-in envelope generator.
- For greater creative possibilities, the best option is to use MPR with poly modules from Cherry Audio and third-party module makers. To convert MPR's MIDI output to poly jack pitch, gate, and velocity signals, use a MIDI To Poly CV module. The MIDI Channel Filter module can also be helpful.
- To play individual monophonic parts, use the Poly CV Converter module to split MIDI note data into four independent pitch, gate, and velocity CV's. Its MIDI *Overflow* output jack can be used to stack The Poly CV Converter instances for additional channels.

57.6 Keyboard Shortcuts

The table below shows all MPR keyboard shortcuts:

QWERTY KEYBOARD KEY	FUNCTION
Z	select tool
X	pencil tool
C	eraser too
A	select all
Q	repeats last quantize operation
H	repeats last humanize operation
D	repeats last duration operatio
T	repeats last transpose operation

57.6.1 Selection

ACTION	FUNCTION
Drag on the timeline (above grid)	selects time region - hold [SHIFT] to inhibit snapping
Drag over notes in grid	selects multiple notes

58 Mini LFO

The Cherry Audio Mini LFO module features two independent, low-frequency oscillators. Each LFO has an adjustable rate and simultaneously outputs a triangle and square wave.

If you are unfamiliar with LFOs it may be helpful to read the documentation for the "standard" LFO module.

58.1 Outputs and Controls

Rate- Sets the frequency, or rate, of the LFO. Frequency values are represented in Hz (cycles per second). A frequency setting of 1 Hz means it takes one second to complete a full cycle of the waveform. The Mini LFOs frequency range is 0.02 Hz - 20 Hz.

Output jacks- These are the output jacks for the LFO signals. Each LFO has a triangle and square wave output which can be used simultaneously.

59 Mini Mono to Poly

The Mini Mono to Poly module converts a standard "mono" CV or audio input signal to a multi-channel "poly" output. The input is copied to each channel, or "lane," of the poly output making it possible to send the signal to other poly module's input jacks.

59.1 Inputs and Outputs

CV In jack- Patch standard "mono" CV or audio signals here to convert them to poly signals. The small LED glows red when voltage is present.

Poly Out jack- This "poly" output carries the signal received at the *CV In* jack on each of its channels, or "lanes."

60 Mini Plug-In Host

The Mini Plug-In Host module lets you use almost any virtual instrument or effect plug-in "inside" of Voltage Modular and allows CV control of four plug-in parameters. It includes four audio ins and outs as well as MIDI I/O.

It's functionally identical to the larger "double-wide" Plug-In Host module, but has four parameter control slots instead of twelve. We provided this to save space when many control slots aren't needed.

Please see the full-size Plug-In Host user guide for information on how to use Mini Plug-In Host.

61 Mini Poly to Mono

The Mini Poly to Mono module mixes, or sums, the individual signals carried by a "poly" CV or audio cable to a standard "mono" output jack. This module can be used, for example, to mix the individual audio signals, or voices, of a poly synth to a standard mono output so that it can be sent to the *Main Outs*.

61.1 Inputs, Outputs, and Controls

Poly In jack- Patch "poly" CV or audio cables here to convert the signals to standard "mono" cables.

Mix Level- Adjusts the level of the combined signals. It is often necessary to attenuate the output to compensate for the increase in amplitude caused by summing multiple signals.

Mix Out jack- Outputs a standard "mono" signal carrying the sum of all signals received by the *Poly In* jack.

62 Mod Wheel Assistant

The Mod Wheel Assistant is a handy "helper" module designed to make configuring mod-wheel controlled LFO routings quick and easy. The module has a full-featured internal LFO whose amplitude is scaled by the voltage received at the *Mod Wheel In* jack. An attenuator is also included at the output to scale the overall depth of modulation.

Typically the *Mod Wheel* output jack from the CV Outs section of the I/O panel will be connected to the module's *Mod Wheel In* jack. Mod wheel messages sent from your MIDI keyboard or DAW will then control the amplitude of the internal LFO which in turn increases or decreases the amount of modulation it imparts on its destination.

A typical application would be to use the mod wheel to adjust the amount of vibrato (pitch modulation) in a patch. To set this up, patch the *Mod Wheel* jack from the CV Outs section of the I/O panel to the Mod Wheel Assistant's *Mod Wheel In* jack and patch the *Output* jack to the Keyb CV or Freq Mod input of an oscillator.

Remember that any type of control signal can be patched to the *Mod Wheel In* jack. An envelope, sequencer, or LFO could be used to modulate the depth of the internal LFO in the same way that a mod wheel allows you to control it manually.

62.1 Inputs, Outputs and Controls

Mod Wheel In jack- Patch the *Mod Wheel* jack from the CV Outs section of the I/O panel to this input to control the internal LFO's amplitude manually with the mod wheel of a MIDI keyboard controller.

Frequency- Sets the frequency, or rate, of the internal LFO.

Polarity- Sets the polarity of the LFO. With the mod wheel and the *Amount* knob all the way up, *Bi* will output a bipolar LFO ranging from -5V to 5V, + will output an LFO ranging from 0V to 5V and - will range from -5V to 0V.

Pulse Width (PW)- Adjusts the width, or "duty-cycle," of the LFO for the pulse wave only. At 50%, a symmetrical square wave is produced, meaning the positive and negative portions of the cycle are equal lengths. As the knob is turned clockwise, the positive portion of the LFO cycle is increased; as it is turned counter-clockwise, the positive portion of the cycle is decreased.

Waveforms- These six buttons select the waveform of the LFO. Shape options are pulse, random (i.e. sample and hold), sine, ramp up, ramp down and triangle.

Amount- This is a second stage of attenuation that limits the maximum amount of modulation imparted by the LFO.

Output- Outputs the scaled internal LFO signal. Connect this jack to the CV input of the parameter whose modulation depth will be controlled via the mod wheel.

63 Mono To Poly

The Mono To Poly module accepts up to 16 individual CV or audio inputs and converts them to a single multi-channel "poly" output. The module can be used with CVs to send different envelopes, LFOs, pitch CVs etc. to each individual voice of a polyphonic patch or to convert multiple mono audio signals, such as oscillators, to a "poly" audio signal.

63.1 Let's look at an example.

In the patch below, four LFOs are patched to a Mono To Poly module's inputs to create a "poly" CV signal that is modulating the cutoff frequency of a Poly Filter module. This results in a patch where each of its four active voices have a different LFO rate modulating its filter.

63.2 Inputs, Outputs, and Controls

Input jacks- Each input jack can receive a unique CV or audio signal that will be "packaged" together and output as individual channels, or "lanes," of the poly output signal. The number of active inputs depends on the *Number of Voices* setting in the I/O Panel. When the patch is set to eight voices, for example, the LEDs for *Inputs 1 - 8* will glow red to indicate that they are active. Signals received by inputs that are not active will not be passed to the output.

CV Type- This is a handy little text box that can be used to label the module. To change the text, click in the box labeled "Click to enter label" and begin typing.

Poly Out jack- Outputs the CV or audio signals received by the active input jacks as a multi-channel "poly" signal.

Out Level- Simultaneously adjusts the output level of all channels of the poly signal.

64 Multiple

The Cherry Audio Multiple is a dual module that “copies” the CV or audio signal received at its input to six output jacks so that the signal can be sent to multiple destinations. When the *Link* button is engaged, the input from module one is sent to the outputs of module two as well, creating a total of 12 copies.

The Multiple module can also be used to help organize or quickly re-route signals. Copying the *Pitch* and *Gate* jacks from the I/O panel, for example, is a common practice for keeping things tidy and versatile. By using the outputs of a Multiple to send pitch and gate CVs to all of the oscillators and envelopes in a patch, the input source can easily be changed, to an arpeggiator or sequencer for example, without having to re-patch every pitch and gate CV.

Remember that all Voltage Modular input and output jacks feature unlimited mults; the idea behind using a mult module is to make it easier to visualize cable routing. Using a dedicated multiple module also lets you take advantage of Voltage's *Add Label* right-click command for better organization.

Finally, if you don't need the Multiple module's two independent jack fields, the *Micro Mult* module offers 1>10 connections.

64.1 Inputs, Outputs and Controls

Input jack- This is the input jack for the CV or audio signal that will be copied to the output jacks.

Output jacks- These six jacks will output a copy of the signal received at the input jack.

Link- When this button is engaged, the input from module one will be copied to the outputs of module two as well, and the input of module two will be ignored. When linked there is a total of 48 possible copies.

65 Noise Generator

The Cherry Audio Noise Generator simultaneously outputs white and pink noise which can be attenuated by its built-in amplifier.

Noise signals are random-voltage signals that can be used as audio or control signals. As an audio signal, noise is often used to emulate the percussive hit of a drum or the breathiness of a voice. As a control signal, it can be a great source for creating randomness in a patch. Noise signals are often used as the input signal of a Sample and Hold module which can adjust the rate at which random voltages are output.

65.1 Outputs and Controls

Level- Attenuates the amplitude of the noise signals.

White Noise- Outputs a random signal in which all frequencies across the frequency spectrum are represented equally.

Pink Noise- Outputs a random signal in which each octave across the frequency spectrum is represented equally. Pink noise will sound duller to the ear than white noise, and as a control signal, will be less likely to output higher frequencies.

66 Notch Filter

The Cherry Audio Notch Filter is a variable-bandwidth audio filter for removing frequencies in a CV-controllable notch. This module can be used "surgically" to target and remove unwanted frequencies from an audio signal or can be used more creatively to create phaser-type effects by modulating the filter's frequency.

66.1 Inputs, Outputs, and Controls

Input jack- Patch audio signals here.

CV Mod jack and attenuator- CV input jack and attenuator for externally controlling the *Frequency.*

Frequency- Sets the frequency where the filter's notch will be. Audio content at and around this frequency will be removed from the input signal.

Bypass button- When lit red, the notch filter is active. When gray, the filter is bypassed. Notch filters are often used to make subtle adjustments. Toggling this button on and off can help identify which frequencies are being removed by the filter.

Bandwidth- Adjusts the width of the notch. Low values create a narrow notch while higher values create a wider notch.

Output jack- Outputs the filtered audio signal.

67 Octagon

Octagon is a mondo powerhouse sequencer. At first glance, it appears to have just 8 steps, but it actually can be set from 1 up to 32 steps, and each step can have up to 8 individual clock "pulses." Add this to myriad step play order options, per-step 303-style slide, and it's easy to see what a beast Octagon is!

We'll begin by explaining Octagon's top section for global controls, including transport, number of steps, and more.

67.1 Global Controls

67.1.1 Steps and Banks

A "step" refers to one set of sequence step controls. Eight steps are visible at all times, but Octagon can actually be set between 1 and 32 steps. Each step, in turn, can consist of between 1 and 8 gate or trigger "pulses." The important thing to remember is that ***each step has its own CV value (typically a musical note), but pulses within a step will all be the same CV value (note).***

As a result, ***individual clock pulses from Octagon's internal clock or external clock pulses correspond to the pulses within each sequencer step*** (not the entire step, like most typical sequencers). This may sound a little confusing, but we promise, it's easy to wrap your head around once you start using it.

Octagon can have up to 32 sequence steps with 8 steps viewable at any given time.

Shift Bank Left/Right buttons- Shifts which sequence step sliders are currently visible in banks of 8. For example, if the *Sequence Length* is set at 16 steps, and steps 1-8 were currently visible, clicking the *Shift Bank Right* button would change the view to steps 9-16. The gold numbers beneath each step change to reflect this.

Shift Step Left/Right buttons- Shifts which sequence step sliders are currently visible by one step. For example, if the Sequence Length is set at 16 steps, and steps 1-8 were currently visible, clicking the *Shift Bank Right* button would change the view to steps 2-9. The gold numbers beneath each step change to reflect this.

Step Play Order button and display- Clicking the button or display opens a pop-up where step playback order can be changed. The step play order modes are as follows:

- *Forward*- Starts at first enabled step and continues to last enabled step then loops. This is the default setting.
- *Reverse*- Starts at last enabled step and continues to first enabled step then loops.
- *For-Rev*- Moves forward until it reaches the last step number then reverses. When it reaches step 1, it plays forward.
- *For-Rev Repeat*- Same as *For-Rev* but plays the first and the last stage twice. This can help to keep sequences playing correctly in 4/4 time, for example.
- *Even Only*- Plays even steps only, in forward direction.
- *Odd Only*- Plays odd steps numbers only, in forward direction.
- *Funnel*- First, last and inward, e.g., if the sequence length is 8 steps, the order would be 1,8,2,7,3,6,4,5, etc.
- *Hourglass*- Same as *Funnel* above, but once it reaches the "center," it works its way back out in reverse, ex: 1,8,2,7,3,6,4,5,5,4,6,3,7,2,8,1.
- *Random*- Starts at a random step and chooses steps randomly within the # of steps setting.
- *Brownian*- Advances in a pseudo-random pattern known as “drunken walk.” Starting at stage 1 it has a 50% chance of moving forward, 25% chance of staying at the same stage, and 25% chance of moving backwards. This results in a sequence that mostly moves forward with some degree of repetition. (Not to be confused with *James Brownian*, which only plays on step one, and fines you if you make a mistake.)

Order CV jack and associated Step Play Order modes

The *Order CV* jack allows CV control of which steps are played when using the *Step Play Order* modes below. Since the *Order CV* jack only affects these modes, it appears "grayed out" for all other modes.

- *CV Cont - Distributed*- The 0-5V CV "spread" is equally divided by the current *Sequence Length* setting. For example, If the sequence length was 8 steps, voltages from 0-0.625 would correspond to step 1, 0.626-1.25 would correspond to step 2, 1.26-1.875 would correspond to step 3, etc. Math is fun, eh?
- *CV Cont - 1V/oct*- Each stage corresponds to 1/12V divisions, allowing a keyboard or secondary sequencer CV to "play" the steps in a consistent fashion. Unlike *CV Cont - Distributed* mode, stage voltages are the same regardless of the *Sequence Length* setting.

67.2 Transport Section

The top area of the module uses standard tape deck-style transport controls.

Stop button and CV jack- Stops sequencer from running. The *Stop* button can be activated via CV using the jack below the button with any trigger or gate CV greater than +2.5 volts.

Start and CV jack- Starts sequencer running. The *Start* button can be activated via CV using the jack below the button with any trigger or gate CV greater than +2.5 volts.

Step- Advances current position to the next step. This is useful for setting pitches for each stage when stopped. The advance button also works when the sequencer is in play mode. Note that we didn't include a CV jack for step advance- the *Ext Clk* jack does exactly this, hence the fanciful arrow.

67.3 Rate, Gate, and More Top Controls

Ext Clk button and CV in- Clicking the *Ext* button disconnects Octagon's internal clock and accepts clock signals from *Ext Clk* jack. Octagon isn't too fussy about external clock sources; pretty much anything that creates rapid (or not-so-rapid) pulses can be used, including LFO's, oscillators, or even the gate or trigger output of another sequencer. Along these lines, note that external clock pulses don't have to consistently repeat either; any pattern of pulses can drive the sequencer *Ext Clk* input.

Reset- This input jack is **really** important for locking sequencer timing to a DAW project or other sequencers. It force-resets the sequencer to the very beginning of step 1 the instant it receives a gate or trigger voltage.

Offset CV- This input jack lets you add or subtract overall voltage from the sequencer's output. Most commonly this would be used to transpose the key of a sequence during playback (from a keyboard CV, or another synced sequencer running at a slower rate), but it can also be used for more esoteric applications, such as routing an LFO to continuously vary the pitch of the entire sequence.

Rate knob+LED/CV jack and attenuator- The rate knob sets the internal clock speed from around 4 - 240 bpm. The LED indicator flashes with each "click" or step advance.
The *Rate* pop-up tooltip is calibrated to display tempos based on sixteenth-notes. For example, setting the *Rate* knob to 120 bpm plays 480 notes a minute (We did this because you'll likely want to play fast tempos such as this, and it's sort of kooky to set the knob to 480 bpm to get sixteenth-notes). External clock signals can used if faster or slower speeds are needed - we won't judge! The CV In jack and attenuator allow CV control of tempo.

Gate Time- Sets the length of the 5V gate signal from 1-1000ms for each pulse in a sequence step. The *Gate Time* setting is not affected by the overall sequencer rate. It also has no effect on the *Trig Out* jack signals (because a trigger signal is always a rapid pulse). The CV In jack and attenuator allow CV control of the gate length.

Sequence Length and display- These up/down buttons set the total number of sequencer steps from 1-32 steps. This defaults to 8 steps and can be altered with the sequencer in stop or play mode. The *CV In* jack and attenuator allow CV control of the gate length.

Pro tip: Holding down the buttons continuously changes the setting, so you won't have to click the button a zillion times.

Slide knob- The *Slide* control causes notes to slide from one pitch to the next, as opposed to discretely jumping from one pitch to the next. It works in conjunction with the individual *Slide* buttons in each step's controls - **slide only occurs if the *Slide* button is enabled for that step.**

Octagon's slide implementation is unique in that it features two types of slide; normal and "303" style.

- **Standard slide-** Technically, this would be referred to as "constant time" slide, where the speed of the glide is fixed and higher settings = longer glide time. This is how the portamento or glide operates in most classic monophonic synthesizers. Depending on the overall *Rate* setting, note interval distance, and *Slide*knob setting, pitches may or may not fully "make it" up or down to the next note in the time between steps.
- **Constant Rate/303-Style slide-** "Constant rate" slide emulates the famous little silver box o' techno heard around the world. It has a few unique idiosyncracies.

- - Pitch changes from one step to the next *always* occurs within the time between the two steps, regardless of pitch interval or bpm setting. In other words, the pitch change will always "arrive" at the following pitch in the time between the steps regardless of whether it's a half-step or five octaves.
- - When the *Slide/Constant Rate* button is enabled, the last pulse of a slide enabled step is automatically set to legato- i.e. the gate is high from last pulse of the step until the beginning of the next sequence step, and the pitch slide happens during the last 16% of the step. (To frame that in little-silver-Japanese-technobox terms, each of its steps lasts 24 pulses, and the slide occurs during the last 4 pulses.)

67.4 Sequencer Step Controls

Each step consists of a big fader for setting the step CV, along with a gaggle of buttons, many featuring fun cryptic abbreviations! (We attempted to keep Octagon's overall size manageable) Here we'll dissect one sequence step and run down what all the buttons do.

CV Slider and LED- Sets the voltage sent to the main CV out for the step. The LED beneath indicates to show the current step is active.

Pulses buttons- Each sequence step can have up to eight gate/trigger pulses. The sequence step counts vertically through the number of pulses for that step; the pulse buttons flash red at each pulse count.

The gate signals behave differently depending upon the current *Gate* mode selection. The *Pulses* buttons will illuminate differently dependent on the current *Gate* mode, and the gate pulse steps may or may not send a voltage depending on their setting, but Octagon will also count up to the highest selected pulse number before advancing to the next step.

Gate mode- These buttons define the behavior of the gate voltages for the pulse steps.

- **Off**- Doesn't output any gate CV's, but lingers on the step for the duration of the current *Pulses* setting. All buttons up to the selected # of "rest" pulses are lit and flash as it counts up. When it reaches the last selected pulse #, it advances to the next step. The # of counts is selected by clicking the top button only. This is essentially a rest with definable length.
- **Single (Sngl)**- Outputs a single gate on pulse #1 then counts up to whatever the # of pulses setting is. In this mode, pulse #1 is always on (hence always green); you only need to click the top button to select the # of pulse counts (how long it rests after the pulse on 1). When it reaches the last selected pulse #, it advances to the next step.
- **Legato (Leg)**- Outputs a continuous gate lasting as long as the # of pulses setting then advances to the next step. The # of counts is chosen by clicking the top button only.

- **Repeat (Rept)**- Outputs a separate gate on each active pulse #. When it reaches the last selected pulse #, it advances to the next step. The # of counts is chosen by clicking the top button only. If you're not hearing individual notes sound, make sure the *Gate Time* knob setting is short enough. If the gate output is patched to an envelope generator, make sure the decay and release times are short enough to hear individual notes.
- **Defined (Def)**- The overall pulse count AND which steps send gates can both be set in any combination. Consecutively selected steps send separate gates (i.e., no legato).

When the *Defined* button is clicked, a pop-up menu opens where you'll select the number of active pulses and the corresponding number of pulse buttons will illuminate in amber. Any combination of the active pulses can be clicked to turn on gates. Active gate steps will illuminate in green, and gray pulse buttons don't do anything. The number of active pulses can be changed at any time by clicking on the *Defined* button and changing the number of pulses.

In the image to the left, the step will count up to six and gates will be sent for steps 1, 2, and 5.

Skip- When a step's skip button is on, Octagon jumps over it as it if wasn't there (in contrast to the *Off* gate mode, where the selected number of pulses will count but no gate voltages are output).

Slide- Adds a CV slide for the step - see *Slide knob* section for more info.

Division- Sometimes referred to as "ratcheting," this adjusts the pulse rate for the step. By default, this is set to the standard /1 rate, but can also be set to /2, /3, or /4 to play pulses more quickly. The *Division* button acts globally for all pulses of a step. Note: The Division function is disabled when using an external clock source (because external clock pulses happen once per sequence pulse or step and can't be "multiplied."

67.5 Output Jacks and Other Right-Side Controls

Play Trig Out- The *Play Trigger Out* jack outputs a 5V trigger spike any time play mode is initiated (from the *Start* button or via CV control). This can be useful for starting ganged multiple sequencers and other functions.

Clock Out- This outputs Octagon's internal clock signal for syncing other modules. If *External Clock* mode is selected, it echoes the incoming clock, thus turning Octagon into the world's most complex mult module.

Trigger Out jack- Outputs 5V trigger signals for active pulses.

Gate Out jack- Outputs 5V gate signals for active pulses.

CV Range buttons- Selects the ranges of voltage for sliders.

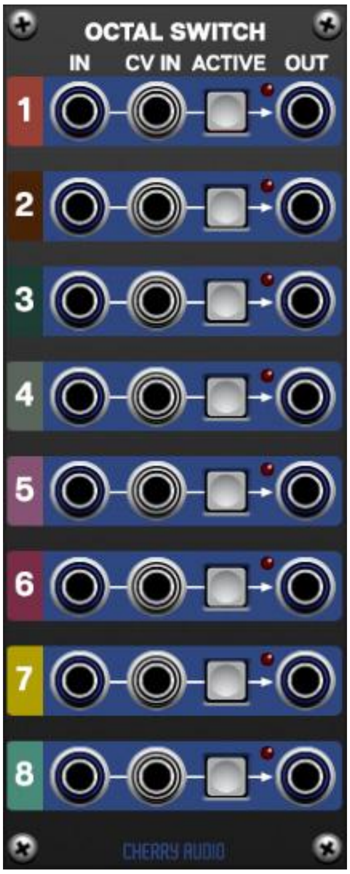
- **1V = 0 to +1V** (one-octave range)
- **2V = 0 to +2V** (two-octave range)
- **5V = 0 to +5V** (five-octave range)

Since Voltage Modular's pitch conforms to the 1V/octave standard, this means a 1V range equates to a range of one octave, a 2V range equates to two octaves, and a 5V range equates to five octaves. Octagon's pop-up tooltip displays will change to reflect *CV Range* selection.

Quantize- Enabling *Quantize* forces fader values to snap to 1/2 step note increments (i.e. 1/12 volt). Without this, it would be difficult to set note values to play in tune. Disabling *Output Quantize* turns off pitch "snap" and allows any value to be set - this is useful when the sequencer is being used to modulate non-pitched destinations, such as filter cutoff or amplitude. The pop-up tooltip displays will show note or decimal values dependent on the *Quantize* button setting.

CV Out jack- Outputs the slider CV for the current step.

68 Octal Switch



Octal Switch is a simple gate switch that allows CV or audio signals to be turned on and off. It doesn't mix signals or do anything fancy, but it makes switching up to eight signals super easy. Each switch can also be CV controlled.

68.1 Inputs, Outputs and Controls

Since there are eight identical instances, we'll go over one set of controls.

In- Input jack for the audio or control signals.

CV In- Accepts a gate (constantly on) CV signal to enable signal flow for as long as the CV is present. Any voltage > 2.5 enables signal flow.

Note that the button always takes priority over CV - if the button is on, then CV's will have no effect. If the button is off, CV will enable signal, and the LED will glow when CV is present.

Active button- Enable signal flow. These are latching buttons, i.e. they remain on when pressed. Press again to turn off.

Out- Signal output. Duh.

69 One To Eight Switch

The Cherry Audio One To Eight Switch module routes an audio or control input signal to eight individual output jacks. Signal is only passed from an output when its respective “step” is active. The outputs can be stepped through sequentially with a manual or CV trigger, or targeted individually via discrete control voltages.

Switches are used to re-route signals without having to unplug or re-patch any cables. As an example, the One To Eight Switch could be used to pass a clean audio signal from one output while sending another output to a distortion module and another to a delay unit. The fun starts when you begin experimenting with different ways to step through the outputs!

69.1 Inputs, Outputs and Controls

In jack- Input jack for the signal that will be routed to the eight outputs.

1-8 output jacks and LEDs- These eight jacks output the signal received at the input jack whenever their respective step is active. The small red LEDs give visual feedback of the active step.

Steps- Sets the number of steps that can be activated. When stepping through the outputs sequentially with either the manual or *Step Trigger* CV input, this sets the number of the last step before it will cycle back to step one.

Step Trigger jack- A 5V pulse or gate received at this jack will trigger the steps sequentially.

Step CV jack- CV input jack for switching between steps in any order. The control voltage range of 0V - 5V is evenly divided between the number of steps making it possible to target specific steps with discrete voltages.

Here are a couple examples of how the voltage is divided:

- If *Steps* is set to two, the 5V range is divided between the two steps. Step one is selected with voltage from 0V - 2.49V and step two is selected with 2.5V - 5V.
- If *Steps* is set to eight, the 5V range will be divided equally between the eight steps. Five divided by eight is 0.625 so, step one = 0V - 0.62V, step two = 0.63V - 1.24V, step three = 1.25V - 1.87V and so on.

If you don't happen to make music with a calculator next to you, we recommend just playing around until you find the step you're looking for!

Reset jack- A 5V pulse or gate received at this jack will immediately force the module back to step one. Note that resetting the module will be unnoticeable when using the *Step CV* input because the voltage received at its jack is constantly updating the active step.

Manual Step- Click this button to manually advance to the next sequential step.

70 Oscillator

The Cherry Audio "standard" oscillator is a full-featured classic analog-synthesis oscillator. It generates all standard synthesis waveforms and can be used as an audio source, or as a control voltage (CV) modulation source. Its waveform outputs are always “on”; you’ll need to use a mixer or amplifier (VCA) of some sort to start and stop its sound.

70.1 Inputs, Outputs, and Controls

Pitch CV jack- Accepts a CV input for pitch. Typically this would come from the PITCH jack in the IO Panel CV OUT section, or from a sequencer pitch CV out.

Frequency Mod attenuator, input jack- This is used for externally modulating the oscillator frequency. It's useful for adding vibrato with an LFO, siren noises, envelope-controlled pitch sweeps, etc.

Mod Type- The *Mod Type* button lets you select linear or exponential modulation. We'll give a couple of examples to clarify how they work:

- **Exponential**- For a given mod input voltage, the mod amount increases as frequency increases. For example, if the base frequency is 1000 Hz, and a bipolar wave is applied to the mod CV input, the cutoff frequency rises to 2000 Hz, and falls to 500 Hz. Because audio frequencies are inherently exponential in nature, the resulting cutoff frequency rises and falls exactly one octave.
- (This is why expo mod is generally used for oscillator mod - so that vibrato will rise and fall an equal amount above and below the pitch center.)
- **Linear**- For a given mod input voltage, the mod amount stays the same as frequency increases (hence the "linear" name). For example, if the base frequency is 1000 Hz, and a bipolar wave is applied to the mod CV input, the frequency rises to 1500 Hz, and fall to 500Hz. In other words, the frequency rises and falls by the same number of Hz in either direction.

  If the intent of the modulation is a vibrato or siren effect, linear mod is likely the wrong choice, because the audible effect will appear to be greater in one direction than the other - in this case, expo mod would be the best choice.

So when would we use linear mod? The advantage of linear mod is that it stays constant regardless of the base frequency, which makes it useful for audio-range modulation (i.e. mod frequencies 20 Hz and faster) when the mod source is to alter tone color. In use, you'll find that expo mod allows notes and scales to play in tune, whereas expo mod in the audio range allow neat ring mod-style sound effects, but doesn't usually allow properly pitched half-step scales.

Hard Sync- Force resets the start of the waveform to the beginning of its cycle. Most often used to create the "sync sweep" oscillator sounds made famous in The Cars' "Let's Go" (or Kraftwerk's "Neon Lights" and No Doubt's "Just A Girl"), by routing the output of a second oscillator to the *Hard Sync* input and sweeping the pitch of the first oscillator.

Hard Sync is also useful when creating drum and percussion sounds to ensure that the wave starts at the beginning of its cycle. Finally, hard water in your sink is a totally different issue that can be remedied with a water softener.

Range- Sets the basic pitch of the oscillator, displayed in traditional organ footage. *LO* will be beneath the audible range and allows the oscillator to be used as a mod source.

Frequency- Fine-tune control for pitch. This can be used to fatten up multi-oscillator patches by detuning a small amount, or for "building-in" a set interval. Its range is a smidge over a fifth, up or down.

Pulse Width- This sets the width or "duty-cycle" of the pulse wave. It has no effect on any other waveform. Its default setting of 50% outputs a perfect square wave, rich in delicious odd-order harmonics. Moving the knob left or right narrows its width as well as the thickness of sound until it almost disappears at its extremes, and we’ve included a nifty “faux-OLED” display to indicate the current pulse width.

PWM Amount attenuator and PWM Mod input jack- You may have noticed that moving the *Pulse Width* knob back and forth creates a nifty sound; instead of wearing our your mouse hand, the *PWM Mod* input can be used in conjunction with an LFO, envelope generator, or other mod source to continuously vary the pulse width. Best of all, the OLED display looks real cool swooping back and forth.

Waveform Output Jacks- These are output jacks for ramp, sawtooth, pulse, sine, and triangle waves. These can be used simultaneously, in any combination.

71 Oscilloscope

The Cherry Audio Oscilloscope is a handy utility module featuring two inputs, range, zoom, and a freeze button for monitoring audio and CV signals. Besides looking cool (who doesn't love looking at waveforms?), this is an extremely helpful tool for learning about modular synthesis or troubleshooting complex patches.

71.1 Inputs, Outputs, and Controls

Input A and B jacks- Patch CV and audio signals here to visually monitor their voltages. *Input A* displays a blue waveform, and *Input B* displays a yellow waveform.

Auto Trig- These buttons are used to "stabilize" the visualization of the signal and are triggered from *Input A* only. When neither of these buttons are engaged, the voltage received is displayed without a consistent trigger or "starting point" within the display window which can sometimes appear jumpy and unstable. By engaging the - button, the oscilloscope will always show voltage starting from 0V at the left hand side and decreasing, while the + button will show voltage increasing from 0V. Having a consistent trigger point for where the voltages are displayed on the oscilloscope creates a stable waveform when monitoring cyclical signals such as oscillators. Try patching a sine-wave oscillator to the input and switching between - , +, and neither to see the difference.

Freeze- Instantly freezes the display. The signal that was present at the moment the button was clicked will be displayed until the button is turned off again.

Display window- A classic analog-style oscilloscope display showing the signal received at the *Input* jack. The horizontal line in the center represents 0V. Positive voltage is displayed above the 0V line while negative voltage is displayed below.

Range- Adjusts the range of voltage shown within the display. Think of this as a vertical zoom.

Zoom- Adjusts the sample length of the display window. Think of this as a horizontal zoom.

72 Panner

The Cherry Audio Panner module is a static or voltage-controllable utility for panning an audio or CV signal between two outputs. Typically the outputs will be connected to a stereo mixer or the *Main Outs* in the I/O panel to pan an audio signal between the left and right speaker but the module is also useful for sending signals to two unrelated destinations. You could, for example, route the CV output from a sequencer to the mod inputs of a filter's cutoff and oscillator's pulse-width. The *Pan Position* can then be used to gradually route the CV sequence to either or both destinations.

72.1 Inputs, Outputs and Controls

Input jack- Receives the audio or CV signal that will be routed to the two outputs.

CV Mod jack- CV input for externally controlling the pan position.

CV Mod Amount- Scales the amplitude of the CV signal received at the *CV Mod* input jack. The + and - LEDs indicate when positive and negative voltage respectively is modulating the pan position.

Pan Position- Controls how much of the input signal is sent to each output. With the knob in it's center position, the input signal is sent at full amplitude to both outputs. Moving the knob to either side gradually decreases the amplitude of the opposite output.

Left and Right Outputs- Patch these to the left and right jacks of a stereo input for traditional panning. When used to route signals to destinations other than the left and right speaker, you can think of these as outputs "one" and "two" if it makes you feel better.

73 Percussion EG

The Percussion EG (Envelope Generator) module is an envelope generator with controls optimized for percussive sounds. It features a continuously variable exponential to logarithmic decay curve, click enable switch, attack peak hold knob, and accent and choke inputs making this the perfect go-to envelope for sculpting a percussion sound from scratch.

73.1 Inputs, Outputs, and Controls

Trig In jack- A 5V gate or trigger received at this jack will trigger the envelope. If a gate signal is used, the length is ignored as this is a "one shot" envelope and has no sustain stage.

Accent In jack- A 5V gate or trigger received at this jack triggers the same envelope shape only with a greater amplitude (set by the *Accent Leve*l knob). This can be used to create a second, "louder" version of the envelope that can be used to create accented drum patterns. The Drum Trigger Sequencer has dedicated *Acc (Accent)* output jacks that are perfect for triggering this. Note that if the envelope is modulating something other than an amplifier, the sound will not necessarily be louder. Instead, the accent will increase the modulation amount.

Click- Adds a distinct "click" at the beginning of the envelope to add presence to the beginning of a drum sound.

Choke- A 5V gate or trigger received at this jack will immediately force the envelope to 0V. This can be used to stop the ringing of a long drum or to "close" an open hi-hat sound.

Attack Hold- Adjusts how long the the envelope will stay at 5V before starting the *Decay* stage.

Accent Level- This knob scales the amplitude of the accent envelope by as much as 400%. The accent envelope is triggered via the *Accent In* jack.

Decay- Defines the length of time it takes for the envelope’s voltage to drop from 5V back to 0V. This can be set extremely short (making it perfect for adding a little “smack” to a sound), or as long as 2.5 seconds.

Decay Curve- Adjusts the shape, or curve, of the envelope's *Decay* stage from logarithmic when turned to the left, to linear at its center position, to exponential when turned to the right.

Env Out jack- This is the output jack for the envelope.

74 Phaser

The Cherry Audio Phaser module is a four-stage phaser audio effect featuring voltage-controlled rate and individual on/off switching for each stage. The stages of this module are made from four notch filters instead of typical all-pass filters creating a unique type of phaser effect.

74.1 Inputs, Outputs, and Controls

L(M) and R Input jacks- These are the mono or stereo audio input jacks. When using a mono input signal, patching it to the *L(Mono)* jack will feed the signal to both sides of the stereo effect. This is a true stereo effect unit with each side having its own phaser effect. The internal LFOs that modulate the frequency of the left and right notch filters are 180° out of phase with each other to create a sweeping motion from left to right.

Stages- These buttons engage each of the independent notch filters that make up the phaser effect. Each of the four notch filters modulates through a different range of frequencies with stage one being the lowest and stage four the highest.

Rate Mod jack and attenuator- CV input jack and attenuator for externally controlling the *Rate.*

Rate- Adjusts the rate, from .02Hz to 10Hz, at which the notch filters frequencies are modulated by the internal LFO.

Feedback- Adjusts how much of the filtered signal is sent back to the input of the effect.

L and R Output jacks- These are the module's stereo output jacks.

75 Plug-In Host

The Plug-In Host module lets you use almost any virtual instrument or effect plug-in "inside" of Voltage Modular and allows CV control of up to 12 plug-in parameters. It includes four audio ins and outs as well as MIDI I/O.

In this guide, we'll interchangeably refer to virtual instruments and effects plug-ins simply as "plug-ins" to save me some typing and you some reading.

75.1 The Big Disclaimer

We've tested Plug-In Host with many plug-ins, and in most cases, it performs fabulously well, but be aware that not all plug-ins will play nice with it.

- If individual plug-ins use a shell host, you may not be able to load them. The most common of these are Waves plugs, and they should work fine.
- Controls for some plug-ins won't appear in the *Parameter Select* menu.
- Because some developers use non-standard preset browsers, presets may not appear in the *Preset Select* window.
- If you're using Softube instruments or plug-ins on a Mac, the VST versions will crash (it's them, not us, baby), but AU versions work fine, even if you're using a DAW that doesn't inherently support AU plugs (because you're hosting them in Plug-In Host, not the DAW itself).
- Some plug-ins may crash our system, but we're continually working to improve third-party plug-in support.

75.2 Top Buttons

Active- This is the "on/off" switch. Green is on, gray is off. In off mode, virtual instruments won't make any sound, and plug-ins will pass signals unaffected.

Select Plug-In- Use this to choose an instrument or effect. When clicked, you'll see a sub-menu with instrument and effects types *(Audio Units, VST, VST3* - depending on whether you're using Windows or OS X). Choosing the desired type will automatically navigate to the appropriate folder where plug-ins can be selected.

View Editor- Opens the editor window for the selected instrument or plug-in. Edit windows can be closed by clicking the *View Editor* button again or by clicking the *X* in the top-left corner of the plug.

Latency- Just like the audio latency setting in DAWs, lower settings result in quicker processing and snappier performance, but require more computer processing power. As with audio hardware, the general rule is to set this as low as possible until you start hearing crackling noises, but the default setting of 128 samples should work well for most applications.

Preset Select- The left/right arrow buttons can be used to cycle through a plug-in's presets. Clicking in the black area to the right of the selection arrows also opens a pop-up menu displaying all presets for the plug-in.

Please keep in mind that presets won't be visible if the plug-in developer uses a proprietary preset browser implementation; presets will only be displayed if the plug-in uses the standard Windows or OS X plug-in preset browser.

75.3 Inputs

MIDI- You'll typically play virtual instruments by patching the I/O Panel *MIDI From Host* MIDI output to the Plug-In Host *MIDI* input jack.

If you'd like to play a virtual instrument using CV's from a sequencer or other source within Voltage, use the **CV To MIDI** module (or the **CV To MIDI CC Converter** for transmitting MIDI CC data).

Audio Inputs- Use these for routing audio into an effects plug-in. Usually the *1L* and *1R* inputs are all you'll need, but we've provided a second set of inputs as well.

75.4 Outputs

MIDI- A MIDI out for plug-ins that use it, typically arpeggiators and some virtual instruments with on-screen keyboards.

Audio Outputs- The plug-in's audio outs. A second set of outputs is included for multi-out instruments and plug-ins.

75.5 Parameter Control Slots

We've arrived at the fun part! Not only do these allow 12 plug-in parameters to be "remote-controlled," but each slot includes a CV in jack and bipolar attenuator. This allows CV control of any parameter of any plug-in, which is kind of awesome.

CV Amount knob - Attenuates and/or inverts incoming control voltages. Center position is zero. Turning the knob right applies positive voltage, turning the knob to the left applies negative voltage.

CV In jack - Patch incoming CV's to this jack.

Parameter Amount - Sets the initial amount for mapped plug-in parameters. Once mapped, it acts as a "remote control" for the selected parameter, but keep in mind that that it only communicates one way, i.e. moving the control in the plug-in's interface will **not** move the *Parameter Amount* knob, but moving the *Parameter Amount* knob **will** move the control in the plug-in interface.

Parameter Select button - Click this to assign plug-in parameters to control slots. A single mouse click displays all of the plug-in's parameters; clicking on one assigns it. Parameters can also be assigned using Plug-In Host's *Learn* function.

To learn a control, select *Learn* from the pop-up menu (if the plug-in editor window is currently hidden, the menu will say *View Editor and Learn*; this initiates learn mode and opens the plug-in editor window). The *Parameter Select* button will say *LEARNING* in red; simply move the control you'd like to assign in the plug-in interface to instantly assign it to the current control slot. Learn mode will automatically disengage.

***Learning vs. direct assignment**: Generally speaking, for plug-ins with a just a few parameters, it's fastest to assign controls by simply clicking the Parameter Select button and clicking the desired parameter, but for plug-ins with dozens of parameters, Learn mode greatly simplifies parameter assignment.*

To clear control assignments and intialize a parameter control slot, click the *Parameter Select* and choose *None*.

76 Poly Amplifier

The Cherry Audio Poly Amplifier is a voltage-controlled amplifier (VCA) for polyphonic audio or CV signals. If you need a refresher on how VCAs work, check out the documentation for the standard Amplifier module.

Below is a simple polyphonic patch where the Poly Amplifier is modulated by a Poly Envelope Generator to control the volume of a Poly Oscillator.

76.1 Inputs, Outputs, and Controls

Input jack- Use this jack to patch in "poly" audio or control signals to be affected by the *Poly* and/or *Mono CV Mod* jacks.

Gain- Adds up to 5 volts of gain. This works *in addition* to incoming *CV Mod* jack voltages. It's also useful for manually "opening" the amplifier.

Poly CV Mod jack and attenuator- This is a "poly" CV input and bipolar attenuator for individually controlling the amplitude of the signals received at the *Input* jack.

Mono CV Mod jack and attenuator- Standard "mono" CV input and bipolar attenuator for simultaneously controlling the amplitude of all signals received at the *Input* jack.

Response - Lin/Expo- These select the "curve" of the amplifier's response as the input CV rises from 0 to 5V. *Lin* or linear response curve is equally proportional across the voltage input range, where as an *Expo* or exponential curve is closer to how the human ear perceives volume. With that in mind, you'll likely want to use the *Lin* setting for modulation or control voltage situations, and use the *Expo* setting when an envelope generator is used to control an audio signal with the amplifier.

Output jacks- The *Output* jack carries the CV-modified version of the polyphonic input signal. The *Inv Out* jack outputs an inverted version of the same "poly" signal. Be careful not to use both at the same level as they can cancel the output entirely.

77 Poly CV Converter

The Poly CV Converter converts polyphonic MIDI input to four individual CV/gate/velocity outs for creating versatile polyphonic patches. Featuring selectable number of voices, MIDI channel, and outputs for converting MIDI pitch bend, mod wheel, aftertouch and volume messages to CV signals. If more than four voices are needed, multiple instances can be chained together using the *Overflow* MIDI jack to expand the polyphony of a patch.

Voltage Modular has a number of “Poly” modules that can simplify making polyphonic patches, but doing it the “old-school” way can give you more versatility by being able to customize each voice individually. It’s possible, for example, to use a different filter envelope on each voice so that every note in a chord sounds a little different from one another.

77.1 MIDI Section

MIDI Channel- Selects which MIDI channel the module will respond to. Set this to the same channel as your controller or external MIDI device. When set to “All,” MIDI notes from all sixteen channels will be received.

MIDI In jack- Input jack for receiving MIDI notes from an external controller or MIDI device. This is typically patched to the *From Host* MIDI jack in the I/O panel or the *MIDI Out* jack of a MIDI Input module.

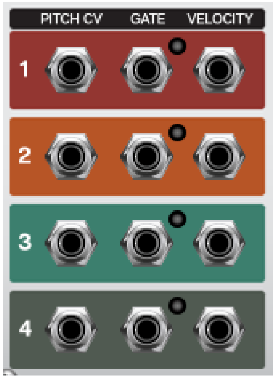
Over Flow jack- Once the module is using all of its allocated voices, additional MIDI notes received at the *MIDI In* jack will be passed thru to this jack. Connecting this to the *MIDI In* jack of a second instance of this module makes it possible to increase the number voices to more than four.

77.2 Polyphony

Number of Voices- These buttons select how many voices will be used in the polyphonic patch. Set this to the maximum number of notes you wish to play at the same time.

If only three voices of polyphony are needed, setting this to three eliminates the need to set up an oscillator, envelope, amplifier, etc. for the fourth voice which can save space and CPU.

77.3 Pitch, Gate, and Velocity CVs



Each time a MIDI note is received, the module evaluates which voices are already being used and allocates the note to one of the four available voices. Each voice has the same three output jacks.

Pitch CV jack- MIDI note-number messages are converted to CVs for controlling the pitch of an oscillator via their *Keyb CV* or *Pitch CV* jacks.

Gate jack- MIDI note-on/off messages are converted to gate CVs which are typically patched to the *Gate In* jack of an envelope. The red LED illuminates to show when each voice is in use.

Velocity jack- MIDI note-velocity messages are converted to CVs which can be used for a number of things, but often will be patched to the *CV In* jack of an Amplifier module to scale an envelope that is controlling the volume of a sound.

77.4 Mod/Performance Controls

Pitch Bend jack- MIDI pitch-bend messages are converted to CVs ranging from -5V to +5V. 0V is output when the pitch wheel is in its center, or neutral, position. This will often be patched to the *Bend CV In* jack of a Bend Limiter module who's output is patched to the *Keyb CV* or *Pitch CV* jack of an oscillator.

Mod Whl jack- MIDI mod wheel (CC#1) messages are converted to CVs ranging from 0V when down to +5V when up.

Aftertouch jack- MIDI channel-pressure messages (how hard the key is pressed while sustaining) are converted to CVs ranging from 0V to +5V. Note that not all MIDI controllers send aftertouch messages.

Volume jack- MIDI volume (CC#7) messages are converted to CVs ranging from 0V to +5V.

77.5 Let's look at an example

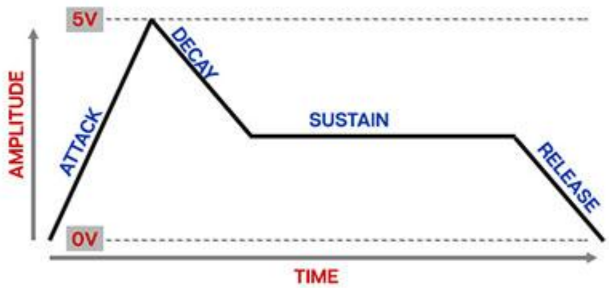
This is a velocity-sensitive patch with four voices of polyphony. You can see that MIDI is received from the *From Host* MIDI output in the I/O Panel. The Poly CV Converter converts the note messages to four separate pitch, gate, and velocity CV outputs. The *Pitch CV* jack for each voice is patched to the *Keyb CV* inputs of the four oscillators. The *Gate* jacks are patched to the *Gate In* jack of four envelopes which are patched to the *CV In* jack of the Amplifier next to them to control the volume of the oscillators. The *Velocity* jacks from the Poly CV Converter are patched to the *CV In* jack of a second Amplifier which scales the overall level of each voice. Then finally the output of each secondary Amplifier is fed to a Stereo Mixer which is patched to the Main Outs.

78 Poly EG Station

Poly EG Station features eight fully independent ADSR envelope generators. Though it's intended for use with the Cherry Audio *FM Station* oscillator module, it can be used with any modules in Voltage. It packs a lot of envelopes into a relatively compact footprint and features velocity CV inputs allowing control of envelope intensity. Other polyphonic capabilities and jacks, it's essentially identical to the "mono" EG Station module.

78.1 How ADSR Envelope Generators Work

If you're not familiar with the operation of standard envelope generators, here's how they work: when a gate voltage is sent to one of the *Gate* jacks, the envelope generator outputs a voltage that changes dynamically according to its four stage settings.



The *Attack* stage defines how long it takes for the output voltage to rise from 0 to 5 volts. Once the attack stage reaches 5V, it moves to the *Decay* phase, which defines how long it takes to fall from 5V to the setting of the *Sustain* phase. Unlike the *Attack*, *Decay*, and *Release* phases, each of which define a time, *Sustain* sets the held voltage level following the *Attack* and *Decay* phases - this usually equates to the envelope output level while holding down a key on a keyboard controller. Finally, the *Release* knob defines the the length of time it takes for the voltage to fall back to 0V when the gate input voltage is removed (typically when you let go of a key on a keyboard controller).

78.2 Inputs, Outputs, and Controls

Control Bars and CV Indicator Bars

EG Station's main controls are a bit of a departure from other Cherry Audio modules. Instead of standard knobs or sliders, its main controls are colored illuminated bars. These work just like the slider controls in other modules. To change values, grab the control bar at its top edge and move it or click anywhere in the control travel to jump to the setting (this is helpful when the control is set to minimum; you won't have to precisely click the tiny visible region of the bar). Hovering the mouse on a control bar causes it to light up. The controls bars behave the same as any standard Voltage slider control- right-clicking them allows all standard operations including Perform and MIDI controller assignments.

The control bars are color-coded to match the *FM Station* module's four operator sections, but otherwise each of the envelopes are functionally identical.

P-Gate jacks (normalled)- *P-Gate* is short for poly gate. This is where you'll patch gate voltages to initiate an envelope generator cycle. The connection from the IO Panel Poly Sources *Poly Gate* CV out is semi-normalled to all eight of EG Station's *P-Gate* inputs - in other words, even though there are no visible cables, it's automatically connected "under the hood." We think you'll find this makes using Poly EG Station really fast and easy - it also dramatically reduces cable clutter.

If a cable or a cable bus is connected to a *P-Gate* input jack, the IO Panel connection is overridden for that jack (i.e., disconnected). This is useful if you don't want to use the IO Panel *Gate* jack output when using an alternate poly gate source.

> **Can I use a "trigger" to trigger an envelope generator?** It would seem logical, but the answer is, "sometimes, but generally, no." First let's clarify the difference between a gate signal and a trigger signal:
>
> - A **gate** is a *constant* voltage. If you're playing a keyboard, it remains high (i.e. +5V) as long as the key is held down.
> - A **trigger** is a *rapid spike* of +5V. It's useful for a number of things (like turning stuff on and off, or triggering "one-shot" drum sounds or modules).

With this in mind, Poly EG Station needs to see a constant gate voltage to move through the *Attack* and *Decay* phases and hold during the *Sustain* phase. Removing the gate voltage following the *Sustain* phase tells it to move to the *Release* stage. Conversely, using a trigger signal will cause the envelope generator to *immediately* jump to the *Release* phase.

P-Vel jacks (normalled)- *P-Vel* is short for poly velocity. These poly CV in jacks allow the overall voltage output of the envelope to be controller via CV. Like the *P-Gate* jacks, all eight *P-Vel* CV inputs are semi-normalled to the IO Panel *Poly Vel* CV outputs, so they don't need to be patched to function. The IO panel connections can be individually overridden by plugging poly cables into them. Poly velocity CV intensity is controlled by the *Vel* control bar.

Poly Envelope Outs - The envelope voltage outputs have an envelope icon beneath them. Their output voltage ranges from 0V to +5V.

"A" (Attack) control bar- Defines the length of time for voltage to rise from 0V to 5V when the gate voltage is applied.

"D" (Decay) control bar- Defines the length of time for voltage to fall from the *Attack* stage 5V peak to *Sustain* stage setting.

"S" (Sustain) control bar- Sets the held voltage level following *Attack* and *Decay* phases.

"R" (Release) control bar- Defines the length of time for voltage to fall from *Sustain* level to 0V when gate is released.

Velocity control bar- This an attenuator controlling the intensity of CV from the *Vel poly* CV jack. Moving the bar up from middle position adds positive CV input.

This *Velocity* bar also displays an unfilled rectangle that displays incoming the sum total of incoming velocity CV's in real-time.

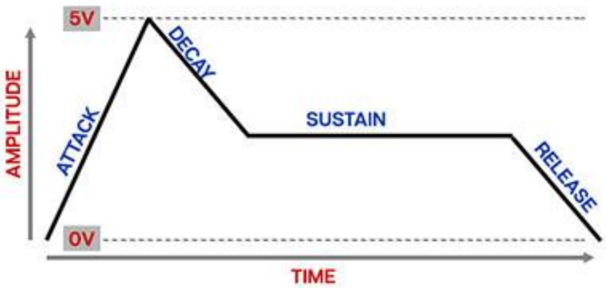
The *A*, *D*, *S*, and *R* control bars illuminate when the mouse is hovering over them.

79 Poly EG-20

The Poly EG-20 modules is a poly version of the envelope generators found in the classic Korg MS-20 monosynth. It's essentially the same as the EG-20 module, but in addition to poly capabilities, it adds sustain pedal inputs for each envelope generator. In addition to their particular envelope curves and times, these include a couple of unique features.

If you're not familiar with the operation of envelope generators, here's an overview of a standard ADSR-style envelope generator:

When a gate voltage is sent to the *Gate In* jack, the envelope generator outputs a voltage that changes dynamically according to the settings of its four stages.



The *Attack* stage defines how long it takes for the output voltage to rise from 0 to 5 volts. Once the attack stage reaches 5V, it moves to the *Decay* phase, which defines how long it takes to fall from 5V to the setting of the *Sustain* phase. Unlike the *Attack*, *Decay*, and *Release* phases, each of which define a time, *Sustain* simply sets the held voltage level following the *Attack* and *Decay* phases - this usually equates to the envelope output level while holding down a key on a keyboard controller. Finally, the *Release* knob defines the the length of time it takes for the voltage to fall back to 0V when the gate input voltage is removed (typically when you let go of a key on a keyboard controller).

Now that you're an ADSR envelope expert, let's go over the specifics of the EG-20 envelopes.

79.1 Inputs, Outputs, and Controls

EG-20 is a dual envelope generator; each envelope generator operates completely independently. Not only do they have different controls, their timing constants are different as well. **We'll refer to the left side envelope as EG 1 and the right side one as EG 2.**

EG1 / EG 2 Poly Gate In jacks- This is where you'll patch gate voltages to initiate the envelope generator cycle. Most often this will come from the IO Panel *Gate* output. The standard gate voltage for Voltage Modular (and most hardware analog synths) is +5V, but EG-20 responds to gate voltages as low as +2.5V.

> **Can I use a "trigger" to trigger an envelope generator?** It would seem logical, but the answer is, "sometimes, but generally, no." First let's clarify the difference between a gate signal and a trigger signal:

Most standard envelope generators need to see a constant gate voltage to move through the *Attack* and *Decay* phases and hold during the *Sustain* phase. Removing the gate voltage following the *Sustain* phase tells it to move to the *Release* stage. With all that in mind, using a trigger signal will typically cause the envelope generator to *immediately* jump to the *Release* phase. However, EG-20's EG 2 can move through multiple stages using by using the *Hold Time* knob - more on this below.

EG 1 / Sust Pedal In jack- Use this to patch a cable from the IO Panel CV Sources *Sus* CV out (or any other 5V CV source); this holds the envelope at 5V following the attack phase.

EG 1 / Delay Time- Delays the onset of the EG1's attack phase by up to about 10.5 seconds when gate is high.

EG 1 / Attack Time - Defines the length of time for voltage to rise from 0V to 5V when the gate voltage is applied. EG1 max attack time is 30 seconds (!).

EG 1 / Release Time- Defines the length of time for voltage to fall from 5V to 0V when gate/key is released.

EG 1 / Sustain switch- When turned on, following the attack stage, this holds the output voltage at 5V until the key is released.

EG 2 / Sust Pedal In jack- Use this to patch a cable from the IO Panel CV Sources *Sus* CV out (or any other 5V CV source); this holds the envelope the sustain stage following the decay phase.

EG 2 / Hold Time- Turning this up holds the gate high wzhen a key is momentarily struck for up to 22 seconds. This allows EG 2 to run through its stages without the need to hold a key for the duration, and also allows very brief gates to be used (i.e. trigger signals).

EG 2 / Attack Time - Defines the length of time for voltage to rise from 0V to 5V when the gate voltage is applied. EG1 max attack time is 14 seconds.

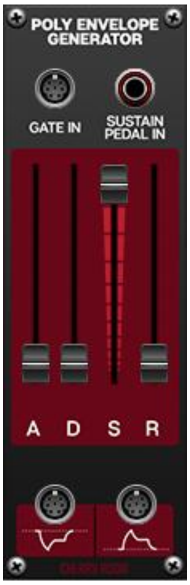
EG 2 / Decay Time- Defines the length of time for voltage to fall from 5V to sustain level

EG 2 / Sustain Level- Sets the held voltage level of held notes following the decay stage.

EG 2 / Release Time- Defines the length of time for voltage to fall from sustain level to 0V when gate is released.

Poly Env Out (top) / Poly Inv Out- These are the envelope voltage outputs. The *Out* voltage ranges from 0V to +5V. The *Inv Out* jack is a bit unusual - instead of outputting voltage from 0 to -5V, its output ranges from +5V to 0V. This mimics the functionality of the original MS-20 envelopes.

80 Poly Envelope Generator



Cherry Audio's Poly Envelope Generator is a polyphonic ADSR with sustain control that drastically simplifies creating polyphonic patches. It takes a polyphonic gate signal as its input and generates control voltage envelopes for each individual note, or voice. If you are unfamiliar with envelopes in general, or need a quick refresher, check out the documentation for the "standard" Envelope Generator which goes over ADSR envelopes in detail.

Below is an example of a simple polyphonic patch with sustain control that uses the Poly Envelope Generator to control the amplitude of a Poly Oscillator module.

80.1 Inputs, Outputs, and Controls

Gate In jack- Patch polyphonic gate signals here to trigger the envelope. Typically this will be connected to the *Poly Gate* jack in the I/O Panel.

Sustain Pedal In jack- A +5V gate signal received at this jack holds the envelope at its sustain level. Typically this is patched to the *Sus* (sustain) jack in the CV Outs section of the I/O Panel which converts MIDI sustain pedal messages (CC#64) from a keyboard controller to a +5V gate signal.

"A" (Attack) slider- Defines the length of time for voltage to rise from 0V to 5V when the gate voltage is applied.

"D" (Decay) slider- Defines the length of time for voltage to fall from the *Attack* stage 5V peak to *Sustain* stage setting.

"S" (Sustain) slider- Sets the held voltage level following *Attack* and *Decay* phases.

"R" (Release) slider- Defines the length of time for voltage to fall from *Sustain* level to 0V when gate is released.

Env Out and Env Out Inv- These are the envelope voltage outputs. The *Env Out* voltage ranges from 0V to +5V, whereas the *Env Out Inv* jack is an inverted version, with output ranging from 0V to -5V.

81 Poly Filter

The Cherry Audio Poly Filter is a full-featured polyphonic synthesis filter featuring lowpass, bandpass, and highpass outputs, 12- and 24-db per octave slopes, with "poly" and "mono" CV modulation inputs. If you are unfamiliar with filters, check out the documentation for the "standard" Filter module.

The image below shows a simple polyphonic patch using several poly modules. The Poly Filter is modulated by a Poly Envelope Generator which results in each individual note, or voice, having its own dedicated filter envelope.

81.1 Inputs, Outputs, and Controls

Audio In jack- Patch "poly" audio signals in here.

1V/Oct jack- This is a cutoff frequency modulation input intended to be used with polyphonic keyboard CV inputs. It allows the cutoff frequency to follow or "track" notes played so that the relative brightness of notes follows note pitch. This will typically be patched to the *Poly Pitch* jack in I/O Panel.

Keyb Tracking- This is an attenuation control for the signal received at the *1V/Oct* jack. At 100%, keyboard CVs are tracked at 1V per octave.

Poly CV Mod jack and attenuator- Polyphonic modulation input and bipolar attenuator for controlling the cutoff frequency of each individual note played. This is useful with the Poly Envelope Generator, for example, to create individual filter envelopes for each voice as shown in the image above.

Mono CV Mod jack and attenuator- Standard "mono" CV input and bipolar attenuator for simultaneously controlling the filter cutoff of all voices with one CV signal.

Cutoff- Sets the frequency where attenuation begins. Attenuation will be above or below this frequency (or both) depending on which output is currently used. Also something I frequently hear at the bar, as in "you're cut off, pal!"

Resonance- Emphasizes sound energy at and around the current cutoff frequency by adding feedback from the filter's output back to its input. At lower settings, this can be used to create mild resonances such as those heard in acoustic instruments. At more extreme settings, resonance can create a pure sine wave at its own frequency (variable via the *Cutoff* knob). Be careful with this knob as it can get loud at extreme settings.

Slope- The nature of how a filter works is such that its affect on frequencies "falls off" above or below the cutoff frequency. Slope adjusts the "steepness" of this slope. A 12db per/octave filter has a shallower slope, giving it a clearer and brighter character, whereas a 24db per/octave filter's steeper slope gives it a tighter and darker tone (as well as more pronounced character with the resonance knob turned up).

Lowpass, Bandpass, and Highpass Output Jacks-

These are the poly output jacks for lowpass, bandpass, and highpass modes, respectively. The icons visually represent the effect each has on incoming signals if the signal were to be viewed in a spectrum analyzer. These can be used simultaneously, in any combination. Combining the outputs with a poly mixer can result in interesting curves.

82 Poly FM Station

Poly FM Station is a digital oscillator inspired by the Yamaha DX/TX-series synths. It is mostly identical to the "mono" FM Station module.

Like the classic Yamaha synths of the 80s, it uses frequency-modulation (FM) synthesis to create tones. In their day, they were giant step forward for and delivered an entirely new palette of sounds. Unfortunately, the original Yamaha synths gained a reputation for being difficult to program and as a result, most users simply used the factory presets.

Though FM synthesis is inherently less intuitive than standard subtractive analog synthesis, most of the difficulty in programming the original Yamaha synths had more to do with the minimal user interface than the actual synthesis method itself. In order to cut costs, almost all synth manufacturers of the era were switching from knob-per-function interfaces (think Minimoog/Prophet-5) to interfaces where a single slider or up/down buttons in conjunction with a small alphanumeric LCD was used for sound programming. This made creating sounds tedious at best, and given their large number of parameters, next to impossible on the Yamaha FM synths.

With this in mind, we designed Poly FM Station to be as easy to understand and program as possible - all of its main parameters are indicated by colored bars, some of which change colors to clarify their current function. In plain English, Poly FM Station is really fun and easy to use. We had a blast developing it and playing it - we hope you enjoy it as much as we do! Because it's basically identical to the mono version of the FM Station module, we'll generically refer to it as simply as "FM Station."

One more note: You'll likely need a fairly large number of envelope generators to optimally use the mono or poly FM Station modules. We contemplated building envelopes directly into the module, but this would've necessitated either hiding them behind tabs or pop-ups, or making a really large module, neither of which sits well with our "make all controls visible and grabbable at once" design philosophy. Instead, we created the *EG Station* module in mono and poly versions. Intended to sit directly beneath FM station, its eight ADSR envelope generators should be plenty to cover any EG mod needs when using FM Station. That said, the *EG Station* modules can of course be used in conjunction with any other Voltage modules.

82.1 FM Synthesis Theory Nerdfest

Here we'll cover the basics of how FM synthesis works. You don't need to read this section to use FM Station, but it will greatly help your understanding, and we promise, no math equations, because as Butthead once said, "If I wanted to do math I'd go to thgool, huh-huh."

The underlying concept of FM synthesis is much like using the wavering output of a sine wave low-frequency oscillator (LFO) to modulate the pitch of an audio oscillator for vibrato. The LFO's rate is below the threshold of hearing (i.e. less than 20Hz), so you're able to clearly hear its up and down effect on the audio oscillator. In the patch below, the oscillator on the left is set to *Lo* range, and its sine wave output is routed to the second oscillator's *Freq Mod* input, creating a basic sine wave with vibrato.

The patch shown below is exactly the same, but the modulating oscillator's *Range* control has been changed to the *32*' setting, aka audio-rate modulation. In other words, its effect on the audio oscillator is no longer heard as vibrato- instead it changes the tonal color of the basic sine wave by creating additional frequencies known as "sidebands." The character of these sidebands can be altered considerably by manipulating the second oscillator's frequency modulation depth as well as changing the modulation oscillator's frequency. This modulating-one-oscillator-with-another-oscillator in various combinations is the basis of FM synthesis.

You may have also noticed the *Modulator* and *Carrier* dymo labels in the example patch. In FM synthesis, *carrier* refers to the oscillators used as audio sources; *modulator* refers to oscillators used to modulate the frequency of carrier oscillators - their audio is not directly heard.

82.1.1 Mod Type (aka, "I always wondered what that button was for.")

When setting up standard mod routings (i.e. vibrato, alternating pitches, etc.) a lot of us have likely clicked the *Mod Type* button both ways, not heard any significant effect on sound and moved along. But **in the case of FM synthesis, linear mod is of paramount importance for creating musical, tonal sounds.**

- **Exponential Mod**- For a given mod input voltage, the mod amount increases as frequency increases. For example, if the base frequency is 1000 Hz, and a bipolar wave is applied to the mod CV input, the cutoff frequency rises to 2000 Hz, and falls to 500 Hz. Because audio frequencies are inherently exponential in nature, the resulting frequency rises and falls exactly one octave. This is desirable for most standard sub-audio frequency modulation, i.e. vibrato, sirens, etc., because the audio rises and falls an equal amount above and below the median pitch.
- **Linear Mod**- For a given mod input voltage, the **mod amount stays the same as frequency increases** (hence the "linear" name). For example, if the base cutoff frequency is 1000 Hz, and a bipolar wave is applied to the mod CV input, the frequency rises to 1500 Hz, and fall to 500Hz. In other words, the frequency rises and falls by the same number of Hz in either direction.
-
- In the case of our vibrato or siren effect example, linear mod works poorly, because pitch will appear lopsided - for example, it would go down an octave but only rise a fifth.

To state it plainly, expo mod causes notes to go way out of tune as the mod amount is increase, whereas linear mod stays in tune and remains musical. (To be fair, audio-rate expo mod is great for creating laser sounds.) Try clicking the *Mod Type* switch in the demo patch above while adjusting *Freq Mod depth* settings and you'll instantly hear the difference.

By the way, besides being super basic, our two Voltage oscillators example doesn't really make for the best FM synth emulation, because the standard Voltage oscillator has some randomness built into its tuning and waves. This is desirable for accurately emulating an analog synthesizer, but not ideal for FM synthesis where it's best to have absolutely perfect waveforms and tuning. This is why you almost never see hardware analog FM synth oscillators - if everything isn't calibrated exactly perfect, it won't sound right or play in tune across a keyboard.

82.1.2 Algorithms and Operators

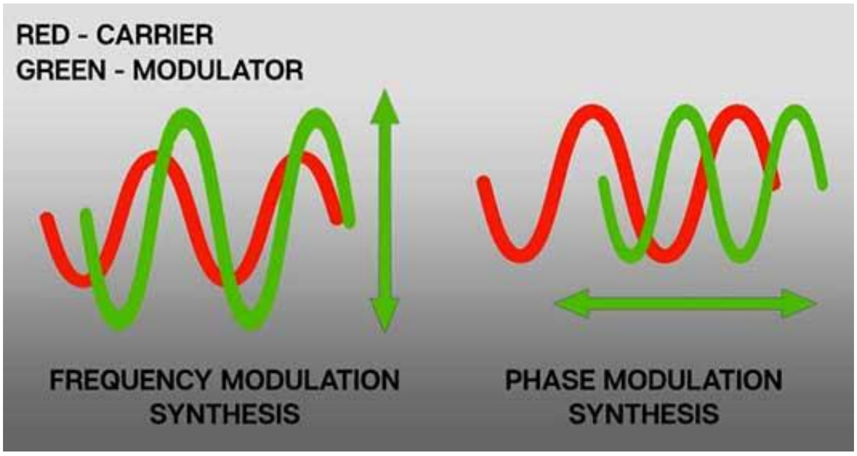
The preceding example patch illustrated the most basic FM synthesis configuration: one oscillator modulating another, aka one modulator modifying one carrier. Lots of great sounds can be realized this way, but it's roughly akin to an analog synth with a single oscillator. In order to expand the FM synthesis tonal palette, multiple modulator and carriers are configured in various pre-patched combinations to create more complex tones. These pre-patched combinations are known as **algorithms**. The carrier and modulator oscillators contained therein are generically referred to as "operators," and often abbreviated to "op."

The original Yamaha DX and TX synths were either "four-op" or "six-op." In other words, they included a total of four or six oscillators which could be configured as various combinations of carriers or modulators, depending on the currently selected algorithm. Like FM Station, the algorithms were graphically displayed on the top panel of the instrument for reference (these are actually selection buttons in FM Station, but we'll talk about that later on).

82.1.3 Frequency Modulation vs. Phase Modulation

The story goes that Yamaha licensed the use of the FM synthesis technology developed by John Chowning at Stanford University (FYI, the patent expired in the 90s). But oddly, Yamaha DX and TX synths don't use FM synthesis at all - they actually use the closely related *phase modulatio*n synthesis instead. What's the difference, why would they do this, and what does it mean to you? First, your bad-at-math help doc writer will attempt to explain the difference.

When an audio signal is frequency modulated, it can be thought of moving the wave up and down. Phase modulation can be thought of as rapidly moving the audio signal wave from side to side, thus changing the *phase* relationship between the two waves. Because both waves are constantly moving up and down, the effect is roughly the same on the resulting output wave.



There is one interesting quirk of phase modulation that's good be aware of - because the *Freq/Ratio* CV inputs affect phase, pitch changes are only audible *when the mod source is moving.* In other words, a constant DC voltage patched to a *Freq/Ratio* CV input (such as a keyboard pitch CV) won't affect pitch. Similarly, modulation waves with "straight" angles (such as sawtooth or triangle waves) will essentially sound the same as square wave mod, because their rate of change remians the same through their cycle, as opposed to a curve that's constantly changing - this is why all of FM Station's waveforms are curved.

A good analogy to understand phase mod is the doppler effect. When a car drives by honking its horn, the pitch of the horn changes as the car travels by, but if the car stops moving at any distance, the pitch of the horn remains constant. If you've ever messed with the time knob on an analog delay, they behave in the same way - pitch change is only audible while adjusting the delay time, and "catches up" as soon as you stop moving the knob. This is because a delay is performing exactly the same function - it's changing the phase (i.e. distance) of two copies of the same audio material.

Note that this only applies to the *Freq/Ratio* CV inputs; the *Fine* control bar and CV inputs are "standard" pitch CV ins (i.e. not phase mod) as is the master *Freq Mod* CV in at the far left of the panel.

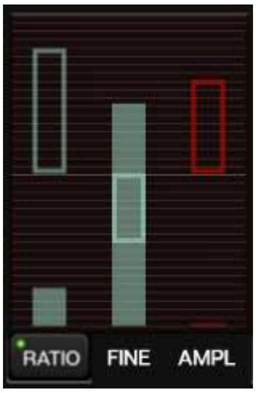
Why did Yamaha choose phase modulation over frequency modulation? As it turns out, it takes less computational horsepower to perform phase modulation synthesis. This was an important consideration in light of the relatively limited computer horsepower available in the 80s. The other reason is that FM synthesis makes use of feedback loops for further tone colors, and with pure FM, these loops cause oscillators to go badly out of tune - with PM, tuning isn't an issue (we'll discuss these feedback loops later on). FM Station Basic Architecture

As discussed in the previous section, an "operator" is FM synth-speak for a single oscillator. FM synths generally include "pre-wired" combinations of oscillators, configured as modulators and carriers, known as algorithms. FM Station features four oscillators, making it a "four-op" FM oscillator. It includes eight selectable algorithms. Its operators each include eight selectable waveforms, and the fourth operator includes an adjustable feedback loop.

Because FM Station includes separate outputs and modulation inputs for all four operators, it can be used "a la carte," that is, if algorithm 8 is selected, all operators and mod routings can be manually patched for unlimited custom algorithm and mod routing, including multiple instances of FM Station for 8-op, 12-op, stereo or quad output routing, etc. etc.... it's Voltage Modular, what'd you expect?!?

82.2 Inputs, Outputs, and Controls

82.2.1 Control Bars and CV Indicator Bars



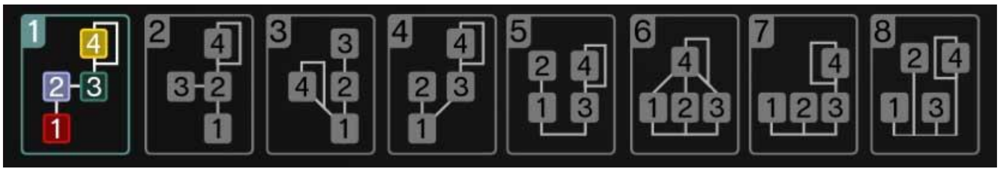
FM Station's main controls are a bit of a departure from other Cherry Audio modules. Instead of standard knobs or sliders, its main controls are colored illuminated bars. These work just like the slider controls in other modules. To change values, grab the control bar at its top edge and move it or click anywhere in the control travel to jump to the setting (this is helpful when the control is set to minimum; you won't have to precisely click the tiny visible region of the bar). Hovering the mouse on a control bar causes it to light up. The controls bars behave the same as any standard Voltage slider control- right-clicking them allows all standard operations including *Perform* and MIDI controller assignments.

You'll also notice the unfilled rectangular CV input indicators bars. These act as meters and display the sum of incoming poly mod CV levels in real-time. Besides being great eye candy, they're useful for quickly determining which carrier operators are making sound while under CV control.

Poly Pitch CV jack- Accepts a poly CV input for pitch. Typically this would come from the *Poly Pitch* jack in the IO Panel *Poly Sources* section.

Range- Sets the basic pitch of the oscillators, displayed in traditional organ footage. Unlike most other oscillators in Voltage Modular, there are only three range settings. This is because the *Freq/Ratio* controls offer a huge range of initial pitch settings, and the *Range* buttons are mainly intended for quick transposition.

Frequency Mod input jack and bipolar attenuator- Used for globally (i.e. all oscillators at once) modulating the base frequency. It's useful for adding vibrato with an LFO, siren noises, envelope-controlled pitch sweeps, etc. The input voltage is increased by turning from center position to the right; the input voltage is inverted and increased as the knob is turned to the left.



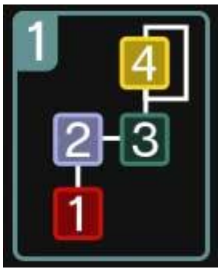
Algorithm selector buttons- The eight large diagram buttons across the top configure FM Station's operators into pre-patched routings known as "algorithms." The colored number boxes within correspond to each of the four operators; the operators in the bottom row of the diagrams represent carriers (i.e. audio sources), and operators above represent modulators (mod sources for the audio oscillators). In algorithms 5-8, all operators hooked to the horizontal line are carriers (i.e. audio sources).

Carrier oscillators are routed to FM Station's *Master Volume* and *P-Mix* (poly) and *M-Mix* (mono) output jacks; modulator oscillators are routed only to carrier *Freq/Ratio* CV inputs. All four operators are **always** routed to the *Op1, Op2, Op3,* and *Op4* individual poly outputs along right side of the panel. The *Amplitude* control bars affect output levels in the *P-Mix* and *M-Mix* outs as well as the individual poly outs.

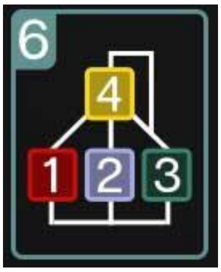
Algorithms are selected by clicking on the diagrams. The numbered operator boxes and outline frame will be colored for the currently active algorithm. A couple of things happen to help clarify which operators are currently configured as carriers and modulators in the currently selected algorithm. The text in the large colored buttons beneath the algorithm boxes changes to display *CAR* or *MOD* to indicate the status of each operator...

... and the *Ampl* control bars turn red for carrier operators, and green for modulator operators:

In following section, we'll go over a few of the algorithms; once you're gained an understanding of these, the others should be easy to grasp.



In algorithm 1, operator 1 is the only carrier. Operator 4 modulates operator 3, and operator 2 modulates operator 1. Although only operator 2 is directly modulating operator 1, the cumulative modulation of operator 4, then 3 can result in operator 2 having a highly complex modulation wave. That said, this algorithm is a good choice for simple patches where operator 2 modulates operator 1 (set operator 3 and 4 *Ampl* sliders to zero). This simple modulation path is essentially the same as the two-oscillator demo patch at the beginning of this guide and is a good starting point for understanding FM synthesis.

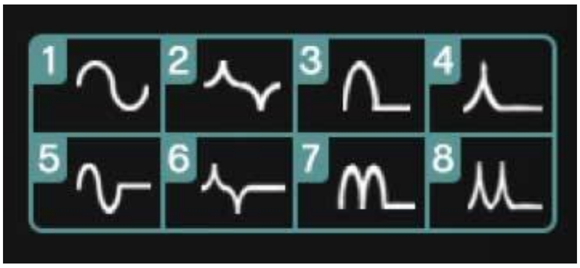


Algorithm 6 configures operators 1, 2, and 3 as audio sources, and operator 4 as a the sole mod source (with feedback loop). Algorithm 6 also resembles the gearshift pattern of my cousin's '71 Jensen Interceptor, which had chronic brake fluid issues.

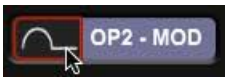
We wanted to specifically discuss algorithm 8, because it's the most interesting algorithm for the serious FM tweaker. When used with the *Mix* output, it's useful for additive synthesis/creating stacked chords, etc. But when used in conjunction with the individual outputs and the freq/ratio CV input jacks, it's the key to using FM Station "a la carte." Because it contains no pre-wired modulator routings, it allows users to roll their own custom algorithm routings.

It's also useful when routing separate outputs to stereo panned mixers and effects, as well as combining multiple instances of FM Station for doubling or tripling the number of available operators for wildly complex FM patches.

Waveforms selector buttons and panel diagram-



The original DX/TX FM synths only used sine waves, but later models added extra waveforms. It should go without saying that a small number of waveforms on an FM synth isn't nearly as much of a limitation as with an analog or sample-playback synth; the original DX7 created a tremendous range of tones with sine waves only, but additional waveforms certainly expand the tonal palette, especially in a four-op configuration (as opposed to a more complex six-op configuration).



The panel display at the top right is simply a graphic displaying all eight waveforms (it's not a control of any kind). Waveforms are selected individually for each operator by clicking the black waveform selector button. These cycle forward with each click.

Freq/Ratio control bar, poly CV input and attenuator- This really a fancy name for pitch. These are tuned to the standard harmonic series of even overtones. They go from 0.25 of the base pitch all the way up to the 32nd overtone. It's unlikely you'll use the higher values as basic oscillator tones, but they higher values are intended for use as modulator sources, and they're the key to getting the signature FM bell-type sounds.

Unlike standard oscillators, the freq/ratio CV mod input isn't used for vibrato; remember that this is actually a phase modulation input, so it won't behave as expected if you patch a low-frequency oscillator or DC voltage source into it (such as a keyboard pitch CV). The main thing you'll use it for is "manually" routing the output of a modulator operator, but remember this is only necessary if the algorithm you've chosen doesn't already contian a pre-wired mod routing path.

Fine freq control bar, poly CV input and attenuator- This acts as a detune control with a range of up or down one octave, and default position at center. The CV input allows control via CV. This is a bipolar control with the middle position representing zero. Negative CV control decreases as the knob is dialed to the left; positive CV control increases as the knob is dialed to the right.

Amplitude control bar, poly CV input and attenuator- Sets the overall volume of the operator. If the operator is currently configured as a carrier, this affects the audio output level in the *Mix* and individual outputs. If the operator is currently a modulator, this affects the amount of modulation signal traveling to the operator it's routed to - this is the most significant sound varying parameter in FM synthesis.

The poly CV input allows control via CV. This is a bipolar control with the middle position representing zero. Negative CV control decreases as the knob is dialed to the left; positive CV control increases as the knob is dialed to the right.

Patching an envelope generator to a carrier operator is how you'll shape the amplitude curve of the sound, exactly as you would use an envelope generator in conjunction with a VCA in analog synthesis. Conversely, patching an envelope or other mod source to a modulator operator is the primary way to achieve dynamically shifting timbre.

Pay attention to the initial setting of the *Amplitude* control bar. If the operator is a carrier, you'll most likely want to set it to zero and patch an envelope generator to "turn the sound on and off" as you play the keyboard. Otherwise, setting the bar position above zero will result in a constant drone. You could run the *Mix* out to a VCA (*Amplifier*) module (controlled by an envelope generator), but there's really no need to, as each operator essentially contains its own built-in VCA.

Feedback control bar, poly CV input and attenuator (Operator 4 only)- FM synthesis makes use of feedback, present in operator 4 in FM Station. This routes the output of operator 4 back to its own *Freq/Ratio* input, with the amount set by the control bar. The resulting audio output greatly varies depending upon the currently selected waveform and amount. If the selected algorithm is using operator 4 as a modulator, this means the resulting modulation wave becomes considerably more complex than the original "unadulterated" wave.

One common FM synthesis trick is to use a sine-wave operator with feedback on its own, and turn the feedback control up about halfway, resulting in a fairly accurate sawtooth wave (hook up an *Oscilloscope* module to check this out). Sonically interesting results occur with different waveforms as well.

The CV input allows feedback control via CV. This is a bipolar control with the middle position representing zero. Negative CV control decreases as the knob is dialed to the left; positive CV control increases as the knob is dialed to the right.

83 Poly Glide

The Poly Glide module is used to smoothly transition between polyphonic CVs. This is typically used to slide between pitch CVs allowing chords to be played with portamento.

This module can be thought of as a stack of standard [Glide] modules. Each module in the stack slides between the CVs received by one of the active voices in the patch. The number of voices used in a patch is set using the *Number of Voices* control in the Poly CV Outs section of the I/O Panel. This setting effects all poly modules in the patch.

The image below shows a simple polyphonic patch using the Poly Glide module to create portamento.

83.1 Inputs, Outputs, and Controls

Input jack- Polyphonic input jack for the control voltages you wish to glide between. Typically this will be patched to the *Poly Pitch* jack in the Poly CV Outs section of the I/O Panel to create polyphonic portamento.

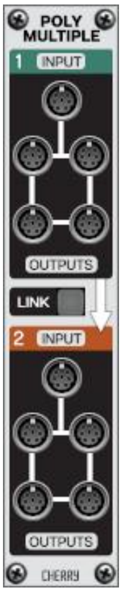
Amount- Adjusts the speed of the glide from 0ms to 5000ms (5 seconds). When the curve is set to *Con*, this is the amount of time it will take to slide from one CV to the next. When the curve is set to *Lin*, the slide time will be different depending on the distance between CVs so this is actually adjusting the speed of the glide rather than the time it will take to complete each transition.

Curve- Selects between two options for transitioning from one CV to the next.
When *Lin* (linear) is selected, the rate of the glide will remain the same regardless of how far apart the CVs are. Therefore gliding between voltages near one another will take less time than voltages that are farther apart. When *Con* (constant) is selected, the amount of time it takes to glide between voltages will be the same regardless of how far apart the voltages are.

External Engage jack- “Mono” CV input that allows the glide module to be enabled and disabled in real time using control voltages. Voltages 2.5V or higher will enable the glide while voltages less than 2.5V will disable it.

Output jack- Polyphonic output jack which outputs multiple "lanes" of CVs that slide from one voltage to the next. Typically this will be connected to the *Poly Keyb CV* input of a Poly Oscillator to create polyphonic portamento.

84 Poly Multiple



The Cherry Audio Poly Multiple is a dual module that "copies" the CV or audio signals received at its "poly" input to four poly output jacks so the signals can be sent to multiple destinations. When the *Link* button is engaged, the input from module one is sent to the outputs of module two as well, creating a total of eight copies. Since every jack in Voltage Modular can have up to six cables connected to it, it’s possible to send a mix of six polyphonic input signals to as many as 48 poly outputs when linked!

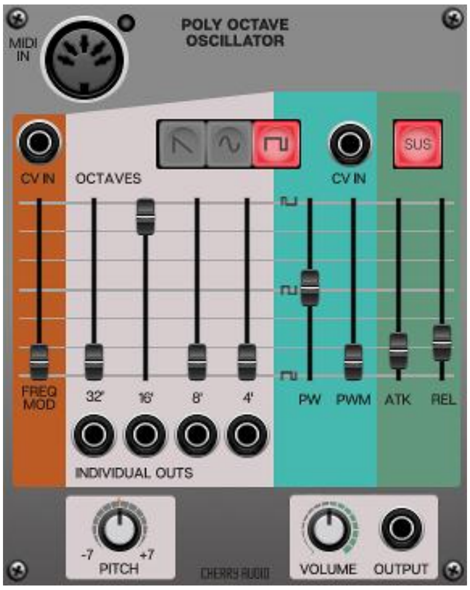
84.1 Inputs, Outputs and Controls

Input jack- This is the input jack for the poly CV or audio signal that will be copied to the output jacks. As many as six poly cables can be patched to this input.

Output jacks- These four jacks output a copy of the poly signal received at the input jack. Each jack can have up to six cables patched to it for a total of 24 possible copies.

Link- When this button is engaged, the input from module one will be copied to the outputs of module two as well, and the input of module two will be ignored. When linked there is a total of 48 possible copies.

85 Poly Octave Oscillator



The Cherry Audio Poly Octave Oscillator is a polyphonic oscillator with four mixable octaves per voice. The MIDI input receives pitch and note on/off messages and internally routes them to the oscillator's frequency and internal amplitude envelope respectively. This module is the perfect starting point for an organ or lush pad! Waveform options are saw, sine and a voltage-controllable variable-width pulse wave.

85.1 Inputs, Outputs, and Controls

MIDI In- This MIDI input jack receives polyphonic pitch and note on/off messages from a MIDI controller or host DAW. Typically this will be connected to the *MIDI From Host* output on the I/O panel.

Freq Mod CV In jack- Control voltage input for externally modulating the oscillator frequency. Useful for adding vibrato with an LFO, siren noises, envelope-controlled pitch sweeps, etc.

Freq Mod- Attenuates the signal received at the Freq Mod *CV In* jack. At 0% no modulation occurs; at 100% a +/-5V signal will modulate the pitch up and down one octave.

Octaves and Individual Out jacks- The sliders act as a mixer for the four octaves of the oscillator. Each octave, labeled in traditional organ footage, is output simultaneously to the main output and its individual output jack at the amplitude specified by the attenuation slider.

Waveform Selector- Selects the waveform of the oscillator. Options are saw, sine and a voltage-controllable variable-width pulse wave.

PW (pulse width)- Sets the width or "duty-cycle" of the pulse wave. This slider has no effect on the other two waveforms. Its default setting of 50% outputs a perfect square wave, rich in odd-order harmonics. Moving the slider up or down will shorten or lengthen the duty-cycle resulting in a thinner sound at either extreme.

PWM CV In jack- Connect an LFO, envelope generator or other mod source here to continuously vary the width of the pulse wave. This is a time-tested analog synthesis trick for adding some serious flavor to a patch!

PWM- Sets the amount of pulse-width modulation by attenuating the signal received at the PWM *CV In* jack.

Attack- Adjusts the amount of time it takes for the internal amplitude envelope to raise from zero to its maximum level.

Sustain- This button activates or deactivates the sustain stage of the internal amplitude envelope. After the designated attack time, the envelope will either sustain at full volume until a key is released or jump immediately to the envelope's release stage.

Release- Adjusts the amount of time it takes for the internal amplitude envelope to fall from its maximum volume back to zero once the release stage is activated.

Pitch- This is a fine tune control for the oscillator. The frequency of the oscillator can be shifted up or down by as little as 0.01 semitones (1 cent) or as much as 7 semitones.

Output jack and Volume attenuator- The main output and volume control for the four mixed octaves.

86 Poly Oscillator

The Cherry Audio Poly Oscillator is a polyphonic version of the "standard" Oscillator module with up to 16-voices of polyphony. This module can be thought of as a "stack" of oscillators that each receive individual pitch CVs so that multiple notes can be played simultaneously.

The number of voices that are used is defined by the *Number of Voices* setting in the Poly CV Outs section of the I/O Panel. This setting is used for all poly modules in the patch.

Below is a simple polyphonic patch created using the Poly Oscillator and other poly modules.

86.1 Inputs, Outputs, and Controls

Poly Keyb CV jack- Accepts a polyphonic CV input for independently controlling the pitch of each oscillator in the "stack." Typically this would come from the *Poly Pitch* jack in the Poly CV Outs section of the I/O Panel.

Poly CV Mod jack and attenuator- Accepts a poly CV input for individually modulating the frequency of the oscillators. A polyphonic envelope, for example, could be patched here so that each individual note in a chord has its own pitch envelope. The bipolar attenuator adjusts the amount of modulation for all voices.

Mono CV Mod jack and attenuator- This is a standard "mono" CV input and bipolar attenuator for modulating the frequency of all of the oscillators in the stack simultaneously.

Hard Sync- This is a polyphonic input jack for force resetting the oscillators. This can be used, for example, to make each oscillator start from the beginning of its waveform every time it is triggered or to create "sync sweep" oscillator sounds by routing the output of a second Poly Oscillator to the *Hard Sync* input and sweeping the pitch of the first oscillator.

Range- Sets the basic pitch of the oscillators, displayed in traditional organ footage. *LO* will be beneath the audible range and allows the Poly Oscillator to be used as a polyphonic modulation source.

Fine- Fine-tune pitch control for all oscillators in the stack.

Pulse Width- Sets the width or "duty-cycle" of the pulse wave. It has no effect on any other waveform. Its default setting of 50% outputs a perfect square wave, rich in delicious odd-order harmonics. Moving the knob left or right narrows its width as well as the thickness of sound until it almost disappears at its extremes, and we've included a nifty "faux-OLED" display to indicate the current pulse width. All oscillators in the stack are controlled simultaneously by this knob.

PWM CV In jack and attenuator- Standard "mono" CV input jack and bipolar attenuator for controlling the pulse width of all of the oscillators. This only effects the pulse waveform.

Waveform Output Jacks- These are polyphonic output jacks for the ramp, sawtooth, pulse, sine, and triangle waves. These can be used simultaneously, in any combination. Each jack outputs all oscillator voices via a single poly cable. This will typically be patched to a Poly Amplifier, Poly Filter, Poly Six-Input Mixer, etc.

Once the signal is ready to be patched to the *Main Outs*, the "poly" signals will need to be converted back to standard "mono" cables. There are multiple modules that can be used for this including the Poly to Mono CV, Mini Poly to Mono CV, and Poly Spread modules.

87 Poly Quantizer

The Cherry Audio Poly Quantizer is a polyphonic version of the standard Quantizer module. This module receives polyphonic pitch CVs via its "poly" input and conforms them to a specific key and scale, or a user-defined set of notes. There are 29 preset scales available and custom scales can easily be created by manually toggling individual notes on or off using the virtual "keyboard." Pitches are tracked and quantized according to the standard 1V/octave scaling.

87.1 Inputs, Outputs, and Controls

Input jack- "Poly" input jack for the CVs to be scale quantized. Typically this will originate from the *Poly Pitch* jack in the Poly CV Outs section of the I/O Panel.

Scale Select- Opens a menu to select one of the 29 preset scales.

Key- Sets the root note or tonic of the scale.

Keyboard Octave- These 12 black and white keys represent each note of an octave in a standard piano keyboard arrangement. The notes included in the selected scale are illuminated in red and can be toggled on and off to create custom scales by clicking on the buttons.

In Offset- Offsets all input voltages up or down in semitone increments by up to two octaves.

In Offset CV Mod jack- Input jack for externally controlling the voltage offset of the input signal. Patching the *Pitch CV Out* from the I/O panel allows pitch transposition of the input signals in semitones via a keyboard or other CV source.

Output jack- Outputs the input signals after being forced to the specified key and scale.

88 Poly Six-Input Mixer

The Poly Six-Input Mixer module is a six-channel mixer for polyphonic CV or audio signals. Each poly input jack accepts up to 16 "lanes" of audio or CVs which are mixed to a single poly output jack. This can be used, for example, to mix the audio outputs of Poly Oscillators or to combine multiple polyphonic CV signals sent from Poly Envelope Generators, Mono to Poly CV modules etc.

88.1 Inputs, Outputs and Controls

S- Solo button for isolating the channel's signal. When engaged, all channels that are not also soloed will be removed from the master output.

M- Mute button for removing the channel's signal from the master output.

Poly Input jacks (1 - 6)- Poly input jacks for polyphonic audio or CV signals.

Level (1 - 6)- This knob adjusts the level at which each channel's audio or CV signals are sent to the mixer's master output.

Level Meter- Visually displays the level of the input signal sent to the master output.

Master Volume- This knob controls the volume of the mixer's master output.

Master Output jack- Poly output jack which outputs a mix of all six poly signals.

89 Poly Stereo Spread

The Poly Stereo Spread module converts Voltage Modular’s "poly" audio signals to a standard left/right stereo output with CV control of width and balance.

When using poly modules, it’s necessary to convert the poly signals to standard "mono" signals before sending them to the Main Outs. This module takes care of the conversion while also allowing all of the active voices to be evenly spread between the left and right outputs to create width.

For example, if the *Number of Voices* in the Poly CV Outs section of the I/O Panel is set to three and the *Width* control is set to 100%, one voice will be panned hard left, one will be centered, and one will be panned hard right. The number of active voices will always be spread equally between the two outputs and the overall width can be narrowed by decreasing the *Width* setting.

Keep in mind that if all of the active voices are not being played, the panning positions may seem random or unbalanced because there will be "gaps" between voices. If you are playing four note chords with four active voices on the other hand, each chord will remain evenly balanced between the two outputs.

89.1 Inputs, Outputs, and Controls

Poly In jack- Patch “poly” audio signals here.

Width CV jack and attenuator- This is a standard "mono" CV jack and attenuator for externally controlling the width of the stereo output.

Width- Adjusts the overall width of the stereo output. When set to 100%, the voices will be spread across the entire stereo field from 100% left to 100% right. This range can be narrowed by turning this knob down.

Bal (Balance) CV jack and attenuator- This is a standard "mono" CV jack and attenuator for externally controlling the balance of the stereo output.

Balance- Adjusts the overall pan position of the voices as a group. In other words it offsets the "center position" which the voices are spread evenly around.

Out Level- Handy little volume knob for controlling the level of the stereo output. When converting poly signals back to standard "mono" signals, the summed amplitude of all the voices can become quite loud and often needs to be attenuated.

L and R Outputs- Outputs a stereo mix (using standard "mono" cables) of all the voices received at the *Poly In* jack.

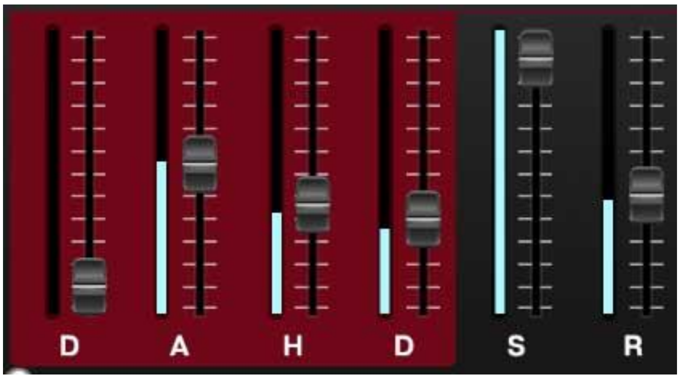
90 Poly Super Envelope Generator

Poly Super Envelope Generator is the dream-come-true envelope for modular synthesists. It starts with a complex DAHDSR envelope (Delay-Attack-Hold-Decay-Sustain-Release). The shapes of the *Attack*, *Decay*, and *Release* stages are individually controllable, morphing from logarithmic to linear to exponential, and these shapes can all be modulated via control voltage. On top of that, the length of each stage (and the sustain level) is CV-controllable as well! Powerful visual feedback is provided every step of the way, so you can see at a glance what's happening with your envelope generator in real-time.

We are going to assume that you understand how a standard ADSR envelope works. If you are unfamiliar with envelopes in general or need a recap, please check out the documentation for the standard Envelope Generator which goes over the basics in detail.

Note that Poly Super Envelope Generator is a polyphonic version of the "standard" mono Super Envelope Generator and is identical in all ways except that the *Loop* Segment Mode has been removed, because it doesn't make sense in a poly module.

90.1 DAHDSR Sliders



"D" (Delay) slider- This is the first stage of the envelope and defines the length of time (after receiving a gate signal) the envelope will remain at 0V before starting the *Attack* phase.

"A" (Attack) slider- Defines the length of time it takes for voltage to rise from 0V to 5V.

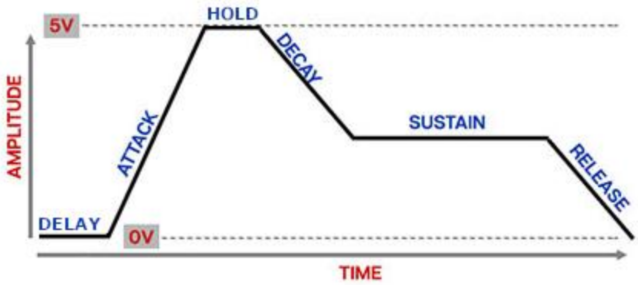
“H" (Hold) slider- Defines how long the envelope will remain at 5V before starting the envelope’s *Decay* phase.

"S" (Sustain) slider- Sets the held voltage level (sustain level) following the *Decay* phase.

"R" (Release) slider- Defines the length of time for voltage to fall from *Sustain* level to 0V when the gate is released.

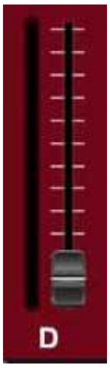
LED stage indicators- These illuminate to show the currently active envelope stage.

Below is a diagram of a DAHDSR envelope to help clarify the individual stages.



90.2 DAHDSR CV Control

Each of the DAHDSR sliders can be CV-controlled using their respective CV input jack and bipolar attenuator. When controlling a slider's value externally, the light blue LED meter to the left of each slider shows the modulation in real time.



It is important to understand that the sliders themselves only show the initial value before any modulation. The blue LED meter displays the actual current setting being used for each stage of the envelope.

90.3 Attack, Decay, and Release Curves

The shape, or curve, of the *Attack*, *Decay*, and *Release* stages of the envelope can be adjusted individually. Each stage has its own curve control which can morph smoothly from logarithmic, to linear, to exponential.

- A logarithmic curve will move quickly at first, then slower as it approaches its destination (as shown in the *Decay* and *Release Curve* displays above).
- A linear curve moves towards the destination voltage at a constant pace.
- An exponential curve will move slowly at first, then quickly “ramp up” as it approaches its destination (as shown in the *Attack Curve* display above).

The shape of each curve can be CV-controlled using its respective *Curve CV* jack and bipolar attenuator and all modulations will be visually displayed in real time.

90.4 Input

Gate In jack- A gate signal received at this jack triggers the envelope to start when in *Normal* or *One Shot* mode.

90.5 Sustain Pedal In

This jack generally gets its input from the I/O Panel CV Sources *Sus* CV out. It holds the env generator at the sustain stage as long as >2.5V is present.

90.6 Multiplier

These buttons multiply all of the slider’s timed values by one, five, or ten making it possible to have seriously long envelope shapes! As an example, if the *Decay* slider is set to 1000ms (1 second) with the *x1* button selected, the decay length will be 5 seconds or 10 seconds with the *x5* or *x10* buttons selected respectively.

Note that these buttons have no effect on the Sustain slider as it is not a time based stage.

90.7 Segment Mode

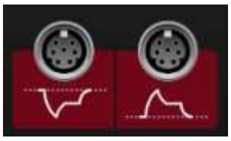
The Poly Super Envelope Generator can be used in two different modes:

Norm- This is the normal envelope behavior where when a gate signal is received, the envelope starts at the *Delay* stage, moves to the *Attack*, *Hold* and *Decay* stages, sustains at the *Sustain* level, then starts the *Release* stage when the gate stops.

One Shot- This mode also only uses the first four stages (DAHD) of the envelope. Each time a gate signal is received at the *Gate In* jack, the envelope starts at the *Delay* stage, moves to the *Attack*, *Hold* and *Decay* stages and then stops. All four stages will be completed regardless of how long the gate signal is held. Note that since the *Sustain* stage is inactive, the *Decay* stage returns all the way back to 0V.

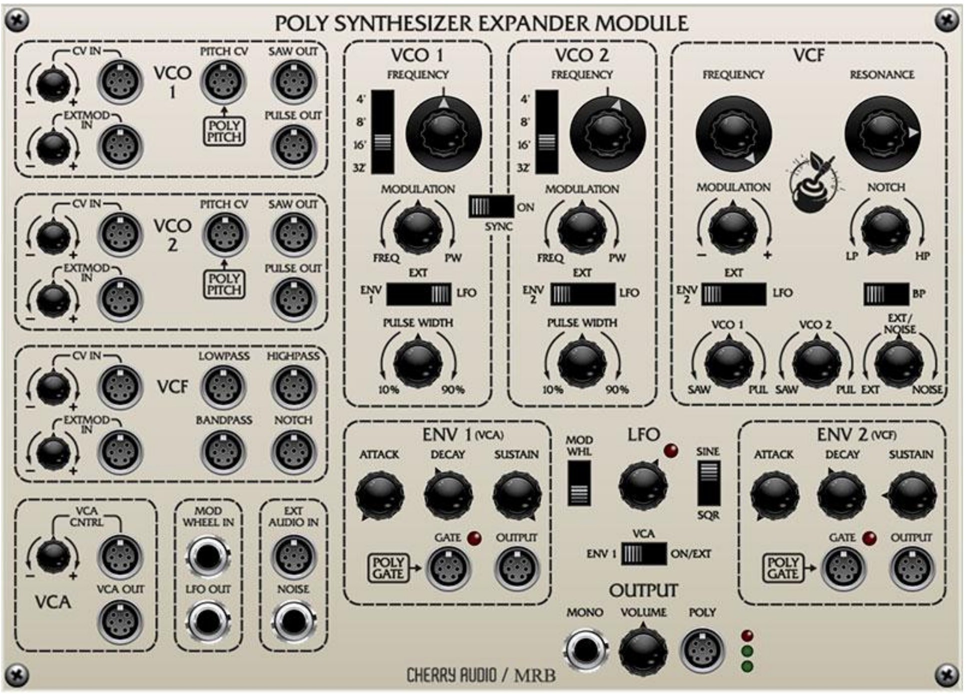
90.8 Outputs

Env Out and Env Out Inv- These are the envelope poly voltage outputs. The jack on the right



has a positive voltage output of 0V to +5V; the jack on the left has an inverted negative voltage output of 0V to -5V.

91 Poly Synthesizer Expander Module



The Voltage Modular Poly Synthesizer Expander Module is a sixteen-voice polyphonic emulation of the classic Oberheim Synthesizer Expander Module, aka the "SEM." Originally released in 1974, the keyboardless, mono SEM module was intended as a companion to the Oberheim DS-2, one of the earliest digital sequencers. Soon thereafter, Oberheim realized they could interface a digitally scanned keyboard, mount the whole mess in a big box, and create polyphonic synthesizers, beginning with the Two Voice, followed by the Four Voice, and finally the beastly Eight Voice (be sure to check out **Cherry Audio Eight Voice**, our super-rad virtual instrument emulation).

Though it was a simple, barebones monosynth, the SEM sounded fantastic, and had a tone quality very different than the common fuzzy, fat Moog sound, thanks to its 12 dB/oct state-variable filter. With lowpass, bandpass, highpass and notch modes, this flexible filter was the star of the show. We've precisely recreated it here with a detailed emulation programmed by award-winning synth designer, Mark Barton (MRB).

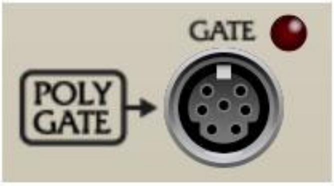
The design is similar to the "patch panel" reissue SEM's released in the 2000's; these were basically identical to the 70s versions but added an extensive patch panel for modular synth flexibility. Because this a polyphonic module, the majority of the patch panel jacks are poly jacks (mono jacks are used where appropriate).

91.1 Semi-Normalled Patching

The Voltage Poly Synthesizer Expander Module would best be described as a normalled semi-modular synth. Unlike most Voltage modules, it contains standard synth sections (i.e. oscillators, filters, etc.) that are internally patched together. Additionally, the Voltage IO Panel *Pitch* and *Gate* connections are "normalled," i.e. invisibly connected to Poly

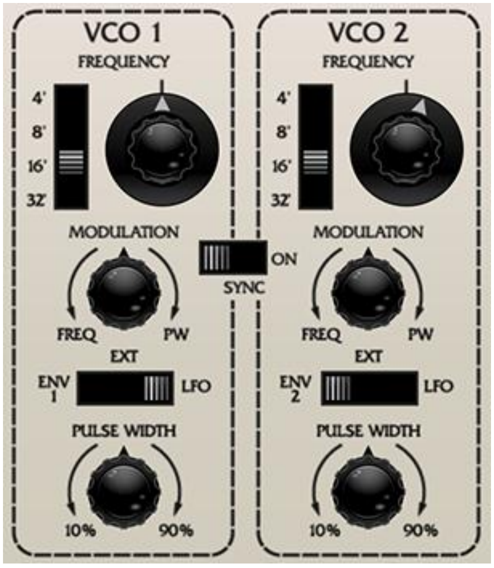
Synthesizer Expander Module's *Pitch CV* and *Gate* jack input jacks. In fact, the only connection that needs to be made to begin playing is the *Output* jack to the IO Panel *Main Outs To Host*.

Any cables patched to the normalled jacks will override (i.e. disable) the normalled IO Panel connections.



Finally, normalled connections are indicated on the front panel by inverted (framed) boxes with arrows.

91.2 VCO 1 and 2



Poly Synthesizer Expander Module includes two almost identical voltage-controlled oscillators. The only difference are their modulation routing options.

Range- Sets the pitch range for each oscillator in octaves. These are at standard organ footage settings of 32', 16', 8', and 4'.

Frequency knob- This can be used to fatten up two oscillator patches by detuning a small amount, or for "building-in" a set interval. Its range is a smidge over a fifth, up or down.

Modulation amount knob and source switch- The mod amount knob is bidirectional; rotating it left from center position adds pitch modulation, rotating it right modulates pulse width of the pulse wave. Center position is off, i.e. no modulation.

The three-position slide switch selects the oscillator mod source from three sources:

- **Env 1 (VCO 1) / Env 2 (VCO 2)**- Modulation source is envelope 1 or envelope 2. Selecting the envelopes as mod source doesn't "disconnect" them from the VCA or VCF.
- **Ext**- Enables CV mod from the *Extmod In* jack and attenuator. The attenuator is bipolar - center position is zero; turn right for positive CV or left for inverted (negative) CV values.
- **LFO**- Enables mod from the onboard LFO.

Pulse Width- Sets the width or "duty-cycle" of the pulse wave. It has no effect on the saw wave. This defaults to 50%, i.e., a perfect square wave. Moving the knob left or right narrows its width as well as the thickness of sound until it almost disappears at its extremes.

Sync- This causes VCO 1 to force reset the start of VCO 2's waveform to the beginning of its cycle, resulting in a wide range of harmonic tones from VCO 2. The range of tones can be varied by adjusting VCO 2's *Frequency* controls.

***Choosing oscillator waveforms**- One unusual aspect of the Oberheim SEM design is that the oscillators themselves contain no waveform controls. Instead, the level of saw and pulse waves is adjusted via mixer knobs at the bottom of the VCF section.*

91.2.1 VCO 1 and 2 Patch Panel

Allows control of VCO frequency mod via patch cables routed from other modules, or the Poly Synthesizer Expander Module itself, as well as separate wave outputs. All attenuator knobs are bipolar - center position is zero; turn right for positive CV or left for inverted (negative) CV values.

CV In jack and attenuator- Allows CV control of oscillator frequency.

Ext Mod In jack and attenuator- Allows CV control of oscillator frequency. **Only active when the three-position modulation source switches in the VCO 1 and 2 sections are set to *Ext*.**

If you need more mod inputs, remember that Voltage Modular allows an unlimited number of cables to be plugged into a single jack, or alternatively, you could mix all the mod sources with mixer module.

Pitch CV in jack and attenuator- Allows CV control of oscillator frequency. These are active at all times and are unaffected by the three-position modulation source switches in the VCO 1 and 2 sections. These are 1V/oct inputs, intended for half-step pitch control input from a keyboard controller or sequencer. If nothing is plugged in, the IO Panel *Pitch* output is normalled to the *Pitch CV* jack for pre-routed keyboard controller; plugging a jack into *Pitch CV* will disable the normalled routing.

The framed *Poly Pitch* box indicates the normalled connection from the IO Panel *Poly Sources/Poly Pitch* output

Saw Out jack- Direct out of the saw wave. This comes before the filter and amplifier stages.

Pulse Out jack- Direct out of the pulse wave. This comes before the filter and amplifier stages.

91.3 VCF

The Voltage Poly Synthesizer Expander Module filter section represents a departure from the 24 dB "ladder" style filter often seen in vintage synths. Like the Oberheim SEM module it's based on, Eight Voice uses a 12 dB state-variable filter (no, this doesn't mean it can sound like Rhode Island or Montana). This refers to its curves - it can function as a lowpass, bandpass, or highpass filter and features a knob allowing a continuous sweep from lowpass to highpass response (with "notch" filtering in the middle position). This gives it a great deal of flexibility, and the 12 dB curve gives it a brighter overall tonality than a typical ladder filter.

If you're not familiar with how filters work, a lowpass filter allows frequencies *below* the cutoff frequency setting to pass through, but blocks frequencies *above* the cutoff frequency. Highpass is the opposite of lowpass mode: high-frequency content remains, but low frequencies are removed as the cutoff frequency increases. Sliding the *Bandpass* switch enables Bandpass mode combining both lowpass and highpass modes, leaving sound only "in the middle." The cutoff frequency lies roughly halfway between the falloff on each side.

A notch filter does the opposite - it removes a middle piece of the audio spectrum but leaves other frequencies intact. (That might not sound useful, but setting the filter to *Notch* mode and slowly sweeping the cutoff frequency with an LFO creates nifty phaser-like tones - great for imitating vintage string synths.)

Frequency- Sets the frequency where frequency attenuation begins with its effect dependent upon the currently chosen lowpass/bandpass/notch/highpass/etc. filter mode.

Resonance- Emphasizes sound energy at and around the cutoff frequency by adding feedback from the filter's output back to its input. This is useful for creating commonly heard synth "wah" tones, especially when the cutoff frequency is modulated with an envelope generator or one of the LFO's.

Modulation amount knob and source switch- Applies modulation to the filter cutoff frequency. The mod amount knob is bidirectional; rotating it right from center position adds positive modulation, rotating it left adds negative modulation. Center position is off, i.e. no modulation.

The three-position slide switch selects the filter mod source from three sources:

- **Env 1 (VCO 1) / Env 2 (VCO 2)**- Modulation source is envelope 1 or envelope 2. Selecting the envelopes as mod source doesn't "disconnect" them from the VCA or VCF.
- **Ext**- Enables CV mod from the *Extmod In* jack and attenuator. The attenuator is bipolar - center position is zero; turn right for positive CV or left for inverted (negative) CV values.
- **LFO**- Enables mod from the onboard LFO.

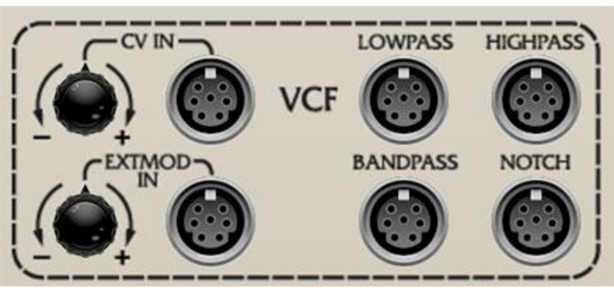
LP>HP/Notch control- Continuously varies the filter response from lowpass to highpass as the knob is rotated from left to right. The middle position creates a notch response.

BP (bandpass) switch- Alters the filter response to bandpass when engaged (i.e. combination of lowpass and highpass leaving frequencies "in the middle." The LP>HP knob disappears when the BP switch is enabled.

VCO 1 / VCO 2 Saw/Pulse level- The *VCO 1* and *VCO 2* knobs adjust the volumes of the saw and pulse waves for VCO 1 and 2, respectively. These are bidirectional knobs as well - rotating them left from center position increase the saw wave level, rotating it right increase the volume of the pulse wave. Center position is off (**if you're getting no sound, check these first**).

Ext/Noise level- When rotated to the left, this sets the level of signals plugged into the patch panel *Ext Audio In* panel jack, when rotated right, it sets the level of the onboard pink noise generator.

91.3.1 VCF Patch Panel



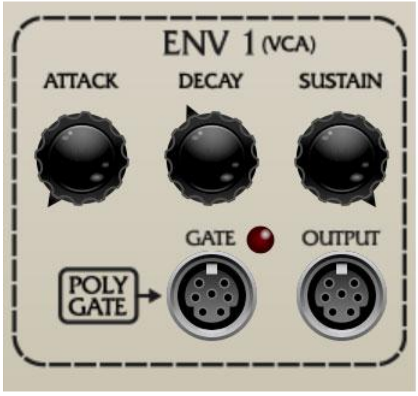
This allows control of VCF frequency mod via patch cables routed from other modules, or the Poly Synthesizer Expander Module itself as well as outputs for all four filter responses. All attenuator knobs are bipolar - center position is zero; turn right for positive CV or left for inverted (negative) CV values.

CV In jack and attenuator- Allows CV control of filter cutoff frequency.

Ext Mod In jack and attenuator- Allows CV control of filter cutoff frequency. **Only active when the three-position modulation source switches in the VCF section is set to Ext.**

Lowpass / Highpass / Bandpass / Notch Out jacks- Separate outputs for each of filter responses. These are all available simultaneously - try routing a couple or all of them to a mixer, and play with their levels and the cutoff frequencies for all manner of awesome formant tonalities.

91.4 Envelope 1 and 2



The original SEM modules included two attack/decay/sustain (ADS) envelope generators. These function much the same as more common attack/decay/sustain/release (ADSR) envelopes, the only difference is that the decay and release stages are combined into a single control.

91.4.1 How They Work

When a voice sees a gate voltage from a note, the envelope outputs a dynamically changing voltage, according to the settings of its stages. The *Attack* stage defines how long it takes for the output voltage to rise from 0 to 5 volts. Once the *Attack* stage reaches 5V, it moves to the *Decay* phase. If the key is released, the *Decay* knob defines how it long takes for the voltage to fall back to zero. If the key is held, the *Decay* time defines how long it takes to fall to the *Sustain* level setting. The note then holds at the *Sustain* level until it is released, and fades to zero at the time set by *Decay* knob - the *Decay* knob effectively does "double-duty," acting as a decay and a release control.

91.4.2 Envelope Controls

Attack- Defines the length of time for voltage to rise from 0V to 5V when a key is played.

Decay- If the key is released, the *Decay* knob defines how it long takes for the voltage to fall back to zero. If the key is held, the *Decay* time defines how long it takes to fall to the *Sustain* level setting.

Sustain- Sets the voltage (i.e. the level) the envelope holds at following the *Attack* and *Decay* phases.

91.4.3 Envelope 1 and 2 Routing

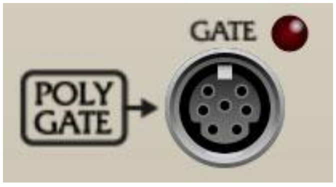
As indicated on the front panel, Env 1 generally affects the VCA, i.e., amplitude. It's hard-wired to the VCA, so no other routing is necessary. If VCO 1's three-way mod routing switch is set to the *Env 1* position, it will also modulate VCO 1's pitch.

Env 2 is mainly intended to modulate VCF cutoff frequency, but because it's not hard-wired, VCO 2's three-way mod routing switch needs to be in the *Env 2* position for cutoff mod to occur. In addition to filter cutoff frequency, Env 2 can also be used to modulate VCO 2's pitch when VCO 2's three-way mod routing switch is set to the *Env 2* position.

91.4.4 Envelope 1 and 2 Patching

Unlike the left-side CV inputs, these are in the same sections as the controls (space considerations!).

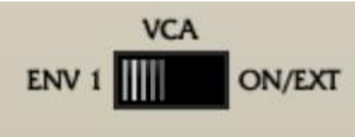
Gate in jacks and LED- This is where you'll patch gate voltages to initiate the envelope generator cycle. Most often this will come from the IO Panel Poly Sources *Poly Gate* output. The LED's next to the gate jacks illuminate when a gate signal is present. If nothing is plugged in, the IO Panel *Gate* output is normalled to the *Gate* jack for pre-routed keyboard controller; plugging a jack into *Pitch CV* will disable the normalled routing.



The framed *Poly Gate* box indicates the normalled connection from the IO Panel *Poly Sources/Poly Gate* output.

Output- Envelope CV signal outputs.

91.5 VCA



VCA switch- Poly Synthesizer Expander Module's VCA has only one control, but it's important to understand. It's located in the middle, directly above the master *Output* knob.

When set to *Env 1* position, amplitude is controlled by envelope 1; this where you'll generally leave it when playing with a keyboard or sequencer controller. The *On/Ext* position latches the VCA open; this useful for drones or when using the *Ext Audio In* jack to process signals with the filter (this way you won't need to hold a key down to hear sound).

91.5.1 VCA Patch Panel

Allows control of VCA amplitude via CV's routed from other modules, or the Poly Synthesizer Expander Module itself, and also includes an output. The attenuator knobs is bipolar - center position is zero; turn right for positive CV or left for inverted (negative) CV values.

VCA Control In jack and attenuator- Allows CV control of the amplitude level.

VCA Out- This is a VCA signal out. It's essentially the same as the master *Output* jack, but with no volume control.

91.6 LFO

The LFO generates sub-audio range signals intended for modulation purposes. Note that Poly Synthesizer Expander contains a single LFO applied to all 16 voices.

Mod Wheel switch- Turning this on normals the IO Panel *CV Sources/Mod Wheel* output to LFO depth control. This makes setting up the LFO with a mod wheel super easy. If the Mod Wheel switch is on and your controller's mod wheel is at zero, the LFO won't have any signal output, thus... **if the LFO doesn't seem to be working, make sure the *Mod Wheel* switch is in the off position (or push up your controller's mod wheel)**.

Frequency (*control unlabeled*)- The *Rate* knob sets the speed of the LFO, from 0.08 to 15 Hz (with *Sync* switch off) or from 8 beats up to 1/64th note triplets (*Sync* switch on). The LED beside it flashes at the current rate.

Wave Select- Chooses between sine and square waves.

91.6.1 LFO Patch Panel

Mod Wheel In jack- The mod wheel is used to vary the depth of the LFO. The IO Panel *Mod Wheel* jack is normalled to LFO depth when the LFO section *Mod Whl* switch is in the up position; patching a cable to the Mod Wheel In jack overrides the IO Panel connection. Remember that the *Mod Wheel In* jack can accept a CV from any source, not just the mod wheel.

LFO Out jack- CV output of the LFO. You may not need it, because the LFO CV out is normalled to VCO 1 and VCO 2's mod routing switch, but patching a cable from the jack lets you route to other desinations in Poly Synthesizer Expander or to external modules.

91.6.2 Ext Audio In and Noise Out Patch Panel Jacks

Ext Audio In- Routes poly audio signals to the filter inputs. To hear external audio, the *Ext/Noise* knob in the VCF section needs to dialed toward *Ext*.

The *VCA* switch (beneath the LFO section) is important when using the *Ext Audio In* jack. When set to *Env 1*, you'll only hear the external audio signal when a key is played (or the gate signal to Env 1 is high). Setting the *VCA* switch to *On/Ext* latches the VCA open so that external signals are always audible. You'll hear a ton of poly notes playing at once when the switch is enabled, but it's useful if you'd like to use a separate module as an envelope generator/VCA. In this situation, we recommend using the unattenuated *VCA Out* jack in the patch panel section.

If you'd like to use the Synthesizer Expander Module's filters to process mono signals (such as other oscillators, drum loops, etc.), we recommend using the mono Synthesizer Expander Module, as it has standard mono jacks.

Noise- Direct output for the onboard pink noise generator.

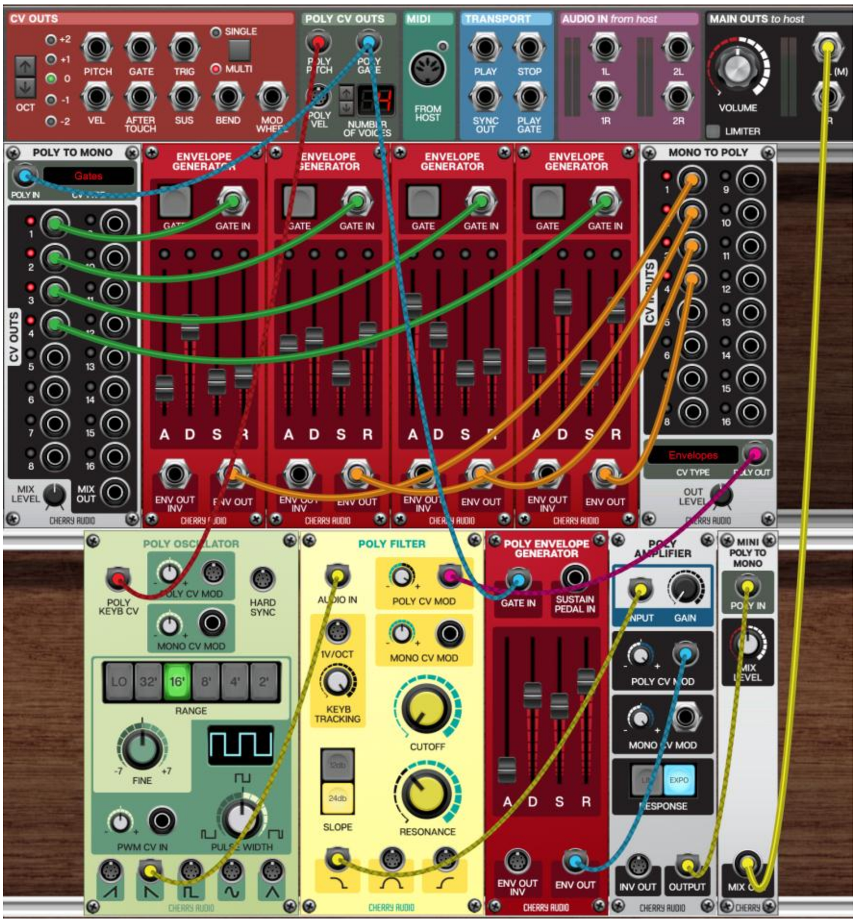
91.7 Output Section

Volume, mono and poly output jacks, and meter- This is post-VCA, and is the final module output with volume control. The red meter LED will glow when things are getting too hot. The *Mono* output jack is the sum of all the poly voices mixed together.

92 Poly To Mono

The Cherry Audio Poly To Mono module receives a "poly" audio or CV signal, and outputs its channels, or "lanes," individually and as a summed mono output with level control.

92.1 Let's look at an example.



In the image acove, the Poly To Mono module is used to create a polyphonic patch that has a different filter envelope for each voice.

The *Poly Gate* jack in the I/O Panel is patched to a Poly To Mono module to separate the gate signals for each of the four active voices. The gate signals are then patched to four envelopes with different settings and converted back to a poly signal using the Mono To Poly module which is patched to the *Poly CV Mod* jack of the Poly Filter. By using four standard envelopes instead of a Poly Envelope Generator, the filter envelope of each voice can be shaped differently which can create interesting variations as the patch is played.

92.2 Inputs, Outputs, and Controls

Poly In jack- Patch "poly" audio or CV signals here.

CV Type- This is a handy little text box that can be used to label the module. To change the text, click in the box labeled "Click to enter label" and begin typing.

Output jacks- Each of these jacks outputs one channel, or "lane," of the signal received at the *Poly In* jack. The number of active outputs depends on the *Number of Voices* setting in the I/O Panel. When the patch is set to have eight voices, for example, the LEDs for *Outputs 1 - 8* will glow red to indicate that they are active. Jacks that are not active will not output signal.

Mix Level- Adjusts the level of the summed signals. When converting signals from "poly" to "mono," it’s often necessary to attenuate the output to compensate for the increase in amplitude caused by summing multiple signals.

Mix Out jack- Outputs a standard "mono" signal carrying the sum of all signals received by the *Poly In* jack.

93 Poly Unison

The Poly Unison module allows monophonic pitch and gate CVs from a keyboard or sequencer to control a "stack" of up to 16 detuned voices playing in unison. The amount of detune can be adjusted manually or controlled externally via the CV input and bipolar attenuator.

Playing a stack of detuned oscillators is a classic trick for creating massively thick sounds! To create the infamous super-saw waveform heard in so many EDM hits, connect the *Pitch* output to the *Poly Keyb CV* input of a Poly Oscillator module and add a healthy dose of detune. The Poly Oscillator's saw-wave output will now be a nice fat super-saw!

93.1 Inputs, Outputs and Controls

Keyb CV In jack- Receives pitch CV signals from a keyboard or sequencer. Typically this will be connected to the *Pitch* jack in the CV Outs section of the I/O panel or the CV output of a sequencer.

Gate In jack- Receives gate CV signals from a keyboard or sequencer. This is usually connected to the *Gate* jack in the CV Outs section of the I/O panel or the gate output of a sequencer.

Detune CV jack and attenuator- CV input jack for externally controlling the amount of detune. The attenuator for this input is a bipolar knob. When at its center position, the detune amount will not be affected by voltage received at this input. When turned to the right, positive voltage received will increase the amount of detune while negative voltage will decrease it. When turned to the left, the CV signal is inverted so that positive voltage will decrease the detune amount and negative voltage will increase it.

Detune- Spreads the tuning of the voices equally above and below the input pitch while keeping the perceived note constant. If the *Number of Voices* is set to one, the detune knob will have no effect. When set to two, increasing the detune amount will lower the pitch of the first voice while equally raising the pitch of the second voice. If set to three voices, the first will be detuned lower, the second will be in tune, and the third will be tuned higher. As more voices are added, their pitches are evenly spread between the lowest and highest voices.

Number of Voices- Shows the number of voices that will be "stacked" in unison and output from the Poly Out jacks. This can be adjusted with the *Number of Voices* setting in the Poly CV Outs section of the I/O panel and will affect all poly modules in Voltage Modular.

Pitch Poly Out jack- "Poly" CV output to control the pitch of a Poly Oscillator module. This jack will output pitch CVs for multiple voices playing the same note in unison. The pitch of the voices can be detuned from one another using the *Detune* knob.

Gate Poly Out jack- This "poly" jack outputs a gate CV signal for each stacked voice. This will typically be patched to the *Gate In* jack of a Poly Envelope Generator to control a Poly Amplifier or Poly Filter module.

94 Poly VCO-20 Dual Oscillator

Poly VCO-20 accurately replicates the tone and functionality of a classic 70s Japanese monosynth, including all waveforms and a white noise source. It also adds CV-controllable pulse width on VCO 1, and hard sync inputs for both oscillators.

Other than being polyphonic, it's functionally identical to the "standard" mono VCO-20 module.

Since Poly VCO-20 contains two independent oscillators, we'll go over the repeated controls and I/O one time, because you're smart, and we don't like typing!

94.1 Inputs, Outputs, and Controls

Pitch CV jack- Accepts a CV input for pitch. Typically this would come from the *Poly Pitch* jack in the IO Panel *Poly Sources* section. Patching a cable to the poly *Pitch CV* input on the VCO 1 side automatically connects to the VCO 2 poly *Pitch CV* input if nothing is plugged into VCO 2's poly *Pitch CV* input (that's what the horizontal arrow is showing). Patching a cable into VCO 2 poly *Pitch CV* breaks the normalled connection and lets the *Pitch CV* inputs function independently.

Hard Sync jack- Force resets the start of the waveform to the beginning of its cycle. Most often used to create the "sync sweep" oscillator sounds made famous in The Cars' "Let's Go" (or Kraftwerk's "Neon Lights" or No Doubt's "Just A Girl"), by routing the output of one oscillator to the other's *Hard Sync* input and sweeping the pitch of the first oscillator.

Wave Form- Yes, we incorrectly split the word in half, just like the real synth. This selects the waveform for each oscillator.

VCO 1 pulse position has a variable duty-cycle (i.e. width), adjusted by the *PW* knob. The jaggedy wave in the last position of VCO 1 indicates white noise.

VCO 2 has two fixed pulse waves - square and narrow. The *Ring* position enables ring modulation between the two oscillators. It won't affect the *VCO 1 Out*, but you'll hear it in both the *Mix Out* and VCO 2 Out. It's actually not a true, technically correct ring modulator - the engineers of the original synth used a common-for-the-era method of achieving a very similar effect, and it sounds pretty wicked! It's most audible when using wide pitch spreads between the oscillators and mucking with VCO 2's *Pitch* knob (or modulating via VCO 2's *Freq CV* mod input if you're one of them fancy types).

PW (VCO 1 only)- Sets the width of VCO 1's pulse wave, from a perfect square to a very narrow pulse. It has no effect on other waveforms.

Master Tune (VCO 1 only)- This is situated on the VCO 1 side, but it affects tuning for both oscillators, up or down about a fourth.

Scale- Sets the basic pitch of the oscillator, displayed in traditional organ footage, i.e. larger number equals lower pitch.

Pitch (VCO 2 only)- Detunes VCO 2 independently of VCO 1. The front panel of the original MS-20 is labeled -5 to +5, but its range was actually around an octave either way, so we've done the same here - Poly VCO-20's detune range is just over an octave up or down. Detune can be used for subtle "fattening," or setting note intervals between the two oscillators for hideous prog rock soloing (or other less offensive uses).

PW CV mod jack (VCO 1 only)- You may have noticed that moving the *PW* knob back and forth creates a nifty sound; instead of wearing out your mouse hand, the *PW CV* input can be used in conjunction with an LFO, envelope generator, or other mod source to continuously vary the pulse width. Note that the attenuator knob is bipolar, i.e. "zero" position is center. Turning right adds a positive modulation, turning left inverts the incoming CV.

Freq CV mod jack- Allows modulation of oscillator pitch. This input is exponential; typically you'd use this for standard vibrato, trills, etc.

Lin Freq CV mod jack- Allows modulation of oscillator pitch. This input is linear and is useful for applications where you want the mod amount to stay constant across the pitch range, such as FM synthesis.

VCO 1 Out / VCO 2 Out- Independent outputs for each oscillator.

Mix Out- Outputs an equal, 50/50 mix of both oscillators. For finer control of oscillator mix level, patch the individual outputs to a mixer module.

95 Poly VCF-20 Filter

The Cherry Audio Poly VCF-20 Filter is an analog-style, voltage-controllable, dual highpass/lowpass filter that recreates the aggressive tones of a classic 70s Japanese monosynth. Its uniquely raunchy sound totally transforms the overall tonality of Voltage Modular! The two resonant filters can be used individually, in series, or manually patched in various configurations and are both capable of screaming self-oscillation. We carefully A/B'd the Poly VCF-20 Filter with the coveted "version 35" filter of the original instrument and we think you'll be delighted with its authenticity.

With the exception of its poly capabilities (and poly jacks), Poly VCF-20 is functionally identical to the standard "mono" VCF-20 module.

95.1 Inputs, Outputs and Controls

Highpass/Lowpass In jacks and Level control- These are the input jacks for the highpass and lowpass filters. Signals input here can be attenuated before being sent to the filter via their respective *Level* knobs. Be sure to try using these to “dial in” the filter. Changing the input level of a signal can drastically change the way the filter sounds especially when using high peak settings.

Series- Engaging this button internally routes the output of the highpass filter to the input of the lowpass filter. This is a quick way to use both filters in series with only one input patched. Note though that this bypasses the input stage level control of the lowpass filter. It is possible however to manually patch the filter in series and use both input stages.

HP Cutoff Freq- Sets the cutoff frequency of the 6 dB/oct highpass filter. All frequencies higher than this will be allowed to pass through the filter while frequencies lower than the cutoff will be attenuated at a rate of 6 dB per/octave.

LP Cutoff Freq- Sets the cutoff frequency of the 12 dB/oct lowpass filter. All frequencies lower than this will be allowed to pass through the filter while frequencies higher than the cutoff will be attenuated at a rate of 12 dB per/octave.

Poly CV inputs and attenuators- Poly CV mod inputs and attenuators for externally controlling each filter’s cutoff frequency.

HP and LP Freq CV inputs and attenuators- CV mod inputs and attenuators for externally controlling each filter’s cutoff frequency.

Peak (resonance)- Emphasizes sound energy at and around the cutoff frequency by adding feedback from the filter's output back to its input. As the peak is increased, any modulations or knob twisting of the cutoff frequency becomes more pronounced and can create the classic "vowel-sound" this filter is known for. When turned up past seven or so, the filter begins to feed back enough to self-oscillate. (Note that unlike the original, a cable must be patched to the filter's input to hear it self-oscillate. This is designed to save CPU when the filter is not in use.)

Peak CV inputs and attenuators- CV mod inputs and attenuators for externally controlling the peak (resonance) of each filter. This is a feature the original monosynth did not have. The resonance of this filter can get out of hand pretty quickly, so it's quite nice to have a little extra control via the CV inputs.

Saturation- Adds distortion to the signal. Used subtly it can add extra harmonics to a smooth bass sound or some tasteful grit to a vocal sample. Higher settings will produce the aggressive character that the original is famous for. Be careful though... when used in conjunction with a high peak setting, this filter will literally scream!

About MS-style Oscillator Distortion: *You may notice that VCF-20 doesn't necessarily distort in the expected way when using the standard green Voltage Modular Oscillator (especially with square waves). This is because the wacky, characteristic MS-style filter distortion is partially the result of the not-exactly-correct-on-an-oscilloscope waveforms output from the original MS synth oscillators. These "incorrect" waveshapes are accurately recreated in the VCO-20 Dual Oscillator module, so try it in conjunction with VCF-20.*

Saturation CV inputs and attenuators- CV mod inputs and attenuators for externally controlling the saturation of each filter.

Highpass Out jack- Outputs the processed signal from the highpass filter.

Mix Out jack- Outputs the sum of both filters.

Lowpass/Series Out jack- Outputs the processed signal from the lowpass filter. When the *Series* button is engaged, this will output the signal sent to the highpass filter's input which is then sent to the lowpass filter.

96 Polymode

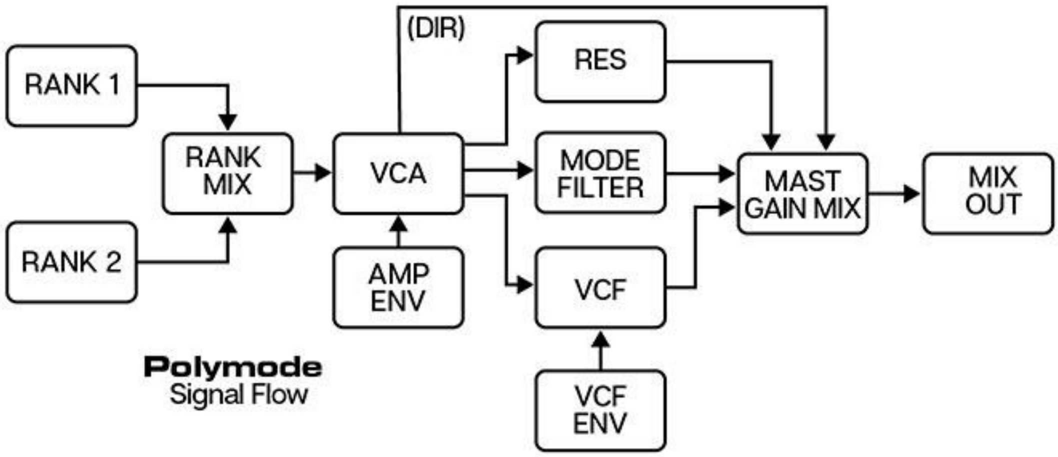
Polymode is a complete instrument module for Voltage Modular, inspired by the infamous 70s-era Polymoog synthesizer. One of the world's first polyphonic synthesizers, it excels at large pads, strings, and vocal-esque sounds, and its unique multiple filter implementation and mod routings gives it a sound like no other synth.

Unfortunately, original Polymoogs are known to be highly unreliable as a result of their elaborate circuitry and the inconsistent quality control of 70s electronic components. They were also really expensive ($5200 in mid-70s dollars!), weighed a ton, sensitive to movement, and relatively confusing to operate. The Voltage Modular Polymode module sidesteps every one of these drawback - hooray for modern computers! Of course we've eliminated all of the Polymoog's reliability issues, we've updated its controls to make it MUCH easier to use, and we've added CV mod ins to almost every control for immense improvement in creative possibilities.

96.1 Polymode Signal Flow

The original Polymoog works like a string synthesizer (or a transistor organ) on steroids. The pitches of two fixed high-frequency square-wave oscillators are "divided-down" (i.e. slowed down) to the frequency of each chromatic note of the very top octave. Another series of IC's halves the top-octave chromatic notes to create the remaining pitches for the entire length of the keyboard. This is how just about every 60s and 70s transistor organ and 70s string synth works (as well as a number of 70s polyphonic sort-of synthesizers such as the ARP Quadra, Korg Delta, etc.).

Though it sounds convoluted, top-octave divide synthesis (referred to as "TOC") was common because it's easy and cheap from an electronics standpoint, but it can only product square waves, which severely limits the tonal palette. To get around this, the Polymoog contained a small circuit board called a "poly card" beneath each key with a custom IC chip (the "Polycom IC") that converted the square wave to a ramp wave and allowed the pulse width of the square wave to be adjusted and modulated. Because of its separate note generation for each key, TOC-based synths can't have a "mono mode," thus keyboard glide can't be implemented. Though we did not implement this "top-down" divide+waveshapers for every note in Polymode (because it's a bear on CPU load), we did go to great lengths to model the resulting idiosyncratic oscillator waves with great precision. Here's what the signal path looks like:



Following the oscillator "ranks" are the VCA/Amplitude Envelopes. This is unlike just about any other analog synthesizer - the VCA is almost always *after* the filter(s), not before. This is because the original Polymoog featured individual VCA/Amplitude envelopes for every note on the aforementioned poly com cards beneath each key. It also means that *every single note has its own independent envelope generator*. The downside of this arrangement it that the VCF cannot be self-resonating because once it started "ringing," there would be nothing to stop its sound - in a more conventional analog synth, playing the keyboard would open and close the VCA, thereby stopping sound from the ringing filter.

Following the VCA/Amplitude Envelopes are three (!) separate filters, all fed in parallel. Mixing and matching these in the *Master Gain* mixer section, along with the unfiltered *Direct* signal, is one of the niftiest things to do with a Polymoog (in turn, this is greatly enhanced by the *Master Gain* mixer's CV mod inputs). The filters are:

- *Resonators*
- *Mode Filters*
- *VCF*

We'll go over each filter in detail later on.

The *Resonators*, *Mode Filters*, *VCF* are fed to the *Master Gain* mixer, and then to the *Mix* out jack in the bottom right *Outputs* section. The direct and filter signals can be individually routed as well.

Polymoog Lower/Upper/Octave Bal Controls- The original Polymoog included "lower" and "upper" duplicates of a number of its controls, allowing independent adjustment of various parameters at a fixed keyboard split point. Because only certain parameters had the dual controls, this resulted in a pretty half-baked attempt at keyboard split capabilities, and along with the already unusual control layout, tended to make the Polymoog even more confusing.

On top of this, the Polymoog had three volume sliders that split the keyboard into three volume zones (with different fixed split points than the upper/lower controls!) for extra bonus confusion. We've eliminated all of this extra-control-splitty madness from Polymode. Trust us, you'll be much happier this way - I can't say how many times these multiple controls have derailed me while making sounds on my vintage Polymoog.

96.2 Polymode Inputs, Outputs, and Controls

Normally we would detail a module's controls from left to right, but because Polymoog/Polymode doesn't exactly have a left-to-right signal flow, we'll start with the top left controls, then follow the actual signal path.

Control/Normalled Cable Connections

Polymode accepts controls signal exclusively from the IO Panel *Poly Sources* section. It doesn't have a "mono" mode per se, but it can easily be played monophonically by setting the IO Panel *Number of Voices* control to *1*.

For convenience, Polymode takes advantage of Voltage Modular's cable normalling feature. Cable normalling allows us to design modules with automatic, "invisible" cables routings. By default, Polymode always has normalled poly cables routed from the IO Panel's *Poly Pitch*, *Poly Gate*, and *Poly Vel* outputs to Polymode's *Poly Pitch*, *Poly Gate*, *Poly Vel* inputs. In other words, it's the equivalent of patching cables like this...

... without having to patch anything. The little green-framed boxes beneath the jacks are a visual reminder of the normalled connections (note how the boxes are the same color as the IO Panel *Poly Sources* section).

If you'd like to patch a signal from a source other than the IO Panel, plugging a cable into any of the *Control* jacks overrides the normalled IO Panel connections. If you'd like to use the IO Panel outputs with additional cables, simply patch a cable from the IO Panel output plus any other cables to the *Control* input jack.

96.3 Control Section

Poly Pitch- Input for poly pitch control. This is normalled from the IO Panel Poly Sources *Poly Pitch* output.

Poly Gate- Input for poly gate control. This is normalled from the IO Panel Poly Sources *Poly Gate* output.

Poly Vel- Input for poly velocity control. This is normalled from the IO Panel Poly Sources *Poly Vel* output.

While we're in that neck of the IO Panel woods, the *Number of Voices* control defines Polymode's maximum polyphony. Dialing this down can be helpful if you're experiencing CPU load issues, but you probably won't, as Polymode is pretty light on CPU.

96.4 Master Tune/Pitch Bend

Mast Tune- Sets overall tuning up or down a half-step.

Pitch Bend and CV input jack- Sets the range of incoming pitch bend data from zero to one octave. Similar to the *Control* section poly jacks, the IO Panel CV Sources *Bend* CV out jack is normalled to Polymode's *Bend* CV in jack (indicated by the dark orange box). This means it's connected by default, but can be overridden by plugging a patch cable into it.

96.5 Oscillators - Rank 1 / Rank 2

In Polymoog parlance, each of the two polyphonic oscillator banks is referred to as a "rank," which is good ol' medieval organ terminology (as are commonly used octave footage numbers). We made the decision to greatly alter the oscillator rank section layout because frankly, the original Polymoog rank controls are really confusing, and offer no improvement in functionality over the more standard layout we've implemented in Polymode. Additionally, the original Polymoog allowed rank 1 to produce *only* square and pulse waves, and rank 2 to produce *only* ramp waves. We felt this was a silly limitation, so we've enabled ramp/square/pulse waves either alone or in combination for both oscillator ranks. The controls for both oscillator ranks are identical with the exception of the Rank 2's *Fine* tune control and are as follows:

Range buttons- Sets the basic pitch range of the oscillator rank, displayed in traditional organ footage. Only one range button can be enabled at any time.

Waveform buttons- Selects ramp and/or pulse waves. These may be used independently or in combination.

Fine *(Rank 2 only)*- Fine-tune control for pitch. This can be used to fatten up patches by detuning a small amount, or for "building-in" a set interval. Its range is a smidge over a fifth, up or down.

FM - Rate/Amount- Each oscillator rank includes a dedicated triangle-wave LFO for frequency (pitch) modulation. The *Rate* slider controls LFO speed, and the *Amount* slider controls LFO depth. The CV jacks and attenuators beneath the sliders allow positive or inverted voltage control of the slider amounts.

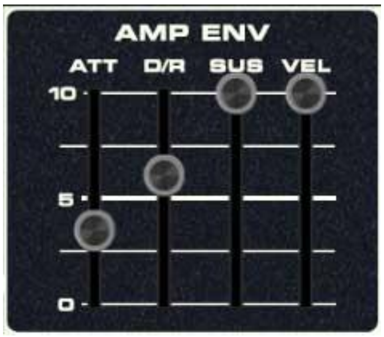
Shape/Mod - Shape- Sets the initial pulse-width of the pulse wave. 5% is narrowest setting and 50% is a full square wave. The CV jack and attenuator beneath the slider allow positive or inverted voltage control of the shape slider.

Shape/Mod - Rate/Amount- Each oscillator rank includes a dedicated triangle-wave LFO for pulse-width modulation (i.e., separate from the frequency mod LFO). The *Rate* slider controls LFO speed, and the *Amount* slider controls LFO depth.

> ***Polymode's Four Independent Pitch and Pulse Width Mod LFO's -*** Most classic polysynths have one LFO, routed via some kind of mod destination selector. Polymode is unique in that it features dedicated LFO's for pitch mod and pulse width, for *each* oscillator (because one of the oscillator ranks was ramp-wave only on the original Polymoog, it had an LFO for rank one pulse-width, and separate LFO's for pitch mod of each oscillator rank). **This multiple and independent modulation is the key to achieving the swirly, warbly sounds the Polymoog is famous for, particularly the "*Vox Humana*" preset made famous by synthpop icon and spiffy dresser, Gary Numan.**

Rank Mix- Balances the level of oscillator *Rank 1* and *Rank 2*. Set to the middle for an equal mix. The CV jack and attenuator beneath the slider allow voltage control of the mix.

96.6 Amplitude Envelope



The Amp Envelope controls Polymode's voice amplitude. Like most modern poly synths, each voice (up to a maximum of 16) has its own independent envelope and associated voltage-controlled amplifier.

Vel (Velocity)- Defines how much the envelope affects amplitude via keyboard velocity. Remember that the IO Panel Poly Sources *Poly Vel* output is normalled to the Controls *Poly Vel* input, so you won't need to patch a cable to utilize amp envelope velocity. A setting of zero gives maximum dynamic range; at a setting of 10, keyboard velocity has no effect on amplitude, and is essentially full blast all the time.

Att (Attack)- Defines the length of time for VCA amplitude mod voltage to rise from minimum to maximum.

D/R (Decay/Release)- Defines the length of time for VCA amplitude mod voltage to fall from the *Att* stage peak to *Sus* stage setting (key held) or fall to zero (key released).

Sus (Sustain)- Sets the held amplitude voltage following *Att* and *D/R* phases (key held).

96.7 Resonators

The Polymoog *Resonators* section is one of its most unique features. It consists of three state-variable filters in a parallel configuration. The parallel routing means the signal is split and runs into all three filters with their collective outputs summed together. Cranking up the *Emph* controls (aka, resonance) for each band can create three individual "peaks" or resonances, hence the "resonator" name.

Another unusual aspect of the original Polymoog Resonators section is that each is band-limited. Most synthesizer filters are configured such that the cutoff frequency covers the entire audible sound spectrum (20-20,000 Hz, give or take), whereas each of the three Polymoog resonators cover a "slice" of the audio spectrum, i.e. low, med, and high frequencies as follows:

Low: 60 - 300 Hz

Med: 300 - 1500 Hz

High: 1500 - 7500 Hz

The original Polymoog *Resonators* section has two major shortcomings: the filter slopes are a little too shallow to create really dramatic resonance effects, and the cutoff frequencies aren't CV controllable. We've addressed both of these issues with a 12/24 db slope selector, as well as bipolar CV inputs for each cutoff frequency

The *Resonators* section excels at creating formant-type sounds, and is also great at emulating phaser-type sounds when its cutoff frequencies are swept via CV modulation.

Pass Mode - Globally selects the behavior of all three filters. *Lowpass* mode allows frequencies *below* the cutoff frequency setting to pass through, *Highpass* mode allows frequencies *above* the cutoff frequency setting to pass through, and *Bandpass* mode combines both lowpass and highpass modes, leaving sound only "in the middle." The cutoff frequency lies roughly halfway between the falloff slope on each side.

Slope- The nature of how a filter works is that frequencies "fall off" above or below the cutoff frequency. Slope adjusts the steepness of this falloff, hence the "slope" terminology. A 12db per/octave filter has a shallower slope, giving it a clearer and brighter character, whereas a 24db per/octave filter's steeper slope gives it a tighter and darker tone and far more pronounced ringing characteristic when the *Emph* slider is turned up.

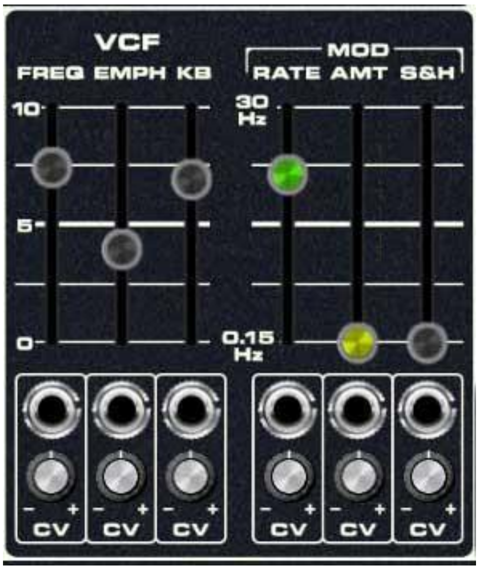
Since the controls for the *Low*, *Med*, and *High* sections are the same other than frequency range, we'll explain them once:

CF (Cutoff Frequency)- Sets the frequency where attenuation begins. Attenuation will be above or below this frequency (or both) depending on the *Pass Mode* switch setting. The CV jack and attenuator beneath the slider allow positive or inverted voltage control of cutoff frequency.

Emph (Resonance)- Emphasizes sound energy at and around the current cutoff frequency by adding feedback from the filter's output back to its input. At lower settings, this can be used to create mild resonances such as those heard in acoustic instruments. At more extreme settings, resonance can create a pure sine wave at its own frequency (variable via the *CF* slider). Be careful with the *Emph* sliders as they can get loud at extreme settings (you can easily control this using each section's *Gain* slider). Note that this "ringing" resonant frequency is much more prominent with the *Slope* switch in the *24db* position. The CV jack and attenuator beneath the slider allow positive or inverted voltage control of emphasis amount.

Gain- This acts as a volume control for each resonator section. Resonator sections can be muted by setting their *Gain* control to 0%. The CV jack and attenuator beneath the slider allow positive or inverted voltage control of gain.

96.8 VCF



This is a standard 24db per/octave voltage-controlled lowpass filter along with a dedicated envelope generator for cutoff frequency control. Being a lowpass filter, it removes high frequencies as its cutoff frequency setting is decreased from max. Note that **this is a single filter for all voices**, sometimes referred to as a "paraphonic" implementation. This means that it's global for all notes, and the VCF envelope will retrigger any time a new note is played. This isn't so great for piano, clav, or other percussive sounds, but it's not too much of an issue for sustained pad, string, or organ-type sounds. The VCF and VCF Envelope controls are as follows:

Frequency - Sets the frequency where high-frequency attenuation begins. The CV jack and attenuator beneath the slider allow positive or inverted voltage control of cutoff frequency.

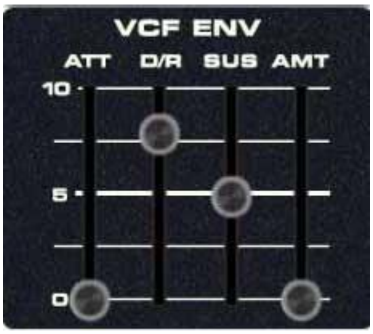
A note about max cutoff frequency: If the cutoff frequency is open all the way, the sound still won't be at full brightness; in other words, some high frequencies are still attenuated. Full filter brightness can be achieved by using the VCF Envelope (detailed in the next section), or by adding CV from the a mod source via the cutoff frequency *CV* jack. This was done to accurately emulate the behavior of a real Polymoog VCF - in fact, a lot of vintage analog synths work this way.

Emph (Resonance)- Emphasizes sound energy at and around the current cutoff frequency by adding feedback from the filter's output back to its input. Useful for typical synth "wah" sounds. The CV jack and attenuator beneath the slider allow positive or inverted voltage control of resonance amount.

Mod - Rate/Amount - The VCF section includes an independent triangle-wave LFO that's hard-wired to frequency modulation. The *Rate* slider controls LFO speed, and the *Amount* slider controls LFO depth. The CV jacks and attenuators beneath the sliders allow positive or inverted voltage control of the slider amounts.

Mod - S&H (Sample and Hold) - Applies sample and hold (aka, random) modulation to the cutoff frequency. The sample and hold rate is controlled by the same clock as the triangle-wave LFO above, thus its rate is set with the same *Rate* control. The triangle-wave LFO and S&H can simultaneously modulate the cutoff frequency as desired. The CV jacks and attenuators beneath the sliders allow positive or inverted voltage control of the slider amounts.

96.9 VCF Envelope



The VCF Envelope exclusively controls the VCF described above. Its controls are as follow:

Amt (Amount)- Defines the depth of envelope control of VCF cutoff. A setting of 0 has no effect on cutoff frequency; a setting of 10 would be maximum control.

Att (Attack)- Defines the length of time for VCF cutoff mod voltage to rise from minimum to maximum.

D/R (Decay/Release)- Defines the length of time for VCF cutoff mod voltage to fall from the *Att* stage peak to *Sus* stage setting (key held) or fall to zero (key released).

Sus (Sustain)- Sets the held VCF cutoff mod voltage following *Att* and *D/R* phases (key held).

96.10 Mode Filters

On the original Polymoog these buttons allowed selection of hard-wired sound presets; in addition to changing waveform, octave, and envelope parameters, the numbered preset buttons also routed audio through fixed filter circuits. Each preset had its own custom fixed filter board, known as a "mode filter." (We weren't kidding when we said Polymoogs have a lot of filtering options.)

Unlike a typical synth filter, they have no external controls and they aren't voltage controllable, so they can't change timbre over time. It's best to think of the *Mode Filters* as preset EQ curves.

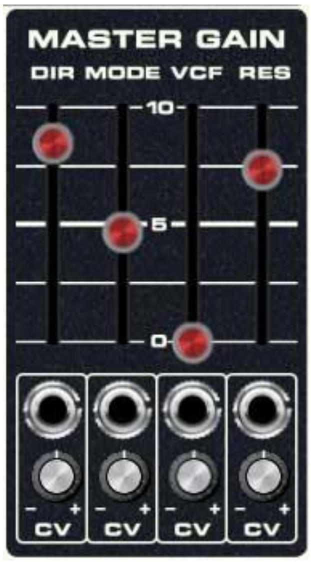
Given Voltage Modular's sophisticated factory and user preset browser, we felt it would be redundant (and potentially confusing) for the *Mode* preset buttons to affect all of Polymodes controls, so please keep in mind that the ***Mode filter names refer only to the mode filters themselves, not entire sound patches*** (the entire factory presets are accurately reproduced and appear as the first nine presets in the standard Voltage Modular Polymode Presets bank).

There were actually two original Polymoog models - the 203A, which is the full-tilt boogie version with numerous controls, and eight sound presets/mode filters. A couple of years later, a less expensive, cut-down version (model 280a) was released., but it increased the number of presets (and mode filters) to 14. Generally speaking, the 280A is less desirable, but it featured the aforementioned "Vox Humana" preset, made famous by new wave icon, Gary Numan. As a result, many Polymoog 203A owners have their instruments modified to eliminate one of the lesser presets and replace it with the Vox Humana settings and mode filter PCB. We've essentially done this in the Polymode module - it retains all eight of the standard Polymoog mode filter presets and adds Vox Humana as a ninth "extra" preset.

To use the *Mode* filters, simply select one and raise the *Mode* slider in the *Master Gain* section. The LED numeric indicator displays the currently selected mode filter. As mentioned, switching mode filters won't affect other parameters.

96.11 Master Gain (Mixer)

The *Master Gain* section is where all Polymode signals are mixed.



Dir (Direct)- Sets *Direct* signal output amount. This signal is the combination of both oscillator ranks, following the *Rank Mix* slider and prior to any of the filters.

Mode- Sets the *Mode Filters* output level. The CV jack and attenuator beneath the slider allow positive or inverted voltage control of *Mode Filter* level.

VCF- Sets the *VCF* output level. The CV jack and attenuator beneath the slider allow positive or inverted voltage control of *VCF* level.

Res (Resonators)- Sets the *Resonators* output level. The CV jack and attenuator beneath the slider allow positive or inverted voltage control of *Resonators* level.

96.12 Outputs

From the *Master Gain* section, signals travel to the *Outputs* section. The *Mode*, *VCF*, and *Resonators* individual output levels are affected by *Master Gain* mixer levels.

Dir (Direct)- This signal is the combination of both oscillator ranks, following the *Rank Mix* slider and prior to any of the filters.

Mode- *Mode Filters* direct out.

VCF- *VCF* direct out.

Res (Resonators)- *Resonators* direct out.

Mix and Volume knob- This the master summed output from the *Master Gain* mix. The master mix level is set with the volume knob above the jack.

97 Quantizer

The Cherry Audio Quantizer module is designed to conform a control voltage input signal to a specific key and scale, or a user-defined set of notes. There are 29 preset scales available and custom scales can easily be created by manually toggling individual notes on or off using the virtual "keyboard." Pitches are tracked and quantized according to the standard 1V/octave scaling.

97.1 Inputs, Outputs, and Controls

Input jack- Input for the CV to be scale quantized. Typically this will originate from the CV output of a module such as the Arpeggiator, Eight Step Sequencer, or I/O Panel *Pitch CV Out*.

Scale Select- Opens a menu to select one of the 29 preset scales.

Key- Sets the root note or tonic of the scale.

Keyboard Octave- These 12 black and white keys represent each note of an octave in a standard piano keyboard arrangement. The notes included in the selected scale are illuminated in red and can be toggled on and off to create custom scales by clicking on the buttons.

In Offset- Offsets the input signal’s voltage up or down in semitone increments by up to two octaves.

In Offset CV Mod jack- Input jack for externally controlling the voltage offset of the input signal. Patching *Pitch CV Out* from the I/O panel allows pitch transposition of the input signal in semitones via a keyboard or other CV source.

Output jack- Outputs the input signal after being forced to the specified key and scale.

98 Random Task

The Cherry Audio Random Task module is a “Turing machine” CV module for generating semi-random voltages with CV control of probability, step shuffling, and pattern length.

This module generates random voltages that are stored at 16 individual steps. Once the steps are initially filled with random voltages, the *Probability* knob controls the chance that the current step will be overwritten with a new random voltage while the *Shuffle* knob controls the chance that all of the stored voltages will be “shuffled” like a deck of cards.

98.1 Inputs, Outputs and Controls

Clock In jack- A 5V pulse or gate received at this jack will advance the module to the next step.

Direction- These three buttons change the order in which the steps are cycled through. Options are *Backward*, *Back* and *Forth*, and *Forward*.

Probability- This sets the probability that the voltages stored at the current step will be overwritten by a new random voltage. At 0% there is no chance that the stored voltage will be changed. At 50%, there is a 50/50 chance that each step will output its stored voltage or be changed to a new random voltage. At 100%, every step will be overwritten with a new random voltage.

Prob CV In jack- CV input for externally controlling the *Probability* setting.

Shuffle- Adjusts the chance that the stored voltages will be “shuffled” like a deck of cards. All of the voltages stay the same but the step number that they are stored at are randomly changed. At 0% there is no chance that the steps will be shuffled. At 50%, there is a 50/50 chance at each step of the pattern that all of the values will be shuffled. At 100%, the voltages are shuffled every step creating a random pattern of the same stored voltages.

Shuffle CV In jack- CV input for externally controlling the *Shuffle* setting.

Steps- Sets the number of steps that are being cycled through as well as potentially changed due to the *Probability* and *Shuffle* settings. With the *Probability* and *Shuffle* knobs both set to 0%, the number of steps can be adjusted and all of the voltages will remain unchanged. If *Steps* is set to three, and the *Probability* and/or *Shuffle* knobs are above 0%, there is a chance that the first three steps may be randomly changed or shuffled, but steps four through sixteen will remain unchanged. Therefore, if the *Probability* and *Shuffle* knobs are set back to 0% and the number of steps is increased again, the voltages previously stored at steps four through sixteen will be the same as they were before.

Steps CV In jack- CV input for externally controlling the number of steps in the pattern.

CV Offset- Offsets the output voltage by adding or subtracting up to 2V.

CV Range- Attenuates the output signal so that its voltage can be limited to a specific range.

CV Out jack- Outputs a control voltage each time the module advances to a new step.

99 Ring Modulator

The Cherry Audio Ring Modulator multiplies two input signals together, typically resulting in a metallic tone with inharmonic overtones often used to recreate pitched percussion instruments such as bells and chimes.

When two signals are multiplied by one another, the resulting signal contains only the sum and difference of the two signals and not the original signals themselves. This often results in tones with unrelated harmonics which can sound harsh or out of tune, but when dialed in carefully can create sounds and timbres hard to create with other methods.

It’s worth mentioning that any two signals can be multiplied by the Ring Modulator. Try multiplying two LFOs to create a more complex LFO shape or multiplying a drum beat and a synth lead!

99.1 Inputs, Outputs and Controls

X / Y Input Level- Attenuators for reducing the level of the input signals. Changing the level of the input signals can help "dial in" the tone or timbre of the output.

X / Y Audio In jacks- Input jacks for the signals that will be multiplied. Although labeled as audio inputs, CVs can also be multiplied to create more complex control signals and audio signals can even be multiplied by CVs.

X / Y CV Mod jacks- Bipolar CV inputs for externally controlling the level of their respective input signals. Center position is zero; turning the knob left adds negative mod, turning it right adds positive mod.

Out jack- Outputs the result of multiplying the X and Y input signals.

100 Re-Animator

Re-Animator adds graphics and visual motion to Voltage Modular patches, and can help to better understand what's happening in modular patches. Still images and GIFs can be loaded via menus or by simply dragging and dropping. Transparency, width, height, and X/Y offsets can all be CV modulated.

100.1 Loading Images and GIFs

The easiest way to load an image or GIF file is to drag and drop it to the center of the display. Images and GIFs can also be loaded by clicking the *Load* button at the bottom right. If an image or GIF is currently loaded, it can be changed either by clicking the *Load* button, or by right-clicking in the display area and selecting *Change*... The display can be initialized by right-clicking and selecting *Clear.*

GIF animations and images are stored with Voltage Modular patches.

100.2 Controls and CV Mod Inputs

All of Re-Animator's controls include mod inputs with jack and a bipolar attenuator. These appear in the blue box adjacent to the control.

X Amount- Shifts the image position left or right horizontally.

Y Amount- Shifts the image position up or down vertically.

Width- Horizontally shrinks or stretches the image.

Height- Vertically shrinks or stretches the image.

Speed - Adjusts the frame playback rate of GIF movies. Center knob position is zero frames per/second - i.e. still frame. Turning the knob counter-clockwise from center causes the GIF to play in reverse at increasing speed. The *Speed* control has no effect on still images.

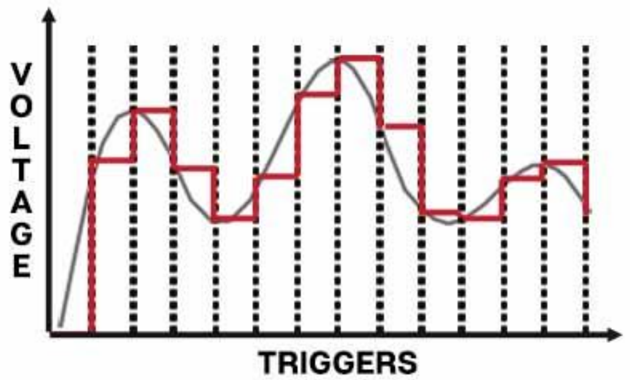
Trans - Adjusts image transparency.

Frame Trig - Sending a gate or trigger signal advances the GIF by one frame. GIG's can easily be synchronized to sequences by patching a clock signal to the input. This input has no effect on still images.

101 Sample and Hold

The Cherry Audio Sample and Hold module is an analog-style synthesis tool that repetitively “samples” an input signal and outputs its voltage until triggered again. This module has an adjustable rate internal trigger source or can be triggered externally with a CV or audio signal.

In the image below, the smooth gray line shows a continuous input signal. Each time the module is triggered the current voltage is “sampled” and “held” until the next trigger. The red line shows the stepped output signal.



White noise is often used as an input source to generate a random stepped-CV signal. Modulating the pitch of an oscillator with this will create the classic sample and hold sound often heard in sci-fi movies. Play with the rate and amount of modulation to create all sorts of bleepy-bloopy goodness! “What was that R2-D2?”

101.1 Inputs, Outputs, and Controls

Input jack- This is the input jack for the audio or control signal that will be sampled.

Ext Trigger jack- This jack can be used to externally trigger the module with a CV or audio signal. Any voltage transition from below 2.5V to 2.5V or higher will trigger the module.

Trigger Source Int/Ext- The buttons *Int* and *Ext* select between the internal and external trigger source.

Rate- Controls the rate of the internal trigger source from 0.02 Hz - 50 Hz.

Output jack- Outputs the stepped sample and hold signal.

102 Sampler I

The Cherry Audio Sampler I is a unique, retro-inspired digital sampler module, designed with creativity at the fore. It's useful for musical tones, drum loops, droning effects and more, and allows instant user sampling, import, and storage of sounds.

We usually try to explain panel controls from top to bottom, or left to right, but here we'll go over things in a way that'll make the most sense for learning how to use Sampler I.

102.1 Concept

Though it bears a resemblance to a classic 80s sampling keyboard, Sampler I works a little differently. It can hold one sample at a time, and can play back one note at a time. And because it has no onboard amplifiers or envelope generators, it's best to think of it more as a sophisticated digital oscillator (as opposed to a self-contained sampling instrument). Since samples don't necessarily play continuously as a standard oscillator would, we've included a few different trigger modes to accommodate different ways you might to use it.

102.2 Audio and CV Inputs and Outputs

Keyb CV- Standard 1V/octave CV input for pitch control.

Gate- Standard 5V gate input for initiating sample playback. This operates a little differently depending on the currently selected *Trigger Mode*.

Master Volume- This is an output volume control. Center position is unity, and turning it adds up to 12db of additional gain, which is useful if a sample is really quiet.

Left / Right Stereo Outputs- These are the audio outputs, but you knew that.

102.3 File Section

This is where existing audio samples can be loaded, and currently loaded samples can be saved and exported. Sampler I can load and play samples in AIF, WAV, MP3, and OGG formats. User samples are in WAV format at 32-bit/48kHz resolution.

Load- Opens a standard dialog for loading raw samples or *.voltagesample* files, which contain samples as well as the following settings:

- Trigger Mode
- Sample Start, End, and Loop Point settings and CV modulation attenuator settings
- Snap To Transient, Crossfade Length, and Loop Enable button settings
- Tune section Octave, Fine, and CV modulation attenuator settings
- Sample Rate/Bit Depth section Rate, Bit Depth and CV modulation attenuator settings

Save- Saves *.voltagesample* files detailed in above.

Export- Exports the entire current sample in WAV format. The exported sample is not affected by *Sample Start*, *End*, or *Loop Enable* control settings (or any other Sampler I settings). The exception is that if a sample has been cropped or normalized, it will export that way.

102.4 Trigger Mode

This affects how samples play back when a gate voltage is received. **It's important to understand how the trigger modes work to best use Sampler I**, so now's the time to turn off Celebrity Big Brother and hunker down.

As mentioned earlier, it's best to think of Sampler I as a sort of big bad sampling oscillator (as opposed to a self-contained instrument), because it doesn't have its own envelope generators or amplifiers. Though it can be used on its own, it's best to create a standard subtractive synth patch where Sampler I's output is patched to filter and amplifier modules, with the amplifier module controlled by an envelope generator, like this:

The way the Trigger Modes function may initially seem counterintuitive, but as you use Sampler I, they'll make more sense, we promise!

Normal- In *Normal* mode with *Loop Enable* off, a sample plays from start to finish when 5V is present at the *Gate* input. The sample plays until the end point regardless of the voltage at the gate input. This is often referred to as "one-shot" playback.

If *Loop Enable* is on, when 5V is present at the *Gate* input, the sample plays from start to finish, then jumps to the loop point and continues loop playback (from loop to end) indefinitely.

In conjunction with the Sampler>Filter>Env>Amp patch shown above, *Normal* mode is useful for playing standard pitched sounds, particularly with looping enabled. The idea is that Sampler I outputs sound continuously while the envelope generator/amplifier combo articulates the amplitude curves of each note. Each time a key is played, the pitch changes to the incoming pitch CV and the new gate voltage restarts playback from the sample start point while retriggering the envelope generator(s).

Gate- Gate mode works much like *Normal* mode, but instead of playing continuously when a voltage is applied to the gate input, sample playback **only** occurs when a gate voltage is present. With *Loop Enable* off, the sample plays from the sample start point until the end as long as a gate voltage is present. If *Loop Enable* is on, playback jumps to the loop point, and the sample plays continuously from the loop point to the end as long as a gate voltage is present.

Gate mode is useful when Sampler I is used by itself, as the on/off nature of the gate effectively functions as a basic on/off organ-style envelope. It's also useful for drum loops when a complex volume envelope isn't needed.

Drone- The sample plays back continuously and the *Gate* CV jack is effectively disabled. This is useful for sound effects beds, background noises, etc. *Loop Enable* is automatically engaged and cannot be disabled in *Drone* mode.

102.5 Tune Section

Octave- Allows transposition of sample playback up or down up to two octaves.

Fine- Allows fine tuning by just over a fifth interval, up or down.

Tune CV jack and attenuator- Allows bipolar CV control of pitch, up to five octaves up or down. You can get pretty crazy with this.

102.6 Sample Rate/Bit Depth Section

Rate- Allows adjustment of sample rate from 48 kHz down to 100Hz. This is a playback parameter, so it won't permanently alter sample data (but it'll temporarily make a real mess). There is no anti-aliasing filtering applied; at lower sample rate settings, you'll hear delicious digital aliasing noise above the playback rate frequency, just like a vintage sampler. The *Rate* setting can be modulated with the *Rate CV* mod jack and bipolar attenuator.

Bit Depth- Sets the bit rate of sample playback from 32 bits down to a grunky 2 bits. As with *Rate*, *Bit Depth* is a playback parameter that won't alter stored sample data. Its setting can be modulated with the *Bits CV* mod jack and bipolar attenuator.

102.7 Sample Edit Section

Here's where the action happens. This where sample start, end, loop points, and other parameters affecting sample playback are set. We'll jump around this section a bit to best explain its operation.

Sample Root- As its name implies, this sets the root position for the current sample. Click on the down arrow button to choose a note value. C2 corresponds to a 2V keyboard CV input, and plays the sample at its native pitch at MIDI note C2.

Sample Start- Sets the sample playback start point. The sample number appears in the tooltip as the slider is moved. As with any control in Voltage, it will move in finer increments by holding down the *Command* key in OS X, or the *Option* key in Windows. *Sample Start* may also be adjusted by grabbing the red start marker in the *Wave View* display. *Sample Start* position can be modulated in real time using the *Start CV* mod jack and bipolar attenuator.

Sample End- Sets the sample playback end point. *Sample End* may also be adjusted by grabbing the red end marker in the *Wave View* display. *Sample End* position can be modulated in real time using the *End CV* mod jack and bipolar attenuator.

Loop Point- With the *Loop Enable* button on, this sets the point at which sample playback loops. Looping always plays from the loop point to the end point, regardless of whether the loop point is set before or after the end point. This potentially translates to all kinds of forward/backward playback shenanigans - see "*Pushin' Forward Back*" section below. *Loop Point* can be modulated in real time using the *Loop CV* mod jack and bipolar attenuator.

Setting Sample Start, End, and Loop points: Note that the *Sample Start* slider's movement resolution stays constant regardless of the *Wave View Zoom* setting, but the resolution of the *Wave View* start marker increases as the *Zoom* size is increased. As a result, it's best to use the *Sample Start*, *End*, and *Loop Point* sliders for big adjustments, and to directly move the *Wave View Start*, *End*, and *Loop* markers for finer adjustments.

Wave View display and controls- Sampler I's *Wave View* screen makes it easy to see and edit samples. The view select buttons have no effect on sound, they only affect the current display.

- *ST*- displays both left and right channels
- *L/M*- displays the left channel, or the entire wave if a mono sample is loaded
- *R*- displays the right channel of a stereo wave
- *Zoom*- rotating clockwise magnifies the wave view. Use the *Zoom* knob in conjunction with the scroll bar thumb beneath the wave display for precise sample editing.

Normalize- This is a fancy term for "increase the overall volume of the sample so that the loudest peak is at 0 db." It's generally a good idea to hit the *Normalize* button if samples aren't very loud. Keep in mind that unlike compression or limiting, normalizing doesn't change the dynamics of a sample, it just proportionally raises the overall gain, and won't affect noise floor. Normalizing does not permanently edit samples loaded from your hard drive, but any exported or saved versions will retain normalization.

Crop- Deletes sample data before the start point and after the end point. This is particularly useful for extracting a region of audio (a drum loop, for example) from a much larger sound file, and will make editing much easier. As with normalization, cropping will not permanently alter samples loaded from your hard drive, but any exported or saved versions will retain cropping.

Snap To Transient- Toggling *Snap To Transient* on makes the start, end, and loop markers snap to detected transient "hits" in the waveform. This is very helpful when editing drum loops, because the markers will instantly snap to drum hits. It's also handy for setting the start point when there's silence at the beginning of a sample. We like it so much that we set to "on" by default.

Crossfade Length- Adds an equal-power crossfade of up to 300ms from the end point to the loop point when looping is enabled. It's very helpful for creating smooth loop transition points with sustained sounds.

Loop Enable- Turns looping on and off. When disabled, the yellow loop marker disappears from the *Wave View* display.

Force LP To Start- Clicking this button instantly sets the loop point to the same location as the start point for situations where you want the sample to repeatedly play from start to end.

Pushin' Forward Back: *The inspiration for Sampler I's horizontal Start, End, and Loop sliders came from a crude 80s hardware sampler. Not only did the sliders make real-time sample editing super easy, its niftiest trick was that samples could instantly be played in reverse by swapping the start and end points. Sampler I lets you do the same- if the end point is set before the start point, the sample will play backwards. As mentioned, when Loop Enable is on, the loop always plays from the loop point to the end point. If the loop point position is modulated using its CV in, it's possible to move the loop point past the end point. Not only does this allow changing from a forward loop to a backward one, it makes very nifty noise as you cross over the end point. To try this, patch the I/O panel Mod Wheel out to the Loop CV mod input (you may need to experiment with positive or negative mod amount settings depending on the loop point location).*

103 Sampler II

The Cherry Audio Sampler II is a unique, retro-inspired digital sampler module, designed with creativity at the fore. It's useful for musical tones, drum loops, droning effects and more, and allows instant user sampling, import, and storage of sounds.

We usually try explain panel controls from top to bottom, or left to right, but here we'll go over things in a way that'll make the most sense for learning how to use Sampler II.

103.1 Concept

Though it bears a resemblance to a classic 80s sampling keyboard, Sampler II works a little differently. It can hold one sample at a time, and can play back one note at a time. And because it has no onboard amplifiers or envelope generators, it's best to think of it more as a sophisticated digital oscillator (as opposed to a self-contained sampling instrument). Since samples don't necessarily play continuously as a standard oscillator would, we've included a few different trigger modes to accommodate different ways you might to use it.

103.2 Audio and CV Inputs and Outputs

Keyb CV- Standard 1V/octave CV input for pitch control.

Gate- Standard 5V gate input for initiating sample playback. This operates a little differently depending on the currently selected *Trigger Mode*.

Master Volume- This is an output volume control. Center position is unity, and turning it adds up to 12db of additional gain, which is useful if a sample is really quiet.

Left / Right Stereo Outputs- These are the audio outputs, but you knew that.

103.3 File Section

This is where existing audio samples can be loaded, and currently loaded samples can be saved and exported. Sampler II can load and play samples in AIF, WAV, MP3, and OGG formats. User samples are in WAV format at 32-bit/48kHz resolution.

Load- Opens a standard dialog for loading raw samples or *.voltagesample* files, which contain samples as well as the following settings:

- Trigger Mode
- Sample Start, End, and Loop Point settings and CV modulation attenuator settings
- Snap To Transient, Crossfade Length, and Loop Enable button settings
- Tune section Octave, Fine, and CV modulation attenuator settings
- Sample Rate/Bit Depth section Rate, Bit Depth and CV modulation attenuator settings

Save- Saves *.voltagesample* files detailed in above.

Export- Exports the entire current sample in WAV format. The exported sample is not affected by *Sample Start*, *End*, or *Loop Enable* control settings (or any other Sampler II settings). The exception is that if a sample has been cropped or normalized, it will export that way.

103.4 Trigger Mode

This affects how samples play back when a gate voltage is received. **It's important to understand how the trigger modes work to best use Sampler II**, so now's the time to turn off Celebrity Big Brother and hunker down.

As mentioned earlier, it's best to think of Sampler II as a sort of big bad sampling oscillator (as opposed to a self-contained instrument), because it doesn't have its own envelope generators or amplifiers. Though it can be used on its own, it's best to create a standard subtractive synth patch where Sampler II's output is patched to filter and amplifier modules, with the amplifier module controlled by an envelope generator, like this:

The way the Trigger Modes function may initially seem counterintuitive, but as you use Sampler II, they'll make more sense, we promise!

Normal- In *Normal* mode with *Loop Enable* off, a sample plays from start to finish when 5V is present at the *Gate* input. The sample plays until the end point regardless of the voltage at the gate input. This is often referred to as "one-shot" playback.

If *Loop Enable* is on, when 5V is present at the *Gate* input, the sample plays from start to finish, then jumps to the loop point and continues loop playback (from loop to end) indefinitely.

In conjunction with the Sampler>Filter>Env>Amp patch shown above, *Normal* mode is useful for playing standard pitched sounds, particularly with looping enabled. The idea is that Sampler II outputs sound continuously while the envelope generator/amplifier combo articulates the amplitude curves of each note. Each time a key is played, the pitch changes to the incoming pitch CV and the new gate voltage restarts playback from the sample start point while retriggering the envelope generator(s).

Gate- Gate mode works much like *Normal* mode, but instead of playing continuously when a voltage is applied to the gate input, sample playback **only** occurs when a gate voltage is present. With *Loop Enable* off, the sample plays from the sample start point until the end as long as a gate voltage is present. If *Loop Enable* is on, playback jumps to the loop point, and the sample plays continuously from the loop point to the end as long as a gate voltage is present.

Gate mode is useful when Sampler II is used by itself, as the on/off nature of the gate effectively functions as a basic on/off organ-style envelope. It's also useful for drum loops when a complex volume envelope isn't needed.

Drone- The sample plays back continuously and the *Gate* CV jack is effectively disabled. This is useful for sound effects beds, background noises, etc. *Loop Enable* is automatically engaged and cannot be disabled in *Drone* mode.

103.5 Tune Section

Octave- Allows transposition of sample playback up or down up to two octaves.

Fine- Allows fine tuning by just over a fifth interval, up or down.

Tune CV jack and attenuator- Allows bipolar CV control of pitch, up to five octaves up or down. You can get pretty crazy with this.

103.6 Sample Rate/Bit Depth Section

Rate- Allows adjustment of sample rate from 48 kHz down to 100Hz. This is a playback parameter, so it won't permanently alter sample data (but it'll temporarily make a real mess). There is no anti-aliasing filtering applied; at lower sample rate settings, you'll hear delicious digital aliasing noise above the playback rate frequency, just like a vintage sampler. The *Rate* setting can be modulated with the *Rate CV* mod jack and bipolar attenuator.

Bit Depth- Sets the bit rate of sample playback from 32 bits down to a grunky 2 bits. As with *Rate*, *Bit Depth* is a playback parameter that won't alter stored sample data. Its setting can be modulated with the *Bits CV* mod jack and bipolar attenuator.

103.7 Sample Edit Section

Here's where the action happens. This where sample start, end, loop points, and other parameters affecting sample playback are set. We'll jump around this section a bit to best explain its operation.

Sample Root- As its name implies, this sets the root position for the current sample. Click on the down arrow button to choose a note value. C2 corresponds to a 2V keyboard CV input, and plays the sample at its native pitch at MIDI note C2.

Sample Start- Sets the sample playback start point. The sample number appears in the tooltip as the slider is moved. As with any control in Voltage, it will move in finer increments by holding down the *Command* key in OS X, or the *Option* key in Windows. *Sample Start* may also be adjusted by grabbing the red start marker in the *Wave View* display. *Sample Start* position can be modulated in real time using the *Start CV* mod jack and bipolar attenuator.

Sample End- Sets the sample playback end point. *Sample End* may also be adjusted by grabbing the red end marker in the *Wave View* display. *Sample End* position can be modulated in real time using the *End CV* mod jack and bipolar attenuator.

Loop Point- With the *Loop Enable* button on, this sets the point at which sample playback loops. Looping always plays from the loop point to the end point, regardless of whether the loop point is set before or after the end point. This potentially translates to all kinds of forward/backward playback shenanigans - see "*Pushin' Forward Back*" section below. *Loop Point* can be modulated in real time using the *Loop CV* mod jack and bipolar attenuator.

Setting Sample Start, End, and Loop points: Note that the *Sample Start* slider's movement resolution stays constant regardless of the *Wave View Zoom* setting, but the resolution of the *Wave View* start marker increases as the *Zoom* size is increased. As a result, it's best to use the *Sample Start*, *End*, and *Loop Point* sliders for big adjustments, and to directly move the *Wave View Start*, *End*, and *Loop* markers for finer adjustments.

Wave View display and controls- Sampler II's *Wave View* screen makes it easy to see and edit samples. The view select buttons have no effect on sound, they only affect the current display.

- *ST*- displays both left and right channels
- *L/M*- displays the left channel, or the entire wave if a mono sample is loaded
- *R*- displays the right channel of a stereo wave
- *Zoom*- rotating clockwise magnifies the wave view. Use the *Zoom* knob in conjunction with the scroll bar thumb beneath the wave display for precise sample editing.

Normalize- This is a fancy term for "increase the overall volume of the sample so that the loudest peak is at 0 db." It's generally a good idea to hit the *Normalize* button if samples aren't very loud. Keep in mind that unlike compression or limiting, normalizing doesn't change the dynamics of a sample, it just proportionally raises the overall gain, and won't affect noise floor. Normalizing does not permanently edit samples loaded from your hard drive, but any exported or saved versions will retain normalization.

Crop- Deletes sample data before the start point and after the end point. This is particularly useful for extracting a region of audio (a drum loop, for example) from a much larger sound file, and will make editing much easier. As with normalization, cropping will not permanently alter samples loaded from your hard drive, but any exported or saved versions will retain cropping.

Snap To Transient- Toggling *Snap To Transient* on makes the start, end, and loop markers snap to detected transient "hits" in the waveform. This is very helpful when editing drum loops, because the markers will instantly snap to drum hits. It's also handy for setting the start point when there's silence at the beginning of a sample. We like it so much that we set to "on" by default.

Crossfade Length- Adds an equal-power crossfade of up to 300ms from the end point to the loop point when looping is enabled. It's very helpful for creating smooth loop transition points with sustained sounds.

Loop Enable- Turns looping on and off. When disabled, the yellow loop marker disappears from the *Wave View* display.

Force LP To Start- Clicking this button instantly sets the loop point to the same location as the start point for situations where you want the sample to repeatedly play from start to end.

Pushin' Forward Back: *The inspiration for Sampler II's horizontal Start, End, and Loop sliders came from a crude 80s hardware sampler. Not only did the sliders make real-time sample editing super easy, its niftiest trick was that samples could instantly be played in reverse by swapping the start and end points. Sampler II lets you do the same- if the end point is set before the start point, the sample will play backwards. As mentioned, when Loop Enable is on, the loop always plays from the loop point to the end point. If the loop point position is modulated using its CV in, it's possible to move the loop point past the end point. Not only does this allow changing from a forward loop to a backward one, it makes very nifty noise as you cross over the end point. To try this, patch the I/O panel Mod Wheel out to the Loop CV mod input (you may need to experiment with positive or negative mod amount settings depending on the loop point location).*

103.8 Sample Record Section

The *Sample Record* section allows real-time user sampling.

Left/Mono and Right inputs- Patch audio to be sampled to these inputs.

Rec Level and Stereo Input Meters- Sets the input level of audio to be sampled. If you've made it this far, you probably know how to set input levels - just into the yellow region for optimum signal-to-noise ratio and overall volume.

Record Button and indicator LED- The *Record* button initiates sample recording. The red LED next to it flashes to indicate "arm" mode, and glows solid to indicate "record" mode. Its operation is closely tied to the *Rec Threshold* control setting. Keep on readin'!

Record Threshold and indicator LED- The sets the level at which Sampler II begins recording. The green LED next to it lights when the recording threshold level has been reached.

If the *Rec Threshold* knob is at zero, the green light is always on, regardless of input level. This means that when the *Record* button is clicked, sample recording begins instantaneously. If the *Record* button is pressed and the *Rec Threshold* LED is not lit, the *Record* button flashes, indicating that it's in "arm" mode - i.e., it's waiting for a signal louder than the *Rec Threshold* setting before it commences recording.

Record Threshold is useful for sampling sources that are quiet and then get loud, such as a drum. In this situation, you'd first set the *Rec Level*, then set *Rec Threshold* so that the green light flashes when the drum sound is played. Now click the *Record* button - its LED flashes to show that it's waiting in arm mode (not recording) until the drum sound is played. When this happens, the threshold has been reached, and recording begins (*Record* LED glows solid). Recording continues until the *Record* button is clicked a second time.

Record Gate- Applying a gate voltage of greater than 2.5V initiates sample recording **for the duration of the gate voltage only**. This is useful for "triggered" on-the-fly recording. Be careful with this, as gate recording will overwrite the currently loaded sample.

104 Six-Input Mixer

The Six-Input Mixer module is a six-input, mono-out mixer for audio or CV signals featuring solos and mutes on each channel. Mixing audio signals is something most of you are probably familiar with but don't forget that this is a CV mixer as well. LFOs, envelopes, pitch CVs, even clock and gate signals can all be mixed together to create unique and interesting control signals.

104.1 Inputs, Outputs and Controls

104.1.1 Input Channels 1 - 6

Input jack- Audio or CV input jack.

Level- This knob adjusts the level at which the input signal is sent to the master output.

S- Solo button for isolating the channel's signal. When engaged, all channels that are not also soloed will be removed from the master output.

M- Mute button for removing the channel's signal from the master output.

Level Meter- Visually displays the level of the input signal sent to the master output.

104.1.2 Master

Volume- This knob controls the volume of the master output.

Output jack- This is the mixer's master output jack.

M- Mute button for muting the master output.

105 Six-Input Stereo Mixer

The Six-Input Stereo Mixer module is a six-channel mixer for audio or CV signals featuring stereo inputs, pan controls, solos and mutes on each channel. This is typically used as a straight-forward stereo mixer for audio signals but don't forget that it can be used as a CV mixer as well. If you think of the *Left* and *Right* output jacks as outputs "One" and "Two," the module can then be used as a routing device to mix CV signals in various ways between the two outputs.

105.1 Inputs, Outputs and Controls

105.1.1 Input Channels 1 - 6

L(M) and R Input jacks- Stereo audio or dual CV input jacks. Use the *L(Mono)* jack to send mono signals to both outputs.

Level- This knob adjusts the level at which the input signals are sent to the master outputs.

S- Solo button for isolating the channel's signals. When engaged, all channels that are not also soloed will be removed from the master outputs.

M- Mute button for removing a channel's signals from the master outputs.

Level Meter- Visually shows the level at which the input signals are being sent to the master outputs.

105.1.2 Master

Master Volume- This knob controls the volume of the master outputs.

M- Mute button for muting the master outputs.

Mono/Stereo button- Selects mono or stereo output. The mono position defeats the pan knobs and effectively turns the Stereo Mixer into a 12-input mono mixer.

L/ R output jacks- These are the mixer's stereo audio or dual CV output jacks.

106 Sixteen Step Sequencer

The Cherry Audio 16 Step Sequencer is identical to the Eight Step Sequencer in feature set and functionality, the only difference is in the number of steps and placement of controls. With that in mind, here's a link to the documentation for the 8 Step Sequencer. Every time you see the number 8, just pretend it's a 16!

107 Splitter

The Splitter module is a MIDI utility module for easily creating "split patches" where the upper and lower portions of a MIDI keyboard-controller send pitch, gate, and velocity MIDI and/or CVs to separate destinations.

This can be used, for example, to play a bass sound with the lower keys while playing a lead sound with the upper keys. The split point can be set at any MIDI note and multiple instances of the module can be used to split the keyboard into any number of "zones."

107.1 Inputs, Outputs and Controls

MIDI In jack- This is the MIDI input jack for the splitter. Typically this will be connected to the output of a MIDI keyboard via the *From Host* MIDI jack in the I/O panel or the *MIDI Out* jack of a *MIDI In* module.

Learn / Split Point- The *Split Point* displays the MIDI note which separates the lower and upper portions of the keyboard and can be changed by clicking the *Learn* button (the button will turn red) and playing the desired note on the MIDI device patched to the *MIDI In* jack.

107.2 Lower section

MIDI Out jack- All notes lower than the *Split Point* will be output from this MIDI jack.

Pitch jack- All MIDI note-number messages lower than the split point are converted to pitch CVs and output from this jack.

Gate jack- All MIDI note-on/off messages lower than the split point are converted to gate CVs and output from this jack.

Velocity jack- All MIDI note-velocity messages lower than the split point are converted to CVs and output from this jack.

107.3 Upper Section

MIDI Out jack- The *Split Point* note itself, and all notes higher, will be output from this MIDI jack. To create more than two “zones,” patch this output to another Splitter module’s *MIDI In* jack. The second module can then be used to split the first module’s upper half for a total of three zones. This can be repeated as many times as necessary to create additional zones.

Pitch jack- All MIDI note-number messages at or above the split point are converted to pitch CVs and output from this jack.

Gate jack- All MIDI note-on/off messages at or above the split point are converted to gate CVs and output from this jack.

Velocity jack- All MIDI note-velocity messages at or above the split point are converted to CVs and output from this jack.

108 Spring Reverb

The Cherry Audio Spring Reverb module is a realistically modeled spring-style reverb with adjustable decay length and true-stereo operation. Spring reverbs can be used in many situations. Short decaying spring reverbs are great for adding a little energy or life into a sound without drastically changing it, while long decaying spring reverbs can take a sound into an entirely new dimension! And let's not forget how cool a classic surf guitar riff sounds drenched in spring reverb!

108.1 Inputs, Outputs, and Controls

L(M) / R Input jacks- These are the mono or stereo audio input jacks. When using a mono input signal, patching it to the *L(Mono)* jack will feed the signal to both sides of the stereo effect.

Input Level- This knob adjusts the level at which the input signal is sent to the spring reverb effect.

Decay (Short/Long)- Adjusts the length of the reverb.

Mix (Dry/Wet)- This knob adjusts the mix between the input signal (*Dry*) and the effected signal (*Wet*) that will be sent to the outputs.

L / R Output jacks- These are the module’s stereo output jacks. When using a mono input signal and wish to keep the reverb mono as well, use the *L(Mono) Output* jack.

109 String Chorus

The Cherry Audio String Chorus module is a quad-chorus, stereo effect modeled after chorus units featured in classic vintage string synthesizers. It can be used in mono, true-stereo, or mono-to-stereo and features *Dry/Wet Mix* and *Stereo Width* controls.

This module uses four chorus effects with varying rates and depths to replicate the sound of a string section where each player performs with slightly different rates and amounts of vibrato.

109.1 Inputs, Outputs and Controls

L(M) and R Input jacks- These are the mono or stereo audio input jacks. When using a mono input signal, patching it to the L(Mono) jack will feed the signal to both sides of the stereo effect.

Stereo Width- While using both the *Left* and *Right Output* jacks, this knob adjusts the width of the chorus effect. With the knob in its fully left position, the left and right sides of the effect are summed to mono while in the fully right position, the four chorus effects are panned apart from one another to create a wider stereo image.

Mix (Dry/Wet)- This knob adjusts the mix between the input signal (Dry) and the effected signal (Wet) that will be sent to the outputs.

In/Out- This button toggles the effect on and off.

L and R Output jacks- These are the module’s stereo output jacks.

110 Sub Octave

The Cherry Audio Sub Octave module is a sub-octave generator that tracks the pitch of an input signal and generates square waves one and two octaves below it. There are individual attenuators for each sub-octave as well as the direct input allowing you to create the perfect mix between the three signals. This module will instantly fatten up an oscillator and is a quick way to add some weight to your sounds!

Although this module is usually used to create lower octaves of oscillators or audio signals, it is worth noting that it will also work on LFOs and can create some interesting patterns when mixing the two sub-octave square waves with the original.

110.1 Inputs, Outputs, and Controls

Input jack- This is the input jack for the signal whose pitch will be tracked.

Direct Out jack and attenuator- Output and level control for the signal received at the Input jack.

-1 Oct Out jack and attenuator- Output and level control for the square wave generated one octave below the input signal.

-2 Oct Out jack and attenuator- Output and level control for the square wave generated two octaves below the input signal.

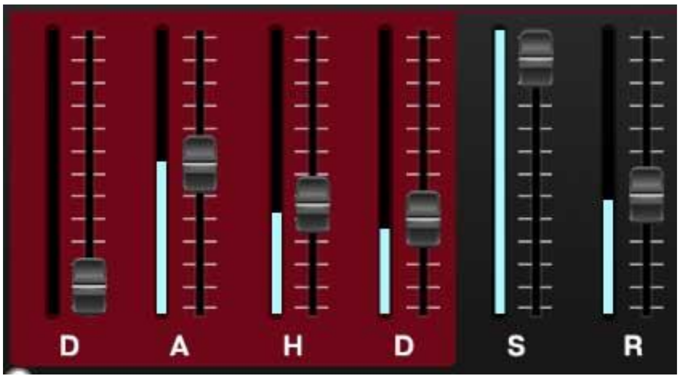
Mix Out jack- Outputs a mix of all three attenuated signals.

111 Super Envelope Generator

Cherry Audio's Super Envelope Generator is the dream-come-true envelope for modular synthesists. It starts with a complex DAHDSR envelope (Delay-Attack-Hold-Decay-Sustain-Release). The shapes of the *Attack*, *Decay*, and *Release* stages are individually controllable, morphing from logarithmic to linear to exponential, and these shapes can all be modulated via control voltage. On top of that, the length of each stage (and the sustain level) is CV-controllable as well! Powerful visual feedback is provided every step of the way, so you can see at a glance what's happening with your envelope generator in real-time.

We are going to assume that you understand how a standard ADSR envelope works. If you are unfamiliar with envelopes in general or need a recap, please check out the documentation for the standard Envelope Generator which goes over the basics in detail.

111.1 DAHDSR Sliders



"D" (Delay) slider- This is the first stage of the envelope and defines the length of time (after receiving a gate signal) the envelope will remain at 0V before starting the *Attack* phase.

"A" (Attack) slider- Defines the length of time it takes for voltage to rise from 0V to 5V.

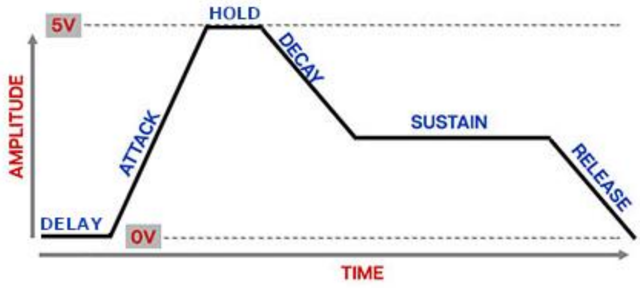
"H" (Hold) slider- Defines how long the envelope will remain at 5V before starting the envelope's *Decay* phase.

"S" (Sustain) slider- Sets the held voltage level (sustain level) following the *Decay* phase.

"R" (Release) slider- Defines the length of time for voltage to fall from *Sustain* level to 0V when the gate is released.

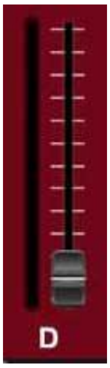
LED stage indicators- These illuminate to show the currently active envelope stage.

Below is a diagram of a DAHDSR envelope to help clarify the individual stages.



111.2 DAHDSR CV Control

Each of the DAHDSR sliders can be CV-controlled using their respective CV input jack and bipolar attenuator.



When controlling a slider's value externally, the light blue LED meter to the left of each slider shows the modulation in real time.

It is important to understand that the sliders themselves only show the initial value before any modulation. The blue LED meter displays the actual current setting being used for each stage of the envelope.

111.3 Attack, Decay, and Release Curves

The shape, or curve, of the *Attack*, *Decay*, and *Release* stages of the envelope can be adjusted individually. Each stage has its own curve control which can morph smoothly from logarithmic, to linear, to exponential.

- A logarithmic curve will move quickly at first, then slower as it approaches its destination (as shown in the *Decay* and *Release Curve* displays above).
- A linear curve moves towards the destination voltage at a constant pace.
- An exponential curve will move slowly at first, then quickly "ramp up" as it approaches its destination (as shown in the *Attack Curve* display above).

The shape of each curve can be CV-controlled using its respective *Curve CV* jack and bipolar attenuator and all modulations will be visually displayed in real time.

111.4 Input

Gate In jack- A gate signal received at this jack triggers the envelope to start when in *Normal* or *One Shot* mode.

111.5 Multiplier

These buttons multiply all of the slider's timed values by one, five, or ten making it possible to have seriously long envelope shapes! As an example, if the *Decay* slider is set to 1000ms (1 second) with the *x1* button selected, the decay length will be 5 seconds or 10 seconds with the *x5* or *x10* buttons selected respectively.

Note that these buttons have no effect on the Sustain slider as it is not a time based stage.

111.6 Segment Mode

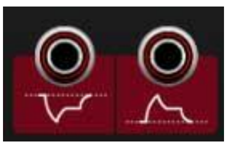
The Super Envelope Generator can be used in three different modes.

Norm- This is the normal envelope behavior where when a gate signal is received, the envelope starts at the *Delay* stage, moves to the *Attack*, *Hold* and *Decay* stages, sustains at the *Sustain* level, then starts the *Release* stage when the gate stops.

Loop- Pressing this button loops the first four stages (DAHD) continuously making the envelope behave more like an LFO. As soon as the button is pressed, the envelope starts at the *Delay* stage, moves to the *Attack*, *Hold* and *Decay* stages, then loops back to the *Delay* stage to start over again. The *Sustain* and *Release* stages are not used at all in this mode, therefore the *Decay* stage will always return to 0V before looping back to the beginning.

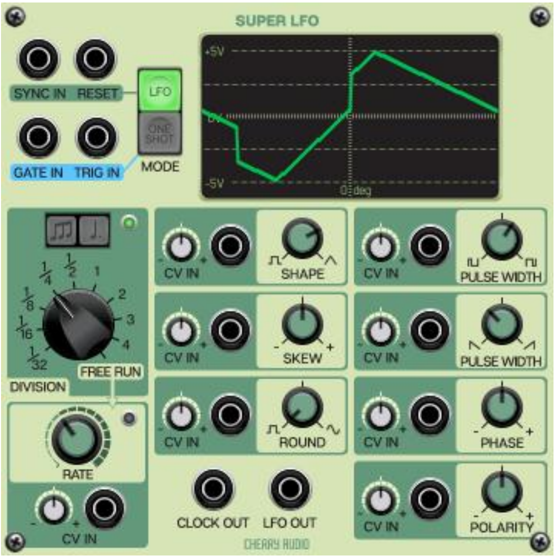
One Shot- This mode also only uses the first four stages (DAHD) of the envelope. Each time a gate signal is received at the *Gate In* jack, the envelope starts at the *Delay* stage, moves to the *Attack*, *Hold* and *Decay* stages and then stops. All four stages will be completed regardless of how long the gate signal is held. Note that since the *Sustain* stage is inactive, the *Decay* stage returns all the way back to 0V.

111.7 Outputs



Env Out and Env Out Inv- These are the envelope poly voltage outputs. The jack on the right has a positive voltage output of 0V to +5V; the jack on the left has an inverted negative voltage output of 0V to -5V.

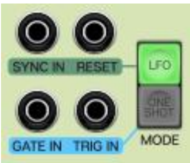
112 Super LFO



We named it "super" for a reason! The Cherry Audio Super LFO (low-frequency oscillator) module is jam-packed with killer features. It can be used as a standard cycling LFO or switched to "one-shot" mode for use as an envelope generator.

It includes a built-in sync divider and reset input for easily syncing the LFO to a host DAW, or can be used in free-run mode with CV-controllable rate. But the real showstopper is its mindblowing custom waveshaping flexibility... with seven waveshaping parameters, a huge real-time display, and bipolar CV control for real-time manipulation, the possibilities are endless. The final word in LFO modulation!

112.1 Sync and Trigger Inputs



Sync In jack- Accepts a 96-pulse-per-quarter-note (PPQN) sync signal for syncing the LFO to a Sync Generator or DAW. To sync the Super LFO to your host DAW, patch the *Sync Out* jack in the Transport section of the I/O Panel to this jack. When *Division* is set to *Free Run*, the sync signal is ignored.

Reset jack- A 5V pulse or gate received at this jack resets the LFO waveform to the beginning of its cycle. When syncing the Super LFO to a DAW, patch the *Play* jack in the Transport section of the I/O Panel to this jack to reset the LFO’s cycle each time the play button in your DAW is pressed.

Gate In jack- When in *One Shot Mode*, a 5V gate signal from a keyboard or sequencer can be used to “play” the LFO like an envelope generator. The LFO will only output while a gate signal is being sent to this jack. Therefore, if a gate signal is shorter than the LFO cycle, only a portion of the LFO waveform will be output. Typically this will be patched to the *Gate* jack in the CV Outs section of the I/O Panel or the *Gate Out* jack of a sequencer.

Trig In jack- When in *One Shot Mode*, a 5V trigger, gate or pulse received at this jack will cause one full cycle of the LFO to output. Typically this will be patched to the *Trig* jack in the CV Outs section of the I/O Panel or the *Trig Out* jack of a sequencer.

Mode- The Super LFO can be used in two different modes.

- When *LFO* is selected, the Super LFO behaves like a standard cycling LFO that is continuously outputting signal. The *Sync In* and *Reset* jacks are used in this mode and the *Gate In* and *Trig In* jacks are ignored.
- When *One Shot* is selected, the LFO can be used as an envelope generator outputting one cycle of the LFO each time a gate or trigger is received at the *Gate In* and *Trig In* jacks.

112.2 Rate

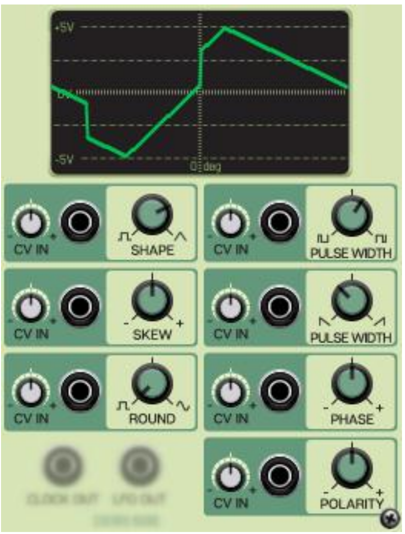
Division- Sets the rate of the LFO to a BPM-specific note-division when synced to an external sync signal via the *Sync In* jack. Triplet and dotted note-values can be selected by engaging their respective buttons above the dial. When set to *Free Run*, the sync signal is ignored and the *Rate* knob is used to set the LFO's frequency.

Rate- When *Division* is set to *Free Run* this knob controls the frequency, or speed, of the LFO.

Rate CV In jack and attenuator- CV input and bipolar attenuator for externally controlling the rate of the LFO when *Division* is set to *Free Run*.

112.3 Waveshaping

The following seven parameters dictate the shape of the LFO waveform and can all be CV controlled with their respective CV input and bipolar attenuator. The waveshape, and any modulations of it, are displayed in real time by the waveform display.



Shape- The LFO's wave shape can morph between a pulse wave, when turned fully to the left, and a triangle wave when turned fully to the right. At any position between, the waveform is a combination of the pulse and triangle waves, both of which can be edited independently of one another by some of the following controls.

Skew- Changes the phase of the triangle waveform independently of the pulse wave.

Round- Gradually "softens" or "rounds" the shape of the LFO.

Pulse Width- Adjusts the width, or "duty-cycle," of the pulse wave independently of the triangle wave.

Triangle Pulse Width- Adjusts the width of the increasing and decreasing portions of the triangle wave allowing it to morph between "ramp down" and "ramp up" waveforms.

Phase- Adjusts the phase of the combined LFO waveform. Changing the phase doesn't change the shape itself, but instead changes the "starting point." This is especially useful when syncing the LFO to a DAW or when in *One Shot* mode.

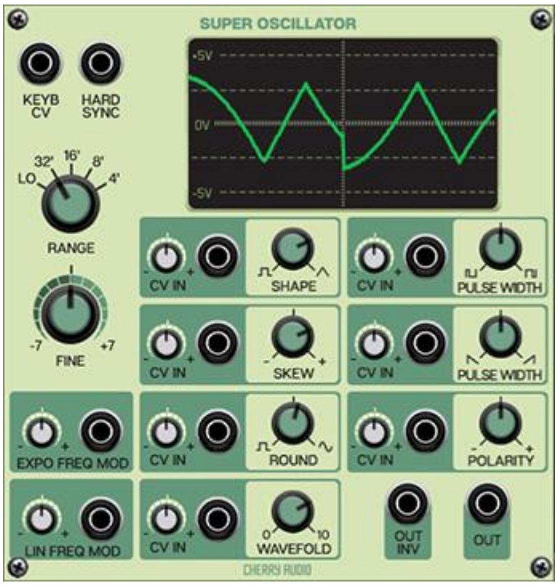
Polarity- This knob changes the polarity range of the LFO's waveform from only negative voltages (-5V to 0V) when turned fully to the left, to bipolar (-5V to 5V) at its center position, to only positive voltages (0V to 5V) when turned fully to the right.

112.4 Outputs

Clock Out jack- Outputs a clock signal for syncing sequencers, switches, other LFOs etc. to the Super LFO.

LFO Out jack- This is the main output jack for the LFO signal.

113 Super Oscillator



The Cherry Audio Super Oscillator is our mondo-powerhouse, pull-out-all-the-stops oscillator module. It's capable of creating an endless variety of waves, many of which you wouldn't expect from a traditional analog-style oscillator. It's surprisingly easy to use, and all of its fabulous waveshaping parameters are voltage controllable. On top of all that, it features a large, real-time animated waveform display for visual representation of the current waveform (plus it looks really neat).

We'll start with the basic controls, then move into the more advanced waveshaping controls.

113.1 Inputs, Outputs, and Controls

Keyb CV jack- Accepts a CV input for pitch. Typically this would come from the PITCH jack in the IO Panel CV OUT section, or from a sequencer pitch CV out.

Hard Sync- Force resets the start of the waveform to the beginning of its cycle. Most often used to create the "sync sweep" oscillator sounds made famous in The Cars' "Let's Go" (or Kraftwerk's "Neon Lights" and No Doubt's "Just A Girl"), by routing the output of a second oscillator to the *Hard Sync* input and sweeping the pitch of the first oscillator.

Most traditional hard sync sounds are created using square or saw waves, but there's no reason hard sync can't be used with the Super Oscillator's more esoteric waveshapes.

Range- Sets the basic pitch of the oscillator, displayed in traditional organ footage.

LO outputs frequencies beneath the audible range and allows the oscillator to be used as a mod source. There's no reason you *can't* use the Super Oscillator for modulation purposes, but we suggest using the Super LFO module instead, as its controls and capabilities are similar, but optimized for modulation purposes.

Fine- Fine-tune control for pitch. This can be used to fatten up multi-oscillator patches by detuning a small amount, or for "building-in" a set interval. Its range is a smidge over a fifth, up or down.

Expo Frequency Mod attenuator and input jack- This is used for externally modulating the oscillator frequency. *Expo* refers to the amount curve across the knob travel - values are finer at the bottom and grow larger exponentially as the setting is increased.

Lin Frequency Mod attenuator and input jack- This is used for externally modulating the oscillator frequency. *Lin* refers to the amount curve across the knob travel - unlike Expo mod, the amount curve is constant or "linear" across the knob travel.

Waveform Display The waveform display window is a real-time animated display showing the current waveform, and any modulation being applied. The center horizontal line represents the 0V point, i.e. zero crossing of the waveform; the center vertical line represents the start/loop point of the waveform. The -5V and +5V lines indicate the 10V peak-to-peak maximum output of the Super Oscillator. The peak-to-peak differential of the waveform is a good general indicator of overall output level.

113.2 Waveshaping Parameters, aka, The Fun Part

Unlike a typical analog oscillator where fixed waveforms are selected using a switch or multiple outputs, the Super Oscillator outputs a single wave, dialed in using its seven continuously variable controls. Each of these controls includes a CV in jack and bipolar attenuator.

Note: If you get really out there, the waveshaping parameter knobs (or any other knobs in Voltage Modular) can be reset to their initial values by double-clicking or option-clicking them (depending on your preference settings in *Gear Settings Icon>Interface>On Control Double-Click*).

Shape- This defines the basic character of the waveform. By default, its position is far right, which creates a pure triangle wave. Dialing the knob to the left audibly and visibly morphs the wave to a pure square by adding a vertical section in its middle. Interestingly, triangle, square, and all points in-between create a wave rich in odd-order harmonics (real-world examples include clarinets and Wurlitzer electric pianos).

Skew- This slides the waveform start/loop point left or right. Its affect on sound will vary depending on Shape and other settings. If Shape is set full right to a pure triangle wave, it won't have much affect; essentially it will just alter wave phase. It will affect the sound more noticeably when the Shape knob is set to create the aforementioned "vertical line" in the center of a wave. As you'll see, experimentation is key.

Round- Gradually rounds off any sharp edges in the wave. Set to its far right, this will change a triangle OR square wave into a sine wave. Its affects are similar to that of a lowpass filter.

Wavefold- The Wavefold control is one of the Super Oscillator's most powerful sound shaping parameters. It sets a threshold at some point in the wave, and "flips" that portion of the wave for dramatic changes in tonality and harmonics. The best way to illustrate the effects of wavefolding is to set up a sine wave by initializing all waveshaping controls and setting the *Round* control at full right position.

Now slowly bring up the Wavefold control amount. You'll see the top curves of the wave "fold" over increasingly and hear its harmonic content grow more intense. Wavefolding's effect is look to see if the wave goes above (or below) a specific threshold. When it does, instead of clipping off the top and bottom of the wave, they create a mirror image of it and reflect that portion of the wave back upon itself, creating more high harmonics and interesting spectra in the process.

To help drive the incoming waveshape into this behavior, they may have amplifiers on their input, or offsets to cause the wave to clip & fold on one excursion such as positive but not the other (this is sometimes referred to as symmetry). They are often adjustable to allow several folds to occur on a single positive or negative excursion beyond the folding threshold, which causes increasingly bright and noisy sounds on the output.

Pulse Width- This sets the width or "duty-cycle" of the pulse wave. It has no effect on any other waveform. Its default setting of 50% outputs a perfect square wave, rich in delicious odd-order harmonics. Moving the knob left or right narrows its width as well as the thickness of sound until it almost disappears at its extremes, and we've included a nifty "faux-OLED" display to indicate the current pulse width.

PWM Amount attenuator and PWM Mod input jack- You may have noticed that moving the *Pulse Width* knob back and forth creates a nifty sound; instead of wearing our your mouse hand, the *PWM Mod* input can be used in conjunction with an LFO, envelope generator, or other mod source to continuously vary the pulse width. Best of all, the OLED display looks real cool swooping back and forth.

Waveform Output Jacks- These are output jacks for ramp, sawtooth, pulse, sine, and triangle waves. These can be used simultaneously, in any combination.

114 Sync Divider

The Cherry Audio Sync Divider can be used to synchronize sequencers and other modules within Voltage Modular to a host DAW. The *Sync Out* jack in the Transport section of the I/O Panel transmits a constant 96-pulse-per-quarter-note (PPQN) signal based on the tempo of the DAWs session.

The Sync Divider is used to “slow down” the super-fast sync signal to musically relevant note-values from 1/32-notes to 4 bars, including dotted and triplet values.

114.1 Inputs, Outputs and Controls

Sync In jack- Patch the *Sync Out* jack from the Transport section of the I/O Panel to this jack to receive a sync signal from your DAW that can be used to synchronize sequencers and other modules in Voltage Modular.

Reset jack- A 5V pulse or gate received at this jack will immediately force-reset the clock.

- It’s important to understand that sync signals have no idea where the "one" is, or whether your DAW is playing or stopped. If a sync signal is routed to the Sync Divider module and its clock is sent to a sequencer, the sequencer will play at the same tempo as the host DAW project, but “shifted” in time by some random (and usually undesirable) amount. To avoid this the Sync Divider, all sequencers, and many other modules have *Reset* input jacks. Patching the *Play* output jack from the Transport section of the I/O Panel to these resets the modules to “one” the instant the DAW play button is pressed, forcing everything to play in time.

Note Value

Selects the note-value of the clock output pulses from 1/32-notes to 4 bars. Any of the selected note-values can be changed to a triplet or dotted note-value by clicking the corresponding buttons which light up green when engaged.

For anyone unfamiliar with rhythmic note-values, a triplet clock will pulse three times for every two regular pulses of the same note-value, while a dotted-note clock will pulse twice for every three regular note-value pulses.

Clock Out jack- Outputs 5V clock pulses for syncing other modules in Voltage Modular. Often this will be patched to the external clock input of a sequencer but can be used for any number of things including advancing switches, resetting LFOs, and triggering sample and hold modules.

115 Sync Generator

The Sync Generator module generates 96-pulse-per-quarter-note (PPQN) sync signals with adjustable rate, tap tempo, and CV control of tempo. When using Voltage Modular as a stand-alone instrument, this module can be used to create a master sync-signal at a specified BPM. The *Sync Out* jack will typically be patched to the *Sync In* jack of a Sync Divider module which “slows” the super-fast sync signal down to a note-value clock signal that can be used to advance sequencers and switches, reset LFOs, trigger sample-and-hold modules etc.

While this module can be used as a rock-solid sync generator to keep everything perfectly in time, the *Tap Tempo* button allows the BPM to be changed “on the fly” and the *Rate CV* jack can be used in creative ways to introduce variation into the sync signal if desirable.

115.1 Inputs, Outputs and Controls

Reset jack- A 5V pulse or gate received at this jack will immediately force-reset the sync signal. It's important to reset the Sync Generator at the same time as other modules so that everything starts at the same instant. Typically this will be connected to the *Play Trig* jack of the "master" sequencer whose play and stop buttons are being used.

Rate CV jack and attenuator- CV input and attenuator for externally controlling the rate of the sync signal.

Rate- Sets the tempo of the sync signal from 1 to 450 BPM (beats-per-minute).

Tap Tempo- Allows the rate to be set by “tapping” the tempo manually. The tempo is set based on the time between two consecutive clicks of the button.

Sync Out jack- Outputs a 96-pulse-per-quarter-note (PPQN) sync signal. Typically this is patched to the *Sync In* jack of one or more Sync Dividers to create clock signals for advancing sequencers, switches etc.

116 Sync To MIDI Clock

The Sync To MIDI Clock module converts a Sync signal to MIDI Clock messages for syncing external MIDI devices such as drum machines, synths, and sequencers to your host DAW's tempo when using Voltage Modular as a plug-in instrument.

The Transport In section's *Stop* and *Play* jacks can be patched to the Transport output jacks in the I/O Panel to send MIDI Stop and MIDI Start messages to external devices each time your DAW is stopped and started.

116.1 Inputs and Outputs

Sync In- Patch this to the *Sync Out* jack in the Transport section of the I/O Panel to receive a sync signal at the tempo of your host DAW.

Transport In Stop jack- Patch this to the *Stop* jack in the Transport section of the I/O Panel to send a MIDI Stop message to external devices each time the Stop button in your DAW is pressed.

Transport In Play jack- Patch this to the *Play* jack in the Transport section of the I/O Panel to send a MIDI Start message to external devices each time the Play button in your DAW is pressed.

MIDI Clk Out- Patch this to the *MIDI In* jack of a MIDI Out module to send MIDI Clock (including MIDI Start and Stop messages) to an external device.

117 Synth Voice

Synth Voice is a powerful, self-contained monophonic synthesizer. It's inspired by the classic ARP 2600 synthesizer, and like the 2600, features "semi-normalled" audio and modulation paths that enable fast and often wild sound creation. It includes two oscillators, a multimode filter, ADSR and AR envelope generators, a low-frequency oscillator, sample and hold, ring modulation, and a simulated spring reverb. Because it's part of Voltage Modular, any and all of its individual sections can be combined with other modules for maximum creativity.

If you don't want to read this entire guide, please have a look at the *"Semi-Normalled Patching*" section below, because Synth Voice's patching works a little differently than most Voltage Modular modules.

117.1 Semi-Normalled Patching

If you've worked with cable patch bays in recording studios, you may be familiar with the concept of semi-normalled connections. This means that certain connections are invisibly patched by default if no cable is plugged into a jack, but plugging a cable into the jack interrupts the "normal" signal flow and replaces it with the patched cable. The orange boxes along the bottom of the Synth Voice panel indicate what the source is if nothing is plugged into the the jack above. In other words, **if nothing is plugged into the jack, it's as if the jack isn't there. If an audio or control signal is plugged into the jack, the normalled source is overridden, effectively disconnecting it**.

This applies to Synth Voice in two ways:

External IO Panel Normalled Connections- The following output jacks in the IO Panel CV SOURCES section are normalled to modulation input jacks at the bottom of the Synth Voice panel.

IO PANEL - CV SOURCES OUTPUT	NORMALLED MOD DESTINATIONS
Pitch	VCO 1 Pitch, VCO 2 Pitch, VCF Pitch
Gate	ADSR Gate, AR Gate (Trig)
Trig	AR Gate (Trig) [if AR Trig switch is enabled]
Mod Wheel	Low Frequency Oscillator depth [if Mod Whl switch is enabled]

Plugging a jack into one of the normalled connections at the IO Panel does not disconnect it from Synth Voice. For example, you could patch the IO Panel *Pitch* and *Gate* outputs to other modules without interrupting the normalled connection to Synth Voice.

Internal Synth Voice Normalled Connections- All of the orange boxes+jacks along the bottom of the panel, as well as the Sample and Hold section *Noise Gen* are normalled connections, with the default source indicated in the orange box. **Plugging a cable into the associated Synth Voice jack disconnects the normalled source and replaces it with the patched cable**.

Almost all of these have an attenuator slider directly above them affecting the amount of either the normalled signal or the currently patched cable.

117.2 Voltage-Controlled Oscillators 1 & 2

Synth Voice includes two super-wide range oscillators that accurately model the imperfect waveforms of vintage ARP synthesizers.

Initial Freq- Sets the basic pitch of the oscillator, displayed in traditional organ footage. *Lo* will be beneath the audible range and allows the oscillator to be used as a mod source.

Fine Tune- This can be used to fatten up two-oscillator patches by detuning a small amount, or for "building-in" a set interval. Its range is a smidge over a fifth, up or down.

Waveform Output Jacks- Voltage-Controlled Oscillator 1 (VCO 1) includes triangle, ramp, sine, and pulse waves. VCO 2 includes triangle, saw, sine, pulse waves, and sub-octave square- this a 50% square wave one octave below the other VCO 2 waves. It is not affected by pulse width settings.

Like a vintage ARP 2600, the oscillators do not have switches for wave selection. Instead, VCO 1's sawtooth wave and VCO 2's pulse wave are normalled to the VCF audio ins and gain sliders (at the bottom left of the VCF section). To select a different oscillator wave, patch a cable from the appropriate VCO wave output jack to any of the VCF *Audio* input jacks. This overrides the "preset" normalled routing.

Note that VCO 1 and VCO 2's ramp and saw waves sound exactly the same at audio-range frequencies, but the reversed shapes are useful for rising or falling modulation when the oscillators are set to *Lo* initial frequency settings and used as mod sources.

Sync *(VCO 2 only)*- Feeding a wave or signal to this force resets the start of the waveform to the beginning of its cycle. Most often used to create the "sync sweep" oscillator sounds made famous in The Cars' "Let's Go" (or Kraftwerk's "Neon Lights" and No Doubt's "Just A Girl"), by routing the output of another oscillator to the *Hard Sync* input and sweeping the pitch of the first oscillator.

Hard Sync is also useful when creating drum and percussion sounds to ensure that the wave starts precisely at the beginning of its cycle.

FM Control/PWM Semi-Normalled Inputs- Mod inputs at the bottom of the panel. The slider control above each jack is an attenuator, affecting the amount of the normalled source, or if a cable is plugged into the jack, the amount of the signal from the patched source.

The table below shows the default sources and destinations for VCO 1 and VCO 2:

MOD SOURCE NAME	NORMALLED SOURCE	NORMALLED DESTINATION
VCO 1 - Pitch CV	IO Panel CV Sources/Pitch	VCO 1 frequency - no attenuation slider
VCO 1 - ADSR	Synth Voice ADSR envelope generator	VCO 1 frequency
VCO 1 - VCO 2 Sine	Synth Voice VCO 2 sine wave output	VCO 1 frequency
VCO 1 - LFO	Synth Voice LFO output	VCO 1 pulse width
VCO 2 - Pitch CV	IO Panel CV Sources/Pitch	VCO 2 frequency - no attenuation
VCO 2 - AR	Synth Voice AR envelope generator	VCO 2 frequency
VCO 2 - VCO 1 Pulse	Synth Voice VCO 1 pulse wave output	VCO 2 frequency
VCO 2 - LFO	Synth Voice LFO output	VCO 2 pulse width

Glide- Also known as "portamento," glide delays the voltage change between pitches for a sliding effect. Synth Voice includes separate *Glide* controls for each oscillator - applying glide to only one oscillator is nice effect.

Pitch CV In- This switch connects or disconnects the normalled IO Panel *Pitch* CV connection. This defaults to on position, for standard keyboard (or sequencer) playing. Turning it off is useful for drones or sound effects when you don't want oscillator pitch to be affected by keyboard CV.

Though the *Glide* slider is situated above the normalled *Pitch* source and jack, it is **not** an attenuator for it, but it is in the signal path, so it will affect the normalled pitch CV or any signal plugged into the *Pitch* FM jack.

PW Initial- This sets the width or "duty-cycle" of the pulse wave. It has no effect on any other waveform. This defaults to 50%, i.e., a perfect square wave. Moving the slider up or down narrows its width as well as the thickness of sound until it almost disappears at its extremes.

117.3 Voltage-Controlled Filter

Synth Voice's filter section models the early "ladder"style filter used in early ARP synthesizers. It can be switched between lowpass and highpass modes, with 12- or 24-db per octave slopes.

If you're not familiar with how filters work, a lowpass filter allows frequencies *below* the cutoff frequency setting to pass through, but blocks frequencies *above* the cutoff frequency. A highpass filter does opposite: it allows frequencies *above* the cutoff frequency setting to pass through, but blocks frequencies *below* the cutoff frequency. In practice, this means a lowpass filter is useful for removing high frequencies, and a highpass filter is useful for removing low frequencies. Modulating the cutoff frequencies via envelope generators, low-frequency oscillators, and more opens the door to endless sound possibilities.

Cutoff Freq- Sets the frequency where attenuation begins. Attenuation occurs above (lowpass mode) or below this frequency (highpass mode).

Resonance- Emphasizes sound energy at and around the current cutoff frequency by adding feedback from the filter's output back to its input. At lower settings, this can be used to create mild resonances such as those heard in acoustic instruments. At more extreme settings, resonance can create a pure sine wave at its own frequency (you can adjust its pitch with the *Cutoff* knob). Be careful with this knob as it can get loud at extreme settings.

Mode- The nature of how a filter works is that frequencies "fall off" above or below the cutoff frequency. Slope adjusts the steepness of this slope. Set to lowpass mode, a 12db per/octave filter has a shallower slope, whereas a 24db per/octave filter has a steeper slope (as well as more pronounced character with the resonance knob turned up).

VCF Out jack- The direct audio out of the VCF section.

Audio and Control Semi-Normalled Inputs- These are the audio and frequency modulation inputs at the bottom of the filter section. The slider control above each jack is an attenuator, affecting the amount of the normalled source, or if a cable is plugged into the jack, the amount of the signal from the patched source.

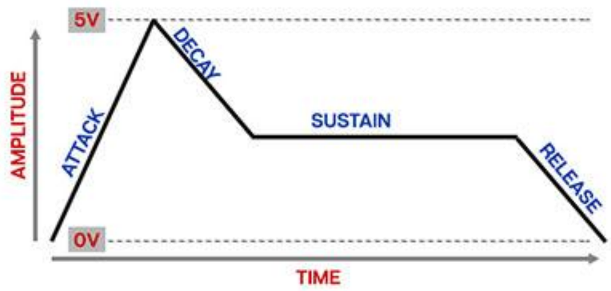
The table below shows the filter's default sources and destinations. The *Audio* section inputs effectively function as a mixer for Synth Voice's oscillators, noise generator, or any other audio sources patched to the semi-normalled inputs.

AUDIO/MOD SOURCE NAME	NORMALLED SOURCE	NORMALLED DESTINATION
Audio - VCO 1 Saw	Synth Voice VCO 1 saw wave	VCF audio input
Audio - VCO 2 Pulse	Synth Voice VCO 2 pulse wave	VCF audio input
Audio - Noise Gen	Synth Voice white noise generator	VCF audio input
Control - Pitch CV	IO Panel CV Sources/Pitch	VCF cutoff frequency
Control - ADSR	Synth Voice ADSR envelope generator	VCF cutoff frequency
Control - LFO	Synth Voice LFO output	VCF cutoff frequency

117.4 ADSR Envelope Generator

A standard "ADSR"-style envelope generator most often used to shape amplitude or filter curves. If you're not familiar with the operation of envelope generators, here's an overview:

When a gate voltage is received, the envelope generator outputs a voltage that changes dynamically according to the settings of its four stages.



The *Attack* stage defines how long it takes for the output voltage to rise from 0 to 5 volts. Once the attack stage reaches 5V, it moves to the *Decay* phase, which defines how long it takes to fall from 5V to the setting of the *Sustain* phase. Unlike the *Attack*, *Decay*, and *Release* phases, each of which define a time, *Sustain* simply sets the held voltage level following the *Attack* and *Decay* phases - this usually equates to the envelope output level while holding down a key on a keyboard controller. Finally, the *Release* knob defines the the length of time it takes for the voltage to fall back to 0V when the gate input voltage is removed (typically when you let go of a key on a keyboard controller).

Keep in mind that the envelope generator moves through its attack and decay segments and holds at the sustain level as long a 5V gate voltage is being received. It moves to the release segment when the gate voltage is removed (i.e., 0V gate voltage).

Its controls are as follow:

"A" (Attack) slider- Defines the length of time for voltage to rise from 0V to 5V when the gate voltage is applied.

"D" (Decay) slider- Defines the length of time for voltage to fall from the *Attack* stage 5V peak to *Sustain* stage setting.

"S" (Sustain) slider- Sets the held voltage level following *Attack* and *Decay* phases.

"R" (Release) slider- Defines the length of time for voltage to fall from *Sustain* level to 0V when gate is released.

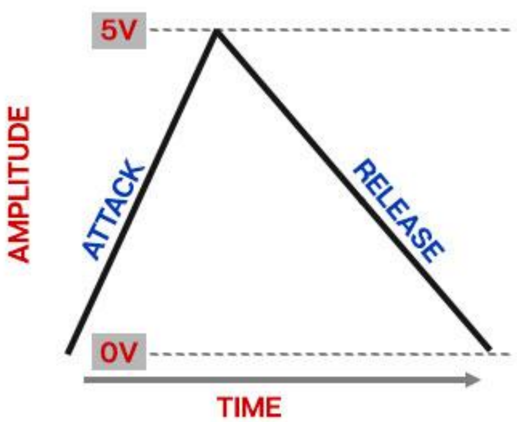
The table below shows the default mod destination:

MOD SOURCE NAME	NORMALLED SOURCE	NORMALLED DESTINATION
Control - Gate	IO Panel CV Sources/Gate	ADSR Envelope Generator

ADSR Out jack- The envelope generator's output voltage.

117.5 AR Envelope Generator

A simple envelope generator with attack and release segments only. It may seem "stripped down," but AR generators are very useful for basic envelope applications, particularly drum and percussion sounds (which Synth Voice excels at).



The AR envelope functions a little differently than the ADSR envelope. If the *Sustain* switch is on, it moves through the attack segment when a 5V gate is applied and remains at maximum as long as the gate voltage is present (equivalent to an ADSR envelope with its sustain control set to maximum).

If the *Sustain* switch is off, it moves through the release segment regardless of whether the 5V gate voltage is present.

Its controls are as follow:

"A" (Attack) slider- Defines the length of time for voltage to rise from 0V to 5V when the gate voltage is applied.

"R" (Release) slider- Defines the length of time for voltage to fall to 0V following the attack segment (*Sustain* switch off) or when gate voltage is removed ((*Sustain* switch on).

Trig- Enabling this causes the AR envelope to move through both of its segments regardless of the gate voltage length. This means trigger signals (i.e. extremely short gates) can be used to run through both stages of the AR envelope. The *(Trig)* label on the orange mod input legend refers to the gate behavior when the *Trig* switch is enabled.

Sustain- Adds a 5V sustain segment following the attack segment for as long as gate voltage is high. The release segment occurs when the gate voltage is removed.

Manual Trigger- Sends a gate signal to both the ADSR and AR envelope for as long as the button is held. This can be useful for testing sounds or playing on the fly.

The table below shows the default mod destination:

MOD SOURCE NAME	NORMALLED SOURCE	NORMALLED DESTINATION
Control - Gate (Trig)	IO Panel CV Sources/Gate	AR Envelope Generator

AR Out jack- The envelope generator's output voltage.

117.6 Voltage Controlled Amplifier

You can think of the VCA as a "gate" to start, stop, and shape the volume of audio or control signals. Applying a simple 5V gate voltage will abruptly open and close the VCA; the CV output of an envelope generator lets you shape audio and control signals with more finesse.

Initial Gain- Sets the static gain of the amplifier; a good analogy would be opening a faucet. For standard instrument-type sounds, you'll likely leave this set to minimum, as the envelope generators control signals will open the amp when notes are played. Turning up the *Initial Gain* is useful for "hands-off" droning sounds.

VCA Out jack- The output of the VCA. The LED next to the jack lights when a signal is present.

This *VCA Out* jack is Synth Voice's master out, and is not normalled to the IO Panel. You'll need to patch a cable from this output to one of the IO Panel Main or Aux Out jacks (or a mixer module patched to an IO Panel out) to hear sound.

AUDIO/MOD SOURCE NAME	NORMALLED SOURCE	NORMALLED DESTINATION
Audio - VCF output	Synth Voice VCF output	VCA audio input
Audio - Ring Mod	Synth Voice VCO 1+VCO 2 ring mod	VCA audio input
Control - ADSR	Synth Voice ADSR envelope generator	VCA amplitude
Control - AR	Synth Voice AR envelope generator	VCA amplitude

117.7 Ring Modulation

The Ring Mod input multiplies both oscillator signals by one another, resulting in a signal containing only the sum and difference of the two signals and not the original signals themselves. This often results in tones with unrelated harmonics which can sound harsh or out of tune, but when dialed in carefully can create sounds and timbres hard to create with other methods. For most dramatic effect, try detuning VCO 1 and VCO 2 using the *Initial Freq* and *Fine Tune* controls.

Ring mod is sourced from the *VCO 1* and *VCO 2* jacks in at the bottom of the filter section. If nothing is plugged into these jacks, this means the ring mod section receives the VCO 1 sawtooth wave and VCO 2 pulse wave. To send different waves to the ring mod section, patch the desired waves into the VCO 1 and VCO 2 inputs *Audio* inputs in the VCF section, just as you would for "normal" alternate wave selection. (sine waves from both oscillators sound particularly good with ring mod).

117.8 Low Frequency Oscillator

A low frequency oscillator or "LFO," is similar to an audio oscillator, but it typically runs at frequencies below the range of human hearing, and it isn't used an audio source. Instead, its cycling waveforms are used as a control source to add modulation to oscillators, filters, and amplifiers.

A typical application would be to slightly modulate an oscillator's frequency to create vibrato (pitch modulation), or to modulate a VCA's amplitude to create a tremolo effect (amplitude modulation). Modulating the cutoff frequency of a filter can create a dubstep-style wobble, or if modulated very slowly, long sweeping tonal shifts.

Frequency- Sets the rate of the LFO from 0.01 Hz-20 Hz. This guy can get REALLY slow.

Mod Wheel switch- Turning this on normals the IO Panel *CV Sources/Mod Wheel* output to LFO depth control. This makes setting up Synth Voice for pitch modulation with a mod wheel easy - turn the switch on, patch a cable from the LFO Out to one or both of the oscillator FM mod jacks, and push up the corresponding FM slider. (Oscillator Pulse Width or VCF mod is even easier, because these have normalled internal routings; simply push up the appropriate fader in the VCO or VCF sections). If the Mod Wheel switch is on and your controller's mod wheel is at zero, the LFO won't have any signal output, thus... **if the LFO doesn't seem to be working, make sure the *Mod Wheel* switch is in the off position (or push up your controller's mod wheel)**.

Waveform Selector- Switches the LFO's output to triangle or square wave. The triangle wave is bipolar (i.e. its voltage sweeps above and below 0 volts); this works well for pitch modulation because the pitch rises and falls above and below the center frequency. The square wave is unipolar, that is, it only goes up then returns to zero; this makes it easy to set up pitch trills or octaves. (Setting up a bipolar square wave for alternating octave pitch modulation is difficult because the high note gets higher and low note gets lower as mod depth is increased.)

Out jack- CV output of the LFO. You may not need it, because the LFO CV out is normalled to VCO 1 and VCO 2's pulse width controls, as well as VCF frequency mod, but patching a cable from the jack lets you select other destinations such as VCO pitch, VCA amplitude, or external modules. The LED next to the jack gives a visual indication of the LFO rate.

Sample and Hold- The sample and hold section repeatedly "samples" the input signal voltage, and continuously outputs that voltage until it is triggered to sample again. This duration between samples is adjustable with the *Frequency* slider source. Sampling can be triggered externally with a CV or audio signal. Note that when we say "sample," we're referring to a simple incoming voltage - this isn't an audio sampler that you can sample and playback beats with. (Though this is the basic principle of how digital audio samplers operate).

In the image to the left, the smooth gray line shows a continuous input signal. Each time the module is triggered, the current voltage is "sampled" and "held" until the next trigger. The red line shows the stepped output signal.

Frequency- Sets how often the incoming signal is sampled. This could also be referred to as the sample or clock rate. Unlike many hardware analog synths with sample and hold, Synth Voice's sample frequency isn't tied to the LFO section; the sample and hold section has its own independent internal LFO clock.

Though it's not at the bottom of the panel with the rest of the normalled connections, the *Noise Gen+In* jack is a normalled connection so here's another swell table:

AUDIO/MOD SOURCE NAME	NORMALLED SOURCE	NORMALLED DESTINATION
Noise Gen	Synth Voice white noise generator	sample and hold sample source

White noise is commonly used as a sample source with sample and hold modules because it creates completely random voltages. If applied to oscillator pitch, this results in note sequences with random pitches. Filter cutoff frequency modulation is another common use (one famous example is the percolating note sequence at the beginning of Emerson, Lake, and Palmer's "Karn Evil #9," better known as "welcome back my friends, to the show that never ends..."). LFO waves can also be used as signal sources - oscillator ramp and sawtooth waves at sub-audio rates can be used to create interesting rising and falling modulation patterns.

Out jack- CV output of the sample and hold. The sample and hold's output isn't normalled to any destinations, so you'll need to patch this to the desired mod destination with a cable.The LED next to the output jack flashes at the current sample frequency.

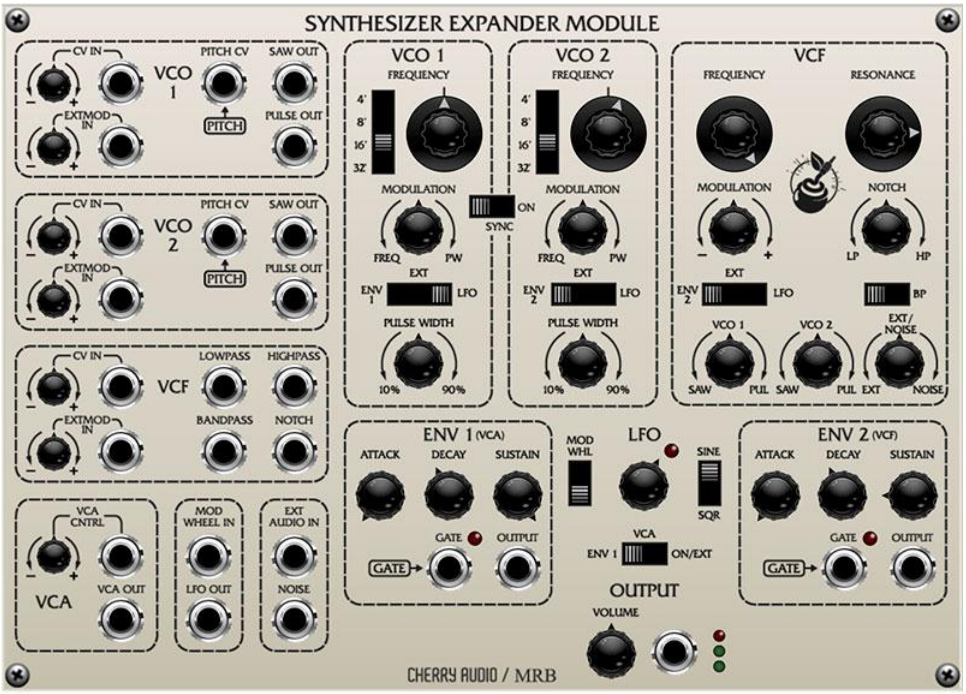
117.9 Master Section

Tune- The master tuning knob affects both oscillators simultaneously and has a range of a half-step up or down.

Reverb- This is an accurately modeled spring reverb. Like a real spring reverb, certain frequencies will excite and "ring" in funny ways sometimes. In other words, it isn't intended to precisely replicate the acoustics of The Royal Albert Hall. Note that dry signal remains constant; the knob mixes the reverberated signal from 0-200%, allowing really wet sounds.

Volume- This is post-VCA, and regulates the volume of all sound coming from Synth Voice.

118 Synthesizer Expander Module



The Voltage Modular Synthesizer Expander Module is an emulation of the classic Oberheim Synthesizer Expander Module, aka the "SEM." Originally released in 1974, the keyboardless, mono SEM module was intended as a companion to the Oberheim DS-2, one of the earliest digital sequencers. Soon thereafter, Oberheim realized they could interface a digitally scanned keyboard, mount the whole mess in a big box, and create polyphonic synthesizers, beginning with the Two Voice, followed by the Four Voice, and finally the beastly Eight Voice (be sure to check out **Cherry Audio Eight Voice**, our super-rad virtual instrument emulation).

Though it was a simple, barebones monosynth, the SEM sounded fantastic, and had a tone quality very different than the common fuzzy, fat Moog sound, thanks to its 12 dB/oct state-variable filter. With lowpass, bandpass, highpass and notch modes, this flexible filter was the star of the show. We've precisely recreated it here with a detailed emulation programmed by award-winning synth designer, Mark Barton (MRB).

The design is similar to the "patch panel" reissue SEM's released in the 2000's; these were basically identical to the 70s versions but added an extensive patch panel for modular synth flexibility.

118.1 Semi-Normalled Patching

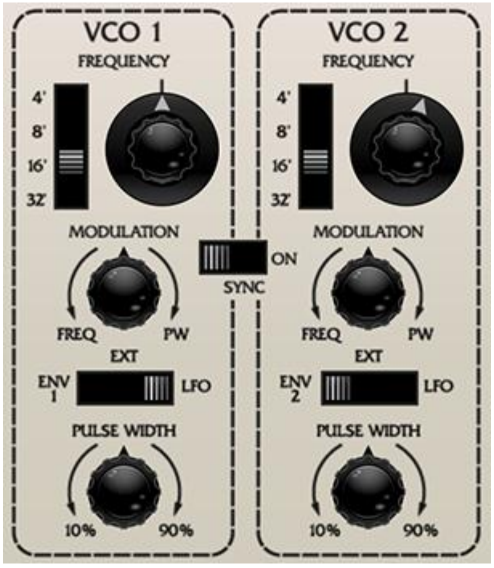
The Voltage Synthesizer Expander Module would best be described as a normalled semi-modular synth. Unlike most Voltage modules, it contains standard synth sections (i.e. oscillators, filters, etc.) that are internally patched together. Additionally, the Voltage IO Panel *Pitch* and *Gate* connections are "normalled," i.e. invisibly connected to Synthesizer

Expander Module's *Pitch CV* and *Gate* jacks. In fact, the only connection that needs to be made to begin playing is the *Output* jack to the IO Panel *Main Outs To Host*.

Any cables patched to the normalled jacks will override (i.e. disable) the normalled IO Panel connections.

Finally, normalled connections are indicated on the front panel by inverted (framed) boxes with arrows.

118.2 VCO 1 and 2



Synthesizer Expander Module includes two almost identical voltage-controlled oscillators. The only difference are their modulation routing options.

Range- Sets the pitch range for each oscillator in octaves. These are at standard organ footage settings of 32', 16', 8', and 4'.

Frequency knob- This can be used to fatten up two oscillator patches by detuning a small amount, or for "building-in" a set interval. Its range is a smidge over a fifth, up or down.

Modulation amount knob and source switch- The mod amount knob is bidirectional; rotating it left from center position adds pitch modulation, rotating it right modulates pulse width of the pulse wave. Center position is off, i.e. no modulation.

The three-position slide switch selects the oscillator mod source from three sources:

- **Env 1 (VCO 1) / Env 2 (VCO 2)**- Modulation source is envelope 1 or envelope 2. Selecting the envelopes as mod source doesn't "disconnect" them from the VCA or VCF.
- **Ext**- Enables CV mod from the *Extmod In* jack and attenuator. The attenuator is bipolar - center position is zero; turn right for positive CV or left for inverted (negative) CV values.
- **LFO**- Enables mod from the onboard LFO.

Pulse Width- Sets the width or "duty-cycle" of the pulse wave. It has no effect on the saw wave. This defaults to 50%, i.e., a perfect square wave. Moving the knob left or right narrows its width as well as the thickness of sound until it almost disappears at its extremes.

Sync- This causes VCO 1 to force reset the start of VCO 2's waveform to the beginning of its cycle, resulting in a wide range of harmonic tones from VCO 2. The range of tones can be varied by adjusting VCO 2's *Frequency* controls.

***Choosing oscillator waveforms**- One unusual aspect of the Oberheim SEM design is that the oscillators themselves contain no waveform controls. Instead, the level of saw and pulse waves is adjusted via mixer knobs at the bottom of the VCF section.*

118.2.1 VCO 1 and 2 Patch Panel

Allows control of VCO frequency mod via patch cables routed from other modules, or the Synthesizer Expander Module itself, as well as separate wave outputs. All attenuator knobs are bipolar - center position is zero; turn right for positive CV or left for inverted (negative) CV values.

CV In jack and attenuator- Allows CV control of oscillator frequency.

Ext Mod In jack and attenuator- Allows CV control of oscillator frequency. **Only active when the three-position modulation source switches in the VCO 1 and 2 sections are set to *Ext*.**

If you need more mod inputs, remember that Voltage Modular allows an unlimited number of cables to be plugged into a single jack, or alternatively, you could mix all the mod sources with mixer module.

Pitch CV in jack and attenuator- Allows CV control of oscillator frequency. These are active at all times and are unaffected by the three-position modulation source switches in the VCO 1 and 2 sections. These are 1V/oct inputs, intended for half-step pitch control input from a keyboard controller or sequencer. If nothing is plugged in, the IO Panel *Pitch* output is normalled to the *Pitch CV* jack for pre-routed keyboard controller; plugging a jack into *Pitch CV* will disable the normalled routing.

The framed *Pitch* box indicates the normalled connection.

Saw Out jack- Direct out of the saw wave. This comes before the filter and amplifier stages.

Pulse Out jack- Direct out of the pulse wave. This comes before the filter and amplifier stages.

118.3 VCF

The Voltage Synthesizer Expander Module filter section represents a departure from the 24 dB "ladder" style filter often seen in vintage synths. Like the Oberheim SEM module it's based on, Eight Voice uses a 12 dB state-variable filter (no, this doesn't mean it can sound like Rhode Island or Montana).

This refers to its curves - it can function as a lowpass, bandpass, or highpass filter and features a knob allowing a continuous sweep from lowpass to highpass response (with "notch" filtering in the middle position).

This gives it a great deal of flexibility, and the 12 dB curve gives it a brighter overall tonality than a typical ladder filter.

If you're not familiar with how filters work, a lowpass filter allows frequencies *below* the cutoff frequency setting to pass through, but blocks frequencies *above* the cutoff frequency. Highpass is the opposite of lowpass mode: high-frequency content remains, but low frequencies are removed as the cutoff frequency increases. Sliding the *Bandpass* switch enables Bandpass mode combining both lowpass and highpass modes, leaving sound only "in the middle." The cutoff frequency lies roughly halfway between the falloff on each side.

A notch filter does the opposite - it removes a middle piece of the audio spectrum but leaves other frequencies intact. (That might not sound useful, but setting the filter to *Notch* mode and slowly sweeping the cutoff frequency with an LFO creates nifty phaser-like tones - great for imitating vintage string synths.)

Frequency- Sets the frequency where frequency attenuation begins with its effect dependent upon the currently chosen lowpass/bandpass/notch/highpass/etc. filter mode.

Resonance- Emphasizes sound energy at and around the cutoff frequency by adding feedback from the filter's output back to its input. This is useful for creating commonly heard synth "wah" tones, especially when the cutoff frequency is modulated with an envelope generator or one of the LFO's.

Modulation amount knob and source switch- Applies modulation to the filter cutoff frequency. The mod amount knob is bidirectional; rotating it right from center position adds positive modulation, rotating it left adds negative modulation. Center position is off, i.e. no modulation.

The three-position slide switch selects the filter mod source from three sources:

- **Env 1 (VCO 1) / Env 2 (VCO 2)**- Modulation source is envelope 1 or envelope 2. Selecting the envelopes as mod source doesn't "disconnect" them from the VCA or VCF.
- **Ext**- Enables CV mod from the *Extmod In* jack and attenuator. The attenuator is bipolar - center position is zero; turn right for positive CV or left for inverted (negative) CV values.
- **LFO**- Enables mod from the onboard LFO.

LP>HP/Notch control- Continuously varies the filter response from lowpass to highpass as the knob is rotated from left to right. The middle position creates a notch response.

BP (bandpass) switch- Alters the filter response to bandpass when engaged (i.e. combination of lowpass and highpass leaving frequencies "in the middle." The LP>HP knob disappears when the BP switch is enabled.

VCO 1 / VCO 2 Saw/Pulse level- The *VCO 1* and *VCO 2* knobs adjust the volumes of the saw and pulse waves for VCO 1 and 2, respectively. These are bidirectional knobs as well - rotating them left from center position increase the saw wave level, rotating it right increase the volume of the pulse wave. Center position is off (**if you're getting no sound, check these first**).

Ext/Noise level- When rotated to the left, this sets the level of signals plugged into the patch panel *Ext Audio In* panel jack, when rotated right, it sets the level of the onboard pink noise generator.

118.3.1 VCF Patch Panel

Allows control of VCF frequency mod via patch cables routed from other modules, or the Synthesizer Expander Module itself as well as outputs for all four filter responses. All attenuator knobs are bipolar - center position is zero; turn right for positive CV or left for inverted (negative) CV values.

CV In jack and attenuator- Allows CV control of filter cutoff frequency.

Ext Mod In jack and attenuator- Allows CV control of filter cutoff frequency. **Only active when the three-position modulation source switches in the VCF section is set to Ext.**

Lowpass / Highpass / Bandpass / Notch Out jacks- Separate outputs for each of filter responses. These are all available simultaneously - try routing a couple or all of them to a mixer, and play with their levels and the cutoff frequencies for all manner of awesome formant tonalities.

118.4 Envelope 1 and 2

The original SEM modules included two attack/decay/sustain (ADS) envelope generators. These function much the same as more common attack/decay/sustain/release (ADSR) envelopes, the only difference is that the decay and release stages are combined into a single control.

118.4.1 How They Work

When a voice sees a gate voltage from a note, the envelope outputs a dynamically changing voltage, according to the settings of its stages. The *Attack* stage defines how long it takes for the output voltage to rise from 0 to 5 volts. Once the *Attack* stage reaches 5V, it moves to the *Decay* phase. If the key is released, the *Decay* knob defines how it long takes for the voltage to fall back to zero. If the key is held, the *Decay* time defines how long it takes to fall to the *Sustain* level setting. The note then holds at the *Sustain* level until it is released, and fades to zero at the time set by *Decay* knob - the *Decay* knob effectively does "double-duty," acting as a decay and a release control.

118.4.2 Envelope Controls

Attack- Defines the length of time for voltage to rise from 0V to 5V when a key is played.

Decay- If the key is released, the *Decay* knob defines how it long takes for the voltage to fall back to zero. If the key is held, the *Decay* time defines how long it takes to fall to the *Sustain* level setting.

Sustain- Sets the voltage (i.e. the level) the envelope holds at following the *Attack* and *Decay* phases.

118.4.3 Envelope 1 and 2 Routing

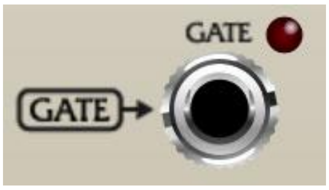
As indicated on the front panel, Env 1 generally affects the VCA, i.e., amplitude. It's hard-wired to the VCA, so no other routing is necessary. If VCO 1's three-way mod routing switch is set to the *Env 1* position, it will also modulate VCO 1's pitch.

Env 2 is mainly intended to modulate VCF cutoff frequency, but because it's not hard-wired, VCO 2's three-way mod routing switch needs to be in the *Env 2* position for cutoff mod to occur. In addition to filter cutoff frequency, Env 2 can also be used to modulate VCO 2's pitch when VCO 2's three-way mod routing switch is set to the *Env 2* position.

118.4.4 Envelope 1 and 2 Patching

Unlike the left-side CV inputs, these are in the same sections as the controls (space considerations!).

Gate in jacks and LED- This is where you'll patch gate voltages to initiate the envelope generator cycle. Most often this will come from the IO Panel *Gate* output. The LED's next to the gate jacks illuminate when a gate signal is present. If nothing is plugged in, the IO Panel *Gate* output is normalled to the *Gate* jack for pre-routed keyboard controller; plugging a jack into *Pitch CV* will disable the normalled routing.



The framed *Gate* box indicates the normalled connection.

Output- Envelope CV signal outputs.

118.5 VCA

VCA switch- Synthesizer Expander Module's VCA has only one control, but it's important to understand. It's located in the middle, directly above the master *Output* knob. When set to *Env 1* position, amplitude is controlled by envelope 1; this where you'll generally leave it when playing with a keyboard or sequencer controller.

The *On/Ext* position latches the VCA open; this useful for drones or when using the *Ext Audio In* jack to process signals with the filter (this way you won't need to hold a key down to hear sound).

118.5.1 VCA Patch Panel

Allows control of VCA amplitude via CV's routed from other modules, or the Synthesizer Expander Module itself, and also includes an output. The attenuator knobs is bipolar - center position is zero; turn right for positive CV or left for inverted (negative) CV values.

VCA Control In jack and attenuator- Allows CV control of the amplitude level.

VCA Out- This is a VCA signal out. It's essentially the same as the master *Output* jack, but with no volume control.

118.6 LFO

The LFO generates sub-audio range signals intended for modulation purposes.

Mod Wheel switch- Turning this on normals the IO Panel *CV Sources/Mod Wheel* output to LFO depth control. This makes setting up the LFO with a mod wheel super easy. If the Mod Wheel switch is on and your controller's mod wheel is at zero, the LFO won't have any signal output, thus... **if the LFO doesn't seem to be working, make sure the *Mod Wheel* switch is in the off position (or push up your controller's mod wheel)**.

Frequency (*control unlabeled*)- The *Rate* knob sets the speed of the LFO, from 0.08 to 15 Hz (with *Sync* switch off) or from 8 beats up to 1/64th note triplets (*Sync* switch on). The LED beside it flashes at the current rate.

Wave Select- Chooses between sine and square waves.

118.6.1 LFO Patch Panel

Mod Wheel In jack- The mod wheel is used to vary the depth of the LFO. The IO Panel *Mod Wheel* jack is normalled to LFO depth when the LFO section *Mod Whl* switch is in the up position; patching a cable to the Mod Wheel In jack overrides the IO Panel connection. Remember that the *Mod Wheel In* jack can accept a CV from any source, not just the mod wheel.

LFO Out jack- CV output of the LFO. You may not need it, because the LFO CV out is normalled to VCO 1 and VCO 2's mod routing switch, but patching a cable from the jack lets you route to other desinations in Synthesizer Expander or to external modules.

118.6.2 Ext Audio In and Noise Out Patch Panel Jacks

Ext Audio In- Routes audio to the filter inputs. To hear external audio, the *Ext/Noise* knob in the VCF section needs to dialed toward *Ext*.

The *VCA* switch (beneath the LFO section) is important when using the *Ext Audio In* jack. When set to *Env 1*, you'll only hear the external audio signal when a key is played (or the gate signal to Env 1 is high). Setting the *VCA* switch to *On/Ext* latches the VCA open so that external signals are always audible. This is useful when using the Synthesizer Expander's filter to process other signals such as drum loops, etc.

Noise- Direct output for the onboard pink noise generator.

118.7 Output Section

Volume, output jack, and meter- This is post-VCA, and is the final module output with volume control. The red meter LED will glow when things are getting too hot. Note that this output is essentially the same as the output in the VCA patch panel section, but with a volume knob.

119 TB Oscillator

TB Oscillator is a super-accurate modular recreation of the oscillator section of the simple-but-beefy oscillator of the world's most iconic acid techno bass synth. Like many vintage oscillators, its waveforms aren't totally accurate on an oscilloscope, which gives them a unique character. In addition to its standard controls, the TB Oscillator's W*ave Mix* control adds new tone colors.

119.1 Inputs, Outputs, and Controls

Keyb CV jack- Accepts a CV input for pitch. Typically this would come from the *Pitch* jack in the IO Panel *CV Out* section, or from a sequencer pitch CV out.

Range- Sets the basic pitch of the oscillator, displayed in traditional organ footage, i.e. larger number equals lower pitch.

Tuning- Fine tune control for pitch, up or down about a fourth.

Freq CV mod input- Allows modulation of oscillator pitch. Note that the attenuator knob is bipolar, i.e. "zero" position is center. Turning right adds a positive modulation, turning left inverts the incoming CV.

Wave Mix and Mix Out jack- Allows blending of the saw and square waves when using the *Mix Out* jack.

Mix CV mod input- CV control input for the *Wave Mix* control.

Waveform out jacks- Output jacks for sawtooth and square waves.

120 Three-Band EQ

The Cherry Audio Three-Band EQ is a straightforward equalizer with 15dB of cut or boost at 150 Hz, 2.5 kHz, and 8 kHz. This is a simple Low/Mid/High EQ useful for changing the tonal balance of an audio signal.

120.1 Inputs, Outputs, and Controls

Input jack- Patch audio signals here.

8 kHz (Highs)- This is a 6dB/Oct high-shelf filter band at 8 kHz with up to 15dB of cut or boost. Turning the knob to the left cuts the "highs" while moving it to the right will boost them.

2.5 kHz (Mids)- This is a 6dB/Oct mid-band peaking filter at 2.5 kHz with up to 15dB of cut or boost. Turning the knob to the left cuts the "mids" while moving it to the right will boost them.

150 Hz (Lows)- This is a 6dB/Oct low-shelf filter band at 150 Hz with up to 15dB of cut or boost. Turning the knob to the left cuts the "lows" while moving it to the right will boost them.

Output jack- Outputs the equalized audio signal.

121 Threshold

The Cherry Audio Threshold module passes voltages received at its input jack to one of two outputs based on whether or not the signal is above or below a specified voltage level. Each output jack also has an affiliated *Gate Out* jack which generates a +5V signal any time it's respective output is passing voltage.

There are many ways to use this module but let’s look at a few fun examples.

- You could patch the output of an Eight-Step Sequencer to the input of a Threshold module to send the low and high notes of a sequence to two different oscillator/envelope/amp setups. The Threshold knob could then be “played” to somewhat randomly change which notes go to which oscillator setup.
- The two *Gate Out* jacks could be patched to the *Stop* and *Start* jacks of a sequencer to have a sequence play only during a portion of a slow envelope or LFO. This could be a fun experiment for a generative patch!
- You can even get really cool results from running audio signals through the Threshold module. Try using a drum loop as the input signal and using the *Over Out* jack to modulate the pitch of an oscillator!

121.1 Inputs, Outputs and Controls

Input jack- This is the input jack for the audio or CV signal that will be tested.

Threshold- Sets the voltage level, between -5V and +5V, that the input signal will be determined to be above or below. This is the setting that defines the "split point" between the two outputs.

Under Out jack- Outputs any voltage received at the input jack that is below the *Threshold*.

Under Gate Out jack- Outputs a +5V gate signal anytime the input voltage is below the *Threshold*.

Over Out jack- Outputs any voltage received at the input jack that is above the *Threshold*.

Over Gate Out jack- Outputs a +5V gate signal anytime the input voltage is above the *Threshold*.

122 Trigger to Gate Converter

The Trigger to Gate Converter module converts short momentary trigger signals to longer gate signals. The gate length can be set between 5 and 5000 milliseconds and is CV controllable.

This could be used, for example, to trigger an Envelope Generator (which only has a gate input) with the trig outs of a sequencer such as the Euclidean Duel or to convert the trig out signal from an Eight-Step Sequencer to a CV-controllable variable-length gate signal.

122.1 Inputs, Outputs, and Controls

Trigger In jack- Patch trigger signals here to create longer gate signals. The small red LED flashes when a trigger is received.

Gate Length- Adjusts the length (from 5 to 5000 ms) of the gate signals created from each trigger.

Length CV jack and attenuator- CV input and bipolar attenuator for externally controlling the *Gate Length*.

Gate Out jack- Outputs a 5V gate signal for each trigger received at the *Trigger In* jack. The small red LED glows when voltage is being output.

123 VCO-20 Dual Oscillator

VCO-20 accurately replicates the tone and functionality of a classic 70s Japanese monosynth, including all waveforms and a white noise source. It also adds CV-controllable pulse width on VCO 1, and hard sync inputs for both oscillators. Because of its unique waveforms (i.e. not-exactly correct), it's the perfect companion for the VCF-20 Filter module.

Since VCO-20 contains two independent oscillators, we'll go over the repeated controls and I/O one time, because you're smart, and we don't like typing!

123.1 Inputs, Outputs, and Controls

Pitch CV jack- Accepts a CV input for pitch. Typically this would come from the *Pitch* jack in the IO Panel *CV Out* section, or from a sequencer pitch CV out. Patching a cable to the *Pitch CV* input on the VCO 1 side automatically connects to the VCO 2 *Pitch CV* input if nothing is plugged into VCO 2's *Pitch CV* input (that's what the horizontal arrow is showing). Patching a cable into VCO 2 *Pitch CV* breaks the normalled connection and lets the *Pitch CV* inputs function independently.

Hard Sync jack- Force resets the start of the waveform to the beginning of its cycle. Most often used to create the "sync sweep" oscillator sounds made famous in The Cars' "Let's Go" (or Kraftwerk's "Neon Lights" and No Doubt's "Just A Girl"), by routing the output of one oscillator to the other's *Hard Sync* input and sweeping the pitch of the first oscillator.

Wave Form- Yes, we incorrectly split the word in half, just like the real synth. This selects the waveform for each oscillator.

VCO 1 pulse position has a variable duty-cycle (i.e. width), adjusted by the *PW* knob. The jaggedy wave in the last position of VCO 1 indicates white noise.

VCO 2 has two fixed pulse waves - square and narrow. The *Ring* position enables ring modulation between the two oscillators. It won't affect the *VCO 1 Out*, but you'll hear it in both the *Mix Out* and VCO 2 Out. It's actually not a true, technically correct ring modulator - the engineers of the original synth used a common-for-the-era method of achieving a very similar effect, and it sounds pretty wicked! It's most audible when using wide pitch spreads between the oscillators and mucking with VCO 2's *Pitch* knob (or modulating via VCO 2's *Freq CV* mod input if you're one of them fancy types).

PW (VCO 1 only)- Sets the width of VCO 1's pulse wave, from a perfect square to a very narrow pulse. It has no effect on other waveforms.

Master Tune (VCO 1 only)- This is situated on the VCO 1 side, but it affects tuning for both oscillators, up or down about a fourth.

Scale- Sets the basic pitch of the oscillator, displayed in traditional organ footage, i.e. larger number equals lower pitch.

Pitch (VCO 2 only)- Detunes VCO 2 independently of VCO 1. The front panel of the original MS-20 is labeled -5 to +5, but its range was actually around an octave either way, so we've done the same here - VCO-20's detune range is just over an octave up or down. Detune can be used for subtle "fattening," or setting note intervals between the two oscillators for hideous prog rock soloing (or other less offensive uses).

PW CV Mod jack- You may have noticed that moving the *PW* knob back and forth creates a nifty sound; instead of wearing out your mouse hand, the *PW CV* input can be used in conjunction with an LFO, envelope generator, or other mod source to continuously vary the pulse width. Note that the attenuator knob is bipolar, i.e. "zero" position is center. Turning right adds a positive modulation, turning left inverts the incoming CV.

Freq CV Mod jack- Allows modulation of oscillator pitch.

VCO 1 Out / VCO 2 Out- Independent outputs for each oscillator.

Mix Out- Outputs an equal, 50/50 mix of both oscillators. For finer control of oscillator mix level, patch the individual outputs to a mixer module.

124 VCF-20 Filter

The Cherry Audio VCF-20 Filter is an analog-style, voltage-controllable, dual highpass/lowpass filter that recreates the aggressive tones of a classic 70s Japanese monosynth. Its uniquely raunchy sound totally transforms the overall tonality of Voltage Modular!

he two resonant filters can be used individually, in series, or manually patched in various configurations and are both capable of screaming self-oscillation. We carefully A/B'd the VCF-20 Filter with the coveted "version 35" filter of the original instrument and we think you'll be delighted with its authenticity.

124.1 Inputs, Outputs and Controls

Highpass/Lowpass In jacks and Level control- These are the input jacks for the highpass and lowpass filters. Signals input here can be attenuated before being sent to the filter via their respective *Level* knobs. Be sure to try using these to “dial in” the filter. Changing the input level of a signal can drastically change the way the filter sounds especially when using high peak settings.

Series- Engaging this button internally routes the output of the highpass filter to the input of the lowpass filter. This is a quick way to use both filters in series with only one input patched. Note though that this bypasses the input stage level control of the lowpass filter. It is possible however to manually patch the filter in series and use both input stages.

HP Cutoff Freq- Sets the cutoff frequency of the 6 dB/oct highpass filter. All frequencies higher than this will be allowed to pass through the filter while frequencies lower than the cutoff will be attenuated at a rate of 6 dB per/octave.

LP Cutoff Freq- Sets the cutoff frequency of the 12 dB/oct lowpass filter. All frequencies lower than this will be allowed to pass through the filter while frequencies higher than the cutoff will be attenuated at a rate of 12 dB per/octave.

HP Freq CV and LP Freq CV inputs and attenuators- CV mod inputs and attenuators for externally controlling each filter’s cutoff frequency.

Peak (resonance)- Emphasizes sound energy at and around the cutoff frequency by adding feedback from the filter’s output back to its input. As the peak is increased, any modulations or knob twisting of the cutoff frequency becomes more pronounced and can create the classic “vowel-sound” this filter is known for. When turned up past seven or so, the filter begins to feed back enough to self-oscillate. (Note that unlike the original, a cable must be patched to the filter’s input to hear it self-oscillate. This is designed to save CPU when the filter is not in use.)

Peak CV inputs and attenuators- CV mod inputs and attenuators for externally controlling the peak (resonance) of each filter. This is a feature the original monosynth did not have. The resonance of this filter can get out of hand pretty quickly, so it’s quite nice to have a little extra control via the CV inputs.

Saturation- Adds distortion to the signal. Used subtly it can add extra harmonics to a smooth bass sound or some tasteful grit to a vocal sample. Higher settings will produce the aggressive character that the original is famous for. Be careful though... when used in conjunction with a high peak setting, this filter will literally scream!

About MS-style Oscillator Distortion: *You may notice that VCF-20 doesn't necessarily distort in the expected way when using the standard green Voltage Modular Oscillator (especially with square waves). This is because the wacky, characteristic MS-style filter distortion is partially the result of the not-exactly-correct-on-an-oscilloscope waveforms output from the original MS synth oscillators. These "incorrect" waveshapes are accurately recreated in the VCO-20 Dual Oscillator module, so try it in conjunction with VCF-20.*

Saturation CV inputs and attenuators- CV mod inputs and attenuators for externally controlling the saturation of each filter.

Highpass Out jack- Outputs the processed signal from the highpass filter.

Mix Out jack- Outputs the sum of both filters.

Lowpass/Series Out jack- Outputs the processed signal from the lowpass filter. When the *Series* button is engaged, this will output the signal sent to the highpass filter’s input which is then sent to the lowpass filter.

125 Vintage Oscillator

The Vintage Oscillator is a fat and warm sounding oscillator that flawlessly replicates the unique waveform discrepancies and minute drift characteristics of coveted classic analog oscillators. It generates all standard synthesis waveforms and features both exponential and linear frequency-modulation inputs.

125.1 Inputs, Outputs, and Controls

Pitch CV jack- Accepts a CV input for pitch. Typically this would come from the *Pitch* jack in the IO Panel *CV Out* section, or from a sequencer pitch CV out.

Hard Sync jack- Force resets the start of the waveform to the beginning of its cycle. Most often used to create the "sync sweep" oscillator sounds made famous in The Cars' "Let's Go" (or Kraftwerk's "Neon Lights" and No Doubt's "Just A Girl"), by routing the output of a second oscillator to the *Hard Sync* input and sweeping the pitch of the first oscillator.

Hard Sync is also useful when creating drum and percussion sounds to ensure that the wave starts at the beginning of its cycle.

Range- Sets the basic pitch of the oscillator, displayed in traditional organ footage. *LO* will be beneath the audible range and allows the oscillator to be used as a mod source.

Expo Freq Mod attenuator and input jack- This jack is used for exponential frequency modulation. This is the "normal" 1V/Oct method used for mapping the pitch of an oscillator across the keys of a keyboard. Positive and negative voltages will raise and lower the pitch of the oscillator in equal musical amounts making this a good choice for creating vibrato or any other low-frequency modulations.

Frequency- Fine-tune control for pitch. This can be used to fatten up multi-oscillator patches by detuning a small amount, or for "building-in" a set interval. Its range is a smidge over a fifth, up or down.

Lin Freq Mod attenuator and input jack- This jack is used for linear frequency-modulation. Linear FM is used for classic FM synthesis where the frequency of an oscillator (referred to as the "carrier") is modulated by another audio-range oscillator called the "modulator." To set this up, patch one of the waveform outputs (typically a sine wave) of another oscillator module to this jack. Patch the *Pitch* jack from the CV Outs section of the I/O panel to the *Keyb CV* or *Pitch CV* input of each oscillator. Now patch one of the waveform outputs of the Vintage Oscillator to the *Main Out* jacks. Changing the modulator oscillator's frequency and the modulation amount using the Vintage Oscillator's *Lin Freq Mod* attenuator will give you a wide range of tones from subtle harmonics to harsh buzzy goodness!

> To see the difference, try using *Exp Freq Mod* jack instead of the *Lin Freq Mod* input. You'll notice that the perceived pitch changes as the modulation amount is increased. In addition to this, the pitch relation between the modulator and carrier oscillators changes as different notes are played on the keyboard making it impossible to tune the oscillator to a traditional chromatic scale. This is exactly why Linear FM is used! Linear FM keeps the pitch relation between the modulator and carrier oscillators intact at different modulation amounts across the whole keyboard.

Pulse Width- This sets the width or "duty-cycle" of the pulse wave. It has no effect on any other waveform. Its default setting of 50% outputs a perfect square wave, rich in delicious odd-order harmonics. Moving the knob left or right narrows its width as well as the thickness of sound until it almost disappears at its extremes, and we've included a nifty "faux-OLED" display to indicate the current pulse width.

PWM Amount attenuator and PWM Mod input jack- You may have noticed that moving the *Pulse Width* knob back and forth creates a nifty sound; instead of wearing out your mouse hand, the *PWM Mod* input can be used in conjunction with an LFO, envelope generator, or other mod source to continuously vary the pulse width. Best of all, the OLED display looks real cool swooping back and forth.

Waveform Output Jacks- These are output jacks for ramp, sawtooth, pulse, sine, and triangle waves. These can be used simultaneously, in any combination.

126 Vintage Resonator

The Vintage Resonator module was inspired by the *Resonators* section of the vintage 70s Moog Polymoog. The "resonators," aka, filters are globally switchable to operate in highpass, bandpass, or lowpass modes, and they're fantastic at imparting an organic feel to otherwise blah patches. If you've ever heard "The Model" by Kraftwerk, almost every sound in the track makes use of the Polymoog's resonators section.

By most accounts, the original Polymoog version has two major shortcomings: the filter slopes are a little too shallow to create really dramatic resonance effects, and the filter cutoff frequencies aren't CV controllable. We've addressed both of these issues with a 12/24 db slope selector, as well as bipolar CV inputs for each filter's cutoff frequency (which allow killer phaser-ish swept sounds). Best of all, Vintage Resonator costs a fraction of the original, and we excluded the endearing Polymoogs-are-ALWAYS-broken "feature."

126.1 How It Works

In the original instrument, Moog got a little fanciful with their use of the word "resonators." In actuality it consists of three state-variable filters in a parallel configuration. In other words, the signal doesn't flow into each filter, one after the other - the signal is split, runs into all three filters and their collective outputs are mixed together. This means that settings on each section won't interact with settings on another, resulting in three separate filter tonalities sounding simultaneously (dependent on *Gain* settings). Cranking up the *Emph* controls (aka, resonance) for each band can create three individual "peaks" or resonances, hence the "resonator" name.

One unique aspect of the original Polymoog resonator filters is that each is band-limited. Most synthesizer filters are configured such that the cutoff frequency covers the entire audible sound spectrum (20-20,000 Hz, give or take), whereas the Polymoog resonator filters are set up more like a parametric EQ where each filter covers a section of the audio spectrum, i.e. low, mid, and high as follows:

Low: 60 - 300 Hz

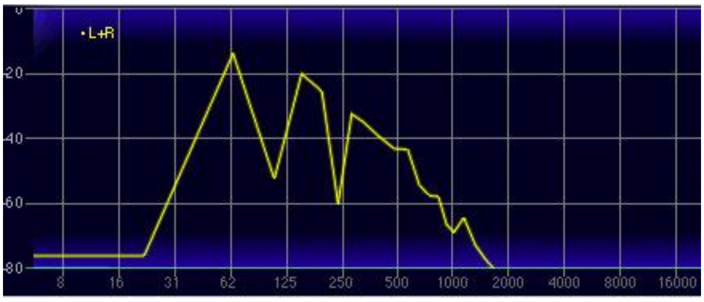
Med: 300 - 1500 Hz

High: 1500 - 7500 Hz

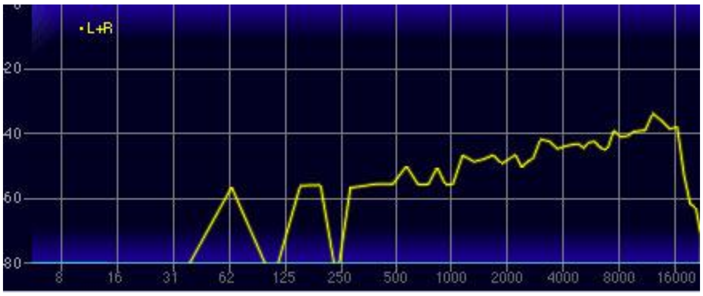
(The Polymoog panel labels the mid frequencies as "Med," because, y'know, "medium frequencies.")

126.2 Pass Mode

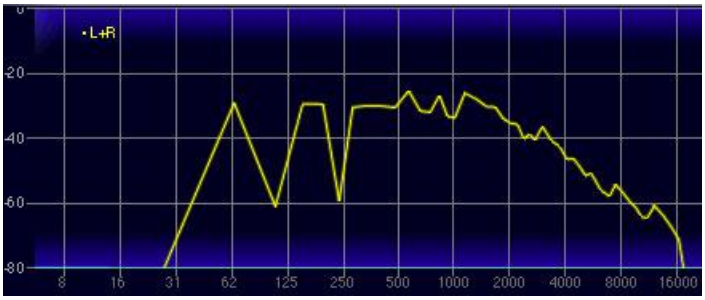
Vintage Resonator has a three-position *Pass Mode* switch that globally selects the behavior of all three filters. If you're not up on your audio filter nerdery, here's an overview of how each mode works:



A lowpass filter allows frequencies *below* the cutoff frequency setting to pass through, but blocks frequencies *above* the cutoff frequency. The frequency plot above shows the effect of a lowpass filter with its cutoff set at 412 Hz on a sawtooth wave (the vertical axis represents amplitude and the horizontal axis represents frequency). Notice how the high-frequency content trails off at the high end of the audio spectrum.



The plot above shows the same oscillator signal and cutoff frequency setting using highpass mode. This is the opposite of lowpass mode: high-frequency content remains, but low frequencies are removed as the cutoff frequency increases.



The plot above shows the same oscillator signal and cutoff frequency setting using the bandpass mode. Bandpass mode combines *both* lowpass and highpass modes, leaving sound only "in the middle." The cutoff frequency lies roughly halfway between the falloff slope on each side.

126.3 Inputs, Outputs and Controls

L/M In and R In jacks- These are the mono and stereo audio inputs. Though it only has one set of controls, Vintage Resonator actually has two complete and independent signal paths "under the hood" for true stereo processing. For mono use, simply use the *L/M In* jack.

L Out and R Out jacks- Stereo output jacks. For mono processing, use the *L Out* jack. (and don't call me Jack)

CF (Cutoff Frequency)- Sets the frequency where attenuation begins. Attenuation will be above or below this frequency (or both) depending on the *Pass Mode* switch setting.

Emph (Resonance)- Emphasizes sound energy at and around the current cutoff frequency by adding feedback from the filter's output back to its input. At lower settings, this can be used to create mild resonances such as those heard in acoustic instruments. At more extreme settings, resonance can create a pure sine wave at its own frequency (variable via the *CF* slider). Be careful with the *Emph* sliders as they can get loud at extreme settings. Note that this "ringing" resonant frequency will be much more prominent with the *Slope* switch in the *24db* position.

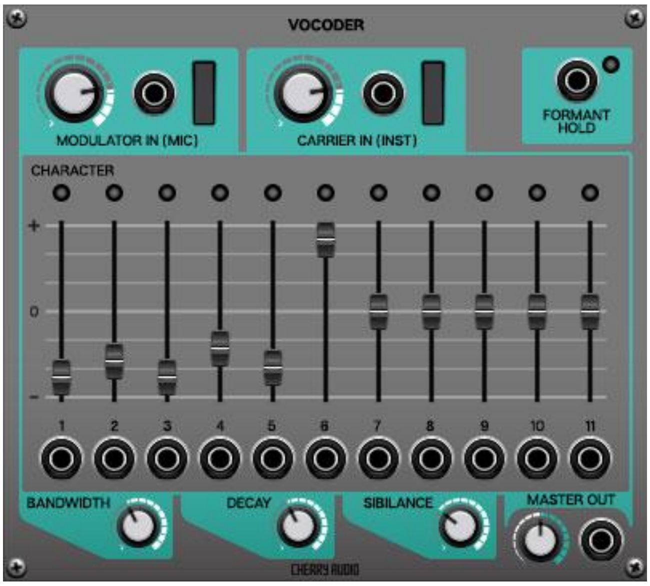
Gain- This acts as a volume control for each resonator section. Resonator sections can be muted by setting their *Gain* control to 0%.

Slope- The nature of how a filter works is such that its affect on frequencies "falls off" above or below the cutoff frequency (see the *Pass Mode* section above). Slope adjusts the steepness of this falloff, hence the "slope" terminology. A 12db per/octave filter has a shallower slope, giving it a clearer and brighter character, whereas a 24db per/octave filter's steeper slope gives it a tighter and darker tone (as well as a far more pronounced ringing sound when the *Emph* slider is turned up).

Freq CV mod input/attenuators- Adjusts the depth of the cutoff frequency modulation, i.e. how much the filters sweep back and forth when a CV signal is applied. The *Freq CV* mod input and attenuator allow negative or positive CV control of cutoff frequency. The center setting correlates to no modulation.

When used in conjunction with one or more LFOs, the CV mod inputs are especially effective for creating all manner of phaser-esque sweeping madness.

127 Vocoder



A vocoder is a specialized type of multi-filter bank that imparts the tonal characteristics of one sound upon another.

Vocoders are commonly used to create "robot" voice or choir effects by imparting the spectral character of a spoken or sung source (aka, the "modulator") to a full-spectrum constant tone, typically a bright sawtooth synth pad (aka, the "carrier").

Unlike most other Voltage modules, getting sound from the Vocoder module isn't immediately obvious, so we'll explain how to configure it for real-time (i.e. with a mic input) or pre-recorded audio control (i.e. playing back an audio track).

127.1 Quick Start

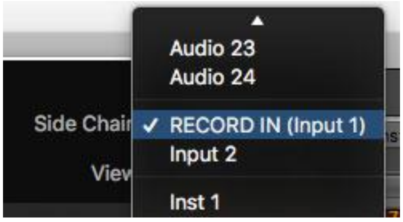
If you're not interested in what's going on under the hood and just want to get your electro *rock-it-baby* on, you and your pal The Egyptian Lover can read this section to get the party started.

The basic idea is that the modulator input imparts its character upon the carrier input. A crude analogy would be humming a pitch while cupping your hand over your mouth to change the sound - the constant hum would be the carrier signal, and modulator would be your hand on your mouth.

In the context of a Voltage Modular patch, the *Carrier In* jack would typically be a constant audio source, such as an oscillator wave or white noise, preferably with rich harmonic content; sawtooth waves and noise work particularly well. The *Modulator* In jack would be the tone "shaper" - live or prerecorded vocals work well, as do drum beats or rhythmic guitar tracks.

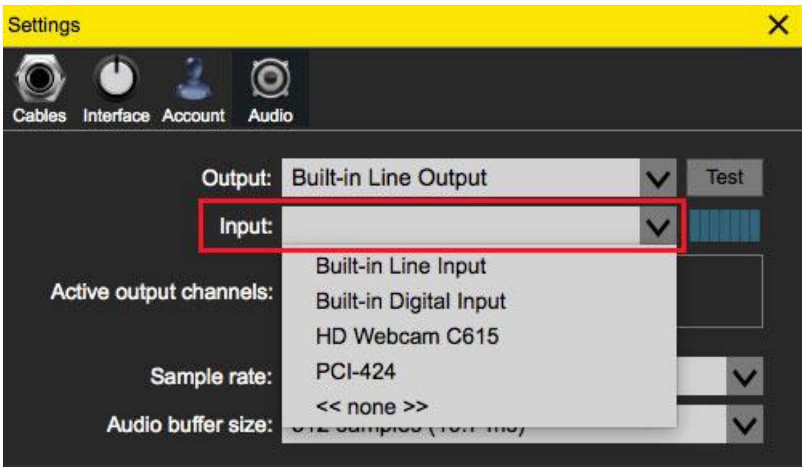
127.2 Configuring Modulator Input and Modulator Audio Source

If you're using the VST or AU plug-in version of Voltage, you'll use its sidechain input to route audio to the *Audio In* from host jacks in Voltage's I/O Panel. The image below shows Logic Pro's *Side Chain* input pop-up menu (located in the upper-right corner of the plug-in window):



The specific location of the sidechain dialog may vary depending on your DAW software, but you'll want to select the audio source being used to modulate the static (carrier) audio. In this example we're using Logic's first audio input, with a live microphone plugged into an audio interface. (You may need to enable software monitoring in your DAW to use live audio sources.) Alternatively, the sidechain source could be an audio track containing pre-recorded audio.

If you're using the standalone version, you'll need to choose the audio input source in Voltage's preferences menu. Clear the gear icon to the left of the *Library* button at the top, then click *Audio*. Click in the *Input* section to choose an audio source.



The sidechain audio is now routed to jacks *1L/1R* in the *Audio In from host* section of the I/O Panel at the top of the Voltage window. When sidechain audio is playing, the *Audio In from host* meters will blink. Depending on which input is routed, click on the *1L* or *2L* jacks to route a cable to the *Modulator In (Mic)* jack. Set the signal level so that the nominal level sits in the full green/occasionally yellow region.

127.3 Configuring The Carrier In

This one's easy - route the output of a continuous audio source such as an oscillator, or noise source to *Carrier In (Inst)* and set the level appropriately. A patch with an amp/envelope generator that gates on and off with a keyboard is fine, just avoid percussive sounds that die away quickly, otherwise you'll be trying to vocode "nothing." Remember that vocoding controls not only the carrier audio's frequency spectrum, but its amplitude as well.

Voltage's Poly Oscillator module set to saw waves makes an ideal vocoder carrier signal source. Its built-in voltage-controlled amp and envelope generator means you won't need to use separate modules for amplitude control. A basic patch would look something like this:

Once you've got the Vocoder up and rockin', try manipulating the Poly Oscillator's octave sliders as well as the Vocoder's Bandwidth, Decay, and Sibilance controls for different effects.

127.4 How Does It Work?

The Voltage Vocoder module consists of two matching sets of ten bandpass filters (plus two highpass filters): one set for the modulator signal, and the other set for the carrier signal. These bandpass filters each cover a small "slice" of the audible audio spectrum - the eleven bands you see on the panel logically go from low frequencies to high frequencies. The number 11 slider is a highpass filter that handles the top end of the audio spectrum from 8k-20k.

The modulator signal is split and runs through all eleven filters. Each filter only allows a small "slice" of the audio frequency range through. Immediately following each filter is an Envelope Follower, which is a special type of amplifier that converts incoming audio levels to a corresponding control voltage. So far, we have the incoming modulator signal being split into eleven separate control voltages, all changing independently and in real-time dependent upon the modulator audio's energy across the frequency spectrum. Heavy stuff, right?

Let's slide on over to the carrier signal side and talk about what's going on with its filters. The carrier signal also gets split, and runs through the second set of eleven bandpass and highpass filters. These filters each route to the audio input of a standard voltage-controlled amplifier (or "VCA" for short). Remember all those control voltages from the modulator's envelope followers? Those are connected to the carrier's individual corresponding VCA control voltage inputs.

If there is no signal at the modular input, the carrier's VCA's are all closed. If the modulator signal contains audio energy in the 400Hz area, the carrier filter/amp combos will "open up" in the same area of the audio spectrum, thus letting carrier audio in the 400Hz area through.

In reality though, this is much more complex and nuanced, because different audio signals contain different energy levels across their frequency spectrum. This complexity is why vocoders can create such unique sounds (and also why a relatively simple set of filters can produce recognizable speech).

Though this covers the basic operation of a vocoder, there are few other things going on under the hood that help the speech intelligibility and general fidelity of a vocoder, including adding highpassed noise to help with S sounds, weird EQ curves, and some other tricks we'll never divulge! Suffice to say, making a great-sounding vocoder is tricky business, and we hope you'll enjoy the fruits of our efforts.

127.5 Now That I (Sort Of) Understand How This Mess Works, What Do The Knobs Do?

Modulator In (Mic)

This sets the input level of the modulator signal input. Use the meter to set it so that nominal sits in the full green/occasionally yellow region.

By the way, this input is labeled Mic for clarity, but any audio signal can be used, such as a pre-recorded vocal track, a drum loop, rhythm guitar, or even an entire song. Signals with constantly changing frequency and/or amplitude tend to work best - in other words, don't use an organ.

Carrier In (Inst)

This sets the input level of the carrier input signal. Use the meter to set it so that nominal sits in the full green/occasionally yellow region. If you've made it this far, we're sure you know how to set an input level, but do keep in mind that vocoders are fairly sensitive to levels (too low and too high), so it's a good idea to make sure levels are in their happy spot for best sound.

Because the constantly changing filter bank is effectively removing different areas of harmonics, the best choices for carrier audio are sources with a full-frequency spectrum and constant sustain.

Formant Hold

If you read the *How Does It Work?* section (of course you did, everyone loves a rambling technical explanation), we talked about how the carrier's eleven VCAs dynamically mirror the modulator's frequency "profile" as it changes in real-time. Applying five-volts to the *Formant Hold* jack freezes the current state of the carrier VCA's. The most common use for this would be if you were singing into a mic to shape a continuous sound, and you wanted to sustain the sound (imagine a vocoded choral "ahh"), but were going to run out of breath. A sustain pedal patched via Voltage's IO Panel to the *Formant Hold* jack allows the sound to infinitely sustain the current carrier filter curve. The LED indicates when *Formant Hold* is active, i.e. when five volts is being received.

Character LEDs, Sliders, and Individual Output Jacks

These controls correspond to the Vocoder module's individual filter bands. The individual Character LEDs display the activity of each filter band's envelope follower voltage output. This can be seen by making low-to-high pitched noises with a microphone (make S noises for the highest bands). In use, these give a good general idea of frequency spectrum activity.

The Character sliders control the volume of each of the carrier side VCA's, and effectively act as a graphic EQ of sorts. Unlike most other vocoders, bands can be turned down to zero, effectively turning them off completely for unique effects. The Character individual out jacks correspond to each individual carrier VCA's output; note that individual out volumes are affected by the sliders. Separately routing specific bands or groups of bands within Voltage allows all kinds of interesting noises.

Bandwidth

Sets the width or "Q" of all bandpass filters. Narrow bandwidths let less audio through, whereas wider bandwidths let more audio through for a denser sound. A good analogy would be to imagine water running through a comb with wider or narrower tooth spacing.

Decay

This sets how quickly the carriers signal envelope followers recover to zero amplitude. Lower settings have a snappier, tighter sound (good when using drums or percussion as a modulator); higher settings are looser (a good choice when using vocals as a modulator).

(Not to toot our own horns too much, but Bandwidth and Decay are both parameters that you'd never see on an analog hardware vocoder, as they would be difficult to implement in analog electronics - these features are comparatively easy in computer world.)

Sibilance

This adds highpassed white noise into the final signal when S sounds are detected. This helps speech intelligibility, because vocoders aren't inherently good at detecting S sounds, and many carrier sources don't have much energy in the S sound spectrum.

Master Out

This is the mix of all the of the bands, with a volume knob and out jack.

128 Voltage-Controlled Mixer

The Cherry Audio Voltage-Controlled Mixer is a four-channel mixer for audio and control signals featuring solos and mutes, individual stereo outs, and CV control of level and pan on each channel.

This module essentially consists of four Panner modules, each with a dedicated VCA (voltage-controlled amplifier), that can be used individually and/or mixed to the master output (which by the way also has a dedicated VCA!).

128.1 Inputs, Outputs, and Controls

128.1.1 Channels 1 - 4

Each of the four individual channels feature all of the same jacks and controls, so we will just look at channel 1.

Input jack- Audio or CV input jack.

Pan- Typical pan control for routing the signal between the left and right outputs of both the channel and master outs.

Pan CV- CV input and attenuator for externally controlling the pan position. The -*CV* and *+CV* LEDs will light up to indicate the polarity of the voltage currently controlling the pan position.

S- Solo button for isolating the channel's signal. When engaged, all channels that are not also soloed will be muted.

M- Mute button for muting the channel.

VU Meter- Visually shows the level at which the input signal is being sent to the channel and master outputs.

Level- This fader adjusts the channel's output level (to both the channel and master outputs) from -infinity to +6.0dB.

Level CV jack and attenuator- CV input and attenuator for externally controlling the channel's output level.

L and R Ch Out jacks- Individual stereo outputs for the channel.

128.1.2 Master Output

Stereo VU Meter- Visually shows the master output level of the mixer.

Master Output Level- This fader adjusts the amplitude of the master outputs from -infinity to +6.0dB.

Level CV jack and attenuator- CV input and attenuator for externally controlling the master output level.

L and R Mast Out jacks- This is the mixer's master output. All four channels are mixed in stereo and output from these jacks.