The Cherry Audio Rackmode Signal Processors bundle represents the first time these renowned analog treasures have been recreated and made available as virtual effects. Rackmode also marks our latest collaboration with award-winning synth designer and DSP legend Mark Barton (MRB), who modeled the circuit-precise emulations based on the original rack-mounted hardware. Further, we have expanded these effects with additional features aimed at today's music producers, including presets, LFOs, stereo processing, and full Apple Silicon M1 native support. And speaking of authenticity, these modules capture not only the deep sounds and capabilities of the original Signal Processor line, but also the elegant look and heritage of the rack-mounted enclosures.

Cherry Audio's Rackmode Signal Processors collection represents a first-time opportunity for producers to explore these studio-quality vintage effects. When combined with today's modern production workflow, experimentation will lead to inspiration, and an old approach to sound can unexpectedly result in something completely new. We hope you enjoy the Rackmode Signal Processors as much as we do!

Pre-Purchase Demo Mode

If you haven't purchased the full Rackmode Signal Processors collection, they will run in demo mode. All functions will work, but bursts of noise will occasionally be heard (the LED next to *Demo* in the top toolbar will illuminate when the noise is sounding). We've also added a handy button to make purchasing easy, so you've got no excuse! As you might expect, all this demo nonsense disappears once the Rackmode Signal Processors collection is purchased.

Technical Assistance

Cherry Audio's unique online store and automatic updating should make operation a smooth experience, but if you run into any issues or have questions, you can discuss issues online at the Cherry Audio forums at:

https://forums.cherryaudio.com/viewforum.php?f=61

... or you can communicate directly with our ~~difficult and easily perturbed except for the new guy~~ outgoing and friendly tech support staff at:

https://cherryaudio.kayako.com/

Top Toolbar and Preset Browser

The purple strip at the top is the same* in all of the Rackmode plug-ins. It's where you'll load, save, and create presets. It also contains utility functions such as undo/redo, UI zoom and *Focus* controls, under-the-hood settings, and more.

*Almost the same - both the instrument and FX versions of the Vocoder add a few icons and functions. We'll address these differences in the **Vocoder** section.*

Getting back to things, let's go over the top menu strip:

New- Opens a new blank preset. If an unsaved preset is currently open or you've modified an existing saved preset, a dialog asks if you'd like to save the preset in its current state. This greatly reduces the possibility of losing an edited unsaved preset.
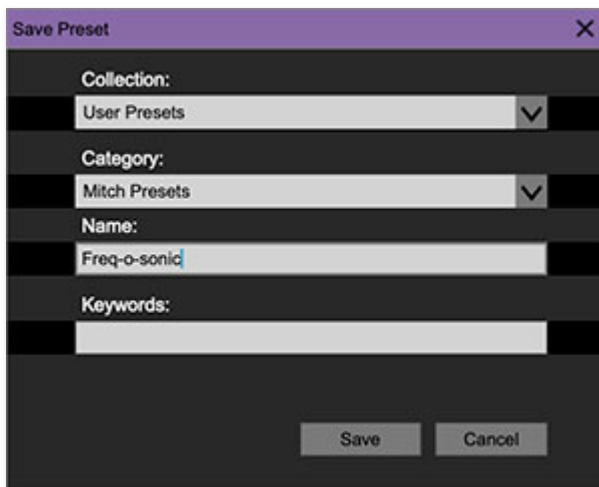
Save- Use this to save presets. There are a couple of levels of hierarchy:

- **Collection**- This is the top level of organization, and contains entire "sets" of presets. We also include a *User Presets Collection* for storing your own presets, but you're free to create your own collections. To create a new collections, click in the *Collection* text field (where it says *User Presets* above) and type a name. User-created presets can be freely saved to any collection; we like to keep 'em separated for organizational purposes.

- **Categories**- Within each *Collection* are a number of preset categories. As with collections, you're free to create as many categories as you like. To create a category, click in the *Category* text field of the *Save* dialog window and type a new category name.

- **Preset**- To save a preset, simply type the name in the *Name* field and click *Save.*

- **Keywords**- You can add descriptive words such as "vocal," "alt" etc., to presets to make them appear when terms are typed in the *Search* field. Use commas to separate multiple keywords entries.
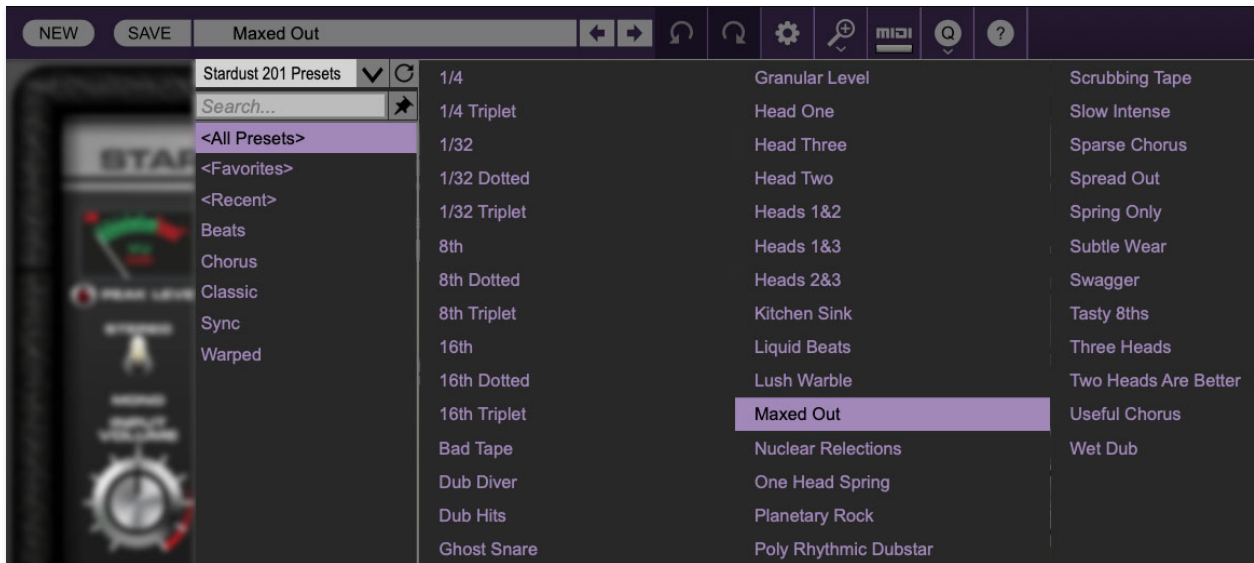
Browsing Presets- Presets can be browsed by clicking the *<Select Preset>* field. To select a preset collection, click in the area that says *<All Collections>* or on the downward-facing arrow next to it.

Clicking on the left-side categories narrows down which presets are displayed.

- **<All Presets>** will show presets from all collections and categories.

- **<Recent>** displays recently used presets.

- **Refresh**- This is the circular arrow button to the right of the downward arrow; clicking this checks the Cherry Audio server for new or updated presets.
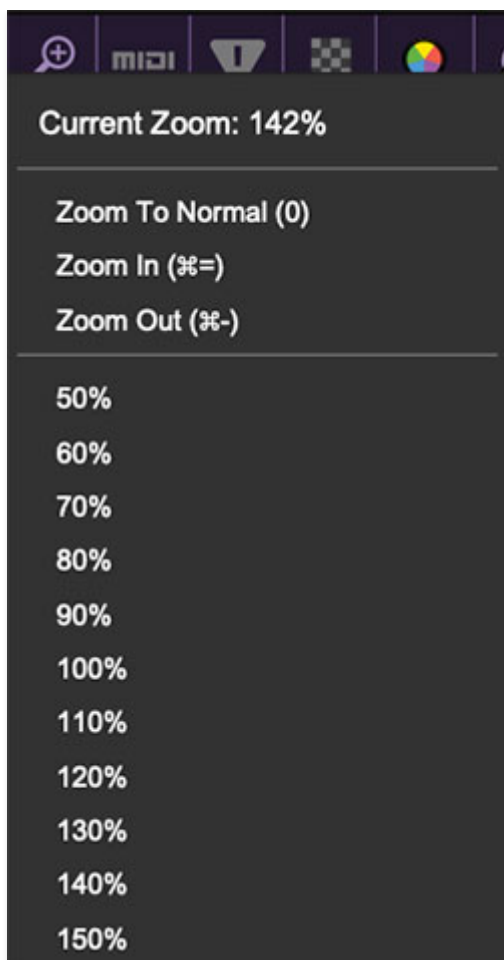
Pin - Clicking the push-pin icon locks the preset selection list open, allowing fast and easy browsing and auditioning of prests. Click the icon again to disable pin mode. when in pin mode, the up and down arrow keys can be used to select presets.

Preset - *Step Back/Forward Arrows* - These step to the previous or next preset. macOS [⌘+left/right arrow key] or Windows [CTRL+left/right arrow key] will navigate through presets back and forth in the currently selected collection/category.

Undo/Redo - These circular arrows will undo or redo the last action. It remembers many steps, so if you really loused something up, keep on clickin'...

Settings - This is where user preferences for user interface, audio interfaces, user account, and more are configured. See the **Settings** section for full information.

Current Zoom: 142%

Zoom To Normal (0)
Zoom In (⌘=)
Zoom Out (⌘-)

50%
60%
70%
80%
90%
100%
110%
120%
130%
140%
150%

Importing Presets with drag-and-drop

Presets can be imported singly or en masse (as a single compressed ZIP file) simply by dragging and dropping from the desktop anywhere on the user interface.

If a single .*preset file is dragged and dropped, the sound is immediately loaded and the standard Save Preset dialog appears; this lets you save the sound to the instrument's preset browser. Note that you don't have to save the sound to preset browser; if you just want to hear and play the sound, click the Cancel button in the Save Preset dialog - the sound will still be loaded.

Compressed zip files containing multiple sounds can also be drag and dropped onto the UI. This works the same as with single sounds, but instead of the Save Preset dialog, you'll see the Import Preset Collection dialog. The presets will be added as a new collection and available in the categories for which they were tagged.

Zoom Magnifying Glass - Click to resize UI interface. Selecting *100%* returns the user interface to native size. You can also grab and drag the diagonal "grab" lines in the lower-right corner of the user interfaces.

MIDI Tab- Opens the MIDI controllers tab for configuring internal and hardware MIDI controls. See the **MIDI Controllers Setup and MIDI Tab** section for full information.

Help- Clicking this launches your web browser and opens the help document you're currently reading (but I'm already reading this... circular logic, man).

Cherry Audio logos - Clicking these displays "about" information, and shows the version number and current registered user ID.

Preset List Right-Click Functions

Show File- This displays the selected preset in the Mac or Windows folder containing it. This is useful for backing up or sending a preset file to another user.

Show In Original Category- Selects the preset within its category, i.e. the category will highlight in the left preset menu. The *Show In Original Category* command only displays if the preset was selected within the *<All Presets>*, *<Favorites>*, or *<Recent>* categories.

Favorites- Favorited presets will show in when the <Favorites> category is selected. A star will display next to the preset name. Right-click on the preset and reselect *Favorite* to un-favorite it. (Or say something nasty about its mother.)

Delete- Deletes the selected preset.

Restore Factory Preset- If one of the factory (i.e. not user) presets is edited and saved, selecting this command restores the patch to its unaltered "factory" setting. This menu will be grayed-out for user bank patches.

Restore All Factory Presets From '<Plugin Name Name> Presets'- If any presets from the "factory" bank for each plugin are edited and saved, selecting this command restores *all* of them to their unaltered "factory" setting.

Restore All From Factory- If any presets from the "factory" banks are edited and saved, selecting this command restores *all* of them to their

unaltered "factory" setting.

The Cherry Audio Rackmode Three Band Parametric EQ is a super-accurate recreation of a late 70s rack mount parametric EQ. Besides having a highly flexible EQ character, it includes a unique *Drive* knob that lends a gentle saturation to signals.

Other than their respective frequency ranges, each of the three EQ sections is identical. Controls are as follows:

Width- Sets the bandwidth of the boosted and cut signal from 1/4 to 4 octaves wide.
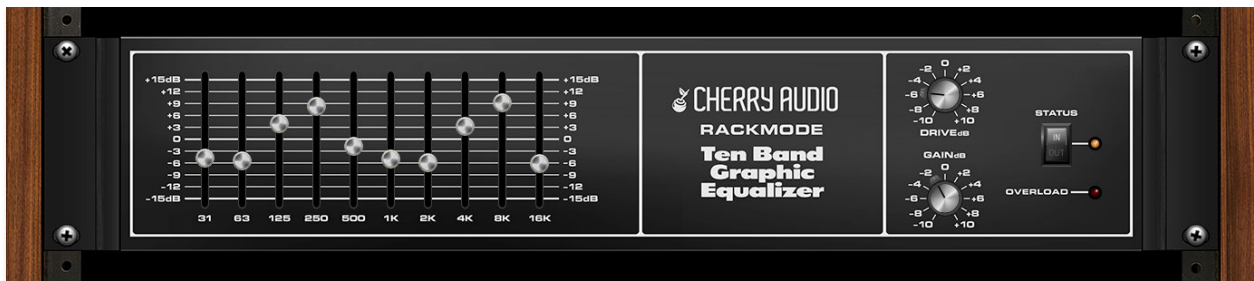
Frequency- Sets the center frequency to be boosted or cut.

Height- Sets the amount of boost or cut, up to 20 dB.

Drive- Cranking this up adds a gentle clipping to audio content.

Gain and Overload lamp- Sets overall output level. It's generally best to set this so the Overload lamp occasionally lights, and then back it off a hair.

Status- Enables or disables the EQ.

The Cherry Audio Rackmode Ten Band Graphic EQ recreates a late 70s rack-mount Ten Band EQ. It sounds great and is really easy to use for "carving" sonic landscapes.

Boost/Cut Sliders- These boost or cut frequencies up to 15 dB at one-octave increments.

Drive- Cranking this up adds a gentle clipping to audio content.

Gain and Overload lamp- Sets overall output level. It's generally best to set this so the Overload lamp occasionally lights, and then back it off a hair.

Status- Enables or disables the EQ.

Rackmode Vocoder

The Rackmode Vocoder replicates and expands on a classic 16 Channel Vocoder, originally released in 1978. Interestingly, the original was something of an "OEM" rebadge, as it was originally marketed as the Bode model 7702. As far as we know, the two are functionally identical, and both sound awesome. The original was used extensively on moustache 'n' shades innovator, Giorgio Moroder's release "E=MC²" - a spectacularly awful-yet-fantastic time capsule of late-70s electronic music.

If you're not familiar with how a vocoder operates, the basic concept is that a vocoder imparts the spectral characteristics (usually a human voice) upon a constant tone source (usually a synthesizer playing bright, static tones). This classic voice-modulating-synth setup results in the classic robot tones associated with vocoders, but vocoders are actually capable of a wide variety of effects dependent upon the audio sources used.

How A Vocoder Works

We're going to simplify the building blocks of a vocoder somewhat in order to make things easy to conceptualize. After this, you'll totally understand how vocoders work and you'll be the life of the party - you can thank us later!
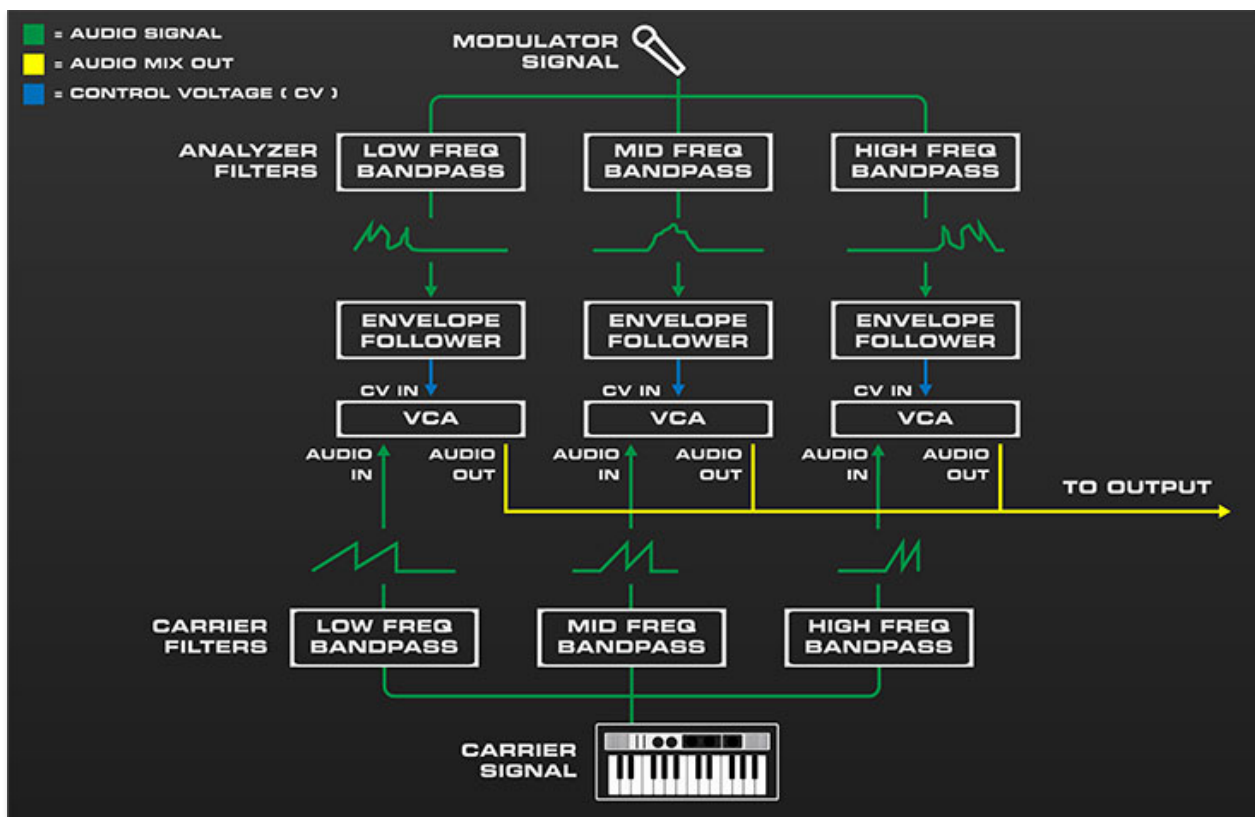
As mentioned, a vocoder imparts the frequency spectrum of one sound onto another. In the diagram above, a microphone's frequency spectrum (i.e., speech), is imparted onto the sound of a synthesizer playing a basic sawtooth-wave patch. Let's break down what's happening:

At the bottom of the image, a synthesizer plays constant notes using a harmonically rich sawtooth wave. This is referred to as the "carrier" signal. The synthesizer output is multed to three bandpass filters (aka, BPFs). Because each filter only allows a certain region of the audio frequencies to pass through, this effectively splits the signal into low-, mid-, and high-frequency components (an actual vocoder has many more individual bandpass filters which we'll discuss later, but for our example, we've simplified the circuit to three bandpass filters). Each of these feeds an individual voltage-controlled amplifier (VCA), which acts as volume control that gets louder or softer depending on a separate incoming control voltage (CV). If the VCA's CV inputs are *not* currently receiving a control voltage, the VCA's are "closed," that is, no sound is allowed to travel through. This is why a vocoder doesn't make any sound unless you're holding down keys *and* speaking into the mic.

Now let's examine what happens when the split-up synthesizer carrier signal is playing into the VCAs and the mic is spoken into. When the user speaks into the microphone, the mic signal is multed into an identical second set of

bandpass filters and also split into frequency bands - this is called the "modulator" signal (the original refers to this as the "program" signal, as does Rackmode Vocoder), and the modulator BPF's are referrred to as "analyzer" filters, because they're effectively analyzing the incoming mic signal. The low-, mid-, and high-frequency components each travel to individual envelope followers. The envelope followers convert the constantly oscillating, up and down AC audio signal voltages to single-polarity DC control voltages proportional to each frequency band's overall signal volume. These control voltages are routed to the carrier signal VCA CV ins and (here's where the magic happens) cause the carrier (synth) bandpass filter volumes to mimic the volume and frequency spectrum of the modulator signal filters. The audio outputs of all VCA's are then mixed together and travel to the vocoder output.

The preceding is a simplified explanation. First and foremost, if we want good fidelity from our vocoder (voice intelligibility is the main factor here), more than three sets of bandpass filters (bands) are necessary. Most commercially available vocoders have at least 10 bands, and Rackmode Vocoder has 16 (we're not impressed though - Woodstock had 32 bands). Taking into consideration that this means 32 individual bandpass filters, 16 VCAs, and 16 envelope followers, and it's easy to see why hardware vocoders have historically been pricey affairs.

There are few other things going on under the hood that help with speech intelligibility and general fidelity of a vocoder, including adding highpassed noise and modulator signal to clarify S sounds (the *Hiss* you see on the front panel), weird EQ curves, etc. Suffice to say, theoretically making a vocoder isn't too complicated, but making a really great-sound vocoder is indeed tricky business.

An important aside is that a vocoder does *not* detect specific note pitches in the modulator signal, nor will they be audible in the vocoder output. This means that if you're using a mic as a modulator source**, it makes no difference whatsoever whether you speak words or sing, on key, off key, or otherwise. The actual notes you'll hear will always be those of the carrier signal. When using a microphone (or recorded speech), the best results are usually obtained by speaking very clearly and exaggerating diction. TL;DR: singing ability has no effect on vocoder results.** (Hooray for lousy singers!)

Rackmode Vocoder Instrument vs. FX Version (Important, please read!)

As mentioned, vocoders make use of two separate audio inputs, a modulator (*Program*) signal, and a carrier signal. As a result, it's very important to understand how these two signals get routed within DAW software otherwise it could lead to frustration, head scratching, tech support tickets, and those clowns bugging me, which no one wants.

Rackmode Vocoder includes two different plug-in versions. **Rackmode Vocoder Instrument** includes a one "rack-space" MIDI-controlled dual-oscillator synthesizer, which acts as the carrier signal source. This is hard-wired to Rackmode Vocoder's carrier input and *Level* knob. It features ramp, square, and a very special "glottal" wave that enables exceptional sounding voice synthesis. The instrument version appears in your DAW's instrument menu like any other virtual instrument. **Its onboard dual-oscillator synth is *always* the carrier signal, and the modulator signal is derived from your DAW's sidechain input** (this routed is configured within a DAW using a popup menu in the effect window). If you're primarily using Rackmode Vocoder to create robot voices or synthetic choirs, the instrument version may be all you'll ever need.

Conversely, the **Rackmode Vocoder FX** version appears in your DAW's effects menu; it does *not* include the single rack-space onboard synthesizer. Carrier and modulator sources are completely up to you; **the carrier signal is the audio content of the current track** (either a prerecorded audio region or live incoming audio); as with the instrument version, **the modulator signal is derived from the sidechain input**. Rackmode Vocoder FX version offers endless possibilities, because *any* audio source maybe used as the modulator or carrier input. For example, a recording of an orchestral string section could be vocoded using a drum loop as the modulator signal.

Your DAW must support sidechain signal routing to use either vocoder version. Most professional DAW software should include sidechain capabilities; *GarageBand* is a notable exception.

Alright, that's enough bold-print and and exclamation points for now, let's get rockin':

Main Vocoder Controls

The instrument version consists of two sections - the top synthesizer section, and the bottom vocoder section. The FX version consists of the bottom vocoder section only and is identical in both versions. We'll review its controls in a way that makes most sense from an operational standpoint (as opposed to moving directly across the front panel):

Program / Level- Sets the volume of the incoming modulator signal from DAW sidechain input. Note that "program" is oddball 70s terminology - in just about any other context, this signal will be referred to as the "modulator" signal, and we will refer to it as such in this user guide.

If you're using the vocoder to create standard "robot voice" effects, the modulator signal should be either a live microphone input or an audio track containing spoken or sung audio. (Drum loops or rhythmic guitars also make excellent modulator signals). Set the level so the overload lamp (OL) only flashes occasionally; optimizing the modulator level will result in the best speech intelligibility.

Carrier / Level- Sets the volume of the incoming carrier signal, aka, the signal that gets modulated. If the instrument version is being used, the carrier signal will be the onboard synthesizer; if the FX version is used, the carrier signal will be the audio on the DAW track (or real-time live audio routed to that DAW track).

By its nature, a vocoder is a subtractive synthesizer - that is, it applies numerous filters to the carrier that constantly remove sections of the audio spectrum. Because of this, **it's best to use a carrier signal with a dense harmonic spectrum** - sawtooth waves are used most frequently because of their rich even-order harmonics. Square and pulse also make good carrier tones, and Rackmode Vocoder's unique "glottal" wave excels at creating realistic vocalesque tones. Conversely, we don't recommend using dull or excessively thin sounds as carrier signals.

Hiss & Buzz

One caveat of vocoders is that they don't reproduce non-pitched sibilant sounds very well (i.e., anything with an "S" sound). These S sounds lie at the top of the audible frequency range, and they're very important for intelligibility (feel free to make a drinking game out of the use of the word intelligibility in this manual).

There are two ways vocoders compensate for this:

- The modulator signal is run through a steep highpass filter and this signal triggers a highpass filtered white noise generator to "fill in" the S

frequencies.

- The modulator signal is run through a steep highpass filter, leaving only the very highest "hissy" parts of the signal. This highpassed direct modulator signal (typically a microphone) then bypasses the main vocoder bandpass filter band and is mixed directly into the vocoder's master output.

The original made use of both of these techniques. Non-pitched sibilants are referred to as hiss, whereas pitched "body" sounds are referred to as buzz. The clever bit (as our English friends say) is that the vocoder automatically detects whether the incoming modulator audio is non-pitched hiss sound, or a pitched buzz sound, and instantaneously switches between these signals, resulting in excellent intelligibility and overall fidelity. This switching is always active, with the current signal mode displayed by the green and red LEDs adjacent to the *Hiss* and *Buzz* mode switch positions. There a couple of controls that enable, disable, and set the mix level of each component:

Switched/Direct - 5080-15000 Hz- When in the *Switched* (left) position, only the highpassed white noise signal is heard when S sounds are detected in the modulator signal (i.e. green *Hiss* LED is glowing)

When in the *Direct* (right) position, highpass-filtered direct modulator audio is mixed into the output in addition to the aforementioned highpassed white noise signal, resulting in more pronounced sibilance. If you're speaking or singing into a mic, you'll hear a faint "ghost" version of your voice in the vocoder output (this is easiest to hear with the *Mode* knob set to *Hiss* position). Note that the highpass filtered direct audio is always on if the *Direct* switch is enabled - that is, it doesn't turn on and off via the hiss detector circuit like the highpassed white noise signal.

Balance- Sets the mix of hiss and buzz signals when the *Mode* knob is in the *Hiss & Buzz* position. It has no effect if the *Mode* knob is set to the *Hiss* or *Buzz* positions.

Mode- Determines whether hiss, buzz, or both are heard in the vocoder output. In the *Hiss & Buzz* position, the hiss detector will alternate between signals depending on the incoming modulator audio content; in the *Hiss* position, the highpassed white noise signal is constantly audible, i.e., it's not switched by hiss detector circuit. (We don't find the *Hiss* position to be super useful for final vocoder audio, but it's an effective way to "solo" and hear exactly what's happening in the hiss component.)

For most situations, we recommend parking the *Mode* switch at the *Hiss & Buzz* position and using the *Balance* knob to dial in the desired amount of hiss vs. buzz signals. The buzz signal is usually where the sonic interest comes from, but hiss can be very helpful for intelligibility.

Formant and Formant Mod

As we've mentioned, a vocoder consists of two identical banks of bandpass filters, and these correlate exactly to each other by default. For example, if you made spoke the WAHHH into the mic with a honky, nasal tone, the four modulator "analyzer" bandpass filters from 317 Hz to 800 Hz might open to varying degrees; this in turn would open the corresponding carrier bandpass filters in the exact same way.

But let's say an offset was applied between the two filter banks. Instead of the 317 Hz analyzer filter causing the 317 Hz carrier filter to open, the 317 Hz would cause the 400 Hz filter to open. This is what the *Formant* knob does - it gradually "shifts" the relationship of the two filter banks up or down relative to each other. We call this *Formant*, because when used for vocal sounds, the audible effect is of the vocal tract growing larger, i.e. more manly, or shrinking, i.e., child-like (or at extremes, like a chipmunk, but a kickass funky robot chipmunk).

This control isn't on the original and it's a very nice effect. In development, we noticed it sounded particularly cool when we moved the knob back and forth, so we added a syncable LFO modulation section to automatically modulate knob movements.

Formant- Adjusts the amount of up or down offset shift between the modulator analyzer and carrier BPF banks.

Formant Mod / Rate- Adjusts the rate of formant modulation, from 0.1 to 10 Hz (with *Sync* switch off) or from 8 beats up to 1/64th note triplets (*Sync* switch on). The flashing red LED indicates the current mod rate.

Formant Mod / Sync- The *Sync* switch locks the LFO to DAW host tempo. The green LED illuminates to indicate sync mode is active.

External Patch and Jacks

Each vertical jack pair represents one of the 16 pairs of bandpass filters. The numbers to the left and right of each jack indicate the lower and upper frequencies it passes (like any analog filters, they have a "slope" on either side, so adjacent BPF frequencies will overlap to some degree depending on

the *Resonance* control setting). For example, the first BPF on the left passes from 50 Hz to 159 Hz, the second BPF passes from 159 Hz to 200 Hz, etc. The LEDs between the jacks illuminate with varying intensity, dependent on the current signal strength of each band, e.g., the VCA control voltage amount.

Rackmode Vocoder's jack section is closely related to the *Formant* knob explained above, but instead of a gradual shift in relationship between the modulator and carrier BPF banks, enabling the *External Patch* switch completely disconnects the modulator-side control voltages from the corresponding carrier-side VCA's (the blue arrows in the *How A Vocoder Works* overview diagram at the top). This allows the modulator and carrier BPF filters to be rerouted or "cross-patched" in any desired configuration, opening many creative possibilities. **External patch jacks will only function when the *External Patch* switch is enabled.**
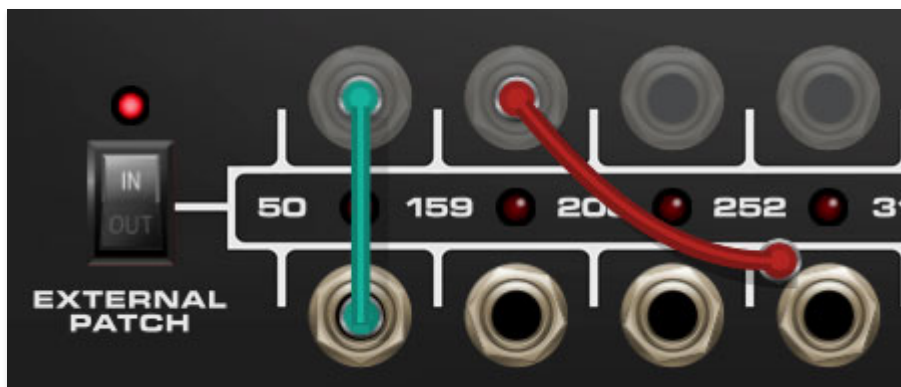
To patch a cable, simply click and hold on a jack then drag to another jack and release the mouse button. Cables can be routed from an input to an output, or from an output to an input. When a cable is being dragged, all potential destinations will appear normal, and non-destinations will gray out - this prevents inputs from being routed to other inputs and outputs from being routed to other outputs.

Note that Rackmode Vocoder utilizes the same highly developed cabling system as Cherry Audio's Voltage Modular virtual modular synth, so it's got a number of tricks up its sleeve that aren't immediately obvious - please check out *External Patch Cabling Tips and Tricks* at the very bottom of this section for more information about cable behavior (or call your local cable operator if it's been on the fritz).

The patching configuration shown above is identical to the normal "hardwired" internal patching when the *External Patch* switch is disabled.

Additional Top Toolbar Controls

We covered the majority of the top purple menu strip controls way back yonder in the **Top Toolbar and Preset Browser** section, but there are a few extra top toolbar controls associated with the patch cables described in the preceding section.
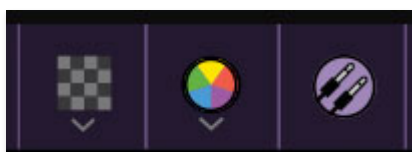
Cable Transparency- Clicking the checkerboard icon displays the cable transparency slider. Slide this to the left for more transparent cables, or to the right for more opaque cables.

Cable Color Select- Click this to select the global cable color, i.e. the color of any newly patched cable. Clicking *Random* randomly chooses a color for each new cable.

Show/Hide Cables- Clicking this hides or shows all cables. It has no effect on sounds, and its status does not save with patches. Cables can also be shown or hidden using the key shortcuts [CONT-D] (Windows) or [⌘-D] (Mac).

Special Cool Cable Color Select Feature- The color of any existing cable can be changed by right-clicking in jack area. Right-clicking on a jack that doesn't have a cable plugged in will change the global cable color (i.e. the same as changing the color with the toolbar button).

_____

Resonance- Sets the width or "Q" of all bandpass filters of both banks simultaneously. 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.

Articulation- This sets the decay/release time of the modulator side envelope followers. In use, this equates to the overall responsiveness of the vocoder to modulator signals - fast articulation times equate to a tight, responsive sound, whereas slower times result in "lazier" recovery from transients. Setting to max *Freeze* position is essentially the same as enabling the *Hold* switch described below.

Hold- Enabling the *Hold* button freezes the current state of the carrier VCAs. This is useful if you'd like to infinitely sustain a carrier filter curve without turning blue in the face endlessly singing a note with a mic. Specifically, it's useful for sustained, choral "aah" type pads.

The *Hold* switch is especially handy when a sustain pedal is mapped to control it (super easy: right-click the *Hold* switch, select *MIDI Learn*, and tap the sustain pedal). The LED illuminates when *Hold* is active.

Chorale

Chorale is a great-sounding stereo chorus effect. It's highly recommend it for choral pad patches (hence the name). The original did not have a built-in chorus, but just about every Roland vocoder ever made does, and it's easy to hear why.

In/Out- Enables and disables the chorus effect.

Amount- Increases intensity of chorus effect as the knob setting is increased.

Output

Mix- Sets the balance of the signal between the dry modulator signal (i.e. incoming sidechain signal) and vocoded sound. Typically this would be set all the way the to the right, but it's useful for monitoring the modulator signal or as an easy way to mix vocoder with "dry" modulator signal.

Level- Sets the master output level. It's a good idea to carefully optimize the *Program*, *Carrier*, and *Output* gain for best fidelity.

Synthesizer Oscillator Section

The instrument version of Rackmode Vocoder includes a synthesizer oscillator section that is hardwired to the vocoder carrier section. Oscillator pitches are controlled by incoming MIDI just like any standard virtual instrument. The Rackmode Vocoder synthesizer oscillator section consists of

two polyphonic oscillators only. It does not contain filters (that's what the vocoder is for), or amplitude envelopes - the incoming modulator signal from the sidechain input controls amplitude via the carrier-side VCA's (and likely, the "signal" coming from your face).

Tune- Allows fine-tuning of master pitch by 100 cents, up or down.

Glide- Also known as "portamento," glide delays the voltage change between pitches for a sliding effect. Keyboard *Glide* works in mono and poly oscillator modes.

Vibrato / Delay- Delays the onset of vibrato from 0.1 up to 5 seconds.

Vibrato / Rate- Sets the vibrato modulation rate from 2 to 18 Hz. The flashing LED above indicates the current rate.

Vibrato / Depth- Sets the vibrato depth from 0 to 100% (about a whole-step).

Oscillators / Mono/Poly- When set to *Mono*, the oscillators will play one note at a time with last-note priority. The most recently played note takes priority. In the *Poly* position, up to 16 notes may be simultaneously played.

Osc 1 Range- Sets oscillators 1's overall coarse pitch range in standard organ footage settings of *32'*, *16'*, *8'*, and *4'*.

Osc 2 Range- Sets oscillators 2's overall coarse pitch range in standard organ footage settings of *32'*, *16'*, *8'*, and *4'*. The *Off* position disables oscillator 2 (hence the clever name).

Detune - Tuning control for Oscillator 2 only. This can be used to fatten up sounds by detuning a small amount, or for "building-in" a set interval. Detune range is up or down a fifth (seven half-steps). The *Detune* knob is disabled if *Osc 2 Range* is set to the *Off* position.

Wave- Selects ramp, variable-width pulse, or Rackmode Vocoder's exclusive glottal wave. The glottal wave selection engages a vocal cord wave model widely used in speech synthesizers. Each cycle of the wave is divided into two sections in time referred to as the "open" and "closed" phases. When air passes the vocal cords which have been brought together to produce voiced speech or song, the cords open and close once each pitch cycle admitting a single puff of air, similar to a trumpeter's lips.

Tone- Emphasizes bass or treble frequencies, center setting is neutral. The *Tone* knob affects all three waveforms.

Width- When the pulse wave is selected, this affects the pulse width, with maximum width (square) at the center setting.

When the glottal wave is selected, *Width* adjusts the relative time between open and closed phases, similar to the duty-cycle of the pulse wave. During the open phase, air passes through the opening between the vocal cords, so at higher settings, you'll hear a lot of "air." When the *Width* is at maximum, the vocal cords are open for almost all of each cycle and lots of air is used. Narrow widths (i.e. lower *Width* settings) are associated with holding the vocal cords in a tense manner, thus the open phase is short and you don't hear much air.

Try this yourself - speak with your throat very open and relaxed and you'll find that you will use up your air supply very quickly. When creating high voices, especially with the *Formant* control turned to the right, you will find that using a wide *Width* and some *Breath* dialed to taste will create a very realistic sounding voice.

If the ramp wave is selected, the *Width* knob is disabled and grays out .

Breath- Sets the volume of the "air" heard at higher *Width* settings. When the ramp or pulse wave are selected, the *Breath* knob is disabled.

The Cherry Audio Rackmode Phaser replicates the original rack-mounted 12 Stage Phaser with incredible accuracy. The original unit is super rare, and it's truly one of the best sounding phasers ever made. It's also very flexible, and capable of a lot of different subtle-to-extreme phasing colors and tones. These include individually switchable stages and feedback insert point, as well as extensive modulation, taking it far beyond the standard two knob-special stomp box.

How Phasers Work

Phasers make use of a special type of filter called an "allpass" filter. Unlike low-, high-, or band-pass filters, an allpass filters allows all frequencies to pass through. What good is that? Though it passes all frequencies, it alters the phase relationship *of a section* of the audio spectrum. By itself, you wouldn't hear any audible difference, but when the signal is combined with the dry signal, phase cancellation occurs in regions of the audio spectrum. In and of itself, this doesn't sound particularly interesting, but when the center frequency of the allpass filter(s) is continuously varied, voilà, we hear the familiar swooshing sound known as "phasing."

When a single all-pass filter is used, the effect is rather subtle - almost inaudible. But stacking multiple allpass filters or "stages" creates a deeper, more pronounced swooshing effect. This is why phaser effects are sometimes referred to as four-stage, six-stage, etc. (Most classic stomp box phaser effects use four stages.) Phasing effects can be further intensified by feeding back an inverted version of the phased signal between stages.

In a typical phaser circuit, the total number of stages and feedback (labeled on the panel as *Resonance*) routing are fixed. The 12 Stage Phaser's stages can be enabled or disabled in groups of two, and its feedback insertion point selected with the *Resonant Stages* selector. ("*Resonant Stages*" is also a hard-to-find Rush album recorded live in Brazil in '92).

Phasing Stages and Feedback Section

Phasing Stages- Selects the total number of allpass filters the signal is fed through. More stages generally equals a deeper phase effect, but don't discount the lower settings - there are a lot of cool sounds to be had here. (The MXR Phase 90, one of the world's most revered phasers, is only four stages.)

Mix- In *Chain* position, only the direct output of the allpass filter chain is audible. *Mix* position combines the allpass filter chain with the dry signal. The *Mix* switch is actually a pretty unusual feature, as most phasers are always in Mix mode and don't offer the ability to disable the dry signal. The allpass filters-only nature of the *Chain* position makes for relatively subtle effects.

Chain position can also be useful for creative stereoization effects when 12 Stage Phaser is used on an effects buss.
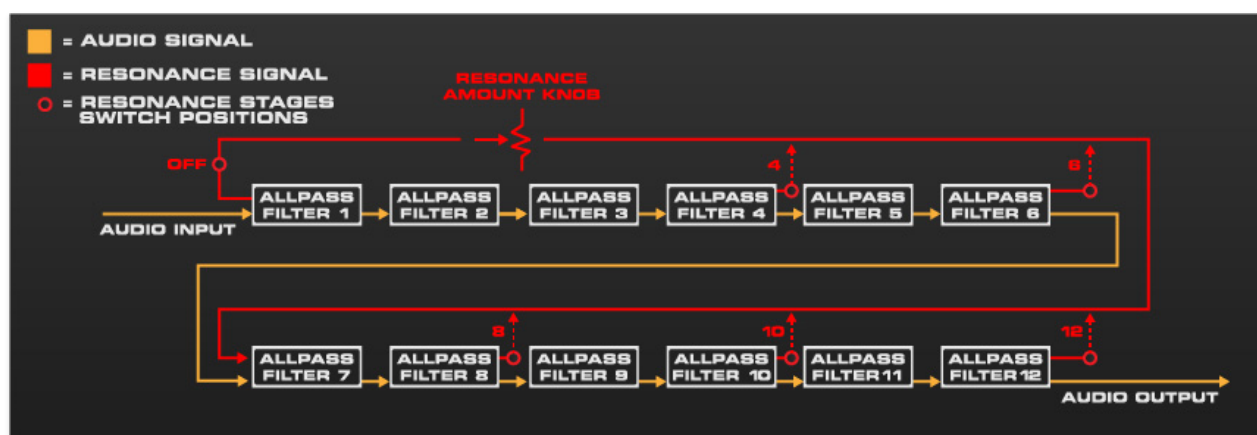
Resonant Stages selector / Resonance knob- The phased signal can be fed back into the allpass filter chain to intensify the phasing effect. The *Resonant Stages* selector chooses where the feedback signal is tapped; the Resonance knob adjusts the amount of signal fed back through the chain.

Sweep Section

The Sweep section is a low-frequency oscillator (LFO) that modulates the allpass filter center frequencies.

Rate- The *Rate* knob sets mod speed of the LFO, from 0.01 to 10 Hz (with *Sync* switch off) or from 8 beats up to 1/64th note triplets (*Sync* switch on).

Amount- Sets the modulation depth, i.e., how far the center frequencies of the allpass filters are varied up and down.

Sync- The *Sync* switch locks the LFO to DAW host tempo.

Waveshape- Selects the repeating modulation pattern. Available waveforms are sine, triangle, ramp, saw, and square.

Stereo Sweep- This enable wide stereo effects by using tapping the LFO at two points 180° apart and applying each to a separate allpass filter chain for each side. When engaged, the *Rate* knob LEDs flash in an alternating fashion.

Stereo Sweep works with mono or stereo signals, and is a fabulous way to stereoize mono signals. It can also be used to create cool rhythmic phasing sounds when *Sync* is used in conjunction with saw and square waves.

If you're using 12 Stage Phaser on an effects bus, make sure it's inserted on a stereo bus channel (or a *Mono>Stereo* channel for Logic users) to properly hear *Stereo Sweep*.

Offset- This is closely related to the *Amount* knob. Whereas the *Amount* knob adjusts the total amount of up and down modulation sweep, the *Offset* knob shifts the entire sweep range up and down across the overall mod range.

Output Section

Fast/Slow- The original 12 Stage Phaser included a "Motor Speed" control, intended to replicate the acceleration and deceleration of a Leslie-type rotary speaker. Unfortunately, the implementation was really confusing (it went into "slow" mode when the *Status* switch was disabled). To make a long story short, we changed this to a simple *Fast/Slow* switch to prevent confusion. Here's how it works:

- **Fast**- In *Fast* mode, the phaser rate is as indicated by the *Rate* knob position. The *Fast/Slow* LED will glow green.

- **Slow**- In *Slow* mode, phaser rate is approximately 1/16 the of the current rate knob position (to be clear, the fraction 1/16, not a sixteenth note). The *Fast/Slow* switch LED will glow red.

When the switch is clicked, the phaser speed gradually speeds up or slows down just like a Leslie-style rotary speaker.

Output Routing- The original 12 Stage Phaser had two rear panel outputs; the A jack outputted the allpass filter "notches," and the B jack outputted the allpass filter "peaks." It's not terribly important to understand their technical implications, but suffice to say, they have different sounds, and the *Output Routing* switch selects their configuration.

- **A**- A output in both left and right channels.

- **B**- B output in both left and right channels.

- **AB**- A output in left channel, B output in right channel.

If the *Stereo Sweep* switch is disabled, *A* or *B* positions result in mono output, whereas the *AB* results in stereo output. All *Output Routing* positions result in stereo output with the *Stereo Sweep* switch enabled.

Gain and Overload lamp- Sets overall output level. It's generally best to set this so the Overload lamp occasionally lights, and then back it off a hair.

Status- Enables or disables the effect.

The Cherry Audio Rackmode String Filter comprises 36 high-Q parallel bandpass filters arrayed throughout the musical range (say that five times fast... go ahead, we'll wait). The original unit was intended to mimic the multiple static resonances of a stringed instrument body such as the violin or cello. When simple waveforms (especially solo waveforms) are played through it, the String Filter imparts a woody richness to the sound making it sound more "stringlistic."

Having so many resonant filters close together has a secondary effect on the amplitude of moving harmonics as well. If vibrato is applied (i.e., frequency modulation), some harmonics will ride up and down their nearby filter slopes creating amplitude modulation of that harmonic. It is this kind of detail that the unit was created for, but it can create interesting effects from many signal sources.

The adjustment of this mix has a profound effect on the resulting tone. **The filters are arranged into odd and even interleaving sets which are associated with the left and right stereo outputs.** The stark stereo spread of the odd/even splitting can be lessened by setting the *Width* control more toward *Mono*. Note that the String Filter is really useful for stereoizing mono sources.

Input Level- Sets the overall level of the incoming signal.

Frequency Offset- The filter frequencies of the original unit were fixed, but we've added a *Frequency Offset* knob that simultaneously adjusts the frequencies of all 36 filters in parallel.

Resonance Offset- Globally adjusts the bandwidth of all 36 filters. Turning this down all the way pretty much renders the effect inaudible (not at all stringy), and turning it up all the way will sound clangy and metallic.

Filter Frequency Levels- The 36 filters are divided into four groups of nine: *Lo*, *Lo-Mid*, *Hi-Mid*, and *Hi*. The four groups are mixed via their respective *Filter Frequency Level* controls. The adjustment of this mix has a profound effect on the resulting tone.

AM Sweep and Sync switch- *AM Sweep* is comprised of four asynchronously running LFOs which produce tremelo on each of the four frequency groups. Very different tremulant effects are available at different rate settings. When the *Sync* switch is enabled, the LFO is locked to the host DAW host tempo and the knob will snap to settings ranging from 8 beats up to 1/64th note triplets. With Sync enabled, the LFO's are no longer asynchronous and will move in lock stop.

FM Sweep and Sync switch- *FM Sweep* is comprised of four asynchronously running LFOs that modulate the frequencies of each of the four frequency groups. When the *Sync* switch is enabled, the LFO is locked to the host DAW host tempo and the knob will snap to settings ranging from 8 beats up to 1/64th note triplets. With Sync enabled, the LFO's are no longer asynchronous and will move in lock stop.

Width- Sets the stereo panning width of the filters.

Status- Enables and disables the String Filter.

The Cherry Audio Rackmode Frequency Shifter is a super-accurate recreation of the vintage Bode Frequency Shifter. A frequency shifter is an audio modifier that "shifts" the entire frequency spectrum of the signal by a given amount. The Rackmode Frequency Shifter can shift the spectrum up or down by up to 5000 Hz, and the "downshifted" and "upshifted" signals are simultaneously available. The shift amount can be modulated via the onboard low-frequency oscillator.

What Is A Frequency Shifter?

It's important to understand how a frequency shifter differs from a modern digital pitch shifter (aka, a "harmonizer"). A pitch shifter maintains harmonic relationships between tones and harmonics of the input signal (or at least, tries to). If you speak or sing into a pitch shifter, this means the voice sounds like a larger or smaller person, because the harmonic relationships are maintained. Conversely, these harmonic relationships are *not* maintained with frequency shifter, and as a result, processed audio starts to sound unnatural beyond very small amounts of shift. Small amounts of shift are useful for phase-y types of sounds (particularly when the *Regeneration* knob is up). Large amounts generally create harmonically rich and clangorous metallic timbres.

In layman-est of layman's terms, even if you don't 100% understand what's going on under the hood, the frequency shifter can make a whole lot of incredible sounds, and it's very easy to use.

Input Level- Sets the overall level of the incoming signal.

Noise Gate Threshold and LED- Allows fine-tuning of input level to prevent bleedthrough of the unprocessed signal to the processed signal. Thanks to the wonder of modern computers, bleedthrough is a far less of a concern than it was with analog hardware frequency shifters, but it's included to accurately replicate the original.

Turning the knob clockwise sets the input threshold higher until the input signal completely disappears. The LED illuminates to indicate signal is

passing.

Scale selector and Shift Amount knob- This is a coarse control for overall shift range, similar to the octave switches in a synth oscillator. The *Scale* selector positions are as follows:

- **Exponential**- The *Shift Amount* knob scale is exponential, with frequency shift amounts represented in the outer concentric circle. In exponential mode, the shift range is 2 Hz to 2 kHz, from full counterclockwise to full clockwise.

- **5/50/500/5k**- The *Shift Amount* knob scale is linear, with frequency shift amounts represented in the inner concentric circle. Shift amount starts at zero (center position) and goes down or up when the *Amount Of Shift* knob is turned counterclockwise or clockwise, respectively. The maximum setting in either direction is determined by the current *Scale* range.

Regeneration- This is a feedback control that sends the output back to the input at an amount determined by the knob setting. It really opens up a lot of sonic possibilities, and with subtle amounts of shift, it's useful for phaser-esque (made a word!) sounds.

Mix- Sets the mix level between downshifted (*A*) and upshifted (*B*) signal.

Stereo/Mono switch- Selects whether incoming signals retain stereo and imaging and processing or get summed to mono prior to processing.

Power- Enables and disables the Frequency Shifter.

Modulation Section

The Modulation section allows low-frequency oscillator (LFO) modulation of Shift Amount.

Waveform- Selects the repeating modulation pattern. Available waveforms are triangle, ramp, saw, and square.

Rate- The *Rate* knob sets the speed of the LFO, from 0.01 to 15 Hz (with *Sync* switch off) or from 8 beats up to 1/64th note triplets (*Sync* switch on).

Sync switch- The *Sync* switch locks the LFO to DAW host tempo .

Amount- Sets the modulation depth applied to the Shift Amount.

Rackmode Ring Modulator

The ring modulator was invented by Frank A. Cowan in 1934 and patented in 1935 as an improvement on the invention of Clyde R. Keith at Bell Labs. The original application was in the field of analogue telephony for frequency-division multiplexing for carrying multiple voice signals over telephone cables. We're not totally sure what that means either, and if it sounds like we cut-and-pasted that blurb from Wikipedia, it's because we did. Anyway, ring modulators (also known as a "balanced modulators") date back to the earliest days of electronic music, and were one of the handful of ways audio signals could be radically manipulated back in the pre-synthesizer days of musique concrète, which is French for "bunch-of-weird-French-dudes-making-a-racket-with-tape-recorders-and-test-oscillators."

When two audio sources are inputted, the output contains only the sum and difference frequencies of the two signals, while removing common frequencies. The audio result is useful for creating sounds with inharmonic frequency content, which is useful for synthesizing bell and metallic sounds.

In order to facilitate processing a single input signal, ring modulators often include an onboard oscillator. Rackmode Ring Modulator's oscillator is not unlike a typical analog synthesizer oscillator, featuring range controls, and multiple waveforms. Its pitch can also be "played" with a standard MIDI/USB keyboard, allowing for (somewhat) more musical exploits in the realm of ring mod madness.

Input Routing

These may look like algebraic equations, but don't let them scare you, they're real easy understand. Just remember that a ring modulator always needs two input signals to do its magic, so the Input Routing selector defines the source of the two signals.

Input Mode Switch-

- **VCO x (L/R)**- Combines the onboard Voltage-Controlled Oscillator (VCO) with incoming left/right signal.

- **Sidechain x (L/R)**- Combines the mono or stereo signals coming from your DAW's sidechain input with incoming left/right signal.

    If you're not familiar with how sidechaining works, the idea is that it allows signals from elsewhere in your DAW's signal path/mixer to be routed to directly to plug-ins. (*The most common application of sidechaining is to route a kick-drum signal to control a compressor in order to make the entire track "pump" on kick drum hits, a trend we hope will go the way of "Got milk?" and pet rocks.*) In our case, it's an easy way to combine the signal on the Ring Modulator track with almost any other signal in a DAW project. Most DAW plug-in windows include a pop-up menu for defining sidechain signals (if you don't see it, please consult your DAW's owner's manual).

- **L x R**- Combines the left and right signals of an incoming stereo audio signal. The Ring Modulator will have no affect on mono signals in *L x R* mode.

Squelch

Squelch Threshold and lamp- Allows fine-tuning of input level to prevent bleedthrough of the unprocessed signal to the processed signal. Thanks to the wonder of modern computers, bleedthrough is a far less of a concern than with analog hardware frequency shifters, but it's included to accurately replicate the original.

Turning the knob clockwise sets the input threshold higher until the input signal completely disappears. The lamp illuminates to indicate signal is passing.

Squelch Switch- Sqwitch? Enables and disables Squelch.

Frequency Modulation

The Frequency Mod section allows low-frequency oscillator (LFO) modulation of the Voltage-Controlled Oscillator (VCO) signal.

Waveform- Selects the repeating modulation pattern. Available waveforms are sine, triangle, ramp, saw, square, and sample and hold.

Keyboard Track- Allows oscillator pitch control in standard semi-tones using a MIDI/USB keyboard controller.
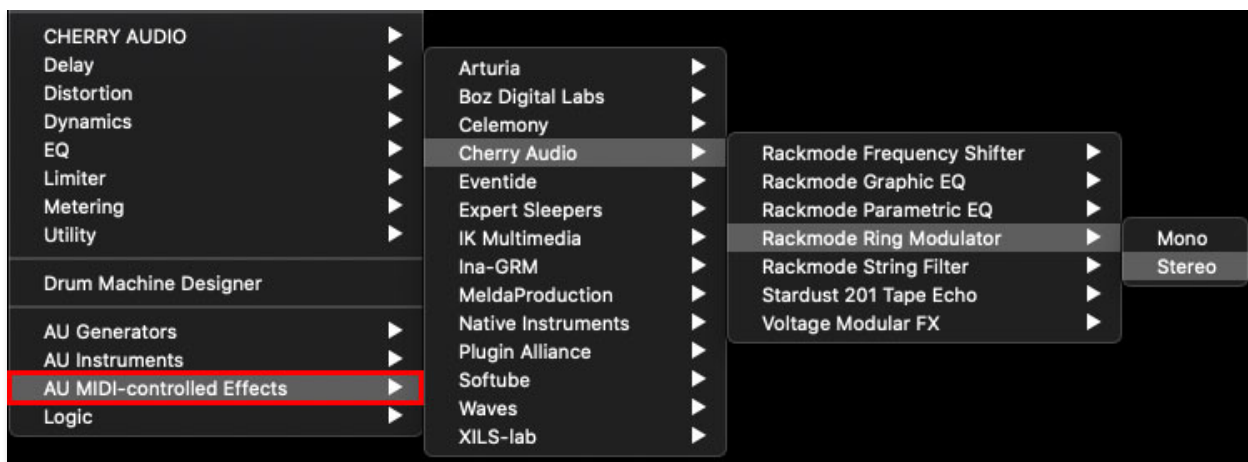
Important Note about using Keyboard Track: Some DAW software, namely Logic Pro, will **not** pass MIDI data to a standard plugin. That means MIDI notes played on aforementioned MIDI/USB keyboard won't make it to Rackmode Ring Modulator. But there's a simple workaround.

Instead of opening Rackmode Ring Modulator in a mixer effects slot, you'll need to open it in an instrument track, an *AU MIDI-Controlled Effec*t in a virtual instrument in slot (see diagram above). Once it's loaded, use the sidechain menu we discussed earlier to choose an audio source. To be clear, **you'll only need to open Rackmode Ring Modulator as an AU MIDI-Controlled Effect if you want to use a MIDI/USB keyboard to "play" notes on the Voltage-Controlled Oscillato**r. Otherwise, use a normal effects slot.

Rate- The *Rate* knob sets the speed of the LFO, from 0.01 to 15 Hz (with *Sync* switch off) or from 8 beats up to 1/64th note triplets (*Sync* switch on).

Sync switch- The *Sync* switch locks the LFO to DAW host tempo.

Voltage-Controlled Oscillator

The Voltage-Controlled Oscillator (VCO) signal is combined with the incoming audio signals when the Input Mode Switch is set to the *VCO x (L/R)* or *Sidechain x (L/R)* positions.

Waveform- Selects the VCO audio waveform. Available waveforms are sine, triangle, ramp, saw, and square.

Coarse- Sets the the VCO's overall coarse pitch range in standard organ footage settings of 32', 16', 8', 4', and 2'.

Fine- Allows fine-tuning of pitch by 1.25 octaves, up or down.

Freq Mod- Sets the modulation depth applied to VCO pitch. Center position equals no modulation; turning clockwise applies positive mod, turning counterclockwise applies negative mod.

Power switch- Enables and disables the Ring Modulator.

MIDI Controllers Setup and The MIDI Tab
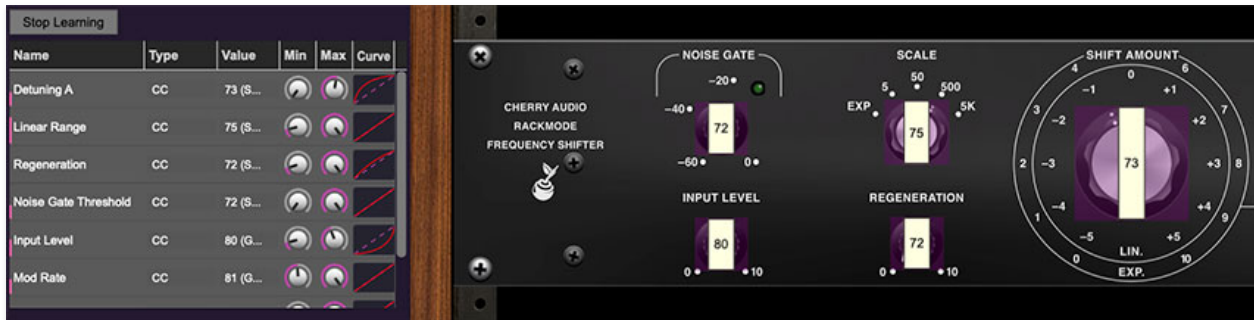
Assigning external hardware controls adds a whole new dimension of control and musicality. The MIDI Tab is where controller assignments can be viewed and tweaked. It's highly flexible and really easy to use.

Note that some DAWs handle MIDI controller assignments for effects plugs differently than others - some DAW software doesn't pass MIDI data to plug-ins in effects slots. Specifically, Logic Pro has its own way of handling MIDI to effects, which we'll explain later in this section. If MIDI controllers aren't assigning to onscreen controls as expected (i.e. nothing happens when you try to assign them), you may need to consult your DAW user guide, or do some Google/YouTube research.

Basic External Hardware Control Assignment

This is the quick, "I just want to assign a hardware control right now!" section. In this example, we'll assign a hardware slider control to the Rackmode Frequency Shifter *Shift Amount* knob.
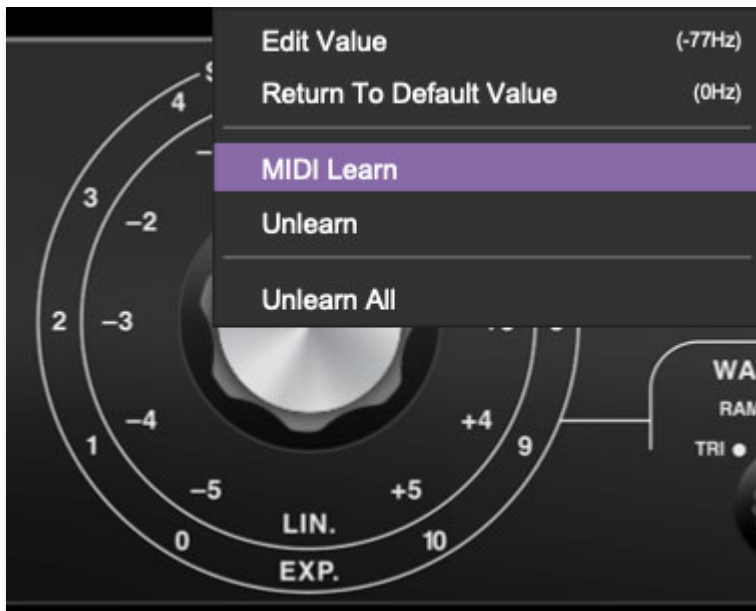
Begin by right-clicking on the *Shift Amount* knob and selecting *MIDI Learn.*
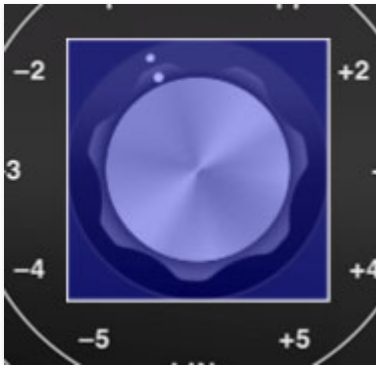
A transparent purple overlay appears over the knob, indicating that it's in learn mode. Now move the desired hardware control device. The purple overlay disappears and the hardware control will move the *Shift Amount* knob. If you get cold feet (or accidentally put the wrong control into learn mode), learn mode can be aborted by right-clicking and selecting *Stop Learning*.

This is the basic procedure for assigning hardware controllers to almost any Rackmode Processors control.

The MIDI Tab

This is command central for all MIDI controller assignments. Here you'll be able to see information about all currently assigned controllers and adjust control ranges. There's also an additional method for assigning controllers which is useful for quickly setting up multiple controls.

To view or hide the MIDI Tab, click the *MIDI* button in the top toolbar.

MIDI Learn button- This is almost the same as enabling MIDI learn mode by right-clicking a control. Click the *MIDI Learn* button to enter learn mode (all controls turn purple). Unlike right-clicking on specific controls, where Rackmode Rackmode Processors automatically exit controller assignment mode, clicking the *MIDI Learn* knob "stays on" to enable assignment of multiple hardware controls. This is handy for quickly assigning a bunch of sliders or the buttons of a grid-style controller.

To assign multiple controls, click *MIDI Learn*, click an on-screen control, move the desired hardware knob or slider, continue clicking and assigning on-screen controllers until all desired controls are assigned, then click *Stop Learning* to exit learn mode.
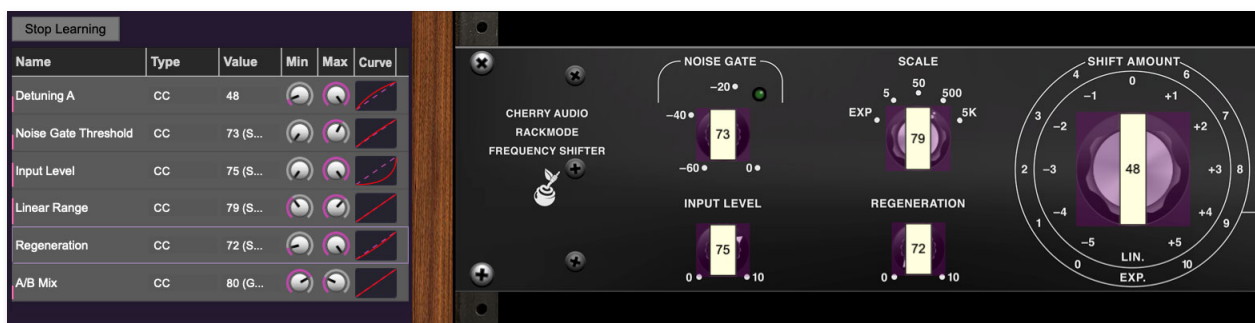
When in MIDI learn mode, previously assigned controller numbers are displayed atop of the relevant control. These indicate the MIDI continuous controller number of the assigned hardware control (these are also displayed in the *MIDI* library tab at left).

Once a MIDI controller has been assigned, in addition to real-time control of a Rackmode parameter, you'll also be able to record and play back controller data from a DAW.

Controlling Multiple Parameters With A Single Hardware Control

A single hardware controller can be assigned to simultaneously operate as many controls as desired. This can be very powerful when combined with *Min/Max* and *Curve* controls in the MIDI Tab (see the **Super Cool Min-Max Tricks** section below).

MIDI Tab Columns

Name- Displays the name of the parameter being controlled.

Type- There are five possible types of controller automation in Rackmode Processors:

- *Note*- Notes played on a MIDI keyboard controller, expressed as C-1 to G9

- *CC (MIDI Continuous Controller)*- The standard 128 MIDI controller numbers as defined in the MIDI spec. More specifically, these are the controllers transmitted by hardware knob and slider controls. MIDI CC's can be used to control parameters in real-time or recorded and played back within DAW software.

- *MMC (MIDI Machine Control)*- The MIDI control protocol for tape machine-style transport controls. Back in the dark ages, this was used to control wonky old Tascam and Fostex reel-to-reel monsters, but it's useful if your MIDI controller has tape machine-style transport control buttons.

- *Pressure*- Some keyboard controllers transmit controller data when keys are pressed and released as they're held down. The vast majority of keyboard controllers with aftertouch transmit "mono" aftertouch only; in other words, aftertouch data is the sum of all keys to one single data stream.

- *Key*- This allows keys of the computer QWERTY keyboard to at as button controls for Rackmode onscreen controls.

Value- Displays the specific automation controller. In the case of a *Note* this would show a MIDI note number (C-1 to G9); for a MIDI CC, this would be the MIDI CC controller number, etc. Clicking on the value opens a pop-up menu where all values can be selected.

Min- Sets a limit on the lowest value any automation control can set a mapped controller to. This recalibrates the range of the automation controller to the remaining parameter range.

Max- Sets a limit on the highest value any automation control can set a mapped controller to. This recalibrates the range of the automation controller to the remaining parameter range.

- ***Super Cool Min-Max Tricks***- Not only can parameter ranges be limited via the the *Min* and *Max* knobs, mapped control destinations can be completely inverted by setting the *Min* knob all the way up and the *Max* knob all the way down (or anywhere in between).

Limiting and inverting parameter ranges with the *Min/Max* controls is particularly useful when multiplexing a single hardware control to operate multiple parameters. Along with the *Curve* control, the customization possibilities are highly flexible.

Curve- These allow the customization of how incoming MIDI CC controls affect the movement of Rackmode onscreen controls, ranging from exponential to linear to logarithmic curves. (We're huge fans of the "Logarythmics," a synth pop duo hailing from the Pacific Northwest.)

MIDI Tab Column Configuration Right-Click Menus
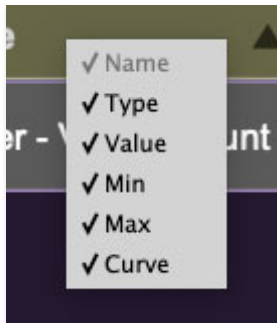
Right-clicking anywhere in the top row (*Name*, *Type*, *Value*, etc.) displays the column configuration menu. Checking/unchecking these allows hiding and display of each column. This has no effect on assignments.

MIDI Tab Parameters Right-Click
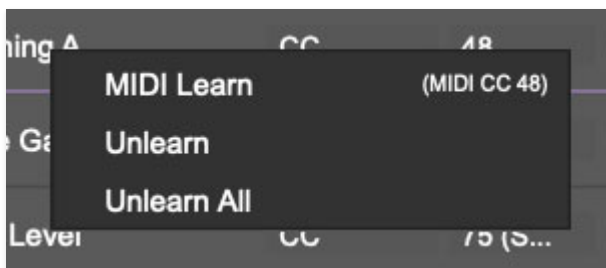
Right-clicking on an assigned parameter opens the menu above.

MIDI Learn- This is used to change the controller assigned to a particular parameter.

Unlearn- Deletes the selected automation parameter.

Unlearn All- Deletes all controller assignments for the patch. Rackmode Processors will display a warning dialog prior to deletion in order to thwart potential unlearn-related disasters.

# Configuring MIDI Hardware Controllers In Logic Pro

Logic does not pass MIDI data to effects opened in standard mixer channel effect slots. This means that MIDI controller assignment as described above will not work (specifically, nothing will happen when controllers are moved in *MIDI Learn* mode). There are two ways to successfully assign MIDI controllers to effects plugs in Logic.

Note that the instrument version of the Rackmode Vocoder is inherently an instrument plug, so you don't need to worry about any of the following workarounds for the effects plugs - just follow the MIDI controller assignment methods at the top of the page and Bob shall be your uncle.

Logic Controller Assignment Method #1

The first method is to use Logic's questionably named *Smart Controls* window that looks something like this (depending on the currently selected template). It creates a new window that (poorly) approximates the plug-in UI.
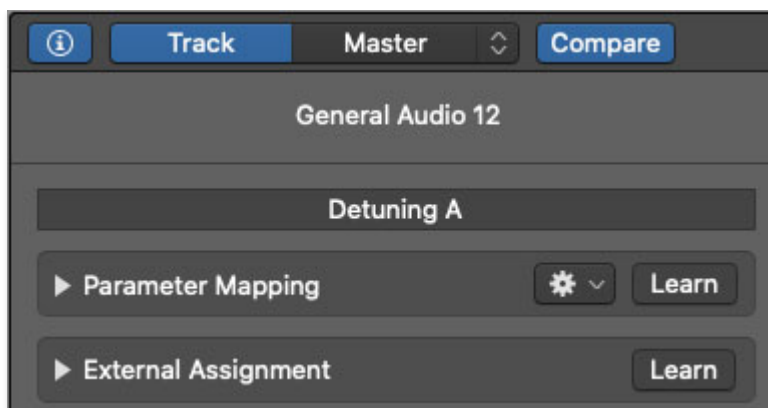
A number of plug-in controls are automatically assigned to the Smart Control knobs and switches; if the parameter you'd like to assign a controller to isn't present, Smart Control knobs or switches can be reassigned by clicking the *i* button, clicking the *Learn* button next to *Parameter Mapping*, clicking on a Smart Control knob, then clicking the desired control in one of the Rackmode UI's.

Assuming that the desired parameter is now a Smart Control, external MIDI controllers can now be assigned by clicking the *Learn* button next to *External Assignment*, and moving the relevant hardware controller.

Logic Controller Assignment Method #2

In case it wasn't clear, we're not big fans of Logic's impressively kludgey Smart Controls- we prefer the following alternative method that lets you use Rackmode's onboard MIDI assignment controls and tab as described at beginning of the section.

As mentioned, Logic does not pass MIDI data to plug-ins *when they're opened in a standard plug-in slot*. But you can "trick" Logic into thinking a plug-in is a virtual instrument - which *will* properly receive MIDI data. Here's how:

- Create a new virtual instrument mixer channel.

- Click on the channel's *Instrument* pop-up at the top of the channel and scroll down to *AU MIDI-controlled Effects>Cherry Audio> (Rackmode Effect of your choice here)*
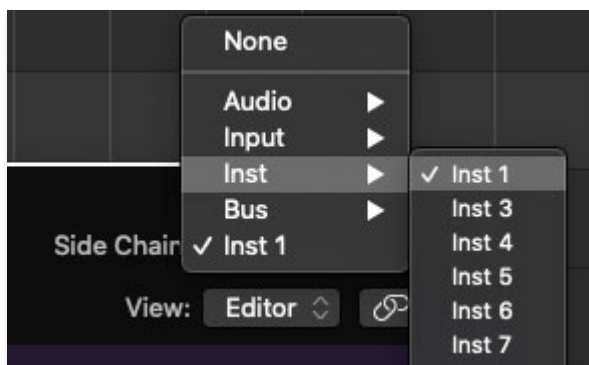
- At this point, we have a plug-in in an instrument slot, so we need a way to get sound into it. This is accomplished using the Sidechain Input pop-up in the upper-right corner - click on the appropriate category and choose the channel you'd like to process. This effectively configures Rackmode Processors as bus effects. Make sure the channel fader for the sidechain input signal is turned down, or else you'll hear the dry *and* effected signal (the channel fader will not affect the level going to the sidechain input, which is always on full blast).

As mentioned, with Rackmode Processors configured as an AU MIDI-controlled Effects, all MIDI learn functionality will work correctly.

Settings

Clicking the settings gear in the purple menu control strip opens a window with multiple tabs for configuring various "under-the-hood" settings. These are mostly set-and-forget kind of parameters - all the good stuff is in the main UI.

General
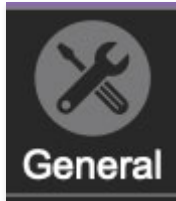
- **Add Undo For Control Changes With The Mouse**- Enabling this allows undo of knob/slider/button adjustments. You'll want this on if you want the ability to undo all aspects of patch editing and programming.

- **Create A Log File For Usage-** This creates a text doc of all of the Rackmode Signal Processors' internal and routines during use. It is mainly intended for our tech staff should you experience any issues. Clicking *Show Log Folder* opens the folder containing Rackmode Signal Processors log file docs.

- **Preset Folder-** Displays the current location of Rackmode Signal Processors sound presets. This can be changed by clicking and typing in the field.

  - **Browse...** - Displays the current location of preset folder in the file manager.
  - **Set Default-** Sets the current displayed *Preset Folder* path as the default location

- **Clear Cache Files-** Deletes all log files, temporary sounds, and the image cache.

Interface

Allows customization of Rackmode Signal Processor user interface settings.

- **Reset To Default Window Size-** Resets the Rackmode Signal Processor UI to 100%. Use this to reset the window size if effect interface workspace somehow becomes too large for your display and can't be resized (it's unlikely to happen).

- **Tooltip Delay-** Tooltips are those informative bits of text that pop up when hovering over a control (go ahead and try it, we'll wait...). The *Tooltip Delay* setting defines how long you must hover before the tooltip pops up.

- **Knob Movement-** Defines which how mouse movements related to turning onscreen knobs. It defaults to *Vertical Drag*, but can be changed to *Horizontal Drag*, or *Rotary Drag* if you're one those folks that cut their teeth on the *Steinberg Model E* VST back in 2000.

- **On Control Double-Click**- Defines what happens when the mouse is double-clicked on a control. If *Edit Value* is selected, an exact number can be entered by typing the number and hitting [ENTER] or [RETURN]. If *Sets Default Value* is selected, double-clicking a control resets it to its default value.

- **Mouse Wheel Adjusts Control Value-** Enabling this lets you adjust knob, slider, and switch values by moving the mouse wheel. This works great with a standard mouse wheel, but you'll want to disable it if you're using an Apple Magic Mouse (which will move the control AND scroll the window).

- **Show Tooltips When Adjusting Controls With Mouse-** Displays parameter tooltips/values when the mouse is hovered over a control or as a control is moved with the mouse.

- **Show Tooltips When Automating Controls-** Displays parameter tooltips/values when a control is being moved via automation.

- **MIDI Program Changes Should Change Current Preset-** Allows MIDI program change messages to select Rackmode Signal Processor patches. This parameter is DAW dependent, as some DAWs don't transmit MIDI data to plug-ins (we're lookin' at you, Logic Pro).

- **Ask To Save Modified Presets-** This opens a dialog window asking if you'd like to save changes if a patch has been edited and a new patch is selected. If you're the type that likes to click through presets and tweak a

control here and there, it can be annoying to have a window pop-up asking if you'd like to save every time you switch presets - if you're that person, keep this off.

- **Remember MIDI Mappings For New Plug-in Instances**- When enabled, Rackmode Signal Processors remember all MIDI Tab controller settings.

Account

Settings for your personal login information and account.

- **Email**- Displays the email address of the current login.

- **Update Login Info**- Clicking this opens the same email and password login screen you'll see when initially launching Rackmode Signal Processors. Click **here** for updates about yacht rock superstar, Kenny Loggins.

- **View Account Settings**- This opens your personal account page on the Cherry Audio Store website containing information about modules purchased and more.

  Software Update

  - We often fix bugs and make improvements; below are options defining how updates are handled.

- **Check For Update-** Click this to see if an updated version of Stardust 201 available.

  Status-

- **Automatically Install Updates**- Updates are automatically downloaded and installed.

- **Ask Before Installing Updates**- By default, Rackmode Signal Processors automatically download new versions when available. Checking this box defeats automatic updates and will ask if you'd like to install updates when they become available.

We'll never make changes that can potentially "break" existing patches but we recommend enabling *Ask Before Installing Updates* if you're using Rackmode Signal Processors for live performances or other "mission critical" situations.

- **Never Install Updates**- Rackmode Signal Processors never automatically installs updates. You can be just like my dad, who's still running Snow Leopard on his Mac and shaking his fists in the general direction of Cupertino.

- **Show Updates On Toolbar**- Checking this will display an icon in the toolbar next to the logo letting you know there's an update available.

Show Notifications From Cherry Audio- We occasionally will fire off in-app advertisements; disabling this checkbox will hide them. We hate repetitive, annoying ads as much as you, so we won't use this feature too often. When we do, it'll be for something super cool (like a killer sale), so we recommend leaving it on.